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MODERN RECORDING & MUSIC

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VOL. 9 NO. 2

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MODERN RECORDING & MUSIC

FEBRUARY 1983
VOL. 8 NO. 5

THE FEATURES

COMING NEXT ISSUE!
Recording with Spyro Gyra
Recording Techniques—Part XI
Profile: Philip Glass

THE STAPLES

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Letters to the Editor

A Hearty Calibration

I have a small 4-track home set-up and would like to find out how to adjust my machines (TEAC 3340's and TEAC 22-4) for bias and EQ, as well as other basic maintenance info. Does TEAC publish any literature on the subject? Could you recommend any other literature on the subject? Could you recommend any other literature available to non-trained people who can follow instructions, but lack the know-how?

—Paul Lenart
Cambridge, MA

We received the following response from Drew Daniels.

TEAC and Tascam service manuals assume the reader has already acquired the basic skills, e.g., reading voltmeters and oscilloscopes—not much more difficult than reading a gas gauge—and can of course, follow step-by-step procedures. The more recent Tascam manuals are combined Owner/Maintenance manuals which include some of the best non-technical writing for user maintenance and calibration I've seen.

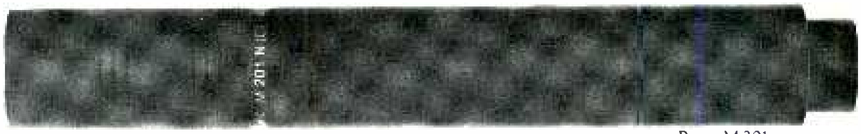
Beyond the Tascam Owner's/Maintenance manuals, TEAC publishes only *The Multitrack Primer*, a basic entry-level primer on the jargon and techniques of recording, sound, and signal producing and carrying elements of a recording system, but the bibliography contained in the *Primer* serves well as a guide to a solid reading foundation in the subject. Books like Everest's "Acoustic Techniques For Home and Studio," and Woram's *The Recording Studio Handbook* advertised here in *MR&M* are books that should be among those in your library.

As far as the actual calibration of the 3340S and 22-4, you'll need the service manuals to locate the adjustment trimmers and find the recommended voltage levels to check on your audio voltmeter, but beyond that, the setup procedure is the same for any (analog) tape recorder. +3 dB over peak bias at 10 kHz for a quarter-mil record head gap (like Tascam) and set the EQ for the flattest response. The necessary test gear can be pricy, especially if you want to use a pink noise alignment tape in addition to your other gear, and you can look forward to a learning period of anywhere from a few hours to several dozen hours, but you may save a substantial amount of time and money over the years not having to rely on repair shops who are not always as highly motivated as you are when your tape equipment needs a tuneup. If you're making tapes for clients to play on other recorders, you will definitely want to acquire an oscilloscope to keep your head azimuth aligned to a standard test tape, and this procedure will require frequent checks that make trips to the service shop prohibitive.

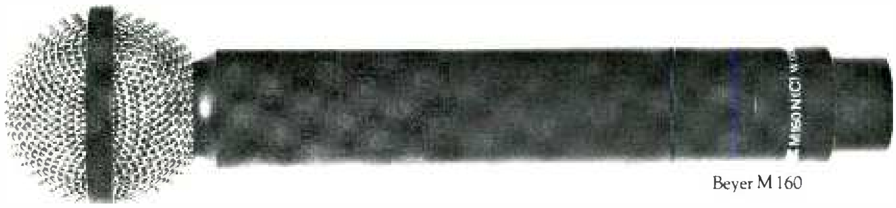
—Drew Daniels
Applications Engineer
Tascam Production Products
TEAC Corporation of America

Why Beyer microphones give you more extraordinary performance for the most ordinary applications.

Beyer M 88

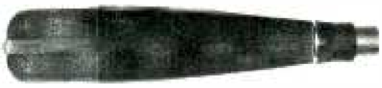


Beyer M 201



Beyer M 160

There are other microphone alternatives when high sound pressure is a factor.

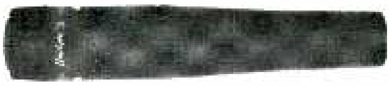


As Sennheiser claims, the MD 421 undoubtedly stands up to extremely high decibel levels and has other features that have contributed to its popularity. But if you're already using the MD 421 to mike loud instruments or voices, we suggest that you investigate the Beyer M 88.

The Beyer Dynamic M 88's frequency response (30 to 20,000 Hz) enhances your ability to capture the true personality (including exaggerated transients) of bass drums, amplified instruments and self-indulgent lead vocalists.

The Beyer M 88 features a matte black, chromium-plated brass case for the ultimate in structural integrity. Beyer microphones are designed for specific recording and sound reinforcement applications.

When you need a rugged and versatile microphone, consider the alternatives.

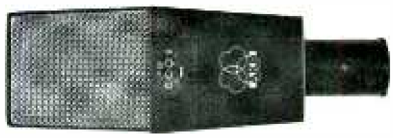


For over 10 years, engineers have used mics like Shure's SM57 for the widest variety of applications in the studio. And we feel that one of the main reasons more engineers don't use the Beyer M 201 in this context is simply because they don't know about it. Those who have tried it in the full gamut of recording situations have discovered how it can distinguish itself when miking anything from vocals to acoustic guitar to tom toms.

The M 201's Hyper-Cardioid pattern means that you get focussed, accurate reproduction. Its wide and smooth frequency response (40 to 18,000 Hz) provides excellent definition for the greatest number of possible recording and sound reinforcement situations.

Each Beyer Dynamic microphone has its own custom-designed element to optimize the mic's performance for its intended use.

You may not always need a condenser microphone for "critical" recording applications.



Some engineers prefer condenser microphones like the AKG C 414 to accurately capture the subtle nuances of a violin or acoustic piano. But should you have to deal with the complexity of a condenser system every time this kind of situation comes up?

The Beyer Dynamic M 160 features a double-ribbon element for the unique transparency of sound image that ribbon mics are known for. While its performance is comparable to the finest condenser microphones, the M 160's compact size and ingenious design offers significant practical advantages for critical applications.

Beyer Dynamic microphones offer state-of-the-design technology and precision German craftsmanship for the full spectrum of recording and sound reinforcement applications.

The Dynamic Decision **beyerdynamic))))**

*Documentation supporting specific comparative claims available upon request.

Beyer Dynamic, Inc. 5-05 Burns Avenue, Hicksville, New York 11801 (516) 935-8000

You Can Always Use Them As Dishes

In response to Drew Daniels' article against "home tapers," which appeared in the December, 1982 issue, I would like to point out the fact that I purchase about 100 albums a year. If Mr. Daniels knew how painful it is to click on my $6000.00 stereo system, clean my stylus, then crack the cellophane and pull out a warped, off-centered piece of garbage I just excitedly had purchased an hour before, I think he would understand why the record business might be in trouble. I have about a fifty to sixty percent return ratio on all my albums. But I still come back for more punishment—even after two to three exchanges on the same album. What happened to the good old flat, healthy, well-centered discs of yesteryear that usually also came with lyric or photo booklets? I guess we've just gotten "sloppy discs" in the computer age.

—Jeff Perry
Santa Clara, CA

We welcome any and all responses to Mr. Perry's impassioned letter.

Uni-Sync: They Stayed Afloat

I have a peculiar problem that I need assistance in solving. For the last three years my electronic work has been in two-way radio and commercial TV, and I have lost touch with sound recording/reinforcement equipment manufacturers. When a local church approached me to assist with upgrading their reinforcement system I began seeking manufacturer's info.

However, there is one company I cannot find a good address for—Uni-Sync, a BSR company of Westlake Village, Ca. Since I used Uni-Sync mixers five years ago, and was pleased with the products, I would like to have up-to-date information. If you have a current address, please print it!

—Mitchell A. Hull
Tulsa, OK

Uni-Sync is now a division of dbx, and they're located at:

71 Chapel St.
Newton, Mass.
02195

The phone number there is (617) 964-3210. The people we spoke to at BSR didn't know anything about the Westlake Village address you mentioned. But I think you'll be able to get what you need with the address above.

The Pink and the White

Again, we are sorry to have to run another correction letter. This time, we owe our apologies to Banner. First of all, in our Product Scene column of the December, 1982 issue, we listed Banner as a division of Optics Corporation. Banner is actually a division of Optronics Corporation, and their full address, which we failed to list in the column, is:

Optronics Corporation
P.O. Drawer 1803
Shelby, N.C.
28150

The other error that was made in the paragraph about the Banner 1232 Real Time Analyzer was in the statement: "Both pink and white noise are provided by any internal digital noise generator." That sentence should read, "Both pink and white noise are provided by an *internal digital noise generator." You see, not just* any *internal digital noise generator can provide pink and white noise, but Banner's internal digital noise generator does. Sorry, Banner, and sorry to you folks who became confused. We hope that's all cleared up now!*

That Dog-Eared February Issue

I have been an avid reader of your magazine for quite some time, and was looking through old issues the other day. I was especially interested in an article in your February 1981 issue, written by Craig Anderton on Beigel Sound Lab's Envelope Controlled Filter. I was going to fill out the reader service card, but was afraid that it might be out of date. Could you provide me with their address and possibly a telephone number?

Thanks for all the work it takes to make a great magazine.

—R. Geoffrey Beckwith
Tucson, AZ

You can write to Beigel Sound Labs at:

Beigel Sound Lab
24 Main Street
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Their phone number is: (914) 986-1699.

Good luck, and thanks for all the complimentary words.

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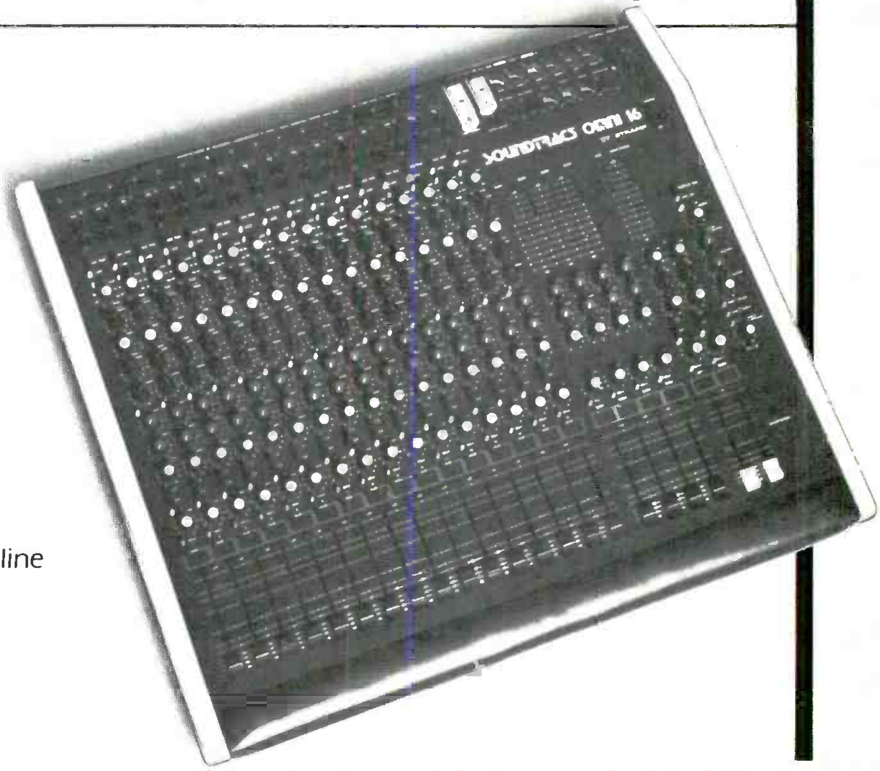

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TALK BACK

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

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

Bias Adjustment

I have a small 4 track home set-up and would like to find out how to adjust my machines (Teac 3340s and Teac 22-4) for bias and EQ, as well as other basic maintenance info. Does Teac publish any literature on the subject? Could you recommend any other literature available to non-trained people who can follow instructions, but lack the know-how?

—Paul Lenart
Cambridge, MA

There really is no mystery in bias adjustment. The tape manufacturers publish an information sheet for each type of tape sold for professional use. These sheets describe the essential parameters you need for proper adjustment of your tape recorder to optimize performance with each tape.

Basically, the rule of thumb is that the record head's gap length determines the amount of bias used. Tape manufacturers recommend that professional tapes be over-biased to a point where the third-harmonic distortion and modulation noise is at minimum. This corresponds to about 1 dB over-bias for a 25 *u*m (1 mil) record head gap, 2.5 dB over-bias for a 12 *u*m (½-mil) record head gap, and 3 dB over-bias for a 6 *u*m (¼-mil) record head gap.

The 3340S and 22-4 have 12 *u*m record head gaps, and the service manuals for both machines also recommend a 2.5 dB over-bias. Over-bias means: starting at minimum, increase the bias until it peaks, then increase 2.5 dB more. This will make the peak drop back, and flatten out the high-frequency response to a reasonable degree. If you're really picky about frequency response, you can fudge the bias to flatten the frequency response some. If you are the type who is fussier about distortion, you can adjust the bias for minimum distortion, but you'll have to get a distortion analyzer and the use of such an analyzer takes some study if you're a non-technician (but the results may be worth the effort).

Tascam Owner's/Maintenance manuals are some of the very best we know of in terms of providing background and useful information to people who read them, for the care of the machines and the understanding of recording basics. They provide very clear step-by-step guidance on bias and equalization setting procedures and other maintenance procedures, and general information on subjects like decibels, volts, impedance matching, etc.

The Teac "Multitrack Primer" is still available, and in fact has just been given an updating with some new information on nanowebers and system operating levels in addition to the acoustical and other very useful information it already had. (Give the folks at Teac's sales department a call (213) 726-0303 to order.) The Primer itself has a good bibliography of pertinent reading material inside.

The initial investment in the basic items you need to calibrate your tape equipment comes as a shock to many people. You'll need a calibration tape for play level and equalization setting, and a good audio voltmeter to read the results. Some sort of oscillator or test tone generator (not your synthesizer) and if you want to save yourself a lot of time, a small oscilloscope. All these things might come to $500, but the investment can save you that much and more if you rely on service shops to calibrate and tweak your equipment for each change of tape brands or types, and a pro studio checks its tape equipment each time a new reel of tape is used.

—Drew Daniels
Applications Engineer
Teac Corporation of America

Demagnetization

Like many people these days, I have learned about recording, not by being apprenticed to a professional engineer but by reading and experimenting with my home studio. As a result, I have lots of questions and nobody to turn to for answers, especially when it comes to the arcane and mysterious process of demagnetizing. I have never seen anybody else demagnetize a tape head, and the written instructions I've encountered are far from comprehensive. I could even be doing something horribly wrong that will destroy my tape heads. So here are a few of the things that have been puzzling me.

1. When moving the head of the demagnetizer back and forth over the tape head (well, actually, under the tape head) should I move only in a direction parallel to

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the head gap, that is, perpendicular to the face plate of the tape head, or should I move from side to side, in a circular manner, or in some other pattern?

2. Should I continue moving the demagnetizer in this "scrubbing" pattern as I draw it slowly away from the head, as one of my friends recommends, or should I stop scrubbing and draw it away in a basically perpendicular line?

3. Will it harm the tape head if the metal tip of my cheapo demagnetizer accidentally comes in contact with the head itself?

4. How close should the scrubbing motion be done, anyway? I have been doing it about ⅛" from the head.

5. Does the tip of the demagnetizer have to be square to the orientation of the head, or is it okay to be angled slightly to the left owing to the fact that I'm left-handed?

6. I know I'm supposed to back the demagnetizer off at least three feet from the deck before turning it off. But do I have to back it away this far between demagnetizing the record head and demagnetizing the playback head, or can I just back it off a foot or so, so that the effective distance is roughly the same to each head?

7. After I've demagnetized one head and am bringing the demagnetizer in to demagnetize another, do I have to approach as slowly as I drew back, or can I come in faster? How much faster?

8. Do I have to demagnetize the little metal posts that the tape travels over between heads? If so, should I demagnetize them before or after I demagnetize the heads? Won't the magnetic field coming out of the side of the demagnetizer's tip have some effect on a nearby head even when I'm demagnetizing one of the adjacent posts?

9. Do I have to rotate the tape guides on the spring-loaded tension arms and demagnetize them from all sides, or will just demagnetizing them from one side demagnetize the whole roller clear through?

10. How often do I have to go through this whole ritual, anyway?

—Jim Aiken
Cupertino, CA

O.K. Jim, let's start with your last question. There is only *one* way to determine the need for demagnetization and that's with a magnetometer. Now before you faint, let me say that this device is both inexpensive and simple to operate. It's like a VU meter except its zero is dead center. An excellent version is made by R. B. Annis of Indianapolis. You just touch its bottom edge (or its clip-on probe) to the suspect part and it'll read negative (left) or positive (right) magnetism in "gauss." A scale of no more than five gauss is necessary and as an explanatory note the earth's magnetism is about ½ a gauss.

In the studios I work out of, the machines are checked daily. This is due to the high volume of work done, as just running tapes can build up magnetic fields. Additional causes of magnetization are things like speakers or transistor radios placed near the deck. Occasionally a meter or even the earth's field can cause problems. Recording tape is very sensitive to extraneous magnetism. Exposure can degrade the signal by limiting or decreasing its high frequency content as well as increasing background noise (hiss) several dB. In other words, your signal to noise ratio goes down the tubes.

Your heads are made up of "soft" metal. They are very permeable and therefore magnetize very easily. However, once the influence of a magnetic force is removed, its retention or memory of that source is slight. Magnetism can build up, but demagnetization is easily accomplished. *All* the metal parts along the tape's path such as capstans, guides, rollers and spring can not only pick up magnetism but pass this on to the tape. These are "hard" metals and their retention is such that they can actually become magnets. If your "cheepo" demagnetizer is rated at less than 300 Oersteds it wouldn't be of much use here. What's an Oersted? It's simply a measurement of magnetic intensity and should be listed on your instrument's specification sheet.

The way that demagnetizers work is easily understood and understanding is the key to knowledgeable use. When you use a demagnetizer, you're exposing a metal part to a magnetic field of reversing polarity (neutral) such as 50 or 60 cycle alternating current circuits, then reducing this intensity gradually by pulling the magnetic field away. This is called cyclical demagnetization and the actual demagnetization occurs *only* during the reduction of intensity (pulling away period).

When starting demagnetization the parts should be rotated or the demagnetizer waved to insure optimum neutralization. You should always pull away to at least one to two feet for each part demagnetized. Follow these simple guidelines and you'll never have to worry about going through this whole ritual again.

1. To avoid occurrence of accidents do not keep demagnetizer in control room or anywhere else tapes are stored.
2. After cleaning deck, check parts with magnetometer.
3. Check to insure no tapes are around the area before plugging in demagnetizer.
4. Turn on at least one to two feet away from deck.
5. Approach to ⅛ inch (do not touch heads).
6. Wave slightly sideways or rotate part.
7. Don't hold in that position, but slowly pull away at no more than three to four inches per second to a point at least one to two feet away.
8. Never interrupt the demagnetizer's power during this cycle.
9. Continue demagnetizing where needed following the same steps.
10. Check all parts again with magnetometer.
11. To clear any magnetic particles out of the head gap, soak a lintless tissue in head cleaning fluid (less than 10% water) and stick this to head. Now when you approach the head, move the demagnetizer once down and once up the length of the gap, then demagnetize as usual. Any particles will adhere to the tissue.

In any case, once you have a magnetometer, you'll be able to answer your own questions about the adequacy of demagnetizing methods.

—Mike Shea
Chief Engineer—
Precision Measurement
Chief Technical Consultant—
IM+RW
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Planet Sound Studios

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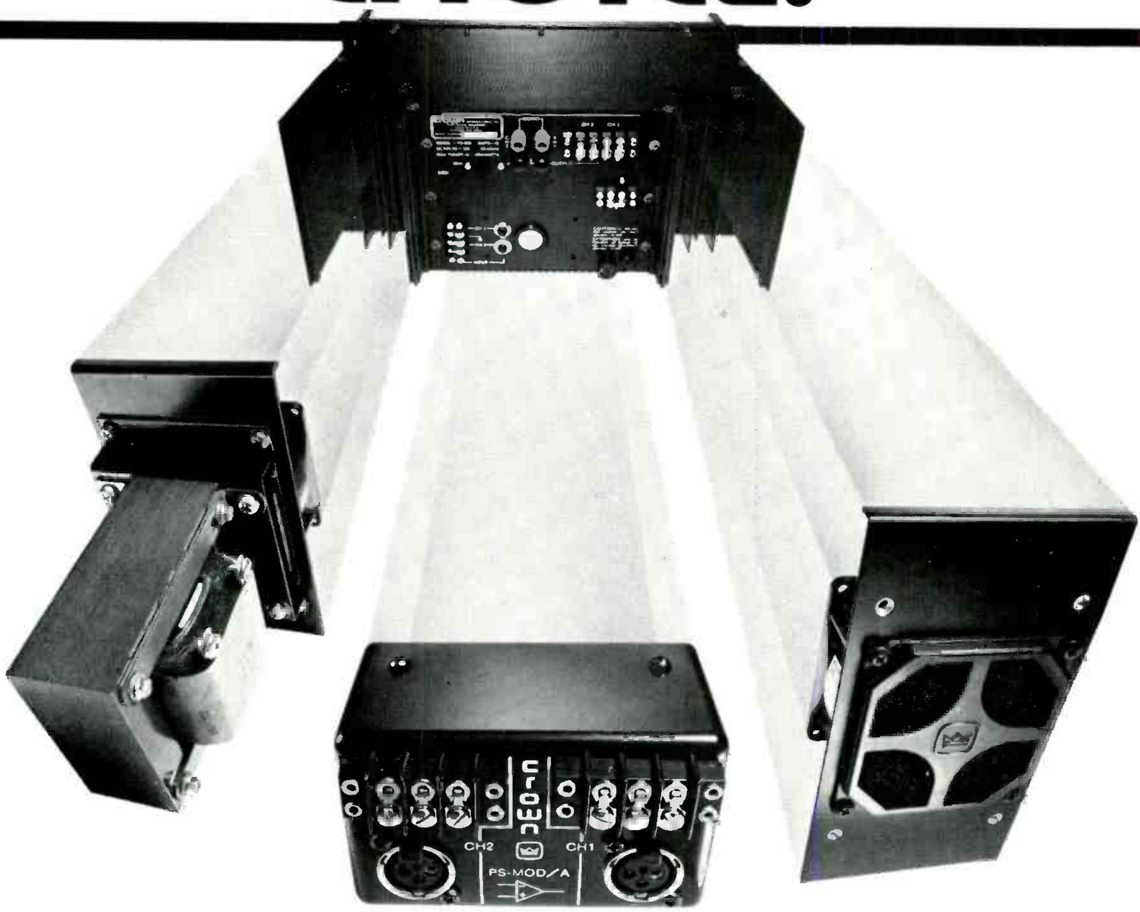

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CIRCLE 17 ON READER SERVICE CARD

Studio Notebook #12

By James F. Rupert

ATTENTION READERS who are not looking to the United States Small Business Administration for assistance in obtaining money for a new studio: this article is for both of you!

In the last episode, we talked about SBA participation loans and financial assistance. Now let's explore the potentially more important programs that SBA can offer the struggling young recording studio. Just because Uncle Sam might have opened the doors to a few dollars, is no reason to assume that the relationship has to end. Business loans are just the beginning.

Even prior to making your pitch to your banker, you could do a lot worse than to pick up the phone and call the closest field office of the SBA. One way, for example, to do worse would be to seek the services of a professional consulting firm that would give you the same excellent advice and counseling that the SBA will give you. The difference, however, is that the consulting firm will charge you many sheckels for the same information the SBA will arrange for you to receive for nothing. (Call me old-fashioned, but I'd rather get the same service for nothing anyday!)

The SBA will aid you in putting together a solid business strategy in a logical step-by-step process. Now before anyone gets the wrong idea, let me stress that the key word in that last sentence is "aid." SBA small business counselors work with you, not for you in establishing your studio's plan. Your job (should you decide to accept it) is to provide them with hard facts, educated conjecture and the wild speculation that goes into drawing up a small-business plan. Almost every aspect of recording studio organization that has been discussed in past "Studio Notebook" installments will be covered in the proper sequence to provide a solid base for your future business.

In some cases, the SBA will work with local universities and business colleges to provide real business counseling experience for the schools' top business administration majors. These students will have the full resources of their schools' faculty, resource materials and computers to work with in helping you lay the groundwork for a successful studio. Although some of them may be amateurs, take it from me that as consultants, these students are anything but rookies.

If the possibility of students advising you still makes you a litle paranoid (shame, shame!) then consider the SBA's SCORE program. SCORE is short for Service Corps Of Retired Executives and the name says it all. SCORE is a non-profit association that provides free business counseling to the small businessperson in need. Any small business that is not dominant in its field may apply for help. Counseling can be either general or about a specific problem. SCORE can review facilities, products, procedures, marketing, financing and a host of other subjects or problems.

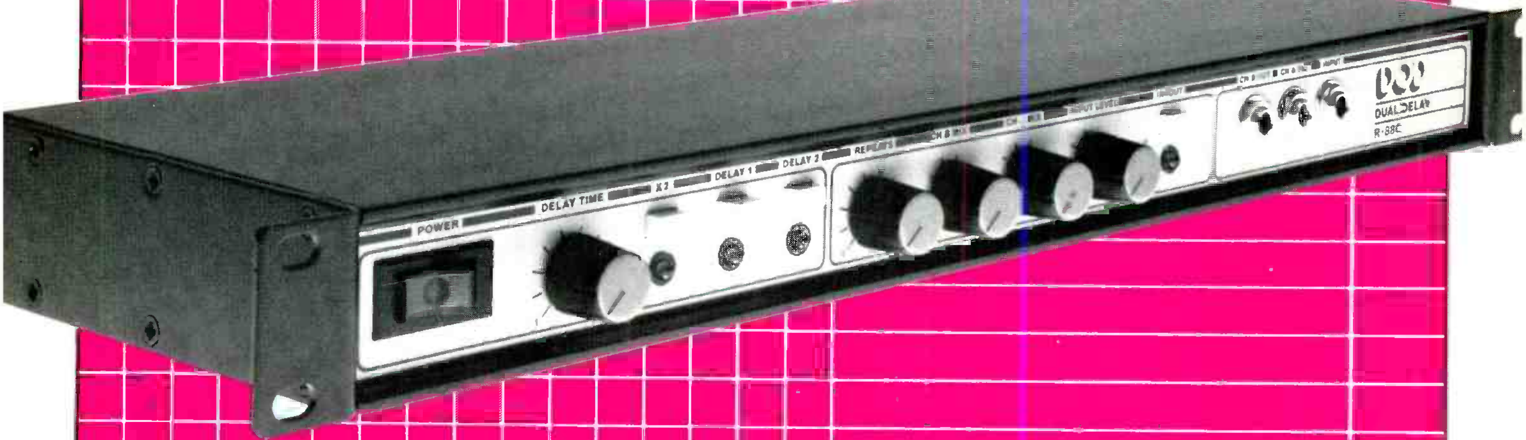
All of SCORE's counseling is on a strictly confidential basis. The counseling sessions can take place in your studio, on a remote recording job, in your unremodeled proposed studio site, in your home or anywhere else that would make sense and give the best picture of what you would like to achieve with your new recording business. SCORE can help you break bad business habits, prevent future problems from occurring and sharpen your perhaps rusty, if not non-existent, management skills.

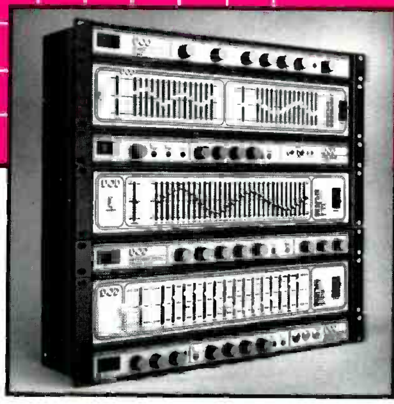
Where it is possible, the counselor's experience and expertise is matched to the applicant's needs and type of business. If necessary, additional counselors and specialists can be called in, or used as a team for those real "stumper" problems. Since SCORE is staffed and run by volunteers, you gain more than knowledge in your counselors, you gain people who care. If they didn't give two hoots or were in business just for the bucks, they would not be involved in the program in the first place. Over 8000 of these good people are ready to give you a hand, and they promise not to have their hand out in return. Can you ask for anything more than that?

Part of the specialty reserves that stand ready to work with SCORE volunteers are the members of the ACE program. ACE stands for the Active Corps of Executives, and is made up of volunteers who are still active in business. The sole purpose of both ACE and SCORE is to help you make it. Their only motive is to lend you the value of their brains, their ideas, their know-how and their experience so that you might walk out a winner from the arena that you are about to enter. No fees, no surprise surcharges, no hidden costs. Any costs of both associations are paid for through the sponsorship of the SBA. (Think about that the next time you grumble come income tax time.)

SBA also publishes an incredible series of booklets to help with almost any facet of your business that is imaginable. Some literature is free, some is available at a minimal cost. Catalogs of all SBA publications are available through two addresses: The listing of free publications (ask for SBA-115A) is available by writing U.S. Small Business Administration, P.O. Box 15434, Fort Worth, Texas 76119 or by calling (toll free, no less!) 800-433-7212. (Texas residents only can call 800-792-8901.) For the catalog of publications that carry a small charge, write to the Superintendent of

Dual Delay R-880





Description

The DOD Dual Delay R-880 is intended for echo and reverb effects. The R-880 is ideal for mono or stereo P.A.

Special noise reduction techniques make the R-880 quiet enough for even the highest gain preamps, and it incorporates some features only available in digital systems.

All this, combined with the DOD reputation for quality and service makes the R-880 an excellent choice for medium to long audio delay applications.

The Dual Delay uses both companding and emphasis to achieve its remarkably quiet operation.

Delay times of 12 ms through 500 ms are easily obtained by adjustment of the simple, straight forward controls. The front panel is divided into three sections: the delay controls; the signal controls; and the signal jacks.

The DELAY 1 and DELAY 2 switches engage each of the delay lines; therefore, at least one must be "in" to produce a delayed signal. The INPUT jack goes directly to the input level pot, so there is no input stage to overload. The CLIP indicator begins to light at about one-half of the actual clip point to allow for more headroom in the program material. The A MIX and B MIX controls are two identical mix circuits that go to separate output jacks. When using two amplifiers, the mix controls may be set differently for greater presence.

Specifications

Frequency Response:
- Dry 20Hz to 20KHz ± 1db.
- Delay 40Hz to 6KHz ± 1db.
- X2: 40Hz to 3KHz ± 1db.

Signal to Noise Ratio:
- Dry 95 db un-weighted.
- Delay 90 db un-weighted.

Input:
- 100K ohm unbalanced.

Outputs:
- Channels A and B are separate and identical. Output impedance is 600 ohms each channel unbalanced.

Indicators:
- All switches have LED lamps to indicate when they are in. The power switch is illuminated when on and the clip lamp lights when a signal over 5 volts PP is present.

Delay Range:
- Delay 1: 12 ms to 125 ms.
- Delay 2: 25 ms to 250 ms.
- Delay 1 × 2: 25 ms to 250 ms.
- Delay 2 × 2: 50 ms to 500 ms.

Size:
- Standard 1¾" × 6" × 19" rack.

Weight:
- 6 lb. 7 oz. (3 kg.)

DOD

Electronics Corporation
2953 South 300 West
Salt Lake City, Utah 84115
(801) 485-8534

Printed in USA

Documents, U.S. Government Printing Office, Washington, D.C. 20402 or write or call the national SBA office in Washington, D.C.

These pamphlets and booklets are constantly being updated to keep the SBA's clientele abreast of what's happening with business and management procedures all over the world. All language is kept simple enough to be easily read by even the most novice entrant into the respective subjects covered. A comprehensive business encyclopedia can be assembled from this literature at little or no cost to you. Once again, the government is offering this service to be used by anyone who needs it. So why not use it?

If paranoia has completely overtaken your life and you don't want any personal involvement with a government agency, then at least force yourself to call or write for a sample starter packet of information from the SBA. This packet will contain outlines of all the programs the SBA is offering as well as several basic booklets covering topics you might need some help with. As usual, the packet is free—without even a postage or handling charge.

Please keep in mind that the SBA is there to help give you a step up, not to climb over the wall for you. That little responsibility still rests on your shoulders. The bottom line validity and worth of your studio plan is up to you. SBA counseling can help point you in the right direction, but you'll have to be the one that will be laying down the shoe leather. The SBA can only recommend: the final decisions rest with you-know-who.

How does one go about getting help, you say? Simple enough, Harvey! Just make that call to your nearest field office or write to the main SBA office for a Request For Counseling Application. (If you're requesting the basic information packet, you'll find the application enclosed anyhow.) Check off the type of business you're shooting for on the form, pop it in the mail, and before you know it you'll be on your way to Entreprenuer City!

As long as you're writing or calling, ask about a subscription to "Small Business Reporter." This is the entry level equivalent of the Wall Street Journal and is (best of all) free, except for a small mailing charge. Or, you can pick this gem of a publication up at any Bank of America branch at no charge. Full details are available through the SBA.

There, in a very incomplete nutshell, are some of the highlights of further services the SBA offers to the American taxpayer. Until you can see for yourself just how genuinely interested and helpful the good people at the SBA are, it's tough to appreciate how much they can do for the new business initiate. So don't be paranoid! Go ahead and put a few of those tax dollars you've shelled out back to work for you. You owe it to yourself, your business and the clients you'll be serving better.

As for me, I've never had a problem with paranoia. And take it from me, I'm surrounded by paranoid people and I don't trust any of them. (You know, the thing I really hate about paranoid people is the way they spy on you!) Nosiree, I'm not paranoid.

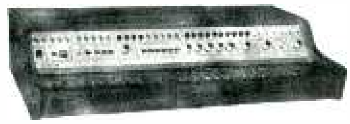
See you next time.

(...unless the little people I hear walking in my walls at night get me!)

MUSICAL NEWSICALS

SYNTHESIZER EQUIPMENT

The latest from Buchla and Associates is the 400 Series of electronic musical instruments. The 400 is essentially a computerized synthesizer using architecturally advanced hardware and specialized, high-level software to achieve a unique level of performance characteristics. Three computer processors are used within the system, each of which has been optimized to its particular task. A host computer (which may reside in or out of the instrument proper) handles user communication, data handling, and executive control, while a second processor is responsible for processing temporal data; a third processor governs the generation of sound. Unusual performance features include dynamic waveshaping techniques, multiple complex envelope generation and advanced instrument definition. Control devices include a specialized, touch-sensitive keyboard which can be organized in traditional or non-traditional fashions (and may be tuned to any imaginable scale), pressure-sensitive joysticks and analog modifiers. The 400 Series is a modular system built around plug-in circuit cards. The latest addition to the system is the model 404A. Featuring an expanded memory capacity of 126 Kbytes, the unit is able to run sophisticated music programming languages for real-time interaction with moderately large data bases. At present, two such languages are available—MIDAS and PATCH V. MIDAS combines multiple instrument definitions, tunings and waveshapes with a comprehensive score editor which



assumes a fairly traditional approach to musical structure, while PATCH V provides generalized facilities for translating input gesture into instrumental response in any complex, arbitrary relationship, without any bias toward traditional musical structure.

CIRCLE 40 ON READER SERVICE CARD

An interesting development in the field of synthesizers is the VSC I Violin Synthesizer Controller from Zeta Systems. The VSC I allows the violinist access to synthesizer technology without compromising his playing style or limiting the expressive capabilities of a violin-family instrument. The instrument itself is crafted from choice hardwoods, and its freedom from acoustical considerations has allowed for a unique and striking design. Each string has a separate, full-range pickup mounted in the bridge for freedom from intermodulation effects. Rather than pitch-to-voltage converters, the VSC

I uses laser-trimmed position transducers mounted in the fingerboard; along with the pickups, these transducers produce all the gates, triggers, and pitch and amplitude control voltages to control a conventional polyphonic synthesizer. An optional card adapts the quad outputs for use with monophonic, lead synthesizers. An unusual feature of the instrument is the use of phase-locked-loop techniques to constantly maintain tuning between the vibrating string and the synthesizer according to the front panel master and fine tuning controls.

CIRCLE 41 ON READER SERVICE CARD

GUITARS

Ibanez has introduced an interesting new series of electric, semi-acoustic guitars for the musician who wants the sound and feel of a semi-acoustic but prefers a more compact body style. The new Stagemaster series has a body which is comparable in size to a solid-body electric, while retaining the sound of a semi-acoustic; actual body dimensions are 16½-in. in length and 14-in. maximum width. Two models are currently available: the AM50 and the top-of-the-line AM205. This latter model has a burled mahogany top and back (finished in Antique Violin) with an ash/mahogany/maple center block. The neck is mahogany with ebony fingerboard and abalone/acrylic inlays, and has a 24¾-in. scale length. On the hardware side, the AM205 has two Ibanez Super 58 pickups with Alnico III magnets, and gold-plated hardware including adjustable bridge, quick-change tailpiece, and 18 to 1 Velve-Tune

tuning machines. The AM50 is a similarly configured instrument, but at a more moderate price. The brown sunburst body of the AM50 has maple top, back, and sides with a solid ash center block, while the mahogany neck has a rosewood fingerboard and dot inlays. The hardware and electronics of the AM50 are identical to the AM205 except that they are chrome-plated rather than the fancier model's gold plating.

CIRCLE 42 ON READER SERVICE CARD

Gibson has announced a new series of modestly-priced guitars under the Epiphone USA banner. The intention of the new line is to provide a high-quality instrument which benefits from Gibson's experience and stringent quality control, and which is American-made—but at a price which is competitive with the imported mid-priced guitar models. The Epiphone USA line is currently comprised of two models, the Spirit and the Special, both of which are double-cutaway, solid-body models. The Spirit is reminiscent in styling of the early Les Paul Special, but with today's electronics and hardware, while the Special is similarly a contemporary interpretation of the Gibson SG styling. The two basic models are available in single- or double-pickup models, the double-pickup version having two volume controls, a master tone control, and a three-position pickup selector.

CIRCLE 43 ON READER SERVICE CARD

New from Ibanez is the Musician 924 bass guitar, a two-pickup model with active equalization circuitry. The 924 has a maple and mahogany neck which runs the full length of the contoured, solid ash body; the body has deep double cutaways to allow

full access to all 24 frets. An exclusive Ibanez feature is the use of steel tone bars embedded in the neck to eliminate dead spots. The instrument has two pickups, an Ibanez Super P5 and a J5, along with a new pickup balancer which allows any blending of the two pickups' sounds with a single knob. The Musician 924 bass features the EQB tone system, which gives the player three boost/cut controls in distinct frequency bands for a wide range of tonal possibilities.

CIRCLE 44 ON READER SERVICE CARD

The latest member of the DAION guitar family from MCI, Inc. is the Mark Jr. model. The Mark Jr. is the least expensive member of the DAION flat-top acoustic lineup and is available as a standard acoustic or equipped with a transducer system similar to that on the DAION 81 acoustic/electric guitar. The Mark Jr. has a cedar top, mahogany back and sides, mahogany neck with 22-fret rosewood fingerboard, and enclosed, geared tuning machines.

CIRCLE 45 ON READER SERVICE CARD

MUSICAL INSTRUMENT AMPLIFIERS

News comes from Mesa Engineering of—would you believe—the Son of Boogie, also known as the S.O.B., or simply the Son (all three names are trademarks, by the way). The Son is basically a re-issue of the famous, original Mark I Boogie amp, featuring that model's basic, straightforward format rather than the more sophisticated design of the Boogie Mark IIB. The Son is a 60 watt combo amp with a single BlackShadow 150 watt rated 12-in. speaker. The Son features a refined version of Boogie's classic three volume control overdrive system which allows easier adjustment of the amp's sound from a crisp rhythm sound to a blistering sustained lead sound.

CIRCLE 46 ON READER SERVICE CARD

Good news for tube amp aficionados comes from Groove Tubes, who have announced the expansion of their line of selected preamp tubes and matched output tubes to include virtually all models required for domestic or European tube amplifiers. Recent additions to their lineup include matched pairs, quartets, and sextets of EL34s (for Hi-Watt, Vox, and English Marshall), EL84s (for early Gibson amps), 7027As (for Ampegs), and 6CA7s (for early Music Man amps). Groove Tubes also offers several rare, antique tube types for antique and classic amps such as the tweed series of Fender amps; these rare types include 12AY7, 12AZ7, 6SN7, 6SL7 and 6SC7. Also available are original-type replacement parts such as ceramic tube sockets and tube shields.

CIRCLE 47 ON READER SERVICE CARD

Unicord, the new American distributor for Dynacord, has announced the availability of the Dynacord BS 412 bass amplifier. This model is a compact combo amp with 200 watts rms of solid-state power and sophisticated performance features including a switchable compressor for distortion-free sustain, and a switchable harmonic distortion circuit for the dirty sound. Other features include high- and low-frequency shelving EQ plus a 7-band graphic equalizer, high and low sensitivity inputs and an effects insertion loop. Also available from Dynacord is the BS 414 extension cabinet and the BS 408 which is the same amplifier in a rack-mount, head-only configuration.

CIRCLE 48 ON READER SERVICE CARD

MUSICAL INSTRUMENT ACCESSORIES

For the sophisticated user of electronic effects devices, the Boss Division of Roland Corporation offers the SCC-700 Sound Control Center, a computer-controlled effects programmer. The SCC-700 is designed to be connected to up to seven effects devices of any type (rack-mount, pedal, or rocker) which it will then access in any combination and, more uniquely, in any sequence. The unit stores up to 32 patch programs, organized in four banks of eight patches, each of which controls which effects are activated, the order of the effects in the system, and the levels of the main output and a sub-output. The effects combinations are programmed simply by touching the corresponding front-panel pushbuttons in the sequence desired, and storing this is one of the memories; these patches are then recalled in the Monitor mode (which displays the patch without actually activating it) or in the Play mode. A Manual mode is also provided in which the unit's selector switches function in real time. The SCC-700 system is comprised of three components; the SCC-700 itself, which has all of the facilities described, the optional SCC-700F Foot Controller, which allows convenient selection of the patch programs and which connects to the main unit via multi-conductor cable, and the SCC-700B, which is a pedal board designed to house and power the seven effects controlled by the system and connect them to the main unit.

CIRCLE 49 ON READER SERVICE CARD

Unicord has announced the availability of the Dynacord VRS23 Vertical Reverberation System. This unit is a fully electronic echo/reverb system in a 19-in. rack mount chassis. The unit uses newly developed "vertical reverberation characteristics" which are said to prevent the "tile bathroom type" of sound while still providing sizzling highs. Besides echo up to 400 milliseconds and reverb, the VRS23 is also capable of producing a variety of spatial stereo effects when used in a recording system or with two amplifiers on stage.

CIRCLE 50 ON READER SERVICE CARD

Also available in the Dynacord line is the TAM21 Stereophonic Studio Phaser/Flanger. This modulated delay line effect is designed to produce four basic effects: Flanging 1, an extreme jet-flanging effect; Flanging 2, a stereo, mirror-image flanging/pitch shifting effect: Double Track 1, a contrary left/right double tracking effect with short delay times for chorus effects and Double Track 2, a synchronous left/right double tracking effect with longer delay times for echo effects. The TAM21 has a control voltage input and output, making it possible to gang two or more units together for more complex synchronous or semi-synchronous effects.

CIRCLE 51 ON READER SERVICE CARD

A most unusual electronic tuning device has been designed by an electrical engineer who is also a French horn player at the Sydney, Australia Opera House. The device, known as the Pitch-Pro Automatic Tuner, is manufactured and marketed by the engineer's company, ESIS, Pty., Ltd. Unlike most tuners, which require the user to pre-select the note to be tuned on a switch, John Dobbie's Pitch-Pro gives an automatic, continuous readout of the note played and its intonation. Obviously, this is most appropriate for single-note instruments such as the inventor's French horn, but it is also of value for any instrument, since it shows when individual notes or entire registers are out of tune in a given instrument and plus provides continuous feedback on the accuracy of a musician's intonation. The tuner's display is a vertical row of lights to show flatness, perfect pitch, or sharpness to within about a 1/100 semitone accuracy; a note display model shows the name of each note as it is played.

CIRCLE 52 ON READER SERVICE CARD

Metone has announced the introduction of the Model 23F electronic metronone. The battery-operated device reliably and accurately produces any tempo from 40 to 220 beats-per-minute. Tempos are monitored visibly via an LED, and audibly via a built-in speaker. The unit features a reverse-side screened Lexan faceplate and a high-impact case, and since it has only one moving part, the reliability is said to be extremely good.

CIRCLE 53 ON READER SERVICE CARD

THE PRODUCT SCENE

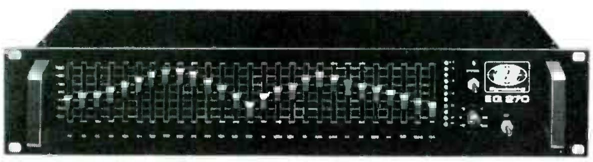
PHASE LINEAR'S NEW GRAPHIC EQUALIZER

Phase Linear announces the introduction of the Model E27 single channel 27 band Graphic Equalizer for professional applications.

According to the manufacturer, the E27 utilizes state variable filters to achieve amplitude change independent of bandwidth. The design ensures one-third octave equalization throughout the adjustment range, and eliminates broadened bandwidths at small adjustment settings.

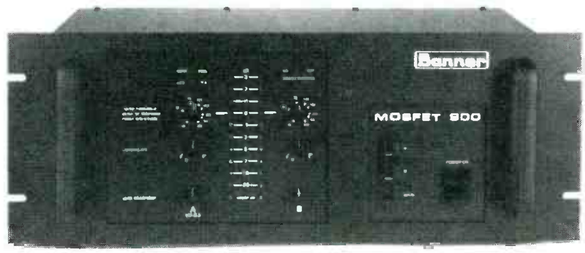
Other features of the E27 include: +12 dB/-15 dB control range; switchable 40 Hz high-pass filter; 12 dB available gain; signal/noise ratio of 111 dB below max output with sliders centered; passive bypass; and balanced input and output. The price of the E27 is $549.

DYNACORD'S THIRD OCTAVE GRAPHIC EQUALIZER



Unicord announces the release of a new one-third octave equalizer in 19-inch standard design for use with P.A. systems of any kind, or for use as a multi-sound filter with high class instrumental systems. The resonant frequencies of the filters of the EQ 270 corresponds to ISO standards, so that one can work with usual AF analyzers without any problems. The unit includes electronically balanced XLR input and output connectors; a bypass switch for electronic switching, and an LED level chain (logarithmic extended display range) which blinks when the input is overloaded. It also contains a specially designed safety power supply to prevent ground loops.

BANNER'S MOSFET 900



The Banner MOSFET 900 features power output ratings of 300 watts RMS per channel into 8 ohms and 450 watts into 4 ohm loads. It utilizes power MOSFETs for increased reliability and improved transient response. In the mono bridging mode the rated power is 900 watts into 8 ohms. It has a dual-speed thermostatically-controlled fan, front panel level controls, LED meters, and a new stereo limiter. The meters can be front panel calibrated to a continuous range of output from 56 to 450 watts rms into 4 ohms. This calibration point provides the "0" reference point of the meter and is automatically the limiter threshold point. The limiter slope of the unit may be varied from soft to hard limiting action and is fully defeatable. The limiters may be set for stereo tracking to prevent image shift. All controls are front panel operated, including circuit breakers and input impedance selection. Its active balanced inputs on XLR and ¼-inch connectors are standard. The outputs have full speaker protection, including DC fault sensing. The distortion typically ranges from 0.005 percent to 0.2 percent, depending on power and frequency. The MOSFET devices provide positive temperature stability. The price is $1,195.

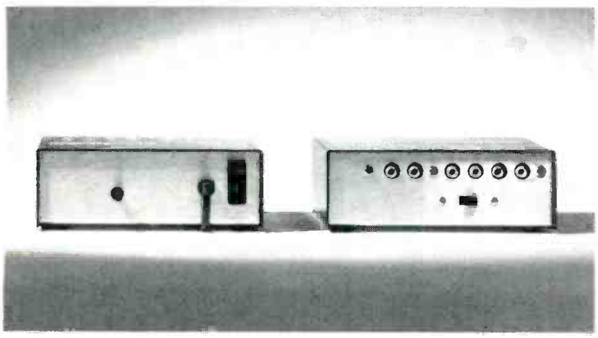
SHURE'S CONDENSER MIKE NOW WITH OMNIDIRECTIONAL CARTRIDGE

Shure's SM81 Condenser Microphone is now available with a built-in omnidirectional cartridge.

The new model, called the SM80, has an omnidirectional pickup pattern which eliminates problems with proximity effect. The SM80 has a selector switch which permits any of three low-frequency responses to be selected to match the application.

The microphone also has an adjustable 10 dB attenuator pad to allow use with very high SPL signals. The SM80 features a backplate structure designed to maximize signal-to-noise ratio and insure long-term charge stability. It is capable of withstanding extreme temperature change, high humidity and physicial punishment (as long as it is not too cruel or unusual). It is available in two versions: SM80-LC (without cable) and SM80-CN (with cable with 3-pin professional audio connector). User net prices are $327 for the SM80-LC and $348 for the SM80-CN.

ACE's CROSSOVER



Ace Audio Company announces the new model 5000-6 crossover, designed to operate with any subwoofer to extend bass response into the 20-35 Hz region.

The 5000-6 features a 100 Hz crossover (other frequencies are optional) and can be used with one or two subwoofers in mono connection. A bridging amplifier is built in, as are a bypass switch for the subwoofer and level control; the level control allows boosting the subwoofer level by up to 10 dB over the main speakers. By using active filter circuits, the distortion is kept in the 0.002 percent region at 2V output level.

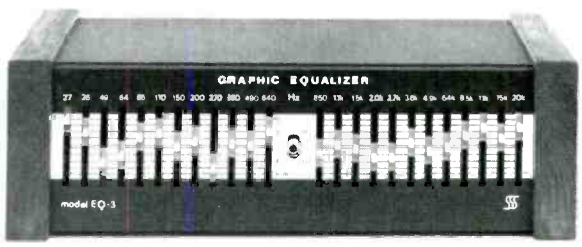
The 5000-6 also features 1 percent metal film resistors, and polystyrene capacitors for an accurate crossover frequency (within 3 percent). Two different slopes are used: for the subwoofer—18 dB octave; for the main speakers—6 dB octave. This gives a smooth transition between speakers, and allows use of an existing full-range speaker with a subwoofer add-on. The price of the 5000-6 is $161, wired and tested.

AUDIO SWITCHER

Modular Audio Products has announced that its eight-circuit switching card is now available in two versions: the 4088 with buffer amplifier and the 4088-1 as a switcher alone. Both units are pin compatible. The 4088-1 MOSFET switcher provides for individual remote double-throw switching of eight line-level inputs. A small jumper plug allows the user to vary the connections between four of the inputs and a group of four of the MOSFET switches without rewiring the frame connector. An optional BTA module (Balanced Transformerless Amplifier MAP Model 5008) can be used in the input circuit of two switches to serve as a buffer amplifier to ensure isolation and non-loading of the inputs. The 4088 also provides up to 14 dB of gain to make up for fader setting loss when the card is used as a channel routing switcher in console applications. A number of 4088 or 4088-1 cards may be connected in parallel permitting channel routing to an almost-unlimited number of output buses or to an assembly of large crossbar switching arrays. Many other custom switching circuits can be devised by custom wiring the jumper plug connections or the connector.

Other features are crosstalk of −80 dB, off attenuation of 100 dB, distortion of 0.05 percent. The 4088 is designed for audio switching applications at peak levels of −10 dBm to 20 dBm. The switching logic is TTL compatible with permissible switching voltages from +4.5 Volts to +15 Volts.

GRAPHIC EQUALIZER KIT



The Symmetric Sound Systems of Santa Rosa, California has introduced a relatively inexpensive graphic equalizer kit. The Model EQ-3 Graphic Equalizer features 24 bands per channel, evenly spaced over the three decade range. Odd numbered bands are in an independent circuit from the even bands, eliminating interaction between adjacent controls for easier high-resolution room and speaker equalization. The range of the EQ-3 is ±10 dB nominal. The distortion is under 0.02 percent at 1 kHz, and the S/N ratio exceeds 88 dB at the 2V rated output. The maximum output is 8.5V rms. It has solid walnut end panels, standard on each channel. A monaural unit measures only 10 inches — 3$\frac{3}{16}$ inches — 4$\frac{3}{8}$ inches. Kit prices are $125 for mono, and $225 stereo. The units area also available assembled.

NEW BASS BIAMP

The Gallien-Krueger 800R bass biamp contains a preamp, electronic crossover, and dual power amps all in one package. It is 5¼-inch rack mountable, and weighs 20 pounds. The low end power amp is rated at 300 watts rms into 4 ohms, while the high end is rated at 100 watts rms, into 8 ohms. Each power amp has its own level control for precise balance of low and high frequency power. The internal electronic crossover has a crossover point that's continuously variable between 100 and 1000 Hz. The preamp controls are: input attenuation (10 dB), volume, voicing filters for low, midrange, and high frequencies, four bands of active equalization, and footswitchable boost. A low-impedance direct out (XLR) and effects loop are included on the rear panel. The price of the 800RB is $899.

TWO-TRACK TAPE HEADS

Complete ½-inch 2-track head assemblies for MCI JH110A tape machines have been announced by JRF Company, Inc. of Hopatcong, N.J. Designed by JRF for retrofit applications, the new heads are claimed to provide substantial improvement over standard ¼-inch 2-track performance specifications. In addition to MCI, ½-inch 2-track heads are also availabe for Ampex and Scully tape machines. JRF, a company engaged in precision head relapping and assembly alignment services, also offers direct replacement heads for most studio, mastering and tape duplicating equipment.

SF-10 SUBSONIC FILTER

Nakamichi has announced the introduction of a new SF-10 Subsonic Filter which has been designed to minimize subsonic interference when making cassette recordings. The SF-10 consists of a pair of passive subsonic filters that plug directly into the left and right channel line-input terminals of a cassette deck or into the corresponding record-output terminals of a preamplifier, integrated amplifier, or receiver.

The SF-10 Subsonic Filter can improve recording quality when taping from a source that contains substantial amounts of infrasonic energy. Such infrasonic energy can emphasize the low-frequency modulation effect and cause a wow-like sound in playback. Under extreme conditions, it can even cause tape overload. The SF-10 is compact, economical, and simple to use, while providing 10 dB attenuation at 10 kHz and progressively greater attenuation at lower frequencies, thus eliminating all infrasonic energy prior to recording.

PURE CHROMIUM DIOXIDE BULK CASSETTE TAPE

Agfa-Gevaert introduced a new chrome cassette tape at the AES Convention in October. The new tape, PE 627 for C-60, and PE 827 for C-90, combines a high-frequency response (generally associated with chromium dioxide) with mechanical precision. It uses a pure chromium formulation, instead of a gamma-ferric equivalent. The PE 627 and PE 827 were developed in response to the growing demands for a higher level of reproduction from standard cassettes. Applications of the new tape include cassette for broadcasting and pre-recorded audiophile cassettes. PE 627/827 is wound onto the Agfa-Gevaert patented stack hub, in lengths of 8,200 feet for C-60, and 11,500 feet for C-90.

FOUR HITACHI DECKS

Among four new cassette recorders from Hitachi is the model D1100M, \$600, a solenoid-operated three-head unit with computer-controlled bias and EQ known as ATRS (automatic tape response system). A two-motor, double-capstan deck, the D1100M uses a 24-LED peak-hold signal metering system. Specs include frequency response (with metal tape) of 30 Hz to 19 kHz; S/N (with Dolby on) of 69 dB; wow-and-flutter of 0.038 percent.

Separate R and P heads also are offered in the DE95. This \$430 deck has features and specs similar to those of the D1100M but it lacks the automatic tape response system.

NEW DISTRIBUTION AMPLIFIER

The Protech Audio Corporation announces the addition of a new distribution amplifier to their product line. Designated the Model DA1528, it is designed to fill the requirement and demand for a high-quality, cost-effective audio distribution system. The unit features eight balanced, 600-ohm transformer-isolated outputs; a balanced bridging transformer-coupled input; a user programmable VU meter; selectable metering of input and output; front panel mounted circuit breaker, and output level control. Another feature of the DA1528 is that the entire electronics unplugs from the front, eliminating the need to go behind the rack after the initial installation. Spare boards are available for multi-station installations.

TWO AND EIGHT TRACK TAPE RECORDER/REPRODUCERS

Teac's TASCAM Division has just introduced two new audio tape recorder/reproducers: the two track ¼-inch Model 52 and eight track ½-inch Model 58. These production machines can accommodate up to 10½-inch reels and operate at 15 ips; the 52 also operates at 7.5 ips with NAB or IEC equalization. Full sync recording is standard on both machines, and there is a choice of optional remote controllers.

The 52 and 58 both have an extremely heavy duty industrial transport to help it survive tough broadcast and recording studio environments. The 52 and 58 have rear panel accessory connectors which are compatible with most popular SMPTE controller/synchronizers. All three of the Series 50 motors—supply, takeup reel and capstan—support remote controller/synchronizer commands. Manual cueing is therefore not necessary in order to remain locked-up during rapid winding and searching.

All transport functions are governed by a microprocessor which detects tape motion by means of five different photo-interruptors. These non-contacting sensors eliminate flutter-causing friction and wear, and continuously inform the computer of the tape direction, speed, tension, and stop or end-of-tape condition. The transport is programmed to protect master tapes. A bipolar, 8-output power supply provides ample headroom while isolating logic, motor, solenoid and audio functions. AC power interruption results in orderly transport shutdown.

A fluorescent tape counter displays positive and negative real time. Coarse and fine speed control sliders are provided for Vari-Pitch mode wherein percentage of speed can be displayed. The Zero Search button rapidly moves tape forward or in reverse to 00 min 00 sec.

All audio and transport trimmers are accessible from the front, even when the machine is rack mounted. The meter panel swings down to reveal the audio trimmers, which are on the front edge of plug-in PC boards (one per channel), and the transport cover plate is easily removed for access to the servo adjustments. Both Series 50 transports use TASCAM's own long-life high-density permalloy heads that typically deliver 40 Hz to 22 kHz response in sync or repro mode; a high-beta staggered-core erase head eliminates the need for bulk erasure prior to initial recording. Hand-lapped heads and low frequency compensation keep head bumps under 1 dB.

Recording Techniques Part 10

by Bruce Bartlett

After spending time and effort creating a beautiful mix, the last thing you want is to lose its quality when committing it to tape. Unfortunately, the sound you crafted during recording can be degraded by tape hiss, dirty heads, misaligned tape machines and mishandled tape. But these problems can be prevented. In this article, we'll tell how to optimize the performance of recording tape and analog tape machines. We'll also describe how to edit recordings into a tight, professional package.

Cleaning

Here's a tape-related problem we've all encountered: oxide shedded from the tape accumulates on the recorder heads. This layer of deposits separates the tape from the heads, causing high-frequency loss and dropouts. In addition, buildup of oxide on the tape guides, capstan and idler wheel can cause flutter. So it's very important to clean the entire tape path frequently.

Use the cleaning agent recommended by the tape recorder manufacturer. Denatured alcohol and a Q-tip are often used, although head-cleaning fluid containing silicon oil is preferred. Avoid tolvene and carbon tetrachloride. Clean your machines after every eight hours of intermittent use, after every four hours of continuous use, before alignment and before every recording session. Allow a few minutes drying time.

Demagnetizing

Tape heads and tape guides also can accumulate residual magnetism which can partially erase high frequencies, add tape hiss and cause clicks at splices. This residual magnetism can be eliminated with a tape-head *demagnetizer* or *degausser*, available from any sound dealer. Essentially an electromagnet with a probe tip, the demagnetizer produces a 60 Hz oscillating magnetic field. By touching the probe tip to the heads and tape guides, you magnetize them; and by slowly pulling the tip away, you diminish the induced magnetization until no magnetic field is left.

The probe tip should be covered with tape, if necessary, to avoid scratching the heads. Generally, only the gapped types are strong enough to be effective.

The technique of using a demagnetizer is critical. Touching it to the heads and quickly removing it may magnetize the heads worse than when you started. Plug in the demagnetizer at least 3′ from the machine and bring it up slowly to the part to be demagnetized. After touching the part with the probe tip, remove the demagnetizer *slowly* to 3′ away so that the induced magnetic field gradually diminishes to zero. Demagnetize each tape head and tape guide one at a time in this manner, then turn off the demagne-

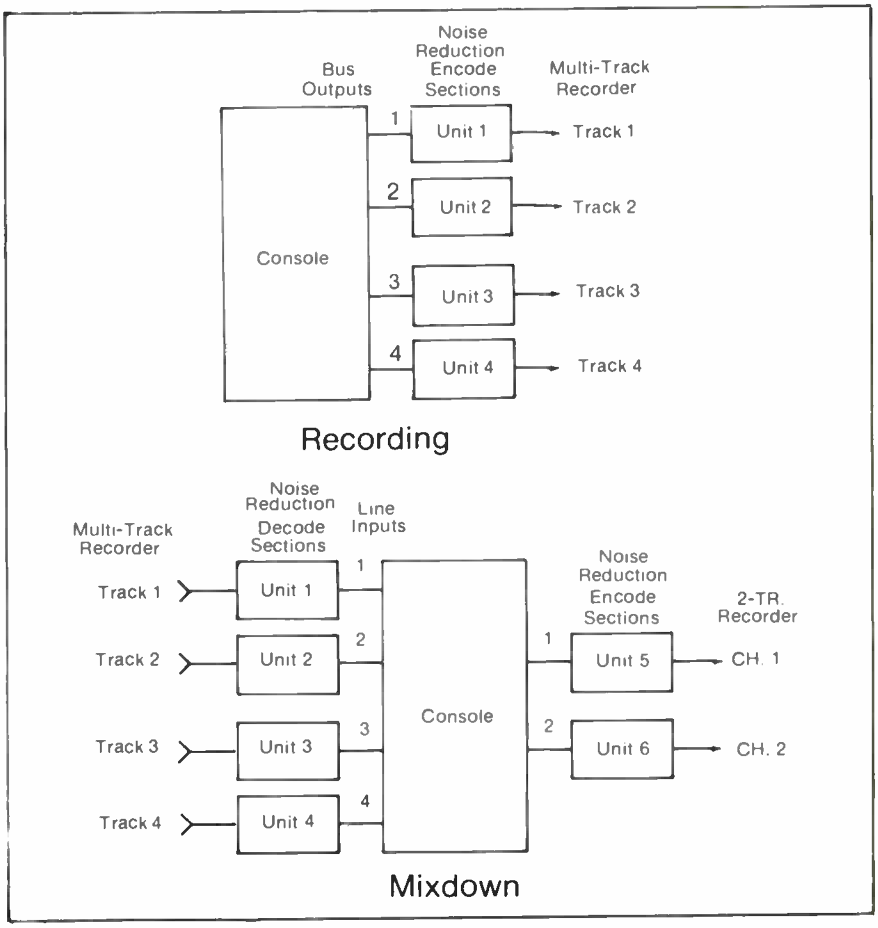


Figure 1: Noise reduction applied to multi-track master tape and to 2-track tape master.

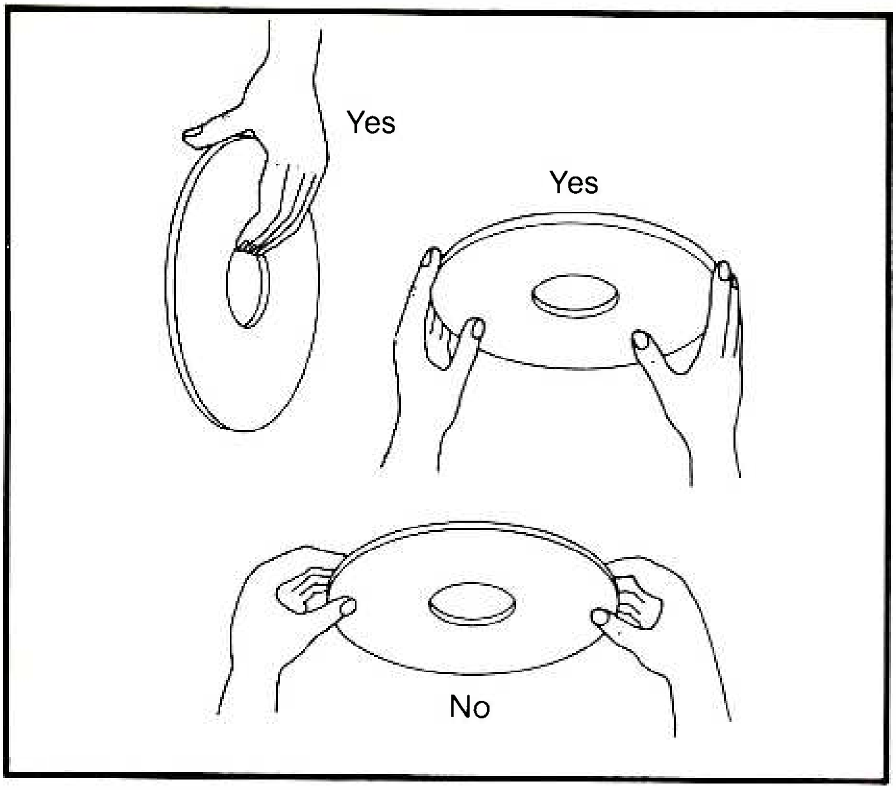


Figure 2: Handling tape reels.

tizer only when it's at least 3′ from the machine. Demagnetize your machines after every 20 hours of use and before playing an alignment tape. The same precautions about slow operation and 3′ turn-off apply to bulk tape erasers as well.

Alignment

Alignment or *calibration* is the adjustment of tape-recorder circuitry and tape-head azimuth for optimum performance from the particular type of tape being used. It's a complicated procedure not recommended for beginners. Professional recording engineers align their machines before every session to ensure flat frequency response, maximum signal-to-noise ratio and lowest distortion.

Some home and semi-pro recorders are not designed for easy alignment. The internal pots to be adjusted may not be easily accessible. In that case, the alignment is usually left alone, and you use the brand of tape for which the machine was adjusted. Some decks have switches for setting the bias and EQ. The bias switch is set according to the manufacturer's directions, usually for maximum playback output of a 700 Hz tone recorded on the machine. Then the EQ switch is set for flattest high-frequency response.

To perform a complete alignment, you'll need a small screwdriver, an audio-frequency generator and a standard playback alignment tape. Information about such tapes is available from: (1) Ampex, 2201 Lunt Avenue, Elk Grove Village, IL 60007; (2) Audiotex, 400 South Wyman, Rockford, IL 61101; (3) Magnetic Reference Laboratory, 999 Commercial Street, Palo Alto, CA 94303; (4) Magnetic Research Laboratories, 229 Polaris Avenue, Suite 4, Mountain View, CA 94043; (5) Nortronics, 8101 West 10th Avenue North, Minneapolis, MN 55427; (6) Standard Tape Labs, Inc., 26120 Eden Landing Road, Hayward, CA 94545; and (7) Taber Manufacturing and Engineering, 2081 Edison Avenue, San Leandro, CA 94577.

Follow the tape recorder instructions regarding calibration. Basically, you'll perform the following operations:

1. Using the alignment tape, play the 15 kHz tone and adjust playback-head azimuth for maximum output or for best phase match between channels (using a scope).
2. Adjust the high-frequency playback EQ to achieve the same output level at 700 Hz and 10 kHz. Or try for the flattest overall response if several tones are on the tape. Don't adjust the low-frequency EQ yet.
3. Play the 700 Hz or 1,000 Hz tone recorded at standard operating level and set the playback level to read 0 VU, -3 VU or -6 VU as recommended by the recording tape manufacturer.
4. Using the desired blank tape, record a 15 kHz tone and adjust record-head azimuth for maximum playback output, or for best phase match between channels.
5. While recording a 1 kHz tone, set the bias to achieve maximum playback level. Or turn up the bias past that point (overbias) until the output drops ½ to 1 dB. Overbiasing reduces dropouts and modulation noise. Consult the tape manufacturer's directions for alternative overbias settings.
6. While recording tones of 10 kHz, 100 Hz and 700 Hz, adjust the high-frequency record EQ and low-frequency playback EQ to achieve the same playback output level at all frequencies. Or use many tones to achieve the flattest overall response. Record the tones at 0 VU for 15 ips, -10 VU for 7½ ips and -20 VU for cassettes.
7. Feed a tone at 0 VU from the console to the recorder. Set the record level so the recorder meter also reads 0 VU on playback.
8. Set the "record cal" so the meter reads 0 VU on "input" or "source."

After calibration, your tape machine will operate as well as possible with the particular type of tape you're using. The playback signal should sound identical to the input signal (except for some added tape hiss).

Reducing Print-Through

Print-through is the transfer of a magnetic signal from one layer of tape to the next, causing an echo or pre-echo. It is especially audible in recordings with many silent passages, such as narration. To minimize print-through:

- Demagnetize the tape path (because stray magnetic fields increase print-through).
- Use 1½-mil tape (thinner tape increase print-through).
- Use noise-reduction devices (discussed in the next section).
- Store tapes at temperatures under 80°F, and don't leave tapes on a hot machine (because heat increases print-through).
- Store tapes tail out. That is, after playing or recording a tape, leave it

on the take-up reel. Rewinding a tape about 15 minutes before playing helps to reduce print-through that may have occurred during storage. (This measure becomes less effective as the storage time increases.) In addition, tail-out storage results mainly in post-echo, which is less audible than the pre-echo emphasized in tapes stored rewound.

- Rewind tapes in storage at least once a year to allow print-through to decay.

Noise Reduction

The recording process also adds undesirable tape hiss to the recorded signal, degrading the clarity of the mix. Tape hiss becomes especially audible during a multi-track mixdown because every track mixed in adds to the overall noise level.

Fortunately, noise-reduction devices such as Dolby or dbx are available to reduce noise and print-through added by the tape recorder. Note that they *don't* remove noises in the original signal from the console. One channel of noise reduction is needed per tape track. These units connect between the console bus outputs and the appropriate tape-track inputs, and also between the tape-track outputs and the appropriate console line inputs (as in *Figure 1).*

These noise-reduction devices work by compressing the signal during recording and expanding it in a complementary fashion during playback, reducing tape noises in the process. The compressor part of the circuit boosts the recorded level of quiet musical passages. The expander part works in a complementary way on playback, turning down the volume during quiet passages, thereby reducing noise added by the tape. During loud passages (when noise is masked by the program), the gain returns to normal.

A compressed tape is described as *encoded;* the expanded tape is called *decoded.* If an encoded tape is played without decoding, the dynamic range and frequency response will sound altered.

With dbx, the compression ratio is 2:1. That is, a program with a 90 dB dynamic range is compressed to 45 dB, which is easily handled by a tape recorder with a 60 dB signal-to-noise ratio. Then during playback, the dynamic range is expanded back to the original 90 dB. Use of dbx improves signal-to-noise ratio by 30 dB and increases headroom by 10 dB. The dbx circuit also includes pre-emphasis (treble boost) of 12 dB during recording and complementary de-emphasis (treble cut) during playback to reduce modulation noise.

The Dolby A system divides the audible spectrum into four separate frequency bands which are compressed and expanded independently. In addition, Dolby operates only on quiet passages—those below about -10 VU. High-level passages do not need noise reduction since the program masks the noises. This system yields a 10 dB improvement in signal-to-noise below 5 kHz and up to 15 dB improvement at 15 kHz. Dolby B, a lower cost system, operates only at high frequencies to reduce tape hiss by up to 10 dB. Dolby C works over a slightly wider range and reduces noise up to 20 dB.

When using Dolby, you must record a calibration signal called a *Dolby tone* on tape before the regular program. During playback, the Dolby input level is set so that the Dolby tone recorded on tape lines up with the "Dolby level" mark on the Dolby meter. Then the expander circuitry will track the recording properly. If the level is improperly set, the frequency response and dynamic range are altered slightly. Fortunately, there is some room for error since these alterations occur in low-level signals and are, consequently, hard to hear.

Both Dolby and dbx have advantages and disadvantages. Compared to Dolby, dbx provides more noise reduction and requires no calibration tone or careful level setting. On the other hand, dbx seems to exaggerate dropouts more than Dolby does. Dolbyized recordings are relatively free of noise "breathing," which is sometimes audible on a dbx'd tape as fuzziness accompanying bass- or bass-drum solos.

Dolby- and dbx-encoded tapes are not compatible with each other, and cannot be played properly without decoding through the appropriate unit. So, if you plan to send your master tapes to another studio, check that the studio has the same type of noise reduction you want to use.

When using noise reduction, avoid saturating the tape while recording. Otherwise the attack transients may be altered during playback through the noise-reduction unit. Encoded tapes can be copied directly from one recorder to another without decoding in between the two machines; just be sure to copy the Dolby tone if the master tape has one.

Tape Handling and Storage

Careful handling and storage of tape reels is essential to avoid damaging the tape and the signals recorded on it.

If you examine a reel of used recording tape, you may see some edges or layers of tape sticking out of the tape pack. These edges can be damaged by pressure from the reel flanges, causing dropouts and high-frequency loss. For this reason, never hold a reel of tape by squeezing the flanges together. Instead, hold the reel in one hand by putting your fingers in the hub and your thumb on the flange edges (as in *Figure 2).* Or hold a reel in two hands with extended fingers on the flange edges.

To prevent edge damage during storage, leave tapes tail out after playing or recording to ensure a smooth tape pack. Repair or discard reels with a bent flange. Reels left out in the open can collect dust, so keep them in boxes. Store tape boxes vertically—not stacked. The preferred storage conditions are 60°-75°F, 35%-50% relative humidity.

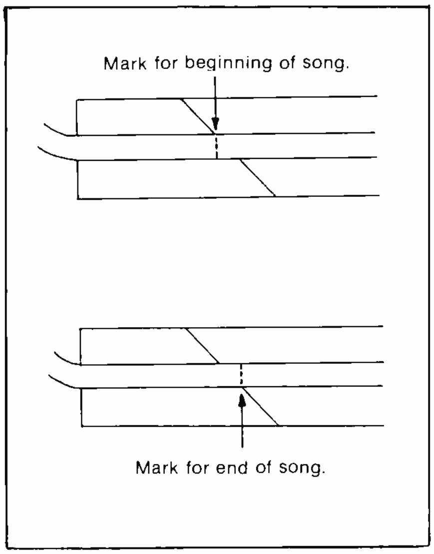


Figure 3: Aligning edit marks with cutting slot.

Editing

Another subject related to tape handling is editing. *Editing* is the cutting and rejoining of magnetic tape to delete unwanted material, to insert leader tape or to rearrange material into the desired sequence. This process requires the following materials: demagnetized single-edge razor blades, a felt-tip pen or yellow

crayon, splicing tape, leader tape and an editing block.

Leader tape is plastic or paper tape without an oxide coating, used for a spacer between takes (i.e., silence between recorded songs). Paper leader is preferred over plastic because plastic can cause static-electricity pops. An *editing block* holds the tape during the splicing operation. It's easier to use than a tape splicer with hold-down tabs and gives more precise cuts.

Before editing, wash your hands to avoid getting oily spots on the tape. Cut several 1" pieces of splicing tape and stick them on the edge of the tape deck or table. Also cut several sections of leader at the 45° slot in the editing block. A typical leader length between songs is four seconds, which is 60" long for 15 ips or 30" long for 7½ ips. While editing, try to hold the magnetic tape lightly by the edges.

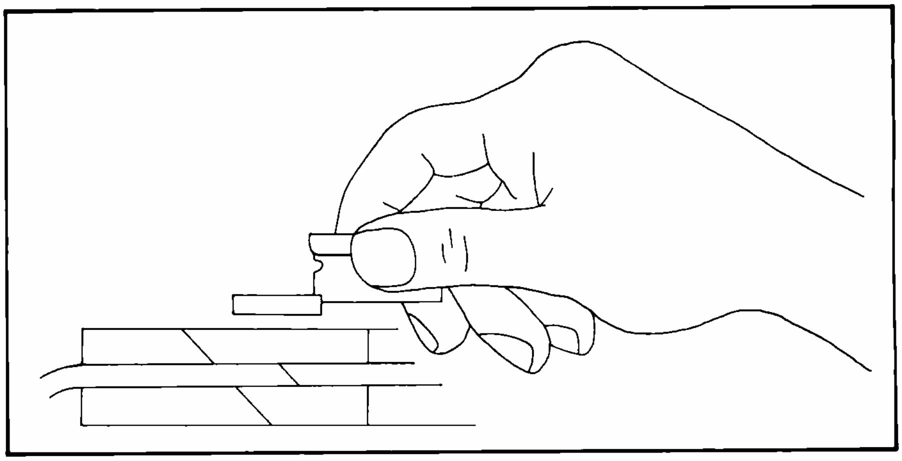


Figure 4: Applying splicing tape.

Leadering

Suppose you've recorded a reel full of takes, and you want to remove the out-takes, count-offs and noises between the good takes. You also want to insert leader between each song. This process, called *leadering*, can be done as follows:

First, wind several turns of leader onto an empty take-up reel and cut the leader at the 45° slot. Remove this take-up reel, put on an empty one, and play the tape to be edited.

Locate the beginning of the first song's best take. Stop the tape there. Put the machine in "cue" or "edit" mode so the tape presses against the heads. While monitoring the tape recorder output, rock the tape back and forth over the heads by rotating both reels by hand, first rapidly, then more and more slowly. You'll hear the music slowed way down and low in pitch. Find the exact point on tape where the song starts—that is, where it passes over the playback-head gap. Align the beginning sound with the gap. Using the pen or crayon, mark the tape about ½" to the right of the gap (that is, at a point on tape just before the song starts).

Next, loosen or "dump" the tape by simultaneously rotating the supply reel counterclockwise and the take-up reel clockwise. Remove the tape from the tape path and press it into the splicing block oxide side down. Align the mark with the 45°-angled slot (as in *Figure 3* top). Slice through the tape with a razor blade, drawing the blade toward you. Don't use the 90° slot since such an abrupt cut can cause a "pop" noise at the splice.

Remove the unwanted tape to the right of the cut and put the take-up reel aside. Slide the cut end of the tape to the right of the editing-block slot (as in *Figure 4*). Put on the take-up reel containing the turns of leader tape, and insert the end of the leader into the right half of the block. Shove together the ends of the leader tape and magnetic tape so that they butt or touch together with no overlap.

Now, take a piece of splicing tape and stick a corner of it onto a hand-held razor blade. Align the splicing-tape piece parallel to the recording tape (*Figure 4*). Apply the piece over the cut onto the non-oxide side, and stick it down by rubbing with your fingernails.

Slide the splice out of the block. Gently pop the tape out of the block by pulling up on the ends of the tape extending from both sides of the block. Twist the tape toward you while pulling. Check that there is no gap or overlap at the splice.

Now wind the first song onto the take-up reel and locate the ending of the first song. As it ends, turn up the monitors and listen for the point where the reverberant "tail" of the music fades into tape hiss. Stop the tape there and mark it at the playback-head gap (at the center line of the head).

After pressing the tape into the block, cut the tape at the mark as in *Figure 3* bottom. Remove the tape to the left of the cut. Splice the end of the first song to a four-second length of leader and again check the splice. Wind the first song and the leader onto the take-up reel and remove it. Then put on the take-up reel containing unwanted material you previously laid aside. Splice it to the rest of the master tape.

Next, locate the beginning of the next good take you want in the program. Mark it and cut the tape. Put the reel containing the first song on the take-up spindle. Splice the tail end of the leader onto the beginning of the second song, and then wind the second song onto the take-up reel. You now have two songs joined by leader tape.

Repeat this process until all the good takes are joined by leader. Then you will have a reel of tape with several songs separated by white leader, which makes it easy to find the desired selection.

Joining Different Takes

What if you want to join the verse of Take 1 to the chorus of Take 2? You'll have to cut into both takes at the exact same point in the song, then join them. It takes practice to make an inaudible splice in this manner, but it's done every day in professional studios.

The two takes must match in tempo, balance and level for the edit to be undetectable. To mask any clicks occurring at the splice, cut the tape just before a beat—say, at the beginning of a drum attack. An alternative is to cut into a silent pause. If you cut into a continuous sound such as a steady chord, a cymbal ring or reverberation, the splice will be noticeable.

Let's run through the procedure. First play Take 1 and locate the point where you want Take 1 to stop and Take 2 to start—say, at the beginning of the chorus. Stop the tape there. Then put the recorder in "cue" or "edit" mode, rock the tape and try to identify a beat or attack transient. At the point on tape where this beat just starts to cross the playback-head gap, mark the tape. Cut the tape at the mark and remove the take-up reel containing the verse of Take 1.

Next, put on an empty take-up reel, thread the master tape and fast-wind to Take 2. Find the same spot in Take 2 that you marked in Take 1. Mark and cut it. Using splicing tape, join Take 2 (in the supply reel) to Take 1 (in the take-up reel you just laid aside). Again, check that there is no gap and no overlap at the splice.

Play the spliced area to see if the edit is detectable. If not, congratulations! It should sound like a single take. If Take 2 comes in a little late, carefully remove the splice and cut out just a little tape surrounding the cut. Re-splice and listen again.

More Editing Tips

Suppose you've recorded most of a good take, but then the musicians made a mistake and stop playing. Rather than repeating the entire song, the musicians can start playing a little before the point where they stopped, and then finish the song. You can then splice together the two segments into a complete and perfect take. Editing is also useful for interjecting comments or sound effects in the middle of a song, or for making tape loops. You can even record a difficult mixdown in segments, then edit the segments together.

Assembling the Master Tape

Let's say you've mixed several songs onto a ¼" master tape. Now you're ready to assemble the tape into a finished format for tape duplication or record cutting. It will contain the songs in the desired order, plus leader for banding the record. In addition, it will have calibration tones for the record-cutting engineer to align his playback machine for flat response from your tape.

First, decide in what sequence you want the songs on the record. For the first song on Side 1, use a strong, accessible, up-tempo tune. Follow it with something quieter. Alternate keys or tempos from song to song. The last tune on each side should be as good as or better than the first one, to leave a good final impression.

Try to keep the total time per side under 18 minutes for maximum level, maximum bass and lowest distortion on the record. A maximum of 24 minutes per side is recommended.

The length of leader between songs depends on how long a pause you want between them. Use longer leader if you want the listener to get out of the mood of the piece he just heard before going on to the next. Use shorter leader either to change the mood abruptly or to make similar songs flow together. Or, using two 2-track machines, you can *crossfade* between two final mixes, copy the result on a third machine and edit the copy into the rest of the program.

Calibration tones should be recorded on the ¼" master tape just before recording the mixes. First record 30 seconds of Dolby tone (if the tape is Dolby encoded) at 0 VU on both channels simultaneously. Then record the following 20 second tones on both channels, without noise reduction, at 0 VU for 15 ips or −10 VU for 7½ ips: 15 kHz (for azimuth adjustment), 700 Hz, 10 kHz and 100 Hz. Finally, record 1 kHz at 0 VU.

If you don't have access to a multi-frequency generator, just put on a 1 kHz tone or a sine wave synthesizer note (2 octaves above middle C) at 0 VU, both channels.

To assemble the master tape, wind onto the take-up reel the following material in this order:

1. Tones
2. At least ten seconds of leader.
3. First song
4. Leader
5. Second song
6. Leader, etc.
7. Last song on Side 1 of album.
8. At least ten seconds of leader.

Then rewind the tape. Play it and time it from the beginning of the first song to the end of the last song (including the leaders between songs). This is the "running time."

Using a piece of masking tape, fasten the leader tail to the reel and print "TAIL OUT" on the masking tape. Type or print a neat label for the tape reel including title, artist, "Side 1" and running time. Label the tape box similarly.

Using another take-up reel, assemble Side 2 of the album (but without tones). Time Side 2; label the reel and box.

On the tape log inside the box for Side 1, print or type the following: (1) label information; (2) tape speed, track format and type of noise reduction used; (3) maximum VU level the program was recorded at; (4) tone frequencies and levels as they appear on tape; and (5) song titles and times.

There are your finished master tapes. If you plan to send them to another studio, be sure to make a safety copy of the master before sending the master tape, in case it is lost or damaged. When copying the master, set the 0 VU tone from the playback machine to read 0 VU on the recording machine. There's no need to reset the program levels since you already set them while recording the master tape.

Tape Tricks

As long as we're on the subject of tape and tape recorders, be sure to read Craig Anderton's excellent three-part series, "Multi-Track Magic," which appeared in the May, July and August, 1981, issues of *Modern Recording & Music*. Craig tells how to create special effects and how to record nine tracks or more using a 4-track recorder. In the July, 1982, issue, Marc William Fallon tells in a letter how to make the best use of 8-track recorders. We'll describe two more special effects here:

- *Reverse echo or preverb:* With this effect, you can make echo or reverberation *precede* the attack of each note. On your multi-track machine, reverse the tape reels so that the tape plays backwards. That is, put the take-up reel where the supply reel was, and vice versa. Play the tape in sync mode. You'll hear the tracks playing backwards. Add reverb or echo to the desired track and record the result on an open track. Again reverse the reels and play the newly recorded track.
- *Backwards tracks:* You can make an instrument or voice play backwards in the mix. This effect is useful for overdubbing lead guitar solos or for adding secret messages (e.g., "Paul is dead"). To do this, reverse the reels as described above, then record the new part on an open track. The musician should monitor the other tracks playing backwards in the sync mode. Then reverse the reels, play the tape and mix the tracks as desired.

Conclusion

You can ensure that your tape will be a faithful copy of your mix by (1) cleaning and demagnetizing the tape path; (2) handling and storing tape properly; (3) calibrating the tape machine; and (4) using noise-reduction units. In addition, editing and leadering your master tapes into a professional format will aid other studios using your tapes.

Profile:

PETER TOSH

Jeff Tamarkin

THE HEAVIEST THUNDER storm of the summer of '82 is pouring down on New York. Peter Tosh is nestled comfortably in an armchair in the sparsely furnished apartment of an associate. Still wearing his shades, he stares blankly ahead. Although the rain falls in buckets, the window overlooking Central Park is open wide. The thunder and lightning inspires him more than sitting through yet another interview does.

He walks over to the open window, thrusts his hand-carved walking stick towards the heavens, and, disregarding the rain, shouts to the sky: "Yes, mon, even the heavens know when a saint comes to town!"

While Tosh may not be recognized as a saint in the U.S., he is certainly a king—a king of reggae music. And although he'll adamantly tell you that he's not interested in stepping into the late Bob Marley's shoes (Tosh even skipped Marley's state funeral in Jamaica), he's arguably reggae's top name in America, and has been since his highly publicized opening act stint on the Rolling Stones tour of 1978.

Tosh is an unlikely candidate for superstardom, and even if his records do begin to sell as well as Marley's did, one gets the feeling Tosh will never become a media darling. The reason for this is that Tosh speaks his mind, unafraid of offending anyone, especially those in control of the media and what he calls the "shitstem." Tosh is an outspoken advocate of equal rights and justice—both in his music and in interviews—and he's unafraid to point a finger. At times, his attitude has brought him trouble, like the time in 1975 when he was badly beaten in the streets of Jamaica and almost died. The incident only made Tosh more committed to his cause.

Another reason that Tosh will probably never be a media superstar is that he refuses to place himself above other reggae performers. He's the music's greatest supporter, and considers the whole of reggae more important than its individual artists, himself included. To Tosh, the message of reggae and the music of reggae is what's important, not its individual practitioners... living or dead.

Peter Tosh began life as Peter MacIntosh in Jamaica. He first reached prominence in the late '60s with the Wailers, a group which began in 1964 and included, among its other members, Bob Marley and Bunny Livingston. As the group gained in popularity and began to gain acclaim outside of Jamaica, where they were already #1, it became apparent that Marley was becoming the acknowledged group leader. Tosh split in 1974 for a solo career.

Tosh, who has been a musician since age three, and had written or co-written some of the Wailers' best songs, had no trouble landing a record contract in the U.S. His 1976 solo debut, *Legalize It* (dedicated, of course, to the Rasta's sacred herb, ganja), was highly received. The follow-up, *Equal Rights*, contained some of Tosh's most poignant songs, including "Get Up, Stand Up" (also popularized by Marley), "Stepping Razor," and the title track.

The big turning point, however, came in 1978 when Tosh signed with the Rolling Stones label and released *Bush Doctor*. Tosh's duet with Mick Jagger on the old Temptations' song "Don't Look Back" was the first reggae song to receive a large amount of airplay on rock stations. That hit, coupled with Tosh's subsequent tour with the Stones, established him as a hero. Reggae and Peter Tosh had arrived.

The following year Tosh recorded *Mystic Man*, followed by 1981's *Wanted Dread And Alive*. His already-widespread popularity was bolstered last summer when Tosh and reggae's other reigning leader, Jimmy Cliff, toured the U.S. as one dynamite double bill. By the time the tour hit New York in September, the show was the hottest ticket in town. And Tosh lived up to the billing, following Cliff with a powerful set that had the entire Felt Forum auditorium on its feet.

Prior to the tour, Peter Tosh stopped off in Manhattan to talk to MR&M's Jeff Tamarkin about his recently-completed album, *Mama Africa* (still without a label at the time of the interview), and to expound on his philosophies regarding reggae and life.

Modern Recording & Music: We haven't heard your new album yet, so can you tell us how it differs from your last one, *Wanted, Dread And Alive?*

Peter Tosh: It's very different, very good. The best is always yet to come. When you hear it, you'll be surprised. It's called *Mama Africa.*

MR&M: Who plays on it? Do you have Sly and Robbie again?

PT: Yes, the same band.

MR&M: You're not on a label yet, although the album is completed. What happened to the deal with EMI and Rolling Stones Records?

PT: We're negotiating now. EMI is still in the picture, but Rolling Stones is out. We're also talking to CBS.

MR&M: Is it important to you to keep selling more records and to reach a larger audience, or do you feel that the people who have to hear your music will hear it?

PT: Yes, mon, it works both ways. I want to reach everyone, but the people who must hear it will hear it.

MR&M: Do you feel that your music, and reggae in general, can only appeal to a segment of the people, or that everyone can enjoy it?

PT: It's the universal music, for every nation—Europe, Asia, everywhere.

MR&M: Why did you and Jimmy Cliff decide to tour together last summer? Your ideologies and music are very different.

PT: I did not decide nothin'. I'm just workin'. Yeah, mon. Managers and promoters came up wit' the idea, 'cause it was better to reach a large crowd dat way.

MR&M: Do you feel that your music and Jimmy's have a lot in common—they're both reggae, true, but he's not a Rasta and his sound is more like a soul-pop reggae than yours.

PT: They have a lot in common; they're all called reggae music.

MR&M: Do you still have a lot of people to conquer in America?

PT: With each new season, reggae music affect more people. It even affect the thunder outside.

MR&M: All Rastas, of course, long to go to Africa. Have you ever toured there?

PT: Never toured, no, but I been there recently. Africa is the greatest place on the Earth, but y'know there's a lot of exploitation and punishments there. A lot of bad shitstems down there.

MR&M: Is it worse there than in Jamaica or the United States?

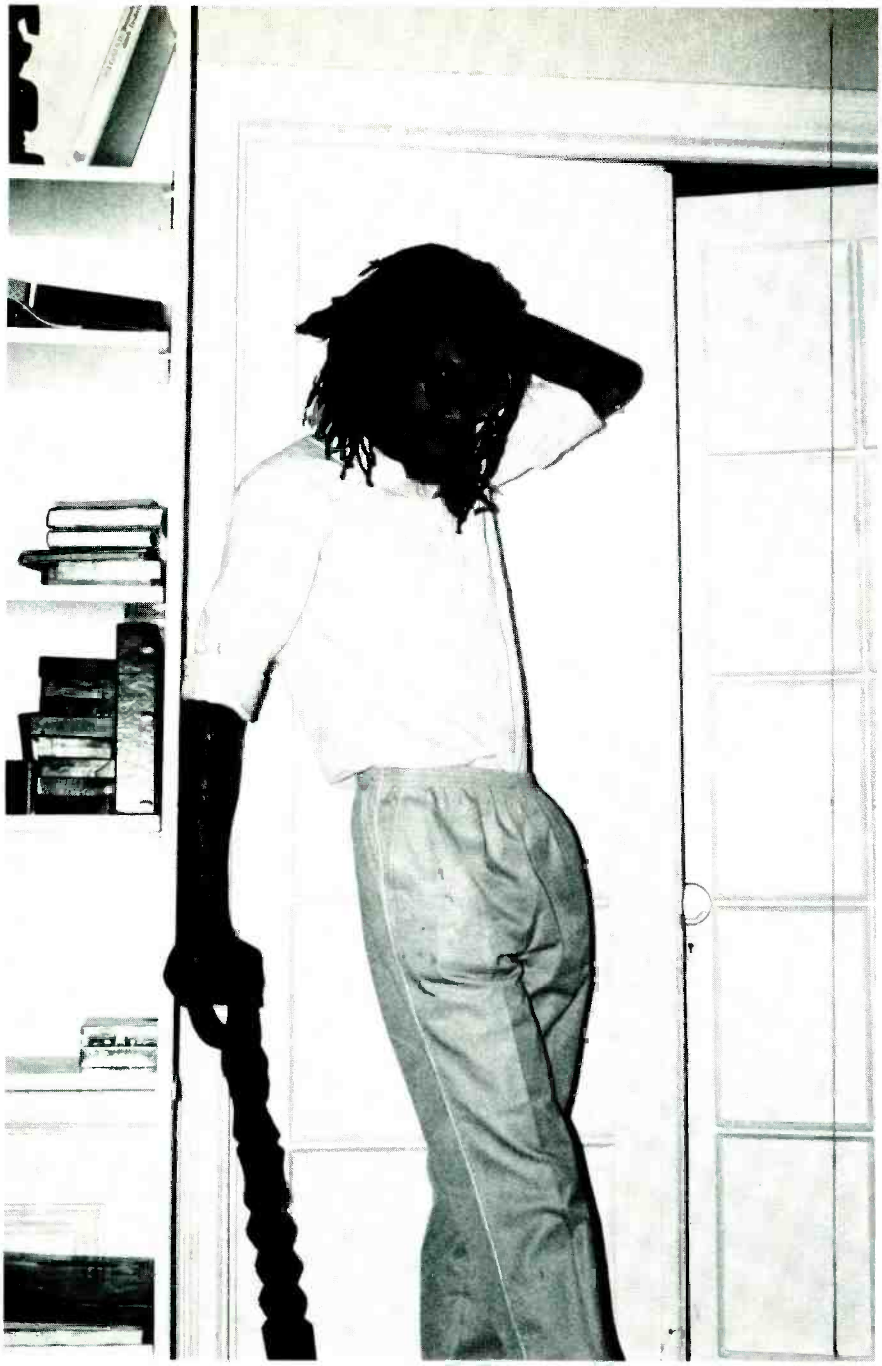
PT: No, it's the same everywhere; the same demons and the same devils.

MR&M: Do you think things are getting better or worse?

PT: Well, it must get worse before it can get better.

MR&M: Reggae still doesn't get played on the radio in the U.S. or sell many records. Whose fault is it that it hasn't caught on?

PT: It's not that it hasn't caught on; it's not the fault of the people. Don't blame it on the people, blame it on the medias. It's the bureaucrats behind it; they're a part of the shitstem. Reggae music is branded political; it's the message. You can't listen to the charts; the charts are a



PT: Yes, we get good play on the college stations, because they're the youths. The youths know the truth.

MR&M: Is there any other kind of music that you think spreads a positive message?

PT: Reggae music is the only music right now speaking the truth. There are other musics where one or two times a person may sing a message, but it's not a continual thing. They just want to try a thing to see if it will work.

MR&M: Do you like what some of the British reggae acts like Steel Pulse and Linton Kwesi Johnson do?

PT: Yes, mon, me has to like what they do. Because reggae is a flower, and the flower is very beautiful, and it takes more than one flower to make the garden.

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MR&M: Reggae singers always sing against oppression, but as the governments of the world get more conservative, how does that affect what you sing about? Are you afraid of what they might do to you?

PT: Well, we all know that the wicked get wickeder, but the righteous is going to get more righteous.

MR&M: All reggae artists are trying to spread the same good message, but is there competition among you to be the most popular? For instance, do you feel that you competed with Jimmy Cliff when you toured together?

PT: No, mon, I don't compete against no man. I have the positive message and so does Jimmy Cliff. My music is the music that breaks down barriers. And when it breaks down these large barriers, then the other artists can walk through the open walls.

MR&M: How did Bob Marley's death affect reggae? Especially in Jamaica.

PT: No, mon, didn't change nothin'. Bob Marley was not the culmination of reggae music. Bob Marley was just a branch on the tree of reggae music.

MR&M: How do your fellow countrymen see you in Jamaica? After all, they are mostly very poor and you've made some money with your records. Are you considered an upperclassman?

PT: My Father owns the world, so how should I be seen, see? That means that either all the world is mine or half the world is mine. Materialistic people can judge me by what I earn,

fantasy. Those people aren't living a life of reality, and reggae is the music of reality.

MR&M: Why are the people in charge so afraid of reggae's message?

PT: Because they are taught lies. (At this point, a huge thunder blast rocks the windows of the room we're in). I bet this is the first interview you ever did with thunder and lightning. That shows you how powerful it is! Yeah, mon, yeah, mon.

MR&M: Do you think it's a good thing that white rock groups like the Police and the Clash and Blondie have used reggae?

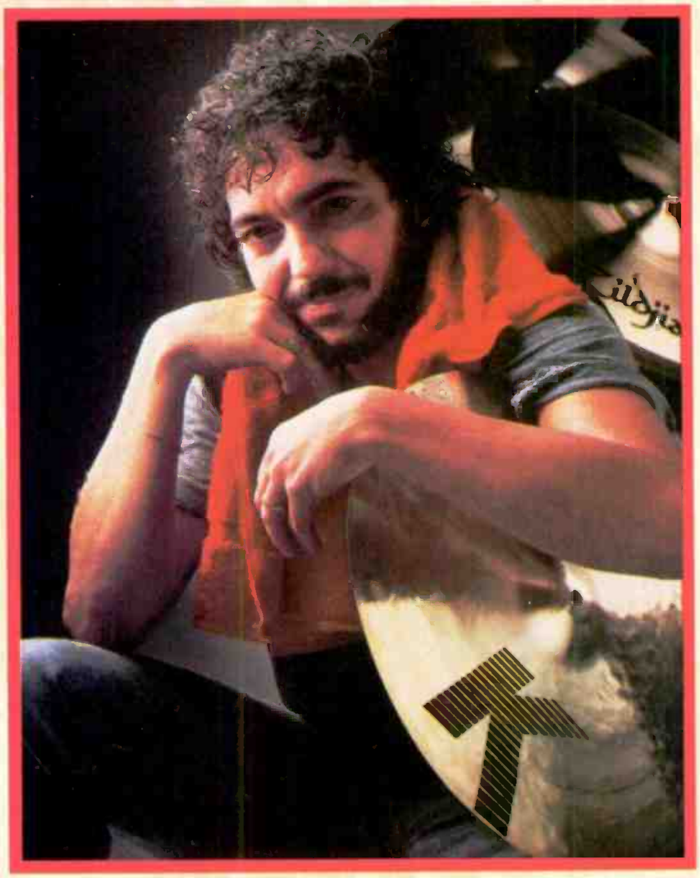
PT: Yeah mon, because it makes more people aware that there's a music in the world called reggae.

MR&M: But they're making all the money by playing a diluted version, and you're playing the roots reggae and not selling as many records. Don't you feel they're exploiting you?

PT: No, they're not exploiting me because, if they sing my song, I get the royalties.

MR&M: Reggae gets a lot of airplay on college stations in America. Why do you think they're not afraid of it?

STEVE GADD. HOT ON ZILDJIAN.



Steve Gadd, one of the world's most innovative musicians, has paved the way toward new playing techniques for today's drummers.

The man is hot! And he should be. No less than Chick Corea put it this way: "Every drummer wants to play like Steve Gadd because he plays great. He plays everything well. He could very well go on to become one of the greatest drummers the world has ever seen."

As you can imagine, between his touring and recording, Steve's not the easiest guy in the world to pin down. But he did stop for a breather the other day and we got a chance to talk with him.

On Practice. "I've been playing since I was a kid. As long as I keep my muscles loose, I don't have to practice a lot every day. When I do practice, I just sort of let things happen naturally and then later on try to work it into my playing. Like on '50 Ways to Leave Your Lover'... I used my left hand on the high hat for the whole section–it was a little thing I'd been practicing and it just worked out."

On Control. "Sometimes I use light, medium and heavy sticks to do the same drills because the sticks affect my muscles in different ways. You have to use your hand and arm muscles differently to control your playing. It's a subtle thing but it helps me tremendously."

On Effects. "After I graduated from Eastman, I played in a rock 'n roll band. It was keyboard, bass, drums and a lot of homemade stuff. I bought 6 big artillery shells, sawed them into different lengths and hung them on a rack that I built. I'd use them for the free sections in the music."

On K's. "Art Blakey gave me my first set of K. Zildjian's a long time ago. I love the feel of them. There's something about the way the stick reacts to the surface...it almost becomes part of the cymbal. They're not cold or edgy. They have a very warm and deep feeling. They've got real character. I use a 20" Ride and an 18" Crash Ride with 14" Hi Hats for recording and live sessions."

On A's. "I love to use A. Zildjian's when I play rock 'n roll. When I want to play louder, I add a 16" Thin Crash and an 18" Crash Ride for a full crash sound. The bells on the A's really project the sound in a clear natural tone."

On Zildjian. "Zildjian to me is the foundation. I play Zildjians because that's what's in my heart. I love the sound, the feel, the history...I love the quality and the status of a Zildjian."

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but there shall be no limit put on what I earn, because when I earn even diamonds and pearls, no one should want that.

• • •

MR&M: You've been beaten up and almost killed a few times in Jamaica. Are you still brutalized today?

PT: I am always wanted, but as my father says, after all the ugliness, there shall be honor, power and majesty. That is my protector. When the thunder roll and the rain fall, then there shall be justice.

MR&M: Do you think we'll see justice in our lifetime?

PT: Yes, mon! See the justice there (points to rain outside)! If America hear the sound of this thunder and lightnin' and earthquakes, for half an hour straight—and I don't mean this little one, but ones 10 times as much—they see even God give justice. Yeah, mon.

MR&M: Do you write your music at home or on the road?

PT: Anywhere. Anywhere I get the inspiration.

MR&M: When does that come to you?

PT: Anytime, anywhere. I don't have to say I'm going to try to make a song. A song is always there; I just have to open my mouth and a song comes out.

MR&M: When you record, do you rehearse and work out songs or is it all spontaneous?

PT: When I go to studio, all I have to do is call my musicians, because the picture is already perfect inside of me. (More thunder) Jah Rastafari-hi-hi!

MR&M: You produce your own records. Why do you like to do it yourself?

PT: Yeah, mon. Because I realize that there aren't much producers that can do it efficiently. They can't make a perfect music. There are not producers who can materialize the pictures from another man's mind.

MR&M: Doesn't your mind ever become clouded when you have to keep it on the technical things and the music at the same time?

PT: My mind cannot be clouded. When a mind is polluted, it is easily clouded. I always have my mind clear. I can focus on one thing or two things.

MR&M: Do you get involved technically with the engineering?

PT: Yeah, mon, all of it.

MR&M: You've been recording since the early '60s when you were with the Wailers. You've seen the change in technology in the recording studios. How has that affected reggae?

PT: It has affected it a great deal. When you hear my new album you'll realize this is not an album that could have been made 10 years ago. In the early days I could just go in the studio and sing and everything had to be balanced perfect, because it was just one track. Now we can do more with the technology.

MR&M: Do your first solo albums, like *Equal Rights* and *Legalize It*, sound outdated to you?

PT: No, mon, my music never grow old. It is timeless. It is always new. Every time I hear new sounds.

MR&M: Do you prefer recording or playing live?

PT: Playing to the audiences, because then they can dance.

MR&M: What do you do if you find yourself playing to an audience that doesn't like reggae?

PT: I can use psychological power over them, to change their minds—to baptize them.

MR&M: There are still some areas of the U.S. where reggae is still not liked by the majority. Maybe it's a racist thing.

PT: Yeah, mon, I know that. But my music has hypnotic powers. I'm always prepared for those kind of people.

• • •

MR&M: What was it like to tour with the Rolling Stones in 1978 and play to 50,000 rock fans at a time?

PT: Greatest experience. I can program my music for 500 people or 10,000 people or 125,000 people. The problem is that many of those people are not easily hypnotized because they are only programmed to rock 'n roll.

MR&M: A lot of them are too closed-minded to turn on to something different.

PT: Yeah, mon, I can tell you that.

MR&M: Is it any different with the black American audience? A lot of them are close-minded, too, and they just listen to the R&B they hear on the radio and in discos.

PT: They are not slow; it's the medias, the people that control the music they hear. (More thunder crashing). Jah Rastafari! Not a stone shall be left standing upon the island. And if Reagan thinks that his missiles and all his ballistic bullshit can destroy the world, he makes a sad mistake. And if Russia thinks the same thing, they make a sad mistake, too. Every time I hear thunder and lightning, I know that is true. (More thunder). Yes, Jah! It is only mind vibrations that create these things. This music that I have on my new album is as powerful as the thunder out there. The way this thunder rolls, that is the way this music comes. (More thunder). Yes, Jah lightnin'!

MR&M: Peter, you've often said that your music deals with truth and rights and not politics, but don't they have a lot to do with each other?

PT: Well, according to the shitstem and grammatical rules, men put form to words to make them soothe the earth. But politricks is the betrayal of the people, and a violation of people's rights. I don't get involved with politricks. My music deals with equal rights and justice for the people: people who are being victimized and discriminated against. Because every man, whether he is white, black, blue, pink or green, has the right to live. And I see no reason why any man must think he has power to violate another man's rights. So because these things have been going on for all these years and all these periods of time, the rulers and leaders of the world act as if they don't care. Because they have been doing so much dangerous things and getting away with it, they forget that thunder rolls and lightning flashes. One earthquake in the next 10 seconds and America was; they don't like that.

MR&M: But haven't people also used religion to gain power throughout history?

PT: Religion is another betrayal of the people's rights. That's selling the name of God in many denominations, and calling it the truth; that's religion. I-and-I concept is a way of life—the way Christ used to live, until Satan came.

MR&M: When did it all start going wrong?

PT: When Satan came. And that's a whole long time, 6,000 years ago. Satan's work has been for a period of time, but the reign of Satan is going down. Babylon is falling. Soon the people shall know the truth, and it shall set them free.

MR&M: And where do you fit in? What's in the future?

PT: Progress; prosperity and life; continual praises of the almighty

MR&M: Thank you.

PT: Yes, this was a great one.

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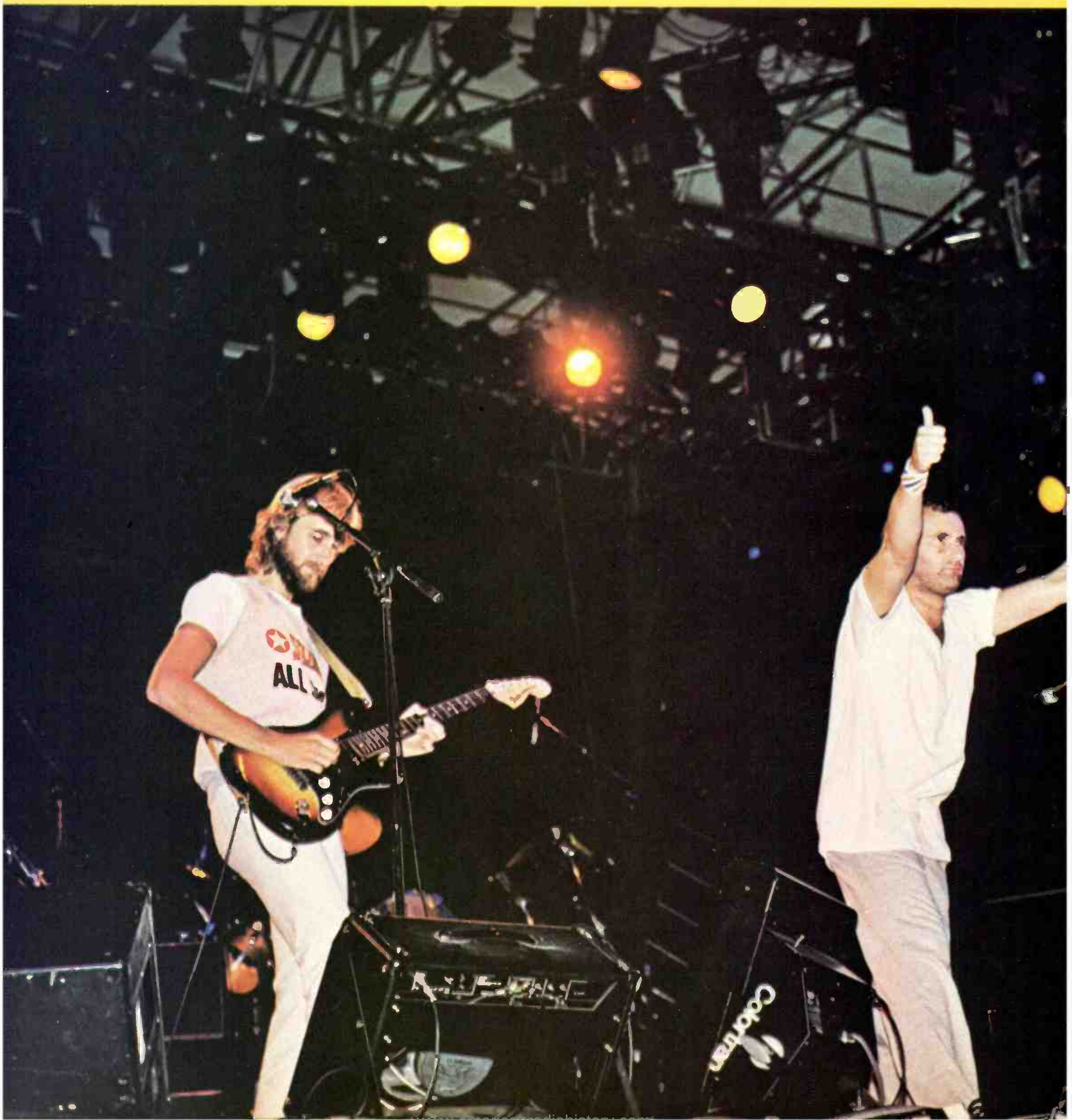
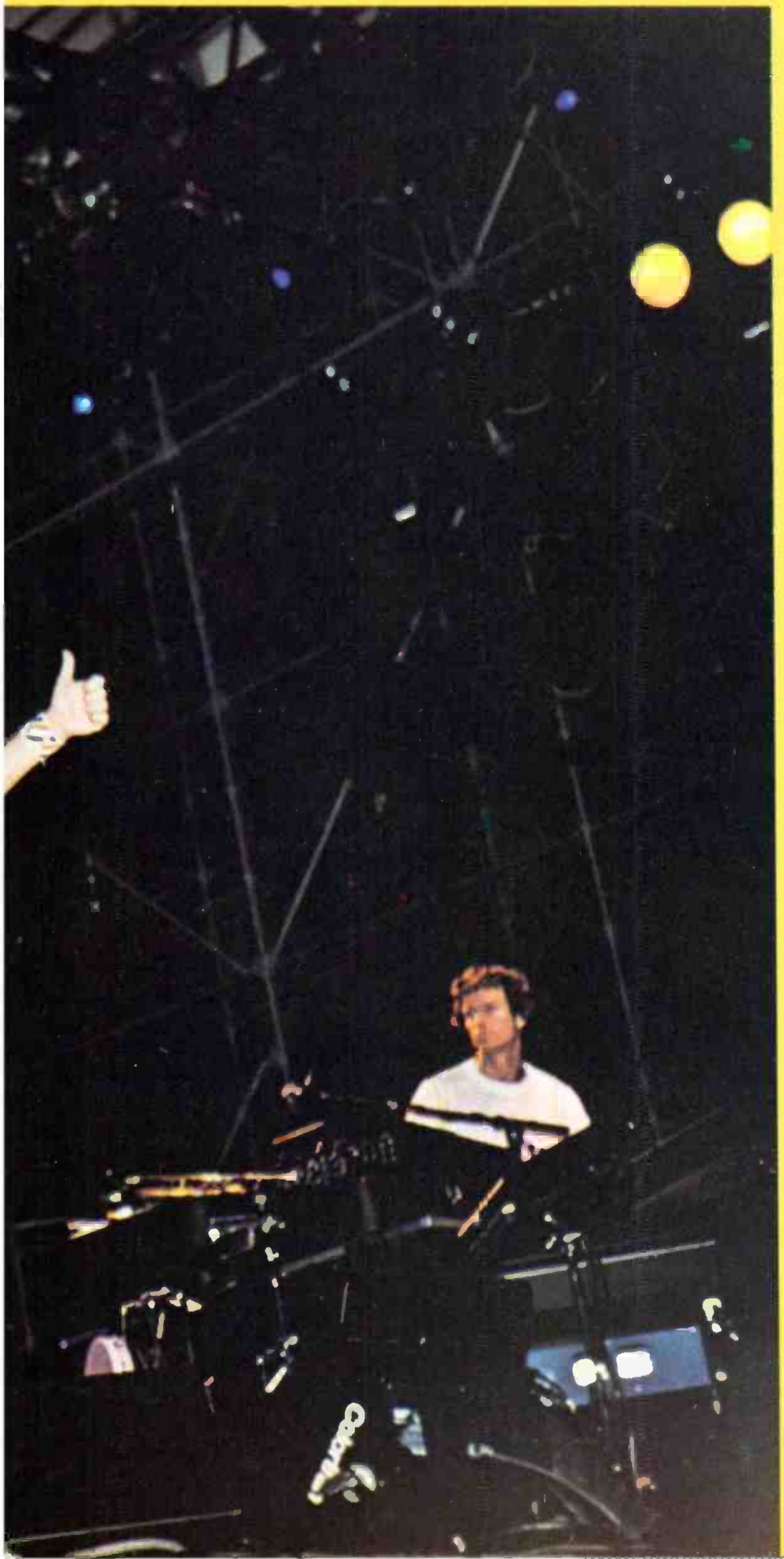


'LIVE'!

by rob patterson



Genesis may best be known for the incredibly rich yet direct studio sound it achieves, but its current "live" show rivals Genesis's albums for clarity and depth. Known as one of the guiding lights of the progressive rock movement, Genesis has grown ever-more pop while building a "live" sound that is precise and technologically advanced.

At Forest Hills Tennis Stadium [New York] in the late summer of 1982, Genesis played a two-night stand that was stunning for its sound quality, especially when one considers all that happens on-stage—a singer who also drums; a second drummer; two guitarists who play a variety of four- and six-string instruments (including double-neck guitars); as well as a keyboardist who plays a variety of 88s. Add to that the horn players from Earth, Wind and Fire, and you have the potential for either a cacophony of conflicting signals or one hell of a layered and complex sound. Through the creative efforts of the three members of Genesis—Phil Collins, Tony Banks and Mike Rutherford—Showco Sound, from Dallas, Texas and their mixer Craig Schertz; HHB Hire and Sales in London; and Noel Mawer, the band has conquered many of the pitfalls of "live" sound with a specially-designed monitor mix and an intelligent approach to amplification.

Modern Recording & Music: How did you come to have such a well-planned and easily workable "live" sound set-up?

Tony Banks: It sort of evolved over the years, really. When you start out and you first play "live," you don't have any [stage] monitors. Then you hook up an extension speaker from the other guy right across the stage, and you have no control on that. Then, even when you have monitors, it's just too much for one guy to do—mix everyone's sound. We used to try and keep it very simple. For example, I just had Mike and Steve [Hackett] come through a speaker with a couple of knobs, and just forget about everything else and just go with that. Now we have a monitor system with 10 channels, which means we each can totally mix our own monitor sound.

But a lot of it is still very simple. If you look at Mike's box, usually he has just me up, because that's all he can't hear from where he stands in the middle. On my side, I need a lot of Darryl [Stuermer, the band's second guitarist "live"], a lot of Chester [Thompson, their concert drummer], and some of Mike.

What it means is that if some night something does go wrong, and all of a sudden the bass comes booming through, you can just turn it down, rather than trying to shout at the monitor mixer: "I can't stand it anymore!"

Mike Rutherford: To me, it's incredibly stupid the way monitoring is usually done. Just one monitor mixer for all five guys in the band... who are all trying to catch his attention. How can it be any good? What sparked us off in getting our own monitors was playing some festivals in Europe with Frank Zappa. The way he had it, the monitor system was part of the band's gear, not the P.A. That's what inspired us to do the same.

Phil Collins: The design for our system came when we did *Duke* at Abba's studio, Polar, in Stockholm. Usually in the studio, it's terrible when the engineer is trying, say, to get the bass right, and you're trying to get his attention: "Excuse me, can I have a bit more of me and a bit less keyboards?" "Yes, in a minute." Then it never gets done and you have to live with this awful noise in your head. So we came to Polar and found they had individual headphone mixes, and thought it was great.

TB: It was the first studio we ever went to where they had 4-channel stereo controls on a box for the monitor headphones. With three of us, we had each of us on one channel and the other channel for echo. And the technology is such that you can also do it onstage.

PC: You basically have total control. Chester has his mixes in stereo. Noel [Mawer] basically does my mix when I sing. What I hear out-front is Chester's bass and snare, because it helps to have that coming right at me. Tony, Mike and Darryl I hear enough of right on the stage. I used to have my voice on an ADT [Automatic Double Tracking]—I was losing my voice, so I figured I'd spread it out with a slow repeat. Back at the drums, I have everyone basically up, with a little more of Chester.

MR&M: What about the wonderful sound you get out front?

TB: Craig [Schertz] has basically worked with us a lot, so credit does go to him for that. I feed him a stereo mix from one of every keyboard, plus a single channel, so if my mix isn't good, he can push me up. Because we've got such a good monitor system, to some extent the sound mixes itself.

MR: I think that basically the band's got a good internal balance. In fact, Craig reckons a good evening is when he doesn't do anything. I used to send everything to him DI'd [direct input], which was crazy. Now I've got all the guitars—the bass is still DI'd—one little amp with a 12-inch speaker with a mic in front, and that's it.

TB: He [Craig] really doesn't have to do much EQ either, because we do almost all of it.

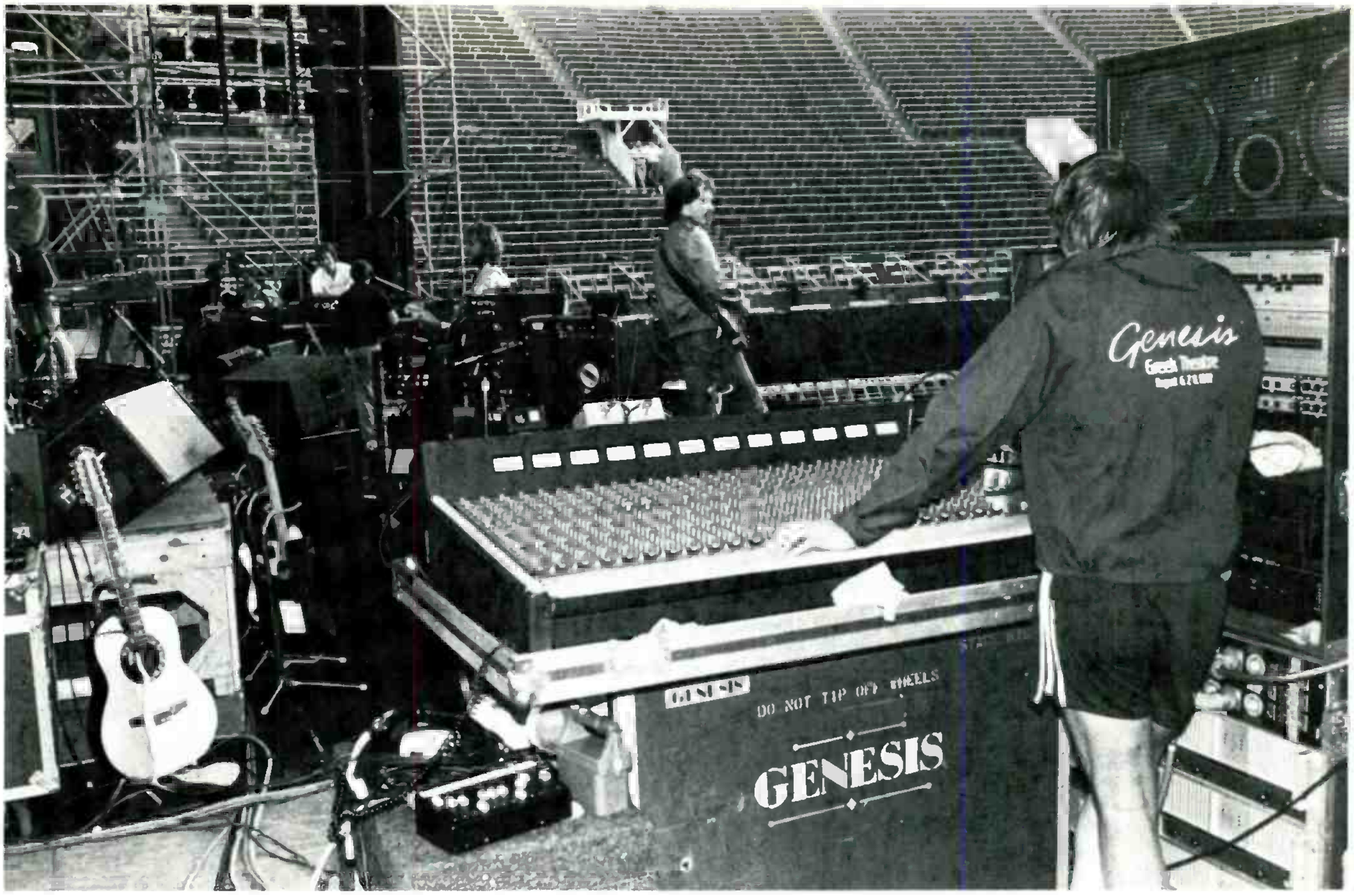
The key component in all this is the band's specially-designed monitor system, which also does double duty as a studio monitor set-up at Genesis's "The Farm" in Surrey. The system was designed by Noel Mawer, whose company first started working with Collins when he was in Brand X, and also later worked with Peter Gabriel. "We just kept cutting in with them," says Mawer, adding, as everyone we talked to stressed, "I really enjoy working for them. They treat you good and look after you."

Noel Mawer: In 1980, the band approached myself and the company I jointly run back in London—HHB Hire and Sales, Ltd.—because they wanted to purchase a monitor system that was part of the band gear rather than part of the P.A. system. One of the requirements was that they could control their own monitor mix, which is fairly logical. So this is what I put together: a 24 by 12 Amek board, slightly modified to give pre-fade outputs as well as post-fade. The pre-fade outputs go out on a multi-core distribution system to these little boxes [mixers]—this one here's for the horn players tonight. Basically you group down. Instead of having each group as a mix point on a conventional monitor system, you treat them as sub-groups on an ordinary board. They come up on these little mixers and return down the same multi-core there into the amps or effects rack. And that's it. All I really do here is look after Phil when he's singing, because I obviously don't want to give him control over it. What he's hearing, that's dangerous stuff. Chester has a stereo box for his mix there, because he needs the separation. Tony has his monitor mixer built into his keyboard mixer, and that's basically it.

MR&M: What about the monitor speakers?

NM: The speakers in these monitors are all ATC, with JBL horns. The drum fills, the top parts are essen-

"Genesis (is) a joy to work for—not only as performers, but as people, too."



tially the same as the floor monitors, with slightly different mid-speakers. The bass units below are basically the same as Darryl's using for bass guitar.

MR&M: And the power amps?

NM: Crown PSA2 amplifiers, and a bunch of jizz in here to make it all happen. The way I use the EQ on it is not actually principle [*Laughs*], because it's too off-the-wall!

MR&M: How is that?

NM: Say for Phil's front pair of wedges, I put pink noise through them and use the vocal mic as a measuring instrument for the analyzer, and I've got it basically flat. I don't really touch that. Phil has three vocal positions. They are sent on group 12 into the echo machine, and returned via a notch filter to the graphic, because it's mainly the front I'm interested in...The notch filter has some very, very, very deep notches in it, approaching 40 dB in some places, coupled with the graphic. In rehearsal before the tour, we just use the graphic to boost the level up on the fader here, which controls the monitors, to notch it out. It looks ragged, and in real terms it probably doesn't sound very nice, but it's what Phil wants to hear, which is the mid, and nothing but the mid. It's chopped off on the bottom, and I try to squeeze as much high in as I can. With the mic in that proximity to the speaker cabinets, you're in for a lot of feedback. On peaks, I can get about 141 dB at the microphone, and I've given up trying to get any better than that, because it's the threshold of pain, isn't it? [*Laughs*]. It averages about 128, which is cool.

MR&M: Mike and Darryl's monitor speakers are a little different, aren't they?

NM: I don't know; I never listen to their mixes. Chester and Darryl are constantly changing their mix very slightly before each number. So if there's anything wrong with their mixes it's not my problem. Chester says it's the best thing that's ever happened. Before they got this system, there was another guy over there just doing Chester's mix, which meant they had a three-way split on the mixes—and all sorts of problems.

MR&M: Basically it seems your EQ on the board itself is fairly flat.

NM: Apart from the vocals and the drums. We put a bit of heavy top boost on the bass drum because they both like to hear the snap sound. But by and large, a bit here and there on the bass, that's about it.

MR&M: What are the mixes each member likes to hear?

NM: When Phil's singing, basically he likes to hear himself, and Chester's kick drum and snare drum...and that's it. Darryl and Chester pretty much have blended mixes like you hear out front, with the absence of themselves, as does Tony as well. I know Mike doesn't use his mix much—a bit of keyboards and a bit of drums, that's it. But I usually don't listen unless I think something's wrong. Otherwise, I'm too busy listening to Phil. Phil likes echo on his voice—about 30 milliseconds.

Much of the system is run on multicore. This one here runs all the drums fills, that's all the amplifiers on the right, and the returns from the mix system. Makes it quick to plug up.

Sound mixer Craig Schertz of Showco has worked a number of Genesis tours, and on this one he's working with a P.A. that came directly from the Rolling Stones tour of Europe. Schertz is especially fond of the band's monitoring, for it provides him with a clear mix he has to do very little to. He also finds

Genesis a joy to work for—not only as performers, but as people, too.

MR&M: Can you fill me in on the board you use?

Craig Schertz: This is the standard Showco "Superboard." This is 30 channels in, 8 submasters, which could be quad-panned. Although, we've only done two shows that ever needed quad panning—David Bowie in 1974, and a thing called Starship Encounters with orchestras where they had some quad. To do a stereo mix, you have to use two submasters.

As far as the electronics go with this band, a guy called Jim Bornhorst and two of Showco's owners were responsible for the design. It was one of the first road boards with parametric EQ on it—three band, fully parametric, including a shelving function on both the low band and the high band. A surprising number of boards don't have that, but I find the shelving function very useful.

They're three sends, all identical, standard pre-EQ, post-EQ, pre-fade and post-fade. The last two of these boards built also have a mic-line position, so you get 30 dB more padding in the line position, since you get so many more line inputs these days.

The only effects I'm doing are on Phil's voice, basically. They have so many effects on their instruments themselves, all my stuff is post-fade. They create the sound they want on stage, and I'm blending that with the voices, mostly.

MR&M: What do you do on the voices?

CS: Phil's really into what the English call ADT, sort of automatic double-tracking type thing. What I do is use a short delay, about 30 milliseconds or so, then do some flanging effect with the [Delta Lab] DL-2. They're a couple of reverb cues during the set, echo cues, and then they use a pitch transposer for an octave-lower effect on one line of one song.

MR&M: What about your outboard equipment?

CS: We've got the dbx 900 power rack, but I don't use any limiters, just three noise gates on Phil's tom mics, because they have a tendency to ring a bit, and you can cut off that last little ring so you can get the wallop without it ringing for twenty seconds or so. The limiters are real easy to use; the gates aren't as easy. They're trickier and not as fast to set, but they're real good.

MR&M: What kind of overall mix do you go for?

CS: This is more of a keyboard band than a guitar band, so you tend to emphasize the keyboards more. Both Phil and Chester are both real good drummers, so I just let them get their drums tuned up to where they feel comfortable on stage, and that's usually pretty close. Chester has eight toms, so I have to make sure in the soundcheck that there's a good balance on the four mics as he goes all the way around; same with Phil. I personally like a pretty fat kick bass sound—a lot of low wallop—while Phil likes a little harder sound which we tried to get on this trip. I'm still trying to get the same wallop with a little more attack on it, if you will. Especially on his kick, because when Phil's not singing and on the drums, he tends to play on the real fast songs with quick kick drum parts, so it needs to be a little harder to cut through the rest of the mix. On this particular tour, it's taken us three or four gigs to get a clean signal and enough level to represent what he does on stage.

Mike, with his basses and guitars, has two basses and bass pedals, all going into one rig. That's unacceptable for Noel and me, because we have to have separate control. If he's up there playing bass, and hits the bass pedals, it's like an explosion. So we had to jizz around, and come up with a separate feed for the basses and the bass pedals.

Darryl has a different set-up, but plays the same things—guitar, bass and bass pedals. I don't really do anything on the instruments, because they've really got it on stage.

Tony, we take the stereo keyboard mix that he does on stage, and we also take individual channels (mono) from each keyboard just for boosters, in case he's not level. But he's probably one of the most consistent players I've ever worked with. He's always there, so 95% of the time we go with the stereo mix coming off his keyboard mixer.

It's all interrelated with the monitor system they have. The basic idea was to get a sound on stage that they were all comfortable with and try to go with that. It's worked out pretty well.

MR&M: What special things do you do with Phil's voice?

CS: His voice is real hard to get. He's not a real powerful singer, but he's got a real consistent level. He's one of the few guys I've seen who's improved his technique over the years.

He's got kind of a pretty thick accent, and a voice that doesn't really cut through, if you've heard him talk. I think the vocal would, just by itself, cut through without the effects on it, but he's real into having the ADT, so we have to find a compromise point. Then you have the problem most mixers have in that you get to know the song so well they tend to forget about the vocals. I try to concentrate on keeping his vocals out there. Sometimes I succeed, sometimes I don't.

MR&M: What about your stereo set-up?

CS: This little mixer down here has all the toms and overheads in stereo, and that returns directly into the stereo master. Then I have a submaster that has Chester's kick and snare and Phil's kick and snare. Tonight we have the horns as extra, but they're playing when Phil sings, so I put them on the same sub. Then we have a vocal sub, an effects sub, then all of Darryl's and Mike's stuff, and Tony's keyboards. So basically you've got the drums stereo-panned to make them sound a little bigger, and then the keyboards, which is a pretty full sound there. The guitars are pretty straightahead. There're a couple of pans on some of the songs, but nothing too exotic.

Tony Banks **Mike Rutherford** **Phil Collins**

MR&M: How is this system amplified?

CS: Well, as you can see, we have a bass cabinet and a high cabinet. There are four JBL E140s in the bass cabinets; the high cabinet has two JBL K120s facing the rear—a folded horn design—and a JBL 2482 and a 2441 in each cabinet that are on a passive crossover. And two Yamaha tweeters. Each amp rack has two pairs—a Crown PSA2 on the low cabinets, then a PSA2 in stereo that runs the low mids and the highs, and then there's a Crown mono DC300A on the mid drivers. I think we may be going to a bigger amp on the mids. That's what we have in mind as a winter project. Each of the amp racks has a transformer in each one with an English connector on it, so in Europe, we use an English distribution system, and you can literally go overnight from the U.S. to Europe with no real problems.

MR&M: The richness of Genesis's sound must make for an enjoyable job of mixing.

CS: They're real good, because they don't play real loudly on stage, which helps a lot. Both drummers tune their kits well. Tony is very consistent on the keyboards. But I swear that 85-90% of any sound [mix] is what the group is putting out anyway.

MR&M: What other interesting aspects can you point out on the system?

CS: Well, the intercom module and solo module let me call any channel, or any send or any output, or the tape feeds. And I can put it in this headphone so the guy on stage can hear what's going on, so if he has to troubleshoot, he knows what to look for. But the amp in that thing isn't very big, so it's not really great fidelity. What I do is take the solo output, and then the left and right front outputs are on a switch there, so I can have the solo which goes through a [Crown] D-75 down here. Consequently I either hear the stereo mix in the left-front/right-front position or in the solo. It puts the solo signal into both earpieces. I use that quite a bit to work on particular instruments. I think we'll put a beefy solo amp in the board for that purpose if we build another. And that's basically it.

John Monforte

The Fostex Model 250

General Description

The Fostex Model 250 is a *four*-channel cassette recorder with a self-contained four-channed mixer designed to be operated as a small multi-track recording studio. Internal switching allows the user to perform the standard operations of recording, overdubbing and mixdown with no patching required, although several points in the signal path are accessible for use with auxiliary signal processing gear or whatever else a user might dream up.

The system is basically intended to be a "personal studio," analogous to the personal computers now proliferating in peoples' homes. Just as Neil Armstrong enjoys playing "Lunar Lander" at home, any musician—seasoned studio session sideman or otherwise—might use the Model 250 to compose, perform, or simply experiment without having the enormous expense of studio time limiting his creativity. On the other hand, when it comes time to make that giant leap for mankind, or at least for an LP, it might be advisable to seek the help of something more elaborate. Still, the Model 250 can be useful for practicing parts and rehearsing special effects, and the studio time saved by being practiced and rehearsed will save enough money to pay for the 250 fairly quickly.

Just a few items are necessary to build a complete recording studio system around the Fostex 250. Tape, a few microphones, mic stands and cables will complete a very rudimentary studio. Most bands will already have these, as well as a few other items that, while not essential, should prove to be useful. These would include a hi-fi system for monitoring the cassette, a reel-to-reel tape recorder used for mixdown, and some special effects devices such as reverb units or an assortment of those little effects boxes. The Model 250 is designed so that the levels are compatible, and the items mentioned above will interface neatly. Weighing in at 19 lbs. (8.5 Kg), this unit could be considered portable save the need for a handle and A.C. power. Our unit was equipped for 120-volt 60 Hz operation, but a universal 120/240 volt 50/60 Hz model is also available.

The Recorder

The recording section consists of what, at first glance, appears to be a conventional cassette recorder. It is, however, not your standard cassette format. The machine moves tape at twice the normal cassette speed (3¾ ips) and will record on up to four tracks at once, in which case cassette records on one side only. The high speed extends the high frequency response and increases the signal-to-noise ratio. Dynamic range is further increased by the built-in Dolby Type C noise reduction. The Dolby is switched in and out by a recessed switch on the system's underbelly, making it almost invisible to the user. The 70 microsecond equalization is wired in, requiring the use of a quality high-bias tape. Fostex recommends using TDK SA or

Maxell UDXL-II tape for best results. (For its reel-to-reel machines, Fostex recommends Ampex 457.) The use of metal or chrome tape is not recommended. The resulting performance can be equivalent to an open-reel machine, while using a tape format that can be purchased even at the Bazaar in Baghdad. It should be noted that the cassette format makes editing almost impossible, but this should not be a deterrent for most people.

The controls on the recording section appear spartan compared to more pedestrian cassette recorders, yet functions are included to assist in the multi-track recording process without adding unnecessary complexity. First, the standard transport functions are included without a pause control—which actually would be unnecessary. Four separate record track-selector buttons arm selected tracks for recording or erasure. Above each button is an LED that blinks to warn of the imminent fate of the corresponding track. Once the tape is rolling and the RECORD button is pressed, the LEDs turn full-on to indicate which tracks are erasing and recording.

It is also possible to go into the record mode using a footswitch (not provided by Fostex) plugged into the rear of the Model 250. This is usually used to insert or "punch in" a portion of a previously recorded track. Of course, this powerful studio technique is usually performed by the recording engineer, but the one-man studio must be able to get into record mode while his hands are occupied making music. In our opinion, a footswitch is essential to a personal studio, and while Fostex does not include one with the Model 250, it is no problem to make one yourself.

A very sophisticated electronic tape counter has been included in the Model 250, along with a return-to-zero button. It is very accurate, and should prove to be useful in multi-track recording where it is necessary to repeatedly go back to the beginning of a tune.

A variable-speed control completes the transport functions. Its range is only ±10 percent, which is less than a whole-tone, thereby limiting its use in making tuning changes. As useful as this control might be, a wider range would be necessary to facilitate fast riffing.

The Mixer

The left half of the Model 250 is a four-channel mixer with a stereo output, two auxiliary mixes and a stereo line level return.

Input signals enter the mixer through four ¼-inch phone jacks on the front of the unit. A preamp trim control adjusts the gain to accommodate low- or high-impedance microphones, instrument pickups, effects boxes, guitar amps, or whatever. The signal then proceeds to a switch that selects either the input or the corresponding previously-recorded tape track. This allows the same channels to be used for both recording and mixdown. Next, the signal level is controlled by a linear slide fader that sets the channel level. After this, the signal passes through a two-band equalizer that can boost or cut 10 dB at 300 Hz or 4 kHz. Center detents on these two controls give flat response. From here the signal gets assigned via a stereo bus-selector switch and a pan pot, then proceeds to the recorder (or to an external recorder in the case of a mixdown). Signals assigned to channels 1 and 2 pass through one more linear slide fader for use in mixdown where a stereo level control may be necessary.

Each channel feeds two other mixes that are essential to multi-track recording. First, a monitor mix control feeds a signal from the preamp before the linear fader. This allows a separate mix for the musicians' headphones without affecting, or being affected by, the levels of the tape tracks. The second mix is called the AUX BUS, which takes its signal after the equalizer and sends it back out the back panel through a jack where it is available for signal processing; this is typically used for reverb. The processed signal can be returned to the stereo mix via the Auxiliary Receive control. This input is stereo and is found on the back panel.

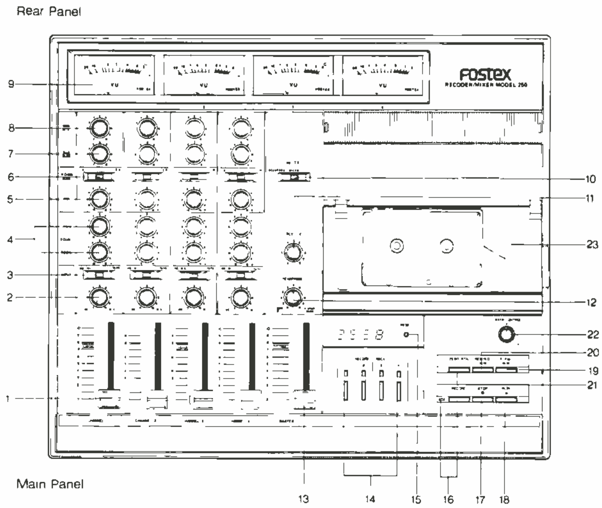
In addition to the previously-mentioned signals, the back panel includes patch points that allow direct connection to the input and output of each tape track, and a direct output from the equalizer before the pan-pot on each mixer channel. This means that external equalizers and other signal processors can be inserted in the signal path so the system can have maximum flexibility. All signals found on the back panel interface through RCA jacks and the level found here are fully compatible with consumer hi-fi equipment.

Fostex has also included four fairly large VU meters with peak overload lights. These meters can be switch-selected between the recorder and the mixer. They are illuminated and are legible even at a distance.

The Manual

For a product as unique as this a very detailed manual is essential, and Fostex has included an excellent one. Printed on heavy paper with holes punched in it so it can be kept in a binder, Fostex correctly assumed that many copies will get dog-eared quickly. To borrow a term from the personal computer field, the manual makes the Model 250 "friendly," or easy, to understand and use, even for the neophyte.

The first page includes the obligatory "safety



1. Input Fader
2. Line/Mic Control
3. Input Selector Switch
4. Equalizer
5. Bus Pan Pot
6. Bus Selector Switch
7. Aux Bus Control
8. MonMix Control
9. VU Meters
10. Meter Function Switch
11. Aux. Receive Level
12. Headphone Level
13. Master 1, 2 Fader
14. Record Track Selector
15. Tape Index Counter & Reset
16. Record Button & Indicator
17. Stop
18. Play
19. Fast Forward
20. Rewind
21. Zero Return
22. Pitch Control
23. Cassette Well & Cover

instructions" that warn against recording yourself in the shower or letting your puppy chew on the power cord. Once past this, however, there are detailed descriptions of how to operate the controls and a very complete section describes the basic fundamentals of recording, overdubbing, and mixdown. Also included are discussions on how to bounce tracks and punch in musical sections. There is even a section on interfacing with other gear which includes discussions of decibels and impedance. The manual winds up with a short bit on how to keep a track sheet and includes a blank one ready for photocopying. In all, the manual goes a long way towards de-mystifying the sorcery involved in engineering a recording. Any musician familiar with these fundamentals would be at a great advantage in a professional recording environment.

Conclusion

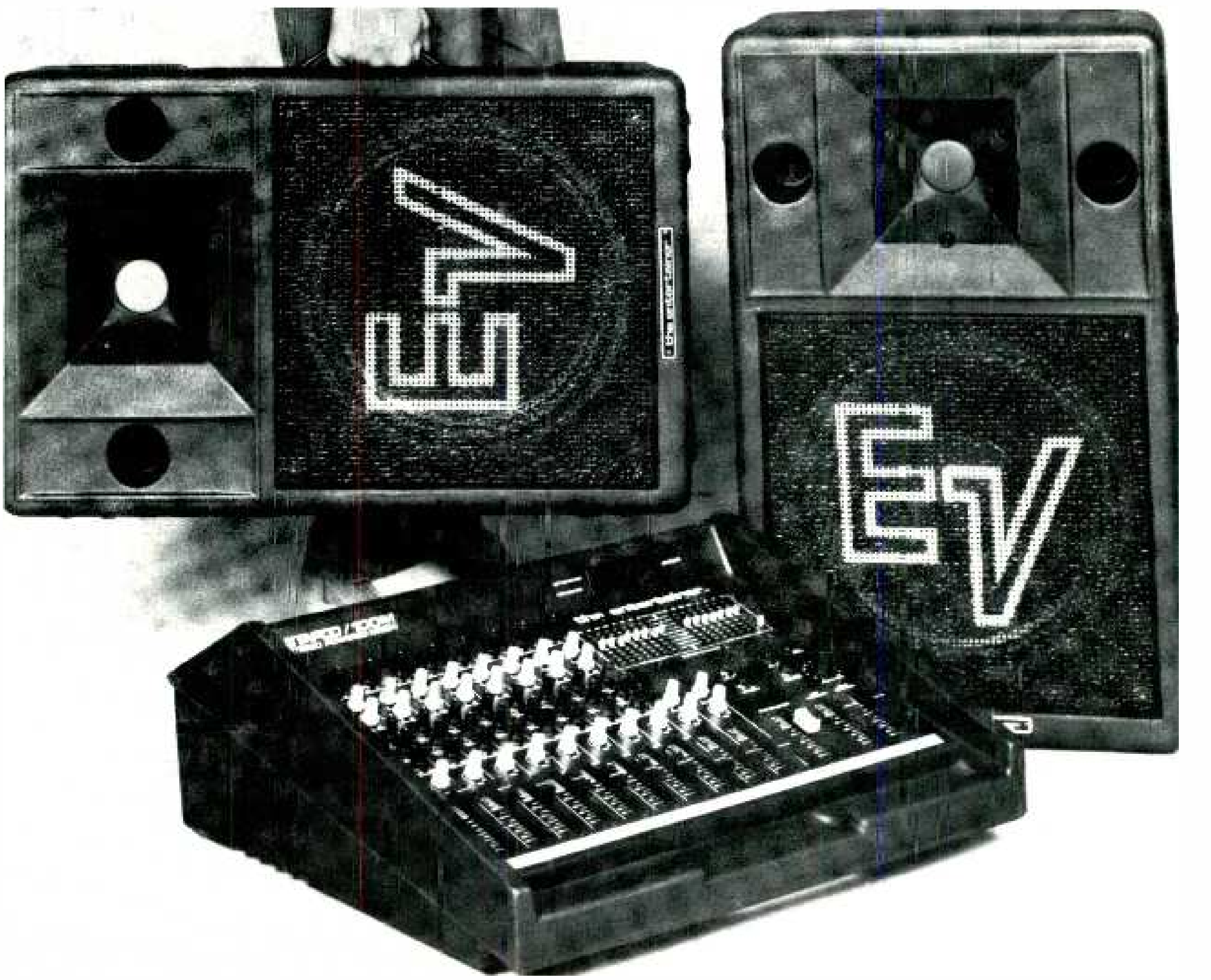
It almost goes without saying that the Model 250 is an extremely versatile device. When used with the sound equipment that most bands have, it can record a demo tape either in performance or in the shed. It can also be used to record slide-show presentations, allowing several projectors to be controlled thanks to its four-track capability. Composers will find it useful for writing parts and experimenting with voicing or orchestration. Also, as mentioned before, it can be a valuable rehearsal tool before a professional recording studio session. Although a standard-speed (±⅞ ips) version is also available, we hope Fostex will consider making a dual-speed version available. With this it would be possible to obtain a rough mix of basic tracks at the studio, using a conventional cassette recorder and then bring the tape home to practice lead and solo parts on the remaining tracks. Also, the Model 250 could double as a hi-fi cassette machine and, with another standard cassette machine, would allow you to make copies of your demo tapes. This objection, however, is relatively minor and may not be of any importance to some users. Overall, the Fostex 250 is very successful in making multi-track studio technology readily available, giving musicians the creative tools for developing and refining their talent.

The Entertainer

Chip Stern

IT'S RATHER FRUSTRATING to consider the priorities of working musicians searching for the ultimate presentation of their sound. It seems that they'll spare no expense when it comes to equipping themselves with an arsenal of guitars, pedals and signal processors—all with the goal of finding the most personal and most pleasurable dimensions of timbre and color to forge a sound signature. And, of course, we're all familiar with the "My band's bigger than your band" syndrome, wherein professionalism (*status*) is somehow synonymous with the number of instruments and cabinets you can cram on a stage, and where music is measured in sound pressure levels. Naturally, everyone wants to have impact and presence, but musicians are so used to being *inside* the sound, they're often unaware of how muddled and unbalanced it can be when dispersed out into the listening field. The end result: thousands of dollars worth of gear and no gigs. After all, no one wants to be subjected to bad sound.

Sensible sound men have been trying to get musicians to turn the volume down for years, a war of attitudes they usually lose. Too bad. A few years back, we got to hear both Parliament/Funkadelic and Bob Marley at Madison Square Garden within a very short span of time, and the disparity in sound quality was striking. P/Funk went for the wall-of-amp gusto and decibel assault, yet it all merged into a muddy woof as the sound ricocheted around the cavernous arena. Marley's Wailers employed almost as many singers and instruments as P/Funk, yet their stage set-up humble "club" gear like the Ampeg B-15 and Fender Pro Reverb was much more downscale. However, *they*



filled the Garden with clean, spacious sound; without sacrificing the electronic intensity of reggae, they managed to make a hockey rink sound intimate.

Simply put, the last link between your music and the audience is the sound reinforcement system. Those musicians and bands who are serious about their craft would do well to consider buying a few less cabinets and guitars and investing in good instrument mics and one of the new generation of self-powered, portable mixing systems like the Electro-Voice/Tapco Entertainer. The Entertainer is a simple, versatile sound system whose main attributes are its lightweight portability (at 36 lbs. for the mixer, and 28 lbs. for the 100S speakers) and its ability to disperse uncolored, musical sound into small-to-medium-sized clubs and theatres. While it is certainly not designed to handle the melt-down levels of hard rock, the Entertainer has excelled in any number of studio and live applications, both acoustic and electric.

It's the superb balance between the Entertainer's power output and its transducers that allows this system to deliver such clean, spacious, uncolored sound. The Entertainer employs dual MOS-FET power amplifiers (EV/Tapco uses the same Hitachi MOS-FETs as Hafler) that deliver 100 watts into 8 ohms (such as the matching 100S Constant Directivity Speakers). In addition to MOS-FET circuitry's inherent stability and cool operation, the onset of clipping is much more gradual than with conventional bi-polar transistors; this in part, accounts for both the smooth sound and light weight of this compact board.

Vital Statistics

With their luggage-type handles, the 100S speakers (at 24-in. H x 15-in. W x 8.5-in. D) are also exceptionally easy to transport. The molded plastic design of the 100S' vented enclosure contributes to sound quality as well as portability. EV/Tapco says that the enclosure is constructed of a "dual layer sandwich with a foam cell structure for the internal wall layer," and claim that "this yields a wall thickness similar to wood cabinets." Well, the proof is in the sound anyway, and this cabinet contains a 1½-in. EV Super-Dome tweeter with a 1.6 lb. magnet structure coupled to a high frequency director that is molded as an integral part of the cabinet, as well as a 12-in. direct radiator woofer with a 10 lb. magnet. In construction of these components, care was taken to use materials that are resistant to environmental influences, with a metal screen to protect the woofer assembly. The integral crossover network is a 12 dB/octave dual-section-type, occurring at 1.5 kHz, and the screen yields a respectable 96 dB SPL at 1 watt/1 meter. Without getting into the technical rationale behind the concept of a *Constant Directivity System*, EV/Tapco claims 100 degree wide horizontal and vertical coverage in midband frequencies from 500 to 10,000 Hz—eliminating dead zones, hot spots, and providing flat, uniform response. What our ears heard was an effortless dispersion of sound, with tight, flat bass, crisp, defined high end (up to 18,000 Hz ±3 dB), and a remarkably open midrange. The sense of stereo imaging and spatial depth is impressive and in the course of our testing we were also impressed by the fact that there was no real sense of speaker location; the 100Ss didn't even sound like they were there, which is as high an accolade as we can imagine for any speaker system. To further tailor the response of this system, a four-position high frequency control on the back panel allows you to attenuate response in 3 dB steps from flat to -9 dB; molded-in inserts allow horizontal or vertical mounting of the 100S on the optional 480A telescoping stand.

Performance Features

The performance features of the Entertainer are what you'd expect from a high quality P.A. system. There are 10 input channels, two of which (channels 9 and 10) are high impedance/line inputs with both a ¼-in. phone jack and isolated RCA phono jack (the isolation allows both jacks to be used simultaneously) designed to accept tape machine playback; turntables; effects returns; electronic drums and keyboards; the output signal from a movie projector; another mixer's output signal and line level inputs. Channels 1 through 8 offer XLR mic input jacks to accept balanced low impedance sources from microphones, direct boxes and instruments, and ¼-in. phone jacks for high impedance line inputs from instruments, audio equipment and unbalanced sources (with an output level from -40 to +26 dBu or 0.01 to 15 volts).

Channels 1 through 8 are equipped with a *Sensitivity* control, which adjusts the gain of the input amplifier to maximize the headroom of the various input sources, optimizes signal-to-noise ratio, and minimizes the possibility of overload distortion. By carefully monitoring the red *Peak* overload LED for clipping, and by achieving the proper balance between the *Channel Fader* and the Sensitivity control, one can produce a clean signal in the overall mix. By backing off on the Sensitivity and compensating with the gain of the Channel Fader, clipping may be avoided. To further balance the sound, rotary *EQ Controls* provide critical cut and boost of *High* (10 kHz Shelving Type ±18 dB), *Mid* (3 kHz Peak/Dip ±12 dB), and *Lo* (100 Hz Shelving Type ±18 dB) frequencies. In addition, each channel contains a *Monitor Send* (independent of all channel controls except Sensitivity), a *Pan* control and an *Effects/Reverb* send control to determine the amount of coloration for each input (the internal reverb and external effects loop share a common send, so one cannot have separate reverb and effects sends for each input—it's one, the other or both).

The Output Section

In the output section of the Entertainer, the *EFX Master* control adjusts the level of the signal appearing at the *Effects/Send* output jacks on the back panel (independent of the signal sent to the internal reverb), with independent control set on each channel's EFX/Rev send; the *EFX Return* control sets the loudness of externally generated effects signals in the main mix. Each channel's EFX/Rev send also switches in spring reverb, determined by the *Rev Level* control. The accompanying *Rev Color* control is a particularly useful feature, allowing the user to determine the reverb timbre that best suits one's individual taste or that complements the acoustical characteristics of a particular room. Rotating the knob to the left yields a Warm sound (which emphasizes the lower frequencies), while rotating it to the right offers a Bright sound (emphasizing the higher frequencies); in situations where a room was excessively pingy or muffled, the Rev Color control proved very effective for mellowing the overall mix or adding presence. Rounding out the output section are *Left & Right Submaster, Mono Master* and a *Monitor Mix* (a mix of the individual channel monitor sends; *Dual 8-Band Graphic Equalizers* (±12 dB at 63, 125, 250, 500, 1000, 2000, 4000 and 8000 Hz), and a meter section employing a *Dual Channel Fluorescent Bargraph Display* for monitoring the output from the internal power amplifiers coming through the graphic equalizers. As the amplifiers may be set for stereo or mono operation by a *Mode* switch on the back panel, the bar graphs can indicate left and right submaster output (in stereo) or the output of the Monitor Master (left bargraph) and the Mono Master output (right).

Powerlock

To the right of the meter section, EV/Tapco's patented *Powerlock LEDs* monitor each individual amplifier channel for the onset of clipping (automatically, the company asserts), reducing the gain just enough to prevent clipping. "Occasional blinking of the LEDs is normal," states the manual. "Sustained operation of the LEDs, while not harmful in itself, is an indication of amplifier overdrive (operating the system beyond its limits) and should be dealt with...." Remember, the Powerlock LEDs do not indicate amplifier clipping. Rather, they indicate the activation of the Powerlock circuitry, and in essence are saying: "If it weren't for Powerlock, the signal at this instant would've been clipped, causing distortion." Based upon our experience with this unit, we're not sure if this is entirely accurate, though it may very well be a matter of semantics. What we heard and saw mirrored in the pulses of the Powerlock LEDs was indeed clipping and distortion; although we could audibly detect the strain and breakup of a signal, clearly some sort of automatic soft clipping, or limiting was taking place, preventing this clipped output from reaching the speakers and preventing the overall volume from rising significantly. In other words, though the signal was degraded, there was no violent spike of sound to tear the tweeter's heart out or dropkick the woofer into your lap.

In addition to the rear panel functions we've already described, there are a pair of left and right amplifier outputs which can accept any combination of speakers with a total impedance of no less than 4 ohms and speaker fuses for both channels; stereo and mono mixer outputs; hi level and lo level effects sends (the former for studio-type effects, the latter for guitar type pedals and effects boxes), and an effects return input; a monitor send output (that functions also as a line level output capable of driving power amplifiers or other line level inputs); amplifier and equalizer patching jacks; headphone jack and reverb footswitch jack; a mode switch that assigns the Entertainer's internal graphic equalizers and power amplifiers, and a *Phantom Power Switch*, which controls the internal 48V (nominal) power supply that allows dynamic or ribbon microphones to be used alongside phantom powered condenser microphones. In addition, the Entertainer may be used for recording via the stereo outputs, mono outputs, equalizer outputs or monitor and effects sends (the monitor output being independent of nearly all the mixer's controls, while the effects send is affected by nearly all of them).

Conclusion

As you may have already gathered, we were quite favorably impressed by the Entertainer as a self-contained package. Though it is *not* inexpensive ($2,818.00 for the system), it offers no-compromise performance and is a real workhorse. While EV/Tapco shies away from recommending it for rock applications, we found it to be a perfectly adequate performer for your basic lounge rock gig. To wit, on a Ho-Jo's gig (Howard Johnson to you) it held its own with a guitar, bass, keyboards, drums, and vocal line-up. Placing mics off-axis on the instrument amps and using an aerial mic and floor mic on the drums, we basically panned the instruments left and right in the mix, using only enough gain to bring out their presence in the room, and centered the vocalist right in the middle, bringing her out enough so that she didn't have to holler to be heard. She was extremely pleased by the quality of her voice in the mix—her singing came out fast, full and warm, testament both to the dynamic headroom of the dual power supply and the speakers' smooth midrange response. When we tried to pump too much instrument gain through the system, the Powerlock LEDs started blinking, so rather than get the band to turn down, we just backed up on the P.A. and EQ'd things to favor the vocalist.

We also found that, in a pinch, the Entertainer made an acceptable instrument amp for guitar or bass (plugging a guitar in through a Scholz Rockman, we were able to create a remarkable stereo effect), and as a multi-keyboard set-up it was exceptional.

But where the Entertainer really shines is in acoustic application. In a studio rehearsal of the World Saxophone Quartet, the musicians (who normally play without any amplification) were surprised at how little the Entertainer altered the basic parameters of *their* sound. Saxophonist/composer Julius Hemphill was so impressed by the unit that he took it on a gig to New York's Public Theatre, with the hope of bypassing the questionable input of the Cabaret Theatre's sound crew. The theatre seats 300 people, and as its main use is for plays, it has very high ceilings, a large stage, and a fairly long distance for sound to travel to the back. Using Hemphill's clip-on Sony microphone and a Roland FET Pre-Amp plugged into one input, and poet Curtis Lyle in another, we faded Hemphill's saxophones to the left and Lyle to the right, placing the speakers on chairs approximately 10 feet from the back of the stage (20 ft. behind the musicians), angled up and slightly to the middle to achieve good spread to the top rows. The sound was bell-like and transparent, and surprisingly enough, filled the whole theatre without strain; we then close-miked the 100S speakers and used the house P.A. simply to provide some ambience for the listeners in the back. We also made a first rate two-track recording going direct from the mixer outputs. Finally, on a test we didn't control, we heard the Entertainer in a jazz club called Lush Life work wonders with a piano trio and vocalist at a press party staged by EV/Tapco. Being familiar with the peculiar amplification problems of this long, rectangular room, we were impressed again by the Entertainer's uncolored projection of the music—nothing more, nothing less.

In concluding, the Entertainer has clearly reached a new level of P.A. performance in a compact package of unprecedented size and sound.

Ambient Sound

By Len Feldman

The Emphasis At AES Was On Digital

Of the many scholarly engineering papers presented at the most recently held Audio Engineering Society Convention, I counted at least eight that had the word "Digital" in their titles. In addition, there were a great many exhibitors at the Disneyland Hotel in Anaheim, California (yes, that's where the AES Convention was held this past October) who had digital, or digitally-related products to introduce and display. Walking about the exhibit floors and sitting in on some of the papers, it became abundantly clear that in professional audio, at least, there was no longer any debate about where audio was going... It was going digital!

Of course, there were still some objectors and detractors. The debate about sampling rates goes on, as does the question of the audible effect (if any) of anti-aliasing sharp-cut-off filters which introduce radical phase shifts at high audible frequencies. The effect of such filters (which are a basic requirement of any PCM system that uses a sampling rate that's barely twice the highest frequency to be recorded) on transient response remains a subject for debate as well. If you have ever tried to record and reproduce a square wave of even low frequencies using a 14-bit or even a 16-bit PCM system with a sampling rate of 44.1 kHz or thereabouts, you know only too well that the "ringing" you will observe in the reproduced square wave would be enough to discourage any true audiophile from venturing near any digital recording or playback equipment.

But when all of that has been said, the truth is, that to my ears at least, digitally recorded and reproduced audio sounds far better, overall, than even the best analog master tapes I have ever heard. I suspect that most of the manufacturers involved in digital audio must agree with me, judging by the raft of digital products on display at the aforementioned AES. A couple of these were sufficiently innovative to warrant more than casual mention in this column.

Sansui Cleans Up The Digital Code

By now, there are quite a few digital audio processors available for consumer use. Not surprisingly, because these 14-bit processors are all formulated to work with the EIAJ (Electronic Industries Association of Japan) Standards on PCM recording with a video tape recorder interface, many small recording studios and broadcasters have latched on to one or another of these lightweight processors and have tied them into either ¾-inch video tape transports (such as U-matic) or, in cases of real economy, to half-inch consumer-format VCRs, such as those which use the VHS or Beta systems. Some small studios have even discovered, to their great surprise, that these inexpensive PCM processors, working with low-price VCRs, can be interfaced with professional digital editors. Often, the professional editors end up costing more than the processor-plus-VCR combination itself, but that's beside the point.

To get to the point, Sansui, along with anyone else who has tried to skimp on tape, discovered that the processor-plus-VCR digital approach works only if you operate the VCR at its fastest speed (Beta II in the Beta format machines; SP in the VHS decks). Try to operate at the slowest EP speed on a VHS machine using any existing PCM processor and when you play back the tape, the dropouts and sound breakup are so bad that you're tempted to rush quickly to the power-off switch. Sansui lists four reasons why this is so in a technical paper which they wrote as part of their introduction to a new PCM system called Tricode PCM:

1. A VCR scans video tracks of a video tape as it records and plays back digital data. At normal speed, track width is 58 microns. At one-third speed on a VHS machine, the track width is only 19 microns. This reduction in width leads to a threefold increase in noise. In addition, the head tends to trace not only the desired track but adjacent tracks.

2. Any misalignment in a VCR transport is likely to show up more prominently at slow speed than at the faster SP speed. If the mechanism is not precisely aligned, jitter or vibrational shifts in the time base can occur. During playback, data must be read on a regular time base. Therefore, any shifts in data time base can lead to reading errors.

3. Since frequency response of the system is narrower at slower tape speeds, variations in amplitude of the digital data are more likely to occur, causing the digital patterns to close in from top to bottom; there is not as much amplitude difference between a "0" and a "1" in the digital data stream.

4. VCR manufacturers, conscious of the degraded video picture quality at slower speeds caused by the problems cited, usually incorporate special circuitry to compensate for such picture deterioration. While such circuitry often achieves its intended purpose insofar as picture quality is concerned, when the VCR is used for digital audio data storage, that data is often damaged beyond recognition.

One form of video compensation circuitry mixes adjacent pairs of horizontal lines in the video picture. Since adjacent lines are quite similar in content, such mixing tends to reinforce the image and improve the video signal-to-noise ratio. In digital PCM audio, however, adjacent lines are *not* interrelated, and such mixing of lines is likely to produce pulses of data which are neither distinct "0s" nor "1s." Such non-standard pulses are illegible to an ordinary digital PCM processor.

Sansui's Tricode PCM processor allows the user to make and play back digital audio recordings using any VCR at its slowest speed. In simple terms, the system reads and corrects data by extrapolating future data from past data. The special circuitry, incorporated in a single chip, operates only during the playback or digital data reading process. Thus, it in no way affects recording procedures which remain standard and in accordance with previously agreed-upon EIAJ recommendations for PCM recording.

dbx Goes Its Own Way

With the imminent arrival of digital audio and its potential for a lifelike dynamic range, it's become fashionable to predict the demise of all noise reduction techniques, not to mention analog tape recorders, phonograph turntables, phono cartridges and a host of other products whose very survival is analog-audio based. I confess that I've been guilty of the same sort of doom-and-gloom predictions. However, such prophecies do not take into account the inventive genius of the companies involved in these various product technologies. A good example is dbx, whose very existence had, until now, been largely based upon noise reduction signal processing. Recognizing this, dbx has boldly stepped into the digital arena with the introduction of their own version of a digital audio processor intended for professional studios that can't afford the "pro" versions of PCM processors that sell in the five-figure dollar ranges.

dbx calls their system CPDM (instead of PCM) which stands for Companded Predictive Delta Modulation (whew!). Delta modulation has been known for years to be a low cost means of analog-to-digital data conversion. In this digital process, the numbers produced by the A/D converter represent differences between successive sampled voltages, rather than the instantaneous voltage of the audio signal at each instant of time, as is the case with conventional PCM processors. Ordinary Delta Modulation, however, produces sound with a dynamic range of only 55 dB—a less-than-acceptable range for any music reproducing system. A modified version of Delta Modulation known as Adaptive Delta Modulation (in which digital numbers produced by the A/D converter represent varying differences between successive audio amplitude samples) offers as much as 90 dB of dynamic range, yet is also considered to be unacceptable. It suffers from noise modulation problems and exhibits a noise floor that has a distinct tonal character.

dbx engineers came up with two major technical innovations which made the use of a Delta Modulation system practical for their new processor: Linear Prediction and Precision Companding. The Linear Prediction Circuit estimates a signal's future by monitoring its recent past history. It does this 700,000 times per second. This portion of the circuit is responsible for increasing the dynamic range of the basic Delta Modulator from 55 dB to 70 dB.

The remainder of the dynamic range increase in the CPDM system results from a Companding Circuit. The compander is a direct link between encoder and decoder and therefore there is no possibility of mistracking, as might be the case were there an intermediate link such as in an analog companding system that employs a tape recorder. This technology increases overall dynamic range of the system to 110 dB—considerably more than is offered by most professional PCM processors now in use.

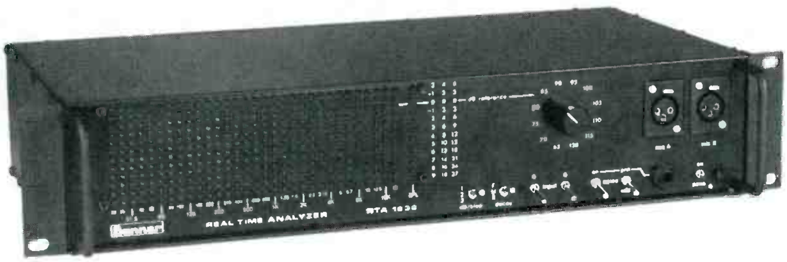
While the dbx digital audio processor is intended to work with professional quality VCRs, it too will work with Beta or VHS type machines as well, putting digital recording within the reach of small studios that simply couldn't afford higher-priced professional PCM processors. Of course, as of this writing, there is no digital editor available for this new machine. Neither is the CPDM system compatible with other existing PCM systems now in use in other studios. Still, for use as a two-channel final mix-down system in a small studio (where outputs from this processor can feed cutting-head amplifiers for disc mastering, for example), the dbx digital system may well enable studios to turn out a lot more digitally-mastered music than they might otherwise, and that bodes well for the future of digital audio in whatever format it ultimately reaches the listening public.

MODERN RECORDING & MUSIC

LAB REPORT

LEN FELDMAN

Banner RTA 1232 ⅓-Octave Real Time Analyzer



General Description: The Banner (division of Optronics Corporation) RTA 1232 is so new that its makers didn't have time to supply us with an owner's manual. Of course, by the time you read this the manual will have been printed. So why bring up the absence of a manual? Simply because the layout of the RTA 1232 is so logical and well thought through that we were able to operate this real-time audio spectrum analyzer just by examining the front panel and rear panel nomenclature. That speaks very well for any piece of electronic test gear, and especially well for an RTA that offers as much flexibility and precision as this one.

The left section of the front panel of this analyzer contains 31 vertical columns of LEDs (12 indicators per column), each corresponding to ISO center frequencies that are one-third octave apart. A separate column of LEDs, somewhat removed from the third-octave indicator columns, serves as an overall SPL meter. Both the analyzer columns and the SPL column are calibrated in dB, with a rotary switch selecting the "0 dB reference" level in 5 dB steps ranging from 120 dB down to 65 dB. Since sensitivity is selectable (by means of a 3-position toggle switch) at 1, 2 or 3 dB per LED step, maximum visible dynamic range can be as great as 33 dB (from +6 dB to -27 dB with respect to "0 dB reference") or as little as 11 dB (from +2 dB to -9 dB). Additional toggle switches select either or both microphone inputs, rate of decay of the SPL-indicating section of the instrument, white or pink noise test signals and power on/off. A standard ¼-inch phone jack on the front panel delivers either of these test signals. Standard XLR connectors are used to connect one or two microphones which may be phantom-powered (15-volts) directly from the RTA 1232 itself. Two line level inputs are located on the rear panel of the instrument and utilize standard ¼-inch phone jacks.

Test Results: Perhaps the most important feature of the RTA 1232, and the one that distinguishes it from other low-cost analyzers, is its double-tuned filter circuitry. The importance of this feature is best understood by examining *Figure 1*. In this diagram you can see that the normal ⅓ octave single-tuned shape has a rather broad "skirt." This response

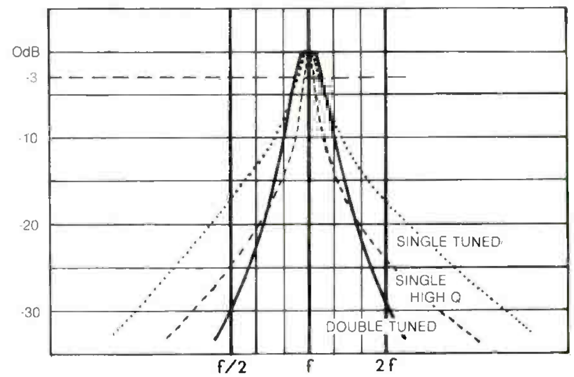


Fig. 1. Comparison of single-tuned, High Q, and Double-Tuned filters used in RTAs. Banner RTA 1232 uses double-tuned filters.

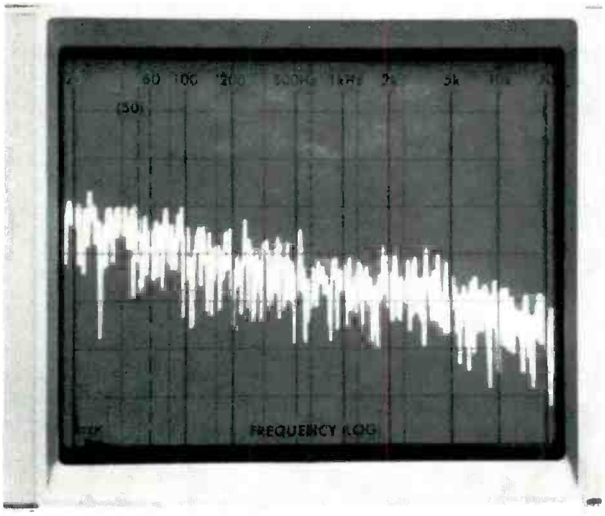


Fig. 2. Pink-noise output of Banner RTA 1232 had typical 3 dB per octave slope, for equal *energy* per frequency band.

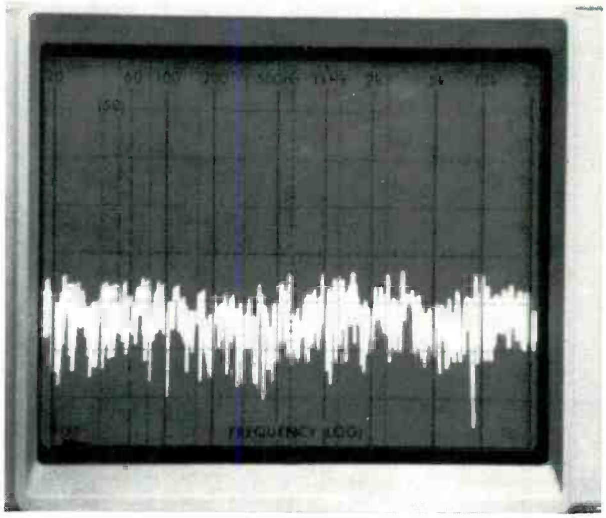


Fig. 3. White noise output of RTA 1232 has essentially flat amplitude response, measured from 20 Hz to 20 kHz, in this 'scope photo taken from spectrum analyzer display.

characteristic tends to mask out frequencies that fall beneath the skirt. To get around the problem, some designers simply increase the Q, or selectivity, of the filter. While this helps somewhat, it also introduces another problem. The sharpness of a high-Q single-tuned filter may leave gaps between adjacent filters such that you can't see the amplitude response of interband frequencies. What then appears to be "flat response" may not be flat at all. The unobserved frequencies between bands may be just the ones that need equalizing because they are causing feedback, amplifier clipping, etc.

The double-tuned shape of the RTA 1232 filters have fairly steep sides which hide very little of nearby frequencies, but at the same time they exhibit a somewhat flat-top shape that catches interband frequencies. Certainly, double-tuned filters for RTAs are not exclusively a Banner idea, but accomplishing this design at approximately half the price of other double-tuned RTAs makes this a rather unique instrument. Of course, there are RTA refinements which can be found on more expensive units which do not appear on the RTA 1232 (such as memory storage of response curves and battery operation or portability). But as with all test equipment (or equipment as a whole), you get what you pay for, and in the case of the RTA 1232, that's quite a bit.

Figures 2 and *3* are 'scope photos of a spectrum analysis of the built-in pink-noise and white-noise generator outputs. Signal amplitudes of these random noise signals were 86 millivolts for each, measured on an average-responding meter calibrated in volts, RMS.

In the slow decay mode, with sensitivity set to 1 dB-per-step, a decay of 10 dB took approximately 19 seconds. With the decay switch set to medium, the same amount of decay took 3.3 seconds, while in the fast mode, decay was virtually as rapid as attack time, and too fast to be timed with a manually-operated stopwatch.

This type of instrument does not lend itself to very much laboratory measurement and analysis, but as you can see from our Vital Statistics chart at the end of this report, all of the specifications and features listed by the manufacturer were confirmed, and a minimal amount of additional data concerning the instrument is provided.

General Info: Panel dimensions of the RTA 1232 are 19-in. by 3½-in.; the chassis is 9-in. deep. Our sample was wired for 120 volt operation but the unit can be obtained for 240 VAC, optionally. Price: $1250.00.

Individual Comment by L.F.: At its suggested retail price of $1250.00, the Banner RTA 1232 Real Time Analyzer/SPL Meter offers straightforward, precise operation with no gimmicks. It is ruggedly built and utilizes high-grade components that should give long and trouble-free service. Banner's one-year warranty (as opposed to the usual 90 days) covering repairs, replacement and labor suggests that they have every confidence in the longevity of the product, too.

One addition which might make sense would be a level control for adjusting the white or pink noise output. Keeping it at a fixed level often forces the user to operate other level controls (such as on the mixing console or power amplifier) at inconvenient settings.

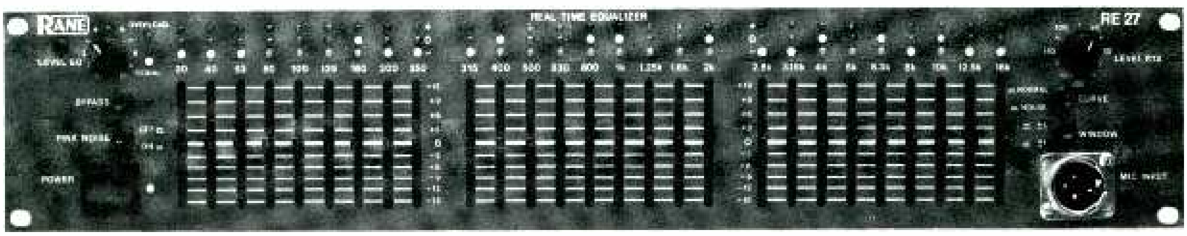
Of course, if this proves to be a problem in your installation, you could, I suppose, always add your own external attenuator between the noise-output jack and the input to the system being analyzed or equalized. The three rates of decay for the SPL meter (which operates completely independently of the ⅓-octave LED displays) were ideally regulated.

As Banner honestly states in their brochure describing this analyzer, there are many things this unit can't do. If you need an analyzer that can be held in the palm of your hand or whose filters are computer controlled, these are available and cost several thousands of dollars. Having examined and used them, as well as this Banner entry, I can honestly state that Banner has managed to give the user about all that could be incorporated in an RTA at this price.

Vital Statistics: BANNER RTA 1232 1/3-Octave Real Time Spectrum Analyzer

SPECIFICATION	MANUFACTURER'S CLAIM	LAB MEASURED
Number of bands	Thirty-one (ISO)	Confirmed
Meter Reference Range	65 to 120 db	Confirmed
Usable range (SPL)	N/A	38 to 126 dB SPL
LED Display Matrix	12 x 32	Confirmed
Number of SPL Decay Rates	Three	Confirmed
Sensitivity, per LED	1, 2 or 3 dB	Confirmed
Bandpass characteristics	See Fig. 1	10 dB/3rd octave
Pink Noise Level	N/A	86 mV
White Noise Level	N/A	86 mV
No. mic. inputs	2	2 (phantom powered)
No. Line inputs	2	Confirmed
Panel Dimensions	19" x 3½" high	Confirmed
Suggested Retail Price: $1250.00		

Rane RE-27 Realtime Equalizer



General Description: The Rane RE-27 is, first and foremost, a well-designed third-octave equalizer with 27 separate band filters set at ISO center frequencies, each capable of providing approximately 12 dB of boost or cut. In addition, the RE-27 combines everything necessary to perform accurate on-site equalization of a sound system. Specifically, it provides a source of random pink-noise, a calibrated microphone, and a clever but inexpensive system of LED indicators that give a clear indication of system response and correct EQ settings.

The Rane equalizer-analyzer (to call it by its appropriate name) is rack-mountable, but can just as easily be positioned on a table top or on the rear apron of many mixing consoles. Despite its relatively low profile, the 27 sliders offer sufficient throw above or below their detented center positions to give the operator precise control of adjustment. Calibration of the sliders is in 3 dB increments and the bank of sliders is divided into three groups of nine sliders, corresponding essentially to the bass, mid-range, and treble regions of the audio spectrum.

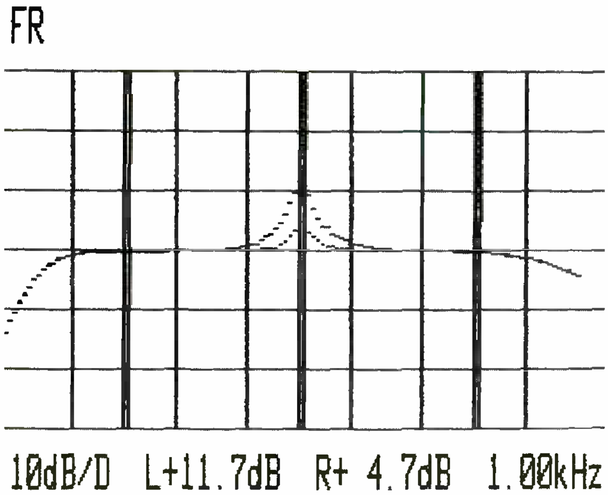


Fig. 1. Rane RE-27: Filters maintain constant bandwidth at all settings of slider controls.

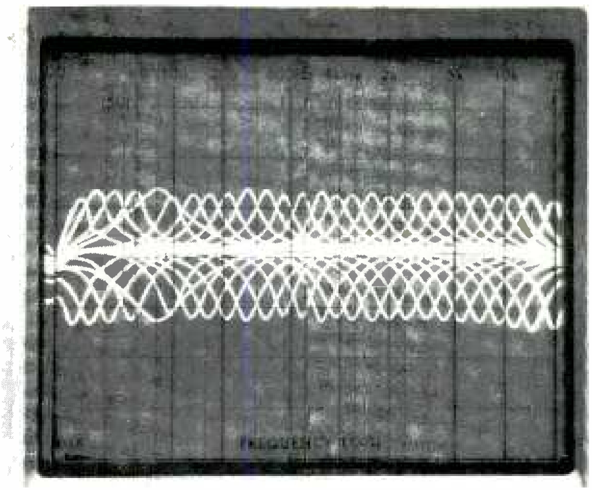


Fig. 2. Rane RE-27: Multiple sweep on spectrum analyzer was used to plot boost and cut characteristics of each of the 27 third-octave filter bands.

An equalization level control at the left of the panel controls volume of signal through the equalizer section and can be adjusted for up to +12 dB of gain. Nearby are an equalizer bypass switch, a power on/off switch and a pink noise on/off switch. Three indicator lights in this area are used to denote activation of the bypass switch, presence of an audio signal (any input above -20 dBm) and overload levels (anything above 4 dB below clipping). The right end of the front panel contains a curve-select switch (normal for flat response or house curve with a mid-range dip in response), a microphone input jack and a switch and control which are associated with the LED analyzer display.

The LED display consists of three LEDs per filter band, located directly above each slider. A red LED lights up when response is too high in that band; green LEDs light when response is within 3 dB or 1 dB (depending upon the setting of a window select switch), while a yellow LED comes on when response for that band is too low. An RTA level control is used to adjust microphone level (or line level when no mic is plugged in) to properly drive the display. This control is accurately calibrated in dB SPL so that any display band whose LED is green has the sound pressure level indicated by this rotary calibrated knob, with either a +/-1 dB or +/-3 dB accuracy, depending upon the setting of the window select switch described above.

The rear panel of the Rane RE-27 unit has standard ¼-inch phone jacks for EQ input, EQ output and pink-noise output. A small hole just above the pink noise output jack permits the user to adjust the pink noise

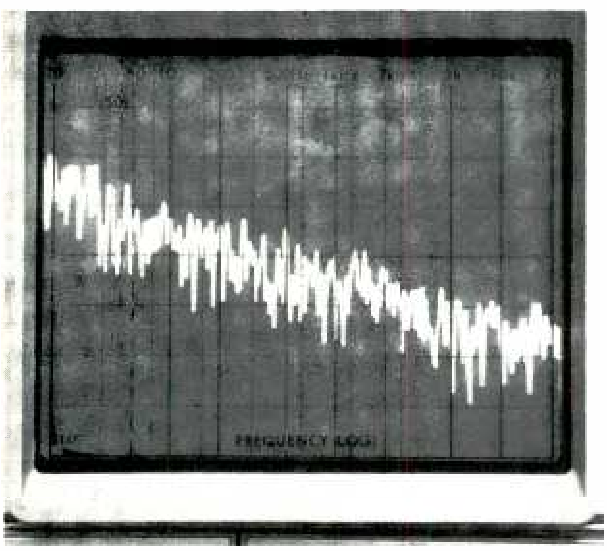


Fig. 3. Rane RE-27: Spectrum analysis of pink-noise output.

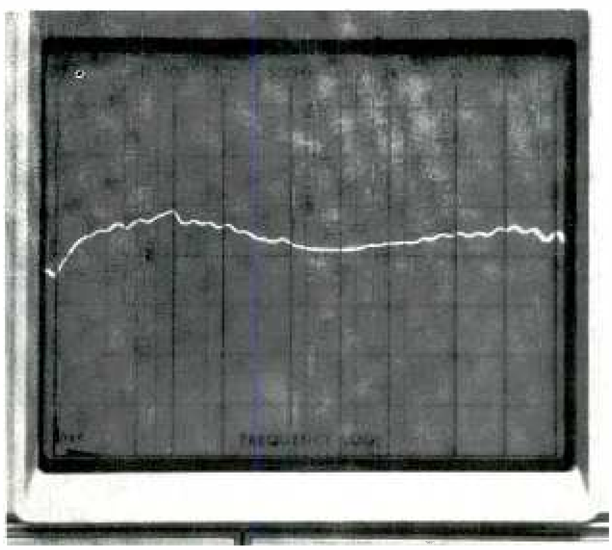


Fig. 4. Rane RE-27: "House Curve" built into the analyzer section.

output level from 0 to 1.2 volts (+4 dBm) to match input level requirements of any mixer, amplifier or other equipment being driven by the pink noise generator. Incidentally, the pink noise produced by the RE-27, while actually pseudo-random rather than truly random, is CMOS digitally synthesized using a 30-bit word length. It has a repetition rate of 2.5 hours and may therefore be regarded as random.

Test Results: Rane makes a big point of the fact that their filters maintain a more constant third-octave bandwidth at all slider positions, unlike other equalizers, which tend to vary bandwidth as slider position is changed. The principle is illustrated in *Fig. 1*, where we sequentially plotted maximum boost and approximately one-third maximum boost of the 1 kHz filter band. Notice that for the lower amount of boost, the total range of frequencies affected is reduced compared with the range of frequencies affected when maximum boost is called for. This does make for more precision in setting up a desired response curve.

Figure 2 is our usual multiple spectrum analyzer plot of all 27 filter bands of the RE-27, showing maximum boost and maximum cut for each slider. We detected a defect in our particular sample during this sequence of plots. Notice that the fifth band from the low-end (the one which has a frequency center near 100 Hz) has a broader-loooking shape than all of the other bands, both in the maximum boost and the maximum cut positions. In other words, the Q of this particular filter band was not as sharp as all of the others and we suspect that this was caused by some component defect in our sample. As usual, this spectrum analyzer plot (as well as those of *Figs. 3* and *4*) is from 20 Hz (at the left) to 20 kHz, with frequencies calibrated logarithmically across the face of the CRT. Vertical sensitivity in these spectrum analyzer displays is 10 dB per division.

Figure 3 shows the spectral makeup of the pink noise output of the RE-27. As usual, the output exhibits a downward slope of approximately 3 dB per octave, indicating equal-energy per frequency spectrum segment, as it should.

Figure 4 is a response plot of the Rane "house curve." This response is achieved when the house curve switch is depressed and the equalizer sliders are adjusted so that all analyzer displays illuminate the green LED. Rane suggests that this particular curve will produce a "warmer," more natural sound for the acoustic environment of smaller clubs and lounges, particularly at lower sound pressure levels. Essentially, their house curve is simply a slight dip in the mid-frequency response of the system, which corresponds, to some degree, with Fletcher-Munson loudness compensation.

Our table of Vital Statistics confirmed most of the published performance claims for the RE-27 equalizer-analyzer. While THD and IM distortion measured for our sample were somewhat higher than claimed by Rane, they were nevertheless at such low level as to be completely inaudible under program listening conditions.

Individual Comment: Rane has come up with a truly unique approach to visual displays for a real-time analyzer. Instead of the usual multiple-LED matrix array commonly found on real-time analyzers, their simple, color-coded three-LEDs-per filter system is not only easy to use but has saved a bundle of money, which results in a price for the equipment that is tempting even to the smallest sound system user or technician. I was actually able to equalize my own listening system to within 1 dB of "flat" (as verified by my much more elaborate and expensive third-octave analyzer that *has* the familiar array of some 250 separate LEDs). I prefer the rather high-Q characteristics of the filters provided on the RE-27 to the broader, low-Q characteristics found on some competing equalizer products. I find it easier to cancel specific feedback frequencies with this type of filter and, in doing so, a minimum of general coloration is added to the overall sound of the system.

The owner's manual is well written, and assumes no previous experience with equalizers or with real-time analyzers. I particularly liked the three diagrams which illustrate how and where to hook in the RE-27 with typical sound systems. All too often, I've found equalizers inserted at points in the signal chain of complex sound systems where they don't belong or where they are likely to be overloaded or fed with too little signal level to work properly.

One word of caution regarding the use of the microphone in conducting the pink-noise test and analysis: *do not* attempt to plug in any other mic but the one supplied with the RE-27. Not only might the calibration of your microphone be different from that of the mic supplied, but it might actually be damaged because the mic-jack is used to power the microphone with a DC voltage that *your* microphone might not appreciate. This warning is, of course, found in the owner's manual—but we all know how few of us bother to read except as a last resort. A printed tag or even a warning inscribed right on the panel near the mic jack might be a good idea for Rane to consider in future production.

It goes without saying (but I'll mention it anyway) that the RE-27 can easily be stored in a stereo sound system, simply by adding another equalizer. The analyzer section, being completely independent of the equalizer section of the RE-27, can be used to analyze the second channel (the one containing a separate equalizer) just as easily as it is used to analyze the channel in which its equalizer section is incorporated. I should mention, too, that although the input and output jacks of the RE-27 are of the phone-jack type, they do have ring-tip-sleeve contacts. Therefore, if you are working with a balanced line input you can wire up your input and output cables from and to three-pin XLR connectors so as to maintain balanced line operation. As you can see, there isn't much that Rane has overlooked in the design and implementation of this multi-purpose instrument. Given its very reasonable asking price, the Rane RE-27 is definitely worth looking into.

RANE RE-27 REALTIME EQUALIZER: Vital Statistics

SPECIFICATIONS	MANUFACTURER'S CLAIM	MR&M MEASURED
Number of filters/frequency range	27/40 Hz to 16 kHz	Confirmed
Frequency Response	31.5 Hz to 20 kHz, +0/-3 dB	34 Hz to 31.5 kHz
Signal-to-Noise Ratio:		
Re: +20 dBm, unweighted	103 dB	101 dB
Re: +20 dBm, A-unweighted	107 dB	107 dB
Re: + dBm, A-weighted	87 dB	87 dB
Maximum Output into 600 ohms	+20 dBm	Confirmed
THD + Noise @ +4 dBm Output	0.005%	0.015%
SMPTE-IM Distortion	0.008%	0.017%
Slew Rate	Above 10V/μsec	Confirmed
Maximum Gain (sliders centered)	+12 dB	+12.5 dB
Input/Output impedance	20K ohms/56 ohms	Confirmed
Sunsonic Filter	18 dB/octave below 31.5 Hz	Confirmed
Ultrasonic Filter	18 dB/octave above 20 kHz	Confirmed
Display Range	+/-3 dB or +/-1 dB	Confirmed
Measurement Range	70 dB to 120 dB SPL	Confirmed
House Curve	-3 dB from 400 Hz to 1.6 kHz	(See Fig. 4)
Microphone Characteristics		
Sensitivity	-65 dB	
Response	20 Hz to 16 kHz, +/-0.5 dB	
Maximum SPL	140 dB	
Dimensions	19" w. x 3.5" h. x 8.5" d.	Confirmed
Weight	11 lbs.	Confirmed
Power Consumption	N/A	17 watts
Suggested List Price	$799.00	

CIRCLE 36 ON READER SERVICE CARD

Rick Chinn

Getting The Most From Your Effects

Like the old saying "Clothes make the man (woman)," effects make the modern musical sound system. Flanging, reverb, echo, automatic double tracking, phasing, chorus, envelope filtering, deliberate distortion, time delay, and many others are tools available to the modern effects maniac. Unfortunately, their operation within the complete sound system is sometimes poorly understood.

For example: I have an MXR Phase 90. Where is the best place in my sound system to connect it? Should I use the patching jacks on my mixer's input channels, or should I use the effects send/return system, or should I put it between the input source and the input jack? What about the impedance match? My mixer says that the output impedance is 60 ohms. Won't that cause a mismatch?

This month, we will explore the world of effects, what makes them tick (kinda non-technically), and how to hook them up to get the best out of them. I'm also going to get up on my soapbox and preach a bit about matching.

First: What is an effect? For the purposes of this column, an effect is a device that does something to an input signal. I am not going to include signal processors like equalizers and compressors. What we are talking about are things like echo, phasing, flanging, pitch shifting, reverb and others.

However, equalization or compression could become an effect, if used in the extreme. For instance, you might use some rather extreme settings of an equalizer to make a "megaphone voice," or some bizarre settings on a compressor to deliberately cause distortion in a bass guitar. Ordinarily though, both of these devices are used more as tools whose presence enhances (improves), rather than drastically changes, the signal passing through them.

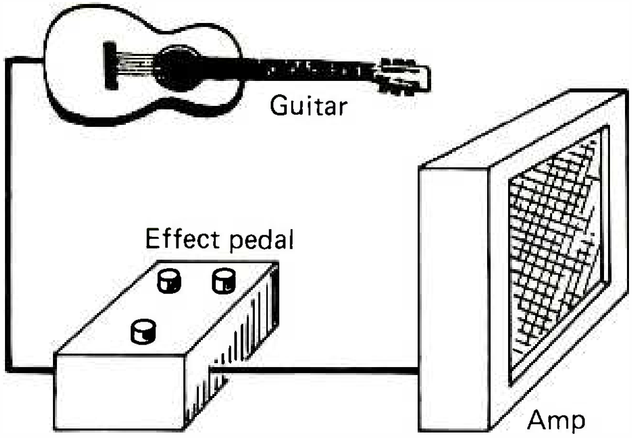


Figure 1.

How an Effect Does Its Magic

Let's talk in very general terms about the treatment of the input signal when passing through a typical effects device.

There are two basic generic effects devices: series and parallel. A series device may be as simple as a pedal-type device intended for use with a guitar. The parallel device is slightly more complicated, but for the purpose of this example, an acoustic reverb chamber will do. There are also effects devices that may operate in *either* series or parallel. These devices are typified by the presence of a "blend" control or its equivalent.

A series effects device is connected by breaking the signal path at an appropriate point, and inserting the device. When equalization or compression is used (either normally, or as an effect), the equalizer or compressor is inserted in series with the signal, perhaps at an insert point within the input channel of the board, wah-wah pedal is another example of a series type device. For those of you who can't separate series from parallel, remember that if you remove a component from a series circuit, nothing works (like some Christmas tree light strings). *(Figure 1.)*

A parallel effects device is usually connected to a mixing board via the effects send and return system. Here, there are parallel signal paths: one for the dry, noneffects signal and one for the wet, effects signal. The mixing console provides the dry signal path via normal output assignments and the effects device creates the wet signal path. *(Figure 2.)* If one of the parallel signal paths (wet or dry) quits, there is still some output signal. The acoustic reverb chamber or an ordinary three-head tape machine used for slapback are good examples of parallel effects devices.

A special case of the parallel effects device is a unit like the MXR Phase 90. Such a device is parallel-structured internally, but is designed to operate within an effects system. Additional examples would include most guitar "pedal" type effects. *(Figure 3A.)*

Other series/parallel type devices may operate properly in either mode, depending on a control setting. The presence of a "blend" control is usually a dead giveaway to the identity of such a series/parallel device. A good example is the TAPCO 4400A Reverb System. *(Figure 3B.)*

For the remainder of this discussion, I will refer to series devices as Type I, parallel devices as Type II, series/parallel pedal devices (like the Phase 90) as Type IIIA, and other series/parallel devices as Type IIIB. Table 1 shows some popular effects and their classifications. See if you can figure out why they are classified as they are.

A good many of these devices let you control the strength or intensity of their effect. In a Type I device, such as an equalizer, the control is one or more of the boost/cut controls. In a Type II device, the intensity

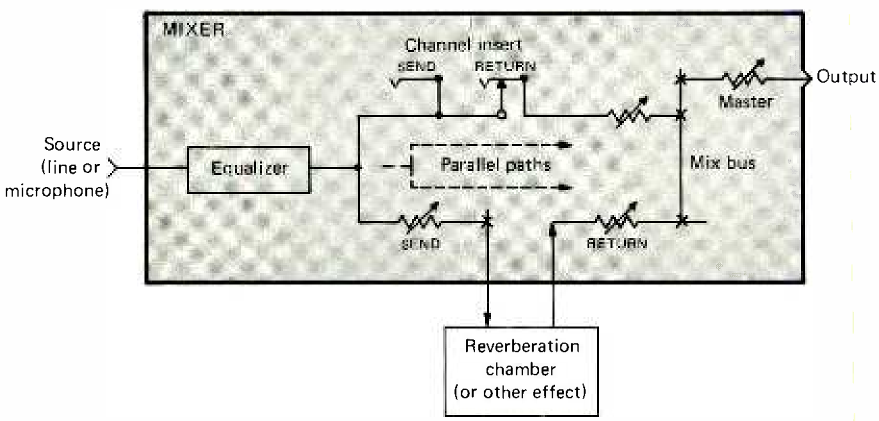


Figure 2.

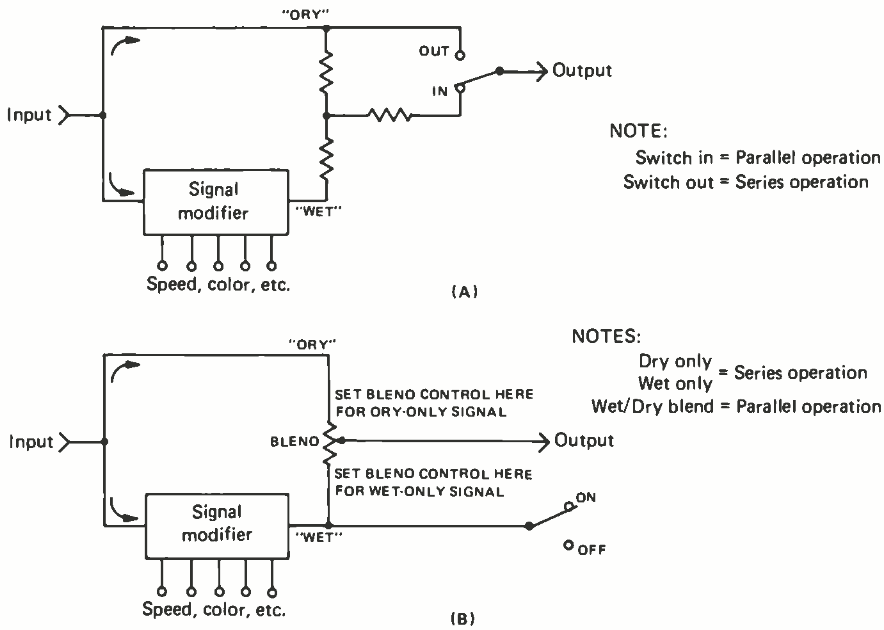


Figure 3.

control might be the effects return knob on the mixing console. For a Type IIIA, it might be just on/off, or there may be an actual control, and for Type IIIB, there is a blend control.

Let's look at what actually happens in a Type II effects box. For those of you who are rusty at reading block diagrams, refer to Larry Blakely's article: "Interfacing Auxiliary Equipment—Where and Why" **MR&M**, April and May, 1979. If not, try reading along and watching the diagrams. It *will* make sense, eventually.

For our Type II effect, let's use an acoustic reverberation chamber. Nice and simple. *(Figure 4A.)* The signal that is to be "reverbed" is applied to a powerful amplifier, which drives a loudspeaker located inside the chamber. The chamber is a moderately sized room with hard plaster or tile walls (kinda like a tile shower). There is, however, one catch: Ideally, for best results, neither the floor nor the walls should be parallel. Finally, a microphone (or microphones) picks up the signal bouncing around inside the room. The microphone's output feeds a preamplifier that finally brings it back to line level. Note that the signal here at the preamp output is pure 100 percent unadulterated reverb.

To use a Type II effect, you must use a mixer. The mixer's job is to combine the dry signal with the wet or reverb signal. Remember that the output of a reverb chamber microphone preamp is pure reverb and used alone would be far too strong an effect for most uses. By mixing the reverb signal back into the dry signal, the reverb effect may be "watered down" to a tasteful level.

The Type IIIA device boils down to nothing more than a glorified Type I or Type II device with a fixed-gain mixer. To connect it, you must break the signal line at the appropriate point and insert the device.

Now, examine the Type IIIB effect. *(Figure 3B.)* Close scrutiny should tell you that this amounts to nothing more than a Type II effect with its own mixer. That's all it is—nothing more, nothing less. Label the mixer knob that has the wet signal on it "blend" or "intensity" (or other suitable name). Most of the time, the "blend" control is actually a type of cross-fader (sort of like a reverse panpot). At one extreme of its rotation, the dry signal is heard, while at the other extreme of rotation, only the wet signal is heard. In between rotational extremes, both signals are heard, in a proportion determined by the setting of the knob.

Connecting Into a System

With an understanding of what happens inside the effects box, the actual interface with an instrument amplifier or PA system should be getting clearer.

INSTRUMENT AMPLIFICATION SYSTEMS. Let's use a typical guitar amplifier (like the Fender Twin Reverb) and a standard electric guitar for this example. As previously mentioned, Type I and III boxes are simply put in series with the guitar signal line, at the appropriate point. The appropriate point is

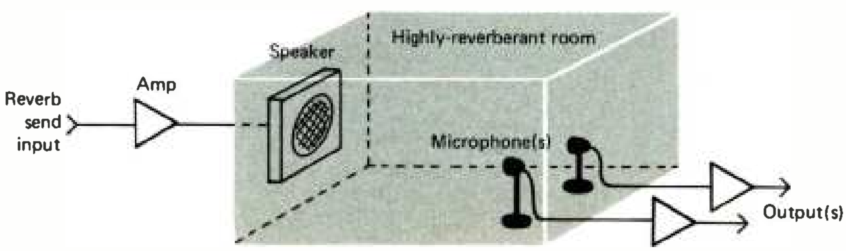


Figure 4.

that point where the signal level matches the signal level requirements of the effects device. More on that subject later. If the guitar amplifier has preamp out/power amplifier in jacks, you can connect line-level Type I and III devices here.

A Type II box must use a mixer of one sort or another. Another problem is that of getting levels to match. (More on that later, also.) If you use an external mixer, the guitar will simultaneously drive one input of the mixer and the input of the Type II effect box. The output of the effect box then feeds another input of the mixer. The mixer output is fed to either input channel of the amp. The wet and dry signal level controls on the mixer are then adjusted for the desired effect.

Another possibility here is to have the guitar drive one input of the amplifier and the input to the effect in parallel. Now the output of the effect drives the other input on the amp. Remember that we are using a Fender Twin Reverb, which has two input channels (normal and vibrato). By doing this, the effect may also have different tonal characteristics for additional sonic variety.

To use a Type IIIB box in this situation, connect it as a Type I series-connected device. (Break the signal line at the appropriate place and insert the device). Set the "blend" control on the device for the desired intensity of the effect. Be sure that you observe the level matching discussed later on in this article.

EFFECTS CONNECTION INTO A MIXER. The mixer specified for this example has a single effects-send bus and one effects return channel. The input channels have both line and microphone input capability, some form of equalization, a single monitor send and channel insert jacks.

Type I and III devices are best connected into the system at the channel insert jacks of the appropriate input channel. They can also be connected between the mixer output and power amplifier inputs, providing the level matching requirement is met.

Type II devices interface directly with the mixer. The effects send drives the effects device input and the device output drives the mixer effects return. As an alternative, an extra input channel may be used, giving the advantage of being able to send the effect to the monitors by simply turning up the monitor send control on the channel driven by the effects device. The effect signal may also be equalized with the channel EQ.

Type IIIB devices may also interface in the same manner as Type II devices. The "blend" control must be set at full effect, or maximum wet signal. Again, the extra input channel may be used, with the same benefits.

THE BLEND CONTROL PROBLEM. When a Type IIIB effects device is used in the console's effects send and return loop, the controls on the console are used to set the wet/dry ratio because the effects send/return system is designed for parallel operation of the effects generators. Thus, the mix or blend controls on the effects unit must be set for maximum effects, allowing *no* dry signal to be returned to the mixer. If any dry signal is present in the effects signal, the loudness of the entire signal will be increased when the effect is used. But if the effects unit just happens to cause a phase reversal, then there will be a point at which the signals from the effects unit will match the main signal and they will cancel! (Half the effects devices available in the world invert phase!) This will not happen if no dry signal is present in the effects loop.

This problem is especially prevalent when attempting to use "guitar pedal" (Type I and IIIA) devices in an effects send/return loop. They can be modified to work, but they really are better off not being used in this situation, or being donated to the Needy Guitar Players Fund. If you really want to use them, the dry signal path within the effect must be defeated (for an example: see "Talkback," *MR&M* June '82, pages 20 and 22). Your service tech will have to do the job for you if you are not electronically skilled. If you use them unmodified, don't expect optimum results.

Matching

Next to connecting up effects devices, matching has got to be the second most-misunderstood aspect of a modern audio system. Part of this comes from the old days of tube amplifiers (yes, I know that some of you still use these antiques). Most tube amps wanted to see (have connected) a load (like a loudspeaker) of a specific impedance in order to work properly. Everyone pounded on you to always "*Match the Impedance.*" In most modern audio systems today, impedance matching has gone by the wayside. What is important, is *level* matching. (Time to climb onto the soapbox.)

SIGNAL LEVELS. Signals exist at many levels or strengths. In audio, weak sihnals are generally referred to as being at mic level. Not-so-weak signals are considered to be line level, and strong signals are called speaker level. To give you an idea of what the actual levels are: mic levels are generally considered to range from around -60 dBm to -20 dBm (roughly 0.001V to 0.1V), line levels range from approx. -20 dBm to +30 dBm (0.1V to 24.5V) and speaker levels from +30 dBm or 1 watt and up. Bear in mind that these numbers are approximations and it is possible for a microphone to put out line level signals.

In the strictest sense, a line-level signal is usually one that is strong enough (around +4 dBm or 1.23V) to drive a basic power amplifier (like a BGW or Crown) to full output. Most synthesizers and other musical electronic instruments do not fall into this strict category. For the purposes of this article they will be considered as "instrument level" devices. In reality, their output level would be classified as "strong mic level." Pedal-type effects that use a 9V power source fall into this classification.

Both inputs and outputs have specific maximum and minimum signal operating levels, and outputs usually have a specified minimum impedance into which they will work. The minimum signal operating level, for outputs as well as inputs, is related to one thing: noise; because all electrical circuits generate some unchanging amount of noise when they operate.

When audio circuits are forced to handle signals that are too weak, their inherent noise becomes evident and you hear hiss. This hiss is at a level usually referred to as the noise floor of the circuit. When signals are kept well above the noise floor the hiss is masked by the desired signal, and the circuit is operating well within its normal range of signal levels. When you try to exceed the maximum level, distortion results.

What does all this mean? Simple. If you attempt to use a pedal type, 9V battery-operated effect at line level, expect distortion. If you attempt to use an effect designed to operate directly from a guitar at line levels, you can also expect distortion. If you attempt to use a line level device at instrument level, expect noise.

LEVEL MATCHING (OR, WHY WE DO BOTHER). It is far more important to worry about level matching than impedance matching. Most modern sound equipment has the impedance problem solved ahead of time (there isn't one to begin with). All that level matching amounts to is trying to match the signal requirements of the source to the input that it is feeding. This means that a -10 dB instrument level source won't drive a typical power amplifier to anywhere near full output. On the other hand, it also means that you can't plug a device with a +4 dBm output into a -50 dBm mixer input and expect it to work well. It can also mean that if you try to use a line-level equalizer (+4 dBm level) at instrument level (-20 dBm), you can expect it to be noisy.

Musical instruments with no amplification built into them can generally be treated like high-impedance microphones. This means that they will require some sort of preamplification before driving a power amplifier. This should also be done before operating line-level effects devices, as most of these are designed to be operated at true line level (+4 dBm or 1.23 volts). If you don't preamplify the signal first, the effects device will probably appear noisy. Many newer effects devices are designed with this problem in mind. In any event, it pays to read the manual or at least check the spec sheet to make sure that you are operating the device at the correct level before you accuse it of being "noisy."

Effects devices using 9-volt batteries or power converters for operation should be considered instrument level or high impedance microphone level. This means that they will not work at true line level without overload.

IMPEDANCE MATCHING (OR WHY WE DON'T BOTHER). There are only two instances where impedances must be matched in the absolute sense of the word. One is where you want to transfer maximum power (remember that power is measured in watts) to a load. The other instance occurs when a passive device such as an equalizer or filter is used and depends upon "seeing" its characteristic impedance (the one it was designed for) at its input and/or output terminals. A high-level crossover network is a good example of this type of matching. If the crossover is used in a situation, like an 8-ohm network with 16-ohm transducers, the characteristics of the crossover will not be what was originally intended.

In a sound system, the loudspeaker and the wall socket (120 volts AC) are the only places where electrical power is important. With modern transistor amplifiers, the impedance is not matched. For that matter, neither is the connection to the wall socket.

As a general rule, if the input impedance of a device is greater than the output impedance of the preceding device, the two may be interconnected. But how *much* greater is a matter of some debate. As a rule of thumb, I would say anything greater than 5 to 10 times the actual source impedance. The actual source impedance is the impedance looking into the output terminals. It is *not* the same as the load impedance. The load impedance, it specified, is a minimum specification. In other words, don't go below it. An active equalizer with a 60 ohm output impedance will operate properly with a power amplifier having a 25,000 ohm input impedance.

It is *not* permissible to interconnect two devices if the input impedance of the receiving device is lower than the minimum load impedance of the sending device. A guitar pickup having a 50k ohm output impedance should not be connected to an input with a 10k ohm input impedance.

Do not terminate a device with a load resistance equal to its actual output impedance unless it is specifically required for proper operation. Do not confuse *actual* output impedance with *minimum load* impedance.

Passive (passive means no battery or AC line power required for operation) filters, crossovers or equalizers must be operated with their input and output impedances matched for proper operation.

The subject of level and impedance matching is complex, to say the least. While the preceding discussion is somewhat sketchy in places, the basic rules are there. A more complete discussion is a subject for a future column.

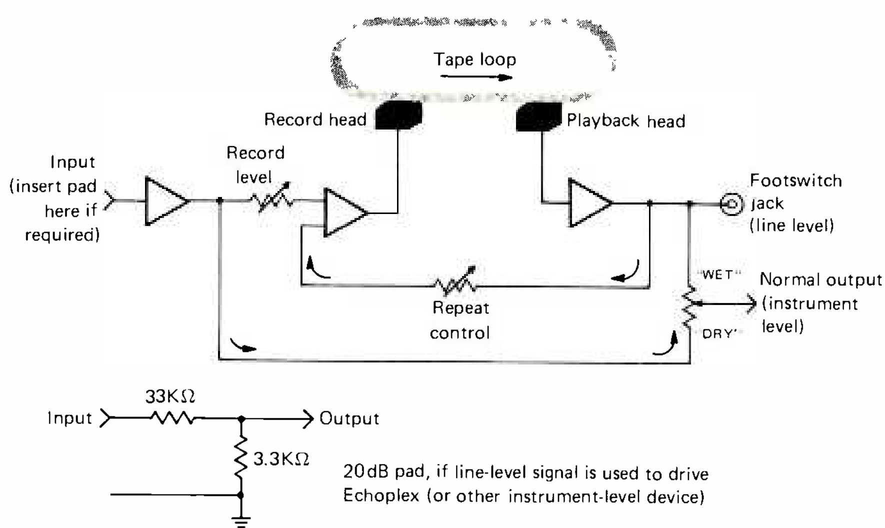


Figure 5.

Putting It All Together

With all this new knowledge under your belt, I'm going to diagram a couple of extremely unusual patches. One involves the Echoplex, an ancient tape echo unit, the other involves psychoacoustic enhancement.

THE ECHOPLEX. This is very definitely a Type IV instrument level device. If you attempt to connect it directly into a mixer's effects send/return system, distortion and/or insufficient return level is the usual result.

Refer to the diagram. (*Figure 5.*) The first problem is level matching to the input circuit. If your mixer has line-level *and* instrument-level effects-send jacks, (the TAPCO mixers do), use the instrument (or LO level) send. If not, you will need to make an external 20-dB pad. If you are clever, you can build it into the phone plug. If you aren't, you can use a minibox.

The second problem is the output. The Echoplex has a passive mixer for its output. I would really like to see a tube guitar amplifier as the next thing in the chain. Connecting a typical solid-state mixer input to the output jack compounds the problem by causing additional loading, lowering the already low output signal.

In order to remotely kill the echo effect, the Echoplex has a footswitch jack. Electrically, the jack is connected to the output of the playback amplifier. Remember that the Echoplex is nothing more than a glorified tape recorder. The signal at the footswitch (echo, not sound-on-sound) jack is line level. Glory be! Just connect the echo footswitch jack to the mixer's effects return or to a channel line input. If you still need a footswitch, use a "wye" cord on the footswitch jack.

Listen to the signal coming from the echo footswitch jack. It should be reasonably undistorted. If not, try adjusting the record level control. The record level control is the screwdriver adjustment near the input jacks. With normal levels coming from your mixer, adjust the record level control for undistorted output from the echo footswitch jack.

THE PSYCHOACOUSTIC ENHANCER. There are several products on the market that claim to stimulate or excite various aspects of our hearing processes beyond simple vibration of our eardrums. To get an idea of the effect, patch a spare 18 dB/octave electronic crossover into the effects send/return loop on your mixer. In this patch, the crossover becomes a Type II parallel effects device. Of course, while this setup is not an exact copy of the circuitry used in any of the commercial devices, audibly it is remarkably similar.

Set the crossover's frequency to 1 kHz or higher and use its high-frequency output. The actual setting will be determined in use. Set all level controls at "normal"; create a mix on the mixer's effects bus of the inputs you wish to "enhance"; try the vocals or drums for starters; be sure that the effects master is turned up sufficiently to drive the crossover at its normal levels. Now, bring up the effects return control on the mixer. Adjust the return control for the desired amount of enhancement. Be subtle. The increase in brightness is the result of the enhancement process. Try various crossover frequencies; the range of 1 kHz to 8 kHz seems to work well.

That's it for this month. If you have a pet subject of general interest, drop me a line care of **MR&M**. In the meantime: "Gimme more guitar in the monitors—and put some echo on it while you're at it."

TABLE 1

Name	Effect	Type	Signal Level
MXR Phase 90	Phase	IIIa	Instrument
MXR Distortion +	Fuzz	I	Instrument
MXR Analog Delay	Delay	IIIa	Instrument + (1)
MXR Digital Delay	Delay	IIIb	Instrument & Line
MXR Stereo Chorus	Chorus	IIIb	Instrument + (1)
Echoplex	Tape Echo	IIIb	Instrument
Roland Space Echo	Tape Echo	IIIb	Instrument (2)
AKG BX10	Reverb	II	Line
Lexicon 224	Reverb	II	Line
Prime Time	Delay (3)	IIIb	Line
DeltaLab DL 1	Delay	II	Line
Acousticomputer	Delay (3)	IIIb	Line
Vox Wah-Wah	Wah	I	Instrument
Yamaha Analog Delay	Delay	IIIb	Instrument
Kepex	Gate	I	Line
Gain Brain	Compressor	I	Line

Type I = Series only devices
Type II = Parallel only devices
Type IIIa = Pedal-type effect with internal parallel structure
Type IIIb = Series/Parallel effect with blend control

NOTES:
(1) Level approaches line level.
(2) Older space echo units were instrument-level only, even at the "H" setting of the level switch.
(3) Complex delay processor.

GROOVE VIEWS

JEFF TAMARKIN
NORMAN WEINSTEIN
GENE KALBACHER
MICHAEL ROBERTS
ROBERT HENSCHEN
JOE KLEE

POPULAR

BRUCE SPRINGSTEEN: *Nebraska.* [Recorded in New Jersey by Mike Batlin on a Teac Tascam Series 144 4-track cassette recorder; mastered at Atlantic Studios by Dennis King. Mastering consultants were Bob Ludwig (Masterdisk) and Steve Marcussen (Precision Lacquer).] Columbia TC 38358.

Performance: **All by himself**
Recording: **Pretty damn good for a homemade job**

To put it simply, Bruce Springsteen's *Nebraska* is a very heavy album. From the minute the needle drops into the title cut, our heart strings are tugged—no, pulled—by the man's ten slices of blue collar frustrations. These one-act plays are created by our storyteller, and listeners feelings are drawn to the surface by little more than that man, his guitar, a harmonica and a random keyboard note or two. The slow, greatly spaced measures of many of these songs can be as draining to a soul as the tales themselves.

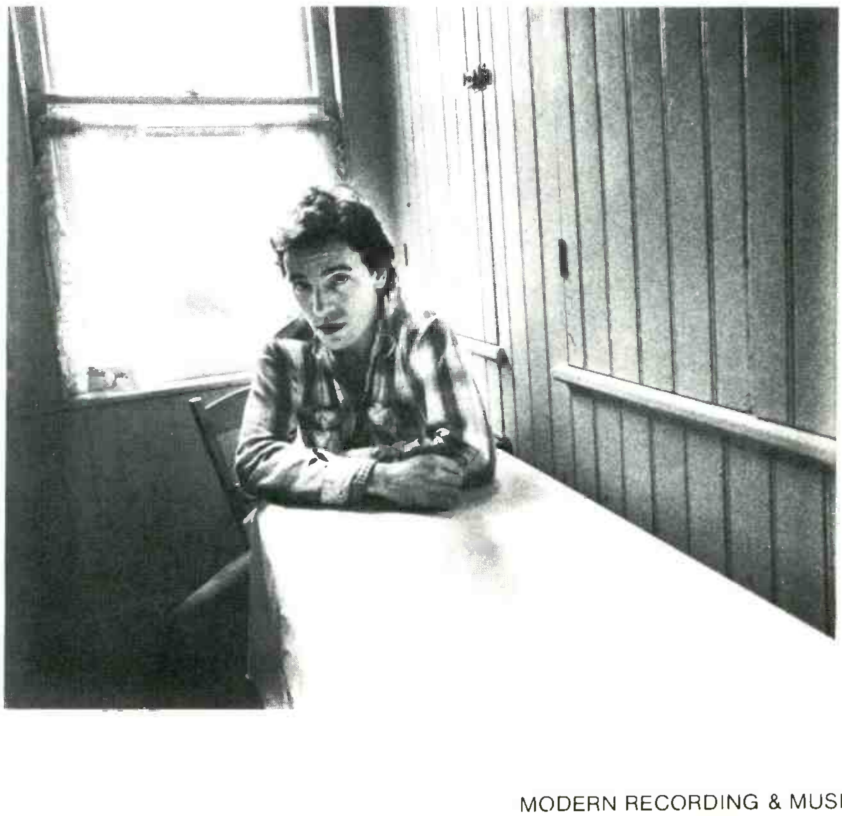
Springsteen produced *Nebraska* at his New Jersey home on a four-track cassette recorder and the resultant "folk" album is truly "the Boss" stripped bare. He's in despair and wants to tell us why; from the shores of Atlantic City straight across this land of ours, we've got troubles in the backlands, and that's just the way things are. Live with it.

Nebraska is basically a lyrically depressing collection. There are numerous accounts of murders, yet the sadness comes from the revelation that the protagonists kill because of feelings that many people seem to be expressing today. This country is filled with hopelessness ("Nebraska"), unemployment ("Johnny 99") and family trouble ("Highway Patrolman"). Even non-death-related stories nurture negative themes like loneliness ("State Trooper") and misguided faith ("Reason to Believe") to their fullest.

Because of the clean mix accomplished on this homegrown collection, one can clearly hear the musical puzzles Springsteen has pieced together. The fact that he throws his gruff voice out in front of the mix is not as important as the other little tricks of vocal and instrumental placement that make these songs work. Something as common as rotating his singing with a rugged guitar strum on "Open All Night" works wonderfully, but his compositional structuring works best on the soft numbers like "My Father's House," where a shadow of guitar, sustained keyboard notes and tambourine jingles gently peak behind Bruce's singing and harmonica playing.

Behind the subtle acoustic guitar picking of "Mansion On the Hill," one can often hear a tasteful harmonica melody lingering to give the string-tapping a little color. And "Atlantic City's" backup vocal is so cleverly laid into the recording that the voice sounds as if it were coming from a bandshell on the other side of town. An effect this simple fills the music with enough character to run alongside the storylines.

Thankfully, the uptempo guitar-only rocker "Open All Night" has been included on *Nebraska*. While it doesn't have the E Street Band behind it, for real full-bodied punch, the song does remind us that there



Jazz, the Fountain of Youth: Al Cohn and Count Basie

By Nat Hentoff

AL COHN: *Overtones.* [Carl E. Jefferson, producer; Edward Trabanco, engineer.] Concord CJ-194.

COUNT BASIE: *Farmers Market Barbecue.* [Norman Granz, producer; Dennis Sands, Greg Orloff, engineers.] Pablo 2310-874.

Jazz, according to the myths of its beginnings, was primarily a young man's game. The Beiderbeckes, the Berigans, the Charlie Parkers—all soared briefly and brilliantly, and then burned themselves out. But for a long time, there has been bountiful evidence of exuberant long-term survivors—jazz elders who stay young in spirit because of the very music that nourishes them. The need to continually improvise; to tell how you feel, how you really feel, through your instrument—this is a way of life that leaves no time for thinking old.

A resounding case in point is tenor saxophonist Al Cohn who has never sounded more personal, more inventive, and more joyful than in his most recent recordings, the newest of which is *Overtones* on Concord. Through the decades, Cohn has always been a dependable swinger, but now his ideas, his tone, and his beat are deeper, fresher, more compelling.

Keeping the legacy going in this set is Al's son, guitarist Joe Cohn; the just about peerless rhythm section consists of bassist George Duvivier, pianist Hank Jones, and drummer Akira Tana.

Although Concord has, from its beginnings, been exceptionally high in its engineering standards, this recording—so bracingly clear and crisp—sets new highs even for this label. It would be illuminating to see a piece by engineer Edward Trabanko on how he prepares for this kind of session.

Nobody has ever dared accuse Count Basie of a fading jazz spirit, but various editions of his bands—particularly in the 60's and 70's—have been charged with being rather mechanical, of lacking the suppleness and resilient energy of bands led by a younger Basie. During the last couple of years, however, Basie's sidemen have taken on more of their leader's evident delight in music-making, and so a collective youthfulness of temperament has returned to the orchestra.

A particularly relaxed and yet marvelously invigorating illustration of this resurgence is *Farmers Market Barbecue* on Pablo. There are passages when the band is so buoyant that it has the feel of a small combo. And there is an imaginativeness in the scoring—which leads to new challenges for the players—that makes even "St. Louis Blues" sound quite surprising.

Furthermore, this Basie band feels the blues more deeply and easily than some of its predecessors in recent years—"Blues for the Barbecue," for example. On all kinds of tunes, there are pungent solos by, among others, tenor saxophonists Kenny Hing and Eric Dixon, trombonist "Booty" Wood, and trumpeter Sonny Cohn. But the most masterful soloist by far is, of course, Mr. Basie, whose every single note is so satisfyingly inevitable.

The recorded sound is so true that I'd like to know how many microphones were used. My guess is the engineers were as judiciously economical as is Basie. This set is a tribute, by the way, to Norman Granz, without whom Basie would not be recording nearly so often. Nor would he be recording, as he does with Granz, whenever he feels the expressive need.

Bananarama

may be better days ahead—for this country and on forthcoming Bruce Springsteen rock 'n' roll albums. Everybody back to Asbury Park; it's too cold in Nebraska! E.Z.G.

BANANARAMA: *Bananarama.* ["He Was Really Sayin' Somethin'" produced by Fun Boy Three and Dave Jordan; "Aie A Mwana" produced by Paul Cook and John Martin; no engineer listed; all tunes remixed by John Luongo; no recording location given.] London LLD 101.

Performance: **The Shirelles go to Africa**
Recording: **Alive and well**

If you read any article about Bananarama, you'll see that they insist that, although they're a group of three women playing a '60s inspired brand of pop, they're not trying to be like the girl groups of the 1960s. That's all well and good, but it's hard to break away from that comparison when one of the two songs on this 12" EP record is a remake of a '60s Motown tune (originally by the Velvelettes, without a significantly altered arrangement.

Giving them the benefit of the doubt, though, the trio—Keren, Siobhan and Sarah (no last names available)—do bring an '80s sensibility to "He Was Really Sayin' Somethin'." Heavy jungle percussion is provided by producers Fun Boy Three, those former Specials who always knew a catchy rhythm when they heard one. A booming bass drum echoes throughout the mix, along with guitars mimicking some kind of animal or another. An extended break in the middle bounds with a Tarzan ambience, to the point that one expects to see someone come swinging on a rope across the living room.

The Banana bunch, meanwhile, do their part admirably enough, although all that's really required of them is to sing in tune and in harmony and to "bop-bop-shoo-be-doo-wah" as cutely as they can without arousing suspicions that they may be the Go-Go's under a *nom de plume*.

A quick flip to the other side ("Aie A Mwana") will erase those thoughts within a few seconds. The beat is just a bit too pronounced to be the L.A. girls, with an accent toward ethnicity that you just aren't gonna find on Hollywood Blvd. Producers Cook and Martin again add a south-of-the-(European) border touch (the Bananaramas apparently don't play their own instruments), and after the extremely hooky chorus that could've come from any Shirelles record, a percussion feast ensues, making one wonder if perhaps Santana wasn't recruited for the sessions.

So, O.K., if they wanna insist that they don't take their cue from their '60s counterparts, fine. But the two sides of this debut single show an enormous potential for development of more than just an image. Bananarama's music is addictive, danceable, hearty and fun. And anyway, people are still listening to records made by the Supremes, Shangri-las, Ronettes, Chiffons, etc., 20 years after they were released. Do you think Bananarama would mind if their records were still popular in 2003? J. T.

RONALD SHANNON JACKSON AND THE DECODING SOCIETY: *Man Dance.* [David Breskin and Ronald Shannon Jackson, producers; Neal Teeman and Akili Walker, engineers; recorded live at Electric Lady Studios, New York, N.Y.; Ron Saint Germaine, "decoder."] Antilles AN 1008.

Performance: **Polyrhythmic jazz rock**
Recording: **live-in-the-studio crispness and clarity**

Just as drummer-composer Ronald Shannon Jackson braids his hair with nuts, bolts, seashells and subway tokens, so he laces his music with the rudiments of free jazz, funk, rhythm and blues, polkas and even bluegrass. Some may call *Man Dance*, the first major-label LP from Jackson and his quintet, the Decoding Society, an exercise in free jazz. Others may call it punk-funk or avant-funk. It's all of these things and it's none of these things. What it is, certainly, is new music—new music that is stretching the boundaries of jazz.

In Jackson's music, which bears the imprint (and no doubt the imprimatur) of his mentors—Cecil Taylor, Ornette Coleman, Charles Mingus and Albert Ayler—the rhythms determine the melodies. Unlike conventional jazz, in which the rhythm derives from the chord changes or melody, Jackson's music doesn't keep the beat—it *is* the beat (rather beats). The distinction between soloist and accompanist is blurred, if not obliterated, as befits Coleman's harmolodic approach. The Decoding Society (Henry Scott, trumpet/flugelhorn; Zane Massey, tenor/alto/soprano saxophones; Vernon Reid, electric guitar/steel guitar/Roland guitar synthesizer/banjo; Melvin Gibbs, electric bass; Reverend Bruce Johnson, fretless electric bass/electric bass) is aptly named: The music is decoded, as in distilled, refined; and societal, as in public, heterogeneous.

The music on *Man Dance* distills the essence of Jackson's native Texas ("Iola"), the Middle East ("Giraffe") and Africa ("Alice in the Congo"), among other places, real and imagined. Jackson calls his sound heavy swing; his band members call it heavy metal world music, and this listener, in keeping with the name game, dubs it rural and urbane funk. If you are what you eat, or more precisely, if you are what you wear, then Ronald Shannon Jackson is a primitive technocrat or a technocratic primitive. Either way, he's arrived with a joyful noise. G.K.

JAMES BLOOD ULMER: *Black Rock.* [James Blood Ulmer, producer; Bill Scheniman, engineer; recorded at the Power Station, New York, N.Y.] Columbia ARC 38285.

James Blood Ulmer

Performance: **Volcanic**
Recording: **Dynamic**

A recent issue of an esteemed jazz magazine featured a blindfold test involving Joe Pass. The editors blindfolded the popular jazz guitarist and then exposed his critical ears to releases by half a dozen jazz guitarists. Pass's comments were transcribed and published sans editing. Now Mr. Pass had something nice to say about *nearly* every guitarist he heard. About James Blood Ulmer's first release on Columbia, Mr. Pass lost his studied cool: "My son is 13 years old and he plays the guitar; he has two amplifiers, and he doesn't know one thing that he's doing, andhe can do a lot of that stuff."

My intention is not to mock Pass's inability to hear the structure and depth of Ulmer's playing. I cite Pass's response to indicate just how *revolutionary* Ulmer's playing is. How a fine (though musically conservative) older player can totally miss the point of Ulmer's achievement. And yet—I am surprised by Pass's lack of historical perception here—Ulmer's playing finds its roots in the earliest jazz (as well as rock and blues) guitarists.

A very quick history of two of the major jazz innovators will help to create a context for comprehending Ulmer's experimentation. To Charlie Christian jazz owes an enormous debt. He created a style of guitar playing ideally suited for bebop. Rhythmically charged riffs. Stunning single note runs. Complex chord transitions. Wes Montgomery took Christian's inventions and carried them a bit further. Single note runs led (imperceptibly) to surprising (yet lyrical) chord and octave passages. All within the tonal framework of bittersweet blues during his most creative (and least commercial) period of recording.

Ulmer's relation to this Christian/Montgomery tradition is both respectful and revisionist. The rhythmically charged riffs invented by Christian are complexified into wild *poly*rhythms filled with unexpected starts and stops. The single note runs go haywire. Chords and octaves are sounded—then shattered beyond recognition. The best analogy for what Ulmer does to the Christian/Montgomery tradition can be found in the visual arts. Compare a landscape of Constable with one of Picasso during his cubist period. (Funny how uninformed museum goers STILL look at Picasso and say to themselves, "My kid can paint better than that!")

Black Rock, Ulmer's second release on Columbia, is probably the most accessible Ulmer record to date. It is surely the recording that most successfully bridges the gap between avant-garde rock guitar (a la Hendrix) and free jazz. Ulmer mixes

The Psychedelic Furs

several vocals into his heady program of instrumentals. His basic trio is augmented by a rhythm guitarist and tenor sax player. But the trio sound—propelled by Grant Calvin Weston's flawlessly timed free-form-funk drumming and Amin Ali's spitfire bass—is dominant. "Open House," the album's rousing opener, is typical of the delights the intrepid listener will discover. Ulmer plays a stinging shower of riffs and runs that are unexpected and emotionally vibrant. Weston and Ali provide a backdrop of jerkily asymmetrical booms and bams. The music is wonderfully sophisticated—yet appealing to the simple minded among us who want to boogie. "Moonbeam" and "We Bop" feature more of the same sound.

"Family Affair" and "Love Has Two Faces" have Ulmer's truly excretable vocals. I suppose that he thinks that his gruff and harsh vocalizing blend with his acerbic guitar stylings. If so, I don't share his conclusion. The songs are only saved by his inventive playing and some perfectly lovely singing by Irene Datcher, his wife. Vocals are definitely the Achilles heel of this release. The "rap" given by bassist Ali on "Fun House" is trite and as dated as old psychedelic poster art.

But these are small flaws on a *major* work. No one involved with the future of guitar playing at the cutting-edge can afford to be without *Black Rock*.

Columbia has lavished upon Ulmer a sound every bit as soaring as his radical solos. *Black Rock* was mastered on the CBS DisComputer System, so there's plenty of bombastic sonics abounding. Lots of sheen and satin and high-tech gloss. Every nuance of Ulmer's guitar is beautifully captured, as is every crash of Weston's cymbals. A better production would be hard to conceive.

And I have a message for Joe Pass. If your kid can *really* play as well as Blood Ulmer, ask him to mail me a cassette, care of this magazine. I'll make him a star. N.W.

PSYCHEDELIC FURS: *Forever Now.* [Todd Rundgren, producer; Todd Rundgren and Chris Andersen, engineers; recorded at Utopia Sound Studios, Bearsville, N.Y., 1982.] Columbia ARC 38261.

Performance: **'Out-weirding the weirdos'**
Recording: **Fast food sizzle**

Quite honestly, I'm not overly impressed by this, the third album from England's Psychedelic Furs. I might go so far as to say that I was not impressed at all, but that would not be altogether true. Let's leave it somewhere around highly unimpressed. On the surface, this Columbia recording would appear to have a lot in its favor, having been produced and co-engineered by the Todd Rundgren of minor "Utopian" fame. And there definitely are some good aspects of *Forever Now*. They just seem to be overwhelmed by the preponderance of the bad.

This album has left me confused. Richard Butler, lead singer of the Furs, said I would be... He was right. My confusion stems mostly from why this self-fulfilling prophecy is of such importance, as opposed to delivering a notable finished product.

The group has been together twelve years, all told. *Forever Now*, as I mentioned, is their third album. It promised a more mature Furs, attaining untouched regions of commercial endeavor. (Untouched by who, and where?) Note: "Love My Way" is the group's biggest single thus far. But by the Furs' own admission it is a "peverse cabaret crooner." It may also speak profoundly of a group whose sole purpose is to "out-weird the weirdos." The refrain in "Love My Way" goes like this: "In Love my way, It's a new road. I follow where, My mind goes." That may be just nifty-keen for these boys, but we are the ones incited to buy their record and thereby solve the mystery of where their minds are going. (That can be an expensive voyage into Esoterica.) As Butler puts it, "I think the best thing you can do is prompt people to think for themselves and work things out in their own way. The only thing that musicians do is open people's eyes up a bit—that's what I've enjoyed about rock music."

The Psychedelic Furs have been noted for their bizarre stage performance. That can get you a lot of attention in pubs and bars and—undoubtedly—a reputation, but it doesn't make you good. Unless you are able to translate your talent to the disc, you can't sell enough records to keep your employers happy. The thing that allows the bizarre groups to attain and/or maintain success has been the ability to appeal commercially on record. Look at the Rolling Stones, Alice Cooper, and to a lesser extent maybe, Ozzy Osbourne and Elvis Costello, who have an appeal to a broader audience than just those who have seen them perform. The reason is, they sound good on record.

Butler's voice is so grating and unpleasant that he must already be using the flip side of a set of second hand vocal cords. The passages of quality instrumental riffs are too few, especially in light of the distracting "quality" of the vocals on *Forever Now*.

There seems to be a greal deal of pride, on the part of the band at the addition of brass and string accompaniment. The purpose (so they say) was to broaden the scope and aural effect. What actually happens resembles a frequency traffic jam much closer to the sound of "french-frying." There is, however, consistency in the drums and some rather creative production, including heavy use of synthesizer runs.

Should there be a ray of hope springing hence, it may be in an awakening of attitudes of the Furs that rebellious, sarcastic and odious attitudes in music must be tempered and directed toward some semblance of commercial, if not social, appeal to be of redeeming value. After twelve years and three albums, they are where they are now. Will Columbia keep them around the necessary length of time to let them fulfill their maturing process? Thankfully, it won't be my decision. M. R.

B. B. KING: *Live In London.* [Steve Goldman, producer; Galen Senogles, engineer; recorded at Royal Festival Hall, London; Doug Hopkins, Tim Summerhayes, Rob Taylor, remote engineers; Bruce Leek, mastering engineer.] Crusaders CRP-16013.

Performance: **B. B. is still the King**
Recording: **Unbelievable for a live recording**

B. B. King's recorded output is just so huge that one would be hard-pressed to know where to begin analyzing the individual albums this great blues guitarist has made. A large percentage of that output is devoted to live albums, not the least of which is King's *Live At The Regal*, a set recorded in Chicago over 20 years ago that has had a tremendous influence on both blues and rock players over the years.

Surprisingly, perhaps, given the passage of time, the warmth and gusto that characterized *Regal* surfaces again on *Live In London*, recorded in September of 1981 at the Royal Festival Hall. Not only is King the musician/singer still in tip-top form, but his band is scorching as well. The horns jump from the grooves in unison, and the rhythm section keeps time like its life depends on it. This is one vibrant performance.

But when it comes right down to it, this is Riley B. King's show. Even though he's sung some of this material every night for three decades, he puts every ounce of talent and feeling into his delivery. King is primarily praised as a guitarist, but even if he never played a note on this LP, it would still have just as much life. He's soft and pleasing on the slow blues ("Night Life") and he belts 'em out on the uptempo ones ("Love The Life I'm Living") as if he was auditioning for his first gig.

But, of course, this would hardly be a B. B. King album without the trademark fluid fretwork that made the master "the King of the Blues." And does he ever show off here! The encore, a blues jam with the Crusaders, shows King to be swifter and more imaginative than most of the upstarts who've stolen the spotlight from him; he inspires his players... who return the favor...and what emerges is a blues bash that is consistently marvelous from top to bottom. King runs through the gamut of his styles at this London gig, and by its conclusion, one is awe-struck at the continued vitality of this legend.

Equal time must be given, however, to the audiophile pressing that houses the performance. Using the best virgin vinyl, digital mastering, etc., producer Steve Goldman and engineer Galen Senogles have given King a product worthy of his talents. The sound quality is simply stunning—noiseless, sharp, and clear as a bell—so that every nuance played by King, his drummer, his keyboardist, the brilliant horn section, etc., is heard for what it is. This just goes to show that the right combination of music and technology can produce a magnificent listening experience even if the songs and the artist are as familiar to us as an old friend's voice. J. T.

JANIS SIEGEL: *Experiment In White.* [Joel Dorn, producer; Gene Paul, engineer; recorded at Atlantic Studios and The Hit Factory, New York.] Atlantic 80007-1.

Performance: **More trial than error**
Recording: **Lab spotless**

There is more than one side to Janis Siegel, and you get at least two of them on *Experiment In White*. Side one is in a pop chanteuse pose, covering Leon Russell remakes and contrived-but-rollicking gospel remakes. Side two is contrastingly adventuresome and jazz-oriented. Make that an experiment in black & white.

As the first of, hopefully, more solo

Janis Siegel

projects by Manhattan Transfer members, Siegel's album fulfills a certain obligation to hip esoterica while also making some obvious attempts at creating broader waves on the radio front. Leon Russell's "Back To The Islands" opens the album with a rendition that starts off great in a kind of bluesy Bonnie Raitt style (with Dr. John and The Meters in support). Unfortunately, doubling up her vocals and soaking this treatment in horns and strings only serves to average out any impact. Another remake, "All The Love In The World," is fair to mellow, helped by David "Fathead" Newman's biting tenor sax solo and more backing by The Meters.

But "Lovin' Eyes," another all too familiar and too recent recycling, lacks any kind of distinguishing marks on the singer's part. Even the side-ending "Hammer And Nails," done in a gutsy gospel spirit, fails to bring out anything new from Siegel—this is perfect material for her, but she has regularly covered similar ground with the Transfer.

Up to this point, you've got big names like Joel Dorn, Ralph MacDonald, William Eaton, and the legendary Les Paul involved on either the production or arranging end—and little to show for it. But side two changes all that, even with such a dubious retread as "How High The Moon," which opens the side. Lit by just simple accompaniment from Les Paul's famous guitar (and a bit of his brushwork too), Siegel begins to shine, multi-tracking her vocal parts and achieving a fast-paced, shimmering effect.

Next comes "Don't Get Scared" by Stan Getz, with lyrics by Jon Hendricks (of Lambert, Hendricks & Ross fame) and things get interesting. This one bops along recklessly, with Janis singing the original Getz solo, Hendricks singing another instrumental part, and musicians including Kenny Barron, Ron Carter, and Frank Foster all contributing greatly. The lyrics are a scream, Hendricks is in rare form, and the excitement level picks up dramatically atop Slide Hampton's chart. Speaking of drama, Siegel follows with a fairly straight job of that classic torcher "Guess Who I Saw Today?," accompanied by the classy trio of Tommy Flanagan, Ron Carter, and Grady Tate.

"To Be With You" is a nice, slow Latin reprieve, with a good flute solo by Pacquito D'Rivera (of Irakere fame) and an interesting bed of overdubbed background chorus. But "Jackie," a relatively obscure song penned by the late Hampton Hawes, caps this LP off with its best jazz cut. This zany monologue is a great slice of life from the bebop era, and proves to be a splendid vehicle for Siegel at her best.

While the jazz side of Janis Siegel here outweighs the rather meager attempts at commerciality, there may still be a stronger pop side to this fine singer that we're not hearing here. The problem is more with the use of uninteresting material. Given an oven-fresh, original rock tune, there's little doubt that Janis could give it more flavor. R.H.

LUTHER VANDROSS: *Forever, For Always, For Love.* [Vandross, producer; Michael Brauer, engineer; recorded at Mediasound in New York, Village Recorder and Record Plant in Los Angeles.] Epic FE 38235.

Performance: **Gourmet vocals**
Recording: **Hot buttered soul**

In the space of one year and two albums, Luther Vandross has risen through the ranks to establish himself as composer, producer, performer extraordinaire. As a member of "the ranks," Vandross found himself to be a hotly pursued backup singer, studionik, and even a featured voice for 7-Up, Burger King, the U.S. Army, Juicy Fruit, and Kentucky Fried Chicken.

But if it's finger-lickin' good soul music you seek, just let this man loose in the kitchen. He's got the kind of voice that sautes a song just nicely, and turns even average faire into a gourmet feast for the ears. The title cut to this LP is a perfect example—it's the same old filet of soul but when cradled in a simmering butter of a voice, it makes your neck hairs stand on end.

Vandross grew up in New York with three R&B divas as his greatest childhood idols—Aretha, Diana, and Dionne. While Luther's velvet touch obviously owes much to his studious observation of such ladies, this record also demonstrates his acute awareness of great male vocalists as well, Sam Cooke ("Bad Boy/Having A Party") and David Ruffin of the early Temptations ("Since I Lost My Baby"). Given time, and more recorded successes like this one, Vandross will be joining this same pantheon.

While the softer textures of the man's remarkable voice cast a golden spell over excellent ballads such as "Promise Me" and "Once You Know How," you can't typecast Vandross as a soul torcher. By far the hottest and most infectious track on this disc is "She Loves Me Back," a fast pop-soul original that is guaranteed to get anyone up and off their duff. Credit a

Luther Vandross

strong support crew for some of the fire, especially arrangers Leon Pendarvis and Nat Adderly, Jr., plus regular supporters Marcus Miller (bass), Yogi Horton (drums), Doc Powell (guitar), Adderly on keyboards, and some of the best percussionists to be had: Paulinho da Costa, Sammy Figueroa, and Ralph MacDonald.

This entire production unit has the kind of smoldering cohesiveness that, when they're really on, is reminiscent of the Hi Records rhythm section that Al Green used to hang his hat on. Attention to detail and professionalism creates a launch pad for fine recordings, which this one is without question. As for *great* recordings—which this one is more often than not—only an inspired singer like Luther Vandross has the power to make those happen. R.H.

JAZZ

BOB WILBER AND THE BECHET LEGACY: *On The Road.* [Joanne Horton & A. J. Sordoni, producers; recording engineer not listed; recorded November 24, 1981 at Strawberry Studios, Stockport, England.] Bodeswell BW 105.

Performance: **Part of the legacy**
Recording: **Clean, comfortable and unspectacular**

They sure found the right name for this band. The dictionary definition of a legacy as anything handed down from an ancestor to a descendant certainly applies here. Bob Wilber served his apprenticeship at Bechet's side, (see the photo on the back of the album), and players like Glenn Zottola have been immersed in the heritage of Red Allen, Louis Armstrong and many others who were out of their time frame. This is a function that Bob Wilber serves better than any musician of my acquaintance. Who but Bob Wilber could take a bop oriented pianist suffering from nearly total disuse of the left hand, as Mark Shane was when he joined the legacy, and in a few short months of tutelage and influence end up with a two handed pianist surely acquainted with, if not quite fully under the spell of, Jelly Roll Morton and Dick Wellstood. We also have Bob Wilber to thank for the expansion of former Count Basie big band drummer, Butch Miles, into a musician who can comfortably fit in with a revival band playing dixieland. I know that Bob Wilber hates those words "revival" and "Dixieland" and yet I find it hard to think of two words more fit to describe a band that goes into the 1980s playing tunes like "Polka Dot Stomp" and "Santa Claus Blues."

Vocalist Pug Horton (it says Joanne on the record but I know that's a psuedonym for Pug Horton Wilber,) has chosen her material more carefully than in the past and it shows. Both "Ghost Of The Blues" and

Bob Wilber and the Bechet Legacy

"Santa Claus Blues" are admirably suited to her highly personal style. Also I'm pleased that she didn't attempt the same breakneck tempo on "Santa Claus" that Eva Taylor did for the recording with Clarence Williams's Blue Five.

This band plays the music of Sidney Bechet—the music he wrote and the music of others which he played. Bechet was not a purist. Not everything he played came out of the New Orleans repertoire. He was not above playing show tunes like "Lady Be Good," "Summertime" and "Love For Sale." Whether I like these tunes played by a traditional jazz band or not (P.S., I especially don't like "Lady Be Good" played by a pre-Benny Goodman Trio type organization) doesn't matter. Bechet did. They were and are part of his legacy, and as such they belong in this band's repertoire—like it or not.

I do take issue with the label credit on "Egyptian Fantasy." I know that Bechet and J.D. Reid did take credit for this tune on the famous Victor recording by the New Orleans Feetwarmers, but unless I'm mistaken "Egyptian Fantasy" was played in vaudeville by the Original Creole Band which would indicate authorship to some combination of Bill Johnson, Freddie Keppard, George Bacquet and/or Jimmie Palao. Still more probable is the story given to me by Bob Wilber which came from Martin Williams of the Smithsonian Institution. "Egyptian Fantasy" is actually a close paraphrase on a tune called "Egyptia" which the Creole band usurped and adapted for their own purposes. Bechet played it wonderfully in 1941 with his feetwarmers—Wilber plays it excellently forty years later with the legacy.

If there is cause to complain, it's minimal. Mike Peters, for all his fine rhythm work on guitar and banjo, as a soloist has listened more closely and frequently to modernists than to Teddy Bunn and others of the generation of Bechet—and it shows in his playing. Wilber also has settled into a very comfortable groove. He doesn't skate out on the thin ice just to see how far he can go the way he used to—but then he doesn't have Eddie Hubble and Kenny Davern hurling challenges at him anymore. If Bob has gotten milder and mellower he's gotten more professional too. I guess seasoned professionals don't feel the need to take the chances they did when they were feeling their way. It really doesn't matter. Whenever Bob Wilber feels moved to play it will be worthwhile listening regardless of whether it's safely near the shore or out in deeper waters.

If your local record store doesn't stock Bodeswell records—shame on them. You'll just have to mail order this wonderful LP from Bodeswell Record Distribution, Box 624, Brewster, Mass. 02631. J.K.

PETR KOTIK: *Many Many Women.* The S.E.M. Ensemble. [Heiner Stadler, producer; Allan Tucker, engineer; recorded at Vanguard Recording Studio, New York City, N.Y. March 1980.] Labor Records LAB-6/10.

Performance: **Austere and enchanting**
Recording: **Uneven**

Many decades ago the American composer Virgil Thomson created musical settings for works by the great literary revolutionary Gertrude Stein. Thomson's music was anything but revolutionary. His orchestration for the Stein opera *Four Saints in Three Acts* was a rambling collage of quotations from various Romantic composers with touches of American folk melodies. RCA has long ago let its recording of snippets from the opera go out of print. Thomson's music for *The Mother of Us All* is still in print and is not much more of a challenge for listeners interested in tracking the history of Thomson/Stein collaborations.

I've been puzzled as to why more serious attempts haven't been made to musically counterpoint Gertrude

Stein's texts. After all, no writer in our time has so successfully experimented with the musicality inherent in our native tongue like Stein. Try speaking aloud her "A rose is a rose is a rose" without an unintentional lilt entering your voice. She possessed a genius for discovering in the everyday syntactic patterns of English the seeds of song.

Petr Kotik understands what Stein is about. He is also a composer of rare inventiveness. His *Many Many Women* is the most original and sweeping marriage of Stein's language with music I could ever imagine. Kotik accomplishes this miraculous marriage of literature and song by juxtaposing some of the oddest and least likely stylistic elements possible.

First, Kotik's work is aleatoric in design. This means that the composer has left to the performers decisions regarding instrumentation, transposition (all vocals harmonize in fifths, fourths and octaves), and entrances. This incorporation of chance elements in the composition might have been inspired by Kotik's study in the 1960's with the most notorious of our chance-oriented composers, John Cage. But—and here's the real shocker for anyone expecting a composition replete with avant garde crash-squeal-screech mannerisms—Kotik's music sounds like Gregorian chant. Mighty odd Gregorian chant, granted. A Gregorian chant filled with eerie moments of dissonance. A chant sung with solemnity and precision by two sopranos, a countertenor, a tenor, a baritone and a bass. The singers are backed by two flutes, two clarinets, and two trombones. The efforts given by the musicians can best be described as promethean.

A promethean effort is required to perform *Many Many Women* for it goes on (at least in this recorded version) for over three hours. Yet there is an uncanny sense of *timelessness* that permeates the music, a quality of sacred time one associates with Gregorian song. A quality, also, that Stein sought to infuse her writing with. Stein's text (written in 1910) which Kotik chose to musically mine is about how certain of our psychological characteristics tend to change and yet seem not to change in time. This music embodies that paradox of seeming always to minutely permutate while remaining essentially the same.

Dora Ohrenstein and Lois Winter particularly shine in their difficult roles as sopranos. The composer joins Susan Sperl on flute and plays with captivating exactness and clarity. It's hard to single out individual musicians for kudos. Everyone performing deserves a rose (and a rose, and a...).

I do have a few complaints about the package. Labor Records has packaged this in an attractive box holding the five record set plus program notes. Problem number one: no libretto (and the Stein book is out of print). Problem number two: three of the records have excessive surface noise. Numerous snaps, crackles, and pops do not contribute to an otherwise otherworldly listening experience. Odd to have a session so cleanly produced and engineered only to have a poor pressing flaw the sound. The sides without excessive surface static sound crisp and full.

These reservations aside, Labor Records deserves considerable credit for recording a project of such quality and magnitude. If you can't locate this set easily, it can be ordered for $25 from Labor Records, P.O. Box 1262, Peter Stuyvesant Station, New York, N.Y. 10009. Consider the purchase of this provocative music a tribute to the mother of us all. N.W.

ERIC ROSS...***Electronics Etudes*** (Opus 18) ***& Songs For Synthesized Soprano*** (Opus 19). [Eric Ross and Dr. J. R. Mitchell, producers; Eric Ross, engineer; recording location and date not listed.] Doria ER-103.

Performance: **Free form electronics**
Engineering: **Part of the performance**

Eric Ross

One the advantages or disadvantages of the synthesizer, I haven't quite decided which it is yet, is that it gives the composer/performer/engineer complete control over his composition/performance/engineering. It is possible for Eric Ross to put out a record of music he composed, played and engineered by himself. Like most modern classical music (and much of modern jazz) it is free form and like much modern classical music and much modern jazz, I find free form is frequently a euphemism for formless.

Eric Ross has set his goals at composition within the special relationships between real, processed and synthetic instruments and sounds. We therefore have combinations including clarinet, bassoon, cymbal, piano, voice, Balinese and Javanese percussion instruments and harpsichord mixed and matched with synthesizers of numerous varieties.

I find the composing most successful in opus 18 particularly the "Etude Quartet for Synthesizer." The "Songs for a Synthesized Soprano" are too detached from words and form to be entirely successful as a piece—an experiment perhaps but not a finished work with an opus number.

The engineering of the record is no longer a question for debate. The composer combines his tapes the way he wishes—overdubbing—echoing—recording directly from the synthesizer onto the tape—whatever means suits him. Therefore, the engineering of the recording becomes part of the composition process. It eliminates the need for a producer or an engineer but it also eliminates another set of ears which can sometimes offer a view or suggestion from a different perspective. By the time the master tape has been composed/performed/engineered by the composer/performer/engineer it's probably too late for any substantial change in the finished recording.

To be sure this is interesting, and at times enjoyable, experimental music. Considering the limited performance facilities and the even more limited audience for free form electronic music, that is very well what it may remain. If someone were to truly integrate the music of the synthesizer with traditional forms of music then there would be an opportunity for electronic music to become part of the concert repertoire. Free form experimentalism like this won't do it.
J.K.

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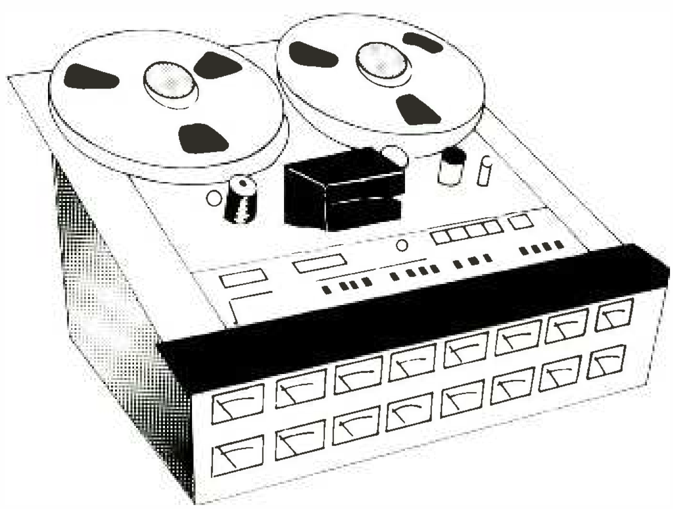

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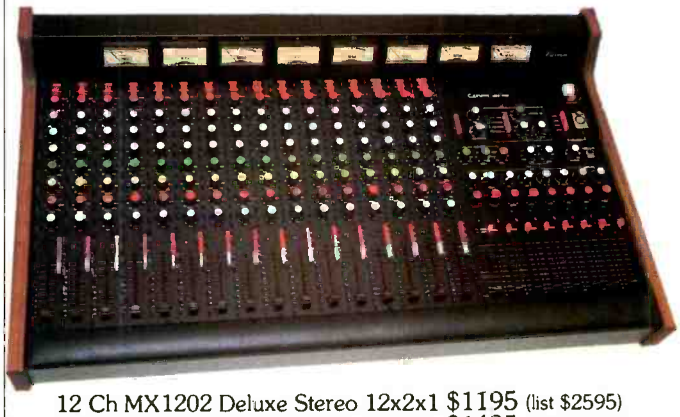

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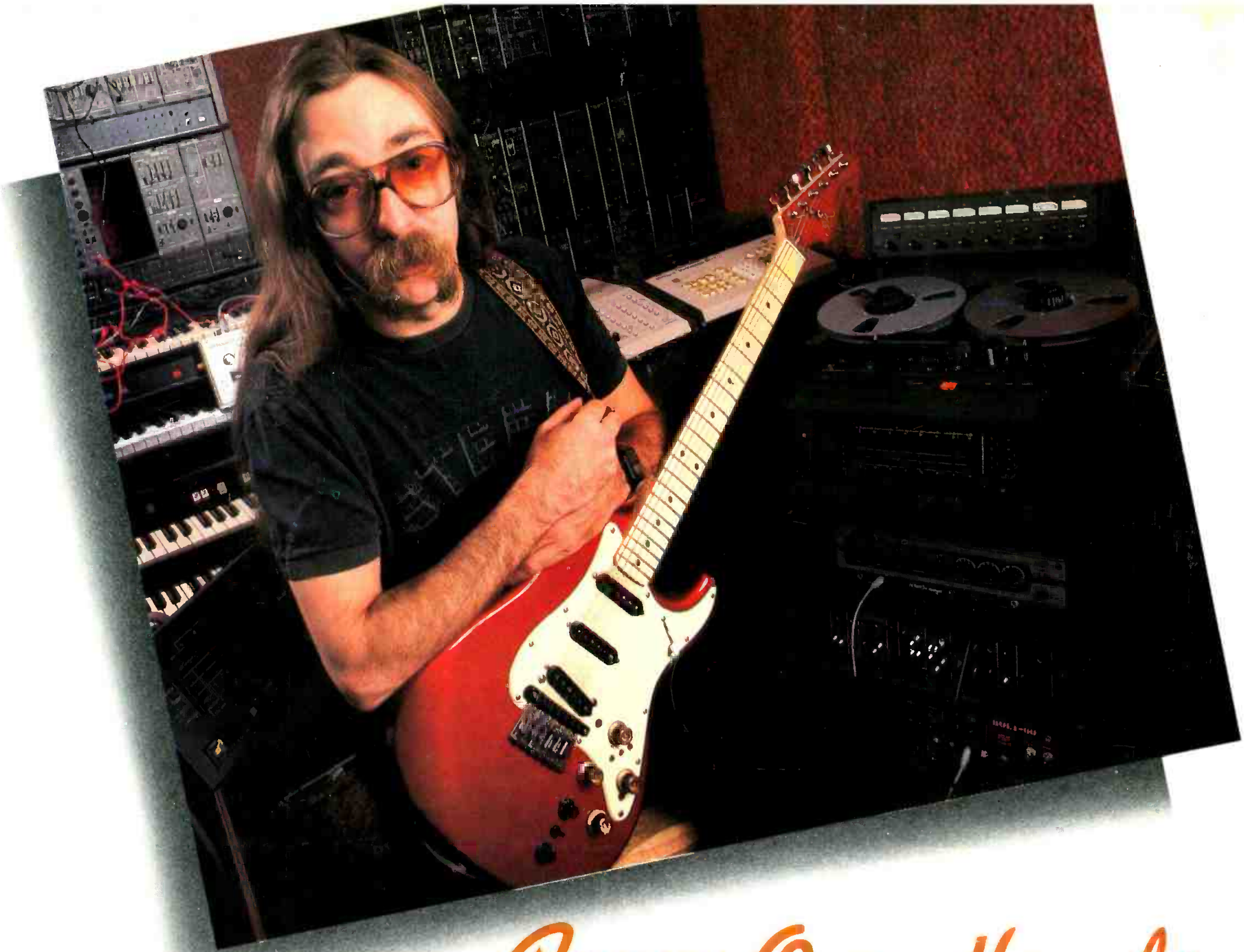

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MK III-8

Before Producer/Artist Jeff Baxter rolls into expensive studio time, he rolls tape on an Otari machine. At Home. In his studio, Casual Sound.

"The Otari saves me a great deal of time and money. A recording studio was never intended to be a $150.00 per hour rehearsal hall, so I work out ideas and refine the tunes before I go into the studio.

All my pre-production recording for the last several years has been on my Otari. That machine has never left my studio, —it's been incredibly reliable.

There's a lot of musical moments that have been captured on that machine ...some of which have been directly transferred to the final multitrack masters... Elliot Randall, Doobie Brothers, on and on. The Steely Dan Pretzel Logic album was mastered on an Otari 2-Track. And, that's obviously a statement in itself... how I feel about the quality of the sound."

Jeff Baxter's always been into instruments that musicians can afford. It's obvious that he's also been heavily involved at the leading-edge of recording technology.

Besides telling you his feelings about Otari tape machines, there's just one other tip Jeff would like to leave you with:

"Try anything and everything and always roll tape."

CIRCLE 12 ON READER SERVICE CARD

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Otari Corporation, 2 Davis Drive, Belmont, CA 94002
Tel: (415) 592-8311 Telex: 910-376-4890