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THE ELECTRIC PRIMER PART X

MODERN RECORDING & MUSIC

VOL. 6 NO. 5
FEBRUARY 1981

THE GRATEFUL DEAD IN CONCERT!

LAB REPORTS:
BPI 7000 Audio Analyzer
GLI 1500 Graphic Equalizer
Studiomaster 800B/C Stereo Power Amplifier

HANDS-ON REPORT:
Carvin MX 1202 Mixing Console

NOTES:
Beigel Sound Labs Envelope Controlled Filter

NEW PRODUCTS
RECORD REVIEWS

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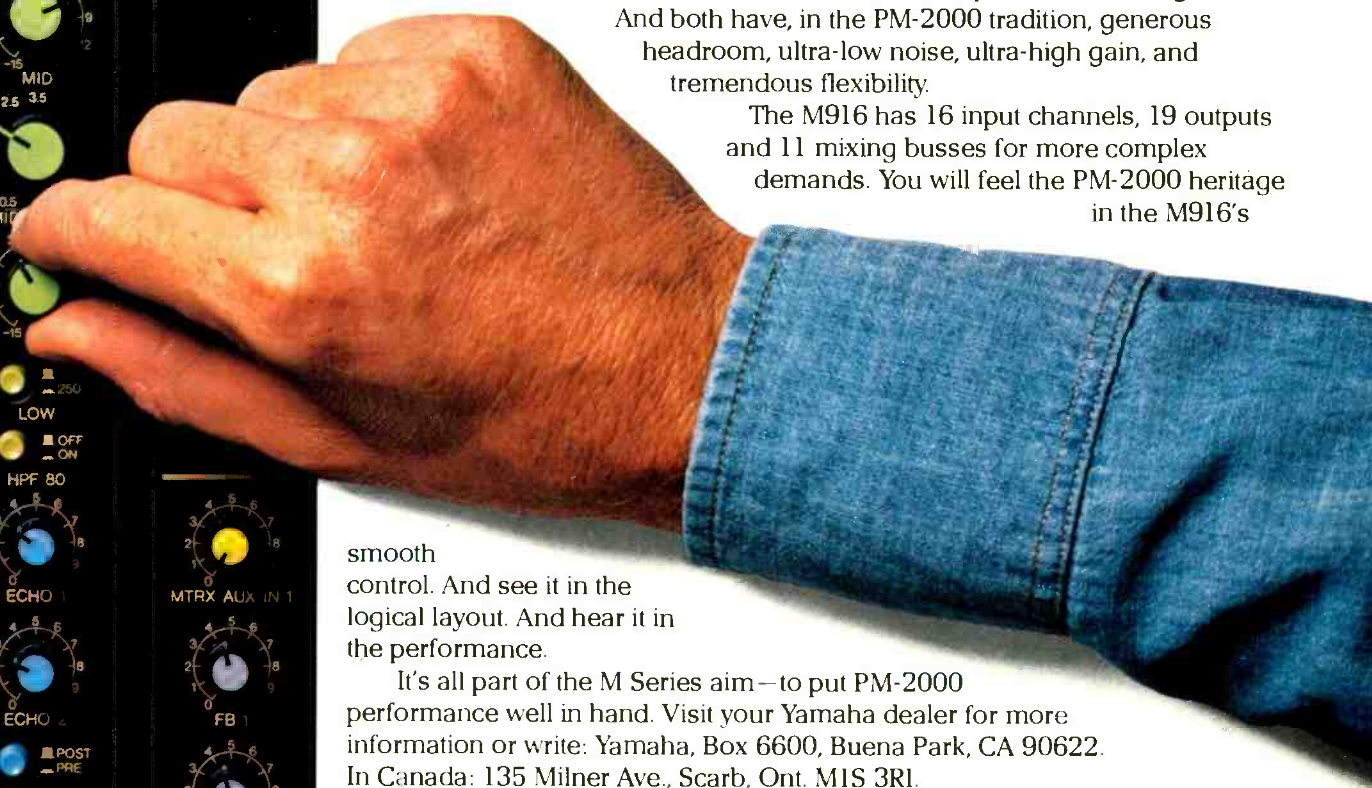

Yamaha's PM-2000 mixer has established itself as an industry standard. Our new M Series mixers maintain that standard of performance—with the features, price, and size to meet your demands.

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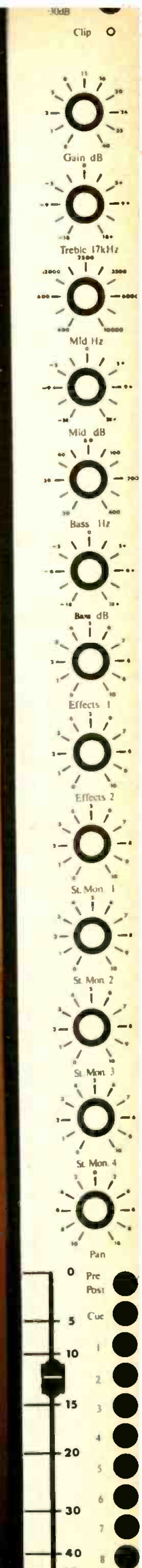

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We are the makers of Studiomixer and the message is *Totally Modular*. Before you ask, we'll answer. *Totally Modular* is a concept . . . our concept of how a mixing console should be built.

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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.

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BOSS

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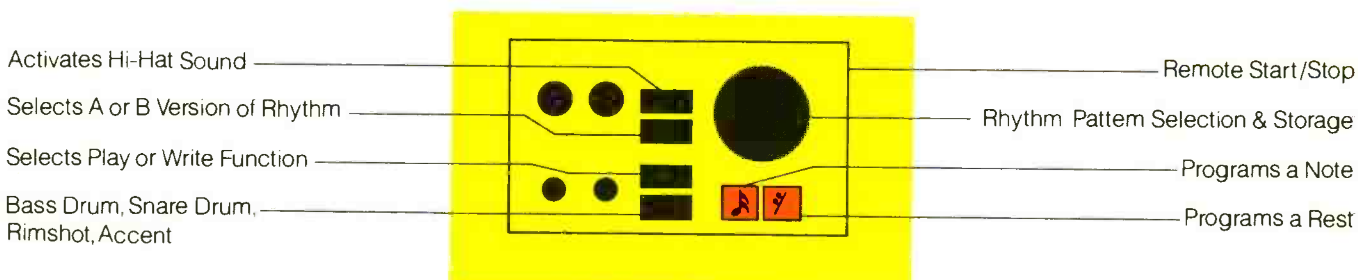
Subject: A Non-Drummer's Guide to Writing Drum Rhythms

DR-55



It's a fact that just about all Rock music is based on the drum beat. So when you're writing or practicing, a good idea can sound awfully flat without the proper beat behind it. The BOSS Dr. Rhythm lets you write up to 16 of your own drum rhythms, and you don't have to be a drummer to do it.

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The Step method used by the Dr. Rhythm breaks each measure into 16 (or 12) steps, each step corresponding to one sixteenth note. To program a rhythm, select a sound (such as the Bass Drum) and walk your way through the measure, pressing the "Note" button on each of those steps where you want to hear the Bass Drum, and the "Rest" button on each of those steps where you don't.

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FEBRUARY 1981
VOL. 6 NO. 5

MODERN RECORDING & MUSIC

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Profile: Guitarist Pat Metheny
Beyond the "Electric Primer!"
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LETTERS TO THE EDITOR

JOHN LENNON
1940-1980

The dream is over, but the music plays on . . .

Overtones of Conflict

I'd like to respond to a response to a letter. The original letter was written by Chris Michaels, and the response was written by Peter Gravina and was published in the December 1980 issue of *MR&M*. His intent may be well-meaning but I'm afraid the misconception is *his*. Chris Michaels' ears are OK.

Peter's position seems to be that (is this letter a ringer?) the *really* important frequency in a musical tone is the fundamental and that the overtones are insignificant "also-rans" that play little, if any, role in its character.

It's important to understand that a *musical* tone is not a *pure* tone or a sine wave but a critical balance of fundamental and overtones. This is what gives a note its timbre. It's what makes a SUNG 783.9 Hz. note (Peter's example) sound completely different from the same note played on, say, a clarinet. The presence of overtines—those applicable to the Overtone Series and those produced by inconsistencies in the instrument—give a fundamental note its quality of musicality.

Boosting midrange, and higher, frequencies is one of the most common methods in an engineer's bag of tricks to add presence to a vocal, or anything else for that matter. Part of the reason is that, because there is little in the way of interfering fundamentals in the 2-4 K range, a vocal can be reinforced in a frequency band where it doesn't have to compete with the bulk of the instrumentation.

Secondly, the human ear does not hear "flat." Its sensitivity falls off rapidly after 2 K (Fletcher-Munson curve). A boost in the 2-4 K region tends to extend this curve creating a psychacoustic presence to the boosted program.

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Making an accurate and faithful recording on most cassette decks requires a lot of practice, a lot of patience and a lot of jumping up and down. After all, with conventional decks, you have to adjust the recording levels as the music varies. But not with Technics RS-M51.

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Finally, since a boost in this frequency band affects mostly harmonics that are consonant with the salient fundamental, you don't affect the harmonic balance to any great degree.

In the end, though, it's the ear that's the judge of any audio debate, not calculators or scientific rationalizations (my own inclusive). *My* "recommended follow-up material" to you, Peter, would be just about any record you care to listen to. For instance, listen to anything recorded by Val Garay (Linda Ronstadt, James Taylor, Jackson Browne) and you'll hear what a difference even a (tasteful) 15 kHz boost makes.

Otherwise, you can look at any standard recording console. They don't put those EQ dials there for appearance.

—Steve Manes, Pres.
ROXY RECORDERS
New York, NY

We received another letter disputing Mr. Gravina:

The other afternoon, my 3rd-year Sound Recording class was discussing equalization techniques in mixing, and a student mentioned Peter Gravina's letter "Singing in the Range" (December issue of *MR&M*). The letter raises several issues worth discussing. Unfortunately, Gravina is off the track, and heading the wrong way.

Gravina suggests that the practice of boosting equalization at 4 kHz to bring out vocals won't work, on the basis that this is outside of singers' ranges, and that the overtones present in the band around 4 khz are "barely audible."

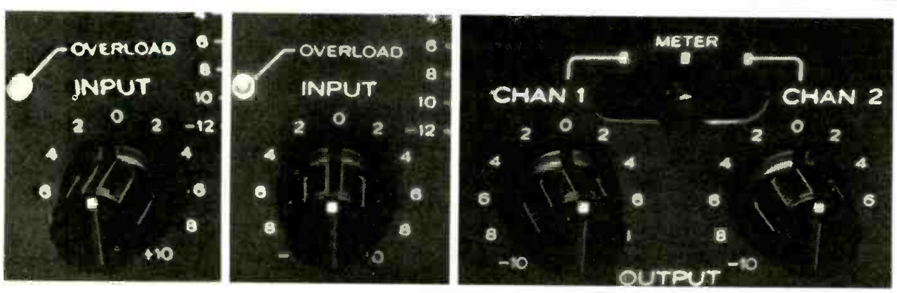
This assumption errs in three respects: (1) the nature of pitch vs. frequency, (2) the nature of speech, and (3) the nature of the hearing.

In the first respect, it is wrong to assume that most of the energy contained in a given pitch exists at the fundamental frequency of that pitch. My own experience is that most often it doesn't especially in musical contexts. The fundamental frequency is quite often irrelevant, except insofar as we use it to identify the pitch. Often (such as in a pulse wave, for instance), energy at the fundamental is insignificant and inaudible. In many routine cases, the energy of even extremely high overtones is quite audible. To cite an extreme case, if the overtones of a Hz square or sawtooth wave that are present in the band between 10 and 20 kHz are boosted or cut by more than 3 dB, the effect is quite obvious. Pitch is not the same as frequency, and it is as wrong to assume that because a singer sings a 440, his/her energy is centered at that frequency predominantly.

In the second respect, speech (or singing) consists of both vowels and consonants. Vowels carry the pitch information, and range from fundamental-oriented (ooooo) to fundamental-rejecting (iiiiiii). Consonants are essentially bursts of noise ranging from LF ("buh") to very high HF ("sss"), and these consonants provide the large majority of intelligibility cues inherent in speech. In general, the noise spectra of consonants tends to be broadband. Thus, such material is best dealt with on the assumption that it is a series of noise transients connected by complex waveform pitches (sort of like a drum kit connected by pitches).

Third, our hearing is most sensitive in the band between three and six kHz. We have an uncanny ability to pull material out of this band for use in discerning echo-location cues and vocal intelligibility. Speech perception is based on the perception of transient noise in this band.

For these reasons, it is useful to boost in this area of the audio spectrum to bring out vocals in mixing. When possible, I reserve the 4 kHz



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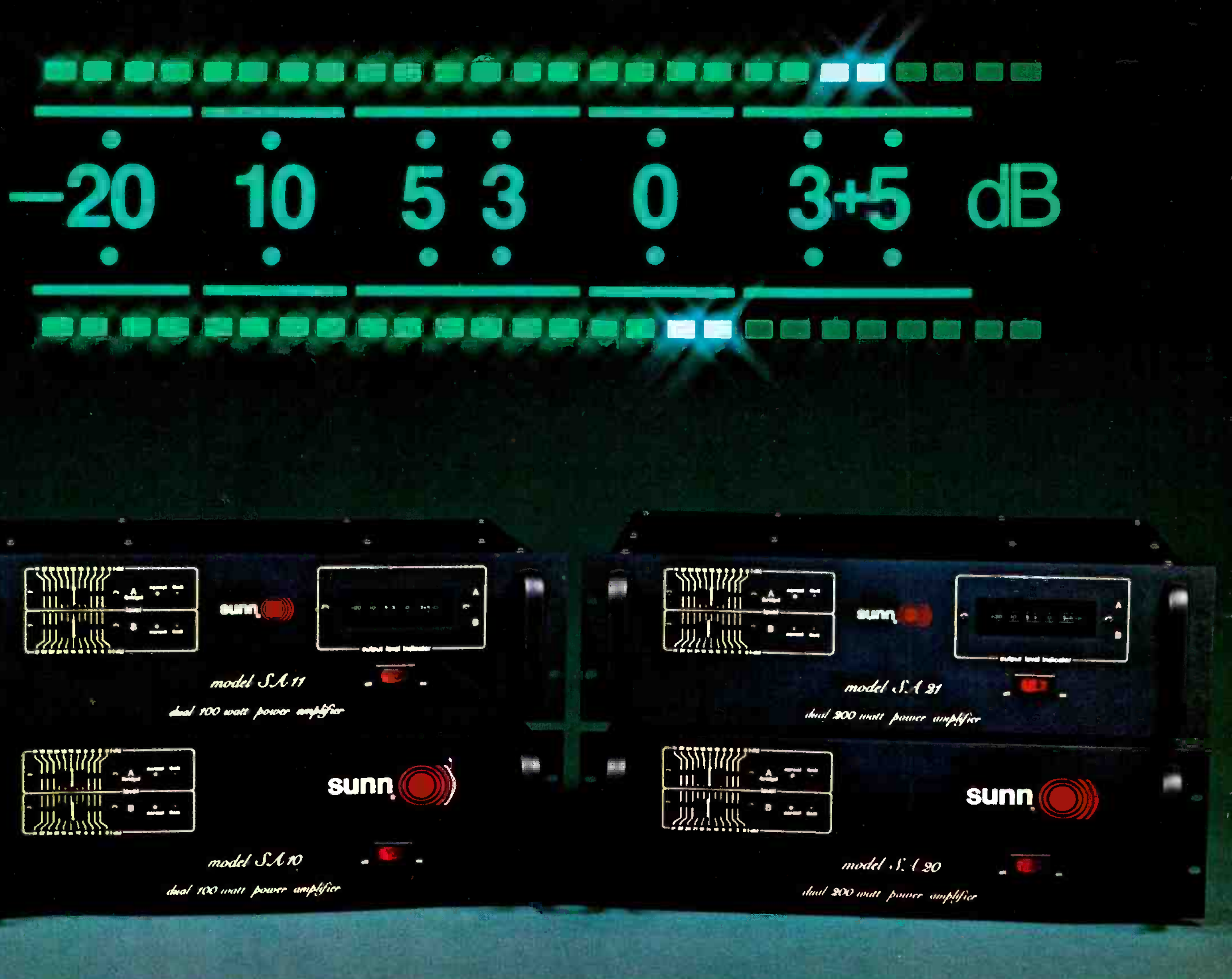

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area for vocal boost (if needed). A big problem with many P.A. mixers is that other HF and upper mid frequencies aren't available for equalization, and if you boost *everything* at 4K, all you have is equalization that sounds harsh.

A final caveat: equalization isn't the whole answer to better intelligibility and presence in mixing. A careful consideration of time domain, stereo imaging, and good basic balances are just as important.

Sincerely,
—David Moulton
Director Sound Recording Technology
SUNY/Fredonia, NY

Sound Advice

This letter isn't commentary, but rather a request for some information. I'm interested in working for a sound and/or light company and becoming an audio or light technician. I am aware of a few firms that provide these services to different musicians and acting organizations and I ask for your help in obtaining their addresses. I am enrolled as a student in a four year audio technical program, but would like to try to become employed in a sound company.

The firms whose addresses I would like are: Showco, Audio Analyst, and Clair Brothers.

—Joe Picuri
Washington, D.C.

The firms you mentioned might give you some leads. Showco's address is: Showco Manufacturing Corporation, 9009 Governor's Row, Dallas, Texas 75247, and their phone number is: (214) 630-7121. Audio Analyst: P.O. Box 33, 27 South Main Street, Terryville, Connecticut 06786. Their number is: (203) 583-2535. Finally, Clair Brothers, at P.O. Box 396, Lititz, Pennsylvania 17543. Their phone number is: (717) 733-1211. I hope that these companies prove to be of some help to you either in terms of information, employment opportunities, or both.

Keep Talking Back

(Soldering iorn? What soldering iorn?
Did you see a soldering iorn?
No, I haven't seen a soldering iorn.
Hey! What's a soldering iorn???)

So you think I've been asleep, eh? Not so, "resin breath"! Actually, I'm very encouraged with the response to my question in regard to phantom powering the Teac model 5. Without a doubt there are model 5 owners, as well as others, who appreciate Eric Breviak, Carl Sandler, Dan Dugan and any future contributor for sharing their "secrets." And, a special thanks to *MR&M* for providing the means by which we can benefit from one another's experiences. Hats off to all.

(Now back to this, whadiyacallit . . . Oh yeah . . . soldering iorn.)

—Ed Perrone
Gloversville, N.Y.

Thanks, Ed, for the letter (and the poem —I don't know which is verse.) Maybe that one Talkback letter will spawn a whole slough of letters and bits of advice. Anyone out there who can share their knowledge, please do so.

Doug Who?

In reference to the letter appearing on p. 6 of the October 1980 issue of MR&M, *entitled "Other People's Mail," we received a response from Jim Rupert, clarifying the point of who, in fact, Doug Dickeson really is.*

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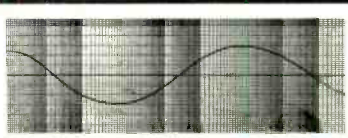
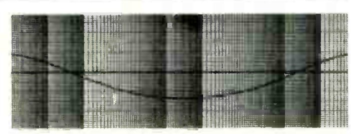
SECTION A



ACTUAL SWEPT-FREQUENCY RESPONSE CHARTS SHOWING THE EXACT CURVES RESULTING WHEN CONTROLS ARE SET AS SHOWN IN PHOTO ABOVE.



SECTION B

DESCRIPTION Exclusive Zero-Gain L.E.D.'s provide instantaneous in-out balancing convenience and optimum EQ performance. They are operated through a high precision Differential Comparator circuit with a read-out accuracy of 0.1dB, to provide true equalization with no change in the signal level being processed.

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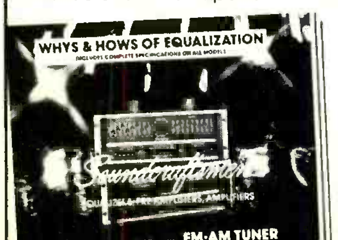

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"Harmonizer" is Eventide's brand name for a special effects device including pitch change.

I would like to thank you initially for the nice plug in the October 1980 issue of *MR&M*. It is nice to be remembered. The editorial aside, "We're not sure who Doug Dickeson is," overlooks a gentleman whom I think deserves a bit of recognition.

Doug Dickeson is an engineer par excellence whom I have had the pleasure of working with personally many times over a ten year period. He has been the chief engineer at several midwestern studios and for several years worked as an audio engineer at Glen Glenn Sound in Hollywood, working with innumerable big names in both the music and motion picture industry. He is innovative, hard working and very good to his mother. He is also my partner, confidant and friend. Perhaps this is the least impressive of his list of qualifications, but it means much to me.

I hope this will clear up the mystery of who Doug Dickeson is. If your schedule permits I hope this letter might be printed to relieve the inferiority complex currently plaguing Mr. D. (Also, whenever he sees his name in print he squeals like a crazy monkey!) If nothing else, may he be thought of and remembered as a personal friend of Melvin Godzilla. What more could any of us shoot for? Till we meet again, may I most cordially remain

—James F. Rupert
Lincoln, NE

Sorry, Baby

We sincerely apologize. The first paragraph of Product Scene, in the December 1980 issue of MR&M should have read, "The new PCM 41 ("Baby Prime Time") from Lexicon, Inc." We had omitted the model number: 41.

Limiting Noise

I built the PAIA Limiter described in the November, 1979 article. Yes, it does indeed limit, but with an unacceptable amount of noise. Feeding a 1 kHz sine wave through it, at the point the limiter becomes active a clipping sound is also present. By adjusting the controls, one can actually count the "ticks." At the limiting point, I can count 6 clicks or pops per second. If the limiter is put into the stereo mode the number of pops doubles. Feeding musical material through the device, any time the input exceeds the threshold there is a dreadful clipping sound in proportion to the amount the input exceeds the threshold. Below the threshold there is no limiting whatsoever; above the threshold there is absolute infinity limiting.

I believe I could use the thing if only I could get rid of the clipping noise. Otherwise it is destined for the trash can. I hope that there are no similar problems with the new spring reverb project.

—Burton E. Hardin
Charleston, IL

Craig told us that for every ten letters he gets praising the Limiter, he gets one with a problem. He feels that this is often due to a misapplication or to a bad component. In fact, we printed a letter in our December issue in which Craig explained how to use the Limiter properly, since he felt that he hadn't really touched on that in his original article. He feels, though, that your problem is something new and different. He suggests first trying a different 4136 IC since that could be the problem. He also suggested that if that isn't the problem, you send the Limiter to him to take a look at it. That's quite generous, I'd say. If the problem is anything more than a bad part, solder joint or something of that nature, then we are promised more information to be printed in this column.

A Credible String Band

I was pleasantly surprised to see a review of an "old-timey" string band (Tasty Licks) in your November 1980 issue, on' the Rounder label. It's truly an encouraging sign for the small minority of us who feel that rock is not the final answer to man's musical search. Thanks for a good, informative, and well-laid out magazine.

—Peter F. Feldmann
Sonyatone Records
Santa Barbara, CA

Glad to hear that we've made "just one someone happy." No, I'm sure you're not the only one, actually. It's good to know what a variety of musical tastes remain.

Believing One's Ears

In the August 1980 issue of MR&M, Chris Michael mentioned that he believed a boost in the 2 to 4 kHz region gave more presence to vocals. In the December 1980 issue, Peter Gravina disputes this claim; however, I feel Peter's conclusions are based on

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Pick a number from 9 to 52!

You've just chosen the ideal DC voltage to phantom-power these new ATM electret microphones.

Introducing four "universal" phantom-powered electret microphones. Designed to work from external power, internal regulation automatically handles any voltage from 9 to 52 VDC without adapters, switches, or rewiring. Just plug in and enjoy. With current drain a mere 0.3 mA at 9 volts (4 mA at 12-52V) a 9V battery lasts thousands of hours, not just the 60 or 70 hours typical of other mikes.

When your power supply isn't available or isn't enough, use ours. The new AT8501 Dual Battery Supply holds two 9V batteries. One to use, and one in reserve. Instant switchover and test LED eliminates guesswork. And spares are as near as the closest shopping center. Neat!

But convenience and versatility are just two of the advantages of the new ATM models. All-new electronics provide plenty of headroom inside the microphone with no more than 1% THD even when used in acoustic fields of 141 dB SPL. Which sets new standards for clean sound even close-up to big brass or inside a powerful drum kit.

And the sound you hear is wide-range and *musical*. Presence without peaks. Highs to 20,000 Hz but without a raspy "edge." Yet despite their responsiveness, these new ATM microphones have the "Road Tough" reliability proved so often on stage and in the studio.

Before you add another microphone, compare our sound, our convenience, our reliability, and our cost. Write for literature and list of nearby ATM microphone specialists. Get great sound...right from the start! AUDIO-TECHNICA U.S., INC., 1221 Commerce Drive, Stow, Ohio 44224. (216) 686-2600.

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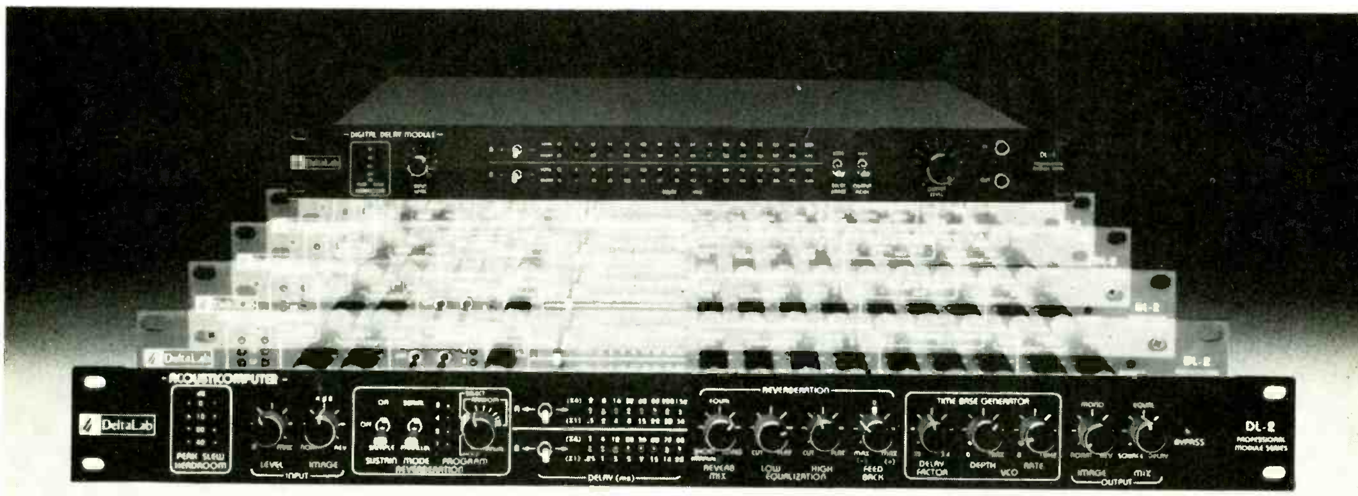
two incorrect assumptions.

The first incorrect assumption is that vocals only contain energy at the fundamental pitch of the note being sung, with lesser amounts of energy present as harmonics. In fact, the voice produces both pitched and unpitched sounds. Unpitched sibilants (such as the "s" sound) contain great amounts of energy in the treble region—even up to 10 kHz and above. Unpitched plosives (such as the "p" and "b" sounds) contribute wideband energy, with a concentration in the bass region (well below the fundamental of the note being sung).

Since these unpitched sounds determine the *intelligibility* of a vocal to a major extent, it is entirely reasonable that accenting upper midrange and treble frequencies would emphasize the higher frequency unpitched speech components, giving an increase in overall intelligibility that would indeed make the vocals appear to "pop out" from the mix.

The second incorrect asumption is that the energy contained in the harmonics of the voice's fundamental pitch are relatively insignificant and that a treble boost would not emphasize these harmonics to any great extent. However, one look at a vocal on an oscilloscope shows that the voice is anything but a sine wave; even though the harmonic components are weak, they are definitely there and can be distinguished from the fundamental pitch. Since our ear can hear tape hiss that is 60 dB below an output signal, it follows that harmonics that are −20, −30, or even −40 dB below a fundamental signal can also be perceived. If you boost the treble on a tape track, the hiss will appear to be more prominent; so it doesn't seem unreasonable that boosting the treble on a vocal will also affect the harmonics of the fundamental pitch being sung, even if those hoamonics ar comparatively weak.

I suppose the lesson to be learned from all this is that the human ear - not an oscilloscope, calculator, or spectrum analyzer - is the ultimate piece of test equipment. If boosting the upper midrange makes the vocals stand out more, then it's better to figure out an explanation for the observed phenomenon than to assume that the ears are hearing incorrectly. As one who often uses a boost in the upper midrange to improve the intelligibility



How's THIS for an encore?

Modern Recording called our DL-1 Delay "probably the best we have encountered" . . . a tough act to follow.* Now after more than a year in development DeltaLab introduces its encore - the ACOUSTICOMPUTER® - a combination digital-delay and special-effects processor designed for use both on-stage and in the studio, providing well-known functions (echo, doubling, chorusing, vibrato, flanging, etc.) plus new effects not available in any other device.

- Pre-reverb delay with two independent delay channels, variable from 0.25ms to 152ms with LED display.
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- Two channels in and out. Built in reverb mixing and stereo imaging controls.
- Foot-switch controlled bypass.

It's impossible to describe in this space everything the ACOUSTICOMPUTER does; you'll have to experiment with it yourself. By carefully minimizing the number of separate controls and grouping them logically, we've made it easy for non-engineers to operate the ACOUSTICOMPUTER.

For further information call or write Phil Markham at DeltaLab Research, Inc., 25 Drum Hill Road, Chelmsford, MA 01824 Tel. (617) 458-2545.

*See Modern Recording "Hands On Report," Sept. 1978.

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JBL

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Flat frequency response. It means accuracy. Naturalness. Reality.

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This new, compact professional monitor produces deep, distortion-free bass. And does it with a newly developed 10" driver. Its massive magnet structure and voice coil are equivalent to most 12" or 15" speakers. Yet it delivers heavy-duty power handling and a smoother transition to the midrange than most larger-cone speakers.

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Working together, these precision matched speakers offer superb stereo imaging, powerful sound levels and wide dynamic range.

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We think you'll agree that its combination of flat response, power and moderate size flattens the competition.

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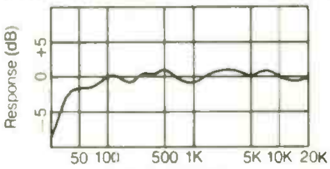

On-axis frequency response, 4313 monitor.

James B. Lansing Sound, Inc., 8500 Balboa Blvd., Northridge, California 91329.

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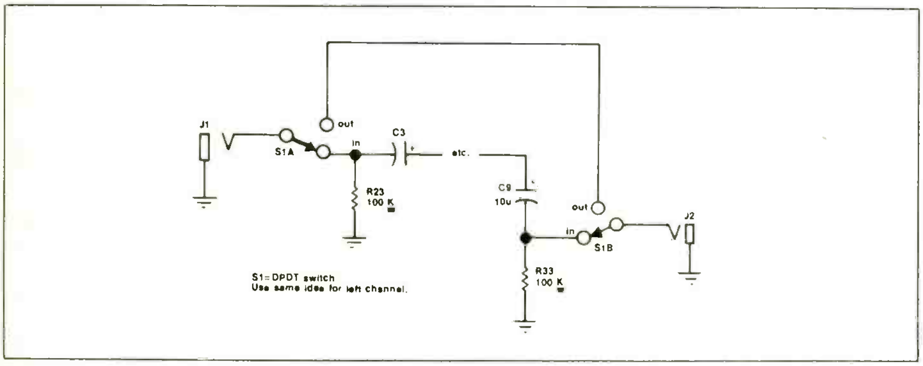
of my vocals (as well as the definition of acoustic instruments, few of which have fundamental pitches anywher near the upper midrange), I can attest that Mr. Michael's ears are indeed hearing what he says he hears.

—Craig Anderton
Contributing Editor
MR&M

For those of you who have been following this sequence of letters, you may have noticed that in printing Craig's letter, "Limiting Yourself" in the December 1980 issue of MR&M, we neglected to include a schematic that he had meant to have printed with the letter. We are belatedly printing that in this issue. Sorry!



Drumming it In

In reading the article on Jack Bruce & Friends in the December 1980 issue of *MR&M*, I noticed that Billy Cobham talks about a Hagar snare. I believe that what he was referring to is the Hinger Space-Tone snare drum, made by people in Leonia, New Jersey. The drum's halves separate by means of four knobs corresponding with slides. Also, the drum expands from 6 ½ inches to 8 inches, not 9 inches. I just thought I'd call this to your attention, as there are probably some readers who are interested in the drum, and would like to know the correct name of the company. Their complete address is: Hinger Touch-Tone, P.O. Box 232, Leonia, New Jersey, 07605. Phone: (201) 342-2858.

Bob Saydlowski, Jr.
Pittsfield, MA

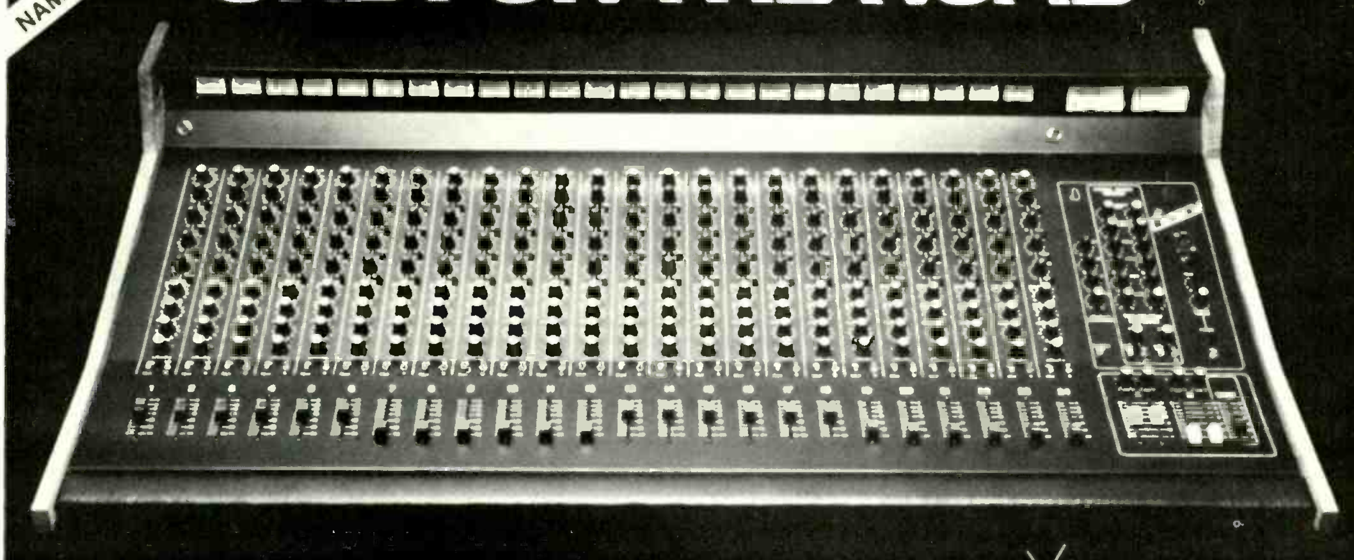
We spoke first to Bert Holman, manager of Monarch Bureau, and he spoke with Bill Cobham, who told us that, yes, you were right, the drum was in fact a Hinger Space-Tone, not a Hagar. Hinger has relocated, and though we appreciate your giving us their address, we must update your files, and let our readers know the new address of Hinger Touch-Tone:

It is: 280 Little Britain Rd., Newburgh, N.Y. 12550. Their phone number is: (914) 565-8870.

The drum Billy has, we were told, opens peripherally around the center, 1 ½ inches, and increases the volume projection by about 30%.

NAMM BOOTH 951

'ONE FOR THE ROAD'



The 24/4 has just about everything.

Primarily designed for Sound Reinforcement use its just as happy making simultaneous four-track recordings. Each channel has its own VU meter to quickly spot any problems, as well as four-band quasi parametric EQ. There are two echo 'bus' and one monitor bus. Monitor has EQ on the output and the two echo returns have EQ also. There is a full mixdown facility for the groups to produce two stereo pairs or four mono sub groups.

Yes the Canary 24/4 has just everything including a rugged flight case. Other models in the range are 16/4, 16/2 and 10/2.

Canary

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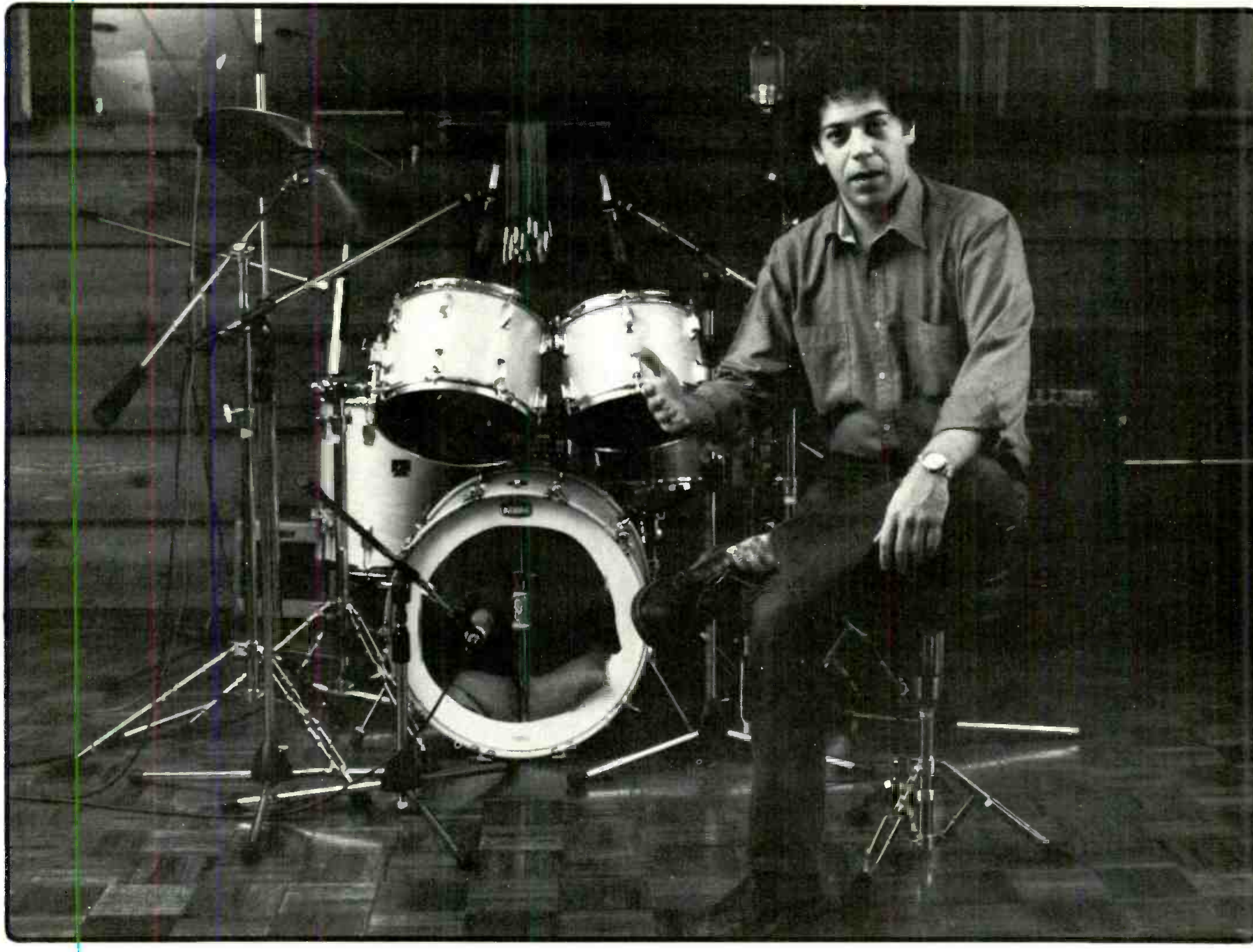
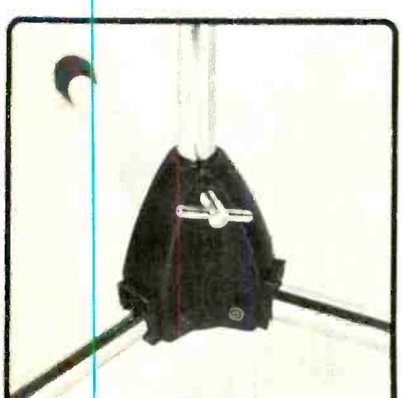
"TAMA MIKE STANDS? -- GREAT!!"

— Dave Starobin, Starr Recording



"We've used Tama drums and Titan hardware for some time now, and they've never let us down. The new line of Tama mike stands exhibits the same thought and attention to detail. Tama stands give us excellent isolation, easy and positive positioning and real stability. All I can say is, they're great!"

When you work with mike stands for a living, you begin to appreciate the fine points. Like extra thick natural rubber feet and boom washers for better isolation, nylon height adjustment bushings and extra-wide-stance legs. When you add up the fine points, they point to Tama.



Wide-stance legs for stability . . . soft rubber feet for good isolation.

The boom that stays put . . . adjusts easily . . . won't sag.

tama

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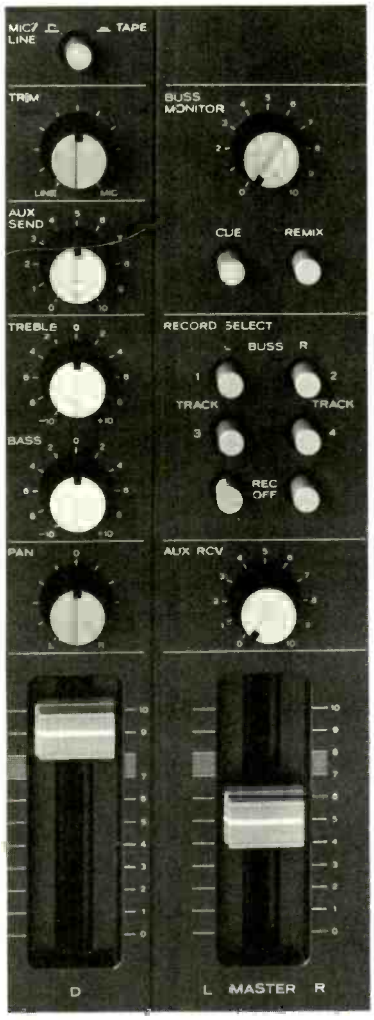
Portastudio gives you all the essential functions of multitrack recording in one compact, self-contained unit.

You can work anywhere you have an electrical outlet. Connect electronic instruments directly. And try out new musical ideas. Will a keyboard work well in the break? Do you want a chorus behind the vocal? With Portastudio, you can find out before you put out hard cash for studio time.



Whether you're recording basic tracks, overdubbing, punching-in or mixing down to another cassette deck, Portastudio helps you get all the signals to the right places.

Portastudio's versatile 4 x 2 mixer section gives you mic/line/tape switching, trim control, high and low EQ, fader, pan and Aux Send for each input. The failsafe group switching matrix lets you record on up to two tracks at the same time. And the master fader gives you overall level control during recording and mixdown.

The full-logic cue system in Portastudio lets you hear everything you're doing all the time. Input and tape cueing, monitoring for recording or mixdown are all available. And every signal can be metered. Coming or going.

TEAC engineers created a totally unique format for Portastudio. Four tracks in sync on cassette tape at 3-3/4 ips. It's fast, simple, reliable and economical.

Portastudio's drive system is built specifically for the rugged needs of multitrack recording. Transport controls are all solenoid-operated for faster, easier switching. And you get a built-in variable speed control that lets you add special effects, fix a flat note or solve timing and cueing problems.

You can work with Portastudio using nothing more than headphones and a microphone. Or send the output through your home audio system. You'll also find the patch points and controls that let you use Portastudio with other equipment like echo units, equalizers and additional mixers.

Nothing else in the world hands you so much multitrack function in such a sensible way. So see your dealer today for a demonstration of the very affordable Portastudio.

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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.*

Alternatives for Synthesizer Sorehead

It seems that every once in awhile, when recording my synthesizers, I get a deflection in the VU meters which causes a great amount of overload/distortion on my tapes.

I'm currently using an Octave Cat SRM and a PAIA Electronics P4700J synthesizer for recording. Both of these units go through Sescom active direct boxes to a Tapco 6300 stereo board into a dbx 154 unit to a Teac 2340 4-track.

An example of my problem is this: When setting up my record levels for the synthesizers, I try to hit the tape with the highest levels possible, around −3 to 0 dB. When I stay in the top two octave range of the keyboard, everything is fine. But every once in awhile, if I do a run in the lower octave range, a note seems to honk and pin all the VU needles, resulting in great pain to my ears, as well as much anger over the time it took me to set up that particular take.

I'd like to know what you think is causing this honking (that's the only word for it). The patches I'm using don't seem to be the cause of the problem as it does this on many different voice patches. How can I overcome this problem without the expense of a limiter and without a loss of quality in my recordings? Any help would be appreciated!

—Gordon M. McAlister
Mac Sound Studios
Ridgewood, N.J.

There are several questions that I would ask you if you were here with me. Do both synthesizers exhibit the same problem? Does the problem occur on runs which are played exactly as rehearsed? In what quadrant was the moon? (But seriously, folks . . .)

On the surface, this would seem to be a problem that relates to the filters in the synthesizers. While this may be somewhat basic for some readers of *MR & M*, we need to examine the basic concepts of synthesis for a moment to see precisely why.

The synthesizers with which most musicians are familiar are based on a "formant synthesis" concept. In simple terms, this means that to produce a specific tone color we start with a waveform which contains a lot of harmonics and then use filters to eliminate or modify certain of those harmonics which are not needed.

The most commonly used filter type in our field is the low-pass filter with resonance. Low-pass because it allows only frequencies below a controllable "corner frequency" (Fc) to pass. At low resonances, the higher harmonics are simply dropped out while those below the corner frequency are passed essentially without modification. At higher resonances, the filter purposely accentuates signals that are exactly at Fc so that more and more the output of the synthesizer contains only that component of the original waveform which is at the corner frequency. At maximum resonance, most filters are designed to self-oscillate and they do so at Fc.

If, due to lack of understanding or in the interest of realizing a special voice, we are driving a filter with a waveform which is fairly low in higher order harmonics (e.g., a triangle) and the Fc is set above the fundamental frequency of the waveform, there is not much there for the filter to extract. In order to compensate for this low signal level the gain has to be cranked somewhere, which is fine as long as the filter Fc is always set well below the fundamental.

But, if the oscillator frequency is raised (or Fc lowered) so that the fundamental of the waveform matches the filter corner frequency, there is suddenly something for the filter to pass. And it does, resulting in pegged VU meters, sore ears and so on.

The most obvious solution to this problem is to alter the patch or parameters so that the problem doesn't occur (perhaps changing the filter Fc or turning the resonance down). If these are not viable alternatives because they too drastically alter the intended voice, then the only alternative that I can think of is just what you said you would rather not do—use a limiter to keep the signal level under control.

—John S. Simonton
President
PAIA Electronics, Inc.
Oklahoma City, OK

"Y," Oh, "Y"

Regarding Mr. Will Parry's discussion in the November 1980 Talkback column ("An Effective Solution," page 18), I would like to caution readers about arbitrarily "Y"-ing the outputs of studio equipment. Most modern equipment uses single-ended output drivers, relying upon balanced inputs to avoid loops, etc. If you short two of these outputs to each other with a "Y" adapter, you will get some heat, smoke, and *maybe* catastrophic failure—but no music!

While there is no substitute for adequate return capability, two single-ended outputs can be summed together by wiring 620 ohm resistors in series with each output, joining the free ends

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together, and taking the signal from the junction. An alternate approach with some mix capability is to connect each end of a 1 K ohm pot to an output. Taking the signal from the wiper, you can pan from unit to unit.

I would like to reply to another comment made by Parry regarding "effects" mixing. In the delay lines that I've designed (Loft, Bozak, and Phoenix Systems), the effects mixes are constant volume. While Mr. Parry's suggestions will be fine for general delay effects such as doubling, slap or even straight flanging, it may not be possible to duplicate exactly all the effects you can get out of the unit's mix control. Without giving away trade secrets, let's just say there's more going on than just a pan for some of the "hairy" effects. Use your head. If you can't get the effect, go back to the unit's mix control.

—John H. Roberts
President
Phoenix Audio Laboratory, Inc.
Manchester, Conn.

I would like to thank Mr. Roberts for bringing to my and the readers' attention an important omission in my November Talkback reply. Due to an error in the transcription of my original and subsequent proofreading, the paragraph concerning "Y"-ing inputs and outputs was incomplete.

The last part of this paragraph should have read: "If your console has only one return, "Y" the outputs and control the level of each unit with its own input. The "Y" can take the same outward physical appearance as an input "Y" but should contain an isolation resistor in series with each "hot" line. Typically, values of between 500- and 1,000-ohm will work well. You may have to spend some time adjusting these various outputs and sensitivities to obtain your proper mix."

I had not had the opportunity to reread my response until Mr. Roberts brought it to my attention. Again, I wish to thank Mr. Roberts for his correction and apologize for any inconvenience that this oversight may have caused the readers.

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

Expansive Advice

At present, I own two Tapco 6200A stereo mixers, which I use for multitrack recording in conjunction with a

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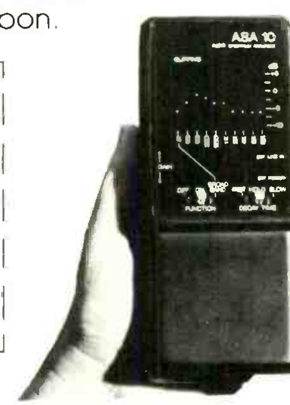

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er 7140 four-channel deck and a 24 series noise reduction unit. I am ased with the results I get with this quipment, however, I find that I need to expand the Tapco's mixing facilities to include two additional foldbacks, preferably pre/post faders, and pre-fader listening facilities.

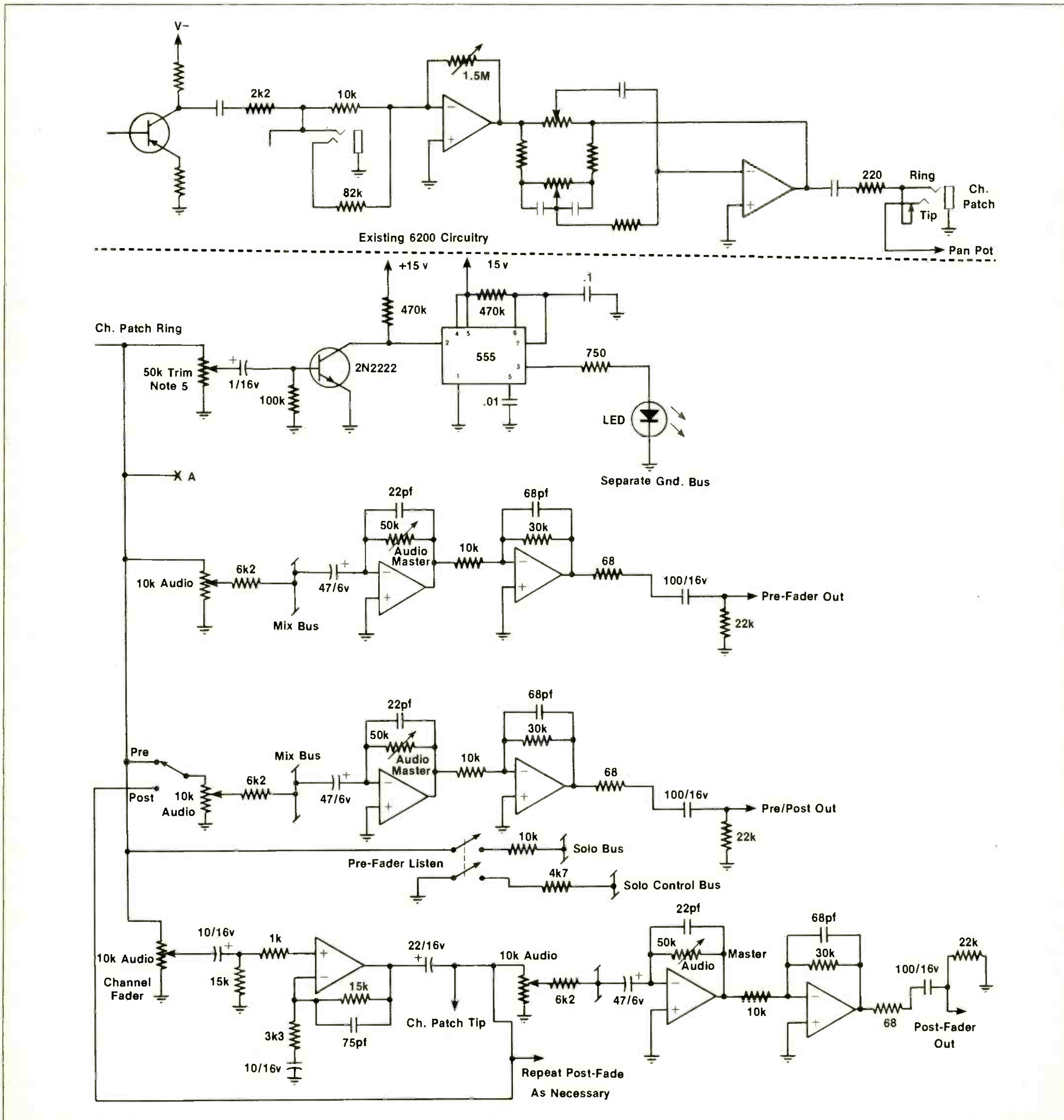
I believe the foldback circuits can be inserted in the channel patching access of the mixer without losing the patching capabilities. Would you know of good quality circuits I could build to include these extra functions without degrading the quality of the mixers?

—Ed Everest
Kingsville, Ontario, Canada

You can add the circuitry you describe in your letter.

The circuitry is simple enough so you should have no trouble constructing it. I am not showing power supply or bypassing other than a simple regulated supply. You should bypass each of the op-amps with a .01MF disc ceramic capacitor (minimum value) to the circuit ground buss.

The choice of op-amps for the circuit is very wide. You can use Signetics type NE 5532 dual op-amps or Texas Instruments type TL072 or National type LF353. These should be available from an *industrial* electronics distributor in your area (like L.A. Varah). They should be low-noise types with a minimum slew rate of 2V/usec. The 741 op-amp will not do. You could also use one of the quad types like TL084, TL074 or TL075. The



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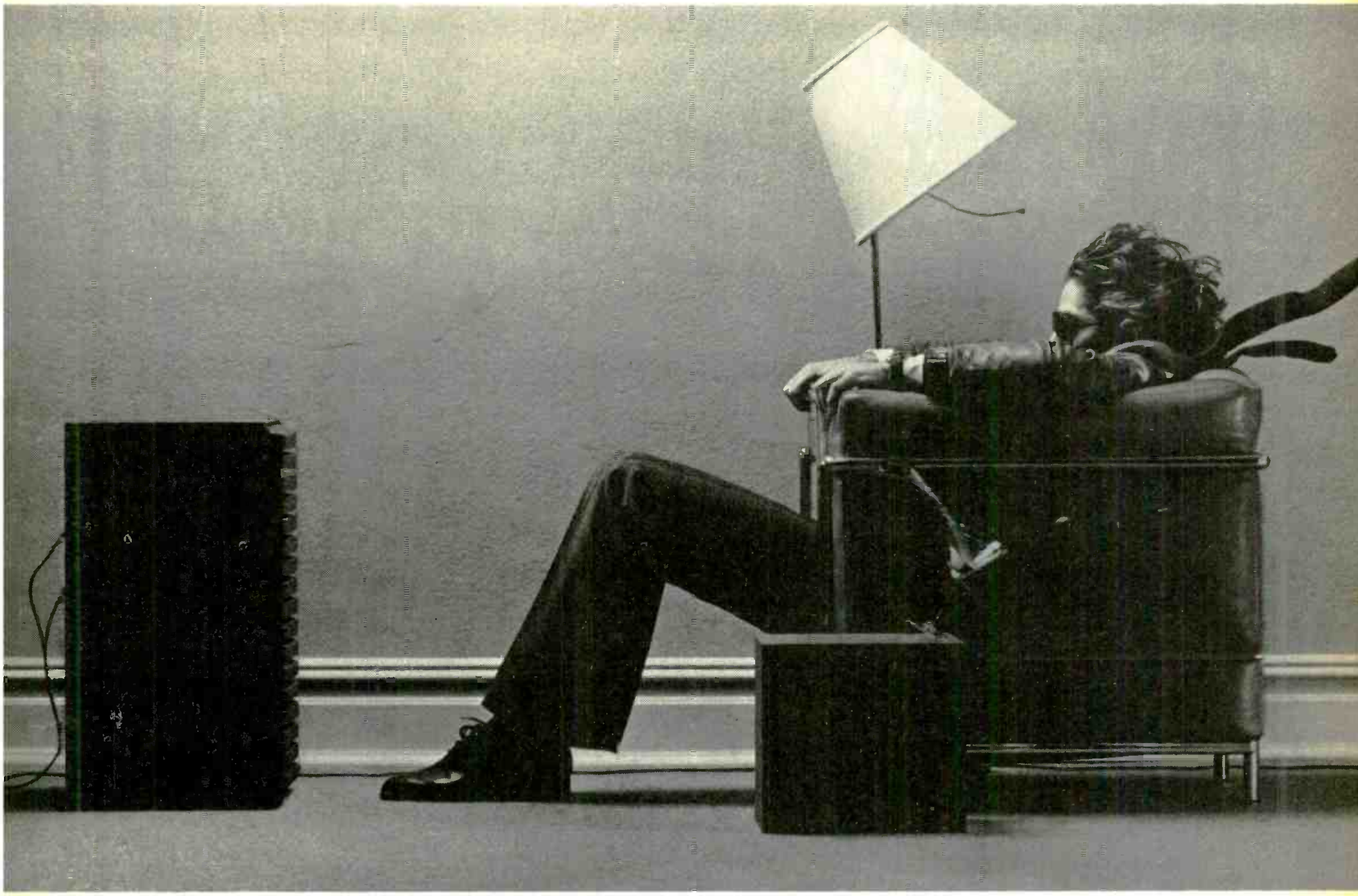

If your old favorites don't sound as good as they used to, the problem could be your recording tape.

Some tapes show their age more than others. And when a tape ages prematurely, the music on it does too.

What can happen is, the oxide particles that are bound onto tape loosen and fall off, taking some of your music with them.

At Maxell, we've developed a binding process that helps to prevent this. When oxide particles are bound onto our tape, they stay put. And so does your music.

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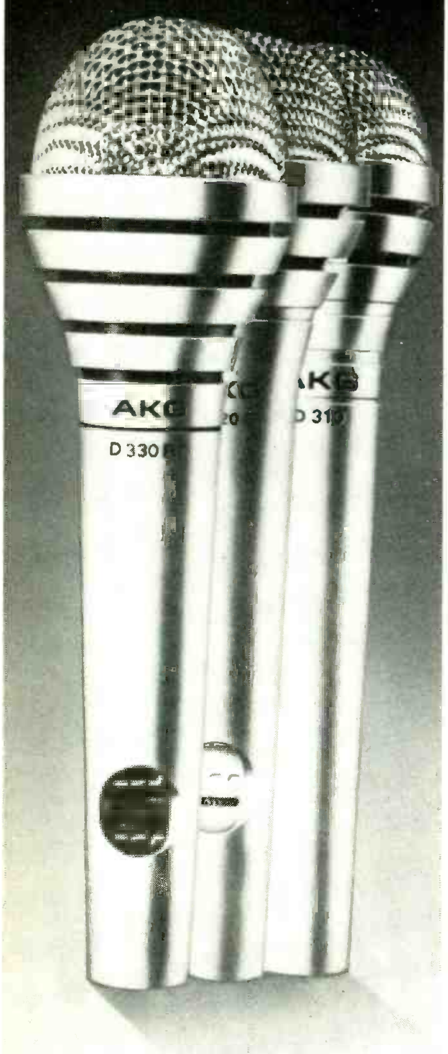
FAIL-SAFE OUT-PERFORMERS



AKG D-300 SERIES MICROPHONES

AKG ACOUSTICS, INC.
77 SELLECK STREET
STAMFORD, CT 06902

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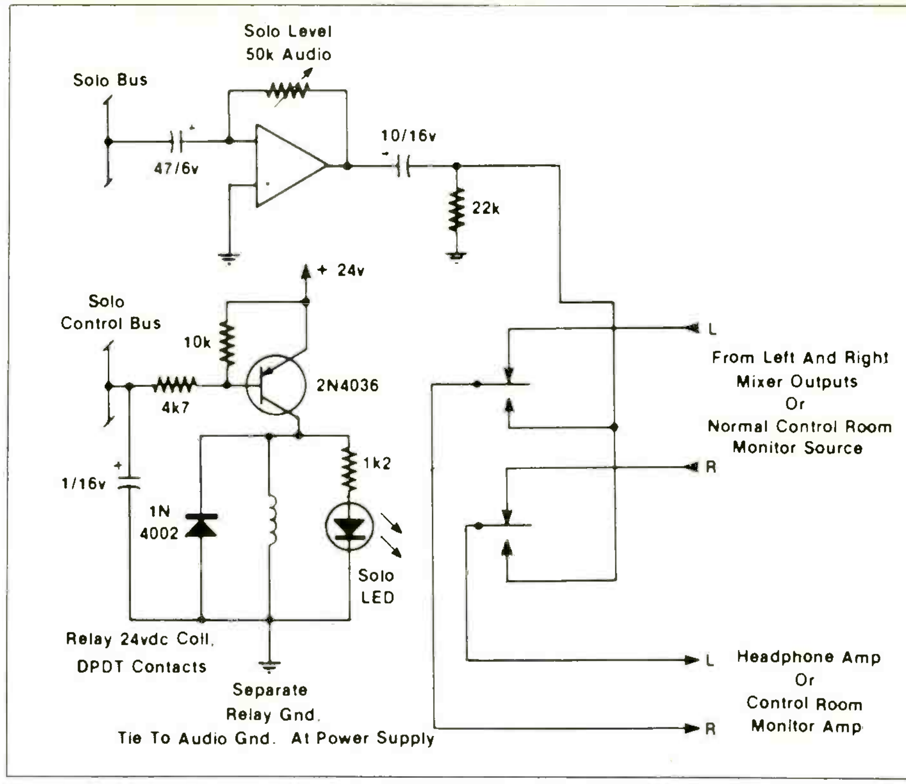


duals all have the same pin-out. The quads are different. The drawing is marked for the dual types.

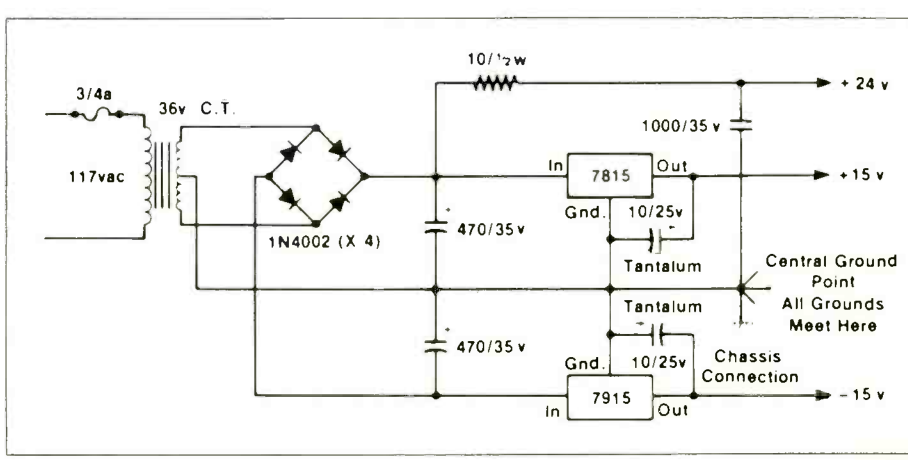
For optimum circuit performance, you will need a good grounding scheme. Contrary to popular belief, *Ground Ain't Ground*. Do *not* use the chassis. Each input channel should have its own ground buss. That buss should tie to the central power supply ground and that point should tie to the chassis. Do not run the ground wire like a snake through the circuitry.

Because of the nature of the 6200's circuitry, it is simplest to change the function of the existing channel gain control to that of a trim control. The channel fader function is then provided on your outboard unit. This could be a high-quality slide fader such as a Duncan or a Penny and Giles.

An optional channel overload indicator is shown. If you don't want it, omit the circuitry from the right of the 50 K control.

NOTES:

1. **All resistors ¼-watt, 5% carbon film (avail. at Radio Shack) except as noted. Value written 6 KHz-6200 ohm-6.2 Kohm, etc.**
2. **All op amps: NE-532—high performance 4558 (*not* 5558) med. performance Acceptable substitute for NE5532: TL072, LF353. Change 68 ohm resistor to 680 ohm. Slight loss of headroom.**
3. **7815, 7915 are on heatsink**
4. **Electrolytic capacitor values are minimum. 10/10V = 10 mfd 16 rdc.**
5. **Set trim so LED is on at +16 dBU at point A.**
6. **Power transformer manufactured by Stancor.**
7. **2N2222 can be any general purpose NPN transistor, 25 V or greater.**
8. **Every chip bypassed to ground, both power supply lines .01 mf ceramic disc.**



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When You're Serious About Your Music!

Gollehon speakers are up front with the best groups for some pretty good reasons. For one, Gollehon delivers all the raw power you need to drive home your music, with all the intensity you can give it.

For another, Gollehon speakers respond with accuracy you can count on. Regardless of the mix, how many mikes, or how hard you're pushing . . . you can rely on Gollehon to separate and define vocals, brass, and keyboards so clean, like you're in the studio, instead of on stage.

What counts is what's inside . . . Gollehon horns, drivers, woofers, and crossovers manufactured at Gollehon with the utmost concern for reliability and performance over the years. On the outside are all the roadworthy features you need. Solid-core plywood for exceptional strength and cabinet integrity. Gray fiberglass resin finish that's tougher than vinyl. Corner protectors, recessed handles, and steel grilles to protect the speakers.

Gollehon has a wide range of speaker systems for sound reinforcement, keyboard and guitar, and on-stage monitoring . . . from our new S2E, a small, light weight, affordable PA speaker to the full S-Stack of bass bins, mid-range and high-frequency sections. And of course, our 280 (pictured) has been around a long time. One of the best 2-way full-range systems you'll find anywhere. The Penetrater 3 (pictured on top of the stack) is a different speaker indeed. Horn-loaded with a 10-inch cone driver and separate high-frequency horn molded into the mid-horn's side walls for full-range reinforcement or as a top section in conjunction with a separate bass bin.

It sounds so simple to say Gollehon sounds good, but it's really that simple. Most bands buy Gollehon because they like what they hear.

Write or call for product information and list of Gollehon dealers. Gollehon Industries, 2431 Clyde Park, S.W., Grand Rapids, MI 49509, 616—247-8287. In Canada: 5980 Westbury Ave., Montreal, Quebec H3W 2W9. © Gollehon Ind., Inc. 1981.

Band: CABARET Club: CORAL GABLES, Grand Rapids, MI

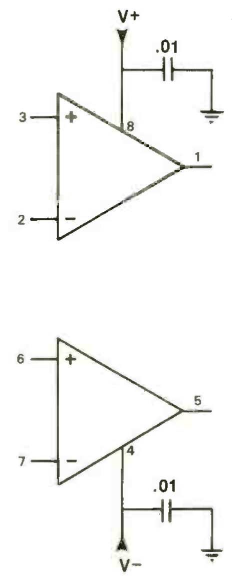
GOLLEHON

CIRCLE 161 ON READER SERVICE CARD

The PFL (pre-fader listen) circuitry is designed to transfer your phones or control room monitors from what they normally monitor to the PFL solo buss.

You can add more pre or post sends (five maximum pre or post). Just add the circuitry to the right of each 10 K pot. You will need one 10 K pot and one 6200 ohm (6 KHz) resistor per input. The circuitry to the right of the mix buss is common to all inputs and only needs to be built once per send.

—Rick Chinn
Product Specialist
Tapco, Inc.
Redmond, Wa.



NE5532, LF353, TL072, 4558

No Problem With Team Work

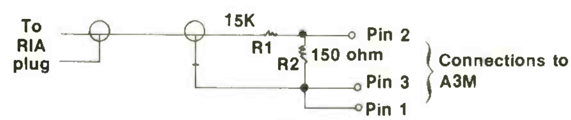
I have just purchased a Tapco 6100 RA six-channel mixer with an expander (the 6100EA), giving me a total of 14 channels. My question is: How can I use my Tapco as a mixing board with a Teac 3340S four-channel? I also use a Kustom X Power Head (stereo) with four Kustom Pro 3W cabinets for PA.

Is the teaming of my two machines only possible with transformers that change impedance? Should my mic's impedance be matched to my Tapco or my Teac?

—Larry R. Pugliese
Utica, N.Y.

In order to use your Tapco 6100RA with your Teac for mixing, you need to match impedance and signal *level.* The level matching is far more important than the impedance part. This could be accomplished with transformers; however, the signal level would still be excessive. Also, transformers are expensive. Instead, do this simply by building four resistive bridging pads.

The pad can be built inside of a Switchcraft adaptor case. If you are clever, you can build it inside of a Switchcraft A3M plug. You will need



Approx. loss $= 20 \log (1+\frac{R1}{R2}) = 40$ dB

The Orban 672A:
A Dream Equalizer at a Practical Price

The 672A is a single-channel equalizer offering astonishing control and versatility. There are eight non-interacting parametric bands with reciprocal curves and the convenience of graphic-style controls. Highpass and lowpass filters with 12dB/octave slopes that tune continuously over a 100:1 frequency range. And, separate outputs that let you use the 672A as an eight-band parametric cascaded with an electronic crossover in reinforcement and monitor tuning applications.

The dream equalizer is usable practically *everywhere* in professional and semi-professional sound: recording studios, cinema, theater, reinforcement, broadcasting, disco — you name it! Yet its price is down-to-earth. And, it's built to full professional standards.

Check it out at your Orban pro-audio dealer.

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San Francisco, CA 94107
(415) 957-1067

CIRCLE 100 ON READER SERVICE CARD

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"Listening tests confirmed what the excellent measurements implied: the Eumig FL-1000 is a superb performer."

JULIAN HIRSCH—STEREO REVIEW, APRIL 1980

What you are about to read is Julian Hirsch's unedited conclusion in his review of the Eumig FL-1000.

"Listening tests confirmed what the excellent measurements implied: the Eumig FL-1000 is a superb performer. Dubbing from FM or phono discs revealed no audible differences between the original and the copy, and even FM interstation noise—our most severe test—could be recorded and played flawlessly up to levels of approximately -5 dB. The Computest adjustment for different brands of tape was not only accurate but contains a built-in rewind mechanism that returns the tape to the precise point where you began your adjustment. The counter was the most accurate we have ever used. And for people who are "into" computers, the one-of-a-kind (so far) Eumig FL-1000 cassette deck opens up endless possibilities."

We couldn't have said it better. We wouldn't even try. For the complete text of the review, write to us. Or, better yet, visit your nearest Eumig dealer and find out for yourself what it takes to make a reviewer rave.

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Eumig (USA) Inc., Lake Success Business Park, 225 Community Drive, Great Neck, New York 11020, (516) 466-6533

CIRCLE 149 ON READER SERVICE CARD

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to find ⅛- or ¼-watt resistors to do this, however. The wattage of the resistors is absolutely not critical (don't you wish everything was?). Also keep in mind that the mics should "match" the Tapco (low-Z balanced, 150 ohms).

—Rick Chinn
Product Specialist
Tapco, Inc.
Redmond, Wa.

Keeping It Crisp

Why is it that high frequencies get lost whenever tracks are mixed together? Whenever I mix tracks down, the final product seems to lose its "crispness." I don't think it's just my equipment (4-track); I've often heard this happen to some extent, even when a stereo setup is switched to mono. Why? Any idea what I can do about it, short of emphasizing the highs, and therefore the hiss as well, on each track?

—John Bartelt
Atascadero, Ca.

I'm not sure what situation you're describing by your phrase "tracks mixed together." Do you mean bouncing tracks within the 4-track machine, or your final mix from 4-track to stereo? Let's first assume that you mean bouncing tracks in sync within your 4-track, i.e. the lead vocal on track 1 and the harmony vocal(s) on track 2 "mixed" together and recorded onto track 4. In order for the entire recording to remain "in sync," you must be playing tracks 1 and 2 off the record/sync head during the bounce procedure. On many ¼-inch 4-tracks, the frequency response of the record/sync head is not nearly as good as the response of the playback head itself.

I'm afraid that in this situation, apart from getting an optimum record level, and the conservative use of equalization, you will always be limited by the playback frequency response of the record/sync head. However, it should be noted that when you are mixing *three* tracks down to one, there is no need to use the sync mode.

If, however, you were originally referring to mix-down from 4-track to stereo reel-to-reel or cassette, you should first check the alignment and bias of your stereo recorder, and *clean those heads!*

Certainly with every generation of tape the audio signal is degraded to a certain degree, but with properly maintained equipment the top to bottom balances should remain intact. The noticeable changes will occur in the areas of transient response and signal-to-noise ratio.

As far as your relating these sub-mix problems to stereo or mono imaging—that's a tough one. In mono, the listener differentiates one instrument from another by the relative volume levels and places in the frequency spectrum. Stereo offers the listener another dimension. Being able to hear an instrument as left, center or right, helps distinguish that instrument from others with similar tone colors and volume levels.

The loss of "crispness" that you refer to when you switch from stereo to mono may be the loss of left to right imaging as a tool in defining instruments and voices in a mix. This stereo-mono effect and your mix-down problems seem unrelated, so clean and align your deck, check phasing on your monitor speakers and try again. Good luck!

—Kevin Kelly
Engineer/Co-owner
The Workshoppe Recording Studios
Douglaston, N.Y.

SHOCK ABSORBERS
Transit Cases from ANVIL®

The next time your instruments and sound gear hit the road, protect them with The SHOCK ABSORBERS™ from ANVIL®. To give your gear the extra protection it needs when it's on the road, ANVIL®'s SHOCK ABSORBERS™ Transit Cases are built to exceed rigid specifications established by the Airline Transport Association to assure safe transport of delicate equipment.

Outside, your gear gets the protection of sturdy, top-grade plywood sidewalls covered with gouge and scuff-resistant ABS plastic. Plus a strong, lightweight aluminum outer frame. And ANVIL® machine-driven, split-steel rivets and steel hardware.

Inside, your gear is cushioned with a custom-fit protective layer of foam padding. So even if your SHOCK ABSORBERS™ shake, rattle and roll on their way to the gig, your gear inside won't.

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CIRCLE 45 ON READER SERVICE CARD

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The logic behind the Revox B77.

The logic is the logic which is built-in.

It's an ingenious and highly sophisticated system—much like the human nervous system—which controls the deck's functions.

You can push any button in any order with no chance of damaging your tapes. Our motion sensing system constantly feeds status reports to the logic circuitry which activates your commands in proper sequence.

The logic also permits full-function remote control, and an editing mode that keeps the playback circuitry live, even when the motors are stopped. You can make your splices right on-the-beat, and our built-in splicing block makes it easy.

The design and construction of the Revox B77 further guarantee smooth and accurate operation. To get the long-life advantage of ferrite without static build-up or heat degradation, we use Revox's exclusive Revodur heads, made of metal to dispel heat and static, and vacuum-coated with permalloy for durability.

The B77 has a unique capstan motor that's monitored by a tacho head to precisely control speed and limit wow and flutter to professional studio standards.

Revox offers many options with the B77 including a full range of speed configurations from 15/16 IPS to 15 IPS, variable speed control, ¼ track record/playback and more.

All this professional quality is neatly engineered to fit in a deck you can carry. After all, if you own a machine this good, it's logical to take it with you.

Experience the B77 and the full line of Revox audio components at your franchised Revox dealer today.

STUDER REVOX

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Offices: Los Angeles (213) 780-4234 / New York (212) 255-4462. In Canada: Studer Revox Canada, Ltd.

CIRCLE 91 ON READER SERVICE CARD

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THE PRODUCT SCENE

By Norman Eisenberg

PARAMETRIC EQUALIZER

An eight-section full parametric stereo equalizer—the model GEM-7—has been announced by Stereographic Concepts, exclusive agency for Superex Electronics. The device features four response-shaping frequency banks per channel. The four frequency sections in each channel are paired to adjust the response regions of 30 Hz to 820 Hz, and 820 Hz to 16 kHz. Though they are complementary overlapping in response coverage, the sections perform logarithmically in that each frequency control has greater control over a specific portion of that range. Each frequency bank control has a gain control providing up to 18 dB of boost or attenuation, as well as a variable bandwidth function for 0.16 to 2 octaves. Bi-Fet circuitry is employed for lowered noise and distortion, and high stability. By cascading this circuitry for each band, as opposed to the conventional method of summing the filters, the design is said to virtually eliminate ripple effects at maximum boost, and completely so at flat response settings. Full tape functions are handled by separate switching for record EQ, playback EQ and standard tape-monitoring. Recommended applications include notch filter, room acoustics compensator, EQ curve modifier, instrument intensifier and noise-reduction. Retail price is $449.95.

CIRCLE 18 ON READER SERVICE CARD

REVOX ENTERS CASSETTE FIELD

The model B710 is the first cassette deck to be offered by the well-known firm of Studer Revox, a manufacturer best-known for its pro-grade open-reel decks. Drive-belts, pulleys, friction clutches and mechanical braking are all eliminated in the B710. The dual-capstan system is direct-driven by two separate magnetic disc drive motors, slaved to a common quartz-crystal reference frequency. The two spooling motors also feature direct drive, and they use optical tachometers in a servo system for constant tension and fast-wind speeds. Says Revox, this not only produces tape-handling "comparable to the finest open-reel designs, but permits the use of a controlled electrical braking system as well."

The deck uses separate record and play heads, mounted on a common alignment plate. Pneumatically-damped solenoids ensure against tape-gate shock. All transport functions are microprocessor-controlled. An internal timer is included. The deck has metal tape capability, Dolby-B, peak-reading LED-bar meters, modular construction, mic/line mixing, automatic or manual bias and EQ adjustments and optional rack-mounting.

CIRCLE 33 ON READER SERVICE CARD

BIB VCR ERASER

A hand-held bulk eraser for video cassette tapes has been announced by Bib, the well-known accessories manufacturer. The new VE-3 is designed to exceed the capability of the erase heads found in today's VCRs and thus ensure that video tapes are signal-free and ready to accept new recordings. The Videophile Edition units' gauss magnetic field far exceeds the erasure capability of erase heads found in today's video recorders. The hand held eraser has a safe thermal protected circuit. Price: $47.50

CIRCLE 20 ON READER SERVICE CARD

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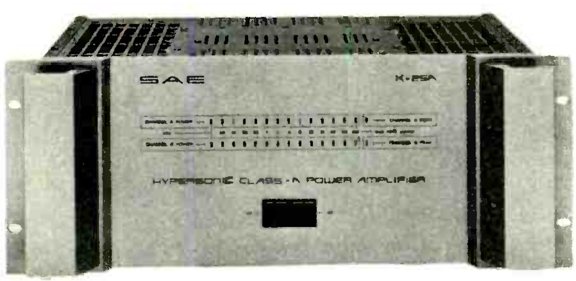
SAE CLASS A AMPLIFIER

Now available from SAE is the new X25A "Hypersonic" Class A power amplifier, rated for 250 watts per channel. The device is said to generate only a fraction of the heat usually resulting from conventional class A design. The X25A incorporates complementary-symmetry circuit design. Each channel employs two independent amplifiers. One handles the positive slope of the waveform, while the other handles the



negative slope. Since both amplifiers operate at all times, any nonlinearity in the output of either is minimized without the need for feedback correction. The design also uses a linear open-loop bandwidth extending beyond the range of audibility. This technique is said to eliminate the need for feedback in controlling frequency or signal level. Feedback is used only to control gain. The panel contains a display made up of separate 15-LED power indicators for each channel. Price is $1500.

CIRCLE 21 ON READER SERVICE CARD

MXR DUAL LIMITER

The new Dual Limiter from MXR Innovations, Inc. functions like two independent limiters that may be strapped together via front-panel switching for stereo applications. Each channel has its own set of controls for in/out; slope; input; output; attack; release; and LED metering. Each channel's detector is accessible via 1/4-inch jacks at the rear. This feature, explains MXR, allows for external tailoring of the detectors' frequency response, for reducing vocal sibilance and to serve other frequency-dependent limiting needs. Rack-mountable, the Dual Limiter is fitted with balanced inputs, can drive 600-ohm loads and is rated for output of +19 dBm.

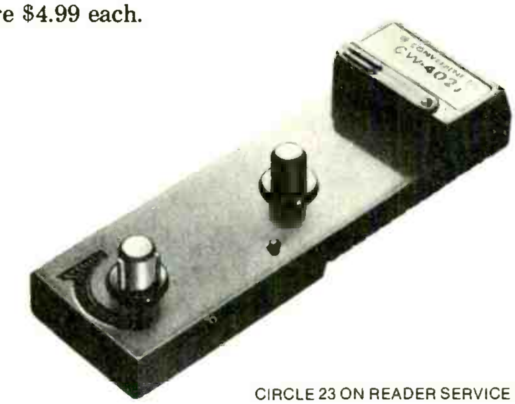
CIRCLE 22 ON READER SERVICE CARD

NEW LINE OF CASSETTE ACCESSORIES

Osawa has introduced a line of cassette recording accessories under the brand name of Nagaoka. The CW-402 is a battery-operated "pocket" cassette winder which rewinds a 60-minute cassette in 35 seconds and then shuts off automatically. The price of the CW-402 is $19.99.

The QC-209 is a head-cleaning cassette priced at $7.99. The PC-507, listing for $24.99, is described as a cassette repair and maintenance kit. It includes a splicing block with 60- and 90-degree cutting slots, tape hold-downs, scissors, tweezers, Phillips and conventional screwdrivers, splicing tape, sensor tape, a tape probe, replacement pressure pads and replacement screws.

Also offered are an index label book with 100 replacement labels and color-coded identification tabs for cassettes, and an index card book with 24 color-coded cassette replacement liners and self-adhesive ID tabs. The prices for each of these items are $4.99 each.



CIRCLE 23 ON READER SERVICE CARD

SHARP MARKETING VCR

The Sharp VC-7400 is a newly designed video cassette recorder with features that, says Sharp, have been until now associated with "deluxe" units. These features include a fully automatic front-loading tape system, touch-button electronic tuning with AFT, soft-touch solenoid controls, tape-remaining LED indicator, a 24-hour clock-timer with automatic stop.

Using the VHS format, the VC-7400 has six-hour record/playback capacity. Price is $895.

CIRCLE 24 ON READER SERVICE CARD

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PEAVEY'S MARK III MIXING CONSOLES

Said to be "new and radically different," the Mark III sound-reinforcement mixing consoles from Peavey Electronics of Meridian, Miss. come in 12-, 16-, and 24-channel versions. Each channel has balanced input circuitry, switchable 48-volt phantom power, two independent pre-monitor sends, 4-band EQ, two post-effects sends PFL/cue button, LED status indicators, pre- and post-send and return. The console also includes a headphone system to allow monitoring of individual channels in any mode or any monitor or mains signals. Five transformer balanced outputs are provided: two mains, two monitors and sum. The signals from the effects and reverb returns can be blended into both monitor mixes. Mark III consoles are available in a rugged "flite case" or in polyester fiberglass end-panel construction.

CIRCLE 25 ON READER SERVICE CARD

THREE-RANGE EQUALIZER

From Modular Audio Products of Bohemia, N.Y. comes word of its new Model 3550, described as a three-range, 21-frequency, reciprocal 12-dB boost or attenuate equalizer. High and low-range EQ curves may be selected independently as either peaking or shelving. A band-pass (50 Hz to 15 kHz) filter may be inserted exclusive of all other EQ settings. The three frequency ranges (low, mid and high) are overlapping and are controlled by the outer knobs of concentric rotary switches, whose inner knobs set the amount of boost or cut in steps of 2, 4, 6, 9 and 12 dB. Designed for panel-mounting, the model 3550 requires a bipolar 15-volt DC supply.

CIRCLE 26 ON READER SERVICE CARD

LITERATURE OFFERINGS

A Replacement Parts Handbook for consumer electronics service technicians has been published by the Parts Subcommittee of the Electronic Industries Association's Consumer Electronics Group Service Committee. Its 69 pages offer a comprehensive inventory control and ordering system, including necessary forms and order sheets. Also included is a list of locations where parts are available.

CIRCLE 27 ON READER SERVICE CARD

A four-page brochure describing tape reproducer calibrators, service aids and illustrative materials tailored to the needs of the magnetic recording industry is offered without charge by writing to R. K. Morrison Co., 819 Coventry Rd., Kensington, Ca. 94707.

CIRCLE 28 ON READER SERVICE CARD

The latest issue of "MXR Discussion +"—issued by MXR Innovations, Inc. of Rochester, N.Y.—contains material on new products as well as a very informative, illustrated section on flanging and distortion.

CIRCLE 29 ON READER SERVICE CARD

YAMAHA POWER AMP

Offering 200 watts per channel output, Yamaha's new B-6 stereo power amplifier boasts a uniquely compact truncated or "pyramid" appearance. Circuitry is said to incorporate two technological innovations. One is the "X" power supply, described as compact, without the bulk common to high-powered amplifiers while still providing "unsurpassed power regulation." The other is the "X" amplifier which avoids the need for large heat sinks by delivering power output stage power at low levels-high and low voltage. The level of the audio signal is monitored and the amplifier switches itself automatically to high or low voltage as required. Longer component life and improved dynamic range are among the claims made for this unit.

CIRCLE 30 ON READER SERVICE CARD

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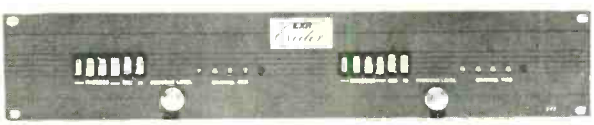
EXCITER PROCESSES AUDIO

Designed for studio, disco, "live" sound and home entertainment use is the Exciter Model SP1 announced by the EXR Corporation of Ann Arbor, Mich. The device is described as a "practical realization of a psychoacoustic audio processing system capable of accurately restoring the natural characteristics of an amplified audio signal." Says EXR, the processing involves preselective 180-degree phase notching; time manipulation; frequency manipulation; and psychoacoustic juxtapositioning.

The first three functions create an "interference signal" which, applied to the original signal, reverses the primary of fundamental buildups and losses caused by multiplier effects and distortion in the audio



reproduction chain. The phase-notching is used to cancel specific frequencies where distortion tends to build up. Volume level is not affected since the device cancels as much as it adds.

The "psychoacoustic juxtapositioning" is EXR's process for filling the holes left by the phase notching. "By using beat pulsing and other long-known but little used processes," says EXR, "information is extracted from one part of the frequency spectrum, processed, then used to sonically replace another part of the spectrum which has beeneliminated or lost." Compatible with previously enhanced program material, the Exciter is claimed to create—for a variety of sound-reinforcement and P.A. setups—a greater intelligibility.

Rack-mountable, the unit has unbalanced inputs and outputs. Each channel's independent controls include mix/bypass select, process level and LED indicators. Price is $695.

CIRCLE 31 ON READER SERVICE CARD

ANTI-NOISE CAMPAIGN

The hot news from Dolby Laboratories right now is the new Dolby-C system, claimed to provide 20 dB of noise reduction above about 1 kHz. By way of reminder, the familiar and widely used Dolby-B system provides 10 dB of noise reduction above about 4 kHz. The new Dolby then not only adds another 10 dB of clean headroom, but it extends the benefit downward on the musical scale by two octaves more. In adition, Dolby-C is said to have a further benefit of reducing high-frequency tape saturation.

According to Dolby, the C-type system will "probably be most appreciated by those who record material of wide dynamic range and listen at loud volume levels." Assuming the use of good tape formulations, Dolby-C is expected to result in tape noise that will be below that of virtually any recorded or broadcast source, and indeed "is often below the ambient noise level of many home listening rooms."

Recorders incorporating Dolby-C will also have Dolby-B, and the latter will be selectable on a switch. Dolby-B encoded tapes will be reproduced properly on Dolby-C decks. At the same time, Dolby-C encoded tapes will be playable on Dolby-B decks without distortion or pumping effects. And, says Dolby, the new tapes also will be playable on "limited-range portable and car stereo players without noise reduction."

No one knows yet how much Dolby-C will add to the cost of a cassette deck. It will add something to the price but since the Dolby-C probably will be available as a "chip" (integrated circuit), the added cost is not expected to go sky-high. Dolby-C encoders also will be made available for use by tape duplicators in producing "prerecorded" cassettes.

Simultaneous with this news is the announcement from Matsushita that this Japanese giant firm has signed a licensing agreement with dbx to start using dbx noise reduction in Technics brand cassette recorders. The new decks also will include a switch to permit playback of dbx-coded discs. Reportedly, Matsushita chose dbx from among five competing noise-reduction systems not only for its performance, but also out of considerations of cost and ease of operation on the part of the user. Other firms already employing dbx noise-reduction are Teac and Marantz, although only the Technics (so far) provides the dbx disc playback capability.

Of course, there are some who say that when full digital arrives, it will wipe out all present-day noise-reduction systems despite their ingenuity. That day may be some years off yet. In the meantime, recordists can pick and choose from several really good systems as the analog anti-noise campaign continues in full swing.

CIRCLE 32 ON READER SERVICE CARD

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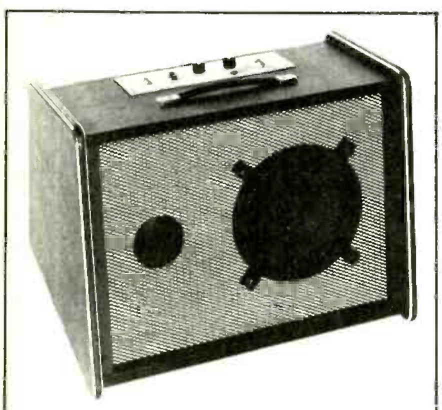
MUSICAL NEWSICALS

MUSICAL INSTRUMENT AMPLIFIERS

Lectrosonics, Inc. has introduced a portable, rechargeable battery-powered amplifier for bass guitar to follow up on its success with a portable guitar amp known as the "Mouse." The new Bass guitar model is known as the "Moose" and weighs in at twenty-nine pounds with battery pack. The unit uses a special long-throw 8-inch speaker in a tuned, ported enclosure to give response down to the low E string of a bass. A unique feature is that the Moose can double as a stage monitor by switching out its built-in bass equalization and connecting it to a mixer. The Moose runs on AC power as well as its rechargeable batteries.

CIRCLE 1 ON READER SERVICE CARD

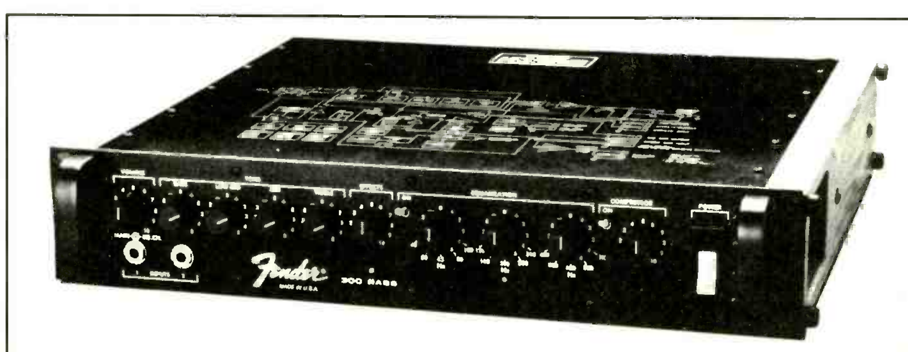


Polaris Electronics Corporation offers a full range of solid-state musical instrument amplifiers in two model lines, the Nova line of basic amps and the Pulsar line of amps with effects and reverb. All Polaris amplifiers employ dual-slope protection circuits for reliability. Features common to all Polaris models include three-band active equalizers with ±15 dB range, a Drive control to vary the degree of distortion independent of volume, a preamp output for feeds to slave amps or P.A. systems, a headphone jack which automatically disconnects the speaker outputs and an LED clipping indicator. The Pulsar models feature reverb plus the exclusive Pulsar effect which is described as a combination of phasing and flanging giving a full Doppler phase shift effect. A variety of models are available, ranging from a 12-watt 1 x 8" amp to a 75-watt 2 x 12" combo and various piggyback configurations.

CIRCLE 2 ON READER SERVICE CARD

Fender recently introduced two new amplifier products, a 300-watt RMS bass amplifier head known as the Fender 300 Bass, and a new 30-watt tube combo amp. The Fender 300 is a 3½ inches high rack-mount package with exceptional capabilities in signal processing. Equalization is particularly flexible with three bands of semi-parametric (sweepable frequency) EQ in addition to four bands of conventional tone control EQ. The amp also has a built-in compressor/limiter with threshold control and bypass switch and an effects loop which will accommodate low or high impedance accessories and has a front panel gain control for level matching. An additional feature is that the unit is bi-amp ready with a built-in electronic crossover with four selectable crossover frequencies. The Fender 30 tube combo amp combines the sound of a tube amp with its wide range of expression with modern features and capabilities. The Fender 30 is a two-channel amp with silent instantaneous switching between channels using solid-state switches activated by a remote footswitch. The normal channel has two input jacks, bright switch and volume, bass and treble controls. The reverb channel has two inputs, a channel switching enable switch, preamp gain with bright boost, bass, mid and treble controls each with a pull-for-boost, and a reverb control. Other features include a master volume control, a rear panel line-out jack with volume control, effects in/out jack and controls for output tube matching and hum balance. The Fender 30 is available in 1 x 12" or 2 x 10" configurations.

CIRCLE 3 ON READER SERVICE CARD



Yet another entry in the mini-amp market is the MiniAmp from Zeus Audio Systems. The Zeus unit is 7½" x 3" x 1¾" and provides 1 watt of clean power or 2.5 watts of overdriven power. The Mini Amp will also function as a preamp for use with a larger amp, a preamp for acoustic transducers or to drive an 8-or 16-ohm speaker cabinet to

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modest levels directly. The power switch is contained in the input jack, while controls are provided for preamp gain, tone and master volume. The unit's housing is of aluminum and the unit is powered by eight AA batteries.

CIRCLE 4 ON READER SERVICE CARD

Synthesizers have always been among the hardest instruments to amplify properly, but with the introduction of their 16-voice electronic piano, the engineers at ARP found that the amplification problem was even more acute due to the accuracy and high quality of the sound of their new instrument. A number of sound reinforcement systems were found to be adequate to the task, but the ARP engineers decided that it was going to be up to them to design a compact, portable speaker system that could do justice to the ARP Electronic Piano. The result is a 75-watt-per-channel stereo amplification system designed to take advantage of and exactly complement the characteristics of both the 4-voice and the 16-voice ARP pianos. The cabinets are designed to fit under the instrument when space is limited or to be used as satellite speakers when space permits. The amplifier has a microphone input and auxiliary inputs for additional instruments, and line level outputs for connection to additional amps or P.A. systems.

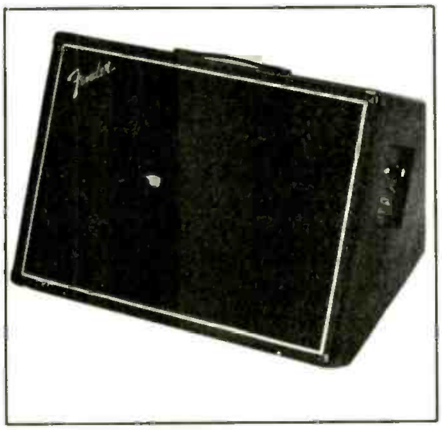
CIRCLE 5 ON READER SERVICE CARD

SPEAKERS AND SPEAKER SYSTEMS

Over the years one of the most popular groups of loudspeakers for musical instrument and sound reinforcement applications was the Electro-Voice SRO series which was updated and re-designated as the EVM series a few years ago. Electro-Voice recently announced the addition of a new 12-inch speaker specifically designed for lead guitar use, expanding the EVM line to five models. The new model is designated the EVM-12S and features a die-cast aluminum frame ½-inch shallower than the popular EVM-12L speaker which remains in the EVM line. The practical consequence of the shallower frame is a lighter, shallower cone with significantly more output in the 2-to 3-kHz range for improved brilliance and bite in guitar amplifiers. The EVM-12S has a massive magnet/heat sink assembly, a beefed-up voice coil and beryllium copper flatwire voice coil leads which contribute to a power handling rating of 200 watts continuous, making the EVM-12S even more blow-out proof than the reknowned E-V SRO speakers. The EVM-12S has high sensitivity, producing 91.5 dB SPL at 10 feet from a 1-watt input, and is rated at a nominal impedance of 8 ohms.

CIRCLE 6 ON READER SERVICE CARD

Fender has expanded its line of Professional Sound Products with the introduction of the Fender Stage Monitors, a compact, two-way speaker system in a multi-tilt cabinet allowing placement at 30, 60 or 90 degree tilts. The system design includes a 12-inch woofer and a 4" x 10" horn with compression driver for the high frequencies. The system crossover uses 18-dB-



per-octave filters and has overload protection for reliability, and has a high-frequency level control for flexibility. The Fender Stage Monitor has high sensitivity, producing 103 dB SPL at 1 meter with a 1-watt pink noise input, and high power handling capability with a 75-watt RMS rating. System impedance is 16 ohms, allowing several speakers to be connected to a single amplifier channel safely.

CIRCLE 7 ON READER SERVICE CARD

Cerwin-Vega currently has two stage monitor models, the SM-12 and the SM-15. Both models are two-way designs using the Cerwin-Vega H-25 high frequency compression driver/horn; the SM-12 uses a 12-inch ER124 drive for the low end and the SM-15 uses a 153EV 15-inch driver with 3-inch voice coil. Both models feature a minimum of midrange peakiness in the frequency response, and high-frequency response that rolls off smoothly off-axis to help minimize feedback problems on stage. The systems are housed in multi-angle cabinets allowing tilt angles of 20, 45 and 55 degrees from vertical; the cabinets are covered in the familar Cerwin-Vega grey carpet and feature recessed handles and expanded steel grilles for roadability. Sensitivity of both models is about 100 dB at 3 feet with a 1-watt input, resulting in maximum output levels of 120 dB at a rated 100 watts for the SM-12 and 123 dB at a rated 150 watts for the SM-15. Both have a nominal impedance of 8 ohms.

CIRCLE 8 ON READER SERVICE CARD

SYNTHESIZERS AND KEYBOARD INSTRUMENTS

New England Digital Corp. recently introduced a new digital synthesizer system called the Synclavier II which has been generating substantial interest among musicians. What makes the Synclavier II different from the various other high-level digital synthesizers is that it was designed with musicians rather than computer buffs in mind. Unlike most digital synthis which are programmed from what is essentially a computer terminal, the Synclavier II is programmed and controlled entirely from a synthesizer keyboard unit which is labeled in terms comprehensible to any musician familiar with analog synthesizers. The Synclavier II system comprises several component parts: the keyboard/controller unit; one (or optionally two) floppy disc drive which mounts below the keyboard unit for accessibility; two custom-built Morley foot pedal units with six foot switches; and the computer unit itself which has no controls other than the keyboard unit connected to it via heavy-duty multi-pin connectors. Synclavier II has a 5-octave, 61-note keyboard which is optionally velocity sensitive to control the volume, harmonic structure, envelope or frequency modulation of the note. Directly above the keyboard is a ribbon controller which can be used to shift the pitch or control the brightness of the note played in either upward or downward directions from whatever point the ribbon is first touched. The Synclavier II uses a unique system of tone synthesis called "partial timbres." Each partial timbre consists of twenty-four separately adjustable harmonics; an amplitude envelope generator; a har-

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monic envelope generator; fully adjustable vibrato, fully adjustable portamento; and special effects. Up to four independently adjustable partial timbres can be triggered from a single key on the keyboard. The system is available with 8, 16, 24 or 32 voices each of which has a corresponding partial timbre. The system comes pre programmed with sixty-four programmed sounds, over three dozen of which are the sounds of real instruments, any of which can be recalled with the push of a button and played as programmed or modified at will. The modified programs or newly-programmed sounds may be stored on floppy diskettes by the Synclavier II for truly unlimited program storage capacity; sixty-four complete programs may be stored on each diskette for later recall. The Synclavier II system has a large number of unique features and functions which are beyond the scope of this space, but prospective purchasers should be forewarned that such sophistication does not come cheap: the prices for the Synclavier II are pretty much on a par with the prices of new Porsches.

CIRCLE 9 ON READER SERVICE CARD

A different approach to a high-end computer-controlled synthesizer system is exemplified in the Audity system recently announced by E-mu Systems. The Audity system is unusual in that the computer controls analog voice cards which are literally complete synthesizers in themselves so that different sounds may simultaneously be assigned to different synthi channels. The system comprises a central computer with dual floppy disc drives; a 16-channel polyphonic keyboard/sequencer with its own floppy disc; a remote computer programming console; and up to sixteen synthesizer voice cards. Each voice card contains two VCOs with pulse width modulation; 24-dB-per-octave highpass and lowpass filters; a multimode resonant VCF; a VCA with linear and exponential response;four ADSR envelope generators with delay; a noise generator; an LFO; four modulation buses; and the circuitry necessary to allow computer control of all the operating parameters. Due to the sophistication of each voice card, the Audity system is *very* expensive, with a full 16-voice system pushing $70,000.

CIRCLE 10 ON READER SERVICE CARD

Moog Music has announced the introduction of Opus 3, a preset polyphonic synthesizer with string, organ and brass voices. The string voice has its own EQ section with a three-mode filter with variable frequency and resonance, while the organ voice provides a mix of five footages plus a tone filter. The strings and organ may be routed in any mixture through a chorus circuit with variable depth, speed and delay. The organ voice and brass voice are routable in any combination through a VCF which has attack/decay/sustain contour controls, contour amount, frequency and emphasis controls in its variable mode plus a preset brass mode. A unique feature of the Opus 3 is a two mode articulator which allows either a cancelling mode, where triggering a new note cancels sustaining tones, or an overlapping mode, where notes with long sustain continue to sound under new notes for a layered sound. Other features of the Opus 3 include an LFO for vibrato, tremolo and "wah" effects, and the famous Moog pitch wheel for monophonic polyphonic pitch bend.

CIRCLE 11 ON READER SERVICE CARD

Rivera Music Services is perhaps best known for its updates and modifications to synthesizers, but with the announcement that they have obtained exclusive marketing rights for the products of Aries Music, Inc. Rivera Music Services has instituted numerous updates and improvements to existing modules in the Aries line to make them the basis of a truly professional quality yet affordable modular synthesizer system. The current Aries line includes: VCOs; VCFs; VCAs; envelope generators; a ten-stage phase/flange module; a state variable filter; an analog sequencer; voltage controlled switches; and a pitch-to-voltage converter, all of which are in stock at RMS. In addition, several new modules will be introduced with the upcoming publication of the 1981 Aries catalog. Rivera's unique experience in modification and updating synthesizers should prove most helpful to anyone considering expanding his or her synthesis system with Aries modules whether the current system is Aries-based or not.

CIRCLE 12 ON READER SERVICE CARD

PAIA Electronics has announced an improved version of their popular Programmable Drum Set which features greatly extended battery life. In the new version, rhythm patterns may be stored for over a year if the unit is not used, while the battery life during normal usage has also been extended to several hundred hours.The improved version of the PAIA PDS retains the original's simple programming of bass, tom, snare, wood-block and clave sounds into two separate rhythm programs each with its own bridge pattern which may be activated from a touch plate on the control panel or from a footswitch.

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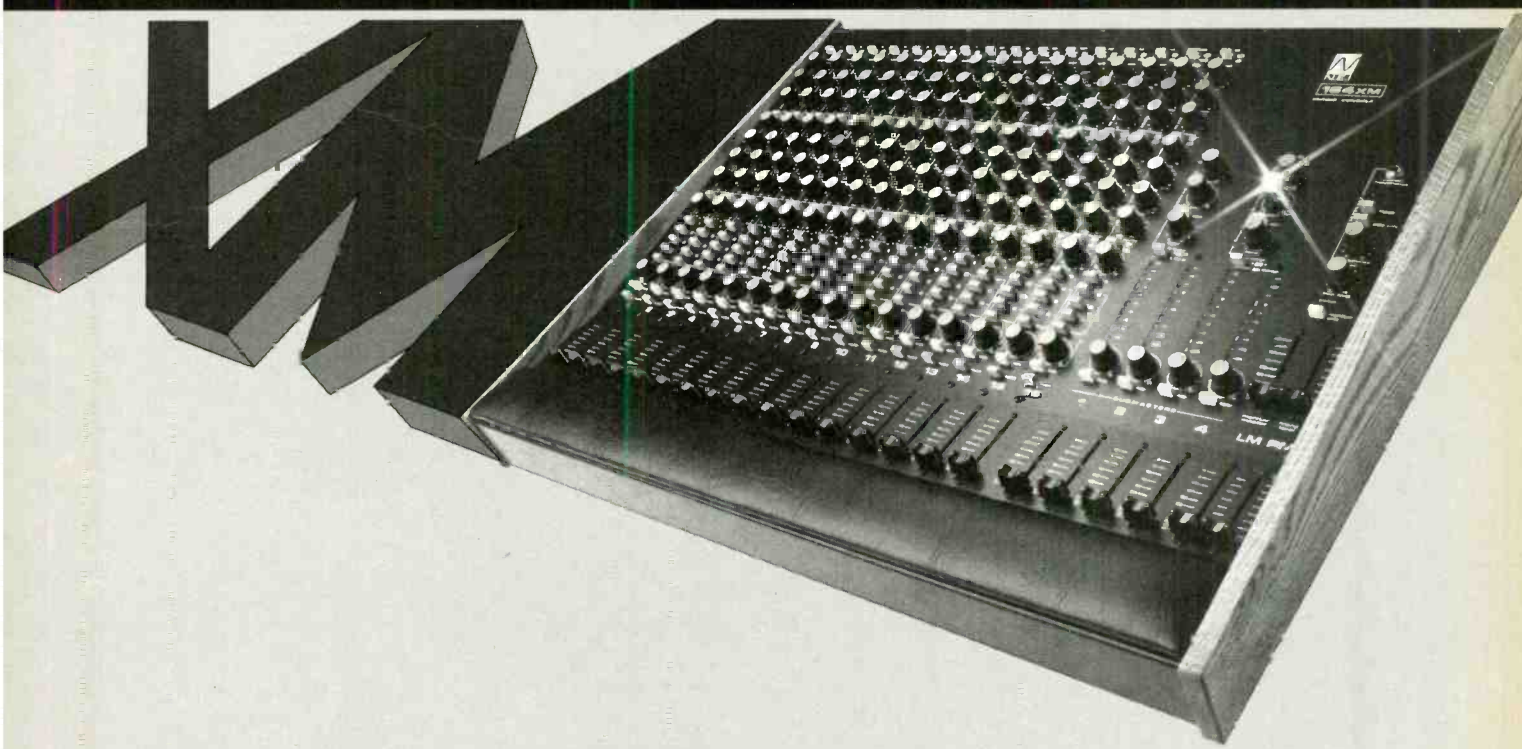

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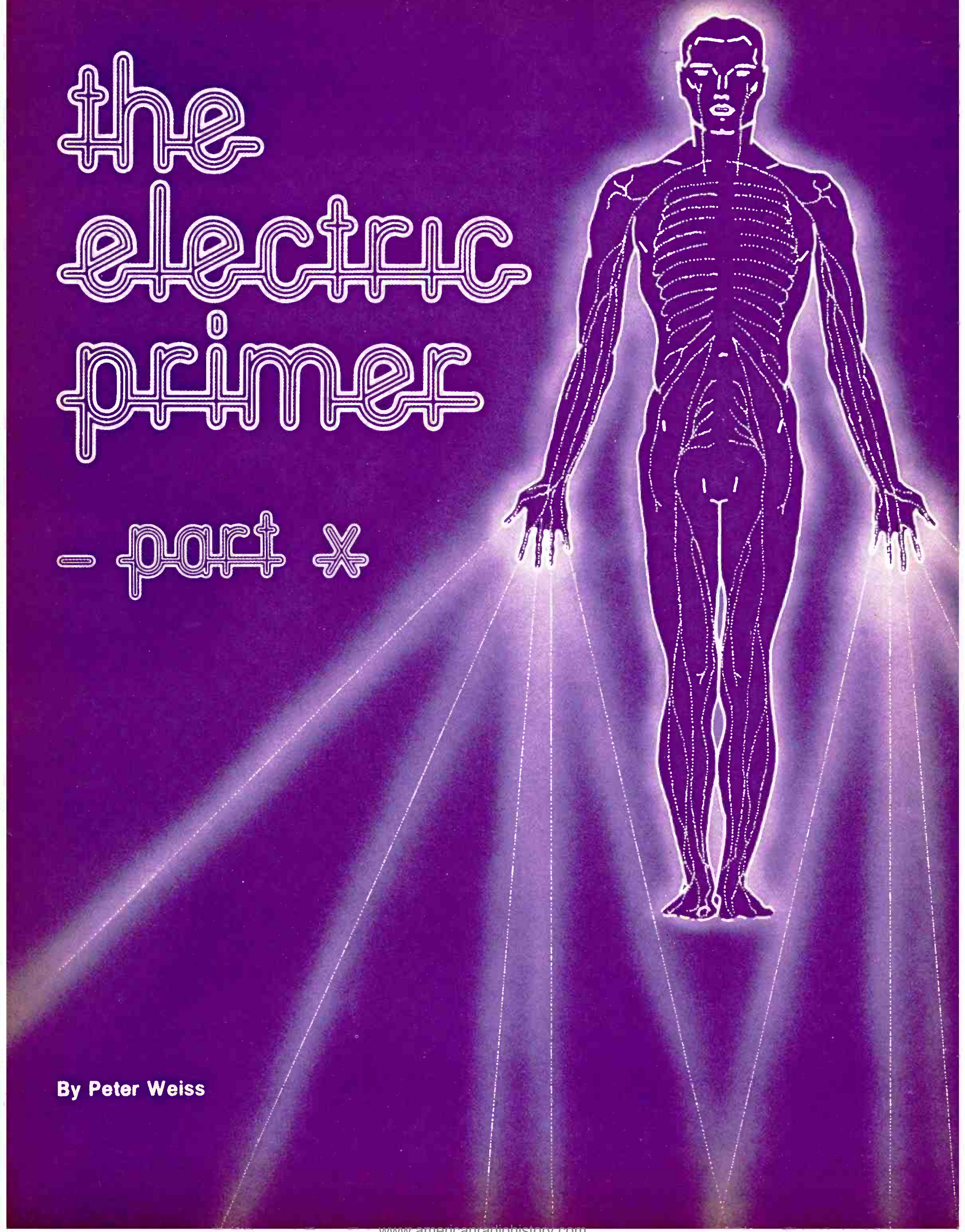
the electric primer – part x



By Peter Weiss

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It would have been nice to have been able to conclude this series of articles smoothly, neatly, with no backtracking or sidetracking. But, in the rush to complete the last installment, the author's typist mislaid a few crucial paragraphs concerning the term "RMS." As a result this term popped up out of nowhere in the next-to-last paragraph of Part IX. (This writer *has* considered firing his typist, but since the typist and the writer's wife of 14 years are one and the same person, patience and discretion are indicated. Besides, having one's dinner served in the garage is no fun. She also works cheap.)

Anyway, back to "RMS." These letters stand for the words "Root Mean Square," which is a method of finding the average of a group of numbers or values when the standard averaging method (adding up all the values and dividing the sum by the number of values) won't work. Finding the average value of a sinusoidal (sine-curve-shaped) signal (either voltage or current, *not* power) is not possible using the ordinary averaging method. The reason for this is fairly simple. In one cycle of a sinusoidal signal half of the wave form is positive and half is negative. Thus, if we choose points at 30° intervals along the horizontal axis, as shown in *Fig. 1*, find the value of the waveform at each point and add the individual values, the resulting sum is *zero*. This is true because for every positive value in the first 180° there is a corresponding negative value in the second 180°.

The zero result achieved above is useless because it gives no information about the possible work, energy or power available from sinusoidal electrical signals. The problem of the positive and negative values cancelling out can be avoided by squaring each value (multiplying the value by itself). When this is done all of the squared positive values remain positive, but since the result of squaring a negative number is *always* a positive number, the negative values are replaced by their positive squares. To get a Root Mean Square average the *squares* of the individual values are added up, this result is divided by the number of values given (as in standard averaging), and the *square root* of the division answer is the Root Mean Square or RMS average value. Here is a quick example using a sinusoidal voltage having a peak value of 1 volt. Below is a table of values taken at 30° intervals in one cycle of this example signal. Also shown are the squares of the values and the result of the rms averaging procedure.

	θ	$\sin\theta$	$\sin^2\theta$
1.	0°	0.000	0.000
2.	30°	0.500	0.250
3.	60°	0.866	0.750
4.	90°	1.000	1.000
5.	120°	0.866	0.750
6.	150°	0.500	0.250
7.	180°	0.000	0.000
8.	210°	−0.500	0.250
9.	240°	−0.866	0.750
10.	270°	−1.000	1.000
11.	300°	−0.866	0.750
12.	330°	−0.500	0.250

$$\text{TOTAL} = 6.000$$
$$\text{TOTAL} \div 12 = 0.500$$
$$\sqrt{\text{TOTAL} \div 12} = 0.707$$

So, the RMS average value of a 1 volt peak-value signal is 0.707 volts. Since all single-frequency sinusoidal signals have the same "shape," we can safely say that the RMS value of any such sinusoidal signal is 0.707 times (or 70.7% of) the peak value of the signal.

The 70.7% figure also represents, as mentioned at the end of Part IX, [*MR&M*, November 1980] the "effective" value of a sinusoidal voltage or current. The term "effective" is applicable because an alternating voltage or current will produce the same heating effect in a pure resistance as will a *direct* voltage or current equal to 70.7% of the peak value of the corresponding alternating signal.

The RMS method cannot be applied directly to electrical (or audio) *power* waveforms, because a power "waveform" is actually the product of a voltage and current. However, if the RMS values of an alternating voltage and associated alternating current are multiplied together, the resulting power value is an "effective" power, as described in Part IX. This effective power value is exactly half the value obtained from *multiplying* the *peak* values of the voltage and current.

Now, on to the de-mystification also mentioned at the end of Part IX. The first concept we will tackle will be the deciBel or dB. It should be understood at the outset that the decibel, by itself, is *not a unit* like the volt, watt, inch, etc. The decibel idea only represents the *relationship* between two values, and these values can be given in *any* appropriate units, as long as the units are human hearing system (ears and brain).

ground. This Earth ground can be a copper spike driven into the ground, a cold water pipe or it can be made available through A.C. power supply wiring. The purpose of ground symbols in these cases is to indicate those points in a circuit that are connected together *and* to a zero potential reference. Again, this zero potential reference may be Earth ground, a metal chassis, just a common point of connection or a combination of these (e.g., a metal amplifier chassis used as a common connection for several zero-potential circuit points, with the chassis itself being connected to Earth ground through A.C. power supply wiring).

Grounds, negative *and* positive potentials can all exist simultaneously in the same circuit, as they do in the circuit of *Fig. 9.* Note the "chassis" and "Earth" ground symbols. This last concept is basic to another topic to be discussed: shielding and balanced versus unbalanced lines.

Before we get to that discussion, we must cover another practical area, the subject of transformers. In our discussions of inductance and self-inductance we mentioned that the property of inductance was due to the presence of a magnetic field (such a field always coexists with any electrical current) that arose from the current moving through a coil of wire, and in particular to the back-EMF or counter-voltage produced by the magnetic field as it alternately built up and collapsed. If an alternating magnetic field (caused by an alternating current) can induce a voltage in the coil carrying the current, couldn't the same field also induce a voltage in a nearby coil? The answer is yes, and this set of circumstances provides the basis for a discussion about transformers. *Fig. 10* will serve to illustrate this part of the discussion. At the basic level, a transformer is a device consisting of two coils of wire (often wound around the same form or iron core) constructed in such a way that when an alternating voltage is applied across the ends of one coil, an alternating voltage is *induced* across the other coil. The actual numerical values of the two voltages are related to each other by the physical nature of the transformer. Each coil of wire is made up of a certain number of turns of wire wrapped around a form or iron core. If the number of turns in the coil to which the external alternating voltage is applied (this coil is called the "primary" of the transformer) is equal to the number of turns in the coil in which the induced voltage is developed (this coil is called the "secondary"), the induced voltage will be equal to the applied voltage (all voltages used in this discussion are RMS values). If, for example, the primary consisted of 100 turns of wire, and the secondary consisted of 200 turns, the voltage induced across the secondary would be *twice* the voltage applied to the primary. At first this fact might indicate that we can get something for nothing, but unfortunately this is not the case. There is nothing in a transformer to add *energy* to a signal, so although a transformer can produce a higher *voltage* at the secondary, the amount of *power* available at the secondary is *equal* to the amount supplied to the primary. Thus, if a higher voltage is available at the secondary, the amount of *current* available is only that amount that will yield the same amount of power as applied to the primary. In actual practice, considerable power is lost in the transformer, and the ideal *primary power = secondary power* condition is just that—an ideal for discussion purposes.

Transformers are not always used to develop higher alternating voltage from lower ones, as described above. Very often a transformer primary winding will contain more turns of wire than the secondary, and in such cases the secondary voltage will be lower than the primary voltage. In either case, the ratio of the voltages will be equal to the ratio of the number of turns in the corresponding windings. The ratio of turns in the primary to turns in the secondary is called the "turns ratio" of the transformer, and gives a direct indication of the ratio of primary to secondary voltages.

Power company wires usually carry voltages that are much higher than the 110 or so volts that are available at household A.C. sockets. These high voltages must be "stepped down" to the required voltage, and this is accomplished by the use of large transformers, usually located on utility poles. The voltage necessary for the operation of a television picture is up into the tens of thousands of volts, and this voltage is developed from much lower voltages by means of a "step-up" transformer.

In audio applications transformers are used in several ways, although the "transformerless" approach is becoming quite popular. Audio transformers are used to "match" the outputs and inputs of devices to each other when the input impedance of one device is not that required by the output of the device to be connected to the input. A good example of this type of transformer application is the "direct box" (featured in a construction article in the April 1978 issue of this publication). The purpose of this device was to enable a console preamplifier, which is designed to accept signal from a source having an internal impedance of approximately 150 ohms, to accept signal from the pickups of an electric guitar or bass guitar having an internal impedance of 15,000 ohms. The transformer used had a turns ratio of 10:1. A transformer will "match" impedances connected across primary and secondary when these impedances are in a ratio that is the *square* of the turns ratio. In the direct box example, the impedance ratio is 15,000:150 or 100:1, so the required turns ratio is 10:1. It is important to point out that a transformer that is specified as a matching transformer capable of matching a 15,000 ohm impedance to a 150 ohm impedance does not have a primary whose impedance is 15,000 ohms and a secondary of 150 ohms impedance. The 15,000 to 150 figure is given to indicate the *ratio* of impedances that can be matched. Specific impedances are mentioned

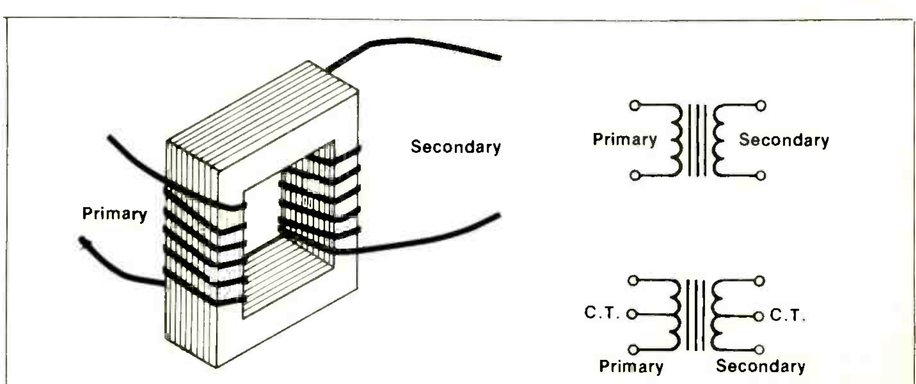

Figure 10

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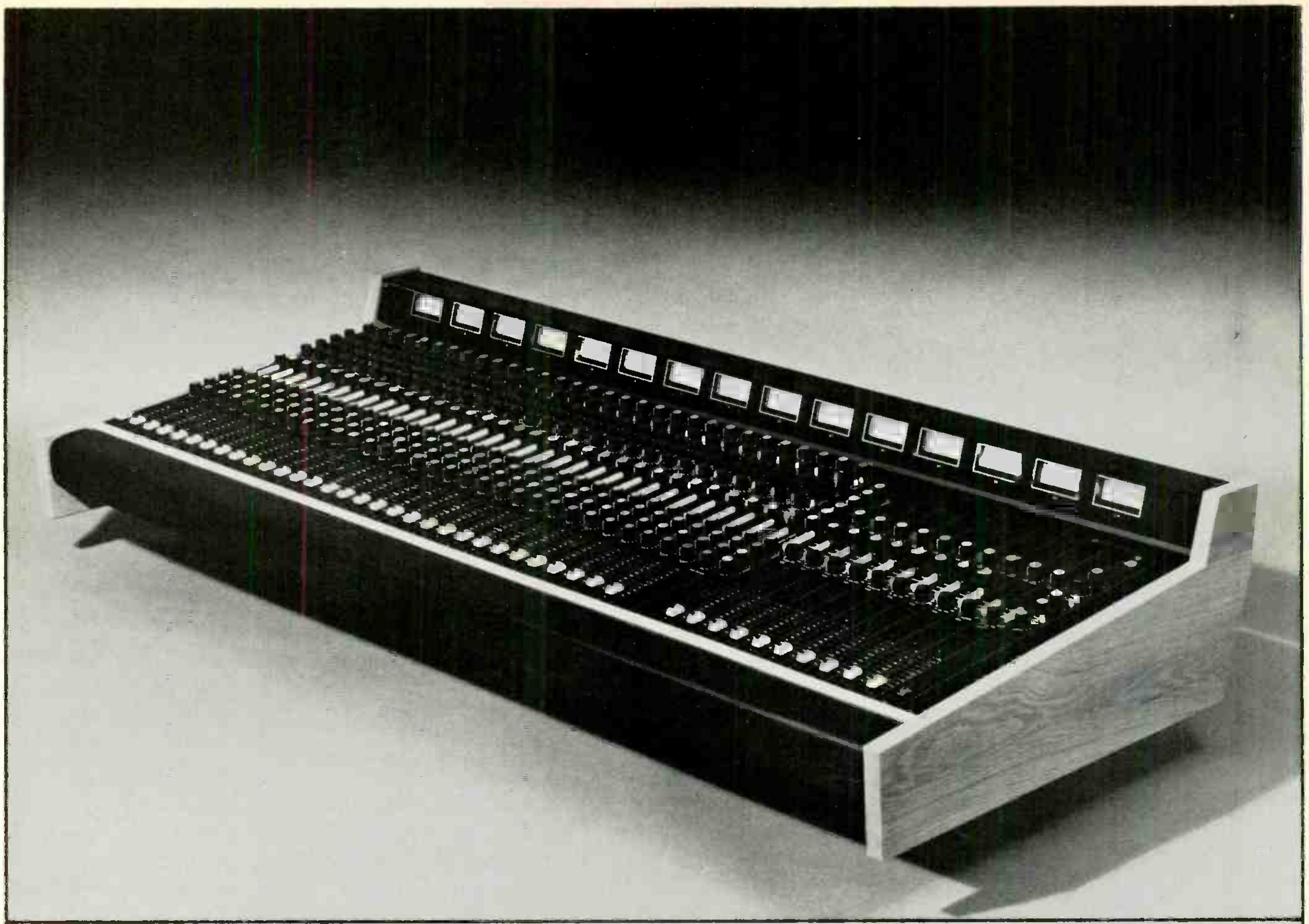

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based on the recommended application for the transformer, which takes into account the power levels expected and the current-handling capabilities of the windings. The actual *resistance* of a transformer winding is likely to be very low, in the tens of ohms, since a winding is just wire. Some windings that are made up of very long lengths of very fine wire may have resistances that are higher.

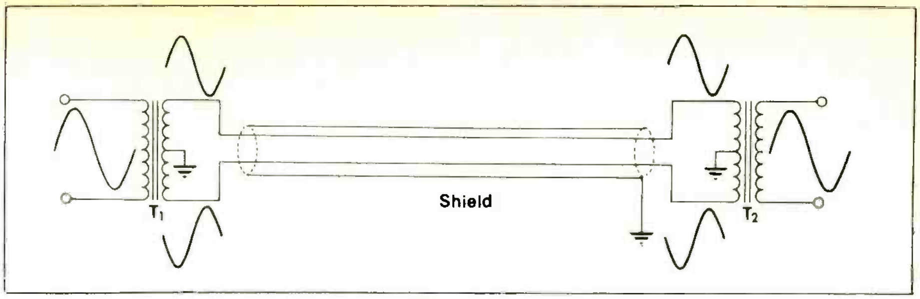

Figure 12

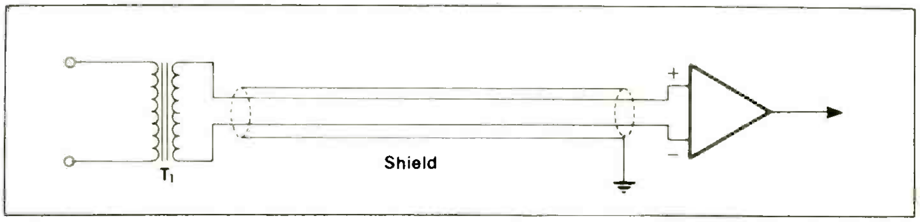

Figure 13

When audio signals are sent over distances of more than a foot or so, some precautions must be taken to prevent unwanted signals from appearing in or across the conductor(s) carrying the desired signal. This is especially important if the desired signal is a very low level signal, like the output signal from a microphone. One method of protecting a signal-carrying conductor is to encase it in a conductive "shield," as shown in *Fig. 11,* and to connect this shield to ground. The shield protects the conductor from external electrical fields (static, etc.) because any field encountering the shield will follow the shield, and any potential that could develop as a result of the external field is immediately reduced to zero because the shield is grounded. However, the shield in this example also carries signal current. Any *magnetic* field in the vicinity of such a one-conductor-and-shield arrangement would induce an undesired current in the internal signal-carrying conductor. As a means of preventing this condition, the arrangement shown in *Fig. 12* was developed. Note that each transformer is grounded at a point that is halfway between the ends of the windings. These points are called "center taps," usually labelled "C.T." in transformer specifications. These center taps are located at the electrical centers of the windings, so that there is an equal number of turns between a center tap and either end of the winding.

The center taps shown in *Fig. 12* are both grounded, so they are at the same (zero) potential at all times. The shield encasing the *two* signal carrying conductors is also grounded, and therefore provides the same *electrostatic* shielding as did the shield in the previous example. Protection against *electromagnetic* interference is provided both by the shield and the two-conductor/center-tapped transformer combination. Starting at the left-hand transformer primary (T_1); when an alternating voltage is applied here, alternating voltages are induced in each half of the center-tapped secondary, but these voltages are 180° out of phase with each other, because of the grounded center tap. When these out-of-phase voltages are applied to the center-tapped primary of the right-hand transformer (T_2), the effect on the secondary of the right-hand transformer is the same as if the voltage were being applied to a complete primary winding with no center tap. That is, since the transformers are all 1:1, the voltage developed in the right-hand secondary will be the same (give or take a small amount due to the losses mentioned earlier) as if the left-hand primary and right-hand secondary were parts of the same transformer. The rejection of electromagnetic interference comes about because any magnetic fields that would exist in the vicinity of this arrangement would create currents in the two conductors simultaneously, and these two currents would be *in phase* with each other, and would develop voltages at the two halves of the right-hand primary that were *in phase* with each other. However, because the center tap of the right-hand primary is grounded, the result of combining two in-phase voltages in this winding is that the two voltages cancel each other out. This two-conductor shielded configuration is called a "balanced line," and the single-conductor shielded arrangement mentioned earlier is called "unbalanced." The balanced method provides what is known as "common mode rejection," which is a fancy way of saying that whatever unwanted signals appear in-phase on both conductors will be cancelled out.

The grounded center-tap method of developing balanced signals is often replaced by the "floating" method illustrated in *Fig. 13.* The method still employs two conductors and a shield, but the center taps have been eliminated. Modern transformerless systems carry things one step further, and instead of terminating a balanced line from a microphone at the primary of a transformer, the two conductors are connected to the inverting and non-inverting inputs of an op-amp type device. High-quality systems of this type can provide ample common-mode rejection while providing transient response that is far superior to that possible with even the highest quality transformers.

This concludes the "Electric Primer" series, since any further exploration of topics in audio do not legitimately come under the heading of "Electricity." We hope that the series has been informative, while at the same time providing a stimulus for interested readers to seek further information from other, more detailed and in-depth sources.

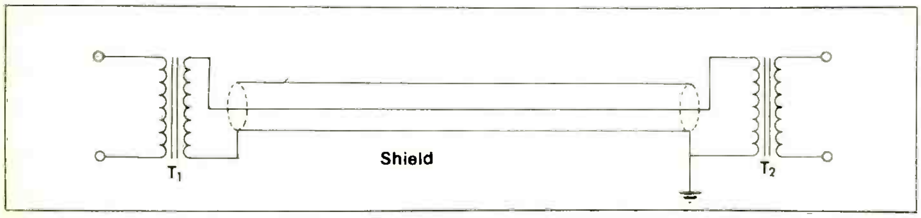

Figure 11

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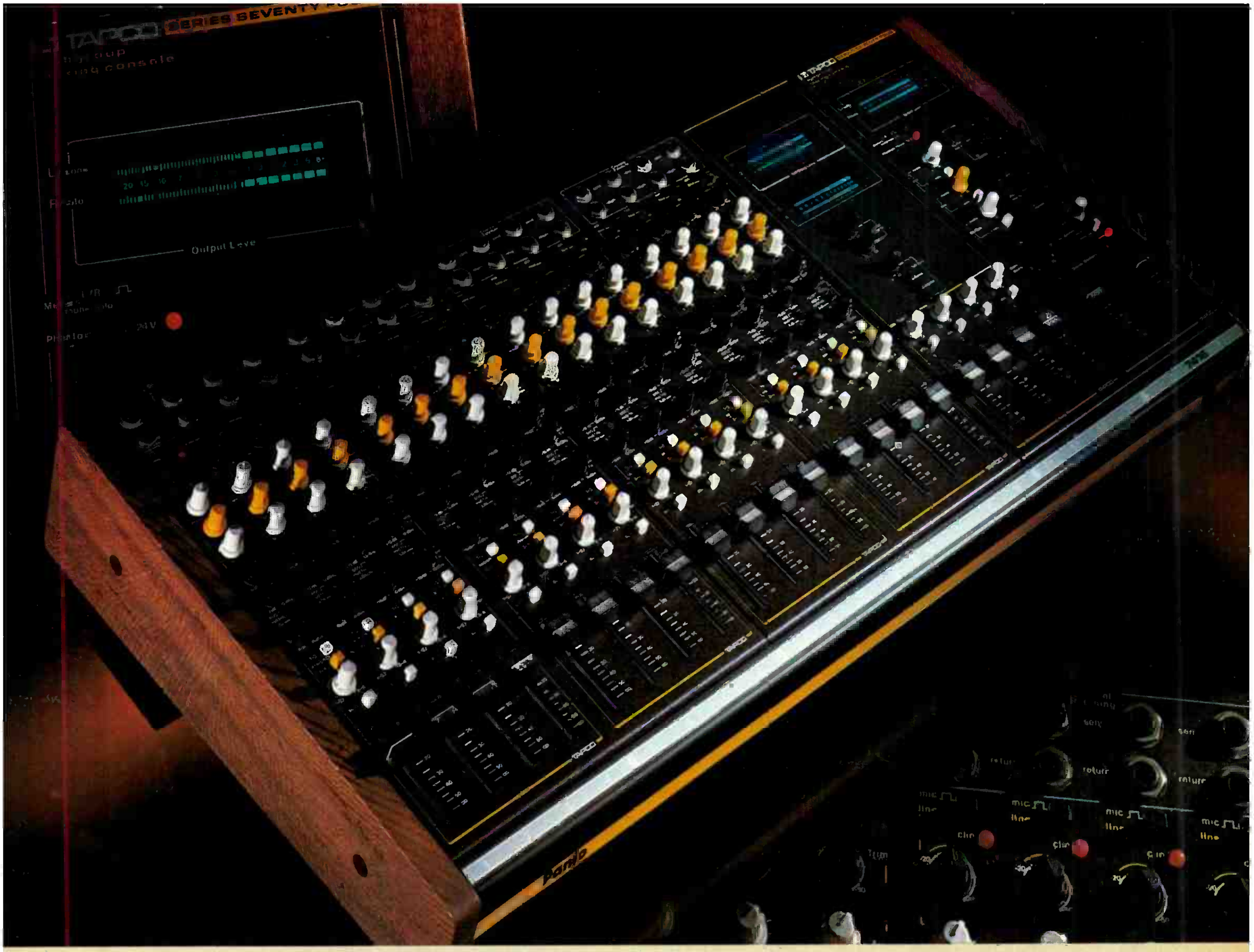

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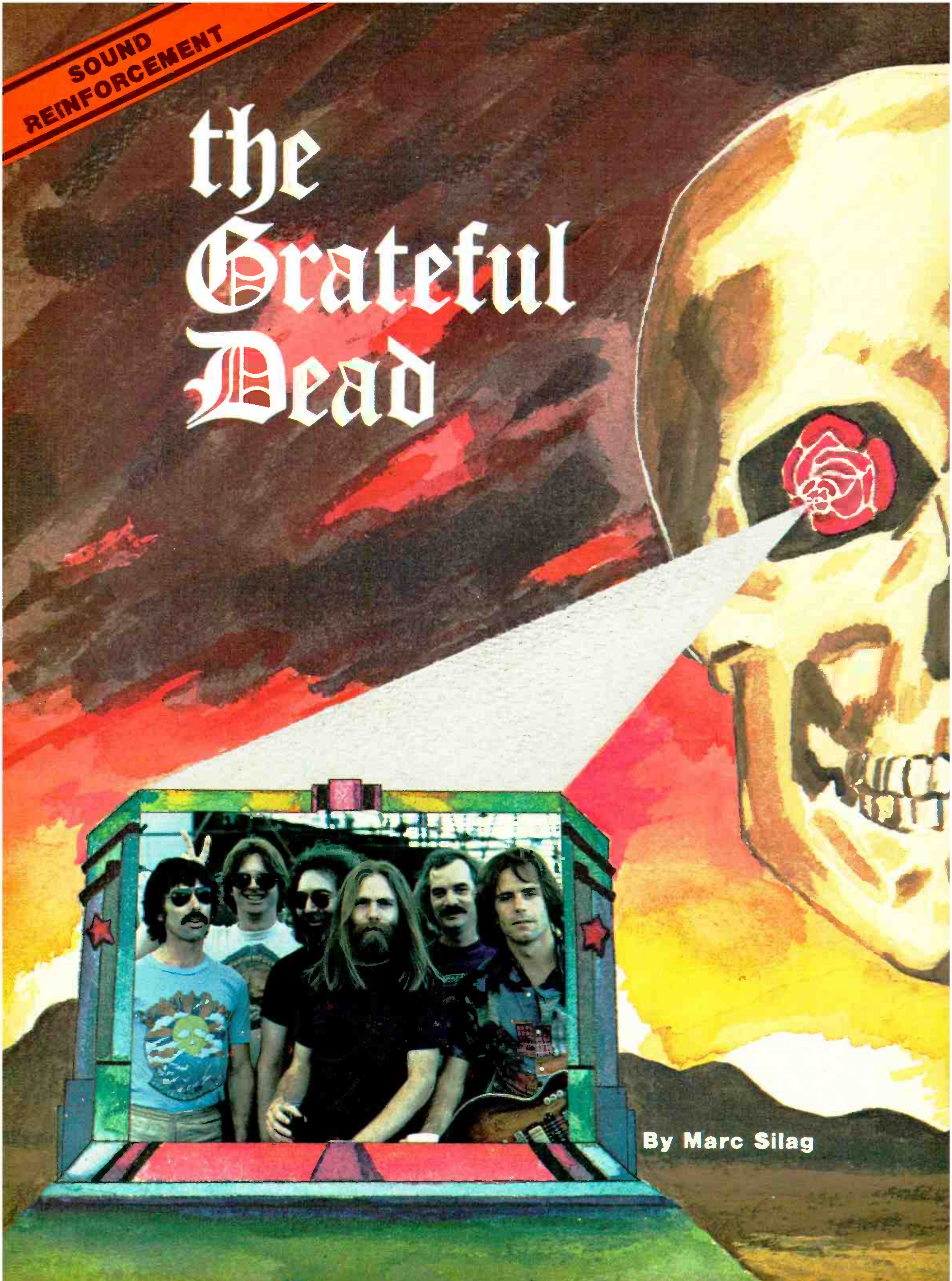
SOUND REINFORCEMENT

the Grateful Dead



By Marc Silag

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IN CONCERT

News of the Grateful Dead's series of eight concerts at Radio City Music Hall in New York spread like wildfire. Three days before tickets went on sale for the October shows, the lines of Dead Heads, those dedicated legions of Deadmania, began to form and wrap around the cavernous show palace. The eight shows sold out in less than two days, the event was covered on the local news, and a month before the Dead's production crew even began to set up, the Dead were the hottest ticket in town. At about the same time tickets went on sale in New York, the Dead were playing a series of shows in their hometown of San Francisco, selling out the Warfield Theatre fifteen times. Yes, Deadmania is alive and well, anywhere the band performs.

For their Radio City shows, the Dead came equipped to handle three main tasks. The first, of course, was the concerts themselves. In addition to the band equipment the Dead normally use, they also employed screen projections and other sundry stage gear for the productions. From a distance, a stage set for the Grateful Dead resembles one which might employ a small electronic orchestra with a *heavy* percussion department. Production effects aside, space on the great stage was at a premium. P.A. services were provided by three companies: McCune Audio of San Francisco, Bill Graham Presents of

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San Francisco and Ultra Sound, Inc. of Larkspur, California. Working in cooperation with Dan Healy, these three companies provided the technicians and the hardware to accommodate Healy's needs for Radio City.

Second, the Dead came equipped with a mobile video production unit that would provide a video simulcast to 19 cities across the east and midwest, for the final show of the series scheduled for Halloween night. This, in effect, turned the apron and the outer periphery of the great stage into a full-scale television studio. Last but not least, the Dead came prepared to record the performances "live," as they had done at the shows at the Warfield a few weeks before. In San Francisco they had set up shop in the basement of the theatre, bringing in their own studio equipment. In New York they decided against the hiring of a mobile recording unit, opting instead to use the recording studio secreted away on the seventh floor of Radio City. However, due to their unique approach to "live" recording, they decided they'd use their own equipment in New York as well. So everything was loaded into two Boeing 747 Cargo Containers and shipped to New York a week before the shows.

The technical aspects of the Dead's "live" and recording productions at Radio City are rather unique. There are a number of people who are responsible from the most embryonic stages of its development right up to mixing the show in the house or changing tape in the "live studio" at Radio City. Dan Healy wears a number of jackets on behalf of the Grateful Dead, ranging from Chief Sound Engineer to Production Manager for their "live" performances. He also wears the jacket of co-producer in the case of the "live" album the Dead plan from the shows on both coasts. Dan is listed as a "Charter Member" of the Dead, having been with the band since their inception in 1965. Healy's co-producer on the "live" album and another veteran of the Grateful Dead's long history is Betty Cantor-Jackson. Betty started mixing two-track tapes for the Dead over twelve years ago and, as can be seen in the accompanying overview of the "live" recording of these shows, her talents and ear are highly valued by the Dead, long renowned for their discriminating attitude towards their sound. Don Pearson is a founding partner of Ultra Sound, Inc. Pearson supplied technical back-up to Dan Healy. Ultra Sound is highly involved with product development of some of the most advanced equipment in use in the reinforcement field, as well as some interesting approaches to "live" recording. Ultra Sound, working in association with engineer John Meyer and his company, Meyer Sound Labs, Inc. (MSLI), is quietly setting the stage for what many consider a major innovative approach to speaker design and theory and electronic processing.

As will be seen, Meyer's technology (as applied by the Dead themselves, Ultra Sound and the two P.A. companies who worked in cooperation at the Radio City dates) is certainly out of the ordinary and prudent in its design; innovative if not revolutionary. It may benefit the reader to recount some basic history as related to *MR&M* by Dan Healy and John Meyer.

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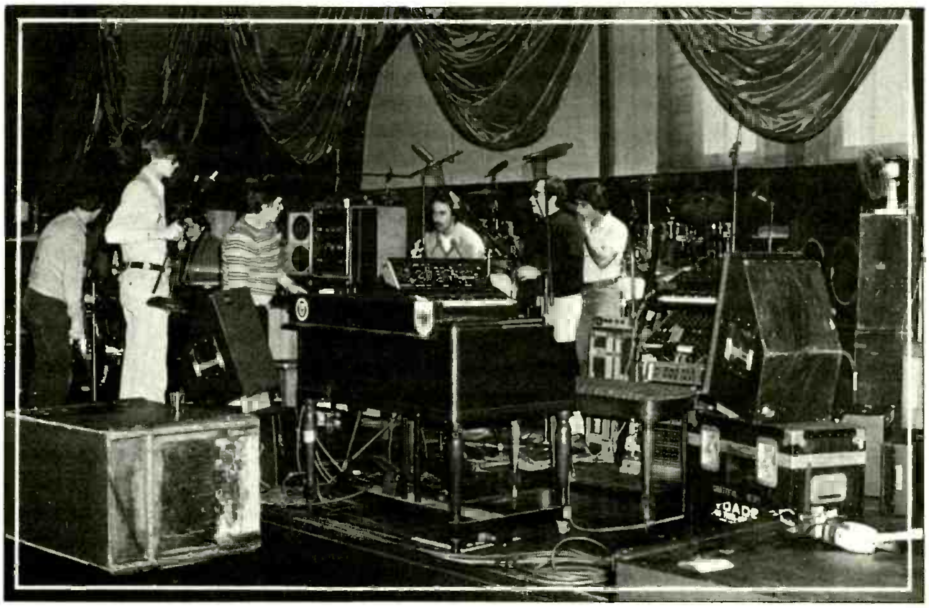
The Dead, perhaps more than any other American band, have always attempted to deliver as much energy to their audience as they possibly can. According to Healy, it was early on in the game that the Dead recognized the role of quality sound reinforcement in their music. For years it's been a known fact that they actively experimented in order to further their grasp of the available technology. Playing at the Carousel Theatre with the Jefferson Airplane and Quicksilver Messenger Service in 1968, Healy was fed up with the garbled, distorted muck projected by the two cabinets that comprised the P.A. Knowing there must be a better way, Healy went out and rented every piece of P.A. gear he could find and set it up in the Dead's rehearsal hall in San Francisco, experimenting with configurations that provided him with a more accessible, cleaner sounding vehicle for reinforcement. It was during this time he articulated his longstanding goal: to develop a system that provided "clear, undistorted sound, that was an accurate image of the source being reinforced."

It was at about this time he first became associated with John Meyer, whose activities in the audio field began as an engineer for Steve Miller. Meyer was dissatisified with the equipment available to the trade, feeling that manufacturers were not applying enough energy to research and development, simply because the market was limited to a select few professional audiophiles. Working with a West Coast company in 1969, he participated in the design and construction of a large P.A. utilizing experimental 8' horns. Meyer seemed to be headed towards the research and development sector, simply by virtue of the fact that he was questioning the status quo and, for the time at least, proving that alternative forms of technical application were readily available. Hooking up with McCune early in the seventies, Meyer worked with the McCune staff in the construction of the JM-3.

In 1974, Meyer traveled to Switzerland to participate in research



The cluttered stage setup with the "heavy" percussion section hidden in the rear.

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being conducted at the Institute for the Advancement of Musical Studies. Meyer remained there for a year, studying various transducers in the hopes of discerning what made one speaker or horn sound different than another. Based on his work in Switzerland, Meyer began to apply his energy in pursuit of a natural sounding transducer that could accurately reproduce the signal as it was meant to sound. Upon returning to the States he again hooked up with the technicians at McCune. The result was the construction of the JM-10, part of the system utilized at Radio City. It was at this time he began to experiment with studio and stage monitor designs. His approach was innovative and Dan Healy was one of the first to see its potential applications.

While Meyer had been pursuing his research, Healy was still in pursuit of the perfect P.A. or at least something akin to that. The Dead turned a few heads when they unveiled their renowned wall of sound in the mid-seventies, but as Healy explained, the economic crunch that affected everybody around that time affected the Grateful Dead as well and it became cost prohibitive to maintain their own system. Healy saw that much of Meyer's technology was the answer to many of the ideals he had been pursuing. He had been trying to find financing for the construction of a system designed by Meyer, but had no luck convincing anyone to supply him with the rather large amount of capital required. Meyer's experimentation had resulted in designs requiring components simply not available in the U.S. The cones are manufactured in Europe and supplied to Meyer through the ACD Company, located in Switzerland. His cabinets utilized wood from Finland and the quality control methods Meyer insisted upon were time consuming and costly. However, thanks to McCune's support, the first JM-10 had proven itself to be worthy of the expense. Healy admittedly liked the system from day one but felt that it required some augmentation and modification to fullfill his needs for the Grateful Dead. So, experimenting with the JM-10 and equipment salvaged from "the boneyards of the Wall of Sound," Healy and John Cutler, a shop technician for the Grateful Dead, constructed what amounts to be the prototype for the second sound system utilized at Radio City, the System 80. This system, like the JM-10, was largely the result of Meyer's research, and it wasn't until Bill Graham contracted Meyer to build the System 80 as it was to be heard at Radio City, that Healy first began to feel he was that much closer to relaxing *the* system for the Grateful Dead.

So, in effect, what the multitudes of Dead Heads heard at Radio City was actually two systems, one horn loaded, the other of direct radiating design. Both were augmented by subwoofer assemblies of MSLI design, the JM-650. These assemblies were provided by Ultra Sound. To this listener's ears, Dan Healy's original concept of what a P.A. should be has, to a degree, been realized.


TV equipment used for the video simulcast arranged on the periphery of the stage.

This was the first time the two systems had been utilized together and Healy admitted he was pleased with the results. Pleased, yes. Satisfied? Never. "What the people at Radio City heard was just the beginning," Healy stated. "What I'm shooting for is to bring a similar system, but bigger, into a 15,000 to 20,000 seat hockey arena and be able to hear the best mix you ever heard, no matter where you're sitting in the hall. But I'm talking about six JM-10s and the System 80. The technology is there. When I first started it was incomprehensible to think about playing to 20,000 seats and hear everything accurately. But that's what audiences can look forward to in the future." A look at the specifics at Radio City may help explain what lies ahead.

McCune's JM-10 system comprised the right and left stage stacks at Radio City. Each side consists of 12 woofers, 6 mid-range horns and 30 tweeters. McCune technician Steve Kadar and engineer Michael Brady provided *MR&M* with the details. Each side is driven by nine Crown DC-300A power amplifiers. The power banks are unique in that McCune's technicians have incorporated an automatic spare switching system. This allows the system operator to monitor the major fault modes that might occur during a performance. A readout located near the power banks signals the operator when an amp blows or failure occurs in the crossovers, etc. In the event of equipment failure the operator can easily switch in a spare allowing the performance to continue "seamlessly."

The crossovers were built to Meyer's design, and here it is possible to see exactly what it is that Meyer is doing that makes his design stand out. As Meyer explained, his systems are built on a theory of linear response.

Through his research work in Switzerland, he became aware that it was possible to make a horn, for example, sound like an electrostatic speaker, or vice versa, utilizing linear theory. Linear theory is dependent on electronic circuitry to control the waveform as it is introduced to the transducer. In Meyer's view, a transducer's reproduction of a signal or impulse is dependent on the *way* the signal or impulse is translated from electronic energy to

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mechanical energy (via the transducer) and then into acoustical energy. By altering the signal via a complimentary circuit before the transducer stage, one can effectively eliminate any of the inherent distortion caused by a speaker's reaction to the signal, relative to the speaker's reaction to air. Under normal conditions, utilizing no circuitry to "predistort" the signal, it is a natural characteristic of a transducer to add its own distortion to the translated signal. An advantage of this linear approach in P.A. system design, is that the sound system is capable of reproducing the information more accurately, imaging the original sound source.

Altering the signal is not an innovation in speaker theory; manufacturers commonly add distortion characteristic to "enhance" the sound of their products; in a sense, cosmetically containing the distortion. Meyer's approach differs in that his circuitry is matched to his transducers with exacting tolerances, allowing MSLI to take advantage of situations that lower harmonic distortion. A speaker designed without electronics to control the wave form, will have certain inherent limitations with regard to achieving a flat response level, one that represents a smooth response to the signal coming from the amplifier. Outboard processing, such as third-octave equalization, detracts from the effect (or lack of effect) Meyer is after. Altering the signal in the processing stage with this type of equipment creates additional phase distortion, which becomes more problematic when it interacts with the transducer. Any hope of hearing what you put into the system, the way it actually sounds, is lost. MSLI seeks out areas in signal processing and reproduction where distortion can be eliminated or suppressed. The distortion affects the phase cohesion, a major consideration in Meyer's designs. He has gone to great lengths to control phase cohesion at every point in the signal path. The JM-10 is time corrected in the high frequencies, to keep arrival times of all wavelengths consistent. In doing so, the operator need not deal with another source of phase distortion. A system not equipped to control arrival times of the spectrum is subject to phase distortion stemming from reflected sound in the hall bouncing back in an irregular fashion, as "a perverted image of the original sound," according to Dan Healy. Time correction reduces the susceptibility to this perverted image, allowing the reflected sound to remain more mirror-like in its image. Meyer's time correction theories are not dependent on standard delay units available to the trade since the available hardware is capable of adding another dimension of phase distortion to the signal all its own. The breakdown of the components in Bill Graham Presents System 80 that follows will help illustrate Meyer's approach to arrival time correction.

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Flying high over center stage at Radio City was a "cluster" of sixty 12" cone drivers, twelve MSLI mid-range horns and twenty-four Heil tweeters. To control arrival time from this array, Meyer constructed what he refers to as the Group Delay Equalizer, basically an all-pass filter, referred to by some as an envelope filter or delay. It s function is to share the waveform in such a way that certain frequencies are delayed while others are not, relative to the transducer stage. Meyer was quick to

The Dead's equipment set up in the seventh floor recording studio.

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point out that this does not imply that an entire bandwidth is delayed, but rather certain frequencies within a given bandwidth are delayed while the others are allowed to pass unhampered. The design technology applied is too lengthy to cover in depth here.

Suffice it to say the circuitry employed here is quite extraordinary, particularly in light of the sound both the System 80 and the JM-10 are capable of producing. Steve Neal, the technician employed by Bill Graham Presents on the gig, responded to a visitor's comment about the SPL potential of the cluster which is driven by approximately 30,000 watts of Phase Linear power. "The system on its own is capable of producing upwards of 140 dB at about 10 yards. At 122 dB, distortion is rated at about 1%. We have to be careful with it because of the lack of distortion. Subjected to high levels of clean sound, the ear does not readily recognize when things start getting too loud. We're very aware that the system is capable of human damage." Looking up at the massive array of speakers, one couldn't help but feel somewhat unnerved by Neal's comments.

Augmenting both the JM-10 and the System 80 were subwoofers of Meyer's design notated as the JM-650U and 650R. The 650U was the first generation of Meyer's subwoofer designs incorporating an 18" ferro-fluid speaker mounted in a cabinet designed to Meyer's exacting specifications. The JM-650R is basically the same subwoofer that has had some modifications in the cabinet bracing and damping, eliminating some resonance Meyer and Co. found undesirable.

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Out in the house, McCune Audio supplied Healy with a 22x4 mixing console that was built in their shop by Steve Kadar. Eight of the board's 22 inputs offer sub-master capabilities. The board incorporates 2 monitor/echo sends, 2 echo returns, 6 inputs switchable to mic or line input. Three-band EQ offers lo- and mid-peaking select and hi-end shelving. Simply designed but effective in that Kadar designed the board with low noise specs in mind, the better to remain faithful to the JM-10's unique characteristics. Two of the main outputs were fed to the house system while the other two fed the video simulcast. Michael Brady explained that the small Tangent board adjacent to the McCune console was a drum sub-mix. "During the Rhythm Devils set I help Dan ride the mix." He explained. "There's just too much going on. It takes two people to ride a mix like that." In terms of outboard equipment in the house, there was little to speak of. Healy does keep two White passive third-octave equalizers handy, but Don Pearson of Ultra Sound offered this consideration as to their use: "95% of all third-octave equalizers in the field are misused. Boosting a certain frequency adds to phase distortion, and that's what you don't want. Instead of boosting a certain bandwidth, the operator is much better off to lower those bandwidths around it and increase the gain."

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Healy pointed out that the only other outboard equipment he was using were effects of his own design, primarily time delay instruments that allow him to indulge in a little bit of three-dimensional sound. "Basically, some of my ex-

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Seven floors above the stage at Radio City, Betty Cantor-Jackson presided over the Dead's "live" recording setup in the studio that was formerly Plaza Sound and remains the in-house recording studio. In a relaxed but conscientious manner, Betty, aided by Ultra Sound's Don Pearson, saw to it that the Dead were captured "live" in this unique environment. The majority of the equipment in the studio had been shipped to New York from the Dead's studio on the West Coast. Betty explained that she and co-producer Dan Healy had also recorded the fifteen shows at San Francisco's Warfield Theatre in September, setting up their equipment in the basement of the hall. They both felt it would be taking undue risk trying to replicate their West Coast setup with rented equipment and due to certain aspects of their approach to this particular live recording, it was unlikely that a mobile unit could have accommodated their needs.

Assisted by other members of the Dead's technical staff, most notably "Wizard" and Bob Matthews, both veterans of the Grateful Dead saga, Betty and Don set up the Dead's 24-channel Neve recording console, augmented by a similar Neve rented from the manufacturer transforming the studio into their control room.

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Radio City Music Hall will probably never be the same after the Dead. Access to the seventh floor studio is nothing less than inconvenient, particularly in light of the equipment Betty brought with her. In the case of the Neve consoles, the house crew had to remove part of the ceiling over one of the staircases to get the console frames into the studio. A splitter on stage sent the mic lines up to the studio and a visitor wondered if the holes in the walls accommodating the snake from the stage had been there before the Dead showed up.

As can be seen in the accompanying diagram, two 2" Studer tape machines were set up adjacent to the consoles, one 24-track and one 16-track. Next to these were four Ampex 4-track machines. A rack of outboard equipment, limiters and a reverb unit had been placed to the left of the console, but Betty indicated that aside from some limiting on the vocals she rarely used any outboard gear.

In front of the consoles, a studio monitor system designed by John Meyer had been set up. Known as the ACD Monitor System, it incorporated the same 12" speakers and MSLI modified horn used in the Ultra-Monitors on stage. Pearson pointed out that the ACD had a "sweeter" sound than its cousin on stage and the Ultra-Monitor was a full 10 dB louder than the studio monitors. Each of these cabinets were augmented by a Meyer subwoofer. Power was supplied by a specially designed amp "block" which in addition to housing four amp modules, also contains the electronic circuitry necessary to facilitate Meyer's linear response design. Each module delivers 125 watts of power when the system incorporates the subwoofers, but is capable of power ratings up to 800 watts. Meyer has been awarded a

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number of patents for his design including one on low distortion compression drivers. Betty also used a pair of small full-range speakers, presumably constructed by Hard Trucker's.

In accordance with the pervading attitude downstairs in the concert hall, the consensus in the studio was "no processing is good processing." The stage mics were patched into the Neves and from there directly to the multi-track machines. Although Betty had 48 tracks at her disposal, plus the 16 tracks represented in the Ampex units, by her estimation she was recording on about 38 tracks. The count varied from night to night as she and the crew made changes on the stage and in the studio. The four track machines were utilized apart from the stage mics as part of a unique approach to "live" recording formulated by Dan Healy.

Healy, who has been mixing the Dead's "live" performances "forever" recounted the evolution of the approach he and Betty planned for the next Grateful Dead "live" album.

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After years of mixing the Dead, Healy felt he had his gig down; Set it up, mix it and make it sound great. With the advent of "The Wall of Sound" Healy's services as a mixer were no longer required. The "Wall" was self mixed; no mixer was required. In 1977, after the "Wall" had been retired, the Dead returned to conventional reinforcement systems and Healy returned to his position behind the mixing console out in the house. Like most reinforcement engineers, Healy would mix in stereo, listening from his position in the house. But Healy began to realize that his singular perspective was not necessarily the only perspective in the concert hall. Throughout the Dead's history, members of the audience have lugged their portable recording equipment to Dead shows, recording the event for posterity (and bootlegs!). Healy began listening to some of these tapes and was appalled at the poor quality of the mix. "If in fact the mics don't lie," thought Healy, "the mix isn't happening in the house!"

So he began to experiment, lending his counterparts in the audience professional mics and mixing the shows with those

The two 24-channel Neve boards and the ACD Monitor System used in the studio.

microphones in mind. And he kept listening to those tapes. After awhile, the quality of the tapes got better, and so did the mix as he heard it at the console. He began to see the potential that such an approach might have for the purposes of "live" recording.

Healy's plan involved using the stage mics in conjunction with mics placed in the house to capture a true "live" sound. In order to justify the delay between stage and house mics he began to experiment with conventional delay equipment.

He achieved the spatial effect he was looking for, but the graininess of the electronic delays used detracted from the quality of the sound. He came upon a solution to this problem by utilizing the SMPTE code, the method used to sync audio tape machines with video sprocketed tape machines. The SMPTE code allows for exacting synchronization, accurate within 1/100 of a frame, approximately 300 microseconds. This would allow Healy to adjust the offset and, in a sense, control the size of the room. By dialing in the right dimensions between the stage mics and the room mics he could control the delay effect without the residual grainy effects of conventional delay methods.

At Radio City, the 4-track machines were each fed the SMPTE Time Code on one track and using a resolver, Betty could lock the four machines together. Downstairs in the house, the crew had placed "clusters" of microphones: two Neumann U87's, three AKG C-414s, a left and right Neumann KM84 and an AKG C-24 over centerstage. These mics were fed to a mic preamp system, built by Ultra Sound, which in turn routed the mics to the remaining tracks on the 4-track machines. In this system, which was built specifically for the recording project, Ultra Sound utilized Jenson input transformers and direct-coupled mic preamps with no series capacitors, input to output. Don Pearson explained that servo techniques were incorporated to know the DC offsets. Because of the level changes in the room, the quiet and low-distortion characteristics of the unit made it ideal for its intended application. In this way Betty and Don have the sound as it was heard in various parts of the room at their disposal to incorporate with the stage tracks recorded on the Studers. According to Healy's plan, he and Betty will be able to account for a number of viewpoints in the house, allowing the recording to reflect the actual sound of the "live" performance more accurately.

Considering there were six tape machines in operation to record the performances, an enormous amount of tape was used; 16-18 reels of 2" tape and approximately 32 reels of ½" tape per

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show. The Dead recorded a total of 23 shows. In order to simplify the cataloging of what reel contained what, when and where, Ultra Sound provided an Apple Computer which printed out identifying labels for the tape reels, and a printout listing song titles, date and an index number to match up to the tapes. The computer also provided printouts of mics used, track assignments and other information pertinent to the recording process.

In order to insure the recording of every note on stage, Betty and Don monitored the amount of tape used on each reel, changing tape between tunes. Although they were in communication with Wizard on stage who could inform the band that a tape change was taking place, with six machines running, there was a lot of tape to change! Assisted by Billy Rothchild, an assistant at Media Sound in New York, and two members of the house crew, each "pit stop" was a study in time and motion. Back-up machines were out of the question. The Dead had already had to justify the great expense of this recording to their accountants. Six more machines as back-up was asking a bit much. To their joint credit, in two nights of observation not a note was missed and nary a reel was dropped.

"All we're doing is creating a stereo tape." With typical understatement Dan Healy summed up the Dead's latest approach towards advancing the state of the art. Although none of the principals involved with the recording process were willing to commit themselves as to the expected outcome of the new theories Betty and Dan planned to use in their mixdown process, all were optimistic. At press time, Betty, Dan Healy and Don Pearson, working with members of the band, were listening to miles of tape, selecting tunes from the "live" shows that might be suitable for the "live" album. Granted, Healy's concept of an "audio hologram" on record remains to be heard, but nonetheless the fact that people like the Dead are willing to experiment with such a concept is the very thing that advances the use of the technology at hand today. Considering the amount of time and attention the Dead's technical staff displayed during the Radio City concerts, one can't help but assume their endeavor will be every bit as successful as the shows themselves.

perimental equipment allows me to move the sound around the room. Using arrival time configurations and considering the phase cohesion of the system, I can create an 'audio hologram' effect. I can actually take the sound and direct it in such a way that I can make the vocals seemingly hover over your head and then 'disappear." Questioned about his interest in psychoacoustics, Healy merely replied, "You got it." Healy refrained from providing a description of the components inside his "toys," claiming they were still in various stages of development.

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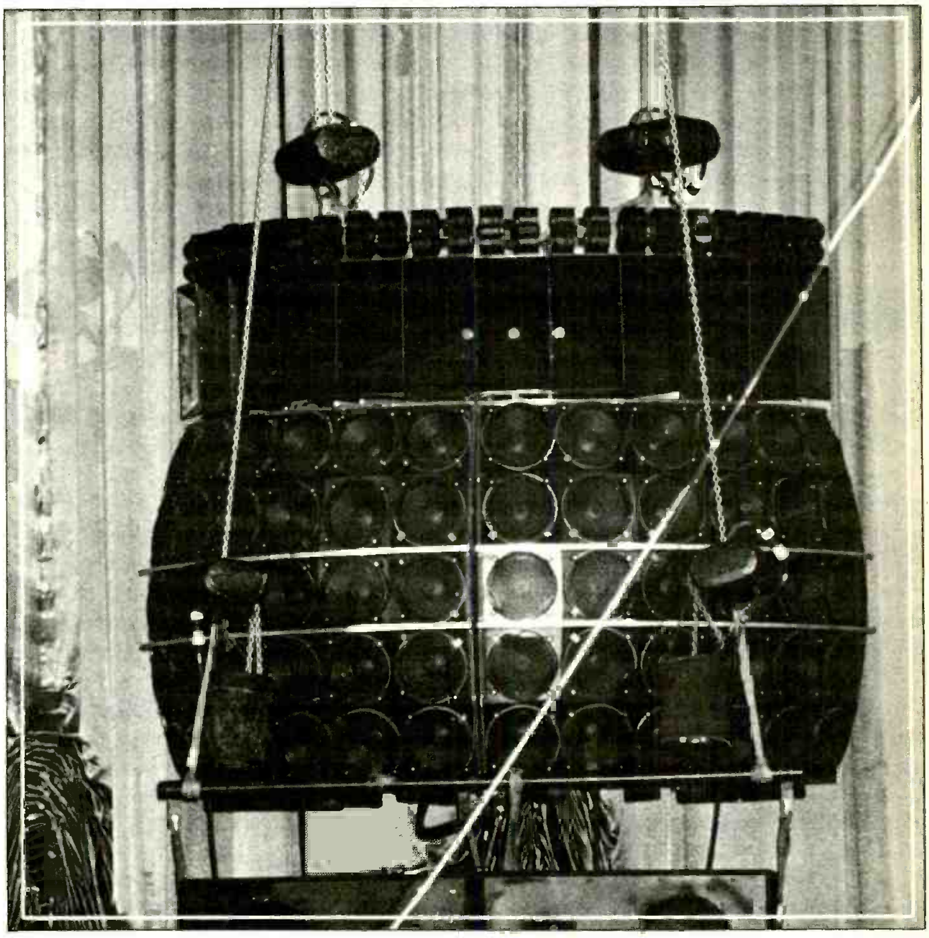
Healy is exacting in his selection of mics and their placement, to say the least. "This is where you're going to get your EQ. We choose the mics for a certain instrument carefully. We'll try anything until we get what we're looking for. The first place to exercise control over the sound you want to record or reinforce, is the place where the sound is converted to electrical energy."

It's difficult to formulate a verbal description of the PA in action. As a visitor stood in a variety of locations inside Radio City during two ferent shows, he was impressed by the imaging, the full body of the sound, and the apparent lack of any P.A. at all. The Dead were in fine form, as was the audience. Guitar lines fluttered from the stage, the vocals were clear and very solid within the mix. The segment of the show known as "The Rhythm Devils" was perhaps the best test for a judgment of the system's ability to handle a signal loaded with transients and the subtle harmonic nuances characteristic of Mickey Hart and Bill Kreutzmann's percussion solos, heightened during the last show of the series—Halloween night—by the guest appearance of Billy Cobham on the drum riser. Out in the house, 6,000 Dead Heads were pleasurably assaulted by the soothing beat of the resonant, gut gripping waves of the counter-drumming of three masterful misicians. Steve Neal of Bill Graham Presents has mentioned earlier that the subwoofers in use had originally been designed to handle the special



The MSLI-designed "cluster" suspended high over center stage.

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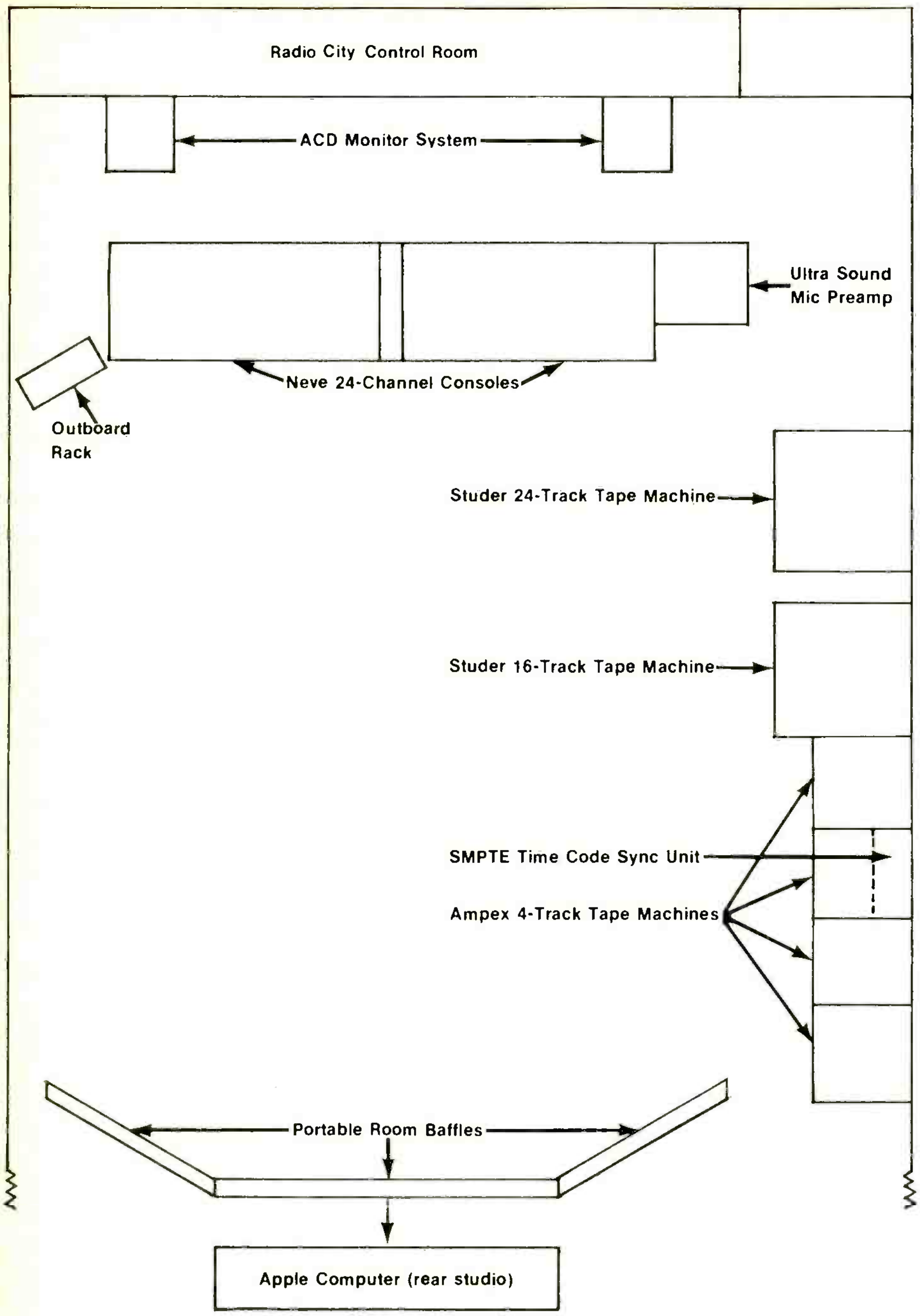

sound effects Francis Ford Coppola had employed in *Apocalypse Now*. This accounted for the very physical sensation during this segment of the show. The sound *felt* good! It was loud, but the lack of distortion managed to lessen the potential of a fatigue that one sometimes experiences after listening to high reinforcement levels over an extended period of time.

On stage, the Dead rolled through their set with a relaxed, confident air. Their stage equipment is a conglomeration of custom cabinets and electronics interfaced with stock Fender and MacIntosh electronics, among other manufacturers and the usual selection of topshelf processing and effects equipment. The Dead's cabinets and woodwork are all done in San Francisco by a company called Hard Trucker's. Hard Trucker's also does the woodworking for McCune's cabinets, as well as Ultra Sound's work for John Meyer's products. All wood used in ¾" Finnish birch plywood. This includes the "Ultra-Monitors" used by the Dead on stage. The Ultra-Monitors were the first joint effort between Ultra Sound and Meyers. They utilize one of the previously mentioned 12" cone woofers of European manufacture, a modified Altec 291 with a conical horn. Ultra Sound is driving each cabinet with 225 watts at 8 ohms. Plus, of course, the electronics that go with it. Referred to as the Processor by Don Pearson, he explained it contains the delays, the crossovers, the equalizers, and limiters to make it a foolproof system. "We've had them on the road now for a year under the worst conditions possible, and we've yet to have a failure." Pearson feels that there is virtually nothing one could do to the system as it stands now to make it any better. He reiterated his feelings about third-octave equalizers stating that third-octave EQ would detract from the sound and was unnecessary. The cabinets are built to Ultra Sound specs using Meyer's engineering, and MSLI manufactures the Processors. Designed to function as a directional, "beaming" monitor, Meyer emphasized the fact that a great deal of quality control is done on each component of the system, with the electronics being finely calibrated to the transducer. The consistency in the European 12" drivers is a consideration that lends itself to Meyer's design. The drivers have to be within ½ dB of each other, but MSLI has not found a stateside manufacturer willing to work at this level. Each driver is screened and if they don't meet this tolerance, they are sold as instrument speakers. The quality control process is strenuous and costly. This is justified by the reliability of the system, and Meyer feels this is just as important as the quality of the sound. Operator efficiency is another consideration, and the system's maintenance record speaks for itself. Harry Poppick, who handles the monitor mix for the Grateful Dead, uses a console designed and manufactured by Chaos Audio (specs unavailable). He also utilizes the White ⅙-octave equalizer for EQ cut. Harry pointed out that Phil Lesh's bass system utilized a recent development of Ultra Sound/MSLI, a cabinet much like the JM-650 subwoofer, employing two 18" speakers. Meyer indicated that they are just starting to delve into instrument amplification, so far with very good results.

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And the sound? Superlatives are risky in horse racing or in the appraisal of sound reinforcement systems. However, there is quite simply no superlative that can adequately do justice to the sound last Halloween at Radio City. Simply Superlative!

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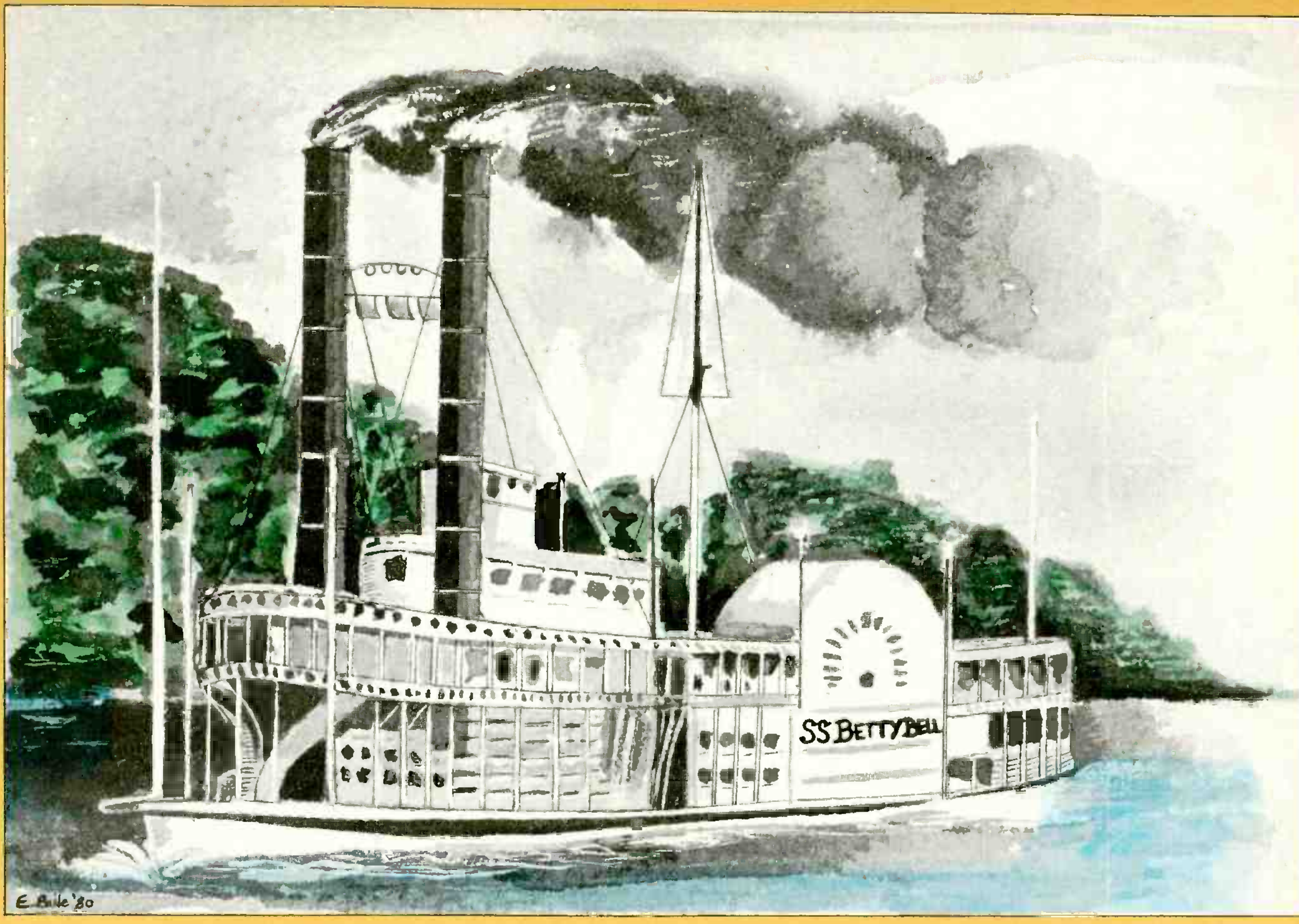

NEW ORLEANS JAZZ & HERITAGE FESTIVAL

by Russell Shaw

As a tourist mecca, New Orleans does not suffer from a lack of hype. Open any brightly illustrated promotional brochure, and pictures of Preservation Hall, Al Hirt and Pete Fountain beckon you to come to the "city where jazz began."

No one should underestimate the vast municipal good will and creative resource these institutions represent, but the full heritage of New Orleans music can not be contained in a night of cocktails, tips, and hot Dixieland on Bourbon Street. How many travelers, for instance, know of the aged blues artists, the black and white Cajun musics, whirlwind ensembles of rhythm that keep Africa and Europe alive and which constitute the roots of much of today's country, rock, and jazz?

If Mohammed . . .

It is entirely probable that a few hardy adventurers do leave the bustle of the French Quarter for an ethnomusical investigation of the nearby Delta of Louisiana. Such inquisitiveness demands an intellectual curiosity and logistical maneuvering ability few of us have, however. What better way, then, to keep traditions alive and popularize them than to bring these musics to the city itself?

All of which brings us to the New Orleans Jazz and Heritage Festival. Held in late April, the Fest recently celebrated its eleventh, and most successful, year. For the better part of two weeks, the whole city throws open its appropriate performance halls for the onslaught of over one hundred artists and nearly two hundred thousand musically starved souls.

The performance sites are wide and varied. The Fairgrounds, a grassy, and ofttimes muddy (this is the rainy season) piece of real estate but a short drive from downtown, features several tents, all operating at once, where the more eclectic acts tend to be featured. It is a special show unto itself, one which demands separate, and lengthy acoustical treatment.

At other venues throughout the city, more established acts are featured. A calculated matching of an artist to a facility where the most souls can watch. Thus, we are likely to witness the fruits of the roots; i.e. mainstream and fusion jazz artists, black contemporary acts, and top name blues

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players, in such diverse settings as the Saenger Theatre, Municipal Auditorium, the Hilton Grand Ballroom, and most uniquely, the S.S. President.

This is the precious, great, and good music. Sound mixing and reinforcement for an event like this transcends all boundaries of mere workmanlike duty. The heritage, the love, the life of the music must come out of those speakers. Much more than just another gig, it is an obligation for any company involved in production.

McCune Sound— Fifty Years Young

Apart from the Fairgrounds shows, McCune Sound handles this duty for all the Fest events. By reliable estimates the third largest sound company in the United States (after Showco and Clair Brothers), the San Francisco- based company is respected in the trade as an ongoing pioneer.

Ken DeLoria, a McCune audio engineer who was intimately involved with all phases of the multi-faceted production, definitely had a key frontline role. We wanted to know a bit about McCune, and how the company fits into the infinite intricacies of bringing off the auditory necessities of such a mammoth production.

MR&M: Tell us a bit about McCune.

KD: Well, McCune is probably the oldest sound company in the world. We are celebrating our fiftieth anniversary next year. That, of course, is a long time, considering that some rock sound companies stick around for two or three years, and even some of the very solid ones have only been open ten or fifteen years. Having that fifty year history has given us that rare financial solidness that has enabled us to put more money into research and development to manufacture our own speaker systems and consoles. We take existing amplifiers—we use Crown quite a bit—and modify them heavily. We do our own cone design, driver and hoist coil design. We design a system from the ground up rather than take what's available on the common market.

Overall it's a very artistic company. We are into other things besides sound, ranging from audio-visual and video work to things like intercom systems for crane operators at high-rise construction sites!

MR&M: How many employees does McCune have?

KD: About forty in our San Francisco shop, fifteen or so in our Anaheim shop, and over a dozen based at hotel accounts in the Bay Area. About seventy employees in all.

The 8,000-seat Municipal Auditorium, one of the six venues for which McCune did the sound reinforcement.

MR&M: Who are some of your top clients?

KD: Probably our biggest client in the concert division is Festival Productions. They produce the New Orleans Jazz and Heritage Festival and Kool Jazz Festival. Another one of our big clients is Abe Jacobs, sound designer for the production company connected with *Beatlemania*, and the *Evita* show, too. We put out a lot for a vast amount of custom equipment which was wired up especially for *Evita*. It will be playing in Los Angeles for about a year. We're also doing the Chick Corea and Neil Sedaka tours. We just finished doing a couple of Grateful Dead and several Devo dates. We've done many, many rock artists in the past from the Beatles to Jimi Hendrix.

MR&M: Are there any distinguishing equipment or operational characteristics in your sound reinforcement systems?

KD: We use processing electronics in all our amplifiers. We don't supply just an amplifier cable and speaker. We have an amplifier system that is tuned to the speaker system. The electronics are built right into the amp rack as a permanent part. It's a tuned, finished system. We don't tune our speakers to the halls like many people do with octave analyzers and pink noise. Our theory is that if you tune for the room, *maybe* you can get one spot in the room. If your direct sound is flat to start with, you've got half the battle won. This is a departure from what I feel is the norm in the sound industry; that is, taking a graphic equalizer and leaving it up to the operator about what to do at the last minute.

MR&M: Before we talk specifically about the ways you worked each of the six rooms and halls at the Jazz and Heritage Festival, it would be appropriate to talk about some of your key personnel.

KD: We have to start with Harry McCune, our founder and present Chairman. He's in his seventies and comes in the shop almost every day. He's got as much energy as any of the newer, younger employees. His son, Harry McCune, Junior is now President of the company. Mort Feld, the General Manager, is an idea man for new products. Bob Cavin, who manages the shop, is an amazing person, an amazing engineer. Bob is responsible for the designs of all the processing equipment, all the mixing consoles.

We have three McCune people here. Mike Neal is the head of our operation here. He came out earlier and spec'd the shows, figuring out what equipment would go where. His responsibility was as crew chief. My role is to be responsible for monitor systems, going along with my electronics background. I'm primarily responsible for any type of technical problems such as anything that breaks down. I've also been taking care of the stage arrangement along with our other McCune person, Flint

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McCune's SM-4 speakers as they were arranged aboard the S.S. President.

Ward, who is also one of our drivers. He did some mixing as well.

Class Acts

MR&M: Six rooms, two weeks, lots of great music. An infinite number of circumstances and requirements from a sound reinforcement standpoint. What we would like to do now is explore each facility, general requirements, and specific variations for certain acts. Let's start with the Theatre of the Performing Arts.

KD: It's a new theatre, with a capacity of about eighteen hundred. There's quite a bit of balcony seating, so there was a need to get sound distribution on a fairly wide vertical axis area, not like a gym where you have tier seating.

The shows we did there were Count Basie and Dave Brubeck. It's a well designed hall and the bands didn't need much in terms of sound reinforcement. They could be heard quite well without any kind of miking at all. We brought in three of our JM-3 speakers for each side for a total of six. We brought them in not mainly for power but to reach the balconies.

We used four or five mics on Basie and eight or ten on Brubeck. Basie had about twelve or fourteen horns. We put three mics in front of the stage by the horn section. When a horn player would do a solo, he would just walk up to the nearest mic. None of his horn players had pickups. Basie is a highly non-technical act.

Brubeck was very simple, as well, two mics on piano, a couple of mics on the drums. With a jazz act like that, you don't close mic the tom-toms. It's not necessary. It distorts the fact that they are playing the way they do in clubs, which is often without any sound reinforcement at all. We're trying to duplicate that. He had an electric bass and a sax player too.

Both these acts were a real great way to start off; professional and classy.

Rolling On The River

MR&M: The next night began a series of concerts on the S.S. President, a huge, 2800-capacity cruise ship that regularly plies the area around New Orelans harbor. Seemingly this would present a limitless series of considerations. Ramble on.

KD: Doing sound on a riverboat going up and down the Mississippi is a unique experience for any sound engineer. Mike Neal came out six weeks or so before the fest and looked things over.

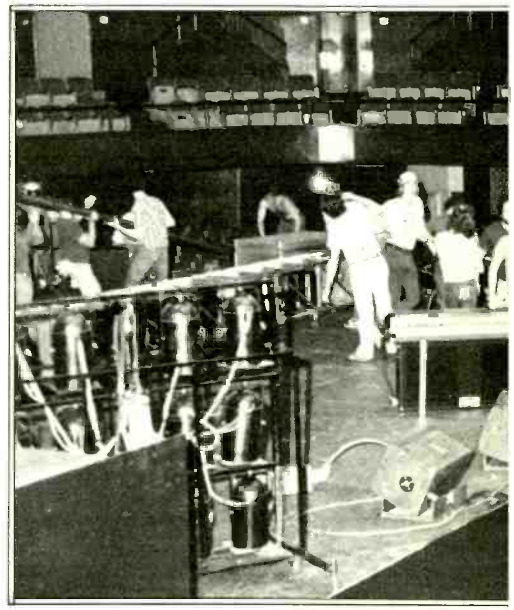
MR&M: How do you get the necessary power to run the show?

KD: We had plenty of help from their engineering staff; not one comprehensive engineer, per se, but people who ran the generators and knew how they worked. First, though, we and the lighting company had to make sure that there was sufficient power, ideally fed from a separate set of breakers.

In tying in the power, we brought in our own single phase distribution system. We had to run about two hundred feet of cable from the ballroom deck to the generator room. We snaked them outside the boat. No problem with weathering; we used large AC cables.

We also had to run cables to the upper decks, and that took a lot of manpower. Then we found out when we plugged into the generator and powered out our voltage that it was way too high, about 135 volts. All our equipment has pretty heavy power regulation in it and 135 volts is too high; the equipment will get too hot. So we checked the second generator of four on the boat. They fired it up, and that was okay.

The logistics of our setup schedule were affected by the fact that the boat is on a cruise every day from two to five. You can't do anything that makes any noise during that time, so we had to do things like soundchecks before and after the cruise. The shows started at seven, so we had two hours from five to seven, or else in the morning. Not everyone wants to do that.



McCune personnel begin the stag

MR&M: Was there any existing PA setup on the boat?

KD: None that affected us. There were high-frequency horns, like the ones you see in a gas station. They are used for tour narration, intelligible for that, but no way for music.

MR&M: Give us a topical description of what was hung, where, and why.

KD: We used McCune SM-4 speakers. This is a dual purpose speaker suited for a variety of duties. We hung a total of twelve of those in the room. Because of the way the boat

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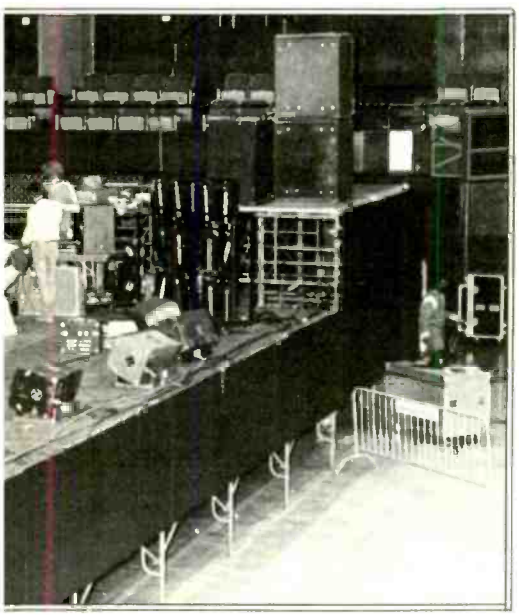
and the stage were set up, we couldn't possibly find a central place for a single source system. If we had attempted that, it would have ruined all the sight lines to the stage. These were hung at various strategic locations.

There were five digital delay outputs feeding different sets of amplifiers feeding, in turn, different sets of speakers.

The speakers and cabinets put up were McCune designed. They are based around an Altec 604, with some of our own parts, such as a voice coil, put in. Some of these shows were packed, with people making a good deal of noise. You had to get on top of that.

MR&M: The way the stage was centered, most of the length of the boat, consequently most of the seats, were to the sides as opposed to the area facing the stage directly. What adaptations did you make for this?



setup at the Municipal Auditorium.

KD: We chose our positions as carefully as possible, avoiding clutter. We had pretty much of a ring around the stage, some speakers around the bow. Our sidefills were placed to fill out the longer part of the bow, where the sound would die out from the stage. Also, we placed small speakers at the extreme ends of the boat.

We also put speakers on the outside at the very top deck by the pilot house, SM-3's worked here and covered it well. We had to turn it down, in fact, because of the harbor sound and noise regulations.

MR&M: Considering that you had this part outside, did you encounter any problems with weathering?

KD: Yes, we did. One night, we left those speakers up during a rainstorm. We covered them up with plastic, leaving enough room for the sound to get out. During that storm, the boat leaked. Rain came down through the roof near the stage where Taj Mahal was playing. It was like a garden hose, just above our stage box.

MR&M: The third deck was the snack bar deck. What did you do on that level?

KD: We had two SM-2 speakers. They were very effective, and not too loud, in case people wanted to just sit and talk.

MR&M: Let's talk specifically about some of the artists who performed on the boat.

KD: Let's start with Taj Mahal, where we had some unusual things going on. There was a steel drum player who had three steel drums. They're not very loud, but they do have a ring. We miked those under the direction of their sound man, top and bottom, six mics on three steel drums. On the bottom we used Electro-Voice 1777s; on the top, Shure SM57's.

MR&M: Did you use different vocal mics for different acts?

KD: Yes, we did. Our basic vocal mic is the Shure SM-58, particularly with unknown artists where we don't know how strong they are. A person with a weaker voice, who might be a bit closer to the mic needs a 57. We had a problem with Lee Dorsey. We tried a 58, but couldn't get any level. So we changed to an Electro-Voice 1777. Generally, though, we stick to the 57 or 58. The 58s have the lowest distortion at high levels of any mic we've tested.

Overall, though, you're held back by the fact that you have 31 or 32 artists for our part of the fest. Lots of set changes, so your choices have to be somewhat limited

MR&M: One night you had Dr. John, Fats Domino, and the Neville Brothers, three very different acts. What was that like?

KD: Dr. John was pretty straightforward. He had a magnetic pickup above the keyboard. It doesn't have the quality of sound a condenser mic does, but it does work. He didn't have a very big band. Fats is a real performer, he attracts attention, and people love him.

MR&M: The last night on the boat was something special; an evening highlighted by McCoy Tyner and Sonny Rollins. What are some of the aspects of each that stick out in your mind?

KD: McCoy Tyner's violinist played very loudly. We had a direct box on his amp, and it came off very nicely. He also has a percussionist with a good size tractor-trailer full of percussion equipment. He must have had a hundred different percussion instruments. We didn't know there would be that much. We planned on one mic, but we had to use more. He was tricky to do. He liked to pick the instruments up

Setup begins on the ballroom deck of the S.S. President.

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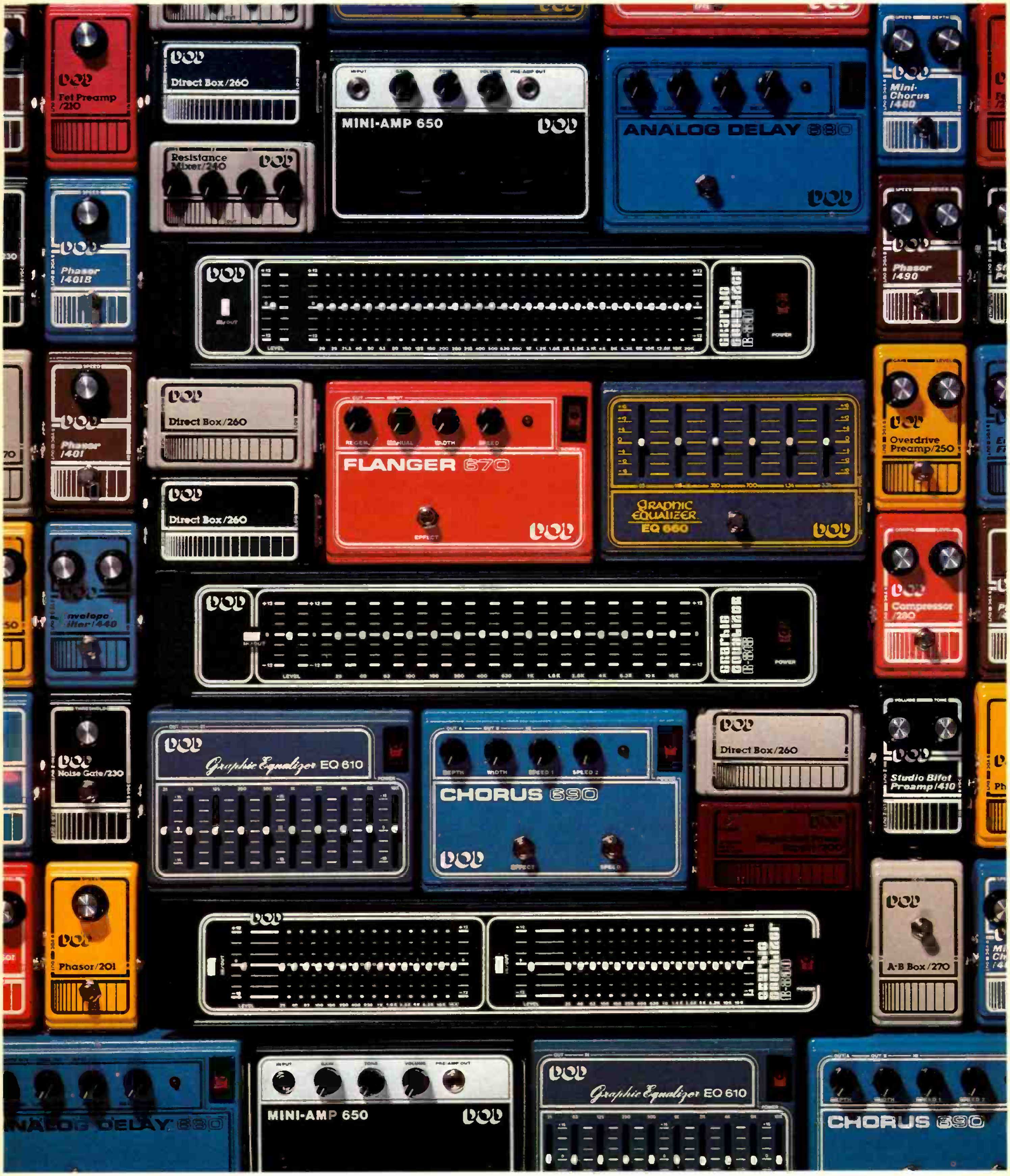
SOUND IMPROVEMENTS



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over his head, as opposed to playing them directly into the mic.

Rollins—Moving At 50

Seizing this moment to inject some personal commentary, festival attendees found musical outlets at various points for just about all emotional states and needs. Blues for anger; party music for moments of revelry, acts to challenge the folklorist maven in the brain. Yet it is fair to say that no one night had greater auditory promises of and fulfillment of ultimate satisfaction than the final night aboard the S.S President.

With a menacing late afternoon sky yielding to a misty twilight the grand boat accepted over three thousand patrons. A most forgettable collection of adept, yet quite ordinary chart readers, the New Orleans All-Star be-Bop Orchestra opened the proceedings. Not creative yet extremely tuneful in a Woody Herman-Kenton mold, the band set the setting for two of the giants of American music.

The ship left moorings, and outside, the rain commenced. Yet all were safe indoors as the only rain was the rain of bountiful, soulful notes from the tenor saxophone of the immortal Sonny Rollins, supported by the always relevant musings of pianist Mark Soskin, electric bassist Jerome Harris and drummer Al Foster.

Rollins, 50, a statuesque man at 6'2" is an imposing sight. His horn, in this case a Selmer Mark VI tenor, is virtually part of his body. Utilizing a Sony ECM-50 condenser mic stationed in the bell of the sax, Rollins was able to utilize all of the stage for mobility purposes. All ends of the creative spectrum were represented during his all too brief fifty minute set. Moments of rhythm found Sonny dancing back and forth; not, of course, in a choreographed jive funk style but in spontaneous steps of the spirit. Long, protracted blue notes, present on ballads such as "My One And Only Love," reveal Rollins in a moderately woeful pose, clamping down on his Berg Larsen mouthpiece with resolute force. At the same time, Sonny's particularly moving, protracted phrases are often matched by the pointing of the sax skyward, as in invocation.

Drummer Al Foster is one of these players who issue rhythm and melody at the same time. Boasting an extremely modest kit, Foster was his rumbling self, always a factor yet never overbearing and obtrusive like other more fusion-oriented jazz drummers. Foster learned his licks with Miles Davis.

Bassist Harris' electric fretless was played flawlessly, save for trivial timing problems during one of his solos. Soskin's electric fills were always appropriate and melodious.

On one unidentified tune, in the middle of the set, Rollins put tenor aside for an excursion with the Lyricon. Looking like a cross between a soprano sax and a clarinet, the Lyrican produces a sound closer to Wurlitzers heard at ice skating rinks. It had its own pickup wired directly through the board, and the skill of the player turned the natively grating sound into another instrument of emotional expression.

The only blotch on an otherwise flawless set were Rollins' brief greetings to the crowd, and a short speech at mid-set. These were delivered not by a stage vocal mic, but speaking through the small Sony in the bell of his sax. The somewhat incomprehensible utterances further underscored the fact that this unsurpassed sax mic was simply not built for vocals.

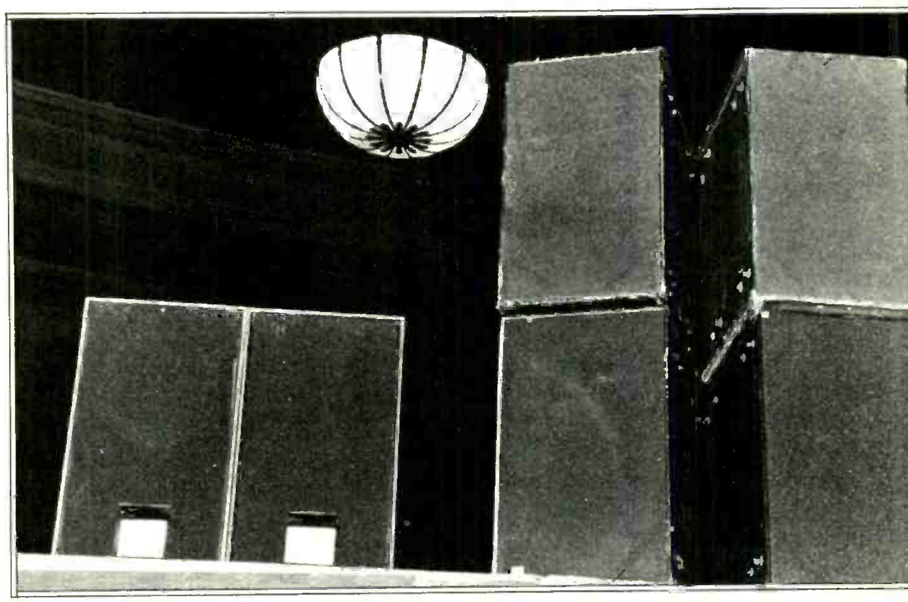
An Electric "Africa"

A surprisingly electric McCoy Tyner followed. He, of course, was and always will be acoustic. He once stated that "there is no electricity in the Africa where my ancestors come from." That should never be taken as self-righteousness because, for many, electric pianos are often shallow and pallid. Such an instrument would of course, do little to capture the musical waterfalls of onomatopoetically picturesque notes that have been identified with Tyner's music for well over two decades.

Acoustic bass, electric violin, drums, a percussionist with a whole array of toys, and a multi-dimensional sax player (tenor, alto, and soprano, on various tunes) joined McCoy and his paintings of gentle fury. While sax was without pickup, the amplified violin gave a sense of futuristic propulsion to a music whose roots, to a considerable extent, ultimately emanate from "the Africa."

Other Nights, Other Rooms

MR&M: On another night, you had one of the new Orleans Hilton Ballrooms offering a presentation of jazz greats on film. Since you worked this event, too, we're interested in details here.



McCune's JM-3 and JM-7 speakers provided the P.A. at the Auditorium.

KD: Wiring a room like that is pretty basic, for an audio-visual installation. We put up four of our SM-3's on risers next to the screen. We took a direct feed from the speaker output of the projector, via a projector bridge. There was a narrator, with a podium and a regular old SM-57 for lecture purposes.

The film was old. The soundtrack was scratchy, but people loved it. You could see the miking arrangements on the film, like one old, old RCA room mic.

MR&M: Next, you had a couple of acts in the Hilton Grand Ballroom. How

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did you work that up?

KD: That was another relatively simple arrangement. It was a relatively small venue.

We used two of our JM-3 speakers. They are a package system; a single cabinet that is triamped low, mid and high; Most of the design is proprietary. The overall package is extremely light. It's a clean setup, with speakers hanging out of sight. They are birch plywood, aesthetically attractive. We used a pair of them in the Ballroom, along with a pair of SM-3's.

It was a fairly standard hotel ballroom setup; acoustically dry and dead to some degree. The snare drum would bounce around a bit.

MR&M: The last night of the Festival, four acts played the Municipal Auditorium: Chic, Patrice Rushen, Gil Scott-Heron, and the Southern University Marching Band. What kind of room is it in terms of acoustics, and what modifications did you make to and for each of the acts?

KD: It's a big, old room with an interesting arrangement, lots of balconies that keep on going up and up. There's a stage about two-thirds down on one end. It's a house stage on an electric lift. It's level with the ground floor, but lifts up three feet if you want it to. It's propelled by an old leather belt that propels a series of shafts that lift the thing off the ground. That's not easy!

The house is large with a capacity of over 8,000. So we used eight of our JM-3's, four on each side. We also had two pairs of JM-7's on each side of the stage, which weren't used during the Chic show because the sound-man travelling with the group thought they were not necessary.

There was quite a bit of monitor mixing for Rushen, Scott-Heron, and Chic. We had our own designs working there, four monitor mixes out of an MM4. The house mixer was an MC-3, not one of our newer designs, but one which works well with this kind of presentation. It has two units; a main board and a side panel. The total comes out to about 34 microphone hooks and four submasters.

The auditorium was very reverberant and "live." If you walked towards the back, there was a pretty good delay, like the snare drum coming back twice. But there was a lot less of that in other places. This is common in arena seating, where at the rear of the hall, by the time you hear a drum beat bounce off the wall, a new one is being hit, and it sounds jumbled. A critical listener that wanted to pick out everything like he was in front of a pair of studio monitors would not have enjoyed it, but the people in the back were just having a good time.

MR&M: The mixing board was situated a good distance from the stage. How did your mixers communicate with onstage personnel?

KD: That was pretty straightforward, really. We have a Clearcom Communication headset and power-pak. We locate it near the amplifiers on stage where our power feed comes up. Most people locate the power pak by the mixing board out in the house. Our theory is not to do that, because if the power fails by the mixing board, what happens then? There's a yellow flashing light to get the attention of the stage contingent.

MR&M: What was it like working with Patrice Rushen?

KD: She was very strong and demanding at the soundcheck. She wanted to hear stage sound without any monitors or sound system. I warned her twice that we didn't have the time for it, but she did it anyway. She's been on a headline tour and she wasn't used to this kind of setup, but we managed to get by.

MR&M: How about Gil Scott-Heron?

KD: Super-simple and easy, with no heavy requirements. They asked for three monitor mixes just for convenience. There was a weird noise from the keyboard end that we managed to stop before the show.

We had pretty much of our basic layout. Gil's acoustic piano player had SM-57's because of his position on stage with monitors and guitar amps around him, we didn't want any condenser mics near him. There were three horn players; the sax player was one of the best at the festival. For these we used SM-57's and on Gil's vocals, an SM-58.

Chic, Chick, Flora and The Future

MR&M: The last band, Chic, was the most ornately staged group at the whole festival.

KD: Yes, it was. Where do we start? Well, they had two wireless guitars; they had transmitters on their shoulder straps and receivers on their amplifiers. For the guitar player, we **took a feed off his receiver, rather than the speaker. The band had violin** players doing parts normally associated with and for horns. They had three direct boxes with three feeds. The drummer had a full compliment of mics—even every tom was miked.

MR&M: The sixth venue was the Saenger Theatre where Chick Corea and Flora Purim played.

KD: Saenger was supposedly remodeled—lot of balcony seating going' way up, not too much house seating. About 2,000 capacity.

Chick was already on the road with two other McCune people on a sixty-day tour. We got together with them and tied in our systems. They used their monitor mixes, and their mics. Everything was compatible electronically; the same snakes, and the same splitters.

The room was real easy, pretty reverberant. Three JM-3s on each side gave us fine coverage. One or two of them on each side could have done in terms of power; the third was primarily for balcony fill. We also used JM-7s, which helped us out with some of the low ends.

The McCune contingent with Chick had some Sennheiser mics that we didn't have; They also had a different house mixing console; about eight or ten direct boxes. This resulted in some of the best "live" sound I've heard in some time.

Flora had something unusual, two microphones side by side on the stage. One was going through an Echoplex; the other went directly into the house mixer. She slid from one to the other. These were her own mics—Electro-Voice PL71s.

MR&M: In wrapping up, what are some of the things you see in the future for McCune and for yourself?

KD: We're a technical company overall. We have our own research and development staff, we have manufacturing capacity. We're in the forefront of high-tech operations and innovation. What I would like to see is a marriage of our technology and some of the rock acts that work with companies that still haven't solved the problem of sight lines. We'd love to expose them so some of our ideas, and our service abilities. I mean bands like Yes, and Emerson, Lake and Palmer. I'm a piano player myself, so I look at things from both a musical standpoint and from my college training in sound engineering. Individually, I would personally love to work with McCune in such a capacity.

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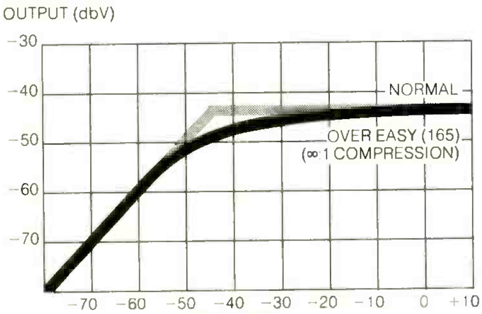

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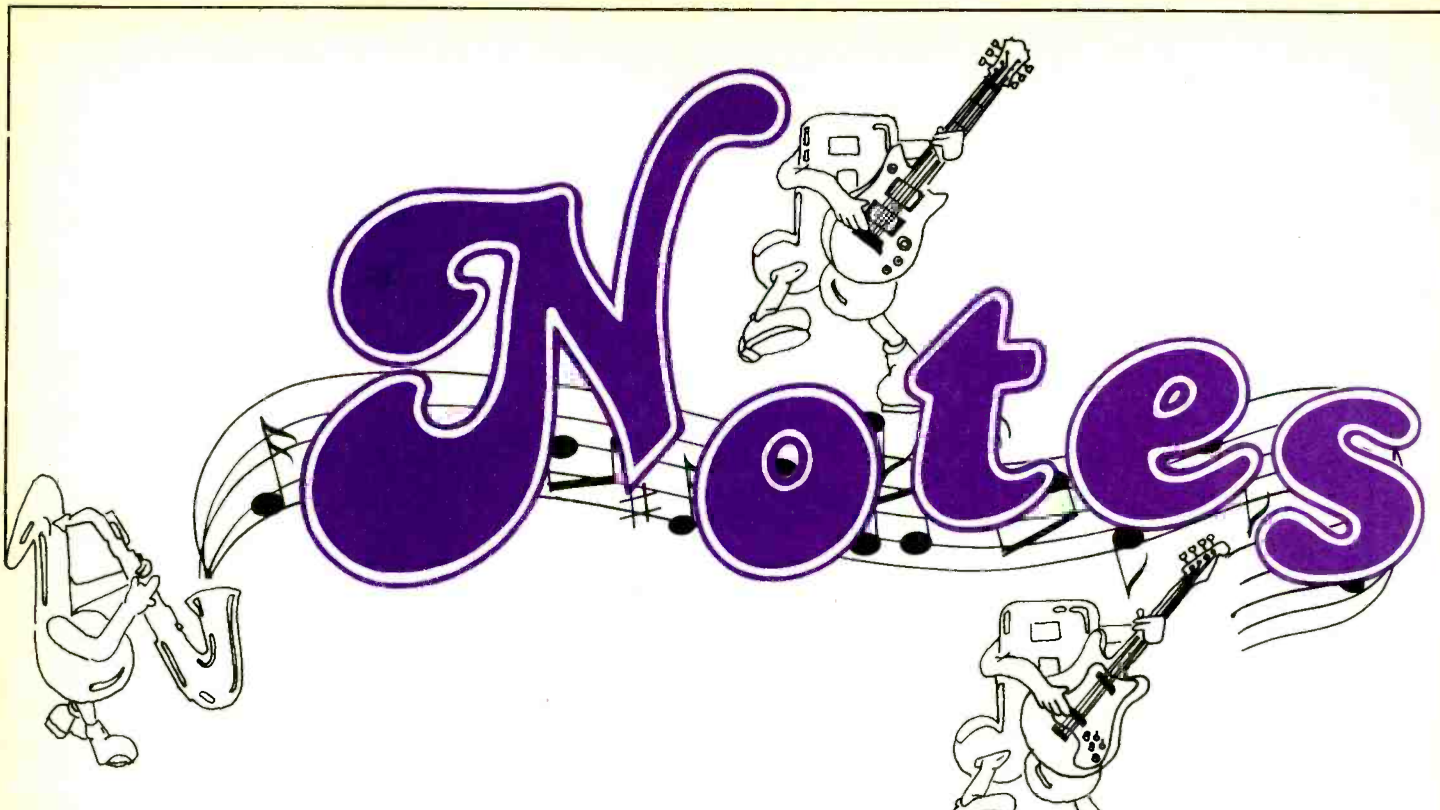

Beigel Envelope Controlled Filter

Virtually all musicians have, at one time or another, played with some type of envelope controlled effect. The most popular of these is the envelope controlled filter (ECF for short), which consists of a filter that can vary the timbre of the sound it is processing, and an envelope follower that senses changes in the dynamics of your playing. The resulting sound is similar to a wa-wa pedal, but with an important difference: unlike a wa-wa pedal, where rocking the pedal back and forth alters the filter's resonant frequency, the envelope follower changes the filter's resonant frequency in synchronization with the dynamics of your playing. For example, playing loud would affect the filter in the same way as pushing down on a standard wa-wa pedal, while playing softer would create the same *effect as if* you pulled back on the pedal.

The advantages of an ECF over a conventional wa-wa pedal are two-fold. First, the envelope follower circuit that senses your dynamics can create more rapid variations in filter response than you could ever have by rocking a pedal back and forth with your foot, and second, the effect is always synchronized to your playing. There are other ways to modulate an effect—the cyclic oscillator used in phase shifters is one example—but I greatly prefer envelope followed effects for the natural, animated quality they impart to an instrument. A good ECF is much more than just a super wa-wa, it can give a whole range of timbre effects that would be impossible to obtain otherwise.

Incidentally, sometimes these effects are simply called "envelope followers." However, this is inaccurate because an envelope follower is only one sub-module of the envelope controlled filter; an analogy would be calling an amplifier a "tone control" just because it happens to have some built-in tone control circuitry. Also, filters are by no means the only devices eligible for envelope control. Phase shifters, flangers, voltage controlled amplifiers and many other basic sound modifiers give excellent effects when controlled by an envelope follower.

WHAT is IT? The Beigel Sound Lab Envelope Controlled Filter combines a sophisticated envelope follower and state variable filter in a single, rack-mounted package. Priced at $495, it seems intended for the high-end musician and studio market.

Probably the most intriguing aspect of the Beigel Sound ECF is the number of controls—how many other ECFs have you seen with six knobs, seven switches and eight jacks? Not very many I would venture to say . . . and these controls aren't just there to impress. Each one serves a valid musical purpose and greatly enhances the versatility of the unit.

The various jacks, all located on the back panel, interface with studio consoles or rack-mounted stage units. There are both balanced (XLR three-terminal jack) and unbalanced (¼-inch phone jack) inputs and outputs. A switch selects between the balanced and unbalanced inputs, so you can only use one of these at a time; however, the outputs may be used simultaneously. This is handy for when you want to send the unbalanced line to a guitar or similar amp, and the balanced line directly to a studio console for direct injection.

Unlike most envelope followers, the Beigel ECF includes an effects loop to allow insertion of another effect (such as fuzz, flanging, etc.) in the signal path. There is an excellent reason for including this loop. With conventional ECFs, the

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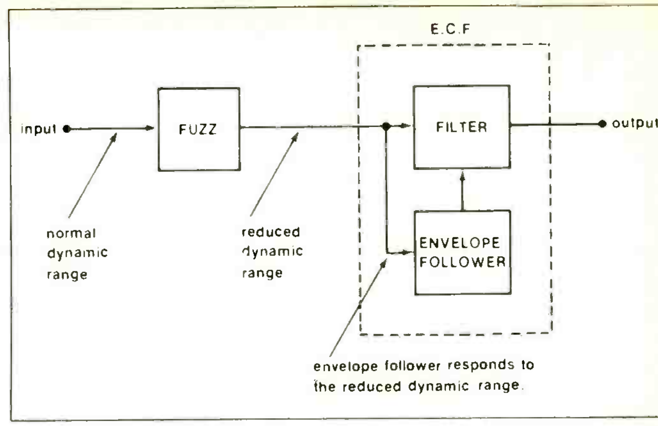
filter is controlled by whatever signal is at the input of the device. This is fine if your instrument plugs directly into the ECF, but consider what happens if you insert a fuzz between the instrument and the envelope controlled filter *(see Figure 1).* The fuzz greatly reduces the dynamic range of the instrument signal, and as a result, the envelope follower has precious few dynamic range variations to follow. This produces a rather uninteresting sound compared to what you would expect from an ECF-type device.

Figure 2 shows what's happening with the Beigel unit. By feeding the envelope follower with the unprocessed input signal instead of the fuzzed signal, the filter reacts to the full dynamic range of the instrument and therefore modifies the fuzz in a more animated manner than would occur if the filter followed the less dynamic fuzz output.

The effects loop is a real nice addition, but we're not finished yet. There is also a stereo footswitch jack (with associated footswitch assembly) that controls the *effect in/out* and *effects loop in/out* functions (these may also be controlled via toggle switch at the front panel, and there is a pair of associated panel LEDs to indicate when each function is selected.) So, if you had a fuzz in the ECF's effects loop, you could select the fuzz alone, the ECF alone or the fuzz filtered by the ECF. Incidentally, I keep mentioning fuzz only as an example—other effects also work well in the effects loop section. The remaining rear panel jack is an envelope output, which provides a 0 to +10 V DC output that is proportional to the input envelope.

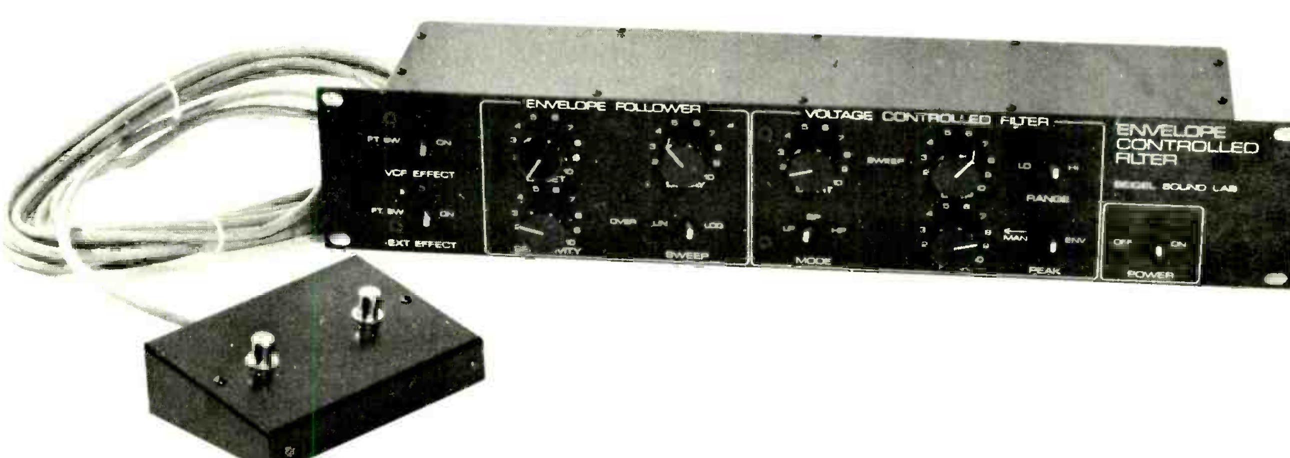
We complete our tour of the back panel with a glance at the fuse post (¼ A type) and the line cord (two conductor). My only complaint about the back panel arrangement is that I prefer to have the most often used jacks accessible from the front panel as well as the back. I know that having patch cords dangling across the face of a piece of equipment is somewhat messy, yet anybody who has ever worked with a modular synthesizer will recognize the value of having accessible patch points. There's nothing wrong with rear panel patch points that are semi-permanent (i.e., preamp to power amp and the like), but patch points that are used often—such as effects loops—should preferably be paralleled with rear panel connections or at least located on the front panel.

The front panel controls will make more sense if we describe their functions as we evaluate the effects obtainable with this unit. Onward to the mechanical evaluation . . .



Mechanically, the Beigel ECF is at the very least adequate and in most respects is outstanding. The analog ICs are a mix of 4558s and LF353s (the former appear to be for the control functions, the latter for the audio sections), and opto-isolators are used to control the filter's resonant frequency and for signal switching. For those of you who are not familiar with opto-isolators, these are four-terminal parts. Pumping current into one pair of terminals (which connect to an internal LED) changes the resistance appearing between the other pair of terminals (which connect to an internal photoresistor that is illuminated by the LED). This resistance varies over a wide range, and you can pass audio through the photoresistor section without picking up any significant noise or distortion. In essence, the opto-isolator is a current-controlled resistor that, while expensive, is less prone to clicking and noise than analog switches or conventional VCAs. In the ECF, one dual isolator controls the frequency of the filter, while four other isolators control the in/out switching for the main effect and the effects loop.

The feel of the box itself is very sturdy. Allen screws are used extensively (including the set screws on the knobs); the transformer and power supply run cool (an important consideration since there are no ventilation holes) and the circuit board is epoxy glass. All ICs are socketed to allow for easy maintenance and upgrading. The pots are not exactly the most costly ones I've ever seen, however, this can be overlooked considering the fact that only one of the controls



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actually carries audio. The other pots carry control signals that aren't affected very much by occasional pieces of dirt and crud on the pot contacts.

PRE-FLIGHT for the BEIGEL ECF: The ECF is easy to set up. Plug your instrument into the input and patch the output to a power amp or studio console. With the amplifier volume control down, plug the line cord into the wall and turn on the power switch (the associated LED indicator above the switch should light). Turn the amp volume back up and you're ready to roll. The Beigel ECF is quiet enough to accept low-level signals, but also has enough headroom to accept high-level signals as well.

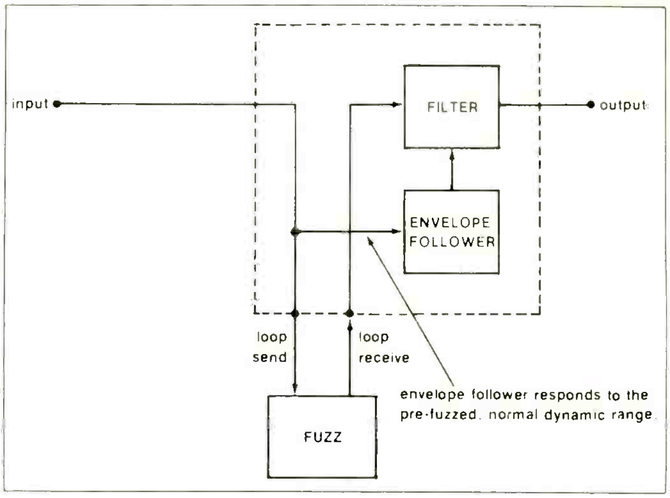
The ECF's many controls give you the flexibility to make some really pretty sounds (as well as some exceptionally ugly ones if you don't know what you're doing). As a result, this not an "intuitively obvious" effect where you can just plug things in and get going. It's necessary to read the supplied instructions (a little too sketchy for my tastes, but adequate) and carefully run each control and switch through its paces to observe the overall effect on the sound. Which, in fact, we'll do right now.

EVALUATING the FILTER SECTION: The filter includes three controls (START, STOP and PEAK) and three toggle switches. One three-positon switch determines the filter mode—LP, BP or HP (lowpass, bandpass or highpass); one switch determines the filter range (LO, 150 Hz to 5 kHz, or HI, 350 Hz to 12 kHz); while the third switch determines whether the resonance is set by the manual PEAK control or whether it follows the envelope dynamics (where a louder input gives more resonance is set by the manual PEAK control or whether it follows the envelope dynamics (where input gives more resonance). Note that is is not possible to obtain a notch response out the ECF's filter; while this is not a very important drawback, notch filters do give some nice effects.

The PEAK control is easy to figure out—it corresponds to the resonance or Q controls found on most filters, and probably should have been named resonance or Q instead to avoid confusion. The START and STOP controls are a little different from the norm. START sets the initial filter frequency when there is no envelope, and STOP sets how high the filter frequency will go with maximum input envelope.

Let's start our evaluation by temporarily ignoring the envelope follower section and simply checking out what the filter sounds like. According to the instructions, "If the START and STOP controls are both set to the same value, the filter frequency does not change at all (as you play): the unit then performs similar to a one-channel parametric equalizer." That sounds like a good place to begin. We switch the effect in/out switch to *on* (noting that the switching action is indeed very quiet and that the associated "effect in" LED lights up) and the ECF is ready to start filtering.

All control settings are labelled so that fully counterclockwise is 1 and fully clockwise is 10. In the LP mode, setting the START and STOP controls at 1 removes most high frequencies from the audio signal. Turning both controls equally clockwise allows more and more highs to



pass through. In the bandpass mode, the response is somewhat different; with the controls fully clockwise, there is a peak in the bass range. This peak moves up through the frequency spectrum as you turn both controls more clockwise.

The highpass mode offers another type of filtering action. With the START and STOP controls at 1, all frequencies are present. Turning the controls clockwise removes more and more low frequencies until at the full clockwise setting, the only energy left is in the high-frequency ranges. So between these three modes, we have quite a bit of latitude as to how we can process the tone of a given input signal. Turning up the PEAK control in any mode (provided that the peak switch is in the MAN position) adds an additional sharpness to the sound that resembles whistling at extreme clockwise settings. Remember, though, that highly resonant sounds are difficult to capture on tape without some kind of limiting or compression, so leave the PEAK control on as low a setting as possible consistent with obtaining the effect you desire.

As a filter, the ECF does just fine except for one reservation: there is no way to parallel the straight and filtered sounds at the unit itself. A filter, by its very nature, removes part of a sound; so after being processed by a filter, a signal is always thinner sounding than it was before processing. With lots of resonance, this can produce a very tinny sound that has few (if any) musical applications. However, adding in some straight signal by Y-ing the input (as shown in *Figure 3*), and feeding both outputs into a mixer, entirely changes the picture. We can now keep the straight signal up at a high level and mix the effect in for extra emphasis. Many filtered sounds that initially appear useless by themselves acquire a whole new significance when paralleled with unfiltered sounds. While it is possible to create these parallel sounds by using the effects loop out jack to provide a straight channel feed (see *Figure 4*), this requires the use of either a mixer (in the studio) or a two or more channel amp (for on-stage use) and also ties up the effects loop. It would be far more convenient if the ECF included some kind of balance control to choose between filtered and unfiltered sounds.

ENVELOPE FOLLOWER CONTROLS: The envelope follower has three controls. A SENSITIVITY control matches your dynamics to that of the envelope follower, and

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there is an associated overload LED that allows for optimal adjustment of this control. A pair of controls, labelled ONSET (attack) and DECAY, determine how rapidly the envelope follower responds to changes in dynamics. With minimum onset, the follower will jump immediately to full on when the input signal goes from full off to full on. Add a little bit of onset, and the follower response will lag behind the signal. The DECAY works in a reverse fashion. With minimum decay, the follower will go immediately to full off when the signal input goes from full on to full off. With maximum decay, the follower will glide down more slowly to full off after the input has gone from full on to full off. Note that the onset can never be faster than the attack time of the signal, nor can the decay time be shorter than the decay time of the signal itself.

Whew! Well, we're not finished yet. A LIN/LOG switch sets how the envelope follower responds to changes in dynamics. The LIN position gives the widest dynamic range response, while the LOG position compresses the response somewhat. In the LOG position, dynamic changes are not as drastic as those that occur in the LIN position. And don't forget that all these controls interact with the START and STOP controls of the filter . . .

PUTTING IT ALL TOGETHER: Now that we've fathomed the inner secrets of these building blocks, it's time to see what happens when we make these various submodules work together. We'll start off with minimum ONSET and DECAY, the START control at 1, the STOP control at 10 and the PEAK control about halfway up (the more resonance, the easier it is to note changes occurring in the filter).

The START/STOP settings specified above ensure that the filter will initially start off at its lowest resonant frequency, and then sweep up to the highest resonant frequency as the input signal hits maximum. Play a few notes and adjust the SENSITIVITY control so that the overload LED flashes on the peaks of your playing. (Incidentally, this control is very sensitive; feeding in a guitar that had been preamped up to line level meant having to turn the control *way* down in order to keep the LED from staying lit virtually all the time.)

Now is a good time to note how the START and STOP settings vary the sound. If the STOP control is set to a higher value than the START control, the filter sweeps from the lower frequency specified by START to the higher frequency specified by STOP before gradually decaying back to the initial START frequency. With the START control set to a higher value than the STOP control, the filter sweeps downward from the higher frequency specified by the START control to the lower frequency specified by STOP and eventually decays *upward* to the initial START frequency. As a result, you can sweep the filter in a narrow or wide fashion, upwards or downwards, any place within the filter's range. The LO/HI switch further expands the range of the filter. The LO position is good for bass, guitar and wa-wa effects; the HI position is good for adding treble enhancement and "sheen" effects to a sound.

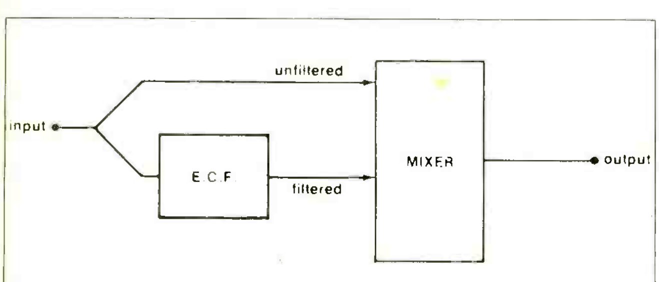
While the ECF did an excellent job with single notes, I started getting terrible distortion whenever I played chords. This usually indicates that the response of the envelope follower is too fast and is therefore responding to all the minor changes in level that occur from strings beating against each other and from harmonics interfering with, and adding to, each other. Turning either the ONSET or DECAY control (or both controls) slightly clockwise from setting 1 damps (slows down) the envelope follower response slightly to eliminate any distortion when playing chords.

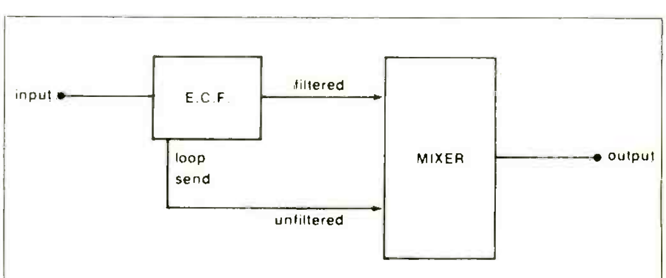
Turning the ONSET and DECAY settings further clockwise produces a sound that is much more loosely synchronized to the dynamics of your playing. While a fast responding envelope followed filter gives primarily rhythmic effects, slower responses produce a dreamier, legato, less percussive sound that is useful for subtle applications.

Of the two choices offered by the LIN/LOG switch, I tended to gravitate more towards the LIN position. However, I can see where the LOG setting would definitely have its uses (for example, if you had an instrument with a wide dynamic range—such as drums—and felt that the LIN position gave too much emphasis to these dynamics).

APPLYING the ECF: Because the ECF responds to a player's dynamics, the overall effect is pretty rhythmic. When used with guitar and bass, and with the envelope follower set for a fast response, you get those "funky" effects featured on many of the more electronically oriented disco and fusion cuts. Since filters are also associated with synthesizers, adding filtered sounds to a stringed instrument (especially if you put something like an octave divider in the effects loop) produces a very synthesizer-like timbre. ECFs also work well with drums, vocals, even tape tracks.

When used in parallel with an unfiltered channel, the ECF provides animation and enhancement. For example, sweeping the upper filter range in the HP position, and mixing that in the background, produces a very lyrical, wistful type of effect. Sweeping a narrow portion of the lower filter range while playing heavy rhythm guitar adds to the heaviness. One of my absolute favorite envelope followed sounds is something I occasionally use when processing voice. I sweep the filter over about an octave of the upper midrange (say, 2.5 kHz to 5 kHz) in the BP mode and mix this in with the straight signal. The filtering action accentuates sibilants and





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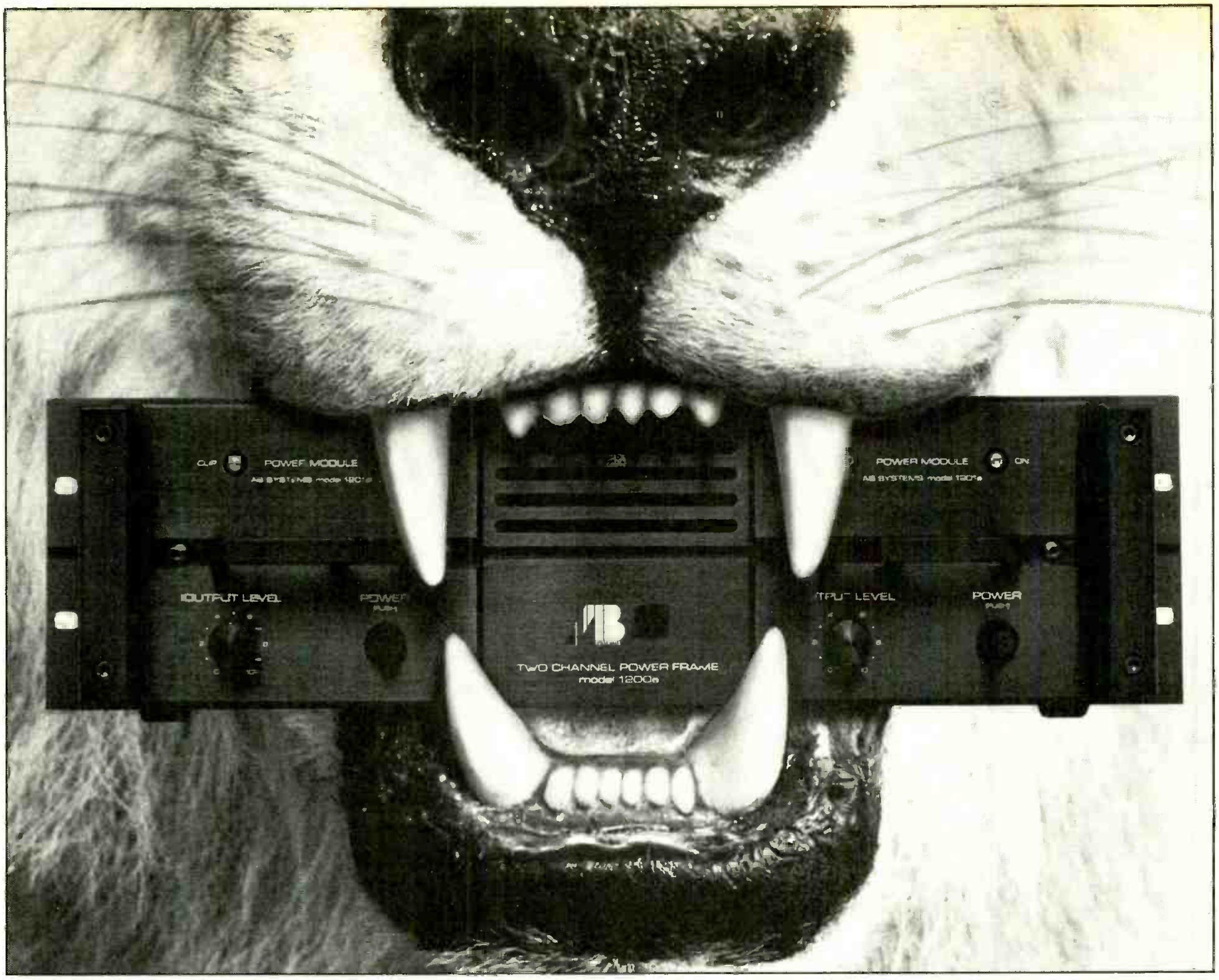

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mouth sounds, creating a warmer, more "human" effect. Subtle envelope controlled filtering on string synthesizers also works very well. By the way, you might assume that you couldn't use an ECF with something like an electric organ, since the organ essentially has no dynamic range. There is a way around this problem. Simply patch a flanger or phaser in between the organ and ECF. The notches and peaks introduced by these effects will alter the dynamics enough to create envelope follower response changes.

OVERALL EVALUATION: One of the most important advantages of the Beigel Sound ECF is that you can add effects into the effects loop. Without this feature, it would be difficult to get a convincing envelope followed filter effect when the ECF is preceded by fuzz, compression or any other device that restricts the dynamic range. I would like to see one extra jack, though, for an external envelope input. With this jack, you could feed one instrument through the ECF's filter, but make the filter respond to a different instrument altogether (such as drums). I have often used this effect while recording, and it is extraordinarily effective for creating dynamic, rhythmic sounds.

Another strong point about the ECF is that you have an amazing amount of control over the sound because of the full complement of controls. From upward sweeps, to downward sweeps, to treble sweetening, to bass popping, the ECF effortlessly handles a number of different effects. The noise performance is superb (especially with higher level signals), all switching is quiet and the construction is excellent.

In fact, there are so many positive things that you could say about the Beigel ECF that it's too bad there is no onboard provision for blending the filtered and unfiltered sounds. I know I've been hitting on this pretty hard, but it bears repeating. Without a doubt, the most useful, least gimmicky, most creative and least stereotyped effects I've ever gotten with any filter involved paralleling the filtered and straight sounds. If you'd like to experiment with parallel sounds, those of you using the ECF in the studio should have no trouble; simply Y the input of the ECF (or take a separate feed from the EXT OUT jack) and plug this into a separate console channel from the standard ECF output. The ECF outputs are in-phase with the inputs, so cancellation will not be a problem. Musicians using the ECF with a two-channel amp can achieve similar results by patching the ECF output into one channel and, again, either Y-ing the input, or taking a separate feed from the EXT OUT jack and plugging this into the amp's second channel.

Those of you who appreciate envelope followed effects will no doubt be impressed by the the Beigel Sound ECF's multiple capabilities. This effect is not for everyone; it will be too complex for some people to handle and too expensive for others. But in the hands of a sensitive operator who appreciates the circuit enough to learn how to properly use the various controls, the Beigel Sound ECF is capable of sounds that range from the subtle to the dramatic. I hope that if nothing else, this unit will turn other musicians on to the world of sounds created by envelope control.

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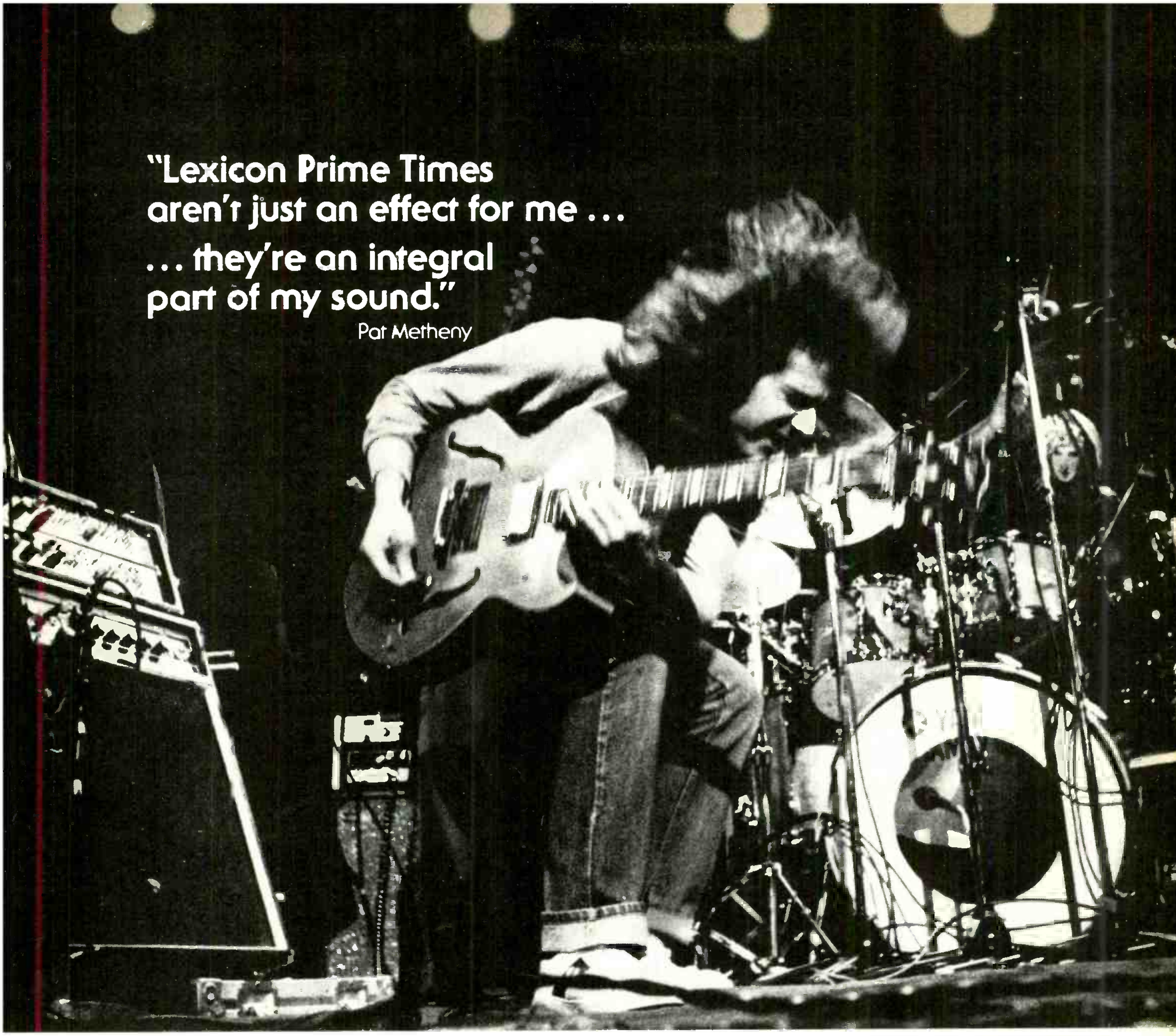

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Ambient Sound

BY LEN FELDMAN

Digital Update

Having recently returned from another trip to Tokyo and a visit to the combined Tokyo Audio Fair 1980 and the Japan Electronic Show (everything other than audio, including video, test equipment, component parts, etc.) I can tell you without much hesitation that if you are in the recording business today, life in your studio won't be the same ten years from now (or perhaps even sooner). If you are an audio enthusiast who simply likes to listen to well recorded music over a good stereo system in your living room, the complement of equipment with which you fulfill that need won't be recognizeable in ten years, either. All of us who write about audio and sound recording have been talking about the coming digital audio revolution for a couple of years now, and perhaps you, our readers, were fast becoming convinced that we were doing so simply because we had nothing else to write about and wanted to show that we were aware of the latest laboratory technology.

Well, we are past the laboratory protoype stage as far as digital audio mastering in the studio is concerned and, from all appearances, we are past the laboratory prototype stage when it comes to digital audio tapes and digital audio tapes and digital audio disc possibilities for the home. About the only thing that can hold up digital audio discs for home use now is the kind of industry in-fighting and competition that killed quad sound back in the mid-seventies (remember?). And from the looks of things, the big fight is going to be between those who favor a combination digital audio/video disc and those who would prefer to see a digital disc dedicated completely to audio.

The Combination Video/Audio Disc

As you probably know, the currently available laser-optical video disc format embraced by Magnavox and Pioneer has the necessary bandwidth capability to handle PCM (pulse code modulation) digital audio. So, too does the disc format which JVC and Panasonic intend to introduce sometime late next year. Infact, at the recently concluded AES Convention in New York City [November 1980], I saw a demonstration of a capacitance pickup grooveless disc from JVC (they call it the "AHD/VHD" format, for Audio High Density or Video High Density) which featured *three* digital audio channels *plus* the capacity to store a still-picture digitally. This disc is some 10 inches in diameter and is housed in a sort of "caddy" to prevent damage to its surface. The same disc would be used for ordinary video which, of course, would *not* not be digitally stored, and for an associated audio sound track (or two), again neither of which would be digital. So, what we have here is a disc that can be used for ordinary video or digital audio. The advantage, as claimed by JVC, is that the consumer would be able to make do with just one new kind of disc player. The disadvantages, not talked about by this proponent, are the fact that a PCM processor would have to be purchased in addition to the player itself in order to enjoy digital audio discs. From a technical point of view, too, the combination audio/video disc raises questions in that the PCM digital audio is being force-fitted into a format which was essentially designed to suit NTSC television signals. The audio signals (in digital bit-encoded form) are literally being inserted "between the lines" (the video lines, that is).

In Favor Of a Dedicated PCM Audio Disc

Until recently, I might have been convinced that a single disc player and a single type of disc for audio and video made sense. That was before I heard the arguments for and saw the new compact digital audio disc now being proposed jointly by Philips of Holland and the Sony Corporation. This newly perfected disc measures only 12 centimeters in diameter (roughly 4¾ inches) and uses about one sixth the area of a conventional 12-inch LP. Playing time for a single side is 60 minutes, in stereo, and since there is no contact between the laser pickup and the disc, records of this sort may be

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expected to last indefinitely, with absolutely no deterioration in sound quality with repeated playings. Just as in the case of optical video discs, these laser-tracked digital audio discs do not need to be stored in any sort of caddy or holder to protect their surfaces. Sony engineers told us that the one remaining problem—the high cost of conventional lasers—can and will be solved shortly through the use of much less expensive semi-conductor lasers.

The player that we saw demonstrated was so small in size that one could easily envision applications in moving vehicles, an application that is not lost on the sponsors of this disc format, you can be sure. The projected price for this player, in Japan, translates to around $500, but would probably be higher by the time it reaches the U.S. market, both because of inflation and because of import duties, shipping costs, etc. Aside from the very attractive packaging of the disc and player, I see another important reason why digital audio should have its own standard form of disc, unrelated to any of the video disc formats. As many readers probably know, different regions of the world employ different TV transmission standards. In Japan and the U.S. we have the so-called NTSC system. In other areas there are the PAL and SECAM systems. Presently, video cassette recorders must be built differently for these different parts of the world and, were we to settle for a combination video/digital audio disc player, we would not only need players for different regions of the world, but there would be no compatibility between discs used in different parts of the world. Can you imagine how the analog record industry we know today would have been stifled years ago if it had had to face such a problem of incompatibility? Many of us can recall the relatively simpler problem of having to inventory mono and stereo records during the difficult years of transition from monophonic to stereophonic sound, but that was a small matter compared with what we are talking about here, I think.

Compatibility at the Pro Level

Things are moving along even more rapidly when it comes to studio digital audio equipment. We are seeing more and more sophisticated editing equipment, multitrack digital equipment, and the like. It shouldn't be too long before we begin to see a transition at the console level itself. Here, too, the question of standardization arises. There are, of course, rotary head recorders (based again upon video technology, such as Sony's U-matic or equivalent) and there are also a growing number of stationary head systems. In a recent survey of available rotary head machines, we counted no less than five different digital sampling rates (from 30 kHz to 47.25 kHz) and quantization (bits per channel) of from 12 to 16 bits. In the case of stationary head systems, we counted eight different sampling rates (from 32 kHz to 52.0 kHz) and quantization of from 12 to 16 bits. Tape speeds for these stationary head machines also varied over a wide range, from 38 cm/sec (15 ips) to 114 cm/sec (around 45 ips).

In Japan, companies such as Sony plunge ahead with more and more elements of their PCM-3300 Series, such as their new PCM-3324, a 24-channel digital audio recorder (several generations removed from their earlier PCM-1600, a processor that worked in conjunction with their U-Matic VCRs).

In the United States, 3M Company ccontinues to build and install their remarkable 32-channel digital recorders which use 1-inch wide tape, and their companion 2/4-channel mastering digital recorders, as well as a whole variety of new fully digital editing equipment.

Error correction systems differ, speeds differ, number of bits differ between systems and the thought of universalization and world standardization seems even further away when it comes to professional digital audio than it does in the case of consumer audio discs or tapes. Of course, from a practical point of view, the self-contained studio may not be as concerned with world compatibility as the consumer is sure to be. It would be nice, however, if one studio of the future could at least be able to transfer signals, while they remain in the digital domain, from their audio recorder and console to ones of a different brand in another remote location anywhere in the world.

In an effort to promote that goal, the people at 3M recently proposed standardization of digital systems in several areas. As they point out, the adoption of a universal machine may be too much to hope for in the foreseeable future, but establishment of a signal standard at an early date would represent a major step forward and one which could head off potential future problems for studios around the world. The areas of standardization which 3M proposes are as follows:
Sampling rate; digital code to represent data; serial or parallel format; system to allow for small variations and time delay; provisions for transmission of extra bits for data other than audio; means of locking the systems together with a common clock; driver-receiver configuration and impedance.

They recommend the following standards to accomplish those goals:

- A sampling rate of 50 kHZ
- Data code to be 16 but 2's complement
- Serial data transmission with MSB first and LSB last.
- Provide quadrature clocks for outputting and inputing data to prevent time delays causing erroneous data reception.
- Provide space in the serial data stream for up to nine extra data bits.
- Two standard clocks to synchronize systems (2.5 MHz for transmission and reception clock; 50 kHz clock to synchronize word rates).
- Use 50-ohm line drivers and receivers.

Efforts on the part of 3M and others to some order into the chaotic world of digital audio is to be commended. Anyone wishing to obtain further information on 3M's proposed standards is urged to contact Robert J. Youngquist, research manager, Mincom Division, 3M Center 236-GN, St. Paul, MN. 55144, or call (612) 736-0286.

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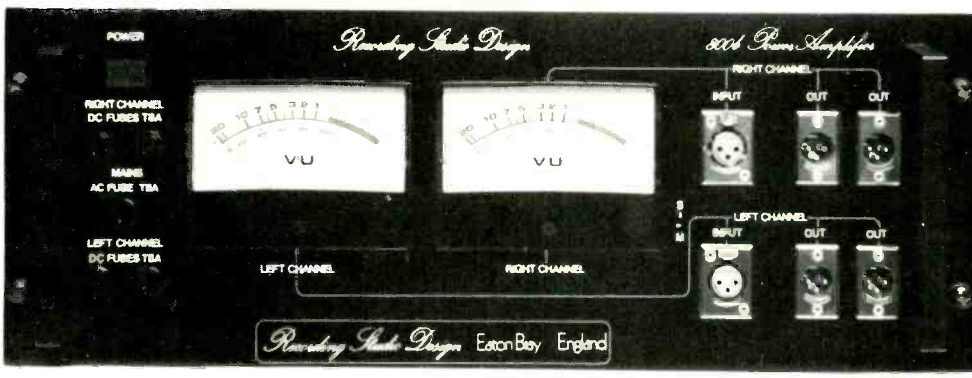
NORMAN EISENBERG AND LEN FELDMAN

Studiomaster 800B/C Power Amplifier



General Description: The model 800B/C Studiomaster (made by the British firm of Recording Studio Design) is a dual-channel (stereo) power amplifier rated for 225 watts per channel into 8-ohm loads, or 400 watts per channel into 4-ohms loads. The front panel contains more than the usual complement of features, including several that are customarily found at the rear of an amplifier.

At the left are the power off/on switch as well as fuse-holders for both signal channels and for the AC power line ("mains" as called in England). At the right are the signal input and output connectors. Each channel has one input and two outputs. These all are XLR three-pin connectors.

Between the switch-fuse group and the connectors there are two amply proportioned VU meters one for each channel, that are calibrated from −20 to +3, with "zero" representing the nominal maximum rated output. Large handles are fitted at either end of the panel.

The rear of the amplifier contains the line-cord connector (a three-prong line cord was supplied with the unit); an operating voltage adjustment with settings for 130, 220, 120, 240, 110 and 230 VAC; and two fairly large ventilating fans.

Test Results: Although relatively few (for an amplifier) specs were supplied with the Studiomaster 800 B/C, *MR&M* did run most of the usual series of lab measurements on the unit. Power specs were confirmed or bettered in the tests; distortion specs were not. In general, the results of the tests add up to a typically good commercial-grade high-powered amplifier. The unit also seemed sufficiently rugged—in terms of its physical construction as well as its electronic reliability and protection circuitry. Each channel was found to have individual thermal cutouts to remove power from the output stages should the temperature of either heatsink exceed 100 degrees C. These cutouts reset automatically when temperature returns to a safe level. Should a DC voltage appear at the amplifier output, a relay disconnects the speaker load from the amplifier within about one second's time. The ventilating fans were judged to be the noisiest we ever have heard from any amplifier, but the amplifier is marketed for commercial rather than home use, so it will be up to the reader to determine if the fans are too noisy for his or her purpose.

General Info: Dimensions are: 19"W x 12"D x 7"H Weight is: 60 pounds. Price: $1590.

Individual Comment by L.F.: I must confess that the Studiomaster 800B/C amplifier, though it lived up to its power specifications, left me less than enthusiastic. To begin with, when I opened the carton I was surprised to find that the manufacturer had failed to include a single printed word concerning this high-powered unit. It was all we could do to hook the thing up correctly, especially in view of the fact that inputs and outputs all use the 3-pin XLR connectors. I have

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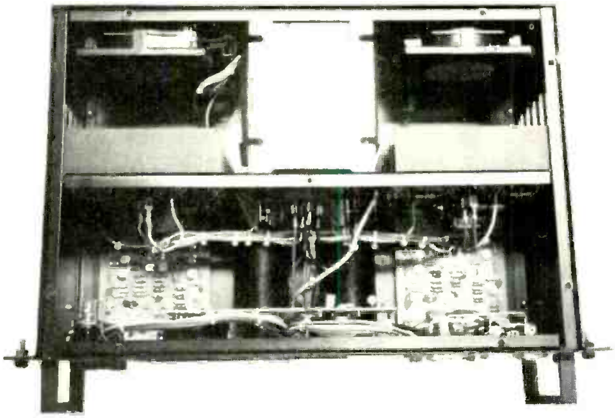

Studiomaster 800B/C: Internal view of the amplifier.

nothing against XLR connectors used for inputs, but I cannot think of one good reason to use XLR connectors for outputs. In fact, I can think of at least one good reason why they should not be used. Most of the commercially pre-wired cables having XLR connectors at each end are generally shielded cables intended for line-level signals, and not for high current signals intended to pump power into 8-, 4-, or 2-ohm loads. The center conductors of these cables are generally of insufficient wire gauge to handle the high currents involved in such a high-powered amplifier without either introducing a substantial IR drop in the cable itself (especially if we are talking about long runs), or at least messing up whatever damping factor the amplifier's own internal impedance might have presented to the speakers being driven by it.

Besides this objection to an XLR output, let me also point out that a shielded wire of the type normally used exhibits a rather high capacitance per running foot and, again, if we are talking about long runs to speakers from the amplifier, we could be talking about goodly fractions of a microfarad of capacitance upsetting the stability of the amplifier and significantly altering the net impedance seen by the output stages.

Finally, consider this: The hot terminal on the input signal XLR connectors turns out to be Pin 3, while the hot terminal for the ouput XLRs is Pin 2. (Pin 1 is "earth"—the U.K. term for our "ground"—for both types of connectors.) As you might well imagine, this took a bit of time to figure out, given the fact that we were not supplied with any instructions whatsoever.

Undaunted, we managed to get signals in and out of the unit. We did find that the amplifier delivered a hefty 253 watts per channel (against the 225 watts per channel claimed) for the first signs of clipping into 8-ohm loads.

On the whole, the measured performance is good for a "pro" amplifier, but it would clearly be inadvisable for any audiophile to try to use this amplifier in a hi-fi stereo system. Of course that use is not implied by the manufacturer. As for its use in professional applications, performance and reliability seem satisfactory.

Individual Comment by N. E.: I think what we have here is a generally good, competently built power amplifier, but one with certain "traits" that set it apart from similar units. The paucity of specifications, the placement of all inputs and outputs on the front panel, the problems implicit in using XLR connectors for output lines to speakers, the noisy fans and so on have all been mentioned. There's something else too that I question. The line voltage adjustment was set for 110 volts, despite the fact that for years the standard line rating in the U.S. has been 120 volts. All this adds up to a certain intransigence and a less-than-complete coming to terms with the U.S. market and its buyers.

STUDIOMASTER 800B POWER AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Continuous power per channel for rated THD, 8 ohms, 1 kHz	225 watts	253 watts
4 ohms, 1 kHz	400 watts	410 watts
THD at rated output, 8 ohms, 1 kHz	0.005%	0.030%
4 ohms, 1 kHz	0.007%	0.045%
8 ohms, 20 Hz	NA	0.100%
8 ohms, 20 kHz	NA	0.350%
IM distortion, rated output, SMPTE	NA	0.020%
Frequency response, 1 watt	5 Hz to 30 kHz ('3 dB)	10 Hz to 70 kHz
S/N ratio re: 1 watt, "A" wtd, IHF	NA	79 dB
S/N ratio re: rated output, "A" wtd	100 dB	101 dB
Dynamic headroom, IHF	NA	1.7 dB
Damping factor, 50 Hz	NA	12
IHF input sensitivity	NA	0.085 V
Input sensitivity re: rated output	1.25 V	1.30 V
Slew rate (volts/microseconds)	15	15
Power consumption, idling/maximum	NA	162/1240 watts

CIRCLE 15 ON READER SERVICE CARD

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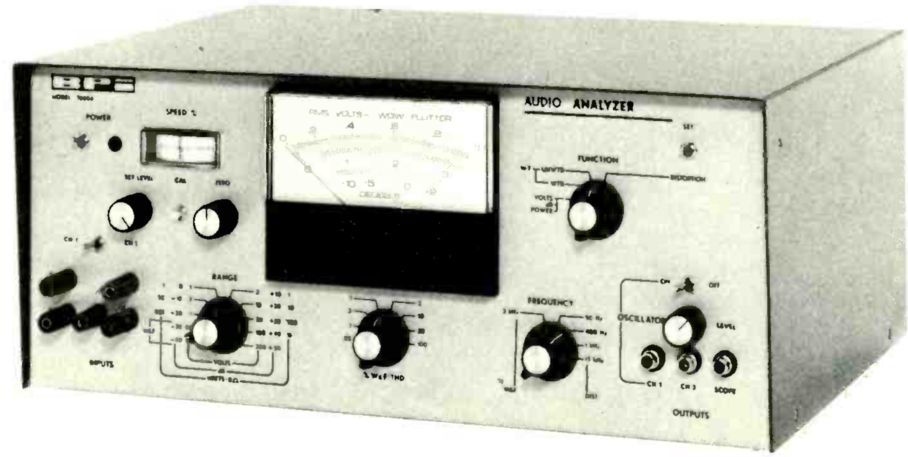
BPI 7000A Audio Analyzer



General Description: The BPI 7000A is a multi-purpose audio test instrument capable of making a variety of measurements, including tape speed and drift, AC voltage, amplifier power output, decibels, wow and flutter (NAB weighted and unweighted), total harmonic distortion, turntable rumble (with the addition of an external weighting network), frequency response (playback only, using a prerecorded standard test tape), signal-to-noise ratio and more.

All operating controls and features, including signal inputs and the device's oscillator outputs, are arranged neatly and logically on the front panel whose black lettering against a light blue background makes for excellent visibility. The two readout devices are a speed-percentage meter and the larger general purpose meter which is amply proportioned and very clearly marked. The various knobs and switches on the panel relate to calibration and test procedures. To the left of the speed meter is the device's power switch and an associated LED indicator. Under the speed meter is a press-to-use calibration button which is used in conjunction with an adjacent "zero" control. Left of the calibration button is a "set-level" control which adjusts the input level to the distortion analyzer.

A small toggle switch selects either of two signal channels which may be connected to the unit and analyzed. The input connectors for the channels are five-way binding posts, and there is a fifth connector for grounding the common leads of the inputs to the analyzer's chassis, if desired.

An input range switch selects the range of the signal being tested and also determines the scale on the large meter. Another switch is used for ranging the meter to the level suited for measurements of distortion and of wow-and-flutter.

To the right of the large meter is the function selector. Near it, another small toggle switch chooses between two functions: one, to permit adjusting the voltmeter for referencing the distortion analyzer input level as 100 percent; the other, for feeding the output of the distortion analyzer into the voltmeter to get a reading as a percentage of that reference level.

Below are the oscillator frequency selector switch, the oscillator level control and the oscillator outputs. There also is an output for feeding an external oscilloscope, and the oscillator off/on toggle switch.

The rear panel of the BPI 7000A contains only the device's AC line cord, fitted with a three-prong (grounding) plug and a fuse-holder for the power line. Two more fuses, protecting the signal input lines, are located on a circuit board within the instrument.

Test Results: The various specifications for the BPI 7000A are listed under specific functional headings, and in our lab tests they were confirmed or exceeded. In use tests, the Analyzer proved to be accurate as well as relatively convenient to set up and put into action. The 'scope output jack on the unit supplies, in effect, whatever the voltmeter is reading at the time. Thus, for instance, if one is reading THD in percentage on the meter, the 'scope output will present the distortion component signal for display on the 'scope face so that the specific components that make up the "total" distortion can be examined at will.

The accuracy of the BPI's meter, by the way, was found to be within 2 percent, as against the 3 percent that was claimed.

General Info: Dimensions are 17 inches wide; 12 inches deep; 7 inches high. Weight is 17 pounds. Price: $1295. Optional 100-watt stereo loads cost $90.

Joint Comment by L.F. and N.E.: This is another "all-in-one" instrument designed to make life easier for the recording engineer-technician, service shop technician or even the very serious audio hobbyist. Since our lab is equipped with a couple of more sophisticated BPI devices, we are perhaps "prejudiced" in favor of this company, but what impresses us most about their designs is their uncanny ability to make measurement work relatively simple in those instances when simplicity is called for. Often, this kind of

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product engineering costs a lot of money since test equipment, unlike mass-produced equipment, cannot always take advantage of volume parts purchasing and volume production. Yet BPI manages to do it within a relatively low-cost (for the buyer) framework.

Like more expensive counterparts, the distortion analyzer section of the 7000A is self-nulling, which makes a big difference when you have to do a lot of repetitive harmonic distortion measurements as you would, for instance, when tweaking up a recorder's bias adjustments.

In all fairness, we would not call this a laboratory-grade instrument since its residual distortion (while low) and its meter accuracy (while good) are not the equals of some costlier lab equipment—made by BPI as well as by others. The 7000A is, simply, a service instrument—ideally suited for use by a service technician. Readers may recall that, in our January 1981 issue, we reviewed an all-in-one test device made in Great Britain which, we felt, was a bit archaic in its design and therefore a bit inconvenient to use (though we could not dispute the accuracy of measurements made with it). The present BPI instrument is far more convenient to use, and it also does more than the previously tested device. And it also costs much less than its nearest competitor.

That's not to say that we couldn't find fault with it—even at its relatively low cost. These days, we question whether the few available test frequencies (50 Hz, 400 Hz, 1 kHz and 15 kHz) are really enough, even for a cursory check of modern audio equipment. Perhaps the addition of 20 Hz and 20 kHz would make all the difference and would elevate the piece of equipment to the status of its higher priced competition. In any event, as a minimum, we would think that anyone wanting to set up audio measurement facilities with this instrument might want to add a couple of high quality load resistors (these are optionally available from BPI), as well as perhaps a continuously variable audio oscillator for those instances when it is necessary to check out response in greater detail than is possible with the four test frequencies supplied on the 7000A.

On the positive side, once more, we appreciated the differential inputs which will permit the user to check out car stereo equipment that doesn't want to be connected to a "common" chassis ground. In addition, we have always found that the nomenclature on the front panels of BPI equipment is so clear and easy to understand that we seldom, if ever, had to refer to the preliminary owner's manual supplied with the equipment in putting the tester through its paces. Since it is a lightweight unit, it could easily be used in the field, providing, of course, that one has access to a source of 110-130 volt 60 Hz power.

BPI 7000A AUDIO ANALYZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Oscillator		
Frequencies	50 Hz, 400 Hz, 1 kHz, 3 kHz, 15 kHz	Confirmed
Frequency accuracy	3 kHz, ±0.25%; others, ±2%	3 kHz, +0.3%; worst case, 2.7%
THD	<0.015%	worst cast, 0.012%
Output level (max)	3.0 V	3.5 V
Distortion Analyzer		
Frequencies	50 Hz, 400 Hz, 1 kHz, 15 kHz	Confirmed
Residual distortion	<0.015%	0.010%
Input level range	1 V to 300 V	Confirmed
Distortion % ranges	0.03, 0.1, 0.3, 1, 3, 10, 30, 100	Confirmed
Auto nulling freq. range	±5%	±6%
AC Voltmeter		
Meter ranges	10 mV to 300 V in 10-dB steps	Confirmed
Input impedance	100 K ohms	Confirmed
Accuracy	±3%	±2%
Frequency response	20 Hz to 50 kHz	20 Hz to 100 kHz
Wow and Flutter		
Input frequency	2.7 kHz to 3.5 kHz	Confirmed
Input level range	50 mV to 10 V rms	40 mV to 10 V
Input impedance	100 K ohms	Confirmed
Ranges % wow-and-flutter	0.03, 0.1, 0.3, 1.0, 3, 10, 100	Confirmed
Residual noise	0.01% unwtd; 0.005% wtd	0.008%; 0.003%
Speed and Drift		
Accuracy	10%	7%
Range	0 to +5% or to −5%	Confirmed
Power requirements	100-130 VAC, 60 Hz	25 W

CIRCLE 16 ON READER SERVICE CARD

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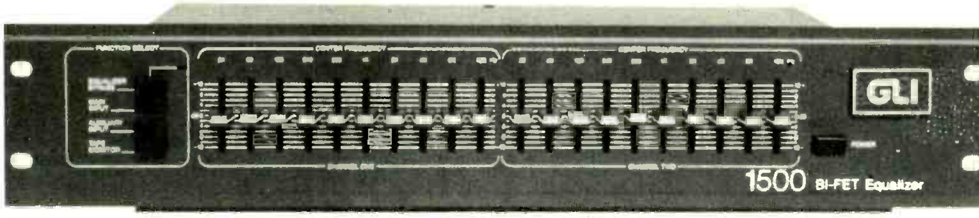
GLI 1500 Graphic Equalizer



General Description: The GLI 1500 from Integrated Sound Systems, Inc. is a stereo graphic equalizer offering ten octave-bands on each of its two channels. Nominal center frequencies are 30, 60, 120, 240, 500, 1 K, 2 K, 4 K, 8 K and 16 K Hz. Each slider control covers an adjustment range from +12 to −12 dB; the ranges are marked off by lines and numbers in steps of 2 dB. Each slider has a detent position at the 0 dB mark.

To the left of the EQ sliders are four pushbuttons for EQ in or bypass; main input; auxiliary input; tape monitor. A similarly styled button at the far right turns power off and on. All control markings are bright white lettering against a black matte background, and are readily visible. The EQ bypass button has a status LED that shows whether or not the device is in the circuit. The front panel overlaps the actual chassis, and, with the predrilled end holes, is suitable for standard rack-mounting. Alternately, the unit may be placed "as is" on any surface since there are four small feet under it.

Signal connectors, at the rear, include—for each channel—the main output, tape output and input and auxiliary and main inputs. Options for interfacing with other audio equipment thus include patching the GLI 1500 into the tape-monitor loop of a preamplifier, using it with a tape deck and running it in conjunction with a disco mixer or a P.A. mixer. These hookups are explained and illustrated in the owner's instruction manual.

Suggested applications for the GLI 1500 are compensating for room acoustics; improving the response of phono cartridges, speakers and headphones; and to remove noise from program sources. Some midrange and high-end peaks also may be smoothed out with the device. General tonal compensation and highlighting of instruments or groups also are suggested.

Test Results: In bench tests, the GLI 1500 met or exceeded its published specifications. Center frequencies were confirmed, as were the boost and cut ranges of the sliders. Distortion was notably lower, and signal-to-noise was better, than claimed.

Fig. 1 shows a composite response plot of each of the octave-by-octave band controls of the unit. Frequencies in this plot extend from 20 Hz to 20 kHz and are plotted logarithmically, so that octave spacings should be, and are, approximately equal.

In *Fig. 2,* we show composite response plots of the GLI 1500 with progressive calibrated settings of a single band control (the 1 kHz slider). Note that the bandwidth of this filter appears to change somewhat (Q varies) with different amounts of boost or cut. On the whole, we think this is a desireable characteristic since, in most instances, where a moderate amount of boost or cut is required, it is also desireable to have a very gradual rising and falling of response. On the other hand, extremes of boost or cut are usually called for along with narrow bandwidth (high Q) such as when a specific narrow band of feedback frequencies—as in a P.A. system—are to be attenuated.

Fig. 3 shows the amount of boost and cut provided by a single band control. To achieve the double response plot on our test instrument we had to arbitrarily call one curve "L" (channel) and the next one "R" (channel), although, in fact, both traces were taken successively on the same channel of the GLI equalizer.

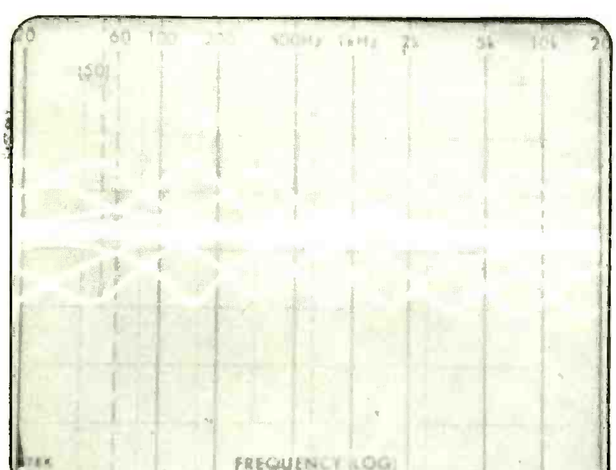


Fig. 1: GLI 1500: Composite of boost and cut range of each band. Vertical sensitivity is 10 dB per division.

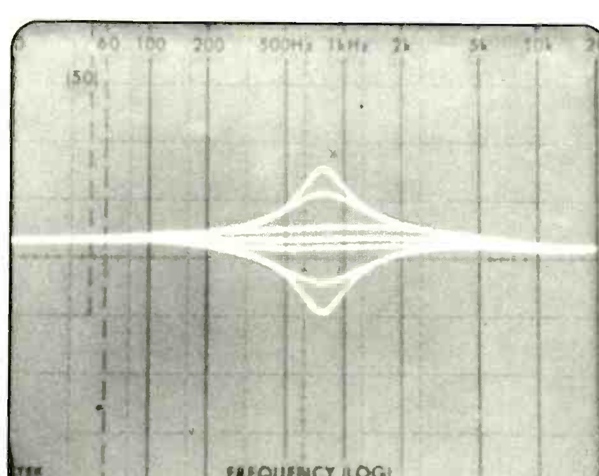


Fig. 2: GLI 1500: Incremental calibrated settings of a single control on the equalizer yield boost and cut characteristics shown in this composite plot.

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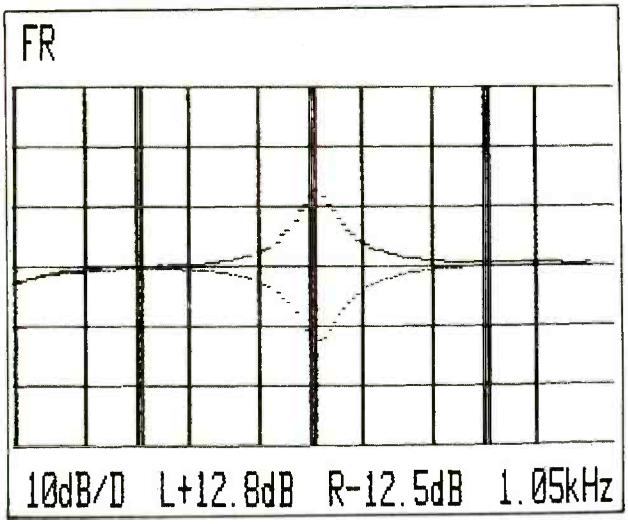

Fig. 3: GLI 1500: Maximum boost and cut for nominal 1.0 kHz control occurs at 1.05 kHz and exceeds claimed ±12 dB by a slight amount.

Note that the maximum boost and cut frequencies for the nominal 1-kHz control worked out to be 1.05 kHz (certainly, accurate enough!), and the amount of boost and cut available did exceed somewhat the amounts claimed by the manufacturer.

General Info: Dimensions are 19 inches wide; 3½ inches high; 7¾ inches deep. Weight is 7 pounds. Price is $250.

Individual Comment by N.E.: Although there are graphic equalizers with longer-throw sliders (making them more "graphic"), the topology on the GLI 1500 is adequate enough to permit accurate and visible filter settings. The manufacturer has managed to offer a compact neatness in style without sacrificing actual readability of the panel. It is easy to get to specific scale settings here, and the center detent certainly helps in orienting your fingers to the scales. Also to the good are the selectable inputs and the status LED for EQ in or out. I would have also liked similar LEDs for the other three switches in this group, not to mention an overall gain control, but then we'd probably be talking about a unit costing more than the GLI 1500 which, at $250, seems like a good buy.

Individual Comment by L.F.: On the plus side for this unit are its separate band controls for each channel, the detented zero-dB settings, the gyrator circuitry, the selectable inputs, the status LED for EQ in or out and the unit's complete tape-monitor loop that can replace the similar loop that you may have preempted in your stereo system by patching in the GLI 1500.

On the negative side, I note the omission of an overall gain control. Boosting or cutting a couple of mid-frequency octaves by more than a few dB each results in a significant overall apparent level shift of program material. Of course, this can be compensated for by other master volume controls in an audio system. However, for one who wants to do an A-B comparison using the bypass switch (EQ versus no EQ in the sound), it is important to be able to make the comparison with approximately equal levels under both listening conditions.

Then too, if input levels are on the high side, and extensive boost or cut is used for certain octave bands, it is not inconceivable that overload might result at certain program frequencies. The maximum undistorted output capability of the GLI 1500 is a comfortable 10 volts RMS, but if a program peak signal of 3 volts were to be applied to the unit at a frequency where you had boosted an octave-band control to its full or nearly full 12 dB, that signal would surely overload the equalizer. With an input level control it would be possible to bring overall levels down so that overload could not occur. Without such a control, the ultimate dynamic range of the unit becomes somewhat limited.

I suppose, though, when all is said and done, its price is what makes this equalizer worthwhile. It is low in cost and it does provide all the flexibility intended in most applications. I, for one, would have been willing to spend a few more dollars for it if the input level control were included.

GLI 1500 GRAPHIC EQUALIZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Band frequency centers	30, 60, 120, 240, 500 Hz, 1, 2, 4, 8, 16 kHz	Confirmed
Boost and cut range	± 12 dB	+ 12.8, − 12.5 dB
Frequency response	± 0.5 dB, 20 Hz to 20 kHz	− 2.0 dB at 20 Hz; − 0.5 dB at 20 kHz
THD	<0.05% at 1 V out; 20 Hz to 20 kHz, any EQ setting	At flat settings: 0.004% at 1 kHz, 0.005% at 20 Hz, 0.0045% at 20 kHz Worst case: 0.015% at 1 kHz, 0.04% at 20 Hz, 0.017% at 20 Hz
Slew rate (volts/microseconds)	14	Confirmed
S/N	90 dB below 2.0 V	95 dB unweighted; 102 dB "A" weighted
Max output before clipping	10 V	10 V

CIRCLE 17 ON READER SERVICE CARD

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Carvin MX1202 Mixing Console

By John Murphy and Jim Ford

This month we're reviewing the Carvin MX1202 mixing console. This is basically a twelve input, stereo output board with a convenient built-in spring reverb among its many features. There are three side mixes in addition to the main stereo output: Monitor 1, Monitor 2 and Effects/Reverb. Each input channel has 3-band EQ (Low, Mid, High) with fixed frequencies of 100, 2.5 K and 10 kHz. The microphone preamplifier is a transformerless type and the unit has low-noise/high-slew-rate gain stages (op-amps) throughout. The board is also available in a 16-input version (MX1602) and both versions are available with optional graphic EQ (9 band) on each of the main and monitor outputs. Other features include 48-volt phantom power for microphones, soloing and talkback facilities, control room/headset monitoring system and attractive solid walnut sides. The price of the basic MX1202 is $1,195; the optional graphic EQ is an additional $250.

General Description: The MX1202 will accept twelve input signals (either balanced or unbalanced) ranging from microphone level through line level up to about +30 dBV. That's really quite a range of acceptable input levels! This is possible because both balanced and unbalanced input signals are routed through the microphone preamplifier which has variable gain (36 dB range) and is preceded by a switchable 20 dB pad. Line-level inputs would normally require the use of the 20 dB pad whereas microphone inputs would not.

After a signal is routed through an input channel where it can be EQ'd and sent to any combination of the effects and monitor mixes, it is assigned to the main stereo output through the channel pan pot at a level determined by the channel fader. Reverb and effects are added to the main stereo output through the use of the reverb and effects receive and pan controls. The main faders are then used to set the final output level of the main mix. The master monitor faders likewise set the output levels of the monitor 1 and 2 mixes. Four large VU meters indicate output levels for the Main A, Main B, Monitor 1 and Monitor 2 output signals. The nominal output level for "OVU" is +4 dBV but the meters can be calibrated to any desired level through the use of the VU calibration controls located near the output faders.

The console operator can monitor any of the main, monitor or effects signals through headphones or, if the board is used for recording, through control room monitors. Now let's look at one of the input channels in detail.

Channel input connections are made at the rear of the unit by way of either a 3-pin, XLR-type connector for balanced signals (usually a low-impedance microphone), or by way of a ¼-inch phone jack for unbalanced signals. In addition to these connections, there are ¼-inch phone jacks at the rear of each input channel labeled "send" and "return" which constitute the channel patch point. The patch point is immediately after the microphone preamplifier and is therefore pre EQ and pre fader.

The controls for each input channel are arranged logically in a vertical column above the input fader. At the top of each input strip, there is a push-button switch which inserts the 20 dB input pad when depressed. Immediately below the pad switch is a rotary control which varies the gain of the microphone pre-amplifier over a 36 dB range. Clockwise rotation of this control results in reduced signal level. To the right of the gain control is a red LED peak overload indicator. This indicator is helpful in setting up the channel's input gain.

Moving down the input strip we next see the channel EQ controls. This is a set of three rotary controls for "high," "mid" and "low" frequency EQ. The channel EQ affects the main and effects outputs but not the two monitor sends. Frequency response characteristics of the channel EQ can be seen in *Figures one* and *two.*

The Monitor 1 and Monitor 2 level controls are located just below the EQ section of the input channel. These controls send pre-fader signals to the monitor 1 & 2 main faders. In a typical sound reinforcement system one (or both) monitor output signals would be returned from the mixer to the stage (through a power amplifier and monitor loudspeaker) to serve as a "cue" for the musicians. It's important to remember that the quality of the monitor mix and the stage monitoring system can have a powerful influence on the performance of the musicians.

Continuing down the input channel, the next control is

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the Effects/Reverb level control. This control has two functions. First, it sends a signal to the master "Effects Send" control where all of the individual channel effects signals are summed. The master effects send signal is then made available at a nominal-10 dBV level at the rear of the console. This signal would normally be routed through an external effects unit (such as a reverb, echo or time delay effects unit) and returned to the console through the "effects return" system. The second use of the channel's effects/reverb signal is to provide an input signal to the mixer's internal spring reverberation system. As with the effects system, there are master controls for "reverb send" and "reverb receive."

The channel pan pot is located below the effects/reverb send and is used to assign the signal in the input channel to either the A or B main outputs or to pan anywhere between them in a stereo perspective. Immediately below the pan pot is a small white writing area for identifying channel signals.

Immediately above the fader near the bottom of the input strip is the solo switch for each channel. This is a locking push button which when depressed interrupts the control room (and headphone) audio and substitutes the solo signal. There is also a master solo level control which is used to adjust the level of soloed audio signals.

Now that we've covered the layout of the twelve input channels, let's go over the master controls for the console. Note that the input channels occupy the left-most two thirds of the console face while the right-most two thirds is occupied by the master controls.

At the far right of the mixer's face and furthest from the operator are the power on/off switch and an LED power indicator. Just below the main power switch is another push button with LED indicator which is used to apply 48-volt phantom power at the balanced input connectors for each channel.

If the MX1202 is purchased with the graphic EQ option, then the four channels of graphic EQ (nine octave-wide bands each) are located in the area below the main and phantom power switches. The four graphic EQs are normally placed in the Main A and B and Monitor 1 and 2 signal paths. Associated with the graphic EQ are four push button switches located above the EQ controls. These are labeled "EQ defeat" and when depressed they switch the individual graphic EQs out of the normal signal paths and make the EQ available for patching into other signal paths by way of rear panel input/output connectors (¼-inch phone jacks). This means that if a graphic EQ is not needed in either the Main or Monitor outputs, it can easily be patched into the signal path of any of the input channels via their patch points. This provides the user with the flexibility to put the EQ where he needs it most. Excellent!

Below the graphic EQ and above the master faders are the master send, receive and pan controls for the effects system. Similarly, below these controls are send, receive and pan controls for the reverb system. The pan controls allow the effects and reverb return signals to be panned between the A and B main outputs.

There is another pair of level controls located to the right of the sends and returns. These are labeled "A" and "B" playback and allow the playback of a tape deck or other line-level program source through the board's main outputs.

The controls for the talkback system are to the right of the playback level controls. There are a pair of locking push buttons which allow talkback through either the Main or the Monitor system. The talkback level control is located above the push buttons and there is a 3-pin, XLR-type connector located to the right of the group for connecting the talkback microphone.

Below the talkback control group and to the right of the four master faders are the Control room selector switches and level control. A set of five interlocking push button switches allows selection of either the Main (stereo), Main (mono), Monitor 1, Monitor 2 or Effects signals. The selected signal is provided at a stereo phone jack (lower right corner of the mixer) for headphone monitoring and also at a pair of ¼-inch phone jacks on the rear panel for monitoring over control room loudspeakers.

In addition to the "A" and "B" main stereo outputs, the mixer also has a mono output which is the sum of the "A" and "B" signals. There is also a Sum (mono) level control located below the control room selector which

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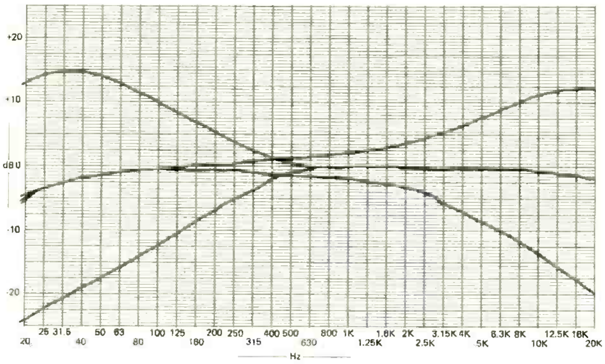

Fig. 1: Carvin MX1202: High and low frequency EQ response curves for flat and full boost/cut.

allows independent control of this signal. Located at the right side of the Sum level control is the solo level control.

On the slanted rear-most portion of the mixer are four large VU meters which indicate the levels of the Main and Monitor output signals.

Let's now turn our attention to the back of the Carvin mixer. At the rear of each channel are female 3-pin, XLR-type connectors for balanced input signals such as low impedance balanced microphones. Above this balanced input is a ¼-inch phone jack for unbalanced input signals. And above that jack are a pair of ¼-inch phone jacks ("send" and "return") for channel patching.

The main mixer outputs are located at the rear of the console's master control section and are provided on both 3-pin, XLR-type connectors for balanced output signals, and on the ¼-inch phone jacks for unbalanced connections.

If the EQ option has been selected then there will be eight more ¼-inch phone jacks toward the top of the rear panel. These serve as the input and output connections for the four graphic EQs when they have been switched out of their pre-assigned signal paths.

The rear panel also contains a connector for the detachable line cord as well as the primary fuse for the unit. Just below the AC line connector is an access through the panel to a slide switch which selects between 110 and 220 volt AC operation. The switch is preset for 110 volt operation at the factory.

Field Test: We used the MX1202 to do a mono sound reinforcement mix with excellent results. Inputs to the board included four vocal mics, bass, electric piano, acoustic guitar and electric guitar. The drum set was not miked.

The vocals and acoustic guitar were returned to the stage via the monitor 1 mix using the Monitor 1 graphic EQ. The mono ("sum") output was used for the main P.A. feed after being patched through the defeated "Main A" graphic EQ. The "Main B" and "Monitor 2" graphic EQs were defeated and patched into the bass and electric piano channels, respectively. The board's output and solos were conveniently monitored by the headphone monitoring system.

Since we were providing a mono house mix we were free to use the "Main A" and "Main B" faders as two separate sub groups. We simply split the mix into two components by panning the instruments left to "Main A" and panning the vocals right to "Main B." Thus the balance between vocals and instruments could be adjusted by the "Main A" and "Main B" faders.

On initially setting up input levels we found that we had to keep reminding ourselves that the input trim controls work the opposite of "normal." That is, clockwise rotation of the trim control *decreases* the signal level rather than increasing it. Also, we occasionally had trouble seeing the peak overload indicators. These LEDs are readily visible from directly above, but at the operator's normal position in front of the mixer the brightness of the LEDs is greatly reduced. These minor points aside, the mixer performed fine and allowed us to do our job without any trouble. The built-in reverberation system was found to be quite convenient although the reverberation quality was only as good as typical inexpensive spring units.

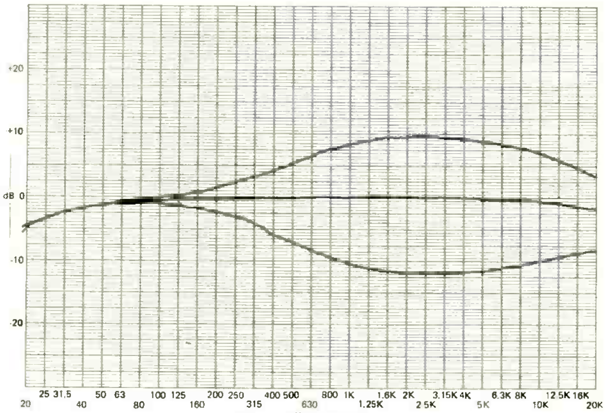

Fig. 2: Carvin MX1202: Mid frequency EQ response curves for flat and full boost/cut.

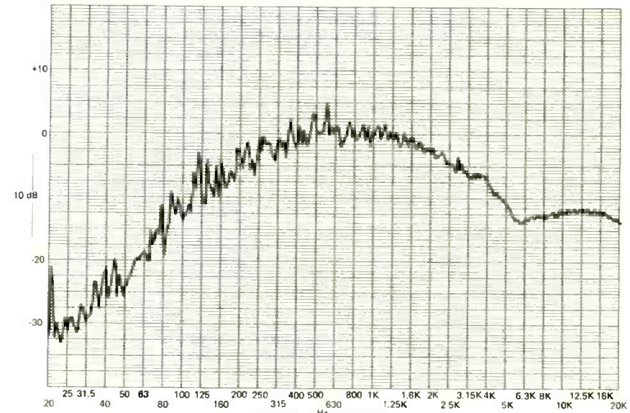

Fig. 3: Carvin MX1202: Response of the reverberation system to pink noise.

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As we usually do in our testing routine, we interfaced the mixer with our reference listening system in such a way that we could listen through the unit and alternately switch it out of the listening chain. We put on a good disc and listened as we alternately switched the mixer in and out of the listening chain. We are pleased to say that we noticed no sonic problems with the Carvin mixer; it was relatively transparent to the audio signal.

Lab Test: The MX1202 was put through our usual barrage of tests and we found nothing to detract from its excellent performance in the field. For the detailed results of our tests, see the "Lab Test Summary" below.

The console's preamplifier was capable of accepting a very wide range of input signal levels (from about −55 dBV to +34 dBV). Likewise, the unbalanced input accepts the same range of signals simply because the unbalanced input is just one half of the balanced input. The only drawback of this arrangement is the fact that even line-level signals must go through the microphone preamplifier and the residual distortion of the preamplifier is greater than any other amplifier stage in the mixer (about .04%). Therefore, the distortion performance of the mixer is limited by the performance of the microphone preamplifier, even for line-level input signals.

For a signal level indication of "OVU" we observed an actual output signal level of −1.3 dBV. This is a little shy of the nominal +4 dBV operating level that Carvin indicates but that's of little consequence since it is a simple matter to recalibrate the VU meters by way of the top panel calibration controls. We noted that the board's output could be driven to + 22 dBV into a high impedance load before clipping and to + 19 dBV into a 600-ohm load.

The noise performance of the mixer was good. Even with all twelve channels set for 40 dB of preamp gain, and all twelve assigned, the noise level at the output was −60 dBV.

With the channel EQ set at zero, the frequency response through the mixer was quite flat. However, we were a little surprised to see the high-frequency bandwidth limited to about 26 kHz, especially since the power bandwidth of the unit was determined to be greater than 100 kHz based on observed signal velocities (slewing rates).

The normalized slew rate limit was determined to be 1.07 volts per microsecond per volt. This is one of the highest slew rate ratios we've observed and indicates a very generous degree of slewing headroom and excellent high-frequency design.

The response of the mixer's reverberation system to pink noise is shown in *Figure 3*. The curve indicates that the reverb return signal is weighted toward midrange response.

The operator's manual supplied with the console, while a little skimpy, was considered to be adequate for most users.

Conclusion: After a thorough evaluation both in the field and in the lab, we found Carvin's MX1202 mixing console to be an excellent unit. Considering the cost of the mixer, it would appear to be a good buy as well as a good performer!

LAB TEST SUMMARY

(Note: 0 dBV = .775 Vrms)

Input Levels

Mic Input:

Minimum input level for "0 VU" level indication with input channel trim and fader at maximum (master faders at nominal (− 10 dB) settings: − 54.5 dBV

Maximum mic input level before clipping the input stage: + 33.9 dBV

(Line input levels are the same as the mic input levels.)

Output Levels

(at main stereo output)

For "0 VU" level indication: − 1.3 dBV

Maximum output level before clipping:

+ 22.5 dBV (into a high impedance load)

+ 19.3 dBV (into a 600 ohm load)

Noise Performance:

(Note: 20 kHz bandwidth, 150-ohm source, unweighted)

Equivalent Input Noise: − 118.8 dBV

With all channel faders at minimum and the master fader at nominal (− 10 dB) noise at the output is: − 68.0 dBV

With one input channel used and the trim set for 40 dB gain, noise at the output is: − 67.2 dBV

With eight channels set as above, noise at the ouput is: − 61.6 dBV

With twelve channels set as above, noise at the output is: − 60.3 dBV

Distortion Performance:

(THD plus noise at + 10 dBV output level, from Mic input to main output)

Frequency	THD & Noise
100 Hz	.046%
500 Hz	.043%
2 kHZ	.042%
10 kHz	.041%
20 kHz	.047%

Frequency Response: ± 1 dB 35 Hz to 20 kHz

Bandwidth (− 3 dB points): 27 Hz to 25.7 kH

Power Bandwidth: greater than 100 Hz

Slew Rate Limit: 16 volts per microsecond

Normalized Slew Rate Limit: 1.07 volts per microsecond per volt

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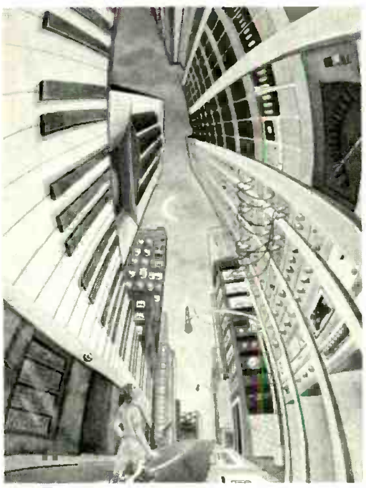
unbalanced? balanced? fader type? meters? sensitivity? input impedance?

CONFUSED?

outputs? metal tape?

parametric? bandwidth? pad? frequency response? bypass? load protection? graphic? balance control?



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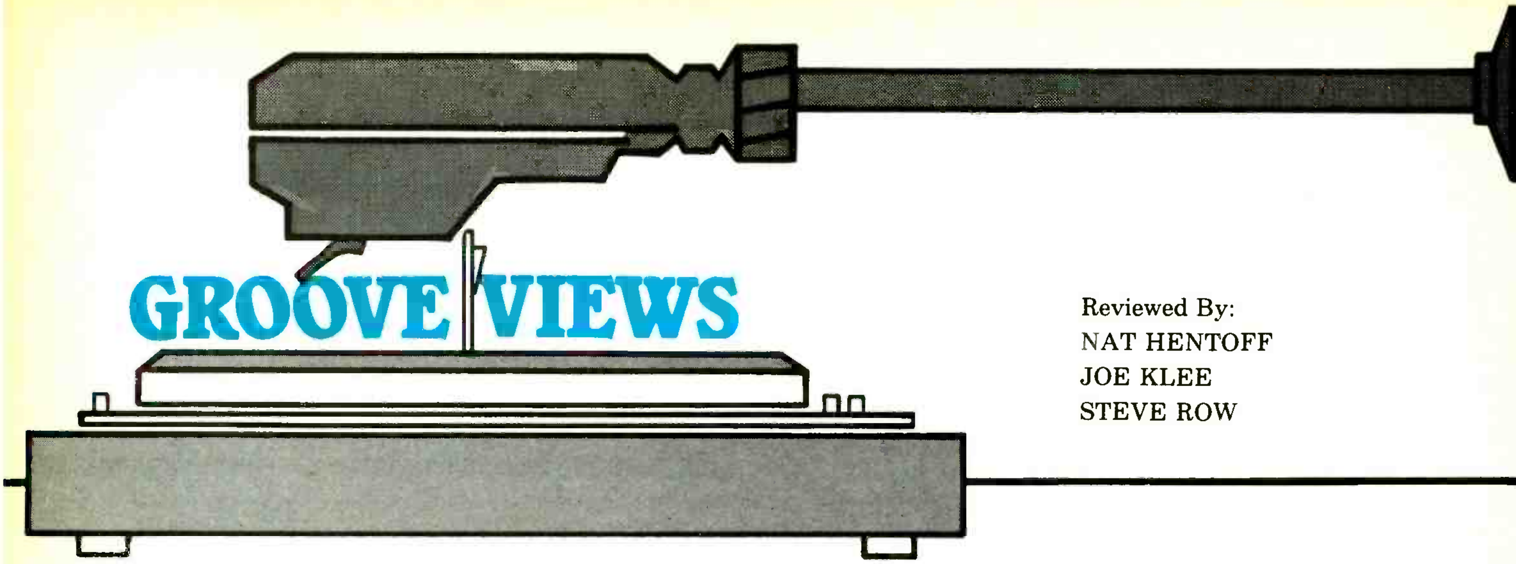

Reviewed By:
NAT HENTOFF
JOE KLEE
STEVE ROW

POPULAR

THE SEARCHERS: ***The Searchers.*** [Produced and engineered by Pat Moran; recorded and mixed at Rockfield Studios, Wales, April-July 1979.] Sire SRK 6082.

Performance: **It's like they never left**
Recording: **Very good**

THE TREMBLERS: ***Twice Nightly.*** [Peter Noone, producer; Bruce Johnston, executive producer; Roger Harris, associate producer and chief engineer; recorded at Conway Recording Studio; Phil Moores, Steve Zaretsky, Chip Orlando, assistant engineers; Rumbo Recorders; Les Cooper, Tchad Thompson, assistant engineers; Black Orpheus Recording; Bob Macias, assistant engineer, Cherokee Recording Studios; Rodney Lovett, assistant engineer.] Johnston/CBS NJZ 36532.

Performance: **Power pop careening into bubblegum**
Recording: **Bright, often crisp, but not always well separated**

These two recordings are not just for rock nostalgia fans, although the recording by the Tremblers, headed by former Herman's Hermit Peter Noone, is much more of a throwback to the bubblegum/bubbleheaded sound of some of the worst of the British Invasion groups of the mid-60's than the new recording by the Searchers. In fact, these albums point up some of the best and the worst of what was popular 15 years ago.

First, the best. The Searchers album is a tightly done, well crafted effort by a group that has been playing together since 1970. Mike Pender and John McNally are founders of the group, which was formed in 1961, and bassist Frank Allan joined in 1964. So the group has been around, together, to grow with its own music and develop its own sound.

The listener will notice instrumentation and arrangements that sound much like the Byrds here, but the Byrds themselves may have based some of their sounds on original songs by the Searchers. Things often come full circle.

The drumming and percussion work by Billy Adamson, who has been with the group "only" since 1970, sparkles throughout, and Pender's playing on the 12-string electric guitar and six-string electric guitar is quite good.

Most of the material is up-tempo and is drawn from both within the group and outside. Included is a song by Will Birch and John Wicks of the Records, "Hearts in Her Eyes," with drums and guitars meshed very well; Tom Petty's "Lost in Your Eyes," a ballad with piano and drums prominent, but with a jarringly small, metallic-sounding vocal; and Bob Dylan's rarely-heard "Coming From the Heart," another ballad that scores well in its musical

THE TREMBLERS: Breathless, gee-whiz sound fills this frantic release.

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"The closest damn thing."

We were recently lucky enough to catch Doug Kershaw on tour. After his show at Harrah's at Lake Tahoe he talked to us about his new Bose® sound system, which consists of four Bose 802 speakers, a Bose PM-2 Powermixer, and a Bose 1800 power amplifier. He's been using the system to amplify his electric fiddle, squeeze box, electric guitar, and vocals.

Q: Doug, you've been playing for a long time. I'll bet you've tried a lot of different kinds of sound gear, haven't you?

Kershaw: Yes, I've used lots of different things and I've spent a lot of time developing my sound. Even then, I could never quite get what I was looking for. But my new Bose system is the closest thing to what I want. The closest damn thing.

Q: What differences have you noticed since you started using the Bose system?

Kershaw: For one thing, it doesn't hurt my ears. You know, I've used some big speakers that have almost busted my ears. I've even put my foot through a few of them. But this is a true sound. It sounds just like my fiddle, no matter how loud I turn it up.

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Kershaw: The attack is easier. It's just easier.

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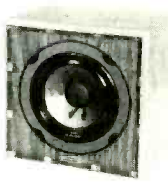

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sound but poorly in its lyrical content.

There also is a nice rockabilly "Switchboard Susan," with a spare accompaniment that creates a nice pop sound, and a very 60's/British sounding "Feeling Fine," with an interesting key change at the end of the stanza before the chorus. This track is dominated by acoustic guitar work except for some fuller ensemble work on the instrumental breaks.

All in all, this is familiar-sounding music, even if the listener is not familiar with the specific tracks. The singing is strong and rarely is crowded out or overshadowed by the instrumental work. The instrumental work is crisply recorded and well balanced, with very little audio clutter to divert attention from the material.

The same praise cannot be given to Peter Noone's effort, however. This recording is a distinct disappointment, either taken by itself, or compared with the Searchers' album. Noone and his four compatriots (Gregg Inhofer on keyboards and guitars, Robert Williams on drums, George Conner on guitar and Mark Browne on bass) have produced a slightly frantic release here, one that boggs down in breathless, gee-whiz sounding material.

One problem is the lack of variety in the 11 tracks. Nearly all of them have the same general tempo (mostly fast), and nearly all of them have the same general theme (the adolescent dreamer on the make, the post-adolescent bloke on the make, the kid who's trying to affect the pose of a punk, on the verge of a big score).

Sadly, the recording doesn't do much to redeem the material, either. The vocals by Noone, who occasionally sounds like a female punk rocker, frequently are distant, and most of the ensemble sound is centered, with little attempt made to open up into the channels, to give the songs breathing room. I switched from "stereo" to "monaural" on occasion and was struck by the lack of difference in where the sound was coming from, even when I listened through earphones.

Some of the material is just terrible, with a shrill sound and trite lyric, while other songs are less objectionable. "I Screamed Anne" is an example of the former, while "Don't Say It" is an example of the latter, a tune with a little more substance, better engineering and less gee-whiz.

I'll put my money on the Searchers.

S.R.

STEVE HACKETT: ***Defector.*** [Steve Hackett and John Acock, producers; no engineer listed; recorded at Wessex Studios.] Charisma/Mercury CL 1-3103.

Performance: **Splendid**
Recording: **Likewise**

Anyone familiar with Steve Hackett's previous solo albums since his departure from Genesis shouldn't be too surprised at the incredibly dense musical and sonic textures that are found on *Defector.* Unlike the jarring, almost cacophonous, sounds that can be heard on former colleague Peter Gabriel's current release, Hackett's *Defector* is a sensual, melodic musical experience that skillfully mixes vocals with instrumentals.

The record also marks a continuation of the effort by Hackett's band to become a strong musical unit, and one can hardly imagine that the group could get any better. With John Hackett on flute, Nick Magnus on keyboards, Pete Hicks joining Hackett on vocals, John Shearer on drums and percussion and Dik Cadbury on bass, the ensemble playing on the 10 tracks here is tight, disciplined and extraordinarily well done.

If alienation is the general theme of the material, then familiarity is the method of musical presentation. Familiarity in the sense that genuinely melodic composition, logical chord progressions, relatively restrained instrumental scoring all are hallmarks of the tracks. Not standard or dull, by any stretch of the imagination, however, because some of the compositions are uncannily beautiful, while others in a more up-tempo rhythm sound very much contemporary.

Some of the music evokes other performers—the rich, dense melodies might as easily be found on one of Renaissance's better albums, for example. Some of the scoring resembles Yes, except that John Hackett's flute lightens the effect. And one song, "The Toast," immediately reminded me of some of the material by Stephen Whynott, a very good singer-songwriter from the Pacific Northwest who has recorded for First American. And the closing song on the album, a whimsical piece called "Sentimental Institution," is done up in "Rocky Raccoon" fashion, complete with megaphoned lead vocal and hotel ballroom orchestra backing (or is it an extremely well-pro-

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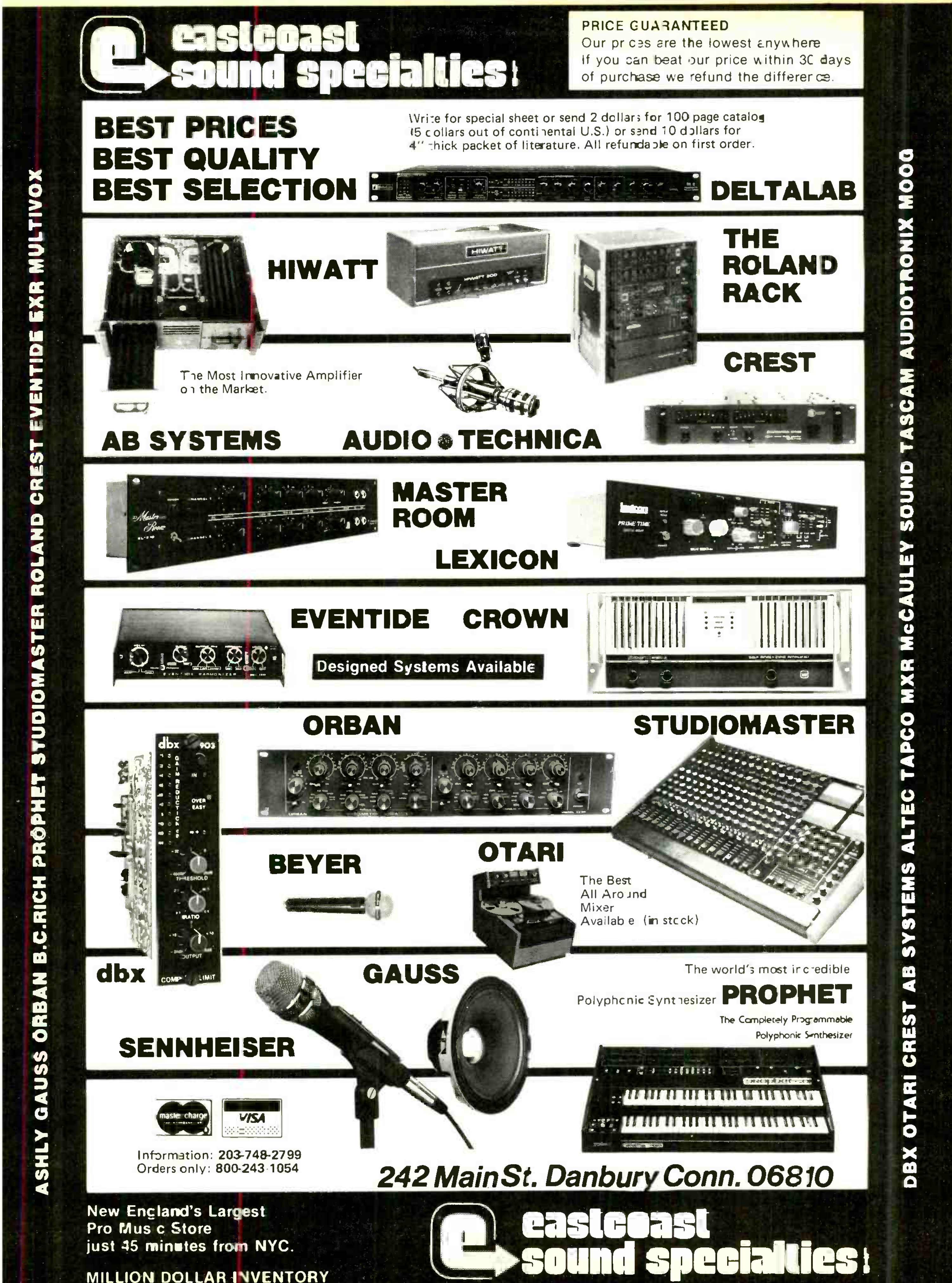

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grammed synthesizer? The accompaniment is so subtle and subdued that it is difficult to tell).

As a recording, *Defector* is very well done, with the vocals rarely overshadowed by the accompaniment, but rarely blaring out either. A nice balance has been struck here, putting the vocal line and instrumental line on about equal footing. The vocal tracks generally feature tight multi-voiced harmonies (Hackett, Hicks and Cadbury are listed as the singers on the album) that occasionally sound muffled and one-dimensional (on "Time to Get Out," in particular).

The sound generally fills the speakers (or ears), but not before allowing the listener to pick out some nice percussion work here, a steady bass line there, and nice keyboard work, too.

Musically, there are some really nice touches evident throughout the album. The flute solo that opens the record, on the instrumental "The Steppes," is quite good, and the contrasts in "Slogans"—moving from a hard, thick, muscular sound of synthesizer, bass and drums in the minor key, to a synthesizer tone near the end of the song of uncharacteristic softness—is very interesting. The fastest, most uptempo compositions on the record never seem to lose their grasp of melody, which is a welcome relief from some of the material that is recorded these days.

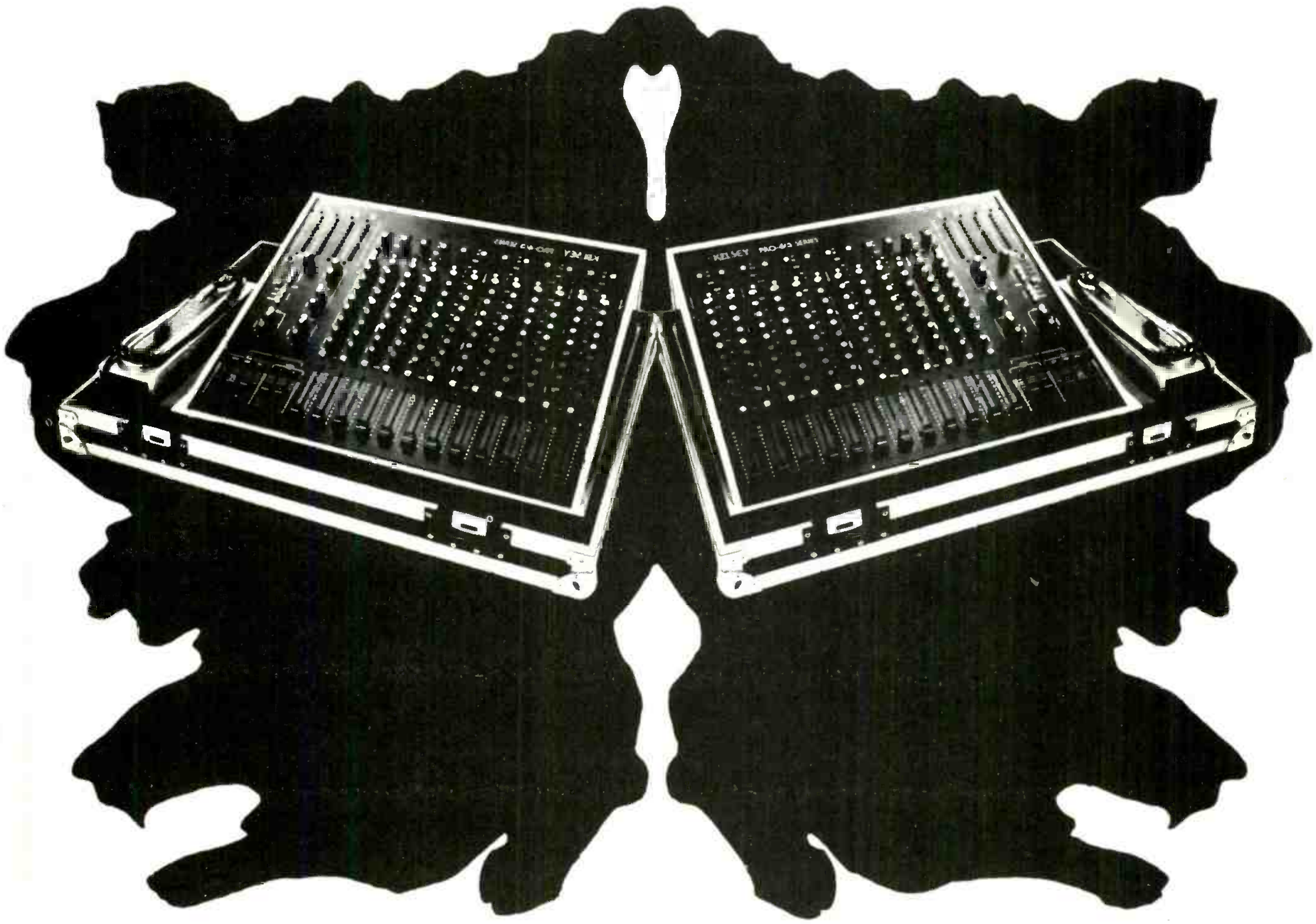
I played Hackett's next most recent album, *Spectral Mornings,* just to see if he was making any great departure musically. He's not, but rather than this serving as a criticism for not growing, this should be interpreted as praise indeed. Hackett once again demonstrates that melodic, well-crafted music does exist in the 80's. S.R.

LIVINGSTON TAYLOR: *Man's Best Friend.* [John Boylan and Jeff Baxter, producers; Bruce Robb and Larry Rebhun, engineers; Phil Jamtaas and Deni King, assistant engineers; recorded at Westlake Sound, Los Angeles, June 1979 through March 1980.] Epic JE 36153.

Performance: **Spirited, relaxed, fun**
Recording: **Very bright, spacious, crisp**

Livingston Taylor always has had to avoid the long shadow of brother

What Do You See In This Ink Blot?



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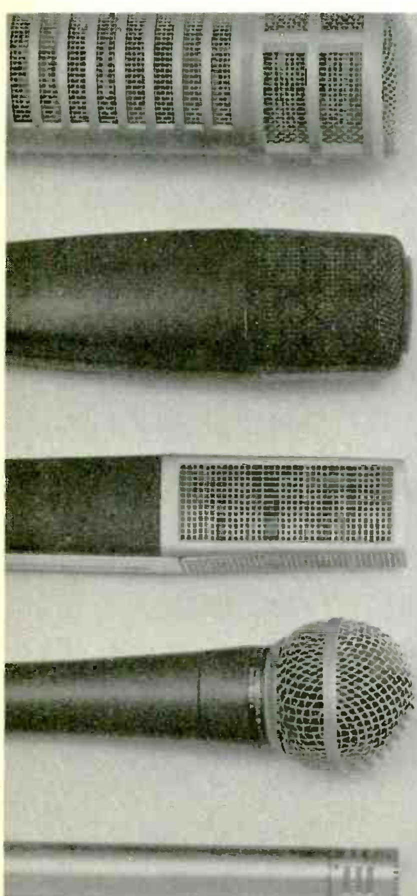

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James in pursuing his slightly more optimistically-oriented musical career, and he has generally survived to produce a handful of interesting, even charming, "little" albums featuring a wide variety of compositions. His voice sounds a little like James', but his outlook is different, and his material a bit more eclectic.

Man's Best Friend continues in the same vein. He has contributed six of the album's 10 cuts, and three of the remaining tracks are familiar songs by other composers that are given a fresh reading here.

Randy Newman's "Marie," one of his most beautiful songs, is given a very sensitive interpretation by Taylor here. He opens in solo with an acoustic guitar backing him, then is joined by a cello and later by a string choir. A concertina can be heard joining as the song moves to a close.

At the other end of the spectrum are two forms of the dance number—John and Joanna Hall's "Dance With Me" and Marvin Gaye's "Dancing in the Street." The former is given quite a slick arrangement, with acoustic guitar, percussion and keyboards sounding like a cushion for Taylor's vocal. Eagle Don Henley joins Taylor on the chorus, and there is a nice sax solo by Gary Foster, but the strings used here are unnecessary, as is the vocal chorus. The latter is actually a duet with soul singer Carla Thomas and features instrumental accompaniment by Jai Winding on piano, Jeff Baxter on organ, Steve Cropper on guitar and the Memphis Horns. This track has a low-down, gutsy feel to it, but it also has one principal drawback—the duets come out of the center of the sound, rather than divided between channels.

Baxter, for former Doobie Brother, generally has acquitted himself quite well, along with John Boylan, in producing the album, particularly on the Latin-tempoed numbers with their bright, shiny percussive embellishments. "Face Like a Dog" has a traditional calypso sound and is highlighted by its use of drums, percussion and steel drums and outstanding sound that manages to fill both channels without slurring.

Some of the material is quite good—"Pajamas" is a delightful kid's song reminiscent of "Knocking Around the Zoo," for example—but some of it is disappointingly mediocre. An obviously heartfelt love song, such as "Out of This World," resembles more an up-

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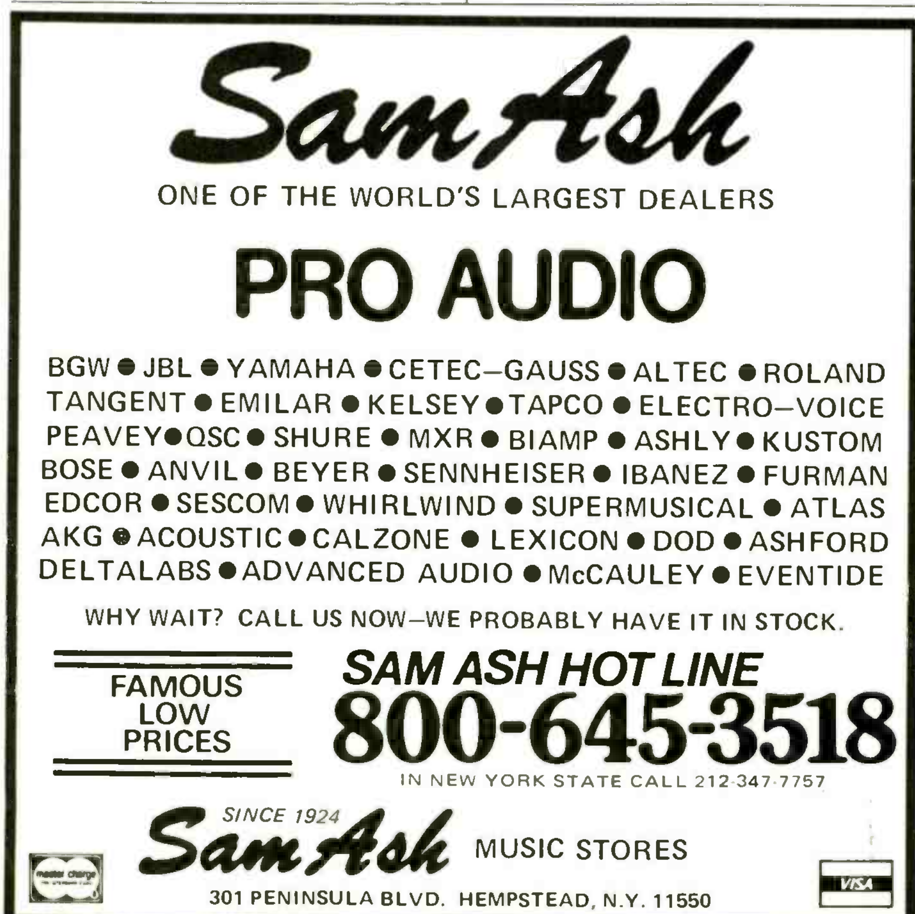
dated version of something like Bobby Goldsboro's "Leaving the Straight World Behind," while "First Time Love," by Pat Alger and Peter Kaminsky, is put into an awkward MOR position with string accompaniment that takes the edge off the song.

One also notices on occasion Taylor's strangely clipped pronunciation that sometimes sounds so gutteral as to resemble a foreign language. This is evident on "Marie," with the quickly rolled "r" prominent, and on "Sunshine Girl," with the line, "And you thought I didn't know nothin' " repeated several times in succession in that peculiar style that tends to submerge the projection of the lyrics.

But Taylor is having fun on this album, no doubt about it, and there is a refreshing, cheerful air of optimism, even naivete, about the selection and delivery of the material that makes one think of "tunes" instead of more ordinary "songs." Brother James is exploring now much the same kind of upbeat material that Livingston has been singing for years, and it might be interesting to see or hear the two of them together one day, playing off each other.

S.R.



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JAZZ

JOHN SERRY: *Jazziz.* [John Serry, producer; David Leonard, engineer; recorded at Studio Sound Recorders, Excalibur Studios, Studio City, and Lyon Studios, Newport Beach, Ca.] Chrysalis CHR 1279.

Performance: **Mostly hot**
Recording: **First-rate**

John Serry, late of the short-lived fusion group Auracle, has produced his second solo album, and one could not ask for much more in the way of uptempo, complex, forward-moving, keyboard-driven jazz. At times, the material almost sounds avant-garde; at other times, more traditional structure is followed.

Serry made an initial splash with a cut from his debut *Exhibition* album that was nominated for a Grammy last year. That album was an auspicious debut for a solo performer, and none of the luster has been lost in this year's effort. Using both standard and electric pianos, and two kinds of synthesizers, Serry has produced an exciting product here. What may be lacking in sheer melody is compensated for in sheer energy.

Jazziz contains eight tracks, including the title cut on which Serry pulls a Paul McCartney and plays all instruments. The title cut swirls about in a rush of textures, with instruments swaying between channels. Especially good percussion work is shown here.

Two of the more interesting tracks are "Doc Holiday" (should be spelled "Holliday"), and "Don Quixote's Hustle: A Disco Nightmare." On the former, a witty insertion of a gunshot, much like that in "The Sheriff" by Emerson, Lake and Palmer, is quite unexpected; on the latter, a Latin-tinged opening unfolds into a nice, syncopated beat reminiscent of small studio ensemble music of several decades ago.

Actually, the latter may be the musical centerpiece of the entire album. A disco-sounding bass line soon leads into a more standard keyboard-and-percussion statement. The bass and Bob Sheppard's flute weave the same melody line, but octaves apart, before Serry takes over with a busy right hand and more subdued bass underpinning in the left hand of the electric piano. The guitars deliver a break that resembles staccato Morse Code. The piece closes with a flourish, with various players shooting off in all directions, and then ends with drum beats—but not necessarily disco beats. Serry has called this a deliberately distorted parody of contemporary disco music, although the majority of the seven and one-half minutes is devoted to the kind of jazz Serry has been making on the other cuts.

My own favorite on the album is "Song For You," not to be confused with a Leon Russell song of the same title. This opens in a much lighter vein, with piano and winds dominant, before exploding into a rather frenzied ensemble sound heavy on guitar. Serry's piano again shows two different moods —exploratory in the right hand, and complementary to the bass line in the left.

Serry's production of his own material calls for tight, bright sounds, rather than expansive, open sounds. The ensemble playing often comes through as if several of the players were crowded around a single microphone, which tends to make the product sound smaller than it actually is. Also, because the compositions generally are up-tempo, the chance for lazy or relaxed, more laid back, improvisation must be forsaken; instead, a busy, sometimes furious, pace is maintained. The quickness in the drumming by Weather Report's Peter Erskine in "Up Start" is a good example of this, as is Serry's rapid fingering on the electric piano.

My principal complaint is that the material lacks good tunes. Jazz need not be tuneless, and too much improvisational material by the same composer/player will tend to drag after a while, no matter how much energy is expended in the execution. Serry's jazzy *Jazziz* has an abundance of energy, considerable skill and even some raw power in the grooves. With an ear toward more melody, what is now palatable could have been really exciting.

S.R.

HANK JONES & JOHN LEWIS: *An Evening With Two Grand Pianos.* [John Lewis, producer; Mike Moran, engineer; recorded at RCA Studios in New York, N.Y., Jan. 25 and 26 and

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Feb. 8 and 9, 1979.] Little David LD 1079.

Performance: **Simply grand twice**
Recording: **Good, clean, clear but I'd have liked even more separation**

According to Leonard Feather, who usually writes good scholarly liner notes, "the use of jazz piano duos on a formal basis for concerts, festivals or records is a relatively recent development." As frequently happens with Leonard, his roots don't go back quite far enough. The precedent for jazz piano duos goes back at least to March 21, 1930 when Thomas "Fats" Waller and Bennie Paine recorded "St. Louis Blues" and "After You've Gone" for the Victor company. There have been other examples of jazz piano duos such as Willie "The Lion" Smith first in 1939 with Joe Bushkin for Commodore and later for Flying Dutchman with Mike Lipskin at the other piano. Duke Ellington also recorded piano duets (with the above mentioned Bennie Paine) on the Ellington orchestra's 1931 recording of "12th St. Rag" and later as a duo with Billy Strayhorn for RCA. If Leonard's point is that the duo piano has been neglected by modern jazz artists I'd have to agree with him, and it is for that reason, as well as the excellence of the playing here, that I'm glad that John Lewis and Hank Jones got together.

The similarities between these two titans of modern jazz piano are amazing. I had always imagined that Hank was many years John's senior, but they were born within two years of each other and both in the middle west: John in Illinois and Hank in Michigan. Both are primarily associated with the bebop era of jazz but Hank goes back further than John does. Hank Jones has given programs of ragtime music and he is thoroughly grounded in the swing era, having played with such giants of those days as Hot Lips Page and Andy Kirk's band. John, on the other hand, was pretty much an unknown until he joined Dizzy Gillespie's first big bop band. If bop is a common denominator for these two pianists then Bird is the common experience. Both served their tenure in Charlie Parker's rhythm section, the younger Lewis being present on many of Bird's pre-California Savoy classics and the elder Jones showing up at a later period when Parker was recorded by Norman Granz for the Mercury,

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Norgran, Clef and Verve labels. If Parker is the common experience, then it's only right that two of Bird's tunes "Confirmation" and "Billie's Bounce" show up here. The other tunes go from standards like "Stompin' At The Savoy" and "I'll Remember April" to John Lewis specialties like "Odds Against Tomorrow" and John's adaptation of Bach's Prelude #8 from the Well-tempered Clavier which Lewis calls "Tears From The Children."

As Feather points out, duo-pianists need not only to complement one another, but to stay out of one another's way. "Non-jazz" piano teams like Eadie & Rack, Arden and Ohman, and Whittemore and Lowe accomplish this end by careful arrangement and painstaking rehearsal. For the improvising jazz piano duo, it's a lot more difficult. As Hank says in the liner notes, "we had to spend some time together and really keep both ears open."

That they did exactly that is the main reason this album works as well as it does. Yet when the listener has to deal with two pianos (acoustic Steinway concert grands, by the way) and especially when played by two artists as close in idiom as Jones and Lewis, it would have been helpful if Hank's left channel was a little more clearly separated from John's right channel. I know that smacks of ping-pong recording, about which I have complained long and often, but when you're dealing with two similarities the gimmick actually becomes a listening aid. Regardless, wonderful things happen on this record and I would highly recommend it to any jazz fan, especially if their taste runs to bebop. J.K.

JOE PASS: *The Complete Catch Me Sessions.* [Richard Bock, producer; Pete Welding, reissue producer; engineer not listed; recorded in Los Angeles in 1963.] Blue Note LT-1053.

Performance: **A completed pass**
Recording: **Fine and dandy**

JIMMY SMITH: *Cool Blues.* [Alfred Lion, producer; Michael Cuscuna, reissue producer; Rudy Van Gelder, engineer; recorded at Small's Paradise, New York, N.Y., April 7, 1958.] Blue Note LT-10540.

Performance: **Cliche-ridden and tedious**

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A FANFARE FOR (AND BY) TWO ALTO SAXOPHONISTS

By Nat Hentoff

A master improviser, whose incisive ideas and emotional force have become ever more compelling through the years, Phil Woods has now made the most thoroughly satisfying album of his career so far. *The Phil Woods Quartet, Volume One* (Clean Cuts, distributed by the Adelphi Jazz Line) is a continually brilliant illustration of Woods's extraordinary clarity-of sound, of structure, and of exhilarating jazz time. There is not a single excessive note, for one of Phil's key characteristics is his exceptionally judicious economy. The same is true of his crisply knowledgeable associates—bassist Steve Gilmore, pianist Mike Melillo, and drummer Bill Goodwin.

And this is a true quartet, not a soloist using a rhythm section primarily for accompaniment. Each player has ample solo space, and the ensemble passages involve everyone in challenging collective improvisation. From Charlie Parker's seldom performed "*Bloomdido*" and Bud Powell's "*Hallucinations*" to Benny Golson's lean, lyrical "*Along Came Betty,*" the Woods quartet engages in a veritable seminar in the art of advanced jazz invention. Phil in particular is wholly absorbing—disciplined passion fused with a constantly fresh, searching imagination.

The sound is the best I've heard on a jazz recording in months. It was a "live" date in Austin, Texas, and as the Edward R. Murrow TV series used to put it, "You are there!" For balance alone, this set should be played for all recording engineers. It can be equalled, but I doubt if it can be bettered for a long time.

While Phil Woods has always been in front of the band as a soloist, Marshall Royal was known—mostly to musicians— for decades as a lead alto player in the section. With Les Hite, Lionel Hampton, and then for more than twenty years with Count Basie. Finally, on the Concord label, Royal is having a chance to show that his former employers erred mightily in giving him so little space in which to stretch out.

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With a most congenial, flowing rhythm section (pianist Monty Alexander, bassist Royal Brown, guitarist Cal Collins, and drummer Jimmie Smith), Marshall Royal is a consistent delight in *Royal Blue.* With a warm tone and a loping beat (reminiscent of but not imitative of Pete Brown's "jump" style years ago), Marshall is supple, reflectively tender on ballads, and joyously assertive on up tempo tunes. His is an utterly distinctive voice, and the seasoning from all his years in the music makes his every note authoritative. From now on, any list of classic jazz altos will have to include Marshall.

The tunes are standards—from "*Avalon*" to "*I Got It Bad And That Ain't Good*"—but Marshall and his colleagues add new dimensions to each. In essence, this album is a reminder of how much sheer fun jazz can be—to hear and to play. The sound, as is customary on Concord, is clean, spacious, and *full.* That is, you get it all, in accurate, enlivening perspective. Like the Phil Woods set, this one is immune from changes in jazz fashion. It's basic, open jazz of invigorating immediacy.

PHIL WOODS: *The Phil Woods Quartet, Volume One.* [Bill Goodwin, producer; Cliff Carter, Fletcher Clark, Mark LeBaron, engineers.] Clean Cuts CC 702. (Available directly from the label, Box 16264, Baltimore, Maryland 21210.)

MARSHALL ROYAL: *Royal Blue.* [Carl Jefferson, producer; Phil Edwards, engineer.] Concord Jazz CJ-125.

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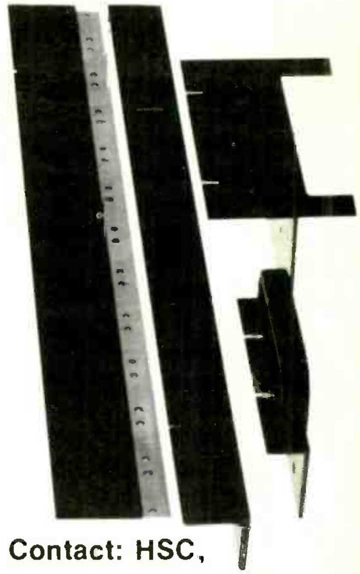

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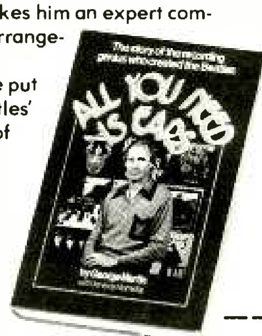

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Recording: **Not up to the Van Gelder standard**

Once upon a time, Blue Note was a great traditional jazz label. It was a bastion of New Orleans, Chicago and Dixieland in a sea of modern labels. The trendiness of style being what it is, today the label is remembered more for the likes of Jimmy Smith than it is for such giants as Sidney Bechet, Frankie Newton, Edmund Hall or Art Hodes.

Pacific Jazz was another label of importance. It didn't have the early giants that Blue Note did, but it had a few of its own like guitarist Joe Pass who was to go on to Norman Granzdier things in years to come.

Pass and Pacific are represented here by what a jazz record ought to be. It's cleanly recorded. It's full of interesting statements by a newcomer, at that time, making his first appearance as a leader on an LP. But Joe Pass isn't the whole story. He's ably backed by West Coast rhythm men like Albert Stinson and Ralph Pena on bass and Colin Bailey and Larry Bunker on drums. Yet the co-star of the LP is an underrated keyboard player/arranger named Clare Fischer. I don't know what happened to Clare Fischer. He did some important arranging for Dizzy Gillespie as well as a beautiful piano with strings album for Columbia, but he never made it big. He's heard here with Pass on some fine standards like "Just Friends" and "Days of Wine and Roses" plus two takes of Pass' first big quasi-hit "Catch Me." (Pass didn't really hit until a generation later when Norman Granz started to guide his career.) Yet even at this early stage, one can hear the consummate technique the man has on the guitar and his sense of good taste, especially when playing a ballad. At that point in his career he tended to rely more on tricks than he does today and the Wes Montgomery octave style crops up more in his playing than it does now. But then Wes Montgomery wasn't the first guitarist to use octave playing effectively in a jazz context; that goes back to Django Reinhardt . . . and so does Joe Pass (as well as going back to Charlie Christian and a few other players who pre-date him by a generation or so).

Jimmy Smith is another matter. I've never been too fond of his playing which I sometimes refer to as an organic waste of time. It's not that I have anything against the Hammond organ. Wild Bill Davis made it swing.

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So did Count Basie and Fats Waller. And there was a guy out of Gary, Indiana named Milt Herth who didn't make it swing—*he* made it stomp and shout and cry for mercy. Jimmy Smith—although I don't know his background that well—sounds like a pianist who took up organ as a double without bothering to really master the pedal work. Organists like those mentioned above and such later descendents as Shirley Scott and Dick Hyman have such a command of the pedals that a bass player becomes unnecessary—gilding the lily so to speak. Smith doesn't use a bass player either but the bass lines just aren't there—at least not the way they are with Wild Bill Davis and were with Fats Waller. His cohorts are no help. Lou Donaldson and Tina Brooks are two of the most uninteresting saxophone players I've ever heard and even the usually hard-swinging drummer Art Blakey doesn't get this album off the ground.

I don't know what obstacles Rudy Van Gelder was up against recording on location at Small's Paradise. All I can say is he's done better both on other locations and in his Hackensack studios. When I see the name Rudy Van Gelder on an album I expect a certain standard of excellence and clarity—not the dim and distant sound of this LP. The fact that this 1958 album was allowed to lie unissued for more than twenty years should tell you something about the quality both musical and technical. I don't go along with Michael Cuscuna's assertion in the liner notes that it lay dormant because it was a mono recording in a stereo world. First of all, the change from mono to stereo didn't happen overnight. Second, many of the early stereo recordings were electronically enhanced mono recordings. If Blue Note had deemed this releaseworthy, they could have put it out mono or compensated for it electronically. The fact, at least as I hear it, is that there isn't that much here worth hearing and so the shoddy recording is no great loss. The fact that it wasn't issued for over twenty years is no great loss. The fact that it finally was issued is, for my taste, a great waste when there are still classic Blue Notes by Sidney Bechet and Frank Newton that could and should be transferred to LP with the best possible up-dating in sound technology, given authoritative liner notes and most importantly, released in complete chronological sequence. J.K.

CLASSICAL

MAURICE RAVEL: *Daphnis and Chloe.* The Dallas Symphony Orchestra, Eduardo Mata, cond. [Peter Dellheim, producer; Edwin Begley, recording engineer; recorded in 1979 in Dallas, Texas by the RCA Mobile Unit, No. 1; digital recording and tape mastering by Sony Corp., Lewis Nanassy, digital audio engineer.] RCA ARC 1-3458.

IGOR STRAVINSKY: *The Firebird Suite (1919); Symphony in Three Movements.* The Dallas Symphony Orchestra, Eduardo Mata, cond. [Same recording credits as above.] RCA ARC 1-3459.

Performances: **Excellent, but not definitive**
Recording: **Digital quality and digital quality explained**

These two discs function on so many levels. They are excellent performances of three important works of the concert repertoire. They are also among the most successful examples of the digital process put to a totally musical use. They are also a remarkable last will and testament to the dedication, the art and the craftsmanship of the late Peter Dellheim who passed away soon after these recordings were made.

To expand upon the first aspect of these recordings, I do not expect you to believe that Eduardo Mata and the Dallas Symphony Orchestra are the ultimate. True, Dallas is a fine orchestra and is in many ways the equal of the Big Five (the New York Philharmonic, the Boston Symphony, the Philadelphia Symphony, the Cleveland Symphony and the Chicago Symphony Orchestra). In other ways they are not equal, but it is not fair to expect them to be. They are, like many other symphony orchestras, quite capable of excellent work, whether they rank as number one or number fifteen. Eduardo Mata is quite young for a conductor (still in his thirties). If his interpretations of Ravel are not as idiomatically correct as those of the late Charles Munch, and if his Stravinsky is not as definitive as the composer's own recordings on Columbia, so be it. Eduardo Mata brings taste and professionalism to his interpretations of these scores. I really think it's

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Hector LaTorre, Publisher

unfair to expect more of one so young.

Mata seems to conduct with an all pervasive feeling of the balletic quality of this music. Although his biography, furnished by RCA, doesn't make mention of ballet work as part of his background, Mata's sense of tempi (a bit slower than the average) and his relation to what is being danced on stage at any particular point in the score would indicate that he probably came to this music first in ballet, rather than in orchestral, form. At no point does he strive for the big splashy effect (a la Solti) or impose the orchestral disciplines to the point where the music becomes undanceable (as Karajan frequently does). This is a dance conductor's conception and as such these interpretations are valid and valuable despite any insights which the more experienced conductor might bring to the music.

In any difference of interpretations concerning Stravinsky, we must turn to the composer's own recordings. Whether or not Igor Stravinsky was or was not a conducting genius, he certainly knew his intent as far as the music was concerned. Whether or not one enjoys Stravinsky's recording of *Firebird* as a purely auditory experience, one cannot doubt the authority of his interpretation. It is my feeling that Stravinsky, by the time he made his last recording of *The Firebird Suite* in its expanded 1945 version, had long ceased to think of this music as accompanying a ballet. Were he to have recorded the entire *Firebird* I don't know what his attitude would have been, but I think his recording which appeared in the gigantic silver boxed set of five LPs issued just after his death, takes a symphonic point of view of *The Firebird*. I also note with interest that even though the 1919 suite has been the most frequently recorded, it was never done by the composer, who always recorded either the 1911 or 1945 version of the excerpts from this famous ballet, if he did not record the complete work.

Stravinsky's *Symphony In Three Movements* presents less of a problem since it exists in only one edition. Once again Mata's interpretation times out slower than Stravinsky's, and although the work was intended as pure orchestral music, Eduardo Mata seems to have had the dance in mind. As a matter of fact, a ballet was choreographed to Stravinsky's *Symphony In Three Movements* by George Balanchine in 1972, the year after Stravinsky's death. It works both as pure music (as Stravinsky recorded it for Columbia in 1961) and as dance (Mata's concept on this recording).

Ravel's *Daphnis and Chloe* is a work of totally another kind. It was conceived as ballet and completed by Maurice Ravel in 1912, placing it only a few years after Stravinsky's *Firebird*. The two suites of orchestral music from *Daphnis and Chloe* have long been favorites in the concert hall and on records but it was not until Charles Munch recorded the complete ballet score with the Boston Symphony Orchestra and the Harvard/Radcliffe Chorus that there was much popularity for the uncut work. The recording was one of RCA's early monuments in hi-fi. It was released as a mono disc and although there was a two-channel pre-recorded tape version available, when the time came to make a stereo disc it was decided to re-record the piece with the same conductor, orchestra and chorus. Now here we have it in digital stereo with Eduardo Mata and the Dallas Symphony Orchestra and chorus. Once again Mata's version will not be as quick as some other conductors. Seiji Ozawa, for example, drives the music a lot harder with the consequence that it often gets into a whipped frenzy and all detail is lost.

As far as the Sony digital recording sound is concerned, I can only marvel at the cleanliness, on both the high and low end of the spectrum, and just how much quieter and distortion free these recordings are. As a part of each LP, there is a detailed explanation of the digital recording process by RCA sonic expert, John Pfeiffer. One has to be a little familiar with computer terminology to understand it fully, but once you get past the basic concepts it all makes wonderful sense. (Lord knows the results are enough to justify anything they want to write about it.) Ten years from now when there will be digital reproducers available for the home (for those affluent enough to afford them), we can look forward to even better sound reproduction than this. For now, this will do wonderfully.

The best thing I can say about the RCA Digital series is that they have resisted the temptation to give us the kind of overblown, overstuffed performances that distinguished early hi-fi, early stereo and the like. These records are as musically excellent as they are sonically superb. The classical record business has matured quite a bit since the days of ping-pong-percussion. J.K.

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