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MODERN RECORDING & MUSIC

VOL. 7 NO. 12
SEPTEMBER 1982

Recording With Al Jarreau

Recording Techniques, Part 6

LAB REPORTS:
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Sony TC-FX1010 Recorder
Technics SH-8065 Equalizer

NOTES:
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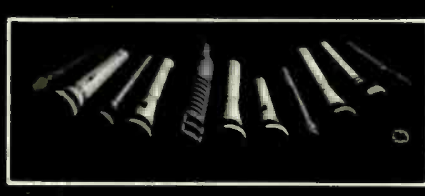

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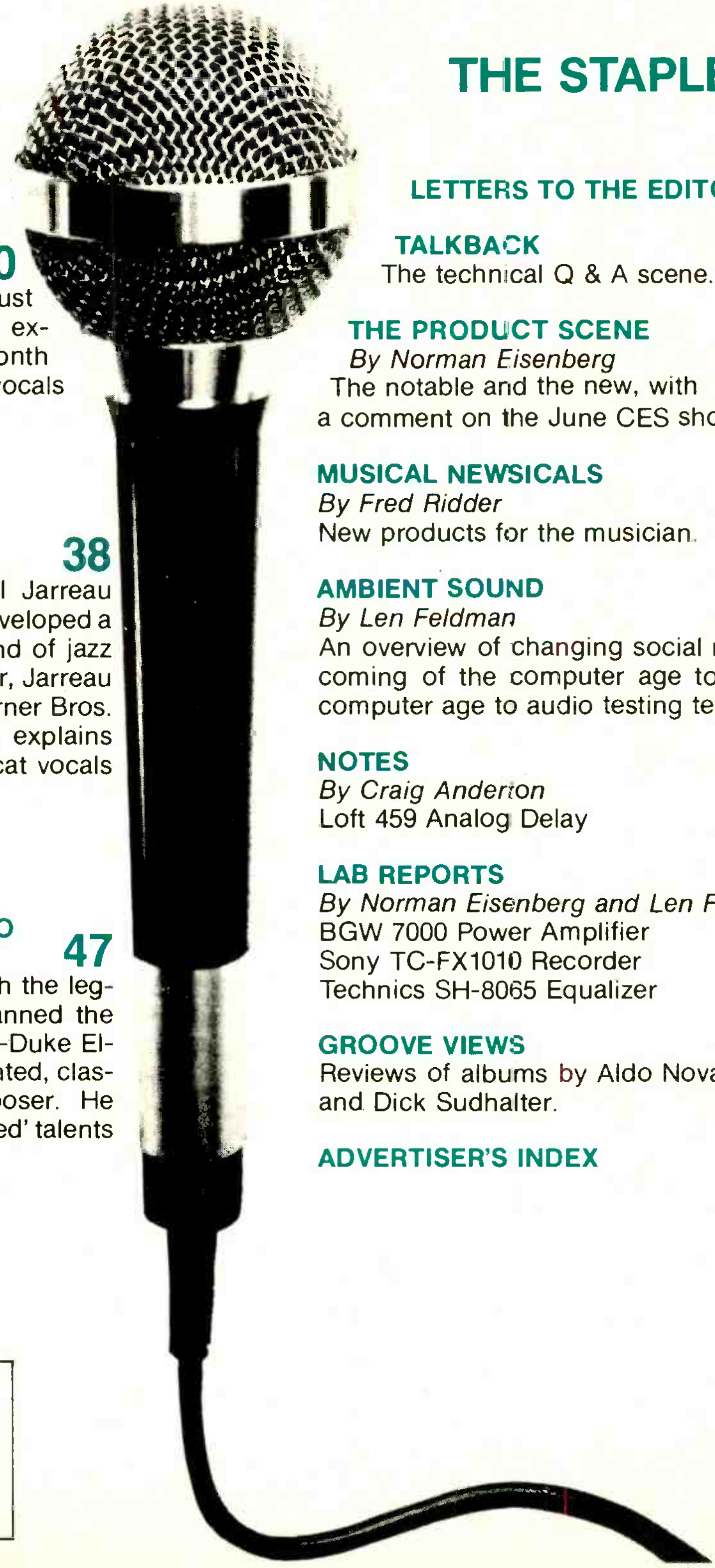

THE FEATURES

THE STAPLES

COMING NEXT ISSUE!
Recording With Squeeze
Profile: Brian Eno
Recording Techniques, Part VII

Cover Photo: Bill King, Courtesy of Rogers and Cowan
Jarreau Color and B&W Photos: Courtesy of Rogers and Cowan
Macero Photos: Bob Sorce

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Letters to the Editor

A Fostex Misunderstanding?

We received the following letter from David Oren of Teac Corporation in regard to the "Fostex Controversy."

Len Feldman's comments regarding the Fostex A8 8-track recorder in the May, 1982 issue of *MR&M* did, as he suspected, miss the point. Len seems to think there was a problem with that machine's ability to record "only" four tracks at a time instead of all eight simultaneously. In fact, when doing multitrack production, one almost never records all the tracks simultaneously. There may be 24 or 32 mic inputs on a big mixer, but you'll seldom see all 24 tracks of a 2" recorder filled on the first pass. Similarly, there may be 8 or more inputs on the mixer in the smaller 8 track studio, but you'll seldom want to fill all 8 tracks at once. (It is true that other 8 track machines such as TASCAM's *permit* all 8 to be recorded at once, but this is more a convenience in copying 8 track master tapes than a requirement for multitrack production).

Len is a talented, knowledgeable audio critic and writer, but when he wrote "I can't imagine breaking up a musical group into two separate sessions...," he expressed his lack of familiarity with the true advantages of multitrack audio production. MULTITRACK WAS CONCEIVED TO ALLOW YOU TO SPLIT UP THE GROUP INTO TWO, THREE OR MORE SESSIONS. The ability to lay down additional tracks in synchronization with previously recorded tracks (the "overdub" process) is what gives multitrack its advantage over the early "one take" recordings on wax, and later on mono and stereo tape machines. You can thus create a master tape with a smaller studio that might not be able to house all the musicians at once with adequate acoustic separation. You can do it when not all the musicians are available for a given recording session. You can do it when the same performer is playing or singing more than one part . You can fix one bad part without having to redo all the parts. And you can rehearse the blending and equalization of these parts down to stereo (and/or mono) many times before committing to a mix. The advantages of multitrack are great, and they don't require that all tracks be recorded at once.

Another of Len's criticisms regarded the use of a single head for recording and playback. Once again, he missed the point. You need a single record/repro head for synchronous recording while listening to previous tracks (e.g., overdubbing). An additional repro head *may* be necessary if sync playback response from the record head is not adequate (playback heads typically have narrower gaps than record heads, particularly in larger format machines). The separate repro head also aids in azimuth adjustment, but in the case of the Fostex A-8 there is no real need for a separate repro head; head alignment is not critical with ¼" wide tape that is played back using the same head with which it was recorded. While TASCAM's 8-track machines have separate repro heads, the repro heads are generally used to check azimuth alignment (TASCAM uses ½" tape for 8-tracks, not ¼", and since those TASCAM tapes are more likely to be mixed in other studios than are FOSTEX's "personal multitrack" tapes, correct azimuth becomes more

Like any fine musical instrument, TDK Professional Reference audio cassettes and open reel tapes are products of genius. In TDK's case, it's the genius of constantly-advancing audio technology. And now, TDK technology has advanced again in the reformulation of our MA-R metal, SA-X high bias, and AD-X normal bias cassettes.

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important). With TASCAM's new machines, such as the 8-track 38, the record head can be used for playback during mixdown because its frequency response is equal to that of the repro head.

Even though the A-8's combined record/repro head can record "only" 4 tracks at once, it will play back up to all 8 simultaneously when recording is not in progress. This fulfills the requirements of mixdown, where the 8-track master tape is transformed into a finished stereo program.

It may seem odd to you that we have come to the aid of FOSTEX, a new company which seeks to compete with TASCAM. However, we cannot gloat over a significant misunderstanding of Fostex's A-8 because such a misunderstanding injures us all. We feel very strongly that promulgating a thorough understanding of the multitrack process is the most important goal for our industry as a whole.

—David Oren
Marketing Manager
TASCAM
Production Products Division

Len Feldman responds to David Oren's letter:

While I appreciate David Oren's gentle manner of criticism of my reaction to the Fostex A-8, I am afraid that Mr. Oren missed *my* point. Honest, David, I'm well aware of the fact that "one almost never records on all tracks simultaneously" during multi-track recording. I am also very well aware of the overdubbing process and all its refinements and variations. What bothered me was the price/performance ratio of the A-8. As Mr. Oren himself points out, Tascam's 8-track machines do permit all 8 tracks to be recorded at once (on their 8 track machines) and Tascam's machines also have separate repro heads.

What Oren ignores in his letter are the very poor test results which we measured for a not inexpensive ($2500.00 is still not a "cheapie" in my book) tape deck. How about the low frequency noise that can't be helped by the Dolby C? How about the low frequency contour effect which is clearly in evidence from about 500 Hz downward? And, yes, I *still* maintain that using a 7-inch reel of tape at 15 ips doesn't give you enough recording time in a lot of remote in-the-field situations for which this machine is ostensibly intended.

I had a long talk with one of the recording engineers who is a regular contributor to this publication and who used the Fostex A-8 in the field after I was finished with it. He loved it and found it a handy tool for certain recording applications. I have no doubt that as a rehearsal deck or where high quality multi-track recording is not vital, it has some merit. But I still question its price and some of the approaches used in its design. After all, it's differences of opinion that make horse races, right? All I really want to get across in responding to David Oren (whom I know and respect, incidentally) is that he give me some credit for knowing what multitrack recording technology is all about after owning and using one of the machines that *his* company makes for more than ten years!

—Len Feldman
Technical Editor
Modern Recording & Music

Praise and Admonishment

I am renewing my subscription mainly for the Studio Notebook series. I would like to see more such dutch uncle stuff about how not to lose your arse in the recording bizz. My aim is towards a production facility for (non-gigantic) commercial spots and miscellaneous voice/record/tape put-togethers, etc., using maybe three halftrack machines, two tables, some dbx noise reduction, compression, and equalization, (all good for stereo), and a board I may have to build myself in order to get input channels that will handle stereo on each pot. I know, I can get one from Ward/Beck for an exorbitant amount of money, but that'll have to wait till I get past paying rent, etc. Any ideas? A TEAC Mod 3 or 5 would be ideal if they had stereo inputs, but nobody (including TEAC) knows about that except folx that make broadcast boards (with all the flexibility of a packaged home stereo.) I am trying to collect all this piecemeal to be available at retirement (13 yrs, 2 months, 6 days away) so I can go out enjoying myself. So, the get-$mart artcles are valuable for rubbing my nose in the practical $ide of it that I don't dig.

I do get out with my Technics RS-1520 recorder occasionally for a field job, and while my Gately Prokit mixer

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is usually needed for middle mic, tone generation, slating, etc., it would be convenient to be able to go right into the 1520 with mic lines, but the mic inputs are unbalanced, and besides my C-451s need phantom power. OK. No sweat. I just add mic input transformers in/ or out/board.

OK, there's a compliment, and some bs. Now here's a gripe: I wish you would get off this trip(e) of reviewing every cassette deck that comes along and stick to stuff of a more (at least semi-) professional level. I could really have used a head-to-head comparison of quality reel machines when I bought my 1520. How about doing stuff on GOOD turntables, parametric equalizers, monitor speakers, microphones, etc. Something on the TEAC 20 series mixers could be useful too, but a goddam cassette deck every month or two? PULEEZE!

Thanks.

—Earl D. McDonald
Dallas, TX

Well, we can't promise that we'll be curtailing those reports on the goddamned cassette decks, but thanks for the strong opinions. It's good to get feedback. And thanks for the compliments.

A Warm Story

I would like to take the opportunity to thank you for printing my article "Track Tips" in the July 1982 issue. I would greatly appreciate any "feedback" from owners of 8-track studios who have other ideas on how to attain "Wall of Sound" recordings on basic 8-track gear. If there is anybody out there who has comparable ideas, but maybe a different technique, let the readers know and maybe we can collaborate on a future article entitled "Advanced Techniques for Home Eight Track Studios." Remember, as the saying goes, two tape heads are better than one!

What I would like to thank you for now is probably the biggest "Thank you" that I will ever give to a magazine in the course of my life. About two years ago, MR&M printed the name and address of a gentleman from Holland who owned and operated a small 8-track Tascam studio. He wanted to know if there was anyone in the states (N.Y., N.J. area) that would exchange tapes as a "penpal" and share different ideas on recording. We exchanged tapes and letters and many different recording ideas getting to know each other as friends more and more as each letter or tape would arrive. Success was, all of a sudden, rampant for this man from Holland as his studio rapidly changed from eight tracks to sixteen tracks, and now twenty-four tracks. The reason for his flourishing success was because the music this gentleman was producing always seemed to get better and better with each cassette sample that I could get my hands on! He is now so popular with his new 24 track studio that he is booked solid for many months in advance, recording only Holland's biggest groups. He is also working on some projects for a big international record company.

On the 22nd of July, a day I will never forget, Jaap Brunner and his lovely wife Anneke from the Ballad Sound Studios, Gorkum, Holland, came to visit, and we were so elated to get together with them that the hours flew by and I was a kid all over again with "show and tell." Jaap is not only one of Europe's better engineers, but what's more important is that is is a great friend indeed! Now, my wife and I can go on a trip to Holland and visit some good friends, thanks to your superior magazine! I might return and write a well documented article on "Recording in Europe." Thank you very much again for helping me become friends with a truly superb sound expert!

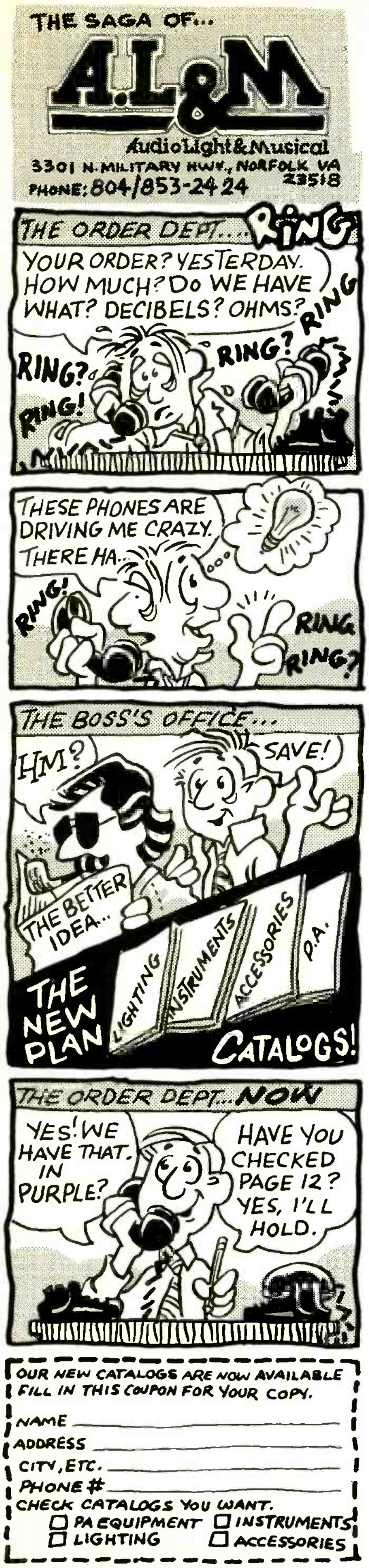
—Marc William Fallon
The Bedroom Studio
Teaneck, N.J.

The Handbook

What is the current price of *The Recording Studio Handbook* by John Woram, and where may it be purchased?

—Richard Weston
Scarborough, Australia

The current price of The Recording Studio Handbook *is $37.50. You may obtain this book by sending your check to ELAR Publishing Company, Inc., 1120 Old Country Road, Plainview, New York 11803.*



The Trap

Could you give me a short explanation of a bias trap? I think I know what it is, but I'm afraid I might have some misconceptions.

—John D. Minkoff
Lawrence, N.Y.

We base our explanation on a short section appearing in John Woram's book, "The Recording Studio Handbook." He explains that the point of the bias trap is to prevent the bias signal from overloading record and playback electronics. The trap itself is a high frequency cut-off filter. Woram points out that the bias frequency is beyond audio limits, but that its high amplitude could overload an audio amplifier and thereby distort the output of the amplifier. Bias traps can be put in two places—after the record amplifier and before the playback amplifier. The one placed after the record amplifier keeps the bias signal from backing up and overloading the record amplifier output, and the one placed before the playback amplifier filters out bias in the playback line.

In Defense of Digital

The following letter was directed to the attention of Todd Rundgren:

I am writing in regard to your article in the April 1982 issue of Modern Recording & Music in which you stated your lack of interest in digital recording.

Because of my great respect for your work over the years, I was disturbed when I read the paragraph in which you told of "weird experiences" people had using digital equipment. After working at Soundworks Digital Audio/Video Recording Studios on digital projects such as Stevie Wonder, David Sanborn, John Denver and currently Steely Dan, I have to disagree with the negative opinions expressed about digital by many misinformed people.

Soundworks 32 Track 3M Digital Mastering System has been easier to work with than multi-track analog machines. With proper pre-session maintenance (about 1 hour each day) there has been no down time for the client. Here are some of the technical pluses of the 3M system: no tape hiss; perfect reproduction of the input signal; non measurable wow and flutter; a 10 Msec crossfade during punch-in to assure a smooth punch; perfect replica transfers for safety masters; digital editing which allows you to preview edit point as many times as needed before making the actual edit.

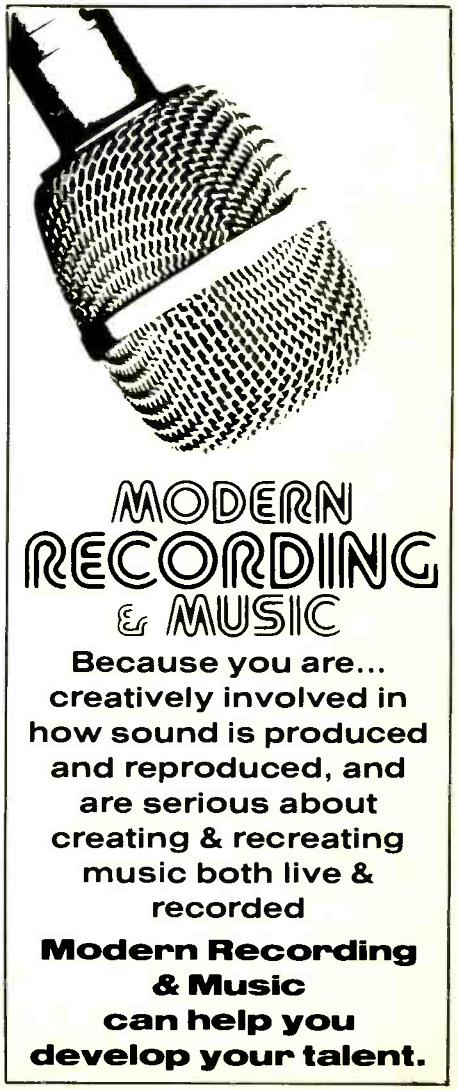
In regard to the "green stuff" it is nothing more than a small two inch roll of green fabric called tape wipe, which makes light contact with the tape at all times and advances when entering play mode. Its purpose is to remove loose oxide and dust particles, which otherwise would build up on the tape and cause dropouts. When properly threaded, the tape wipe can be left alone until the roll runs out (about every 6 hours).

Seeing is believing, so the next time you're in Manhattan, please feel free to call and stop by for a demonstration.

One last point...50% of Soundworks business is video related and 100% is done on digital recording machines.

Thanks for your time.

—WAYNE YURGELUN
Chief Maintenance Engineer
Soundworks Digital Audio/
Video Recording Studios, Ltd.
New York, NY



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Speaker Search

In your October issue of *MR&M*, you had a small article about the P.A. system Jackson Browne was using for his 1980 "Hold Out" tour. I believe the article was in the Musical Newsicals sections of your magazine.

The article had stated the name of the company that had manufactured the speaker cabinets and, also the name of the company that was distributing them in the United States. I heard Jackson Browne perform at "Summerfest" on July 30, 1980 in Milwaukee, Wisconsin. He used these particular speakers that night in concert.

Could you please help? I have lost my October 1980 issue. Would you please send me a copy of the article or a name and address so that I may obtain more information about these speaker cabinets? I would like to build or purchase a pair.

Keep up the good work in your magazine.

—John Spaeth
Oconomowoc, WI

The article you refer to was actually a letter that appeared in the October 1980 issue, entitled "Not For Sale." The speaker system you asked about is manufactured by Audio Analysts. Their address is:

943 Montee de Liesse
St. Laurent, Quebec,
Canada H4T1R2

Their phone number is 514-735-5557.

The only other is Clair Brothers. Their address is:

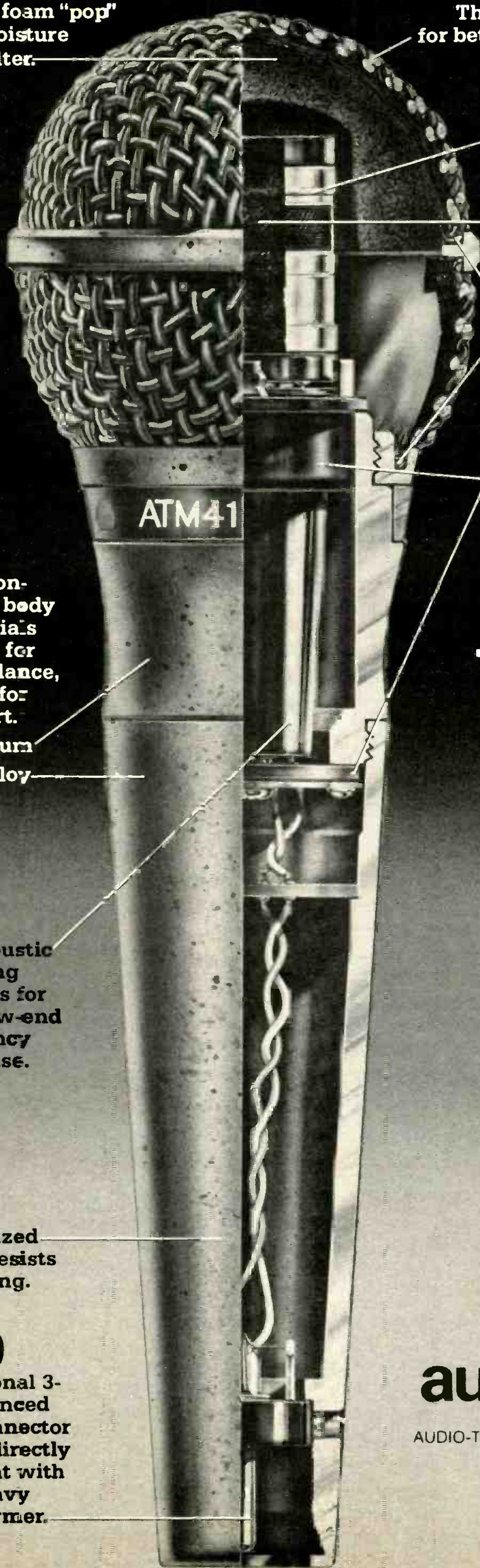
P.O. Box 396
Lititz, Pennsylvania 17543

Their phone number is 717-733-1211.

Good luck getting in touch with either of these companies.

Who Took the Pictures?

We neglected to list photo credits in our August issue, and so we are trying to make up for that now with apologies and the information. Pictures of Blondie were courtesy of Harold Bloom Organization. The pictures for "Direct to Disc All-Stars" were taken by Robert Pierce Mercer. We won't let it happen again.



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TALK BACK

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

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

Ring Around the Collar?

I have a Teac 80-8 and I am having a problem with heads, capstan, etc. gunking up constantly. I have to clean the tape path every 15 minutes and a lot of crud comes off each time. I use Ampex 456 tape and I recently purchased new tape, thinking that would cure my problems, but I am still having the same trouble. I have a Teac 2340 sitting right next to the 80-8 and that doesn't gunk up at all, so I'm ruling out the environment, (dust, dry air, etc.). However, I do use Maxell tape for the 2340. Also, if I'm recording on one track, let's say track 5, over and over, trying to get a part just right, after 4 or 5 passes a thin dark ring forms around the capstan, obviously from track 5. Something in the record process is gunking things up. Any suggestions?

—Dennis L. Chamberlain
Cuba, N.Y.

It is odd indeed, that many rolls of 456 in its ½-inch incarnation have prompted questions from many 80-8 users. In tests here at Teac, we have noticed what you have—that the ½-inch tape seems to shed and foul the tape path while identical machines using other tape widths (40-4's) seem somehow immune. We *can* rule out any peculiarity of the 80-8, however, since 1: the parts in the tape paths of the 40-4 and 3440 are made of the same materials and in terms of machining are also the same geometry as the 80-8, and 2: all the complaints we receive on tape-shed problems come in distinct waves from the field. We don't get complaints about the quarter-inch 456 and only occasional complaints about the 1-inch, and in our many frantic discussions with Ampex, we have been mutually unable to pin down a cause.

The thin ring of oxide on your capstan that looks like track 5, is just that—track 5. When you notice a roll of tape doing this, you have bad binder composition and you are advised to contact Ampex. Ampex has told us (Teac) that tapes like this will gladly be exchanged for new so that they may study the possible causes of binder failure. Of course, you should not wait until your tape is a nearly finished master, with only a few thousand passes left to undergo overdubs, and if this is the case, you should hire studio musicians who can play the parts in one take (humor).

There are cleaning products for tape oxide shed, like Scotch 630 cleaning fabric, which can be used to gently clean off loose oxide as the tape is fast wound through a piece of it. This will greatly reduce the amount of oxide that accumulates on the recorder's metal guides and heads. Also beware of the tapehead cleaners you use: avoid toluene, it does not completely evaporate from the tape path even when it appears to have dried. Fumes linger, clinging to metal parts and can attack the binder chemistry as the tape passes until enough tape travel removes the fumes. Avoid alcohol as a cleaner, it has two nasty properties: it removes all the lubricant left by tape on metal tape-path parts, and it can leave water condensation behind, which is not good for the sensitive tape binder chemistry. Many people have commented that head cleaners and other solutions that contain silicone oil seem to reduce the effects of tape oxide abrasion. Several Tascam studios have switched to SP stainless polish because they ran out of head cleaner, and have noticed a reduction in head wear—some say as much as 4 to 1—we are investigating. SP is the same chemical, Trichlorotrifluroethane 1, 1, 1, Trichloroethane, with silicone added.

Carefully check the temperature of your 80-8's tape path to determine if the tape itself is being abnormally heated. Above room temperature operation can accelerate shedding. If your room layout dictates poor ventilation around the recorder, you can place a whisper fan nearby to cool the tape.

Finally, we have determined no outstanding differences in batch reliability between any manufacturer's tapes among the premium tapes we have tested, but all brands of tape do exhibit changes in binder integrity from batch to batch. If you have time, it's a good idea to play a new roll of tape you intend to use for multitrack mastering, from end to end, even a couple of times to check for abnormal binder integrity.

Remember, as with any technology, the more sophisticated the user's requirements are, the more care that must be taken with details. This annoying detail can mean the difference between good and bad masters when many passes must be made for overdubs. Since the 80-8 is so simple to recalibrate, you might consider trying another brand of tape for a while and then going back to 456. Scotch 226 is such a close match in bias requirement, that a roll can be set up from the settings for 456 in about 5 minutes. In a non-critical session, 226 can even be put onto a machine calibrated for 456, with no adjustments, and sound just fine. Always *play* your master tapes off onto the takeup reel for storage. The tight pack this provides will help preclude atmospheric damage, and oxidation of the binder. Keep the tape storage area dry and cool—below 80°—and avoid any unnecessary playings of the tape once tracks have been recorded onto it.

—Drew Daniels
—Applications Engineer
Tascam Production Products
Teac Corp. of America
Montebello, Ca.

Get It Right The First Time

I presently own a small "bedroom studio" consisting of a Teac A3440 plus dbx, a Teac A3340, a Teac 2A mixer, a dbx 118 variable compressor/expander along with various mics and signal processors. This is combined with a fairly good playback system comprising a large Pioneer amp and four JBL 4311 monitors. The problem is that I need a ½-track deck for mastering and need to know how possible (and expensive) it would be to convert an existing Teac A6300 ¼-track low-speed deck into the likes of a Teac A6100 high-speed ½-track deck. I should mention at this point that these two decks look almost identical except for the capstan width and logic controls. Also I already own an A6300 and it's in perfect condition.

Up until recently I mastered all my recordings onto one of the 4-track machines (tracks 1 and 2, left; tracks 3 and 4, right) and it seemed to be good enough for playback on half-track machines for both demo and even broadcasting purposes. However, I recently recorded a few songs to be pressed into an album (only 500 copies) for a local artist and have been advised by an engineer friend that the master must be half-track or there may be phasing problems in the final product.

Assuming that my tape decks are aligned properly, should I really go back and remix everything onto a half-track, or is my friend being too critical, considering that this is a small, low-cost production?

**—Chris Broadhurst
Brodieville, Bermuda**

Phasing problems might result in the mastering process if you don't give the cutting engineer tones on your master tape which he can use to align his head azimuth to your tape, but it is unlikely that any phasing problems would occur as a function of using multiple tracks—in this case 4—to simulate a half-track track format. The biggest problem is a phenomenon called "fringing." Fringing is the tendency of a larger track width head to "look at" smaller track width magnetic tape tracks differently at long and short wavelengths (low and high frequency sounds, respectively), and produce an output that is not equal to the input in terms of voltage versus frequency, or what engineers call "flat." You would probably notice a change in the bass region of your recorded sounds, which might prompt the cutting engineer to reach for his equalizer, but the nature of the change (most likely a boost) is not something the typical studio type equalizer would be able to correct easily, since the boost in low frequencies is proportional to the wavelengths and track spacing geometry.

Of course, it is possible to modify a quarter-track, low speed deck—and in the past many models from different manufacturers were simply modifications like this—but you will not get the desired results from changing speed alone, since the recording and playback amplifiers in the deck employ equalization to achieve a flat frequency response recording based on tape speed and must be modified also to reflect the change in speed. Any competent tape recorder technician should be able to locate and replace the necessary components in the record and play amplifiers to facilitate the EQ change, and to adjust the bias current to compensate for the new wider track width heads and their concomitant current requirement.

My personal feeling about making records is that if it's worth making plastic from, it's worth doing properly. If you make a good record at the beginning, folks might ask you to make more later on. If the audio quality isn't that important, I suggest you make up cassette copies to distribute—it might be cheaper in the long run, and with cassettes, you'll have an excuse!

—Drew Daniels
Applications Engineers
Tascam Production Products
Teac Corp. of America
Montebello, Ca.

Compliments and Confusion

Congratulations for a consistently fine and perceptive publication. Thank you also to Mr. Harold Cohen of dbx, Inc., for his information about dbx I and II (see TalkBack, "Playing the Numbers," July 1982 issue, pages 16 and 17), something that had been confusing to me for many years. Now the differences are perfectly clear. What is not

clear now is what is the "knee effect"?

—Louis J. Rankin
Executive Manager
Supersound USA
Eagle Rock, Ca.

Hmmm. Louis, your question was just a bit confusing. Indeed, it sent us back to our books to see if we had missed something. (Does this require an answer from dbx? Did Harold Cohen inadvertently leave something—"knee effect"— unanswered or ill-defined in the July issue?—although we seriously doubted this last thought). After deciding that our confusion was due to the lack of a segue of sorts between your thoughts, we decided to forge ahead and hope the following information will answer your question.

Contribution Editor Craig Anderton defined "knee effect" to us on the telephone as being a characteristic or property of limiters, literally the way *a limiter limits over the signal range. Knee effect can further be defined as being 'hard' or 'soft.' A 'hard knee' limiter instantly clamps down on the signal, whereas a limiter with 'soft knee' gradually limits the signal.*

If you are interested in dbx's compressors and limiters, and what knee effect has to do with them, *check out Harold Cohen's reply in "TLC for Comp/Limiter" in the May 1981 TalkBack column.*

P.S. In all our confusion we forgot to thank you for the compliment—Thanks!

A Song and Dance Mic?

A year ago I bought a Beyer M-360 microphone for a song and a dance. I have written to Beyer twice for technical information concerning this mic, but to date I haven't received anything from them. The mic is in perfect condition, but I'd still like to have its statistics (directional pattern, etc.).

—Bill Montillo, Jr.
Warwick, R.I.

In response to your inquiry regarding the Beyer Dynamic Model M 360 microphone. Here is the pertinent information regarding both the application and specifications of the mic:

The Beyer Dynamic Model M 360 microphone incorporates a double ribbon element designed specifically for speech and vocal applications. The M 360 features a vocal-music switch. In the music position (out) the full frequency response of 30-20 KHz is realized. In the vocal position (in) the frequency response is attenuated 10 dB at 50 Hz. The frequency response of 30-20 KHz is measured at +2.5 db. The polar pattern of the M 360 is cardioid, with a side attenuation 180° off axis—25 db measured at 125 Hz.

The sensitivity of the M 360 is 0.14 mV/u bar—77 db (reference 0 db = 1 V/u bar).

The impedance of the M 360 is 200 ohms in the music position or 50 ohms in the speech position.

The M 360 can be used with many instruments (with the strong exception of percussion) or even for certain sound reinforcement applications. However, as mentioned, its primary application is for vocal use.

I hope this will cover all areas of concern regarding the Beyer Dynamic model M 360. If I can be of any further assistance, please feel free to call.

—Paul Murphy
General Manager
Beyer Dynamic, Inc.
Hicksville, N.Y.

Pulling the Rug Out from Under

I have an audio production studio and I had a free-standing booth built in my control room. It's of staggered stud construction and is *real* quiet except for low frequencies. The problem seems to be that the three layers of rugs that are between the bottom of the booth and the floor of the control room act to "couple" the booth to the room, i.e., footsteps transmit into the booth.

I want to jack up the booth and replace the rugs with "acoustical isolators." Where do I find these pads and how do I determine which ones to use? (If you have, by any chance, answered this question already in Talkback, please let me know the issue.)

—Arklay King
Silver Linings Audio Production, Inc.
Boston, Mass.

[*We haven't, Arklay, so here goes:*]

Herb Schwartz, of John Storyk and Associates in Manhattan, who has proved so helpful in the past (Check out, if you have them, Herb's TalkBack responses in the January 1980, May 1980 and September 1981 issues; even if the questions don't apply specifically to your situation, the information imparted is valuable just to have.) *had this solution at the ready as well. The "acoustical isolators" you speak of he further described as "neoprene pucks." These you would glue to a sheet of plywood which you would then slip* *under the booth. This would raise it off the ground a good solid inch, and the airspace could be filled with insulation for even better separation. This should quiet those thundering footsteps once and for all!*

The acoustical isolators mentioned above are available for $2.25 each, prepaid, from Mason Berger East (a division of Mason Industries of Hauppauge, New York). Request stock number EAFM 6534 and mail your order to Mason Berger East, P.O. Box 356, Floral Park, New York 11002, and they will be more than happy to fill your order.

Shielding for a Small Home Studio

In my small home studio, I am running my Micromoog synthesizer's +12 dBm output through a DeArmond volume pedal to a Sansui AX-7 mixer. I've been picking up unwanted signals from a local AM radio station through my recording system which are appearing on my tapes. I've isolated the pickup point to my volume pedal. Can I shield the pedal, or ground it? I've already tried lining it with aluminum foil, with no effect. The pedal housing is all metal although I'm not sure what kind. The design is an older one; the date on the inside of the pedal is 1974.

—Greg Smith
Marshfield, Wisc.

Shielding is the answer, Greg. Indeed, better shielding would probably eliminate your problem; the essential question is what to shield. A call to Steve Tosh, President of DeArmond in Toledo, Ohio, confirms this. While it may appear that it is the pedal that's picking up those local transmissions, look closer at those cables snaking their way across the studio. Chances are (Steve says they've encountered this situation many times before) that the cable you are using is what is acting as the receiver. Many "shielded" cables only offer 70%-80% shielding; good quality shielded cable will offer you at least 95% shielding. This difference could mean much quieter tapes to you! While Steve doesn't know what cables you are currently using, he suggests the standard that DeArmond uses, and one which will give you that 95% shielding. It is Belden 82-18 shielded cable with a braided shield. Before you go any further, re-run those lines, and see if "WXYZ" doesn't experience permanent radio silence in your studio.

THE PRODUCT SCENE

By Norman Eisenberg

EQ COMBINED WITH SPACE EXPANDER

A ten-band graphic equalizer is combined with a spatial expander in the Yamaha GE-5. The EQ section, which handles both stereo channels simultaneously, may be used for recording or for playback. Sliders provide ±10 dB adjustment on ten octave frequency centers from 30 Hz to 16 kHz. The expander is suggested for use in both playback of recorded material and in "live" performances. Price of the unit is $245.

CIRCLE 50 ON READER SERVICE CARD

NAKAMICHI HEADPHONES

The SP-7 headphones from Nakamichi employ a novel driving system that uses a powerful ferrite ring magnet, oversize voice-coil and a large-diameter low-mass polyester diaphragm. The diaphragm is formed with a "tangential edge" for high compliance and good linearity. A newly developed earpad is credited with reducing variations in "subjective response" while also providing a comfortable fit. Offered for critical listening, including monitoring, the SP-7 weighs 150 grams (5.25 ounces) and lists for $70.

CIRCLE 51 ON READER SERVICE CARD

DOLBY-B AND C IN AKAI DECK

Akai's new top-of-the-line cassette deck is the model GX-F91 which includes both Dolby-B and Dolby-C noise reduction. A three-head recorder, the Akai uses a two-motor full logic system with quartz-lock direct drive and a double capstan closed-loop transport. The built-in microcomputer automatically sets EQ, sensitivity and bias levels with four memories. Tuning is performed over a series of 64 steps, each step equal to 0.26 dB. Other features include an automatic fader control which will bring the desired recording level up at the beginning of a piece and then fade out at the end. Also offered is "Introscan" which will play the first ten seconds of each selection to provide a quick review of a tape and the location of specific sections. The instant program location system (IPLS) may be used to locate the beginning of any selection in fast-forward or in reverse modes. The front panel has a flip-down front which covers the less-often used controls, but which also slides down and under the deck to reveal the solenoid controls on the inside of the panel. Meters are peak and VU switchable. The counter is a digital real-time type. Line and headphone outputs are adjustable. Dynamic range compensation is built-in, and input and output terminals are gold-plated. Price is $750.

CIRCLE 54 ON READER SERVICE CARD

HAFLER DEBUTS EQUALIZER

A stereo graphic equalizer, the model DH-160 from David Hafler Co., is available in kit-form or factory-wired. The device offers graphic equalization on each of its two stereo channels of ±12 dB on frequency centers at octave intervals from 32 Hz to 16 kHz. In addition to the ten sliders for each side, the DH-160 has separate sliders for level control on each channel as well as switches for monitor, record, bypass and power off/on. The meter display range covers the range from -20 dB to +3 dB, with the 0-dB level adjustable via a rear-panel control. Side brackets permit rack-mounting. Mic and line inputs are adjustable. Assembled, the device costs $399; in kit form, the price is $299.

CIRCLE 55 ON READER SERVICE CARD

HITACHI "DUAL" CASSETTE DECK

Hitachi's D-W700 is a "dual" cassette deck with one section for playback and the other for record and playback, so that programs may be dubbed from the former to the latter. Microcomputer control facilitates random program dubbing as well as random programming playback. The automatic dubbing option allows the recordist to choose selections on the original tape by number and record them in any order onto a new tape. The random playback feature sets up a program with the number of each selection keyed in for playback in any order, and for repeats. The D-W700 also includes both Dolby-B and Dolby-C noise-reduction and a three-switch tape-selector. Price is $390.

CIRCLE 56 ON READER SERVICE CARD

TEAC DECK USES DBX

Heading the line of new Teac cassette decks is the V-2RX which incorporates dbx noise-reduction as well as Dolby-B. A three-head deck, the V-2RX is powered by two motors and has block-repeat, memory stop and memory play features. Also built-in is a parametric EQ switch designed to tune in on a specific portion of the response to make EQ changes. This feature is said to be especially useful when working with frequencies from 60 to 500 Hz which can be varied by as much as ±10 dB. A pre-post switch is provided. Other features of the deck include fine-bias adjustment and pro-type recording level sliders. Level indicators are 14-segment peak-reading fluorescent displays, and the tape counter shows 4-digit electronic readout. Price of the V-2RX is $520. A remote-control accessory, the RC-90, is optional.

CIRCLE 57 ON READER SERVICE CARD

ADCOM AMP, SPEAKER SELECTOR

New from Adcom is the GFA-2 power amp which features dual power supplies for its two channels, plus a multistage protection system including safeguards for speakers, transient elimination in turn-on and turn-off and peak-current limiting with thermal sensing. Rated for 100 watts per channel (8-ohm loads) at less than 0.05 percent THD, the GFA-2 has front-panel LEDs that show peak power output. Rack-mountable, the amp costs $360.

Adcom also is showing its GFS-1 speaker selector, an outboard accessory that can accommodate up to three pairs of speakers, with switching to allow selection of one, two or all three pairs separately or simultaneously. In addition, headphones may be jacked into the device. With bridged amplifiers, (such as Adcom's GFA-1A), a switch on the selector box may be activated so that the user will not "common ground" the amplifier outputs. A built-in protection circuit also enables the playing of three sets of 4-ohm speakers at the same time without danger of abuse to the amplifier. Price of the GFS-1 is $89.

CIRCLE 58 ON READER SERVICE CARD

BSR UPDATES ITS EQUALIZERS

Featuring what BSR calls a new streamlined design are four new equalizers, all with enhanced specifications vis-a-vis former models, including S/N ratio, THD and IM. Top model is the SS30 which features a built-in spectrum analyzer and pink-noise generator with calibrated microphone. This unit provides ten bands of control on each of two channels; LED slide position indicators; subsonic filter and two-way tape dubbing. The ten frequency centers are spaced an octave apart from 31 Hz to 16 kHz. Price is $400.

The SS20 offers 12 bands of EQ on each of its two channels, also with two-way tape dubbing, subsonic filter and a dual LED seven-segment-per channel dB meter. Price is $330.

The SS10 provides 10 bands per channel and one-way tape dubbing. It has the 7-segment LED metering but lacks the subsonic filter. Price is $230.

The SS5 is a 5-band per-channel unit with no metering or subsonic filter and no tape-dubbing feature. Price is $130.

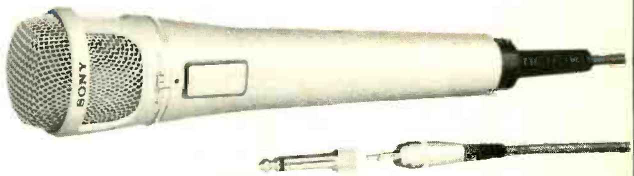
CIRCLE 59 ON READER SERVICE CARD

CHANNEL MASTER EARTH STATION

In addition to its new dish antennas, Channel Master has announced the model 6128 earth station receiver, a 24-channel synthesized unit. The receiver uses a single-conversion downconverter installed at the dish, and a receiver unit located indoors. The 24-channel number format is shown on the receiver's LED digital channel display. Channels are power-selected by pushbuttons, aided by a fine-tune knob with metering. Audio channels (6.8 and 6.2 MHz) are selected on two priority audio buttons. Additional audio channels can be manually tuned. Automatic polarity switching allows one-button selection of any channel without additional polarity adjustments. Capable of receiving normal or inverted video signals, the set has a built-in modulator which eliminates the need for a separate modulator. An optional model 6192 remote-control accessory provides remote channel selection and fine tuning.

CIRCLE 60 ON READER SERVICE CARD

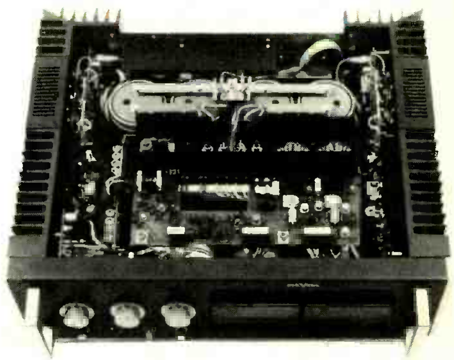
SONY MIC PRODUCES ECHO EFFECT



New addition to Sony's line of "Unimatch" microphones is the model F-V7ET "Echo Mic," a dynamic cardioid vocal microphone which contains active circuitry that produces vibrato and echo effects. Switching activates the echo and also controls the speed of the effect. A high-impedance mic with rated response of 100 Hz to 12 kHz, it weighs 6.7 ounces. Complete with stand, it costs $65.

CIRCLE 61 ON READER SERVICE CARD

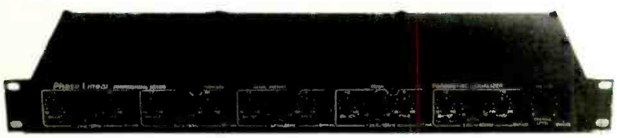
REVOX POWER AMP



The Revox B740 power amplifier, said to be "designed to meet stringent professional standards for sonic performance and long-term reliability," uses internal construction and circuitry similar to the A68. New feature on the B740 is a "current inrush limiter" designed to prevent overload as line current pours into the unit's beefy (60,000-microfarad buffer capacity) power supply. Panel features include calibrated level controls in 3-dB steps; large peak program meters; headphone output; on-off switching for A and B speakers. Both XLR and RCA inputs are provided. Conservatively rated at 100 watts per channel into 8-ohm loads with no more than 0.04% THD, the Revox B740 lists for $2,299.

CIRCLE 62 ON READER SERVICE CARD

PARAMETRIC EQ BY PHASE LINEAR



Said to be the first five-band parametric equalizer to be introduced in a 1¾-inch rack-space size is the Phase Linear model E51. Key feature of the new unit is the option of switchable peak or shelf response on bands 1 and 5. In addition, the E51 offers automatic balanced/unbalanced XLR ¼-inch phone inputs/outputs; system gain of +20 dB for low-level sources; overall level control and bypass switch with LED; signal present, power ready and system overload LEDs; output relay control; and line drivers. Price is $549.

CIRCLE 63 ON READER SERVICE CARD

PRO VOCAL MIC FROM PHILIPS

Designed for professional entertainers is the new Philips 7401 dynamic cardioid microphone. A ball-head type, the 7401 has an internal, locking off/on switch and is available with a detachable cable in any one of three variations: the 7401E with XLR-XLR cable; the 7401P with XLR-to-phone plug; and the 7401T with XLR-to-HI-Z phone plug. Characteristics include an extremely tight cardioid pattern and a response that is described as being "fine-tuned to the exacting requirements of a professional singing voice."

CIRCLE 64 ON READER SERVICE CARD

CES—SUBWOOFERS AND NOISE REDUCTION

There was, of course, a lot to see and hear at the monster Consumer Electronics Show in Chicago this past June. Two things in particular impressed me from the standpoint of the pro or semi-pro sound person.

One is the increasing evidence of a bow in the direction of the subwoofer...not just for "more" bass but for clean bass and, as a concomitant, clean everything-else in the sound. This applies, I feel, whatever the kind of music and whether it is playback over a studio monitor, or in sound-reinforcement and stage speaker applications, or just listening for pleasure.

Of course, a condition of using a subwoofer is the need to bi-amplify, and perhaps this technique is as relevant as the subwoofer itself. Bi-amping does accomplish some specific benefits. For one thing it eliminates IM distortion—there's no deep bass to interfere with the sound from the "main" speakers. These, in turn, become easier to design for smooth response above, say, 200 Hz, and for octave-to-octave spectral balance.

I heard these effects on two widely different and widely varying-in-cost systems—one was the Fried monitor setup with subwoofers on each channel plus midrange and tweeters on each channel ($4,000). The other was a Phase Tech system using two PC 60 bookshelf systems and a PC 50 subwoofer. The latter is a double voice-coil woofer which handles the lows on each channel within one box. Total system cost is only $500.

The other item worth mentioning to *MR&M* readers is that the whole anti-noise race seems to have settled down to Dolby versus dbx. Dolby of course has evolved to Dolby C with a sort of side-excursion into HX which, in the "Professional HX" variation, is espoused most prominently by Bang & Olufsen. The C version itself is showing up in more and more cassette decks.

As for dbx, it now has some twenty equipment manufacturers using it, and the dbx "chip" is going into some car stereo brands as well as into in-house brands such as Teac, Technics, Yamaha, Kenwood, Matsushita and Marantz. And of course there's the growing library of both discs and cassettes in with dbx-encoding.

As for Columbia's CX system, there's some more hardware but still no significant number of CX albums. And, in any event, CX—unlike dbx or Dolby—is really intended for disc reproduction and not for use with tape in either recording or playback.

So, for the serious recordist, it seems to be Dolby versus dbx. Stay tuned in.

MUSICAL NEWSICALS

DRUM MACHINES

One of the most popular brands of drum synthesizers has been the Roland CompuRhythm line. RolandCorp US has announced the introduction of two new, updated models to replace the popular CR-68 and CR-78 units. The new models are designated the CR-5000 and CR-8000, and are oriented toward "live" performance use while retaining many of the same realistic drum sound simulations found on Roland's Rhythm Composer. Both models have twenty-four preset rhythms which can be altered via the Arranger section by adding cymbal, hi-hat, conga or shuffle variations. The individual drum sounds may be independently adjusted for level to further tailor the sound of the pattern. Additional functions include an Intro/Fill In mode which inserts one of eight fill in patterns manually or automatically every 2, 4, 8, 12 or 16 bars, and a Crash function which automatically inserts a cymbal crash on the first beat following a fill in roll. In addition to the twenty-four preset patterns, the CompuRhythm units have eight rhythm patterns and four fills which are user-programmable in a step-by-step programming method.

CIRCLE 80 ON READER SERVICE CARD

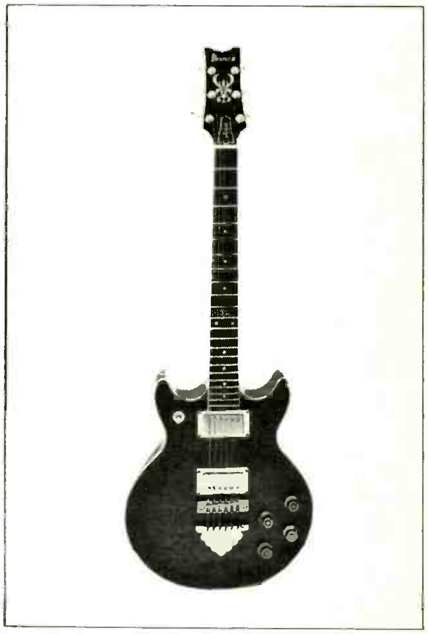
SYNTHESIZERS

News comes from RolandCorp US of the Juno-6 Polyphonic synthesizer, an inexpensive polyphonic keyboard with full six-voice capability. The unit uses a new Digitally-Controlled Oscillator (DCO) which functions the same as a VCO but with much improved stability and repeatability, and has new functions such as an arpeggiator (with variable rate and mode), and a digitally controlled transpose function. The DCO features Pulse and Sawtooth waveforms plus a square sub-oscillator waveform which may be combined simultaneously; LFO modulation is available as is pulse-width modulation via manual wheel or envelope generator. Other functional blocks include an LFO with rate and delay time controls, a VCA with four-part (ADSR) envelope generator and a chorus effect similar to Roland's famous chorus units. Left-hand performance controls include a return-to-center bend wheel which effects either or both the DCO and VCF, with individual range sliders for each, a modulation pad to bring up the LFO and a three-position octave transposer switch.

CIRCLE 66 ON READER SERVICE CARD

GUITARS AND BASSES

Ibanez has announced two new guitar models—one an addition to its Artist series and one added to the Musician series lineup. The Ibanez Artist AR105 features a burl mahogany top for a unique, highly figured grain pattern. Beneath the burl top one finds a mahogany body and a three-piece maple neck. The body shape features a smooth heel for comfortable access to



the higher frets. On the hardware side, the AR105 uses VelveTune tuning machines with a 17:1 ratio, and a Gibraltar bridge and tailpiece. Pickups for the new model are a pair of Super 58 pickups with Alnico III magnets for maximum output but with a warm, pleasing sound. The other new model is the Musician MC150, a solid body, twin pickup electric guitar. The MC150 has a maple neck which runs full length through the mahogany body. Hardware includes gold-plated Smooth Tuner machines, and a Gibraltar locking bridge. Pickups are the same Super 58 pickups but wired for either humbucking or single-coil operation via Ibanez' Duo-Sound push pots which switch between modes with a push of the top of the pots. The MC150 is available in dark stain, fire red and Ibanez' latest color option, Polar White, a custom finish which picks up and reflects stage light in a unique way.

CIRCLE 70 ON READER SERVICE CARD

St. Louis Music Supply Co. is celebrating its 60th anniversary in the music industry, and has announced the introduction of two special, commemorative instrument models as additions to its exclusive Electra line. The first of these new models is the X935 six-string electric guitar which has been added to the Endorser series of Electra's Vulcan line. This new model was designed to be a straightforward, no frills guitar incorporating numerous suggestions from dealers and performers. Construction of the X935 includes a one piece mahogany body with three piece, carved maple, arched top and a five-piece, laminated, hard rock maple neck mounted to the body via an exclusive "bayonet" technique. On the hardware front, the X935 features two double-coil Magnaflux pickups, brass neck nut and bridge for sustain and chrome plated 14:1 machine heads. The other new model from Electra is the X635 Phoenix bass guitar, which features a solid wood body laminated from Canadian ash, maple and walnut. The neck of the X635 is fashioned of Canadian maple and has a rosewood fingerboard and a brass plate on the back of the headstock bearing the production number of the instrument and indicating the limited edition nature of the model. The X635 also features a custom brass bridge with through-the-body string anchoring for increased sustain.

CIRCLE 71 ON READER SERVICE CARD

Phase Systems is noted for its D'Mini series of scaled down guitars with familiar shapes. Two series are offered, Series II which are two-thirds normal size and Series III which are three-quarter size instruments. The newer, Series III models range between 31" and 34½" in overall length, and are offered in three familiar shapes: Les Paul, Strat and Flying V. Scale length for Series III is 19¼", with 22-fret necks, versus 16¾" scale length and 17 frets for Series II. Both the Les Paul and Flying V versions have a single pickup, while the Strat model has two single-coil pickups with a three-way switch and single volume and tone control. Also new from Phased Systems is the D'Mini Bass, a 36" long axe style after the Precision bass. The D'Mini Bass has 21 frets based on a 25½" scale, and has a single pickup for the same split/staggered style as the P-bass.

CIRCLE 72 ON READER SERVICE CARD

Ovation Instruments has recognized the growing trend in the guitar industry toward collecting instruments as a hobby, and the generally increased interest in limited edition models, and has introduced the first of a new series of Collector's Series guitars. The new model is designated the 1982-8, and is distinguished from standard Ovation guitars in both features and cosmetics. On the features side, the 1982-8 has the same built-in stereo electronics package as the top-of-the-line Adamas models, while on the cosmetics side, the instrument combines a blue sunburst top and dark blue peghead face with a natural finish neck. The package is rounded out with an abalone rosette, abalone inlays and 24K gold-plated tuning machines with pearloid buttons. This Ovation model will only be made through the summer of 1982, and each guitar will come with a special label in the bowl, and a serially numbered Certificate of Authenticity signed by Bill Kaman, the inventor of Ovation guitars.

CIRCLE 73 ON READER SERVICE CARD

"Vintage" and "classic" guitars—reissues or copies of early electric guitar models—have been increasingly popular within the last several years. Fender, which was, of course, one of the companies which pioneered electric guitars in the first place, has entered the vintage market, but with a difference. In the case of Fender, many of the same team of designers and production workers who were responsible for the originals are still with the company and are overseeing the production of the reissue models, ensuring their authenticity down to the shape of the knobs, the color of the lacquer finish and the type of wax used to dip the pickups. At least six of the people involved in the project are 20-year-plus veterans of the company and have firsthand knowledge of the original instruments, including Freddie Tavares, Fender's senior designer and elder statesman whose first design project with the company was the Stratocaster. The end product of this project at

Fender is to be exact reproductions of six vintage Fender models: the '52 Telecaster, the '57 and '62 Stratocasters, '57 and '62 Precision Basses and the '62 Jazz Bass. All Fender vintage replicas will come complete with a leather-trimmed tweed case.

CIRCLE 67 ON READER SERVICE CARD

MUSICAL INSTRUMENT ACCESSORIES

New from Shadow of America Electronics is the Shadow Recording Bridge Pickup System™, with models for use on classical guitars, western guitars, madolins and banjos. The new system represents a new generation of six-transducer bridge pickups and features individual, replaceable, slip-fit, single-string saddles, each of which is a complete transducer unit to convert the vibration of its single string into an electrical signal. The body of the pickup is gold plated for maximum shielding and long contact life. Pickup/saddle models are available to replace original bridge saddles without major modifications in most instruments, and makes both natural-sounding amplification and the possibility of using electronic sound modifiers available to the player.

CIRCLE 75 ON READER SERVICE CARD

Shadow of America Electronics is now offering the Shadow Buckeroo 44DP magnetic pickup for round hole guitars. The new model is basically similar to Shadow's popular model 44, but adds a female jack into which a second pickup can be plugged. When a second pickup is plugged into this jack, the knob which is normally a tone control for the pickup becomes the volume control for the second pickup, allowing the sound of the two different pickups to be blended in any proportion. Examples of this function would include using a transducer on the guitar body to combine an electric, magnetic pickup sound with an acoustic sound, or using a Shadow harmonica pickup with the 44DP to allow the guitar/harmonica player a convenient way to mix and control the volume of each instrument independently.

CIRCLE 76 ON READER SERVICE CARD

Multivox has expanded its Big Jam concept of a modular effects pedal system with the introduction of three new devices which can be used independently or mounted in the Big Jam System Case. The new additions include a Preamp pedal with treble, bass and volume controls, and a Limiter pedal which features a distortion circuit as well as clean limiting/compression. Both pedals feature noiseless FET switching, recessed slide controls, an easy-access battery compartment and a color-coded, die-cast metal housing with a non-slip rubber sole. The third addition to the Big Jam system is a Quartz Dual Readout Tuner. The new unit features both an illuminated VU-type tuning meter and a bi-color LED which indicates red when the pitch is sharp and green when flat. The six normal open-string tones plus A-440 are available audibly for reference, and the unit may be left connected in line in an effects chain for quick tuning references during a performance.

CIRCLE 77 ON READER SERVICE CARD

An effects switching center known as Digipatch is the new offering from PNP Manufacturing. Digipatch accepts up to three effects devices and allows them to be switched in and out independently or to be series mixed, for a total of eight different combinations. The unit includes a triple footswitch and three front panel LEDs for unambiguous indication of which effects are in the path, if any.

CIRCLE 78 ON READER SERVICE CARD

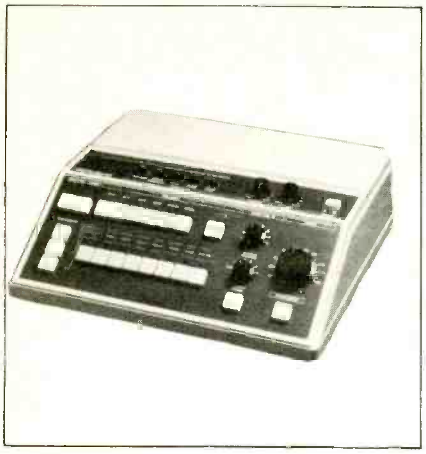
One of the latest additions to the MXR Musical Products Group is the MXR Micro Flanger. The new box, like the rest of the MXR Micro series, is a very compact, battery-powered device in a rugged yet affordable package. Like MXR's very popular Micro Chorus, the Micro Flanger has been carefully designed for a high degree of user orientation in its controls. A Rate control is provided which automatically decreases the width of the flanging effect as the speed is increased to allow a wide range of effects which are all musically usable. The other control on the unit is a Regeneration control with a range from 0% regeneration for a "classic" flanging sound to 100% regeneration for very metallic, resonant effects. The MXR Micro Flanger was designed for very low noise and low current consumption (said to be one-third of that of competitive units), and will accept an AC adapter.

MUSICAL INSTRUMENT AMPLIFIERS

A new amplifier head and complementary speaker cabinet have been announced by Peavey Electronics. The new amp head is known as the MXFC Top Box amp, and produces 130 watts RMS from a "tube-type" configuration The amp features Peavey's Saturation and Thick circuits to produce virtually any distortion/overload sound texture, plus 3-band passive EQ circuits, an active presence control, pre and post gain controls, a normal (clean) volume control and the usual high and low gain inputs. Other features include reverb, a high power/low power switch, forced-air fan cooling and a rugged ¾-inch flight case type housing with aluminum extrusions, handles, and removable front and rear covers. The companion speaker cabinet is designated the 412FC, which features four 12-inch Scorpion speakers housed in a flight case style cabinet of infinite baffle design, and comes complete with a removable transport cover for the front of the cabinet.

CIRCLE 79 ON READER SERVICE CARD

A line of compact, rack-mount or free-standing amp heads has been announced by Gallien-Krueger. The new R-Series Component Amplifiers are available in both 100- and 200-watt RMS versions, with separate versions for guitar and bass. All four models feature the same functions and technology as Gallien-Krueger's Combo Series, including A/B switching between clean and overdrive channels, three-stage overdrive, four-band EQ, and reverb for the guitar versions. G-K's R-Series amps combine preamp and power amp circuitry in one 3½" high unit with detachable rack ears making it very convenient for the musician to carry all his electronics in one rack system.



RolandCorp US has entered the low cost amplifier market with the introduction of the Spirit Amplifier Group. This new line includes a 10-watt guitar rehearsal amp, 30- and 50-watt guitar amps and 30- and 50-watt bass amps. The Spirit 10 is a compact rehearsal amp delivering 10 watts RMS to an 8-inch speaker, from an amp featuring volume and master volume controls, three-band EQ, and jacks for normal/overdrive switching, line out and a headphone out for silent practice. The Spirit 30 and Spirit 50 amps both feature 12-inch speakers, high- and low-gain inputs, reverb (with footswitch), three-band EQ, a master volume control with a pull-on presence switch and a headphone output jack. The Spirit 50 additionally has a pull-on switch for overdrive integrated with the input volume control, a footswitch jack for overdrive switching and a preamp out/main amp in patch point for connecting effects devices. The two bass amps feature high and low gain inputs, bass, mid and treble EQ plus a fully parametric EQ section with frequency and boost/cut controls, a Q select switch, a footswitch jack and an LED indicator. The Spirit Bass 30 delivers its 30 watts into a 12-inch speaker while the Spirit Bass 50 boosts a 15-inch speaker in its cabinet.

CIRCLE 68 ON READER SERVICE CARD

Roland has expanded its line of Cube guitar amps with the addition of two new models designed specifically for keyboard use. Like the rest of the Cube line, the new models offer sophisticated features and performance in very compact packages. The new models are designated as the CK-40, or Cube-40 Keyboard, and the CK-60 (Cube-60 Keyboard), and are rated at 40 and 60 watts RMS, respectively. Both models feature two input channels—each with individual volume controls and a switchable input attenuator—to handle two keyboard inputs simultaneously regardless of their relative levels. Both models have controls for treble, mid, and bass and reverb; the CK-60 additionally allows the reverb to be assigned to either or both of the input channels. The CK-40 uses a self-contained, two-way 10-inch speaker, while the CK-60 uses a 12-inch speaker and a horn tweeter for a wide frequency handling range. Both amps feature effects patching, line outputs, headphone outputs and an external speaker jack.

CIRCLE 69 ON READER SERVICE CARD

St. Louis Music has announced the latest addition to its prolific Crate amplifer line. The latest model is the CR-165B, a 60-watt bass amp with a 15-inch Magna Projecter speaker in a ported enclosure. New features for this model include a peak limiter (with LED indicator) to eliminate overdriving the speaker, a Bright switch with 8 dB of boost and both line output and external speaker jacks.

CIRCLE 81 ON READER SERVICE CARD

CIRCLE 35 ON READER SERVICE CARD

Recording Techniques Part 6

by Bruce Bartlett

In the August 1982 issue, we looked at several ways to record amplified instruments and drums. This article will cover some typical miking techniques for acoustic guitar, piano, strings, horns and vocals. While these methods work well in many applications, they are only suggestions. Feel free to experiment.

Acoustic Guitar

The acoustic guitar has a carefully tailored tone quality which we try to capture through judicious microphone selection and placement. First prepare the guitar for recording. Use strings designed to reduce finger squeaks, if possible. For maximum brilliance, replace old strings with new ones—already played and stretched. Experiment with different kinds of picks and finger picking to achieve a timbre suitable for the song.

A condenser microphone with a smooth, extended frequency response from 80 Hz up is often preferred for acoustic guitar. Such a microphone typically gives a clear, detailed quality, in which the plucking of each string is audible within a strummed chord. The reproduced sound usually has all the crispness of the "live" instrument. However, the clear pickup of string noise can be distracting in some songs. You can diminish this fine detail by using a dynamic microphone, which usually has a slower transient response (the ability to follow sudden changes in acoustic pressure).

If you've ever miked an acoustic guitar close to the sound hole *(Figure 1)*—a popular microphone position—you've probably noticed that the recorded guitar doesn't sound much like the real thing. The recording sounds too bassy, boomy and thumpy. That is mainly because the sound hole and the air inside the guitar resonate at low frequencies (around 80 to 100 Hz). A microphone placed close to the sound hole (or in it) picks up and emphasizes this resonance, giving a bassy character to the recorded guitar. To achieve a more natural sound in this microphone position, roll off the low frequencies on your mixer (say, -10 dB or more at 100 Hz) or use a microphone with a frequency response that rolls off at the low frequencies. An example of such a microphone is shown in *Figure 2.* It clips onto the sound hole of a guitar and provides a good starting-point tonal balance in this position. It also allows the performer freedom of movement.

Why then is a guitar commonly miked close to the sound hole? On stage, this microphone position provides maximum loudness before feedback occurs. In the studio, it provides maximum isolation (minimum leakage pickup). The acoustic guitar, being a relatively quiet instrument, often

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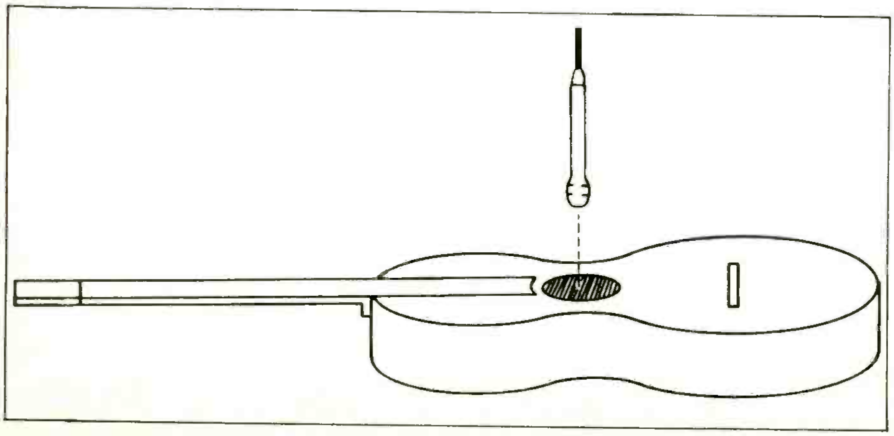

Figure 1: Acoustic guitar miked close to the sound hole.

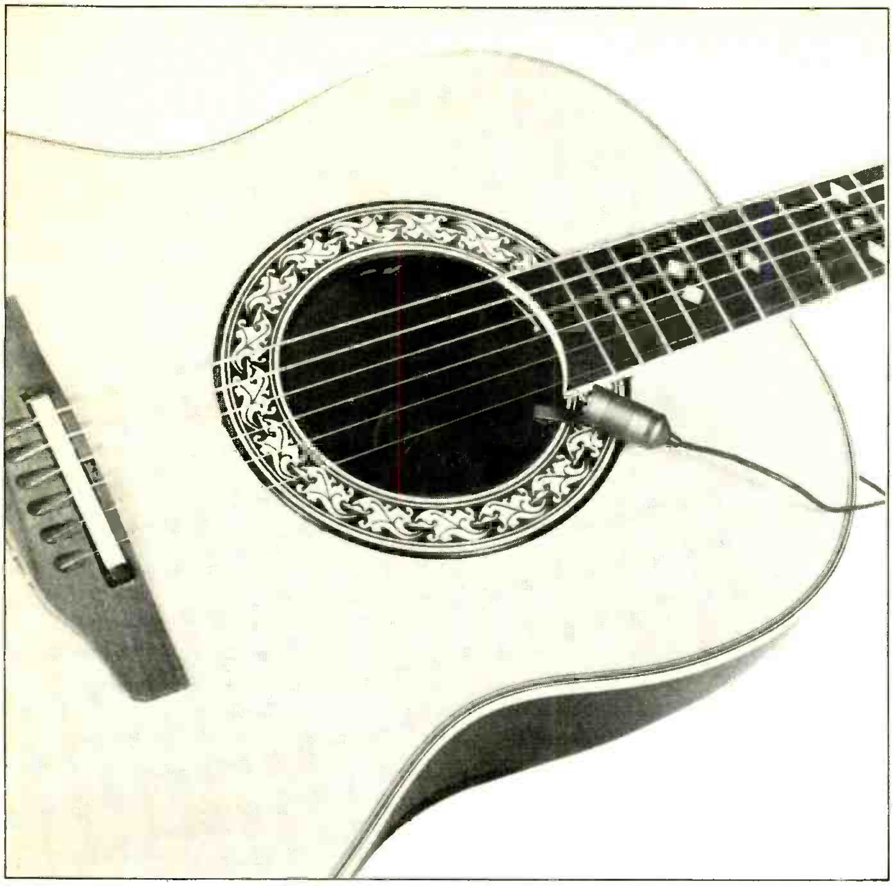

Figure 2: Example of a clip-on mic for guitar.

requires such a technique to prevent feedback and reject leakage.

Best isolation—sometimes at the expense of fidelity—is achieved with a *contact pickup*, which attaches to the body of the guitar. For a starting point, place the pickup on or next to the bridge and adjust the position from there. Positioning a contact pickup is critical—a movement of a fraction of an inch can drastically change the sound. Each instrument has a different "best" location for the pickup, and every brand of pickup sounds different. Multiple pickups, or a pickup and a microphone, can be mixed.

If leakage is not a problem (as during overdubs), a more natural sound can be achieved by miking the guitar at a distance—say, 1½ to 3 feet from the sound hole *(Figure 3)*. At this position, the microphone picks up a well-balanced blend of all the parts of the guitar—strings, soundboard and sound hole. A closer placement that also provides a bright, realistic sound is 6 inches over the top, over the bridge and even with the front soundboard *(Figure 4)*. You may be pleasantly surprised with the sound you get with this technique.

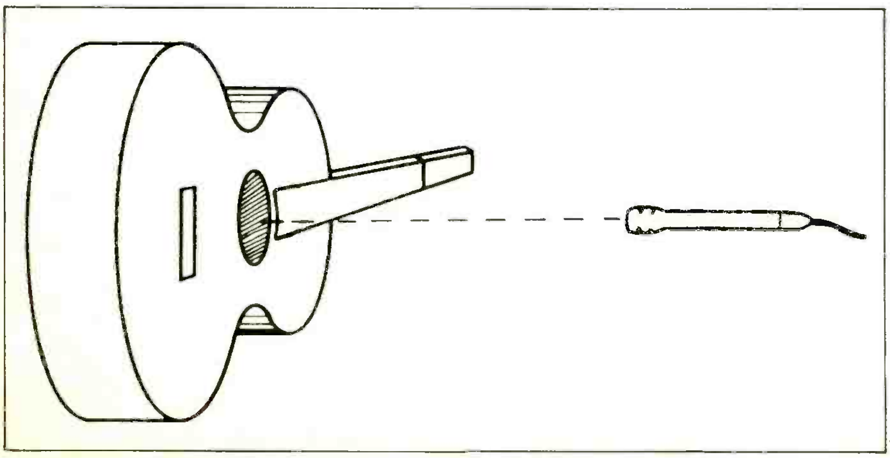

Figure 3: Acoustic guitar miked 1½-3 feet from sound hole.

A woody, mellow timbre is picked up by a microphone about 4 inches in front of the bridge *(Figure 5)*. Here the vibrations of the soundboard (starting around 200 Hz) are emphasized. This position also reduces pickup of string and pick noise.

When recording a classical guitar solo, distant microphone placement is called for to capture the room acoustics or reverberation (a desirable part of the sound of classical music). Record in a recital hall or other warmly reverberant room. Place the microphone about 3 to 8 feet away—closer to reduce reverberation, farther to increase it. For a more realistic sense of "space" or "air" surrounding the soloist, record in stereo. Angle two cardioid microphones 90 degrees apart and space their grilles about 8 inches apart *(Figure 6)*. If you are forced to record a classical guitar in an acoustically dead

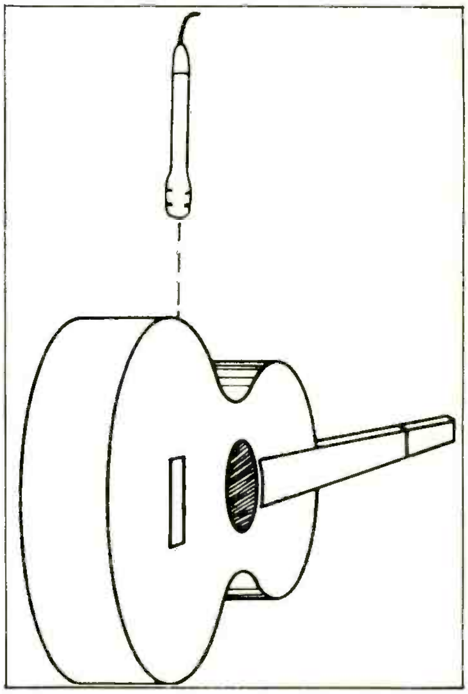

Figure 4: Acoustic guitar miked 6" over the top, over the bridge and even with the front soundboard.

room, try miking the guitar over the top, over the bridge and even with the front soundboard *(Figure 4)*. Add artificial reverberation with your mixer.

Banjo

The banjo uses a "drum head" to couple the string vibrations to the air. The center of the head vibrates mainly at the head's fundamental frequency, while the harmonics of the head

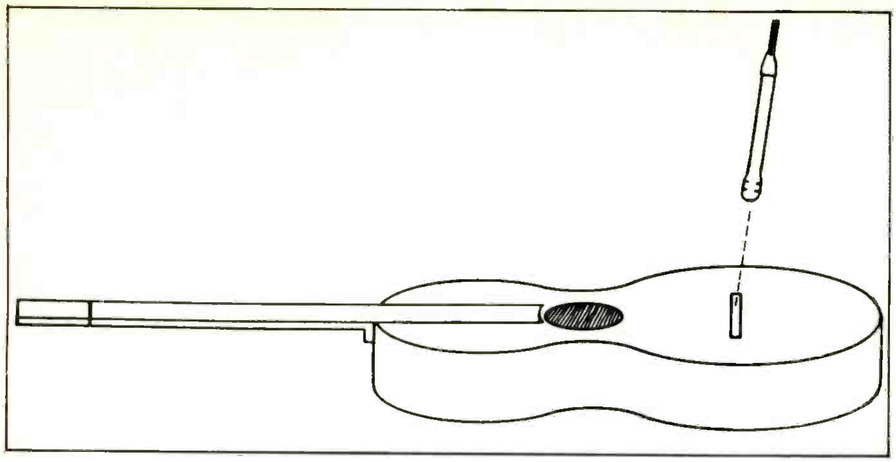

Figure 5: Acoustic guitar miked 4" in front of the bridge.

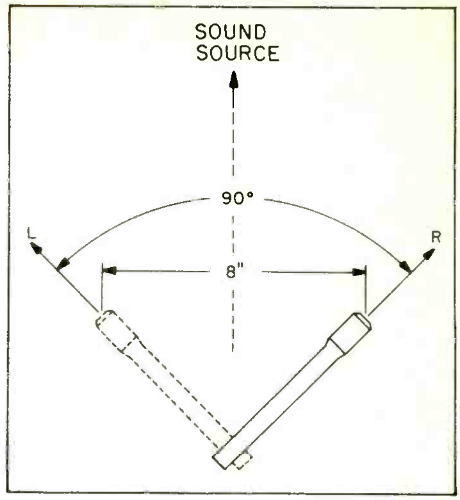

Figure 6: A stereo miking technique.

vibration are strongest near the edge. Sometimes the lower notes are reinforced by holes in the flange surrounding the head. To pick up a natural blend of all the parts of the banjo, place a flat-response microphone about 1 to 2 feet away. Positioning the microphone close to the center of the head produces a rather harsh, thumpy sound (unless you roll off the bass), but provides good isolation. The sound becomes thinner toward the edge of the head. You can mount a miniature microphone inside the banjo for maximum isolation, or clip the microphone onto the tailpiece aiming toward the bridge. As a starting placement for a contact pickup, wedge the pickup between the strings and the head behind the bridge. The pickup should be flat against the banjo head.

Mandolin, Dobro, Fiddle

These instruments are constructed somewhat like an acoustic guitar, so many of the microphone techniques for guitar are applicable.

The fiddle or violin radiates high frequencies mainly at right angles to the front surface. Audiences usually hear a duller sound from the violin than the violinist hears. When close miking the violin, you can avoid the harsh, bright sound the violinist hears by aiming the microphone at the side of the violin. A microphone response down to 200 Hz is sufficient. Another technique that works very well is to mount a miniature microphone with a bass rolloff near the bridge *(Figure 7)*. A suggested pickup placement for a fiddle is on the left side of the top, on the player's side of the bridge.

Grand Piano

The piano is difficult to record and reproduce so that it sounds realistic. One reason is that it is such a big, complex sound source (5 to 9 feet long). The natural sound of a piano heard at a distance is a blend of the individual sounds of its many parts—strings, hammers, soundboard and lid. Close miking, however, emphasizes the part of the piano that the microphone is near. An unnatural recorded timbre can result. To further complicate matters, combinations of sounds from various areas and sound reflections from the lid cause acoustic phase cancellations that vary with microphone placement. Lid reflections arrive off-axis to the microphone, sometimes producing coloration. In addition, a piano has sharp attack transients that can saturate the recording tape unless recorded at lower-than-normal levels. All these factors make the piano a challenge to record without distortion or coloration.

Figure 7: A clip-on mic for violin.

One way to record the piano as an audience hears it is to set the piano lid on the long stick, and place a pair of flat-response cardioid condenser microphones about 6 to 12 feet away from the open lid, about 6 feet high, and record in stereo (as in *Figure 6)*. This method is useful for taping piano solos or overdubs. A classical piano solo should be recorded in a reverberant locale such as a recital hall or concert hall.

In pop-music recording, the piano is miked closer to reduce pickup of room ambience and leakage, and to increase clarity. Try not to mic the strings closer than about 8 inches away because that can overemphasize the strings closest to the microphone. You want equal coverage of all the notes the pianist plays.

There are many ways to close-mic a piano. Experiment to see what works best for the particular song and instrument.

One popular method uses two spaced microphones inside the piano, with the lid on the long stick or removed. One microphone goes over the treble strings, about 8 inches high and about 8 inches horizontally from the hammers, and the other microphone is placed over the bass strings, about 8 inches high, about 2 to 4 feet from the hammers *(Figure 8, position Ⓐ)*. These microphone signals are panned partly toward the left and right for a stereo effect. Alternatively, two surface-mounted microphones can be taped to the underside of the piano lid over the bass and treble strings, or taped to the inside of the front edge. In another technique, a single microphone or a pair of crossed cardioids is positioned about 1 foot over the middle of the piano, about 8 inches from the hammers (*Figure 8, position* Ⓑ). The closer to the hammers the microphones are placed, the more percussive is the attack in the recording. So, if the sound is too "bangy" and lacks tone, move the microphones toward the tail of the piano away from the hammers.

Putting a microphone just over a sound hole, with the lid on the long stick, provides good isolation and yields a punchy, constricted sound which can be effective for rock music

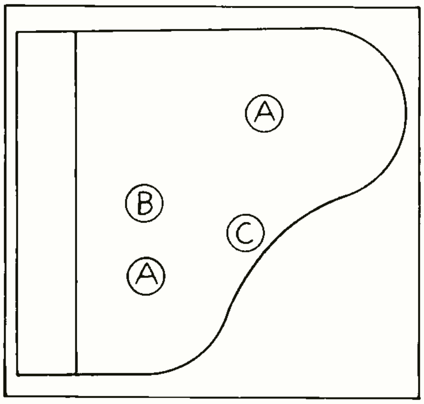

Figure 8: Some close-miking positions for piano.

(*Figure 8, position* Ⓒ). Each hole emphasizes the strings closest to it.

For best isolation, aim a microphone into a sound hole, close the lid, and cover the piano with heavy blankets. The tone quality will be unnatural, so you'll have to experiment with equalization if you want a more realistic sound. Special contact pickups for piano are available to further increase isolation.

Often a bright piano sound is desired. You can improve clarity, sharpness, and attack by boosting frequencies about 5 kHz on your mixer, by using a microphone with a presence peak, or by sticking thumbtacks into the hammer felt.

Upright Piano

As with the grand piano, each microphone placement for the upright piano produces a different tone quality. For a natural sound with clear hammer attack, place two microphones just over the open top, one over the treble strings and one over the bass strings. Record in stereo and pan the signals left and right for the desired piano width. If you can spare only one microphone for the piano, just cover the treble strings. An alternative placement is aiming at the hammers from the front, a few inches away, with the front panel removed. Place the microphones inside the top of the piano to minimize leakage. Excessive hammer attack can be softened by placing a pair of microphones about 8 inches from the soundboard, covering the bass and treble sides. In this case, the soundboard should be facing into the room—not into a wall. A surface-mounted microphone or two, about 1 foot from the soundboard on the floor, or on the wall next to the soundboard, gives a natural sound with good presence.

Strings

Not many home studios record string sections. But if you want to add some string sweetening during an overdub session, here are some suggested techniques:

Place the strings in a large, hard-surfaced room that has noticeable reverberation, and mic the strings at a distance to pick up a natural acoustic sound. Condenser microphones with a flat frequency response are usually preferred. For two violins, try one microphone about 6 feet off the floor, aiming down between the players. The viola and cello each can be miked about 2 feet from the f-holes or from the side. For added definition on the cello, mic it about 1 foot from the bridge.

The acoustic bass (string bass, double bass, upright bass, bass viola) can be recorded in several ways. This instrument goes down to 41 Hz, so use a microphone with an extended low-frequency response. Place the microphone a few inches out front, above or below the bridge, for a well-defined sound. Aim it into the treble f-hole for a fuller sound. As always, watch out for proximity effect with closely placed cardioid microphones. To increase isolation and to allow the performer freedom of movement, wrap a miniature condenser microphone in foam rubber (or in a foam windscreen) and mount it in an f-hole. Or, wrap a regular microphone in foam padding (except the front grille) and squeeze it behind the bridge or between the tailpiece and the body.

Large string ensembles can be covered with one microphone for every four violins and violas, one for every two celli and one for the acoustic bass.

When you mix all the signals of the strings to stereo, pan them evenly between the monitor speakers. Spread them left, center and right to achieve a "curtain of sound." A simpler stereo recording method uses just two microphones spaced 3 to 5 feet apart to pick up the whole ensemble. If you can spare only one track for the strings, you can make that track "pseudo-stereo" in the mixdown. To do this, (1) pan the track to the left channel; (2) simultaneously send it through a 20 msec. delay; and (3) pan the delayed signal to the right channel. Adjust the relative levels of the direct and delayed signals to achieve a stereo spread from speaker to speaker.

A string quartet can be recorded in stereo as in *Figure 6*. Place the microphones about 6 to 10 feet away from the quartet to capture the room ambience. A limited stereo spread, rather than a speaker-to-speaker spread, is sometimes preferred for a string quartet. To reduce the width of the stereo stage, reduce the angle or spacing between the microphones.

A harp can be covered by a microphone aiming toward the treble part of the soundboard from the front, about 1½ feet away (if the harp is playing with an orchestra) or at a greater distance (for a harp solo). Tape a lavalier microphone to the soundboard for best isolation.

Horns

Trumpets, cornets, trombones, tubas: These instruments radiate strong high-frequency harmonics directly out of the bell, but do not project them to the sides. A microphone placed close to, and in front of, the bell will pick up a brighter, more "edgy" tone than an audience usually hears. To soften the tone and restore

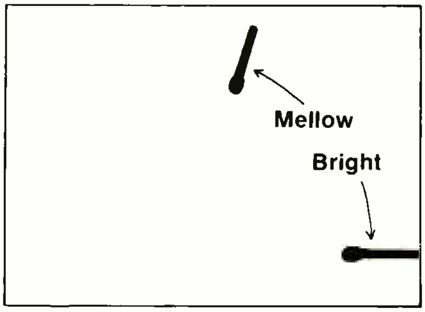

Figure 9: Trumpet tone control (top view).

the natural horn sound, try miking the bell at an angle (*Figure 9*) with a flat-response microphone. Or mic it on-axis with a ribbon microphone, which typically provides a smooth sound. Use a condenser microphone to reproduce a lot of sizzle. Close microphone placement (about 1 foot) gives a tight sound; distant placement (about 5 feet) yields a fuller, more dramatic sound. For best isolation, tape a miniature microphone (dynamic or condenser) inside the bell and adjust mixer equalization for a natural timbre.

It's common to put two or more horns on a single microphone. Several players can be grouped around a single omnidirectional microphone or around a cardioid microphone placed below the group aiming up. Alternatively, the musicians can play to a surface-mounted microphone taped to a hard reflective wall or large baffle.

Woodwinds: Woodwinds and brass are both often called "horns" in studio terminology. With woodwinds, most of the sound radiates not from the bell, but from the holes. So aim a microphone at the holes about 1 foot away. A flat-response dynamic is typically used.

Saxophone: The sound of a sax miked near the bell is bright, breathy and rather hard. Mic it there for best isolation. A fuller tone, plus key noise, can be picked up near the holes. Try miking both the bell and the holes with two microphones a few inches away, for a sound combining presence and warmth *(Figure 10)*. A compromise position for one microphone might be near the bell but aiming at the holes. A sax section can be grouped around a single microphone.

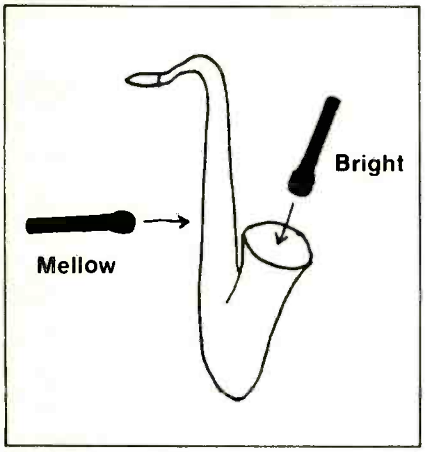

Figure 10: Miking a saxophone.

Flute: One effective microphone placement is a few inches from the area between the mouthpiece and the first set of finger holes. A pop filter may be needed. If you want to reduce breath noises, place the microphone behind the player's head, aiming at the finger holes *(Figure 11)*. His head will prevent the high-frequency breath noises from reaching the microphone. Increasing the microphone height or distance also can reduce breath sounds. For classical-music solos, try a stereo pair 5 to 8 feet away.

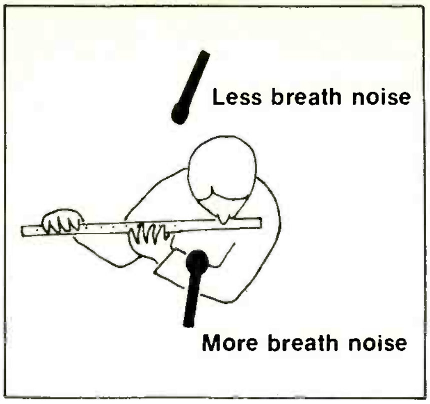

Figure 11: Flute (top view).

Harmonica: A popular technique uses a cardioid dynamic microphone with a presence peak placed very close to the harmonica (sometimes held by the player). You may need a windscreen. For a bluesy, dirty sound, you may want to play the harmonica through a miked guitar amp.

Lead Vocal

Vocal recording presents a number of problems. Among these are proximity effect, pop, wide dynamic range, sibilance and sound reflections from the lyric sheet. Let's look at these in detail.

Minimizing Proximity Effect: A vocalist on stage has to sing with his lips touching the microphone grille to reduce feedback. Singing or talking close to a cardioid microphone boosts the low frequencies, due to proximity effect. The result is a bassy, boomy tone quality that we've come to accept as a standard sound-reinforcement vocal sound. During a recording session, this effect may add robustness to a weak voice; but normally the vocalist should back off at least 8 inches from the microphone to restore a natural tone quality *(Figure 12)*. Vocals are typically overdubbed from about 8 inches to 2 feet away with a flat-response condenser microphone.

If you must record the vocalist simultaneously with the instruments, you'll probably have to mic him or her very close to avoid picking up the instruments, leaking into the vocal microphone. A cardioid microphone with a pop filter is useful here. To reduce the boominess caused by this close placement, roll off the excess bass on your mixer. Some microphones have a bass rolloff switch built in for this purpose. Aiming the microphone up toward the singer's nose will avoid a "closed-nose" or "nasal" effect.

Minimizing Pop: When a vocalist sings a word emphasizing the letters "p" or "t," a turbulent puff of air is forced from his or her mouth. A microphone placed near the mouth is hit by this air puff and generates an undesirable thump or little explosion called a "pop." It can be reduced by placing on the microphone a foam-plastic or metal-screen *pop filter* (also called a *windscreen* or *ball grille*). Some microphones have a ball screen already built in.

Although these devices reduce pop, they do little to minimize breathing sounds or lip noises. Distant miking or some high-frequency rolloff can help with these problems.

Foam pop filters should be made of special open-cell foam to allow high frequencies to pass through. For this reason, it's better to use a commercially made foam screen than to make one yourself from packing foam, cloth

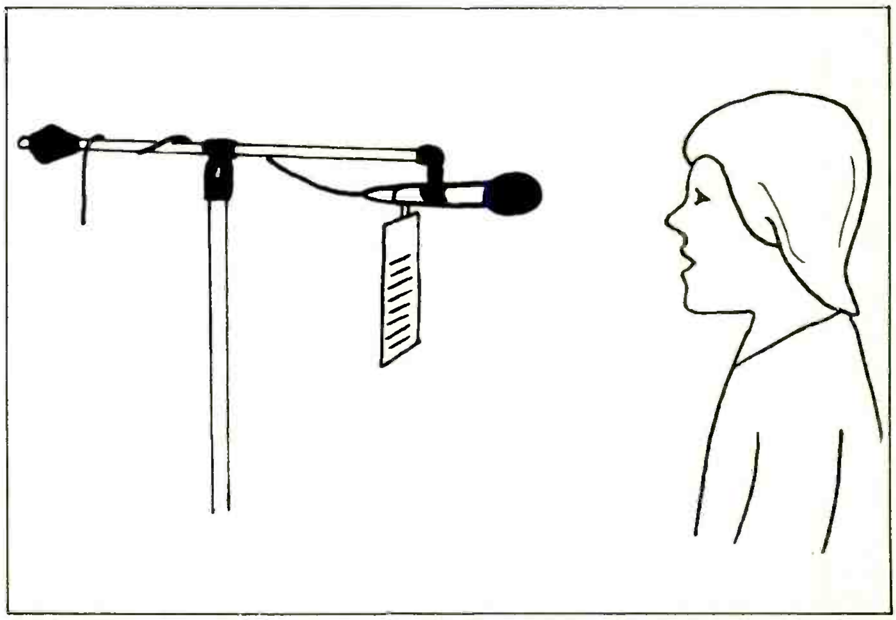

Figure 12: Typical miking technique for a lead vocalist.

or socks. Allow a little air space between the foam front and the microphone grille for best pop rejection.

Since pop filters slightly change the frequency response of a microphone, they should be left off microphones intended for instruments, except for outdoor recordings or dust protection.

A very effective way to eliminate popping is to place the microphone well *above* the singer's mouth level *(Figure 12)*. This way the puffs of air shoot *under* the microphone and miss it. Or you can place the microphone off to one side of the mouth *(Figure 13)*.

Reducing Wide Dynamic Range: Vocalists often sing too loudly or too softly during a song, either blasting the listener or getting buried in the mix. That is, singers generally have a wider dynamic range than their instrumental backup. To even out these extreme level variations, the vocalist should use proper "mic technique"; backing away from the microphone on loud notes; coming in closer for soft ones. Or you can ride gain on the vocalist: gently turn him down as he gets louder or up as he gets softer. Alternatively, the vocal signal can be passed through a compressor or limiter, devices that automatically reduce dynamic range.

A microphone placed close to the mouth is very sensitive to small changes in miking distance. The singer's loudness will jump up and down if he fails to keep a constant distance from the microphone, or if he fails to use the mic technique mentioned above. For this reason, it's better to mic the singer at least 1 foot away. Small movements of the singer affect loudness less at that distance. If you must mic close to prevent leakage, have the singer's lips touch the pop filter to maintain a constant miking distance.

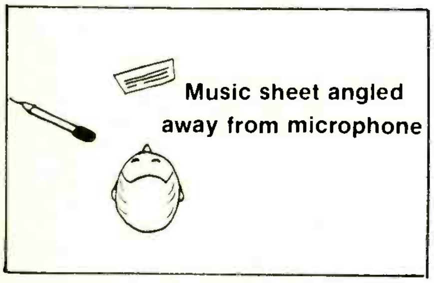

Figure 13: Miking a vocalist from the side (top view).

Minimizing Sibilance: Sibilance is the emphasis of *"sss"* or *"shhh"* sounds. These sounds are strongest in the 5 to 10 kHz range, and can easily saturate a tape running at 7½ ips if not controlled. To do this, use a microphone with a flat response (rather than one with a presence peak) or reduce the highs around 5 kHz on your mixer. A "de-esser" device does this automatically whenever sibilant sounds occur. Also, mic the vocalist from the side rather than in front *(Figure 13)*. The "s" sounds are projected more out front than they are to the sides. A dull-voiced singer, on the other hand, may need a small presence boost to be clearly heard.

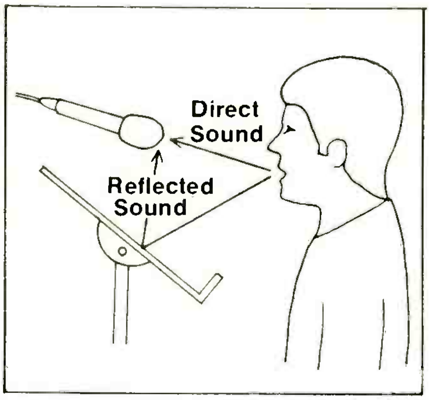

Figure 14: Reflections from a music stand causing interference.

Reducing Reflections from the Lyric Sheet: Sound reflections from the lyric sheet and music stand can bounce into the microphone along with the direct sound from the vocalist *(Figure 14)*. The reflections interfere with the direct sound, creating a colored tone quality similar to mild phasing or flanging. To eliminate this effect, place or tape the lyric sheet at the rear of the vocalist's cardioid microphone, perpendicular to the microphone axis *(Figure 12)*; or mic the vocalist from the side, and angle the lyric sheet slightly away from the microphone *(Figure 13)*. In the first arrangement, reflections entering the rear of the cardioid microphone are rejected. The second method makes reflections bounce away from the microphone.

Some effects often used on lead vocals are reverberation, echo and doubling. Room reverberation sometimes can be recorded "live" by miking the singer at a distance in a hard-surfaced, echoey room. Tape-echo (or electronic time delay) gives a 50s rock 'n' roll effect, and sounds less mechanical if some highs are rolled off the echo signal. Doubling a vocal provides a fuller sound than a single vocal track. You record a second take of the vocal on an empty track at a slightly different miking distance. During mixdown, you mix the second vocal take with the original, at a slightly lower level than the original. Or you can run the vocal signal through a *chorus* device to double it.

Vocals typically are boosted slightly in the "presence" range around 1 kHz to 5 kHz to help them stand out against an instrumental track. This boost may increase sibilance as well.

When recording a classical-music singer, place the microphone(s) about 3 to 8 feet away to pick up room reverberation. You may want to use a surface-mounted microphone on the floor for this application.

Backup Harmony

When overdubbing backup vocals, you can group two or three singers in front of a microphone. The farther they are from the microphone, the more distant they will sound in the recording. Barbershop or gospel quartets with a good natural blend can be recorded in stereo as in *Figure 6*, about 2 to 4 feet away. If their balance is poor, try miking them individually up close, with omnis, and blend them properly using your mixer.

Conclusion

We've covered some popular techniques for recording acoustic instruments and vocals. In general, if leakage is a problem, you place the microphone where the sound output is loudest. Otherwise, you place the microphone in several different positions until you find a location where you monitor the desired tone quality and amount of ambience. There is no single "correct" microphone technique for each instrument because the timbre and ambience *you* want to hear determines the microphone placement. Understanding microphone characteristics and instrument sound-radiation patterns will help you find a spot more quickly.

The accuracy of the monitor speakers and their environment is critical because the sound you hear from the monitors affects your microphone technique. Monitoring will be covered in detail at a future date.

You may want to make yourself a demo tape comparing the sounds picked up at various microphone positions. On the tape you might say, for example, "Here's the sax miked near the bell..." "Here's the sax miked near the holes..." and so on.

All the microphones you've set up so carefully plug into a mixing console for blending, balancing and further sonic tailoring. We'll explain the components and systems of mixing consoles in the next installment.

RECORDING WITH...

By Vicki Greenleaf and Stan Hyman

In the past, Al Jarreau's music has been categorized as R&B or jazz and was mainly limited to those listening audiences. Although lacking a large following—"My audience grew slowly," he says—each of his first five releases has been well-lauded by critics and peers; he holds numerous, prestigious European and American music awards. However, it wasn't until the release of his sixth LP, Breakin' Away *[Warner] last year, that he found recognition—as well as his third and fourth Grammy Awards for Best Male Pop Vocal and Best Male Jazz Vocal of 1982—as a fusion singer, blending pop with his jazz background and R&B influences. With that recognition came the commercial success that made* Breakin' Away *his first platinum effort and the singles "We're in This Love Together" and the title track hits on the pop charts.*

Formerly a rehabilitation counselor who sang in Milwaukee clubs after hours, the 42-year-old vocalist went into music fulltime 14 years ago. While performing at the Troubadour in Hollywood in 1975, he was heard by Warner Brothers executives and signed to a recording contract.

Unable to play a musical instrument, Jarreau utilizes his voice as an instrument. In the ensuing interview, he discusses his vocal abilities, his work as an improvisational singer, his songwriting efforts, much of his recent work in conjunction with keyboardist Tom Canning and producer Jay Graydon and his recording techniques. Graydon adds a few more technical aspects to the conversation. Jarreau also describes the sound that his latest album retains from Breakin' Away *and the improvements he feels he has added to an already successful formula.* MR&M *spoke with Jarreau in Los Angeles prior to some late-night studio work. The conversation lasted until the early morning hours.*

Modern Recording & Music: Is your new release similar to *Breakin' Away?*

Al Jarreau: I think it's similar to *Breakin' Away*, but this album has some stronger material. It's at least as interesting as *Breakin' Away* in terms of the musical structure, maybe a little more interesting. Lyrically, I say what I tend to say, so it's similar in that regard. The rhythm feels will also be similar. So, we could say it's similar, only I think it's going to be better. I'm really knocked out.

MR&M: Did you use the same studio musicians?

AJ: Yeah, most of them.

MR&M: Did the phenomenal success of *Breakin' Away* intimidate you when you went back into the studio to record this album?

AJ: Yeah. There's always that kind of pressure. I have felt that within myself with each successive album. I want to do something that is at least as good, and I'm striving for something better. I want to grow. The audience and the performer hope for and look for growth. Because of the nature of the industry, I think we're using up artists and leaving the wastes somewhere. Unless you have something substantial and strong, you're gone. Even if you do, you've got to work in order to survive. The industry has had a lot of fat around it's middle for a long time. I think we've spent a lot of money on very mediocre stuff for too long. Well, the cutbacks are here and the music industry is feeling it as well as the country. We have to trim the fat now and I hope the effect will be that things of value and substance remain.

MR&M: Do you feel that your musical approach has changed over the course of seven albums?

AJ: I think that it's safe to say that it has changed a bit. I've learned a lot more about me as a singer, about what it is that I do and how to make what I do more effective. The change, in large part, is a result of that process. I think that I tended to be real potpourri-ish in my earlier approach to songs. I put everything into one song. I still tend to

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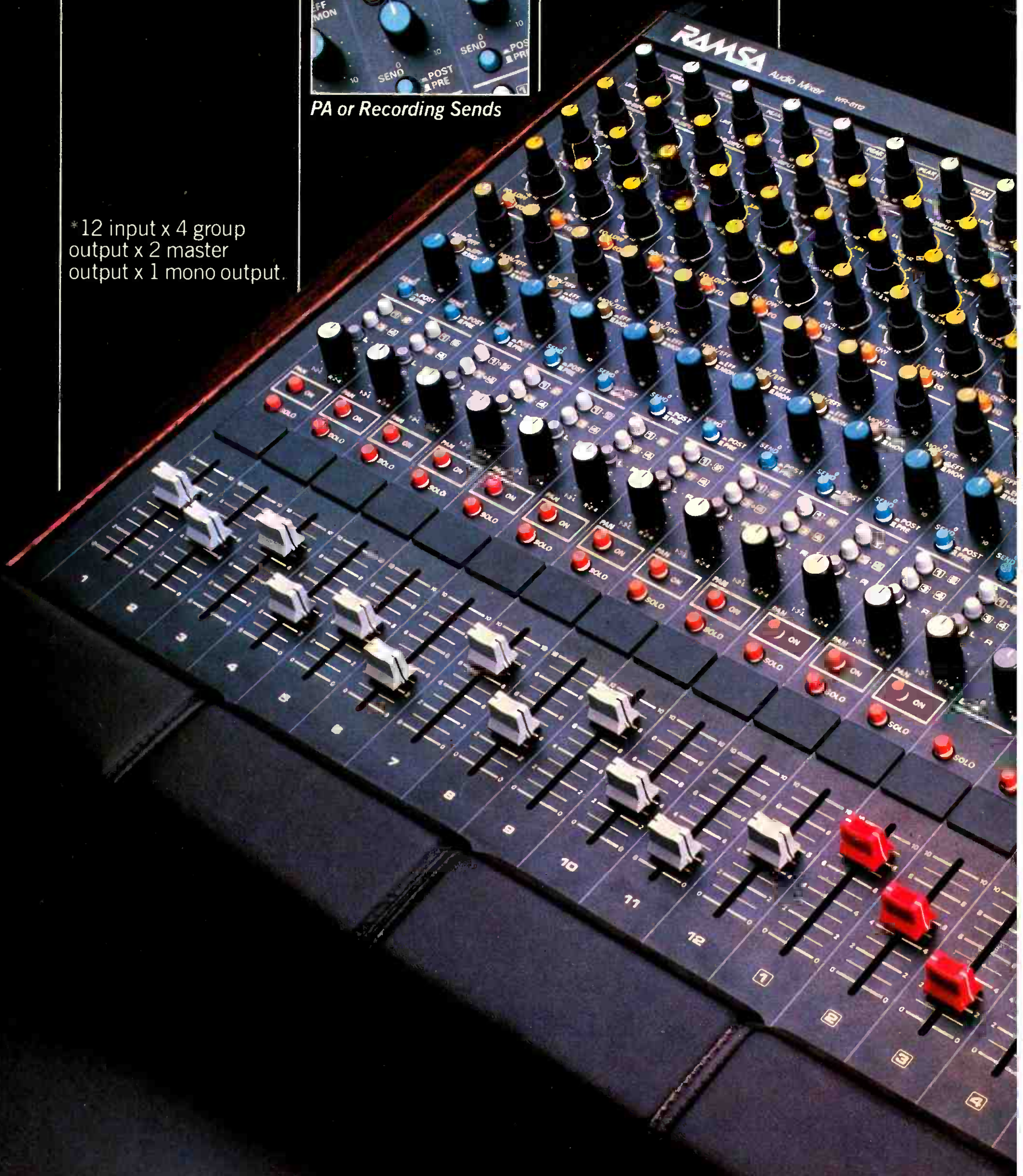

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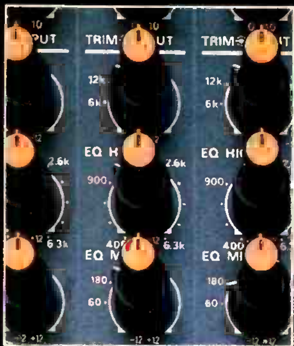

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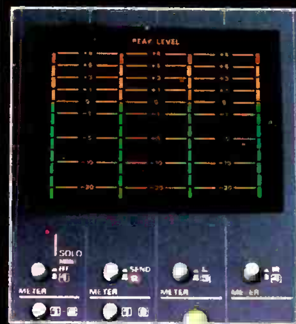

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be undisciplined on stage, but in the studio, I've really tried to focus on the various things that I like to do. I'm an R&B singer, a jazz singer and a pop singer. I even like to do things that are almost sacred, with that kind of message. So, what we're doing now—instead of taking every tune and putting all that inside of it—is spotlighting the various music types and making it more simple. When there's a jazz tune, we just wail; we do "Spain" or "Blue Rondo" or whatever. Then I'll take an R&B tune and just do it straightahead R&B, flavoring it the way I naturally flavor things with my jazz background. I sing a ballad like Frank [Sinatra] would, without too many bebop lines in it. I just sing the pop tune and leave the scat choruses for the jazz things. So I think I've just brought a little better discipline into the studio. It's made me a lot more accessible. I've reached a lot more people, but I'm still able to do basically what I want to do, especially "live."

MR&M: Can you categorize your musical style?

AJ: I suppose I'm a fusion singer. I'm a fusion of a whole lot of things. I enjoy it all. I like performing lots of different kinds of things.

MR&M: Have you retained an identifiable style throughout your work?

AJ: Oh yeah. I think that Al Jarreau fans know me anywhere.

MR&M: Who are your fans?

AJ: I think that my audience is a lot like me; they come from lots of different areas of music appreciation. As far as musical tastes, I think I've got people who like jazz, people who like R&B and listen to the R&B stations all day and people who like pop and listen to the pop stations all day long. But they go to the Al Jarreau concert and listen to other kinds of things. They basically appreciate my approach to other kinds of music that they may not even listen to most of the time. So, it's a real varied audience and I'm real knocked out to say that the audience is mixed ethnically and age-wise.

MR&M: You've been critically acclaimed for years. What were the hooks that made *Breakin' Away* such a commercial success?

AJ:I think I was trying to describe some of that when I was talking about discipline. There's a quote in an interview with George Benson—and he zeroed right in on it—that "probably the biggest problem with Al Jarreau's career, keeping him from greater success than he's had, is that he's giving the people too much at once, more than they can deal with..." So, I think we've found a way that people can isolate the various things that I'm doing and say, "Oooo, I don't know if I like that so much, but I do like this part and I can really understand that." Then the other guy says, 'Well, I understand that, but this is really interesting over here." I mean, that's the comment about songs like "Spain," "Blue Rondo" and "Take Five," "God, that's so interesting."

MR&M: Do you feel that the music industry has changed and become more receptive to your jazz and R&B influences?

AJ: R&B as a market, as a quote, unquote separate part of the industry, and the pop industry have both been very, very successful. There's a lot of money being turned over in either of those areas. I don't think that jazz has had that kind of success. There have been some jazz artists who have merged, done very well and become popular. I think the industry is changing in that it is more accepting of R&B and jazz—basically Black music which has had an immense influence on the music that's being done in the world today. Look back at the early 50s, all that music was called race music and you could only hear it on a few Black stations. Some White artists, who had a lean in that direction, brought it into a mainstream kind of focus.

MR&M: It seems that the two Grammy Awards you received for *Breakin' Away* were an important aspect of your recognition as more than a jazz artist?

AJ: A very important aspect, even to be recognized for a nomination. That meant to me that we were reaching other people; that a part of me that is like all those people out there is now being heard, felt and seen. And there is a part of me that is like all those other people out there, who are basically involved in pop music. I'm glad that's being heard, recognized and appreciated. It was really a feather in my cap [to win]. Jay [Graydon] and I just grinned, laughed, hugged and slapped each other on the back. I've had nice critical acclaim since the beginning, which I think has been a real help to my career, but I haven't had a large massive following. My audience has grown very slowly.

MR&M: Due to the nature of jazz—which entails improvisation—are you striving more for feel in your music than note-by-note perfection?

AJ: I would like to think that they aren't mutually exclusive. Obviously, in a "live" situation, it's totally feel; simply because once played, it's heard. But in the studio, I've come to be, through Jay's influence, probably one of the most-meticulous people doing music. We feel that it's possible to have a really moving performance, but at the same time have real perfection. But you have to know what you're doing in the studio and basically, what you're doing in order to get it right, is performing it over and over and over. Honestly, something *is* lost, but something's gained too. What you have to be careful of is your mental attitude. You have to mentally be able to go back, do it again and find a new freshness. You'll get worn out from doing it, but you have to color the way you do it in some way so that each moment is new. That requires mental toughness, but you can do it and I think we are [doing it]. I was losing things before, because I didn't know how to be painstaking and tough and the people around me, helping me record my early records, were a little bit too much like me. We were going for a jazz attitude: "Hey man, the thing is in the performance. When you get the energy in the performance, that translates and people sense that. That's what we should be going for." Well, I agree, but I think you can attain both.

MR&M: What are your strong points as a performer?

AJ: I think that I have a certain vocal flexibility, but I'm not really sure that you can assign the rest to what's happening vocally. It's really the thinking; what a person has learned as well as experienced in various areas, however broad or limited. There are lots of singers who have the same kind of vocal equipment, but who do things entirely differently. My thinking is really expressive, unafraid and willing to take risks. Also, my feeling is knowledgeable about what I'm doing and where I'm going, which you can only gain from experience. Specifically, I'm interested in music and care about expressing things that are interesting, that really go somewhere, that aren't mundane or commonplace, that allow some room for personal expression. But even if it doesn't, I'll take it [*Laughs*] if it gives me the opportunity to participate rhythmically. I really like to participate rhythmically in music and not float the melody out there like a ballad singer would.

MR&M: Is your inability to play an

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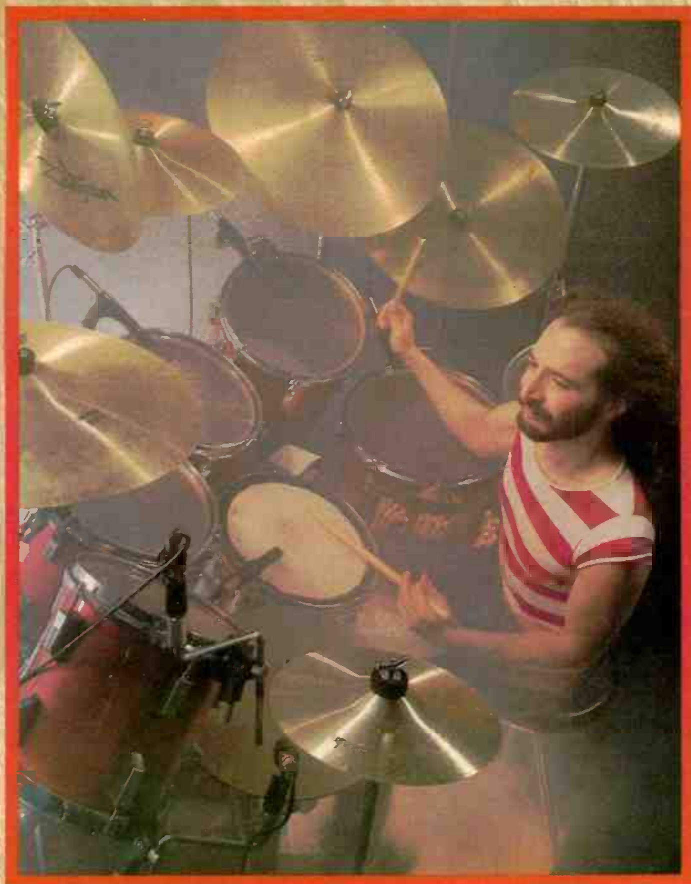
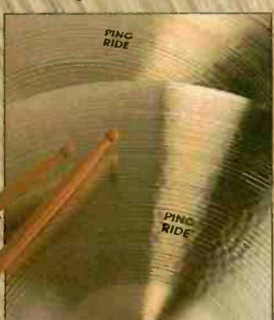


Flying high with the success of Journey, Steve Smith is one of the most versatile and talented drummers in music today.

Steve grew up just around the corner from The Zildjian factory. Of course, for the past few years he hasn't been around all that much, what with his touring with Jean Luc Ponty, Ronnie Montrose and of course the enormously successful group, Journey. However, recently Steve took a break in his wild schedule and had a chance to sit down and talk with us.

On Starting Out. "I started out playing in the fourth grade when I was nine years old and had a really good teacher. When I was in high school I got serious about playing and I got a job as a paper boy to save money to buy cymbals. My teacher used to bring me to the Zildjian factory so I could go in and pick out my own set of cymbals."

On Rock and Roll. "After college I had a lot of experience playing jazz and fusion and I had virtually no experience playing rock and roll professionally except for some high school rock things. I really wanted to follow that direction, because nowadays a drummer has to play rock and roll as well as jazz in order to be well-rounded as a musician."



On Zildjians. "The kind of music we play with Journey demands a lot of power. I've found that the cymbals in the Zildjian rock line are the only ones that can really do the job for me—that can carry the big halls and not sound thin. Zildjian cymbals have extraordinary projection but at the same time they have this wonderful, full musical tone. I also particularly like the Ping Ride—I got my first one back in the eighth grade and I've been playing one ever since."

On Career. "You know if you should get into music. It's something you can just feel. If you have to ask yourself the question, then don't bother. Being a musician isn't just a career it's a way of life.

"I find that most successful musicians don't think about success as much as they think about being a good player or songwriter. To try to focus on success is a little too contrived and usually just doesn't work."

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instrument a detriment to your songwriting?

AJ: Oh, it definitely is. I think I'd be a better writer if I were a player too. However, I think that when I do get off my duff and learn an instrument, I'll be better for having learned how to write without that instrument. I don't think I'll be as confined by the instrument as some people find themselves.

MR&M: Any particular instrument that you would like to learn how to play?

AJ: It would be piano. That's probably the instrument that the majority of music is written on.

MR&M: Since you are unable to play an instrument, what is your songwriting process like?

AJ: I hear a line in my head that feels good to me. Maybe I'll go around singing that line for 15 or 20 minutes or maybe for most of the day, but usually by the time I've sung it for 5 or 10 minutes, I like it pretty much. I'll go and turn on the machine (tape recorder), and I'll just put that line down. Usually what happens, or what I like to happen, is to match a line of that sort with a lyric that feels good for that line. I don't care what the lyric is 'cause I'll work with it and take it where I want it to go. I'll just use it as a guide and maybe change it a little bit. I'll get together with Tom [Canning] and he will actually write it out and sit down to the piano. That will be the first time that I actually hear the chords played that I'm hearing in my head.

MR&M: So your voice is your instrument?

AJ: Mmmmmm. My voice in my head.

MR&M: You seem to be doing a great deal of writing as of late with Tom Canning and Jay Graydon.

AJ: Working with other people is basically new for me. Tom and I didn't start working together until after *Look to the Rainbow*, which was recorded "live" in Europe. Working as a threesome began with *Breakin' Away*. Up until those points, I was really a solitary writer.

MR&M: What are your strong points as a songwriter?

AJ: I'm not sure. Probably, that I really like working with lyrics. I like working with verbal ideas. I have strength there. I think my background as an improvisational singer has really opened me up to melodies, lots of melodies and how melodies can go. Improvising requires that you have that. I've learned some of that along the way and that's important as far as writing is concerned.

MR&M: What are your weaknesses?

AJ: I think my weaknesses are that I don't know an instrument, 'cause with an instrument, what you learn is many varied combinations of directions that a chord can go in. Although you can hear that, respond to it and sing it—later, after it's been structured—it's a little more difficult to think of all those things. If you have the piano at your fingertips, you can play the various possibilities of how a chord should be played and what the next chord should be.

MR&M: Is there a message in your music?

AJ: Yes, about the goodness and potential of the human being and that we really do have a certain amount of control that we can exercise in our lives to make life a good, beautiful thing.

* * *

MR&M: You were originally a rehabilitation counselor before you went into music fulltime. Over the years, did you ever think of quitting and returning to your former job?

AJ: Absolutely never. I would do music at Howard Johnson's if Warner Brothers and all the other record companies decided to make shoes instead of records. I want to do music. It wasn't like a decision to go and get into music. I had two careers. I was going to school during the day... when I finished school I worked as a counselor during the day, and at night I was singing. It was like I was riding two horses. Well, I just got off that second horse and went straight at the music and stopped getting up at eight o'clock in the morning. I *loved* it!

MR&M: Who were your biggest influences?

AJ: Probably the biggest was Les Czimber, a Hungarian man, who came over during the revolution, in '56 or '57, settled in Milwaukee, took me under his wing and started teaching me to really listen to the jazz players who were really happening; the cats to listen to and think like. My older brothers and sisters who brought basically jazz music into the house and provided my early listening experiences, that foundation. Probably the biggest influences as singers—cats who were real professionals—were Johnny Mathis and Jon Hendricks. They really represent two extremes. Johnny Mathis was a jazz singer during his early years, but he developed into a ballad singer, a real romantic singer. So, him for the romance and Hendricks for the energy, jazz technique, scatting lines and improvisation.

MR&M: Did scatting come naturally, or did you have to master it?

AJ: You have to want to and you have to try it for a long time. It's largely the question of musicality. Where does that come from? I don't know. I think you come here [are born] with some of it and I think you develop it through listening, playing or singing. You do have a sense of value of certain notes; how to get to them, how they work in a chord meter. You either know that from technical studies or you've listened so much that you've felt the effect of a certain chord placed against a background of certain chord changes. You develop the facility to get to those notes in any particular sequence or in any written pattern. That's what you go for and you get better and better at it. I thought that after I got through learning the piano, or maybe even before, I would go and study with a horn player. That would really help my scatting. That's how cats learn.

MR&M: Is there a key you prefer to sing in?

AJ: As long as the key is not too high or too low, why do I care? The song will move around, whatever the case. I like to alter the melody, improvise and make it different every time I come to the same passage. But I am partial to a certain kind of instrumentation. When I'm on the road, I use two keyboardists and one bass player.

MR&M: What type of keyboards?

AJ: One guy plays an acoustic piano and he also plays a Rhodes, a synthesizer and a string machine, which is also capable of other things. The other piano player has a similar setup: a Rhodes and a string machine, but a more complete synthesizer system for various sounds and effects. The basic keyboardist has the acoustic piano and the other guy is the secondary keyboardist.

Jay Graydon: We have a Yamaha C7 in the studio, the B series, I think. We use the Fender Rhodes a lot as well.

MR&M: What about the bass?

AJ: For bass and drums [for a rhythm section], I'm using a percussionist. I've been doing that for two years now. But I've never used a guitar player on stage. Maybe sometime I will, but I like how it makes the sound of the group a little different.

MR&M: What type of amps?

JG: None. Only for guitar overdubs

in the studio. I have a Fender Deluxe. But on Al's records, it's mostly direct guitars.

MR&M: Do you use a lot of overdubbing and special effects in the studio?

JG: No, we really don't. We hardly ever use special effects. Sometimes we double Al's voice with a Harmonizer™, with some tricks, some random tricks. It's not the random that's built into the Harmonizer™. We do it with a synthesizer. But we haven't even done that as much this time. We'll probably do it on a few tracks, but basically we don't have a lot of effects on Al's records. It's pretty much straightahead.

MR&M: Can you detail the studio?

AJ: [*Laughs*] This is my first album in this studio. Except for the basic tracks, which we did in a commercial studio, we did *Breakin' Away* in a home studio. We did the entire album in a room in this house. We worked in a spare bedroom with cubbyholes with guitar amps and guitar cases and radio-operated airplanes hanging there. Someone was sleeping nearby in the bedroom. Eight or ten steps from the bedroom door and you were in the control room. Out in the hall and up three stairs was another bedroom which was where the musicians played and I sang. Unbelievable.

MR&M: Any acoustical problems working in a house?

AJ: Well, you can basically do the same things to a house. You don't have to have a vaulted ceiling or that kind of thing. If you're recording strings with a nine-piece orchestra, you do need that spacious kind of sound, 'cause that does affect the phonics. But if you're recording basic things, you can do it. They just soundproofed it, added insulation.

MR&M: Can you describe the new studio?

AJ: It's not just a home studio, it's the real thing. It's top-of-the-line, state-of-the-art equipment chosen by someone [Graydon] who knows. But he basically supervised the construction of the building and bringing in all the stuff that went into the studio.

JG: It's MCI, a 24-track, the new one without transformers and an MCI JH 528 console with automation. It's 28-in, a 500 series, with automation.

MR&M: Can you detail the new building's acoustics?

JG: The whole building is about 2,000 feet square with 16-foot ceilings. We have a "live" chamber as well as a bunch of electronic chambers. The reason we have a 'live' chamber is because you have to have a bathroom in a building. When we built the building, there was no bathroom, so we made it double purpose. We figured, that if we were going to have to put a bathroom in, we might as well make an echo chamber.

MR&M: Jay, is there anything you like in particular about the recording equipment you chose?

JG: It's hard to describe. It's just a working tool. All consoles sound different, but this just happened to be in my price range when I bought.

MR&M: What are the different mics you're using?

JG: We've used different mics on each album. The last album was a [Neumann] U47 tube. On this album we're basically using a [AKG] 414 [Sony] 48 and a Telefunken 251. We did use the U47 on one song.

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MR&M: How are you laying down the basic tracks?

JG: We basically cut with a trio, sometimes four players; piano, bass, drums and maybe guitar on the basic track. But we usually cut with a trio. The musicians are [Steve] Gadd, (drums); Abe [Laboriel], bass; and the piano player will vary on whether the player wrote the tune with us or not.

AJ: What we do is bring in the basic rhythm section: piano, bass and drums, and sometimes a guitar. I go in and work with these guys on what we want the thing to sound and feel like. I get in the booth while they're out in the studio and we do it. We do it until we have a real good-sounding rhythm track. That will not be my finished vocal. It's a work vocal for them, so they get a chance to interact with me and communicate. What we get out of them is a rhythm track that's been played with me. So it's inspired by interacting together. Then I go in and do the vocal later. Then the process goes to the sweetening, to the next element that we think is important.

MR&M: Does the "sweetening" entail a great deal of technical adjustment?

JG: Well, I'm just real careful when it comes to mixing. I'm real careful with getting the vocal even and just trying to get the drums up front if it [the song] calls for it. Each tune is different in that area. Basically, we don't do any special, technical stuff. The tunes pretty much speak for themselves. We're trying to leave the stuff as simple as it can take it. We're trying to overkill with overdubs. Some tunes call for that and get it. But we're trying to be sparse with the tunes that don't call for it, which means that during performances we're more careful to get real good-feeling tracks, instead of just okay tracks, so they don't call to add a lot of stuff. We work carefully and spend time on detail. There's a lot of pre-thought.

MR&M: Can you detail any tracks on the album?

AJ: There's a tune written by Richard Page and Steve George of Pages. The title is Swahili for "I will be here for you." It's a middle grooving tune with a nice rhythm. A very pretty tune.

MR&M: Anything unusual about how it was recorded?

AJ: I'm doing all the vocals and there are pretty extensive background vocals. It was really intense. It took hours.

MR&M: Because of the language?

AJ: No, because of the particular structure of the song and what it required vocally; singing the chorus line many times in an octave above the melody, a falsetto and trying to match the melody that I sang in laying down the basic vocal. When we do that, we usually do four voices for background voices. So, any time that chorus came up, there were five of me. Matching that all in that high octave and getting the intonations and inflections just right is a very painstaking procedure. By the time we were finished with it, we were very happy to be done working on that song.

MR&M: Any other tracks?

AJ: "Surrender Love to Love Today," a nice song. It has a kind of new Brazilian feel, some kind of contemporary Brazilian bordering on the bossa samba. It's basically a tune by one of the piano players whom I've worked with on and off over the past few years, but Tom, Jay and I rounded it off and I wrote the lyrics for it.

MR&M: Any scat material on this LP?

AJ: I did one of the best scat solos that I've ever done on a tune called, "I'm Calling."

MR&M: Why do you feel it was one of your best scat solos?

AJ: I feel that it satisfied the criteria. First of all, the criteria prescribed by the song's feel—what the song needed. And then basically, just as a scat, or an improvised, solo. It contained melody and rhythm feel. I just think that it covered the ground that I like a vocal scat solo to cover.

PROFILE: PRODUCER TEO MACERO

By Gene Kalbacher

Meeting Teo Macero for the first time, one might mistake him for a truck driver or a construction worker. That's not to say he's shabbily dressed, only that his working-class demeanor is utterly devoid of pretension. Sitting on the couch in his mid-Manhattan office, this roly-poly character engages visitors with his anecdotes, occasional off-color language and alternately gruff and congenial manner.

Then one sees the numerous gold and platinum disks that adorn his walls, disks he produced for such superstars as Miles Davis, Duke Ellington, Thelonious Monk, Dave Brubeck and Johnny Mathis, to name but a few. His appearance perhaps to the contrary, Teo Macero is arguably the most accomplished and versatile producer in modern music.

But to call him simply a producer is insufficient. A classically trained graduate of Juilliard, Macero is a visionary, yet often overlooked, alto saxophonist-composer who works as comfortably with a small jazz combo as he does with the London Philharmonic.

Besides being one of the first jazz composers to win a Guggenheim grant, he has earned the friendship and respect of such classical giants as Leonard Bernstein and Aaron Copland. Moreover, Macero's 1955 record, Time Plus Seven *(reissued in 1979 by Atlantic Records), has been acclaimed as the forerunner of the "Third Stream," a genre that marries the improvisation of jazz with the structure of classical music.*

Although his technical skills—mixing, engineering and editing—are as highly developed as his musical abilities, Macero emphasizes people *skills, the artist-producer bond that is essential to a successful recording. A longtime staff producer at CBS before becoming an independent in the mid-70s, Macero supervised the mixing and editing, beginning in 1959, of the first 200 long-playing albums for the company.*

The following interview, conducted in two sessions over the past half-year, begins with trumpeter Miles Davis, whose Macero-produced "live" album [We Want Miles (CBS-C2 38005)] *has recently been released. Nobody speaks for Miles Davis but Miles Davis (when he speaks to the press at all, which is seldom), but Macero, by dint of his quarter-century tenure with the trumpeter, can speak* about *Davis with more insight and authority than anybody in the business.*

Modern Recording & Music: A number of critics contend that *The Man with the Horn* is an extension of Miles's earlier jazz-rock fusion concepts. Do you agree? Does he?

Teo Macero: He doesn't and neither do I. The critics who review this record, like so many people, have to have a frame of reference. And if something is familiar to them, even the least little bit, they say it's a repeat. It's *not* a repeat; it's not a continuation—it's a total departure.

MR&M: If one listens to the record as is, there seem to be a lot of new, innovative things from the standpoint of the tonality, the rhythm, the structure of the pieces. There's nothing like it on the market today. There's a similarity but there isn't a similarity. The similarity is Miles. There's a different playing concept on the record.

TM: He does little throwbacks, which is fine, but always with a little new twist. He does a little blues, a little bossa nova, a little *Sketches of Spain.* It's like a guy who paints a picture. He uses the same colors, but it's a different picture. the critics hear the same colors, so therefore they think it's the same picture. It's not the same picture.

Musically, he stops, he starts, he spurts, but in context it's a very harmonious collection of pieces. The compositions are much different; they're not like any earlier things, not even like *Jack Johnson.* Miles views it the same way I do.

MR&M: A frequent criticism of *The Man with the Horn,* as well as Miles's comeback shows at Avery Fisher Hall, has been the electric guitar playing. Barry Finnerty's power-chording on "Back Seat Betty" reminds me of Eddie Van Halen trying to sound like Dave Davies. It's okay for what it is, but it sounds, to use your word, out of "context."

TM: ...Like all new players with Miles, they will grow. I remember [guitarist, John] McLaughlin when he was with Miles the first time. There wasn't anything happening. He knew it. It wasn't any great shakes.

You know that thing [on *Bitches Brew*] called "John McLaughlin"? It was a *throwaway.* It was a throwaway take and I found it. I said, "We need a little something in here to fill up the album." It was just a little vignette.

MR&M: How would you describe your working relationship with Miles in the recording studio?

TM: When Miles plays, I listen. Sometimes I'll say, I'll use that later on, but maybe in some other place." He may tape 10 or 15 takes, and I tape everything. Something he did on take one may be put in the middle of the last take. The musical concepts are Miles's. I certainly can't make the thing happen unless there's something there to make happen. If he didn't play it, I couldn't put it in.

MR&M: Prior to last year, Miles hadn't recorded an album in seven years, nor performed in five. Suspicion abounded that he had lost his chops. Were you in close contact with Miles during his period of inactivity?

TM: I was with him *all* the time for the last five years.

MR&M: Some have presumed that you exert a Svengali influence over Miles.

TM: *Me?* Not according to him. [*Laughs*] He says he made *me.* Anybody who has any influence over Miles is Miles. I may have a personal influence, maybe an emotional influence, over musically what he does but...

I've been a friend of his for 20 some years. Sure he was sick, sure he had [hip and throat] operations. But everybody goes through that. Everybody ends up in the hospital sooner or later, right? If you're lucky, you don't go; if you're unlucky, you go.

I remember when he came back from his trip to Japan [in 1974]. He was *tired*—physically, mentally. I never thought he was going to make it back. He plays with such energy and fire and enthusiasm...he gives *everything.* When he plays for an hour [as he did at the Kool Jazz Festival in New York City in 1981], the audience thinks he should play for *two* hours. Well, man, when you're running at 10 *miles* per hour for any length of time, you're going to *collapse.* The man is absolutely *fatigued* [when he leaves the stage]. He just has to recharge his battery.

MR&M: From listening to the album, it's obvious that he hasn't lost his chops.

TM: He never did lose his touch. It's a funny thing, I don't have to play my horn for five years, either. And I'd be a son of a bitch if I couldn't pick it up and play it. My lips may be a little sore, and I may have a little less technique, but given a couple weeks, you're almost back to where you were before. But mentally, you're in a different place.

I knew when he came back [from Japan] that he'd have to retire for a little while. All great men in history have withdrawn for a period of time, and Miles is no exception.

MR&M: Besides the length of his Avery Fisher concerts, Miles was also criticized, mostly by concertgoers seeing him for the first time, for not facing the audience when he played. Is it shyness? Aloofness? Or is he just working closely with the band?

TM: He's shy, but he isn't aloof. He was working very closely with the band. Miles's conception of playing has always been to be right there with the guys. You may not be aware of it, but when they're playing, he's over there telling them what to do. Sometimes he'll take the saxophone out of the guy's mouth, or put his hand on the guitar, and tell the guy to stop. He's conducting the band.

MR&M: His direction is so absolute that with a simple wave of the hand, all the music stops.

TM: If you listen to his music and listen to him play, you'll notice that nothing is on the beat. You could compare him to the painter Jackson Pollack throwing colors on the canvas. All of a sudden, there it is. That business of starting and stopping—a lot of guys are doing that today. Miles did that many years ago. It's just Miles conducting, that's all.

MR&M: How was the recording of *The Man with the Horn* different, if at all, from the thirty other recordings you've made with Miles?

TM: There were a few overdubs [on "Shout" and the title track, specifically], but normally he doesn't do that. We don't like to do it, but we do it if we have to. He isn't one for overdubbing. He wants to do it all "live."

MR&M: One gets the impression that "The Man with the Horn," the album's one vocal and title track, was written about Miles, for Miles, and as such was put on the album. Is that so?

TM: It was. But it wasn't put on the record for that reason. Randy Hall [composer of the piece] was the guy with [Davis's nephew] Vince Wilburn's band. The song [originally] was done for a vocal album.

MR&M: The title track contrasts markedly with the tone and substance of the rest of the record. Was it indeed intended for another record?

TM: It was part of an earlier album. I don't know whether it was Miles's decision or mine, but we all liked the song; we liked the way it was done; and we felt it was a great title.

As for it being different from the other tracks, *thank God!* If an album has some contrast, that's what makes it interesting. It does fit the album. It takes you [the listener] out of something and puts you into another area for a short period of time.

A lot of critics say, "Well, it doesn't belong." If you listen to it five years from now, you'll understand that it was right for the album. An album like this needs some release. An album that's always intense can sometimes be a little boring. And I don't think this album is boring at all. I don't think any of Miles's albums have been boring. Miles is a believer in tension-and-relaxation.

Let me tell you about *In a Silent Way* [released in 1969], because it may help you to understand what we're trying to do today. *In a Silent Way* was a gigantic record of about nine or ten reels of master tapes from different sessions. I mixed it all and boiled it all down to five or six reels.

I kept cutting it back. Meantime, the record was still three or four reels, and we only wanted to do one record. These were all pieces that had to connect.

You see, with Miles, we never throw anything away. Everything Miles does in the studio is put on a tape and mixed. Everything is laid end to end. We go through the tape and condense it.

So one day I called Miles and said, "Look, the record is too goddamn long, and I've made all the cuts I'm going to make without you. You've got to get yourself down here, and you and I will finish the goddamn thing." I said, "I know what to do, but I think you should be here."

So we cut each side down to 8½ minutes. Miles, as he was walking out the door, said, "That's the record." I said, "Miles, if you were ridiculous before, you're *insane* now." [*Laughs heartily at the recollection*] I said, "First of all, I'll look for another job tomorrow. Secondly, we can't put out a record with 8½ minutes on one side and nine on the other. Your fans, paying five or six dollars, are gonna scream. They're gonna say, *'Cheats! Robbers!'* We can't do that.

So Miles turned to me and said, "I know what you're gonna do." [*Mimics the trumpeter's hoarse, hacking voice*] I said, "You'd better believe it!" I said, "I'll see you in two days."

I took the two records and listened to them over and over again. Then I told the engineer to start copying them. I had it copied four times and I had 32 minutes. Then I started cutting it back. If you listen to the record, you'll see that it's basically the same material [*Laughs*] for the 8½ minutes that it is for the 16½ minutes.

The first section gives you the clue, because it stops. It's like a little pivot. Then it picks up later on, then it picks up again, and again. And another little phrase is repeated and repeated. I smoothed it all out and had one side done. Then I did the same thing for the other side. Then I sent it up to Miles. I said, "This is what I want to do."

I don't think it was dishonest. It was there. It was just another way of making the point so obvious. So many musicians have used this as a rule of thumb for a lot of records. The starts and stops he does today could've been affected by that. I don't know. I never ask those questions.

I talk to Miles about a lot of things, but I don't talk about yesterday's records. Yesterday's news is over and done with. But the *technique* of putting it together, like a good editor, is like hitting the same eight notes over and over [*Punches his fist into his palm*] until you drive it home that this is a great record.

MR&M: In the liner notes to *In a Silent Way*, the writer says: "...it will probably be five or ten years before they really understand his creativity, his compositions, his mastery of musicianship." Now, about 13 years later, do present-day musicians understand what Miles was doing on that record?

TM: I think so. I think they finally realized that here's a man with a tremendous amount of talent. The producer can only do so much. If the material isn't there, there's nothing he can do. If a clever editor doesn't have a musical mind, and he doesn't understand the artist, forget about it.

There are a lot of clever editors who go in and mix it, and leave all the bullshit in there. We don't do that. There's no bullshit on Miles's records. Listen to Miles's old records and you'll hear no bad solos, at least none that I can remember. If there was a bad solo it was edited out.

MR&M: How closely does Miles work with you on the editing? From what you've said about *In a Silent Way*, it would appear that he trusts you implicitly.

TM: He leaves it all up to me.

MR&M: And he's always satisfied with the end result?

TM: Well, [*Laughs*] I wouldn't say that. I haven't yet done him in. I probably could have, I guess, if I were a lousy, stupid editor. Whatever is there I try to make a little better.

MR&M: When did your association with Miles Davis begin?

TM: I worked on a record with [producer] George Avakian called "Sweet Sue" [in 1955]. I was working on Leonard Bernstein's *What is Jazz*. I worked on [Davis's] *'Round About Midnight* and I did all the editing on *Porgy and Bess* with Gil [Evans]. I did half of *Kind of Blue*, but in those days I don't think the producer's credit was listed on the record.

Kind of Blue is a good record. "All Blues" and "Flamenco Sketches," I think, are the wrong titles on the wrong pieces. I think it should've been the other way around. At the time we put it out, I remember Miles calling me up and saying, "You f----- up! You got the titles wrong. 'Flamenco Sketches' is 'All Blues' and 'All Blues' is 'Flamenco Sketches.'"

MR&M: Was that ever corrected?

TM: I don't know if it was.

MR&M: What was the first record you produced for Columbia?

TM: [Dave] Brubeck's *Gone with the Wind* was actually the first. Then I got Miles, Duke Ellington, Thelonious Monk. Before long I had twenty artists, so I had a big expense account because I was entertaining everybody. It was great.

MR&M: Have you ever performed with Miles?

TM: Never.

MR&M: Why? Did you ever want to? Has he ever performed any of your compositions?

TM: Oh yeah. He's played my compositions. He thinks I'm a great saxophone player. I guess I wanted to [make music with him], but I never impose my own thing on anybody.

MR&M: Now that *The Man With the Horn* has been out for a while, how does Miles view it? Is he satisfied with it?

TM: I don't know if he really likes it. He may like it. There are a few records he's liked.

MR&M: Such as?

TM: He liked *Jack Johnson, Pangaea* and *Agharta* [the latter two released in the mid-70s]. I think he's pleased with the new record, but like a true creative artist, he's never really satisfied. That's why it's so difficult to work with some of these great people.

It's probably the first impact that's most difficult to take. When you haven't been on stage for a long time, you don't even know if you're dressed properly, for crissakes; you think you're going out nude. You wonder, "Is everything covered?" It's the same with making a record.

MR&M: Getting back to Miles's stop-start playing technique, it seems like his career has gone the same way. He's had periods of tremendous creative bursts followed by periods of entrenchment, and then, sometimes, detachment and withdrawal.

TM: I really don't have an answer for that. I never dwell on those things. He's always conscious of what he's doing.

He does things spontaneously. That's what artistry is all about. There's 20 percent inspiration and 80 percent work. Sometimes it takes a long time to get inspired in this business.

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MR&M: Speaking of inspiration and inspired music, over the years you've produced numerous giants of music—Davis, Duke, Monk, Charlie Mingus. Those four must be considered the most innovative song structuralists in modern music.

TM: Look at the four giants—Miles, Mingus, Duke and Monk. Even Charlie Parker wasn't as creative in terms of songs as these four guys. He used other songs and adapted other melodies. They did it with "Cherokee" and "Hot House," which was [based upon] "What is This Thing Called Love?" The lines are very creative, but they took the basic chord structure from something else.

So did Dizzy Gillespie. Dizzy did a few original things, but not many. It would be interesting to go back and see just how many *original* compositions Charlie and Dizzy wrote. I'm not putting them down, I love them both, but I think Miles, Mingus, Ellington and Monk really were writing music, besides playing. Mingus, Miles, Duke and Monk took the raw material and made their own things.

MR&M: In terms of their recording needs, how were they similar or different?

TM: All of them had a genius for creating, for not standing still. That's the first thing. Mingus, Miles and the others were striving to do something different all the time.

The Lounge Lizards and Kip Hanrahan [punk-jazz musicans with whom Macero has recently worked] are trying to do something different. But with established jazz musicians, they're often doing [standards such as] "Autumn in New York" instead of a stretched-out piece. Stan Getz stretches out from time to time; he'll play with a string quartet or a symphony orchestra. I think that's what they should do; they should really branch out.

MR&M: Speaking of branching out, one of your most interesting productions of late has been the Lounge Lizards, who covered such Monk tunes as "Epistrophy" and "Well You Needn't." What attracted you to this group?

TM: It's a record with a sense of humor. Monk always had a sense of humor. They brought their own thing to it, their own crunch chords and everything. It's so unlike them, and so unlike the original [Monk] records, that I thought it was delightful. It's hard to find those [Monk] melodies in there... I myself was taken aback.

These kids have really got something. If they stick together long enough, they're really going to make it big.

MR&M: On an individual basis, what were the specific recording needs of the four great structuralists you mentioned? Did one need more direction in the studio than another?

TM: I always let the artist do all the work. I let them lay it down. Whatever they want to do, they do. I help them in any way I can. In Mingus's particular case, I used to go out and conduct the band. I might sit and play.

With Duke, I did the same thing. I conducted the band. He'd say, "It's *your* band! You take it over and do whatever you want to do." I almost froze to death. I didn't know what the hell to do with those guys. But I took the pencil, went out there and started to work with them.

MR&M: What made Duke say that?

TM: We were doing a single. I said to Edward [Ellington's Christian name], "Look it, this piece is seven pages long. We don't have a 45 that will play for seven pages. Can we tighten it up? Can we make it like a single?"

I don't know if he appreciated it or not, but one night he said, "Here it is, it's your band. Cut it up any way you like." So I did. I think that particular piece was "The Asphalt Jungle."

Mingus was the same way. He'd say, "You want to fix it, fix it." Sometimes it's very helpful if there's another pair of ears that's in tune with what they're doing. I always appreciate if some guy helps me. Miles is the same way.

MR&M: Obviously, the fact that you yourself are a composer and musician must have been a great advantage.

TM: I played with all these people, so they knew I wasn't there to jive them. I always, *always* consulted with the artist. I *never* took something on my own and said, "Too bad, I'm doing what I want to do." I might mix something and say to the artist, "Is this a possibility? Do you hear it this way?" "They'd say yes or no; if they said no, I'd pass it by.

Every one of those guys knew I could write, knew I could play. I used to write for Mingus, and Mingus and I used to play together. Monk the same way. I think it really helps. But if you're not careful, you can go the other way and become very domineering and implant your wishes on them. I try not to do that. I try to be as objective as possible, so the end result is *theirs*, not mine.

The only way to get a topnotch performance sometimes is to force somebody to do just a little better, maybe one more take. I remember the time I wanted Monk to do something over again. He fought the bejesus with me; he didn't want to do it again. I said, "Thelonious, if *you* want it, that's it. I'm happy with it, but I think you can make it better." He came back later and did it again.

I've never been satisfied with one performance. If I have the luxury to do it again, I do it again.

I wouldn't say that we always got along 100 percent. I'm sure they got angry at me, and I got angry at them. Nothing in the studio is personal. It's not a personal thing between the artist and myself. The personalities don't come into play; the music is the first concern.

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MR&M: Although, as you say, Teo,

nothing in the studio is personal, obviously the personal, human touch is essential. People make music and people have problems. As for the studio itself, what do you look for in a prospective room?

TM: I've used different studios over the years, and I check the personalities out, find out what kind of people they have. That's very important, especially with engineers.

Sometimes when you get into a new studio, you find a hotshot engineer who's going to push you aside. We don't want that. We want a guy who's going to work with us, who's not going to snowball us or the artist, and jump in and take over your job.

If you get a three-way situation with the the artist, the engineer and the producer, it becomes very awkward. You try to get guys who are compatible with you. You want three, not two and one. And you have to make sure the equipment is good.

MR&M: Before you produced your first record, Teo, did you have any role models, and other producers whose work you respected and wished to emulate?

TM: No, I think I approached producing a record as if it were my own record. As an artist, how would I like that record to sound, if it were me out there?

MR&M: Earlier in our conversation you mentioned the importance of editing out bad solos. Have you ever run into the problem of other people, other producers, going into the archives and tampering with material you edited and mixed years before?

TM: *They can!* I hate to have people touch my things. I know they're doing this now with Duke Ellington. I said, "Please let me do it!" Okay, they may have to pay me a few dollars, but I made it [originally], I know what's there. And some of the stuff you can't put out as is, because it makes the artist look like an idiot.

The Mingus album [*Nostalgia in Times Square*] is a big f----- joke to me. They went back and restored the solos Mingus and I *cut out!* They said, "It's great! It's historical!" I said, "That's bullshit!" I can't even listen to most of it. The guy [Mingus] was just learning it [the music].

Can you imagine what would've happened if we'd put out *Sketches of Spain*, for instance, without any editing? I would've shuddered! There were fifteen sessions for that particular record, and it took us a long time to edit it.

MR&M: Teo, in line with your work on the upcoming series of Monk packages, didn't you once propose a thematic series of Miles material that was recorded but never released?

TM: Yes. It's not fair to the artist, after he dies, to put the stuff out. The artist really deserves to be a part of it. I've always said to the companies through the years: "Please, the artist is alive. He may have some insights that I [the producer] don't have. The artist's insight is *better* than mine."

MR&M: So what you're saying is that the record company should take stock of its holdings and consult the artist and producer as to eventual, possible releases.

TM: Right. I don't believe in putting all these things out posthumously. There are things in the can with Brubeck, for instance, that have never come out. One reason they didn't come out is because he had too many hit records going for him all at once.

We did two albums with clarinet and the trio that never came out. It's *fabulous* stuff! We did a large thing at Tanglewood that was fabulous—probably the best drum solo [Joe] Morello ever made. I remember that. They should go back to Brubeck and say, "Look, we have some stuff here. Let's put it in some respectable form."

MR&M: Do you have any secret fears that somebody somewhere will go through the can and tamper with all the great, unreleased music you've made over the years? Do you ever lose sleep worrying about that?

TM: It's *happened* to me. The guy who made it should be the guy who puts it together. Let him give some input into the project: "You should cut here, edit there." Okay, so they have to pay me a small honorarium. I'm sure it's happened to a lot of other producers....

What they've done is give it to the critics and various other people to do. I can understand that from the standpoint of sales or whatever, but they should give it to the guy who did it originally. I don't know whether the critics really know how to do that kind of work [editing, mixing]. I don't know whether the critics are capable of editing a solo or taking four bars of this and moving it around. I've done it for 20, 25 years and it's become second nature.

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MR&M: In the mid-70s, after almost 20 years with CBS, you went out on your own as an independent producer. Have you ever regretted that decision?

TM: Being an independent producer is much more fun because I'm able to come and go as I choose. I don't have to put up with corporate nonsense and I can pick and choose what I want to do. I do a lot of writing for television and everything, so I've really got the best of both worlds.

MR&M: As a staff producer, you're reputed to have assisted artists with their contract negotiations and other financial matters. Weren't you, in effect, in the middle between the artist and the record company?

TM: I always took the side of the

artist. I always thought of the artist first. I worked within the company and I thought of the company, too, because they were paying my salary. But if the artist wasn't happy and I wasn't taking care of him, I wouldn't have a job.

MR&M: If you were to start over again as a young producer in the music business, given your talents, do you think it would be easier or harder to succeed in 1982 than it was in the 50s?

TM: It would be easier for me because I would approach it entirely differently. I would, first of all, be a multi-millionaire. There's no question about it. With all the gold records I've made, with all the big hits and everything, I'd have so much goddamn money.

There's fifteen gold records [on the wall]. There's some here and some [in my home] in the country. Plus, there are other records that have sold in the hundreds of thousands over the years. I would have managed two or three of them [artists] and said, "That's it." It would be a little easier today.

* * *

MR&M: Let's turn to your songwriting, a talent which has been underrated, if not overlooked, during your career. How would you describe your writing habits?

TM: I write every day, except the days I drive into New York. I just write for the sake of writing. I just finished a string quartet, and I'm doing a ballet at Juilliard. People ask me to write music all the time, and I do, often for nothing.

I've just embarked on a big project. I'm going to record every piece of music I've ever written—ballets, symphonic pieces, jazz pieces, children's pieces, whatever. I plan to go over to England and do some things with the London Philharmonic. So 80 percent of my pieces have been recorded, while almost 99 percent of them have been performed.

MR&M: You mentioned earlier, Teo, that when you record Miles, you never throw anything away. Does that hold true for your own compositions?

TM: I never throw anything away. I just put it in another bin. Eventually I may come back to it. I try to write something different every time, because you can't paint the same picture. That's why I like Miles. That's why I like Monk and Mingus. Every time they'd do something it was something different. If an artist is predictable, he's a bore.

MR&M: Over the years have you set any goals that, for one reason or another, you've never been able to achieve?

TM: There's one thing I've always wanted to do but never been able to do—record Frank Sinatra.

MR&M: Have you made any overtures?

TM: Everybody I know who knows Sinatra told me he didn't want to be bothered by anybody. They never wanted to offer an introduction. When Sinatra comes to town, everybody runs the other way. I said, "I just want to meet the guy; I don't want to make love to him [*Laughs*]."

Producing Sinatra would be a gas. He's always been my idol anyway. He's the greatest of all the greats. I've met practically everybody in the [music] world, worked with them or seen them socially, but Sinatra's the only guy I've never come in contact with. He's surrounded by all these people. I just want to say hello to him and thank him for who he is and what he's done. My ambition is to do one record with him.

Ambient Sound

By Len Feldman

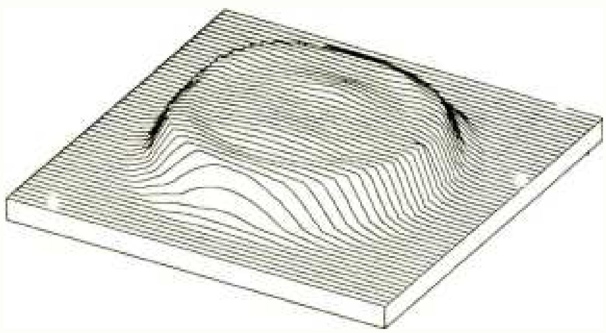
CAD is No Longer a Nasty Word

In my grandfather's day, to call someone a CAD was to imply that he was a bounder (so what's a bounder?), an unscrupulous fellow who mistreats the "gentler sex" and is unfaithful to his wife. Now, CAD has become an acronym with an entirely different meaning. It stands for Computer-Aided Design, and computer programs are taking the drudgery out of technically complex and lengthy design equations in many fields. Our own field—audio and audio engineering—is full of design problems which involve lengthy sets of equations (some twenty-eight are involved in the design of a vented-box speaker enclosure alone), and creating computer programs to solve these equations has not only reduced the amount of repetitive, tedious pencil pushing that speaker design engineers formerly had to go through, but has resulted in much-improved speaker designs besides.

If computers have been a blessing in fundamental design work, they have also contributed significantly to audio equipment measurement, to some of the empirical design work which some speaker system manufacturers still prefer and even to the design of such fundamental speaker components as the drivers and crossover networks themselves. In my own laboratory, the transition to computerization has been very gradual indeed (publications tend to pay by the word, rather than by the time I take in my laboratory to measure a piece of audio equipment I review in print, or in terms of the cost of my capital equipment). Still, while my entire lab may be a long way from computerized, if you have been following the reports prepared by Norman Eisenberg and myself for the past year, you will recall that all of our test reports concerning tape hardware or software have been greatly facilitated by a piece of microprocessor-controlled and programmed test equipment manufactured by Sound Technology. This test equipment has not only substantially reduced the time it takes us to measure the performance of a piece of audio tape equipment under test, but has enabled us to come up with more accurate test results and with new tests which would have been impossible (or too lengthy) to perform "the hard way."

During a recent visit to England, the importance of CAD and CAM (Computer Aided Measurement—my own acronym. If others can make them up, so can I!) was once again brought home to me. Some of the techniques I saw were not totally new to me. One was, and it was particularly fascinating. All were demonstrated at the research and production facilities of B&W Loudspeakers Ltd., located in the town of Worthing, along the Southern coast of England, in Sussex County. As has been stressed many times in these and other pages, it is virtually impossible to obtain meaningful frequency response measurements for a loudspeaker system in the room in which it is to be used using simple sine-wave signals of varying frequency. While relatively accurate response measurements *can* be made in large, anechoic chambers which absorb all reflections, these measurements are really somewhat academic, since no one listens to music in an anechoic chamber.

A system of measurement now employed at B&W (and at KEF as well as at other well-known reputable speaker manufacturing companies in the U.S. and abroad) is the so-called Fast Fourier Transform (FFT) method. In this method, a short-term pulse is used as the signal to be applied to the speaker terminals. A properly positioned, calibrated microphone picks up the sound of the pulse, as reproduced by the loudspeaker system. Viewed on an oscilloscope, for example, the signal amplified from the microphone would tell us little about the actual frequency response of the system. But that signal, analyzed and dissected by a properly programmed computer, provides enough information for an accurate plot of overall frequency response of the speaker system. And because the signal itself is of such short duration, room reflections or room acoustics have little or no influence on the measurement, which can of course be rendered as a "hard copy" printout for permanent storage of data concerning that speaker system. Without computers, none of this would be possible.



Laser vibration interferometer scan of B&W's new TW20 LM tweeter used in their LM1 "Leisure monitor" loudspeaker system. The plot shows the measured amplitude of the motion over the tweeter dome surface for a sinewave input signal at 10 kHz. This graphical representation of the dome behavior was derived from the velocity impulse responses measured at 180 points distributed over the dome surface.

Measuring Speaker Cone Motion with A Laser Beam

A less familiar and perhaps more fascinating measurement technique which I learned about during my visit to B&W is called Laser Interferometry. For the first time it allows design engineers to accurately measure the nature of speaker cone vibrations at high frequencies without having to attach anything to the speaker cone itself, and therefore without modifications that might alter the resultant measurements. While several optical and mechanical methods of investigating the amplitude and phase of the cone vibration of a loudspeaker are available for low and mid frequencies, at higher frequencies cone displacement becomes very small, and therefore cone displacement measurement becomes more difficult as the frequency of the driving signal increases. This difficulty was partly overcome during the 1970s, when some manufacturers began to use holography to record images of cone breakup patterns. In holography measurements of that kind, cone parts that are at rest appear light in the developed hologram plate, while those cone parts which are in motion appear dark. One major drawback of this holographic method is the difficulty in obtaining quantitive data of the actual cone displacement amplitude from the hologram produced in this way. Correct interpretation of cone displacement at any point is, therefore, a function of the observer's personal interpretation of the contrast between light and dark areas surrounding the point in question.

Like the holographic analysis method, the vibration interferometer, developed and built by the Atomic Energy Research Establishment at Harwell, England and purchased by B&W in 1978 also makes use of the interference of laser light. However, it differs considerably from the holographic technique in its principle of operation. In the interferometer, a polarized light beam from a helium-neon laser is passed through an electro-optic frequency-shifting device and a lens and is focused onto the point of interest on the loudspeaker cone's surface. Without getting into too much technical detail, in normal operation of the device, a constant *shift* in frequency of about 5 MHz is applied to a reference light beam. When the cone surface is not in motion, the scattered light and the reference beam differ in frequency by the value of this shifted frequency only (5 MHz). If the cone surface is in motion, frequency of the scattered light is frequency modulated as a function of surface velocity, because of a Doppler effect that occurs with light waves much as it does with sound waves. The modulation of the scattered light produces a corresponding modulation of the 5 MHz beat signal output from a difference-amplifier. This signal is finally converted into a signal voltage which is therefore proportional to the velocity of that point on the cone on which the light beam impinged.

The interferometer approach can measure the motion at any point on a speaker cone's surface as a function of the driving signal applied to the voice coil.

To make a more complete investigation of total cone motion requires that the motion be measured and recorded at a number of discrete frequencies for each of many, many positions or points on the cone's surface. The complexity of cone breakup usually exhibited by a speaker cone (especially one designed for use as a mid-range or high-frequency driver) is so great that the minimum number of points on the surface needed to correctly define cone motion can become very high. That's where a computer comes to the rescue once more.

B&W's PDP-11 computer system, for example, is equipped with data acquisition facilities together with the software required to enable the measurement and storage of time-varying voltages. Both the input voltage to the loudspeaker's voice coil and the velocity output voltage derived from the interferometer laser measurement system are interfaced with the computer system. Simultaneous acquisition of input and output signals is possible at a rate of up to 50,000 samples per *second*. Measurement and storage of the cone breakup pattern by the computer can be done using sine-wave signals and even pulse-type signals of the type described earlier. In addition to measuring and storing the cone vibration data and controlling the precise position of the laser beam's point of illumination on the surface of the cone being analyzed, the computer also processes and combines the data and displays it in a form which can be easily interpreted. A "three-dimensional" type of graph generated by the computer can show the variation of the amplitude of measured cone acceleration along one radius of a loudspeaker cone as a function of frequency. Another type of display (*see illustration*), shows the measured amplitude of the motion over the entire surface of a speaker cone at a given frequency (in this case, B&W's new TW20 LM tweeter used in its latest LM-1 "Leisure Monitor" speaker system, for an input frequency of 10 kHz). A perfectly rigid loudspeaker cone would have the same amplitude of acceleration at all points on the surface when driven by a constant amplitude sinusoidal force for frequencies above fundamental resonance of the system. Being able to "see" any breakup patterns in this unique way must surely have been of tremendous help to the designers of this system and must have reduced the design time enormously.

The increasing use of computers in the design of audio products can be viewed with some concern by those who think about innovation and inventiveness in terms of the solitary, impoverished inventor working in his or her basement or garage workshop. To be sure, the days of that type of inventor may be over. On the other hand, with the prices of mini- and micro-computers coming down as fast as they are, it's possible that before long, the sole-entrepreneur type of inventor will be able to afford a skilled "assistant" in the form of a totally obedient and silent computer after all.

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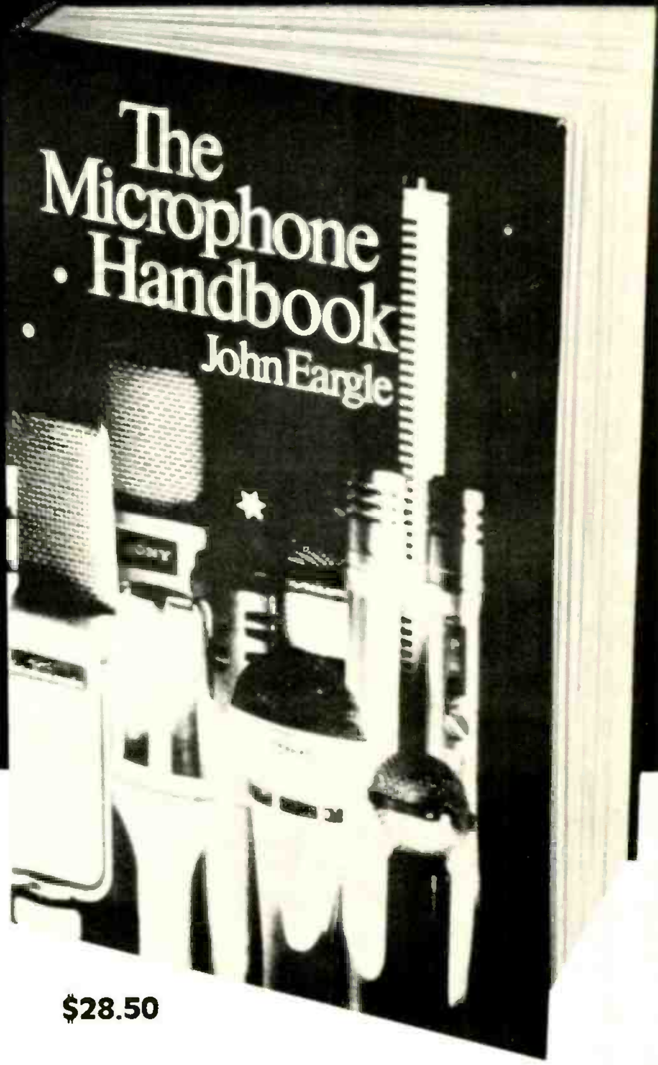

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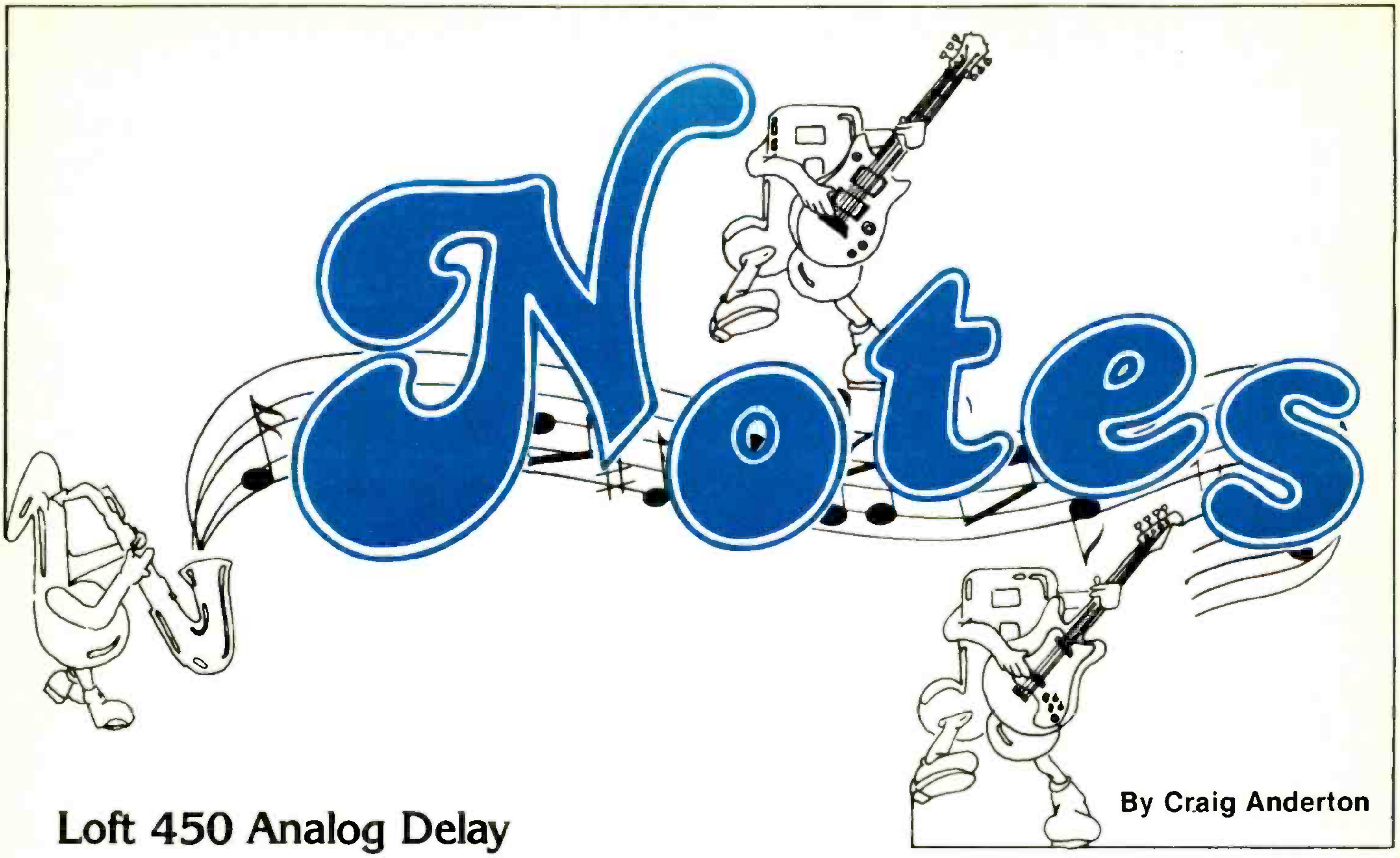


By Craig Anderton

Loft 450 Analog Delay

I bet you think it's easy doing these reviews: I just get some juicy piece of equipment, play with it in the studio for several weeks, and talk about it in print. Simple, right? *Wrong!*

What makes it so hard? Well, I'm not your basic "meter reader," but rather, a musician trying to evaluate a piece of equipment from a musical, rather than technical, standpoint. While there can be a correlation between musical usefulness and technical sophistication, this is by no means a certainty; one of the best examples of what I mean is the fuzz. When fuzzes first came out, I remember engineers scratching their heads and saying "but we've been working for decades to get distortion *out* of amplifiers," while musicians were grooving away to the sounds of 100% distortion. But that's not all—musicians even talk about different types of distortion, such as "warm" distortion, "gritty" distortion, and so on.

That's just one example, you say...okay, here are some more. How about the company that made one of the first phase shifters? When they changed their design to eliminate what they thought was undesirable distortion, sales dropped off until they went back to the original design. What about "pre-CBS" Fenders? And people who prefer a $49 Silvertone guitar they picked up in a pawn shop in Eau Clair to a gold-inlaid Superpicker III? And the folks who use tape saturation as an effect, and those who prefer the sound of a one transistor wa-wa than a state of the art 24 dB/octave synthesizer filter? And lastly, what about that Anderton guy who has to write articles which make sense to the engineers who only know what a scope says, as well as to the musicians who don't know the difference between an Ohm and a Volt—while expressing exclusively musical qualities via the printed page?

These thoughts came into my head as the well-meaning representative of a company (not Loft, incidentally) was trying to convince me his delay was better than virtually every other delay in the world by virtue of its specs. What makes a musician buy a particular piece of equipment is highly subjective, and may or may not involve specs. Those in the audio world try to keep phases matched, yet one of the most popular modifications for guitar is putting the pickups out of phase with respect to each other.

Let's get into this matter of specs a little further. Suppose you have two delays. Both have a dynamic range of 85 dB, but one claims flat response to 10 kHz, the other to 5 kHz. Which one would you choose?

In my opinion, the correct answer is "I would choose whichever one sounded better in a given application." First off, a dynamic range spec doesn't mean a whole lot. Manufacturers often seem to equate dynamic range with a signal-to-noise spec, but that's not entirely accurate and here's why.

Suppose you have an analog delay with a compander. When there's no signal going through the unit, there's virtually no noise, and no noise contributed by the delay line. That sets the bottom of your dynamic range. When you pump lots of signal through, the maximum undistorted signal level sets the top of the dynamic range. This yields an impressive dynamic range figure, but doesn't take into account that the delay line noise often rides along with the signal, getting louder as the signal gets louder. If you put cymbals through the delay line, this noise will probably be unnoticeable. But if you put a bass through the delay line, there isn't a lot of high frequency energy to mask the noise, and the end result could be Niagara Falls rising and falling along with the dynamics of the music.

And do you always want the maximum possible bandwidth? Sometimes yes, sometimes no. For example, I wanted to add a little delay to a drum track to add a sort of double time effect. Using an expensive do-it-all digital delay line gave me a signal which had excellent bandwidth and low noise, but which didn't do the job. Why? Because both drum signals (straight and repeat) were so close sonically that the delayed sound

There can be a correlation between musical usefulness and technical sophistication...this is not a certainty...

stepped on the straight sound. No problem, you say; simply pull back on the delayed sound level—but that still didn't give the effect for which I was looking. While filtering out some of the highs helped, I thought I'd try another device.

So, I put the signal through the Loft 450 at its longest delay. The bandwidth closed way down, there was noticeable noise, and the quality of sound was definitely inferior to the digital delay wonder. But—and this is the moral of the story, I suppose—the sound coming out of the Loft *sounded like an echo in a "live" environment* and was the effect I wanted. [I have a theory as to why listeners often prefer delay lines with restricted bandwidth. When trying to create a feeling of ambience, delay effects usually concentrate exclusively on the time element (some have high-frequency cut switches as well, although sometimes the action of these isn't drastic enough). However, there are also many frequency response alterations which occur to a signal as it travels in an acoustic environment, as well as other possible phenomena which haven't been identified yet. A time delay device which also includes these phenomena in addition to time delay would therefore produce the most "natural" sound.]

The question of *this* column is not specs, and it's not "does Unit A deliver $\frac{1}{4}$ dB more high end or low end than Unit B." Which brings us, in an admittedly roundabout way, to the Loft 450. It's not cheap, the bandwidth is limited at longer delays, it can't delay longer than a third of a second, and it has some limitations. Yet, it also makes some very lovely sounds, and I can see it inspiring loyalty among the same crowd who will never use solid state amps as long as tubes exist...and it's interesting to look at why.

WHAT is IT? The Loft 450 is an analog delay unit which covers a total range of 0.5 ms to 160 ms and lists for $750. An expander module, which adds $125 more to the price, extends the delay to 320 ms (this is the model I reviewed). Probably the easiest way to become familiar with the unit is by investigating the various controls.

Frankly, I'd rather have drama... that's what I use an effect for...

One main pushbutton selects between *flange* (which lights up a corresponding amber LED) and *delay* (which lights up a green LED), while four interlocking pushbuttons select four different delay ranges. In the flange mode, the buttons select ranges of 0.5 to 5 ms, 1 to 10 ms, 1.5 to 15 ms, and 2 to 20 ms. In the delay mode (with the version which does not include the expander module), the buttons select ranges of 4 to 40 ms, 8 to 80 ms, 12 to 120 ms, and 16 to 160 ms. With the expanded version, these ranges become 8 to 80 ms, 16 to 160 ms, 24 to 120 ms, and 32 to 320 ms.

A manual delay control varies the delay over a 10:1 range, within the setting selected above via one of the four interlocking pushbuttons. Note, however, that this control works "backwards" (like the MXR Delay System II we reviewed recently [*see Notes, July 1982 issue*])—in other words, turning the control clockwise lengthens the delay. I hope this isn't a trend; I've sure gotten used to turning knobs clockwise for shorter delays. Anyway, what with the main pushbutton, the row of four pushbuttons, and the manual delay control, you've got delay selection pretty well covered.

A *clock mix* control lets you add in modulation from the Loft 450's built-in LFO (low frequency oscillator). With this control fully counterclockwise (CCW), the delay is controlled totally by the manual delay control; when turned fully clockwise (CW), the delay range is swept by the LFO over a 10:1 range. In between these two points, both the manual control and LFO affect the delay modulation. Note that the more CCW the *clock mix* control, the less effect the modulation has on the delay line.

The *auto* control (which has nothing to do with cars, but rather controls the LFO speed) goes from a properly slow and majestic sweep to a not-quite-fast enough sweep. It doesn't seem like it would be that hard for Loft to design in a little more LFO range; sometimes faster LFO speeds are important (like for vibrato and certain types of rotating speaker simulations).

With respect to input and output controls the Loft 450 does a good job of matching levels. There are balanced XLR connectors as well as ¼" phone jacks for both inputs and outputs. However, I would not feed a guitar directly into the input, since the input impedance isn't really all that high (around 100 K). You'd be better off going through a buffer, preamp, compressor, or other electronic effect first.

The input level control goes from unity gain to +20 dB (fine for handling low level instruments which need a little gain). The maximum input level is specified as +18 dBm, and there are three LEDs to help you set levels. A green LED turns on when you have 20 dB of headroom remaining, an amber LED with 10 dB, and a red LED when 5 dB of headroom remains. I found that signals as low as 1.5 V peak-to-peak could drive the unit to maximum dynamic range with the gain at +20 dB, while I could pump greater than 20V p-p signals into the device at 0 gain.

The output control is an attenuator which can trim back the output should you add a lot of gain at the input, but still need to preserve unity gain throughout the system. The output impedance is under 10 Ohms.

Other controls include a signal mix control, which pans between straight and delayed sounds; a regenerate control, which adds feedback for a more resonant sound; an EQ shift switch, which changes the frequency response of the regenerated signal; a power on-off switch with associated indicator; and finally, an effect bypass pushbutton which activates FET switching inside the unit as well as a red panel LED.

The Loft 450 also has quite a few jacks on the back. In addition to the balanced/unbalanced inputs and outputs, there is another output which generates a synthesized stereo output. This is excellent for "live" use (P.A., etc.), since it spreads the sound out. Remember, though, that if you record with synthesized stereo on two tracks of a tape and then that tape is later played back in mono (say, on AM radio), any time delay effects disappear and you are left only with straight signal. This synthesized stereo output also has ¼" phone unbalanced and XLR balanced connectors.

There is a control voltage input jack which accepts a non-standard control voltage (+1 to +11 VDC). However, it's there if you want it, and making an appropriate interfacing circuit for dealing with standard 0 to +10 V synthesizer control voltages is not difficult if you're into that sort of thing.

There is also an interesting control voltage out jack, which is designed to slave two Loft 450's together. This stereo jack offers two options: a tip voltage which tracks the internal LFO, and a separate ring voltage which produces an inverse control voltage (in other words, when one Loft 450 is at minimum delay, the "slaved" unit will be at maximum delay).

Finally, there's a jack which accepts a footswitch for remote in/out switching.

PRE-FLIGHT for the LOFT 450: This effect is simple to set up, once you've decided which of the input and output jacks to use. Plug into the input, plug the output into your amp (or outputs into your amps for stereo), and you're ready to go. Adjust the input control so that the red LED flashes occasionally, set the output control for a suitable level, and start playing.

EVALUATING the LOFT 450: Let's consider the flanging range first. Like most delays, the longer the delay, the lower the fidelity. Specifically, although the 0.5 to 5 ms range gives response ±3 dB out to 18 kHz, the bandwidth of the 1 to 10 ms range extends to 9 kHz or an octave lower, the 1.5 ms to 15 ms range to 6 kHz, and the 2 to 20 ms range's bandwidth only extends out to 4.5 kHz—not exactly a crispy high end at these delays. Also note that the perceived noise changes with delay. With the manual delay control set for shortest delay, the noise is no problem. But at longer delays, the noise becomes more prominent. Although the effect of the noise is tempered by all that low pass filtering, its presence can be heard riding along with the signal.

Yet—specs don't tell the whole story. One very nice flanging feature of the Loft 450 is the way in which the sweep occurs. Now, this may not seem significant, but there is more than one way to sweep a delay line. With the Loft 450, the sweep seems to "decelerate" as the delay gets longer and "accelerate" as it approaches the high end of the flanging range. This means that the flanging sweep appears to stay longer at the lower, more sonically interesting part of the range, accelerate up to a peak, and decelerate back down to the low end of the flanging range. (Interestingly, the flanger which I have designed for a future *Modern Recording & Music* article also generates this type of sweep, which I feel is musically more desirable). Being able to flange over a 10:1 range—a little over three octaves, as opposed to the one or two octaves offered by most digital types—means you get a more dramatic flanging effect than you would with any digital delay I've seen.

Another unusual feature of the flanging sound involves the EQ shift switch. This filters the feedback path, and produces a really beautiful effect (possibly because there are phase changes which create interesting feedback information, although I'm not sure). When you crank up the regenerate control and switch in the EQ shift, the flanging sound is *very* tasty. However, the Loft 450 does not give you a choice of in-phase or out-of-phase regeneration, something which I feel is highly desirable in a flanger. Also, remember that at longer delays, the bandwidth isn't all that great. So, if you're expecting a real crisp flanging sound on something like drums or lead vocals, you'll probably find yourself reaching for a different box; however, for guitar, most keyboards, and other instruments, this bandwidth limitation—at least to me—is not a serious problem. After all, the straight signal still retains its fidelity, and as mentioned earlier, sometimes limited bandwidth gives a more, not less, realistic sound.

About the delay range...well, I've never felt that analog delay techniques were suitable for delays much over 100 ms or so, and the Loft 450 has not changed my mind. Paradoxically, I liked the sound of the two longer delay ranges better than the two shorter delay ranges; the bandwidth of the longer delays is quite restricted, which really kicks any noise in the background. At shorter delays (with higher bandwidths), the hiss comes through with greater clarity.

OVERALL EVALUATION: The first Loft 450 I got from Loft to review arrived DOA. It didn't take long to figure out the thing wasn't working as intended, so Loft sent me another one (that one worked just fine). The construction certainly looked adequate, and the people at Loft assured me that what happened was a fluke and not at all the general experience of their customers.

It's difficult to give the unit itself an overall rating, because I think some musicians (particularly those who are interested in flanging/chorusing) would love it, while other musicians who are mostly concerned with high bandwidth, long delays would not be impressed at all. And there are limitations: although you have a great flanging sweep and that nice EQ shift option, there's still no way to select between positive and negative flanging. Also, the delayed signal is out of phase with the straight signal—which I feel is the wrong way to go, although others don't always agree and this is probably a matter of taste.

The flanging sounds much better to me than digital units in terms of drama, but inferior in terms of fidelity. Frankly, I'd rather have drama since that's what I'm using a special effect for in the first place. Some of the chorusing effects towards the lower end of the flanging range are also extremely full-sounding and warm. With respect to the inspiration factor—does playing the thing get me off—the Loft 450 brought several smiles to my face as I fooled around with the flanging and chorusing sounds.

The delay range doesn't knock me out; I've gotten spoiled by digital delays which give long echoes with good bandwidth. Yet even so, I'm sure those musicians who like the "warmth" of analog delays might find these echoes not at all objectionable when mixed in the background (which is where echoes usually end up anyway).

I also feel that the Loft 450 might have a bit of an identity crisis by wanting to be all things to all people. It flanges, it delays, has unbalanced inputs for the budget guys, balanced inputs for the pros, some jacks on the back for wire freaks (but nothing that gives you access to the regeneration loop), and so on; maybe Loft should be a little more specific, and either strip the thing down and sell it for less with the same basic characteristics, or really soup up the unit's greatest strength (flanging) into an excellent flanger/doubler.

The 450's specs are less impressive than some of its digital brethren (and sisteren!), but it's a musically useful, general purpose device which gives a good account of itself in certain popular applications. If chorusing and flanging are big parts of your sound, and you want something that's got more options than the average floorbox—and you want echo capability built-in—look into the Loft 450.

NORMAN EISENBERG AND LEN FELDMAN

BGW 7000 Power Amplifier

General Description: The BGW 7000 is a sturdily built, heavy-duty professional-grade power amplifier rated for the following power outputs: In stereo use, 200 watts minimum into 8-ohm loads, or 325 watts into 4-ohm loads; in mono use, driving an 8-ohm load, 650 watts. Designed for standard rack-mounting, the model 7000 can be stacked fairly close to other units since it incorporates a fan for forced-air cooling which comes on automatically when large amounts of power are being drawn from the amplifier for extended periods of time. The amplifier's output stages employ eight transistors in each channel, each transistor being a 150-watt power device, which suggests a certain amount of "conservatism" in the design, and an emphasis on reliability.

The front panel contains a feed-through opening for the fan, level controls for each channel, the AC power off/on switch and an LED power-on indicator. At the rear are found the other opening for ventilation, quarter-inch jacks for inputs, five-way color-coded binding posts for outputs, a circuit-breaker reset, the mono/stereo switch and the amp's AC power cord which is fitted with a 3-prong (grounding) plug. The stereo/mono switch may be used to convert the model 7000 to a fully bridged mono amplifier in which case only the channel-A input is to be used, while the output is taken only from across the two "plus" binding posts. Minimum recommended load impedance for this mode of operation is 8 ohms.

The model 7000 is shipped from the factory wired for 120-volt operation. Alternate power-line voltages may be used, but the manual cautions that such conversion be done only by a BGW authorized service station. Also included in the manual are helpful data concerning choice of output leads, and of fuses for loudspeaker protection.

Test Results: In our tests, specifications for the BGW 7000 were confirmed or exceeded, and it was apparent from bench tests, and further underscored by listening tests, that the thorough ruggedness of this amplifier was combined with very low-distortion, clean-sounding performance on all manner of program material.

It should be noted that BGW was careful to state power output capability in accordance with FTC rules for power disclosure (although they really didn't have to since this is a pro amp and not one intended for home consumer use), and in accordance with the newly adopted EIA Amplifier Measurement Standard RS-490. On the other hand, they did not choose to measure input sensitivity or signal-to-noise in accordance with that new Standard. As a result, you may find it difficult to correlate our measured results for these parameters (see "Vital Statistics" in this report) with BGW's figures. Actually, however, the S/N we measured—94.5 dB IHF—is an excellent one since it is referred to a mere 1 watt of output. We also measured S/N with respect to rated output, and used the "A" weighting to arrive at our figure of 110 dB, whereas BGW chose to provide an unweighted figure here which would account for the difference between the two readings. The ventilating fan came on during some of the bench tests (power-handling capacity, etc.) but remained off during listening tests.

General Info: Dimensions are 19 inches wide; 5¼ inches high; 15½ inches deep. Weight: 41.5 lbs. Price: $779.

Individual Comment by L.F.: I have always wondered why professional amplifiers equipped with ventilating fans allow the fans to run (often noisily) all the time. It has seemed to me that when an amp, professional or home type, was called upon to deliver a couple of watts of average power (as many amps do most of the time when handling typical program material), the heat generated in the output stages or in the output stages or in the power supply section would be so low that forced cooling would not be needed. Evidently BGW agrees with me, for in this well-designed and ruggedly built model 7000, they have included a fan that comes on only when forced cooling is really needed. It was not needed during listening tests and so I was not disturbed by the mechanical noise that is generated by the fan when it does operate. All of which impressed me as a happy compromise to a problem that has bothered me for years with pro amps.

I was somewhat surprised to find that input provisions are only available in unbalanced form, and no instructions are provided in the owner's manual in case you have to connect balanced line level inputs to the amplifier. On the other hand, instructions concerning bridged operation (in which mode more than 650 watts of total power into an 8-ohm load is available) are complete and very detailed.

I listened to a variety of program material via the BGW model 7000 amplifier—everything from classical music to hard rock which makes greater demands for steady-state power upon an amplifier such as this. The model 7000 handled everything with no apparent strain, and it seemed to have plenty of reserve headroom even at very loud levels. Above all, I somehow felt it was never on the verge of component failure or overheating problems. If one can describe an amplifier as "coasting along," that phrase would fit the BGW 7000.

Individual Comment by N. E.: The model 7000 is the third power amplifier we have had the opportunity (I should add, and pleasure) of testing from BGW Systems of Hawthorne, California, a company that is a specialist in producing professional amplifiers. The earlier amps were the model 500D [*Modern Recording & Music*, September 1977], and the model 750B [August 1978]. The present model 7000 is priced lower (and weighs a bit less) than the previous amps but it is in every way the kind of no-fooling-around, solidly built, dependable kind of amplifier we have come to expect from BGW. That is to say, it proves very well-mannered on the test-bench (no problems with special grounding, loads and so on), and it performs equally reliably and very clean-sounding too when connected to speakers and fed with program material.

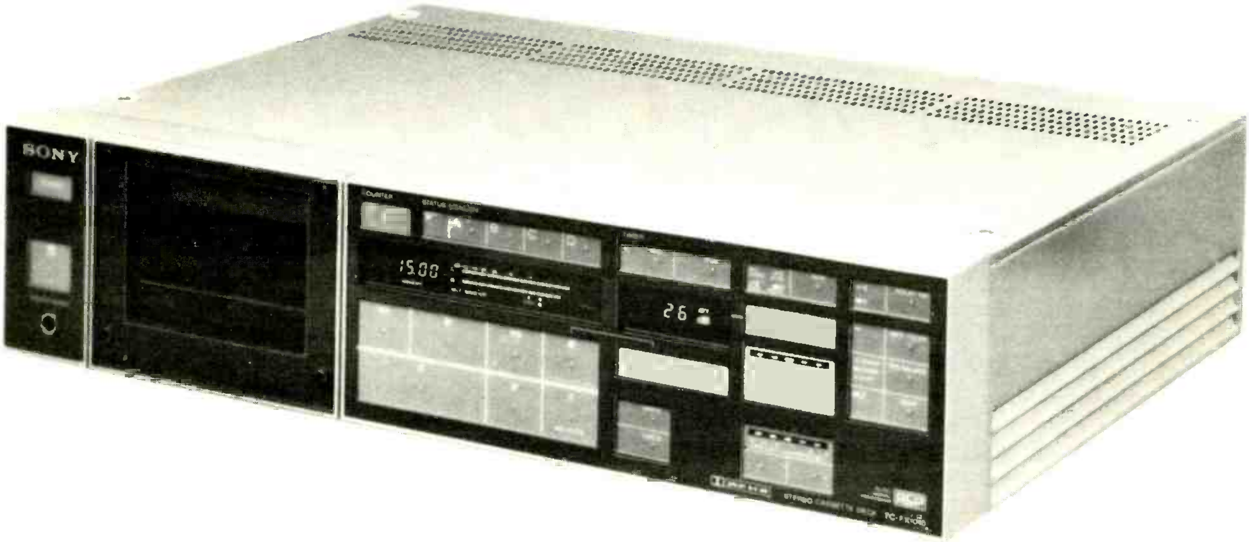
The amp's lack of a balanced input is, I am told, purely a matter of keeping its cost within the realistic reach of the budget-minded segment of the music market. That is to say, the model 7000 is designed to serve the needs of musical performers who want to put together a high-quality P.A. system in today's world of tight cash and high-interest loans. For this reason the extra inputs were omitted. Of course, if balanced lines must be used with the amplifier, they can always be managed with external transformers.

BGW 7000 POWER AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Continuous power for rated THD		
8 ohms, 1 kHz	200 watts	207 watts
4 ohms, 1 kHz	325 watts	325 watts
FTC rated power (20 Hz to 20 kHz)	200 watts	202 watts
THD at rated output		
1 kHz (8 ohms)	0.1%	0.03%
1 kHz (4 ohms)	0.15%	0.15%
20 Hz (8 ohms)	0.1%	0.09%
20 kHz (8 ohms)	NA	0.40%
IM distortion, rated output		
SMPTE	0.02%	0.02%
CCIF	NA	0.0131%
IHF	NA	0.030%
Frequency response at 1 watt (for -3 dB)	1 Hz to 100 kHz	1 Hz to 65 kHz
Signal-to-noise re 1 watt, "A" wtd, IHF	NA	94.5 dB
Signal-to-noise re rated output	96 dB unwtd	110 "A" wtd

Dynamic headroom, IHF	NA	1.4 dB
Damping factor at 50 Hz	100	144
Input sensitivity, IHF	NA	0.08 volt
Input sensitivity re rated output	1.23 volts	1.15 volts
Power consumption, idling; max.	NA; NA	62; 870 watts

Sony TC-FX1010 Cassette Recorder



General Description: Sony's new TC-FX1010 cassette deck incorporates some novel features not the least apparent of them being its front panel which has no knobs or switches in the usual sense. Instead, all operating functions and adjustments are made by pressure-sensitive electronic switches that are flush with the panel, many of them working through the deck's built-in microcomputer. This includes even those controls that are not in the "either-or" category, such as level adjustments, which are made by pressing the designated "block" continuously. Actually, all mechanical switching has been replaced with these electronic keys, and this design concept is credited with shortening and simplifying the signal path through the recorder.

A three-head configuration is used, with the record and play heads sharing a common mounting so that there is no need for periodic readjustment of azimuth. However, the play head in this recorder does not monitor the recorded signal while it is being taped. The use of a separate play head here is strictly in the interest of improved sound on playback. Monitoring as such is carried out by the deck's built-in "brain." This self-monitor system, as Sony calls it, automatically compares the source with the recorded sound when in the record mode. If signal levels are the same for both the source and the recorded sound, a white indicator will blink occasionally, signifying that all is well. If signal levels are off, or if the heads are dirty, a red indicator also will start to blink and a beep-alarm will sound. (See "Individual Comments" by L. F. and N. E. for further discussion of this feature.)

The deck can automatically adjust bias, equalization and recording sensitivity for different kinds of tape when the cassette is inserted. One key selects any of four tape types (I for normal bias: II for CrO_2 or other high-bias formulations; III for ferrichrome; IV for metal). A touch of the automatic calibration key then activates the adjustment procedure which takes 7 seconds. The computer-derived data from this process, as well as several other chosen parameters (recording signal level; channel balance; line output level; the use of Dolby B or Dolby C; whether the MPX filter was in or not) all can be stored in the deck's "status memory" for up to four different combinations (A, B, C and D); recalled for checking; and re-used for recording or for playback. Settings and combinations also can be changed and re-entered into the memory.

The two-motor transport in the Sony TC-FX1010 employs dual capstan drive. Transport functions are all governed by the electronic touch-switches, and complete fast-buttoning (including going into the record mode from any other) is provided. In addition to turning on the deck by its AC power switch, you also can activate it by touching any of the transport keys, status memory keys, tape counter memory or reset keys, or the cassette eject key.

At the extreme left of the front panel are the main power switch, the cassette-eject button and a stereo headphone jack. To their right is the cassette compartment behind its see-through swing-down cover. To its right is the function control area—a top row of keys for the tape counter (with reset and memory keys) and for the status memory. Below this group is the signal-level metering which consists of twin rows of fluorescent bars that show peak program signal levels (input during recording; recorded levels in playback mode). A peak-hold feature is included. The self-monitor indicators are just under the bar-graph indicators, while to the left is the tape counter readout, a digital linear display of real time in minutes and seconds. Below this group are the transport control keys (rewind, forward, fast-forward, record, stop, pause, and record muting).

Continuing across the panel, there are timer keys for setting the deck to record or play when unattended with the use of an external timer; the main tape select keys and their indicators; the Dolby (B or C or off) selector; the MPX filter key; the MOL (maximum output level) selector; and a key that activates the automatic beep-alarm.

Just below the tape selector is the recording level control which is adjusted up or down by pressing its "plus" or "minus" side, respectively. As signal level changes a separate indicator at the left of the key shows digitally the amount of recording level attenuation in dB. Below this indicator are the keys for starting the automatic tape calibration process, and for activating an automatic attenuation which will keep recording level below whatever preset level you have chosen. And below these keys are two more: The "write" key stores recording and playback settings in the status memory, while the "check" key recalls the stored settings (in playback mode only).

In addition to the recording level key, there are separate keys for channel balance when recording, and for playback or output level. Each of these keys has its own set of indicators. The output control adjusts volume at the headphone jack as well as at the line output jacks which are at the rear of the deck together with the line input jacks. Also at the rear is a connector for use with an optional wired remote-control unit. A wireless remote-control unit also is available from Sony. The final control on the deck is a rear-panel switch to suppress interference when recording medium-wave or long-wave AM. Just above the deck's AC power cord insert is a switched AC convenience outlet.

Test Results: The basic electrical performance of the Sony TC-FX1010 was tested in our lab using three tape types: Normal-bias (Maxell XL-1-S); high-bias (Sony UCX); and metal (Sony Metallic). Results were generally fine, if somewhat mixed, vis-a-vis the three tapes. For instance, the metal tape produced the best overall response (20 Hz to 20 kHz within ±3 dB), but its headroom was a dB less than that observed for the other two tapes. Metal tape also had the highest distortion. Its signal-to-noise characteristics were better than those of the normal-bias tape but slightly less than that for the high-bias tape. In any event, the use of Dolby-B did improve S/N for all three tapes; the use of Dolby-C improved it even more, going significantly better than 70 dB in all cases.

Figs. 1, 2 and *3* show the record/playback response for the three tapes, each taken at both the 0 dB and the -20 dB record levels. It is important to note here that Sony has elected to use a 250 nWb/m magnetization level as its 0-dB reference. That level is fully 2 dB above Dolby calibration level, and a good 4 to 5 dB higher than the "0 dB" used by many cassette deck

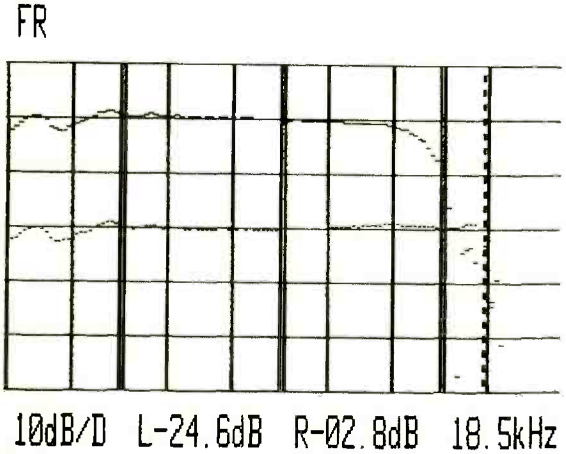


Fig. 1: Sony TC-FX1010: Frequency response at 0 dB and -20 dB levels (Maxell XL-IS tape).

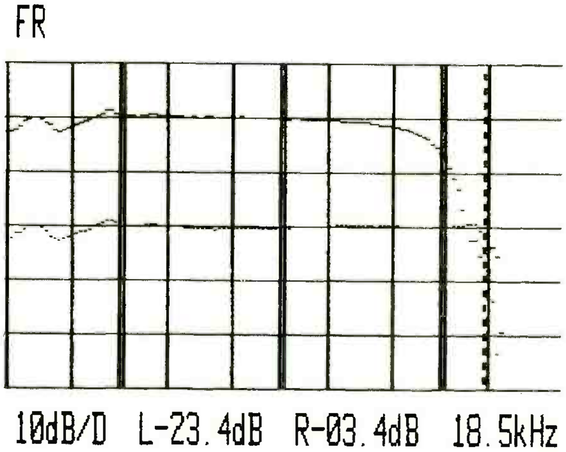


Fig. 2: Sony TC-FX1010: Frequency response at 0 dB and -20 dB levels (Sony UCX-S tape).

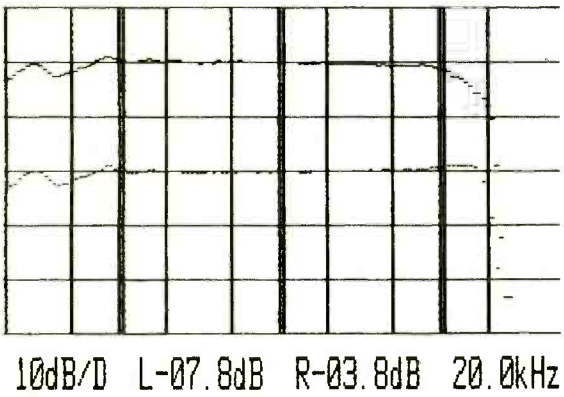


Fig. 3: Sony TC-FX1010: Frequency response at 0 dB and –20 dB levels (Sony metal tape).

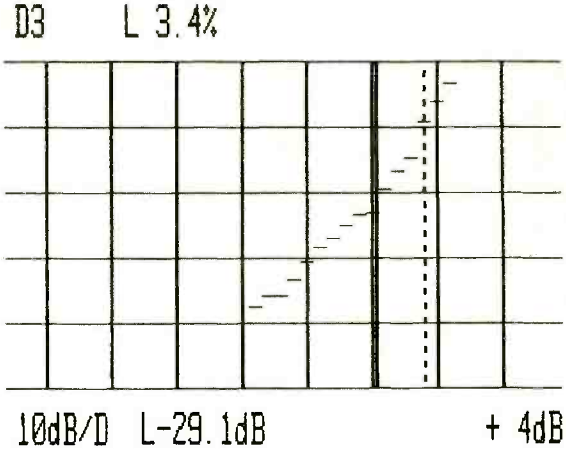


Fig. 5: Sony TC-FX1010: Third-order distortion vs. level (Sony UCX-S tape).

manufacturers. In *Figs. 1, 2* and *3*, therefore, the upper curves of response are taken actually at levels that are generally higher than those reported for many other decks in the past.

Similarly, when examining the 3rd-order distortion results we obtained *(Figs. 4, 5* and *6)*, the "headroom" figures shown (+4 dB for normal-bias and for high-bias samples; +3 dB for the metal sample) are really much more impressive than they might appear at first glance. And the same holds true for the "0 dB" level 3rd-order distortion readings listed in the "Vital Statistics" table. An exception, however, is the 1.6 percent reading we got for the metal tape which does seem a bit high, even after taking into account the higher absolute magnetization level which Sony uses as its 0 dB reference. In *Figs. 4, 5* and *6*, by the way, "0 dB" is represented by the double vertical line in each figure.

Signal-to-noise test data for the three tapes are shown in *Figs. 7, 8* and *9*. Note that each of these figures has an "A" and "B" display. In the A display, the upper set of numerals shows S/N without Dolby (designated as the "L" channel readings), while the numbers next to the "R" show S/N with Dolby B. Plots of third-octave noise spectra are shown in the graphs themselves, over a frequency range of 20 Hz to 20 kHz. The numbers below the graphs are the noise values for the particular frequency at which the cursor (dotted vertical line) has been set. For example, in *Fig. 7A*, noise of the third octave centered at 6.3 kHz comes to –61.5 dB (below the 3-percent distortion level of the tape), while with Dolby-B the noise for that third-octave is reduced to –69.8 dB below the same 3% distortion level.

In the "B" displays for *Figs. 7, 8* and *9*, S/N without Dolby is compared to S/N with Dolby-C. The Dolby-C results are shown, in each case, next to the "R" designation at the top of each figure.

The low 0.036% WRMS wow-and-flutter for the deck is shown in *Fig. 10*, together with a plot of specific wow components in the frequency range from 200 Hz downward. The percentage (0.019%) shown below the graph represents the individual wow contribution at 10 Hz.

Fig. 11 is a plot of tape-speed error, taken after running the tape for 3 minutes. At the end of that time, speed was off by an insignificant 0.246%, and had become so stable that there was no point in continuing this test.

The owner's manual for the TC-FX1010 states that

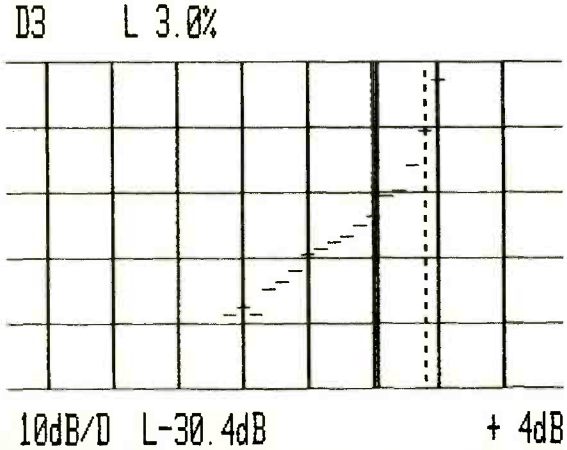


Fig. 4: Sony TC-FX1010: Third-order distortion vs. level (Maxell XL-IS tape).

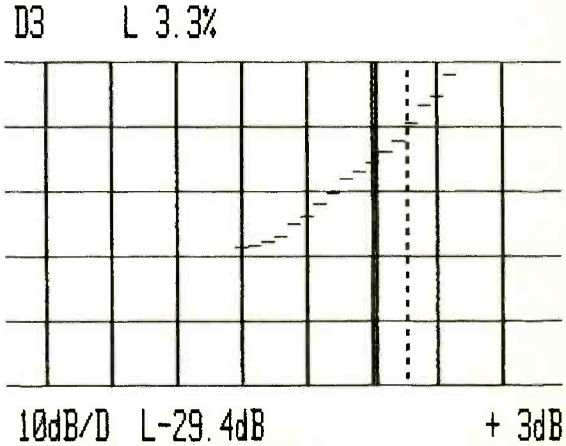
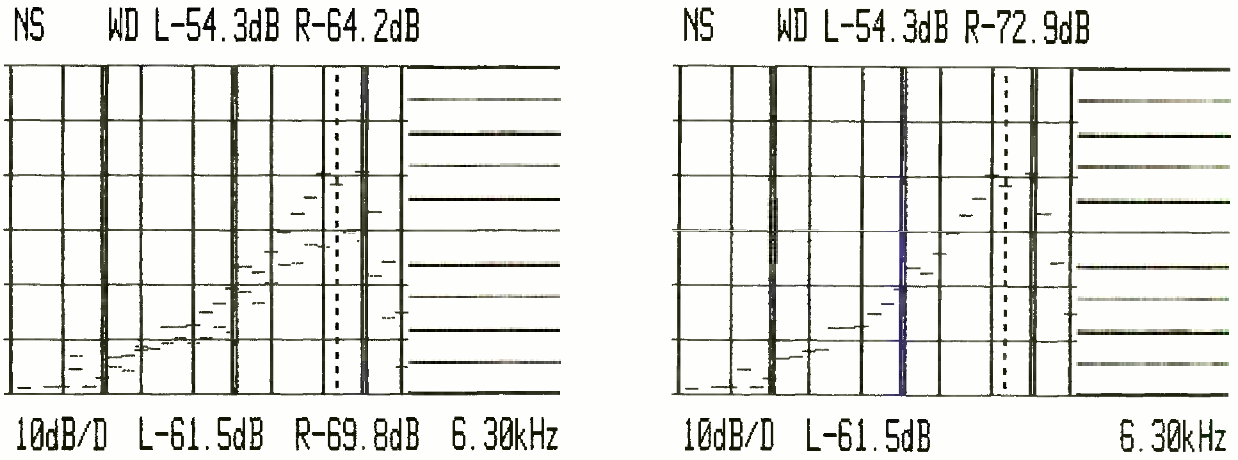


Fig. 6: Sony TC-FX1010: Third-order distortion vs. level (Sony metal tape).



(A) (B)

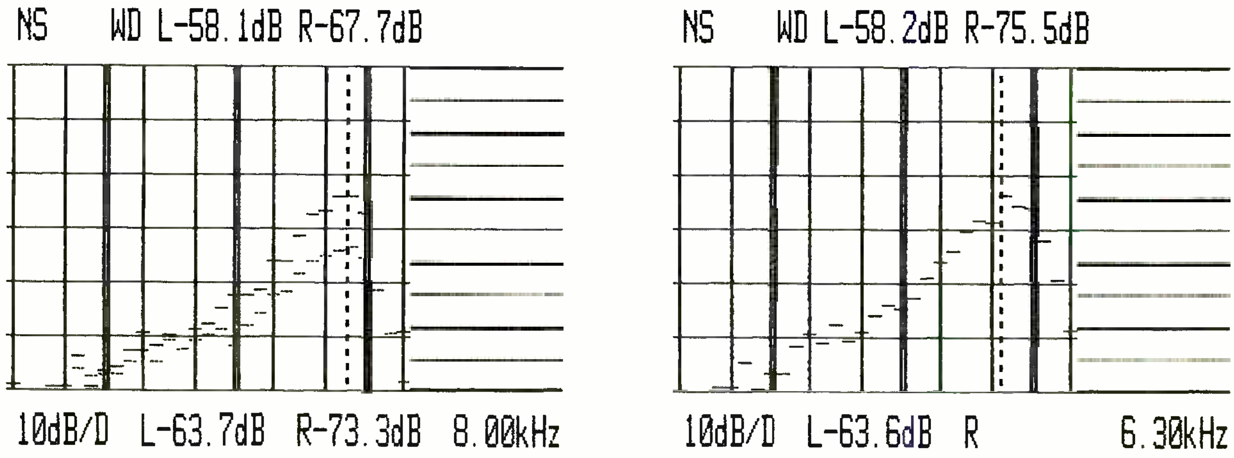
Fig. 7: Sony TC-FX1010: S/N ratios using Maxell XL-IS tape. Upper figures in (A) are for Dolby "off" and Dolby "B" (shown as "L" and "R," respectively). Upper figures in (B) are for Dolby "off" and Dolby "C."

Dolby-C, in addition to providing as much as 20 dB of noise-reduction, also incorporates a high-frequency "anti-saturation" network. During recording, high-level high-frequency signals are reduced, while in playback such signals are automatically restored to their correct levels, in a sort of "reverse Dolby" effect (that seems very similar to the system known as Super-ANRS introduced some years ago by JVC). In any event, to check out this aspect of Dolby-C, we ran a pair of frequency response sweeps at 0 dB record level (where high frequency saturation normally occurs), and we plotted one curve without Dolby and another curve superimposed on it with Dolby-C activated. The results, shown in *Fig. 12*, do confirm Sony's statement. It is clear that for the upper curve, whose results are designated in dB below the graph for a frequency of 12 kHz, there is more than 5 dB of output than there is for the curve produced without Dolby-C in the circuit.

General Info: Dimensions are 17 inches wide; 4½ inches high; 13 inches deep. Weight: 17 lbs., 11 oz. Price: $650.

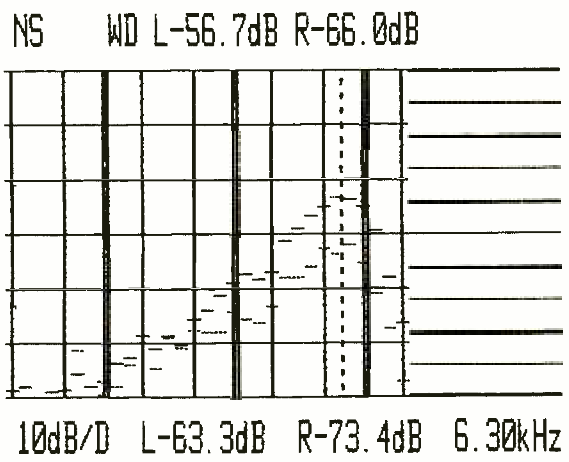
Individual Comment by L. F.: I must admit that my first reaction to the Sony TC-FX1010 cassette deck was not overly positive. Until I worked with the deck for a while both on the lab bench and in actual musical recording, I could not for the life of me figure out why Sony had gone to the trouble of incorporating separate, optimized record and play heads, and then had *not* gone to the trouble of providing the user with off-the-tape monitoring capability such as one might find on any "respectable" three-head machine.

I soon realized that the features which Sony chose to incorporate in this sophisticated deck more than compensated for the lack of off-the-tape monitoring. In fact, in some ways they were actually better for the user who wants to be able to set up a recording operation on the deck, walk away, and be assured of a near-perfect recording when he or she returns to the machine later. Sony's "self-monitor" system makes that possible. It constantly compares the input signal level with the recorded (off-the-tape) signal level, and warns (by indicator lights, and by a beep too if you

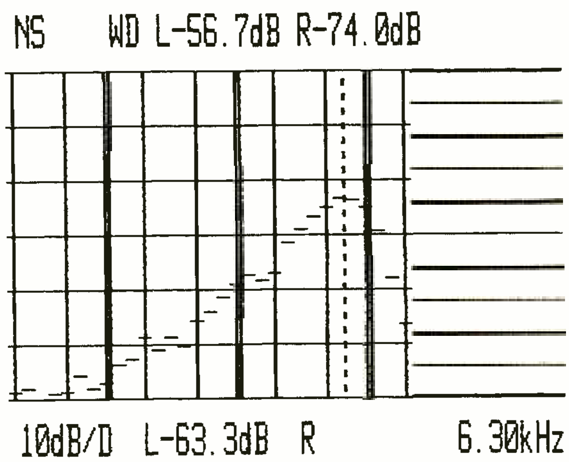


(A) (B)

Fig. 8: Sony TC-FX1010: Same data as in Figure 7, but using Sony UCX-S tape.



(A)



(B)

Fig. 9: Sony TC-FX1010: Same as in Figure 7, but using Sony metal tape.

choose) if any significant discrepancy occurs. For some recordists, that's a better form of monitoring than they could do themselves.

The self-calibration of the deck to meet the needs of just about any type or brand of good tape occurs so fast (about seven seconds) that you won't want to skip this important operating each time you insert a new cassette. Then there's the ability to "store" operating parameters for up to four different types of tape (that includes not only bias and sensitivity settings, but tape selection, type of Dolby desired (B or C), record balance settings, MPX filter on or off and even line-output levels). With all of these things optimally adjusted, there really is very little need for off-the-tape monitoring, and the separate record and play heads are there to perform their more vital role—that of being optimally designed for the record or play functions themselves.

In my opinion, the Sony TC-FX1010 thus is an extremely well-designed deck that has utilized modern microprocessor technology to best advantage. This deck should find favor with those who want to achieve good cassette recordings with a minimum of manual effort, and who want to get the most out of the tapes they use regardless of brand or generic type. The suggested retail price of $650 seems to me quite fair for a deck with these capabilities.

Individual Comment by N. E.: There are times when I get the feeling that the whole business of new product design, at least in the cassette tape recorder field, has become an ongoing contest to see who can come up with the most intriguing application of computer-age automation to what is essentially a "convenience" format. In other words, while I can appreciate or even marvel at the cleverness of product designs such as this new Sony deck, at the same time I cannot but wonder if we are not also witnessing some kind of technological over-kill in which an effort is being made to impress certain potential buyers, but without necessarily advancing the art in terms of basic audio performance. There is just so much that can be gotten from the cassette format, from the standpoint of the serious recordist, and it seems to me that with the new tape formulations, the advanced noise-reduction systems such as Dolby C or dbx or Dolby HX and so on, we are just about "there." Whether these and similar audio advances (and perhaps a few innovations or refinements yet to come) are best realized in terms of familiar format design or as embodied in more and more computerized formats remains—at least for me—a very arguable point. The "hard facts" aspect of our report on the new TC-FX1010 add up, of course, to a very competent cassette machine (a few obvious departures from published specs notwithstanding), but the audio performance of this deck is not any better or worse than that of many of its contemporaries, computerization or no.

As to some specific thoughts on the TC-FX1010: The fact that the deck "monitors itself" is significant only in terms of the electrical parameters it can measure. This is fine (as long as the system continues to function properly), but it does not let *you*, the active recordist, really know how the tape is taking down the performance in more artistic terms—for instance, special microphone techniques or special tonal effects and so on that you may have set up to make a recording that has your personal stamp or that of the artists are

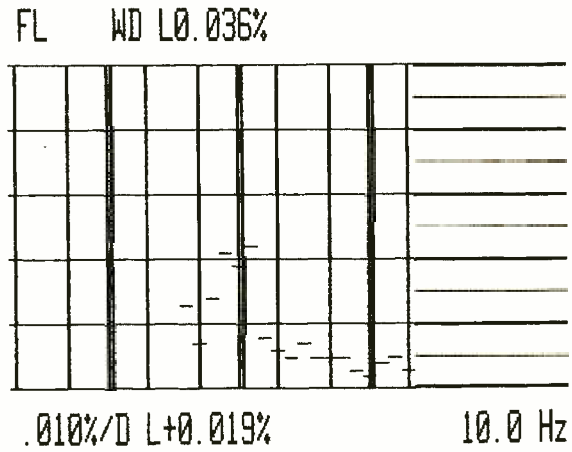


Fig. 10: Sony TC-FX1010: Wow-and-flutter plot and readout (upper figure is total) for the recorder.

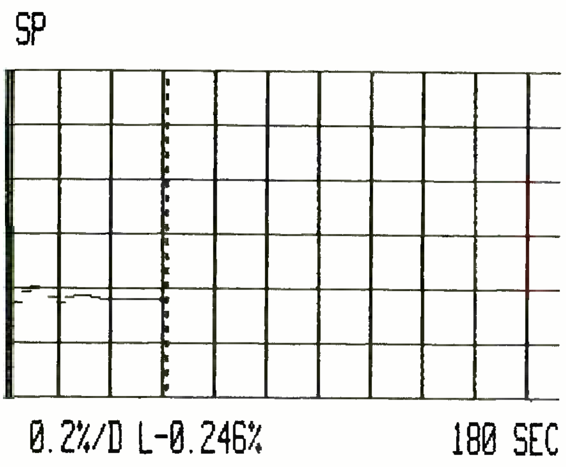


Fig. 11: Sony TC-FX1010: Speed accuracy, plotted for 3 minutes of operation.

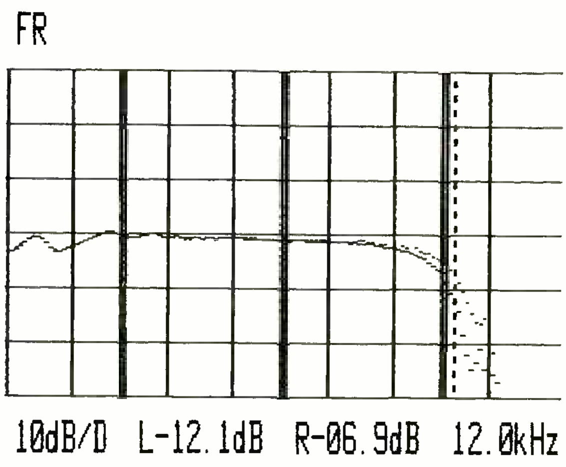


Fig. 12: Sony TC-FX1010: Comparison of response at 0 dB record level with and without Dolby C confirms Sony's claim that Dolby C also adds dynamic headroom at high frequencies. Upper curve (R) shows 5.2 dB improvement at 12 kHz compared with no-Dolby (L) plot.

just not going to be verified during the recording, as they would be if you were able to monitor off-the-tape during the recording. The notion of being able to walk away from a recorder during a recording hardly applies, I think, to a "live" session situation; it is more relevant for dubbing or copying from an already recorded source or broadcast, and so if that is your main interest in owning a recorder, okay. But if you are into serious "live" recording, the "self monitor" system may just not be quite adequate.

You also may feel, as I do, that it still is easier and more "secure" to operate a slider or a knob than it is to press a small square or rectangle among other small squares or rectangles (somewhat faintly labeled at that) in order to adjust levels. These considerations would seem to indicate that this deck's primary appeal would be to the quality-minded "home" sound buff rather than to the active or semi-pro recordist. If so, there are two things that seem to modify that appeal. One is the lack of a microphone input. The advanced recordist does not mind the lack of a mic input on a tape deck; that's what mixers are for. But the convenience-oriented, have-it-all-together home enthusiast might be put off by the absence of a microphone input. The other thing is an item the deck does have which, again, would mean more to the semi-pro than to the average home user—and that's the prominent display of dB attenuation which changes as you adjust the recording level. How many home hi-fi fans, including those who spend small fortunes on their equipment, really know what a dB is in recording terms, let alone what "ATT dB" actually means?

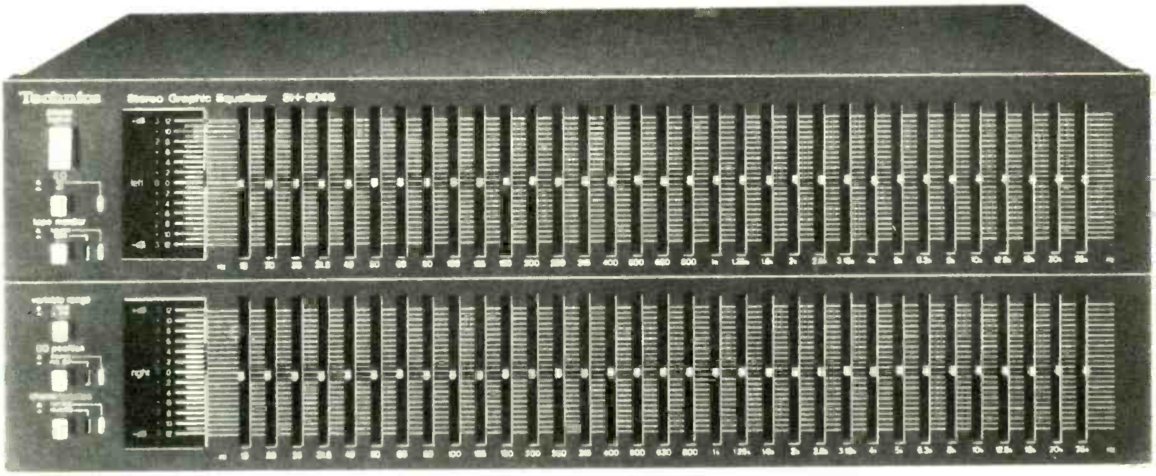
Now for some kind words. While I find the "product philosophy focus" of this deck somewhat blurry in terms of who it is aimed at, I also have to nod at it in acknowledgment of its utterly clean, full-range sound; its smooth and flawless operation; its low wow-and-flutter; its handling of tape; and the fact that, within its admittedly unique product design philosophy, it does everything it is supposed to do. Visually, of course, its designers have taken advantage of the computerized technology to present a new kind of front panel—one that eschews traditional knobs and switches and creates its own Mondrian-like esthetic with colored squares and rectangles. You take it from there.

SONY TC-FX1010 CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response		
normal bias tape	20 Hz to 18 kHz (no dB tolerance stated)	±3 dB, 20 Hz to 18.5 kHz
high-bias tape	±3 dB, 25 Hz to 17 kHz	±3 dB, 20 Hz to 18.5 kHz
metal tape	±3 dB, 25 Hz to 18 kHz	±3 dB, 20 Hz to 20 kHz
Signal-to-noise ratio w/o Dolby re 3% 3rd-order HD		
normal bias tape	56 dB	54.3 dB
high-bias tape	57 dB	58.1 dB
metal tape	60 dB	56.7 dB
Signal-to-noise ratio with Dolby B/C re 3% 3rd-order HD		
normal bias tape	63; 69 db	64.2; 72.9 dB
high-bias tape	64; 70 dB	67.7; 75.5 dB
metal tape	67; 73 dB	66.0; 74.0 dB

Record level for 3% 3rd-order HD (0 dB = 250 nWb/m)		
normal bias tape	NA	+4 dB
high bias tape	NA	+4 dB
metal tape	NA	+3 dB
3rd-order HD at 0 dB record level normal; high-bias; metal	NA (THD spec is 0.8% for metal)	0.66; 0.71; 1.60%
Wow-and-flutter, WRMS	0.04%	0.036%
Speed accuracy	NA	-0.246%
Line output at 0 dB	0.435V	0.718V
Headphone output level, 0 dB	NA	0.130V
Line input sensitivity, 0 dB	77.5 mV	136 mV
Fast-wind time, C-60	80 seconds	80 seconds
Bias frequency	105 kHz	confirmed
Power consumption	40 watts	35 watts

Technics SH-8065 Graphic Equalizer



General Description: Thirty-three bands of equalization on each of two stereo channels are provided by the Technics SH-8065, which makes it the most elaborate graphic equalizer we have yet encountered. Nominal center frequencies extend from 16 Hz to 25 Hz. The 16-Hz slider, and every fourth slider after it are on standard ISO center frequencies; the added sliders between them and above 16 kHz divide the spectrum into what are called approximate "fourth-octave" frequency bands. The full list of frequency bands is: 16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630 and 800 Hz; 1, 1.25, 1.6, 2, 2.5, 3.15, 4, 5, 6.3, 8, 10, 12.5, 16, 20, and 25 kHz.

The normal ±12 dB range is provided. In addition the range of the sliders may be changed to ±3 dB for a very precise, but obviously limited, equalization characteristic which Technics recommends for correcting comparatively small frequency discrepancies such as those related to phono pickups.

Connectors and switching on the SH-8065 permit the device to be patched into a sound system at either "high level" (preamp out, main amp in, nominal 1-volt signal) or "low level" (tape monitor loop, nominal 150-mV signal). The former interface would be used when the equalizer is to serve only for playback. The low-level connection would additionally permit recording the equalized signals. With two tape decks, it is possible to dub from one to the other with deliberate compensation introduced, if desired.

The vertical scales (+12 through 0 to -12 dB, and +3 through 0 to -3 dB) for each group of 33 sliders are shown in contrasting color-tones, while the horizontal lines next to the numbers change to the appropriate color as per the setting of the variable-range selector switch. Each slider has a center detent. Turning on the power switch (at the top left of the panel) lights up the numbered scales and also various LED indicators depending on what setting their particular switches have. These switches include EQ on/off; tape monitor; the range selector; EQ position selector (used when making a tape recording with frequency correction); and "characteristics normal/reverse." This last switch may be used to reverse the polarity of the sliders, which is intended to serve as a convenient way of reducing tape hiss, and so on.

The rear of the unit contains, in addition to the pin-jack input and output connectors, a signal-level switch (1V or 150mV), the unit's AC power cord, and an unswitched AC convenience outlet.

Test Results: In *MR&M*'s tests, both on the lab bench and in actual use, the Technics SH-8065 proved to be an equalizer that offers an unusually fine degree of frequency response control, which was matched by its ±3 dB option of total slider excursion for extremely precise amplitude control. The normal ±12 dB range, of course, would be well suited for the usual chores of an equalizer, while the ±3 dB range could handle some more specialized needs.

The extreme response control flexibility of the device is shown in *Fig. 1*, which was made with the aid of our own laboratory spectrum analyzer. If you count the curves on this photo you will not see thirty-three separate boosts and cuts. The Technics has its lowest and highest center frequencies at 16 Hz and at 25 kHz, but our spectrum analyzer display sweeps only from 20 Hz to 20 kHz.

Maximum range of boost and cut for one of the thirty-three available filter bands is shown in *Fig. 2*. The ±12 dB and the alternate optional ±3 dB settings are both displayed here.

The action of the "characteristics" selector (normal/reverse) is shown in *Fig. 3*. A given boost setting is changed to an equal, but opposite, cut setting. One possible application for this novel feature is suggested by our graph. The upper plot is the equalizer's response (shown here from 20 Hz to 40 kHz) when the switch is set to its "normal" position. We arbitrarily set up a gradual treble boost characteristic which conceivably might be used to record some program material onto tape. During playback of that tape, by simply moving the "characteristics" switch to its "reverse" position, we would obtain the inverse of that characteristic, shown by the lower curve of *Fig. 3*. In this way, the treble cut would reduce tape hiss, for example, while restoring flat frequency response to the program material (since it is an exact mirror image of the first EQ curve). In effect, this option can be thought of as a sort of non-dynamic "Dolby-like" noise reduction system.

We also checked for interaction between adjacent bands of the equalizer. As shown in *Fig. 4*, despite the large number of bands, there is really little interaction between adjacent bands.

General Info: Dimensions are $16\frac{15}{16}$ inches wide; $6\frac{1}{32}$ inches high; 13 inches deep. Weight: 14.6 lbs. Price: $500.

Individual Comment by L.F.: When I first unpacked this graphic equalizer and counted the number of filter band controls spread across its front panel, my immediate thought was that here was a real case of overkill. After I had worked with the unit for a time, however, I wasn't so sure that my offhand criticism had been justified. It became clear to me that what Technics was trying to do with the SH-8065 was to offer a two-channel equalizer that would give users a degree of control that they never had been offered before, not by an octave-by-octave equalizer, nor even by a third-octave equalizer. In addition to the ±12 dB adjustment on each of thirty-three sliders per channel, the SH-8065 offers an alternative of ±3 dB for the total excursion of each slider. That is fine control indeed. For a perfectionist who might be troubled by even fractions of a dB of error in response of, say, a phono playback equalizer, this would be an ideal way to compensate for that error. And, in addition to the noise-reduction possibility already suggested for the unit's "reverse" option, I am sure other applications

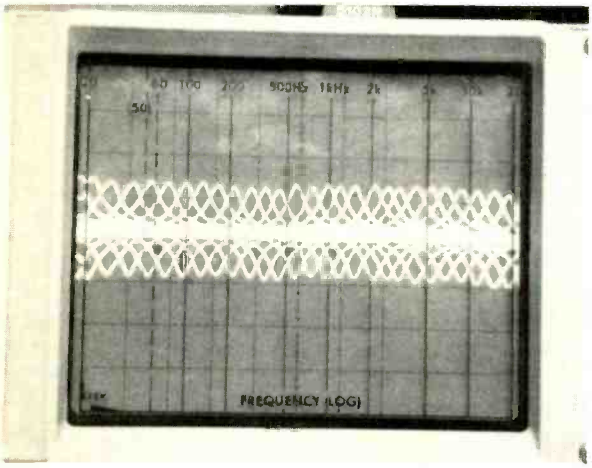


Fig. 1: Technics SH-8065: Spectrum analyzer photo is a composite of the boost and cut characteristics of each of the thirty-three available bands on the equalizer.

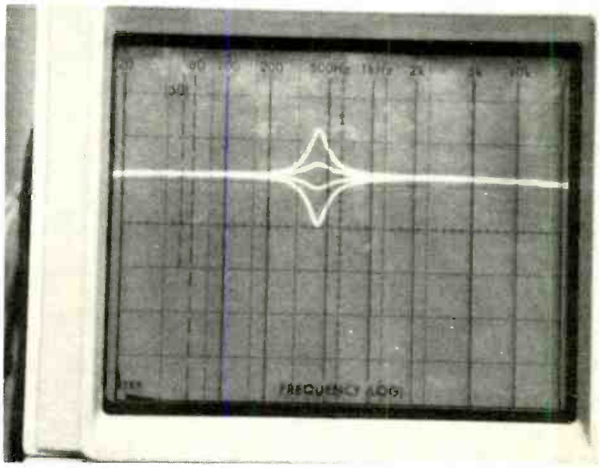


Fig. 2: Technics SH-8065: Equalizer can be set for wide adjustments (12 dB cut or boost) or finer adjustment (3 dB boost or cut per slide control) depending upon requirements and applications.

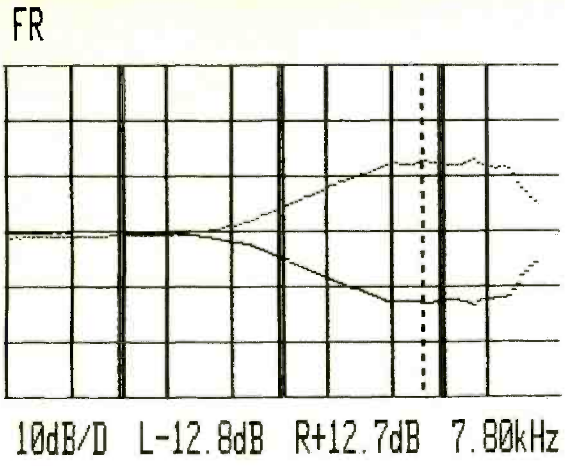


Fig. 3: Technics SH-8065: "Normal/reverse" feature on the unit makes it easy to apply one type of EQ during recording, and inverse EQ during playback. *(See "LF Comments.")*

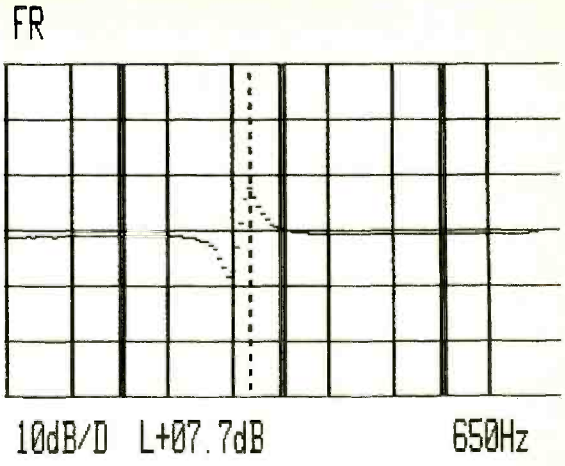


Fig. 4: Technics SH-8065: Despite the large number of filter bands on the SH-8065, there is little interaction between adjacent filter bands.

will come to the minds of those who are involved with equalization on a daily basis.

I am certainly not suggesting that anyone thinking of buying a graphic equalizer needs one with thirty-three bands of control per channel. In most instances, far fewer frequency increments are needed. But for those who, for one reason or another, do require such elaborate and precise frequency-tailoring capabilities, I know of no other available equalizer at this or any other price that offers quite as much as this very interesting and unusual unit from Technics.

Individual Comment by N. E.: The enhanced versatility of this equalizer, vis-a-vis those with fewer bands, is given due consideration in the owner's manual, which suggests among its possible applications the correcting of room acoustics; cutting noises from "live"-recording situations; correcting speaker characteristics; special adjustments used in making tapes for car stereo systems; adding some power to disco music; correcting the frequency characteristics of phono pickups; reducing tape hiss; cutting disc noise; introducing your own loudness contour and noise-filtering characteristics if your own amplifier lacks them or if you disagree with what is provided by your amplifier; and some help in producing clear vocals with different settings suggested for male and female voices.

It seems apparent that the SH-8065 can help in all of these uses not only by virtue of its unprecedented number of control bands but also thanks to its clean response which is quite wide-band, has negligible distortion and has extremely good dynamic range. The only question that might be raised has to do with the very number of controls provided which could make for more chance for error unless they are used carefully. But I suppose whoever bought this unit would know to be careful, and approach its use with a healthy respect for its versatility and no-skipping of the instructions in the owner's manual.

TECHNICS SH-8065 STEREO GRAPHIC EQUALIZER: Vital Statistics

PERFORMANCE CHARACTERISTICS	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response (controls centered)	−1 dB, 5 Hz to 100 kHz	−1 dB, 5 Hz to 45 kHz ±3 dB, 5 Hz to 100 kHz
Maximum output voltage	8 V	8 V (0.1% THD, 1 kHz)
Rated output voltage	1 V	Reference value
Total harmonic distortion	0.0025% (20 Hz to 20 kHz)	0.0023% at 1 kHz 0.0027% at 100 Hz 0.0014% at 10 kHz
IM distortion (SMPTE)	NA	0.0068%
Input sensitivity	1.0 V	1.0 V (unity gain)
S/N ratio, "A" wtd	110 dB	100 dB (re 1 V)
Input impedance	47 k ohms	Confirmed
Output impedance	600 ohms	Confirmed
Band level control range	±12 dB; ±3 dB	Confirmed
Power consumption	28 watts	26 watts

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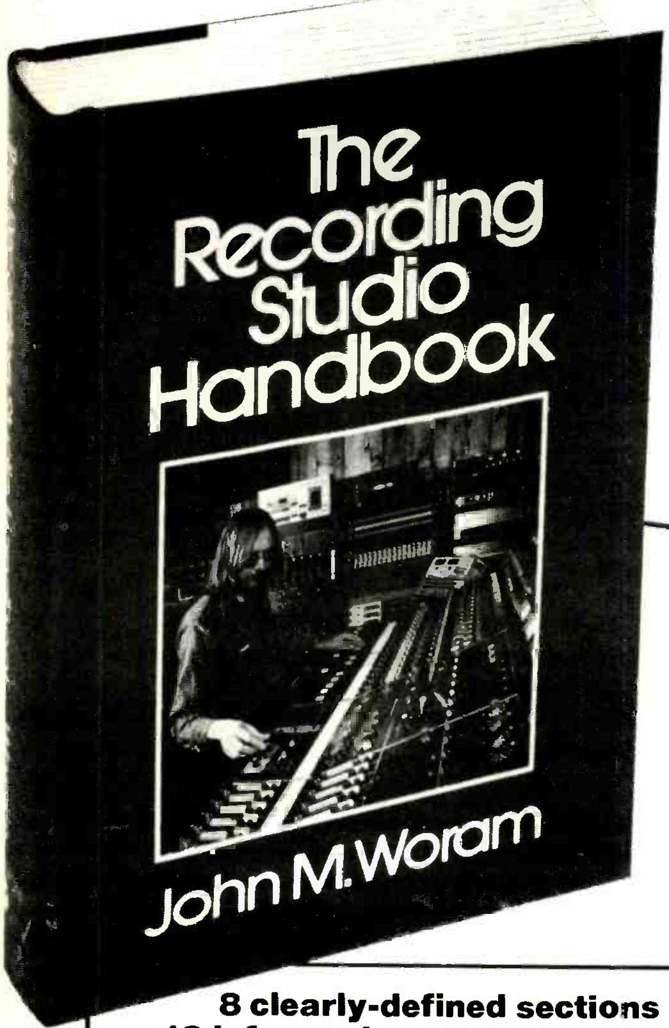

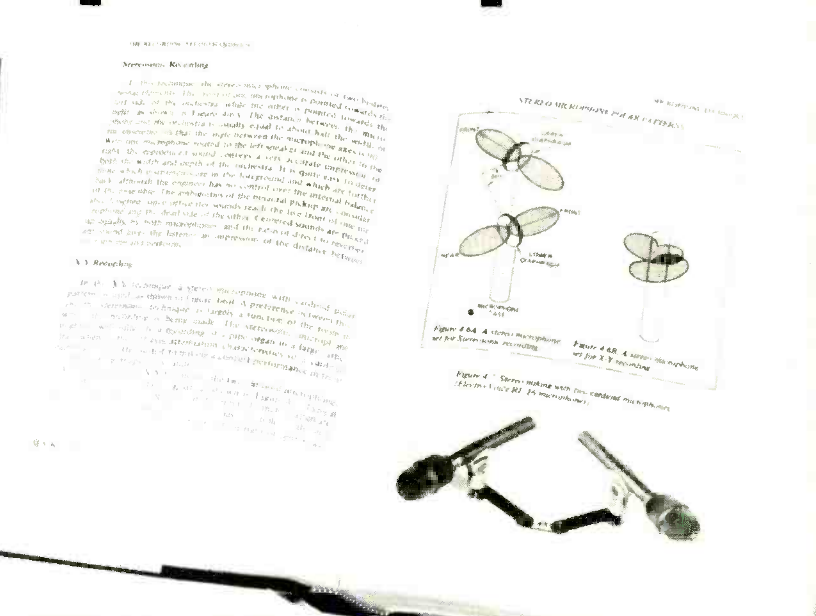

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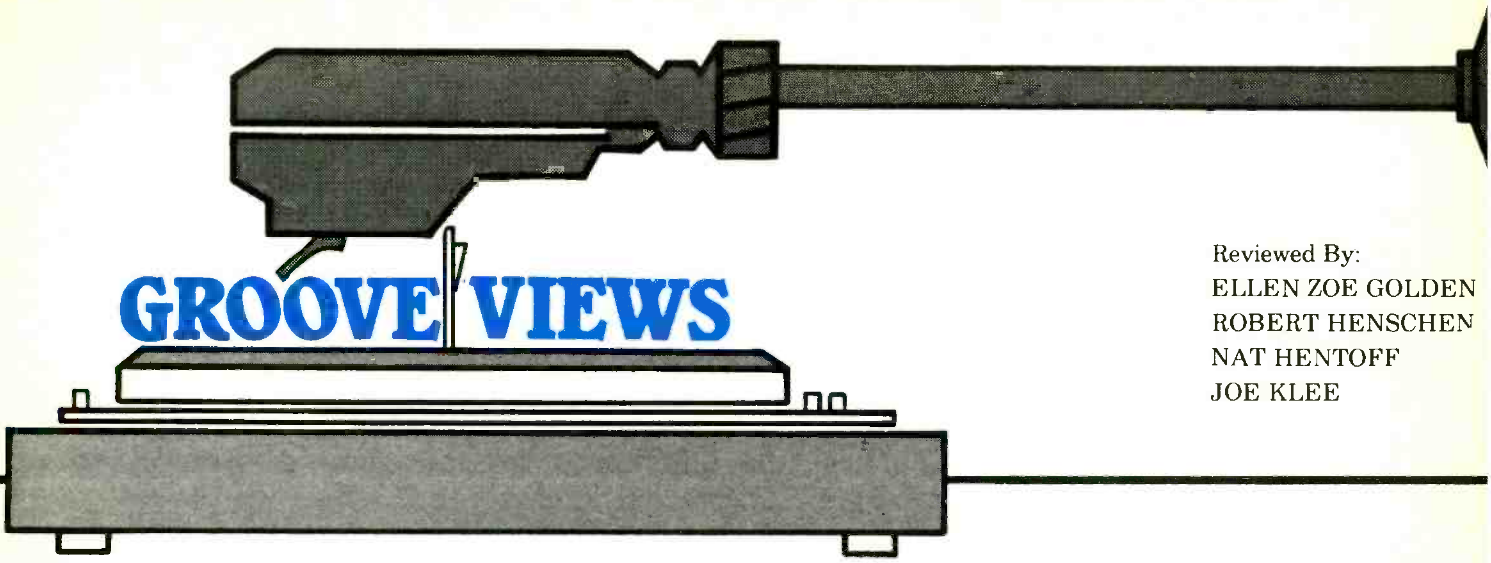

Reviewed By:
ELLEN ZOE GOLDEN
ROBERT HENSCHEN
NAT HENTOFF
JOE KLEE

POPULAR

ALDO NOVA: *Aldo Nova.* [Aldo Nova, producer; Aldo Nova, Louis Mercier, and Billy Szawlowski, engineers; recorded at Bobinason Studios in Montreal's Kingdom Sound, Syosset, New York, and Power Station Studio A, New York, N.Y.] Portrait ARR 37498.

Performance: **Hard pop**
Recording: **Arena-sized impact**

This disc has "hit" written all over it, despite the fact that it comes from a veritable unknown. Someone has made a thorough study of everything that works in today's hard rock and pop (hard pop?) marketplace, drawing on the expertise of those who have gone before, and regurgitating the hit formula with powerful precision.

That someone is a French Canadian of Roman parentage named Aldo Nova, a flamboyant rocker with a catchy name and a musical style full of familiar strains. To his credit, Aldo makes it all look easy: you'd have to be searching pretty hard to recognize a bit of Tom Petty's swagger in the vocals, a Toto piano hook on a chorus, or Francis Ford Coppola's helicopter effects. Aldo Nova has assimilated these influences and probably many others into a musical package that, while not at all groundbreaking, transcends run of the mill rock & roll with an uncommonly bright treatment of a rather simple heavy metal medium.

There are no real astounding studio tricks involved, but Nova gets the maximum out of even the power chording, dual guitar stuff such as "Hot Love" and "You're My Love." Much of the album has a "live" sound to it, an energy and vibrancy not always felt from other studio efforts. "Fantasy" is the big single success from *Aldo Nova* thus far, but the debut disc is long on saleable songs—"Heart To Heart," "Foolin' Yourself," or "Ball and Chain"—to a point where an element of sameness almost begins to creep in.

While Michel Lachapelle (drums) and Michel Pelo (bass) and others help out, Nova has generally played most of the instruments himself, overdubbing chorus vocals and so on. Maybe it is a hint of echo that gives this four-walled studio ambience an arena-sized impact. Or maybe it's just the carnivorous twin-guitar leads that Aldo sends splitting across the stereo signal. This guy is a hard rocker striking the classic pose, and he's playing the kind of ax that American crowds invariably eat up.

Aldo Nova: Your new main man.

There are several young rock groups currently doing material with perhaps more imagination and depth—The Sherbs, Asia, Squeeze—but Aldo Nova has chosen to expand on an established form and do it spectacularly. If that's your bag, this could be your new main man. R.H.

JOY DIVISION: *Closer.* [Martin Hannett, producer; Martin Hannett and John Caffery, engineers; recording site unknown.] Factory Records #6 (Distributed in the U.S. by Rough Trade).

Performance: **Appropriately terrifying**
Recording: **Appropriately reverbed**

Let's invent an entirely new musical genre to place *Joy Division* in: Poe Rock. Poe Rock aims for the same emotional impact as Edgar Allen Poe's tales: chills up the spine, intimations of unspeakable terror, a mardi gras of sinister sensations. The long lamented Doors certainly perfected an American form of Poe Rock with their first album. Imitations of the Doors have certainly proliferated in the years since lead singer Jim Morrison's death. Joy Division is the first band I've heard in the last decade that doesn't simply copy the characteristic apocalyptically angry tone of the Doors. They go an eerie step beyond.

Joy Division consisted of three Manchester youths who developed a very distinctive sound on guitar, synthesizer, bass and drums. The impact of the English punk scene is definitely evident in their recorded output. Lead singer Ian Curtis had a rough-edged voice perfectly suited to communicate the dark undercurrents of songs entitled "Atrocity Exhibition" and "Isolation." The band's drummer could play ferocious blends of disco/funk/rock/jazz rhythms. The bassist had a jackhammer insistence. The band released only two albums (*Closer* is their last) and two singles. Yet their mark on English and American new wave rock is indelible.

Closer explains why. Nine songs about gloom and doom (which seem to blur into each other forming a continuous song cycle) propelled by Curtis' icy vocals. Bass and drums are pushed very high in the mix and create a churning and dynamic backdrop for the vocals. Subtle touches from guitar and synthesizer reinforce the emotional impact of the lyrics. Add to these factors producer Martin Hannett's glossy production. The man has a fascinating sense of how and when to use reverb. Reverb is so painfully easy to overdo. But Hannett knows just when to put Curtis' voice under echo. Hannett's production adds to the ghostly presence of Curtis—who was a *literal* ghost by the time *Closer* was released. The band ended when Curtis committed suicide a few years ago (shades of Jim Morrison and the Doors again?)

In spite of the demonic darkness of *Closer*, there is a solid musical intelligence evident throughout. Pleasure can be derived in spite of the horrors of the subject matter illuminated. That's the nature of Poe Rock at its best. Joy Division was blessed to have so excellent a producer. The recording quality is simply smashing. I particularly appreciate how crisply the drummer is recorded. Curtis' voice seems to mysteriously zoom and menacingly dart over the other instruments. However low the electric guitars get in the mix, they are never totally lost. They hover at the edges of my consciousness like the low-throated threnodies of phantoms.

This isn't an album for all occasions. It isn't the stuff of AM radio. This is one of the ultimate soundtracks for that never to be forgotten interior movie in the collective unconscious: the dark night of the soul. N.W.

JAZZ

DICK SUDHALTER: ***Friends With Pleasure.*** [Richard Sudhalter, supervisor; George H. Buck, Jr., producer;

James Mooney, engineer; recorded at the Stage and Sound Studios, Hollywood, Ca., March 4, 1981.] Audiophile AP-159.

Performance: **Tightly arranged Dixie with a couple of jams thrown in for good measure**
Recording: **Clean, clear, distinct, but without reservations**

Dick Sudhalter or Richard M. Sudhalter (the names are interchangeable throughout the liner notes by Leonard Feather), is as close to a renaissance man as one is likely to encounter in the field of traditional to mainstream jazz. He wrote *the* book on Bix Beiderbecke, *Bix Man & Legend*, and his recreations of the Beiderbecke sound George Wein's New York Jazz Repertory Company were at once accurate and innovative. His journalistic items in the New York Post are so positive and constructive that one hesitates to call him a critic. As a bandleader, he has fronted groups that went all the way from recreating The California Ramblers-sound to bringing about a new fusion of the traditional dixieland and mainstream Fifty-second Street styles of jazz.

As an artist Dick Sudhalter has chosen to grow rather than fit the pigeon holes into which past circumstances have attempted to place him. Still he has not burned his bridges behind him. While much of this new album is the mainstream-dixie fusion style of today's Dick Sudhalter, the Richard M. Sudhalter I first met in 1975 often shines through the cracks in the arrangements.

The pure Sudhalter, as those who prefer our old memories choose to call it, shines through primarily in the smaller ensembles such as the trio rendition of "Home" and the duo with pianist Dave Frishberg on "Blue River." To these ears, prejudiced with the mold of time, "Blue River" represents the finest Sudhalter on wax to date. Frishberg's full, yet not overfilled, piano makes an ideal cushion for Sudhalter's luxurious Bix-influenced yet not imitative cornet playing. The absence of modern rhythmic impediments and the sparseness caused by the lack of other horns makes the duo statement all the more poignant.

Dave Frishberg is the only other member of Sudhalter's Primus Inter Pares Jazz Ensemble who would be known to the average record listener. This does not change the fact that although totally unknown to the East Coast, such musicians as Bob Reitmeier on clarinet and saxophone and Dan Barrett on trombone are first class jazz musicians suffering only from under-exposure to national audiences. Reitmeier seems to have found that particular little groove of common ground in between Frankie Trumbauer and Lester Young that is often written about by jazz historians but seldom, if ever, heard in contemporary playing situations. Guitarist Howard Alden is an enigmatic performer. His single string electrified playing reminds me greatly of the late George Barnes. Alden is also a very strong rhythmic player on both guitar and banjo. I find his solo banjo work not to my liking. I much prefer the chordal style solos of the majority of jazz banjoists of the '20's to the more guitaristic single-string work which juxtaposes the sound of the banjo with the ideas and licks of Charlie Christian and later players.

This also is a tune freaks record, restoring to the repertoire such gems as Hoagy Carmichael's neglected "Boneyard Shuffle" and "Come Easy Go Easy Love" as well as such unusual gems as Johnny Mercer's "Jamboree Jones" sung here by Dave Frishberg with an excess of hipness where I would have wished for a little more humor.

Daryl Sherman, whose vocals on "Home" and "Come Easy Go Easy Love" are both engaging and true to the Mildred Bailey genre, is a new name to me but I've been assured by generally reliable sources (Mr. Sudhalter himself) that she's been around and has paid her dues. Newcomer or oldtimer, her vocal choruses are musically sound and refreshing.

While the recording technically captures everything that was there and has real presence and a decent balance it does not seem to me that the engineering could have been done a bit more carefully. Not only are the bass and guitar occasionally miked a little hot by an engineer with an ear for the rock sound but there is a tinniness to the piano sound at times which is not appropriate to the sophisticated approach which Sudhalter and the band bring to this music.

If Audiophile records aren't sold at your local record store—a distinct possibility since it's a specialist's label—this record can be mail ordered from George H. Buck of 3008 Wadsworth Mill Place, Decatur, Ga. 30032. I don't know the price but the music is sure worth the price of any LP I've seen on the market. J.K.

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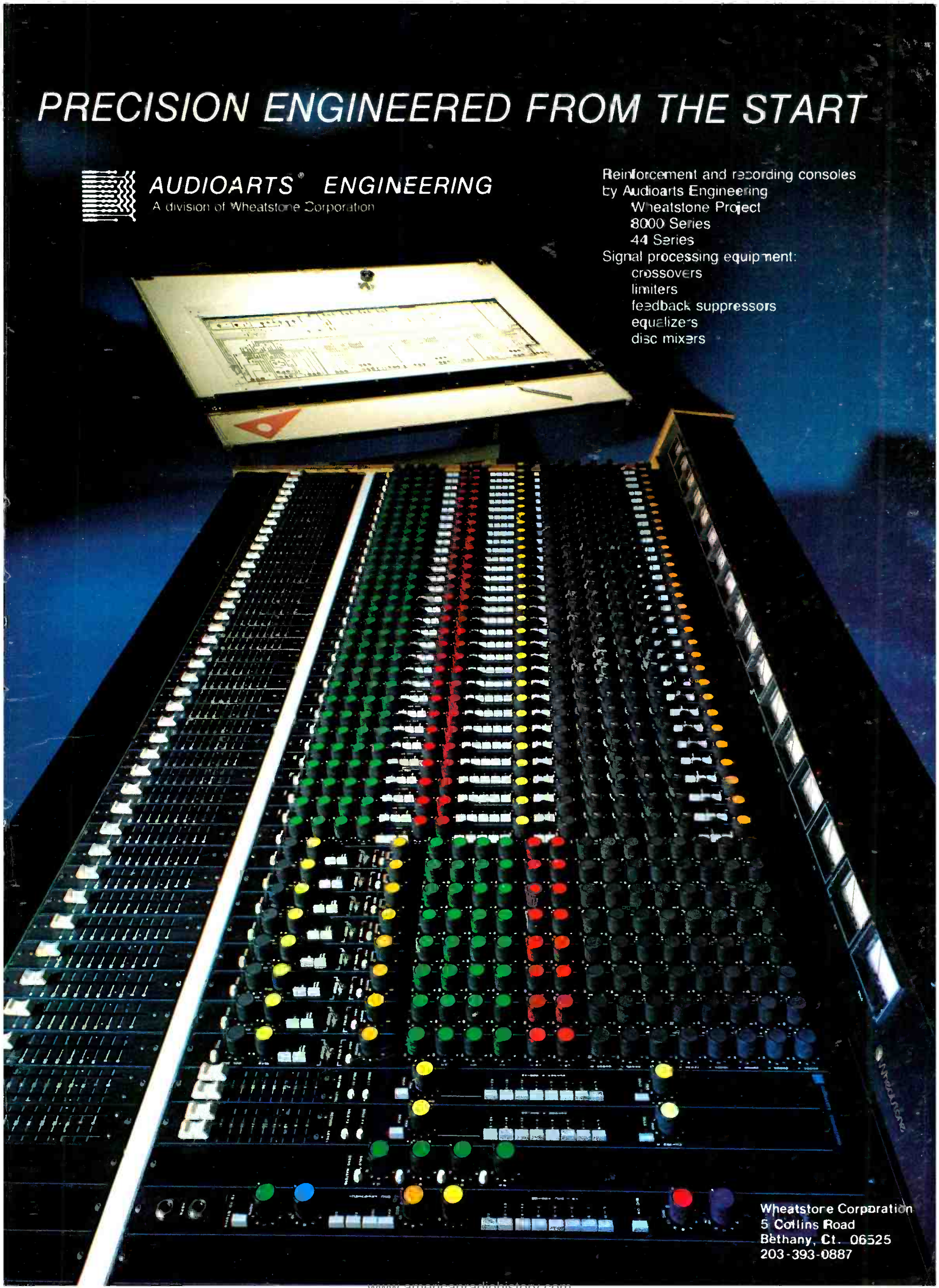
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