Stalking the Wild Studio Customer

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

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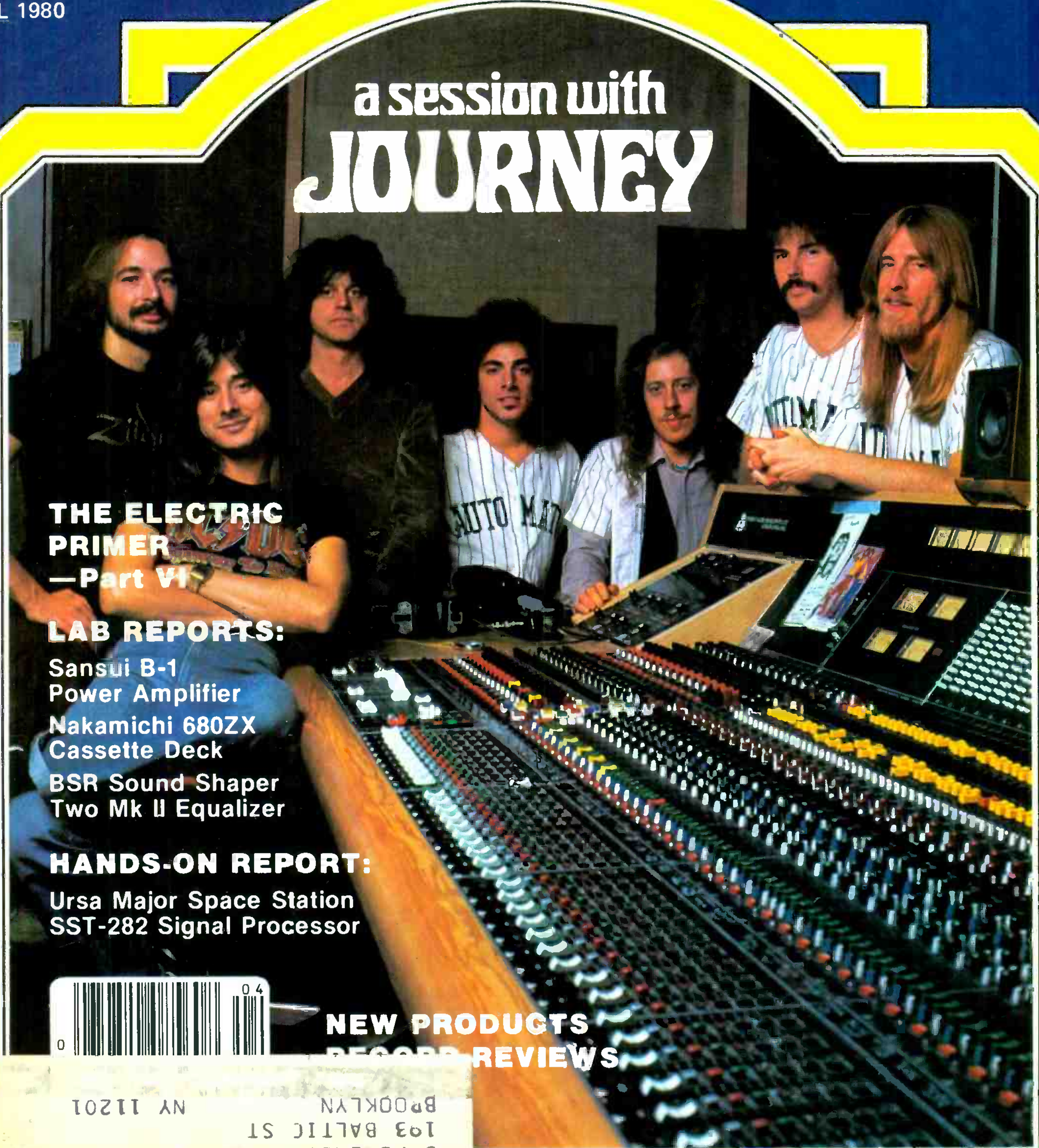
VOL. 5 NO. 7
APRIL 1980



a session with JOURNEY

THE ELECTRIC PRIMER —Part VI

LAB REPORTS:

Sansui B-1 Power Amplifier

Nakamichi 680ZX Cassette Deck

BSR Sound Shaper Two Mk II Equalizer

HANDS-ON REPORT:

Ursa Major Space Station SST-282 Signal Processor

NEW PRODUCTS

RECORD REVIEWS

How serious are you about a power amp?

We build our Professional Series power amplifiers as if our reputation were at stake. Because it is. And so is yours, when you select an amplifier. That's why you should consider Yamaha power amps. They come through for both of us. Because we both designed them. Comments and suggestions from professionals like yourself were incorporated into the final design. As a result, Yamaha power amps excel in the areas that can make or break a power amp—performance, reliability, and flexibility. Take the P-2200 for instance.

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adaptor-free connection to any mixer. A polarity switch satisfies DIN/JIS or USA wiring practice. The P-2200 is readily suited for monaural operation as well as 70-volt commercial applications.

The P-2201 is identical to the P-2200 except it does not have the peak-reading meters. The P-2100 and the P-2050 differ primarily in rated power output and size. Each model offers the maximum in performance, flexibility, reliability and value for the dollar in its category.

We have a technical brochure covering all four models. Write Yamaha, P.O. Box 6600, Buena Park, CA 90622. Or better yet, visit your dealer for a demonstration of the Yamaha power amps that take their job as seriously as you take yours.

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YAMAHA

CIRCLE 99 ON READER SERVICE CARD

Studiomaster is Expanding

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

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

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

CIRCLE 83 ON READER SERVICE CARD

No More Waiting for Louie

Louie's a great drummer, and you've been playing together for years, so he must be good. But in all the time you've known him, he's never been on time for practice. So you've always had to wait for him to arrive before you can start to play; and now that you're in the studio that's even more of a problem, because as we all know—nothing can happen without the rhythm track.

Well, now you can spend your waiting time a little more productively by using a Roland Computer-Rhythm. The Compu-Rhythms are two astounding devices that produce accurate and dynamic rhythm tracks that can be used in recording, performing, or just practicing.

Both the CR-68 and the CR-78 provide a variety of rhythms that can be combined and varied over a wide range of tempos—but the real beauty is in their sound. Though the sound is electronically created, it is unbelievably rich—the snare drum

crisply pops and the bass drum punches as if they were heavily compressed.

Two unique features are the Accent control, which gives an added realism to the program tracks, and the Variation control, which produces 11 different drum fills that can be programmed to play at desired intervals.

In addition to all of the other features, the CR-78 Compu-Rhythm contains a micro-computer that allows you to program your own (or Louie's) drum tracks into the machine by tapping them out on the programmer pad, and a built-in battery keeps them in storage even if you turn the Compu-Rhythm off.

The Compu-Rhythms will find a lot of places in your music—from writing music to laying down click tracks for multi-track recording. They can even provide trigger outputs for external control of synthesizers of digital sequencers.

The Compu-Rhythms may not replace the drummers of the world, but they're going to make it a lot easier to live with their little inconsistencies.

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CIRCLE 144 ON READER SERVICE CARD

APRIL 1980
VOL. 5 NO. 7

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

THE FEATURES

THE STAPLES

COMING NEXT ISSUE!
Chuck Mangione "Live!"
Profile: Synthesist Larry Fast
The Electric Primer—Part VII
Plus MR's other great monthly features

Cover Photo: Pat Johnson
Journey Photos: Pat Johnson and M. Marks
Primer Illustrations: Peter Weiss

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The Publisher's Track

Notes is Coming!

For some time now, I have been sitting around talking to musicians and recording engineers about a subject near and dear (how "dear" where the engineers are concerned is debatable!) to them: special effects, specifically for the guitarist.

In the past two years especially, footpedals and other units have proliferated at a mad pace. It has become difficult for even the media to keep tabs on what's available. Mind you, I am not complaining—the flexibility offered by these units is wonderous, and the technology, quite simply, is the future now. *But*, the musician's knowledge of these special effects is falling well behind the technology. While phasers and flangers abound, no one has taken the time to explain their functions and capabilities to the general (albeit selective) musician audience. The musician has been stuck between wanting to buy everything in sight and knowing that if he doesn't get to really know the product he will never fully realize the effect's potential.

Recently I walked into a rehearsal of a fairly well-known band expecting to hear the band blasting away in high gear. No such luck. Silence. Except for what sounded like mild whimpering. I looked around to see the lead guitarist on the floor surrounded by a multitude of new footpedals and guitar cords moaning, "What do I do now? Things are so fouled up that I can't get even the guitar to work, let alone the damn effects!"

Anyway, the musicians and engineers have been telling me that the arrival of an informative *and* critical source that describes the new effects, guitar synthesizers, *plus* the musical instrument amplifiers through which the units play, would mark a great day in their lives. And I listened, because I had been thinking the same thoughts for quite some time.

That is why *Modern Recording* will be introducing in the next few months a column dedicated to the musician who needs to know what all the new equipment is all about. Now, because of its nature this was a tricky piece of editorial to work up. Unlike in hi-fi equipment, for instance, there is no way to be totally objective. Very few hi-fi standards are applicable to the musical instrument field. In the MI area we very often deal more in subjective terms. Distortion figures for music amplifiers and noise specs for foot pedals are *not* going to be what you have become accustomed to seeing in our "Lab Report" tests.

So, what can you expect from this new column? You can expect a realistic approach to testing of MI equipment. A column that explains the use and functions of musical instrument amplifiers, special effects units and other auxiliary gear. A review yes, but also a technically objective *and* musically subjective analysis of the latest and best in music products.

Our reviewers are already well-known to *Modern Recording* readers: technical editor Brian Roth and contributing editor Craig Anderton. Both highly respected young individuals whose work you have seen in these pages, *and* who *actually use this equipment on a day-to-day basis.*

We at *Modern Recording* are very happy to be bringing *Notes* to you. We hope that you will send us your feelings on it (readers and manufacturers alike) and that you also will send us your thoughts after seeing the first installment. Which, as previously mentioned, is just around the corner.

Until then, keep looking to *Modern Recording* to continue as the innovator. And keep playing, recording and bemoaning the fact that you didn't . . .

Buy gold.

Sincerely,

H.G. La Torre
Editor/Publisher


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CIRCLE 85 ON READER SERVICE CARD

Letters to the Editor

Balanced Output

I've just finished reading "Small Studios: The Lighter Side of Business" by James Rupert (*MR*, February 1980) and from my brief experience at running a 4-track studio in the red, I can say that he has hit all the need-to-knows that a body might, except those soon to be revealed: "Stalking the Wild Studio Customer." Articles such as this give a good balance to the plethora of watts, dBs, Hz and ips's in your other articles.

—Ken Upham
Brookline, Ma.

A body will be pleased to know that an issue in a body's hands incorporates the revelations of "Stalking the Wild Studio Customer." See page 38.

Impressions Unlimited

Thanks for a great magazine. It fills a definite need for us semi-pro recordists and I really enjoy reading it. As a musician, I enjoy making music and the challenge of capturing those sounds on tape. Being a certified electronics engineering technician, I naturally enjoy working with equipment—almost as much as making the music.

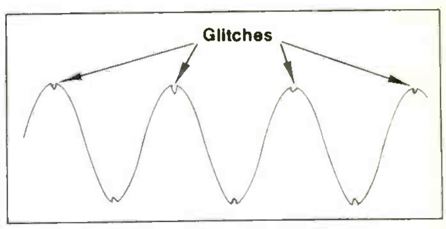
Let me congratulate you on your efforts to supply us budget-minded semi-pros with plans and kits to build some of the fancy gear used by professionals, at a fraction of the cost. I was delighted to see the dual limiter article in your November '79 issue. In fact, I purchased the kit listed in that article from PAIA. It does work as presented, unlike some plans carried by other electronics magazines which shall remain nameless. I did have a few minor problems which I would like to share with you and with your readers so that they will be able to anticipate them.

I was a little displeased to find substitutions made in my kit. An NE571 was included instead of the slightly superior NE570, but since the parts list says either can be used, I suppose that is understandable. Also, an MC3302P was substituted for the LM339. These substitutions did not seem to affect the performance. I measured distortion and observed the outputs on an oscilloscope with the ICs provided with the kit in the circuit. I then obtained an NE570 and an LM339 and did the same. I did not find any significant measurable or visible differences. Distortion was not discussed in the article, so I did not know what to expect. I found it to be slightly under 1%.

There were a couple of problems with the switch wiring diagram provided by PAIA. They reversed the connections on S1b in the "Figure B" wiring diagram, as referenced to the schematic. But since this section acts only as an on-off (SPST) switch, this change has no functional effect on the circuit. It is merely an inconsistency.

Now the real problem: In order for S1 to function as labeled on the front panel provided with the kit, it must be turned 180° (upside down) and then attached to the front panel after being wired as shown in Figure B.

There was a final problem I observed which I was not able to cure. The output is very clean when the dual limiter is below the limiting threshold. However, when the unit enters the limiting mode, there is a "glitch" at the very top of the sine wave (See accompanying sketch.) It seems to be generated by the comparator kicking in, but it is always at the very top of the sine wave at the limiter output. Even with this component, though, the distortion remains below 1%. Maybe one of *MR*'s readers can explain this occurence or instruct on how to effectively remove it.



I hope my comments help other *MR* readers to cope with their kits. This is a very worthwhile circuit and I hope to see many more like it in the pages of *MR*. Thanks again.

—J.L. Whaley
Sevierville, Tn.

Introducing TDK's Optimum Dynamic.™
Normal bias tape taken to the optimum.

TDK's answer to the need for a normal bias reference standard.

Optimum Dynamic is the outcome of the same, sophisticated technology which set the high bias reference standard with TDK's SA cassette. Its tape formulation consists of Optima Ferric particles. A needle-shaped, pure iron oxide that has been ultra refined to cover the tape surface evenly and densely. The result is a cassette with a sensitivity and MOL audibly superior to any normal bias cassette available in the market today.

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Optimum Dynamic has the same Super Precision Mechanism as the SA cassette, protected by TDK's full lifetime warranty.*

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We believe we've been highly successful in fulfilling the need for a normal bias reference standard. But there's a simple test. Listen to an Optimum Dynamic just once. Compare it to anything else you've been using. From then on, you may want to use it as a reference.

TDK

The machine for your machine

*In the unlikely event that any TDK cassette ever fails to perform due to a defect in materials or workmanship, simply return it to your local dealer or to TDK for a free replacement.

CIRCLE 67 ON READER SERVICE CARD

First let me offer my congratulations on having such an information-packed magazine. My bookstore sells them out within three days of displaying each new issue—in fact, to insure I'd get my issue, I've *had* to take out a subscription. And my issue is read wholly or in part by about six other people. In sum, it's quite a publication.

I do have a couple of comments on the dual limiter article. The first is that Mr. Anderton didn't mention in the "How it Works" section, what function IC2A performs. Also, with regard to the Signetics literature entitled "Signetics Compander Mini-Manual," an address for Signetics might have been included for those who'd like the additional information.

Aside from these two small items, the article itself was clear and well-written, and I liked the inclusion of the component layout with the foil artwork. I personally would like to see more articles in this area as I'm sure there are countless other "poor man" audio people who just can't afford a dbx at $175 per channel. May I suggest articles on building the compander the NE570/571 was designed as, two and three way crossovers of the electronic type and even a timer with hall effect switches for the studio.

—Henry Walters
Lancaster, Pa.

The Craig Anderton article "Build a Dual Limiter" was much appreciated, and I immediately assembled one, utilizing a Bill Godbout HK-16 power supply. I thought you and your readers might be interested in some impressions, so here goes:

Construction details were generally adequate, but no mention was made of the orientation mark on the ICs, potentially a source of trouble for novice builders. Performance is fine, but watch out for the loud thumps which occur when the power supply is turned on or off. They are severe enough to damage speakers if precautions are not taken. The design would have been improved if the "input monitor" LED had been utilized as an "over threshold" indicator and the control labeled "limit point" in the kit should have been labeled "threshold" in accordance with standard industry practice. In all, a very useful project. Many thanks.

—Bill Bennett
Bill Bennett Sound
Baton Rouge, La.

I would like to appropriate the manual mentioned in the dual limiter piece; "Signetics Compander Mini-Manual." How can I go about this?

By the way, I would like to tell you that I think your do-it-yourself projects are fantastic. I have never before seen a magazine that published such a project and explication. Almost everything was perfect—though I propose that you modify your "Construction" part with particulars of the construction procedure and also give a few references of books that could help.

If you could also introduce a simple block diagram of the circuit to help a bit on the comprehension of the project, it would be perfect. Thank you!

—Andre Lajoie
Hull, Quebec

Replies from the author, and from the president of PAIA, follow:

While it's always nice to get letters concerning a project, it's even better when those letters come from people who are clearly quite knowledgeable about and

involved with the field of musical electronics. These letters raise some interesting points, which we'll cover one at a time.

Substitutions: A substitute IC is not a cause for displeasure, since many IC types are equivalent but have been assigned different numbers from different manufacturers. Also, availability problems account for some substitutions (in this particular case, the NE570 was virtually impossible to obtain due to industry-wide shortage problems; to avoid back-ordering kits for at least two months, the more readily available NE571 was substituted in the PAIA kit).

Errors in PAIA instructions: Thanks for catching these discrepancies; a copy of your letter has been sent to the people at PAIA so that future instructions can be corrected. These things happen sometimes . . .

Glitch: Sorry, there's not much you can do about this, as it is an inherent part of the way this limiter works. However, as you point out, even with this glitch the distortion is under 1%, which is certainly better than the distortion problems you might get if the limiter were not there at all! I agree the glitch doesn't look pretty on a 'scope, but luckily it is barely audible (if at all) under normal operating conditions.

IC orientation: Bill is quite right that I should have mentioned IC orientation for the sake of novice builders, although frankly this project was not intended as a beginner level sort of thing. Nonetheless, since many neophytes have successfully assembled the dual limiter, I'll be more careful in the future. For the record, the pin 1 end on each IC is indicated with a dot or notch etched into the epoxy case. These notches should be lined up with the IC notches shown in figure 4 in the article (the component side layout).

Thumps: There are a lot of capacitors in this circuit and DC level shifting, which accounts for the thumps at turn-on and turn-off. There is really nothing that can be done about this at the circuit level: the only solution is to turn down the console volume if the limiter is in line with a signal and being turned on or off (which is standard practice anyway).

What IC2A does: IC2A simply inverts the reference voltage set by R9 and feeds this to IC3b. This is required for the limiter to respond to over-threshold conditions on both positive *and* negative peaks of the signal (not all signals are symmetrical—which is one reason why sine wave test oscillators don't always show the full story on how a circuit performs).

Signetics: Their address is 811 E. Arques Ave., Sunnyvale, Ca. 94086; I'm not sure, however, that the compander Mini-Manual is still available at this time.

Finally, if you wish to expand the limit threshold range set by R9/R10, reduce the values of resistors R11/R12 and R19/R20.

I appreciate hearing your comments very much; criticisms are always so much easier to accept when they're constructive (and offset by so many compliments!). You'll be seeing several more do-it-yourself projects for the budget-minded studio in these pages in the months ahead.

—Craig Anderton
Contributing Editor
Modern Recording

Craig's response on parts substitutions accurately presents the situation on that topic. For a variety of reasons, continuing tight supply would seem to be a fact of life for the early eighties, at least. This problem affects small and large manufacturers alike, but it's a lot harder to hide a parts substitution in a kit than it is in a piece of manufactured equipment.

My personal apologies to J.L. Whaley for the reversal in S1, as I personally cleared the dual limiter instructions and failed to notice this error. We are in the process of correcting the instructions.

—John Simonton
President
PAIA, Inc.

AEIOU, But Never "Y"

We fear we may have spawned some confusion in the opening "Talkback" letter from Larry Auman in our February '80 issue. That letter included a view of the Tascam Model 5 which illustrated an INCORRECT procedure—"Y"-ing outputs. Mr. Auman's last major paragraph, and Dale Dalke's final paragraph (p. 25) refer to the diagram, which many readers who only skimmed the actual letter and its answer might have deduced was Teac's suggested procedure (supported by the credit to Teac regarding the use of the back panel view itself, onto which Mr. Auman had indicated the wiring in question). The legend under the diagram should have read: "Back panel

CIRCLE 78 ON READER SERVICE CARD

view of the Tascam Model 5 showing incorrect *"Y"-ed outputs.*

To restate with brevity: Do not *apply the wiring configuration pictured on page 24 of* MR's *February 1980 issue.*

Toward Saturating Tape . . .

I've been meaning to write to you for some time, possibly hoping someone else would, concerning this matter.

I subscribe to and enjoy your magazine, yet the title of it is *"Modern Recording,"* so how come there are so many feature stories on "live" music? Could it be that sessions are too hard to get into? Sure P.A. work is interesting, but I want to know more about *studio* techniques. Yes, the "live" does relate to recording, but usually only as, ". . . and to record, they use a 24-connector 2-way splitter, sending one of the feeds to the recorder . . ." or something like that.

I do like the way the magazine is written, and I especially appreciated the how-to article on the dual limiter (which a more adept friend is building for me) but like I said, how about more on *recording*?

If the material and staff is available, why not just branch out into *two* magazines, one for "studio" and one for "live," the way *Guitar Player* has (acoustic and electric)? That would be more revenue for you, I'm sure, 'cause I'd probably subscribe to both anyway, as would a lot of others, even if not a "main" interest.

In the meantime, though, I'd like to see more on recording, more with producers, artists, engineers, the guys who are actively involved in producing recorded product. Thanks for hearing me out.

—Robert Koenigsberg
Torrance, Ca.

We promise you more on recording. Read the next letter.

. . . And Keeping It "Live"

I've finally done it! After years of trying to find a job in the audio business, I've finally scored a job with a professional sound and light company. Since I have been trained and taught within the confines of a studio, my recording experience is limited to what goes on in the studio and control room, though. Now I've been asked to do "live" P.A. sound. Help! I realize I've a lot to bone up on in this field in spite of the similarities, but there are definite differences between the two. I feel confident that I can do the job with the help of forthcoming experience and the monthly support of your magazine, but I'd also like to get my hands on an article called "The P.A. Primer," mentioned in so many of your issues. If it's available, how can I get it?

Your magazine has been and still is a trendsetter. All the best to you!

—Louis Helbling
St. Paul, Minn.

We promise you more on "live" sound.

"The P.A. Primer" has not been available for some time, but we are thinking of reprinting it, along with other select gems that are in demand. Send no dough but watch for future announcements.

At present we are sticking to the one Modern Recording *and we're gonna cram as much of everything that will please everybody into each and every single monthly issue. Remember, if you didn't O.D. on your audio/taping object of fanaticism this month, there's more coming next month.*

A "Close" Reader

Your magazine has been of extreme value in researching new equipment as well as keeping tabs on the latest gear from the latest manufacturers. After I get my monthly issue, I read every advertisement, and then I read through your articles with my dictionary in one hand and my handbook on electronic terms in the other. When I come across some articles, those obviously for the advanced of your readership, I nod my head in agreement and pretend to understand everything.

After reading the rash of letters in the February issue, I was relieved to see the number of other beginners in the same ball park. Could we have a section in the back devoted to us? A short glossary of terms and explanations of specs would also be helpful!

—Billy Jonns
Columbus, Ohio

We're flattered that you'd like to get all your answers from us, but the books we named in the answer to those folks in your ball park will really serve you better than anything we could prepare on a periodical level. No doubt you still have the February copy; those books, and others, will have you truly understanding—not just pretending—advanced material.

How the Pros Got There

I have just read your February issue (excellent as always!) which inspired me to write this letter.

After reading the many letters requesting information on getting into professional audio, it occurred to me that you could probably best help your readers by asking the pros how they got started. The interviews you conduct for the "Profile" section of the magazine would seem to be an excellent opportunity to uncover an engineer's or a producer's early history in professional audio and to get some first-rate advice for readers like myself seeking a career in this field.

Great magazine! Keep the much-needed information coming and *please*, let's hear from those pros about their first jobs in audio and how they got them.

—Wayne Garvin
Mississauga, Ontario

Are we reading the same magazine? Seems to us we generally start off interviews with questions directed towards the artist's beginnings in the industry. But true, sometimes not. We vow to pry deeper, in the future. (Around here, the award for the studio entree of not little luck goes to Richard Dashut, a co-producer with Fleetwood Mac, who related to our interviewer of his first nine months in the recording business: sweeping studio floors.)

Eenie, Meenie, Mynie

I am hoping you can help me make a decision. I would like to purchase an 10½" open-reel deck, and I've narrowed my choice down to three models; Studer Revox B-77, Teac X-10R, and Tandberg TD20A. I know that your magazine is concerned primarily with the pro and semi-pro sound person, but could you advise this budding recordist with regard to these quarter-track machines?

—Todd M. Spencer
New Orleans, La.

You've hit the jackpot, Todd; we've lab-tested all three of the decks you're evaluating. (The Revox was tested in its half-track configuration, but you can apply the results to your concerns easily.) The "Lab Report" by Len Feldman and Norman Eisenberg for the Tandberg was printed in our October 1978 issue; the Revox was written up by the same folks in November 1979; the Teac was given the same treatment in our March 1980 issue.

Nifty little bonus here—all three issues are in print and waiting for you to redeem them from our Back Issue Dept. with proper payment.

FYI

Your magazine is right on. I'm a songwriter who does 4-track recording for demos and your articles and interviews have given me much knowledge.

Could you tell me what is meant by "AM" and "AOR"? These terms were used in the intro to "A Session with Bob Welch" in the January 1980 issue. Write on!

—Don Mason
Norman, OK.

You're probably not into labels (we don't mean record, but descriptive), so being unaware that "AOR" means "album-oriented rock" ain't real important. But "Amplitude Modulation" is what "AM" stands for—that is, AM radio. You must have been on another wavelength.

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.*

Acoustic Amplification Advice

I've got a real good one for Talkback: a sound reinforcement system for an acoustic band. An old-timey country band, bluegrass band or a jug band using only acoustic instruments doesn't fare very well with equipment designed for rock and roll! Can you suggest types of mics, pickups, mic stands (and placement, too, since our performers like to move around a bit), mixers, amps (I think a 2-channel, tube amp provides a sweeter sound), speakers and enclosures appropriate for this sort of music?

—Chuck Small
Glen Cove, N.Y.

As we all know, the "old-timey" acoustic band was one we most enjoyed "live." Only in recent years has the studio been able to not only faithfully reproduce the true sound of acoustic instruments, but to make them sound "bigger than life." As a consequence, it is now the "live" band that has to live up to the quality of the recording.

We should approach the "live" sound reinforcement of an acoustic band with the studio sound as an ideal without limiting the spontaneity of the players. The equipment should attempt studio clarity but not encumber the "live" group, i.e., baffles, headphones, massive mic stands and such.

It may seem unorthodox to suggest using any condenser microphones in a "live" situation, however, we feel that it accentuates the top end and adds presence to the instrumental sound. More expensive mics such as the Electro-Voice 451, Neumann KM84 and U87 may be too expensive but there are less costly mics such as the Sony ECM22 which would be a reasonable substitute. All mics should be shock-mounted on their stands.

Here is an example of mic use and placement for "live" reproduction of an average bluegrass band:

Vocal miking cannot be done better than with the Shure SM56 in this situation. Put one near each vocalist and be certain the levels are lowered whenever any vocal mic is not being used.

For guitar, start miking with an E-V 451 where neck and body join, adjusting mic between this point and the sound hole until best sound is produced. Moving the mic away from the sound hole will produce clearer but thinner sound, while miking closer to the sound hole itself will produce more resonant and, consequently, muddier reproduction. Look for a crisp sound with a natural guitar richness.

The fiddle presents a slight amplification problem in that miking is preferable to using attached pickups on the instrument. The optimum results will be obtained with an overhead Neumann KM84 adjusted, as with the guitar, between where the bow plays the strings and the f hole. A Barcus-Berry or Frap pickup will allow freer movement for the player but must be run direct to the P.A. for acceptable results. On-stage amplification will just not reproduce a natural sound.

Banjo miking requires placing a dynamic microphone such as the Shure SM56 as close as possible to the head without impairing the player's performance. Do not mic strings.

The Dobro is relatively easy. Mic it directly from the sound hole with a Shure SM56.

The upright bass creates the most difficult problem in sound reinforcement. The sound engineer must depend upon his own ears when mixing for best placement and it will vary from environment to environment. In the studio the instrument would be miked with a Neumann KM84 between the strings and f hole, depending on the individual bass. Once again, the problem is resonance, and a dynamic mic such as the Shure SM56 may prove most practical in a "live" situation. Electric bass should simply run direct to the P.A. with a small, on-stage amp for the player to monitor.

For the overall amplification system, a small mixer such as Tapco or Tangent with a stereo output and one monitor send should be sufficient. Use a tube amp, such as a McIntosh for power. Solid state circuitry does not effectively reproduce the richness of acoustic instruments. Actual studio monitors will best deliver the highest fidelity and we would suggest either the Century 3's or 5's, JBL 4340's or the Altec 604-E's. Excellent results in on-stage monitoring can be expected from the JBL 4311's.

The sound reinforcement engineer should be careful to mix mono with slight emphasis to left and right for fiddle or banjo. This will give greater instrument clarity while not creating pockets of unbalanced sound in the "live" environment.

Most of the techniques listed above are studio procedures with compensations for a "live" environment. This is because there are no visuals in a studio and the only objective is to come as close as possible to the ideal, "bigger than life" quality. With any acoustic band this should also be a prime objective when presented "live," with an allowance for player-audience rapport.

—Mark Evans and
Larry Dunlap
Engineers
City Recording Services
Hollywood, Ca.

A New Generation of Recording

A few issues ago, you discussed some of the disadvantages of several of the new scientific breakthroughs in recording technology. I have a question that I hope you can provide with an answer.

One of the things you discussed was the fact that we cannot take advantage of digital and direct-to-disc recording. In direct-to-disc recording, we cannot use some of the advantages of multi-track recording, such as overdubbing. We lose much of the improvements of digital recording because at some point it has to be translated back into analog. Wouldn't mating these two processes eliminate, or reduce some of these problems?

Why can't a two- or four-channel digital master be used to feed the cutting lathe? If the digital signal is so lifelike, a properly mixed master should have (almost) the same sonic qualities of a "live" performance. Even if a mixed-down master is unavailable, you still should be able to mix a 24-channel digital signal down directly to the lathe. This would simulate a direct-to-disc recording even more, but without the pressure on the performer. Outside of the expense, am I missing something?

—Robert Gershenfeld
Galveston, Tex.

In answer to your letter, it may be best to review direct-to-disc and digital mastering separately as some confusion seems to exist between them.

Direct-to-disc mastering (how records were first made) was re-adopted in the early '70s for some audiophile productions to eliminate tape recording as a major source of noise and distortion in the studio's portion of the record-making process. This move to upgrade sound purity, however, required some significant production and quality compromises. For instance, elimination of tape precludes the producer's ability to mix the sound at other than real time and to edit, both of which can lead to better sound balance and tonal quality, expanded composition possibilities and the removal of any musician fluffs.

Tape also permits studio-signal processing, such as overdubbing, and more precise optimization of groove pitch and cutting amplitude for particular sound levels at the time of disc cutting, which can also frequently mean more minutes of music on the record. The tape master provides the opportunity to correct for any lack of quality in groove cutting, disc electroplating, replicating, as well as the resource for making additional pressings. Direct-to-disc records by necessity are limited-edition productions, which is partly reflected by their higher price.

Digital technology reinstates tape recording by eliminating the negatives associated with both direct-to-disc and analog recording and providing unexcelled sound quality. With the proper word size and sampling rates, etc., a digital recorder can faithfully recreate the frequency range from 10 Hz to 20 kHz +0.5, −3.0 dB with a signal-to-noise ratio in excess of 90 dB. It does this without distortion, noise, wow and flutter, multi-generation degradation and other factors inherent in analog recording. And, because distortion is virtually eliminated from tape recording, sound can be put onto the disc at significantly higher levels than with analog.

Regardless of the number of original channels (tracks) which record the session (32, 16 or four, in the case of 3M's digital equipment), music is mixed down to two-track stereo to feed the cutting lathe. At present, the digital signal is converted back to analog just before the cutting lathe, so that conventional, but improved records may be produced. Within the next five years or less, however, it is predicted that digital masters will be producing totally digital discs for a new generation of home disc players, so that the full sound now heard only in the studio or concert hall can finally reach the living room.

—Clark Duffey
Public Relations Department
3M
St. Paul, Minn.

Noise Reduction: The Intelligent Decision

I'm setting up a home studio to record demo tapes, and since my budget is very tight, it is extremely important that I spend my money where it will do the most good.

For the past few years, I have been doing remote recordings, and have gotten very satisfactory results using a TEAC 2340, a TEAC Model 2 mixer and Shure 545 and Audio-Technica 811 microphones (low-Z, balanced, with step-up transformers). Final mixes and dupes are made on a TEAC 2300S. With "live" recordings, noise introduced into the recording chain was usually masked by room ambience (audience noise, etc.), but now that I am going to a studio format, it appears that problems will start to arise, at least during quiet passages or songs, since the S/N ratio of the equipment that I'm using in no instance is greater than 55 dB (according to the equipment spec sheets).

It appears that I am an obvious candidate for Dolby or dbx noise reduction, but I am hesitant about investing the limited funds I have available because it seems to me that having 85-90 dB S/N on the tape deck isn't of much value when the maximum I can expect from the mixer and mics is only 55 dB. I would appreciate some pros and cons on the subject so that I can make an intelligent decision. One of my alternatives is to invest in a larger mixer (such as a Tascam Model 3) since I am rapidly approaching the limits of the

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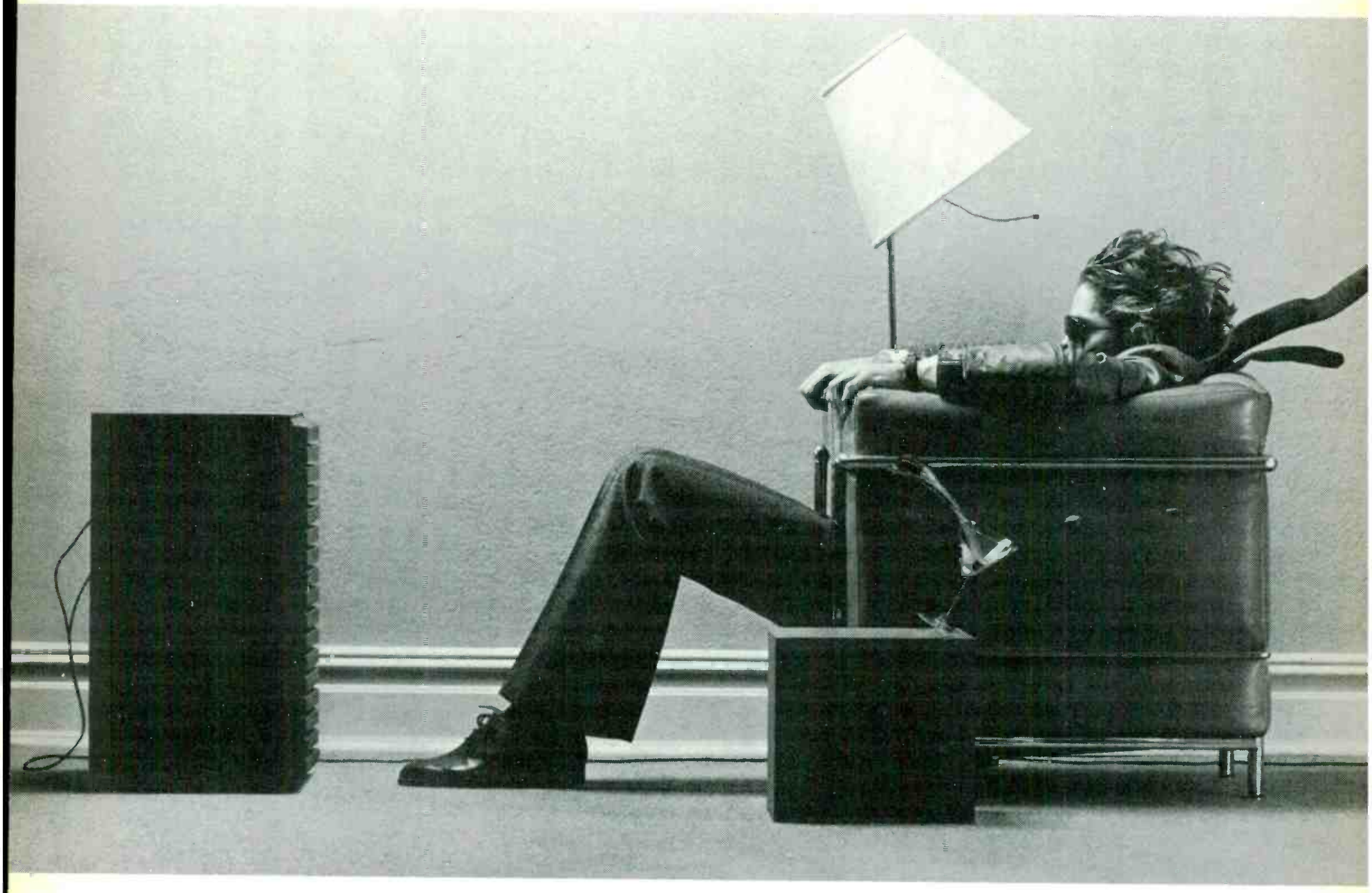

If your old favorites don't sound as good as they used to, the problem could be your recording tape.

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

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

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Model 2's flexibility.

I have also been thinking of using unbalanced mic cable (Belden 84-21) to tap off the extended speaker jacks of instrument amplifiers and using this to feed the line inputs of the Model 2. Before I try this stunt, though, I'd like to know if it's feasible without doing damage to the mixer. Also, would cable length present any problems in terms of signal loss or RFI? (The maximum run would be about 50 feet.) If a direct feed is not practical, is there a workable direct box configuration that I could build?

— Mark Simmons
Brunswick, N.Y.

The most "natural" way to reduce noise is, of course, to use quieter equipment. However, with a signal-to-noise ratio of 55 dB you may be relieved to know you're in good company. The noise figures for a good deal of the equipment used in full blown professional recording studios hovers in this area. This is particularly true of some of the older (and some of the more coveted) pieces of equipment. It is important to note here that even the best of professional analogue tape machines has signal-to-noise ratio of 62 dB, this figure being determined primarily by the noise generated by the blank tape itself.

There are a number of ways of coping with noise. Some engineers will manually mute a channel when it isn't producing useful program, in turn eliminating the noise from that channel. An electronic noise gate will perform this function automatically: muting a channel when no program signal is present above a preset threshold and allowing signal to pass when it rises above that threshold. This takes advantage of the psycho-acoustic rule of thumb that in most instances the program will "mask" the noise. This depends on the kind of program and the kind of noise.

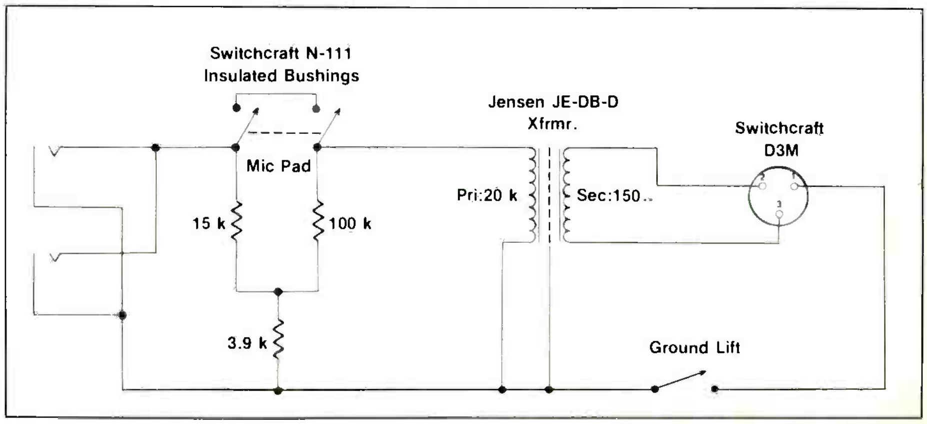
Noise reduction systems, like Dolby and dbx, are essentially a sophisticated version of the noise gate. Each system uses a somewhat different approach but they all have the same overall effect, which is to selectively mute noise when (and in some instances, in the part of the audio spectrum where) program falls below a certain threshold. Most of these systems mute noise across the entire audio spectrum, this includes hums, thuds, clicks, hisses, and buzzes. The exception is a system like Dolby "B" (as opposed to Dolby "A"). This is the type found on most cassette machines that have noise reduction and is designed primarily to reduce tape hiss. I will avoid doing a full dissertation on noise reduction operating theory by saying that there are a number of good books on multichannel recording technique that cover noise reduction systems thoroughly.

The pertinent point is that noise reduction devices don't just deal with noise generated by tape and the tape deck. The device is inserted between the console and the tape deck, consequently it processes the signal from the mixer during recording and the signal from the tape during playback. You still have a theoretical limit of 55 dB signal-to-noise, since that is the specification for the mixer with which you are mixing these wonderfully noiseless tracks. What you have done, however, is to greatly reduce the noise buildup that would have been generated by the early stages of recording; and 55 dB is, after all, a fairly workable figure. The alternative you mentioned of going to a larger console, in and of itself, won't solve your noise problem, so let the issue of versatility be the guiding factor as to when to upgrade the console.

By funneling the output of a power amplifier into your console's line inputs you're doing something akin to watering a potted plant with a high pressure fire hose. Your Tascam Model 2 likes to see about 0.25 volts at its line inputs and a typical musical instrument amplifier can dish out 20 volts or more at full tilt. It is possible to pad down the output and match impedances, but it's a little tricky and not usually done. Most engineers prefer to take direct signals from a more tame place in the signal chain, that being the instrument itself, using a direct box (see the figure below). This is fed to a microphone input and all the signal processing is done at the console (tone shaping, echo, distortion, etc.). The amplifier's controls have no effect on the recorded sound. Also, some amplifiers have a jack marked "Pre-Amp Out"; the signal from this point can be fed directly to the line input of a console.

This brings us to cable lengths. In the kind of high-impedance (typical line input termination) and unbalanced configuration you have mentioned the problems you would face are susceptibility to noise and RFI, and a loss of high frequency in any cable run longer than about fifteen feet. Low-impedance, balanced lines can be run as far as five hundred feet without significant signal degradation.

— Ken McKim
Chief Maintenance Engineer
RPM Sound Studios
New York, N.Y.



Curing Church Organ "Illness"

A church in my area has been trying to solve a problem they have had for some years now, with no satisfactory results thus far. The local experts have not been able to help, perhaps someone out there can.

When the organist plays their four-manual Moeller organ, annoying "clicks" and "pops" are introduced into the P.A. system when changing from one manual to another, or while setting certain stops which is accomplished by electric relays. The P.A. system consists of a Bogen MX 30A mixer amp and all mics are low-dynamics connected via heavy duty Belden 2-conductor, shielded cable.

Now there is a possibility that the problem is being transmitted through the mic cable since it is routed near the organ relay bank. However, I have done some recording there with a totally separate system and I have experienced the same difficulty, although it was not consistent. (I've used both

tube-type and transistorized recording equipment.)

This situation has become something of a 'head scratcher.' Any suggestions as to how to cure this "illness" would be appreciated.

—Ed Perrone
Gloversville, N.Y.

To eliminate the "clicks" and "pops" that occur in the church P.A. system when the organ settings are changed, we would suggest the following methods be tried. This type of noise, when associated with the operation of electric relays, can be traced to two possible sources: transient surges from relay or switch contacts opening and closing, or transient pulses generated by the collapse of the magnetic field of a relay or solenoid coil when it is turned off. Provided that the sound system is properly shielded and grounded, these transients must be suppressed by attenuating them at the source(s).

D.C. coil relays and solenoids can be "de-popped" by connecting a diode across the coil, positive (cathode) of diode to positive side of coil, in order to short-circuit the pulses created when the coil is turned off (Fig. 1). As the pulse generated here is of opposite polarity to the coil supply, the diode shunts the pulse out. Use a diode with an adequate P.I.V. rating and a high-peak current capacity. For a relay with a D.C. coil rated at 28 volts or less, a 1N4004 or 1N1764 type diode should suffice.

For A.C. operated relays a different method should be used. We recommend the use of a symmetrical transient suppressor such as the GE-MOV series varistor or a similar device. The varistor, which is available in a wide range of voltage and current ratings, is a voltage dependent, symmetrical resistor. When encountering a transient exceeding its varistor voltage rating, it changes from a very high standby resistance to a very low value

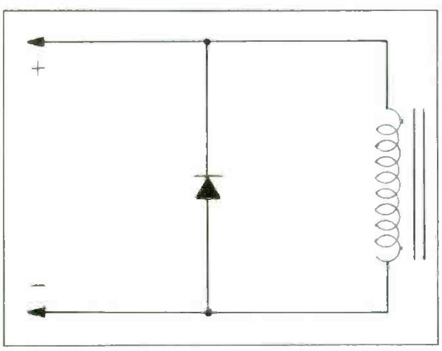


Figure 1

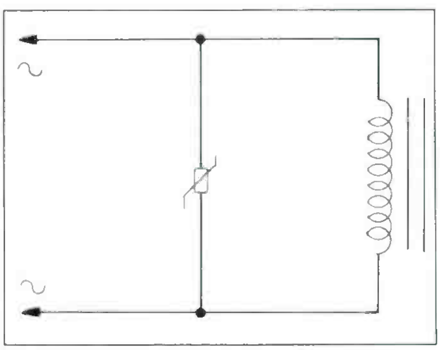


Figure 2

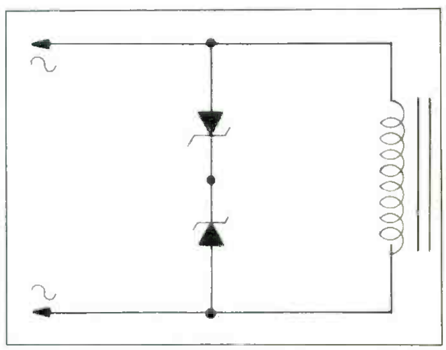


Figure 3

which shunts the transient, absorbing its energy. Connect the varistor across the coil leads (Fig. 2). For a 24-volt A.C. coil, a GE-MOV number V47ZA1, with a peak current rating of 250 amps, should do the job nicely.

A similar result can be obtained by the use of two Zener diodes in a back-to-back configuration, although higher pulse current ratings can be gotten with the varistor (Fig. 3). Try a Zener with a voltage rating twice the A.C. voltage applied; i.e., for a 24-volt A.C. coil try a pair of 48 volt, one watt Zeners. Whichever method you use, the suppressor circuit should be connected as close as possible to the actual coil, preferably to the coil contacts on the relay itself.

Transient surges from the contacts of a relay or switch may be treated in a like manner, again as close as possible to the contacts themselves. In the case of D.C. switching contacts, all that may be needed is a capacitor or resistor-capacitor series combination across the switched terminals (Fig. 4).

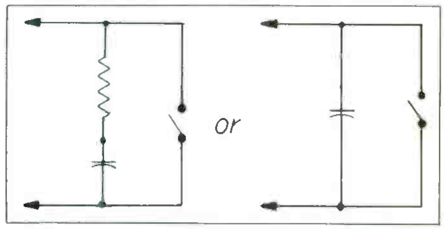


Figure 4

Try values of 0.1 to 1.0 MF for the capacitor; 1 to 10 ohms for the resistor. The D.C. voltage rating of the capacitor should be 400-600 volts, and resistors should be carbon composition types.

Hope these ideas are of some help.

—Paul McGalliard
Field Service Engineer
Capron Lighting & Sound
Needham Heights, Mass.

The Shape of Things to Come

I have a small home studio that I wish to expand. The new studio will be an addition to my home and the only prerequisite for its shape is that it share a common wall with the present control room. I am thinking of an area 400 to 500 square feet, and I would like to ask your advice on its basic shape.

Is there an ideal shape for a studio (i.e., square or rectangular) and is a very high ceiling (greater than 10 feet) desirable? Am I better off with a sloping

ceiling (if so, what number of degrees) or a flat one? Where in this "ideal" room should the monitors be placed and where should they be directed? I have some experience with double walls, "floating" floors, etc., but still wish to begin with a good basic shape.

Any advice or suggestions that you can offer would be most appreciated. Thank you for an excellent magazine every month. I find it most helpful. One cannot possibly test out every new piece of equipment that arrives on the market and I find your reviews and articles most helpful when choosing new equipment.

—Frank E. Lancaster
Ottawa, Ontario

There have been several books written on the subject of studio design and I have referenced some of them at the end of this article.

Determining the ideal shape for a home studio will be dependent on the existing space available and budget. There are several dimensional ratios which can be used to develop room proportions. Studies have determined that there is no optimal ratio only a range of certain acceptable parameters. Quoting Michael Rettinger, "If one as much as enters an empty rectangular enclosure or places a chair therein, one changes the ratio of room dimensions from an acoustic point of view. Hence it becomes impractical to design a room to have optimal ratios of dimension although certain dimensional bounds should be observed." The ratios based on the studies of Richard Bolt or Ludwig Sepmeyer are among the most popular to date. (See reference.)

Avoiding parallel walls should be a primary consideration. Nonparallel walls help to eliminate unwanted echos and standing waves. Unfortunately, most home studios have existing parallel walls. A simple solution to this problem would be to build one or more new walls which are offset slightly, so that no two opposing surfaces are exactly parallel. An offset of one foot for every ten feet of wall space is adequate.

In small studios, bass trapping is usually needed in either the wall or ceiling treatment to help dissipate standing waves. The most cost effective trapping method is the membrane absorber. The membrane is simply a sealed air space behind a rigid material such as plywood. The size of the membrane's face, density of the material and volume of air behind it determine the frequency to be absorbed. The membrane absorber also conserves on valuable floor space.

Certainly a studio with a floor space of 400 to 600 square feet would benefit by having a ceiling height of 10 or more feet. Budget permitting, the added height will give you the room volume and the option of trapping the ceiling to assist in smoothing out the low frequency response of the studio.

The size of your room will dictate monitor placement. Ideally, monitors should be located 8 to 12 feet from the critical listeners area and spaced the same distance between them as to create an imaginary equilateral triangle. This is only an approximate method for monitor placement which will vary depending on personal taste.

As to any suggestions I could pass along, I think the more time you spend reading and asking questions the better off you'll be. Proper planning will save you many dollars and headaches. It is much easier to change your design on paper rather than to be faced with major structural changes on down the road.

For background and some solid, basic information, I suggest you look up the following: Jeff Cooper, *Building a Recording Studio,* Recording Institute of America Press, Inc., New York, N.Y., 1978; Michael Rettinger, *Acoustic*

Design and Noise Control, Chemical Publishing Co.; Howard M. Tremaine, *Audio Cyclopedia*, Howard W. Sams & Co., Inc., Indianapolis, Indiana, 1978; F. Alton Everest, *Acoustic Techniques for Home and Studio*, Tab Books, Blue Ridge Summit, Pa., 1973.

—Jesse Walsh
Engineer
The Express Sound Co.
Costa Mesa, Ca.

A Handful of Problems

I own a TEAC A-3440 tape deck. While specs and performances on this machine are fantastic, I find that the location of the level controls in relation to the transport controls is very inconvenient. Many times I have had to recalibrate the system because my clumsy hands turn a knob when the rewind button is pressed or when splicing a tape, etc. Any ideas on how these controls can be "immobilized" once they are set?

I'd also like to know if TEAC makes a head assembly dress cover with an editing block that lifts up for better access to the heads. Splicing and head cleaning are two chores that would go much more smoothly and efficiently with such a dress cover.

—Sgt. Anthony A. Silva
APO, N.Y.

Dale Dalke, Technical Correspondent for TEAC Corp. of America (or "Mr. TEAC" as he's affectionately known to consistent readers of the Talkback column), was a bit hard put for an absolute solution to these. Basically, there is no replacement knob available for the A-3440 with locking or "click-stop" operation. More careful handling is the ticket here, or you might even consider a bit of new body positioning while working at the deck—approaching the transport controls from above, perhaps, instead of below would minimize your opportunity to nudge the level controls out of position.

As for your second question, there is no head assembly dress cover such as the one you describe offered by TEAC. Dale points out that mounting a basic, readily available editing block on the standard cover should provide you with adequate work space at negligible cost.

Simple solutions, these—but then, we never advocate spending vast sums of money where a little invention or hard work might do the trick, do we?

Peavey equalizers have been designed using the latest computer assisted design techinques and precision components to offer the musician, sound man, and home audiophile flawless performance without extravagant cost or compromises in quality.

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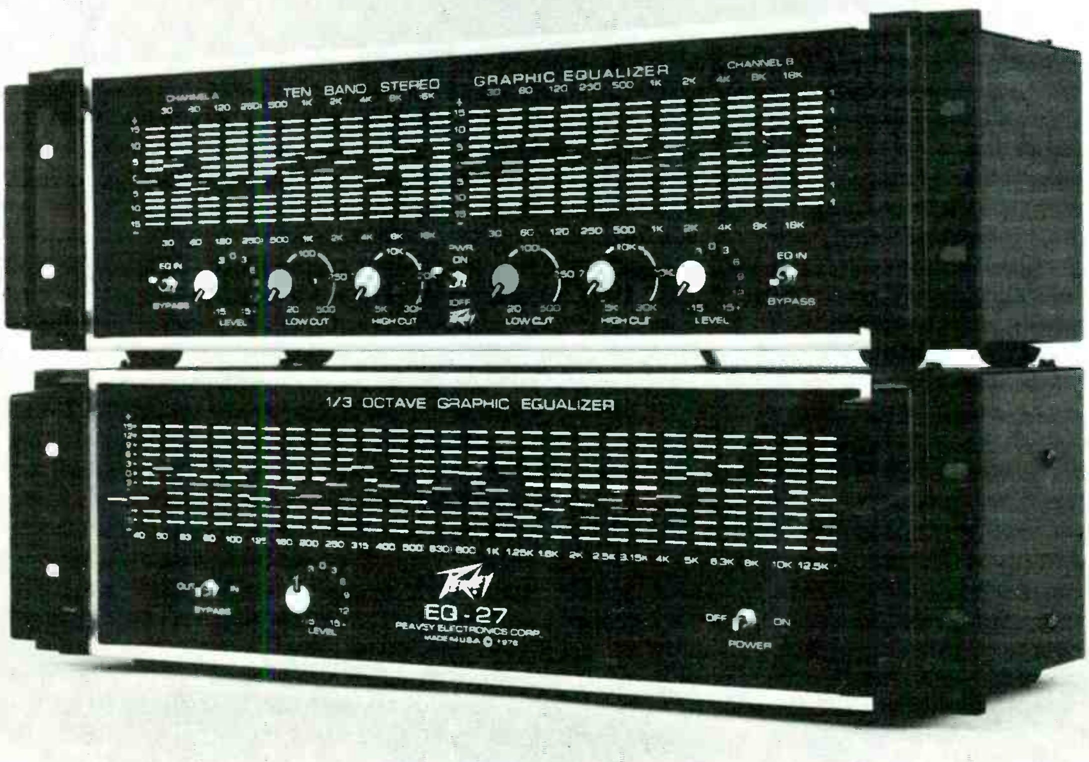

Complete specifications and descriptions of the Stereo Graphic and EQ-27 are available upon request by writing our Literature and Promotional Department, Peavey Electronics; 711 A Street; Meridian, Miss. 39301.

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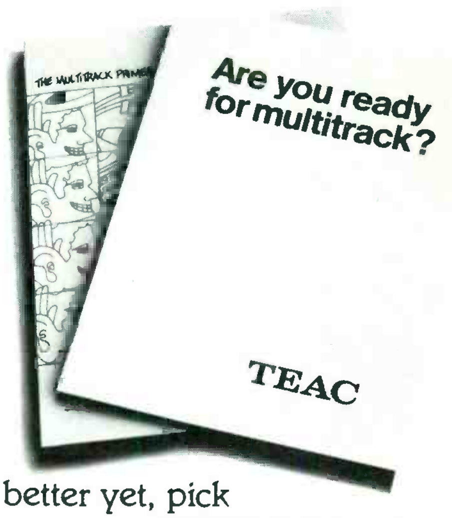
The process starts with the multichannel recorder. Specifically, our A-3440 —the new standard for four tracks on ¼-inch tape with sync. Rugged, reliable and very fast to operate, the A-3440 uses one button per track for Record/Playback status and dbx* Encode/Decode switching. It has a built-in 4x1 headphone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

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THE PRODUCT SCENE

By Norman Eisenberg

60-BAND GRAPHIC EQUALIZER

The GE-60 from Court Acoustics Ltd. of London, England offers stereo equalization in two rows of thirty one-third octave bands on standard ISO center frequencies from 25 Hz to 20 kHz. All sixty sliders provide ±10 dB adjustment, and have center "click stops." A level control provides up to 20 dB of gain, while a by-pass switch disables the equalization. This switch also allows the signal to pass through the device even with its power turned off. The 25-Hz filter acts as a shelving network to permit its use as a rumble filter or as a subsonic filter to eliminate room resonances or unwanted low-frequency elements. The 20-kHz filter acts as a low-pass filter to minimize supersonic interference and random noise. The GE-60 is rack-mountable.

CIRCLE 9 ON READER SERVICE CARD

LIGHTWEIGHT STEREOPHONES

Superex has entered the super lightweight stereo headphones field with its model TRL-88, weighing a bit over 4 ounces. Driver elements use micro-thin mylar film diaphragms in conjunction with high energy samarium cobalt magnets, a combination that is said to extend high frequency response within ±2 dB to 20 kHz besides increasing response speed for transient detail. Slot tuning the ear shell, and decreasing the foam thickness between driver element and ear is held to be responsible for extended low-frequency response down to 18 Hz. Price of the TRL-88 is $49.95.

CIRCLE 10 ON READER SERVICE CARD

ENCODE/DECODE DBX UNIT

The model 224 Type II from dbx is a simultaneous encode/decode noise reduction system that allows monitoring of the noise-reduced signal off the tape while recording. It may be used with two-head recorders, although only with three-head recorders will the full monitoring function be possible, of course. The 224 also provides up to a 40 dB increase in usable dynamic range. The system can be used for recording "live," tape-to-tape, disc-to-tape and off-the-air-to-tape. It also will play the new dbx encoded discs.

CIRCLE 11 ON READER SERVICE CARD

SANYO ANNOUNCES DIGITAL PROCESSOR

Sanyo's new Plus 10 Digital Audio Adapter, used in conjunction with a video cassette recorder, converts signals into PCM form for recording and then playback. The Plus 10 is said to offer a dynamic range of 85 dB. LCD meters, with a range of −50 dB to 0 dB, are switchable between peak-reading and peak-hold. Built into the Plus 10 are two mic preamps with level and attenuation controls. The device also features a preemphasis control for the industry-standard high-frequency boost as needed. Dropout is handled by a built-in error correction circuit with LED indicators. Price will be approximately $4000.

CIRCLE 12 ON READER SERVICE CARD

NEW AMPEX MULTITRACK RECORDER

Said to incorporate the best features and performance of its popular ATR-100 series along with the most advanced technology available today is an all-new analog multitrack recorder from Ampex. The model ATR-124 is a 24-track recorder employing a closed-loop DC-servoed transport designed to maintain constant tape tension at each reel in all operating modes without pinch rollers. "Flux gate™" record heads combine the record and sync playback windings on one head, said to give the user Sel Sync™ response that approximates normal reproduce response. A variable speed shuttle control lets the operator control the forward and reverse motion by sliding a finger along the switch. Shuttle speeds can be regulated from slow to 300 ips. The recorder has 16-inch reel capability. Ampex points out that this feature makes the system ideal for double-system recording in a quad videotape recorder environment. The system also has a "unique transformerless I/O capability that eliminates annoying distortions while offering excellent frequency response." The I/O bus feature also allows evaluation of each channel without continual moving of the I/O cables.

Flashing lights on the VU meters alert the recordist to a malfunction. A "pick up recording capability" permits editing or dubbing of new material without creating errors at either end of the new insert. Also provided are four assignable record, playback and Sel Sync equalizers per channel. Interfacing with various professional noise-reduction systems is possible. A new multipoint search-to-cue provides a capacity of ninety-nine memories. A complete remote control panel is available. The ATR series also includes 16-track models, and mastering mixdown machines.

CIRCLE 13 ON READER SERVICE CARD

ALTEC ANNOUNCES "STANLEY SCREAMERS"

New professional speaker systems, designed and built in conjunction with Altec sound contractor Stanal Sound (headed by Stan Miller), have been announced by Altec Lansing. Models include a dual subwoofer system and a three-way tri-amp-ready system. The "Screamers" are available in two versions, road-finished and utility-finished.

CIRCLE 14 ON READER SERVICE CARD

TAPE CLEANING KIT

Enough material for over 900 cleanings is said to be included in a new kit—TX 250—for tape heads available from Texwipe Co. of Hillsdale, N.J. A major feature of the kit is its cleaning pens—one for pressure rollers and the other for the heads. Each pen holds an adjustable, lint-free, absorbent wick that can be cut to the exact shape needed. Other items in the kit are a cleaning solvent, ten replacement wicks, a wick cutter, a machine-tooled open-front cartridge shell pressure roller activator (this may be used to position tape rollers for cleaning) and a rack to hold the kit's contents. According to Texwipe, its kit will remove oxides, dust, airborne contaminants, lubricant residue and oil from fingertips. In addition to cleaning heads, the kit is suggested for use in cleaning capstan shafts, hubs and other critical elements. Price is $27.50.

CIRCLE 15 ON READER SERVICE CARD

KENWOOD SHOWS METAL TAPE CASSETTE DECK

A three-head cassette tape recorder with metal tape capability has been announced by Kenwood. The new KX-1060, priced at $450, features a variable bias adjustment with built-in oscillator to help calibrate the deck for metal, chrome and normal tape. Tape-to-head contact is increased in this model, says Kenwood, by the use of a "double-back" tension system which maintains constant signal output and also extends the frequency response. The transport utilizes a two-belt drive system with rated wow and flutter of only 0.045 percent. S/N ratio (metal tape, Dolby on) is given as 65 dB. Other features include memory index, dual VU meters with peak-level LED indication and a timer standby device that starts operating at pre-determined time.

CIRCLE 16 ON READER SERVICE CARD

COMPACT REVERB UNIT

AKG's model BX-5 is a compact reverberation unit that uses their tortional transmission line principle. The BX-5 is said to be the first professional reverb unit that possesses the sound qualities inherent in AKG's BX-20 and BX-10E2 models, compactly

designed for rack-mounting. Stereo inputs and outputs are balanced. Levels and impedances are compatible with those commonly used in broadcast and recording. The patented TTL principle provides, AKG says, accurate reverb made possible by a series of springs whose transmission properties have been controlled by statistical variations of the spring parameters. Moreover, the TTL is claimed to be the only reverb device, including "live" chambers, that does not contain any of the dry input signal at its output. To maintain optimal reverb quality, the TTL system also provides a frequency-dependent decay-time characteristic. Price is $1195.

CIRCLE 17 ON READER SERVICE CARD

NUMARK MIXER

The model DM-1100 mixer from Numark Electronics has facilities for four stereo program inputs and two microphone inputs. Built-in features include a low-noise preamp for magnetic phono pickups and low-Z microphones; a headphone circuit to monitor each input; stereo slide controls; two-position LED output indicators with reference graph; and more. The DM-1100 is said to be capable of handling any quality high-powered amplifier without the need for an external preamp. Listed as a new "DJ" feature is the fact that both channel 1 and 2 are at maximum output with the fader at center position. Price is $199.50.

CIRCLE 18 ON READER SERVICE CARD

PARAMETRIC EQ

From Orange County Electronics of Winnipeg, Manitoba, Canada comes word of a new DEQ full parametric equalizer module. A four-band device, its center frequencies are variable from 20 Hz to 20 kHz in overlapping five octave (32:1) ranges. Each section tunes over an 80-dB control range (60 dB cut and 20 dB boost). Bandwidth is variable from 0.15–3 octaves (Q = 10–0.33). All controls are claimed to be non-interacting which means, for instance, that when bandwidth is changed, there is no change in level. With all sections in 20-dB boost, S/N is rated as 110 dB. Distortion, for 18 dBm output, is listed as 0.05% THD. Standard balanced or unbalanced operation is available. The DEQ is designed for use as a stereo parametric equalizer in Orange County's standard FR-1 rack frame or as a mono unit in the FR-2 desk housing. It also can be used as part of the VS-1 stressor.

CIRCLE 19 ON READER SERVICE CARD

JBL ANNOUNCES NEW LINE OF CINEMA SPEAKERS

A new generation of motion picture speaker systems announced by James B. Lansing Sound, Inc. includes seven models, all designed for high efficiency, wide dispersion and low distortion. The systems, of varying size and types and numbers of drivers, are intended to meet the needs of different size theatres, including surround-sound applications when needed.

CIRCLE 20 ON READER SERVICE CARD

NEW SUPER DISCS

Discwasher has announced its initial release of three albums of Chalfont/Varese Sarabande Soundstream digital recordings. One album includes selections by Shostakovich, Ravel, Ginastera and Weinberger performed by The London Symphony, Morton Gould, conductor (Chalfont SDG 301). Another is *Danzas Fantasticas* (compositions by Falla, Albeniz, Turina and Granados), same orchestra and conductor (Chalfont SDG 302). *Carlos Curley Goes Digital*, the third album, is a world premiere recording of a digital organ in works by Bach, Clarke, Mozart, Pierne and Widor; Carlos Curley, organist (Chalfont SDG 303).

CIRCLE 21 ON READER SERVICE CARD

Denon's PCM library includes well over a hundred albums covering all styles and types of music from Beethoven's Ninth Symphony to the Bridgewater Brothers.

CIRCLE 22 ON READER SERVICE CARD

Conventional recording and cutting, but via remastering at half speed and custom-pressed on "super vinyl" continues to be used by Mobile Fidelity, whose latest albums include *Fly Like an Eagle* by Steve Miller, *The Manhattan Transfer Live* and the Beatles's *Abbey Road*.

CIRCLE 23 ON READER SERVICE CARD

YAMAHA 3-HEAD CASSETTE DECK

Three heads, low noise operation, closed-loop dual capstan drive and built-in variable echo are some of the features included in the new Yamaha TC-720 cassette recorder. In addition to the usual switches for adjusting bias and EQ, the new Yamaha has a rotary control for fine-tuning bias current up to ±15 percent. Also included is a calibration (pink noise) signal generator which, together with the recording-level adjustment, enables the user to adjust the recording sensitivity for either standard (LH) or chrome tape. The model TC-720's mixing circuit enables mic and line mixing during recording and playback. The mic amplifier has two independent power supplies to provide wide dynamic range (54 dB against 0.3 mV inputs). Price is $450.

CIRCLE 24 ON READER SERVICE CARD

TAPE NEWS

The long awaited industry "thrust" re metal tape apparently is on. Fuji is showing its metal tape in three cassette sizes, C-90, 60 and 46. Up to 10 dB higher output level is claimed for high-frequency performance, while a gain of 3 to 5 dB is claimed for the lows. Fuji says that its metal tape cassettes, when properly recorded on a deck that has metal-tape capability, can be played on any deck with the EQ set in "chrome" or 70 μsec position.

CIRCLE 25 ON READER SERVICE CARD

TDK's metal tape cassette (MA-R) is a C-60 for which up to 7 dB greater output level than CrO_2 is claimed.

CIRCLE 26 ON READER SERVICE CARD

Sony, which started in metal tape with the 46-minute size, plans to introduce 60 and 90 sizes.

CIRCLE 27 ON READER SERVICE CARD

Hitachi's metal tape will be available in 46 and 60 minute lengths.

CIRCLE 28 ON READER SERVICE CARD

3M, which started it all, seems to be keeping a low profile as regards metal tape. Its latest news bulletin talks about head-cleaner video-cassette tapes for the VHS and Beta formats, and a new Scotch-brand Master XS open reel "audiophile" tape, said to be bias-compatible with most high-end open-reel decks. Master XS is described as a high-efficiency ferric-oxide with exceptional print and maximum output properties. 3M also has a Beta-format video-cassette recording tape, its L-750, which offers 4½ hours of running time on Beta III recorders.

CIRCLE 29 ON READER SERVICE CARD

Denon's new DX cassette tapes are double-coated. The upper layer consists of high coercivity cobalt-doped ferric-oxide; the lower layer is a high-sensitivity ferric-oxide. According to Denon, the upper layer supports the high frequencies while the lower layer "gives strength" to the low frequencies. As with the metal tape cassettes, a good deal of emphasis is also placed on the cassette housing and its accoutrements.

CIRCLE 22 ON READER SERVICE CARD

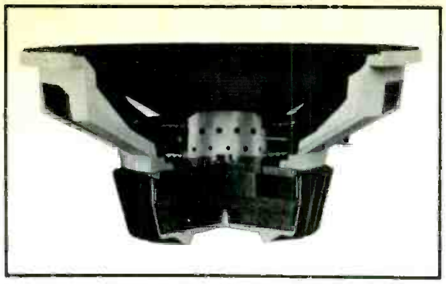
MUSICAL NEWSICALS

SOUND REINFORCEMENT SPEAKERS

News comes from James B. Lansing Sound, Inc. of the new JBL Cabaret Series of portable speaker systems designed for club-size sound reinforcement applications. The Cabaret Series currently comprises three systems, each designed for a specific application. All three models are loaded with JBL's well-known K Series musical instrument loudspeakers for high efficiency and power handling capabilities along with the greatest possible accuracy of sound reproduction. The model 4602 Stage Monitor is a compact, wedge-shaped system designed for wide frequency response and controlled directivity. The enclosure shape allows vertical, 30° or 60° orientation of the baffle board for great flexibility in stage setup. The 4602 uses a K120 12-inch loudspeaker, a 2402 high-frequency ring radiator and a specially-designed crossover network to provide high efficiency and a power handling rating of 100 watts RMS continuous. The model 4622 Lead Instrument speaker system incorporates two K120 12-inch loudspeakers mounted in a closed-back enclosure specifically designed to complement and enhance the clarity and distortion-free high power capabilites of the K Series speakers. The system was designed for lead guitar and keyboard applications, and has a power handling rating of 200 watts continuous sine wave power. The 4680 Line Array is a column-type P.A. speaker system featuring four K110 10-inch speakers and a 2902 High Frequency Power Pack. The 4680 is basically a JBL Professional Series 4682 Line Array housed in a special Cabaret Series enclosure, and provides uncompromised sound quality over a wide range of applications. The packaging and hardware of the Cabaret Series systems deserves special mention. All three models are completely self-contained in use, and are constructed from ¾-inch finish grade plywood, extensively braced inside, and finished with an extremely rugged black polyurethane paint. Speaker protection has been given high priority in the Cabaret Series with the drivers protected by a metal grille when in use and a flush-fitting cover panel when in transit; an additional feature is a special input jack which shorts the speaker voice coils when there is no plug in the jack to minimize speaker cone bounce in transit by electrically damping the cone. For convenient handling, all three models feature flush-mounted handles which lock in position to prevent rattles and vibrations in use; in addition, the 4622 and 4680 feature virtually indestructable polycorbonate corners which interlock to allow secure vertical stacking of several enclosures.

CIRCLE 1 ON READER SERVICE CARD

For years one of the best-known names in high-power loudspeakers has been Gauss, who has developed a reputation for being virtually blow-out proof. Now, word comes from Cetec Gauss of design improvement to the entire line of 12-inch, 15-inch and 18-inch speakers allowing the units to handle twice as much power as the existing models. Although both the cone/voice coil and the magnet structure have been revised, the outward speaker parameters (such as sensitivity, free air resonance, impedance, dimensions, etc.) remain unchanged except for the power handling rating which is now 300 watts RMS for lead guitar versions of the models, and 400 watts RMS for the bass guitar and low-frequency versions. Cone and voice coil dimensions are also unchanged, allowing interchangeability of recone kits with



existing models. The design changes in the Gauss speakers were developed for the 10-inch speaker which the company introduced two years ago and which may well be the most powerful and efficient 10-inch loudspeaker on the market. The important design innovations include the use of a 4.75 pound ceramic magnet within a 20 lb. magnet assembly with cast bottom plate and finned aluminum cover for effective heat dissipation, the use of an anodized aluminum voice coil support with special breathing holes to maximize heat dissipation and a special technique of winding the voice coil directly onto its support to improve roundness and uniformity.

CIRCLE 2 ON READER SERVICE CARD

MUSICAL INSTRUMENT ACCESSORIES

Countryman Associates is perhaps best known for its electrostatic piano pickups, although its hottest product at this moment is the Type 85 direct box. The Type 85 is an active direct box using a low-noise FET preamplifier section to virtually eliminate loading down the pickup with a too-low load impedance while maintaining excellent noise and distortion performance. Power for the Type 85 direct box comes either from a 9-volt battery supply or from a 48-volt phantom powering scheme (the 48-volt circuitry

uses a DC to DC converter for complete ground isolation for the two sides of the direct box. Input impedance of the unit is 10 Megohms, response is 20 Hz to 20 kHz $\pm$.5 dB, and distortion is a low .05% at 1 volt peak-to-peak.

CIRCLE 3 ON READER SERVICE CARD

DBJ Laboratories, Inc. is the manufacturer of a new combined effects unit, the EC-301 Effects Console, which combines distortion, wah-wah and volume pedal in one compact, steel-cased unit. The distortion circuitry includes controls for depth of distortion, blend of clean and distorted signal and output level allowing a range of distortion effects from soft, tube-like distortion to extreme fuzz with long sustain. The distortion and wah effects both have foot switches independent of the foot pedal, so that effects may be preset and switched in and out without disturbing the position of the pedal. The output volume of the wah effect is set by adjusting the system gain control, and the balance between distortion and wah when used simultaneously is set with the distortion level and system gain controls. In the volume control mode, the pedal position controls the output level through a voltage controlled circuit stage to eliminate the possibility of noise generation from a worn or dirty pedal potentiometer.

CIRCLE 4 ON READER SERVICE CARD

Music Technology, Inc. has introduced an accessory device called Tubes, which, logically enough, produces the much-desired tube-type distortion. This new unit does this the most direct way possible, by using a tube preamp. The Tubes unit is styled to resemble an old MacIntosh, and includes bass and treble controls and volume and master volume controls for complete flexibility in getting just the "right" tube amp sound.

CIRCLE 5 ON READER SERVICE CARD

The PW-6 Power Attenuator from Altair Corp. is a device designed for musicians whose amplifiers sound great when turned up loud but who must actually play at lower volume levels where their amp may not sound so great. The Altair PW-6 is connected between the amplifier output and the speakers and serves to attenuate the amplifier power so that less power actually reaches the speakers even if the amp is turned up to get the right sound from the output stage. The attenuation of the PW-6 is variable in 4 dB steps from 0 to 44 dB, and is adjusted with a large front panel knob. Output jacks are provided for two speaker cabinets, plus there is a 150-ohm balanced line output with up to an additional 30 dB of attenuation which can be used to feed a post-amplifier signal directly to a mixing board or recording console without the need for an external matching transformer or "direct box."

CIRCLE 6 ON READER SERVICE CARD

Ranger Leather Products may not have built a better mouse trap, but they have designed a better guitar strap. The Slinger is a unique, three-piece guitar strap which was designed to be significantly more comfortable thanks to a better balance and weight distribution. The center section of The Slinger is a wide, contoured shoulder support which is lined with suede for freedom from slippage, and which is designed to ride closer to the spine to distribute the weight of the guitar more effectively. The end straps of The Slinger are adjustable by means of two independent Velcro adjustments to fit individual playing positions and improve balance. Heavy, top-grain cowhide stitched with heavy linen thread is used throughout the strap, with double thicknesses of leather at all connection points, and double button tabs with offset slots for maximum security in attaching the strap to prized guitars.

CIRCLE 7 ON READER SERVICE CARD

The Micro Amp, a battery powered preamp for transducers or electric instruments, is the latest addition to MXR's Musical Product Group. The Micro Amp uses a bi-fet IC op-amp for its very high input impedance (6.8 megohm) which eliminates the effects of loading down an instrument's pickups. The unit has a low 470 ohm output impedance to allow connection to virtually any amplifier, mixer or signal

processor without loss of level or pickup of hum and noise. A gain control adjusts the voltage gain of the Micro Amp from unity (output voltage equal to input voltage) to about 26 dB (output voltage twenty times the input voltage). Frequency response of the Micro Amp is given as 12 Hz to 24 Hz ±3 dB, and the equivalent input noise is −103 dBV. The unit was designed for extra-long life from its single 9-volt battery, 1500 hours being given as the typical figure.

CIRCLE 31 ON READER SERVICE CARD

MUSICAL INSTRUMENT PICKUPS

DiMarzio Pickups has announced a new pickup model for acoustic instruments called the Quick-Mount. The Quick-Mount is a high-output humbucking design which features full shielding, Belden cables and a Switchcraft metal jack for minimum signal loss and minimum hum pickup.

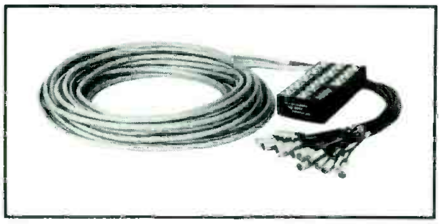
CIRCLE 32 ON READER SERVICE CARD

CABLE SYSTEMS

Coast Wholesale Music Co. has announced the introduction of a premium quality line of audio cables known as Signal-Flex. For the instrumentalist, the Signal-Flex line uses special custom ¼-inch phone connectors in one of four configurations: straight brass, right-angle brass, straight brass with collar-actuated shorting switch for silent plugging and unplugging and a straight military-type. All four connectors feature a patented screw-bush-end which clamps down on the cable at its exit point as the connector shell is screwed on tightly. The microphone cables in the Signal-Flex line all use the latest Neutrik XLR-type connectors which offer more reliable connections and superior strain relief than other similar mic connector brands. The wire used in Signal-Flex cables is also special, being extra-flexible at all temperatures, twin-shielded for lowest noise and highly resistant to heat, cold, oil and ozone. Instrument cables are available in straight lengths from 1 foot to 25 feet in regular and extra heavy-duty versions, and in coiled 20-foot lengths with a variety of connector combinations. Mic cables are available in five colors with either male/female XLR-type connectors or female XLR-type/¼-inch phone connectors; standard length is 20 feet.

CIRCLE 33 ON READER SERVICE CARD

New from Musimatic Inc. is a line of 100 foot microphone snake cables in eight different configurations to suit any mixer. The Musimatic snakes use 100% shielded cable of a low-impedance, balanced type suitable for carry-

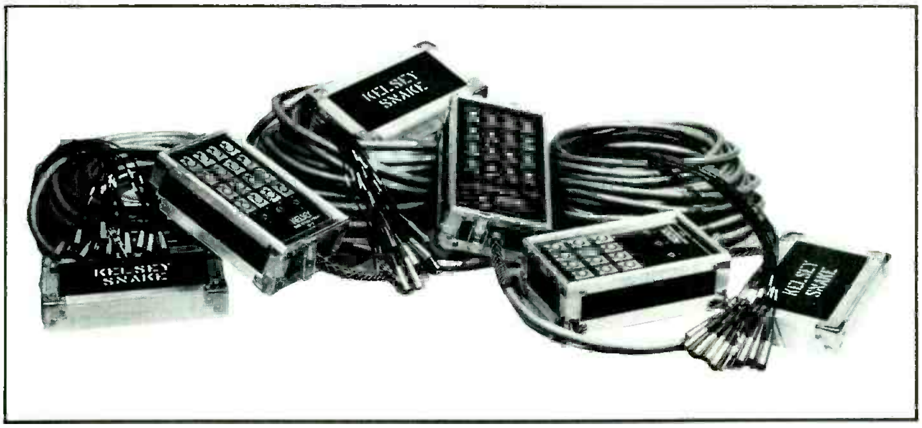


ing mic level signals from stage to mixer and/or high level signals from mixer back to stage. Switchcraft connectors are used throughout for reliability, and are individually labeled on the mixer end and labeled in an easy-to-read manner on the stage box.

CIRCLE 34 ON READER SERVICE CARD

Dallas Music Industries has brought their years of engineering experience to bear in the design of their new Kelsey Stage Return System. These new snake cables are available in a standard 100 foot length in 8-input, 12-input and 16-input configurations each of which includes an additional three lines for high-level returns to the stage from the mixer. The stage box in the Kelsey system is constructed from half-inch high density plywood, lock-cornered and braced, covered with wear-resistant vinyl and protected with heavy metal corners. The connector panel is 1/8-inch aluminum with black anodized finish and silk-screened designations. Switchcraft connectors are used throughout, 3-pin XLR-type for the mic lines and 2-conductor ¼-inch phone connectors for the return lines.

CIRCLE 35 ON READER SERVICE CARD



Synthe-Sound Musical Products is a very specialized company which makes cables and cable assemblies. The company's standard line includes an extensive selection of ready-made cable assemblies featuring Belden cable and Switchcraft connectors. Various connector combinations are featured using XLR-type mic connectors and standard and military-type ¼-inch phone connectors along with Belden wire or cable to suit the particular application. Various lengths are available depending on the configuration, ranging from 1 foot to 100 feet. For users whose needs can't be met by the standard line, Synthe-Sound offers a "U-Pick-A-Cable" system of cables custom-built to the customer's own specifications on an individual basis. Another unique service from Synthe-Sound is the Cablematix Specialized Strain Relief Process, a kind of ultimate strain relief. For extra-heavy duty applications, the Cablematix process incorporates a potted or encapsulated connection; a liquid rubber compound is poured into critical areas of the connector to immobilize the cable relative to the connector, resulting in a termination with the advantages of a molded type cord but using first quality component parts. Additionally, the cable shield is folded back, soldered and threaded directly into the connector for positive, trouble-free ground connections.

CIRCLE 37 ON READER SERVICE CARD

For more than a decade we've pushed the limits of electronic amplification and sound reinforcement equipment. We've established ourselves as the sound innovators. Our commitment is to develop imaginative concepts that give you the opportunity to discover your own unique sound. But it's really more than that. Our amps and sound systems *are* different. They're built for you. But, we go at it like they're built for us.

That's why we applied C-MOS (Complimentary MOS) circuitry to our Beta amplifiers. C-MOS was first used in the computer industry. We had no idea there was music in C-MOS until we tried it. The result is a unique overdrive circuit that is rich in harmonic content, much like tube distortion. But C-MOS does more than tubes. Distortion can be added in any amount, at any volume, so you don't destroy your ears to get that hot sustained sound you want. Yet in the clean mode the Beta is clearer, tighter than tube sound, so the characteristics of your instrument shine through.

Most amplifier people use standard speakers. We don't. We make our own so we can capture the subtle sounds that make each of our products unique. And that is what makes your sound unique. Because you can count on those differences to bring out your best. Play the fantastic Beta Series at your Sunn Sound Dealers.

Sunn. The Sound Innovator.

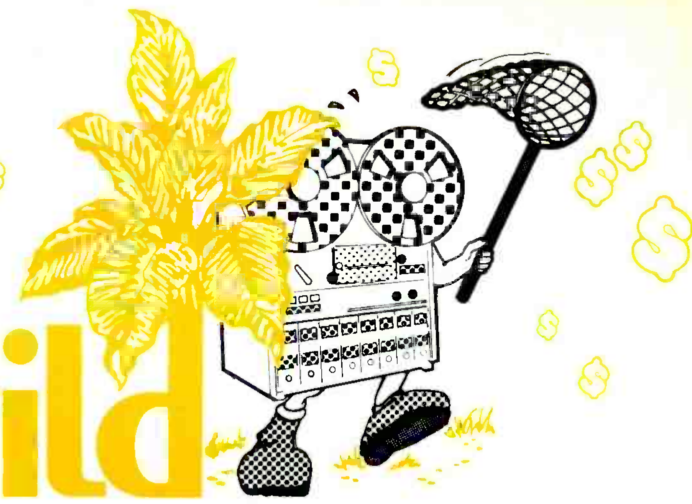
Beta Series Self-Contained Amps

Sunn endorsees include Chris Squire of Yes (shown above). John Entwhistle with The Who, Steve Busiowe with Meat Loaf, John Deacon with Queen, Joaquin Lievano and Jamie Glaser with Jean Luc Ponty.

Sunn Musical Equipment Company A Hartzell Corporation Company Amburn Industrial Park Tualatin, Oregon 97062

CIRCLE 76 ON READER SERVICE CARD

STALKING THE WILD STUDIO CUSTOMER



by JAMES F. RUPERT

In the deepest corner of the Valhalla, of good intentions and high aspirations, lie the corpses of literally hundreds of small recording studios that lived, gasped and died in the past ten years. Etched upon their bleaching bones are dozens of excuses for why they went under. Excuses as lame as "I didn't have enough gear," and as absurd as "My studio was just ahead of its time for this podunk town!" The cries of the damned and dying wail from Hollywood to Muscle Shoals, fingers pointing wildly in every direction and at every reason upon which blame might be placed. Yet still the perplexion goes on. Why, why, why?

Most of these studios had a good enough selection of equipment. Engineers for the most part knew their stuff well enough. Owners dumped more than sufficient buckets of money into the operation. Repair hassles? A few but not really enough to cause these kinds of problems. Bad locations? Keep trying. Unrealistic pricing? Shortage of help? Too long hours? High stool and seat cramps?

Now you may be asking yourself, "If it wasn't any of these, what *was* the major reason these studios didn't have anybody coming through the doors?"

And you've just answered your own question.

A recording studio is just like any other business. It has to solicit interest in itself. I don't know of any business that can just sit there and still have customers come to it—short of a pay toilet. There will come a day very quickly when you will soon discover there is more to the business of a recording studio than sitting behind a console, riding gain and occasionally yelling, "More balls," into a cue microphone. Whether you're in L.A. or Potato Junction, Iowa, you have got to reach out and round up the customers you need to stay alive.

Easier said than done? You bet it is. Impossible? Not on your life. You might find out that the promotion of your operation is the most enjoyable aspect of your studio. Perhaps the key is: *Imagination.* Tape recording is evident in far more aspects of our life than the 25-cent hunks of warped plastic they sell for ten dollars at the corner record store. Recording rock groups may be a lot of fun, but depending on their demand for studio time, they may not put much cholesterol on the table. (Worse yet, you might not make any money.) Remember Rupert's First Law: *He who lives by the basement band shall die by the basement band.* Amen.

So where do you start? How about with yourself. Your outlook must change from a consideration of what you are offering to enthusiasm over something you are trying to sell. Determine exactly what you can and can't do. If all the other studios are boasting "No job's too big," what's wrong with you pushing, "No job's too small." Maybe a lot of clientele is tired of the long waits from the big boys and the high price tags when the recording is finally completed. If you can pride yourself on your fast, good quality, reasonably priced product, why not make it your specialty?

What to Stalk

This slides right into our next step, that is, after you come to know yourself, you must then learn what type of customer you wish to shoot for. Are you looking for "live" musical group recording or are maybe some commercial and jingle production more up your alley? Ever think about soundtracks for slide/tape shows? Recording some books or magazines for your local library's program for the visually handicapped?

All this wraps up to mean that there might be more recording opportunities staring you in the cake-hole than you might have originally realized. If you've got the time, money, manpower and facilities you could conceivably achieve the highly improbable and become everything to everybody. If you are the garden variety humanoid biped with normal resources and scheduling, the value of specializing will become clearer and clearer. This does not mean you should end your quest for knowledge or your striving to do everything well. It just means you have to start someplace. Many studios have ended because they did not have any idea how to handle starting. As stated in a previous article[*see* MR, *February 1980 issue.*], look for your strong points. Then shoot in the direction of those particular stars. Your customers will indeed find out about all the other things you have the potential to shine at, but they have to become your customers first. Once they come through the doors they will begin to appreciate all the services you may be able to provide them with both immediately and also further down the pike. In the words of my ex-boss, "You first have to earn the public's respect, confidence and trust. Then you can stick 'em in the monkey-joint!"

So much for deciding on the audience you are aiming for. So what next? How about now taking a moment to set down and have tattooed on your brain that you are operating a *business*. (Note the emphasis.) A business is not like an English literature class where you can blow off your homework. There are two ways to legally gamble with big money stakes in America. You can either go to someplace like Las Vegas chock full of big plans to walk away wealthy in no time, or you can go into business with the same attitude. Every job imaginable requires certain skills, or you just aren't qualified for the job. Starting a business requires (unbelievably enough) a knowledge of business. If you are not willing to admit this cosmic truth and accept the responsibilities of the statement, then you might as well have your EQ tweaking fingers bronzed because you're done. You could be the Muhammed Ali of the recording console only to discover you are the Joe Palooka of the tax manual.

Now before the line forms at the registration office of the local business college, there are several options available to you in getting the necessary information on the intricacies of dat mean ol' world known as capitalism. One of the most helpful is the Small Business Administration. This is a service offered by the U.S. Government to anyone currently engaged in or even thinking about starting a (what else?) small business. Government pamphlets are available either free or for a very small charge at your local S.B.A. office which explain almost every aspect of the agony and the ecstasy of small businesses in this country. Counseling is available through this agency for personal advice and guidance on questions you might have about starting and maintaining your studio. The S.B.A. will also continue with their guidance after you have started your operation, as well as aid you in obtaining loans and financial assistance. Special interest rates and time repayment terms on loans are available if you qualify. Sound too good to be true? Then let me toss you the cherry on the sundae. It's all free. It's your tax dollars at work and it costs not an additional dime to use. Don't be afraid to admit that you don't know anything about business to them, that's why they're there.

For those of you who don't have a nearby S.B.A. office, don't forget about the public library. It's free too, and a great place to go when you have a headache. Actually, if you don't have a headache when you go in to read a few business manuals, you will by the time you are done. Almost any decently-stocked library will have a full complement of books available on tax laws, bookkeeping, inventory management, business ethics (yes, they exist!), operational techniques and a coupla dozen other etceteras. I once paid an arm and part of a leg to a lawyer for answers to questions I found three weeks later in the library for free. There's a moral in there somewheres. . . .

If you still feel unsure about leaving the line in front of the business college registration office, then don't. A few night classes couldn't hurt any of us. And though your questions might not receive the personal attention you want, to be fair, it might also be one of the most rewarding and revealing experiences of your life concerning yourself and your upcoming studio enterprise. Even I am going to take a few night courses one of these days when my rapidly leaking supply of excuses can no longer hold water.

The summary point of the last few paragraphs is that even a basement studio is a business. If your palm is crossed with silver for your services, you are a professional. Running your studio professionally keeps your customers coming back. Running it as a business keeps you able to invite them back. It is nearly impossible to have one without the other for long. Hence Rupert's Second Law: *If you want to drink with the big dogs, don't expect to pee with the pups.*

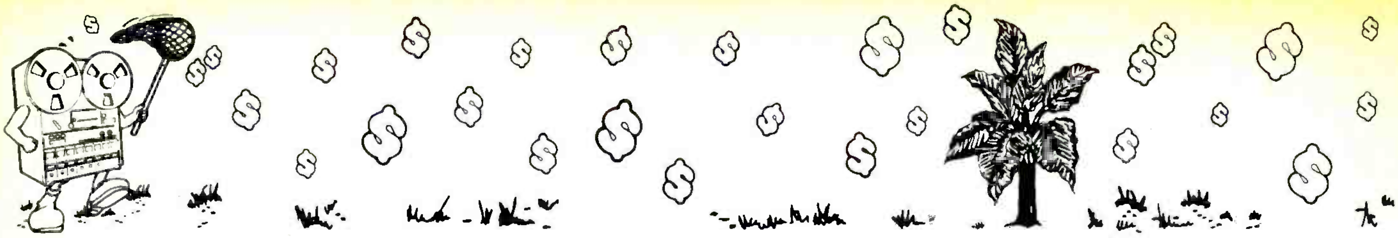
Checking the Competition

After you've done the obligatory homework before you open, you will know that once the marketplace is analyzed for demand, it's time to take a look at the competition. What services are they offering and how much are they charging? If you are going to undercut their prices, your newfound business expertise will allow you to figure if you are charging enough to at least break even. Price wars in any business can hurt everybody involved and are generally started only by saps who don't know what they are doing. I know, I started one myself. Picking a random figure to charge for your studio fee without thinking it out logically, realistically and competitively will soon reduce you to crouching in a padded-walled room playing with plastic vegetables.

(Another thing for the new studio owner to remember is the fact that he is indeed new. The idea of a planned gradual build to the full anticipated price goal is not a new one. Why not offer a grand opening sale with limited time introductory reduced fees? Part of any studio's hourly rate is for that particular studio's reputation. Racks of equipment sooner or later hopefully become secondary to the final product. Initial high pricing and overcharging can erect walls between your efforts and the public's appreciation of it. You have to take the time to earn a good reputation before your prices can reflect it. A further pox on you if your final product comes nowhere near to justifying the high price you are charging for it. Studio rates cannot make caviar promises and then deliver beans and weenie quality.)

We've now started to touch on an important aspect of building your clientele. That being the image projected to the public who will be using your services. We've talked a great deal about how to handle the customer inside the building. Now let's concentrate on how to get him there in the first place.

A discussion of image can not be complete without a discussion of promotion. Advertising and promotion,



however limited, can make or break your studio. Word of mouth is still the best advertising possible, but it has to start somewhere. And it has to be maintained. This is where promotion rears its complicated yet necessary head. If you do not promote your studio, you will probably be down the dumper faster than the Shah riding a bicycle through downtown Teheran.

For those of you out there thinking, "I can't afford a big advertising budget," let me say, "You don't need one." As with everything else when you are first starting out, ingenuity has to substitute for dollars. There are many options available to you that will put you in the public eye without letting the air out of your sock purse.

Stalking Advertising

Recording studios sell a service. An intangible. Perhaps it's time more of us sold that intangible as a product, albeit an end product, and not a service. This might entail adapting a little retail store sales philosophy.

There is no law—written or unofficial—that states a new recording studio cannot have a George Washington's Day sale. Or any other day for that matter. Cut your rates for a limited time and see if it affects your schedule. Throw in free tape with "X" amount of hours contracted. Offer a quantity discount to people buying so many hours of studio time. A half-dozen free cassette dupes of the final master. Free treatment for high stool and seat cramps.

Use the newspapers to your own advantage. A small ad for your studio in the classifieds might be lost in the shuffle. But how about a small ad for studio musicians in the classifieds? Try advertising for musicians to come in and make a free five-minute tape of themselves to be kept on file in the studio. The result is that you get a rapidly growing file of musicians on tape to present to clients who need backup people for their recordings and the musician gets a free five-minute session. You're able to offer a fuller service to your customers and the musician gets to see what it is like to work in a real live professional studio. You could be surprised at how many of these five-minute wizards get the studio fever and amble over to inquire about longer bookings for themselves. And why not? If you've done a creditable job with their audition tape, the hook should be firmly planted. Reel that fish in before you lose him to the current.

Many areas of the country have specialized musician's newspapers. This is an excellent resource to use. Ad rates are relatively cheap and are aimed specifically at a market you might want to tap. These newspapers also highlight the better known and more successful groups and bands in the area. If you're into recording bands, who better to choose to approach about your studio than someone working steadily enough to afford it? Once you get a line on them and where they are playing, go to them with your sales pitch. Approach them on a weekend, when crowds are better, egos are massaged and moods are more likely to be receptive to the idea of your studio being their first step on the ladder of stardom. Most importantly, don't just wait around for them to come to you!

If there are any local talent agencies, perhaps a deal can be worked out so that all the groups they book might receive a special discount for your services. This could provide incentive for groups to contract with the agency that you have reached the agreement with. This is one of several ideas where the studio owner is not the only person to benefit from the innovation. Another would be to work with any music composers available to you to come up with a free musical radio commercial "bed" for public service announcements used by local and national charitable and nonprofit organizations. Groups like the Big Brother organization, the Red Cross, Salvation Army or March of Dimes might not only provide you some free promotion, but you might feel pretty good about yourself for doing it too!

Past employers are also an excellent contact for potential recording contracts. They might not know how reasonably their firm might obtain a jingle or series of commercial spots until you tell them. Go straight to the guy at the top. The fact you once worked there will usually get you in the door. The rest is up to you. Once your foot is in the door, that same former employer could be the lead to the suppliers, jobbers, distributors and manufacturers that service him. Even if all you can get is addresses, don't be afraid to send some free sample spots to these behind-the-scenes companies. Thousands of dollars have been lost by small businessmen thinking certain companies are just too big to bother with the products and services of a small fish like him. The loss is his own. Small thinking is reserved for microscopes and politicians.

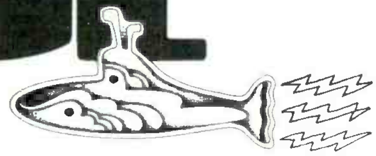
How about hitting every school in your reachable area with a form letter about your services. Every school has choral festivals, class plays, band concerts and the like, and there is always a ready supply of parents ready to slap down yankee dollars for the chance of having junior immortalized on record or tape. Once again, if you don't do it, somebody else will. Check with your ever-helpful library for lists of schools.

Send a sample tape to every advertising agency within a hundred miles of you. The tape should be 7½ ips and four to seven minutes long. Ad agency tapes should be professional, as creative as possible and above all clean. Do *not* send Dolbyized tapes. If the agency would rather have a cassette copy or more samples of your work, they'll let you know. Once again, what's good for your business is also good for theirs.

First Class Tapes

If you have a particular skill or system that can be committed to a cassette tape, don't overlook the mail-order market. Mail-order products are a multi-million dollar industry in this country. Even the classified section in the back of the magazine you are holding (sacred as it is!) has sported

TAKE A FANTASTIC VOYAGE



Inside Tip:
The filters can be modified just by changing capacitor values to "roll-off" or "roll-on" at virtually any frequency. Result: *A Built-In Electronic Crossover.* Graphs for these modifications and others are in the owner's manual. We even made the owner's manual small enough to fit in a pocket and printed it on waterproof (and beer proof) paper.

Easy Access
6 screws hold the main board to the chassis. Only Velcro® could be quicker.

Gold Plated, Locking Connector
No "jiggle" quotient.

Overload Indicators
These start to flash 1dB prior to clipping at any load, at any frequency.

Toroidal Transformer
High current drive capability allows easy 2 Ohm performance. The Toroidal design also has no stray hum field, so you can put low-level stuff like preamps and digital delay lines right on top of the P50.

All Discrete, Fully Complementary Circuitry from Input to Output
If you're tired of an amplifier that sounds like a chicken being chased by a steam-roller, give the P50 a listen. . .soothes ravaged ears.

Relay
D.C. sensing protection circuit eliminates turn-on and turn-off thumps.

Fan
The P50 not only meets F.T.C. specs at 2 Ohms, but does it with no thermal cycling.

High Pass Filter
With this filter "in circuit" the response is 3dB down at 20Hz. Gets rid of rumble, and works very well with cinema noise reduction systems. Remove the filter and the response is flat to 0.5Hz.

Mono Input
Inserting a 1/4" phone plug into this jack disconnects the left and right stereo inputs and automatically bridges the amp for mono operation. No switches. No jumpers. No headaches.

Pem Nuts
Instead of using sheet metal screws that come loose, we use Pem Nuts. Pem Nuts are threaded pieces of metal that, when bonded with the chassis, provide extra thickness and strength. Plus, we can now use a machine screw instead of the self tapping sheet metal type. . . you can take the P50 apart and put it back together as often as you want. We use Pem Nuts. . .Obviously.

Chassis of .090" Aluminum
We even have an .090" Aluminum L-Bracket running down each side to give the amp extra rigidity when rack mounted.

Low Pass Filter
A 6dB per octave filter gives the amp a 3dB down point at 25kHz to keep R.F.I. from passing through the amp and frying tweeters. If you *are* interested in frying tweeters, remove this jumper and the response goes out to a couple of hundred kilohertz. (By the way, we give you a dummy pin to store the jumper on when you want it out of the circuit.)

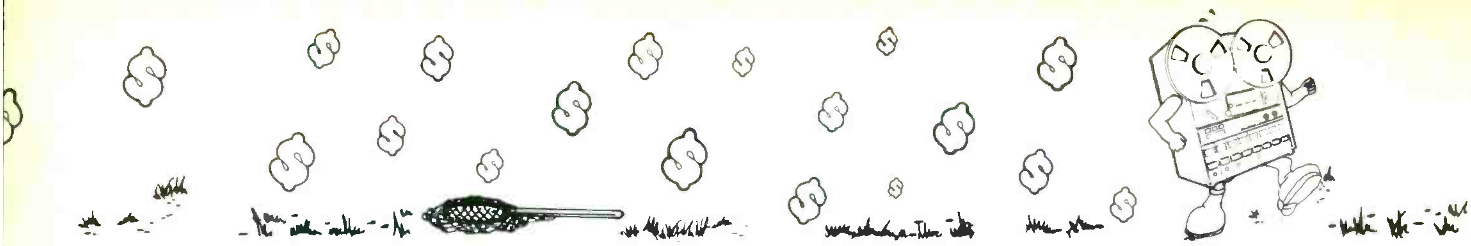
Power Output:
At Least 70 Watts per Channel in Stereo, @ less than .05% T.H.D.
300 Watts in Mono

THE P50 PROFESSIONAL 70 WATT PER CHANNEL POWER AMP

SAE
PROFESSIONAL PRODUCTS GROUP

For information write: SAE Professional Products Group, Dept. FM, P.O. Box 60271, Terminal Annex, Los Angeles, California 90060

CIRCLE 137 ON READER SERVICE CARD



ads for enterprising young companies with instructional techniques of recording for sale on tape. Mail-order taped courses on how to learn to play different instruments are visible everywhere. And this leads us to an oftentimes overlooked point. Even the smallest doggiest looking ads you see in almost any magazine would not be there month after month if they were not making money. So why not also use your studio as a mail-order center? Once the master tape is made, minimal amounts of time, money and effort are expended to make the dupes. If your tape is about cars, stick with the specialty automotive magazines at first. This specialty-book formula holds true regardless of the subject chosen. As usual, hit your library for details. There are several excellent "How to" books available on the art of setting up a mail-order business. Just remember to stay within your local laws and regulations concerning mail-order businesses, keep the price fairly budget-minded (mail orders are notoriously impulse oriented) and turn out a good product. Ask yourself if *you* would pay the price listed for what the buyer finally receives. If not, head for the bulk eraser and try again.

I hope by now some of you are beginning to see that your best customers may want and need your services and not even know it. And they won't ever know it unless you do something about it. Once again, the best advertising you can have is that which you don't pay for. Namely the aforementioned word-of-mouth recommendations. An excellent reputation for quality is worth far more than a full-page advertisement in any publication. But you have to start the mouths talking. Explore your options. Is there a market that no one is tapping? Do you have the ability to tap it?

The points we have listed here are obviously just a few of the ideas the new studio owner can try in order to establish his position in the marketplace as well as in his own head. If you have a few of your own, why not share them with the rest of the readers of *Modern Recording*? Life is a garden and knowledge is like manure. You've got to spread it around before it can do you any good.

Don't send those ideas to me, however. For the next few weeks I'll be tied up in bankruptcy court. That's what you get for trying to start a studio in this podunk town. Just forward my mail to the Institute for the Prevention and Treatment of High Stool and Seat Cramps. Boy, if only I had had more gear. . .

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

PERFORMANCE:
Overall Signal-to-Noise: 66 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.

Overall Frequency Response (15 ips): 30 Hz to 22 kHz ±2 dB.

Playback Frequency Response (MRL test tape): 31.5 Hz to 20 kHz ±2 dB.

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 $2,050 (USA).

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

OTARI®

Call Ruth Pruett Ables on 415/592-8311 for the name of your nearest Otari professional dealer.
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San Carlos, CA 94070 TWX 910-376-4890
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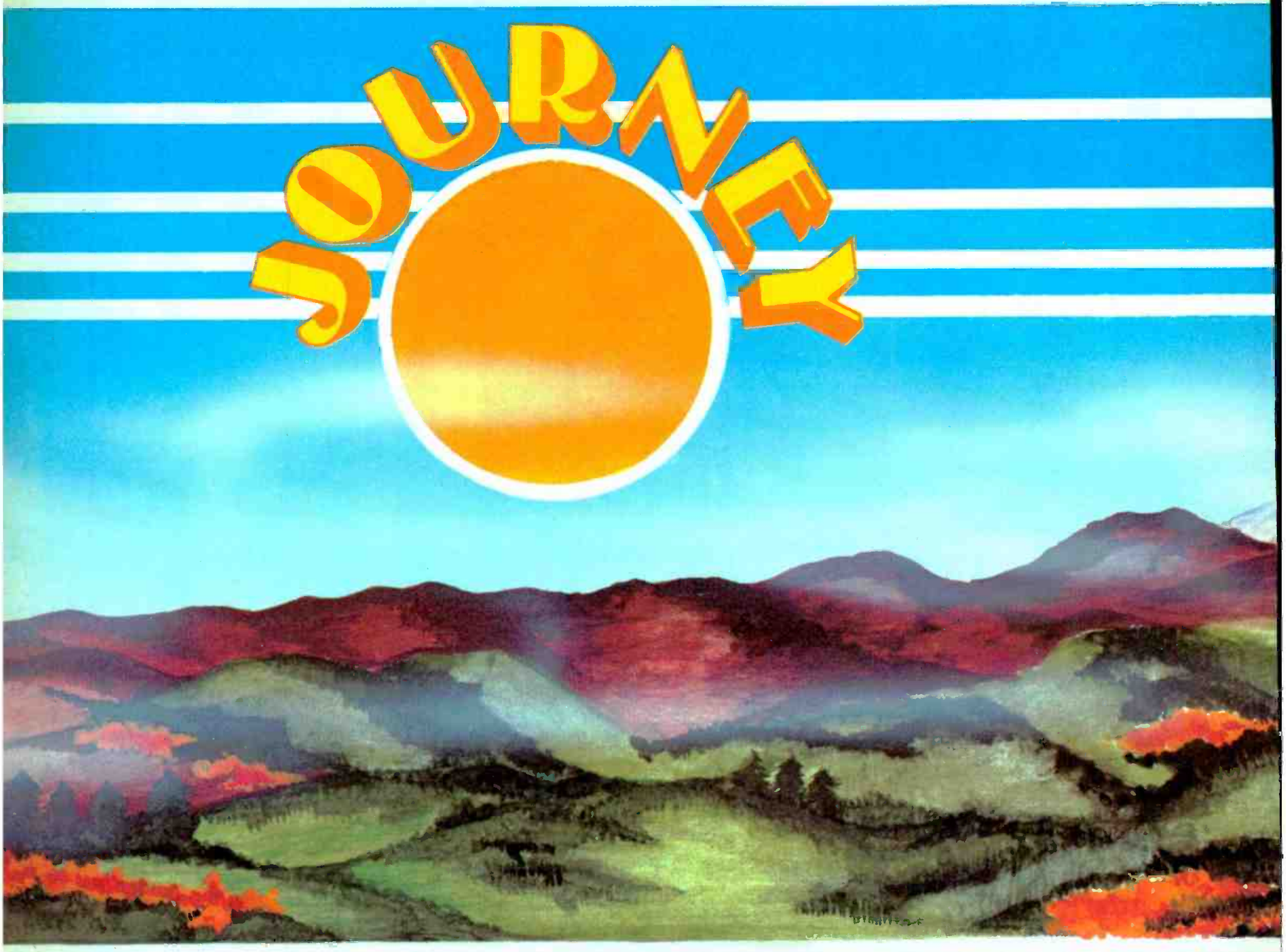
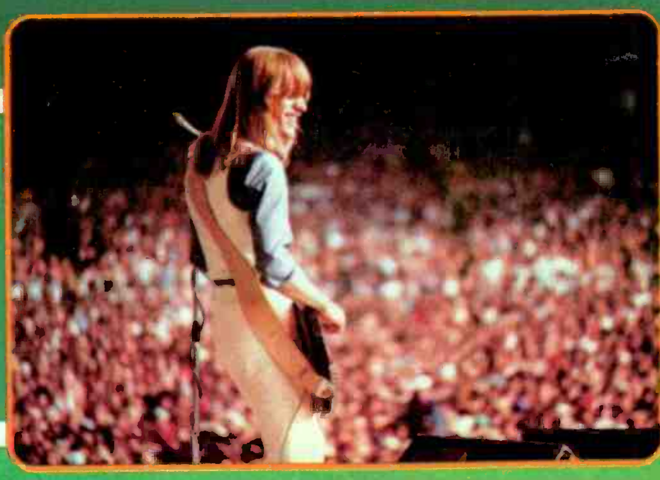
P.O. Box 7003 Station B, Rexdale, Ontario M9V 4B3
416/675-2425

a session with

JOURNEY





By Craig Anderton

Journey is a five-piece band, with Neal Schon on guitar; Gregg Rolie, keyboards; Steve Smith, drums; Ross Valory, bass; Steve Perry, vocals. Journey is also much a team effort, not only on a musical level, but also on a recording and management level.

The first feeling I had when walking in to do this interview was that I was dealing with professionals: no one was stoned out, appointments were promptly kept, and even during the interview the recording process rolled smoothly along.

Production and engineering chores are shared by Kevin Elson and Geoff Workman, although it seemed to me that Kevin is more production oriented while Geoff concentrates on the engineering. While Kevin had never produced any albums entirely by himself prior to this project, his engineering work on Lynyrd Skynyrd's *Street Survivors*, demo work with .38 Special and Molly Hatchet and particularly his handling of Journey's "live" sound for the past several tours certainly qualified him for working on *Departure*, the group's sixth album. Geoff Workman has worked extensively with producer Roy Thomas Baker; his list of engineering credits includes an amazing array of albums from early King Crimson to Queen, Foreigner, Journey, the Cars, and numerous others. Second engineer Ken Kessie rounds out the production team.

Kevin is articulate, deliberate, and clearly gives a lot of thought to how to best realize a particular sound. Geoff, on the other hand, has that glint-in-the-eye characteristic of any mad scientist. He would much rather twiddle knobs than talk about *how* he twiddles knobs. In fact, the first thing he said when he came in for the interview was that if Kevin had told me what I needed to know, that was fine with him and he wouldn't be miffed if he were left out of the story. He keeps a low profile, and being called something like a "super-engineer" would probably embarrass him more than anything else. Before letting him get back to recording, though, I did manage to get some interesting comments concerning the recording process. Geoff is not a person whose words translate easily onto the printed page. The words come out in big enthusiastic clumps that may, or may not, have anything to do with the question being asked. But this didn't strike me as a sign of flakiness; rather, this guy is a likable pro who doesn't give a damn about publicity and is secure in his own abilities and attitudes.

The members of Journey were happy with the way things were going in the studio. There was no evidence of friction or malaise, and everyone seemed pretty loose. When Neal walked in with some new guitar gadgets, everyone paused long enough in their work to be properly appreciative. Considering that Journey has developed its following through constant touring and persistent management as opposed to cashing in on a hit single and burning out, this type of closeness and professional attitude seems appropriate. But now they have a hit single too ("Lovin', Touchin', Squeezin'," which became a hit eight months after it had been released), and the future is looking very bright indeed for this San Francisco-based band.

Modern Recording: Did you feel any need to change the Journey "sound," or is the current project a continuation of previous albums?

Kevin Elson: The last couple of albums have been mainly vocal oriented. My background is more as a straight ahead rocker. In past Journey albums, I've felt the bass and drums didn't stand out well enough and were not properly heard in the mix. This album also represents more facets of the band, because they enjoy playing lots of different music. There's real good rock, some ballads, a little blues, some jazz—this album is a departure in the sense that there's more variety of material. A lot of solos, and even some vocals, have been done "live" so there's more of a feel. We're not as picky as in the past; if something feels good, we go with it. All the musicians are happy with the results of this approach.

MR: Journey had a lot of success with [producer] Roy Thomas Baker at the helm, and I would think there would be a tendency to go with a proven winning combination. How did you come to be the band's producer for this project?

KE: I came into it because I was doing their "live" sound, and I started doing some "live" tapes. They came out really well, so CBS decided to put out some "live" material for the FM stations, and the response was good. Also, the band didn't care for the mastering on *Evolution* and some of the singles, so I re-edited "Lovin' Touchin' Squeezin'" and re-mastered it. I got a different sound that they were real happy with. When it came time to talk about producers, the band decided that they wanted myself and one other person to collaborate closely on getting the album together. The other person was Geoff Workman, who at that time was working with Roy Thomas Baker on the Foreigner album.

MR: Was Geoff co-producer with Roy on that album?

KE: No, he worked strictly in an engineering capacity and Roy was the producer, period. Geoffrey had no say in the production, but he did the bulk of the work and we felt he was responsible for the sound. I've worked with some really good producers and have seen how the input of a producer works, so when it came time to make a decision, I definitely wanted to work with Geoffrey—not only because of his talents, but also because of his all around attitude and good humor. We finally put the production package together in September of '79.

MR: So you feel much of production involves attitude . . .

KE: Yes. Both Geoff and I come from a non-technical background; in fact, we both started out as keyboard players. I've always been concerned primarily with the sound, and didn't care about what it took to get the sound I wanted.

MR: Do you feel your musical background is helpful, or does it get in the way of being a more effective producer?

KE: It's helpful. I feel that the people who have written a song know what they're looking for better than anyone else. I try to bring out the best possible performance from the players. The main thing is that with a musical background, there's an easy relationship with the musicians. Say, when Gregg is looking for a unique sound, I can understand what he's talking about because I've been in that position before. It's also easier to relate musician-to-musician in terms of getting a good bass and drums sound.

MR: So you're pretty much in an equal position with the players, it's not "the producer as God?"

KE: Oh yeah, exactly. We discuss everything. If Geoff and I have an idea, we'll try it, and if it doesn't make it, we'll move on. I was involved in the pre-production, so things were well prepared before we even went into the studio. We haven't had to stray much from our original ideas.

• • •

MR: Why did you choose to record at the Automatt [Recording Studios, San Francisco]?

KE: Journey had done the last two albums in Los Angeles, but didn't feel real comfortable because they had to be so scheduled. So, last summer their management talked with David Rubinson (the Automatt's owner). One of the best things about the studio is that it has an excellent "live" room that we wanted to use. At that time, that room had a rented Neve [console] and was in the process of being refurbished. They ended up selecting a Trident console, which also worked in our favor since Geoffrey was very familiar with it. The Automatt had what we wanted: a good room, good board, good outboard equipment and good people. The only change I made . . .

well, I didn't particularly care for the monitors. So I brought in some JBL 4350s, which I'm really used to hearing. That was the only real change.

MR: You don't feel you have to go to L.A. or New York to get a good sound?

KE: No, not at all. The Cherokee room in L.A. is a good-sized room, but it's a little smaller than the one we're using now and we had to do things like tear down equipment between sessions. Here we don't have to worry about that.

MR: How did you prepare for the recording of the album?

KE: It all started on the road with collecting songs and album ideas. Once the ideas were down, we'd rehearse them during sound checks. We came into the studio with nineteen songs and put down fourteen basic tracks. We'll select eleven for the album without getting into too much of a time problem.

MR: Was any material worked out in home studios beforehand?

KE: Yes, Neal (Schon) has an 8-track system at his house using a Teac 80-8 with a Sound Workshop board. We endorse Electro-Voice products, so they supplied us with microphones and such, which helped keep the cost down.

With what we're doing here mixed on a cassette, Neal can transfer the cassette over to his 8 track and experiment with guitar leads, check out harmonies and things like that to see what will work. That saves time, which is one thing you want to do with expensive studios. It really is important for a band to be well prepared—that's what pre-production is all about.

MR: Are you trying for any particular "sound," like a thick sound or a bright sound?

KE: Not really, we're just trying to come up with the best sound for each song. Without a doubt you'll know every cut is a Journey cut, but we're not looking for anything anyone has done before. Some songs need more vocals, some more bass; we concentrate on whatever makes the song stand out.

MR: Do you therefore approach recording as more of a documentary of the "live" sound?

KE: The attitude of this band is that we want to be able to reproduce this material "live" like it sounds on the record. We don't want to do outrageous sound effects, or overdo it on banks of vocals or guitars. Our approach is towards making the best

Producers Geoff Workman (left) and Kevin Elson at the Trident console.

quality record, so we do use the studio where appropriate; after all, playing "live" compensates for any little extras you use on an album to get the sound across. But we won't have a symphony orchestra come in to do string parts, we'd rather use synthesizers.

MR: I understand Roy Thomas Baker has a very different approach, going for a big sound with massed vocals and such. Did the band not like that? Or did they go for it initially, then change their minds when they had to play it "live?"

KE: They liked it at the time, because it fit the material they were writing then. We could try that now and it would probably work well with some of the cuts. They were happy with the ideas Roy had . . . that's a hard question to answer, really. What I think he did best was handling the vocals, since he brought out the vocal lines very well. Steve (Perry) has a lot of his own ideas, but Roy's contribution was to really integrate the voice well with the band.

MR: Journey has always struck me as going for a powerful type of sound. How do you translate power into the restricted range of an album?

KE: Basically, we try to record very hot; we use a lot of room miking to get the ambient quality of the room, which is another reason why we go for the big room. We put as much level as we can on the tape, which makes it a bit stronger and puts the sound right up front. With close miking sometimes the sound is too separated. It works well with some records, but it's not the right approach for this time around.

MR: Geoff, how do you feel about close miking?

Geoff Workman: I'd sooner have a band utilize the whole room and have close visual contact so that they don't feel like they're sitting in a box somewhere. In that respect, you sacrifice separation . . . but I like to use a lot of room mics anyway, maybe four room mics on a drum kit. Obviously that's going to pick up some bass and guitar, but that's an integral part of the overall sound, and to try to eliminate it seems rather pointless.

I've been using distant miking techniques for as long as I've been engineering; one mic in front of an instrument doesn't always capture the sound you want, although it's fine for some material. I couldn't see, for example, a Steely Dan or Earth, Wind and Fire record approached in the same way I approach recording. My technique matches the bands I work with.

MR: Do you mix the ambient sound on to separate tracks?

KE: We have stereo front and stereo back ambience tracks, and we also

Steve Perry on Recording Vocals

While Journey is undeniably a team effort, perhaps the most visible member of the group is lead singer Steve Perry, who also shares vocals with Gregg Rolie and other members of the band. He had some interesting comments on the recording process, which follow below.

MR: How did you decide to be a singer?

SP: I played drums until '71 . . . I was a lead singer/drummer, like Buddy Miles, or the guy in Rare Earth, or the drummer in the Standells if you want to go back that far. Finally, my voice ended up surpassing my drum technique so I decided to be a singer. I play enough guitar to write songs, but would never play it on stage.

MR: How do you psych yourself up for singing in the clinical confines of the studio?

SP: That's a major problem, and that's why I don't like working with certain producers; things sometimes seemed so sterile with them.

MR: You mean they just sit there and say, "Roll the tape"?

SP: No, that isn't the problem. You open your mouth first time of the day, and Lord knows your voice would be filled with cobwebs; they push the talkback button with comments like, "That was dreadful." Two or three of those, and my balloon's out of air. The confines of this medium have enough things to psych you out . . . I didn't need to accumulate any new ones.

MR: Why would you work with such producers if you didn't get along all that well?

SP: Well, I'm just the singer! [laughs] There's more than one guy in this band, plus the record label has a secure feeling with a certain type of producer.

MR: Do you prefer doing a "live" vocal and keeping it, compared to over-dubbing?

SP: I do three complete vocal tracks, mix down the best parts to a compilation on a fourth track, and let that give direction to the song. Then I'll go back later and re-sing the track; sometimes it will be kept, sometimes it won't.

MR: Do you find that bouncing destroys some of the continuity?

SP: Well, you have to keep that in mind as you're bouncing.

MR: How did you get interested in studio techniques?

SP: I was working in a studio in Los Angeles as a tape operator, and did that for about two years because I couldn't find a gig. The producer working with me was an engineer there, so after hours we'd work on vocals to try to get me a demo. In the meantime, I was learning a lot about the studio. Prior to that, in my home, I had a two track and learned a lot about editing and vocal stacking. I had an early sound-on-sound unit, and I'd do tons of ponging . . . back and forth . . . I'd found out about sibilance building up, tiny timing errors, that type of thing . . . what worked and what didn't.

MR: How do you feel about limiting?

SP: I'm very opposed to too much limiting. Geoff will limit a bit, but he gives me the most dynamic range possible. He's also very talented at riding the fader, which helps. The limiter doesn't have any smarts when riding gain, but a human does.

MR: Does the whole group sing on choral parts, or is it mostly you?

SP: The entire band does a lot of singing. I don't have the same tonal quality as anyone else, so there's a real plus to the sound when you have a number of different singers.

MR: You have a remarkably smooth high range. How do you keep it in shape?

SP: I don't smoke cigarettes or pot to speak of. It's not that I don't dig pot, it's just that smoking has no place in my vocal instrument. Also, drinking [alcohol] dehydrates you, and that's the enemy of your vocal cords. Your cords have to be lubricated, so I drink a lot of water.

MR: How did you extend your range?

SP: I can't really explain how my range increased. One day it just was there, and I have no idea how it happened. Sometimes the key to doing something is not thinking about it, and just letting things happen naturally.

MR: Are there any effects you like to have on your voice?

SP: The only thing I really like is a "live" chamber with a warm bottom. I like sibilance through an echo chamber; I prefer sibilance on the chamber sound instead of the basic track. It gives a kind of personal feeling. If you push a note real hard and put it into the chamber, it will hang over for a bit. You can then sing a soft note and the two will harmonize nicely. As far as electronic effects go, I'm not that interested. We'll do things that accentuate what I am, but we don't try to make me something I'm not.

MR: Have you ever felt limited by being self-taught? Do you want to take additional training or voice coaching?

SP: I'm just going to expand on what I'm dong now. If I start getting non-productive, or if my singing slacks off, then I might do something. Voice is difficult; it's temperamental, and it has so many variables. An instrument can sometimes be impersonal, but there's no way you can consider a voice impersonal.

have an ambient guitar track in addition to the close miking track.

MR: Geoff, do you have problems with print-through or leakage with high recording levels?

GW: I've never run into those problems, even with the Stevens 2", 40 track. If you isolated every single track, you might find something bleeding through; but no more so than you'd get from the mic bleed. No one's going to hear it.

MR: Do you agree that tape saturation sometimes gives character to a sound?

GW: If you record a snare drum and really crank it on there, the limitations of the tape and tape electronics become part of the sound. A lot of times this sound is desirable.

MR: So you'll trade off feedthrough for feel anytime . . .

KE: Well, we figure that if we still feel it after hearing the thing so many times during recording, then when people hear the album hopefully they'll feel what we felt. Most of the basic tracks are done "live," and so are some of the solos and vocals.

MR: How do you cope with hearing the same pieces of music over and over again?

KE: It does get a bit ragged sometimes, which is a good time to walk out of the room and play a game of pinball! We decided to do the basics by giving a couple of shots at the songs we had planned to do that day; if it didn't work out after a couple of tries, we'd move along to the next song. If it didn't happen, we figured we'd get it another day. We moved along really quickly, and did fourteen basic tracks in fourteen days.

MR: No "take 39?"

KE: No, a lot of what we have is first- and second-take stuff. We'd go ahead and try a 3rd or 4th take to see if we could get a better feel, but things would usually decline after that point. We took a pretty carefree attitude. We'd see how the band felt when they came in; if they were in a joking mood, we'd do a rocker. If they were half-asleep or kind of relaxed, we'd do more of a ballad type of thing. It all relates back to a feel and closeness between producer and artist. We hang out together, we go out drinking afterwards together.

MR: Do you think your attitude has been influenced by the New Wave backlash that regards rawness and intensity as more important than perfection?

KE: Well, I don't know . . . maybe after the very disciplined type of approach Roy used in the past, I think the band was ready for a change. I've always had that type of attitude anyway, and so has Geoff.

MR: When do you schedule your sessions?

KE: Our rehearsal schedule was 10 am to 3 pm, although we go a little longer in the studio. Journey is comprised, pretty much, of early people; they're not real blazers, where they go to 3 or 4 in the morning. They like to use the daytime, when there's the most energy, and have the night off to do whatever they want to do. If you've got business to do during the day and then go into the studio, you'll be worn out. We felt a daytime schedule would give the most productivity.

MR: Since you don't have much of a technical background, what do you do to keep up with the state-of-the-art? Do you find yourself omitting options because you were not aware of them at the time?

KE: I constantly deal with people who are updating equipment because of the "live" work I do, and it's much the same thing as working in the studio in terms of equipment. I learn more from working with things than I do from reading about them. I'd like to know more, but just don't have the time to really study up on that type of training. I think if I did, I might tend to record in a way where I would think too technically. You know, not trying something because I've been told it's the wrong thing to do. A lot of the people I associate with will say try this or try that, so I get a lot of word-of-mouth learning. Geoff, of course, works year round in the studio, so he's up on all the latest gear. Neither of us can tell you how something works, but we know how to use it.

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MR: Geoff, how do you feel about electronic effects?

GW: The electronic effects that are available now at the touch of a switch are very good, but don't come close to the effects you used to get by working over a 4 track all day. Nothing will ever re-create the sound of true tape phasing (flanging), or a tape machine going at 60 ips for a DDL effect. Electronic effects just don't seem to have the character of what you are able to get with tape.

Ohm's law is of no consequence to recording music, and I even lose track of microphone model numbers and all that. I'm always willing to prostitute a certain piece of equipment, short of damaging it, such as creating a fuzz sound by cranking the mic gain. Engineers with degrees would tell me you can't do that, it will distort! But of course, that's what I wanted to do . . . that was the noise I wanted. What's happening now with university courses and all that is that you're going to end up with a glut of engineers who are brilliant theory-wise—but it's the two flaps on the sides of your head that count. Your ears, not a machine, should analyze the sound of a room. I always carry a reference tape with me of something I know well, like the last project I worked on, then I let my ears tell me how the studio affects that tape.

MR: So it's not that important what type of equalizers and other equipment you use, then . . .

GW: Oh, I use a lot of EQ, sometimes more than a lot of people would think is necessary. I'll boost a particular frequency on record and on playback, for example. But if it sounds right, it's fine. I don't go back and feel ashamed because I added 19 dB or something. The way I adjust the parametrics on this board is I just pull everything out, and sweep everything until I find the frequency I'm looking for so I can boost or cut it. I don't predict, "That must be between 800 Hz and 1.2 kHz" and work from that premise.

MR: Did your career begin with 4-track recording?

GW: I started engineering in 1968, and that was straight stereo on a lot of dates. Then 4 track, ping-ponging and all that. The first King Crimson album was all done on an 8 track, with humongous amounts of bouncing back and forth, back and forth, then mixing it down into stereo and hand syncing it back into the track again. That's probably why I find splicing tricks and stuff like that to be real easy. My forte was chopping up pieces of tape, and putting them in backwards, or using tape phasing, or whatever.

MR: Many engineers who started on 16 track can't do things like that . . .

GW: People still don't realize the easiest way to get backwards tape on some machines is to lift the tape and put it around the back of the capstan and in front of the pinch roller. As long as you have someone there paying attention to the tape motion, and you disable the erase head, this is a lot easier than flipping the tape over, changing the monitor balance and figuring out which track is which when the tape is backwards.

MR: Are you using any noise reduction on this project?

KE: We're not using any at all. Running the tape at 30 ips and recording louder music means you don't really need noise reduction all that much.

GW: When Dolbys first came out, they were a necessity because recorders and tape weren't that good. But as tape became better and better, and machines became better, the need for noise reduction was reduced—especially with rock material. There's nothing recorded at a low level that's going to be prominent in the mix later.

MR: What about limiting?

KE: We use a combination of Urei, Compex and dbx.

MR: How do you choose one over another for a given application?

KE: They all have readily indentifiable "sounds." The Urei has a harder sound that works well with guitar and keyboards. Dbx on the bass gives a rounder, punchier sound, and the Compex works great with vocals. Why, I don't know.

MR: How do you define a "good sound" for limiting vocals?

KE: Well, you don't hear the real limiting happen. It's very smooth as it limits; you don't hear any sucking. Steve (Perry) has a lot of range and can hit some real strong notes, so you need a limiter that's not obvious.

MR: I take it you're not a purist who feels limiting destroys the sound.

KE: I'd prefer to ride gain with someone like Ronnie Van Zandt because he sang in pretty much the same dynamic range. That's what I do with Steve "live," but it doesn't work out too well in the studio. With the Urei you could really hear him hitting. I don't think it was our doing. I just think that different devices have different sounds.

MR: How do you record the drums?

KE: We're using Sennheiser 421 microphones on the toms, because they accentuate the sound. A lot of people will use the Shure SM56, but for the toms we like the Sennheisers better. We used 56s on the snare.

MR: How does your mic selection process work? Do you have certain mics you like, or do you try different ones all the time?

KE: There are certain mics I'll use all the time on drums because they do a good job, but for room miking and guitar miking it will vary quite a bit. I picked up an old AKG D12 on tour in Europe, and it's got a kind of ribbon sound . . . there's a boost at 100 Hz, and it's perfect for a kick drum. Just put it on flat, and it's got an amazing sound.

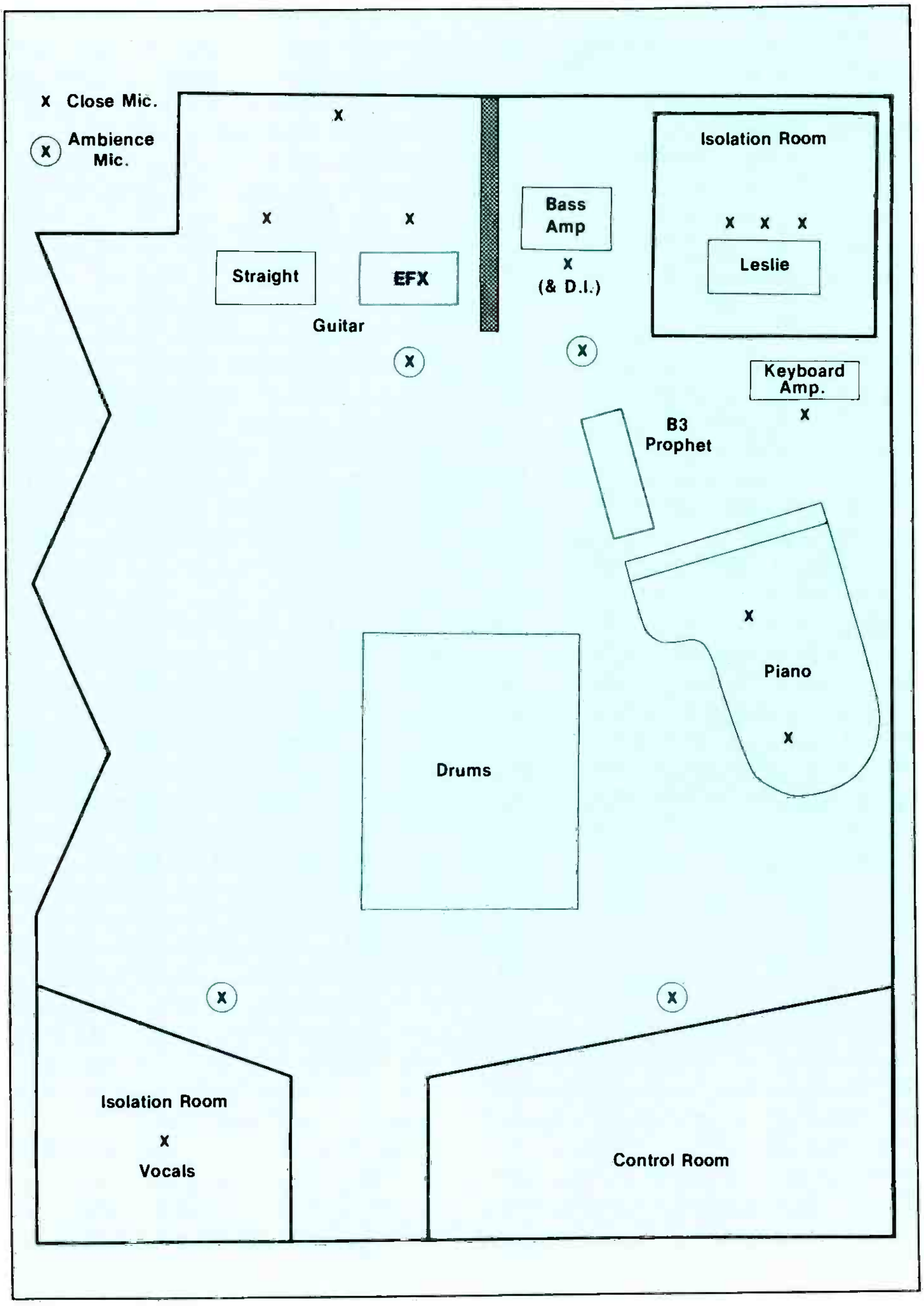
MR: So you'd rather take a mic with personality than use a fairly colorless mic and add equalization?

KE: Yes, although it also depends on, say, how the drums are tuned. Steve (Smith) is very particular about the tuning of his drums and heads, so that makes the mic selection just that much more important. Most of the problems I've found with drummers involve getting snare sounds; some want a real "snare" sound, and others want a boxy midrange sound. I like a fat sound, myself.

MR: Do you use any direct injection?

KE: On the synthesizers we have mics on the amp system he's going through and we have directs. That way we have a choice of what we want to do with the individual sounds. Synthesizers taken direct tend to sound kind of buzzy, whereas a speaker will tend to smooth all that out. The main synthesizer we're working with on this album is the Prophet 5, and it's amazing.

MR: How do you feel it compares to



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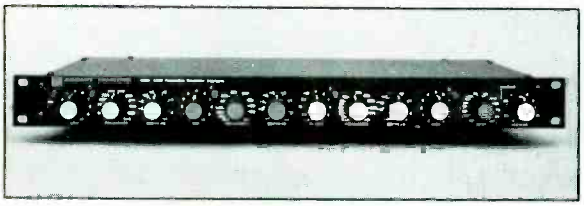

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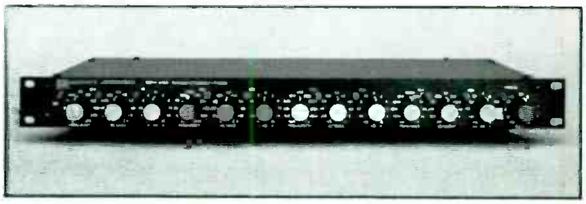

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other synthesizers for recording?

KE: Basically, you know what kind of sound it has before you have to start laying it down; there's no patching or other complications. I've never worked with modular systems much.

MR: Does the Prophet have tuning or drift problems?

KE: Yeah. In the higher registers, when you're playing full chording you have to be careful, since the high octave note might flatten out a bit. I don't know if it's the nature of the instrument, or whether this particular one just wants a trip to the shop.

MR: You can always set everything an octave lower and record at half speed to avoid upper register problems.

KE: Well, single notes come out just fine, and Greg's still using the Minimoog for certain sections. It's pretty much a trouble-free instrument.

MR: Are you using Neal's Roland GR-500 guitar synthesizer much on the album?

KE: Yes, we're using it on a couple of background filler-type sounds. When we were in Japan, we got some extra equipment that extended the versatility of the unit. Neal likes gadgets a lot, and loves the GR-500.

MR: Have there been problems in integrating the synthesizers into the tracks?

KE: Actually, it's all falling together pretty naturally. The keyboards and guitars are pretty much clean and up front, and the synthesizer stuff is more or less in the background.

MR: Sort of like a string synthesizer.

KE: Yes.

MR: Are there any special effects you like to use while recording?

KE: Greg likes an Eventide Harmonizer set at 1.01 with his voice; that's one example. I personally like using tape recorders for delay, because the sound is ultra-clean and natural. Instead of phasing the cymbals, we'll use variable speed delay on the MCI multi-track to give a "wash" of a sound. We use more natural things than actual special effects.

MR: Are you using 24 tracks, or two 24-track machines synched for 48?

KE: 48 tracks was an option, but we felt that if the band knew they had two machines they could keep playing with, things could go on forever. We haven't had to sacrifice anything with the 24, except that we've had to put some vocal tracks on early so we could mass them together without running into track space problems.

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MR: What sorts of things do you look for in a board?

KE: One of the most important features to keep the musicians happy is a good cue system. Otherwise, it causes synchronization problems with the basic tracks if they can't hear each other properly.

MR: Doesn't the Automatt have some kind of special cue system?

KE: Yes, it's called Automix, and it gives each musician four different cues that he can mix. The Trident has a separate monitoring board, with four cue sends that are either echo or monitor sends. We'll give the bass one control on the musician's mix box, drums another control, guitar/keyboards the third control and finally vocals get their own control. This allows each musician to get his own personalized cue mix. Our hardest problem in recording the basics was that what came through the headphones sounded like a transistor radio compared to the sound the band was getting in the room. But for a board, as long as the muting is quiet and there aren't any bad faders . . . as long as the board is clean, you can adapt to different EQ setups and any other minor differences between boards.

MR: Is there any type of EQ you particularly like?

KE: One thing I like about the Trident console is that the slide pots on the parametric EQs are faster to set up than rotary pots. On Harrison consoles, I like the fact that you can do a lot of basic EQ with the various low and high pass filters, and you don't even need to touch the EQ modules until mixdown.

MR: What type of reverb do you favor?

KE: I personally like a natural room chamber. The best I've ever heard is at Studio One in Atlanta. I can hear a song on the radio and tell by the reverb whether it was done at that studio or not 'cause the reverb sound has a certain edge to it. I asked Rodney Mills (the designer) how he got that sound, and he said it was the first one he had ever done and he had no idea how to recreate it. But if you listen to the Atlanta Rhythm Section, Alicia Bridges, or Skynyrd's *Street Survivors,* there's a certain quality to the chamber that is unmistakable.

Here at the Automatt we have stereo EMT, Lexicon digital reverb and a "live" chamber. When we get into mixdown, we'll use the EMT on some of the guitars and room reverb for the vocals.

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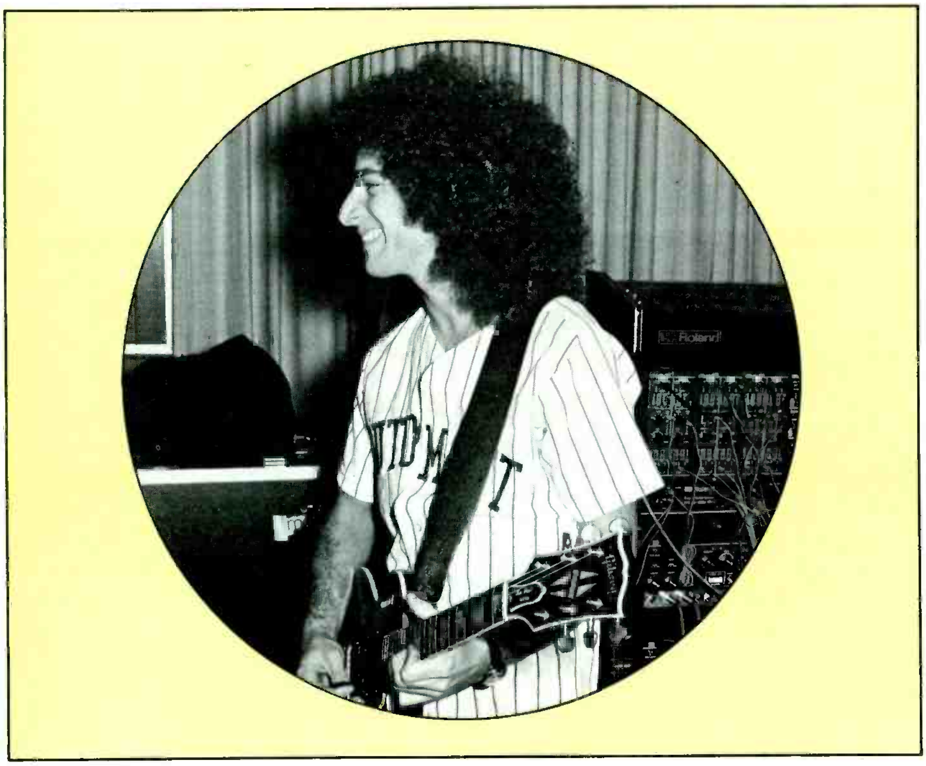

Journey lead guitarist Neal Schon while at work at the Automatt.

MR: You mentioned a "live" album earlier. Is this something you've been recording all along, and you'll just have to cut and paste what you want to make the LP?

KE: The "live" tapes started out as sound for a radio show, but the tapes turned out really good. We won't have to do any fixing in the studio. We have about seven reels of tape, and we'll sit on those while we record this tour. Then we'll pick the best from the lot. We're using the same truck for all our recording to get a consistent sound.

The most important thing to me is that I believe you can record a "live" album almost anywhere. To tell you the truth, I prefer a big, boomy hall to get the excitement of the crowd and the excitement of the building. The ambient mic placement is particularly important, and I spend a lot of time on that. Also, instead of doing post-production effects, I feed all the effects lines to the mobile truck so I can do all the effects in real time. This means that when it's time to mix, I mix just like a "live" show because all the tracks and all the effects are there. I think a "live" album is the feel of the audience and the band, so I'm careful not to make the sound too clean.

MR: This relates to the earlier question of how you fit a powerful sound into those little tiny record grooves . . .

KE: It *is* hard. Mastering is the whole key to the thing, actually. Whatever you do here [in the studio] doesn't matter if you are going to end up with a rotten mastering job.

MR: Do you do the mastering yourself?

KE: No, I supervise the process but wouldn't know how to do it myself. George Marino at Sterling Sound [in N.Y.] does most of the things. I used Artisan in Hollywood for a re-package compilation we just did. I like to get people who want to be the best, who aren't just doing it as a job.

MR: What production tool would you like to see invented from a technological standpoint?

KE: To me, just the ultimate cue system. Studio equipment now is generally superb, and making the musicians happy is the most important consideration because that's where everything comes from.

MR: When mixing, do you ever use little speakers like Auratones?

KE: Yes, for sure. When we listen to the final basics to pinpoint problems, the Auratones don't hide anything. At high volume with good monitors, it's bound to sound pretty good no matter what. You don't have that happen with Auratones.

JOURNEY MIC LIST

Drums	
Kick (primary)	AKG D-12
Kick (secondary)	Electro-Voice RE20
Snare (top)	Shure SM56
Snare (bottom)	Shure SM56
Toms (except rototoms)	Sennheiser 421
Rototoms	Sennheiser 421; Shure 56
Ride cymbal	AKG C-414
Ambience mics	Neumann U87
Hi-hat	Sony C-22
Elec. Guitar	
Close mic	Shure SM56
Distant mic	Neumann U47 (tube)
Bass	
Mic	Neumann U47
Direct	Jensen DI box
Keyboards	
Piano (with isolation box on top for tracks)	Neumann KM84
Prophet 5 (direct)	Jensen DI box
Prophet 5 (amp mic)	Electro-Voice RE20
Minimoog (direct)	Jensen direct box
Minimoog (amp mic)	Electro-Voice RE20
Hammond B3 (top mics)	Neumann KM88 (2)
Hammond B3 (bottom mic)	Electro-Voice RE20
Vocals	
	Telefunken U67
Outboard Limiters	
Guitars	Urei 1176
Bass	dbx 160
Vocals	ADR Compex

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MR: Now that you've become so involved in the process of recording, will you continue to do Journey's "live" sound?

KE: I really enjoy doing six months a year in the studio, and the other half of the year on the road doing "live" shows. The "live" side of it is real challenging.

The P.A. system I use is like a giant studio monitor system—no radial horns, just a big paper system that's really smooth. Certain groups sound good on this system, but other groups couldn't use it. Real raunchy groups need 18-inch horns and that overkill radial horn sound. The system I use is from Canada, and was built for Emerson, Lake and Palmer. It's the same system that Yes, Billy Joel and Fleetwood Mac use. From those groups, you can see what's needed in the sound: punch on the drums, good vocals and a clean sound. You can be sitting in the audience on-axis with one of those cabinets, and it won't destroy you like a horn would.

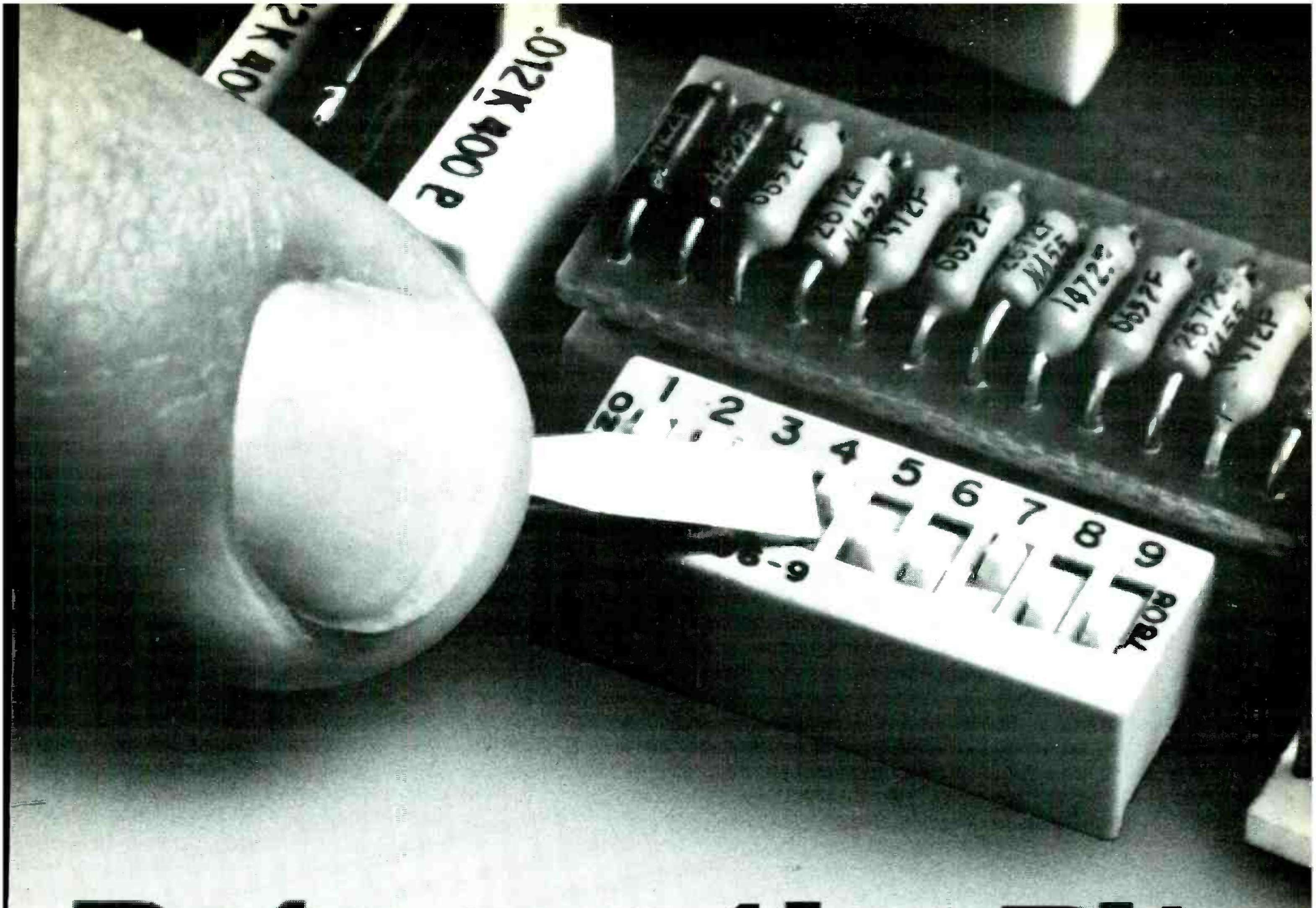
MR: That makes me more inclined to see a Journey concert. I'm the type of person who wears cotton in my ears at concerts.

KE: I don't blame you. How much I like a group depends a lot on the system. I've had to mix Journey on horn systems overseas, and Steve's voice hits right in the horn range. It can not even be loud and still kill you.

MR: What would the two of you advise musicians to do to make life easier for the producer?

GW: Oh dear, there's a question. Well, the guys in Journey and me have a lot of running gags going! Seriously though, there are three different categories of producer/client relationship. One is purely business relationship between band and producer, where they don't really meet except through their lawyers. Then there's the producer/engineer, but that's a little split for me because I have to do two things at once. The third situation is what we have now, where Kevin works with the band all year round, he knows the equipment, the members, the material, everything. I've worked with Journey on two previous albums, so we all know each other well and there is a particular kind of trust. They know I'll record them well, I know they won't come up with a bunch of bad songs and Kevin can keep an overall eye on everything. Every project is totally different, so different musicians need to do different things to adjust to different producers. Trust is really a terrible word to use, but that's what it comes down to. The band is putting its career in my hands, but for me, if the album doesn't come off that doesn't reflect well on my talents.

KE: One thing that would really help the producer is that in rehearsal, the band members should closely examine the rhythm section—bass and drums. After getting familiar with the material, it's good to let everyone else sit out and just listen to the bass and drums, to see if they're locking together right. As soon as things get tense because of uncertainty, the situation gets rough and frustrating. The saving grace is to have every person knowing what each other member is doing. Otherwise, you have to start restructuring in the studio . . . at $150 an hour. If everyone comes in well-prepared, it makes life so much easier.



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The ELECTRIC Primer -Part VI

By Peter Weiss

This month, we start off with an apology. Although we are usually very careful about mentioning the names of the scientists responsible for discovering or developing the electrical and magnetic principles we've been covering in this series, Part V did *not* mention Michael Faraday (1791-1867). Michael Faraday was a championship-class scientist and experimenter, and his work led to all the motional electromagnetic effects we discussed in Part V. So, apologies to our readers and apologies to the memory of Michael Faraday.

In the previous issue at the end of Part V, we were just beginning to talk about an electrical property possessed by certain coiled conductors, and we called this property *inductance.* The proper term for this property is actually *self-inductance,* but after that business with right-hand rules last month, we'll just stick with inductance. Inductance comes about as a result of the electromagnetic phenomena that occur in and around a coiled conductor. One of the main features of inductance is the "back-EMF" or opposing voltage that is developed when changes occur in the amount or direction of a voltage applied across an inductance. The polarity of this back-EMF is always such that it opposes the change in current (see Part V). So far, we have discussed only those conditions that result from the application or interruption of a direct current. Before we move on to a description of alternating current effects, there are some points to cover concerning inductance and direct current.

In our discussion of capacitance in Parts III and IV we investigated the time-related energy-storing properties of a combination of capacitance and resistance. An inductance (or *inductor*, a component having inductance) also stores energy, not in accumulated charge like a capacitor, but in the electromagnetic field that surrounds a current-carrying inductor.

The property of inductance for a coil of wire varies with the shape of the coil (not all inductors are cylindrical coils; see *Fig. 1* for examples of different inductor shapes), how many turns of wire make up the coil, spacing between turns, the physical dimensions of the coil and whether the coil is wound around any magnetically active material such as iron. The unit of inductance is the *henry*, named for Joseph Henry (1797-1878). The amount of back-EMF developed by an inductor in response to a given change in current is directly proportional to its inductance.

In our investigation of combinations of resistance and capacitance, we mentioned that it takes a certain amount of time for a capacitance to become charged (to 63% of full charge) in a circuit containing both capacitance and resistance. This charging time (t, in seconds) could be found simply by multiplying the values of the resistance (R, in ohms) and the capacitance (C, in farads), or

$$t = R \times C$$

This result is also the time it takes for an R-C combination to discharge to 37% of full charge.

Inductance in combination with resistance also demonstrates time-related effects, and we will trace the process verbally first, then state the formula that applies. *Fig. 2* is a schematic of a circuit consisting of a battery (E_B), a switch (S_1), a resistance (R) and an inductor (L). The resistance represents the series combination of the internal resistance of the inductor (usually very low) and any other "pure" resistance in the circuit. Therefore, in this discussion the inductor itself can be a "pure" inductor, with no internal resistance. At first, it might seem that there would be no time-related effects in the circuit of *Fig. 2*; the inductance is just a coiled conductor, having zero resistance, so the current, I_T, should rise to its Ohm's Law value,

$$I_T = \frac{E_B}{R}$$

instantaneously. This is not the case. At the instant the switch is closed, current begins to flow around the circuit. However, *at the same instant*, as the current begins to flow through the inductor, a back-EMF develops across the inductor, its value being proportional to the value of the inductance and the rate at which the current is changing with time. Since going from a no-current to a current-flow condition represents an instantaneous change, the back-EMF is sufficient to prevent current flow completely at the instant the switch is closed. This opposition slows down the rate of change of the current, and the back-EMF is reduced as a result. This process of accommodation continues as the current rises smoothly towards its full Ohm's Law value. *Fig. 3* is a graph of current versus time in the circuit of *Fig. 2*. The time it takes for the current to reach 63% of its Ohm's Law value can be found by dividing

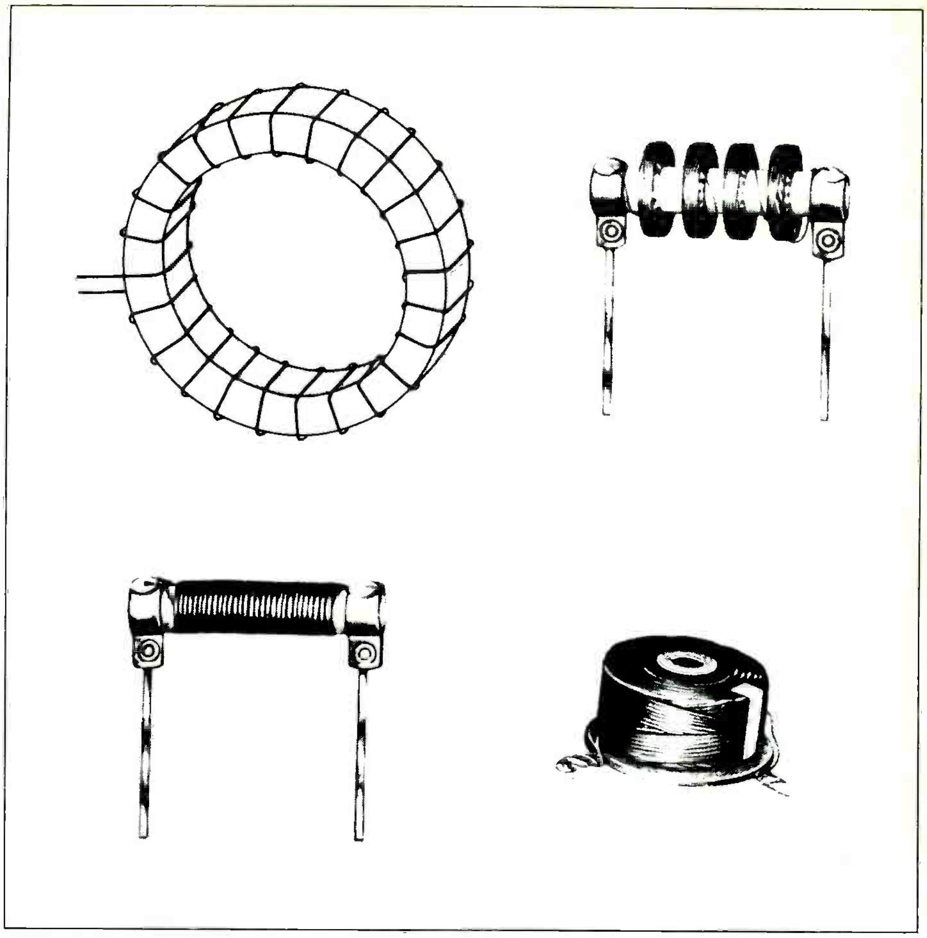

Figure 1

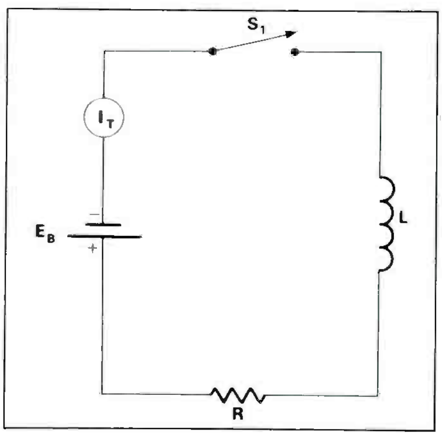

Figure 2

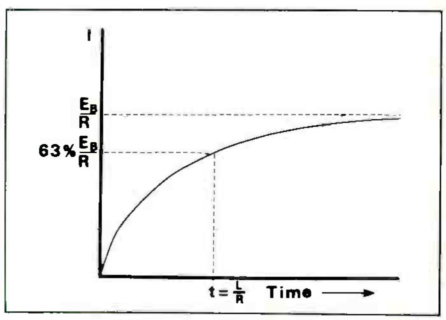

Figure 3

the value of the inductor (L, in henrys) by the value of the resistance (R, in ohms). Stated as a formula:

$$t = \frac{L}{R}$$

The opposition to changes in current flow that characterizes inductors makes them very useful in "traditional" DC power supply circuits. In this application, inductors called "chokes" are used to smooth out rapid variations in the output current of the power supply. Inductors are used in a wide variety of audio applications, both as simple circuit elements and as electromagnetic components in devices such as transformers, dynamic microphones, phono pickups and speakers. Before we go on to these important applications we must first examine how inductors and other circuit elements behave and interact in alternating current circuits.

The subject of alternating current circuit analysis (at least our approach to it) is ultimately not much more complex than the DC circuit analysis that we've discussed so far. However, before we get to the easy part, there is a wide and deep swamp of math-type information that we have to wade through. So wide and deep that instead of providing a separate set of Math Notes, as promised at the end of Part V, we will take the remainder of this installment to discuss relevant mathematical concepts. A warning: mathematics, at any level, is a dry, undramatic topic. The reader should not expect the material in the remainder of this segment of this article to read like a novel. It's really tough to fit sex, violence, greed, corruption, sex, passion, betrayal, sex, history and sex in among a bunch of formulas and graphs. We've tried it and we know. Anyway, the material that we present is necessary to a better understanding of the alternating current theory that will be given here and in Part VII. For those of our readers who are familiar with the mathematical topics covered here, this segment of Electric Primer will serve as a refresher and review. To those of our readers who are new to these subjects, it will be a (hopefully) gentle introduction. Everybody got his swamp boots on? Here we go.

So far, in all of our discussions, the only kind of number we have encountered has been the positive number. We never had any need to call these numbers positive, or anything else, because there wasn't any other kind of number with which to compare them. Now, however, to help us progress through the topics to be covered in Part VII and the remainder of Part VI, we have to identify and define two kinds of numbers: *positive*, and *negative*. Positive numbers are already familiar to us, and we use them to express weights, lengths, resistances, age, etc. Negative numbers are also in everyday use, but not quite as noticeably. For example, when the outside air temperature falls below 0° Fahrenheit, we say something like "the temperature is 5° below zero, Fahrenheit," or "minus 5°, Fahrenheit," or write "−5° F." The written form given last actually demonstrates the basic definition of negative numbers. That is, if "zero" is a selected reference point on a scale (like a thermometer scale), than any point to one side of the zero point can be *assigned* a positive value, while any point on the *opposite* side of the zero point can be *assigned* a negative value. In general, the selection of a zero point and the assignment of positive and negative portions of the scale are arbitrary, and any choice is good as long as everything is labeled and defined clearly. In the case of the Fahrenheit thermometer scale, a reading of −5° is indicated when the top of the thermometer mercury column is at a point five divisions (degrees) below the zero degree point.

Positive and negative numbers can be used to locate a point on a line or linear scale, not just by number of divisions or distance from a reference point, but also by *direction* from the reference point. *Fig. 4* shows a simple number line with a zero reference point (called the *origin*) and locations identified by positive and negative numbers. Negative numbers must always be written with a minus sign (−) in front of them (e.g., −1, −4, −½, −0.25). For positive numbers, the plus sign (+) is "understood," and does not have to be present, except in cases where including it would clarify a statement.

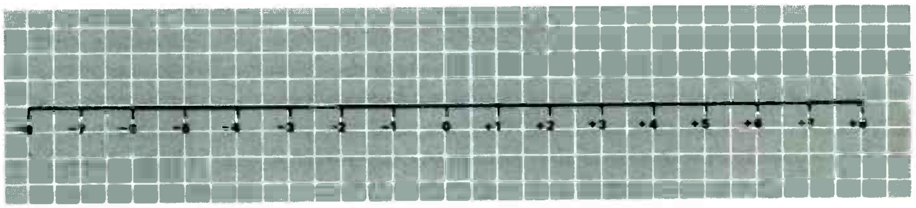

Figure 4

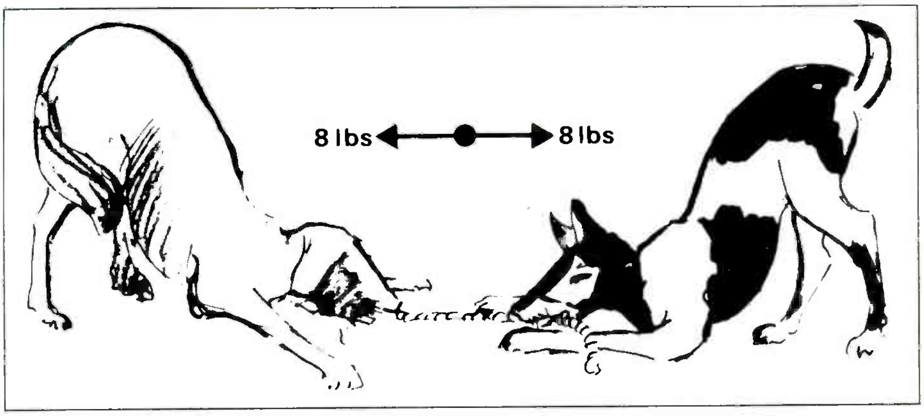

Figure 5

Another use of negative numbers is to indicate actual directions in space. For example, the laws of motion tell us that if an object is at rest, not moving, the *sum* of all the forces acting on it must equal zero. If no forces act on a stationary object, then common sense tells us it will remain at rest. But what about an object that has two forces acting on it, as in *Fig. 5* (we will forget about friction, the force of gravity and any other forces arising from it)? There is a force of 8 lbs. to the right, and a force of 8 lbs. to the left. Just from looking at the situation we can tell that the object won't move, because the two forces are equal in strength but opposite in direction. How do we fit these observations into our law of motion which says the *sum* of the forces on a stationary object is zero? After all, the sum of 8 lbs. and 8 lbs. is 16 lbs., not 0 lbs. By *assigning* negative values to all leftward forces and positive values to all rightward forces, we can now safely say that the sum of all forces on the object is zero. Stated as an equation:

$$F_R + F_L \stackrel{?}{=} 0$$

$$(+8 \text{ lbs.}) + (-8 \text{ lbs.}) \stackrel{?}{=} 0$$

$$8 \text{ lbs.} - 8 \text{ lbs.} = 0$$

In order to explain the transition from the second equation to the third, we must define a new term, *absolute value*, and some rules for handling negative numbers. The absolute value of a positive or negative number is just the number, free of any sign. For example, the absolute value of -8, written $|-8|$, is 8. The absolute value of $+4$ is 4, or,

$$|+4| = 4$$

Other examples;

$$|0| = 0$$

$$|-\tfrac{1}{2}| = \tfrac{1}{2}$$

We use absolute value as a means to isolate, verbally and mathematically, the numerical portion of a signed (positive or negative) number.

With the definition of absolute value as a tool, we can now explain the transition from

$$(+8 \text{ lbs.}) + (-8 \text{ lbs.}) \stackrel{?}{=} 0$$

to

$$8 \text{ lbs.} - 8 \text{ lbs.} = 0$$

as well as other rules concerning signed numbers.

■**Rule 1**: When *adding* numbers of the *same sign*, add absolute values and retain the sign.

Examples:

$$(+4) + (+3) = +7$$

$$(-1) + (-4) = -5$$

■**Rule 2**: When *adding* numbers with *opposite* signs, *subtract* the smaller absolute value from the larger, and retain the sign of the number that has the larger absolute value. If the absolute values are equal, the result is zero.

Examples:

$$(+8) + (-8) = 0$$

$$(-1) + (-2) + (+4) = +1$$

$$(+9) + (-12) + (+3) = 0$$

■**Rule 3**: When *subtracting* signed numbers, *reverse* the sign of the subtracted number and *add*, following Rule 1 or 2.

Examples:

$$(+5) - (-1) = +6$$

$$(-6) - (+4) = -10$$

$$(+5) - (+5) = 0$$

$$(-3) - (-2) = -1$$

$$(+4) + (-2) - (-2) = +4$$

■**Rule 4**: The order that signed numbers are in does not matter in addition or subtraction. Also, parentheses and plus signs (except those indicating addition) can be dropped, as long as the statement is clear and Rules 1, 2 and 3 are followed.

Examples:

$$(-4) + (+5) = (+5) + (-4)$$

$$(-4) + (+5) = -4 + 5$$

$$-4 + 5 = 5 - 4$$

■**Rule 5**: When *multiplying* or *dividing* numbers of the *same* sign, the result is *always positive*. When multiplying or dividing numbers of *different* signs, the result is *always negative*.

Examples:

$$(-1) \times (5) = -5$$

$$(-4) \times (-3) = +12$$

$$\frac{-4}{2} = -2$$

$$\frac{-9}{-3} = +3$$

Positive and negative numbers can be used to locate points on a flat, or two-dimensional surface. Such a flat (or *plane*) surface is called two-dimensional because only two numbers are required to determine the exact location of any point. *Fig. 6* shows two number lines that are perpendicular to each other. They are identical to the

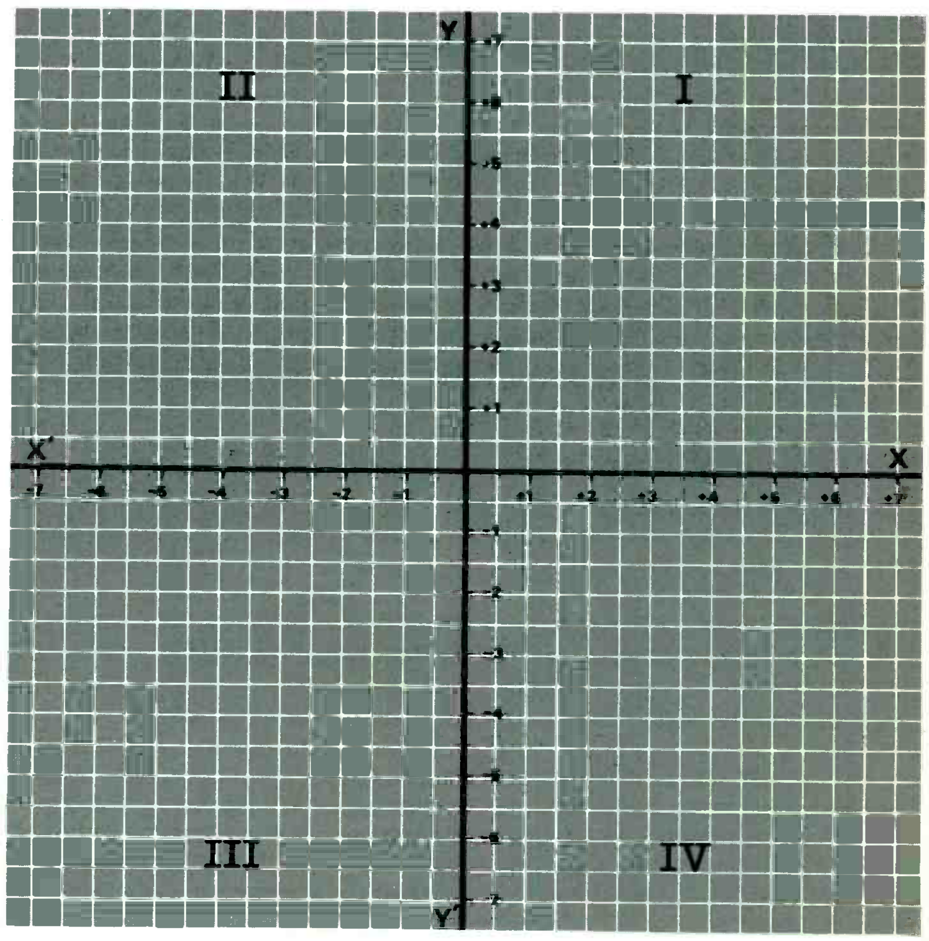

Figure 6

number line in *Fig. 4*, except that one is vertical. The point at which they cross is called the *origin*, and is labeled with a zero, since it is the zero point for both number lines. We will call the segment of the horizontal number line lying to the right of the origin the x axis. The segment of the horizontal number line lying to the left of the origin will be called the x′ (read "x prime") axis. Similarly, the vertical number line is divided at the origin into the y axis and the y′ axis. The x and y axes are labeled with positive number points and the x′ and y′ axes are labeled with negative number points. The Roman numerals in each *quadrant* (area between two perpendicular axes) are conventional labels for numbering these areas.

In order to locate a specific point, all we have to do is to specify an *ordered pair* of numbers—a pair of numbers in which position is significant. The significance we attach to the positions of the numbers in this ordered pair is as follows: the first number will always denote the point on the x or x′ axis that is directly (perpendicularly) under or over the point in question. The second number will always denote the point on the y or y′ axis that is horizontally across from the point. If we specify an ordered pair of numbers, (+3, +1)*, we can locate a point that is one unit above the x axis and three units to the right of the origin. This point, with others, is shown in *Fig. 7*. The numbers in the ordered pair are called the *co-ordinates* of the point. The first number is the x co-ordinate, the second number is the y co-ordinate. In *Fig. 7*, note the signs of both co-ordinates for points in different quadrants. In Quadrant I, both co-ordinates are positive, in Quadrant II the x co-ordinate is negative and the y co-ordinate is positive, etc. The system we have just outlined is generally known as "a system of rectangular co-ordinates," or, in honor of the man who first developed the idea, the French mathematician-philosopher Rene Descartes (1596-1650), Cartesian co-ordinates.

* Two signed numbers in parentheses, separated by a comma, is the standard written form representing an ordered pair.

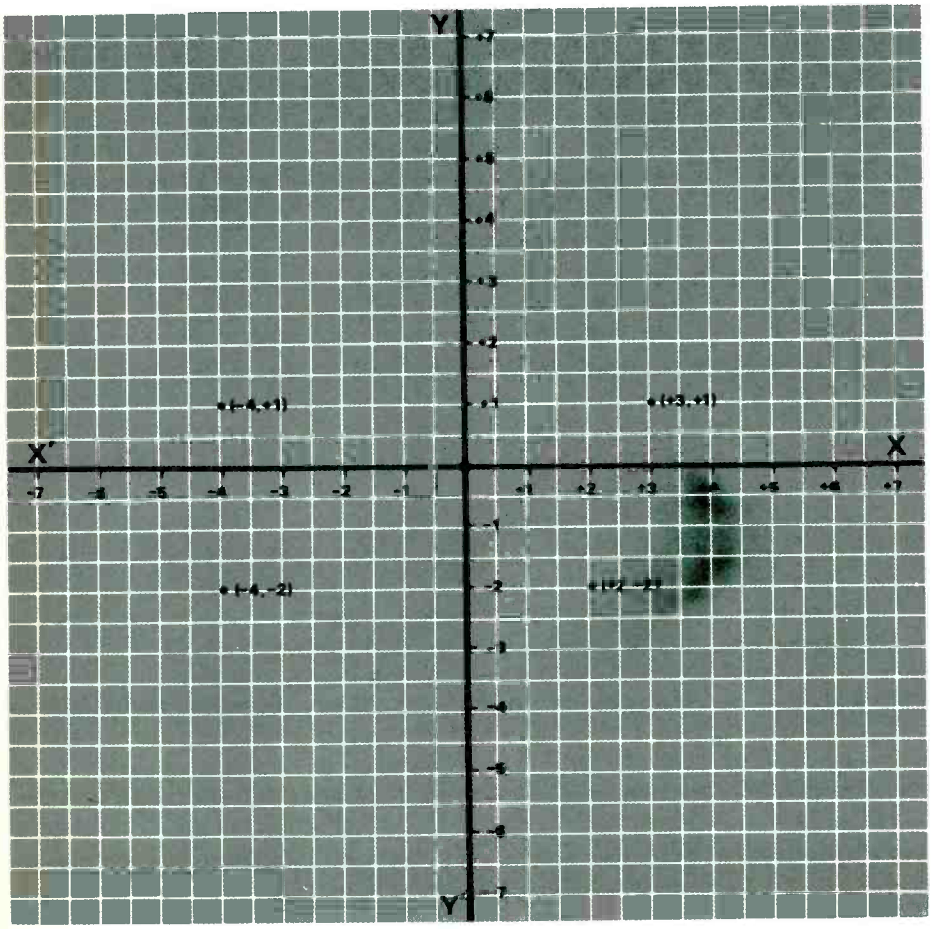

Figure 7

Very often in mathematical descriptions of things or events the notation or symbology used is a form of shorthand. One of the most frequently encountered forms of mathematical shorthand is the *exponent*.

In the expression

$$3^2 = 9$$

the small ^{2} is the exponent, and the 3 is called the *base*. The ^{2} indicates that the number it is written above is to be multiplied by itself, i.e.,

$$3^2 = 3 \times 3 = 9$$

Similarly,

$$4^3 = 4 \times 4 \times 4 = 64$$

and

$$5^1 = 5$$

Exponents written over the number 10 are extremely handy and easy to manipulate:

$$10^1 = 10$$

$$10^2 = 10 \times 10 = 100$$

$$10^3 = 10 \times 10 \times 10 = 1000$$

$$10^4 = 10 \times 10 \times 10 \times 10 = 10{,}000$$

$$10^5 = 10 \times 10 \times 10 \times 10 \times 10 = 100{,}000$$

$$10^6 = 10 \times 10 \times 10 \times 10 \times 10 \times 10 = 1{,}000{,}000$$

An expression like 10^4 is read "ten to the fourth power," or "ten to the fourth," or "ten to the four." Notice that the number of zeros after the 1 in each case equals the value of the exponent. This is true for powers of ten only, of course. Powers of ten can be used to illustrate other rules concerning exponents and exponential notation. Remember, these notational forms are arrived at "by convention," that is, they are agreed upon, and they work consistently as a system.

$$10^0 = 1$$

$$10^{-1} = \frac{1}{10} = 0.1$$

$$10^{-2} = \frac{1}{10 \times 10} = 0.01$$

$$10^{-3} = \frac{1}{10 \times 10 \times 10} = 0.001$$

etc.

For bases other than 10,

$$3^{-1} = \frac{1}{3}$$

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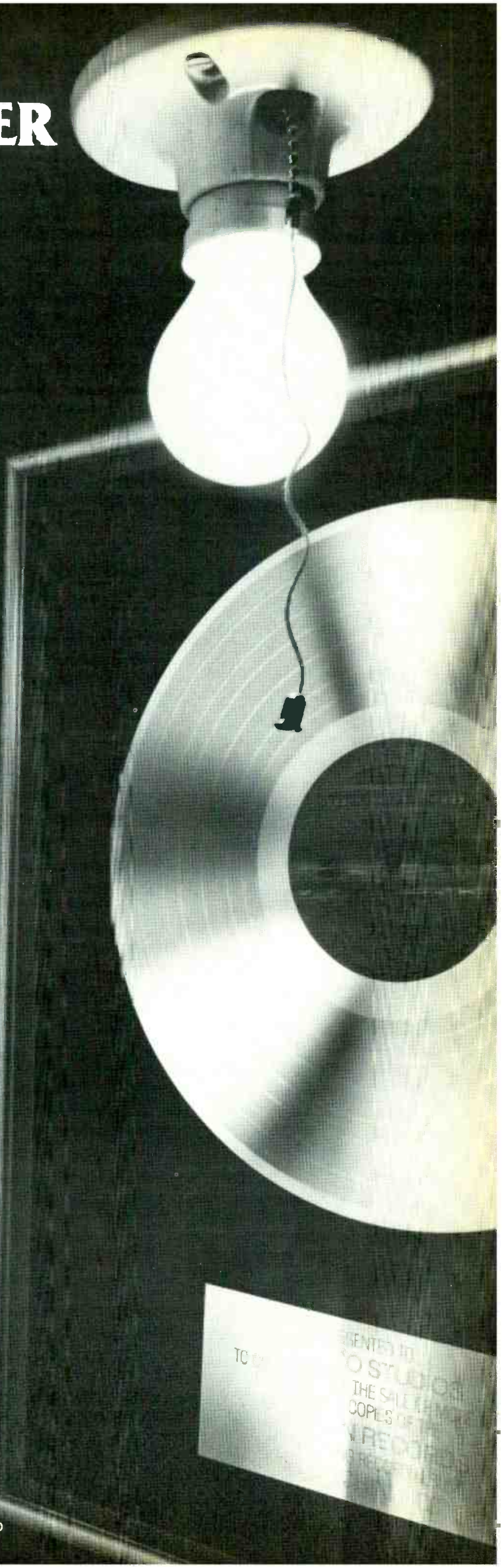

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$$\frac{4}{3}^{-1} = \frac{3}{4}$$

$$\frac{1}{4}^{-2} = 16$$

Exponential powers of ten, because of their simplicity, are used in another system of mathematical shorthand called "scientific notation." Scientific notation expresses any number as the product of a power of ten and a decimal number between 1 and 10. Here are some examples:

1500 ohms, in scientific notation is 1.5×10^2 ohms.

10,000 Hz, in scientific notation, is 1.0×10^4 Hz.

635 miles = 6.35×10^2 miles.

0.0000067 farad = 6.7×10^{-6} farad.

In examining these statements, note that the exponent indicates how many places, and in what direction, the decimal point must be moved to convert from scientific back to standard notation. Negative exponents indicate movement of the decimal point to the left, and positive exponents indicate movement of the decimal point to the right. Most of the time, a number written in scientific notation is more compact and easier to manipulate than the same number written in standard notation. However, in some cases it is easier to leave a number in standard notation. For example, 10 is more compact than 1×10^1. In this and future articles, scientific notation will be used when it is the more compact form.

In order to obtain a "feel' for working with scientific notation, and at the same time introduce a new rule for working with exponents, here is a multiplication example:

$$230 \times 0.3 = ?$$

Re-writing the terms in scientific notation,

$$(2.3 \times 10^2) \times (3.0 \times 10^{-1}) = ?$$

We can remove parentheses and rearrange terms, since order doesn't matter in multiplication:

$$2.3 \times 3.0 \times 10^2 \times 10^{-1} = ?$$

We can take care of 2.3×3.0 easily enough,

$$2.3 \times 3.0 = 6.9$$

but what about $10^2 \times 10^{-1}$? We could go back to the definitions of these exponential expressions, i.e.,

$$10^2 = 100 \text{ and } 10^{-1} = 0.1$$

Then,

$$10^2 \times 10^{-1} = 100 \times 0.1$$

or

$$10^2 \times 10^{-1} = 10$$

But

$$10 = 10^1$$

and this leads us to another rule of exponents: *When multiplying exponential expressions having the* same base *raised to different powers, simply add the exponents.* As in the example above. Completing this example,

$$2.3 \times 3.0 \times 10^2 \times 10^{-1} = 6.9 \times 10^1$$

When dividing exponential expressions having the same base, *subtract* exponents. For example,

$$\frac{6.0 \times 10^3}{2.0 \times 10^2} = ?$$

According to the rules stated regarding negative exponents and the rules for handling fractions given in the first installment of Math Notes, this statement is equivalent to

$$\frac{6.0}{2.0} \times 10^3 \times \frac{1}{10^2} = ?$$

or

$$\frac{6.0}{2.0} \times 10^3 \times 10^{-2} = ?$$

Carrying out the indicated division and adding exponents,

$$\frac{6.0}{2.0} \times 10^3 \times 10^{-2} = 3.0 \times 10^1$$

Certain exponential expressions have special names. For example, x^2 is often read "x squared." This is because the area of a square with sides of length x is x multiplied by itself, or x^2. The expression x^3 is read "x cubed" for a similar reason. The volume of a cube having sides of length x is x multiplied by itself three times, or x^3.

Sometimes it is necessary to determine what number a particular given number is the square of. That is, given a number, what number multiplied by itself will result in the given number? If the given number is 4, after a little investigation it is found that

$$2 \times 2 = 4$$

So we can say

$$2^2 = 4$$

or, "4 is the square of 2," or more commonly, "2 is the *square root* of 4." Similarly,

$$3^2 = 9$$

and so the square root of 9 is 3. It is easy to find the square root of a number like 4, 9, 16, 25 or 36, because for any of these given numbers we can easily find a whole-number factor that, when multiplied by itself, results in the given number. The square roots of the numbers just listed are, respectively, 2, 3, 4, 5 and 6. Numbers whose square roots are whole numbers are called "perfect squares."

But what if we have to find the square root of a number like 17? What number multiplied by itself will result in 17? There is such a number, it is a little more than 4, but we cannot find out exactly what it is by any method we yet have covered. To determine the square root of a number that is not a perfect square, we must use a table of square roots, a slide rule or a calculator. Square roots are expressed notationally in either of two ways:

$$\sqrt{4} = 2$$

or

$$4^{1/2} = 2$$

The $\sqrt{\ }$ symbol is called a "radical." Following our rules for negative exponents we can handle various situations involving square roots. For example,

$$9^{-1/2} = \frac{1}{9^{1/2}} = \frac{1}{3}$$

Notice that the square roots in all of these examples are positive numbers. But isn't it true that

$$(-2) \times (-2) = 4\ ?$$

Therefore, doesn't

$$\sqrt{4} = -2\ ?$$

Yes. Actually,

$$\sqrt{4} = 2$$

or

$$\sqrt{4} = -2$$

This last statement is compressed and written as

$$\sqrt{4} = \pm 2$$

The part of this expression to the right of the equal sign is read "plus or minus 2." In general, when the radical or exponential ½ is used alone, the understanding is that the positive square root is indicated. Otherwise, the ± sign or just the minus sign is used.

Returning to our Cartesian coordinate system for a while, sometimes it is necessary to find the distance from the origin to a plotted point. As shown in *Fig. 8*, the distance (r) from the origin to the point (+4, +3) is the length of a side of a triangle whose other two sides have lengths of 4 and 3. We know the lengths of these two sides because they correspond directly to the x and y co-ordinates of the point, and to the perpendicular distance of the point from the y axis and x axis, respectively. These relationships are indicated in *Fig. 8*. The kind of triangle formed by the line joining the origin and the point (+4, +3), and the two lines representing the distances of the point from the two

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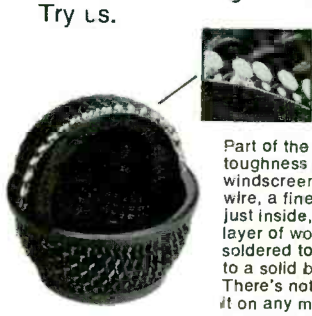

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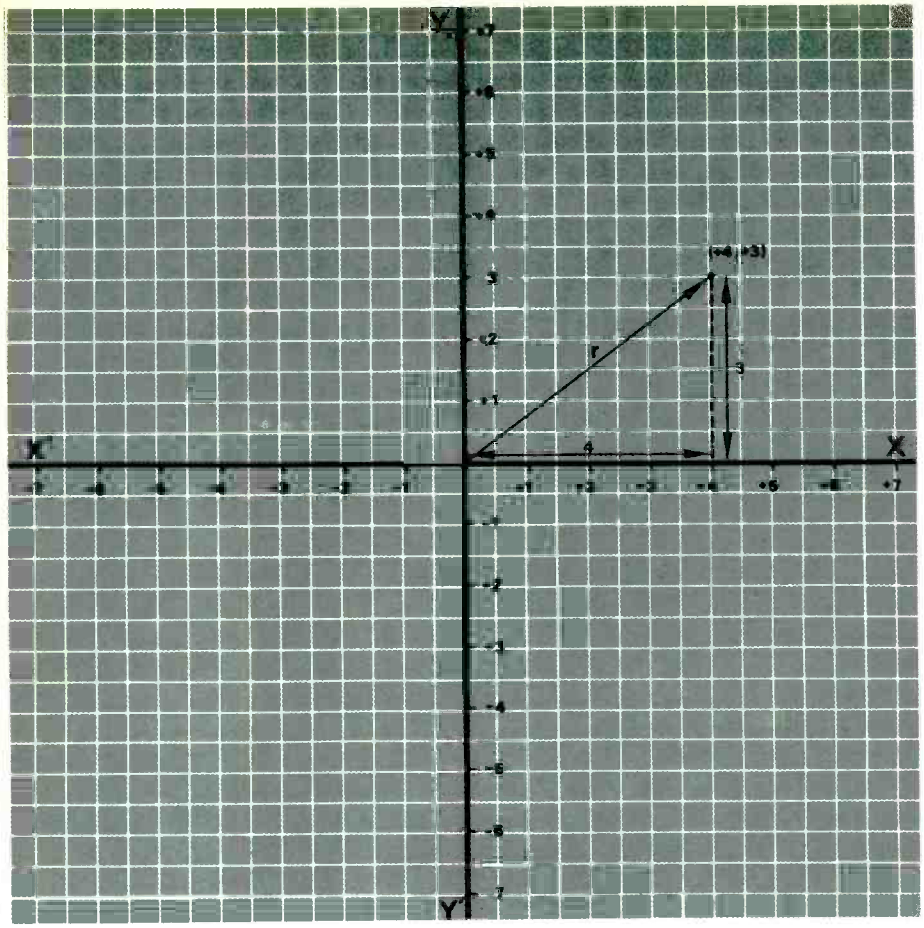

Figure 8

axes is called a *right triangle.* A right triangle is any triangle having two of its sides at a right angle (90°) to each other. The perpendicular sides are called "legs" or just "sides," and the side opposite the right angle (always the longest side) is called the *hypotenuse.*

In ancient times the study of the geometrical and mathematical properties of right triangles became an economic, political and religious necessity, since it was through the use of these properties that land holdings and building outlines could be surveyed. The Greek mathematician-philosopher Pythagoras of Samos (c. 500 B.C.) arrived at a relationship that holds true for *all* right triangles laid out on a flat surface. This relationship is known as the Pythagorean Theorem and states (remember, right triangles only): *The square of the length of the hypotenuse is equal to the sum of the squares of the lengths of the other two sides.* Using the notation from *Fig. 8,*

$$r^2 = x^2 + y^2$$

In words, the square of the distance from the origin to a plotted point is equal to the sum of the squares of the co-ordinates of the point. Again, using the information from *Fig. 8,*

$$r^2 = (4)^2 + (3)^2$$
$$r^2 = 16 + 9$$
$$r^2 = 25$$
$$\sqrt{r^2} = \sqrt{25}$$
$$r = 5$$

So, the distance from the origin to the point (+4, +3) is 5 units. We picked values that worked out neatly, but this is not always the case in practice, when the Pythagorean Theorem is used to determine certain values in A.C. circuits. The Pythagorean Theorem enables us to find the length of *any* side of a right triangle when the lengths of the other two sides are known. Remember, if

$$r^2 = x^2 + y^2$$
then
$$x^2 = r^2 - y^2$$
and
$$y^2 = r^2 - x^2$$

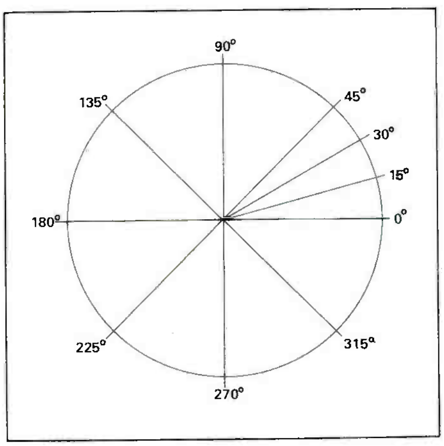

Figure 9

Sometimes only the length of one side (or the hypotenuse) of a right triangle is known. In order to find out any more about the triangle we need to know the measure of at least one angle other than the right angle. (Actually, since all the angles of any triangle must add up to 180°, if we know one angle in a right triangle other than the right angle, we know all the angles.) Angles can be measured in degrees, as shown in *Fig. 9.* There is a set of relationships, known as trigonometric ("triangle measuring") functions, that relate an angle and two sides of a right triangle. These functions are called *sine, cosine, tangent, cotangent, secant* and *cosecant.* We will be concerned only with the sine (sin), cosine (cos) and tangent (tan) functions, and we'll take a look at these topics in Part VII.

No, we're not going to probe the mathematical mysteries of the Pyramids, or measure the distance to Andromeda. Our purpose will be to equip ourselves with the basic mathematical tools for describing (and a little later on, designing) audio circuits that consist primarily of resistors, capacitors and inductors. Circuits such as passive filters and crossovers, and also such devices as impedance-matching transformers, dynamic microphones, speakers and musical instrument pickups rely on these basic circuit elements. Also, many of the principles that were explained in discussing the basic circuit elements apply directly to devices like the ones just mentioned. Once we have all the basics we can go right on to explaining these items in depth without having to pause every paragraph or so to insert a definition, or having to digress or backtrack.

So, in Part VII, we will continue with, and wrap up, our discussion of the use of trigonometry in A.C. circuit analysis, and begin, at last, to apply our findings to real, practical devices that we encounter in the process of playing, recording and enjoying music.

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The total Revox system delivers virtually distortion-free music from every source and in any mode—music reproduction so true that it is unmatched by any other system available today.

All the components in the Revox system are designed to meet the professional quality standards that have been set by the Revox B77 open reel recorder with its ingenious logic control system. The B750 integrated amplifier, for example, is renowned for its fast transient response to all musical signals and for its low TIM. It is paired with the incredible B760 digital synthesizer FM tuner with the unusual capability of programming 15 stations which you can recall at the push of a button. The B790 direct drive, quartz controlled turntable reduces tracking distortion to an absolute minimum using a revolutionary new system called Linatrack.

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CIRCLE 91 ON READER SERVICE CARD

Ambient Sound

BY LEN FELDMAN

Back From Las Vegas!

Traditionally, audio journalists just back from a Consumer Electronics Show (be it the Summer version in Chicago or the mammoth Winter event in Las Vegas) are supposed to reveal to their readers all the wondrous new products that they saw. The semi-annual CES wrap-ups are generally written as if the reporter had a crystal ball in one hand and a dictionary full of optimistic cliches in the other.

Well, here I am back from the annual trek to Las Vegas, and, while there are several new items that relate to the serious recordist and to audio in general that I want to tell you about, I must confess that when my colleagues ask what impressed me most in Las Vegas this time I tell them it was the brief helicopter ride that I took to view Hoover Dam! The truth is that this was simply not an audio dominated show. To be sure, most of the regulars were there, but some of those companies whose products are especially oriented toward the *Modern Recording* reader were not. Teac did not exhibit, but did treat the press to a Las Vegas-type revue and casual conversation at its hotel suite. Crown was absent (though they held forth in a private hotel suite, about which more in a moment). B.I.C. wasn't exhibiting either. Neither was Nakamichi.

I have no doubt but what the state of the national economy had something to do with the relative absence of new products. It is entirely possible that many of our favorite audio companies, though sitting on top of new products, have decided to hold off showing them until the June CES in Chicago, by which time they will have sold off their 1979 inventories. Still, all was not gloom and doom, so let's take a look at some of the more interesting products and technological breakthroughs that we encountered in Glitter Gulch.

A For-Real PCM Audio Processor

Until now, all of the PCM audio processors I have worked with have either been of limited production prototype runs or one-of-a-kind samples from several high-technology companies who wanted to be sure that they were up-to-date. Readers may recall that I previously reported on a Toshiba PCM processor [*see "Lab Report,"* MR, *March 1980*] as well as on the earliest Sony PCM-1 models. Prices for these early PCM units were never really firmed up nor were projected delivery dates. Well, now one company at least has made a firm written commitment with respect to a home PCM unit. Readers will recall that these PCM processors (now pretty much following standards set by the Electronic Industries Association of Japan) all require a VCR as a tape transport and information storage component. So does the new Plus-10 PCM Audio Processor just announced by, of all people, Sanyo. When you consider the fact that little more than a year ago, this company hadn't even entered the serious high-fidelity audio marketplace, that's quite an accomplishment. Of course, whatever your previous image of Sanyo in the American marketplace, bear in mind that this company is the sole owner of Fisher Corporation and certainly has enough technological capability to come up with whatever latest audio products it chooses to market. In

These Toshiba components respond to voice commands, but don't look for them on dealer's shelves just yet.

The Sanyo Plus-10 PCM Audio Processor.

any case, I loved the suggested retail price of the new Plus-10 PCM—"just thirty-nine-ninety-five" is the way their people told it to me—meaning $3,995 with no decimal points anywhere in between. Still, the company plans to have units available at that price before the end of the first quarter of 1980 and that means that the digital audio revolution is ready to begin in earnest.

Last June, Dolby Laboratories introduced a dynamic headroom extension system which they called Dolby-HX. It was based upon the idea of continuously varying tape recorder bias in such a way that when high-frequency, high-energy signals are to be recorded, bias is lowered (instantaneously) and record EQ is altered to compensate. The sensing or control signal used works out to be the one already available from Dolby B noise reduction circuitry, which makes it very nice for Dolby licensees. Six months later, we were able to find at least one manufacturer who had already incorporated the new Dolby wrinkle into a commercially available deck. That company is Harman-Kardon who, in addition, introduced a complete line of mid-sized separates that seemed extremely attractive to me and well priced.

Computer-Programmable Cassette Deck

Eumig, the Austrian-based firm perhaps best known for their movie and lens equipment, introduced their model FL-1000 cassette deck which was demonstrated to open-mouthed visitors. Up to 16 decks can be interconnected through a single 8-bit computer and can

The Eumig FL-1000 cassette deck.

individually controlled, simultaneously or singly, to play or record any section of any tape. One machine can play a selection while a second machine searches for another musical selection on a different tape, stops and waits for its turn (based upon instructions fed to the computer) to play the next selection.

Under direction from a connected computer, titles and index locations of all the musical selections on a given tape can be digitally recorded, using the first few seconds of tape on a cassette. Then, by inserting a programmed cassette into the FL-1000 and punching a few computer buttons, the user can obtain a readout of the entire contents of the tape, displayed on the computer's screen.

Last June I got pretty excited about a new kind of "noiseless disc" introduced by dbx, Inc. The disc did for records pretty much what their 2:1/1:2 companding system was able to do for tape for some time: increase dynamic range to a potential 100 dB and decrease residual tape noise by approximately 35 dB. The early discs that I heard were truly phenomenal, but were flawed only because they had been made from conventional analog tape masters. However carefully these were selected for re-issuance under the dbx format, once the disc surface noise vanishes, we become increasingly conscious of the master-tape's own hiss level, since it is now no longer masked by the much louder surface noise.

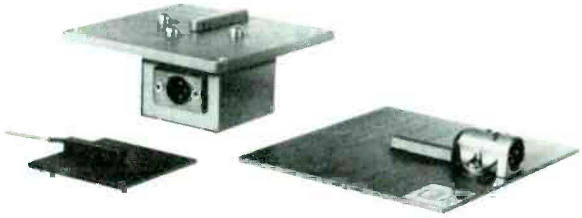
You can probably guess what the next step by dbx was. Their new recordings now feature music that was originally recorded on tape *digitally*. Many of these new releases were done by M & K Sound, using a Sony PCM-1600 digital processor (the professional model) which, using a 16-bit system, provides in excess of 90 dB of dynamic range. Take such digitally recorded masters and encode them onto discs using the dbx approach and you end up with the world's first records that actually can reproduce the full dynamic range of just about any musical performance and that exhibit no audible surface noise.

What may be of interest to the recordist, above and beyond the availability of these interesting discs, is that now dbx also offers a complete companding unit or tape noise reduction system, Model 224, for a suggested retail price of only $275. Not only is this considerably less than the price of the previously available companders (Models 122, 124 and 128) but the new unit has a "disc" setting for playing the new dbx discs.

New Microphone Technology

Crown International, one of the companies that did not officially exhibit at the show, held a press conference in a hotel suite in order to introduce a brand new concept in microphone configurations that may change a lot of thinking about microphone placement and mic technique in general. Dubbed "Pressure Zone Microphones" (PZM™), these microphones are, according to Crown, the first fundamental advance in microphone technology in more than 45 years. These new microphones, unlike conventional types, respond to sound in a hemispherical pattern, so that response is unaffected by the motion or direction of the sound source. The hemispherical pickup pattern is created by mounting a pressure calibrated element facing an acoustic boundary, such as a flat plate, wall or floor, so that it responds to the pressure zone developed at the surface of the boundary. Since this construction allows open access to the element from essentially any direction, no single response axis is favored.

Crown's microphone introductions were the result of a cooperative effort between them, Edward M. Long of E. M. Long and Associates and Mr. Ken Wahrenbrock of Wahrenbrock Sound Associates Limited who developed the pressure zone concept and investigated it at the instigation of the well known audio engineer Don Davis of Synergetic Audio Concepts. Crown has an exclusive license to manufacture and market a complete line of Pressure Zone Microphones. Besides the

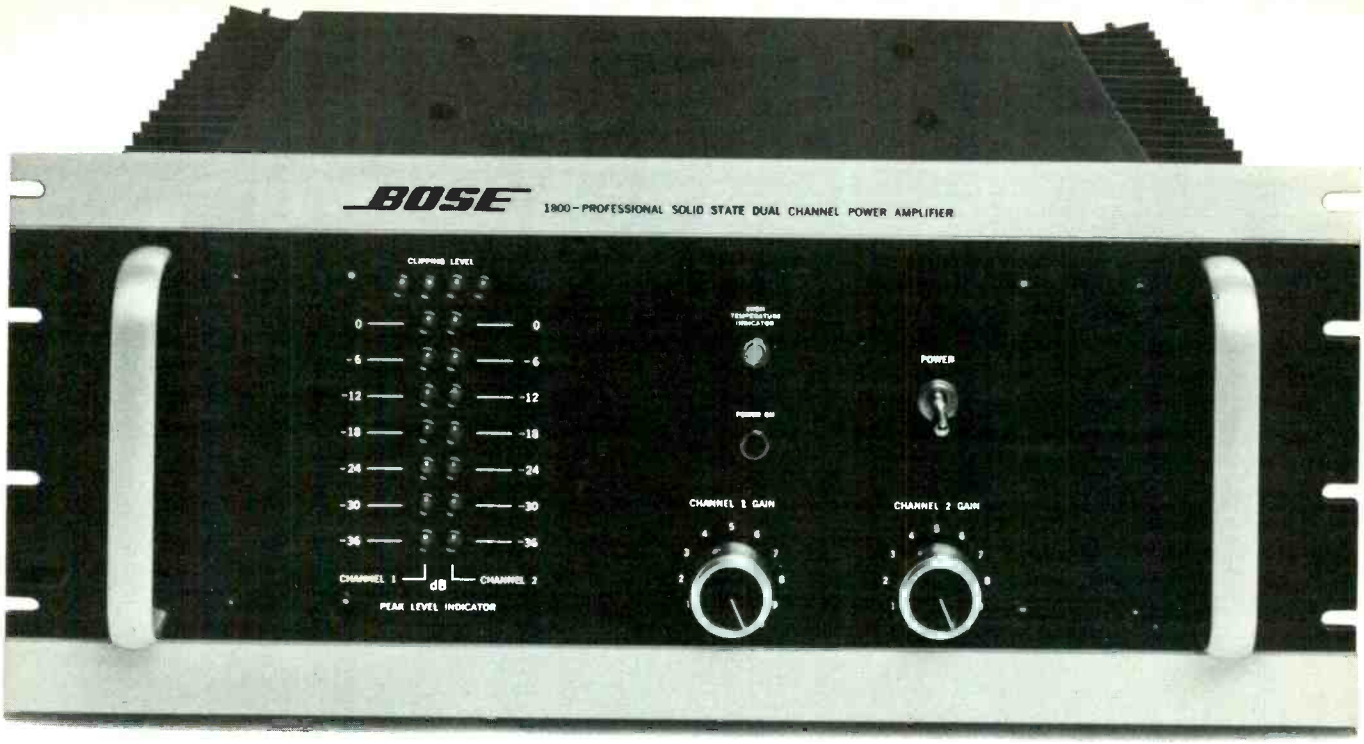


Three examples of the configurations of the new Crown PZM microphones.

elimination of directional discrimination characteristics, additional benefits claimed for the new microphones include simplified miking techniques, increased subject visibility (PZMs can often be mounted in such a way as to be completely unobtrusive) and amazingly high sound pressure handling capability. Pressure Zone Microphones from Crown will be able to handle sound levels of 150 dB. Thus they can be placed inside a drum or directly in front of an electric guitar amplifier or other instrument amplifier.

The Future Now

If awards were to be given for the most futuristic of products displayed at WCES one would certainly have to go to Toshiba who demonstrated an Acoustic Remote-Controlled high fidelity component system that will respond only to the registered owner's voice! The voice control system can perform any of nineteen different operations, such as power on/off, volume up, volume down, play, record, rewind, fast forward and stop (in the case of cassette decks); channel one, two, three, four, scan up or scan down (in the case of the tuner component). At the heart of this system is Toshiba's microcomputer, an 8-bit CPU, 2KB ROM (Read Only Memory) and 3KB RAM (Random Access Memory). These units, though displayed at WCES, were prototypes only, and when I jokingly asked if next year's models would also talk back the booth attendant directed me to a voice-activated TV set made by the same firm which already *did* talk back, with the word "okay" when a command was accepted and with the word "repeat" when the command was not properly received by the TV set. As I intimated earlier, this was more of a video show than an audio show. Perhaps audio will take the spotlight in Chicago, in June. Whether it does or not, I hope to be there too, and to report to *MR*'s readers on what's happening.



Unadvertised Special

Sometimes our mailbox brings us a better ad than we could write ourselves. Here's the complete text of a letter we received from Rick Stalnecker, a musician and experienced traveller. Rick's letter says more about the Bose® Model 1800 Amplifier than we could say in a book.

"Enclosed is a picture of our present amp case. These are the original Bose amps which we put on the road in the summer of 1973. In the past six years, these same amps have played in over 500 cities and done at least 3,500 concerts from Anchorage, Alaska to Key West, Florida. This is the fourth road case the amps have outlived, and we use the finest cases available! One week they'll be in a football stadium, through several rain storms, and the next week in a studio or auditorium somewhere. We figure that they have traveled around 500,000 miles and although we have worn out 3 equipment trucks, we have yet to have the first problem with one Bose amp ever! I can't believe it! We have never even replaced a 15-cent fuse! As if that wasn't enough for these work horses, when I get home to our studio I use them for playback, mixdown, and even headphones. The last time they were out of a case, I thoroughly checked them and there wasn't even a casing screw that needed tightening.

I say all of this for one reason. Right now, everybody and their great uncle is claiming their amp to be the best, and I don't think your advertising has been saying enough about your amps. Personally, I can't say enough about their reliability, power, and inaudible distortion.

There is one bad thing though, I probably will never need to buy another amp from you!"

Thanks, Rick! Letters like yours make all of our work seem worthwhile and rewarding.

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NORMAN EISENBERG AND LEN FELDMAN

Sansui B-1 Power Amplifier

General Description: The Sansui B-1 is a stereo power amplifier rated conservatively for an output of 250 watts per channel into 8-ohm loads, or 350 watts per channel into 4-ohm loads. Loads lower than 4 ohms are not recommended. Of rack-mount width, the front panel has slotted ends and handles.

Peak-power meters monitor the output on each channel. Meter scales provide dB and wattage readings, with "zero dB" representing 250 watts for 8-ohm loads. With other load values, the indicated wattage reading must be mentally changed. That is, 4-ohm loads double the wattage scale; 16-ohm loads halve it.

A peak-hold switch may be used to keep the peak reading on the scale for about 15 minutes. Below each meter is an input level control for that channel. Between the meters are individual channel overload indicators, a thermal indicator and an indicator for the built-in two-speed fan. This fan will come on automatically at a given temperature level. In addition, it may be turned on manually at any time by a switch on the front panel.

Additional front-panel features include a switch to activate a subsonic filter; a speaker selector switch that chooses one or both pairs of speakers that are connected at the rear; and the power off/on switch. Associated with the power switch is an indicator that blinks when the amplifier is first turned on and then glows steadily. In the event of a malfunction, a short in the output load, or DC voltages appearing in the output, this indicator will start to blink again.

At the rear, the input and output connections for each channel are mirror-images of each other. Inputs are unbalanced but include ¼-inch phone jacks as well as XLR sockets. Speaker outputs, for main and extra stereo pairs, are knurled-nut screw posts. There also is, for each channel, a set of screw-terminals marked for 70.7-volt output. These terminals do not furnish the 70.7-volt output but are provided as a convenience in adding an optional transformer (Sansui model B-LT-1) that will furnish the 70.7-volt output if needed for multiple speaker hookups on each channel.

Each channel also has a "DC/AC" switch that permits the Model B-1 to operate either as a direct-coupled amplifier, or with the necessary coupling capacitors in the input stages that would block any DC voltages present in the input signal. Each channel also has its own grounding terminal. Centered between the two connector areas is a metal screen that covers the fan. The amplifier's power cord is fitted with a three-prong grounding plug.

Test Results: In our lab tests, the Sansui B-1 easily met or exceeded its published specifications. Power output went well above rated output for rated distortion, while distortion for rated output was lower than claimed. One of Sansui's specs is for TIM (transient intermodulation distortion), for which Sansui has developed its own method of measuring. It differs from the so-called Otalla method that has become popular with several manufacturers. While TIM is not included in our "Vital Statistics" chart, our lab is equipped to measure it using the Sansui method. On the individual data sheet enclosed with our test sample, TIM was listed as 0.0100 percent and 0.0088 percent for left and right channels, respectively. Our results were amazingly close, with readings of 0.0100 percent and 0.0093 percent for left and right channels, respectively.

The subsonic filter, activated by the front-panel switch, had its 3-dB cutoff point set precisely at 20 Hz. This option is a nice touch, over and above the action of the "DC/AC" switches at the rear. Combined with the other safety and self-ventilating features built into the amplifier, it should assure against various possible bugs or subsonic garbage interfering with normal operation.

General Info: Dimensions are 19 inches wide; 7 11/16 inches high; 19 15/16 inches deep. Weight is 60 pounds. Price is $1200.

Individual Comment by L.F.: A few issues back, I devoted my "Ambient Sound" column to the subject of "professional" versus "hi fi" audio equipment, trying to point out some of the subtle design philosophy differences between the two classes of equipment. Sansui's B-1 amplifier makes an excellent illustration of those differences, because it arrived just a few weeks after I had checked out another Sansui amplifier, the model BA-F1, intended strictly for home audiophile use.

Both units take full advantage of Sansui's latest thinking about "high-speed" amplifier circuitry, fast rise time and high slew rate. Both amplifiers employ what Sansui calls its "Diamond Differential DC" output circuit which is one approach to minimizing TIM and achieving a high slew rate—better than 200 volts per microsecond—for both amplifiers. The "Diamond Differential DC" circuit is designed to insure that there always is adequate current at the input circuitry, no matter how demanding or pulsive the input signal. With early-stage clipping virtually nonexistent, TIM is reduced to nearly unmeasurable values.

Getting back to differences between hi-fi and pro amplifiers as exemplified by the Sansui BA-F1 and the B-1, it should be noted that the B-1 is a much-higher powered amplifier than the other. As such, it requires a cooling fan. This quiet two-speed fan will operate automatically as needed or it can be turned on manually—a nice touch that I have not seen before. In our tests the fan never went into high speed of its own accord, probably because our bench afforded more ventilation than the amplifier would likely have when installed in a professional sound system.

The rear panel inputs, instead of using RCA pin-jacks, consist of phone-jacks in parallel with XLR connectors, so that either type of plug may be used for interfacing. Do not make the mistake, however, of assuming that you can make either balanced or unbalanced connections here. In the three-pin XLR connectors, pins 1 and 2 are wired together and to chassis ground, while only pin 3 goes to the "hot" side of the input circuit. Input impedance measured around 20 K ohms, unbalanced, as specified in the manual.

An internal view of the B-1 amplifier is shown in the accompanying photo. The unit is a heavyweight, no doubt about that. Still, the listening tests confirm what I said in that earlier column: the mere fact that an amplifier is externally configured for professional applications does not mean that it need sacrifice sonic purity, or any of the new-found circuit advances that are presumed by many to be confined to high-fidelity applications. External differences aside, the difference in sound quality between the earlier-tested BA-F1 "hi fi" amplifier, and the present model (aside from about 3 dB more power output from the B-1) are indeed trivial. As for that 3 dB higher, or "double-power" rating, it all works out in the pricing, inasmuch as the B-1 costs almost twice as much as the "hi fi" BA-F1 made by the same company.

Individual Comment by N.E.: I think what we have here is an amplifier that started out as a "super audiophile" unit and then was beefed up in terms of power output, and fitted with certain trappings designed to appeal to the professional user. Let's see—the XLR input sockets will accept a balanced line from another source but they convert it to an unbalanced input. In many instances, you might just do that yourself, and so in that regard, the 3-pin connectors on the chassis are at least a convenience. But if your particular setup is such that you need full balanced inputs right into the amplifier, this unit will not do it.

The speaker output connectors are screw-on posts with holes for poking stripped leads, but they are not the "5-way binding posts" usually found on pro amps and they will not accept the standard banana-plug type of connector.

The "70.7-volt" feature is a convenience in hooking up to an external transformer that itself will supply the 70.7 volt output to drive many speakers down the line. But that specific output is not provided by the amplifier itself. The "70.7-volt" terminals are simply wired in parallel to the regular speaker outputs and chassis ground. Which means of course that even without those terminals you could, if you wanted to, connect a requisite transformer to the normal speaker outputs and get your 70.7-volt operation. And by the same token, you could do this with just about any normal, respectably functioning power amplifier. What Sansui has done here is make that hookup more convenient, and take the guesswork out of choosing the proper 70.7-volt transformer for use with a particular amplifier.

There is no provision evident in the B-1 for combining the two channels to create a super-powered mono amplifier, and the instructions caution against using loads lower than 4 ohms. Both of these options we have seen in other "pro" amps.

So whether the Sansui B-1 is only flirting with the professional sound man, or really means business on his terms, is perhaps a debatable question that is best answered by the individual prospective purchaser, based on his needs. Whatever, and very much in favor of the B-1, is its utterly clean sound, its ample power reserves, and the ample provisions built into it for safe and reliable operation.

SANSUI B-1 POWER AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Continuous power for rated THD,		
8 ohms, per ch.	250 watts	312 watts
4 ohms, per ch.	350 watts	480 watts
FTC rated power (20 Hz to 20 kHz)	250 watts	270 watts
THD at rated output, 1 kHz, 8 ohms	0.02%	0.008%
1 kHz, 4 ohms	0.02%	0.013%
20 Hz, 8 ohms	0.02%	0.01%
20 kHz, 8 ohms	0.02%	0.015%
IM distortion, rated output, SMPTE	0.015%	0.010%
CCIF	NA	<0.03%
IHF	NA	<0.03%
Frequency response at 1 watt		
(for −3 dB)	DC to 200 kHz	DC to 220 kHz
S/N ratio re 1 watt, "A" wtd, IHF	81 dB	90 dB
S/N ratio re rated output, "A" wtd	105 dB	121 dB
Dynamic headroom, IHF	NA	1.9 dB
Damping factor	450 (1 kHz)	>200
IHF input sensitivity	NA	0.079 volt
Input sensitivity re rated output	1.23 volts	1.23 volts
Slew rate (volts/microseconds)	200	210
Power consumption, idling	NA	180 watts
Power consumption, maximum	1085 watts	1500 watts

CIRCLE 38 ON READER SERVICE CARD

ADC "Sound Shaper Three" Equalizer

General Description: ADC's "Sound Shaper Three" (which we will refer to here as the SS-3) is a stereo graphic equalizer that offers what may be called a "parametric-like" feature. That is to say, in addition to the usual sliders for boost or cut of specific frequency bands, there are additional sliders to vary the center frequency up or down by fixed amounts of about 20 percent either way to new center frequencies. Thus, for example, the 32-Hz center frequency (for the first slider) can be changed to either 26 Hz or to 39 Hz center frequency, and so on up the scale.

The bandwidth or "Q" is fixed, but there is a method suggested for increasing it, although not for decreasing it. This is explained below in "Test Results."

Each of the two channels handled by the SS-3 is divided into twelve frequency segments. Inasmuch as each segment can be used with three different center frequencies, a total of thirty-six possible frequency-band adjustments is available, of which any twelve may be used at a time on either channel. The controls are totally duplicated for each channel so that the user need not select the same center frequencies for channel A as may have been chosen for channel B. ADC has coined the term "paragraphic" to describe this design.

The main sliders for each channel are marked from +12 dB to −12 dB in steps of 2 dB, with detents at the +6 dB, 0 dB and −6 dB markings. Beneath each of the twelve sliders on each channel is that slider's corresponding bandwidth slider, a three-position toggle with the "up" and "down" frequency markings indicated accordingly. The area between the two groups of sliders is given over to gain sliders for each channel with individual vertical rows of LEDs for each channel that indicate signal levels in increments of 2 dB. Suggested uses of this metering system are for balancing stereo channels, monitoring signal levels and making frequency-response measurements in conjunction with an external sound-level meter and test signal source. Below each LED meter row is a level control that adjusts the display's sensitivity for each channel.

Additional features on the front panel, found at the extreme right, include buttons for EQ by-pass; meter in or out; line/record; tape monitor; and AC power off/on. There also is a jack marked SLM which may be used to connect an optional sound-level meter (ADC SLM-300) which the manual suggests using together with ADC's pink-noise record and a 20-foot signal cable. This hookup permits the right-channel LED meter to read the output of the sound-level meter directly so that the user may adjust the SS-3 at the equalizer position during a room-equalization process.

The white markings on the black panel make for good legibility. The panel, of standard rack-mount width, is slotted and fitted with handles.

Signal jacks at the rear consists of four pairs of pin-jacks for left- and right-channel tape monitor, tape out, input and output. There also is an unswitched AC convenience outlet. The SS-3 can be interposed "in series" between a separate preamp and power amp, although the owner's manual emphasizes its interfacing via the tape-monitor loop typically included in a preamplifier, integrated amplifier or receiver. This recommended hookup permits, together with the use of the controls on the SS-3, the maximum versatility of application of the device, including normal room or system equalization as well as equalizing tape recordings as they are made. With this hookup, it is possible to listen to programs equalized, while recording them unequalized or equalized, as desired. It also is possible to listen to tape playback and add equalization as desired. It also is possible to listen to programs unequalized and record them the same way, and to play them back unequalized. Finally it is possible to monitor (on a three-head tape deck) the signal being recorded with equalization. These options are explained clearly in the owner's manual with the aid of drawings that show the signal paths for various combinations of control settings.

Test Results: In our tests, the ADC SS-3 met or exceeded its published specifications. In addition to clean performance across the audio band, the device also offered a degree of versitality that, while not of the order of a full-fledged parametric equalizer, was

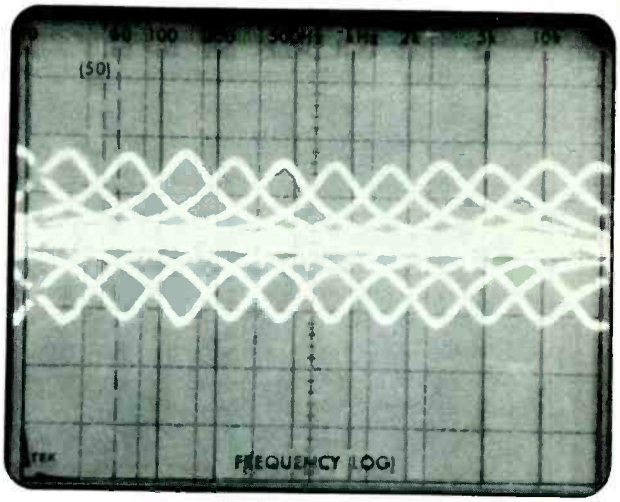


Fig. 1: ADC Sound Shaper Three: Boost and cut range of each of the 12 band controls.

definitely more versatile than that of conventional graphic equalizers. In addition, the 13-segment LED metering display (for each channel) proved to be an ideal and accurate metering system. With the impressive dynamic range of this device (S/N measured was 94 dB re: 1 volt, and maximum output before clipping was 10 volts, all of which adds up to total dynamic range from noise-floor to clipping of about 114 dB in our test sample), ADC wisely provides a pair of controls solely to adjust the sensitivity of the meters. With these controls, the operator can keep the audio signals activating the display within its most useful center-of-range regardless of what input level is being fed into the equalizer.

The range of each band control is shown in Fig. 1 with each band adjusted for its nominal "mid" center-frequency. Figure 2 shows what can be done to shift the center frequency up or down—in this example, the 1-kHz slider was used.

In Fig. 3 we took a cue from the owner's manual which suggests that when a wider band of frequencies

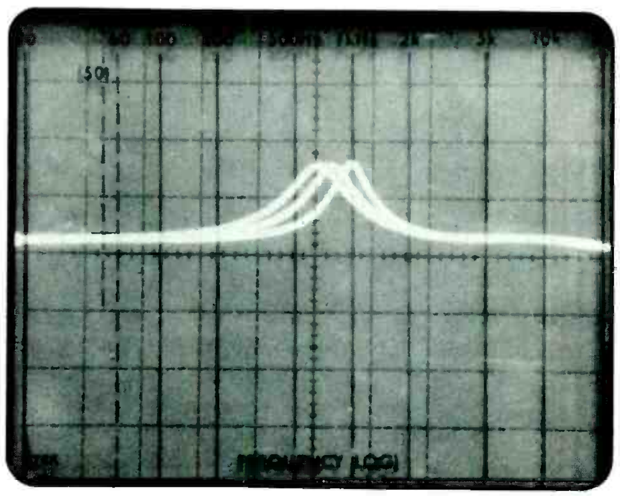


Fig. 2: ADC Sound Shaper Three: Center frequency of each control band can be by ±20%.

needs to be boosted or cut, the thing to do is to shift adjacent-band center frequencies closer to the center-frequency of the band being adjusted. Thus, in the 'scope photo of Fig. 3, we are comparing the response obtained when the 1-kHz slider is boosted fully, but all by itself, with the response obtained when that boosted slider is accompanied by a boosted 1.8-kHz slider that has been adjusted for its lower center frequency of 1.5 kHz, and with the 560-Hz slider adjusted for its upper center-frequency of 680 Hz. This technique actually does vary the bandwidth; at least it widens it. However, there seems little the user could do to narrow the bandwidth (increase the "Q") for attenuating say, a troublesome narrow frequency segment that might be causing feedback in a sound-reinforcement situation.

General Info: Dimensions are 19 inches wide; 6⅑₁₆ inches high; 12 inches deep. Weight is 22 lbs. Price: $499.95

Individual Comment by N.E.: As equalizers in general become more popular, we can expect variations and innovations with respect to the basic format, even as we have been seeing among cassette recorders. This model from ADC offers the interesting and useful added facility of being able to vary the nominal center frequency (and to widen the bandwidth to a degree) of the individual frequency segment adjustments in an otherwise normal graphic equalizer. This added facility, while hardly as "all-out" as in a full-fledged parametric equalizer, does enhance the signal-shaping capability of the device beyond that of a conventional graphic equalizer, and it is the basis for the ADC-coined term "paragraphic."

In addition, the switching built into the device makes it relatively easy and convenient to patch it into a sound system and use it in various possible applications without changing the hookup. That is the real reason for the recommendation in the owner's manual to use the SS-3 "only" in a system that has the tape-monitor feature. From a purely electrical-sound standpoint, it could be patched simply between any preamp and any power amp—but that of course would either decrease its functional versatility or entail changing patch cords for various applications.

The SS-3 provides a unity gain adjustment for each channel, and an accurate and helpful LED metering display. All told, well-conceived and nicely executed.

Individual Comment by L.F.: ADC—a BSR company—calls its most sophisticated equalizer "paragraphic" and has applied for a trademark for that term. The implication is that the Sound Shaper Three is a cross between a parametric and a graphic equalizer. Indeed, the SS-3 does have some of the attributes of each type of equalizer, but if I had to categorize it as one or the other, I would still be inclined to call it a graphic equalizer, although a most versatile one. The center frequencies can be adjusted for one of three settings, some 20 percent apart from each other, but to

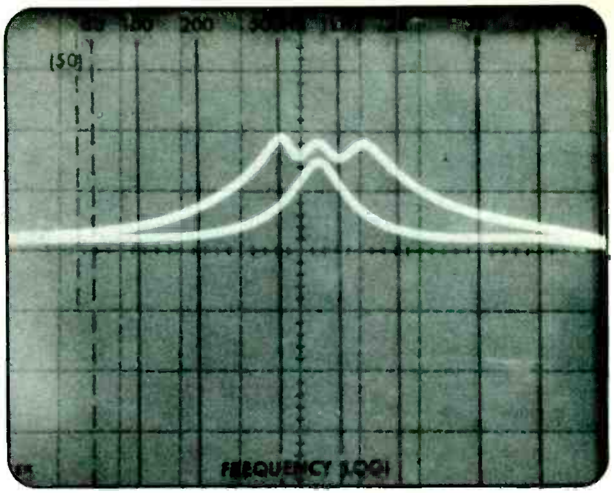


Fig. 3: ADC Sound Shaper Three: Bandwidth can be broadened (upper trace) by switching center frequencies of adjacent band controls towards desired center frequency of primary band control (lower trace).

be technically precise about it, a parametric equalizer usually is one that has continuously variable center frequencies rather than fixed settings. In addition, most of the parametrics I have dealt with also have continuously variable bandwidth or "Q" for each of the available bands. The facility in the Sound Shaper Three for altering "Q" is very limited.

This quibble about terminology aside, I will state unequivocally that—whatever name the SS-3 is given—I really liked its action and performance. I have always favored equalizers having gain controls (to adjust for unity gain regardless of input level or of the settings of individual control bands), and the SS-3 has them. I also have always felt that there should be some sort of indicating device to let you know when gain is adjusted correctly, and this unit provides that too with its LED indicator "arrows" that tell you which way to adjust the master gain controls for unity gain and optimum distortion performance. Finally, I always have felt that a good equalizer will incorporate a metering system to let you know what's happening as you adjust individual controls. The 13-segment LED display (per channel) of the SS-3 is ideal for this.

Internal construction and layout are neat and clean, but I was somewhat surprised at the use of actual inductors in the tuned circuitry as opposed to op-amp gyrator circuits. I suppose ADC had its reasons for doing so, and as long as the signal-to-noise ratio does not suffer (it doesn't), and as long as harmonic distortion is no problem (it isn't), I have no serious objection to the use of discrete inductors in an equalizer. With each inductor requiring three different values of precision capacitance (for the shifting of the center frequencies), you can well imagine the number of precision capacitors found inside the unit. More often than not, it was necessary to use a pair of capacitors in parallel to come up with the calculated value required for all those center frequencies, all of which—according to our tests—were just about perfectly calibrated.

As with all equalizers, I maintain that the user had better equip himself with some means of adjusting these devices properly. ADC offers a sound-level meter plus a pink-noise test record and a 20-foot interconnecting cable that may be plugged into the equalizer directly and thus provide meter readings right on the LED display of the SS-3. If you do not own a real-time analyzer or if you can't borrow one, this combination (known as the ADC Model SLM-300)—while somewhat tedious to use—is the next best thing and isn't too expensive. Come to think of it, for all the flexibility built into the Sound Shaper Three, the unit itself isn't very expensive at just under $500.

ADC SOUND SHAPER THREE: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response	5 Hz to 100 kHz, +0.5, −1.0 dB	3.5 Hz to 200 kHz, −1.0 dB
Center frequencies of controls:		
low:	26, 47, 84, 150, 260, 470, 840, 1.5 K, 2.6 K, 4.7 K, 8.4 K, 15 K	Confirmed
center:	32, 56, 100, 180, 320, 560, 1 K, 1.8 K, 3.2 K, 5.6 K, 10 K, 18 K	Confirmed
high:	39, 68, 120, 215, 390, 680, 1.2 K, 2.15 K 3.9 K, 6.8 K, 12 K, 21.5 K	Confirmed
Control range	± 12 dB min.	+13, −13.5 typ.
THD at 1 V output		
20 Hz	0.018%	0.008%
1 kHz	0.018%	0.01%
20 kHz	0.018%	0.022%
Hum and noise re: 1V	90 dB	94 dB
IM distortion	0.02%	0.03%
Dynamic range (max, out)	10 V	10 V
Gain (controls flat)	0 dB, ± 1 dB	Confirmed
Output impedance @ 1 kHz	10 ohms	Confirmed
LED metering tolerance	± 0.5 dB	± 0.5 dB
Power consumption	25 watts	23 watts

CIRCLE 39 ON READER SERVICE CARD

Nakamichi 680ZX Cassette Recorder

General Description: The top model in Nakamichi's recently introduced "ZX" series, the model 680ZX is a three-head cassette recorder of high performance and several innovative, and worthwhile, features. It is metal-tape capable; it operates at two speeds (15/16 ips in addition to the standard 1⅞ ips); it includes the random access music memory (RAMM) system for automatic search and play of specific selections on a tape; and it provides for automatic alignment of the separate recording head. Unattended playback and recording can be accomplished with the use of an external timing device, and remote control is possible with the use of Nakamichi's model RM-200 r-c accessory.

The tape drive uses a double-capstan, closed-loop system, and the two capstans are of different diameter and are driven at different speeds. This "asymmetrical" design is credited with minimizing any mechanical resonances that could degrade the response. One motor (operating through flywheels of different inertia) drives the two capstans; a second motor is used for reel drive; and a third motor drives a cam that positions the heads against the tape and also engages the reel brakes and other solenoid-controlled operations.

The automatic head alignment is handled by a fourth motor that actually moves the record head while a built-in 400-Hz test tone is applied to the head, and picked off as a difference in phase between the left and right channels at the playback head. Indicators show that the alignment process is under way, and reference cursors appear on the signal meters to indicate when azimuth alignment has been completed. Associated with this procedure is the record-calibration via screwdriver adjustments found on the front panel below the meters. There are twelve such adjustments: two groups of six for the two speeds at which the deck operates, and each group containing left- and right-channel screws for each of three basic kinds of tape—EX (standard); SX (high bias); and ZX (metal). These designations also appear on the separate bias and EQ switches provided elsewhere on the panel for tape-type selection.

A cueing system, not unlike that found on many open-reel decks, is provided on the Nakamichi 680ZX. When the deck is in rewind or fast-forward, pressing the pause button reduces the winding speed and also brings the playback head closer to the tape so that a high-pitched signal may be monitored as an aid in rapidly locating a desired portion of the tape.

For locating a known portion of a tape, the memory switch may be used in conjunction with the tape counter so that during rewind the transport can be made to stop at any desired spot.

The RAMM system itself operates by sensing and counting the number of silent spaces between selections on a recorded tape. In a sense, it "listens" for you while the tape is being moved. You can "instruct" the RAMM to count up to eighteen blank spaces.

The model 680ZX is strictly a line-level input deck. It has no microphone jacks or microphone preamps. For "live" recording with mics, Nakamichi explains that an external mic-mixer (such as its own MX-100) must be used. The only signal jack on the front panel is a stereo headphone output jack; the line-in and line-out jacks are at the rear.

The front panel, which is dimensioned and slotted for rack-mounting, is "busy looking" but neatly laid out and legible. The cassette compartment is at the left. To its right are the memory switch, the tape counter and its reset button, and a display area that includes the RAMM indicator, and the deck's signal meters. The latter are horizontal fluorescent bars calibrated from −40 to +10. They may be switched to show peak or VU levels. They show signal levels for record and play, and they are used during the azimuth alignment and record calibration procedures. The row of adjustment screws for the calibration is just below the meter panel. To the left of these adjustments is a pitch control, operative on playback over a range of ±6 per cent.

Further down on the panel are the transport controls, the tape speed selector, an output level control, dual-concentric input controls for left and right channels and a master input control. This last control may be used as a fader for the program source once the basic level and channel balance have been set on the dual-concentric pair. The transport controls provide for fast-buttoning from one mode to another, except that to go from play to record the stop button must be pressed first. That particular operation is facilitated by the action of these light-touch, fast acting buttons.

At the right end of the panel is a vertical row of switches for tape bias, tape EQ, Dolby system (with a setting for multiplex filter), the metering display, the optional timer, tape/source monitoring and power off/on.

In addition to the line jacks at the rear, there are special sockets for connecting the remote-control accessory, and for supplying a regulated DC voltage to power one or more of Nakamichi's "Black Box" series components, such as the microphone mixer, a subsonic filter, a line amplifier, a bridging adaptor, a booster amp or an electronic crossover. Any combination may be used that does not exceed the socket's rating of 125 mA. For higher current drain, the model PS-100 power supply must be added.

Test Results: The Nakamichi 680ZX was tested at both of its operating speeds and with three kinds of tape (requiring normal bias, high bias and metal bias). In every test, the deck either met or exceeded its published specifications. These were relatively high to begin with, and their confirmation leads to one inescapable conclusion. With this unit, Nakamichi has gone beyond its own model 1000 and has produced what must be acknowledged as—if not the best-performing cassette deck we have yet encountered—then surely one that is unsurpassed by any other. Significantly, it is the only "metal tape capable" cassette deck we have tested to date that actually does perform better *in all tested parameters* with metal tape than with other varieties. Even at the unprecedented "half speed" of $1\frac{5}{16}$ inches-per-second, the Model 680ZX offers performance that rivals that of other cassette decks operating at the $1\frac{7}{8}$ ips speed.

Fig. 1 shows a plot of playback-only frequency response, for the 120-μsec EQ setting, using TDK test tape AC-337. The plot extends only from 40 Hz to 12.5 kHz simply because these are the extreme spot frequencies on this test tape. More revealing are the record/playback curves (*Figs. 2, 3 and 4*) which were made—thanks to the three-head configuration of the 680ZX—by means of continuous sweeps using our spectrum analyzer. *Figs. 2, 3 and 4* show the results (plotted from 20 Hz to 20 kHz, logarithmically swept) at the −20 dB and the 0 dB record levels, using Naka-

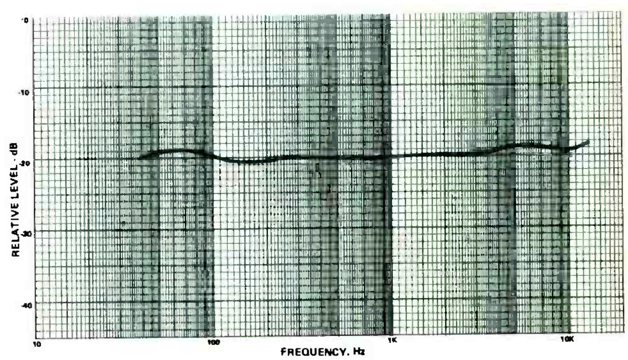


Fig. 1: Nakamichi 680-ZX: Playback-only response at $1\frac{7}{8}$ ips, 120 μsec EQ, using TDK test tape.

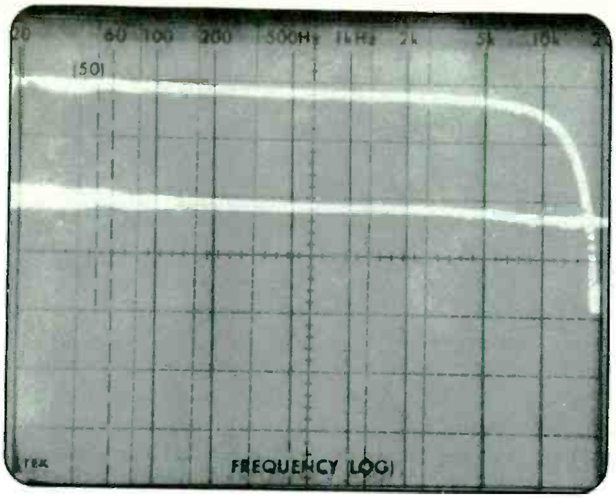


Fig. 2: Nakamichi 680-ZX: Record/play response at 0 dB & −20 dB record level, using Nakamichi EX-II cassette tape (1⅞ ips).

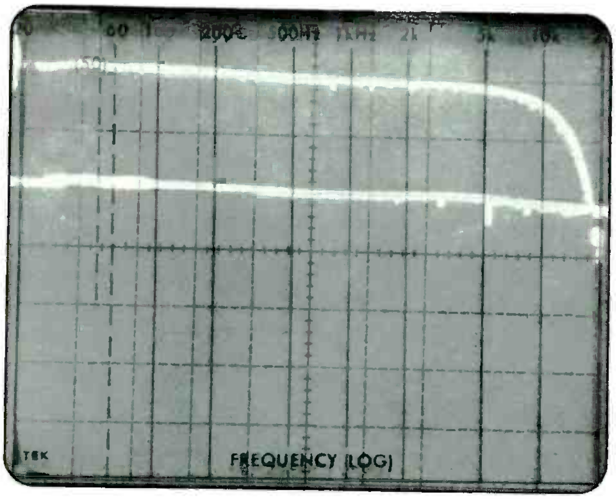


Fig. 3: Nakamichi 680-ZX: Record/play response at 0 dB & −20 dB record level, using Nakamichi SX cassette tape (1⅞ ips).

michi's EX-II standard tape (*Fig. 2*), SX high-bias tape (*Fig. 3*) and ZX metal tape (*Fig. 4*). The advantage of metal tape, when used on a deck that is properly designed for it, may be readily discerned by comparing the 0 dB (upper) curves in each picture. At 10 kHz in *Fig. 4*, response is still virtually flat, while in *Figs. 2 and 3*, response at 10 kHz is already down more than 5 dB owing to tape saturation.

In order to realize the best results from metal tape at the slower ¹⁵⁄₁₆ ips speed, Nakamichi instructs the user to set the EQ for 120 μsec instead of the 70 μsec usually required for metal tape used at the standard 1⅞ ips speed. Following this instruction, we obtained the record/playback curves seen in *Fig. 5*. As might be expected, at this slow speed the 0 dB response shows saturation at the high end (much like previous tapes at the higher speed), but at the −20 dB record level, the response extends to a bit past 15 kHz before rolling off sharply. This, to repeat, is at ¹⁵⁄₁₆ ips.

Using our distortion analyzer we plotted 3rd-order distortion at 1⅞ ips. The results, shown in *Fig. 6*, indicate that the 3rd-order distortion component (using ZX tape recorded at 1 kHz at the 0 dB level) is a full 54 dB down from the fundamental (each vertical division in the photos equals 10 dB). That is equivalent to only 0.2 percent! This measurement, by the way, makes a point often overlooked. If you read THD (total harmonic distortion) on a simple meter-reading distortion analyzer when testing tapes or tape cassettes, what you really are reading is the sum of the various distortion components as well as the contribution to the reading made by random noise. Note that in our "Vital Statistics" chart, this reading turned out to be 0.6 percent. But, in fact, it is 3rd-order distortion that is most important in program reproduction from tape and in determining maximum tape output level or recording level. And in the case of the ZX tape used with the 680ZX deck at the 0 dB record level, that 3rd-order component is an incredibly low 0.2 percent.

General Info: Dimensions are 19 inches wide, 5⅝ inches high, 13⅜ inches deep. Weight is 19 pounds, 13 ounces. Price is $1550.

Individual Comment by L.F.: I can recall, with some nostalgia, when I bought my Nakamichi 1000, a three-head stereo cassette deck introduced in 1972. Imagine being able to monitor while recording with cassettes! Or, even more remarkable, getting record/play response to 20 kHz and beyond, and transport motion with less than 0.1 percent wow-and-flutter!

Well, friends, Nakamichi—the acknowledged leader in cassette deck technology—has been regularly outdong its (and my) pride-and-joy with ever better performing decks. But as far as I am concerned, the new 680ZX offers greater value per dollar than anything Nakamichi has come up with before. Here is a deck that, at the push of a switch or two, actually aligns its completely separate record head so that it is parallel with the playback head for any given tape.

Essentially that's what the 680ZX offers over the 680 which was introduced a year or so earlier. Both models have bias and EQ settings for all sorts of tape, including metal. While that may not seem like much of an accomplishment, I would hate to tell you how many manufacturers of cassette decks have "incorporated" metal tape handling capability that doesn't really show up the new pure metal alloy tapes to fullest advantages. As you can tell from our "Vital Statistics" chart, Nakamichi manages to extract every last ounce of performance out of every grade of tape used with the machine.

Not having had an opportunity to examine the earlier model 680, I was equally impressed with some of the special features carried over from that machine. Example: the Random Access Music Memory (RAMM) which, using the transport controls on the

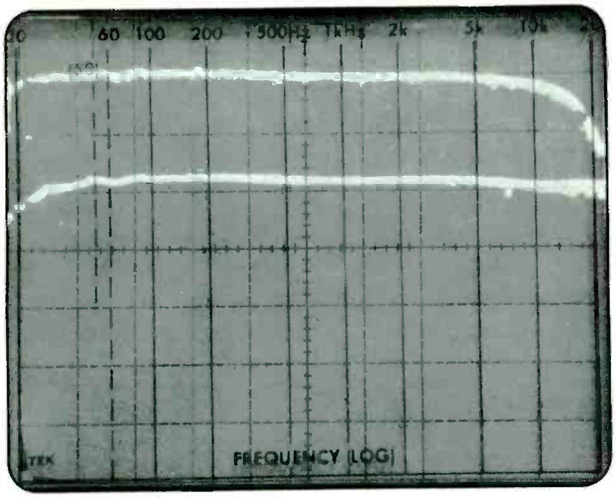


Fig. 4: Nakamichi 680-ZX: Record/play response at 0 dB & −20 dB record level, using Nakamichi metal tape ZX ($1^7/_8$ ips).

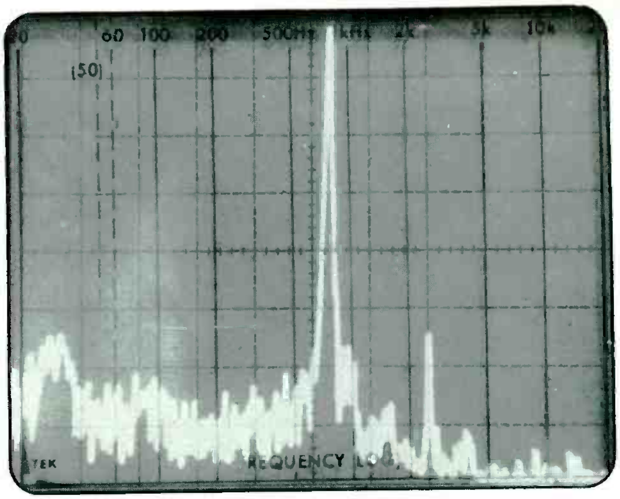


Fig. 6: Nakamichi 680-ZX: 3rd order harmonic distortion (peak at right half of display) is 54 dB below fundamental (center peak) for an equivalent percentage of 0.2%, using ZX (metal) tape recorded at 0 dB, 1 kHz signal.

front panel, lets the machine count the pauses between selections, advancing or rewinding the tape to the selection of your choice. Example: the cueing mode which enables you to hear what is on the playback tape at high forward or rewind speeds without in any way injuring those playback head gaps.

Only Nakamichi would have bothered to provide six separate controls for record calibration (left and right channel adjustments for each of the three basic types of tape), and a calibration method that is so foolproof and easy to perform that you won't find yourself omitting this important step when making an important recording. There are, of course, twelve adjustments—that is, six for each speed. Which brings me to what is perhaps the most interesting feature of all: both the 680 and the 680ZX operate at $^{15}/_{16}$ ips as well as at the standard speed of $1^7/_8$ ips.

Why did Nakamichi choose to go downward in speed when several competitors have begun to offer a higher ($3^3/_4$ ips) speed? According to Nakamichi, the technology of cassette recording has advanced so rapidly and to such a degree that it is now possible to obtain performance at $^{15}/_{16}$ ips that was similar to the best performance obtainable at the higher standard speed of a few years ago. So, reasons Nakamichi, why not provide *more* recording time per tape (twice as much) when the material to be recorded is not super-critical. With the cost of metal tape being what it is, the use of this new tape becomes more reasonable at the slower speed. A C-45 becomes, in effect, a C-90. Nakamichi claims response to 15 kHz (for the −3 dB rolloff point) with metal tape at this incredibly low speed, and we did confirm the claim with two of the tapes we tested (Nakamichi EX-II, and their metal alloy ZX tape). The SX tape (a cobalt-treated ferric-oxide formulation) got out to a very respectable 13 kHz for the −3 dB point.

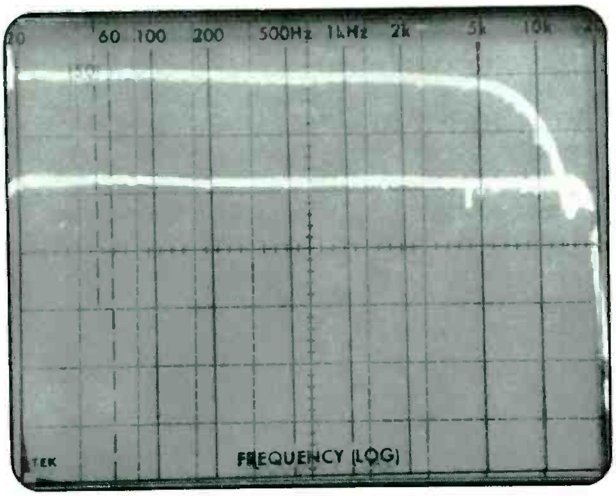


Fig. 5: Nakamichi 680-ZX: Record/play response at 0 dB & −20 dB record level, using Nakamichi metal tape ZX at half speed ($^{15}/_{16}$).

Refer to the captioned photos for a closeup of that section of the panel that includes the feather-touch solenoid-control transport switches and the speed-selector knob. Also check out the closeup view of some of the toggle switches at the right of the intelligently laid-out panel. Note the separate EQ and Tape (bias) switches—a feature even my old 1000 didn't have.

There really is only one succinct way for me to sum up my reactions to this amazing machine from Nakamichi, and that is a simple question: Anybody out there want to buy my slightly used Model 1000?

Individual Comment by N.E.: Kudos to Nakamichi! Once again this manufacturer has come up with a genuine innovation in cassette recording that really pushes back the technological frontier, and advances the state of the art in a meaningful way. While other recent cassette decks may be noted for special features and ingenious twists, the model 680ZX is truly outstanding in demonstrating that a high level of both audio and mechanical performance can be achieved at half the tape speed of the existing format. This makes of the 680ZX not only a great piece of audio equipment, but something of an historic event no less noteworthy in its way than was the fabled model 1000 deck introduced eight years ago. One is tempted to suggest an analogy between the "half-speed" cassette and the long-playing disc, and the comparison does seem fairly valid. Here is a tape machine that provides twice as much program time for a given length of tape and at a performance level that was considered satisfactory at the faster speed only a few years ago. As for the Nakamichi's performance at the standard speed, it is—as far as I can determine—unsurpassed by any other cassette deck I know of.

The model 680ZX is important in another, though related, way. For the first time, since Len and I have been evaluating this class of equipment, we have found a machine that really does justify the use of metal tape on all (not only one or two) counts—in terms, that is, of response, lowest distortion, best signal-to-noise and maximum headroom. Obviously, in grappling with the usual design trade-offs involved in tape-recorder design, Nakamichi engineers have emerged as the winners, so to speak, against the laws of physics.

While endowing this new machine with the alternate slow speed, and with the automatic azimuth alignment feature, the RAMM system, and some other desiderata, Nakamichi has deliberately omitted on-the-panel input mixing and for that matter, inputs for microphones. This doubtless has to do with keeping the deck's cost at a desired level, but it also relates to Nakamichi's design philosophy as exemplified in its "Black Box" line of "accessory components." These include, among other products, a microphone mixer with inputs for three mics. This device, in turn, relates to Nakamichi's idea of the tri-mic "live" recording setup which is explained in a booklet they have issued. So from this particular standpoint, the model 680ZX also represents a deliberately planned element in a larger overall recording philosophy.

NAKAMICHI 680ZX CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
	At Standard Speed of 1 7/8 ips	
Frequency response, standard tape	± 3 dB, 10 Hz to 22 kHz	± 3 dB, 10 Hz to 23.5 kHz
high-bias tape	± 3 dB, 10 Hz to 22 kHz	± 3 dB, 10 Hz to 22 kHz
metal tape	± 3 dB, 10 Hz to 22 kHz	± 3 dB, 10 Hz to 23.5 kHz
THD at 0 dB		
standard, high-bias; metal	1%; 1%; 0.8%	0.75%; 0.8%; 0.6%
Record level for 3% THD		
standard; high-bias, metal	NA	+ 7.5 dB; + 8 dB; + 10 dB
S/N ratio, Dolby off		
standard, high-bias; metal	NA	57 dB; 59 dB; 61 dB
S/N ratio, Dolby on		
standard; high-bias; metal	NA; NA; 66 dB	66 dB; 68 dB; 70 dB
Wow-and-flutter (WRMS)	0.04%	0.04%
Speed accuracy	NA	+ 0.3%
	At Alternate Speed of 15/16 ips	
Frequency response, standard tape	NA	± 3 dB, 10 Hz to 15 kHz
high bias tape	NA	± 3 dB, 10 Hz to 13 kHz
metal tape	± 3 dB, 10 Hz to 15 kHz	± 3 dB, 10 Hz to 15.5 kHz
THD at 0 dB		
standard; high-bias; metal	NA; NA; 1.5%	1%; 1.3%; 0.8%
Record level for 3% THD		
standard; high bias; metal	NA	+ 3 dB; + 2.5 dB; + 5.5 dB
S/N ratio Dolby off		
standard; high-bias; metal	NA	54 dB; 54.5 dB; 57 dB
S/N ratio, Dolby on		
standard; high-bias; metal	NA; NA; 60 dB	63 dB; 63 dB; 66 dB
Wow-and-flutter (WRMS)	0.08%	0.07 %
Speed accuracy	NA	+ 0.3%
	At either speed	
Line input sensitivity	50 mV	50 mV
Line output level	1000 mV	950 mV
Headphone output level	600 mV (8 ohms)	570 mV
Fast-wind time, C-60	NA	58 seconds
Bias frequency	105 kHz	Confirmed
Power consumption	30 watts	32 watts

Ursa Major Space Station

By John Murphy and Jim Ford

The Space Station SST-282 from Ursa Major is a highly sophisticated audio signal processing unit which employs a digital delay line with multiple taps (outputs) to produce a wide variety of reverb, echo and comb filter effects. There are sixteen Audition Delay Programs which provide four basic families of effects: Rooms, Combs, Delay Clusters and Space Repeats. There is also a switch for selecting either reverb or echo mode of operation. The Space Station provides the user with front panel control over a wide variety of operating parameters with the result that many totally unique sound effects can be generated—in addition to the more familiar sounding reverb and echo effects.

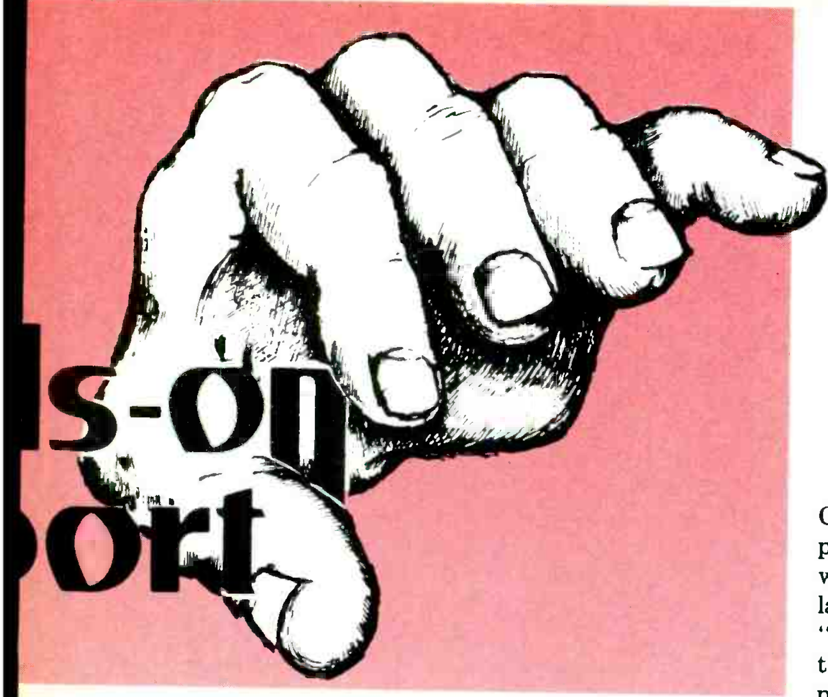
Because of its solid state electronic construction the unit is insensitive to both high sound pressure levels and mechanical shock, this combined with the standard 19-inch rack mount packaging helps make the Space Station at home both in the studio and on stage. The Space Station is currently priced at approximately $1,995.

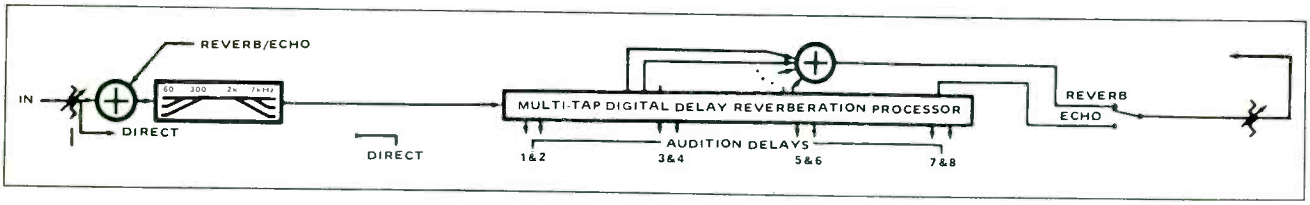
General Description: As is true of most complex signal processing equipment, it is necessary for the user to have some understanding of the unit's operation in order to employ it to maximum advantage. So before going over the front panel controls let's first consider how the Space Station works.

Ursa Major has provided a block diagram on the front panel of the Space Station for quick reference when in use (see photo of front panel). This diagram provides a good starting point for learning about the station. As you can see, the heart of the processor is a multi-tap digital delay line. The delay line provides a maximum of 255 milliseconds (about ¼ second) of time delay with a signal bandwidth of 7 kHz. It has one input and two distinct groups of outputs as well as a separate echo output.

The first group of delay line taps (shown at the top of the delay line in the diagram) are summed together and fed back to the input to generate reverberation when the unit is used in the reverb mode. Reverb decay time is determined by the setting of the feedback control, greater feedback providing longer decay times. The unit is programmed internally to "randomize" the placement of the reverberation taps and avoid the harsh electronic sound of simple fedback delay line reverberators.

The other group of taps (shown at the bottom of the delay line in the photo) are used to audition the signal content of the delay line. There are a total of eight audition delay taps which are mixed in pairs through





four front panel level controls along with the direct signal (on its own level control) to make up the output signal. Taps 1, 3, 5 and 7 are summed and sent to the unit's left output while taps 2, 4, 6 and 8 appear at the right output. The position of the eight audition delay taps along the delay line determines the time delays and relative spacing of the initial echoes from the processor. These eight time delay values are set by selecting one of sixteen Audition Delay Programs. There are four groups of programs named "Rooms," "Combs," "Delay Clusters" and "Space Repeats." The first of these, rooms, uses semi-randomly chosen delays spaced to sound like the early reflections in the natural reverberant field of real rooms. The longer time delays are from the later taps so the user can readily adjust the signal level for the earlier or later taps to emphasize the earlier or later room reflections as desired. Rooms 1 through 4 provide increasing amounts of delay to simulate increasingly larger rooms.

The four comb programs are intended for special effects. Closely spaced delays are combined to produce the series of notches in frequency response commonly known as a "comb" filter (thus named because the closely spaced notches resemble the teeth of a comb). The comb programs provide good sci-fi effects on voices and can be used to add a tonality to percussive sounds.

The next group of programs is referred to as "Delay Clusters" and includes five programs. This family provides eight delays that are closely spaced in time with the cluster of repeats occurring at progressively later times going from the programs "Fatty" to "Echo." In the program labeled "Fatty" the delay of the echo cluster is less than 40 milliseconds. These programs provide extra richness for slap echo type effects.

"Space Repeats" is the name of the last group of programs. Three programs provide 2, 3, or 4 echoes evenly spaced over the 255 milliseconds of available delay. They provide left/right motion and are especially recommended for use with percussive sounds.

When the "echo" mode of operation is selected the reverb taps are disconnected from the feedback loop and the single "echo" tap is connected into the feedback control. The "Echo Delay Time" control is then used to establish the location of the echo tap anywhere along the delay line. Long delay times give rise to decaying repeating echoes, whereas short delay times produce comb filter effects. These effects can be further enhanced by auditioning them through more than one audition delay tap.

Now that we have some idea of how the Space Station works let's go over the front panel controls and rear panel connections. At the far left is an input level control which is used in conjunction with the LED peak level indicator below. The input signal level should be set as high as possible without flashing the 0 dB indicator, as signal levels above this will cause digital overload. To the right of the input level control are two smaller rotary controls labeled "equalization." These controls act on all signals entering the delay line including the feedback signal, providing up to 10 dB of high or low frequency cut. Since the feedback signal is affected, these controls have the effect of reducing the decay times of the highs and lows.

The next five rotary controls constitute the output signal mixer with the first control providing direct signal at the output. The next four controls allow mixing of the signals from the eight audition taps. Each control adjusts the level from a stereo pair of taps with longer delays on the higher numbered taps.

Continuing across the face of the unit next there is a pair of push buttons, one for selecting either mixed or dry output signals, and the other for selecting either the reverb or echo modes of operation. With the first control in the "Dry" position the output consists only of direct signal still under control of the "Direct" level control. In normal operation this switch would stay in the "Mixed" position except to bypass the effects. The reverb/echo feedback control is located at the far right of the front panel. By varying the level of the reverb or echo signals fed back to the input of the delay line, the rate of decay of either type signal can be controlled.

The lower central portion of the front panel is occupied by a set of nine pushbuttons for selecting one of the sixteen audition delay programs. The first switch is an alternate action pushbutton which selects alternate groups of programs for the remaining eight (interlocking) pushbuttons. When the first switch is depressed the eight switches select any of the "rooms" or "combs" programs as indicated by the labels above the pushbuttons. When the first switch is *not* depressed, the "delay clusters" or "space repeats" programs can be selected as indicated by a second set of labels below the pushbuttons.

At the right of the program selector switches is another alternate action pushbutton which when depressed selects a "long" reverb program as opposed to a "medium" reverb program when not depressed. The medium program is said to provide for a normal build-up and smooth decay of reverberation as compared to a slower build up and longer decay time for the long reverb program. Continuing to the right there is a pair of controls for setting the echo delay time when the processor is in the echo mode. A rotary control is labeled 0 to 255 milliseconds and is first set to the desired echo time. The echo delay time is not affected until the associated "set" button is pushed, at which time the echo delay time is updated to the value selected. Finally, a power on/off pushbutton is located in the lower right corner of the front panel along with a pilot LED.

Input/output connections to the Space Station are by way of 3-pin, XLR-type connectors on the rear panel. The units accept either a balanced or unbalanced input signal and provides unbalanced stereo outputs. The pin connections for the input and output connectors are provided for easy reference on the rear panel. The only other item at the rear is a connector for the detachable line cord.

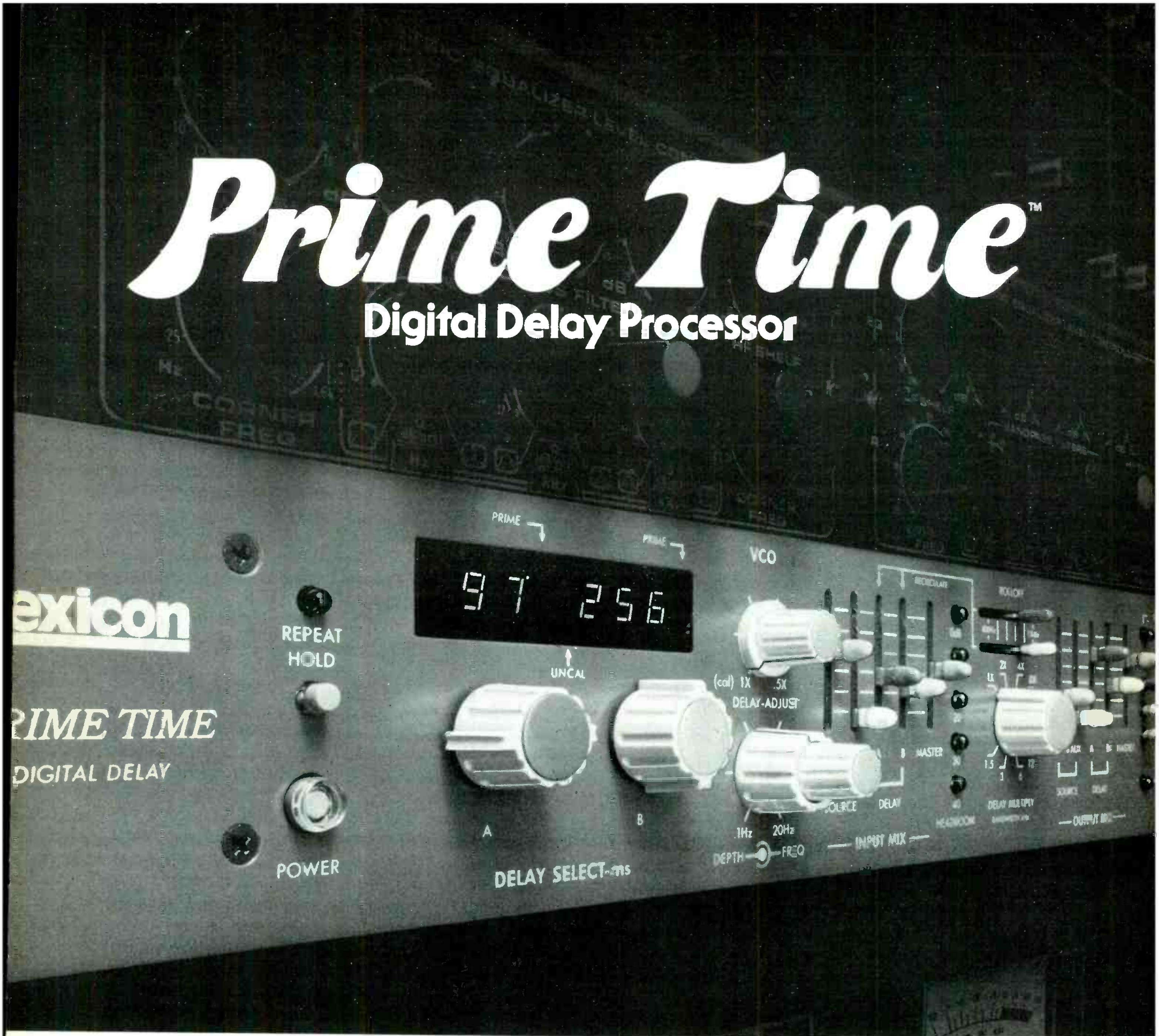
Field Test: The Space Station was field tested at a local 24-track studio where we had the freedom to hear it on a variety of tracks from multi-track master tapes. The Station was interfaced through the console patch bay so that its reverb quality could be readily compared with that of several other reverberation systems on hand. We compared the quality of the various reverb systems on a variety of program material including solo vocal tracks, solo acoustic piano, and solo drum tracks.

We heard some pretty good reverb from the Space Station, but it was not quite as "natural" sounding as either of the two spring reverbs (studio quality types) or the standard plate reverb with which it was compared. However, the Space Station provided reverberation clearly superior to that of another digital reverb unit to which we compared it. In contrast, the Station seemed more diffuse and less colored than the other digital unit. Reverb was most natural with the moderately long decay times that result from moderate amounts of feedback. Higher feedback levels seemed to result in an electronic sounding "hash" towards the end of the decay. We don't feel that this would be a serious problem in most applications however, as the low level hash would typically be masked by program material.

As a special effects unit (as opposed to a reverberation unit) the Space Station truly excels. We found the effect of the delay cluster programs especially appealing on percussive sounds ("fatty" handclaps were great!) while the space repeats programs gave an interesting "machine gun" effect on snare pops. The operator's manual provides over twenty "recipes" of control settings for various effects, most of which we auditioned. Needless to say, most of these have to be heard to be appreciated but they included such effects as: doubling, comb filtering, thickening, slap, ricochet reflections, resonance, fattened echo and pure delay. All in all we heard quite a wide variety of effects and especially combinations of effects (reverb through a comb filter for example). The unit's noise was not noticeable and there was no subjective impression of limited high frequency bandwidth.

We performed our usual listening test back at the shop by connecting the Space Station into a tape monitor loop of the preamp in our reference system. Listening to high quality discs through the direct signal path of the processor we noticed only a slight "softening" in the high frequency range when the unit was switched into the listening chain.

Lab Test: For specific results of the lab test see the accompanying Lab Test Summary. The input of the Space Station was found to be relatively insensitive with no less than a −0.6 dBV input signal level required for a 0 dB peak level indication. Thus, an auxiliary line amp may be required for interfacing with systems standardized on −10 dBV signal levels. Inter-



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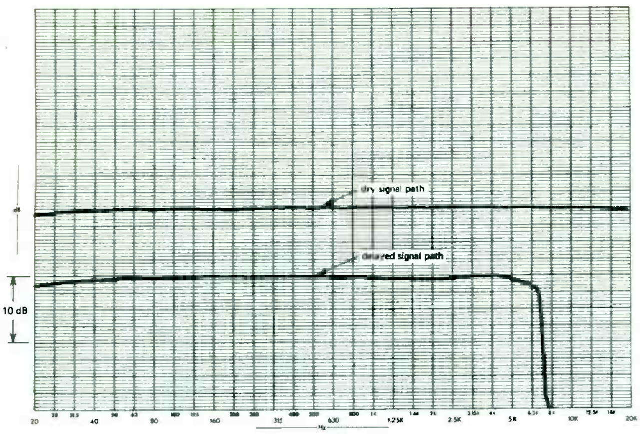


Fig. 1: Ursa Major Space Station: Frequency response of dry and delayed signal paths.

facing with +4 dBV systems should present no problem.

Noise performance through the dry signal path was not quite as good as some professional audio equipment we've encountered and the noise level for the left output was about 6 dB higher than for the right channel. The noise level increased when the four audition taps were set at maximum level but was still a respectable −66.8 dBV. THD through the direct signal path was quite low (about .003%) increasing about tenfold at 10 kHz (to about .03%). For 0 dB level signals through the delay line THD was about 0.12%.

Bandwidth through the dry path was 6 Hz to 20 kHz. The high frequency bandwidth through the delay path was limited to 7 kHz by a very steep anti-aliasing filter *(see Figure 1).* Although more extended high frequency bandwidth would be desirable we observed that the highest octave wasn't really missed in the reverb and effects (maybe because of the full bandwidth of the direct signal).

In testing the direct signal path of the unit we observed that the unit could be driven into slew limiting by high level, high frequency tones. The observed slew rate limit was 0.4 volts per microsecond which implies a slew rate ratio (slew rate limit divided by peak output voltage swing) of 0.054 volts per microsecond per peak volt out. When compared to the recommended minimum slew rate ratio for freedom from slewing induced distortion (0.5)[1] it appears that slewing headroom is marginal. A quick calculation reveals that the slew rate ratio will be maintained above 0.5 for input signal levels up to −19.3 dB peak level. However, operating the unit with such a low input signal level would degrade its noise performance. The best compromise might be to operate the unit with somewhat reduced input levels, say by allowing the −6 dB LED to only rarely flash.

We noted that the unit is excellently constructed using modular printed circuit cards with all the integrated circuits (and there are lots of them!) in sockets for easy maintenance. The user's manual for the Space Station provided sufficient discussion of the processor's operation to allow a new user to obtain good results.

Conclusion: The Ursa Major Space Station is an audio signal processing unit which employs a multi-tap digital delay line to generate a wide variety of special effects as well as high-quality reverberation. The Space Station maintains a good level of audio quality. But a sound effects unit like this has to be heard to be appreciated. So if you're looking for a time delay effects unit we suggest you make sure to audition the Space Station.

LAB TEST SUMMARY

(Note: 0 dBV is referenced to .775 Vrms)

Input/Output Levels

Minimum input level for 0 dB peak level indication:	−0.6 dBV
Output level for 0 dB indication: (dry signal only with the "direct" level control at maximum)	+7.0 dBV
Output clips at:	+16.6 dBV

Noise Performance

(20 kHz filter, unweighted)

For a dry signal only, noise of the output is:	−74.2 dBV (R) −67.9 dBV (L)
With the four audition taps at maximum, noise at the output is:	−66.8 dBV

Distortion Performance

(THD plus noise at 0 dB peak level indication

Frequency	Dry Output	Delayed Output
10 kHz	.037%	–
1 kHz	.0036%	.12%
100 kHz	.0032%	.12%

Bandwidth

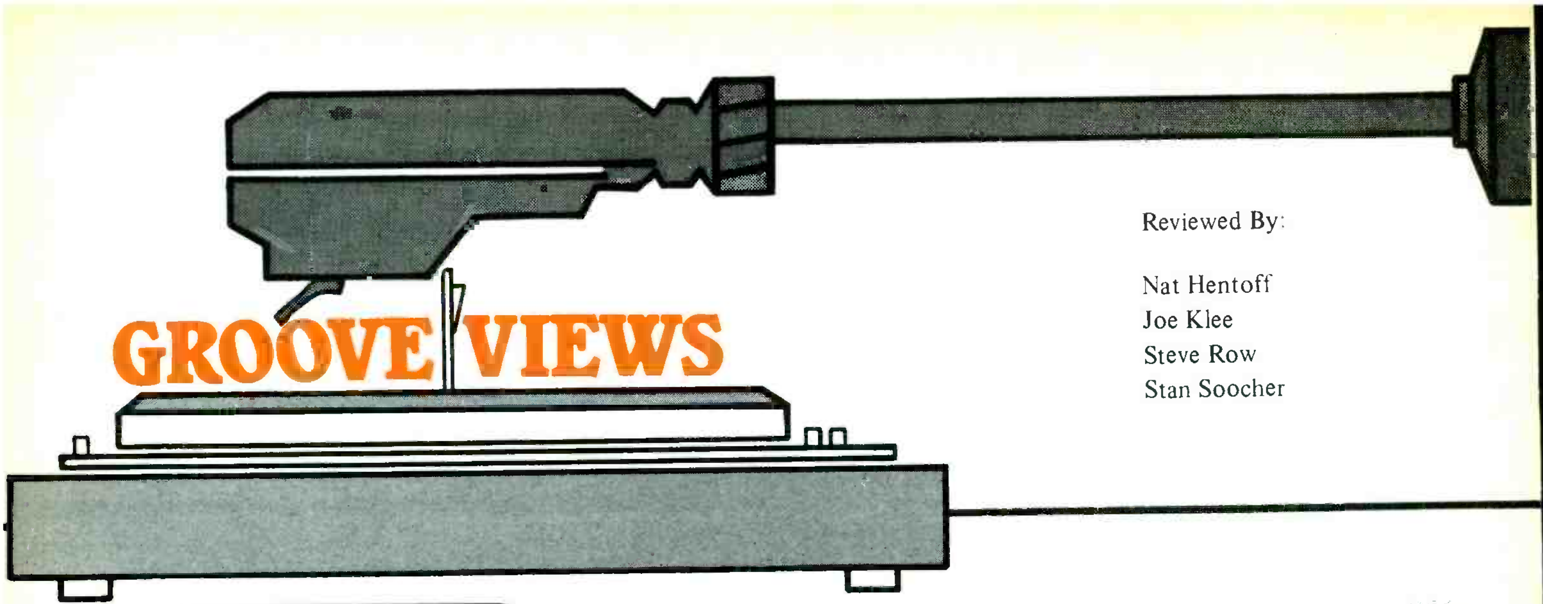
(−3 dB points)

Dry signal path:	6 Hz to 20 kHz
Delayed signal path:	10 Hz to 7.1 kHz

Slewing Performance

Slew rate limit: (dry signal path)	0.4 volts per microsecond
Slew rate ratio (at 0 dB peak level, dry signal path)	0.054 volts per microsecond per peak volt

[1] J.G. Jung; M.L. Stephens; C.C. Todd, "An Overview of SID and TIM, Part II." *Audio,* July, 1979



GROOVE VIEWS

Reviewed By:

Nat Hentoff
Joe Klee
Steve Row
Stan Soocher

POPULAR

FLEETWOOD MAC: *Tusk.* [Fleetwood Mac with Richard Dashut and Ken Caillat, producers; Ken Caillat, Richard Dashut, Hernan Rojas, and Lindsey Buckingham, engineers; recorded at Village Recorder, Lindsey Buckingham's home, and Dodger Stadium, Ca.] Warner Brothers 2HS 3350.

Performance: **Unpredictable method in their Macness**
Recording: **A delightful digital mix**

What a wonderful idea! One of the biggest acts in the business spends close to a million dollars recording a double album of which a large portion sounds like outtakes—from a *demo* session. Now, that's class. *Tusk* intentionally steers Fleetwood Mac away from the super-glossy, squeaky-clean sound of 1975's *Fleetwood Mac* and 1977's *Rumours* in favor of rough-hewn, under-produced bits and pieces that start and stop in unlikely places.

Sure, we still have the comfortable, familiar songwriting hooks, seamless and angelic vocal harmonies, and reliable, yet explosive, interplay between Mick Fleetwood's drums and John McVie's bass. But, especially, songs recorded at Lindsey Buckingham's home reduce the expected impact of these tools by turning Fleetwood's drums, for instance, into Oatmeal-box percussion on "Save Me A Place." Even the relentless beat of "Not That Funny" is dispelled by disconnected cymbal crashes near the end of the song. Apparently, new wave's primitivism has seeped into Fleetwood Mac's sophisticated craft and challenged the band's resourcefulness with positive results. Where else can you find a dose of realism like "Tusk," that, with the help of the U.S.C. Marching Band recorded "live" at Dodger Stadium, conjures up the image of a herd of elephants on safari so effectively it should make Tarzan swing through the trees with envy.

FLEETWOOD MAC: A class act in need of some restraint

Christine McVie sticks closest to Fleetwood Mac's established formulas. "Never Forget" and "Honey Hi" are her structural bows to Buckingham's risky approach. "Over And Over" and "Brown Eyes," however, spotlight the same wistful crooning and manicured melodies heard on *Fleetwood Mac's* "Over My Head." "Never Make Me Cry, with its haunting refrain, reestablishes Christine as the clearest, most convincing voice of Fleetwood Mac.

Stevie Nick's singing does not bear up as well, but this is part of her mystique. "Storms," a gothic romance in the tradition of Emily Bronte's *Wuthering Heights*, finds Stevie's voice faltering, like the relationship, unable to give a dramatic little push to the end of each note. But the mesmerizing and sometimes tormenting power that women and men hold over each other which she writes about in her songs can only be pierced by this vocal vulnerability, making her material attractive rather than prohibitive to the incurable romantic she appeals to.

Tusk's digital mix brings the album's flaws to the forefront. Nothing is hidden or buried. But strengths are heightened, too. With the acoustic guitars, for example, the pick attack becomes as impor-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tant as the sound of the guitar itself.

One unenviable flaw is that some of the material here does sound repetitious, so perhaps Fleetwood Mac should have used some restraint and released *Tusk* as a single album. *Rumours*, after all, produced four hit singles. But who's to say that good things always come in small packages. At least in this case—with all due respect to piano players and elephants—Fleetwood Mac isn't simply tickling ivories. S.S.

SUSIE ALLANSON: *Heart to Heart* [Ray Ruff, producer; Jerry Hall, engineer; recorded and mixed at Heritage Studios, Hollywood, Ca.] Elektra 6E-177.

Performance: **Cloying**
Recording: **Not bad, but what cliches!**

I have this picture of Susie Allanson performing on television, lip-synching her way through a couple of songs from this album, and getting to the end of each number, not really knowing how to lip-synch a fadeout. We've seen this too many times, and it's just too laughable—does one slowly step backward as the music fades, does one mouth the words through a smaller opening, or what?

Susie Allanson is going to have trouble appearing on television to lip-synch these songs because each of the ten tracks on the album ends with a fadeout. Even used sparingly, that recording gimmick can get old fast; used on each track, the device becomes painful to listen to. The artist and producer need something to overcome that glaring aspect of the recording; unfortunately, neither the artist nor the producer here have much else up their sleeves.

This is pure 1970's female pop-oriented country music, with everything intact that some listeners find enjoyable and some critics find so objectionable. The arrangements are fully-blown, with serious, sad-sounding string sections behind the basic ensemble of guitars (including pedal steel), bass and keyboards. The songs are, for the most part, ballads about love: true love, lost love, found love.

Allanson has a musical voice, but it is tiny, tinier even than Emmylou Harris', and there are times when she seems overwhelmed by the grand arrangements. When the accompaniment is toned down—in "Two Different People," with its nice acoustic guitar duo opening, for example—she is much more of a presence on the album, and one does not

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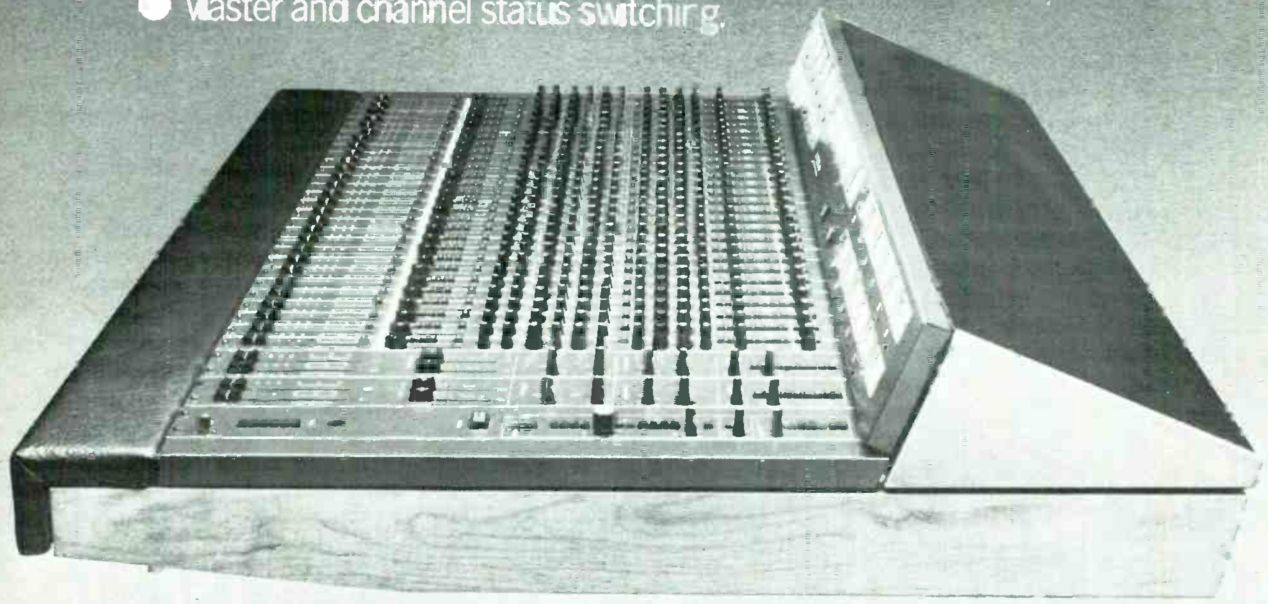

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notice the thinness of her voice as much.

But there are other problems. Some of the material sounds contrived, unnatural. "Heart to Heart" has a big chorus and overdubbed harmonies, many strings, and sounds important at first. But by the end of the track one has the impression that this was a tune written especially for a TV sit-com or drama or perhaps a soap opera about a singer. Rather than pay royalties to someone for the use of a real song, the show picks up a little number cranked out by some songwriting factory especially for television shows.

One will hear all the tricks of the studio arranger's trade here, used frequently to mask the lack of substance in the material or inadequacies in the artist. I'd like to think that Allanson has few drawbacks by herself, and that this album fares so poorly because of the selection of material and the over-hyped way of presenting that material. But cascading strings, piano techniques right out of the Ferrante & Teicher school of kitsch, bouncing bass drum sounds that skip joyfully from one channel to another a measure or two before a critical point in the lyric, and those fadeouts! We deserve better.

Is there anything salvageable about this recording? The country music fans seem to have found this a much less objectionable release than this reviewer has found, and it has been settled in the top 50 or 100 country recordings for a good portion of 1979. But there must be something better than this. The lip service paid to Phil Spector and the Brill Building (in "One Fine Day," for example), and the slightest shade of "blackness" (spelled s-o-u-l, as in the suspiciously-titled "What's A Matter?") don't add much to what is basically a pretty weak recording effort. S.R.

JAZZ

ART FARMER: ***To Duke With Love.*** [Tashimari Koinuona, producer; Ben Taylor, engineer; recorded at Vanguard Studios, New York, N.Y., on March 5, 1975.] Inner City IC 6014.

Performance: **Uncommon performances of uncommon Ellingtonia**
Recording: **Typical Vanguard studio sound**

For those of us who are getting a little tired of tributes to Duke Ellington that carry on as though "Satin Doll," "C Jam Blues" and "Mood Indigo" are the only tunes that the Duke ever wrote, here's an album of fine renditions of some of Duke Ellington's lesser known masterpieces, plus at least one by Billy Strayhorn, Duke's right-hand man. And who better to do these numbers than flugelhorn player par excellence, Art Farmer? While "In a Sentimental Mood" and "It Don't Mean A Thing" aren't unknown Ellingtonia, they certainly haven't been done to death. Billy Strayhorn's "Lush Life" has been somewhat overexposed lately, but there's enough freshness in Farmer's interpretation that it becomes a worthy addition to the growing list of interpretations of Strayhorn's second most popular composition. "Love You Madly" and "The Brown Skin Gal In The Calico Gown" are heard too seldom, indeed, and the Duke Ellington/Billy Strayhorn collaboration, "The Star Crossed Lovers" (for which the album cover mistakenly lists only Ellington as composer), is so seldomly heard that it ranks as a find. The piece was part of the "Such Sweet Thunder" suite which Ellington and Strayhorn collaborated on and was recorded for Columbia by the Ellington Orchestra. It actually had a previous life as a Billy Strayhorn composition on a Johnny Hodges date for Norgran Records (9/8/75) under the alternate title of "Pretty Little Girl."

Farmer, as always, is a master musician. His accompanists do well when they accompany him but as a soloist, in this sort of music at least, I do not feel Cedar Walton fits the role too comfortably. His boppish flurries don't quite fit the mood of Ellington's music—neither does his obvious attempt to emulate the Duke on the intro to "Brown Skin Gal." Also I could have done with a drummer who is less busy behind a soloist than Billy Higgins. But then the best rhythm section for this kind of an album would be Duke Ellington on piano, Jimmy Blanton on bass and Sonny Greer on drums, but only Sonny's still around and he's not playing the same way he did back when he was with Duke's band. So like the song says, things ain't what they used to be, but this apparently is good enough.

Art Farmer's horn playing is better than good enough. As far as I know, Art Farmer never played with the Ellington band, yet I hear a lot of Ellington's

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the (more than) survivors: art pepper and duke jordan

By Nat Hentoff

Art Pepper's current autobiography, *Straight Life* (Macmillan) could be a stunningly dramatic novel —fame and the blazingly fast night life and then a long, deep fall, being strung out, doing time. But it all did happen to this former star of the Stan Kenton band, an alto saxophonist so compelling that for a while, he was topped only by Bird in the *Down Beat* poll. After many wasted years, Pepper straightened out, and now he's playing with more disciplined force, incisive originality, and sheer emotional directness than at any time in his career.

As evidence of Pepper's regeneration, there is *Straight Life* in music (Galaxy). With a nonpareil rhythm section—pianist Tommy Flanagan, bassist Red Mitchell, drummer Billy Higgins—Pepper swings mightily on such tunes as his own "Surf Ride" and reveals how freshly penetrating a ballad player he has become in "September Song." His sound is vibrantly clear and lean, as if scoured of all soft immaturity by his hard times. His phrasing is utterly personal but never eccentric, everything cohering in a firm, driving unerringly logical direction. Whatever the polls may say these days, Pepper is again unmistakably among the very top rank.

The engineering is as impressive as the playing. There is complete presence of the ensemble and the soloists, with a depth of clarity and aural perspective that makes this a model of what a jazz date should sound like.

Duke Jordan has also had an uneven career, though not to the traumatic extent of Pepper's tribulations. He first became renowned for his work with Charlie Parker in the 1940s, going on to Stan Getz in the 1950s, and in more recent years, spending considerable time in Europe, particularly Scandinavia. Except among musicians, Jordan has never received sufficient recognition for his extraordinarily luminous, lyrical playing which can also be as briskly energizing as the very best of bop. Furthermore, Jordan is an unusually inventive and lucid composer, though in this sphere too he has been unaccountably overlooked.

For a sense of the contemporary Jordan, I unreservedly recommend *Lover Man*, recorded in Denmark on SteepleChase and now available here through distribution by Rounder Records, (186 Willow Ave., Somerville, Massachusetts 02144). Accompanied with authoritative, crisp ease by bassist Sam Jones and drummer Al Foster, Jordan ranges from Mile Davis' "Dig" to revivified pop standards to his own tunes ("Dancer's Call" and "Love Train"). Among the many pleasures of his playing is a unique dancing beat, creating an infectious resiliency of mood that might lift almost anyone from gloom.

The engineering, while not quite as expansively "natural" as on the Pepper session, is clean, vivid, and especially illuminating of Jordan's playing. This is a set that will never be out of fashion.

ART PEPPER: *Straight Life*. [Ed Michel, producer; Wally Buck, engineer.] Galaxy GXY-5127.

DUKE JORDAN: *Lover Man*. [Nils Winther, producer; Chuck Irwin, engineer.] SteepleChase SCS-1127 horn section in his playing. I hear the roots that came out of Rex Stewart and Ray Nance, with a tinge of Clark Terry, who probably came from the same roots. Now that Rex and Ray are gone, there's nobody better than Art Farmer to play this pretty music.

Maybe it's my imagination but it seems like a lot of the records that come out of Vanguard's Studios sound pretty much the same. Whether it's the engineers or the studio sound, I couldn't say for sure but they seem to lack any distinctive characteristics such as one hears on records that come out of the CBS facility on 30th Street, or their previous studios at Liederkranz Hall. It's a small complaint. The sound's serviceable, just not outstanding.

The music, on the other hand, is outstanding. Dig it! J.K.

RAMSEY LEWIS: ***Ramsey.*** [James L. Mack, producer; some tracks produced by Wayne Henderson; George Butler, executive producer; Allen Sydes, Mike Evans and Stu Walden, engineers; recorded at Universal Recording Studios, Chicago and Hollywood Sound Recording Studios, Hollywood.] Columbia JC 35815.

Performance: **Where have you gone, Sloopy?**
Recording: **Bright and splashy**

GEORGE DUKE: ***Follow the Rainbow.*** [George Duke, producer; Kerry McNabb, Dave Rideau and Dave Palmer; engineers; recorded at Westlake Recording, Los Angeles, and Electric Lady, New York.] Epic JE 35701.

Performance: **Spirited, but cliches everywhere**
Recording: **Also bright and splashy**

Put these two albums on back to back and (1) you almost couldn't tell them apart, and (2) you almost wouldn't know that they are by two of the more inventive keyboard players in contemporary jazz. As a result, both albums are distinct disappointments.

Lewis's release contains only one cut that is really exciting—the extended medley to close the album that features music by producer James L. Mack sandwiched around the Anthony Lloyd Webber-Tim Rice song from *Evita*, "Don't Cry for Me, Argentina." This is a darkly beautiful piece of music, all done in a Latin or Spanish tempo, with tastefully muted strings and an underpinning of winds, basses and cellos. Although the melody is not very daring or imaginative, the arpeggio work by Lewis on the grand piano is quite good, and the whole feeling of the piece conveys the same kind of mood as Ravel's *Bolero* or music from *Carmen*. The piece then moves into a lightly jazzier mood with Lewis' playing complemented more by drums, guitar and bass than by full orchestra. He moves into his distinctive jazz style briefly before the piece darkens again and comes to a close.

The sound of this piece is gorgeous, both musically and technically, and this should have provided a theme for the selection of much better material for the rest of the album than *Ramsey* now contains. Unfortunately, we are left with a forgettable collection of R&B/disco/funk that will leave most listeners bored to tears. A principal example is the leadoff cut, the 60s evergreen "Aquarius/Let the Sun Shine In," in which the uptempo arrangement of the former is devastated by squeaky strings, and the latter suffers from a case of wimpy female chorus. And this song tends to set the tone for the whole record.

The chorus slurs the title of another track, "Wearin' It Out," to almost one syllable, and the immortal lyrics, "Are ya ready?" pop out with alarming frequency in a song entitled "Every Chance I Get." The album also contains the obligatory song that refers to dancing—this one called, strangely enough, "Dancin'"—and a couple of love ballads that try to rise above the dismal quality of most of the album. "I'll Always Dream About You," with Lewis on the Fender Rhodes, is very smooth, slick, polished, and marred only by the overbearing presence of a string section. "I Just Can't Give You Up," has a Mancini sound to it, with brass and strings prominent.

The principal fault with the album is the lack of dominance by Lewis' keyboards, and one must fault the producers for that. In some songs, one tends to forget that this is a Ramsey Lewis record, because one can't detect too much presence by Lewis in the material. The drums and percussion just don't let up, taking the place of Lewis' piano work of yesteryear in today's emphasis on insistent, monotonous rhythm.

Duke's album suffers from the same malady—too much R&B/disco/funk and too little imagination. The listener should be forewarned about this album by some of its titles—"Party Down" and "Funkin' for the Thrill," for example. The former sounds like so many other 1970s black oriented records, with predictable rhythm, male and female vocal leads, then chorus, then instrumental break, then whispered lyrics. The latter is dull, dull, dull, with the immortal tag line: "Get up and boogie."

The album has a few redeeming features—the nicely arranged "Festival" that closes the first side, for example. This track features sparkling instrumentals, nicely separated in the recording, that sounds something like Sergio Mendes' best work. Some very splashy percussion also is featured, along with a shift from Latin to African tempo, heavy on congas and unusual percussive devices. Listen for a shift in the recording of the bells from one channel to the other here, too.

But a lot of the record is pretentious gibberish, either in the music or in the lyrics, or both. "I Am for Real" uses the worn device of a spoken chorus to open then moves into what sounds like soulful Chipmunks to continue. And it all sounds like contrived nonsense.

The title track closes the album, and Duke's synthesizer literally unfolds in an elaborate opening that is quite interesting to hear. But at only 1:23, the song ends much too quickly, and before one can savor the nice use of the synthesizer the song has ended, almost as if it were an afterthought to the album.

Both albums suffer from commercial blandness and a lack of innovation that is really alarming. It seems almost as if the artists involved have placed whatever skill and imagination they have built up over the years in a box and have substituted pursuit of trendy, plastic mediocrity.

S.R.

CLASSICAL

JEAN SIBELIUS: ***Concerto in D Minor for Violin and Orchestra.*** Gidon Kremer, violin with the London Symphony Orchestra; directed by Gennady Rozhdestvensky. [Hans Richard Stracke, producer; Horst Lindner, engineer; Fred Miller, remastering engineer; recorded at the University of Salzburg, Austria on August 12 and 13, 1977.] Vanguard VSD 71255.

Performance: **Slow and dark and Russian**

Advertiser's Index

Recording: **Soloist not in the forefront**

ALFRED SCHNITTKE: *Concerto Grosso for Two Violins, Strings, Cembalo, and Prepared Piano.* Gidon Kremer and Tatjana Grindenko, violins with the London Symphony Orchestra, directed by Gennady Rozhdestvensky. [Same production credits as above.] Vanguard VSD 71255.

Performance: **A definitive performance of a transient work**
Recording: **Contemporary but reserved**

Few LPs that I have heard couple works less similar than Jean Sibelius' romantic, almost 19th-century D Minor violin concerto and Alfred Schnittke's angular, modern (if not ultra-modern) revitalization of the concerto grosso form. Yet when Maynard and Seymour Solomon founded the Vanguard Recording Society they *did* promise us the unusual.

To begin with the familiar, though not as familiar as it deserves to be, the Sibelius *Violin Concerto in D Minor* (D is the fiddler's key because the instrument is so constructed that the key of D utilizing the open middle strings of D and A bring out the full resonance of the violin to its best advantage) hasn't exactly been neglected by violinists when it comes to records. The Schwann catalogue lists versions by Perlman, Zukerman, Stern, Heifetz, Oistrakh, Francescatti and most of the other great concert violinists. Yet, considering how many times a season the average concert goer has the opportunity of exposure to the violin concerti of Beethoven, Brahms and Tchaikovsky, the Sibelius comes in a weak fourth. It's a shame too because there is much of beauty to this, the last of the true romantic violin concertos. Written in 1903, and revised in 1905, the Sibelius concerto harks back to earlier influences (many cite Edvard Grieg, but I also hear shades of Tchaikovsky, Brahms and especially Rimsky-Korsakov). It received its first major exposure via a classic HMV recording by Jascha Heifetz with Sir Thomas Beecham and the London Philharmonic. Even today this is the standard for comparison of new versions of the Sibelius concerto. And yet these comparisons are not fair. Say what you will about Heifetz' cold and unemotional playing there is no arguing with his virtuoso technique, his sureness of intonation or the brilliance of his sound. Perhaps sensing that competition with the Heifetz/Beecham rendition is futile, Gidon Kremer and Gennady Rozhdestvensky have opted for a totally different version of the Sibelius *Violin Concerto.* It is slower. It is darker. The musical dialect is far more Russian. This is not surprising for a Russian conductor and violinist who would naturally emphasize those elements which they hear in the music which are common to their experiences. Between that concept and the engineering, which places the soloist in the orchestra rather than in front of it, the Sibelius *Violin Concerto* emerges as a symphonic work with violin cadenzas, obligattos, etc. It's a valid conception and one which can coexist with the Heifetz/Beecham recording in which Heifetz is the star soloist and Beecham and his orchestra are content to be the accompanists.

As for the Schnittke *Concerto Grosso* it is difficult to evaluate a work so young. It may well earn its place in the repertoire and certainly this rendition by the two violinists for whom it was written and to whom it was dedicated will be largely responsible for any eventual success the work may achieve. While it is clearly an avant-garde work of the 20th-century it takes the form of a classic Concerto Grosso of Bach, Vivaldi, Corelli and Handel. To be sure it is a modernist's idea of the form but the classic form is there.

The thing that strikes me most about this recording is how the science of the LP record has served the art of the contemporary composer. Jean Sibelius wrote his *Violin Concerto* in 1903 and revised it in 1905. If there was a complete recording of the work before the famous Heifetz/Beecham HMV of 1935 it was certainly eclipsed by that definitive version which was first released as part of the Sibelius Society Project which the late Walter Legge instituted at HMV. The technology of recording in the early part of the century was such that Sibelius had to wait thirty years for a recorded interpretation of such majesty. Schnittke, on the other hand, wrote his *Concerto Grosso* during the period between 1976 and 1977. It was not even a year old before Schnittke could hear a recording of the work performed by the very players for whom he had intended the piece. There's no guarantee that in thirty years we are going to look back on Schnittke's *Concerto Grosso* as a masterpiece but at least we've had the opportunity to hear it and study it at our leisure. An opportunity which was unimaginable to the followers of Bach, Beethoven, Brahms, Mozart, Wagner and even so recent a composer as Sibelius. J.K.

SHOWS and SOUNDTRACKS

ORIGINAL MOTION PICTURE SOUNDTRACK: *Apocalypse Now.* [David Rubinson, producer; Leslie Ann Jones, Bill Steel, Ken Kessie, Richard Beggs, Katherine Morton, Jim Austin, Steve Mantoani, Stacy Baird, Emil Flock, Dan Healy, Bob Matthews, Brett Cohen, Betty Cantor-Jackson, Phil Yeend, Charles Garsha and David Rubinson, engineers; recorded at The Automatt, San Francisco; Omni Zoetrope Studios, San Francisco; Different Fur Music, San Francisco; Club Front, San Rafael, and CBS Studio Center, Los Angeles, Ca.] Elektra DP 90001.

Performance: **Sorry, but we've lost our picture**
Recording: **Hollywood big as life, sometimes bigger than life**

I remember when I was a kid and we used to go to those Saturday matinees (the inevitable cowboy picture, a Mickey Mouse and one episode of Don Winslow, Buck Rogers or Batman) when a projector light would go out (equipment wasn't as stable back then as it is now) and we'd sit there hearing the soundtrack and watching a darkened screen. Pretty soon we'd start whistling and shouting and stamping our feet and we wouldn't stop until the malfunction had been remedied. I get something of the same sensation listening to this recording direct from the soundtrack of Francis Ford Coppola's latest epic, *Apocalypse Now.*

To be sure the film's here, more than three-fifths of the whole 150-minute film, but does it work on a strictly audio level? To check that out, I listened to these two LPs both before and after I went to see the film. It doesn't work. Hearing the soundtrack without seeing the visual presentations left loose ends and unanswered questions galore. Even after seeing the film the record becomes an experience only when you let your mind go back to what *was* on the screen at that moment. This is not to take away credit from Mr. Coppola or Mr. Rubinson. They did their jobs—it's the concept that I find fault with. It's not like the soundtrack to *A Star Is Born* where all you need is Streisand singing the hits again for you. There's a lot of music in this film but seldom, if ever, is it music for music's sake. It is truly background music underscoring a dramatic situation. That doesn't make for an easy transfer to records. Whoever's decision it was to make this a two-fer and include enough of the dialogue for the listener to get the jist of the plot came up with the best solution they could have in this situation and the decision to concentrate heavily on Michael Herr's journalistic narration is a big help. But still no words can conjure up the images that Francis Ford Coppola, who uses a camera the way a composer uses staff paper and pen, manages to put on film. They tell me that the music, with the exception of a couple of rock records interspersed throughout the film and a brief snip from Wagner's "Ride Of the Walkuries," is done only with choir and synthesizers. If that's true then Patrick Gleeson, Richard Beggs and Bernie Krause (I remember when he took Pete Seeger's place with The Weavers) deserve a lot of credit. I've never heard truer string sound from any synthesizer.

The sound itself is big in the theatre and very directional. Perhaps trying to make up for the lack of a visual image and for the lack of directional focus of the sound Rubinson sometimes tries too hard for the *big* effect which doesn't quite come off and ends up being more distracting than helpful. But I guess having set himself an impossible task we can only marvel at how close he came to doing what he set out to do.

The one cautionary word I would like to drive home in this review is to warn against the dangers of taking the record for the whole experience. It did me no good to try to prepare myself for the film by listening to the record; in fact I think I might have gotten more out of *Apocalypse Now* just walking into the theatre cold. However, once you've seen the film the record will help you relive the experience. It's a strong film and there is strong language. There are sexual obscenities, bathroom obscenities and one racial obscenity which had been deleted from the film when I saw it (it didn't slip past me because I was waiting to see how it would be handled so I paid close attention) but still remains on the record. How "up tight" explicit language makes you will be a determining factor on how up tight this film may make you—likewise the recording. J.K.

CLASSIFIED ADS

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