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Profile:
Producer/Engineer
Eddie Kramer

# MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

VOL. 5 NO. 4
JANUARY 1980

## A Session With Bob Welch

**APOCALYPSE NOW:**
The Music Behind
The Images

**LAB REPORTS:**
LT Sound TAD-4
Delay/Reverb Unit
QSC A42 Power Amplifier
TEAC 144 Porta-Studio

**HANDS-ON REPORT:**
Allen & Heath/Brenell
Mini 8 8-Track
Open-Reel Recorder

NEW PRODUCTS
RECORD REVIEWS

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We have never been ones to sit around and watch others progress. stead, we prefer to be the leaders. And in our field, that is how iany regard us. So we apply our energy to expansion . . . growth in very phase of the development of our products.

Our mixers are expandable. *Studiomaster* was first with the add-on iodule concept in affordable professional grade mixers. Whether you elect our 12X2b or 16X4 units, neither will ever limit your group or tudio to its original capacity. By our four-channel-at-a-time expander iodules, a 12X2 can become a 24X2 in minutes. A 16X4 will change ito a 20X4 almost immediately, and without the hassle of dangling ables that can cause shorting or transporting difficulty.

Our features are expanding. As pioneers in the more-feature-per-ollar area, *Studiomaster* was the first to offer 5 way equalization on n affordable mixer. We still are. Our 16X4 mixer was the first studio uality desk that allowed the operator to have a separate stage (or udio) mix for the performers while still sending an independant, naffected mix to the tape recorder. No competition yet. And now, r mixers come with direct in/out channel patching as standard quipment, just one example of our expanded features.

Our product line is expanding. *Studiomaster* now has a 16X8 mixing console designed for 8 track studios and live performance. We offer the professional touring band the most affordable 20X8 monitor mixer available. And we now manufacture three and five way stereo crossovers which solve the complex speaker system problems of large concert arenas. We even make the coolest running, smoothest sounding 225 watt/channel amplifier around, too. Did we say we're expanding?

The only thing that really hasn't kept pace with our other phases of expansion is our price tag. It has escalated only a fraction in two years . . . and that only to justify the extras we offer. So we are still the most affordable 16X4 mixer that money can buy in our performance category.

Maybe you should consider expanding your studio's or group's ability to create. Give us a call at (800) 854-6219 nationally or (714) 528-3360 in California. We will give you the name of the *Studiomaster* dealer closest to you. If you visit him this month for a *Studiomaster* demonstration, you can receive a *Studiomaster* T-shirt free from us. Visit him soon to see the continually expanding possibilities of *Studiomaster* products.

For more information about Studiomaster products, please write to Craig Bullington, National Sales Manager, Studiomaster, Box 55, Atwood, California, 92601.

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# Silence. The Step Beyond.

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

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

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

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

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

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

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JANUARY 1980
VOL. 5 NO. 4

# MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

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# LETTERS TO THE EDITOR

## Bird Brood Take 2

In the July 1979 issue of *Modern Recording*, I reviewed Savoy 5500, a five-LP boxed set that was supposed to be the *"Complete" Charlie Parker Savoy Sessions*. Lo and behold, here comes another LP (Savoy Sessions. Lo and behold, here comes another LP (Savoy SJL 1129 *Charlie Parker, Bird Encores, Vol. 2)* with three fragmentary takes of "Marmaduke" and one complete take of "Marmaduke" which didn't get onto the boxed set.

Perhaps I should apologize for reviewing an incomplete "Complete" set but I *thought* it was complete... and *they* thought it was complete. Then all of a sudden, somebody found that one of the acetate safety copies that were used on the Savoy boxed set was recorded on both sides.

It's really nobody's fault and no harm was done except that anyone who wants these four takes of "Marmaduke" now has to buy an LP that duplicates material they already have on the boxed set to get that missing two minutes and fifty-three seconds of music. I think it would have been more fair to those purchasing the boxed set if the missing tracks had been put on a 45 for the benefit of those who need them to complete their Bird collection. But Savoy, they tell me, doesn't do 45s, so folks are just going to have to fork over $6.98 for just under three minutes of music.

Fortunately, take 9 is a full, completed take with another chorus of Charlie Parker's magnificent alto sax, Miles Davis on trumpet and John Lewis on piano. Also, the trading lines on the last bridge between Bird and Roach make $6.98 not an exorbitant price to pay for such beauty. And take 7 gives an extra four bars of Bird on the first chorus bridge and three bars of a solo before some fool yells "cut!"

This LP then gives us takes 6,7,8, and 9. The takes labeled 6,7, and 8 on the boxed set are in reality takes 10, 11 and 12. And I sure hope they don't find any more takes to add to the confusion. Although if they do, I'm sure they'll be good and I'm sure they'll cost you another $6.98 to get them... but then I'm a born cynic.

—Joe Klee
Music Reviewer
*Modern Recording*

## Dokorder Found; Creek Saved

I am writing in regard to an inquiry made by Randy Carrigan in your September 1979 "Letters to the Editor" as to where he might obtain schematic sheets for a Dokorder 7500.

I too am a Dokorder owner. I have a used 7500—it's about four to five years old—which I purchased for a great price. Buying used equipment has its risky points, and while small performance insufficiencies in a prospective purchase may not attract attention before the sale, I have found that they always do after—to the chagrin of the buyer.

I bought my Dokorder, and a week later I found out that the company was deceased, as you listed them in your *'79 Buyer's Guide*. It was shortly after that I began noticing the small things wrong with my machine. I knew then that a service manual might be quite important to the longevity of my recorder and to

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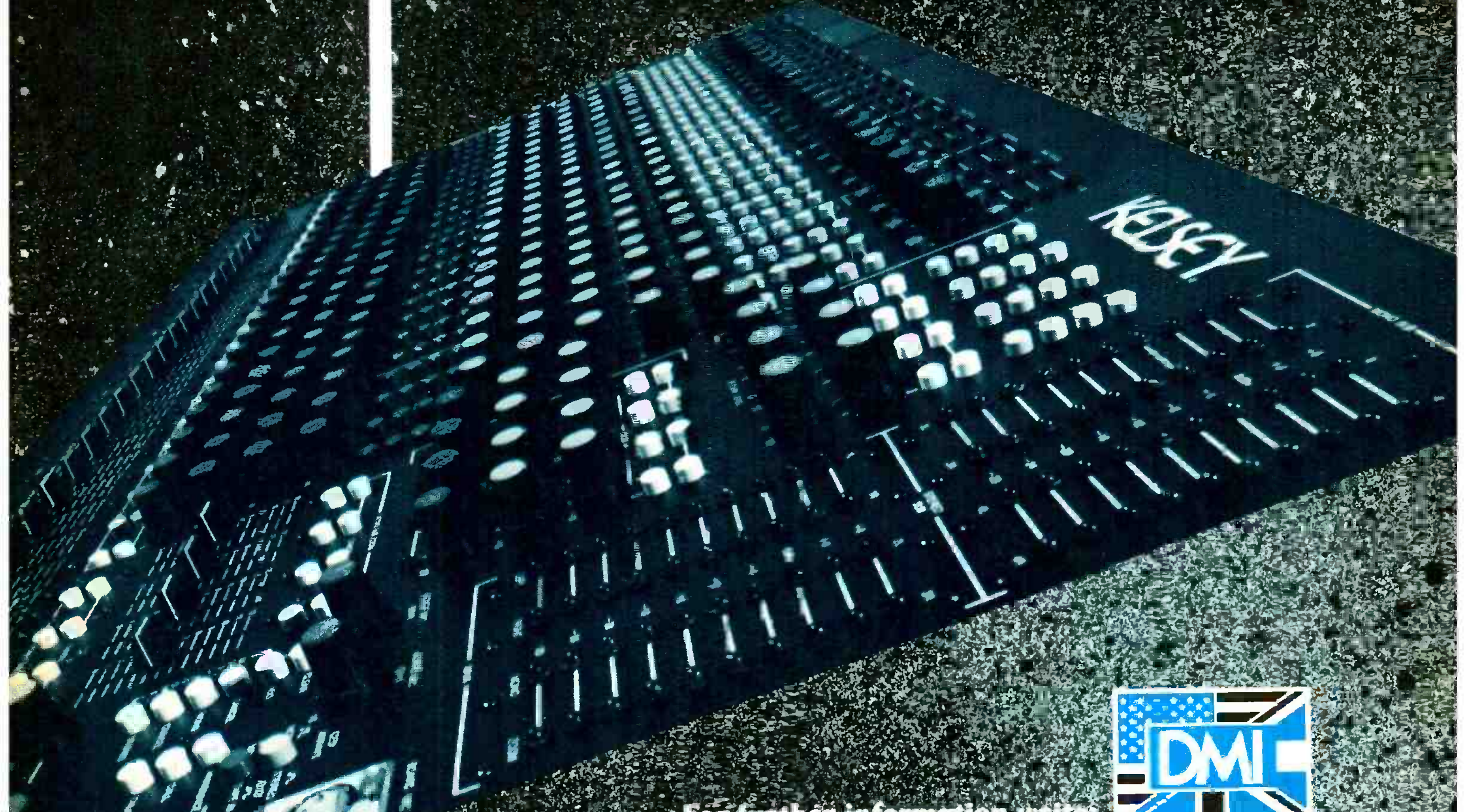
Combine the above with our traditionally superior SMF Road Case, and the new Kelsey Pro-Tour 8/4 Series is equally at home on the road or in the recording studio.

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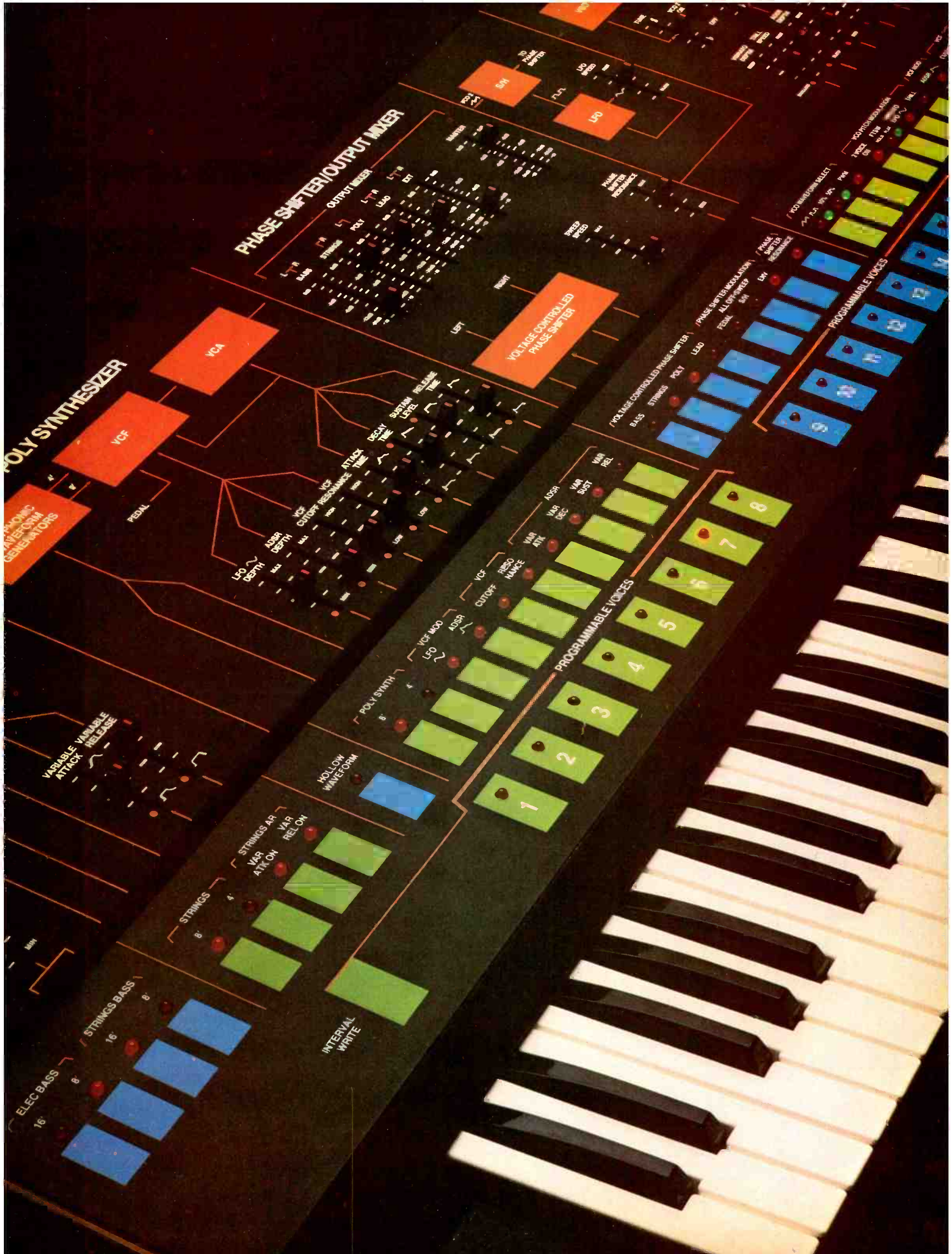
POLY SYNTHESIZER
PHASE SHIFTER/OUTPUT MIXER
OUTPUT MIXER
VCA
VCF
S/H
LFO
VOLTAGE CONTROLLED PHASE SHIFTER
LEFT
RIGHT
PEDAL
LFO DEPTH
ADSR DEPTH
VCF CUTOFF
VCF RESONANCE
ATTACK TIME
DECAY TIME
SUSTAIN LEVEL
RELEASE TIME
SWEEP SPEED
VARIABLE ATTACK
VARIABLE RELEASE
ELEC BASS
STRINGS BASS
STRINGS
STRINGS AR
VAR ATK ON
VAR REL ON
HOLLOW WAVEFORM
INTERVAL WRITE
POLY SYNTH
VCF MOD
ADSR
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RESO NANCE
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VAR SUST
VAR REL
PROGRAMMABLE VOICES
1
2
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11
12
13
14
VOLTAGE CONTROLLED PHASE SHIFTER
BASS
STRINGS
POLY
LEAD
PHASE SHIFTER MODULATION
ALL OFF=SWEEP
PEDAL
S/H
ENV
PHASE SHIFTER RESONANCE
VCO WAVEFORM SELECT
PWM
VCO PITCH MODULATION

# Take a closer look at the differences between the ARP Quadra and all other polyphonic synthesizers.

At first glance, you may believe that all programmable polyphonic synthesizers are the same.

Look again. The ARP Quadra is fundamentally different in design, function, and musical application.

It's a difference you can hear, see, and feel the minute you play the Quadra. And it's a difference that demands close examination.

## Many voices vs. many synthesizers.

All other programmable polyphonics offer 4, 5, up to 10 voices. The result of chording these types of synthesizers is similar to playing an organ or piano—all notes played have the same tonal characteristics.

The new ARP Quadra places *four separate synthesizers* under your control. Chording the Quadra is much like playing an entire orchestra. Each key depressed on the Quadra can produce up to four completely distinct sounds simultaneously. The Quadra's four separate synthesizers are tailored to certain kinds of sounds—string synthesizer, poly synthesizer, bass synthesizer, and lead synthesizer.

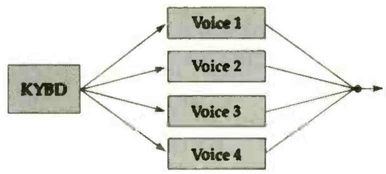
The effect created by mixing and blending these four sections is known as "layering," and is at the heart of the Quadra's tremendous commercial success. It explains why musicians and composers like Joe Zawinul, Ramsey Lewis, Styx, Billy Cobham, Kansas, Neil Diamond, and Electric Light Orchestra, have selected the Quadra for performing and recording.

## Programming and live performance.

Programming—the ability for a synthesizer to store pre-determined sounds in memory for instant recall—can be a great benefit in live performance. In essence, programming allows you to change quickly from one sound to another. Yet, if not implemented in a sensible manner, programming can lock you into an inflexible group of preset sounds that cannot be changed easily in live performance.

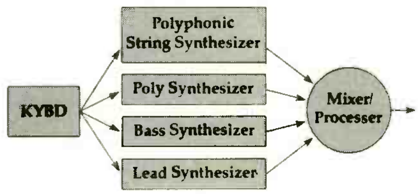
No matter how many programs a synthesizer can store—and some allow as many as 40—it seems there are never enough programs to allow for the subtle changes in texture that live performance ensemble playing requires.

ALL OTHER POLYPHONICS




1 Sound, 4 Notes
1 Sound, 5 Notes, etc.

ARP QUADRA




4 Sounds, fully polyphonic

*An illustration of the basic design difference between the new ARP Quadra and other programmable polyphonics.*

Here again, the Quadra is different. With 16 programs for *each* of its four synthesizers, the Quadra is uniquely suited to live performance control of the "final touches"—the balance between sections, animation, articulation, and so forth. A simple change in the mix produces a dramatically different sound from the same program position. Multiple use of each program, plus the multitude of live performance controls *always active*, makes each of the Quadra's program positions a source for numerous variations and textures.

## How to control programming before it controls you.

When there are too many things to do in succession, *programming can control you.* For instance, take the following typical sequence: "Press button one, press button two, press button three, play the keyboard." To change sounds, repeat this operation. Such serial operations seem more akin to computer programming than musical performance. ARP learned long ago that the best operational concept is the direct approach. That's why the Quadra makes extensive use of "parallel" control—you can get at any part of the instrument, change any sound, any aspect of any sound, *directly.* No sequence of complicated operations is required on the Quadra. And by using the 59 LED status lights, you will know what's going on inside each program you are using. You can be spontaneous, and change with the flow of the music.

## The Quadra's microprocessor works harder.

All the big programmable polyphonics use microprocessors to scan the keyboard and operate the programming. The Quadra's microprocessor does much more.

For instance, try out the Quadra's live performance sequencer. Play a chord on the keyboard, hit the footswitch, and suddenly you have a sequence of the notes in the chord. You can transpose the sequence, extend and modify it, and alter its notes without missing a beat.

The Quadra's microprocessor also makes intelligent decisions, like splitting the keyboard so you never get bass and lead synthesizer parts mixed up, or helping you with phrasing on the string parts. The microprocessor plays trills, intervals, and transpositions, controls the phaser and stereo animation, and even determines what the foot pedals do. In other words, the ARP microprocessor is programmed to let you concentrate on the music.

## Creative outputs (and inputs).

The Quadra's rear panel has 24 jacks for uncompromised flexibility. There's an XLR mono output, animated stereo outputs, and even quad outputs for studio work. Systems interface jacks will make the Quadra "control central" for slave units and remote synthesizers. Five audio inputs bring outside signals into the Quadra or allow processing of the individual sections of the Quadra with outside effects devices.

## Sound is the bottom line.

No question about it, four synthesizers sound different than one. It's a difference you'll appreciate the first time you put your hands on a Quadra.

Take a closer look at the new ARP Quadra at selected ARP Dealers throughout the United States and Canada.

ARP

For the names of selected ARP Quadra dealers, write:
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Lexington, Massachusetts 02173.
Or call: (617) 861-6000.

the integrity of my investment.

The scarcity of Dokorders seems to be high, and the scarcity of Dokorder service manuals even higher. I was able to reach Dokorder at the address you listed in the '79 Guide [there is now a new address], and from the Service Center, I obtained a manual that includes the schematics and the works on the 7500. I'm sure making my Dokorder last, and keeping it properly adjusted will be possible with this manual. I would advise Mr. Carrigan not to dispose of his in the creek before he gets a manual, and send away soon. Happy Dokording,

—Michael R. Dearth
Toledo, Ohio

I have some advice to offer Randy on his Dokorder in question in *MR's* September "Letters."

I used to own a Dokorder 7140 which had the exact same problem as his. I was about to begin semi-pro recording when the damn thing fizzed out on me. Since I had had minor problems with it in the past and also had trouble getting it repaired, without hesitiation I tucked it into a corner and went out and brought a Teac 3440 on credit. This machine paid for itself in a few months.

I managed to sell my expired $600 unit to a friend for $200. She is now having problems getting it repaired, and I am wondering if she still considers me a friend. I hope she can forgive me.

My advice to Randy is to yes, take your Dokorder and throw it in the creek behind your house, and go out and get something reliable.

—Taylor Sappe
Hazelton, Pa.



There is a beautiful creek which winds its way off nearby hills by my house, and on to the Hudson River. I would like to request that Randy Carrigan paddle his Dokorder up my creek rather than toss it into his.

I have a similar problem to Randy's in that I own a Dokorder 8140 4-channel machine. But mine is definitely *not* a lemon. The only problem that I've had in four years is one of the buttons popping off when I press "stop" and that was fixed with one drop of glue. I've made some superb recordings with the machine, including the mastering of an album.

What then, you might ask, is the problem? I want to keep my machine for a long time, and like Randy, I've been unable to obtain servicing information. It's just not practical to keep the machine without the ability to fix it should something go askew. I therefore repeat the request for help on reaching Dokorder or some representative. (Help!)

As far as Randy's machine goes, it is my experience that Dokorder machines have excellent electronics and most often fail mechanically. It may be worthwhile for him to locate another 7500 and rob parts. But if not, don't toss it away—ship it to me.

—Jim Brown
Katonah, N.Y.

*Ah, once again, our readers come through! Along with these letters, we got a phone call from a Jim Moorhouse in Bethel Park, Pa.—also one from an anonymous caller—with the updated info that the Dokorder Service Center, Inc. had moved to 1117 West 190th St., Gardena, Ca. 90248, (213) 515-1798. We checked that number out, and it seems that Dokorder is maintaining offices and a warehouse at this location—and will be pleased to give any help they can (even more so, we imagine, as an anti-pollution measure).*

*Now, Taylor, we're hoping you let your friend know she might be able to repair that machine you sold her at a loss, rather than offer to take it off her hands and fix it up yourself, okay?*

### A Fair Hearing for All

I'm absolutely amazed, though not surprised at the response toward Disco music. I've read each letter concerning the subject, and still can't figure out why these individuals dislike the Disco music format so much.

A few responses to responses, if I may: Sure, any fool can sit around all day and write two-chorded tunes—but this doesn't make Disco tunes alone; listen to what's considered "good" music by some contemporary rock groups—count them chords—one ... two ... hmmm (and the guy who wrote *that* letter is from my home town, too!).

Stop already with the "Disco lover-low intellect" connotations; it's simply not true. Die-hard Rock fans are as apt to fall into promotional schemes and fads as anyone—it all depends upon the individual's character, not his/her musical preference.

The most important point I'd like to make is that each response which I read treated Disco as less than music. So, okay, let's define music. Aha! Just as I thought, as many definitions as there are people. If that's so, how can anyone comment on a subject in music without expressing an opinion?

Here's my opinion. I like Disco, I like Rock, I like Classical, I like Bluegrass, I love Jazz, I love Avant-garde. I've played various stringed instruments and keyboards for almost fifteen years now, and while I'll admit that I'm no expert or maestro on any of these, I've found that there are many Disco tunes which are far more technically challenging than are some Rock tunes, and vice versa. There is good in both. Maybe that's why I like Jazz.

A final response. As an amateur recordist, I can vaguely (maybe even better than vaguely) deduce recording procedures and techniques when I hear them. There are obvious differences, and I see nothing wrong with using the studio to complement a particular composition—think of it as another instrument at the artist's disposal, in his or her pursuit of "art" and creativity. If creativity is influenced by dollars, it's likely that technique isn't. After all, who out there (especially of those who wrote in) really knows how the bucks are tossed around, anyway? All experts please step forward.

As a single individual, with a single opinion, and a single subscription (renewed yearly, of course), I vote *yes.* Lets see more on one of today's most popular musical media, yes, let's hear more on Rock, too, Jazz and Classical, as well. Any other type of music that I've missed mentioning. Let's hear about it, too. Let's open ourselves up and listen to what's around us, before we declare, "*Sound* Sucks!"

And since this is my first letter to an editor or magazine, I just want to say, "Love it." Thanks for the ear,

—Steve Dumler
Ellicott City, Md.

### Radical is Not Right

*We're hoping that our use—improper, in this case—of radical signs in "Math Notes" of "The Electric Primer—Part II," which ran in the October 1979 issue (pp. 44-5) did not serve to muddy the topic that author Peter Weiss was outlining in crystalline clairity.*

*The radical sign normally indicates*

*computation of square root. What we should have drawn in each case there, was a simple division bracket or sign.*

*More crucially, though, on page 42, we misrepresented the versions of Ohm's Law: In the left-hand column,* $I = ExR$ *should read* $I = \frac{E}{R}$. *In the center column,* $R = \frac{E}{I}$ *is correct, but* $E = \frac{I}{R}$ *should read* $E = IxR$.

### Wildey about Allman

I greatly enjoyed your interview with Steve Gursky in the 4/79 issue ("A Session With The Allman Brothers"). It is always interesting to know what the pros are using, especially when one of them is Tom Dowd!

In addition to my lifelong committment to playing music, I have recently developed a rather large interest in the engineering side of the industry. I took a special interest in the mics Gursky used in the sessions written up in *MR*—from firsthand experience (sitting under the P.A. in New Haven, the mics used in concert sure pick up.

I am also designing a tri-amped P.A. system featuring two Altec 9440s for lows and mids and an AB Systems 205 for highs, an Ashly 3-way crossover, and a Uni-Sync 100 for monitors. As far as speakers are concerned, I'm using two Gauss 18-inchers on each side for lows, but I'm not sure of what kind of cabinet to use. I was thinking of a JBL scoop design, but I'm afraid it would be too short-throw. I'd really like a folded horn, but I don't know of anyone who makes them. I know there's a company on the West Coast who makes them for Zappa and Yes.... Now we're talking about twenty cycles! That blows the people away!

Anyway, for midrange, I'll either go with one 15, one 12, or two 10s in some sort of cabinet. I'm kind of undecided at the moment. For the highs, though, it will be JBL horns with a 60-degree dispersion and 16-ohm drivers.

To round it off, I like the Studiomaster 16 into 4 board. Right now I've got a Peavey P.A. 1200, not to mention a couple of Altec A7s. That's basically my system.

Again, I was highly intrigued with the Allman article. I liked the much simpler and straightforward way of recording that Dowd and Gursky use. I'd also like to mention that I thought Tom Dowd's job on Lynyrd Skynyrd's "live" album was one of the cleanest "live" recordings I've ever heard. Like Gursky said, you're making records for people; not other engineers. I like to keep that in mind when we play our music.

—Joe Wildey
Ridgefield, Ct.

### Correspondents Quested

I originally sat down to ask some questions that I had about mixers which weren't entirely answered in Peter Weiss' answer in "Talkback" of your June '79 issue. While I was going through back issues of *MR* looking for the name of the guy who wanted to know if you could drop equipment from a scaffold and expect it to work (is he for real?), I found the answers to my mixer problems in another back issue. Very valuable, those back issues.

My reason *now* for writing is this: I do a different kind of recording. I work mainly on re-recording pre-recorded musicians in putting together mood tapes. Also, I do different things to songs, such as echo, flanging, doubling, phase shifting, combining songs, etc. I like to create different audio experiences with music; it is really a sort of "creative re-recording."

If there are any readers of *MR* who do the same or similar things, or who would like to find out more about what I do (this is a hobby for me), I would like to correspond with them, by cassette or by letter. I'm interested in pooling knowledge on these techniques. Please print my address.

By the way, I greatly appreciate *MR*. When I saw it on the stand, I flipped my bonza beans! (Thanks much.)

—Jon Seitz
4425 Wing View
Kettering, Ohio 45429

## In Which We Bare Our T-shirt

I was very surprised to see you take the attitude that you did when it came to Ken Rapoza's letter (*MR*, September 1979) concerning Disco music. Does your staff wear "Disco Sucks" T-shirts, also?

I looked back at some of your earlier issues and noticed that musically you seem to be Rock oriented, basically. Until Mr. Rapoza's letter, I always thought that your publication took the attitude that it is best to learn as much as possible no matter what the source. In the real world, people in sound reinforcement and engineering may be called upon to face many different musical situations, and the spoken word as well. There are types of music that I personally dislike, but knowing that different types of music are approached differently, I also realize that there may be something that I can use from these approaches for my own work.

I hope that you will take this letter as a constructive suggestion and look more into the backgrounds of Disco, movie scoring, classical and other types of music as well.

—D.K. Taylor
Bayside, N.Y.

*Honestly, we weren't copping an attitude when we buttoned our mouth and asked for reader response way back in September: While we felt obliged to comment in some way, we also wanted to present the views of the inveterate* MR *community. And so, we offered an open forum for democratic response.*

*Most of those readers who took the time to respond (many, many thanks to you all) also expressed firmly—and we quote a Virginian whose letter we printed in the November '79 issue—that "there's always something to be learned." Quite so. Personal taste is not a valid criterion by which to pass judgment on a genre . . . whether in fine art, literature, theatre, music or any art form. Fear not! We will continue to investigate and give you important articles on audio engineering in as many diverse applications—including Disco—as we can.*

*To reassure the virulent anti-Disco faction, we are not becoming a "Disco groupie mag" (as neither have we been "Rock groupie"). Mr. Taylor has, in our estimation, put this topic in proper perspective: Thank you, D.K., and thanks again to all of you who submitted your comments.*

## Truth in Advertising Welcomed

We subscribed to your magazine as a gift for my promising sound man husband. Though it is a primarily secular publication, we felt it would be a technically informative asset to our developing Christian music ministry. We were not disappointed. You produce a quality work and we hope to avail

ourselves of it for years to come.

We were prompted to write and compliment you upon seeing the Bose advertisement on page 19 of the September '79 issue. It caught our attention because the Christian group Truth shown in the ad took the time and effort to perform in our small community last year. Their superb message in music left a lasting impression in our hearts and the Bose equipment left a lasting impression in our ears!

We hope this ad is just the first of many acknowledgments of Christian music groups in your future issues. We would also encourage you to do a feature article about a Gospel group from time to time. We believe you have an untapped market in this particular aspect of music and trust the Lord to honor your attention to some of His "instruments" of love and peace in your magazine. God bless you,

—Bob and Barb Gille
Soldotna, Alaska

### Vital Statistic

*Please note this correction to the Vital Statistics chart in the Lab Report on Ashly Audio's SC-50 Peak Limiter-Compressor (October, 1979): Attack time as specified by Ashly and also as measured by L.F. and N.E. is actually 200 msec to* 20 *msec (not to 2 msec, as through a typographical error, we published). Our apologies for the error.*

### Send to the Hills

There is a great deal of information to be found in them there hills, but the unescapable reality that we must deal with is that New Mexico is hardly the music center of the nation. For this reason, most of our customers can hardly afford fancy, versatile, foreign-made microphones, and we've had to be satisfied with matching *the* right lower-priced microphone to any one job. As one might suspect, this gets quite satisfactory (excellent) results.

To help people do this, we have prepared an annotated listing entitled "Favorite Mics," featuring U.S.-made mics exclusively. Their domestic origin adds important advantages not only in price, but serviceability, as well. This circular is available at no charge through our mailing address, Rocky Mountain Audio, 1101 Santa Rita St., P.O. Box 2577, Silver City, New Mexico 88061.

Thank you for your fine magazine. Please keep up your campaign to improve vinyl pressing quality. The more defective records returned by the public, the sooner it will become unprofitable for the larger distributing companies to release them.

—Bob Mathis
Rocky Mountain Audio
Silver City, N.M.

*And thank you, Bob. Be prepared for a stuffed P.O. Box. Please, folks, send Bob a self-addressed stamped envelope along with your request for the listing.*

### Not Quite Primed

On page 63 of *Modern Recording* Volume 4, Number 1, you begin a series of articles concerning the building of an 8 x 24 active microphone splitter, a spring type echo unit, an analog delay, a graphic equalizer, and in that issue, a power supply to power these devices.

I was in jail when I got a hold of that October '78 issue, and could not for the life of me get the following issues containing the rest of the projects.

Having already acquired the parts for the power supply, I would like to know if you still have the back issues containing the remainder of these articles. Thanks.

—Greg Morgan
Washington Court House, Ohio

What happened to the electronic projects you were going to have each month? I made the power supply and am interested in making the equalizer.

—R. Charles Rownd
Brooklyn, N.Y.

*We've gone as far as the mic splitter, Greg, and that was in our December '78 issue, which is in plentiful supply. Send $2.50, and it's yours. In our November '79 issue, though, we showed how to build a dual limiter—although this was not on our initial list of projects, you might be interested in it as well. That back issue is also available.*

*In the meantime, though, we discovered that many of our readers were still seeking the really basic stuff that has to be digested before getting into building electronic projects, that stuff being mathematical, electric and electronic principles. Therefore, we called a partial moratorium on "building" articles, in order to publish first "The Electric Primer," in installments (three of which have appeared to date, in the* MR *September, October and December 1979 issues). So stay tuned, we haven't forgotten those projects.*

# "Once you get your hands on this machine . . . you'll see what we mean."

**PERFORMANCE:**

Overall Signal-to-Noise: 65 dB unweighted at 520 nWb/m (30 Hz to 18 kHz audio filter).

Playback Signal-to-Noise (electronics): 72 dB unweighted (with audio filter).

Headroom: +24 dB. Maximum Output: +28 dBm.

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**RELIABILITY:** An unmatched four-year track record of on the job performance for the original compact professional recorder. Day in, night out. Just ask someone you trust.

**ALIGNABILITY:** Any tape recorder must be aligned to achieve maximum performance. With the MX-5050-B, all primary alignments are on the front panel. So is a 1-kHz test oscillator. Secondary alignments are inside the bottom panel. You or your maintenance people can align it fast and easy. This saves you time, money, and enhances your reputation.

**INTERFACEABILITY:** With a flick of the output switch you can plug-in to any system: +4 dBm 600 ohm or −10 dB high impedance. No line amps or pads to mess with. A perfect match everytime.

**ADDITIONAL BENEFITS:** Three speeds, dc servo ±7%, ¼ track reproduce, full edit capability, over-dubbing, noise free inserts, XLR connectors, NAB/CCIR switching, unique three-position alignment level switch.

**PRICE:** Suggested retail price $1,945 (USA).

## MX-5050-B: The best value in a professional tape recorder.

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Call Ruth Pruett Ables on 415/592-8311 for the name of your nearest Otari professional dealer.
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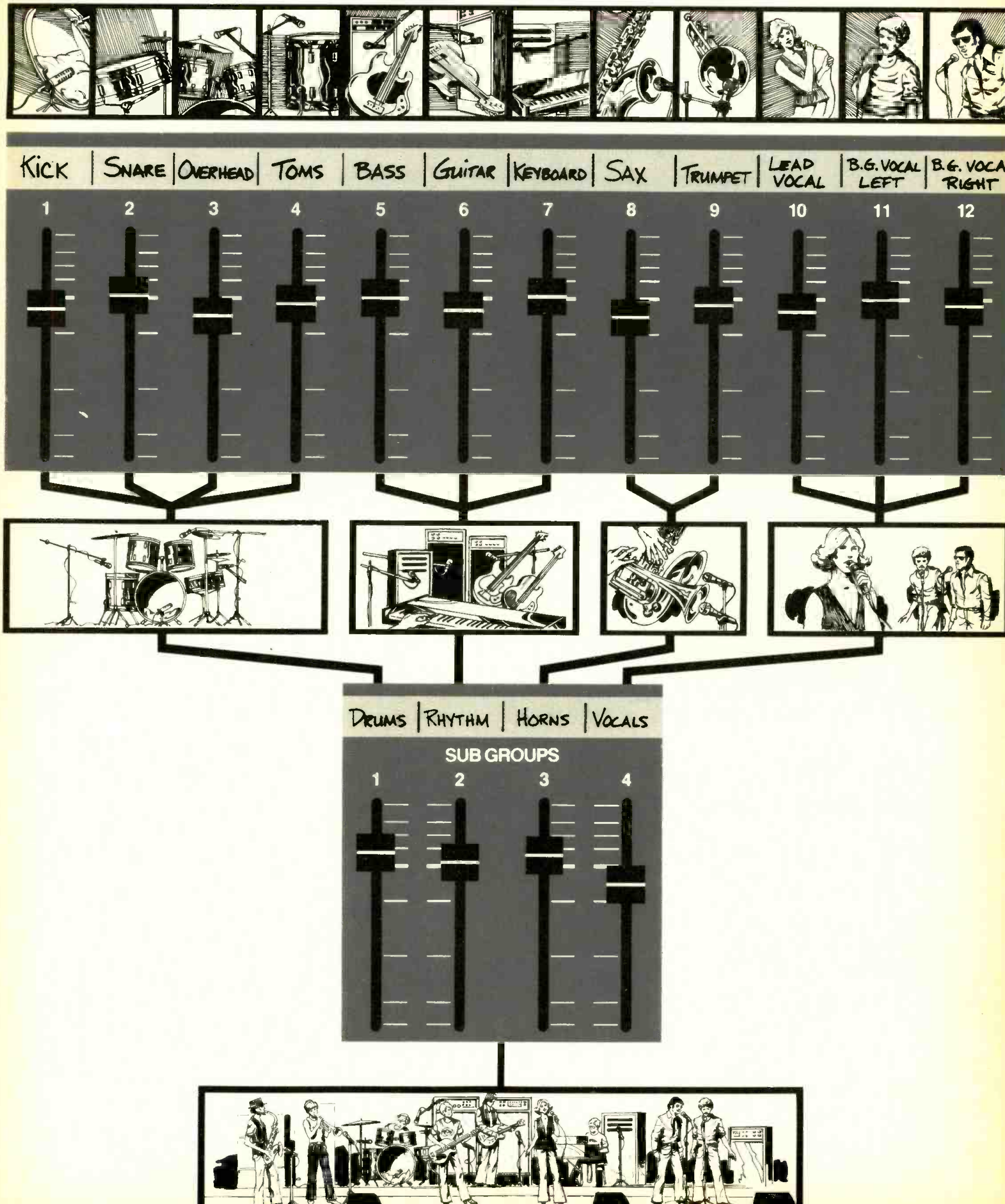
# Sub Group Mixing



# with TAPCO C-12

The TAPCO C-12's unique Sub Group mixing capabilities bring increased control in mixing. For example, when using a number of microphones on drums, with conventional mixing consoles, each microphone level is first adjusted for the proper musical balance (mix). Then, if the level of the entire drum mix is to be increased or decreased, all drum faders must be moved up or down in direct proportion to each other. This will often result in the loss of the proper drum balance or mix. When Sub Group mixing with the C-12, each of the drum mics would be assigned to a Sub Group. Now the level of the drum mix can be increased or decreased with a single fader. Because the C-12 has 4 sub groups, each section of the band (drums, rhythm, horns, and vocals) could be assigned to its own sub group fader.

The C-12 Mixing System can easily be expanded to 20, 28, or even more inputs with the addition of the C-8E Expander. The C-12 is a professional mixing system that is tailored to meet the needs of performing musicians and recording engineers. This uniquely designed mixing system boasts a list of outstanding features, some of which include: Sub Group Mixing • Extensive Front Panel Patching • 3 knob EQ with fixed high and low frequencies and sweepable mid-range • Monitor Buss • Effects • Aux Buss • Two Effects returns with level controls and Pan pots • Overload LED indicators on each input • Pan pot and Gain Trim control on each input • Stereo Headphone jack with internal headphone amplifier and automatic solo • Illuminated Vu meters with meter switching • +48V Phantom power • Mic Level XLR inputs (Transformer balanced) and line level inputs ¼" phone (Unbalanced).

See the exciting C-12 Mixing System for yourself. Ask to see its additional features that make it the most versatile and economical professional mixer available to date.

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Gulton Industries Ltd.
Box 520 340 Herbert Street
Gananoque, Ontario
Canada K7G 2V1

CIRCLE 77 ON READER SERVICE CARD

# 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 *reader's technical forum.*

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## The Cable Connection

**I have a few basic, but important questions concerning cables.**

**First, what cables are used where? It's my guess that shielded cables should be used everywhere except between amplifiers and speakers. If this assumption is correct, why are the three sends supplied in most snakes unshielded?**

**Thank you for doing such a fine job with *Modern Recording* and for giving people like me an opportunity to get answers to all our questions, no matter how basic or unimportant they might seem.**

**—Andy L. Vernon**
**Green Brook, N.J.**

*"The arteries of the sound equipment." That's how Brian Roth described cables. To Lothar Krause, problems with grounding numbered among the most frustrating encountered by soundmen on the road or in the studio.*

*Obviously, in knowledgeable circles cables are considered anything* but *unimportant.*

*For answers to the questions you have put forth, and to allay the myriad of others that are bound to occur to you, we have put together a short reading list that should set you on your way to more efficient use of cables and connectors.*

*For basic definitions of terms used in selecting and interfacing cables, refer to Roth's "Spaghetti Sonata: Or, Understanding Cables and Connectors" which was printed in June 1979. In the course of this article, Brian tells you what to use where—and why.*

*Not that we're anticipating any problems in your application of Brian's material, but just in case of unexpected mishaps, Krause's "And Now A Word About Grounding Problems" (May 1978 vintage) should help you pinpoint why things have disintegrated into static, and how proper "plugging in" can alleviate said noise.*

*Realizing that both these articles are based on some basic (electronic) laws of nature, and will no doubt provoke some soul-searching on your part, give the "Electric Primer, Parts I, II and III" a thorough reading (September, October and December 1979 issues). This series by Peter Weiss explores electronic theory and is important to anyone who will be dabbling with these principles in his or her day to day activities. It will also serve as a lead-in to a future article on practical electrical applications for sound. (We have no definite projected print date yet on this last piece, but hope it will prove to be a valuable source of info for soundmen.)*

## Compression Confusion

**I own a dbx 155 noise reduction system and a dbx 163 compressor/limiter. The 163 has a fixed compression ratio, but I do not know what it is. Is this a fixed ratio of 2:1? If it is, could the 163 be used as an encoder for noise reduction, and then can the encoded tape be decoded through the 155 since the 155 encodes at 2:1 compression and decodes at 1:2 expansion?**

**If I could use the 163 in this manner, how would I know when I have the right amount of signal passing through the 163 for proper encoding? Is there any way to set the amount of gain reduction with an oscillator?**

**—Taylor Sappe**
**Hazelton, Pa.**

Dbx noise reduction systems are based on compressing program materials 2:1 before recording, and expanding program material 1:2 upon playback. While it appears that a compressor, such as the dbx 163 or other 160 series compressor/limiters would perform the tape "encoding," it cannot. The encode "compression" portion of the total encode/decode cycle also contains frequency weighting pre-emphasis curves. These curves are needed to obtain the maximum tape signal-to-noise ratio possible without causing saturation. A complimentary set of curves is employed during the decoding. Frequency weighting is necessary in order to achieve the 30 dB of tape hiss noise reduction without encountering negative side effects. The same curves are also used in the dbx 158, 208, K9-22, 216 and other dbx professional tape noise reduction systems. Thus, the dbx 155 tapes are compatible with them.

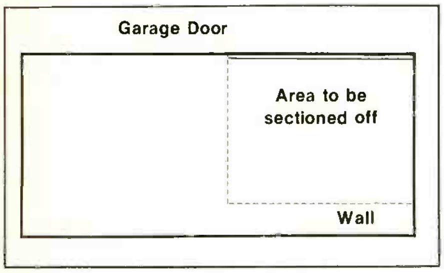
Please note that the compression curve of the dbx 163 is fixed, but the compression ratio varies with the level of program material relative to the threshold. Large amounts of gain reduction give a compression ratio approaching infinity, which therefore cannot be complimented in the expander.

—Harold Cohen
Marketing Manager,
Professional Products
dbx, Inc.
Newton, Mass.

## Soundproofing for Garage Studio

**I am presently designing and building a studio for rehearsal and recording purposes in one section of a three-car garage. In trying to obtain the lowest possi-**

ble amount of sound reflection, I have had no problem in insulating the ceiling, floor and the walls. However, I must isolate the area further by enclosing the two sides which are open (please see diagram). My question concerns how these "walls" (divisions?) should be built. Ideally, I would like to be able to achieve both a bright sound for some recordings

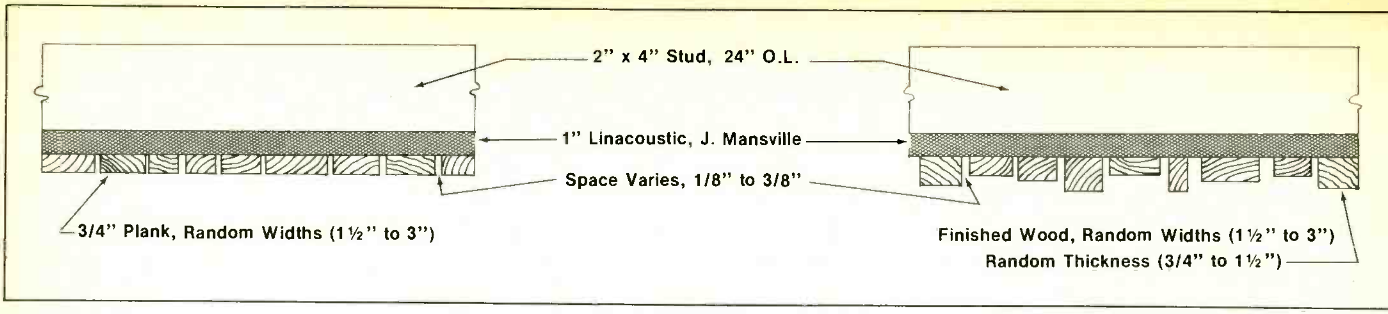



and a dead sound for rehearsals and other recordings. I have lumber, carpet remnants, egg cartons, etc., available to me for soundproofing, but not enough knowledge of carpentry to make any major renovations. Any help you can give me will be greatly appreciated.

—Tom Sumner
Moraga, Ca.

You actually have two problems to solve as does anybody who undertakes to "soundproof" a rehearsal or recording area. The first problem is to design and construct the isolation acoustics—the keeping of unwanted sounds from your space as well as the keeping of the sounds from your space from the outside. The second problem is internal acoustics, namely the creating of an environment which within your means will deliver to you a reasonable sound for the purpose you described, i.e., rehearsing, recording, etc. Let's deal with problem one first.

Not knowing the exact construction of the garage or, more importantly, where the garage is located, it's hard to give you accurate information for the sound isolation acoustics. If it is a detached structure, has few neighbors surrounding it, and has a relatively low outside ambient noise level, you can probably get away with single wall construction. In other words, you may take the existing structure itself and simply skin it with as high a mass material as your budget can afford—two layers of 5/8" sheetrock with a layer of soundboard sandwiched in between, on either side of the studs. Dense insulation between the studs should also be installed. Provided that all the seams are tightly caulked and the layers of soundboard and sheetrock are cross-layered, this con-

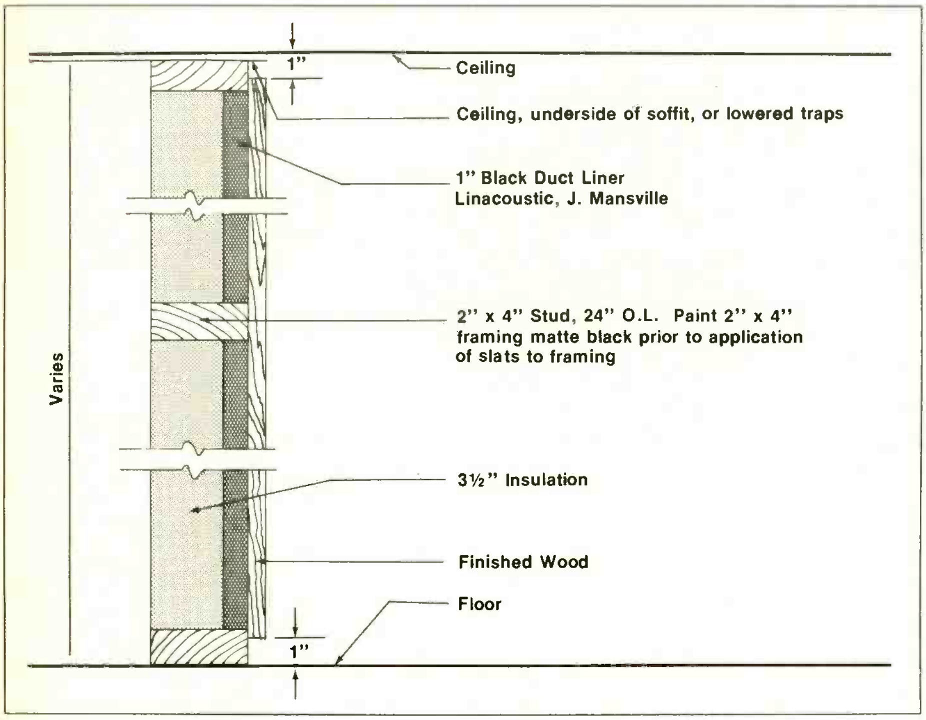



struction will give you a general system STC in the high 40's, depending on the exact structure itself. This will keep out quite a bit of outside noise and will also prevent your sound from bothering neighbors.

If the garage is next to a main road or in a dense residential area, then I'm afraid that you will have to go to double stud construction which is a much more complicated task. But a few reference books are available to show you this basic type of construction (*Graphic Standards; Noise Control* by Rettinger). I might point out that no matter what sound isolation construction system you choose, the weakest link is generally the seams and connections. Pay careful attention to the connections, use plenty of caulking and good craftsmanship. If there is one secret in sound transfer construction, this is it. The next weakest link is usually the doors and windows. You may not have any windows, but you'll certainly have a door. Again, there's plenty of reference literature available and I advise using two doors instead of one heavy door. It turns out to be cheaper, gives you an airlock and is usually much easier and quicker to build.

Problem two—internal sound acoustics—which in your letter you refer to as "dead" and "bright" sound. Of course, the very nature of your request is a bit of a contradiction but let's see if we can suggest a solution. The garage you describe has a number of internal sound acoustic problems which is, of course, why most studios don't look like garages. The most obvious is that it's a low volume, parallel-wall space with (at present) no absorption. The simple solution would be to absorb in a broad band nature low frequencies and at the same time defract mid-and high-frequencies. The one obvious thing that comes to mind would be a good slat resonator solution tuned at about 120 cycles. This is not difficult to build and I have drawn a little sketch of the type of construction that would be involved. They look real good and can be done out of almost any kind of wood (i.e., inexpensive) and turn out to be a material light but labor intensive solution—which appears to fit your budget. My advice would be to cover anywhere from 30-50% of the walls within this solution. Once doing this, a number of more "obvious" acoustical treatments such as rugs, carpets, egg

crates (if you wish), splayed walls, etc., might be employed to give high frequency absorption and more importantly broad band diffraction. One final word, do not cover every surface with mid-high frequency absorption, i.e. insulation, egg cartons, etc. An overly "dead" room, especially at the mid- and high-frequencies where vocalists sing and trumpet players play, etc. will not give you what you *think* it will give you. It will frustrate the musicians and although it might give you one or two more musicians worth of separation, you're only going to be able to rehearse 4 or 5 people




in the space due to square footage limitation so it would be better to intelligently absorb low frequencies and diffract and scatter mid- and high-frequencies.

In passing, the best single and rather simple book that would cover many of the points you might need to know is *Acoustics Techniques for Home and Studio* by Alton Everest. Although incomplete in some areas and no longer 100% accurate, it's an excellent reference book, easy to understand and would help you immensely. And, don't forget lighting and don't forget air conditioning.

—Herb Schwartz
Sugarloaf View, Inc.
New York, N.Y.

## A Rose By Any Other Name . . .

**I would appreciate it very much if you would answer the following question concerning noise reduction.**

**Is there any *audible* difference between the dbx 155 and the TEAC/Tascam DX-4? Please note that I am not asking you the different characteristics of each unit—only the audible outcome. In other words, which unit would be quieter with Tascam's 40-4?**

**The reason I ask is that I'm planning on purchasing the 40-4, but one particular and very knowledgeable friend/salesman insists that the dbx 155 is superior in quality to the DX-4. He notes there is much more noise build-up when layering tracks with the DX-4, especially in the case of cymbals. He also says you're paying for automatic switching and LEDs instead of quality.**

**I don't care about the luxury of automatic encode/decode, I just want the best for my money.**

**—Collin G. Heade**
**Portland, Or.**

*We have all come to accept the fact that a rose by any other name would smell as sweet, but does this premise hold true in the more exacting world of noise reduction systems? We don't mean to cheapen the words of the immortal bard, or to bandy them about lightly, but they do seem to apply to the situation here.*

*To get an objective answer to the question posed above, we contacted Harold Cohen, Marketing Manager, Professional Products, dbx., Inc., and Dale Dalke, Technical Correspondent, TEAC Corp. of America. What we learned from our conversations with these*

*gentlemen was quite informative to us, and will hopefully assist you when it comes time to make your purchase.*

*Both Dale and Harold informed us that TEAC is a licensee of dbx's professional noise reduction format. That is to say, they license the circuitry and are required to manufacture their units as dbx does. Their design is exactly the same, but the component usage differs. In the DX-4, for example, more integrated circuits are used than in the dbx 155, which primarily uses discrete circuitry. Thus, from a practical standpoint, the performances of the two units can be said to be identical.*

*The DX-4 is designed for use exclusively with the Tascam 40-4, not as a general purpose noise reduction system to be used in the field and, as such, is not in direct competition with the dbx 155, whose use is not limited to any one piece of recording equipment. Do keep in mind, however, that the dbx 155 is compatible with the 40-4.*

*As for your question of whether there is an audible difference, we feel that is a very subjective area, and depends to a great extent on what your desired sound is. Our advice on this point is to use your ears, and don't get hung up on labels.*



## Reconing Recommendations

**Having repaired studio and stage sound reinforcement equipment for some time now, I am continually asked by musicians (especially guitarists) to recone their rattling or blown speakers. Can this be done successfully? If it is a procedure that can be attempted, any hints? Also, can you pass along any information about driver diaphragms; specifically, their design and replacement? Information concerning the feasibility of these tasks will be appreciated.**

**Keep up the good work!**

**—Ken Durham**
**Pasadena, Tex.**

Woofers can be reconed. However, there is very little information available from manufacturers on the correct procedure to use when reconing. First, you should try to use the exact recone kit for the model speaker you are reconing and this should be available from the original raw driver manufacturer. Unless you plan to recone a number of speakers, I would recommend sending the speaker back to the manufacturer for reconing. If you have the need to recone a lot of speakers, you might want to try your hand at learning how. Most likely, the first two or three speakers you recone will not be successful and will have to be cut out and thrown away. Tolerances are very close in the speaker gap and great care must be taken to recone a speaker properly. A speaker recone kit consists of a number of parts, which include the spider, coil, cone, surround, dust cap, and gasket. These all must be aligned and glued into exact position with different adhesives and epoxies.

Compression driver diaphragms may be replaced very easily with a few precautions. First, remove the screws from the back cover of the driver and then the screws holding the diaphragm. Check to see that the gap is clean before installing the new diaphragm and very carefully align and tighten the screws.

—Ron Fox
Vice President/Chief Speaker Engineer
Showco Manufacturing Corp.
Dallas, Tex.


# KING OF THE JUNGLE

You're face-to-face with the most magnificent beast in the Pro Sound Jungle—the all new Model 1200A Power Amp from AB Systems.

And while many of the faces in the jungle are looking more alike every day, the 1200A not only looks different—it is different.

The 1200A is the first totally modular Power Amp. It features independent power supplies for each channel, as you'd expect in a high-end professional amplifier. But what really sets the 1200A apart is its completely interchangeable output sections. Each of the "output tunnels" is independent of the other. And each output-module packs its own whisper fan to make sure your sound system keeps its cool—even when the music's hot.

The King of the Jungle measures a slinky 5¼"x19"x 15", so it fits all standard rack-mount installations. Yet it pumps out a full 500 watts per channel into a 4 ohm load. (300 watts into 8 ohms.) And when you compare its slender price—it'll bring out the beast in you too!

The 1200A from AB Systems. Long live the King!

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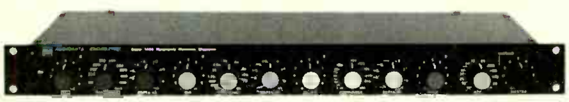
CIRCLE 133 ON READER SERVICE CARD

# THE PRODUCT SCENE

By Norman Eisenberg

## 4-WAY PARAMETRIC CROSSOVER



From Audioarts Engineering comes word of the model 1400, described as a monophonic four-way parametric electronic crossover intended for use with high-power four-way and three-way speaker systems. Front-panel crossover frequency controls enable the device to be matched to virtually any combination of drivers. Crossover depth controls (−7 to +1 dB) enable the operator to compensate for speaker-frequency adnormalities in the crossover region. The unit also includes four front-panel level controls, four phase-reversal switches, a system-overload indicator and a master level control. It also incorporates a variable high-pass filter. The model 1400 mounts in a standard rack space and measures only 1¾ inches high.

CIRCLE 1 ON READER SERVICE CARD

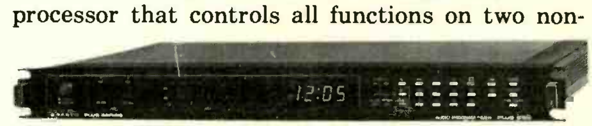
## UNI-SYNC MIXER

From Uni-Sync comes word of its PMS-2 (the letters stand for professional mixing system). The low-profile, rack-mountable unit automatically calibrates the 100-percent light to the peak output of any amplifier under any load condition. The peak-reading meter is adjustable from 0.125 to 1250 watts RMS into 8 ohms. Input is balanced high-Z (16 K ohms, min). There is no common ground, which means the PMS-2 can be used to monitor balanced lines or power amplifiers with floating grounds or in bridged mode without affecting the balanced line or causing shorts between two different bridged amplifiers.

CIRCLE 2 ON READER SERVICE CARD

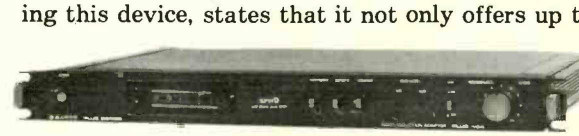
## TIMER; ANTI NOISE UNIT FROM SANYO

The PLUS E55 microprocessor programmable timer from Sanyo is designed to control an audio system and to perform the following functions too: record programs off the air (when tuner and tape deck are unattended); switch lights on and off at preset times; serve as an alarm clock; switch on preselected programs. The unit consists of a fluorescent 24-hour digital clock with a single-chip microprocessor that controls all functions on two non-




audio channels. Channel 1 contains two AC outputs each with one on and one off switching contact. Channel 2 has two outputs, each with up to nine separate on/off contacts. Front-panel controls permit easy setting of on and off times, with a digital readout for those of Channel 1. Memory recall and memory-clear buttons enable the user to reveiw what the unit has been instructed to do, or to cancel previous instructions. Price is $250.

The PLUS N55 Super D is a noise-reduction unit claimed to cut tape noise by 40 dB. Sanyo, in offering this device, states that it not only offers up to



10 dB more noise-reduction than is available from the best existing consumer units, but that it also overcomes their problems. Basically a compander, the Super D compresses signals during recording, and expands them on playback. Price is $360.95.

Both units are low-profile designs that may be rack-mounted.

CIRCLE 3 ON READER SERVICE CARD

## RECORDING PROS FORM GROUP

Recording studio representatives met recently in Ft. Lauderdale, Fla. to discuss the state of the art and to set guidelines for the newly formed Society of Professional Audio Recording Studios (SPARS). The group, says a spokesman, "has dedicated itself to the achievement of excellence in the craft, establishment of a code enumerating professional standards, quality and expertise, the advancement of engineering hardware and techniques. It will also serve as a platform for statements on technical matters affecting the industry."

Elected as interim chairman, pending official elections, was Joseph Tarsia, president of Sigma Sound Studios (Philadelphia and N.Y.). Tarsia stated that SPARS will serve as a "communications link with studios around the nation and thereby be in a position to act as the focal point of valuable dialogue with manufacturers of audio equipment."

Founding company-members of SPARS include: A&R Recording Studios, New York; Atlantic Studios, New York; Criteria Recording Company, Miami; Filmways Heider Recording, Hollywood; Group IV Recording Studios, Hollywood; House of Music, New Jersey; Howard M. Schwartz Recording Inc., New York; Kendun Recorders Inc., Burbank; Larrabee Sound, Hollywood; Media Sound, New York; Record Plant, Los Angeles; Regent Sound Studios, New York; Sigma Sound Studios, Phila.; Soundmixers Inc., New York; Studio 55, Los Angeles; Woodland Sound Studios, Nashville.

Membership is open to any *recording studio* agreeing to follow and maintain the standards postulated by SPARS. Applications should be sent to Kent R. Duncan, Kendun Recorders, 619 South Glenwood Pl., Burbank Ca. 91506 (213) 843-8115.

CIRCLE 4 ON READER SERVICE CARD

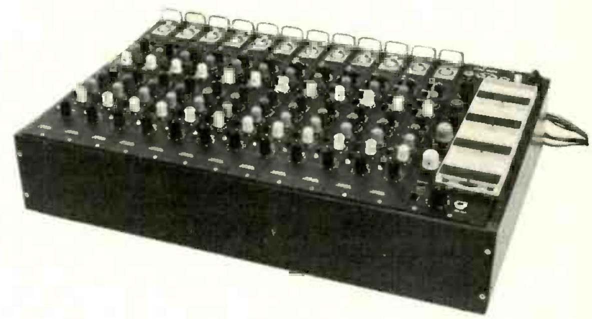
## WHISTLES WHILE IT WORKS

A new cassette tape-head degausser whistles while it removes residual magnetism from tape heads. When the whistling stops, the job is done. Known as WhistleStop, the device is made by Robins Industries and is priced at $26.50. Self-contained and automatic, it is inserted into the deck and the play button is pressed. Supplied with it are two standard mini wafer batteries said to provide enough energy for over 1000 operations.

CIRCLE 5 ON READER SERVICE CARD

## NEW MIXER SERIES

A new mixer in its Series 400 is Interface Electronics' 400L. Available in 8, 12, 16 and 24 input mainframes, the device makes four output mixes with pot sends from each input to each output. Each input module has a XLR-3 type, 200-ohm



balanced input with four-position pad and four-position input gain set. Together, these adjustments permit a range of inputs from −50 dBm to +10 dBm. Each input also has an LED overload indicator, three equalizers with 12-dB boost or cut with a four-frequency switch on the mid-frequency equalizer, a four-position low-frequency 12 dB/octave rolloff, solo to operator monitor, on/off switch and input channel master. The output panel features four masters, four VU meters and operator monitor with mix-select switch and solo. Series 400 mixers are fully modular, with only the mother board in the box; the power supply is in the master module. The 8-input version is priced at $1550; the 12-input version costs $2250.

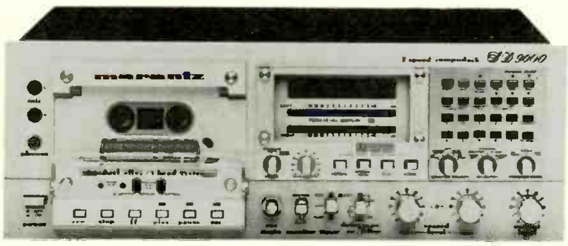
CIRCLE 6 ON READER SERVICE CARD

## ORBAN UPDATES REVERB SYSTEM

Orban's model 111B dual-spring reverb system is now being delivered with six, instead of four, springs per channel. The increase is said to provide lower flutter, higher echo density and a smoother, more natural sound. All of the standard Orban signal-processing is retained, including the company's exclusive "floating threshold" limiter to minimize "spring twang," and a versatile equalizer with quasi-parametric midrange and shelving bass sections. The new version carries no cost increase.

CIRCLE 7 ON READER SERVICE CARD

## MARANTZ OFFERS TWO-SPEED CASSETTE DECK



The standard $1\frac{7}{8}$ ips speed and the faster speed of $3\frac{3}{4}$ ips are included in the new Marantz SD-9000 cassette recorder, which also features three heads plus a microprocessor that allows for direct keyboard entry of a maximum sequence of nineteen music selections into the unit's memory. "RAM" (random access memory) then allows playback of selections as they are sequentially entered. Alternately, "SAM" (sequential access memory) allows for playback of selections in numerical order. Direct keyboard entry also is available for counter, clock and timer functions.

The SD-9000 is metal-tape compatible; it also handles normal, chromium-dioxide and ferrichrome tapes. Included are mic-line mixing, a built-in timer, bias fine adjust and more. Price is $775.

CIRCLE 8 ON READER SERVICE CARD

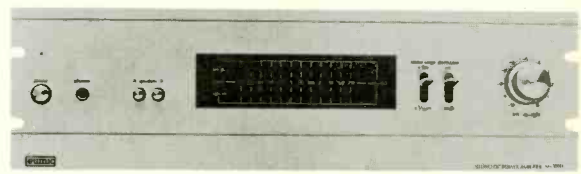
## MONO ELECTRONIC CROSSOVER

Musimatic of Decatur, Ga. has brought out the model C-35 mono two-way electronic switchable crossover designed for use in 2-way and 3-way professional sound reinforcement, stereo and disco systems. Of rack-mount size, the C-35 features all front-panel controls for input level, three-position frequency selector, horn phase switch and lighted power switch. Input and output connectors are ¼-inch jacks. Price is $253.

CIRCLE 9 ON READER SERVICE CARD

## EUMIG EXPANDS U.S. OFFERINGS

Eumig, whose cassette deck was reviewed here some issues back, now offers some new products, including the M-1000, a stereo power amp with minimum output per channel of 110 watts (RMS, 8 ohms, no more than 0.0075 percent THD). S/N is claimed to be 105 dB. Dual calibrated arrays of twelve LEDs monitor the level in each channel. Eumig also is offering a low-noise DC preamp control unit, and a digitally-synthesized FM tuner.



CIRCLE 10 ON READER SERVICE CARD

## GAS AMPS AND OTHER ITEMS

Heard along the pipeline from GAS (that's Great American Sound Co., Inc. of Chatsworth, California) is word of the Ampzilla IIa, a stereo power amplifier rated at 200 watts per channel for 8-ohm loads, and 360 watts per channel into 4-ohm loads. Rack-mountable, the amp boasts a slew rate of 50 volts-per-microsecond and its front panel features twin meters plus a pair of stereo headphone outputs. A ventilating fan is integral. For lower power needs (80 watts per channel and 150 watts per channel into 8-ohm and 4-ohm loads, respectively), GAS offers an amp known as "Son of Ampzilla." For even lower power requirements, there's an amp called "Grandson" (40 and 80 watts per channel into 8-ohm and 4-ohm loads).

To electrically couple two power amps into one, by way of increasing total power output, GAS has a device known as "The Bridge" which just about quadruples the power output of GAS amplifiers. For instance, two 40-watt Grandsons linked by The Bridge result in an amplifier rated for 160 watts per channel into 8 ohms. Two 80-watt units bridged produce 300 watts per channel. Two Ampzillas bridged can produce 720 watts per channel.

GAS offers preamps too, plus a line of m.c. phono pickups. It also has a furniture-style rack which it calls—you might have guessed—the Gas Station.

CIRCLE 11 ON READER SERVICE CARD

## ALMOST PRODUCTS

Some products are so new they haven't materialized yet. We are assured, however, that they will. Two such items that are making news among audio insiders are "turntable systems"—one announced by Technics and the other by Philips.

The Technics product seems closer to realization so let's look at it first. It is the model SL-10 turntable, complete with moving-coil pickup and built-in head-amp. The pickup moves in a straight-line (no pivoted tone arm) via a carrier that is part of the turntable's cover. At the center of the cover is a round section that holds the disc in place on the platter. The unit operates only when the cover is closed, and when it does—says Technics—playback conditions are so stabilized that you can stand the whole thing on end without mistracking. Unusually compact, the forthcoming SL-10 is a bit larger in width and depth than a 12-inch disc itself, and is just a few inches high with the lid down.

The turntable uses a quartz direct-drive motor while the "arm" is powered by another motor with the aid of an optoelectronic guide system plus a micro-computer. As with all straight-line tracking systems, there is no lateral angle error, no need for anti-skating, and presumably the ability to use the lowest possible vertical tracking force. Since the head-amp for the m.c. pickup is built-in, the signal from the SL-10 can be plugged directly into the normal magnetic phono input on any amplifier. As a final fillip, the SL-10 can be used as a true portable since it will run on any 12-volt DC supply as well as on regular AC line voltage. Expected market date is April 1980; price will be about $600.

CIRCLE 12 ON READER SERVICE CARD

Even more far out is the Philips CD (compact disc) system. This is completely revolutionary in that it involves the use of a new disc, "only 4½ inches in diameter and slim as a dime." The disc is digitally-encoded in the form of microscopic pits and flat sections along a 2½-mile long track that runs in a spiral on one side of the record. This track is played by a miniature laser-beam system mounted *under* the platter (the disc is placed track-side down on the turntable) from the inside toward the outer edge. A decoder, built into the player, converts the light pulses into electrical signals. These signals may be connected directly into the standard magnetic phono input on a conventional amplifier. In other words, Philips is making digital discs available in the form of an optional "attachment" to an existing analogue sound system. Advantages claimed for the CD system—besides compactness—are more accurate reproduction; the elimination of background noise; no more rumble, wow and flutter; and no record wear since there is no physical contact between the disc and the means by which it is tracked. Even in its compact size, the one side of the new disc carries a full hour of program.

All Philips will say as to when this prodigy may hit the market is "in a few years—1982 or 1983." As to price, they are even less definite—"about the cost of a good conventional turntable." Stay tuned to this station for late-breaking details.

CIRCLE 13 ON READER SERVICE CARD

# BUY THE TIME YOU NEED TO DO IT WRONG.

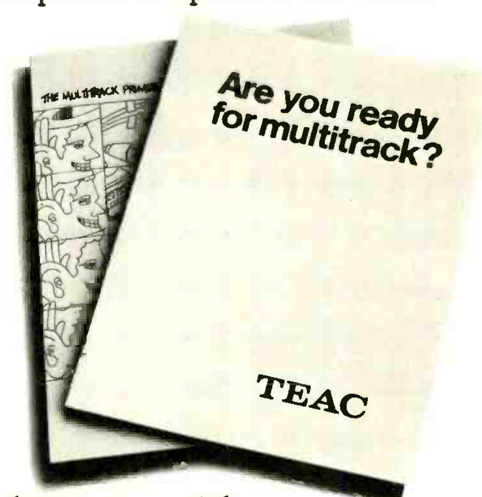
Sometimes you get exactly the sound you want. Other times, it's a bust. That's why you go through the endless hours of practice and rehearsal. And that means you need the time.

More than anything else, owning a multitrack recording rig gives you all the time you need. To practice. To make mistakes and change your mind. To experiment and develop.


The process starts with the multichannel recorder. Specifically, our A-3440—the new standard for four tracks on ¼-inch tape with sync. Rugged, reliable and very fast to operate, the A-3440 uses one button per track for Record/Playback status and dbx* Encode/Decode switching. It has a built-in 4x1 headphone mixer for selective monitoring and cueing, and a pitch control for added production flexibility.

The key to controlling your sound for recording and mixdown is the mixer. For the right balance between real multichannel recording flexibility and low cost, try our Model 2A (shown here with optional MB-20 meter bridge and sideboards). Six inputs drive four separate outputs. Each input has switchable mic/line mic attenuation (to reduce overload distortion), bass and treble controls (±12dB at 100Hz and 10kHz), color-coded channel assign buttons, pan (for stereo balance) and slide fader level control. There's a master fader for overall level control. And lots of mixdown flexibility with the Model 2A's patch points. You can hook up external equalizers (like our GE-20), reverb units, any signal processors that will help you get the results you want.

If you're just getting started, get our free 16-page introduction to multitrack recording called "Are You Ready For Multitrack?" And if you're already cutting tracks, invest $4.95** in "The Multitrack Primer," our 48-page guide to setting up and using a multitrack studio, with special emphasis on never before published ways to conquer acoustic problems typically found in the home studio. Just write us. Or better yet, pick them up at your TEAC Multitrack dealer.




Then get your hands on the tools that give you all the time you need.

*dbx is a registered trademark of dbx, Inc. The dbx unit is available optionally. **Retail prices are determined by individual TEAC Multitrack dealers.

# MULTITRACK SERIES

TEAC Professional Products Group

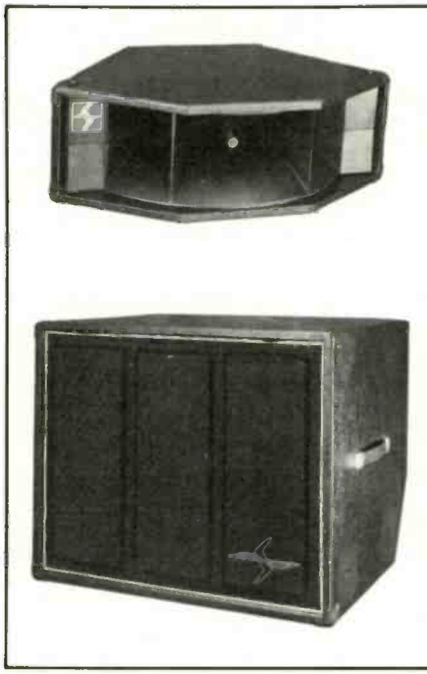
# MUSICAL NEWSICALS

## P.A. AND MONITOR SPEAKER SYSTEMS

A new name in the world of sound reinforcement speakers is Bullfrog. Bullfrog offers a full line of musical instrument, P.A. and monitor speaker systems. The full-range P.A. speakers incclude four column-type cabinets (two with piezo-electric tweeters) and various compact two-way systems based on single 12-inch, 10-inch plus 12-inch, single 15-inch or single 18-inch drivers with various high-frequency drivers. The monitor speakers include a 10-inch plus piezo system and two single 12-inch plus treble systems. All Bullfrog models include heavy wire grilles for speaker protection (heavy expanded metal grilles for monitor models), chrome plated hardware corners, a clip-on vinyl travel cover for the speaker openings, parallel-wired input jacks and T-nut mounting.

CIRCLE 17 ON READER SERVICE CARD

New from Shure Brothers, Inc. are a pair of modular loudspeaker system components designed to be used together as a versatile, compact P.A. system. The new models are the Pro-Master 707 low-frequency speaker and the Pro-Master 708 high-frequency speaker. The model 707 low-frequency unit has a single 15-inch speaker in a bass reflex enclosure. Frequency response extends from 50 Hz to 2600 Hz, and rolls off acoustically above that point. Nominal impedance is 8 ohms, and power handling is rated at 150 watts. The Model 708 high-frequency unit is a unique design with a variable dispersion horn and a high-power compression driver. The radial horn has a basic dispersion of 120° horizontal by 90° vertical, but by turning a knob in the horn's mouth the outer two sectors of the horn are blocked off and the total energy from the driver is concentrated in the center section of the horn



for a 60° horizontal coverage angle for "long-throw" applications. The 708 has a frequency response of 2 kHz to 15 kHz with a built-in filter that rolls off at 18 dB per octave below 2 kHz. Nominal impedance is 8 ohms, and power handling is 150 watts. For those who prefer a single-cabinet system, or who don't need the flexibility of a modular system, Shure offers the same components in a single cabinet as the Pro-Master Model 701.

CIRCLE 18 ON READER SERVICE CARD

McCauley Sound Co. manufactures a full range of radial horns, acoustic lenses and throat adapters for commercial sound reinforcement applications. All of their horn assemblies are made from cast aluminum coated with Aquaplast to damp out any ringing or resonances, while the lens assemblies are formed from sheet aluminum. 40°, 50° and 60° exponential and 90° radial designs are available in various models to accommodate ⅞-inch threaded drivers or 1-inch or 2-inch bolt-on drivers. Various crossover frequencies are recommended for the various models ranging from 800 Hz to 2 kHz for the six exponential models and from 375 Hz to 1.2 kHz for the six radials. Among the horn/lens systems, units are available to cover 100° horizontally with a 1.2 kHz crossover and accommodating any of the three driver mountings, or to cover 130° with an 800 Hz crossover powered by a 1-inch or 2-inch driver. Also available from McCauley are various passive frequency dividing networks (speaker crossovers) and prefabricated loudspeaker enclosures for monitor or P.A. applications.

CIRCLE 19 ON READER SERVICE CARD

Peavey Electronics has introduced two new stage monitor models, the 1245 and the 2445. Both models utilize new driver units which Peavey designed specifically for monitor speaker application. Both models are said to have excellent clarity and projection, controlled dispersion to help reduce

feedback, and high sound output. Both models are compact and feature a low profile for minimum obstruction of view, and come equipped with a flight-case-type cover to protect the drivers in transit. The 2445 is the larger of the two models and features double the low-frequency speakers for a higher maximum output level.

CIRCLE 20 ON READER SERVICE CARD

## MIXING CONSOLES

Walker Audio Visual Engineering has announced the availability of several new models of sophisticated audio mixing consoles. At the top of their line is the model 2440, a 24-input quad output mixer which is also available in a 16-input version as the model 1640. Each input channel of the 1640 or 2440 features both a transformer-balanced, low impedance (200 ohm) microphone input and a high impedance, unbalanced line input which are selected from a mic/line switch on each channel's panel. Mic connectors are 3-pin XLR type and line connectors are ¼-inch phone jacks, plus there is a stereo ¼-inch phone jack for a high impedance, high level patch point in each input channel. Each input module features a preamp gain control (0 to 46 dB of gain); a five-band equalizer section providing ±12 dB at center frequencies of 100 Hz, 300 Hz, 1 kHz, 3 kHz and 10 kHz; two post-EQ, post-fader effects sends (send 1 normally feeds the built-in spring reverb unit); two post-EQ, pre-fader monitor sends; a momentary pushbutton for a post-EQ, pre-fader send to the Cue bus; four output bus assign pushbuttons with a pan pot which pans between the even-numbered and odd-numbered busses; an LED, peak-reading level indicator with green, yellow, and red LEDs to indicate signal levels of −15, 0 and +15 dB, respectively; and a 90 mm (3½ inch) linear fader connected in a feedback configuration to provide signal level control from 15 dB of gain to 60 dB of attenuation. Two effects return modules are provided which feature the same functions as the input modules with the exception of the mic input and effects sends. These modules may be used as effects returns, line input channels with full EQ or as a submaster fed from the effects sends on the input channels. The four output modules of the Walker 1640/2440 feature an unusual design which allows considerable versatility in operation; each output module has the expected main fader and an eleven LED output level indicator which displays signal levels from −36 dB to +4 dB in 4 dB steps, but in addition has an auxiliary mixer which allows the four main output signals to be re-mixed into four independent Aux Mix outputs. The output modules also have return controls for the two effects busses and the two monitor busses, allowing those signals to be mixed into either the main output or the Aux Mix outputs depending on the position of a Main/Aux routing switch. The Walker console is built to accept optional plug-in units containing either four compressors or four graphic equalizers. The compressors feature a bypass switch, input and output level controls and LED display of the amount of compression from 2 dB to 25 dB. The equalizers are 9-band models with ±12 dB in each band, plus a switchable low-frequency roll-off with −3 dB points at 40 Hz. or 80 Hz. Equalizer center frequencies are on octave centers from 63 Hz to 16 kHz.

CIRCLE 21 ON READER SERVICE CARD

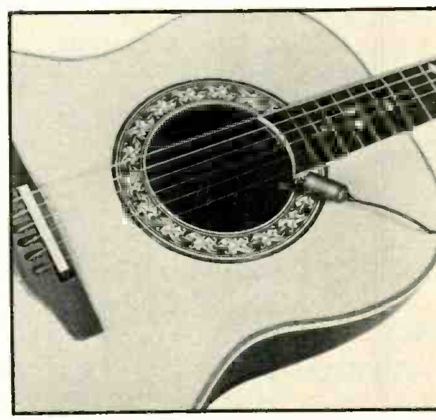
## CABLE AND CONNECTOR SYSTEMS

Anyone who has worked on the road knows what a disaster it can be when a cable breaks and you don't have a replacement connector for it. To help prevent this situation, Foxx Electronics, Inc. now offers four assortments of Switchcraft audio connectors known as "Survival Kits." Kit 1 is a "Phone Plug Assortment" featuring eighteen pieces of various styles of ¼-inch phone plugs. Kit 2 is called the "Roadie Emergency Kit" and includes cable and panel-type 3-pin microphone connectors in both "sexes," plus an assortment of ¼-inch phone connectors. Kit 3 is the "Microphone Necessity Kit" which includes a 3-pin female mic connector with built-in on/off switch, mic connector to phone connector adapters, male-to-male and female-to-female mic connector adapters and an assortment of cable- and panel-type mic connectors. Kit 4 is billed as the "Professional P.A. Kit" and includes a variety of adapters and couplers for mic-, ¼-inch phone- and phone-(RCA) type connectors as well as an assortment of each of those connector types. Each kit includes only genuine Switchcraft parts in a compartmented plastic box. Foxx also markets a full line of professional guitar and microphone cables manufactured with Belden cable and Switchcraft or ADC connectors. Models are available for guitar using either the popular blue Belden 9271 cable or the latest 100%-shielded Belden type which uses a special noise-reducing tape wrapping, and for microphones using a similar noise-reducing type or Belden Twinax cable.

CIRCLE 22 ON READER SERVICE CARD

## MICROPHONES AND ACCESSORIES

New from Shure Brothers Inc. is the SM17, a miniature dynamic microphone designed to be mounted directly on acoustic instruments. The SM17 is a lightweight omnidirectional mic

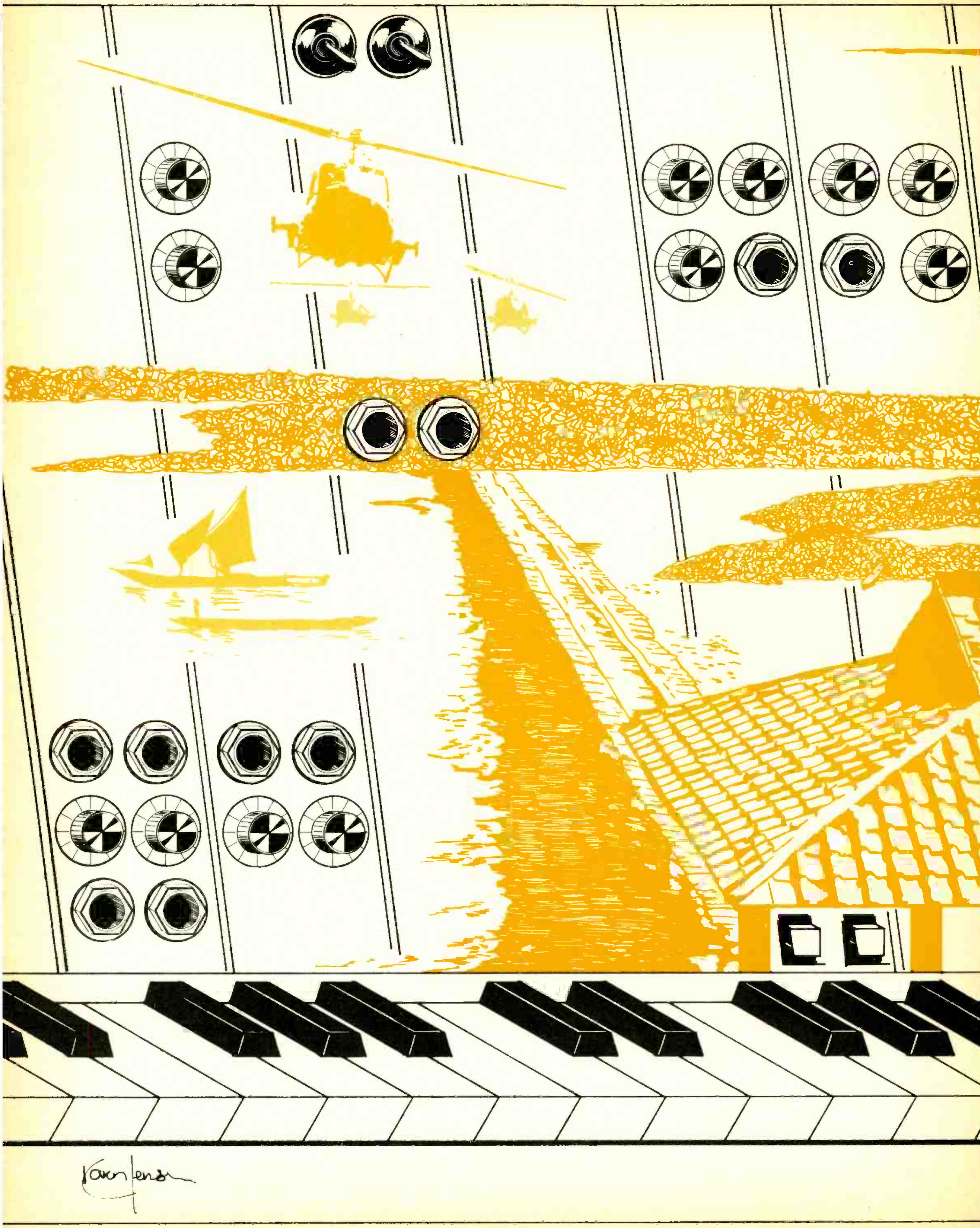


which mounts by one of two means, an expansion mount designed for the string holes of a violin, viola or cello, or a small clip which fits the sound hole of guitars and edges of various other instruments. This new Shure mic is a low impedance design and comes complete with an attached ten-foot cable.

CIRCLE 23 ON READER SERVICE CARD

Burns Audiotronics Inc., the new U.S. distributor for Beyer microphones and headphones, has announced the availability of the Beyer M400 professional microphone. The M400 is a moving coil dynamic mic with a supercardioid pick-up pattern, making it well suited for use in "live" performance since its pattern helps minimize feedback and pickup of unwanted sounds. The mic is housed in a rugged black anodized aluminum case and has a built-in blast filter. Frequency response of the mic is 50 Hz-15 kHz, and features a low-frequency roll-off to minimize proximity effect, and a gentle presence boost. The M400 has a nominal 200-ohm impedance, and uses a 3-pin XLR-type connector.

CIRCLE 24 ON READER SERVICE CARD



# Apocalypse Now:

## The Music Behind The Images

*Filmmakers generally don't hire music producers in the making of a film, but then, again, Francis Ford Coppola is not really known for doing things by the book. The producer chosen for the project was David Rubinson, owner of San Francisco's "Automatt" studio, and producer of acts including the Pointer Sisters, Herbie Hancock, and several others. His function in* Apocalypse Now *was to contract the musicians, realize the written score using multi-track technology, and collaborate with Coppola on mixing the tape in 6-channel quad and synchronizing it to the film.*

*I was interested in doing this interview because according to the press release I received prior to the interview, some of the methods used to create the sound track were considered "unprecedented," and the score was also heavily dependent on synthesized sound. Stuff like that whets my curiosity, so I went over to the Automatt one afternoon to find out what was going on. At the time of the interview, the movie was scheduled to open in a couple of weeks in New York and Los Angeles; David and his associates were working on completion of the sound track album, and everyone seemed relieved and excited that a long stretch of work was near its end.*

**By Craig Anderton**

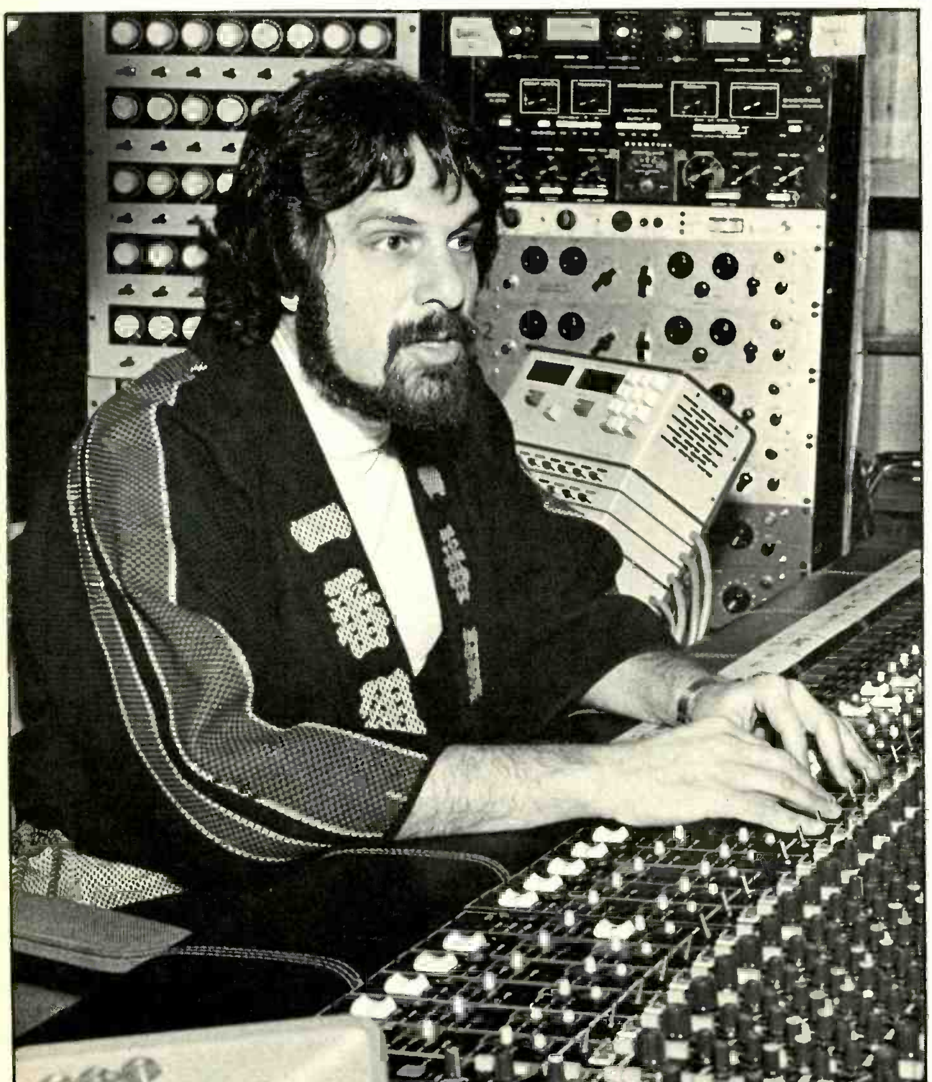
**Modern Recording:** Why is the process of making the soundtrack considered "unprecedented?" What makes it different from other film soundtracks?

**David Rubinson:** Well, the making of the film itself was unprecedented; the concepts that Francis (Ford Coppola) had in making the film were unprecedented, in the most literal sense. While I don't want to speak for him, generally when you make a war film you have the complete support of the armed forces and the Pentagon . . . they supply you with aircraft carriers, tanks, soldiers; they roll out the red carpet. But Francis was making a film that was basically not pro-Pentagon, so he didn't get support; if anything, he got interference. He had to essentially create the entire army that was in Viet Nam from scratch, including helicopters, pilots, tanks and tank drivers. So, there was an incredible amount of logistical work that had to be done for the movie, and the soundtrack was no exception.

Francis and his people decided early on that they couldn't really record the explosions, machine guns, and the like on-site; they had enough trouble just getting the dialogue recorded properly, what with all the explosions and gunshots. So for three years, he literally had a small platoon of sound effects people creating every single machine gun burst, bullet ricochet and helicopter rotor.

**MR:** Were any of these effects realized with synthesizers?

**DR:** No, they were all actual sounds that had to be synched together and woven into a cohesive tapestry. There were hundreds of tracks—and you have to understand, they really went for authenticity on all this. If the Viet Cong in a given part of the delta was using a Russian-made machine gun, the sound people went and recorded that Russian-made machine gun. All of this meant that the entire texture of the film—the sound effects—*always* came first in any discussion of the score or track. Every single sound in the 2½ hour movie, from single rifle clicks to explosions, was created totally from scratch. It took twenty to thirty people working for years.



Producer David Rubinson while at work on the soundtrack recording.

**MR:** Were you actively involved in this particular process?

**DR:** No, I wasn't; the reason why I'm bringing it all up is because Francis' design for the film was *not* to have dialogue, sound effects, narration and music as separate entities. He wanted to knit a fabric that included all of these so that, for example, the motorboat sounds became the bass lines of the music . . . then the music turned into helicopters, and the helicopters turned into a string section which developed into screaming children . . . a napalm burst became the percussion section. The idea was to create a totally integrated, totally seamless type of sound.

**MR:** So you had to work with the sound effects that existed in a very intimate way.

**DR:** Yes, the music that I was producing for the film had to be part of the sound tapestry that already existed. By the time I got involved in January (1979), there were already hours and hours of finely tuned sound. In Francis' concept, the music was merely part of the aural environment, since there were four levels of sound environment (dialogue narration, sound effects and music) going on simultaneously. Therefore, the music was the last thing to be put in . . . although it was one of the first things Francis thought about, it was the last part to be realized. It's not a soundtrack or score in the traditional sense.

**MR:** Were these raw sound effects stored on loops, or what?

**DR:** Oh yeah, hundreds . . . thousands . . . of loops. They were doing premixes the first day I walked into the studio, and had 68 channels of sound effects laid out on the floor. Each channel had maybe 75 events on it, and the interrelationship was so complicated that the track sheets were about 20 feet long and taped on the floor of the studio. Although they had an automated board, it was still a staggering job. They'd take all the effects that went with a particular scene, put them up on the dubbers, and make a sound effects premix of the scene. They'd do maybe 10 seconds, punch in on the automation track, go back, do another 10 seconds . . . they literally worked through the film inch by inch.

**"...there wasn't any of the old stereo ping-pong stuff; it's too literal and doesn't give the audience any credit at all..."**

**MR:** Why did you go for an almost totally synthesized musical approach? I understand almost all of the music was created by synthesizers.

**DR:** When Francis and I talked about how to best realize a musical score, we came to the conclusion that synthesizers were the best instruments for the job because they are the only instruments that can make the transition from total musicality, all the way to machine gun and bullet sounds, without any loss of musicality. This cue I'm mixing right now, for example, has the wind playing the melody. Then, a synthesized oboe takes over the melody line. The wind changes into an electric guitar, which then changes into motorboat whines which eventually turn into helicopter rotors, and so forth.

**MR:** It's almost as if the sound effects are various lead and rhythm musicians . . .

**DR:** Exactly. Francis' concept was that the whole movie had to be organized like a big band, and everybody got his solo. Sometimes it's a solo for an actor, sometimes a synthesized trumpet, sometimes a wind solo.

**MR:** It sounds very John Cage in nature; sound as music . . .

**DR:** *Musique concrete* is one element we're talking about. But the other fascinating thing is that Francis has an encyclopedic knowledge of classical music. As you may know, his father Carmine was Toscanini's flute player and played in the N.Y. Philharmonic for 18 years; he composed the actual score for the film that was later synthesized. We had T-shirts made up that said "Verdibuspucciwag," which stood for Verdi, Puccini, Debussy and Wagner—the main sources of inspiration for the music. It was very interesting to be dealing with synthesizers and insect sounds, but relating to them in terms of classical music. To deal with the concept, we had to evolve some different technologies.

**MR:** What were some of these?

**DR:** Well, there was one major problem we had to solve. Film sound and synching concepts used today are very obsolete and plodding. The whole sync tone, synching machines together with common tones and waiting till they get into synch . . . it's all ludicrous. Just trying to synch eight machines . . . well, you can see for yourself what's going on here, we have five machines (two 24 tracks and three 2 tracks) just to make the album. If you go over to the mix studio where they're doing the final dialogue and so forth, they have maybe ten 6-track dubbers and two 24-track machines synched together with sync tones; it's really off the wall, very obsolete in concept. It's like trying to run faster by building bigger shoes instead of inventing the wheel.

We therefore had to develop a way to have instant creative response to the visual elements of the picture. We didn't want something where you'd roll back a bunch of dubbers and sit there with cue sheets . . .

**MR:** Before going any further, how does traditional scoring work?

**DR:** Basically, let's say you have a scene that's 60 seconds long, and at the 35th second some guy gets hit on the head with a bottle—so the producer wants a "crash" at that point in the score. So you write music that's 60 seconds long, and then notate that at bar 17, for example, you have to hit the tympani. You figure out the tempo of the music you're going to do, and put clicks on the film that you're scoring to that represent eighth notes, quarter notes, or whatever. You then measure how many clicks it takes to cover 35 seconds, and right *there* on that beat the crash occurs. So you end up writing little cues all over the score, like "this is where the door closes," "the 3rd beat of bar 33 is where the wife makes a face," and so on. It's a very mechanical, very mathematical type of thing. The musicians come in and do the score while watching the picture, but they basically play to the conductor more than anything else.

However, here we weren't dealing with conducting a group of musicians, but instead the process of building up each sound a track at a time using individual synthesizer players. So if the score called for a string section, we'd have to build it up track by track. Another problem was that most of the music was rubato (deviating from a strict tempo), and not conceived at all with a rigid mechanical tempo. Compounding the problem, how do you get someone like Francis—who is not a musician per se—to convey his musical concepts to the musicians? The whole movie hinged on Francis, and his input was absolutely mandatory.

The method we worked out goes back to the silent film days when the piano player would sit and play along with the film. What we did was transfer the entire movie, cue by cue, to U-matic video tape, with sync tone and dialogue on separate tracks of the U-matic. We then transferred the dialogue and sync tone to 24 track audio tape; Francis would sit with his father Carmine, me and Shirley Walker (the piano player) and describe exactly what he wanted. Carmine would go home and write out the score note for note; we'd go back in the studio with the notes, and Shirley would play the piano while Francis made whatever changes he wanted. Then we'd project the videotape, roll the 24 track (which was in sync), and the pianist would record onto the 24 track—exactly as Francis wanted it—right along with the picture. We called this an "expression guide track," since it expressed exactly what Francis wanted. Normally you go into a studio and there are 68 guys you're trying to conduct, and they can only relate to what's written on a piece of paper. This way there was one person, Shirley Walker, and all the music was expressed through her piano playing instead of being written out on paper in an antiseptic fashion. We ended up with a 24 track tape, and on the tape would be sync tones so it matched the video, dialogue and the piano track. We could then give the score to a player, and as they were supposed to realize the music, they always had the piano track as a reference so that they knew exactly where the notes were supposed to fall, and what kind of feel they were supposed to have.

We also went a step further. Shirley used a Yamaha CS-80 (polyphonic synthesizer) to make basic sound patches; when Francis wanted French horn sounds, she'd patch up a French horn sound. So, this track (we called it a "slop track") indicated the sonorities and timbres to be used, and gave the synthesizer players a really good idea of what was needed. This constant communication helped make things more efficient; I mean, what happens when you have 130 people sitting in a studio getting scale, and the director comes in and says, "Gee, that's not quite what I had in mind"?

**MR:** What could have helped the process along, in terms of technology, that doesn't yet exist?

**DR:** Instant communication between all the people involved would have helped tremendously. Trying to coordinate all these people over regular phone lines was just too unwieldy.

I would also like to have had the ability to store the picture medium of the film on the audio tape. If we could have stored the visual elements by some means on the audio tape, maybe by adding a different head or something, it would have saved an enormous amount of time. Hopefully in the future, digitally recorded tape will contain the picture, narration, instruments, everything.

No matter how you look at it, trying to sit and mix music to a picture when everytime you make a mistake, you have to stop and rewind eighteen machines . . . and some will be late, and some will be out of sync, or someone will hit fast forward instead of play . . . it's insanity.

**MR:** This is obviously a pretty dense soundtrack in terms of texture. How do you go about maintaining the quality of that sound as the movie goes from theatre to theatre?

**DR:** Omni Zoetrope (Coppola's production company) has a staff of technical people; their job is to outfit the theatres and make sure that the quality control is up to spec. For example, they've been putting subwoofers in every theatre for extreme low frequency reproduction. The soundtrack is 6-track quad . . .

**MR:** What do you mean by "6-track quad"?

**DR:** Left, right, left rear, right rear, hard center and low frequency subwoofers. As you know, there are no tone tapes at theatres, no tweaking or calibrating the playback response; it takes quite a bit of work to get it right.

**MR:** Are you using noise reduction on the film?

**DR:** Yes, Dolby and dbx used throughout—Dolby in the theatres; it is a good system for movies.

**MR:** Did you get into any innovative location modulation techniques (panning, etc.) or devise any new gadgets to do this more efficiently?

**DR:** One new development was a five-position, diamond-shaped quad panner, because you had to assign some sounds to center. But really, we wanted to avoid gimmicks because the texture of the sound is so dense—gimmickry had to be at a minimum.

When you sit in a theatre, true quad is a very elitist *(sic)* thing; you have to sit in just the right spot or it sounds terrible. Stereo is not that critical, at least not like quad. If you're off to the right side of the back of the theatre, with quad you won't hear the sound you're supposed to be hearing. You can't have a situation where you have 1000 people in a theatre, and only those in the 10th row in the exact middle hear things right. So it had to be a democratic mix, you couldn't put something in the right rear just because it was a groovy effect.

**MR:** So you mixed it as a large space with variations . . .

**DR:** That's right, no pinpoint quad stuff; although of course we used the potential of spatial location as much as possible. One example is a scene where Bill Graham plays a rock promoter, yelling at a stadium of GIs. He comes out and faces the audience, so you see him from the audience's point of view. So he says, "Ladies and gentlemen" and the echo comes out from behind you. Then the camera angle reverses and you see the audience from behind the promoter. So, the echo had to come from the front.

But there wasn't any of the old stereo ping-pong stuff; it's too literal and doesn't give the audience any credit at all . . . it's stupid. I wish, though, I had had more time to do phase location things where you can have sounds coming out from behind people's heads. I've done a lot of that. The problem is that if the person's playback system is not in phase, the sound disappears. Location of sounds, though, was not so much a spatial problem; rather, it was a depth problem. We had to do things like separate the battle sounds from the narration without having the narration being yelled . . . it wouldn't make any sense.

**MR:** *Apocalypse Now* had already received a great deal of publicity, and was eliciting certain expectations, since before you were involved. Did you come in as a troubleshooter because of the difficulty in getting the movie completed?

**DR:** (Pause)

**MR:** Is that a clear enough question?

**DR:** Oh yeah, it's a clear enough question . . . I'm debating the political nature of the answer. The truth of the matter is that years ago Francis had a concept of the music which he had wanted realized. He employed some people to give that to him, who in the end did not deliver what he wanted. He got down to January of '79—the picture was supposed to come out in May—and he had no music that he could use. He needed a producer, someone who would just get him what he wanted without any problems. Basically, I was not brought in as a troubleshooter, but as someone to get him what he wanted. There were tremendous budgetary considerations, since millions had already been spent, but the money spent on the music was astronomical, nonetheless . . . .

But back to your question on expectations. What I want to make clear to you, though, is that it wasn't the money or expectations that got me involved; it was the opportunity to work on the music with Francis.

**MR:** So for you it didn't amount to just another job . . .

**DR:** Certainly not, I could make more with one album than I could with three *Apocalypse Nows*; I had to stop work on other projects for four months.

**MR:** Was it the challenge that interested you, or the politics?

**DR:** It had nothing to do with either. I wanted to be of service to Francis . . . and to learn too . . . to get involved technologically was a challenge, but the motivation was to work with that man, on his project, and deliver.

**MR:** Did you attempt any experiments that just plain failed?

**DR:** Sure . . . Jesus, look at those reels of tape *[gestures to boxes of tape on the floor, more than enough for several movies].*

**MR:** What was the biggest problem you couldn't anticipate?

**DR:** The biggest problem—and we're

still trying to cope with it—is that your appetite for expression and perfection exceeds the requirements of what's really needed for the music in a given situation. Francis made so many demands on the music, that he would often hear the music and demand more—more expression, more feel, the ability to fulfill more things. So we'd do a 7 or 8 minute cue that was perfectly wrought, I mean, every little detail was realized and over-realized but without consideration for how it would fit into the whole puzzle. So a big problem was in lowering expectations of the music itself, and trying to keep perspective of where the music would fit in the final product. Francis would get carried away, I'd get carried away . . . but gunshots had to be heard, and *Apocalypse Now* is not a musical. Some pieces would sound great by themselves, but when you heard them with the dialogue and everything else you'd have to take them down, and take them down some more, and sometimes even throw some things out entirely. Maybe it was great music, but it didn't fit into the movie.

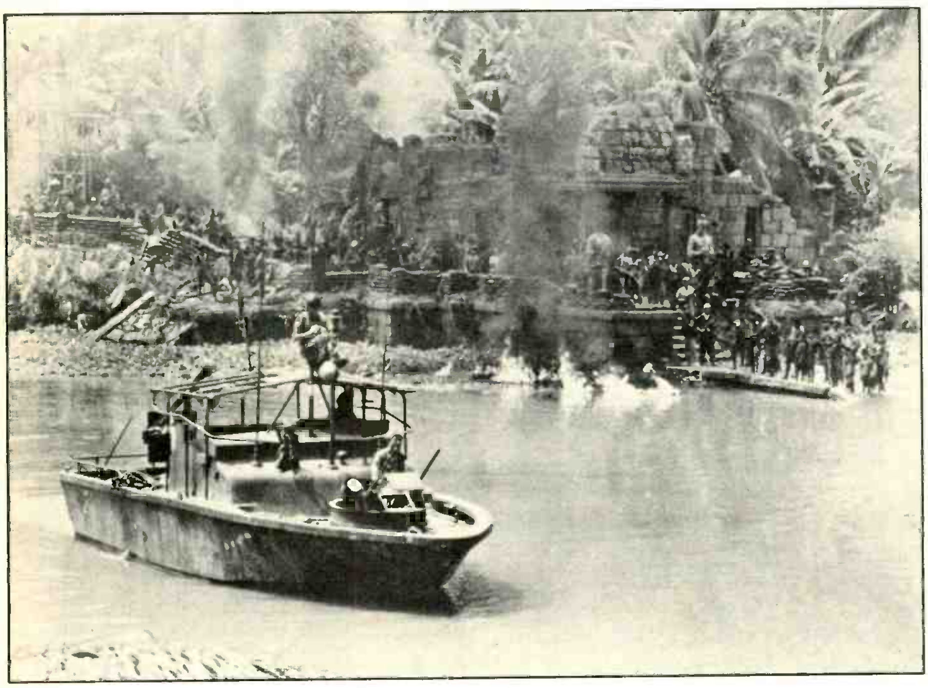
**MR**: It sounds as if you needed a special breed of studio musician . . .

**DR**: Well, you really needed a special breed of people for anyone working on the movie, and that included the musicians. They had to read music fluently, but also had to relate conceptually and emotionally to the content of what was on the screen.

**MR**: Was it your duty to contract the players?

**DR**: Yes. For example, the whole last 18 minutes of music was going to be a vocal choir—70 voices, singing Carmine's score, in 12 tone *a capella* with 16-part harmonies. That takes really fine singing. This wasn't rock, or jazz, or disco, for me; I had to reach back to a lot of abilities I hadn't used in a long time. It was quite a challenge.

**MR**: Tell me something about the synthesizer players.



**DR**: Pat Gleeson was terrific. We've worked together a lot; we go way back. He was the first synthesist I chose. He's a fine synthesizer player, reads music brilliantly and is very creative. But he particularly knows how to get a good sounding recording of a synthesizer.

Nyle Steiner [of Steiner-Parker synthesizers fame] was recommended by Shirley Walker and Pat. He [Steiner] had come up with this incredible electronic wind instrument, the EVI (Electronic Valve Instrument). It is a truly expressive solo instrument, but still has that synthesized sound. Nyle played most of the solos in the picture, playing cellos, wind, helicopters, French horn. The expressiveness of the EVI was absolutely essential to the film. I got an EVI and it's terrific, unbelievable. If Miles Davis could see that instrument, he'd have a whole new career.

**MR:** I understand you used Randy Hansen for guitar . . .

**DR:** I brought him into the project because the prototypical music being played in Vietnam at the time was Jimi Hendrix. Francis wanted the components of the score to reflect the sound environment that was in the trenches at the time, but we didn't get Randy just because he can get Jimi Hendrix sounds. He conceives of his guitar in a very synthesizer way . . . flangers, and lots of special effects. He provided sound effects as well as playing. Francis asked him to do a motorboat sound that sounded like "impending doom," and you don't just call up the sound effects lab and ask for "one motorboat with impending doom."

**MR:** What about the other synthesists, Don Preston and Bernie Krause?

**DR:** Really, all the musicians were great. There were some sessions that were 3 AM to 3 PM with Don, and then Nyle would come in at 3 PM, while in another studio I was working with Randy Hansen, and Pat would be putting in an 18 hour day elsewhere.

One conclusion you can draw from all this is that there are as many different styles of synthesizer playing as there are of saxophone or guitar playing. Most of the cues had everybody playing, but it doesn't sound like sixteen different styles. My hardest job in terms of mixing the music was to make it all have a wholeness, to sound like it was all made in the same room at the same time. But in fact, it was done by seven different people in eight different studios on fifteen different kinds of synthesizers.

**MR:** Are you satisfied with the emotional content of the sound track as a piece of music, divorced from the technology or the tricks that went into making it?

**DR:** Yes, and you just hit the nail on the head. When you have so much technical capability, it's hard *not* to use the technology as a crutch and instead of making it work for you, becoming a slave to it. That's probably the dilemma of the modern world.

**MR:** Expressed in another way, do you believe something just because it comes out of a computer . . .

**DR:** Or, just because there's no hiss, no wow, and no flutter, does that make it good music? If it's perfect, does *that* make for good music? The genius of the synthesizer players in this picture was that they had to make their synthesizers expressive. The hardest thing to make a synthesizer do is create an expression, something with emotional content. Touch sensitive keyboards notwithstanding, whether you play the keyboard hard or soft everything comes out the same; it's like a harpsichord. How to get a synthesizer to cry, or weep, or get angry . . . expression . . . it's very difficult. In fact, the essential problem was while we were involved in all this technology, chips and PCMs and sync and all that crap, we could not become enslaved to it. The music had to come out human and warm; it was really hard to add human elements to the synthesizer.

**MR:** Is there going to be a conventional soundtrack album (just music), or are you going to be including parts of the sound effects or what?

**DR:** The soundtrack will be a double album on Elektra records. The concept of the album is to re-create the experience of the movie, sound effects, dialogue, narration and music.

**MR:** Was it disorienting to be working on a movie instead of an album?

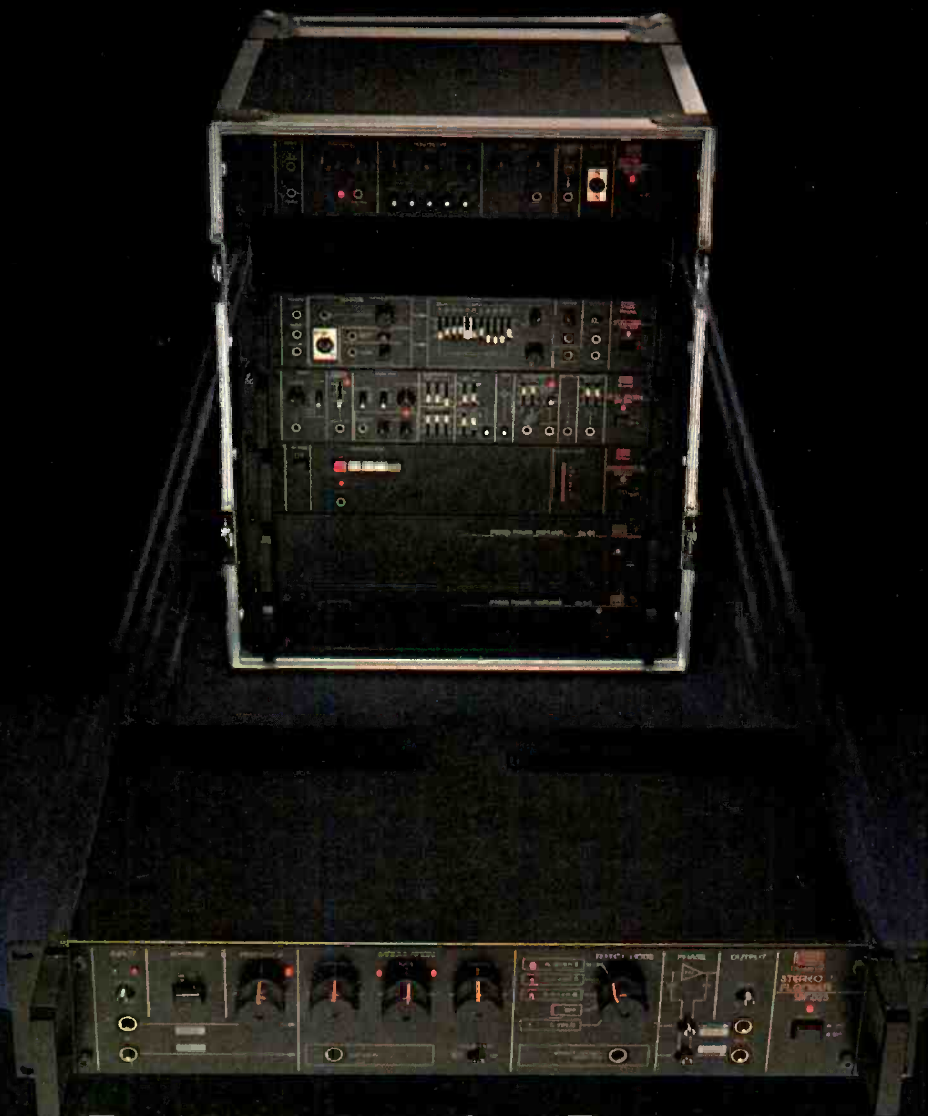
**DR:** Not at all. In either case, you strive for a balance of business-like attitude and creative sensitivity. The disorienting thing to me was not the difference between making a film or an album, the difference was that I'm normally in Francis' position. If I produce an album, I say, "Get me this," or "Do that," or whatever. I get what *I* want. It was a very important exercise to take on this [different] persona, and to discipline myself to work for a "boss." I had to remember it was his picture, and that I was just one of the people working on it. This time, I was part of a team. It was an interesting experience, something I hadn't done in a long time.

**MR:** Was working on *Apocalypse Now* frustrating?

**DR:** Murderous. This was the hardest project I've ever worked on.

**MR:** Would you do it again?

**DR:** Absolutely.



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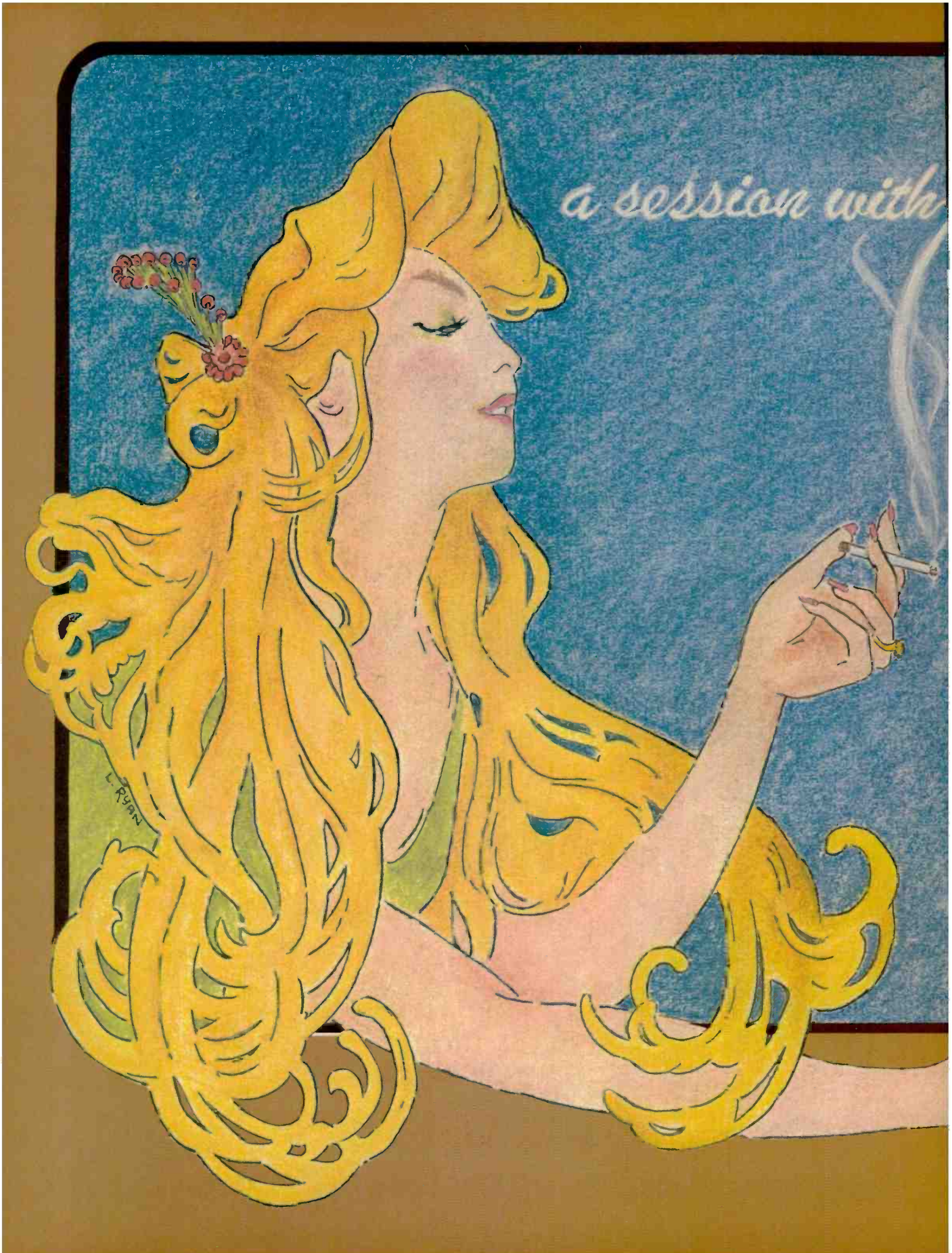
a session with
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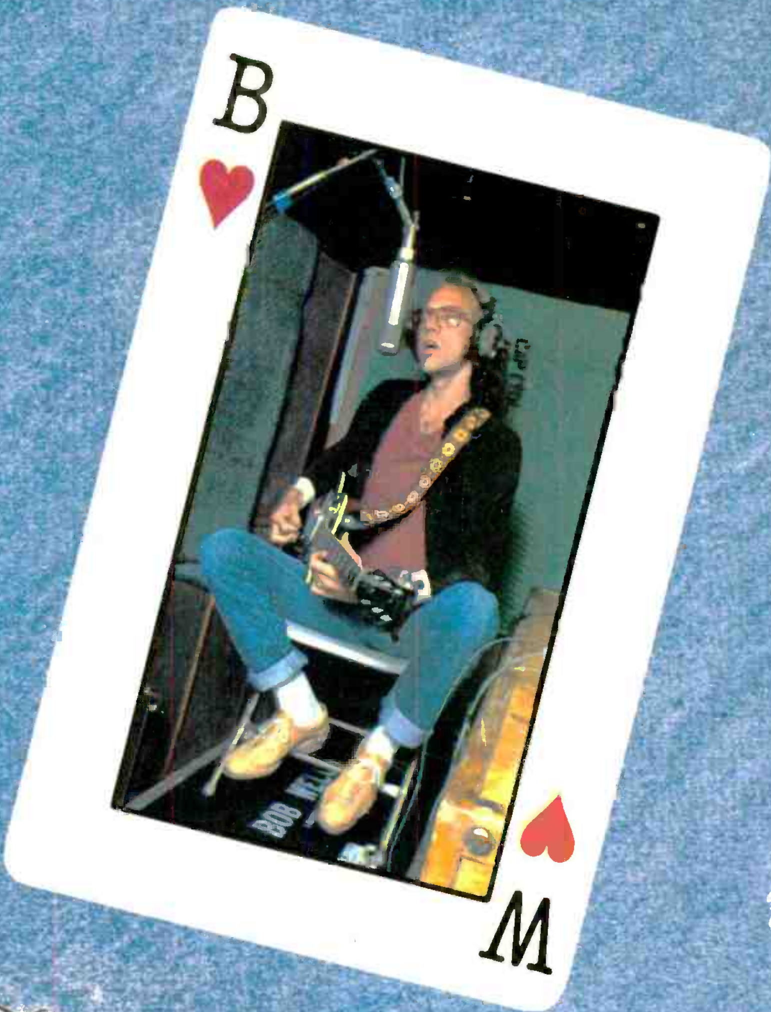
# Bob Welch




*By Rob Lewis*


Since the 1977 success of his first solo single "Sentimental Lady," ex-Fleetwood Mac guitarist Bob Welch has released two best-selling LPs, enjoyed a handful of AM and AOR hits and carved a secure niche in the world of mainstream pop/rock. His intelligent yet accessible songwriting, hook-filled arrangements and sophisticated delivery have made him a favorite to an enviably wide and diverse spectrum of fans.

Against this background, Welch and his entourage moved into Capitol Records' Studio B in Hollywood to record what it was hoped would be his third platinum album in a row. Central to the project besides Welch were producer John Carter (his second Welch LP), road engineer Randy Ezratty (assisted by David Cole and John Stream of the studio engineering staff) and Welch's touring band.

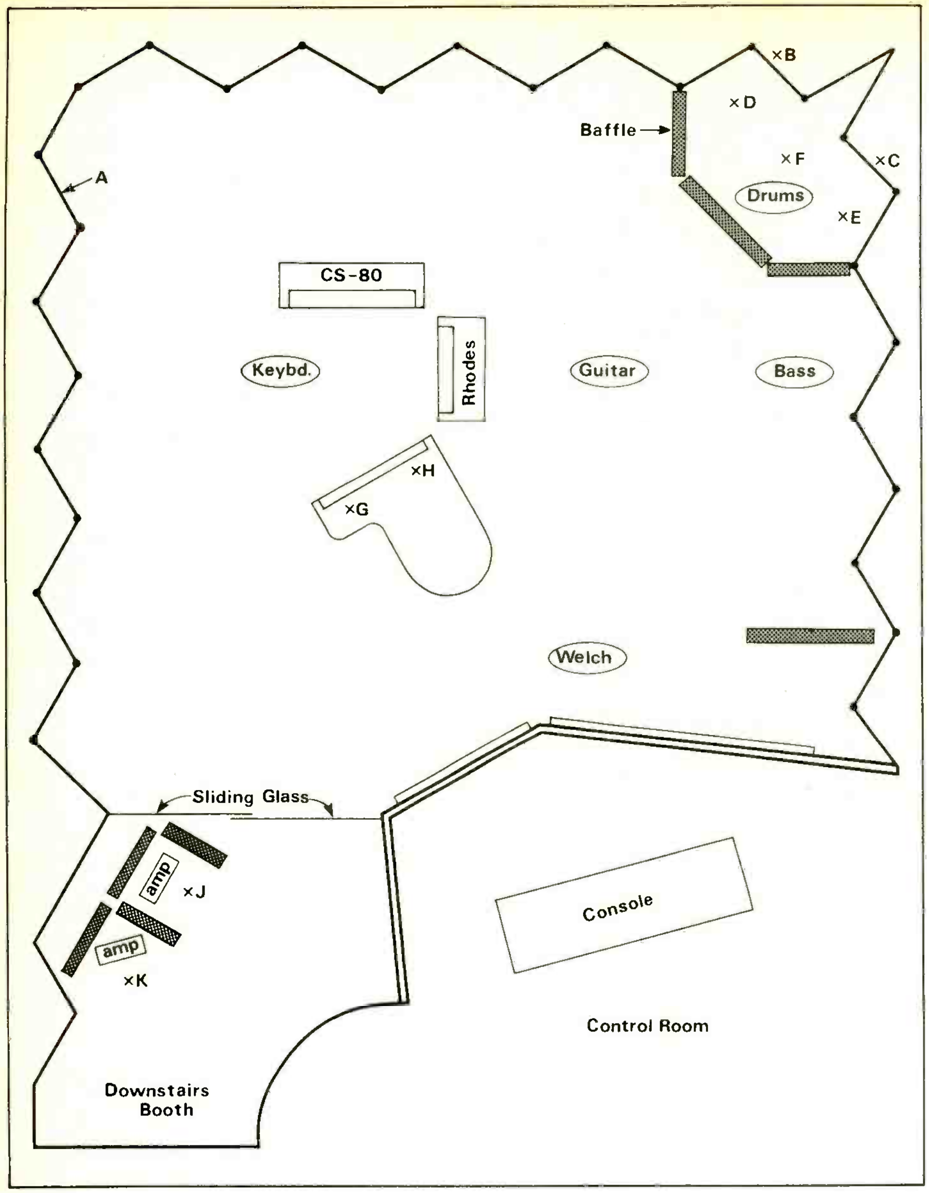



**Fig. 1: Studio diagram; Capitol Recording Studios studio B.**

Besides the advantages of recording on "home territory" (Capitol is Welch's label), there were several other factors that led to the choice of studios. According to engineer Ezratty, the group was after a natural, "live" sound, with lots of natural room ambience rather than artificial echo (as opposed to the previous album, *Three Hearts*, which Welch describes as "a very tight, midrange-y, 'radio' record"). The availability of Capitol's legendary basement echo chambers (about which more later) was an extremely strong attraction in this regard, as were the acoustics of the recording room. Studio B is larger than most typical rock studios, and posseses a ceiling over 20 ft. high. In addition, three of the room's walls are fitted with adjustable sound baffles *(see Fig. 1)*. These baffles are hung like doors on hinges in a zigzag "accordion" pattern. One side of each baffle is flat hardwood, while the other side is upholstered with sound absorbent material. By adjusting the positions of the baffles, the acoustics of the room can be varied from very "live" to quite dead, or anywhere in between. If desired, parts of the room can be made more "live" than others.

For the Welch sessions, most of the baffles were set with the reflective wood side out, to create an unusually "live" room sound. (Note that if the walls had been simple flat panels, severe standing wave problems would have resulted; the zigzag pattern of angled reflecting surfaces helps to break up standing waves and produce a smoother, "incoherent" reverberation characteristic.

To take advantage of this environment, Welch's engineers were experimenting with two new pressure zone microphones for ambience pickup. These specially designed microphones are intended to be placed against a hard, flat surface (wall, floor, baffle, etc.) near the sound source. This placement prevents the common problem of severe frequency response aberrations caused by phase cancellation between the directly arriving sound and the slightly delayed sound reflected from nearby surfaces. In the Welch setup, the pressure zone microphones were attached to two adjoining walls of the studio, above the drum set, about two-thirds of the way up the wall. In this position, they not only picked up room ambience, but captured a supplemental stereo image of the drums, which could be blended with the conventional drum mix for additional depth.

Besides the favorable acoustics, Capitol's array of hardware and facilities offered additional attractions. In the studio's inventory is a great deal of older tube-type equipment, including condenser microphones (principally Neumann U-47s and U-67s, renowned for their mellow sound with vocals and guitars), and a "workhorse" limiter, the Teletronix LA-2A, equally renowned for its unobtrusive action and pleasing sound. When this highly prized, vintage equipment is teamed up with Capitol's modern, state-of-the-art hardware such as the Neve NECAM automated console, digital delay systems, etc., the result is unusual flexibility of sound.

Finally, one other facility at Capitol provided an additional incentive to record there. A few feet down the hall from Studio B are two fully equipped

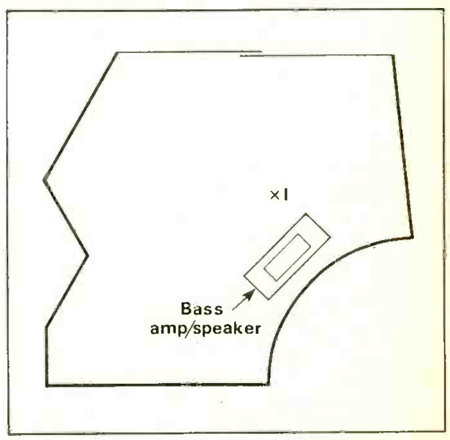



**Fig. 1a: Upstairs isolation booth (directly above downstairs booth).**

**"...the faders automatically move up and down during the mix—unaided by human hands..."**

cutting rooms, currently busy almost around the clock turning out master lacquers both for Capitol and for outside clients. With cutting facilities so close, it is a simple matter to cut a reference lacquer from a two-channel rough mix and play it back for a final check on how the disc will sound.

The studio setup for recording basic tracks is diagrammed in *Fig. 1*. Of the seven instruments in the studio, only the purely acoustical ones—drums and grand piano—were actually miked; everything else was either taken direct, or miked remotely via speakers in isolation booths, or both. An instrument-by-instrument rundown follows:

*Drums*—Drummer Don Francisco's kit was captured with a more or less conventional mic setup. The possible exception was the kick drum. While most engineers seem to prefer the tendency of a dynamic microphone to produce a slightly "rounded," compressed kick drum sound, Ezratty opted for a Neumann U-47 tube condenser mic mounted about a foot outside the drum shell, with its output padded 20 dB and routed through a Teletronix limiter.

Snare was taken with an AKG 414 condenser mic, and a pair of the same microphones set left and right served for overhead cymbal pickup, supplemented by a Neumann SM 69 stereo microphone suspended directly over the center of the drums. The hi-hat was picked up by an AKG 452 condenser mic, and both ride and floor toms were close-miked with Shure SM-57s. To minimize leakage and phasing problems with multi-mic setups, the tom mics were patched through Kepex units, adjusted as "gates." Thus, the mics were effectively off until a tom was struck, when the Kepex opened up to let the signal on tape. Additional processing with Urei 1176 limiters was added for a tighter tom sound.

*Keyboards*—Keyboardist David Adelstein's instruments during the recording of basics included acoustic grand piano, Yamaha CS-80 polyphonic synthesizer and Fender Rhodes electric piano. Everything was taken by direct electrical feed except the piano, which was picked up in stereo by a pair of AKG 414 condenser microphones positioned over the bass and treble strings. The Fender Rhodes was recorded both in mono and stereo, using its built-in stereo outputs.

(Left to right) Bob Welch, producer John Carter and engineer Randy Ezratty.

Keyboard parts were overdubbed later using the Yamaha synthesizer (at times run through a Boss Chorus effects unit) as well as the new Oberheim OB-X synthesizer. Besides its built-in, pushbutton selectable voices, the Oberheim allows a number of user-programmed patches to be "dumped" onto cassette tape, then read back into the machine at any time, saving the task of reprogramming.

*Bass*—Bassist Brad Palmer used a combination of direct and amp/speaker pickup for his Fender Precision. The direct feed was taken immediately at the instrument output, using an active direct box to prevent signal loss. The bass signal was split in the box, with one feed going directly to the recording console, and the other to an Alembic F2B instrument preamp (a modern tube unit designed to serve as a "front end" for any power amplifier).

From the Alembic preamp, the line-level signal was routed to the upstairs isolation booth, containing a Norlin 313A solid-state guitar amplifier head and a Flag Systems speaker cabinet. The Norlin was used as a power amplifier only, with its own front end functions replaced by the Alembic's. The sealed speaker cabinet contained two 12-inch and two 10-inch speakers, and was miked with a U-67 placed on-axis about two feet in front of the cabinet.

The bass signal used for the record mix was primarily the direct pickup track; the miked speaker sound was recorded on a separate track, where it could be mixed out-of-phase with the direct signal for additional tonal effects, using the phase cancellation and distortion effects of the preamp/amp/speaker/mic chain.

*Guitar*—Welch's guitar for these sessions was a new Firebird-style Gibson with two pickups and built-in active preamp. The tracks (primarily "scratch," to be re-done later), were taken directly from the instrument's built-in electronics.

Guitarist Todd Sharp was playing a hollow-body Epiphone for the sessions *MR* attended, but can draw on a collection of other instruments depending on the job at hand. His guitar was fitted with a low impedance output to match the long signal cord to the amplifiers, which were set up in the downstairs isolation booth. In the booth, the signal was split and sent to two Fender Twin Reverb amplifier chassis, mounted in custom speaker enclosures. Both had sealed backs, with one housing a single 15-inch Altec guitar speaker, and the other two 12-inch Altecs. The cabinets were partially isolated with baffles, and separately miked with U-67 Neumanns; however, the incomplete baffling and small, irregularly-shaped booth with its hard, reflective walls contributed to a substantial amount of blend between the two signals.

Tone modification capabilities were provided in several ways: a Boss Chorus unit with stereo outputs was available to drive the two amplifiers for a "Leslie" sound, along with an Echoplex for repeat echo. When more distortion than the Fenders naturally provided was desired, an Alembic preamp could be patched in between the guitar and the amps to overdrive the amplifier front ends.

With this setup, the only instruments audible in the studio were the drums and acoustic piano, so the cue system had to be depended on for the musicians to hear each other. Since different members of the band have different preferences as to headphones, levels, etc., they brought a custom cue setup into the studio, consisting of Yamaha power amplifiers driving whatever headphone setup each player opted for (including AKG open-air sets, Stax electrostatics and Sony dynamics, among others). Four separate cue mixes from the console were used to suit each band member's needs.

In the control room itself, the focal point was the Neve NECAM automated mixing console, one of the first such units to be imported from England. The automation system, controlled by a minicomputer with "floppy disc" data storage, memorized the positions of all 32 channel faders during a mixdown, and can duplicate them at will, while allowing the engineer to make adjustments of one or more faders to "update" the mix on successive passes. Unlike other console automation systems, the mixer can actually watch the faders automatically move up and down during the mix—unaided by human hands!

Disregarding the automation, the Neve is a more or less conventional 32-in, 32-out console with 3-band equalization on each channel, sixteen mixing bus assigns on each module and a total of eight auxiliary sends, which may be used for cue mixes, echo sends, or any other purpose.

Besides the above-mentioned limiters and digital delay units, outboard equipment includes several Urei and Inovonics compressor/limiters, three ADR F300 expander/gates and a battery of equalizers including 3-band fixed Q peak/dip equalizers and 3-band fully parametric equalizers from ADR, a Urei 1/3 octave graphic and two ITI parametrics.

Also among the "outboards" (though not in the conventional sense) must be included Capitol's "live" echo chambers—a group of eight specially designed concrete rooms located in the bowels of the Capitol tower, and perhaps the most famous echo facilities in the world of recording. In fact, the sound of the Capitol chambers is considered so desirable that other studios frequently transmit echo send and return signals over specially conditioned telephone lines from studio to studio, just to get the "live" echo sound that can't be duplicated by artificial means.

With minor differences, the rooms are shaped as shown in *Fig. 2*. As shown in the diagram, the chambers are completely devoid of parallel walls to avoid standing wave resonances, and each is furnished with its own combination of speakers and microphones. The chamber used by the Welch group was equipped with two opposed-firing JBL cabinets containing 15-inch woofers in horn/port loaded enclosures, with LE85 drivers and acoustic lenses for high frequencies. Signals from the echo send bus are reproduced through these speakers, and picked up by a pair of floor-mounted Altec 21D condenser microphones with switchable 120 Hz low-cut filters to eliminate "boominess." To demonstrate the resulting effect, Ezratty soloed the echo return during a take; the pure reverb signal coming back in stereo from the chamber microphones had a depth and naturalness that the writer has never heard from an artificial echo device.

The sessions were recorded at 30 ips on a 3M 24-track machine, using Ampex 456 tape without noise reduction. Mixdown was to a 2-track MCI deck. The "house" monitor speakers were custom biamplified units designed by George Augsberger, utilizing twin JBL 15-inch low frequency drivers, crossing over to

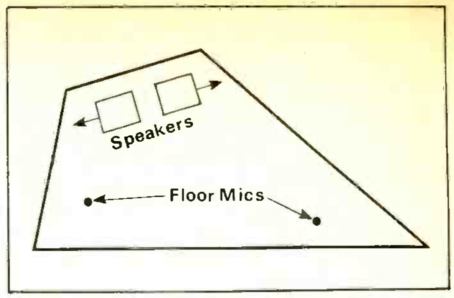



**Fig. 2: Basement echo chambers.**

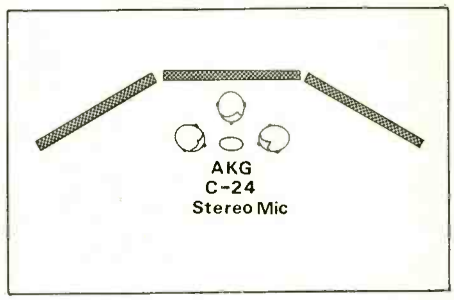



**Fig. 3: Stereo recording of background vocals.**

special wooden wide-dispersion horns energized by Pioneer drivers. The monitor system was tuned with Klark-Teknik 1/3 octave equalizers. In addition, the Welch group brought with them a pair of small Yamaha 2-way home speakers and a pair of British-made Rogers BBC monitors for a further check on the sound.

Welch's lead vocals were recorded primarily with a vintage AKG C-12 condenser microphone, while a somewhat unique setup was used for background vocals. As shown in *Fig. 3*, the three singers were grouped around a single AKG C-24 microphone, which is the stereo version of the C-12. The mic's two outputs were recorded on separate tape tracks, then the whole process was doubled. Heard over the monitors, the six voices spread across the stereo image gave an unusually natural, pleasing effect—one which Ezratty and Cole felt was well worth the extra tape tracks required. In the final mix, a bit of left/right blending will be used to narrow the vocal image slightly if full separation should tend to produce any unnatural, "bigger than life" effects.

In sum, the Welch sessions provided an interesting combination of old (vintage tube equipment and acoustic echo chambers) and new (computerized mixdown and digital synthesizers), proving, one supposes, that if yesterday's audio engineers didn't have all the answers, they at least had plenty that was worthwhile to show us.

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PROFILE:

# EDDIE KRAMER

BY JOEL SIEGEL

*What do the Rolling Stones, the Beatles, Led Zeppelin, Jimi Hendrix, Kiss, the Animals, Traffic, Joe Cocker and Carly Simon have in common? At one time or another they all utilized the engineering/production skills of one person by the name of Eddie Kramer.*

*At age 36, Eddie Kramer is a legend in the recording business because of his rare ability to translate the designs of rock music's greatest artists onto tape. Most noteworthy, Eddie recorded the lion's share of Jimi Hendrix's work, which in itself is a major section of the guitarist/record maker's bible.*

*Eddie was born in South Africa to parents of Russian and English descent. During his childhood, Eddie often traveled with his family to England. He recalls: "I had a classical [music] upbringing from the age of six to about eighteen. I went to the South African Conservatory of Music and was going to be a classical pianist."*

*One can safely surmise that somewhere along the way Kramer deviated from the classical path. During the* Modern Recording *interview, Eddie spoke of how he was attracted to the jazz music that was happening in South Africa in the late 50s (e.g., Dollar Brand and Hugh Masakela). Although interracial socializing was prohibited, Eddie made his way to the few illegally run jazz clubs of Cape Town to hear the music that had captured his imagination.*

*Like the black musicians he so enjoyed, Eddie felt the need to leave his country which he calls "the most beautiful in the world" to pursue an artistic/financial future. He landed in England in 1960 and got his first job in a recording studio in 1962 at London's Advision Studios. The next year he moved over to Pye Studios. At Pye, he worked on sessions with artists such as the Kinks, the Searchers and Petula Clark.*

*(He recalls an audition at Pye Studios at which the Beatles were rejected for a recording contract.)*

*Around 1966-67, and a few studio jobs later, Eddie Kramer began working at Olympic Studios which had recently been built. At Olympic, Eddie engineered for the Beatles, the Rolling Stones, the Animals and Traffic. In addition, he did films, commercials and classical work. It was at Olympic that Eddie met Jimi and a very special relationship began.*

*In 1968, Eddie moved to New York and began work at the newly-opened Record Plant studio. Not long afterwards, he decided to go independent as an engineer/producer. When Jimi Hendrix decided to build his own studio in Greenwich Village, Eddie was asked to help design it. Electric Lady Studios remains today one of the major studios in the music recording business.*

*Many gold and platinum albums down the line, Eddie decided to "cool out," and watch the business for a while. He has recently resurfaced to do a number of albums. Kramer is above all a gentleman and a man of taste. This writer was fortunate to have tracked him down for an interview.*

---

**Modern Recording:** Tell us about your latest projects.

**Eddie Kramer:** I am in the midst of putting together a bunch of new projects. I specifically took some time off to see what was happening and to see where the music scene was going, and also to clear my head out because I had been going at it for three or four years, pretty much without a break. I just wanted to relax and take it easy for a while, and get into my family.

I think the New Wave thing is extremely important, and I've found two bands that . . . I wouldn't really say are New Wave bands, but they are certainly in the vanguard of musical development. Actually, one of them you could call New Wave, and the other is sort of a very different, original sounding band, which is very hard to find these days. The other two projects I'm doing are established rock musicians, one of whom is extremely exciting. I can't give you any names, unfortunately, at this point.

**MR:** Eddie, how did you ever happen to meet up with Kiss?

**EK:** That's a long story; allow me to bore you with the details. It started off as a small project that [engineer/producer] Ron Johnson brought to my attention. He was producing a group called Wicked Lester, of which [basist] Gene [Simmons] and [guitarist] Paul [Stanley] were the mainstays. It was sort of a "Pop-y," "Beatle-y" kind of group—very pretty sounding. At the point where they were making a change musically, Gene came up with this concept of a rock 'n' roll band that wore makeup and had costumes. Ron asked, "Could you make their demo, because it's not really up my street."

We did a four-track demo at Electric Lady [in studio] B, a small room in the back. It was one of the best demos I had done and the one that got them the record deal. And to this day they still play it in their houses. I still have a copy of it. It's very raw and ballsy. We did the first two or three albums and then *Kiss Alive* came along and they asked me to be involved with it. I don't think the first few albums sold more than a few hundred thousand.

They had been touring steadily throughout the Midwest to develop a following, so that by the time we did the recording there were hundreds of thousands of fans out there. When the album came out, it just went *bang!* So, my association with them goes back to the very early days. That "live" album really put Casablanca records on the map. I love Kiss and I loved working with them. It was really a challenge to work with them in the beginning because they were so raw, rough and ready. Everybody would look at you sideways: "You're working with Kiss? Oooooooh! Yech!"

We proved that you can do it, take a raw band like Kiss and make them commercial. They were a bunch of guys that were really determined. They had a fantastic stage act. There's still no stage act around to top it. You may not agree with their music. You may not think their music is the greatest in the world—they're the first ones to admit it—but their stage show is incredible and they have continued to grow as musicians over the years, particularly Ace.

**MR:** What special things did you do in the studio to enhance the sound of Kiss?

**EK:** A lot of them. I'd double-up the voices where necessary. There was a lot of thickening of guitars, double tracking lead lines, double rhythm tracks, doubling with different types of amplifiers to get different types of layers of sound. Certain instruments would be tucked underneath other instruments.

We're getting into a tricky area here, where I'm bordering on giving away trade secrets. Engineers and musicians can decipher most of what's on a tape or record, but there are certain things that happen within the mixing stage which only the mixer knows for sure what he did. I can examine a record and say, well yes I think they did this, that and the other, and I can match it pretty closely, but I could never mix it the same way Phil Ramone or Ken Scott did because they have their particular touch. Every producer/engineer has a distinctive style. I have my own particular style the same way other individuals do. But to say, hey, I put a 40 ips tape delay on his voice and mixed that in with a digital delay at 42 milliseconds would be ridiculous, because then I would be saying this is how I get my sound. If you want to know how to get my sound you're going to have to pay for it. That's what they pay me for.

**MR:** Jimi Hendrix started to do something just before his death . . . he was on the verge of some great music as evidenced by the *Cry Of Love* and *Rainbow Bridge.* He was getting into really developing the art of layering and tracking. The art of layering is something that has hardly been tapped. It seems people in general are very regimented about the use of the studio. Although the Beatles really explored the boundaries of the studio.

**EK:** You see they had the benefit of having George Martin as a producer, who's a total genius in his own right, anyway. The Beatles and George Martin was a magical combination. You couldn't go wrong. The fact that he had so much experience in composing and arranging went a long way toward interpreting what they wanted to say musically.

There's a parallel that you can draw between what George did with the Beatles and what I did with Jimi. I was the person who would interpret what he was hearing in his head onto tape. That was my function all the way through, right from the beginning when I was engineering, right through to the end when I was co-producing with him. The most important function that I could serve that human being was an interpretive function.

If he heard the sound of a guitar under water, I would go to the length of actually putting a plastic speaker in a bucket of water to see what it sounded like. You think I'm kidding, but I'm not. I'm dead serious. It sounded terrible, but the point is we did it. It just went *garblebleble gurglegargle.* There was very little that we didn't attempt within the confines of what we had to work with.

He would come in to the studio with some new toy. He was always coming up with some new toy; something he had bought or some kind of squawk-box that someone had made for him. He'd try anything. We'd put it in there, turn it around backwards, put it in different amplifiers, put it through two Marshalls, two Univoxes, whatever. There was always some kind of experimentation, which I think there's a lack of today. I feel that everything is very formulated today, as you have said. I think that the age of experiment is dying; it needs an injection of chance. Some of the greatest things that have happened to me in the studio have been completely by chance, by fluke, by accident.

**MR:** Doesn't it cost a lot of money to do that kind of experimentation?

**EK:** Yes, if you're talking about working within a structured budget that a record company gives you. If you've got $75,000 to do an album, then you haven't got much time for experimentation. But what has actually happened is there's this movement with everybody having tape machines and mixing boards. There's plenty of room for experimentation at home, and I think there's a lot of experimentation going on in basement studios. Many people have sent me demos that are absolutely amazing; four-track demos that really floor me.

In the old days, and I don't mean to harp on this, but there was something about the way we used to record—stereo, mono, four-track, three-track, whatever it was . . . I mean Jimi Hendrix's albums were all four-track. The first two and a half albums were four-track. The Beatles' albums were four-track.

**MR:** Not at the end . . .?

**EK:** No. Jimi Hendrix's albums in the end were sixteen track. *Axis, Bold As Love, Are You Experienced* and the first part of *Electric Ladyland* were four-track.

**MR:** How about *Rainbow Bridge*?

**EK:** That was on sixteen track. What I'm trying to say, though, was [that] because we were restricted by the number of tracks we had, everything had to be done perfectly. You had to get *the* balance, *the* mix, because you had drums in stereo, and the bass across two tracks, a track for lead guitar, and one track was open. Rhythm guitar was actually the first guitar. Then you'd stick a voice on, and bounce that all across to another four-track machine, and then you'd have two more tracks for another lead guitar or maybe another voice. That was it. That was the only chance you had. In other words, every time you did a mix, you were making part of the final product. You had to think ahead: "If I do this now, what's going to happen, eventually, four generations away, two generations away, three generations away?"

> **"...I found that digital is just too 'clinical,' too precise for rock and roll the way I feel it should sound."**

This was very good training. You had to keep this in your head all the time. You were continually listening in the stereo mode. You were placing all the instruments, getting the relationships correct at the time, which today does not happen.

We've computerized mixing, and have anything up to 48 tracks. We have every goddamn device known to man to help us and it's pretty easy to make a record today. It really is. The science has become easier. The component parts have made things a little complicated to understand first time around, but once you understand what a Necam computer does, what a Necam does, what automated mixing is, then it's a snap. It's like clockwork. You can just sit back, program a few mixes and away you go.

There aren't that many variables in the studio, which is why I prefer to do "live" recording; there *are* many variables there. You are always taking a chance. Your ass is always on the line. What I try to do in my recordings of groups is to inject that degree of tension that's happening. I don't go for the total perfection approach of having every note in its place. I'd much rather have a great feel and the odd mistake. That's my philosophy.

**MR:** Most recorded jazz through the Fifties was like that.

**EK:** Yes, but I don't think that rock 'n' roll musicians as a whole are as qualified musically to pull that off. There are exceptions to that rule. I can't think of a "live" recording, maybe one or two, of rock 'n' roll on which there has been no overdubbing.

**MR:** Isn't that a bit fraudulent?

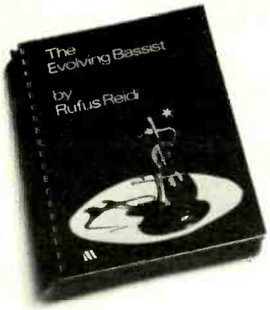
**EK:** It is and it isn't. There's nothing wrong with it. If you can utilize overdubing to better your product, why not? As a purist, you could argue that that stinks, but could you sit down and listen to an album with mistakes in it? I would let a few slide, I definitely would, but if the vocalist were singing flat I couldn't tolerate it. If the feel of the track is really good [and], the bass player and drummer are really kicking ass but the lead guitarist makes one flub . . . if I could find another piece from another night that's the same tempo and it will slide in I'll do it. If I can't do that, then I'll try to punch it in. It's essential. Economics dictate there. Unless you're a band like the Rolling Stones, you cannot afford to record the, say, 62 concerts on a tour and have the luxury of listening to each of them and then picking the best.

**MR:** Getting back to our new technology discussion, Eddie, how do you feel about digital recording?

**EK:** It's definitely the way in which we are going to be recording in the next five years. There has been so much press, and so much publicity on it from the corporations that have pioneered it, that all the studios that have them have put them aside and are going back to analog recording. It all happened too quickly. There are still a lot of problems. People saw it as the savior and they had to have it immediately. They didn't give it the chance to be developed. The machines are very expensive. When the price on them is lowered a bit, within the next few years, and the bugs in them have been worked out, then you'll really see them come out. The major studios that have digital machines aren't using them that much just yet.

I recently had the experience of working with digital equipment out at the Record Plant in L.A., and I made an A/B [analog vs. digital] comparison. Basically, what I found was that digital is just too "clinical," too pre-

# Rufus Reid on Bass and Bose®

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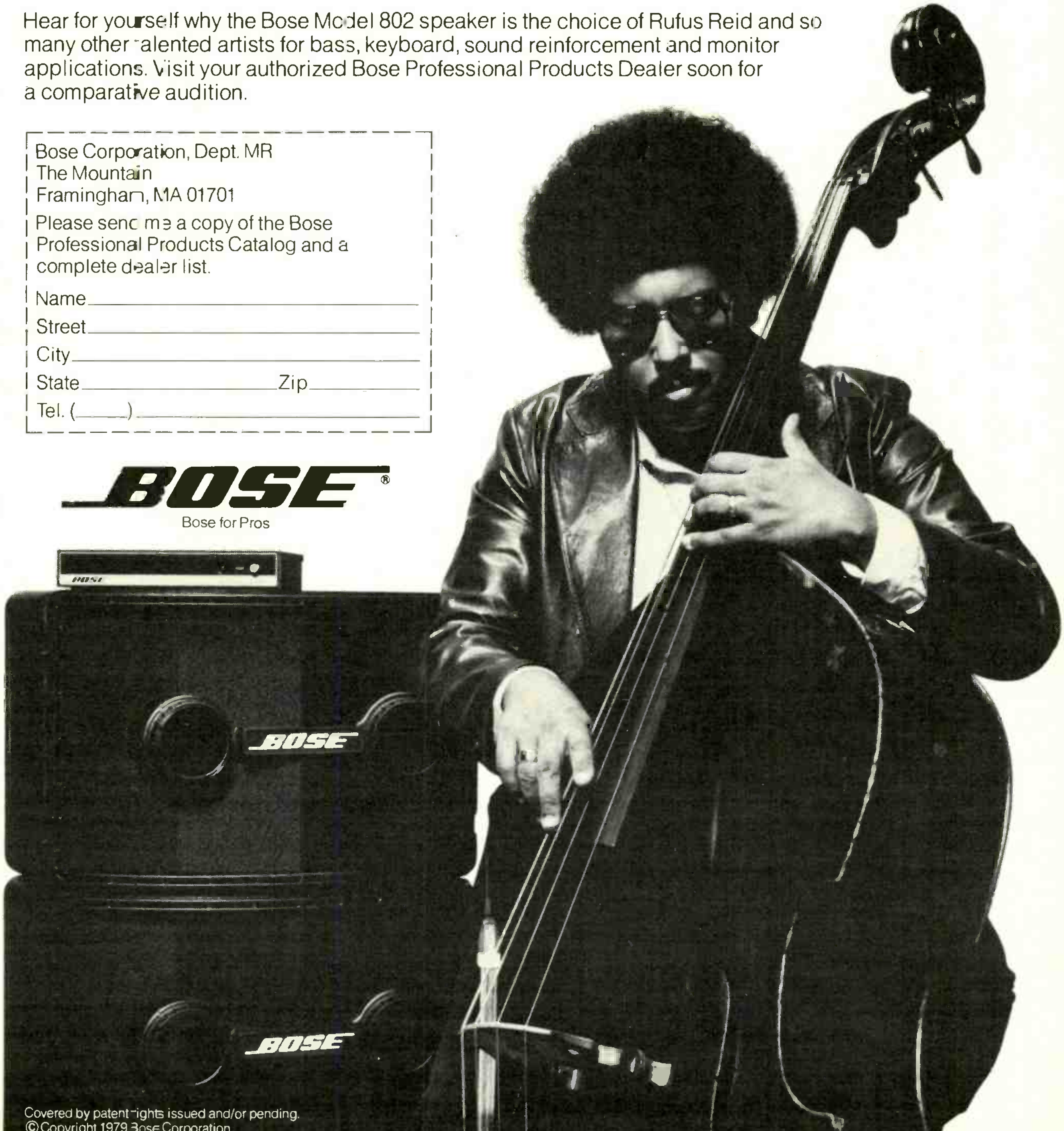


cise for rock and roll the way I feel it should sound. Analog machines—much like rock—are inherently "dirty," a quality that's lost with digital recording. What I think would be marvelous would be to use the digital machines for the vocal tracks, let's say, and resort to analog machines for the instrumental tracks. Combined in this way I think they'd sound fine, but it would be prohibitively expensive at this time to do it.

**MR:** Unlike many of the people in the recording field, Eddie, you seem to have less of an attachment to the equipment itself.

**EK:** To me, the equipment is a means to an end. I was raised as a musician first; the technical part was always secondary. I have never professed to be a great technician. I'm not. I use the equipment to get the sound that *I* want. I don't let the equipment dictate [the sound].

You know, there's one thing that I want to make a particular point of in this interview: monitor levels. It is so dangerous, and I cannot emphasize the word enough, *dangerous,* to be in the studio where the engineer is totally insensitive to levels. Driving 4-way monitor systems to their maximum is *committing aural suicide.* The only time you might need high levels is when you've got a great take down on tape and you want to crank it up—nobody minds it for a few minutes, but *don't* leave it there.

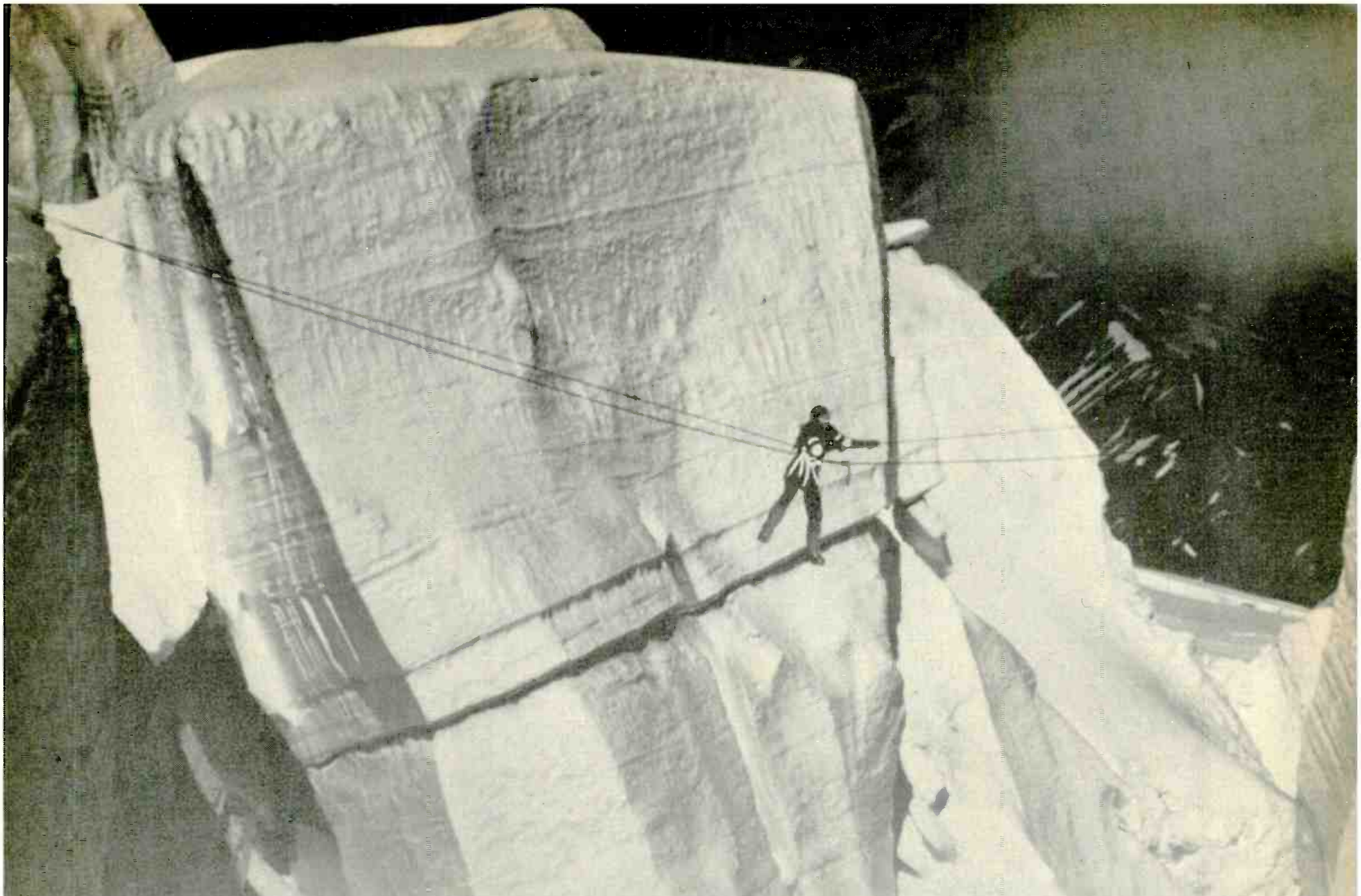
I feel that you must hear the tape through all different speakers. I constantly use three different speaker systems in the studio: I use the studio monitors, which are usually the Urei Time-Aligns, Eastlake-Westlakes, Altecs, whatever they have; JBL 4311s, a medium-size speaker; and the little Auratones. I am constantly switching between speakers and am constantly changing the monitor level—high volume, medium, low. I never monitor the same way throughout a session for fear of ear fatigue.

The same way with mixing. I try not to mix for longer than six or seven hours at a time. I'm not talking about cutting tracks because there you get a break in between. I am talking about consistently sitting behind the desk, in front of the speakers for long periods of time. It is very wearing, both physically and mentally. If you take great care with monitor levels, your ears will last for the rest of your life. There are many engineers around who have some nerve deafness in one ear or the other, and some in both. Certainly their thresholds are down significantly . . . on the high end they cannot even hear 15,000 cycles.

When I go to a rock concert, I stick a lot of cotton wool in my ears. I know then that I can at least make it through the concert. I was monitoring the levels in a club the other night, and the peaks were 120 dB, the average level was 110 dB. This was in a club that was maybe 50′×40′. That's insane. I had to make this point in this magazine because maybe someone will listen. *Your records will be better.* There is a volume above which clarity disappears.

You know how the government always seems to get in on the act? Well, I have a feeling that the EPA is eventually going to lay down some edict about volume levels. That's what's going to happen. I'm not for that. I'm for people deciding what they want, but I have this funny feeling this is what will happen.

**MR: Is there a quality of sound that could make it *seem* like it's loud, but not have such a high decibel rating?**



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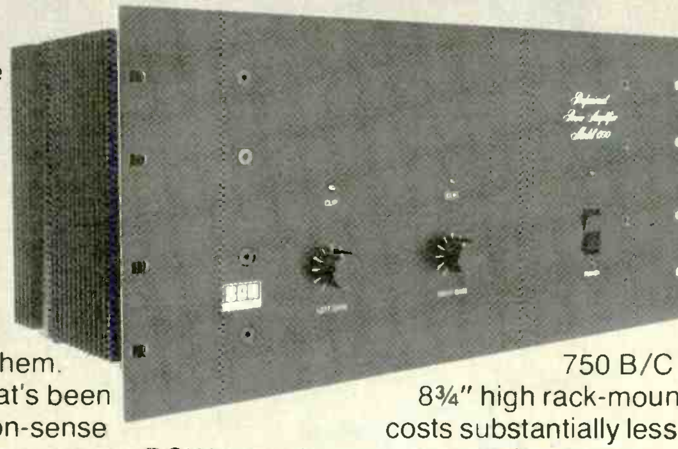

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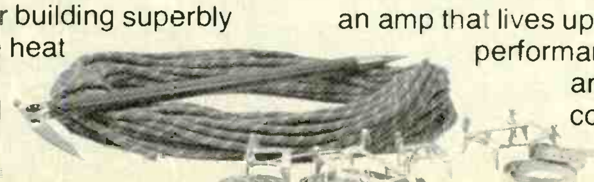

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**EK:** There are some theories [on that]; it's called "apparent loudness." I don't know much about it. All you would-be engineers out there should get on it.

**MR:** Let's hear your comments on the state of today's musicianship. So many so-called "great" musicians are these highly regimented, practiced players whose sessions all sound solid but stiff. You've worked with some great ones, players whose technique may have been lacking, but whose work has influenced so many.

**EK:** You know, Jimi Hendrix understood the solo to the extent that he could play the solo backwards in his head. With the tape turned around, he would know exactly where each note was, backwards as well as forwards, and he could play equally as efficiently backwards as well as forwards. That takes quite some doing. In other words, if he was considering putting on a solo backwards he would know what it was going to sound like before he played it. He was phenomenal. There are very few people who can take on that challenge of doing layer upon layer, as you were talking of before. He was one of the few people who could.

Jimi would do six solos, one after the other, and each one would be far superior to the one that came before. He would ask me at the end of each solo, "How was that one?" I would say, "Fantastic," and they were. They would consistently get better. You would be "stuck" with six masterpieces. What in the world do you do with six masterpieces? You do what that . . . ehh . . . I better not say what I was going to say.

Jimi was on the brink of new things; experimentation with horns. He was talking with Miles Davis. He was talking with a lot of musicians. Jimi was very into expanding his musical awareness. It would have been incredible. It was a great loss, both musically and personally.

I really hope that there's a new generation, a new breed of musicians coming through the ranks who will surpass our ideas of what "live" playing is all about. There's bound to be in the next ten or fifteen years, a musician who will come to the fore, thrown up through the ranks, that will blow our minds as Jimi did.

I know what you're saying, that we've reached a high level of mediocrity and it doesn't seem to be ascending beyond that. We can only hope.

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# Ambient Sound

BY LEN FELDMAN

## The Audiophile Versus the Audio Pro

The volume of mail that both Norman Eisenberg and I receive as a result of our contributions to *Modern Recording* magazine suggests that the most vocal readership of this publication is divided into two camps. When we become ecstatic over a new cassette deck that does everything but whistle "Dixie," we are often beseiged by letters from readers who tell us that they couldn't care less if the deck has a micro processor that can optimally set bias and EQ for a given tape. These readers generally don't trust anything that's electronically automated and would prefer to set up a machine by tweaking it manually, the old fashioned way, as they do with their studio open-reel tape decks.

On the other hand, when we report about a rugged amplifier with limited power bandwidth and deliberately curtailed frequency response that is intended primarily for P.A. work, we are swamped with letters from irate audio buffs who tell us that they wouldn't be caught dead listening to such an amplifier because its limited bandwidth, slow rise time, miniscule slew rate and general circuit design are out of the middle ages and couldn't possibly please their "golden ears." So, the question I'd like to address myself to this month is: What are the real differences between an audiophile and an audio pro? More specifically, how do these differences reflect themselves in the kind of equipment favored by each.

## Dynamic Range and Signal-to-Noise

Both "camps" would agree that high dynamic range in audio equipment is highly desireable, but adequate dynamic range and signal-to-noise ratio for the audiophile may mean something quite different from what it means to the pro audio person. Suppose, for example, that you are responsible for the sound reinforcement at a "live" concert. In terms of sound pressure level (dB SPL), you'll probably find that if the concert is taking place indoors, the ambient noise level of the audience may be as high as 60 or 65 dB SPL. Any sounds that your reinforcement system produces that are significantly below that SPL level will be buried below the noise floor. Yet, at the other end of the scale, you know that if you pump out sound pressure levels much beyond 120 dB peaks, you are either going to deafen your audience or be closed down by the Environmental Protection Agency. That leaves you with a maximum dynamic range requirement of around 60 to 65 dB maximum — not at all difficult to obtain from any pro-designed power amp. The real problem is not so much adequate dynamic range but power output capability of the amplifiers you use. For, even if you couple the amps to highly efficient speaker systems (horns, bass reflex units or what-have-you), you are trying to develop those high sound pressure levels in a large acoustic space. And, when you begin to talk about high power in pro amplifiers, your next immediate concerns are reliability, ruggedness, heat dissipation and the ability to draw such power levels continuously (or nearly continuously) from the amplifiers you select.

The audiophile, listening to his or her stereo music system at home, on the other hand, will probably never exceed SPL peaks of 110 dB or so. (The neighbors might not even tolerate that sound pressure level for very long, especially if the audio buff is an apartment dweller.) But — assuming that the audiophile resides in a relatively quiet place — ambient sound levels may be as low as 30 dB SPL. If hum and noise levels produced by the audiophile's amps are even 70 dB below rated output, sound levels produced by the noise and hum will be clearly heard above the ambient noise level in the room (110 minus 70 equals 40 dB SPL, well above the 30 dB ambient noise level). So, the audiophile demands wide dynamic range not so much because there are program sources that contain such extremes of musical information (we'll have to wait for all-digital program sources for that) but because he or she will not tolerate background hum and noise as an accompaniment to music. Also, a cooling fan which is often a necessary adjunct to a high powered audio amplifier used in professional applications would not be tolerated by an audiophile because of the additional noise it generates itself.

## Multiple Speakers and Damping Factors

Many professional audio amplifiers are equipped with 70-volt line distribution systems. These terminals permit the installer to add speakers (each equipped with a step down transformer) without having to worry about the "net impedance" that is then presented to the output terminals of the amplifier. So long as total power from the amplifier does not exceed its ratings, you can just keep adding speakers, each designed to draw a specified number of watts from the system depending upon how you tap into each speaker's transformer.

The system is analogous to plugging in more and more lighting fixtures or other electrical appliances to AC receptacles in your home. So long as the current drawn from these outlets is not high enough to blow the basement fuse or circuit breaker, the *voltage* supplied to each appliance remains constant at 120 volts. The advantage of a 70-volt line in audio systems is that you don't have to worry about impedance matching when using a great number of speakers and losses along long lines are minimal. (At 70 volts, there's little current needed to produce a given number of audio watts, compared to low-voltage-high current operation when you use 8-ohm or 4-ohm speaker combinations). All of this being so, why would audiophiles reject such a system when they want to use multiple speaker systems at home? Simply because the 70-volt distribution system results in lowered damping factors. Those matching transformers ahead of each speaker represent finite impedances and, to some degree, isolate the speaker from the low-impedance of the amplifier source. And since most hi-fi speaker systems depend upon high damping (from the amplifier) to deliver tight, unmuddied bass, audiophiles prefer to wire their speakers (hopefully with ultra-low resistance speaker cables) directly to the amplifier's outputs with no intervening matching transformers.

## EQ or No EQ

To the professional audio recordist or sound reinforcement contractor, selective equalization is an accepted fact of life. In sound reinforcement work, EQ is used to increase gain before the onset of feedback, make speech more intelligible in poor acoustic environments and provide better sound coverage. To the pro recording engineer, EQ can make the difference between a hit recording and one that is bland and unexciting. Yes, EQ can even make a not so-great vocalist sound better than in real life, though few superstars would ever admit how much their recording engineers contribute to their success.

Not so with the dedicated audiophile. The real purist looks with disdain upon any tone controls, preferring to push that button that bypasses them entirely or not to have them incorporated in the preamp circuitry in the first place. A great mystique has developed concerning the "damage" that tone control circuits do to a "pure" signal by way of phase shifts, added distortions of all sorts, and the like. In truth, there are modern tone control circuits that do none of these terrible things, but the myth persists.

Low-pass filters, so vitally necessary in certain microphone pickup applications and for other acoustic reasons in the pro field are confined to sub-sonic filters in most hi-fi equipment, and even this concession comes about only because of the new DC amplifiers that are so popular in high-end audio systems. The high warpage of many commercially pressed discs forces even the purest audiophile to introduce such sub-sonic filters into the signal path to reduce IM distortion that would otherwise be caused by the wild non-linear excursions of woofer cones.

## Tape Recording Requirements

When it comes to tape deck features (whether open reel or cassette), the difference in attitude between audio pros and audiophiles becomes even more pronounced. We have discussed these differences on more than one occasion, both in this column and in our comments concerning specific equipment that we have tested over the past few years. Audiophiles look for extended frequency response ahead of anything else. Pros are more than willing to give up wide bandwidth in return for better signal-to-noise and lower distortion. Clearly, the audio pro is concerned with the number of dubs or successive "generations" that will have to be made from the multi-track master—each of which degrades S/N performance by a few dB. The audiophile has no such concern. He or she is going to make *one* master recording—generally a copy of something that already exists in recorded or broadcast form—despite all those "dubbing" switches on amps and receivers which are rarely if ever used in real life.

## The Economics Of Pro and Audiophile Gear

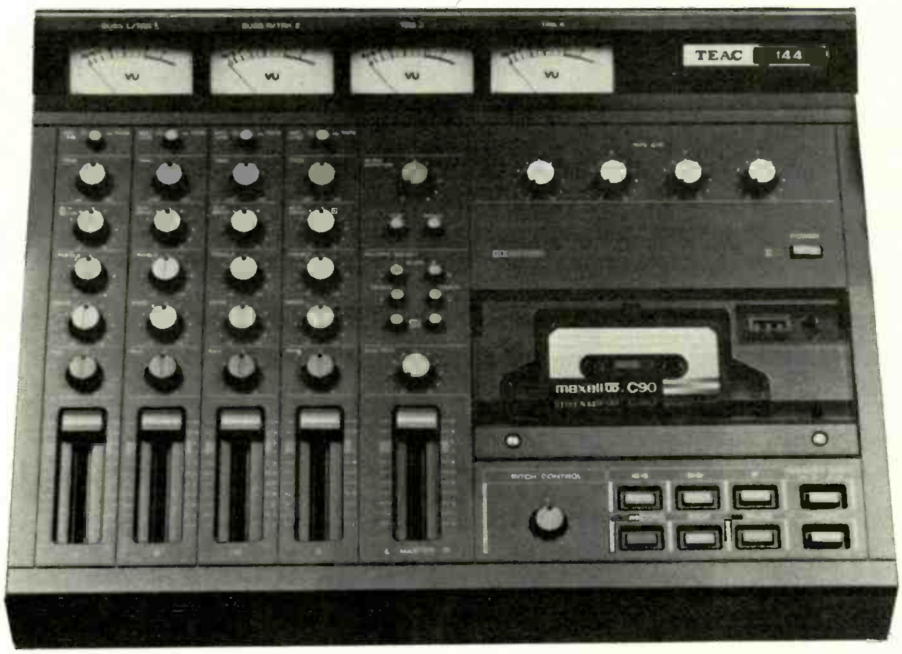
I'm sure that many a recording or sound engineer, entrusted with the job of purchasing audio gear for his or her operation, has often looked with envy at his audiophile counterpart who can choose from a wide selection of seemingly good-performing home gear in every price category. Why is it that the audiophile can buy a high-powered all-in-one integrated amplifier or receiver at a fraction of the price that he or she must pay for just a good professional power amplifier? One factor, of course, is reliability. The QSC A-42 tested for this issue, for example, costs just under $1000, but from the looks of it, you could rely upon it to keep working, day after day, year after year. It's no great tragedy if a home audio amplifier pops its fuses (or even its output transistors) every now and again, but such malfunctions cannot be tolerated by the professional whose livelihood and reputation depends upon reliability and continuous, uninterrupted operation. Then, too, consider the quantities involved. Home audio systems number in the many millions. In the case of pro audio gear, we deal in thousands and when you break that down to individual specialized manufacturers, you're talking about almost-customized production runs of, at best, hundreds, and at worst, dozens. Clearly, mass production techniques and attendant economic advantages do not apply in the case of pro equipment.

Will the pro audio field and audiophile field ever share common goals and common equipment? Perhaps some day. But for the moment, they remain pretty well isolated from each other and, unfortunately, somewhat contemptuous of each other as well.

MODERN RECORDING LAB REPORT

NORMAN EISENBERG AND LEN FELDMAN

# Teac Model 144 Portastudio




**General Description:** Teac's new model 144, called the "Portastudio," is a complete multi-track recording system built around the cassette format. It contains a full-function 4-channel mixer, a 4-track cassette recorder with overdub capability, functional record/reproduce signal routing and monitoring systems and built-in Dolby-B noise reduction. Considering all that is contained here, which amounts to a rudimentary basic studio facility, the complete unit is surprisingly compact and lightweight.

The cassette recorder itself is a departure from the standard format. It runs at 3¾ ips speed. The head configuration is four-channel and not the standard stereo type. The device can record a maximum of two tracks at once, and a cumulative total of the four tracks, with the tape running in the same direction. It will play up to four tracks at a time. Even the track-spacing differs vis-a-vis what is used on standard cassette decks. Thus, to get a compatible two-track tape for playback on a standard cassette deck requires that a tape made on the Portastudio be remixed and dubbed onto another deck. The use of a C-120 size cassette is strongly advised against, which means that the maximum running time for a single cassette (assuming a C-90) is 22½ minutes at the 3¾ ips speed. The cassette recorder is optimized for high-bias ferric tapes (types recommended in the owner's manual are TDK-SA and Maxell UD-XL II, or equivalents). There are no tape selector controls for other kinds of tape. The Dolby circuitry is always on.

Transport controls—feather-touch, logic-controlled—permit fast-buttoning and run-in recording. The lineup includes rewind, fast-forward, normal forward, memory stop, record, stop, pause, and a final button to open the cassette compartment door. The index counter and its reset button are just to the right of the cassette well. Above is the unit's power off/on switch. To the left of the transport keys is a pitch control for varying tape speed over a ±15 percent range.

The left portion and an area over the cassette section are given over to the "board." To begin with, there are four rows of identical controls, one row per track. The controls include (from the top): a button to select mic/line or tape; a trim knob to vary the gain produced by the first amplifier; an auxiliary send control; a treble knob; a bass knob; a pan knob; and a slider-type input fader at the bottom.

The fifth row of controls contains a record-select switch matrix. There are buttons to assign only to track 1 or 3, and only to track 2 or 4. The device has two buses (mixing networks) for recording, at most, two tracks at a time. An aux-receive knob sets the level of whatever inputs are plugged into the left and right aux jacks (at the rear). This section also includes a left-right master fader, a bus monitor control, and cue and remix buttons. The bus monitor and the last two buttons affect what you hear when monitoring with stereo headphones; these three controls do not affect the line outputs.

In the record select group are two additional buttons to disable the recording circuits separately on tracks 1 or 3, and on tracks 2 or 4. This feature prevents accidental tape erasure during playback of either an initial track or an overdub.

The Portastudio's four VU meters, arranged on a forward sloping panel at the top, are calibrated from −20 to +3, with an uncalibrated area beyond the +3 mark. Each meter monitors its track; the first two also provide readout for the left and right buses.

The rear of the unit contains a total of fourteen signal jacks. There is a ¼-inch phone jack for stereo headphones (mono headphones or any device using a single tip-ring connector are forbidden). Next there are pin-jacks for right and left aux out, right and left line out, right and left aux in, tape cue and aux send. Finally, there are four ¼-inch input jacks. These may be used for connecting microphones or—since their sensitivity is adjustable—any medium-level line source up to a maximum of 2.5 volts. Recommended microphones are any types with an impedance rating of 50 to 10 K ohms, and a minimum level of 0.5 mV (−66 dBv).

The owner's manual (of which we saw only a first draft of the text, minus the illustrations) contains detailed instructions for using the Portastudio, including overdubbing, remixing, ping pong, stereo/2-track transfers and recording, auxiliary signal processing, punch-in recording during an overdub, using only one mixed bus, and so on. Also included are hints on microphone placement, equalization, signal levels, impedance and simple owner maintenance.

**Test Results:** Although the Teac model 144 has more versatility than can be fully evaluated in the time needed to prepare a report for publication, we did put the device through most of its indicated paces, in two separate setups, and found that everything we attempted worked "as claimed." The device is both unique and effective.

Performing specs were easily met or exceeded in our bench tests. Frequency response went better than claimed at both the low and high ends. Although the response falls below that of a conventional hi-fi cassette deck, it should be emphasized that—as shown in the r/p response curves of Fig. 2—the results at 0 dB record level are essentially the same as the −20 dB record level. This is something never found in a home cassette deck, even one using metal tape. In other words, what has been gained in this design is headroom within the overall bandwidth. The deliberately restricted bandwidth in this unit gains signal-to-noise ratio and dynamic range capability—factors that are especially important when the Portastudio is used for multiple dubbings and mixdowns, where each successive dubbing of a previously recorded track onto another track results in some degradation of S/N. The emphasis, obviously, in this unit is on its usefulness as an audio tool, rather than as home hi-fi gear.

This design philosophy also is seen in the action of the equalizer (treble and bass) controls, as plotted in Fig. 1. Note how these response curves differ from those obtained from typical hi-fi controls. Instead of the boost or cut continuing to increase at the frequency extremes (to form the usual "bow-tie" pattern), the Portastudio's controls exhibit a definite shelving effect, which we believe to be more desirable for recording-EQ applications of this sort.

**General Info:** Dimensions are 18½ inches wide; 4¾ inches high; 14⅝ inches deep. Weight is 20 lbs. Price of the model 144 is $1,100.

**Individual Comment by L.F.:** On page 10 of the Portastudio's 61-page preliminary owner's manual, it

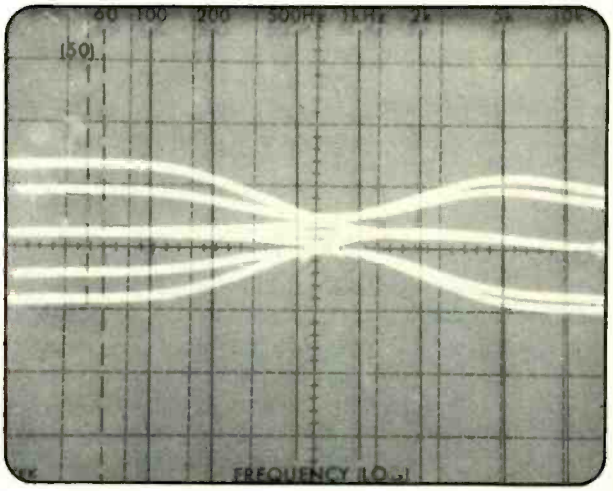



**Fig. 1: Teac Portastudio: Photo illustrating the range of EQ controls (bass and treble) in each input module.**

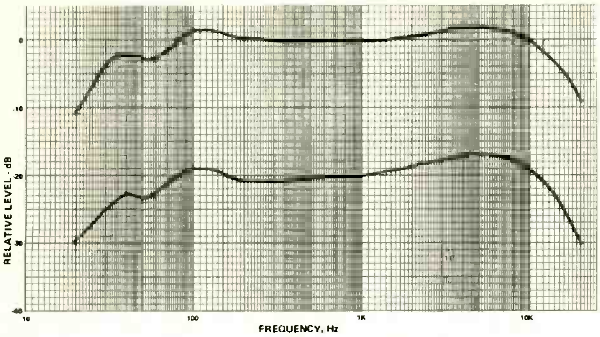



**Fig. 2: Teac Portastudio: Record/play response at 0 dB and – 20 dB record levels.**

says: "Because the Portastudio is so versatile, no manual could describe all the possible applications and hookups." I couldn't agree more. I had to read the manual very carefully in order to appreciate all the things I could do with this unique instrument—or rather "instruments" since the device is no less than a complete multi-track recording facility built into one lightweight package.

The device is aptly named. Its weight and dimensions make it ideal for using as a "studio away from the home studio." And since headphone monitoring capabilities are totally flexible (and electrically separate from what is happening at the line and mic inputs), you don't have to cart along tons of monitoring amplifiers, speakers, etc., when you want to venture forth into the field for multi-track experiments or audition recording work.

If you are a multi-track recordist who owns an open-reel deck plus a mixer, you won't have any trouble understanding the arrangement of the four input modules. Each can be assigned to a given *track* (left or right channel) on the tape, but only two tracks can be recorded at one time. In recording terminology, the Portastudio has only two buses or mixing networks that can be used to record a maximum of two tracks at a time. The left bus assigns *only* to track 1 or 3, while the right bus assigns to track 2 or 4. Of course, the panning controls available in each input module let you route signals to "points in between" extreme left or right by assigning variable amounts of input signal to both buses if you so desire.

The Portastudio obviously is an audio-recording tool, rather than a piece of hi-fi gear. To begin with, its standard operating speed is 3¾ ips. While that's no longer unusual even in the hi-fi field (at least three other makers of hi-fi cassette decks now offer two-speed decks) it does mean that recording time using a C-90 cassette is limited to 22½ minutes (and Teac specifically warns against trying to use thin, C-120 cassettes for extra time).

Nor should you expect to turn the cassette over for an additional 22.5 minutes of recording time, since the 4-track arrangement which is inherent in the unit's multi-track, dubbing and overdubbing capabilities precludes such use of the "second side" of the cassette.

The audiophile will also probably be discouraged by the absence of any tape selector switches. Teac has wisely optimized the machine for one basic type of tape: high bias cobalt-ferric (such as TDK-SA with which we tested the machine, or Maxell UD-XL II, etc.) since the serious recordists will quite naturally select one tape and stick with it (unlike the audiophile who keeps trying different kinds of tape in hopes of finding one that surpasses all others). As for the "always on" Dolby circuitry, why not? So long as the factory has calibrated the Dolby circuits so that they track well from record to play, the user is better off not fooling with Dolby calibration.

In summary then, the Portastudio is intended strictly for the [musician and the] multi-track recordist who wants to be creative, intends to do more than transcribe discs or FM programs onto cassettes, and either can't afford or isn't inclined to spend a small fortune on equipment for use in mobile or portable recording assignments. As such, it does its job extremely well and with greater flexibility than anyone would have a right to expect at such a low cost.

**Individual Comment by N.E.:** Teac, a company that has been very active—both in product design and in its literature—in fostering the creative recording trend has again come out with a highly interesting product that is unique and which also shows many favorable signs of its larger antecedents in the Teac "product family."

The Portastudio is, essentially, a scaled-down version of a mixing board with modifications that suit it for use with the cassette format, and the latter is supplied built-in so that the whole setup is laid out in front of you on one well-designed surface. The "porta" part of its title derives from the system's relative compactness and light weight; it does not, however, imply that the Model 144 can be operated on batteries.

The manual accompanying the device is virtually a primer course in creative recording basics, although it cannot possibly cover all the ramifications and techniques which an experienced multi-channel recordist develops over a period of time. In a sense, the product is a tribute to the ingenuity of Teac product engineers who have been able to elicit so much versatility, and genuine "studio practice" applications, from the cassette format. As such, it could open new vistas for the sound enthusiast who has not yet gotten into recording beyond tape-copying on a simple machine from other pre-engineered sources such as discs or broadcasts or other tapes. For the seasoned recordist, the Portastudio could serve as a handy take-it-along setup, although in this use one should be aware of its admitted limits (vis-a-vis an all-out board and open-reel machine) and the fact too that the most running time you can get from it with one cassette (a C-90 size) is 22.5 minutes. Come to think of it, that much time should be quite enough for takes of short numbers.

Basically, however, the Portastudio is aimed at the recording activist who hankers after a generous measure of studio-type hardware but who lacks the studio itself not to mention the small fortune it takes to equip one. The Portastudio also would make a very effective training aid for use in schools teaching the art of multichannel recording and the use of pan pots, faders, overdubbing, send techniques, and so on.

Finally, there is the musician. I showed the Portastudio to my friend, the guitarist John Lehmann-Haupt. John—who has recorded commercially (Physical World Records) and who also performs "live" professionally—said he felt the Portastudio would be an ideal system for use by a small combo. They could try out various performing techniques and sound combinations on their own, not only to prepare their own creative showcase, but as a sonic dress-rehearsal prior to doing their gig in a bit-time studio.

**TEAC MODEL 144 PORTASTUDIO: Vital Statistics**

| PERFORMANCE CHARACTERISTIC | MANUFACTURER'S SPEC | LAB MEASUREMENT |
|---|---|---|
| Frequency response, high-bias tape | ± 3 dB, 40 Hz to 12.5 kHz (at 0 dB) | ± 3 dB, 35 Hz to 13.5 kHz (at 0 dB) |
| Wow and flutter (WRMS) | 0.04% (0.06% peak wtd) | 0.025% (0.05% peak wtd) |
| Speed accuracy | within 1% | within 0.8% |
| Crosstalk | 50 dB | 59 dB |
| Signal-to-noise, w/Dolby, re: 3% THD record level, high-bias tape | 63 dB | 64 dB |
| Record level for 3% THD (0 dB = 250 pWb/m), high-bias tape | NA | + 3 dB |
| THD at 0 dB record level, high-bias tape | 2% | 1.6% |
| Line output at 0 dB | 0.3 V nominal | 0.3 V |
| Headphone out level, 0 dB | 100 mW/8 ohms | 205 mW/8 ohms |
| Mic input sensitivity for 0 dB | 0.5 mV | 0.44 mV |
| Line input sensitivity, 0 dB | 0.3 V nominal | 0.29 V |
| Fast-wind time, C-60 | 70 seconds | 80 seconds |
| Bias frequency | NA | c. 65 kHz |
| Power consumption | 28 watts | 30 watts |

CIRCLE 14 ON READER SERVICE CARD

# QSC Model A42 Power Amplifier


**General Description:** The Model A42 from QSC Audio Products is a stereo basic or power amplifier rated for 8-ohm and 4-ohm operation (200 and 300 watts, respectively, per channel). The amp has a built-in two-speed cooling fan plus very effective thermal cutout provisions. Cross-patch or "daisy-chain" hook-ups are possible, as well as bridged-mono operation. The latter application includes using the amplifier to drive 70-volt lines. Special instructions for these hook-ups are included in the owner's manual.

Fitted with handles and ready for standard rack-mounting, the QSC amplifier features, on its front panel, separate controls for gain and for power limit on each channel. The gain controls are calibrated in dB, with "0 dB" corresponding to the 26-dB standard. The ability of the A42 to reach higher levels is indicated by the + 7 dB marking on each gain control.

The power limit knobs enable the operator to put a ceiling on the maximum power delivered to either speaker system. Output levels and power-limit levels are displayed on LED indicators above the knobs. The former indicator is a four-step graph plus a TDI (for "true distortion indicator"). This display is calibrated at five points, over a 15-dB range from 3% to full power. Power levels shown are for 4-ohm loads; for 8-ohm speakers, the readings must be divided in half. The same arithmetic applies to the power limit display. A rocker switch to the left of the gain and power limit knobs turns the amplifier on or off.

Outputs at the rear include 5-way binding posts and dual ¼-inch phone jacks for connecting one or two sets of speakers to each channel. For input signals, each channel has a male-female set of 3-pin XLR jacks, and dual ¼-inch stereo phone jacks. Balanced-line or unbalanced-line inputs may be used; the unbalanced-line inputs require ¼-inch mono plugs and single-conductor shielded cable.

As may be seen from the photo of the amplifier's interior, all the output devices are directly affixed to the multiple heat sinks (metal to metal) for maximum transfer of heat, with half of the power devices mounted on heat sinks that are actually part of the unit's top cover. When the cover is in place, the top heat sinks nest with the lower ones, and the whole assembly lines up very nicely with the air path created by the cooling fan.

**Test Results:** The QSC amplifier was inadvertently subjected to a "torture test" during the initial stages of bench measurements and it came through admirably. Here is what happened: For some inexplicable reason (perhaps just to change the pace), it was decided to measure the amp before photographing it. (The usual procedure is: photograph inside and out, bench-test and, finally, listen).

That the QSC A42 had a cooling fan was obvious from the front vents and even more so from peering through the rear vents and noting the plastic blades of the fan. Thus, when power was turned on for the first time, and we did not hear the fan, it was assumed that the fan was the type that came on only when needed. (This erroneous assumption was also due to not having studied all the instructions first which *do* state that the fan runs on slow speed under normal use.)

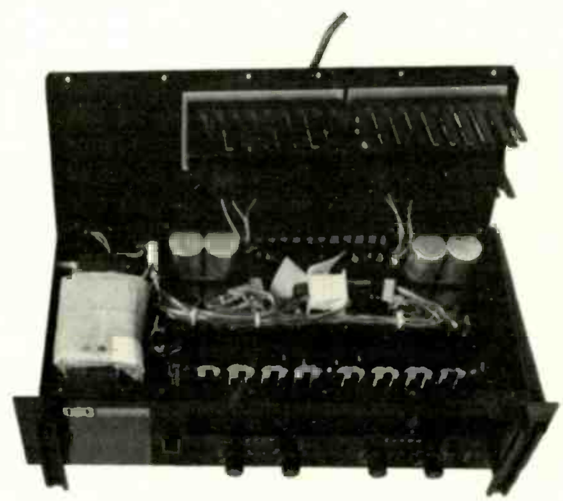
Blissfully unaware that the fan should have been operating, we started measuring. As you might expect, during some of the preliminary high-power tests, thermal cutout took place—again and again. Happily no damage resulted. Each time thermal cutout occurred, we simply waited for the amp to cool down, and resumed measurements.

We finally decided to investigate. Lo and behold, the bracket that holds the fan in place had apparently shifted during shipment, and the plastic blade of the fan was resting against a circuit board. Not only wasn't the fan rotating but by being jammed against the circuit board, the current applied to the fan motor had caused it to heat up, thus adding to the overall heat produced by the audio circuitry.

The fan was quickly restored to its proper position, and all measurements made up to that point were repeated. No thermal cutout was encountered after this correction.

There are three morals to this story. One: Even "experts" should practice what they preach, and really study the instructions for every new piece of equipment. Two: the QSC A42 is very obviously a fail-safe amplifier. Instead of blowing a couple of its twenty-four output devices, we'll settle for thermal cut-out any time! Three: since the dislocation of the fan occurred without our knowing it, it could conceivably happen to other units, and so anyone using this amplifier should, before turning it on, check to see that the fan is indeed correctly seated.

In addition to living up to its fail-safe claims, the QSC also met its power and distortion specs. As for frequency response, although the −1 dB rolloff point



**QSC A42: Half of the power solid-state devices used are mounted to a heat-sink attached to the cover of the amp, which for this photo has been swung aside.**

in our sample occurred at 29 Hz instead of at 20 Hz, we do not regard this as terribly important. It should be kept in mind that this is a pro-amp rather than one designed for super-audiophile home hi-fi; the energy below 30 Hz can roll off by a little more than 1 dB and still not impair the kind of reproduction this amp would logically be used for. (Whether it would be noticeable even to a hi-fi aficionado is also a debatable question.) The bandwidth limiting at both the low and high ends does result in a rather low slew rate (10 volts per microsecond). Some listeners probably will claim they can hear this as TIM on certain program sources. However, if you stick to the outputs of most microphones, mic mixers, conventional discs made from analog master tapes or even possibly the master tapes themselves, chances are you never will miss the extra volts-per-microsecond, and will not hear any significant TIM or transient distortion of any kind.

**General Info:** Dimensions are 19 inches wide; 5¼ inches high; 10¼ inches deep. Weight is 38 lbs. Price of the amplifier is $948.

**Individual Comment by L.F.:** Our "Vital Statistics" table pretty much tells the story of the performance of this amp. The emphasis here is on high power, reasonably low (but not ridiculously low) THD and IM and rugged reliability. The outputs are AC-coupled—that is, a capacitor with a low cut-off frequency isolates the output devices from the speaker terminals. I do not object to this in a pro amplifier where sound is being amplified for all but the so-called "golden-eared super-audiophiles." Not every amplifier in the world has necessarily to be a "DC" design. What I question, however, is QSC's implying in their literature that their use of AC-output coupling is some sort of technological breakthrough. Come on fellows, why not own up to the fact that using a blocking capacitor at the output simply was easier and less expensive than using a full complementary output design. We wouldn't have faulted you for it, honest!

The fact that the $-1$ dB rolloff occurred at 29 Hz in our tests, as against the 20 Hz spec'd, is covered in "Test Results" above; it is not a "big deal." I didn't have a schematic diagram but I suspect that the bandwidth limiting of the A42 occurs at the input as well as at the output, and if that is true you just can't get into slew limiting, even with the relatively low slew rate we measured for the unit.

So, having taken a simple but reliable route in the circuit design of the A42, I am left to wonder why the unit has to cost nearly $1000. Maybe it's that failsafe thermal cutout feature, or the LED power indicators, or the front-panel knobs that let you set the wattage level at which you want peak limiting to take place. Actually, that last feature is one of the best ways I know of to protect under-power-rated speakers—even better than fusing the speakers with slow-blow fuses. Whatever the reason, the QSC A42 is a rugged, reliable power amp that should be able to take whatever punishment you give it, continuing to operate under all but the most difficult circumstances—and that's worth a fair amount too if you are a professional whose livelihood depends upon equipment holding up and performing though under stress.

**Individual Comment by N.E.:** Many of the audio units we have tested recently, especially amplifiers, seem to straddle the fence between two main areas—that of the demanding home system audiophile user, and that of the professional user. Maybe that's because their various manufacturers are not sure of enough of a market in either area and want to hedge their bets, so to speak. Whatever, it is plain that this amplifier—the QSC A42—is aimed at the professional market. There is no apparent "concession" here to the home audiophile user—even to the use of XLR and ¼-inch phone connectors. The XLR connectors even offer the option of female or male mates, so whichever you have on hand from your input sources, you are in business immediately. Apropos of these inputs, by the way, since all four on each channel are parallelled, you can go into one with the input signal, and come out with the same signal from any other jack to feed the same signal to the other channel of the same amplifier, or to an additional amplifier. This is one way to get mono service from the QSC A42 without bothering to use the internal bridging switch (located, by the way, beneath a small plastic hole plug next to the channel 2 input jacks). For whatever it may be worth, in some possibly kinky-mix applications, you also can feed several different inputs at the same time into either channel of the A42.

**QSC A42 AMPLIFIER: Vital Statistics**

| PERFORMANCE CHARACTERISTIC | MANUFACTURER'S SPEC | LAB MEASUREMENT |
|---|---|---|
| Continuous power for rated THD | | |
| (8 ohms; 1 kHz) | 200 watts | 204 watts |
| (4 ohms; 1 kHz) | 300 watts | 306 watts |
| FTC rated power (20 Hz to 20 kHz) | 200 watts (8 ohms) | 200 watts |
| THD at rated output, 1 kHz; 8 ohms/4 ohms | 0.1%/0.1% | 0.04%/0.08% |
| THD at rated output, 20 Hz; 8 ohms | 0.1% | 0.1% |
| THD at rated output, 20 kHz; 8 ohms | 0.1% | 0.1% |
| IM distortion, rated output, SMPTE/CCIF/IHF | 0.05% /NA/NA | 0.043%/0.060%/<0.03% |
| Frequency response at 1-watt output; (for $-1$ dB) | ±1 dB, 20 Hz to 25 kHz | 29 Hz to 32 kHz |
| S/N ratio, re: 1 watt, "A" wtd, IHF | NA | 88 dB |
| S/N ratio, re: rated output, "A" wtd | 95 dB | 93 dB |
| Dynamic headroom, IHF | 2.0 dB | 0.63 dB |
| Damping factor at 50 Hz | 200 | 200 |
| IHF input sensitivity | NA | 0.055 volts |
| Input sensitivity re: rated output | 0.85 volt (8 ohms) | 0.75 volt |
| Slew rate (volts/microsecond) | NA | 10 |
| Power consumption, idling/maximum | NA/1625 watts (4 ohms) | 68/870 watts (8 ohms) |

# Thompson Model TAD-4 Analog Delay


**General Description:** The TAD-4 (the letters stand for Thompson Analog Delay) from LT Sound combines spring reverb units (one per stereo channel) with fairly complex electronics to produce a sophisticated analog delay and reverb system in a single, compact unit.

Provision is made for connecting two stereo amplifiers for driving "front" and "rear" stereo speaker systems. Inputs to the unit may be normal stereo (two-channel), or four-channel. Separate controls permit processing the signals fed to the front speakers independent of the rear speakers.

There are two identical sets of knobs at either end of the front panel; those at left (direct, echo and reverb) handle front sound while the three at the right end handle the rear sound. The ambience controls in the middle handle delay time, echo repeat, echo EQ and reverb EQ. There also are four small buttons. One turns power off and on. Another selects stereo or quad signals. The "cross" button, if used, places the echo and reverb signals for the left channel to the right side and vice versa. The "hi pass" button facilitates choice between a normal signal and a high-pass signal on the rear channels.

Just to the left of the left front level direct control is the LED level indicator, while centered between the ambience knobs is the power off/on indicator.

The rear panel contains stereo pairs of pin-jacks for front and rear inputs, and front and rear outputs; the unit's AC power cord; and a 1-amp fuse holder. The TAD-4 has a black matte panel with bright, very legible markings. It is fitted with walnut side pieces; a version for rack-mount also is available.

In addition to its applications in playback, the TAD-4 also is suggested for providing ambience in recording situations, and for creating recording effects. Included in the effects applications are doubling (creating an illusion of twice the number of performers); slap echo (a lower echo volume and a longer delay time are used); pitch bending (moving the delay time control alters the musical pitch slightly); delayed reverb; and—with the use of an external signal generator—voltage controlled effects of vibrato and chorus. This last application involves a circuit modification for which Thompson charges $15.

**Test Results:** The bandwidth/delay relationships achieved by the TAD-4 were quite impressive. In the most often used delay ranges up to 100 milliseconds, this unit offers nearly 10 kHz of top end for the delayed signal—far greater bandwidth than we have encountered with either digital delay units or other analog delay lines. And since different acoustic environments will produce delays having different bandwidth content, the TAD-4 permits you to actually equalize the "echo" (or delayed) sounds over the range indicated in the plots of Fig. 2. The bandwidths for various delay times are shown in Fig. 1, they conform very well with the data specified.

It is, of course, difficult to show graphically all that a unit of this type can really do, but an idea may be had by using tone bursts, and photographing the results on a 'scope. The input burst is shown as the upper trace on our series of 'scope photos, while the output signals in each instance are represented by the lower traces in Figs. 3, 4 and 5.

In Fig. 3 we see an output consisting purely of a delayed tone burst. The delay was set for minimum (20 msec) simply to keep both input and output traces on the 'scope face. We could have increased the delay to its maximum, and while the waveform would not have changed in appearance, it would have been difficult to correlate actual delay with the 'scope's sweep rate.

In Fig. 4, the delay has been increased somewhat, and also some direct sound has been mixed in.

In Fig. 5, we let everything happen. The lower trace here contains some direct signal, the same delayed signal as in Fig. 4, and a generous helping of reverb

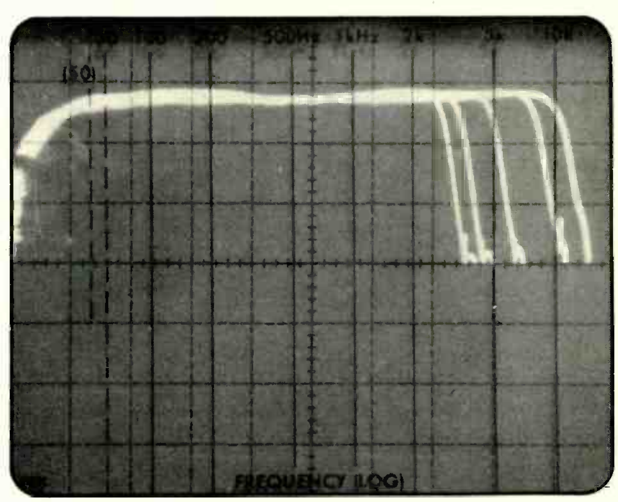



**Fig. 1: LT Sound TAD-4: Bandwidth of "Echo" (delay) circuitry depends upon delay time selected (see "Vital Statistics").**

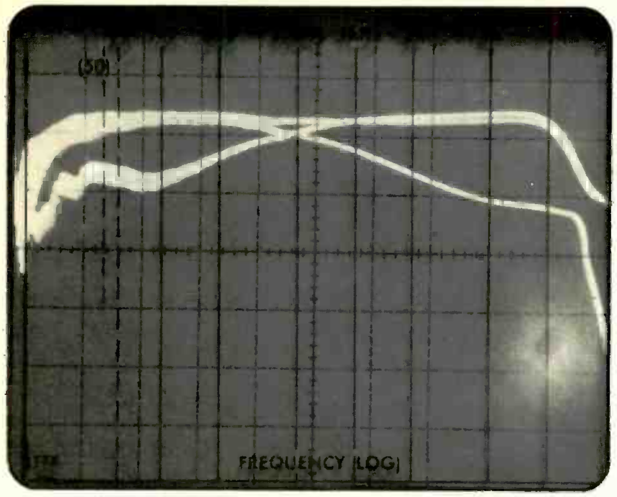



**Fig. 2: LT Sound TAD-4: Range of Echo EQ control at extreme settings.**

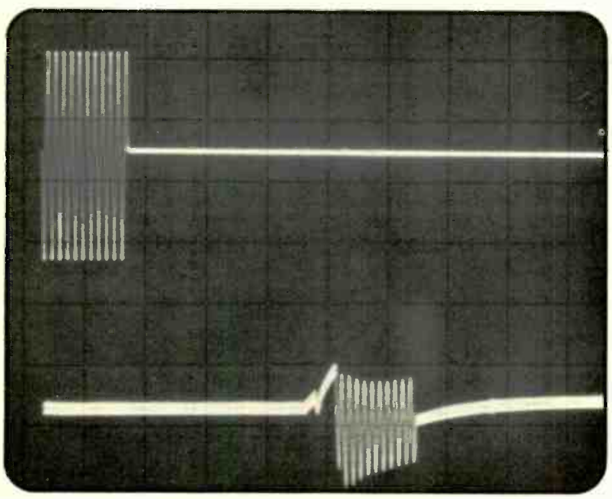


**Fig. 3: LT Sound TAD-4: 'Scope photo shows input burst (upper trace) and delayed output only (lower trace).**

which, as may be seen, is highly random and incoherent—as it should be, and as it is in a real acoustic situation.

The complexity of signal paths and processing in the TAD-4 almost defy verbal description in the normal space here allotted, and so for our more technically inclined readers we have included a block diagram (Fig. 6) of this highly versatile and well-built device.

**General Info:** Dimensions (with wooden side panels) are 18¼ inches wide; 2½ inches high; 7½ inches deep. Rack version: 19 inches wide; 2 inches high; 7 inches deep. Price: $650. Optional rack mount, $20 additional.

**Joint Comment by N.E. and L.F.:** In a recent "Ambient Sound Column" (July 1979), Len indicated a preference for electronics when it comes to straight time delay equipment, and for properly designed spring assemblies when one seeks good reverberation effects. Apparently LT Sound agrees, because that is exactly how the TAD-4 is designed. If you look at the photos of the unit's interior you can see how neatly they have managed to combine the spring reverb units for both channels with the complex electronics required for producing an analog delay with the equivalent of 4096 stages.

In addition to the bandwidth/delay relationships (see "Test Results" above), we also were impressed with the unit's facility for controlling what happens in the front channels completely independently of what is fed to the rear (or ambience) channels. Thus, some delay, echo and reverb can be added to the direct front sound. This feature should appeal especially to those who

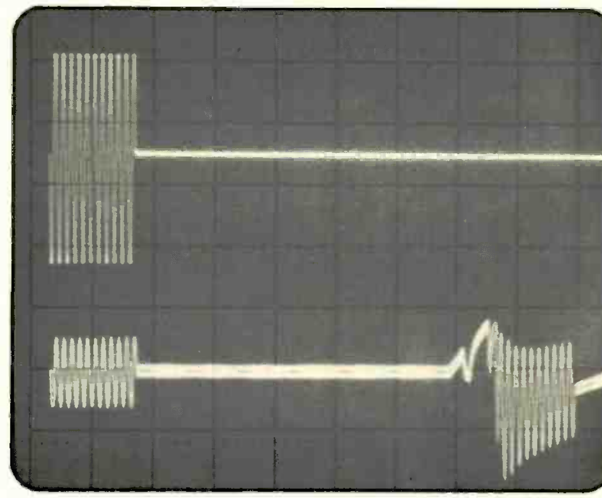


**Fig. 4: LT Sound TAD-4: Direct and delayed burst combined (lower trace).**

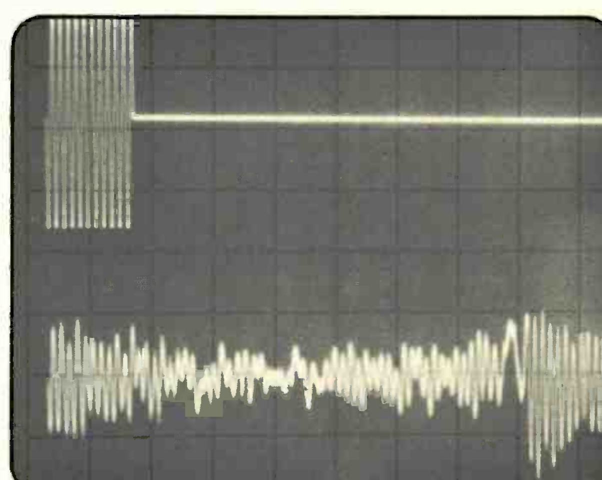


**Fig. 5: LT Sound TAD-4: Lower trace shows combination of some direct sound, plus delayed signal, plus reverb, all created by TAD-4 from single input burst (upper trace).**

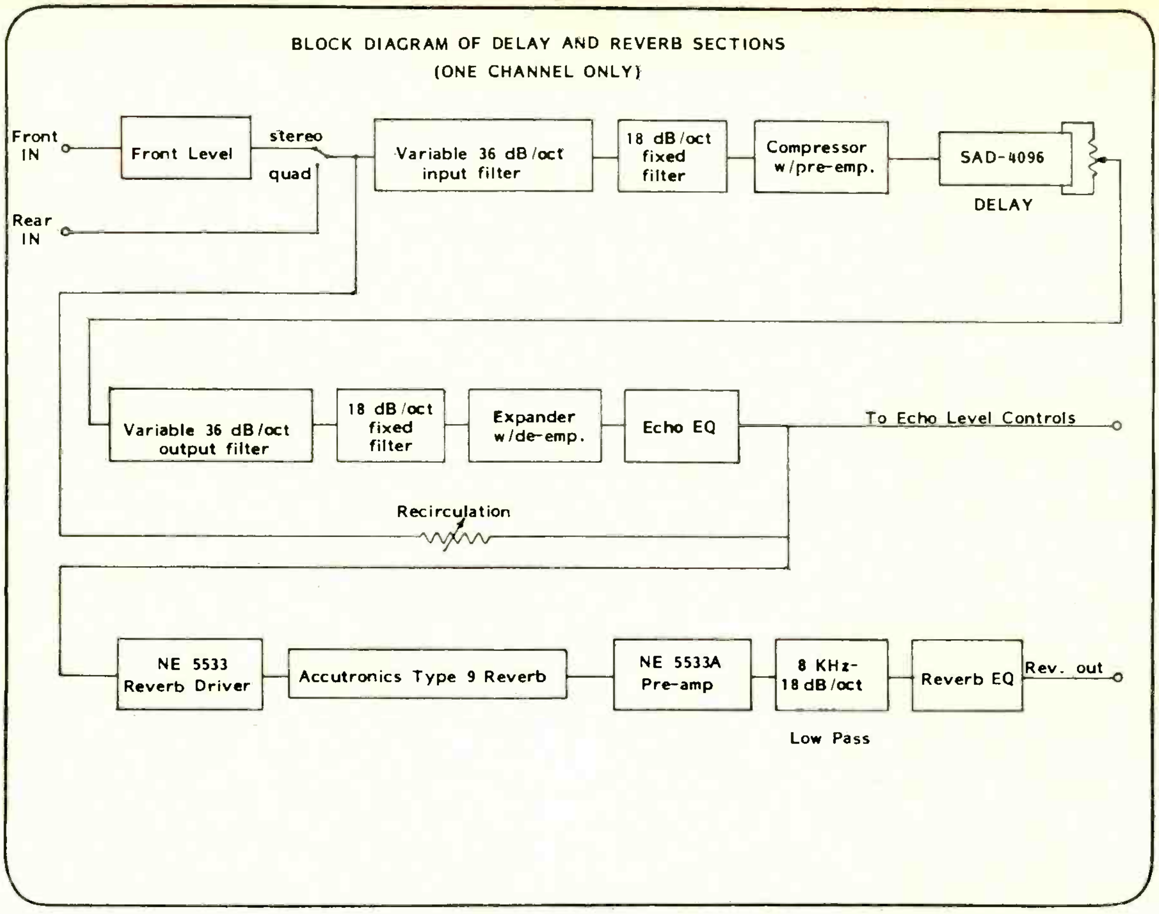



**Fig. 6: LT Sound TAD-4: Block diagram of delay and reverb sections (one channel only).**

would use the device in multi-track studios where dry close-mic tracks are the rule.

The TAD-4 also would be equally at home in a sophisticated hi-fi setup as in a well-equipped recording studio. True, these applications would require different use of the many front-panel controls but the controls are there, and once you have mastered them, the possible combinations are almost infinite.

In experimenting with the TAD-4, using music signals that ranged from classical to hard rock, it was possible to create a variety of listening "spaces" to suit each type of program source. In all of these experiments, except for the most exaggerated time delays of which the unit is capable, we detected none of the artificiality or twanginess commonly associated with less sophisticated reverb or delay units.

**THOMPSON TAD-4 ANALOG DELAY: Vital Statistics**

| PERFORMANCE CHARACTERISTIC | MANUFACTURER'S SPEC | LAB MEASUREMENT |
|---|---|---|
| Direct signal-to-noise, re: 1.0 V | 90 dB | 88 dB (95 dB "A" wtd) |
| Delay dynamic range, re: 1 V | 85 dB | 83 dB (88 dB "A" wtd) |
| THD (direct) | 0.006% | 0.004% |
| THD (delay), for 0.775 V output | 0.5% at 1 kHz | 0.25 to 1.0% (depending on delay time) |
| Direct frequency response | ±0.2 dB, 20 Hz to 100 kHz | ±0.2 dB, 20 Hz to 50 kHz |
| Delayed frequency response | | |
| 20-84 msec | 30 Hz to 10 kHz | Confirmed |
| 100 msec | 30 Hz to 8.4 kHz | per Fig. 1 |
| 150 msec | 30 Hz to 5.6 kHz | |
| 200 msec | 30 Hz to 4.3 kHz | |
| 240 msec | 30 Hz to 3.5 kHz | |
| Input impedance | 47 K ohms | Confirmed |
| Output impedance | 200 ohms (to 2 K ohm loads or higher) | Confirmed |

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# Brenell Mini 8
# Multi-Track Recorder

**By John Murphy and Jim Ford**


**General Description:** The Brenell "Mini 8" is a compact eight-track recorder featuring a professional one inch tape format and sufficient flexibility to serve as a master multi-track recorder. The record/play electronics are constructed as eight separate plug-in modules with the channel controls located on the front panel of each module. The transport is equipped with full logic and motion sensing and provides tape speeds of 15 or 7½ inches per second. Power is supplied from a separately provided power supply which interfaces with the recorder by way of two multi-conductor cables. This allows the power supply (and its noise fields) to be located remote from the transport and audio electronics. In addition the machine is provided with a remote control unit and a remote digital tape counter. The Mini 8 sells for $8,695; remote unit, $625 (approximate retail value).

The recorder stands upright and comes attractively finished with a black front panel and walnut end pieces. The transport accepts up to 10½-inch NAB reels and occupies the upper half of the front panel. There are three distinct 8-track heads for the erase, record and play functions. Each of these has adjustments for azimuth, zenith and skew, and a manually operated head shield is provided for the playback head. A high speed synchronous motor is used with a belt drive to the capstan flywheel. There is a remote drive socket on the rear of the unit which may be employed with an optionally available variable speed control.

Transport controls are arranged horizontally just below the heads and to the left side of the front panel. The controls are light-touch push-buttons and are color coded. To the right of the record button there is a red LED which illuminates whenever the machine is in record mode independently of whether or not any of the individual channels are being recorded. At the right side of the machine opposite the transport controls there is a push button for selecting tape speeds of either 15 or 7½ inches per second. To the right of the speed select switch is a power on/off push button and its associated pilot light. The remainder of the front panel is taken up by input channel controls over the leftmost three-fourths and by VU meters over the right one-fourth.

The input modules are arranged side by side so that the input, record and output controls for each channel form a vertical column. There is a red record light at the top of each module which flashes whenever the channel record switch (located just below the light) is in the "on" position but the machine is not yet in the record mode. When the machine is placed in the record mode (by depressing the transport record and play buttons simultaneously) the record light goes on solid for as long as the channel and the machine are in record. With the machine in the record mode, the individual channels can be switched in and out of "Record" as desired. The record level is set by a rotary pot under the record switch.

Continuing down the channel module the next control is a rotary output level adjust. There are no calibration marks on either the input or output level controls. Below the output level control there is a three


position line-out selector switch labeled, "Sync," "Line" and "Replay," for the up, center and down switch positions, respectively. The setting of this switch determines the source of the signal at the channel output. With the switch in the "Sync" position the source of the output signal depends on the record status of the channel. If the channel is not in record, the output will be the "Sync" signal (i.e., the playback signal from the record head). However, when the channel is in "Record" and the line-output selector is set to "Sync" then the line-in signal is provided at the channel line-out. That is, with the line-output in the "Sync" setting the source/tape switching is taken care of automatically on punch-in/punch-out. This provides the convenience of one button punch-in recording. With the line-output switch set to "Line" position the channel input signal appears at the output. The "Replay" position of the switch provides the playback signal from the playback head at the channel output. The switch would normally be left in the "Sync" position during record operations so that source/tape switching would be automatically performed as the various channels are alternately recorded and monitored. For mixdown, the channels would be placed in the "Replay" mode so that the tape playback could be monitored through the playback head.

The only remaining controls on the channel module are printed circuit card trimmers which set the record/play alignment. These trim pots are screwdriver accessible through small holes distributed up and down the module front. Adjustments include bias, record/play high frequency EQ, and record/play level calibrations, each of which is neatly labeled next to the access hole.

The eight channel modules span from left to right across the bottom of the recorder leaving space for the VU meters to the right. These (the meters) are arranged in two vertical columns of four meters each and have a standard VU response.

All of the inputs and outputs are single ended (unbalanced) by way of ¼-inch phone jacks located on the rear panel. Also located on the rear panel are the connectors for the power supply unit and remote control. Located above these connectors and the input/output jacks is the capstan remote drive socket and capstan switch. The latter may be used with an optional variable speed control. The back panel also contains a vent for a small cooling fan mounted on the inside of the panel. The fan operates whenever the unit is powered on.

The recorder's power supply unit and master bias oscillator are contained in a separate black box. In addition to the cables which interface with the recorder, the power supply has a line cord connector, line voltage selector, three fuses and a pilot light. Housing the power supply separately from the recorder is a good idea because it gets the power supply transformer and its rather strong noise fields away from the relatively sensitive audio electronics and tape heads.

The unit we received came with a remote control unit and remote tape position display. The remote control connects to the recorder through a multipin cable and connector, the remote tape position display then plugs into the remote control through a multipin connector. All of the recorder's transport controls are duplicated on the remote and in addition there is an auto return to zero feature and a counter zeroing button. The remote also duplicates the channel record on/off controls thus allowing the channels to be switched in and out of record at the remote. There is also a master line-out select which simultaneously switches all channel outputs between "Sync,' "Line" and "Replay." Most of the controls on the remote are either lighted or have an associated indicator light. The remote tape position indicator has a four digit electronic display which reads minutes and seconds (two digits each) from zero at 15 ips (at 7½ ips the counter reads half of real time). A small push button on the rear of the counter returns it to zero or it can be zeroed from the remote.

**Listening and Handling Test:** In order to prepare the Brenell for our listening test the heads were first carefully cleaned and demagnetized, then we loaded some Scotch 206 tape (for which the machine had been previously aligned) onto the recorder and patched it into our reference listening system. The unit was interfaced through a tape monitor loop in our preamp so that we could easily (at the push of a button) insert the Mini 8 in our listening chain or alternately bypass it. We initially set our record levels so that peaks occasionally drove the VU meter past "0," but we were careful to avoid "pegging" the meter. The output level was set so that there was no change in listening level when the recorder was switched in and out of the listening chain. Then we got out our favorite test discs and proceeded to listen.

The recorder's electronics were auditioned with the output select switch for the channel under test set to "Line" (source). There were some slight (but audible) changes in the high end when the recorder was switched in and out. The recorder's line out select switch was set to "Replay" (playback) to audition the record/play chain. It was observed that the record/play chain "changed" the high end more than the electronics alone. The effect was slight but some of the transient detail (particularly on snare drum and hi-hat) seemed to get lost when listening through the recorder. We also noticed a trace of harshness introduced with the recorder.

Thinking that these differences might be related to the onset of tape saturation we reduced the record level about 3 dB (and readjusted the output level for no level change when switching) and repeated the previous comparisons. This seemed to reduce the harshness and high frequency changes but did not audibly eliminate them. This tends to suggest that the best results may be obtained by using somewhat conservative record levels on the Mini 8. As an attempt to place our criticisms in perspective let us say that the audible discrepancies that we noted were slight. Indeed, some listeners might consider that the Mini 8 provides a virtual "exact copy" of the source.

The Brenell handles smoothly, its controls are clearly labeled and its operation can be learned quickly. However, we were a little surprised at the omission of a "safe" setting for the channel record switch. Once the machine is in the record mode, it is only necessary to move the channel record switch from "off" to "on" (or punch a channel record button on the remote) to start recording on a channel. We would like to see some record safety provisions that would make it harder to accidentally record over tracks of a master tape. Punch-in recording is quite easy on the Mini 8 since (with the machine in the record mode) it's only necessary to switch the appropriate module into record at either the channel module or at the remote. The unit avoids placing audible pops on the tape during punch-in/punch-out operations by turning the record bias on and off smoothly.

**Lab Test:** When the Mini 8 was brought to the lab we first performed a complete alignment for Scotch 206 tape according to the instructions given in the owner's manual. The unit was put through our test routine and the results are detailed in the "Lab Test Summary." All tests were performed at 15 ips.

With the input level control at maximum, an input level of −6.3 dBV was required for a 0 VU indication. This is fine for interfacing with +4 dBV level equipment but it may make interfacing with some −10 dBV equipment (such as many "semi-pro" mixers) a little tricky. The output delivers up to +9.5 dBV for a 0 VU level, so interfacing the output should be no problem.

Frequency response for playback of a standard alignment tape was excellent. Record/play frequency response was good with the only significant aberrations being the result of head fringing effects *(See Figure 1).* For "sync" playback through the record head the head bump was worse than normal playback giving a peak of +2.5 dB at 50 Hz. Also, the high frequency response was not as extended for "sync" playback as for normal playback. This should be of little concern to most users, however.

Noise levels were measured through a 20 kHz filter to eliminate the ultrasonic components and were not weighted. The record/play noise was fairly low being about 54 dB below 0 VU. Noise from playback of virgin 206 tape was only 1 dB lower at −55 VU and

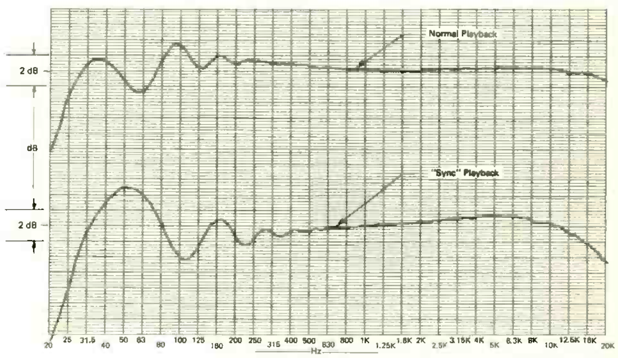


**Fig. 1: Brenell "Mini 8": Record/play response (typical of 8 channels).**

the noise from the electronics only was about 57 dB below 0 VU. These noise levels are shown graphically in Figure 2 which shows the record/play input/output transfer curve.

The total harmonic distortion through the record/play chain with record levels of 0 VU was about 1%. For a 1 kHz tone, the THD reached .3% with a +7.5 VU record level. The signal-to-noise ratio referenced to the 3% THD record level is therefore 54.2 + 7.5=61.7 dB.

The recorder's electronics have plenty of headroom and very low THD. In order to check the slewing performance of the electronics we injected a sine wave and increased the level to a point just below output clipping. The output waveform was observed on our oscilloscope as the frequency of the sine wave was increased and we observed that the waveform maintained its shape up to the small signal bandwidth (68 kHz). At higher frequencies the amplitude of the output simply decreased with increasing frequency while the waveform maintained its shape (i.e., it didn't change into the familiar triangle wave that results from slew limiting a sine wave). Thus we could never drive the unit into slewing and could not observe its slew rate limit. This indicates that the electronics have a power bandwidth at least as great as the small signal bandwidth. Based on this we calculated a slew rate

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In desperation, we agonized over what we might do to satisfy our customers and to match the increasing sophistication of our electronics. After examining all the alternatives, we decided that we must apply an old adage...."**If you want it done right, then do it yourself.**"

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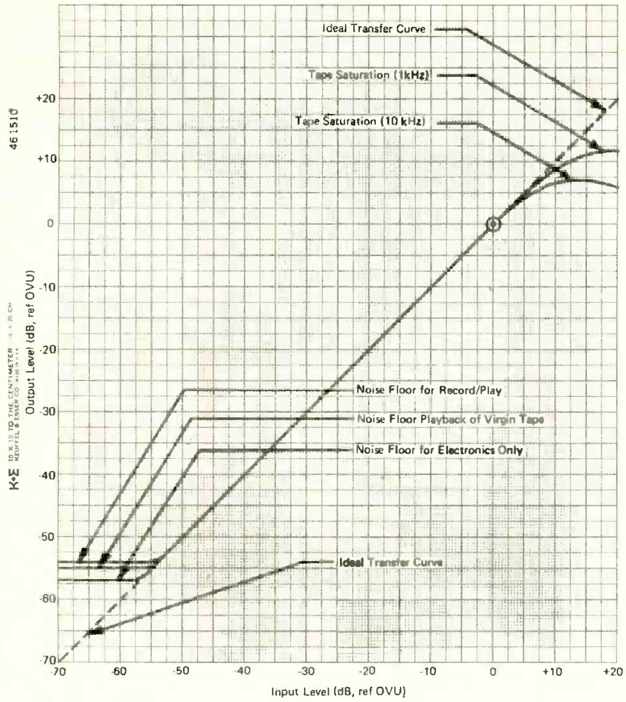


**Fig. 2: Brenell "Mini 8": Input/output transfer curve for record/playback on the unit.**

limit for the electronics of *at least* 7.17 volts per microsecond. We say "at least" because the power bandwidth could well be higher than the 68 kHz small signal bandwidth and therefore the slew rate limit could be higher than that calculated for 68 kHz. Based on this slew rate limit we calculated a slew rate ratio at 0 VU of at least 2.2 volts per microsecond per volt. We note again that a slew rate ratio of 0.5 (minimum) to 1.0 (conservative) has been recommended by researchers studying the phenomenon of slewing induced distortion.[1] Another calculation reveals that an output level of at least +12.8 VU is required before the slew rate ratio drops to 0.5. From this it seems that the Mini 8's electronics should be pretty much "slew proof." If you find the discussion of "slew rate" and "slewing" to be confusing and you'd like a better understanding of what it all means then we strongly suggest you read (and study!) the referenced article.

The owner's manual supplied with the machine contained sufficient information on installation and operation to allow a new user to interface and operate the recorder with minimum difficulty. In addition, the manual includes appendices on alignment and servicing as well as a complete set of schematic diagrams.

**Conclusion:** The Brenell Mini 8 appears to be a good little eight-track recorder. Its one-inch tape format and modular electronics provide a high level of performance with a maximum availability. We would like to see better record protection, and although we did observe some anomalies in our listening test, we feel that the audio quality of the recorder is very good.

[1] J.G. Jung, M.L. Stephens, C.C. Todd, "An Overview of SID and TIM, Part II—Testing," *Audio*, LXIII (July 1979), 38-47.

### LAB TEST SUMMARY

(Note: 0 dBV is referenced to .775 Vrms;all tests performed using Scotch 206 tape at 15 ips.)

**Input Level**
Minimum input level for 0 VU indication: – 6.3 dBV (375 mVrms)

**Output Level**
Maximum output level at 0 VU: + 9.5 dBV (2.31 Vrms)

**Playback Frequency Response**
(from standard alignment tape)
Play response through play head: + 1, – 2 dB 30-15 kHz
Play response through "sync" head: + 1.5, – 2dB 30-15 kHz

**Record/Playback Frequency Response**
Frequency response at 0 VU, 15 ips: ± 1 dB 112 Hz to 23.5 kHz
Bandwidth ( – 3 dB points): 23 Hz to 25.8 kHz

**Noise Levels**
(20 kHz filter,unweighted, 15 ips)
Record/Playback: – 54.2 VU
Playback of virgin tape: – 55.1 VU
Noise from electronics only: – 57.1 VU

**Record Flux Level**
0 VU is *320 nWb/m* ( + 4.76 dB with respect to Ampex operating level of 185 nWb/m)

**Total Harmonic Distortion (plus noise) for Record/Play at 0 VU**
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Record level for 3% THD @ 1 kHz: + 7.5 VU

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Output waveform clipping occurs at: + 23.7 dBV

**Total Harmonic Distortion (Electronics) at 0 VU**

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|---|---|---|---|
| 0 VU: | .085% | .049% | .049% (mostly noise) |
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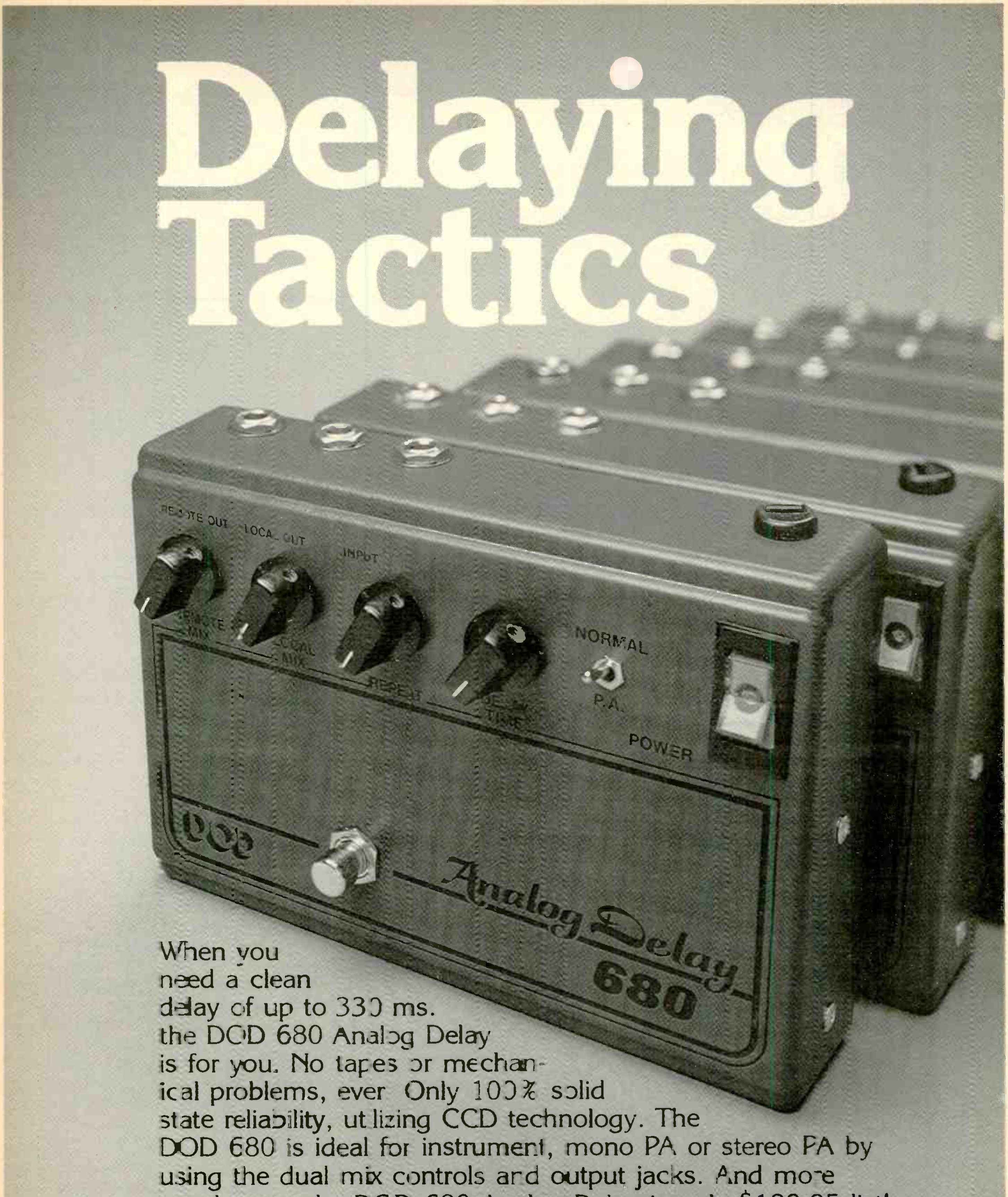


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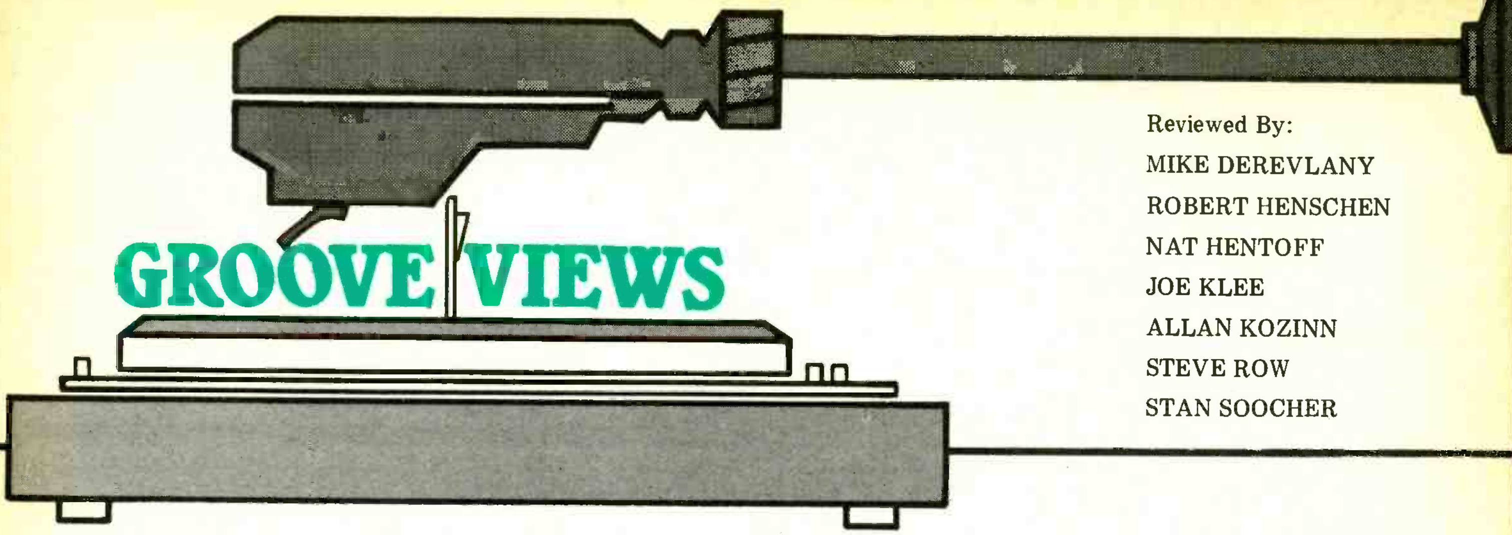

# GROOVE VIEWS

Reviewed By:
MIKE DEREVLANY
ROBERT HENSCHEN
NAT HENTOFF
JOE KLEE
ALLAN KOZINN
STEVE ROW
STAN SOOCHER

## POPULAR

**AVIARY:** ***Aviary.*** [Produced and engineered by Gary Lyons; David Giorgini, assistant engineer; recorded at Caribou Ranch, Nederland, Co. winter 1978.] Epic JE 35716.

Performance: **Intelligent, though somewhat derivative**
Recording: **Very polished**

The latest American entry into the pseudo-British art-rock field is a quintet from Seattle producing a complex mixture of musical material that seems to be drawn heavily from Beatle influences. At present, Kansas may be the only American group to be successful with the arty type of rock that Yes, Genesis and Pink Floyd have practiced; Aviary could become a contender with some more seasoning.

The album contains nine tracks, each one rather intricately plotted and charted to show off considerable skill at arranging and composing. The resulting mix is occasionally dense, occasionally thick and not always in keeping with the airy implications of the group's name.

The singing could be the one drawback to the success of the record, because too much of it is done in falsetto harmonies. The chorus of the title track, for example, is sung in such a way as to begin to grate against the audio nerves by the time the song concludes. Coupled with the crashing accompaniment, the song takes on a pretentiousness that the group probably did not intend.

Some of the effects in the album are very interesting. The listener will find a little bit of Supertramp or Electric Light Orchestra in the campy "Anthem for the USA," and some vintage George Martin-guided Beatles sounds in the whimsical "Puddles." "Anthem" contains such devices as English accents and a chorus sounding as if it were sung through megaphones. The whole song is very busy and could be viewed as a counter to "Back in the USSR"—musically, if not philosopically. "Puddles," with its witty oboe (?) introduction a la "When I'm 64," is really an oddball song, dragged down ultimately by its whiney falsetto chorus (again!), but still not without some interesting musical elements.

The quintet consists of Brad Love, principal lead singer and composer of all the tracks; Paul Madden, keyboards; Ken Steimonts, bass; Toby Bowen, lead guitar, and Richard Bryans, drums. Bowen's guitar work is uniformly good, and shines especially on a break during the title track. Bryans' drums are prominent in nearly every work as well.

Presentation aside, the group may rise or fall on the strength of its material, and Love's material has a degree of sameness that one notices by the end of the album. The rhythms tend to sound alike in the majority of songs—very choppy, leading to insistent singing and playing—and the instrumentation is thickly layered. Love, whose early influences are listed as Led Zeppelin, Uriah Heep and Deep Purple (watch out for such a combination), apparently changed directions during college and began to favor romantic classicists (Chopin, according to the studio publicist).

If this is the case, the Aviary's music (really, Love's music) has yet to achieve some of the delicacy and simplicity that classical composers achieved. The material shows a debt to musical skill and knowledge, but not necessarily to creative innovation.

Some of this criticism could be brushed off in the context of a debut album,

AVIARY: A nice surprise—imagination and an abundance of talent

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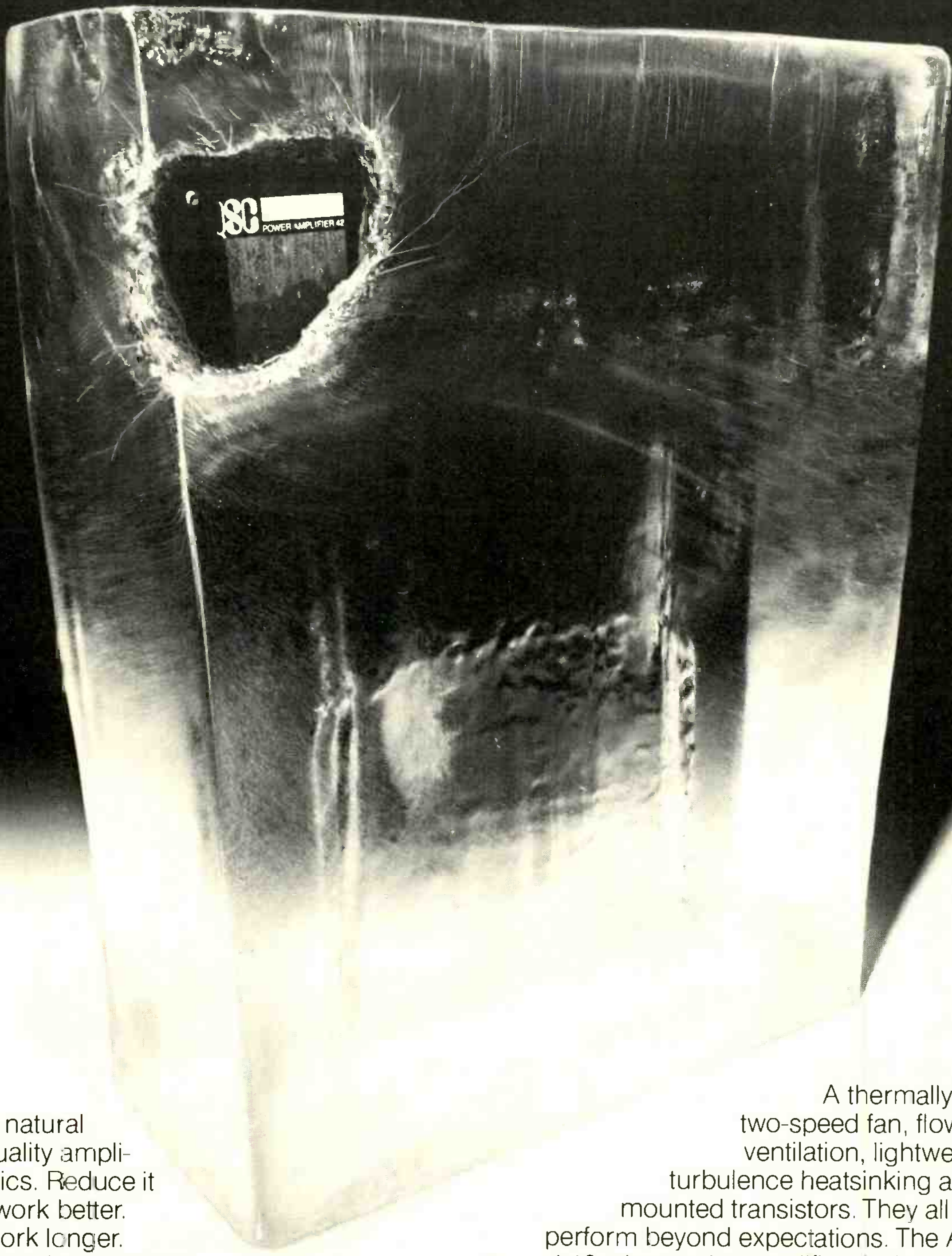

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however. Those who haven't played their Pink Floyd or early Yes albums for some time, for example, might be pleasantly surprised at what they hear in Aviary's work.

The group's debut album is interesting, well-recorded and slightly flawed. But there is considerable promise here, an abundance of talent and enough imagination to warrant our following the group in the future. S.R.

**BILL BRUFORD: *One of a Kind.*** [Bill Bruford, producer; Steve W. Tayler, engineer; recorded at Trident Studios, Soho, London, January-February, 1979.] Polydor PD-1-6205.

Performance: **Spectacularly dull**
Recording: **About 95% of what it could be**

The image most usually associated with drummers is the limousines-in-the-swimming-pool, hotel-wrecking rowdyism of the late Keith Moon. On the more subdued side, there are those drummers who despite somewhat less disruptive natures are no less demanding of attention, particularly in their solo works, as evidenced by Levon Helm, whose solo efforts demonstrated an unescapable ability to obscure every instrument with percussion. It's the absence of antics and the lack of egotistical excesses that distinguish Bill Bruford from most other drummers.

Bruford further distinguishes himself by his tireless efforts to thoroughly expose his weaknesses and failings. Take, for example, his inability as a writer. Bruford is responsible, either directly or as a co-conspirator, for eight of the ten cuts on *One of a Kind.* The remaining two, written or co-written by members of his recording backup group, are equally unpalatable.

On the positive side, it must be mentioned that despite its dearth of artistic accomplishment, the album is an outstanding recording. There is a readily observable element of calculated effort that is based on much prior studio experimentation. Bruford, whose work with Yes and King Crimson has undoubtably heightened his awareness and influenced his style as a producer, has put together a work with great technical strengths and capably demonstrates his ability to use the studio. The mix, although fairly standard, is clean and well-balanced, without any awkward instrumental emphasis.

About the only negative technical aspect is the fact that Bruford relies heavily (and somewhat lamely) on electronic instruments and effects; there is much synthesizer and many instruments are electronically altered and distorted, often with spasmodic results. Under the right circumstances these can be effective; however, neither Bruford's methods nor their effects are original, unusual, or even appropriate. *One of a Kind* is a misnomer; it's one among many of the wrong kind. M.D.

**JONATHAN RICHMAN AND THE MODERN LOVERS: *Back In Your Life.*** [Matthew King Kaufman, Glen Kolotkin, and Kenny Laguna, producers; Glen Kolotkin, engineer; mastered at Electra Sound Recorders.] Beserkley BZ-10060.

Performance: **Pointedly unpolished**
Recording: **Intentionally sloppy**

Is the Beserkley ideal a minimalist reaction to the technological excesses of modern recording? Are anti-stars like Jonathan Richman, the Rubinoos, and Greg Kihn prime examples of rock's current return to basic instrumental values? Or is this music all a joke?

If this new Modern Lovers album is any indication, the answer to all of the above questions is a blanket YES. This is intentionally sloppy engineering of pointedly unpolished material. Jonathan Richman is a worse singer than either David Bromberg *or* Wildman Fisher, and damn proud of it. His bandsmen are equally unpredictable.

But seriously folks, this music is a joke from the outset. "Abdul And Cleopatra" is an obvious farce, "(She's Gonna) Respect Me" is sort of a reverse psychology Fifties doo-wah torcher, and "Lover Please" is a crude attempt at one of those old California twang guitar epics. These are just the first three songs!

Richman's oft-demented humor takes over completely, poking fun at nearly every conventional song form, from outrageous rockabilly ("Party In The Woods Tonight") to mock Sixties sensitivity ("Back In Your Life") and ridiculously clichéd love ballads like "My Love Is A Flower (Just Beginning To Bloom)." Richman writes absurd novelty lyrics ("I'm Nature's Mosquito") and covers golden oldie oddities such as "Buzz Buzz Buzz" and "Lydia."

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But if Richman is going strictly for laughs, the plan backfires on cuts like "Affection" and "Emaline." Delivered in an offhand style with pouting voice, "Affection" is a makeshift lament about the lack of love in the world. Somehow, the song is so humanly fallible and so musically imperfect that it works. The rockers here have plenty of subliminal appeal too.

This seems to be the hidden import behind much of what Beserkley does. On one level, the Modern Lovers are amateurish comedians, neither overly witty nor overtly avant garde. Their portrayal of bygone music forms is outlandish and interesting, possibly defamatory as well. But if you don't take Richman and his colleagues too seriously, and see this music as half parody (half tribute), *Back In Your Life* is recklessly enjoyable and lotsa fun.

R.H.

**THE FARAGHER BROTHERS:** ***Open Your Eyes.*** [Vini Poncia, producer; Joe Bellamy, engineer; recorded at Mama Jo's Recording Studio, North Hollywood, Ca., mixed at Sound Labs, Inc., Hollywood, Ca.] Polydor PD-1-6167.

Performance: **Southern California assembly line**
Recording: **Clean but unadventurous**

I didn't have to look at the credits to know that *Open Your Eyes* was recorded in southern California, and if I could have had five guesses as to who produced the album, Vini Poncia's name would have been one of the five. The Faraghers are the latest in a seemingly endless line of soundalike California bands which have defined the music of that area in the late '70s—or at least the commercial side of it. Those airy harmonies, laid-back percussion/funk guitar interchanges, and subdued, bouncy rhythms are all here, and if Polydor promotes this record as they might, they should have a couple of AM hits on their hands.

There isn't much to a record like *Open Your Eyes*, but there isn't supposed to be. It follows the formula, is clean and pleasant to listen to, and is about as inviting as a pair of roller skates in a snow storm. Blandness is the key, and if one simply requires his music to be unthreatening and unchallenging, then it's a safe choice.

That all said, *Open Your Eyes* is not a

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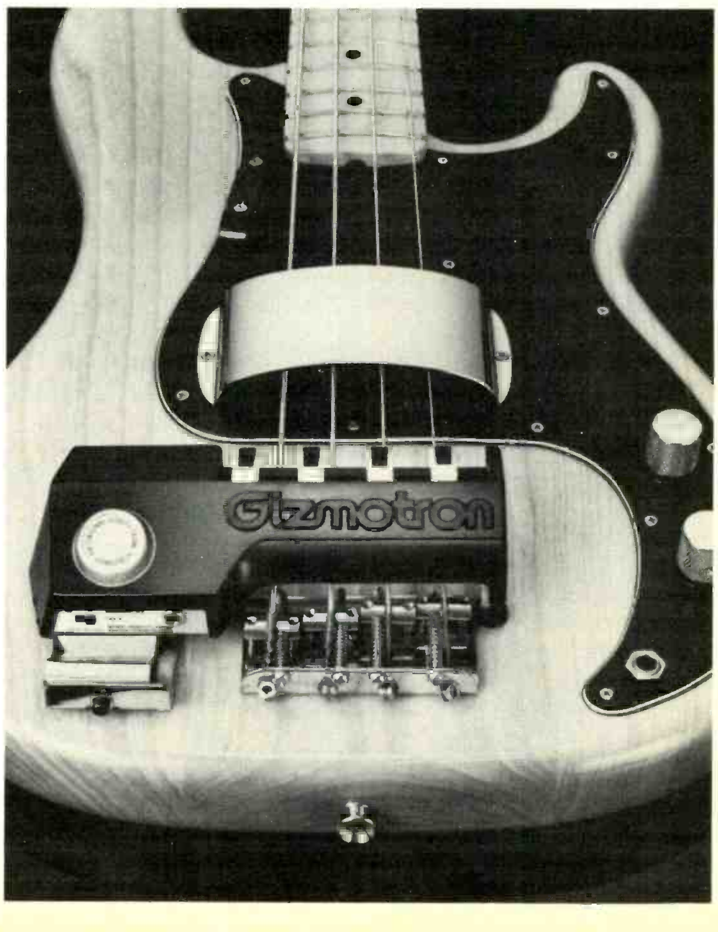


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bad record. The five brothers and occasional sister possess naturally smooth, if less than dynamic voices, and their songs are all well-crafted and melodic. The musicianship doesn't display any particular virtuosity, but it's as competent as need be. Lyrically, there's not much to qualify the Faraghers as great songwriters of the '70s, but they're nice songs—pretty, and mostly about love.

Vini Poncia has made a name for himself as a resident hitmaker in the southern California studio wars, and he hasn't done it by stepping too far off into space with his ideas, but by sticking to the notion that people will buy records that are consistent with what they're already used to buying. And so, record after record, Poncia stays close to established hit lines, and if he fails once in a while in his judgement, he'll just know better next time to make it sound closer to the guaranteed hit sounds that *have* worked. Poncia may not go down as a producer with much daring, but if he keeps churning out these formula discs, he's going to be too rich to care what he goes down as.

The Faraghers' album combines standard MOR tricks with some classic soul music staples and the naturally complementary voices of a bunch of siblings to create its distinct non-identity. Traces of the music of Boz Scaggs, the Spinners and assorted Motown acts pop up, and there's a toned-down disco beat here and there. But mostly, *Open Your Eyes* is just a nice record—unobtrusive and uninventive—and one which will make you happy without making you stretch your lips too far to smile. J.T.

**PATTI AUSTIN: *Live At The Bottom Line.*** [Creed Taylor, producer; David Palmer and David Hewitt, engineers; recorded by Record Plant at The Bottom Line, New York, N.Y., August 18-19, 1978.] CTI 7086.

Performance: **Controlled soul**
Recording: **Small problems**

CTI is open for business again, and Creed Taylor unveils a whole new storefront in this Patti Austin performance. A well-rehearsed act, with plenty of high octane personnel in support (David Spinozza, Michael Brecker, Pat Rebillot, Leon Pendarvis, etc.), Austin's show is nevertheless a non-jazz display of out and out vocal soulfulness—not the overly slick kind of pseudo-jazz production CTI was once notorious for.

Plenty of vaunted "background

singers" have been getting their chance to step out front and make their own albums of late. Lani Hall (A&M), Thelma Houston (Tamla), The Jones Girls (Philadelphia Int'l), Tasha Thomas (Atlantic), May McCreary-Russell (Paradise), and Randy Crawford (Warner Bros.) have all been adding to their blossoming solo credits, and Valerie Carter, Dee Dee Bridgewater, and Leah Kunkel could be on the verge of bonafide stardom. Patti Austin's vocal chops are not only comparable to the above, they're established beyond the shadow of a doubt through many years of studio work, primarily in the New York area.

Austin belts with the best of them on "Jump For Joy" and "Let It Ride," and is particularly fine on soulful ballads such as "Love Me By Name" and "Let's All Live And Give Together," her dusky-voiced tenderness flirting with Brecker's sultry sax. But even Patti Austin can't transform an average tune like "You Fooled Me" or the too-familiar "One More Night." Despite the prestigious support musicians, there is painfully little room for any kind of instrumental stretching. The entire date is very professional, but so expertly arranged that limits are set on how far the music can go.

Austin gets pretty loose herself, since the charts are understandably catered to her abilities. Perhaps the most intimate moment on the album comes on Patti's spoken intro to "Riders In The Rain," Randy Newman's cowboy dream song. "I wanna prove to everyone here tonight," Austin deadpans over a twanging guitar, "that black women *can* sing country-western music." Her humor gets the audience up and this comic relief is conspicuous by the flexibility it gives to the entire session.

The Bottom Line may be an outstanding club, but it is not necessarily the ideal recording environment. Background noise and echo is a problem on the tunnel-sounding "Jump For Joy." A suddenly spliced ending to audience applause on "You Fooled Me" is downright amateurish. In all fairness, Record Plant's remote engineers have gotten a decent mix on Austin, her three background singers, and a seven piece band, as well.

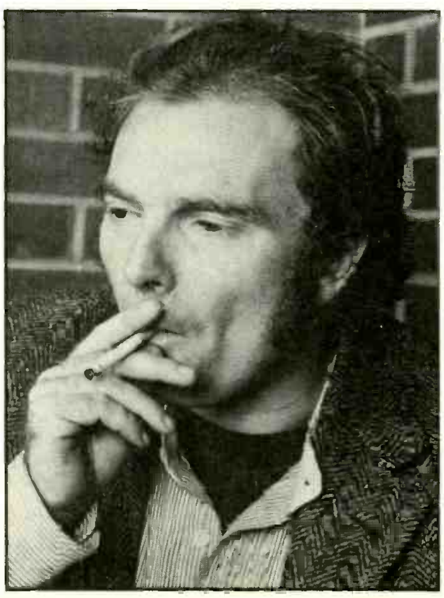
This is a good showing by Patti Austin as a solo vocalist, and an occassionally entertaining assortment of tunes. But weak material intrudes on what could have been a real blockbuster. R.H.

**VAN MORRISON:** ***Into The Music.*** [Van Morrison, producer; Mick Glossop, engineer; recorded at the Record Plant, Sausalito, Ca.; mixed at the Automatt, San Francisco, Ca.] Warner Brothers HS 3390.

Performance: **Spunky, but not always inspired**
Recording: **Busy mix; catchy results**

At the peak of an outrageous early '70s performance, I remember seeing Van Morrison vomit between a stack of amplifiers during a rousing rendition of "Domino." Then last year, after turning the intimate setting of New York's Bottom Line into his own near-hysterical revival meeting, he walked off the Palladium stage the next night, only a few songs into the set. In dealing with this inconsistency in his character, Lester Bangs suggested in his *Rolling Stone* review of Van's *Wavelength* album that Van's intent is to "make the edge the center." And indeed, when listening to *Into The Music* it becomes necessary to look past the overall effect of the body of music Van has produced here and concentrate on the exhilarating moments that make this his strongest album in the last few years.

Admitting to his inconsistency on *Into The Music*'s opening cut, Van sings about "the dark end of the street to the bright side of the road," and is perhaps referring to the contrast between the "bright," uptempo material on side one and the "dark," drawn-out side two. Side one includes the playfulness of Van's harmonica fills on "Bright Side Of The Road;" his testament of religious redemption in "Full Force Gale;" his belief in music as a liberating force in "Troubadours;" the tying together of religion, writing, dancing, and love in the pastoral "Rolling Hills;" and the clever double entendres of "You Make Me Feel So Free." Side two starts out strong with the haunting "Angeliou" and the reassuring "And The Healing Has Begun," which is reminiscent of Van's classic *Astral Weeks.* But segueing the standard "It's All In The Game" into "You Know What They're Writing About" only makes a bad interpretation into an ordeal. In most of Van's songs, the bluesy, roaring undercurrent of his voice repeating one line or syllable over and over can move the listener as forcefully as his best lyrics. But on these last two songs, lyrics and syllables disintegrate into an aural hangover.


VAN MORRISON: An aural hangover

As a producer, Van throws in a little of everything, from Ry Cooder's slide guitar to Robin Williamson's penny whistle, while still managing to capture a bright, clean brass sound. The violin, viola, and stroviola of Toni Marcus marks Van's most prevalent use of strings since his "live" album, *It's Too Late To Stop Now*, with the Caledonia Soul Orchestra. While effectively drawing upon the "gypsy soul" for most of the string arrangements, Marcus's playing is partly responsible for making "It's All In the Game" too monotonous.

*A Period Of Transition* in 1977, Van's first album in three years, was a safe return to the music scene. *Wavelength* was consistently good, but too glossy. So, considering its problems, *Into The Music* is the most rounded, sobering look we've had at Morrison in a while. As he explains in "Stepping Out Queen": "Well you go through the drama/And you work in the dharma/Then you stand up and wipe your mirror clean." S.S.

## JAZZ

**PETER DEAN: *Only Time Will Tell.*** Peter Dean, producer; Mike Barbiero, engineer; recorded November and December, 1977, at Mediasound Studios, New York, N.Y.] Inner City 4002.

Performance: **The Dean of good time, old-timey music**
Recording: **Good, but not really honest**

Peter Dean, former manager for Peggy Lee, Dinah Shore and Paul Whiteman and famed jingle producer, is also a performer in his own right. His style as a purveyor of fun-type jazz goes back to the tradition of Cliff "Ukulele Ike" Edwards. Those of you who haven't heard Edwards can hear him on a Yazoo reissue (L 1047). Peter Dean's repertoire is the same sort of dandy old tunes of the 20s and 30s such as "True Blue Lou" and "I Want A Little Girl." The first noticeable difference in performing styles between Peter Dean and those who did these numbers originally (such as Ben Pollack's band and McKinney's Cotton Pickers) is a tendency on Dean's part to adopt slower tempos. It works marvelously at the slower tempo especially on tunes like "True Blue Lou," a marvelous hit that Richard Whiting, Sam Coslow and Leo Robin wrote for film star Hal Skelly in the 1929 Paramount film, *Dance Of Life*. My only objection to Dean, as a performer is his tendency to play down his artistry on the ukulele. I've heard Peter really get off some virtuoso hot licks on the instrument in a class with those Cliff Edwards played on his old Columbia and Pathe/Perfect 78s. Peter, too often for my taste, defers to his all-star accompanists (Dick Hyman, Sam Parkins, Mike Peters, Ed Polcer, etc.) who, while they are the cream of the crop, still don't have that old-timey feel that Peter's uke does.

Another interesting facet of Peter Dean is his ability to write songs in the 20s and 30s style. There are two excellent examples here: "Only Time Will Tell" and "Baby Baby Baby." In fact, the only songs I don't particularly like on the album are from the '40s, a notoriously low decade for Tin Pan Alley, "On A Slow Boat To China" and "Talk To Me." But the wonders Peter Dean and company work with Harold Arlen's 1934 classic "As Long As I Live" (and if you've never heard Harold Arlen's record of this tune backed by Eddy Duchin's Orchestra, too bad for you) more than makes up for the inclusion of these '40s hit parade epics.

The sound is good enough for all intents and purposes but it's not really an honest recording. Peter Dean and the rhythm section tracked their parts long before the horn section of Sam Parkins, Ron Odrich and Ed Polcer were called in to sweeten the icing on the cake. Parkins, in particular, plays his heart out and sounds closer to the late Chu Berry than any other living saxophonist I can think of. But how much better it all would have come off if Peter Dean could have had the inspiration that Sam's booting playing can offer, right there in the studio with him. I guess over-overdubbing is okay for your rock superstars but a jazz artist is better if he can interact with the band playing around him—and how's he going to interact if they're playing a month later in a different studio? J.K.

**FARRELL MORRIS:** ***Bits of Percussion and Jazz.*** [Tom Semmes, producer; Peter Jensen, Terry Tobias and Louis Nanassay, engineers; recorded at Studio A, Woodland Sound Studios, no date listed.] Audio Directions AD 102.

Performance: **Bits of percussion and jazz, literally**
Recording: **No argument with the digitals**

If ever a recording was aptly titled, it's this one. There are bits of percussion played by Farrell Morris—either overdubbing himself or he must be a centipede—and bits of jazz, mostly from guests like Stan Getz and Ron Carter but also from a new name, to me at least, trumpeter George Tidwell. Most of the rest of the group falls into the category of Nashville session players and they, for the most part at least, do their job well, but without particular distinction.

One exception to this is also my only real complaint with this recording, so let's get it over with. Buddy Spicher is a much admired and in demand country & western fiddler but he's like a fish out of water trying to deal with Charlie Parker's "Moose The Mooche," Bird's dedication to his West Coast drug connection. First of all, Bird's tune ill-fits the fiddle and there also seems to be a total misconception on the part of Spicher, and perhaps Morris, that this tune is some sort of Latin carnival piece. That's certainly not what Bird wrote—for an example of how Bird wrote it, felt it, played it, check out Warner Bros. 2WB 3198 *The Very Best of Bird*

# the texas tornado and a swinging depression

**By Nat Hentoff**

Arnett Cobb was the rocking phenomenon of the Lionel Hampton band in the 1940's, swinging even more exultantly than the leader. In recent years, he's been based back home in Houston, although he occasionally works Europe and such places as Sandy's Jazz Revival, a club in Beverly, Massachusetts. It was at Sandy's in August, 1978, that an historic jam session took place. For the first time, Cobb was in the same front line as fellow Texan Buddy Tate and Eddie "Cleanhead" Vinson (the latter playing alto rather than singing). And the rhythm section was one of the most powerful and mellow ever recorded—pianist Ray Bryant, bassist George Duvivier, and drummer Alan Dawson.

Fortunately, the deep-diving pleasures of that date were not limited only to those in attendance. It was taped and now has been released on Muse as *Live At Sandy's: Arnett Cobb and the Muse All Stars.* The indestructible Cobb is the dominating horn soloist; and on the opening "Just A Closer Walk With Thee," he creates a classic performance of jazz "preaching"—riding enormous waves of excitement that keep getting higher and higher. On the other tracks, Cobb is characteristically hot in sound and attack, still playing a "talking" horn in that he's telling stories, not just running notes. Also extraordinary throughout is pianist Ray Bryant who plays here with an energy and crisp cohesiveness that have been absent from some of his other recent recordings.

The sound quality has the impact of a truly "live" session—clear, spacious, and vivid.

Like most jazzmen of his generation, Arnett Cobb broke into music by playing in territory bands. Few are left, but one of the more unusual is The Widespread Depression Orchestra which, for the past seven years, has been playing college and other dance dates throughout New England and New York State, as well as in the Apple itself. What makes this nine-piece combo unique is that its repertory is composed of vintage numbers associated with Duke Ellington, Jimmie Lunceford, Cab Calloway, Count Basie and others of the truly original bands during the so-called swing era.

This, to be sure, is revivalism, but the playing is not imitative. Instead, these relatively young musicians are primarily recreating the spirit of such legendary numbers as "Topsy," "Reefer Man," "Downtown Uproar," and "East St. Louis Toodle Oo." The lithe, flexible scores allow a lot of room for solos, and the improvisers, while mindful of the roots of this music, clearly sound their own voices. They are particularly skillful in getting at some of the essence of Ellington—the hardest orchestra in the world to approximate in any way at all. They are no substitute for Duke, of course, but they do show that his scores can be interpreted and played with knowledgeable affection and verve.

The engineering is first-class, a model of balancing without in the least blurring or overemphasizing any element of the band. Considering the vast possibilities in the repertory of the 1930's and 1940's, the Widespread Depression Orchestra should thrive for decades to come.

**ARNETT COBB: *Live At Sandy's!*** [Bob Porter, producer; Pat Costa, engineer.] Muse MR 5191.

**THE WIDESPREAD DEPRESSION ORCHESTRA: *Downtown Uproar.*** [Bernard Brightman, producer; Les Paul, Jr., engineer.] Stash ST-203.

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(*Modern Recording*, December 1978). Mercifully, Spicher's misconceived version of Parker's "Moose" is the shortest track on the LP.

Stan Getz is the sort of self-contained player whose complacency can often lead to boredom except when an unusual challenge is thrown at him. Farrell Morris' imaginative use of percussion, such as his marimba solo on "A Child Is Born" elicits some of the most exciting playing I've heard from Getz since his *Focus* album on which Eddie Sauter wrote the charts for the string section.

It was also rather daring of any jazz act to tackle a piece so closely identified with another jazz artist as "Someday My Prince Will Come" is with the Dave Brubeck Quartet. The connection does not escape Getz whose solo opens with a particular feeling of dedication to the late Paul Desmond—like Getz, a lyrical player rare to jazz circles.

If you want to really appreciate the advances brought about by digital recordings, try comparing some of the new gems with some of your old classic favorite records, even the early LPs. As great as Brubeck and Desmond played "Someday My Prince Will Come," there isn't the natural presence to their 1956 LP, even when rechanneled for stereo, that you'll hear on this latest rendition by Farrell Morris and Stan Getz. They tell us this was recorded on a Sony PCM 1600 digital recorder and, when they start to identify the recording equipment on the LP sleeve, one must assume the age of technocracy has arrived. There's nothing wrong with that until we start considering the Sonys and the 3Ms and the whatevers more important than the creative artist whose work it is their job to preserve. If it weren't for Farrell Morris and Stan Getz and George Tidwell, the Sony PCM 1600 wouldn't have had anything to record. J.K.

## CLASSICAL

**CHAVEZ: *Piano Concerto; Piano Preludes No. 1,2,5,8,10.*** Maria Teresa Rodriguez, piano. The New Philharmonia, Eduardo Mata, conducting. [Howard Scott, producer for the concerto; recorded in Mexico. No other information given.] RCA ARL1 3341.

Performance: **Not always up to the music**

Recording: **Should have been better**

Carlos Chavez is to Mexican contemporary music what Aaron Copland is to music in this country. Both seemed to really flourish in the Forties, when nationalism in music struck a responsive chord, even if what was thought of as modernism didn't; and both were therefore able to forge paths for new music, bringing composers and audiences closer. Chavez gained a certain amount of support in this country—from Copland among others—but his output, while it contains some masterpieces, was not consistently great, and with a very few exceptions, one rarely hears much of his music nowadays.

The *Piano Concerto* dates back to 1940, and when it was given its New York premiere, two years later, it evoked a split reaction: to some it was a piece of miraculous orchestral manipulation, while to others it was offensive noise. Today, of course, it strikes us as a conservative work, and this new recording throws some light on both of those early reactions. Some of it—like the intensely building second movement, which is reminiscent of sections of Messiaen's *Quartet for the End of Time;* or the rhythmically active finale—is quite attractive. Elsewhere—in parts of the first movement—orchestral complexity seems to have been taken too far, leaving the pianist to play with extreme aggression in order to be heard at all.

In the present recording, actually, that's not the problem. We hear the pianist when she plays, and when the harp plays, it too is right out in front, louder than the rest of the orchestra no matter how thick the scoring. And that sort of thing seems to call the whole production into question. Surely this isn't what the work sounds like in the hall—it is, rather, an artificial sounding mix in which instruments are highlighted at the expense of true balance. Which is too bad, because this is an interesting enough work to hear on its own non-tampered with terms.

As for the performance, the New Philharmonia sounds a bit scratchy, although Mata holds the proceedings together well. Maria Teresa Rodriguez's playing in the concerto seemed largely monochromatic. However, in the five *Preludes* that round out side two, she sounds much more agile. These are interesting little pieces, sometimes inward looking, but mostly rather on the rollicking side. Unfortunately, the piano sound is on the fuzzy side. A.K.

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