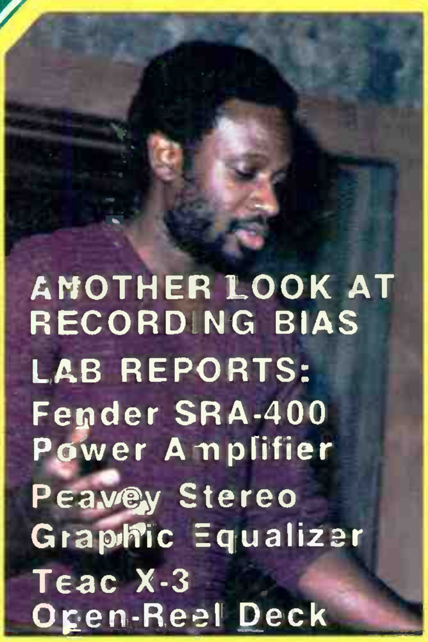
Profile: GRATEFUL DEAD'S MICKEY HART

MODERN RECORDING & MUSIC

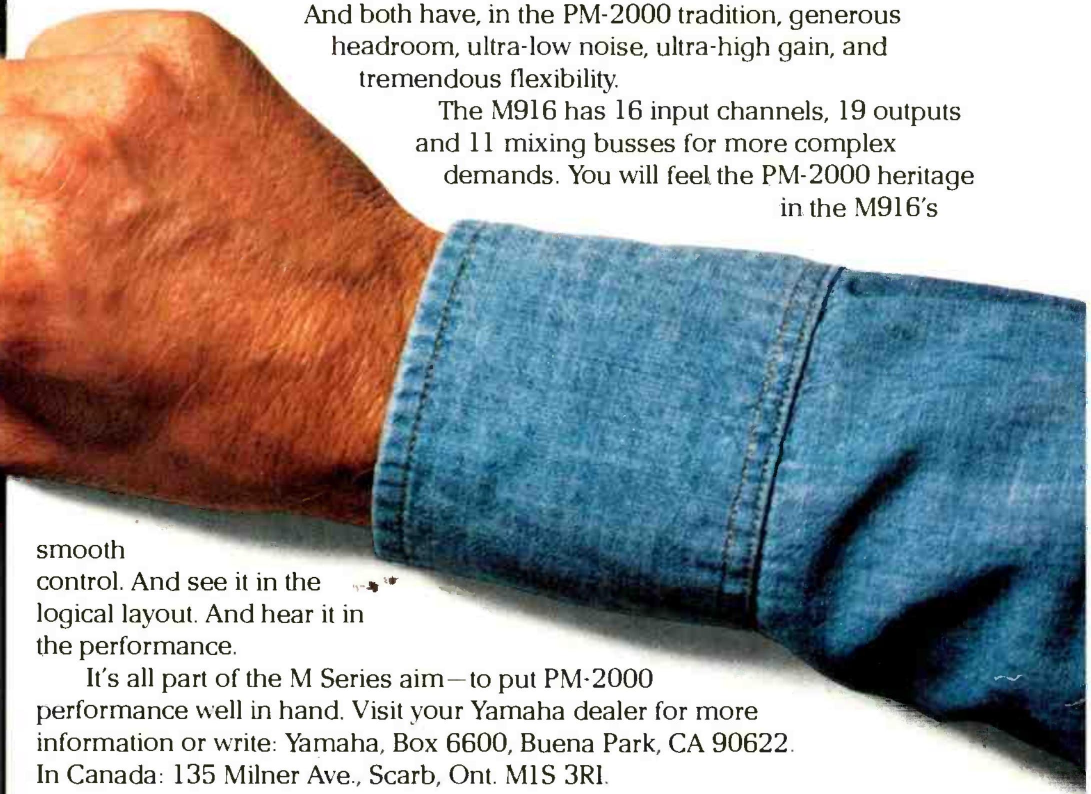
ICD 08560
$1.75

VOL. 6 NO. 3
DECEMBER 1980

a session with Jack Bruce & Friends



Our new M SERIES puts PM-2000 performance within your reach.



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

The M Series makes an excellent choice for live sound reinforcement main or submixers, as well as for theatrical, church, and broadcast uses.

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It's all part of the M Series aim—to put PM-2000 performance well in hand. Visit your Yamaha dealer for more information or write: Yamaha, Box 6600, Buena Park, CA 90622. In Canada: 135 Milner Ave., Scarb, Ont. M1S 3R1.

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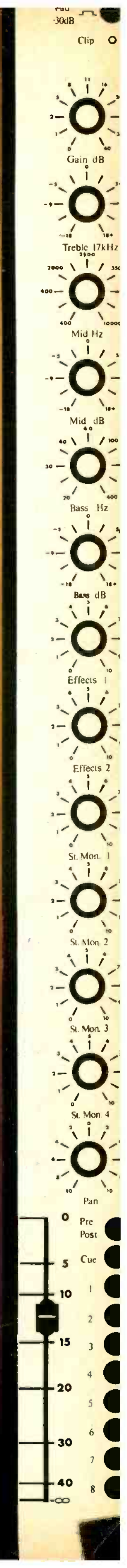

Get the Message?

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

First, get our mainframe. Add to it as many input channels as you need, each one including such features as an overload input light, phantom powering, semi-parametric equalization, 2 effects and four monitor sends, a 100mm fader and many others. Then add as many output channels as you require for your application (up to 8 independent tape, 4 monitors, 2 effects and 2 master outputs), choose a VU meter section to meter the output section you select, and *go for it!*

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We think you will be surprised to find that beside being the most versatile mixer on the market through our *Totally Modular* concept, **Studiomixer** truly offers the *best* performance specifications of any mixer in its class . . . and, at the lowest price. How does $4,750 sound for a 16X8? That's right, 16 mic channels, 3 tape channels, 4 monitor and two effects sends, and 2 master channels. Unbelievable? Not any more!

To get the message, write to us for a free, illustrated brochure explaining our *Totally Modular* concept. We will also provide you with the name of your closest **Studiomixer** dealer. And, if you are as excited about *Totally Modular* as we are, then send us $4, your size and sex, and we'll send you postpaid a brochure *and* a Studiomixer *Totally Modular* T-shirt like we're wearing. Either way, you'll get the message.

For information on our products, please write to National Sales Manager, Amerimex Co., P.O. Box 55, Atwood, California, 92601 . . . or phone 714) 528-3360.

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For All Intensive Purposes

Flanging is a pretty intense effect. And probably the best thing about it is that properly controlled, it can be used in a lot of different musical applications. But if your experience with flanging has only been in mono, you've really only experienced half of the effect.

Presenting the next step, the Roland Stereo Flanger. The Roland Stereo Flanger is an effects device that will produce a myriad of time delay effects from classic flanging to chorus and doubling. But it doesn't just stop there. The addition of the second channel makes possible stereo panning in all modes, which dramatically intensifies the flanging effect, and makes it the most versatile time processor on the market.

The Stereo Flanger has four operating modes—three of which are used in flanging, the other being used for chorus and doubling effects. Each mode is indicated by an LED for constant monitoring. LEDs also indicate sweep rate and overload status. While in each of the time delay modes there is complete flexibility over the sweep width and depth, as well as manual delay control (from .5 to 20 ms.) and full regeneration capabilities.

The Roland Stereo Flanger is just one in a series of very special components in the system we call The Roland Rack. The Roland Rack system includes two instrument Pre-Amps (one for Guitar, one for Bass), two Stereo Power Amps, a Vocoder, Pitch to Voltage Synthesizer, the amazing Dimension D, a Digital Delay, and as we said before—the most versatile Flanger on the market.

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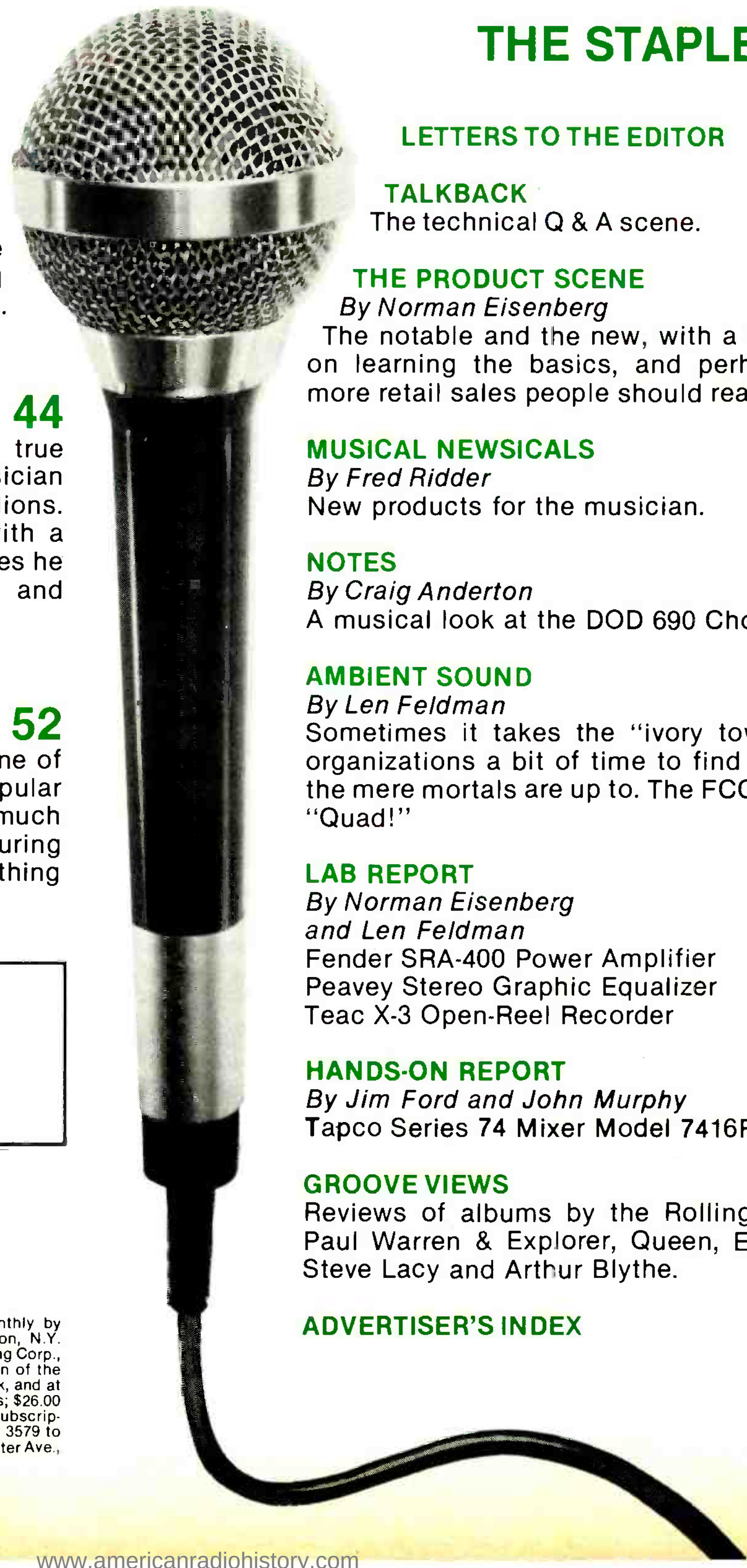
DECEMBER 1980
VOL. 6 NO. 3

MODERN RECORDING & MUSIC



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Bob Seger "Live!"
Plus lots more!

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Letters to the Editor

The Hunt for a Seller of Songs

I'm partially blind and have composed and written over 250 songs and poems for all ages. I do not have the money to get them published and recorded. Living in the small town of Aldrich, Montana, I was inspired to write by people who came to the lake here to fish. I've played my songs in Mississippi, Nebraska, Missouri, Kansas, and a number of other places. I wondered if you could help me out in finding out how to get my songs published and recorded.

—Ann Walker
Aldrich, MO

This question has come up before from our readers, and maybe it's about time we handed down some rough guidelines about getting music published.

The biggest publishing firms are located in New York City, Nashville, and Los Angeles, but there are others in smaller cities. Check the classified pages of your local phone book to find possible local publishers. Also a listing of the big publishing firms appears every September in Billboard's Annual Buyers' Guide. Cash Box *and* Record *are two magazines which contain listings, also.*

The next step is to decide which publisher would be appropriate for your song. Some publishers deal best with only certain types of songs. To find out which firms are most suitable for your music, try reading music trade weeklies.

Publishers will probably not print up sheet music of your songs. This is a practice that was pretty much discontinued about a quarter of a century ago. Now a publisher rarely prints up sheet music until after a record has been made which has been fairly well accepted.

The music publisher really gets the song to the right record company or the manager of an appropriate recording artist. There may then be an album cut, a hit single, and maybe income from the licensing of the performing rights.

In cases where the songwriter is not also the performer, money from licensing the performing rights exceeds record income. The U.S. Copyright Act of 1909 limited the fee that recording companies paid to the owners of copyrighted songs. It was then 2 cents per single sold. The Copyright Law of 1976 revised this to 2¾ cents or ½ cent per minute of playing time, whichever turns out to be larger. Performing rights, though, can bring in much more income than that. Writers and publishers generally split the income from performing rights equally, and usually make the same split for record sales.

A good publisher will try to get more recordings of your songs made by other performers. You should constantly recheck to see what is happening to your song. To present your song, make a clear tape on a good tape recording machine. You must be sure that the music and the lyrics come across clearly. The accompaniment of one instrument, e.g. a guitar or piano, is generally enough. You don't need studio production or an expensive presentation. If your voice is not good enough, get someone else

To sound the best, use the best.

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With the introduction of the new Series 72 and 74 mixing consoles, we have reinforced our position as the leader in the music electronics industry – a position that only comes from being sensitive and responsive to real needs.

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Asheville
Dunham's Music House, Inc.
Gastonia
Howren Music
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Portland Music
Showcase Music & Sound

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Dauphin
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Burlington
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VIRGINIA

Falls Church
Giant Music
Rolls Music Co.
Richmond
Giant Music
Springfield
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WISCONSIN

Eau Clair
UMS Pro Audio
Milwaukee
Uncle Bob's Music
Racine
The Music Store

with a good voice to sing it for you. The publisher may later make a more professional demo tape to send to an artist or record company. Publishers usually prefer reel-to-reel tape, and they want the lyrics typed up, each song on a separate sheet. You can send "lead" sheets of the actual musical notation of your music, but that really is not necessary. Also, don't put more than three or four songs on a tape. Leave 12 to 15 inches of silent tape up front and between songs.

It is very important to write to the professional manager or his assistant at a particular publisher, requesting an appointment. You can find that name for a particular publisher by looking in the music trade weeklies or by telephoning. In your letter, ask for a chance to play the tape, or play the songs live. You might tell them that you'll telephone in 6 to 8 days for an appointment. Persist, and do not be persuaded to drop your tape off at a publisher's to be listened to at his convenience. Some publishers will never get around to it. So try to persuade them to listen on the spot. During those weeks you should try elsewhere. If you choose to copyright before bringing your song to a publisher, you can get some copies of the required form, Form PA, with instructions, free, by sending a postcard to the U.S. Copyright Office at the Library of Congress in Washington, D.C.

You'll at some point have to decide which performing rights organization you want to license your music. The publisher may have a separate firm, one in ASCAP, and one in BMI or SESAC. Phone or write them on your own to discuss matters, e.g., royalties.

The American Guild of Authors and Composers is a good songwriter group. Also, we recommend that you read over your publisher's contract carefully before signing. In reference to an assumption you seem to make in the second sentence of your letter, let me clarify the point that certain publishers ask you to pay to have your songs published. Steer clear of these. Some are just "song-shark" rip-offs. Anyway, ASCAP's policy is not to accept this type of publication as compliance with requirements for joining, so don't bother trying to gain entrance in that way. If you have more detailed questions, you should go to the source from which I drew this guideline: the Contemporary Music Almanac 1980/1981, *published by Schirmer Books. They have a whole section devoted to publishing songs. Also helpful might be the graphic drawn by G. A. Bowley, in our "Letters to the Editor" column, p. 10 in the August 1980 issue of* MR&M. *It shows various connections between songwriter, publisher, performing rights organizations, performers, radio stations, etc. Good luck, however you decide to proceed.*

Fostering Magic

I enjoyed Len Feldman's recent article, the September 1980 issue of *MR&M.* In fact, I have enjoyed many of your articles in the "Ambient Sound" column. After reading many articles in many magazines condemning broadcasters/engineers/program directors, I feel compelled to write that our station is one that uses the contemporary format without the equalization quirks and compression/limiting features that some use. We keep an eye on the Telco

Fender's new 300 Bass Slimline Amplifier.

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The state-of-the-art bass amp

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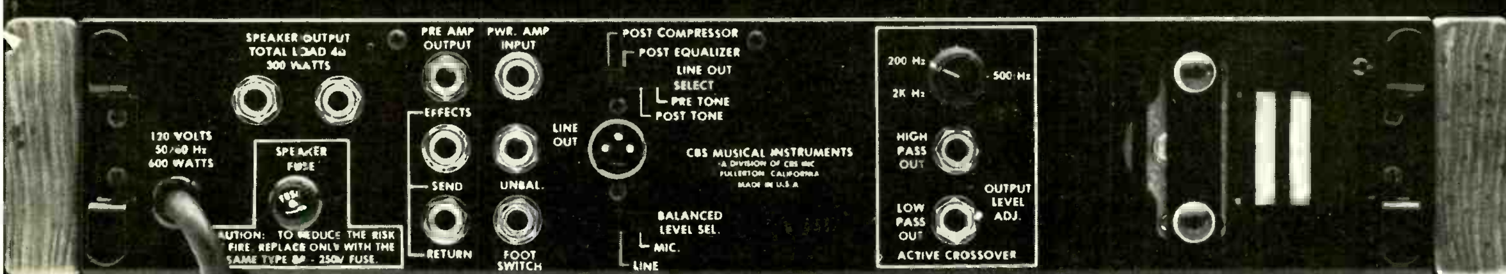
Check the specs, check the sound at your authorized Fender dealer and major band stands. Because no matter how far your playing's taken you,



separate parametric controls let the player create any sound he wants to hear on stage. And, with a selectable line level output, the sound man can have a flat signal he can mix for the house sound. That's versatility you need today!

The built-in internal limiter/compressor with variable threshold control and on/off switch allows high-volume playing without distortion.

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lines with a general expectation that they will give us a frequency response 20—20,000 Hz ±½ dB. We keep our compression to a minimum in order that the sound be fairly uniform, but not "restricted." Our release time is slow as a further aid to the cause of dynamic range. We attempt to keep our production facilities in top shape, and our staff is "encouraged" (hounded) to watch levels.

We stay alert to other ways of improving or maintaining a quality sound in this small market. I believe we sound better than many I've heard in larger markets. We also have started a brief program, "Fidelitalk," to help educate our listeners. In short, we try to get the best signal and audio possible to our listeners, in order that they can enjoy the "magic of music."

—Gary J. Kimm, Program Director
KQRN-FM
Mitchel, SD

We are pleased that you are pleased with "Ambient Sound." It's also encouraging to hear of such conscientious broadcasting and a commitment to avoiding mediocrity. Good luck to KQRN-FM. Just one thing, though. Wasn't "Fidelitalk" originally a Communist propaganda program broadcast from Havana?

Singing in the Range

After reading Chris Michaels' letter in the "Talkback" section of last month's *MR&M,* I came up with this little offering in my hope to clear up what I believe to be a misconception. I was fascinated by Mr. Michaels' mention that his finding was a "couple of dB boost in the 2 to 4 kHz range really makes the vocals pop out." Unless you've already gone and experimented with this, you would do well to note this letter. I have come into contact with many groups that have tried similar things using equalization (graphic and parametric) to force the vocals out in front of the mix. I must point out, however, that the vocal range of "most singers" doing even the highest rock & roll lyrics (e.g., Yes, Zeppelin, Kansas) or any other type of music for that matter (excluding falsetto harmonies) *rarely,* if ever, would exceed G^5, which vibrates at 783.9 Hz. Now, the first and most prevalent *overtone* in a signal of that G^5 would vibrate at 1568 Hz (one octave above the fundamental) and the second most prevalent overtone would be D^7 at 2349.3 Hz (an octave and a fifth above the fundamental).

These "harmonic overtones" as they are referred to, are audibly such a small portion of the original signal that they are virtually indistinguishable; certainly not having anywhere near the amplitude of the fundamental signal, regardless of the frequency. I mean to say that these harmonic overtones are virtually inaudible in the specific environment of a club band playing in a local bar with so much (musically and otherwise) going on at once. Very few, if any ears in the crowd at a club would be even marginally sensitive enough to perceive these harmonics.

How could a boost in the stated range of "about 2 to 4 kHz" add anything to the way the vocals "pop out" (I will assume that by "pop out" Mr. Michaels means, if somewhat loosely, the place of

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Gollehon quality at an affordable price for PA applications, keyboard reinforcement, on-stage monitoring, or playback evaluation. The S2E does it all!

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A high-temperature voice coil, strong cloth-edge diaphragm, and high-flux 60 oz. ceramic magnet provide greater control of cone acceleration and protection from electrical or mechanical failure.

The 4690 high-frequency horn/driver combination now features a Kapton[1] voice coil using a new material that can handle temperatures as high as 700°F. Wide-angle, constant level dispersion insures uniform coverage in front, at the back, and to the sides. Your audience hears you perfectly, all the frequencies...and that's important!

Solid-core plywood and tough fiberglass resin distinguish the S2E from vinyl covered particle board cabinets usually built for a beginner's PA. The S2E's plywood panels provide ten times the dimensional strength of cheaper particle board by virtue of its laminations and cross-graining.

Power handling of the S2E is 125 watts RMS, a true rating not to confuse or mislead the user. Impedance is 8 ohms. Sensitivity is 100 dB, 1 watt at 1 meter for high efficiency. Dimensions are 25" high, 20" wide, and 13" deep.

Lifting the speaker is a delight! It's only 35 lbs!

If you've been waiting to purchase new sound reinforcement equipment in the professional market, get ready. The new S2E from Gollehon really delivers!

Write or call for product information and list of Gollehon dealers. Gollehon Industries, 2431 Clyde Park, S.W., Grand Rapids, MI 49509, 616–247-8231
in Canada: 5980 Westbury Ave., Montreal, Quebec H3W 2W9; *prices optional with dealers, higher in Canada.

GOLLEHON 2431 CLYDE PARK, S.W., GRAND RAPIDS, MI 49509 616–247-8231

 [1]Dupont trademark

CIRCLE 161 ON READER SERVICE CARD

the vocals in the overall mix, as well as the vocals relative amplitude)? Right now, if you think I have any validity in my subjective presentation of range approximation, you can see how this small boost would only affect barely audible parts of the total vocal spectrum. Of course, just in terms of physical acoustics, if the overtone is there at all (and it is) it *will* be boosted a bit. But, I would venture to say, probably not a sufficient amount of amplitude boost to make a harmonic overtone of the aforementioned quality "pop out" at all. Recommended follow-up reading would be Charles Culver's "Musical Acoustics," published by McGraw-Hill.

—Peter Gravina
Teaneck, NJ

Limiting Yourself

The following is a letter we recently received from Craig Anderton, a man who knows his limiters.

A couple of people have written me recently who are having problems with the Limiter described in the November '79 issue of *MR&M.* The symptoms in both cases were described as distortion, buzzing, and rasping while the limiter was actually limiting.

One possible cause of this problem would be a defective IC, and in fact one case of an unhappy limiter owner was traced to that. Another user thought that the "glitch" that occurs while limiting could be the problem, but that glitch is not at all prominent—certainly not audible enough to be considered really buzzy.

I eventually found out that these people were having problems because they didn't know how to *apply* the limiter. I looked back over the article, and by golly, I had spent lots of words telling how to build the thing and very few on how to use it! Hopefully the following info will help those who are not quite sure about how to use limiting properly.

First, while the limiter has an amazing capacity to absorb peaks without creating audible side effects, when confronted with constant, high-level signals it can overload and give those nasty buzzing sounds. You'll find most inexpensive limiters (such as the kind used on better cassette decks) perform in the same way; they have a point beyond which they just can't cope (don't we all).

Levels should be set so that most of the time, signals fall within an acceptable range, then you add the limiter to eliminate peaks and reduce the chances of accidental distortion. Because the limiter has a very natural sound, many people who just plug into the thing don't realize it's limiting, and keep cranking up the input going to the device (or lowering the limit point) until they eventually exceed the capabilities of the unit. Adding the in/out bypass switch shown in the schematic will allow for A-B-ing the limited and unlimited sound. Any attenuation that occurs while limiting indicates the degree of limiting; if the limited sound is considerably softer than the unlimited one, you're doing a lot of limiting. If you feel the need to add this much limiting, and you need to bring up the gain, add the gain after the limiter—not before. Another point worth remembering is that the LED in-

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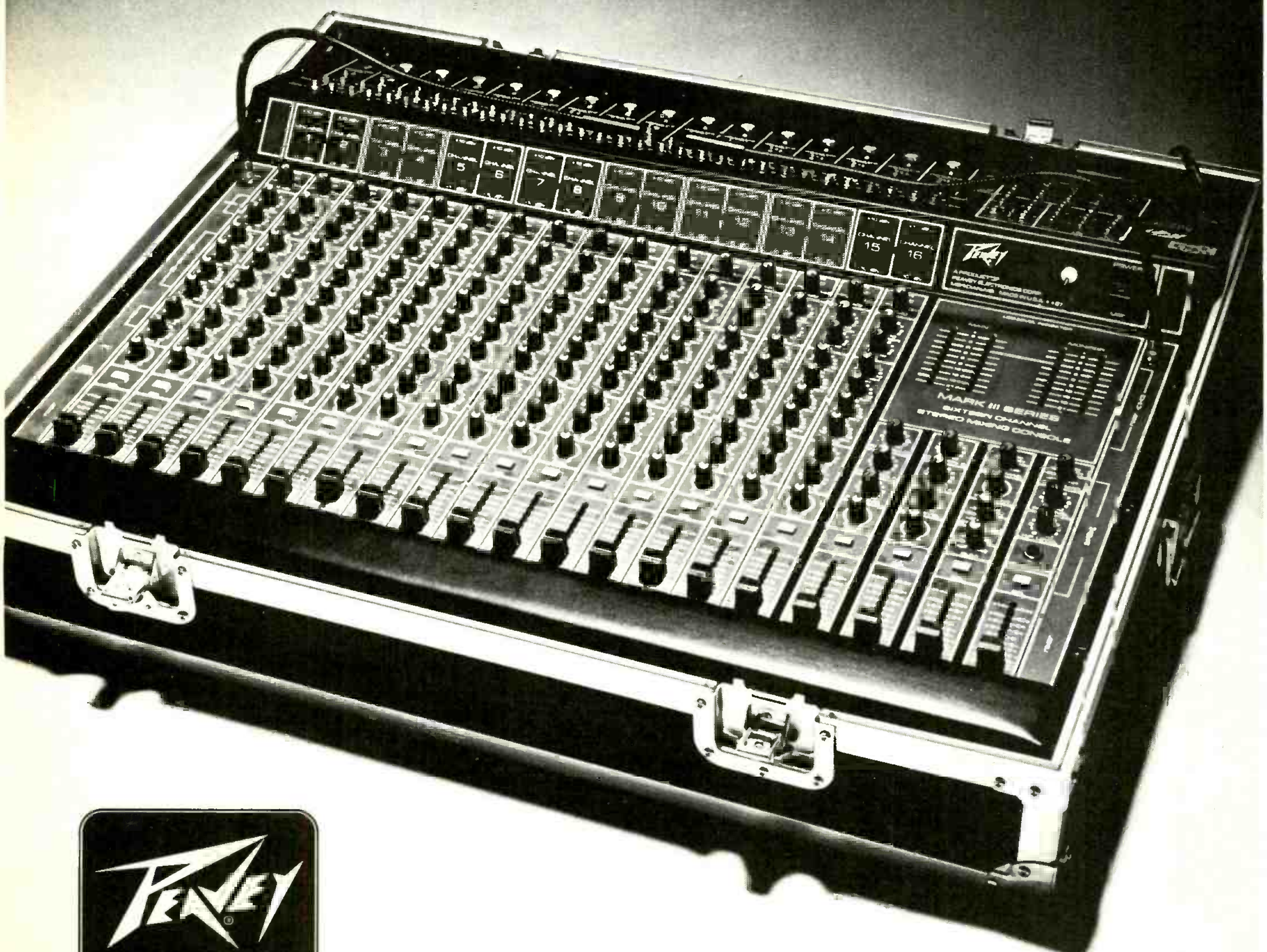
Peavey's extensive experience in electronic design, packaging, and reliability studies along with the utilization of the very latest semiconductor technology have created the Mark III Series sixteen channel mixer,...just the right combination of superb performance and useful features in a highly mobile and durable sound reinforcement console.

The Mark III's transformer balanced inputs feature virtually infinite dynamic range and include switchable 48-volt phantom power for capacitor microphones. Two independent pre monitor sends, 4-band EQ, two post effects sends and a PFL/cue button compliment each channel.

The Mark III's master section includes all the usual professional facilities along with many features previously unavailable in sound reinforcement mixers. A complete patch panel features transformer balanced line outputs on both mains, both monitors, and sum.

We've listed just a few of the Mark III's many features to illustrate its performance,...to get the complete picture, you'll want a "hands on" demonstration at your Peavey dealer. You'll see again why incredible values are still a reality at Peavey.



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CIRCLE 89 ON READER SERVICE CARD

dicator only shows what kind of level you're putting into the device. It has nothing to do with the amount of limiting, and will not glow brighter if you lower the limit point. If enough people want some kind of blinky light display that shows the amount of limiting in dB, it can be arranged—but then it's no longer a $25/channel limiter.

By the way, the limiter is not really intended to act as a sustainer type special effect for guitar or bass, although it does do a reasonably good job in this application. For special effects where you need intense squeezing of dynamic range rather than simply protecting against peaks, use a compressor. This will give the sound you want much more readily than a limiter.

It has also been pointed out that the limiter has other applications which were not mentioned in the article. Placing it before an analog delay allows you to put more average signal into the delay, thus seeming to reduce the noise. You can also patch a limiter before a spring reverb, since sudden peaks may "bong" the reverb and the limiter will keep these under control.

Finally, I'd like to thank all those people who took the time to write and say how much they enjoyed the limiter and were looking forward to more projects. Rest assured there are plenty more projects where the Limiter and Hot Springs came from.

—Craig Anderton
Contributing Editor
Modern Recording & Music

The Matter of Truth

I would like to take this opportunity to compliment you on a publication that I consider to be the best in the field. In the two years since I first subscribed to *MR&M*, I have found the articles to be both informative and useful from a practical viewpoint.

I'm writing to congratulate you for printing a "Letter to the Editor" in the January 1980 issue entitled, "Truth in Advertising Welcomed." I also saw the Bose advertisement featuring the Christian group Truth in the September 1979 issue and was gratified to see them receive the recognition they deserve. Having personally heard two of their concerts, I can only give them my highest compliments.

As I write this letter I am preparing to leave for a six day sound engineering seminar (Workshop 1980) being conducted by Truth in Mobile, Alabama. This promises to be an excellent learning experience, since Bill Gaither, Ron Huff, and several other well known Christian recording personalities will be in attendance. I'm sure many readers beside myself would enjoy reading a feature article about Truth in an upcoming issue of *MR&M*. In the meantime, keep up with the excellent articles you have been printing.

—Dennis Perkinson
Oak Ridge, TN

Thank you for your supportive letter. Though we really have no immediate plans to do an article on Truth, we will keep the idea in mind. If enough readers evince interest in seeing a piece on them, we will most certainly get plans underway for an article. Right now, though, honest as we are, we cannot give you Truth.

Jacks That Go Downhill

Regarding your request in "Sound System Sniffles" in the Product Scene column by Norm Eisenberg in the July 1980 issue of *MR&M*, I have en-

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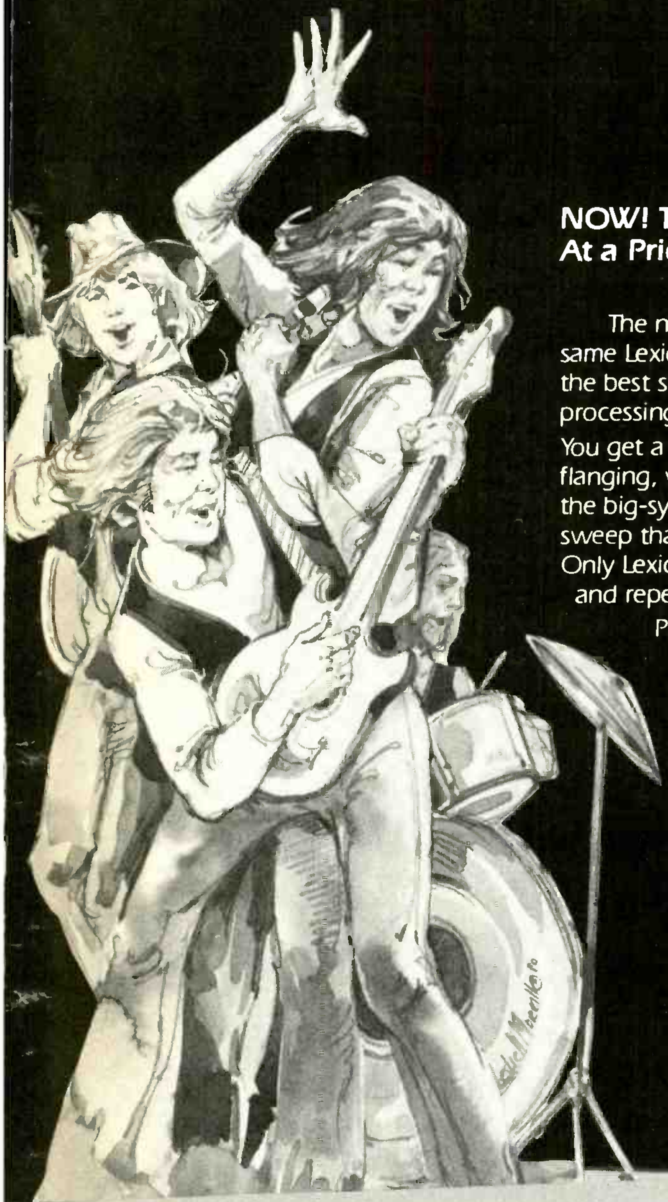
Introducing Lexicon's PCM 41 Digital Delay



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The new PCM 41 (we call it "Baby Prime Time") is based on the same Lexicon technology preferred by world class studios. So you get the best studio-quality sound with PCM (pulse code modulation) audio processing.

You get a broad repertoire of creative effects — double tracking, flanging, vibrato/tremolo, arpeggio, doppler pitch shift, slap echo. All the big-system stuff plus some dramatic new goodies. Like articulated sweep that automatically keys time delay to the type of audio signal. Only Lexicon has it! And infinite repeat that locks up a musical segment and repeats it indefinitely for background rhythm and counterpoint.

PCM 41 is human engineered for easy on-stage (or small studio) use. All major functions can be foot-switch controlled. But the big advantage is the sound quality!

Compare! Listen to Lexicon's PCM 41 for yourself and you'll never settle for second best. Not when you can get Lexicon quality for about the same dollars. The PCM 41. Available from leading pro-audio and musical instrument dealers.

**Lexicon PCM digital delay — 20 Hz to 16 KHz bandwidth.
Less than 0.05% distortion.
Up to 800 ms delay effects.**

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CIRCLE 148 ON READER SERVICE CARD

countered the problem of oxidation building up on RCA jacks also. An intermittent contact in my sound system which caused a crackling in my right speaker drove me crazy for over four months. Part of the problem was an assumption on my part that it was caused by either the speaker, cartridge, or their associated wiring. The fact that it was there one day and gone the next didn't help either. By crossing channels from one component to the next and living with each change for three or four days, I finally traced the problem to my model CD-5 Dubie. On close examination I have noticed that the sleeves on the RCA jacks used on the Dubies appear to be aluminum and may oxidize more quickly due to the interaction of dissimilar metals. After consulting the only real technician I have found in my area (there are lots of mechanics here) and being told that I was The Lone Ranger, I have kept this problem to myself and instituted a routine cleaning of these jacks. I'm glad to hear that someone else has had the same problem. I've considered using the conductive silicon grease used on RF connections, but have been put off by not knowing how they would work at lower frequencies. Until someone checks it out with test equipment, I guess I'll never know. In the meantime, routine cleaning of jacks, plugs, and speaker connections is a small price to pay for good sound.

—Yours for better fidelity,
Jeffrey W. Howell,
Mission, Texas

Though you didn't exactly ask, I'm going to respond to the implied question in your letter. In regard to using the conductive silicon grease that is normally used on RF connections at lower *frequencies, you can try it. We were told by our sources that it probably would do no harm, but also would probably do no good, because the silicon grease is more a thermal transfer medium than an electrical medium. The periodic cleaning you yourself suggested still seems to be the best bet.*

Home Stereo

Please print the address for the Hafler DH-200 Power Amplifier. I am considering building a home stereo and I'm convinced that this amplifier would definitely be the way to go. I know that home stereo is not exactly your forte, but I would also like to build some nice stereo speakers that possibly could be used later on in a beginning studio. Would you know of any books on the subject?

—Clif Coleman
Boston, Mass.

The address you need regarding the Hafler Power Amplifier is: The David Hafler Company, 5817 Roosevelt Avenue, Pennsauken, New Jersey 08109.

Perhaps the following book might be of use to you in building stereo speakers: How to Design, Build and Test Complete Speaker Systems *by David B. Weems. It's published by TAB Books, Blue Ridge Summit, Pennsylvania 17214. Its date of publication is 1978. It's good to see that you're following Billy Joel's advice to not waste your money on a new set of speakers—but instead, are building your own. Thrift is swift.*

'58 was a very good year...

The '80's will be even better!

The PL80 is going to be the hottest vocal microphone of the '80's. It is the microphone every vocalist wants because it has the sound every vocalist wants. The sound of the PL80 results not only from extensive user field testing with rock superstars like Steve Perry of Journey, but also from side-by-side product comparisons and interviews with many of the most highly respected sound men in the business. Most of all, the PL80 is the result of an entirely new application of computer-design technology called "fast Fourier transform" that allows the design engineer to predict, as it's being designed, precisely how a microphone will sound in use, not just in a sterile test environment. The PL80 is a performing vocalist's microphone that has been called the best new microphone design in years.

The Electro-Voice PL80 not only gives you the exact sound you want, it does a whole lot more. The PL80 tops the competition in just about every performance category. Its style sets it apart from any other mike. Its sensitivity is higher than the current best-selling microphone; and when it comes to gain before feedback, the monitor speakers are likely to give out before the PL80 will.

Even the "feel" of the PL80 is impressive! It has the weight and the "heft" to give you confidence. The new snow-gray finish and contrasting charcoal gray grille screen make a striking impression on stage, but the colors are subtle enough not to detract from your performance. E-V's exclusive Memraflex grille material resists the dents and knocks common to other microphones. This will keep the PL80 looking like new for years while other mikes look old after one or two accidental drops.

Use the PL80 at your Electro-Voice PL Microphone Dealer. Test it against any other mike. If you want your sound to be the sound of the '80's, the PL80 – *the* "Sound of the '80's" – is the only mike you'll buy.

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CIRCLE 82 ON READER SERVICE CARD

TALK BACK

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

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

An Ordinary Guitarist Battles the Fading Battery Blues

I'm just an ordinary guitar player and I have a problem that I've never been able to solve. I have five foot pedal accessories: wah-wah, phase, compressor, distortion, and octave. They all need those demon 9 volt batteries. Some have two 9 volt DC jacks for adapting to AC. Do I still have to use five different separate adapters? What are the appropriate milliamperes for each unit? In fact, how many milliamperes does a fully charged 9 volt battery have?

That problem solved—how do you get away from buying (and connecting and dragging around) five separate adapters?

Simple hassles, but I hope you can help.

—Ron Vickery
Miami, Fl.

Indeed, those nasty 9-volt batteries are a pain. Their favorite stunt is passing on to dry cell heaven just when your licks are at their hottest!

An AC adapter can eliminate this problem, and the most obvious solution is to utilize one adapter per device. However, this will create a nifty bowl of wiring noodles. Additionally, many AC adapters are not well filtered and hence, are likely to induce unwanted 120 Hz hum into your effects devices. So, perhaps a better remedy for the "fading battery syndrome" would be to use one common power supply for all the units.

I attempted to determine how much current a typical 9-volt battery can be expected to supply, but my reference sources were incomplete in this area. So, I will guess that 25 mA (milliamperes) would be about tops for a zinc-carbon battery. Thus, if you were to use a power supply capable of producing around 8.5 volts (a 9-volt battery actually does not produce a full 9 volts when "loaded" by a circuit) at 150-200 milliamperes, your problem should be solved. The power supply's output also must have a hum, ripple and noise content of less than 1 millivolt peak.

There is a real price war going on now in the area of so-called open-frame, *linear* (*not* switching) power supplies, and many are also showing up in industrial surplus stores. This route is probably better than building a supply from scratch. All you will need to do is mount the unit in an enclosure, since this economical type of power supply does not come in the usual metal box (hence, its economy).

A few manufacturers of these include Kepco, Inc., 131-38 Sanford Ave., Flushing, N.Y. 11352; Power/Mate Corp., 514 S. River St., Hackensack, N.J. 07601; Power One, Inc., Power One Drive, Camarilla, Ca. 93010. Other manufacturers include Solar Electric and Acme Electric of Cuba, N.Y., whose addresses were not available at press time. Your best bet is to contact several wholesale electronic parts houses in your area and ask for a linear, open-frame power supply adjustable to 8 or 9 volts, and with the lowest current rating available (the smallest models will typically supply ½- to 1 ampere in this voltage range).

One final remark—if the effects device does not include a power supply jack, it is very easy to correct this problem without having to drill holes. Simply purchase a 9-volt battery connector (just like the one already in the effects unit) from a parts house and "mate" the two together. The red wire goes to the positive output, and the black goes to the negative output of the power supply. Also, when wiring a unit with an existing power jack, be *sure* to correctly connect the positive and negative leads (refer to the manufacturer's literature).

—Brian A. Roth
Technical Editor
Modern Recording & Music

Three Great Minds, But With a Single Thought

I found the article, "Footswitching Your Teac 3340" (July 1980 issue) very interesting, but unfortunately, of no use to me since I own a Teac A-3440, which has a different remote control configuration. Since I am very much interested in this modification, can you tell me how I might adapt this footswitch for my 3440?

—Daniel Huneault
Gatineau, Zuebed

I'm writing to make some inquiries pertaining to Craig Anderton's article in the July 1980 issue ("Footswitching Your Teac 3340," page 54). My praise goes out to your staff for accommodating the semi-pro recordist. However, being the proud owner of the newer Teac A-3440, I am not able to utilize this footswitching idea since the 3440 does not use the same 11-pin plug for remote as the old 3340. Is the plug

needed for the 3440 available and adaptable to the punch-in box in the article? Since the 3440 has the improved "Function Select" switching, I am not sure if the circuitry would be compatible.

—John R. Mobilio
Waterbury, Conn.

I greatly enjoyed Craig Anderton's article, "Footswitching Your Teac 3340" (July 1980). But I own a Teac A-3440 and the plug on the back of this unit is different than the one on the 3340. Can you please tell me how to wire a footswitch for the 3440?

—Michael W. White
Whitesong Studios
Memphis, Tenn.

Those interested in adding footswitching to Teac machines other than the 3340 will be glad to hear that Teac offers a footswitching unit designated the SP-70. This gives remote punch-in and pause capabilities for the 3440, 40-4, and 80-8 recorders (as well as the recently introduced 22-4). List price is under $35, so you wouldn't save much (if at all) by doing it yourself. However, note that there is no commercially available device for footswitching the 3340—if you want to add footswitching capabilities in this instance, you'll have to roll your own.

—Craig Anderton
Contributing Editor
Modern Recording & Music

Special Considerations

I'd been shopping for a noise reduction system both for making "live" tapes and for recording from other sources. I finally purchased a dbx 224. One thing I noticed right away was that when I used the dbx to encode a program—even though I use tape from three different manufacturers, in three different price ranges—the tapes all sounded the same at low and high speeds. This from tapes ranging from BASF Professional back coated at 19 cps to Maxell LN35-90 at 9.5 cps! Since my results don't vary, even on good cassette equipment, how will I know when to use back-coated or other special formulation tape? Or should I bother to use it at all?

—John Buford
Philadelphia, Pa.

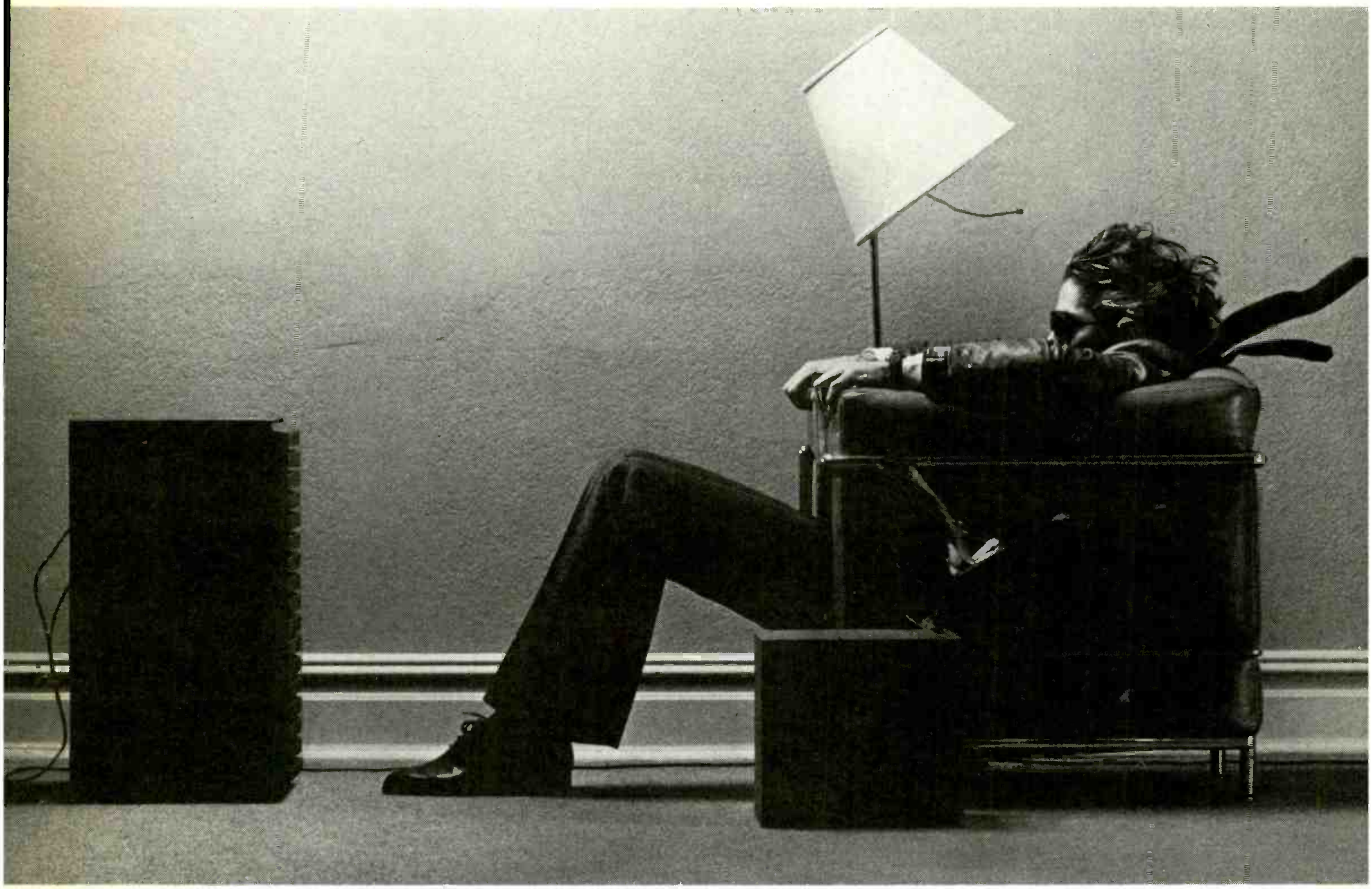
If you hear no difference between premium grades and less expensive grades at 9.5 cps (3¾ ips) and 19 cps (7½ ips) speeds, then the tapes must be fairly compatible with each other and with your recorder's bias and equalization settings so that the frequency responses match closely enough to disguise subtle differences. If they all sound exactly the same to your ear, it would make sense to use less expensive tape at the slower speed in order to save money. The use of a sophisticated noise reduction compander such as the dbx 224, however, adds another element which should be considered—the susceptibility to dropouts.

A noise reduction system using signal compression and subsequent expansion at a ratio of 2:1:2, such as the dbx 224, will expand a momentary loss of signal caused by a tape flaw and double the loss. A 2 dB dropout without noise reduction becomes 4 dB with it. It is essential that the tape used be free from flaws not only *before* recording, but also that it not be damaged during storage.

Dropouts of virgin tape are most likely due to coating imperfections. Quality tape manufacturers have improved coating techniques so well that lesser grade tapes generally do not vary a great deal from premium grades in this respect, although high grade tape may be less prone to rubbing off and leaving a deposit which may cause a desultory dropout or so. What may be a more significant difference is what happens after the tape is recorded and stored. The smoothness of the tape wind can determine the amount of edge damage caused by contact with the reel flange and, therefore, the amount of dropout. As long as the heads, guides, and reel tables are properly aligned, a tape should wind so that no edges rub against the side of the reels. If rubbing occurs, it can deform the tape until it no longer maintains proper head contact at the deformed points; and the signal "drops out." The damage is cumulative; a tape which played back several times without dropouts can finally develop enough edge damage to cause some very audible dropouts.

For this reason I would suggest using a tape at least 1 mil thick (1800 ft. on 7-inch reels; 3600 ft. on 10½-inch reels) for extra durability when using dbx. I would also suggest recording at the faster 19 cps speed, even if there is no audible difference, simply to be sure that any dropout is as brief as possible.

AFTER 500 PLAYS OUR HIGH FIDELITY TAPE STILL DELIVERS HIGH FIDELITY.



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

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

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

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

So even after a Maxell recording is 500 plays old, you'll swear it's not a play over five.

maxell

IT'S WORTH IT.

Maxell Corporation of America, 60 Oxford Drive, Moonachie, N.J. 07074.

CIRCLE 79 ON READER SERVICE CARD

A back-coating tape will give additional protection for two reasons: 1) The back-coating provides smoother winds than untreated tapes and 2) It drains off static charges so that a metal reel can be used in place of a plastic one in order to provide better, warp-free protection for the tape edges during handling and storage. If you are careful about how you use unwarped plastic reels and keeping your recorder transport in alignment, then back-coating is not necessary.

—Terence D. O'Kelly
Supervisor, Field Technical Service
BASF Systems
Bedford, Ma.

When using dbx tape noise reduction equipment, there are many specifications that distinguish one type of recording tape from another. The most important of these are the signal-to-noise ratio, which reflects dynamic range record capabilities, and frequency response, which affects tracking.

The dbx noise reduction system, being a compression/expansion system coupled to frequency weighting curves, compresses program materials during recording and expands the recorded materials on playback. Through this companding action, there is a decrease in the dynamic range requirements of tape while still allowing wide dynamic range to be reproduced.

A poor tape's signal-to-noise ratio is less obvious and, since recordings are made at lower levels, tape frequency response is usually flatter. Also, when using dbx noise reduction, the high level, high-frequency roll-offs of cassette machines are less evident.

Your question does not describe the program source used for testing. We suggest that you use a "live" recording session to determine which tape works best with your machine because "live" recording requires more from a tape system than program sources such as FM broadcasts or discs that typically have been previously compressed. It will take an unusual set of circumstances to reflect any differences in tape performance.

When comparing tapes, listen for musical details and high-frequency distortion. The differences between tapes may be slight. Another thing to compare is the durability of the tape after a few plays. Using these techniques should permit you to determine which tape is best for your machine.

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

The Swiss Obsession With Silence

Can I plug two different kinds of noise reduction into my stereo? I own separate Pioneer components (turntable, tuner, amplifier, tape deck, equalizer) and I would like to get either the SAE 5000, the Burwen DNF 1207 or the Burwen TNE 7000A noise eliminator to get rid of (I hope!) those medium volume clicks and pops. I have about 400 LPs and about half of them have these small ticks. I even get these darn noises on new releases! I would insert one of these devices between the

Honeycomb speakers inspired by today's recording technology.

To hear the full potential of today's digital tape and direct-to-disc recordings, our Professional Audio Division presents a speaker that's just as advanced, just as accurate. The Technics R&B Series Honeycomb Disc Speakers.

The special diaphragm is constructed like a flat, rigid honeycomb disc to eliminate the traditional acoustical problems of cone-shaped diaphragms. Best of all, phase linearity occurs automatically because the acoustic centers are distinctly aligned on the flat diaphragm surface.

The axially symmetric "nodal driver" is designed to vibrate in a more accurate pistonic motion. This provides an extremely wide, flat frequency response, broad dynamic range and extremely low distortion.

And the flat leaf tweeter in the SB-7 and SB-10 is much lighter than many dome tweeters, for incredible high frequency response, up to 125 kHz.

The Technics R&B Series is available at select locations. Call our Professional Audio Division at 800-447-4700 (in Illinois, 800-322-4400) to hear the entire line of Honeycombs—the speakers that don't hold back today's recording technology.

CIRCLE 98 ON READER SERVICE CARD

Technics
R&B series

turntable and the amplifier.

Between the amplifier and the tape deck I'd like to insert either the Sanyo Super D or the new dbx 224. Can I patch two noise reduction units into my system in the manner I have outlined? Do I need any additional devices to accomplish my goal? Can you advise me as to the best way to go about this?

I hope this letter isn't relegated to the trash basket. It's very hard to find knowledgeable help here, and to hear my albums without these awful noises has become something an obsession.

Greetings from Switzerland!

—R. Meneses
Zurich, Switzerland

Greetings from the more or less United States of America!

You can connect both noise reduction units to your system in the manner you describe with no difficulty whatsoever. As a matter of fact, you can connect (theoretically) any number of noise reduction devices in series or cascade fashion if you want to, as long as you think of them as separate units and decode them in the same order as they were encoded. If you decide to get the dbx 224, you will be able to use it to play back any dbx encoded discs you may purchase. But you will have to disconnect it from the tape deck and reconnect it to the turntable. In that case, you will want to put the dbx after the turntable preamp and, if you still wish to use the tick and pop eliminator (or any other device), that would have to come *after* the dbx as the dbx must see straight through to the encoded medium. Actually, though, using more than one noise reduction gadget might be overkill.

You should be alright with the setup which you describe. When you get whatever equipment you decide on, the instructions will most likely give you all the information and direction you should need.

Offhand, I would suggest the TNE 7000A for the ticks and pops. The DNF 1207 is a dynamic noise filter and, while it is excellent for cleaning up really rotten sounding records such as old 78s and such, it does tend to detract from reasonably well recorded material. In my opinion, it should be used as more of a last resort for a terminally ill record or tape. The dbx can serve a dual purpose as I mentioned above.

Whatever you do, before you buy you should take a few records from your collection, one very noisy, one moderately noisy and one relatively quiet, to the store with you and try the various units through equipment similar to your own. There are a number of competing devices on the market and you should make an informed choice as they all have their own characteristics which you may or may not be able to live with. If the store won't let you make comparisons then go to another store.

One last thought. No gadget can get rid of noise altogether. Your obsession (which is shared by many, including myself) will be a bit difficult to realize. We in the U.S. tend to drool over European recordings after listening to our own. Perhaps you could save yourself a bit of money by listening to a bunch of records pressed in the U.S. for a month or so. Then your pressings will sound immaculate, as they to do us!

—David Moyssiadis
Contributing Editor
Modern Recording & Music

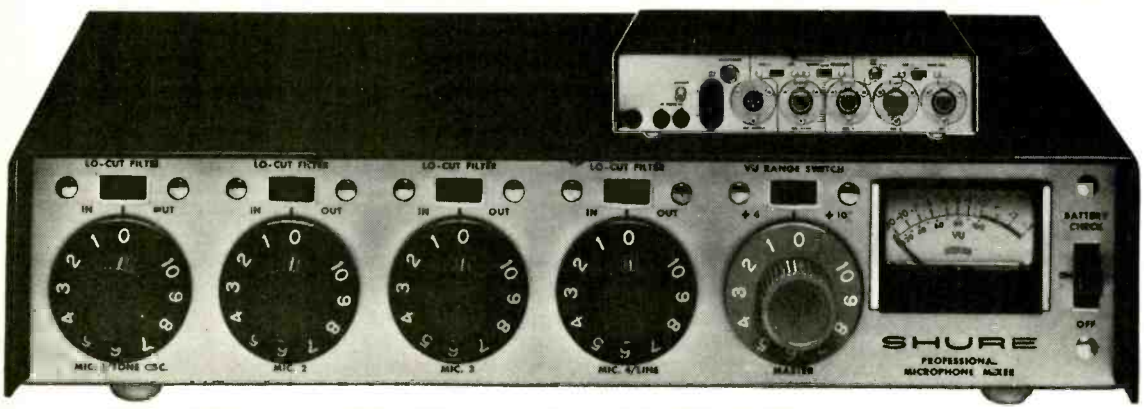
fact: some veteran products do their job so well, for so long, for so little money it's a pleasure to reintroduce them.

Through the years, Shure microphone mixers have gained the reputation of being the practical, efficient, economical way to increase the flexibility of public address, sound reinforcement, and paging systems, as well as tape recorders using multiple sound sources. In fact, they are used in almost twice as many studios as the next most popular brand. . . with good reason.

For Example:

The Utterly Simple M68 Microphone Mixer. . .

The original high-performance, low-cost mixer for professional and semiprofessional applications. Excellent for most sound system and tape recording requirements. Portable (less than 4 lbs), ultra-simple in operation and gratifyingly modest in price. Four high- or low-impedance microphone inputs plus an additional auxiliary high level input, with a master volume control and individual controls. A wide range of accessories lets you customize the M68 for almost any special installation.



The Thoroughly Professional M67 Mixer. . .

Specifically designed for professional recording, TV and radio studios, "remotes," sound reinforcement and audio-visual installations. Four low-impedance transformer-coupled mic inputs, one convertible to line input. Ideal as a self-contained compact console or as an "add-on" for existing facilities. VU meter; built-in tone oscillator for sending level test signal. Extremely low noise and RF susceptibility; two-level headphone monitor jack. . . ac or battery operation with optional battery pack—even switches automatically if ac line fails.

Microphone Mixers by

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Manufacturers of high fidelity components, microphones, sound systems and related circuitry.

CIRCLE 118 ON READER SERVICE CARD

THE PRODUCT SCENE

By Norman Eisenberg

DIGITAL DELAY PROCESSOR

The new PCM ("Baby Prime Time") from Lexicon, Inc. is a digital delay processor that uses pulse-code modulation encoding. The device—made specifically for performers and small studios—offers a full repertoire of effects (automatic double tracking, flanging, vibrato/tremolo, arpeggio, doppler pitch shift, slap echo, infinite repeat, etc.) and is said to be "human factored" designed which makes it easy to operate. Major functions can be footswitch controlled. Price is $1095.

CIRCLE 1 ON READER SERVICE CARD

NEW BGW AMPLIFIERS

Two new commercial power amplifiers have been announced by BGW Systems, Inc. of Hawthorne, California. Both units are dual-channel models, said to be the first products in a new line whose design is based on suggestions received by BGW. The units are intended expressly for distributed systems applications. BGW claims improved performance, reliability, serviceability and cost savings of as much as 50 percent. The model BGW 320 is a 100-watt (minimum sine-wave power) per channel unit that can drive any load impedance from 4 ohms and up, including 25 or 70-volt distributed systems. The BGW 620 offers 200 watts per channel into the same loads. Both amps use modular construction and fit standard rack mounts.

CIRCLE 2 ON READER SERVICE CARD

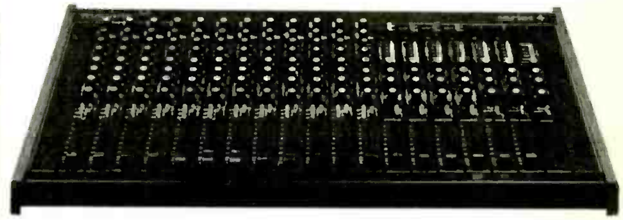
FOUR-WAY CROSSOVER

Four-way parametric electronic crossover is provided by the Power FEP 204 whose "Comelentric" design is said to offer improvement on conventional Butterworth, Bessel or Tchebytcheff configurations. User selectable crossover frequencies are: bass to low midrange, 155, 190, 270 and 340 Hz; low midrange to midrange, 800 Hz, 1.2, 1.8 and 2.7 kHz; midrange to treble, 4, 5, 7 and 10 kHz. Nominal input and output levels are +4 dBm (1.22 V).

CIRCLE 3 ON READER SERVICE CARD

TANGENT SYSTEMS MIXING CONSOLES

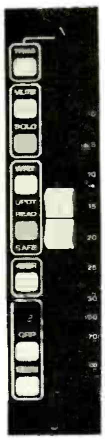
The Series 4 designates a new line of mixing consoles from Tangent Systems, Inc. of Phoenix, Arizona. A four output buss design, the Series 4 is offered for sound reinforcement, and four- and eight-track recording in either 12 input or 20-input fully modularized mainframe. Among its features are transformerless input circuitry; three-band continuously variable EQ in each channel; peak LED and 20-dB pad on each input; eight independent returns, PFL and six-out buss assign through submasters (4) and R/L stereo busses; three independent foldback sends; full multi-track monitoring and assign; muting, 100-mm faders, phantom mic power; external power supply; options for reverb and expander modules. Prices start at $2800.



CIRCLE 4 ON READER SERVICE CARD

NEW PRO AUDIO COMPANY

Fax Audio, Inc. is the name of a newly established company, formed by the former director of engineering at Automated Processes, Inc. (API) together with several other former API engineers. The new company states "Fax will design and produce a complete line of state-of-the-art products and systems geared for the recording and broadcast industries."



Fax already has introduced a "highly innovative programmable fader featuring touch switches, a sophisticated array of visual displays and a wider range of functions than previously available in modular automated faders." Fax also has developed a discrete op-amp whose specifications and performance criteria are claimed to be the finest of any op-amp on the market. The new op-amp can be used to upgrade any API model 2520 circuit, and it also is employed in a complete series of plug-in amplifier cards.

CIRCLE 5 ON READER SERVICE CARD

THE WORKS FROM WIREWORKS

Wireworks Corp. has announced a new set of technical data sheets detailing an extensive offering of cabling products and accessories for use in professional audio interconnect systems. Covered in the data sheets are hardwired microphone multicables, pro mic cables, the multicable components group (a "mix and match" system) and the model TE-2 mic cable tester.

CIRCLE 6 ON READER SERVICE CARD

CERTRON OFFERS NEW TAPE

Ferex I is the name of a new tape introduced by Certron Corp. as its first premium ferric-oxide audio cassette aimed at the high-end audiophile, music and recording studio markets. Described as a standard-bias, high-energy, high-output tape, Ferex I cassettes are marketed in C-60 and C-90 sizes in a smoky-clear 5-screw housing with pressure-sensitive labels and index card. Cassettes—cellophane-wrapped in a clear Philips-styled outer storage case—are available in individual and multi-pak configurations.

CIRCLE 7 ON READER SERVICE CARD

QUINTET FROM NEPTUNE

Five new products have been announced by Neptune Electronics Inc. of Portland, Oregon. The new units include two stereo mixing consoles, two graphic equalizers and a real-time analyzer.

The model 1420 console has fourteen input channels and a host of features which include an extensive headphone monitoring system with a 5-watt headphone amp. The model 821 has eight inputs and can interface with another stacked 821 for more input channels.

One-third octave equalization is offered by the model 2711 whose center-detented, oil-damped slide controls provide ±12 dB on each of twenty-seven ⅓-octave bands on ISO frequency centers. Ten-band octave equalization on each of two stereo channels is provided by the model 1021.

The model 2709A, Neptune's real-time analyzer, checks response in every part of a sound-reinforcement, recording or stage instrument system. The device is designed to employ the user's microphones, loudspeakers and equipment instead of only the standard calibrated mic or line source. Response is displayed on an LED matrix consisting of twenty-seven ⅓-octave bands with nine steps of amplitude.

The models 821, 2711, 1021 and 2709A all may be rack-mounted.

CIRCLE 8 ON READER SERVICE CARD

PATCH BAYS

Jack fields (patch bays) for broadcast and sound studio applications are described in a 12-page catalog from ADC Telecommunications of Minneapolis, Minnesota. Included are bantam patch bays, central patching, console patching and long-frame bays.

CIRCLE 9 ON READER SERVICE CARD

MODULAR SIGNAL PROCESSING SYSTEM

The dbx 900 series is a new modular signal processing system. Up to eight modules can be fitted into a rack-mount space 5¼ inches high, uses standard connectors, and the modules may be slipped in and out in seconds. The dbx model 902 can be used as a broadband de-esser or for attenuating only a user-determined portion of the high frequencies. Gain reduction is adjustable from zero to 20 dB. The 902 also continuously analyzes the input signal spectrum to provide the exact amount of de-essing selected, regardless of signal level. The device requires no recalibration for signal level changes. The user can set it and forget it.

The model 903 compressor begins reducing output volume once the threshold is exceeded. The new 903 is an "Over Easy" compressor—it has a soft-knee threshold that increases compression ratio gradually over a range of several dB. It also features true RMS level detection, continuously variable compression ratios and an adjustable (−40 dB to +20 dB) threshold.

The model 904 is a noise gate claimed to have a combination of features not found on any other noise gate at any price. It offers adjustable attack and release rates, threshold adjustment from −30 to +10 dB, attenuation limit adjustment from 0 to 60 dB, with dbx "Over Easy" downward expansion for smooth sound. The 904 also has a KEY input that allows gating of one instrument by another. The special GATE mode of the 904 allows users without automated consoles to put threshold programmed muting on solo channels. After the user sets the correct solo level on the console, the 904 automatically keeps the channel muted, eliminating spurious signals that may precede the solo itself. When the solo begins, the 904 un-mutes the channel, allowing the solo into the mix at the pre-set level.

CIRCLE 10 ON READER SERVICE CARD

CM LABS PRODUCTS

CM Labs—a division of Audio International, Inc. of Danbury, Connecticut—has announced several products. The CM920 is a stereo power amplifier rated for 250 watts per channel into 8 ohms; 400 watts per channel into 4 ohms. The amp, says CM Labs, will drive all types of reactive loads up to 6 mfd from 20 Hz to 20 kHz. Hum and noise are listed at 100 dB below 250 watts; harmonic distortion, less than 0.15 percent; IM, less than 0.1 percent. The amp fits the C-size relay rack-panel. Output levels are shown on vertical LED scales.

A smaller power amp is the CM914c, also with LED metering, and rated for 125 watts per channel into 8 ohms, or 175 watts per channel into 4 ohms. Other major specs are similar to those of the CM920.

CM Labs also has an FET preamp, the model CM301a, rated for 2 volts output into an IHF load, or 1 volt into 600 ohms.

For use in permanent or portable disco systems, or as an audio mixer for commercials and A/V, as well as a preamp in home stereo, there is the CM607 console which features inputs for two turntables, plus one tape deck, with cross fading. The unit also has full pre-cue facilities and talk-over. Outputs for amplifier and tape are provided.

CIRCLE 11 ON READER SERVICE CARD

HEATHKIT PROCESSOR

An active audio processor that combines a dynamic range expander and a noise-reduction system is the model AD-1706 available from Heath Co. Heath says the AD-1706 increases dynamic range of source material up to 17 dB, through 7 dB of gain and 10 dB of noise reduction. A single-ended design, the AD-1706 may be used with any program source without pre-processing. Rack-mountable, the unit is priced at $249.95.

CIRCLE 12 ON READER SERVICE CARD

LOW COST REVERB SYSTEM

Biamp Systems, Inc. has announced its new MR/140, described as a low-cost ($249) reverb system designed for use in recording, portable sound reinforcement, permanent installations, monitor systems, and so on. Its features include transformerless balanced inputs, "improved cosmetics," EQ blend control system, automatic limiting and drive set indicator. Preliminary specs indicate a slew rate of 13 volts per microsecond dry signal; signal rate of minus 80 dB from zero dB with frequency response to 100 kHz; a decay time of 1.2 seconds. Unity gain is listed as minus 70 dB from zero dB with blend control at maximum reverb, all EQ controls set at 50 percent. THD is rated as less than 0.01 percent.

CIRCLE 13 ON READER SERVICE CARD

WHO'S MINDING THE STORE?

An old story tells of the dying shopkeeper surrounded by his family and loved ones. Through half-closed eyes he tries to discern who is present, and he keeps asking for everyone by name. Reassured at last that everyone is at his bedside, he then rears up and shouts: "So who's minding the store?"

This same question occurred to me recently, though for different reasons, when I tried, in one store, to buy some spade lugs; and tried, in another store, to buy some double banana plugs with standard spacing. In both instances (and these were branches of large, national distributors) the clerks did not know what I meant. In one store I found the damn spade lugs myself after poking around a while. When I showed one to the clerk he acted as if he couldn't care less. He didn't know what this simple basic part was and he didn't want to know.

In store no. 2, the clerk tried to be more helpful but he didn't have what I wanted and he didn't know he didn't have it. He showed me some single banana plugs and thought that the "spacing" referred to the length of metal protruding from the insulated holder. I finally ended up buying two single plugs; he looked mystified when I said I would use two singles instead of one double.

I must say that, aside from the sheer frustration of driving a few miles to a place that is supposed to be a radio parts supply outlet and not finding what I need, I am developing a real concern about what the hell is happening to the level of competence, and the ability to serve customers, in the retail area—where the action is, as they say. It's okay to peddle calculators and telephone-answering gadgets to suburbanites, but what has happened to the rapport between the parts dealer and the working technician? Spade lugs and banana plugs are little things, to be sure, but when you can't get them or when the sales people don't even know what they are, what kind of confidence can be built up in that most critical area of the industry where the manufacturer meets the customer, so to speak?

This ignorance, by the way, is not confined to retailing. I talked on the phone to someone at a radio station who did not know what I meant by "station frequency." And I saw someone at a well-known shipping company spell "phase" as "faze." That's enough to faze anyone.

I could go on too about the p.r. man for a hi-fi company who had never heard of Robert Schumann, the editor of a music magazine who referred to Stravinsky's "Rites" of Spring and the TV announcer who said "in-fec-tyou-uss" when he meant "infectious."

It is fashionable these days to talk about computer error and likely most of us can tell our own tale of screwed-up bills, statements or even car registrations. What bothers me is the increasing incidence of human error generally, and examples of it in our industry specifically. Doesn't anyone in authority give a damn? Who the hell is minding the store?

MUSICAL NEWSICALS

SYNTHESIZER ACCESSORIES

The Bottom Line is the name of a new bass pedal unit from Peterson Electronics. The system has a 13-note pedal unit and a compact control unit with controls for Volume, Sustain and Voicing, and an on/off switch for a plucked attack sound. The control unit either mounts on the upended carrying case or it may be mounted on the edge of a keyboard instrument using the bracket included with the unit. The Bottom Line has a wide range of bass sounds from a flute-like tone through string bass sounds to synthesizer sounds.

CIRCLE 17 ON READER SERVICE CARD

Sequential Circuits Inc., the Prophet synthesizer folks, introduced two new accessory items for synthesizer players. The first of these is the Model 700 Programmer, a 64-memory unit which is intended for use with any synthesizer that has provision for control voltage interface to give fully programmable operation. The Model 700 has two five-function envelope generators (Delay, Attack, Decay, Sustain and Release) with Amount controls to set the envelope amplitude, and three general-purpose voltage sources. All of these controls are, of course, programmable in that their settings can be stored in any of the 64 memories for later instant recall. Envelope 1 is intended for VCA control while Envelope 2 is intended for VCF control and has a programmable Offset control to set the filter's initial frequency. The three voltage sources are intended for oscillator tuning, and have a five volt range quantized in semitone steps for more stable tuning. The other new product from Sequential Circuits is the Model 800 Sequencer, a keyboard-programmable digital sequencer capable of storing up to 256 notes divided into sixteen banks of 16 notes each. Each memory bank has a toggle switch for playback selection and an LED indicator to display the currently selected bank while the note within the bank is displayed on a numerical LED display. Besides conventional programming, individual notes may be reprogrammed without disturbing the remainder of the sequence, and the timing of a sequence may be programmed or reprogrammed separately from the pitch. When playing back, the sequence may be played back at normal speed, half speed or double speed by flipping a switch, or it may be played back in the variable speed mode which has a range from 1/15 to 15 times normal speed. The sequence may also be stopped and started at will or played back one note at a time.

CIRCLE 18 ON READER SERVICE CARD

GUITARS AND BASSES

Bunker Systems, Inc. offers a guitar and a bass guitar of a very original design. Bunker's theory is that the sustain and sound of a guitar depend primarily on the neck, and its guitars are designed around a unique neck design in which the strings are anchored to a spring steel bar which supports them totally while the wood of the neck and fingerboard are not stressed and make no contribution to the structural strength of the neck. Another unique aspect of the Bunker design is that the strings are anchored at the head stock end of the guitar while the tuning machines are mounted on the end of the body; this is said to have certain mechanical advantages as well as making tuning easier, especially while playing. The strings are supported by a brass nut on one end and a massive brass bridge and guide piece at the other end for the utmost in sustain. Bunker guitars use two humbucking pickups and special electronics to produce a very wide range of tonal colors in the output.

CIRCLE 19 ON READER SERVICE CARD

There are two new additions to the Gretsch Beast guitar line by Kustom/Gretsch. The first of these is the BST-1000, a mahogany-body, maple-neck, single-cutaway guitar of all-new design. The all-new electronics feature two humbucking pickups, pickup selector switch and volume and tone controls. A new, fully adjustable bridge with through-the-body string mounting is used for excellent sustain, and the rosewood fingerboard is fitted with 24 frets. The other new Beast from Gretsch is their top-of-the-line BST-5000 which has several unique features. The neck of the BST-5000 is a maple-walnut-maple laminated design and extends the full length of the body for maximum sustain; it is reinforced with a unique, patented, geared truss rod. The body of the guitar is made of walnut laminated onto the straight-through neck piece, and is a double cutaway design for maximum playing access to all 24 frets. On the electronic side, the BST-5000 uses two special humbucking pickups of low impedance design for minimum treble loss and hum pickup. Separate volume and tone controls are provided for each pickup along with a pickup selector switch and a special pull switch which disconnects one coil of each pickup for the thinner, single coil sound instead of a humbucking sound.

CIRCLE 20 ON READER SERVICE CARD

A fresh name on these pages is that of the Matao Corporation, which offers a full line of acoustic and electric guitars. On the acoustic side, Matao offers some ten models of classic style guitars ranging from a low-cost, three-quarter-size student model up to a beautiful, Rubio-style model with hand-selected, book-matched, German spruce top, and ten folk and western style steel string models including a twelve-string and the top-of-the-line model MW-45H dreadnaught which features a book-matched, three-piece Jacaranda Rosewood back, fancy decorative bindings and neck inlays, and a special interior bracing pattern for uniform response. On the electric side, Matao offers Les Paul- and Strat-style guitars and Fender-style basses in addition to their Artist Series guitar and bass models of original design. Of particular interest are the ME2000 guitar and ME2001 bass which offer sophisticated electronics inside the guitar. Both the guitar and the bass are two-pickup instruments with twin-winding pickups, and each instrument has a five-position preset selector switch which chooses various configurations of single-coil and humbucking connections and in- or out-of-phase pickups to effectively simulate the characteristic sounds of several familiar guitars. These presets may additionally be changed using five trimmers which are accessible from the rear of the guitar if the musician has some other basic sound in mind. Both the guitar and bass have active electronics built into the instrument including active high-frequency and low-frequency equalization affecting both pickups and three-position effects switch. The Matao ME2000 and ME2001 are powered by an internal 9-volt battery or by an external power supply if the 3-pin XLR-type connector is used for the guitar's output rather than the standard ¼" phone jack also provided.

CIRCLE 21 ON READER SERVICE CARD

Erlewine Fine Electrics produces two premium-quality electric guitars: the Erlewine, a flat-top model, and the Automatic with a deep-dished carved top in the 50s tradition. Both models have single humbucking pickups and heavy chrome-plated hardware. Both models have neck and body of Honduras mahogany with a laminated top of flame maple finished in a subtle sunburst style. The neck is dovetailed into the body for improved sustain, has all six tuning machines on the top of the headpiece for tuning ease and has an adjustable truss rod for reinforcement.

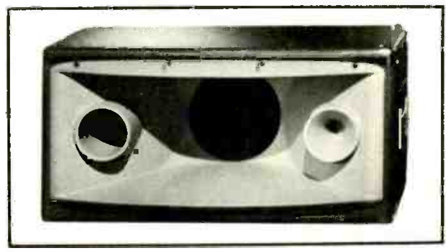
CIRCLE 22 ON READER SERVICE CARD

MUSICAL INSTRUMENT AMPLIFIERS

Acoustic Control offers two new guitar amplifiers and a new bass amp. The Model 123 is a 125-watt RMS combo guitar amp with a single 12-inch speaker in an open-back cabinet. The 123 has a preamp gain control plus a drive control which is enabled remotely by a footswitch; an LED is provided to indicate when the drive control is enabled. Other controls on the 123 include a bright switch, bass, mid, treble and presence controls, reverb level and master volume, plus a five-band graphic EQ as is customary on Acoustic amps. The graphic EQ and the reverb are also controlled by footswitches along with the drive function. The Model 121 has 125 watts of power driving a single 15-inch speaker and does not have the drive control but is otherwise identical to the 123. The Model 122 is also a 125-watt amplifier, but is designed for bass use with a 15-inch speaker in a bass reflex cabinet. Low and high gain input jacks are provided, and other features include volume, bass, midrange and treble controls, a bright switch, a five-band graphic EQ, an effects send/return patching loop between preamp and power amp and a power boost switch. The same circuitry used in the 122 is also available in amplifier-head-only form as the Model 120.

CIRCLE 23 ON READER SERVICE CARD

Gollehon is now offering the Penetrater 3—a full-range loudspeaker. The unit contains a new 10-inch speaker to drive a radial horn, while utilizing a radical design whereby a ducted vent and high-frequency horn are molded in-



to the mid-horn's side walls. The manufacturer states that: "the P3's greatest asset is the smoothness of high level generated by the mid-range horn." The Penetrater may be used as a full-range speaker P.A. for clubs, etc., or solely as a top section for mid- and high-end response when incorporated with separate bass cabinets. Frequency response is listed as 70-20 kHz; sensitivity at 1 watt, 1 meter is 104 dB; dispersion is 60° horizontal, 40° vertical.

CIRCLE 24 ON READER SERVICE CARD

A high-power, direct-coupled bass amp head and two complementary speaker enclosures are the latest from Gallien-Krueger. The 400B is rated at 200 watts RMS into 4 ohms at less than .1% distortion from 20 Hz to 20 kHz. The FET preamp circuitry has normal and bright input jacks, and switches for a 10 dB attenuator, contour on/off and bright on/off. Controls are provided for preamp volume, bass, lo-mid, hi-mid and treble EQ, boost (controlled by a footswitch) and master volume, and two LEDs indicate power on and power amp clipping. To handle the power of the 400B, G-K offers the 4412H, a two-way, horn-loaded cabinet rated at 320 watts of power handling and the 215B, a labyrinth-loaded 2 x 15" cabinet rated at 250 watts. The 4412H uses four 12-inch speakers in a pair of folding horns plus two direct-radiating 10-inch speakers in a sub-enclosure for added bite.

CIRCLE 25 ON READER SERVICE CARD

Pignose Industries has added to its line with two new, medium-sized amplifiers called the 150R Crossmix series. These amps have two complete preamp channels which are remotely switchable from a footswitch operating silent FET switches. Channel 1 is the more flexible channel with a preamp gain control with a pull switch for boosted gain, passive bass and treble and active midrange EQ, a reverb send control and an external effects patching loop. Channel 2 is a somewhat simplified version of the Channel 1 circuitry featuring preamp gain (without boost switch), passive bass and treble EQ, reverb send and effects send/return loop. Additional controls include master volume and master reverb level controls. On the power side, the 150R Crossmix amps have 75 watts of power and deliver it into either a single 12-inch or twin 10-inch speakers.

CIRCLE 26 ON READER SERVICE CARD

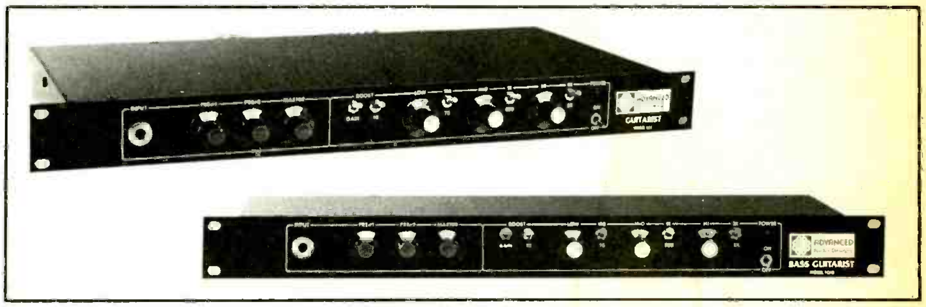
MUSICAL INSTRUMENT ACCESSORIES

DiMarzio Musical Instrument Pickups, Inc. now has three new pickup models along with its clever Big Amp pocket amplifier. The Big Amp is a twin-speaker, battery-powered, miniature amplifier which is styled just like a conventional twin amp, but small enough to fit in a pocket or the string compartment of a guitar case. The unit has a volume control, and on/off switch and a pilot light, and produces an amazing amount of sound considering its size, but if more sound is needed the musician can hook up Big Amp's line output to a larger amplifier. The new DiMarzio pickup models include the DLX-1, a direct replacement for miniature humbucking pickups which features much the same sound as the DiMarzio Super Distortion pickup and which is pre-wired with a four-conductor cable for any wiring configuration. The VS-1 is a direct physical replacement for Stratocaster pickups and produces the classic 50s single-coil sound with an upper mid-range bite achieved through calibrated magnets and precision coil-winding. Finally, the TDS-1 is a tapped single coil pickup which can be wired to a switch to allow selection of the full, high output of the pickup or a slightly lower level but brighter sounding output. The TDS-1 is fully shielded, comes with a shielded output cable and has adjustable pole pieces.

CIRCLE 27 ON READER SERVICE CARD

The Model 101 is a compact, rack-mount instrument preamplifier from Advanced Audio Designs. The unit has a single input with two series-connected preamp gain stages which may be foot-switched with an accessory switch unit for total control of overdrive and harmonic distortion characteristics; a master volume control is provided in addition to the two preamp gain controls. The Model 101 has comprehensive equalization with low-frequency and high-frequency boost switches and low-, mid- and high-frequency equalization. The EQ section allows 15 dB of boost or cut at switch-selectable frequencies of 125 or 250 Hz for bass, 1 or 2 kHz for midrange and 3 or 4 kHz for the treble range. Also available from Advanced Audio Designs is the Model 101B preamp which is designed specifically for bass guitar and has EQ frequencies approximately one octave lower than the standard Model 101.

CIRCLE 28 ON READER SERVICE CARD



D'Addario has long been an innovator in the field of guitar strings, and now they have introduced a new line of strings wound with a specially tempered 430 stainless steel alloy for longer life. This particular alloy is said to yield crisp overtones, perfect string-to-string balance and flawless intonation when wound with D'Addario's computerized winding equipment. Gauged sets are available for guitar in light (.010 top), super light (.009) and extra-super light (.008), plus a light-top/heavy-bottom set, and bass guitar sets are available in super soft, soft, regular and heavy for long scale basses and in soft only for medium and short scale instruments.

CIRCLE 29 ON READER SERVICE CARD

Sescom, Inc. is a familiar name to many musicians for the extensive line of transformers and direct boxes the company makes. Sescom has made several additions to its line of circuitry products. The SH-100 is an AC-powered 10-watt per channel stereo headphone amplifier which can be used as a practice amp or small monitor amp. The SH-100 becomes even more useful with the addition of a SHB-100 headphone splitter box which will accommodate up to six pairs of headphones. The PM-100 is an AC-powered three-band parametric equalizer which can be used directly with musical instruments thanks to a built-in preamplifier circuit. Each band of the unit has sweepable frequency, boost/cut, and bandwidth controls, plus there is an overall level control. The three frequency controls have ranges from 20-230 Hz, 200-2300 Hz and 2-23 kHz. The PC-100 is a battery-powered compressor housed in a compact, die-cast box with a bypass footswitch. Two controls are furnished on the unit, one for the amount of compression and the other for output level. The last new product from Sescom is the LD-100 noise gate/line driver, which is also battery powered and encased in a die-cast box. Controls are provided on the unit for gate threshold and for the amount of gating; this latter can be used to nullify the effect of the gate since there is no bypass switch on the unit. A slow/fast switch is provided to control the reaction speed of the gate, and two LEDs indicate whether the gate is open or closed.

CIRCLE 30 ON READER SERVICE CARD

Only Custom-Tailored Sound meets your taping needs.

If tape is the only sound that's right for you, to maximize your taping requirements, an ADC Sound Shaper® Two MK II frequency equalizer is a must.

When they designed the Sound Shaper Two, they had you in mind. Because, aside from being a superb all-around equalizer, it lets you work with tape the way you want. For example, now you have two-way tape-dubbing capability, a feature many receivers don't offer. You can "custom-tailor" a record and then record it the way you would have engineered it. And that includes your golden oldies because, with the Sound Shaper Two, you can virtually eliminate the surface noise which has accumulated over the years.

The entire ADC Sound Shaper line is impressive. The basic Sound Shaper One is a great introduction to frequency equalizers. And the top-of-the-line Sound Shaper Three, the *Paragraphic*™ equalizer, combines the ease and control of a graphic equalizer with the precision and versatility of a parametric. And, all Sound Shaper equalizers, except the Sound Shaper One, feature LED-lit slide controls, allowing for visual plotting of the equalization curve.

With the Sound Shaper Two MK II, you can appreciate the difference custom-tailored sound makes—over and over again.

Custom-Tailored Sound

ADC
A BSR COMPANY
BSR (USA) Ltd., Blauvelt, NY 10913.
BSR (Canada) Ltd., Rexdale, Ontario.

CIRCLE 155 ON READER SERVICE CARD

Another Look at Recording BIAS

By Peter Vogelgesang

Perhaps the least understood aspect of magnetic sound recording is the function of bias in the recording process. All persons skilled in magnetic recording know precisely how bias is adjusted, and they extract maximum performance from their equipment through long association and experience with recorders of many kinds. But not all understand how bias works.

The model most commonly used to explain bias depicts an audio signal carried into the linear region of a DC magnetization curve on the peaks of the bias waveform, as shown in *Figure 1.* In reality, it can be shown, however, that the DC magnetization curve of a tape is not the transfer characteristic of the tape. Also, this model would mislead one to believe that distortion recurs as bias amplitude exceeds an optimum level. These false assumptions give the model limited value.

A second model which has been proposed for the recording process is the so-called "bubble model" which treats recording primarily as a geometrical process[1]. This model asserts that the bias frequency lays down a series of overlapping cylindrically shaped magnetized regions in the tape. It also implies that the bias frequency must be recorded. But audio frequencies can be recorded using a bias frequency so high that the bias waveform does not appear on the tape even when the tape is played back at very low speeds.

The most rigorous but technically accurate model employs the hysteresis loop of a tape to explain linear magnetization. But, this model is difficult to visualize and comprehend.

A model of bias recording is needed which is both technically correct and easy to understand. The model presented in the following paragraphs contains no new information; it is merely a different way of looking at what is already known. Hopefully it will be both interesting and enlightening.

The Nature of Magnetic Particles

The starting point in understanding bias is an examination of the nature of a particulate magnetic material such as the oxides or metal particles used in magnetic tape. These particles are little needle-shaped magnets so small they cannot clearly be seen with our most powerful optical microscopes and must be examined with electron microscopes to reveal significant detail. Before being mixed into solution in preparation for making magnetic tape, this magnetic material is an extremely fine powder which sifts through the tiniest cracks and can be carried everywhere in the air.

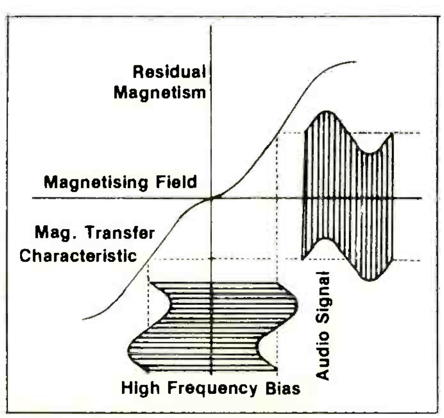


Fig. 1: The model most commonly used to explain the function of recording bias.

Each tiny magnetic particle contains only a single magnetic domain; that is, the crystalline molecular structure that gives rise to magnetism exists in only a single, highly organized pattern throughout the length and breadth of each particle. This feature of a single domain particle contrasts with, say, a piece of iron which may contain thousands of crystalline structures and thousands of domains. Under the influence of an external magnetic field the domains in iron can grow large and small, strong and weak, and can shift about in the iron[2]. But domain size and magnetic strength remain constant in the single domain particle.

Each particle therefore produces a field of constant magnitude. If a particle is brought under the influence of an increasing external field of opposite polarity, at a certain critical intensity the particle is observed to "switch," or, in other words, the field generated by the particle is reversed. But the magnitude of the reversed field is always equal to the field which existed in the previous polarity state. Thus magnetic particles justifiably can be thought of as tiny binary magnetic switches which exist always in one of two magnetic states. Just as a certain amount of finger pressure is needed to flip a toggle switch, a certain amount of "magnetic pressure" is required to cause a single domain particle to reverse magnetic polarity. The field intensity at which a particle switches is called the "coercivity" of the particle, and it is measured in Oersteds. Particles of iron oxide used in magnetic tape generally have an average coercivity of 300 Oersteds.

Not all particles have the same coercivity in a batch of magnetic material, and the population of coercivities within a batch follows the familiar bell shaped curve shown in *Figure 2.* This curve is termed the "distribution curve" of a magnetic material. When a magnetic tape is said to have a certain coercivity, the value stated pertains to the peak of this distribution curve. Only a small percentage of particles has the specified coercivity; nearly fifty percent has a higher coercivity and the other fifty percent has a lower coercivity.

In the tape manufacturing process magnetic particles coated onto the plastic backing are passed through a strong magnetic field before the coating is allowed to dry. Still in a fluid state, the suspended particles are physically rotated so, like a multitude of compass needles, they all lie in a direction parallel with the length of tape. They remain permanently locked in this position when the coating is dried. In this state the fields generated by the particles also point in the same direction, and so add and subtract more perfectly when the tape is magnetized during recording. An "oriented" tape, as it is called, can produce about 3 dB greater output than an unoriented tape.

Distribution Versus Magnetization

A magnetic tape in an erased state has one-half of the particles magnetized in one polarity and the other half in the opposite polarity. Because these oppositely polarized particles are intermixed and closely packed, the tiny fields from the particles cancel each other. In this state a field external to the tape cannot be measured, so the

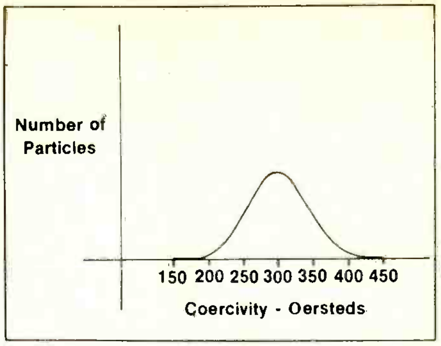


Fig. 2: Distribution of coercivity in a particulate magnetic material.

tape is said to be demagnetized. It is important to recognize, however, that while the individual particles remain magnets even in an erased tape, no external field is present because of the 50-50 ratio of particle polarization.

Suppose a short length of erased tape is slipped inside a coil of wire and a magnetic field is generated by passing an increasing direct current through the coil. How does the magnetic material respond to an increasing magnetic field? The response is illustrated in *Figure 3.*

Magnitude of the applied field is shown along the H axis, and the magnitude of the field produced by the magnetic material in the tape is shown along the M axis. Starting with zero applied field, the magnetic material is demagnetized (50-50 ratio) and no external field exists. The applied field begins increasing, but the magnetic material does not respond because none of the particles has a coercivity low enough to be switched. Soon the applied field reaches a magnitude equal to the low end of the distribution curve, and particles with this low value of coercivity begin to switch. Magnetization of the material starts. When the applied field reaches a magnitude equal to the center of the distribution curve, a maximum number of particles are switched at this point and the magnetization curve has the greatest slope. Then, as the population of particles with higher coercivity begins decreasing, magnetization also increases more slowly causing the magnetization curve to level off. When all the particles are switched in the same polarity, no amount of increase in the applied field will increase magnetization of the material. The saturation level has been reached.

It should be remembered that through this entire magnetization process only one-half of the particles were switched. The other half already existed in the final polarity state.

It can be seen that the initial magnetization curve is very non-linear, so a magnetic field from a recording transducer will not magnetize the tape in proportion to the applied field. In other words, magnetization of areas of the tape cannot be proportional to the instantaneous sound levels which caused their magnetization. Distortion re-

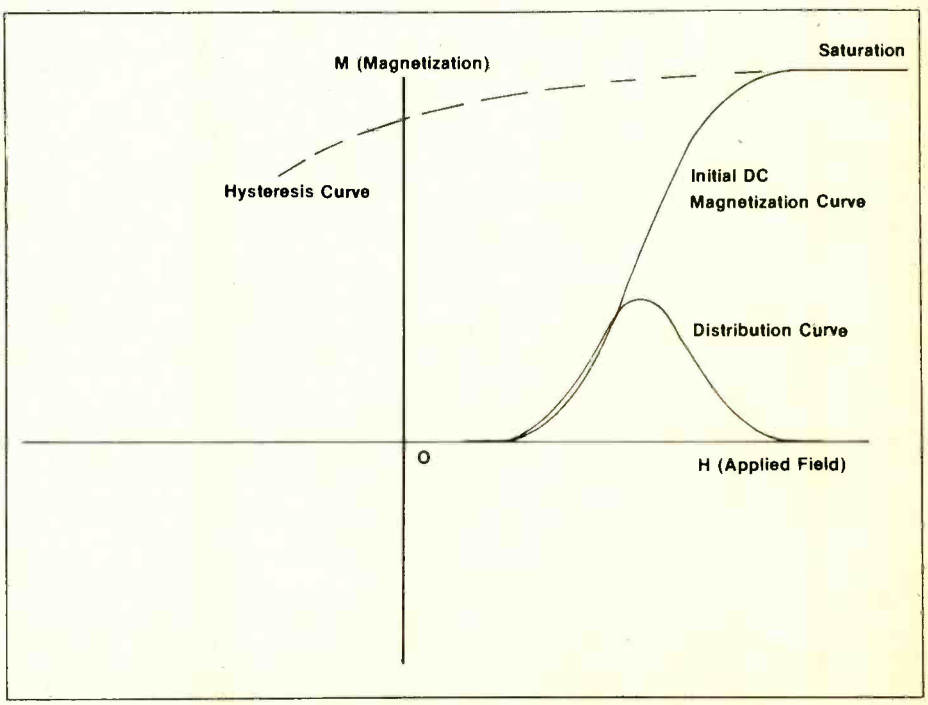


Fig. 3: The initial DC magnetization curve derived from distribution.

sults. Magnetic tape would not be a suitable medium for analog recording of sound if it were not for the fact that bias greatly reduces this non-linearity.

Returning to *Figure 3*, having reached saturation from a demagnetized state, the material follows a characteristic hysteresis curve if the applied field is diminished and then reversed. The hysteresis curve is most often used in explaining the effects of recording bias. But wait. Note that the hysteresis curve is the integral of the distribution curve. Why use the integral? Well . . . let us momentarily forget the hysteresis curve and deal directly with the distribution curve.

Erasure and Recording—Same Thing

A tape can be erased by passing it over an erase head or it can be erased by recording it with no audio signal. What is the difference? None. To see how bias linearizes the magnetization curve of a tape, you must first examine the erasure process.

Figure 4 shows the distribution curve of a typical tape, and it also shows an erasure field which decreases linearly with time. This erasure field could originate from an erase head, recording transducer or even from a bulk degausser. In the case of bulk degaussing, the erasure field diminishes with time because the degausser is slowly withdrawn from the vicinity of the tape reel or cassette. The erasure field from an erase head diminishes slowly because any given area of tape moves away from the head gap at a rate which is slow compared to the erase frequency.

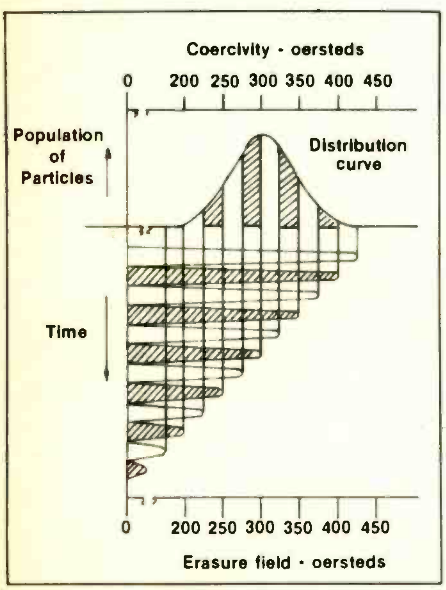


Fig. 4: Relationship of a distribution curve and an erase field.

Normally, the negative half-cycles of the erasure field would be shown to the left of the M axis in *Figure 4*, but for convenience of explanation the negative half-cycles have been folded over and shaded so they can be compared with the positive half-cycles.

The erasure process begins with the first positive half-cycle reaching a magnitude which causes all the particles within the distribution curve to be switched in the positive polarity. At the end of this first half-cycle the magnetic material is saturated positive. The first negative half-cycle reverses polarity of all the particles, with the exception of a very few high coercivity particles at the upper end of the distribution curve. These particles remain switched in the positive direction. The second positive half-cycle once again reverses most of the particles within the distribution, but leaves untouched the upper end of the distribution curve, part of which is positively switched and part of which is negatively switched.

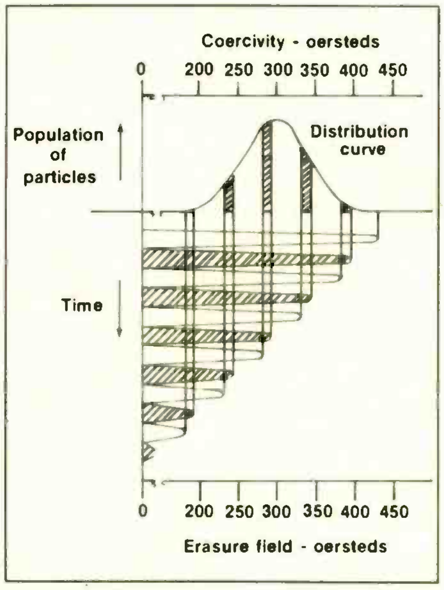


Fig. 5: An erase field with "offset".

This process continues as each half-cycle diminishes in magnitude until finally the erasure field has grown so weak that it does not equal the coercivity of the bottom-most particles of the distribution curve. Observe that at this point one half of the particles throughout the distribution is switched positively while the other half is switched negatively. This condition causes the fields from the particles to cancel, and since no external field can be detected, the tape is erased.

Ideal erasure takes place when a sufficient number of half-cycles is used to obtain a statistical average of 50-50 in particle polarization. Too few half-cycles can leave an imbalance in the ratio, which, of course, leaves the tape with a residual level of magnetization.

From this simple analysis it can be seen why an erasure field must be symmetrical and free of distortion (equal amplitude positive and negative half-cycles). It also shows why the erase field must decay to a low level before the field is terminated. Persons accustomed to using hand-held degaussers know that the energizing button must not be released until the degausser has been moved away from the tape. If the degausser field is terminated before the field though the tape is almost zero, then the series shown in *Figure 4* is stopped partway through, and the tape is left with a lower portion of the distribution curve switched in a single polarity.

Linearization Using Bias

An identical analysis explains how a bias field at the recording transducer linearizes the magnetization curve of a tape. *Figure 5* shows the difference in magnetization which occurs when an "offset" is introduced in the relative magnitudes of the positive and negative half-cycles of the bias field. The figure illustrates positive half-cycles greater in magnitude than negative half-cycles, producing a distribution which is magnetized predominantly positive. Predominant negative half-cycles will result in a negatively magnetized tape.

Magnetization of the tape by this means is proportional to the offset in the bias waveform. In other words, the magnetization curve of the tape has been made linear with offset amplitude.

Figure 6 shows a mixed bias and audio waveform which typically appears at the terminals of a recording transducer. In this case the audio signal merely provides the momentary offset for the bias waveform. Since the offset determines the level of tape mag-

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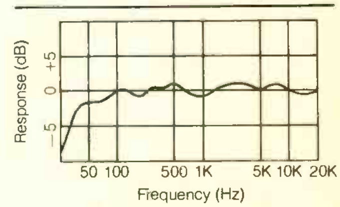

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netization, the tape is magnetized proportionally to the instantaneous magnitude of the audio waveform.

The bias frequency must be high relative to the highest audio frequency being recorded so that an accurate statistical average can be developed for each of the shortest possible audio signal components. If the upper audio frequency is 20 kHz, then a bias frequency of 200 kHz will produce ten polarity reversals for each cycle of the highest frequency recorded. Contrary to a view held by some, it is not necessary that the bias frequency itself be recorded on the tape.

The foregoing analysis shows why the bias level in a recorder must reach a certain magnitude before distortion is reduced. If the peaks of the bias half-cycles do not switch the highest coercivity particles within the distribution curve, then linearization of the magnetization curve is not possible because the magnetic particles in the upper part of the curve are not affected. The analysis also shows why overbias of a tape does not affect the shape of the magnetization curve sufficiently to cause an increase in distortion. If distortion in a recorder increases markedly with excess bias amplitude, the cause is more likely an insufficient reserve of undistorted bias drive.

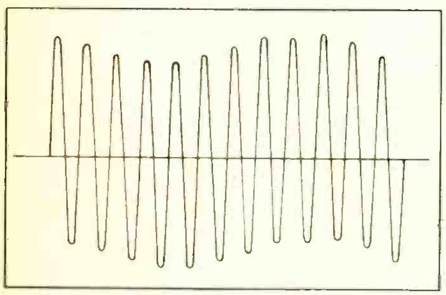

Fig. 6: A composite of the audio and bias waveforms where the audio signal provides "offsets" for the bias waveform.

The magnetization curve of a tape recorded with bias is called a "modified anhysteretic magnetization curve." The term "anhysteretic" refers to the elimination of hysteresis, and the term "modified" is used to convey the fact that the bias field and the audio field diminish together as recorded areas of tape move away from the gap of the recording transducer. (An anhysteretic curve normally is developed with a fixed signal level and a decaying bias field.)[3]

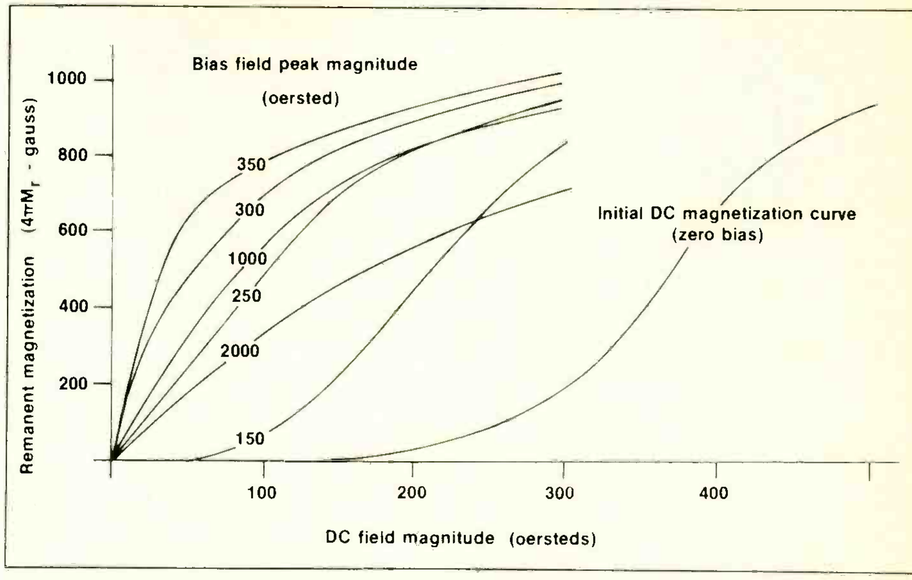

Fig. 7: A family of modified anhysteretic magnetization curves.

A family of modified anhysteretic magnetization curves is shown in *Figure 7* for different levels of bias. The idea of using bias, of course, is to obtain tape magnetization which is precisely proportional to the magnitude of the applied audio field. This is obtained only when the magnetization curve is a straight line. Note that the very non-linear DC magnetization curve is converted by bias to a comparatively straight line. Once converted, the line remains straight even though bias is increased well beyond an optimum level. Excess bias merely decreases the slope of the curves. Insofar as distortion is concerned, a recording system is forgiving of excessive bias but demands a certain minimum bias to produce a low distortion recording.

Interim Summary

To summarize thus far, a tape is magnetized by switching a multitude of "binary" magnetic switches to a preferred polarity. Magnitude of magnetization is established by the ratio of particles of one polarity to those of the opposite polarity. The DC magnetization curve of a magnetic tape, which is the integral of the coercivity distribution curve, is very non-linear. Severe distortion results from an attempt to record sound using the DC magnetization curve. A high-frequency bias field produced at the gap of the recording transducer linearizes the magnetization curve by developing statistical averages of particle polarities which correspond to instantaneous magnitudes of the audio signal. This "modified anhysteretic magnetization curve" remains linear even in an overbiased condition, with the slope of the curve decreasing as bias is increased.

The Complications of Field Geometry

To this point the analysis has assumed that all the magnetic material in a tape experiences the same magnitude of bias field. Indeed, the magnetization curves of *Figure 7* were obtained by measuring tape samples inside a large solenoid coil which produced uniform field strength over the entire length of the samples.

We usually think of a tape as being so thin that little variation in field intensity through the thickness can exist; but if a scale drawing is made of the cross-section of a tape as it relates to the length of the gap in a recording transducer (as shown in *Figure 8*), it is easily seen how a substantial variation occurs. The field standing out from the gap of a recording transducer is semi-cylindrical in shape, and is also very much more intense at the surface of a tape than at a point deeper in the magnetic layer[4]. In a practical recording system the surface of a tape will be overbiased while the deepest region experiences a less than optimum bias level.

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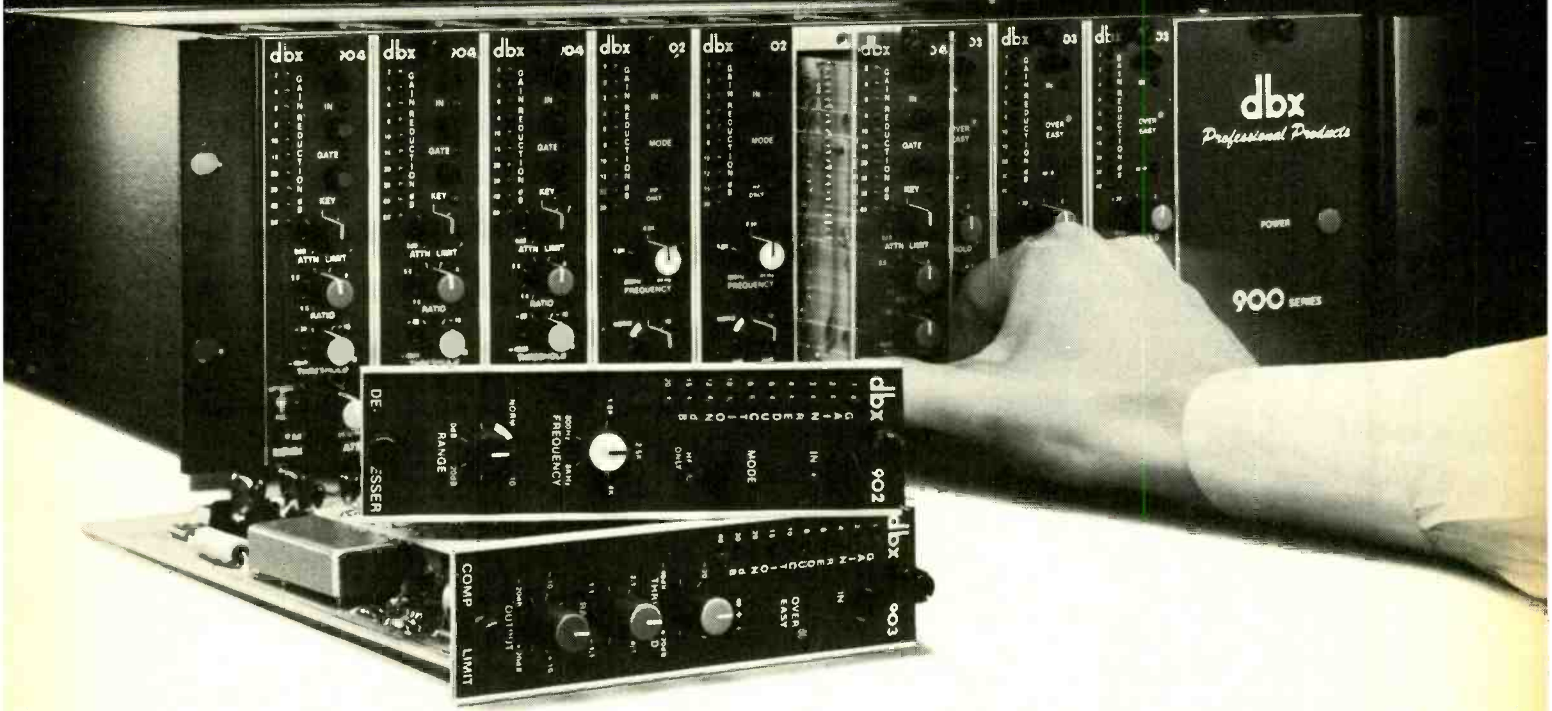

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cording system to obtain minimum distortion, it is necessary that the peak amplitude of bias through the *full depth* of the magnetic coating be sufficient to switch the highest coercivity particles in the tape. Distortion will originate from the deepest region of the magnetic coating if this condition is not obtained because the full depth of the coating will be operating on a curve akin to the DC magnetization curve.

If we imagine that the magnetic coating of a tape is divided into layers that are, say, ten millionths of an inch thick, each successively deeper layer experiences a lower level of bias field amplitude. Consequently, each imaginary layer operates with a different magnetization curve. Because the bias field must be adjusted to a level which linearizes the magnetization curve of the deepest layer, the upper layers are always over-biased and will operate with the straight but more sloping magnetization curves shown in *Figure 7.*

The magnetization curve of the entire thickness of tape, then, is not any of the curves illustrated, but is a composite of all the adequately biased and over-biased curves. To determine the level of magnetization of a tape for any magnitude of applied signal field, one must determine the magnetization of each layer by using the appropriate curve, and then sum the magnetizations of the layers. (This kind of analysis is too complex to perform except by computer simulation.)

The problems of geometry occur not only in the recording process, but are manifested also during playback of a recorded signal. The "spacing loss" equation[5] is a well known equation for determining the amount of signal loss during playback which occurs as a result of imperfect contact between the tape surface and the pole pieces of the playback transducer. Imperfect contact reduces magnetic coupling between transducer and tape by introducing the high reluctance of air (or other material causing the space) in the magnetic circuit. Loss is wavelength dependent, and while a small spacing is not significant at long wavelengths (low frequencies) the same spacing can produce excessive signal loss at short wavelengths.

Going back to the imaginary layers through the thickness of a tape, the top-most layer in contact with the surface of the playback transducer produces a spacing loss for the second imaginary layer. Consequently the second layer contributes less to the playback signal than its level of magnetization would otherwise provide. Output of the third layer is even more attenuated, and attenuation increases as the layers go deeper into the tape.

Since attenuation is dependent upon recorded wavelength, it can be seen that the whole thickness of a tape can contribute to long wavelength output, but only the surface of the tape can contribute significantly to short wavelength output. This fact can be appreciated by choosing two widely separated wavelengths, selecting a separation of small dimension, and then computing the loss for the two cases, as shown in Table I.

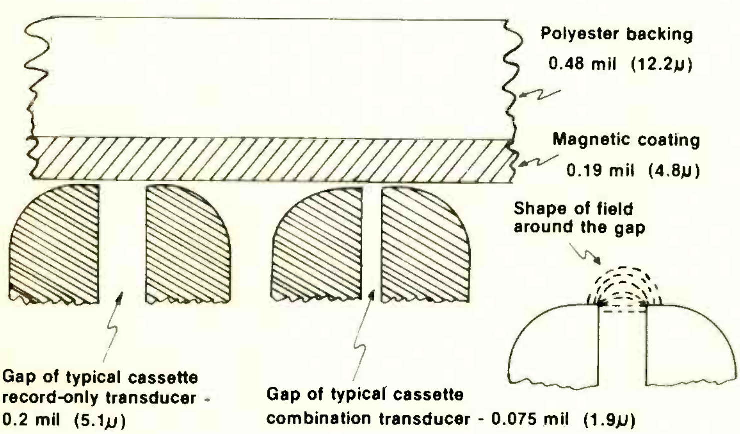

Fig. 8: Scale drawing of tape thickness and transducer gap.

We saw previously that each imaginary layer operates on a different modified anhysteretic magnetization curve of *Figure 7* because the bias field intensity diminishes as it penetrates deeper into the magnetic coating. Combining this fact with the realization that only the surface of a tape contributes to high frequency output, it can now be seen why a single setting of bias level cannot obtain simultaneous short and long wavelength optimization. If the bias level of a recorder is slowly increased, a short wavelength signal quickly reaches the magnetization curve of steepest slope and greatest sensitivity (see *Figure 9*). Distortion at long wavelengths is still high at this point because the lower frequencies are operating on the under-biased curves. Then, as the bias level reaches a point where the deeper regions of the magnetic layer are optimized, distortion diminishes, long wavelength sensitivity increases, but short wavelength output falls because the surface of the tape is operating on a magnetization curve of lesser slope. At a point where third order harmonic distortion is at a minimum, short wavelengths are over-biased and high frequency output is considerably beyond the point where it peaks.

The inability to obtain simultaneous bias optimization of short and long wavelengths is one of the frustrations of magnetic recording, and the problem is caused simply by the three-dimensional geometry of the recording and playback processes.

New metallic particle tapes derive much of their superior performance by obtaining a better compromise between short and long wavelength bias

	Wavelength	Frequency	Loss (dB) (loss $= 54.6 \frac{d}{\lambda}$)
#1	0.0001 inch	18,750 Hz at 1⅞ ips	5.4
#2	0.01 inch	187 Hz at 1⅞ ips	0.054

Table 1. Signal Loss vs. Wavelength for Spacing of 0.00001 Inch.

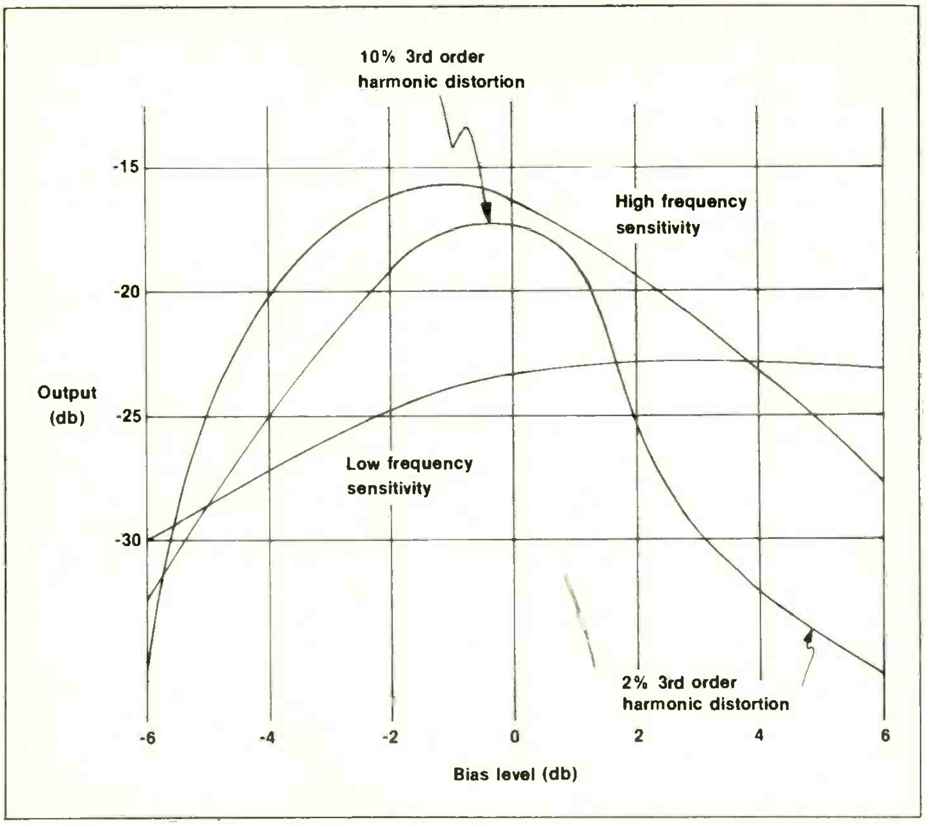

Fig. 9: Distortion vs. high and low frequency sensitivity for a typical magnetic tape.

optimization. The metallic particles have a higher value of retentivity than oxides, and a thinner coating of material will produce an output equivalent to thicker oxide coatings. The resultant reduction in coating depth alleviates the geometry problem, and optimum bias levels for short and long wavelengths are brought closer together. Greater high frequency output results, and signal-to-noise ratio is improved when post-emphasis is reduced.

Final Summary

Linearization of the magnetization curves of magnetic tapes is achieved by using bias at the recording transducer. Such linearization occurs as the result of developing statistical averages of particle polarities for each instantaneous level of the audio signal. The resultant magnetization curves are "modified anhysteretic magnetization curves," and these curves are easily measured by stimulating tape samples in uniform magnetic fields. But since recording and playback are three-dimensional processes, an accurate analysis cannot be made without taking into account the distribution of field intensity through the thickness of a tape. Such an analysis is made convenient, both for visualization and for measurement, by imagining the tape being composed of thin layers of magnetic material. The three-dimensional nature of the recording and playback processes explains the problem of short and long wavelength bias optimization occurring at different bias levels.

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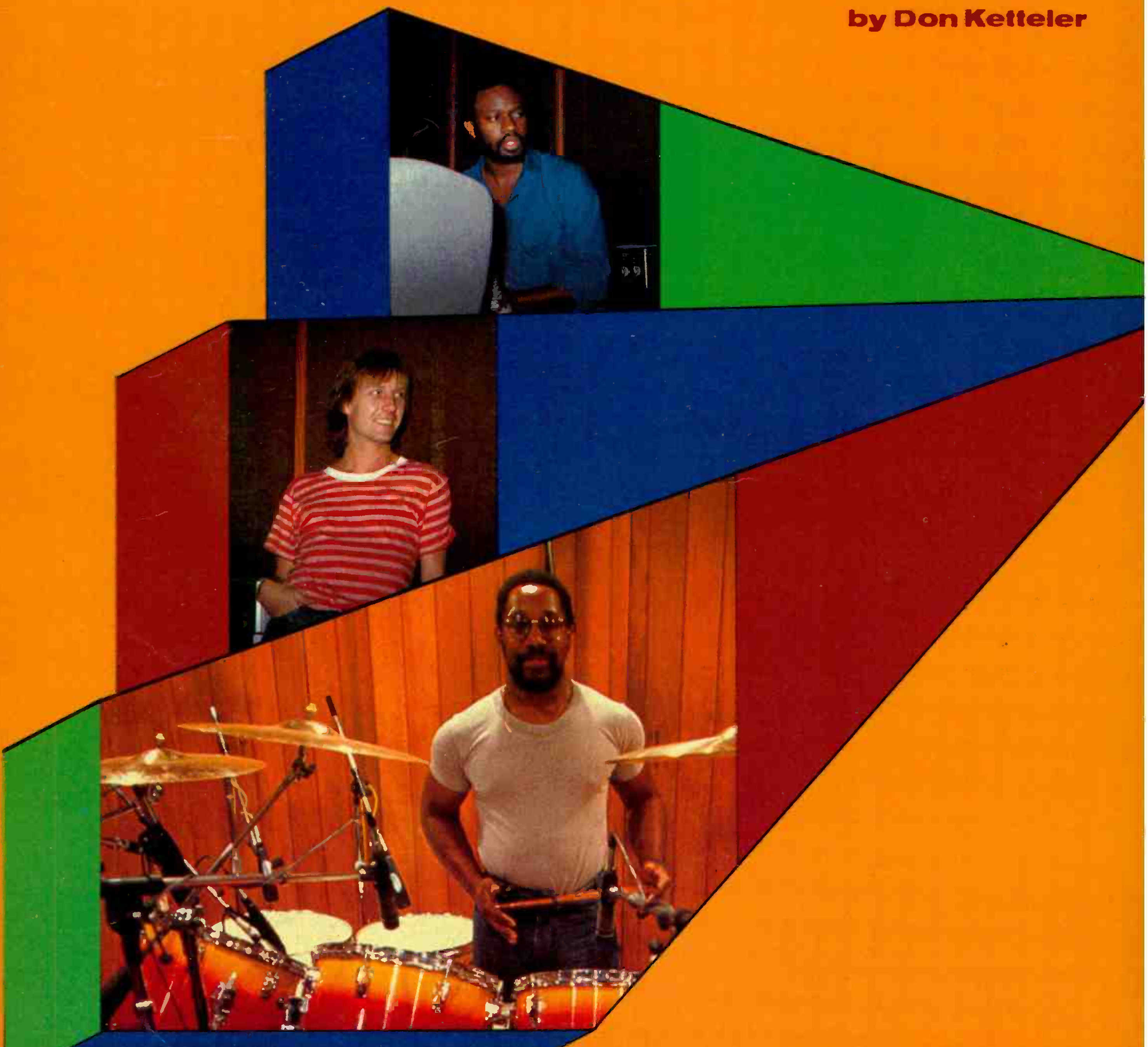
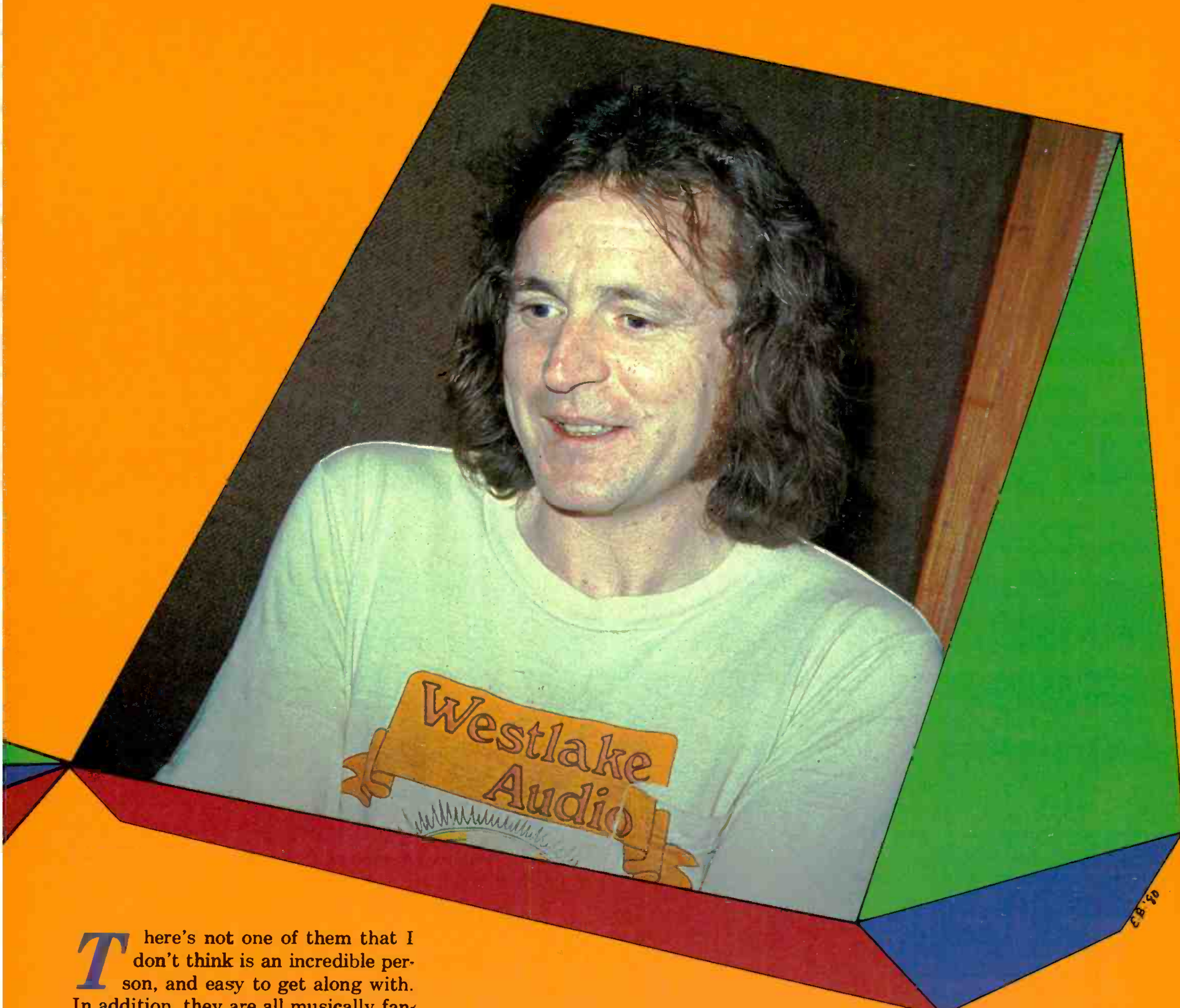
a session with Jack Bruce & friends

by Don Ketteler





There's not one of them that I don't think is an incredible person, and easy to get along with. In addition, they are all musically fantastic. You could tell any one of them to bring [his levels] up or down, or change his technique, in any section of a song. It was like pushing a fader up or down, or tweaking a tone control on the console. You ask them to do something, and it's programmed into them forever. It's a fantastic advantage to have that sort of control as a musician."

Just a couple of sentences of reverent testimony from Stephan Galfas, engineer of the Jack Bruce and Friends album, *I've Always Wanted To Do This*, as to his impressions of this relatively new group of experienced and certainly highly respected musicians: Jack Bruce, Billy Cobham, Clem Clempson and David Sancious.

To devote space to the abilities and backgrounds of these gentlemen would at once be both laborious and redundant to any but those, as the saying goes, born yesterday—either figuratively or literally. Let comment begin and end with saying that it would be difficult to conjure four individuals whose talents and roots touch so close to the essential bedrock of their respective musical idioms. Their various and distinctive work, to a great extent, has been both critical, and perhaps in the short life span of rock and roll, pivotal in establishing the references for the integration of music with technology and technique. In short, they've been around.

You can figure out for yourself just what all the above means, but it has not been stated idly. If this were still the era of "super groups," this would be one, and deservingly so for numerous reasons. Amen.

For now, since this prologue has undoubtedly whetted your appetite for this all but immortal quartet, let's get our feet on the ground and talk about from whence they came to be where they are, what they're doing, and where they're headed.

★ ★ ★

The impetus of Jack Bruce obtaining a new record deal via showcasing with an as yet non-existent, 'temporary' band sowed the seeds of this ongoing and permanent group. The occurrence was hardly happenstance, as Jack and Clem recently worked on a Cozy Powell album together; Jack and Billy recently toured Europe together with John McLaughlin; and Jack's manager

(John Scher of Monarch Entertainment) was booking solo piano concerts for David. They talked, they rehearsed three days, and they knew they had a good thing going.

As a reluctantly interviewed ("Tell them I was sick") Jack Bruce put it as I badgered him into commenting on whether he considered Friends a consciously combined group to relate to an ongoing evolution of a specific musical direction, or pot luck: "I wouldn't say this band was pre-ordained, but it certainly was a band that happened more than was planned. Friends just sort of came together, and in a sense that puts more of a responsibility on you to allow the direction to unfold naturally, without intellectual or generally artificial and contrived restraints." As to direction, he added, "We did start out with the idea of making the songs accessible. That was the one guideline we had, as opposed to making the album clever, which everyone in this band is capable of, or esoteric to serve what purpose?"

As Billy Cobham stated it more simply: "It was potentially an interesting situation which I wanted to try. I am always game to try something intriguing to see if I can fit in. If it's going to be boring, though, I don't want to have any part of it."

In the formative stage of this hardly boring band's existence, it seems that a generally casual mood was prevalent. This was fortunate in that it allowed for a more relaxed initiation, and a rather comfortable development of musical ideas, direction and personal relationships. It all adds up to a mutual learning experience and an environment within which backgrounds and reputations facilitated the making of new, good music, rather than hampered it.

Enough of the preliminaries; let's get on with the album particulars.

Probably the most striking element, at least the most unusual, of the musical equipment used on the album was Billy's drums. There are three bass drums and three snare drums on a special metal rack/holder which Tama made for him (he endorses and helps design for them, so don't go to your neighborhood music store and expect to find one of these around). The bass drum pedals have a special linkage directly in front of him so that he can switch between them at will.

The snare complement includes: a 14" piccolo snare, a regular 14" Tama Rosewood snare and an 8" Hagar snare which splits in the middle (to change the depth from something like 6½" to 9"), to vary the tonality. Billy filled in the gaps in my percussion background as I queried about this snare's legitimacy.

"They're not new. They've actually been on the market some time for classical applications. They are hard to find because it's a specialized company with a limited production." And as to its applications, he continued, "I don't change its height very often. I find a setting for a certain tone or color I like and pretty much stick to it. The point of providing myself all these options in gear is to be flexible in terms of texture and feel both within and between different songs."

Other points of interest from the Cobham corner were special heads for the toms called Fiberskin 2's, which have just been released to the public, and a new muffler called a Deadringer.

The Deadringer is a foamrubber ring about 1" thick which fits inside the drum, under the head, on the ring and serves as a muffler. Billy used the "ringers" on the bottom head (you can put them on the top as well) and says that they make for a much more compatible recording situation for the subtleties of playing only what is necessary and sometimes merely implying the feeling—which was his approach with Friends.

Billy reflects on that music and the influence between it and the equipment: "The music that we played with Jack and the way the drum set was designed were a result of one another. That is, it was rather a continual chicken-egg-chicken scenario in that the music influenced what equipment I used and, in turn, what equipment I used affected the music. For instance, I used the triple bass and snare set-up and tom-tom approach because of the music and style I had in mind, and my equipment then determined the music's character. In any event, everyone felt that this was the sound that we were looking for, and I'm real happy with it."

Not far from Billy's equipment rested the no fewer than seven keyboards of David Sancious—set up so that all could be played "in combination" (real time) in the studio. (He actually only did one piano overdub, the rest were takes on basic tracks.)

The "live" as opposed to overdub approach was utilized because the musicians felt it gave them a better perspective on the music. And perspective is all important as David elaborates:

"You have to consider more than

Engineers Julian Robertson and Peter Roulinavage planning their next move.

just what your instrument it. That is, I can't just see it from a keyboard point of view. I have to think of the other things in the band, and what kind of music it is. Sometimes the most tasteful idea you can come up with is not to play at all. That might be the absolute perfect thing to do."

Well, so far we've got two individuals ready to finish basic tracks and overdubs in one pass (the record company will be glad to read this), on a bevy of keyboards and drums. What lurks beyond?

Try Clem Clempson. Mild-mannered guitarist with his three guitars (Stratocaster, Gibson 235 & Washburn) and two "little" Roland 40-watt Cubes. Sounds reasonable to me . . . *too* reasonable, I'm skeptical. Clem comments on his equipment from his inflatable raft chair in the House of Music's swimming pool. [The album was recorded at the House of Music recording studios in West Orange, New Jersey.]

"The Rolands have a great variety of sound. They're small and produce only 40 watts, so there is not a lot of spread. On stage I'll use four of them and if you stand exactly in line with them they sound really powerful and ballsy, but because there's not much spread, you don't get as much leakage into other instrument and vocal mics. When I first brought them down to rehearsal everyone was skeptical because they only look like little transistor radios, but they've worked really well."

He continues and talks about his overzealous comrades, "Everyone does so much in one pass that it's taken all the work and drudgery out of making the album. I was actually getting nervous about how *well* the backing tracks went. A lot of the basic tracks sounded finished—it wasn't as if we'd just come up with a bare skeleton that would require lots more work."

In rounding out these introductions to personnel and gear we now arrive at Jack Bruce. Jack's bass set-up included: a two-way speaker system with 1600 watts of power (Cerwin Vega 18-inch, two 12s and a horn and tweeter), stereo crossover, parametric equalizer and compressor. These were both miked and taken direct.

His Aria bass guitar has active electronics and state-of-the-art designs in general. "I try to keep up with the technology and the new ways of recording things. I really don't know a lot about any of it; I just know I can get a better sound this way, and I'm always interested and concerned about the sound."

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Speaking of sound, or "the getting of" sounds as it's referred to by engineers and producers, and before we talk more about the album's musical and personality particulars, let's talk tech.

Stephan Galfas, who is co-owner of the House of Music, as well as album engineer, is definitely not found wanting or lacking an opinion on tech talk. From the monitors to mixing to miking, we talked on and on of the technical considerations and approaches to recording the Friends album.

The control room monitors are a design collaboration between the House of Music and designer John Gardner. They each contain two Gauss 15s, JBL drivers on the tweeter and mahogany midrange horn and their own crossovers. These components are mounted in the cabinet so that they are physically in-phase rather than relying on electrical time aligning, and are powered by Crown DC300s on the bottom and Sony 100-watt VFETs (with super fast switching) on the top. Stephan relates, "The Sonys allow us to get away from that hard sound we were getting when we used Crown 150s for the high end. The tightness you get from them may or may not be desirable for low end (we went to the Gauss speakers to compensate just a bit for that)—the trade-off between a hard and a tight bottom being in the amount of ear fatigue, and therefore just how long I can listen and work without becoming inaccurate or sloppy . . . or deaf."

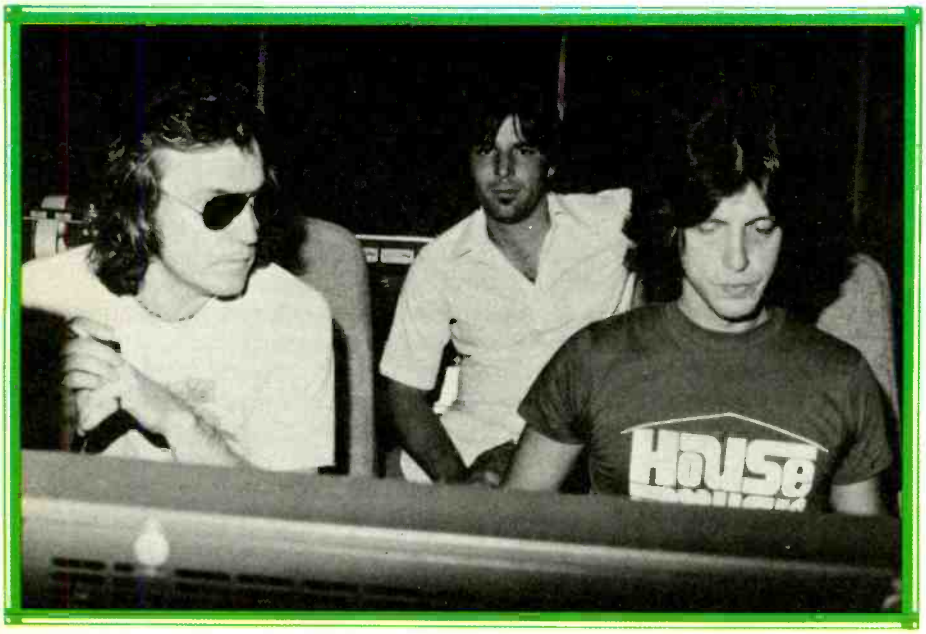
Stephan feels ear fatigue is a big problem in studio monitoring because your signal is not flat until you're above 90 dB—in most studios that is where the room is aligned or tuned. And when you work at those levels for ten or twelve hours you are going to have ear shock beyond belief. "After getting sounds, which I do at high levels where it is totally flat, I prefer to work at much lower levels on overdubs and general work. Very often we listen at levels you can talk over because I've learned over the years that working too long at too high levels (on anybody's monitors) hurts not only your ears, but also the end product."

The approach in terms of the listening chain on the Friends album was to start on the big studio monitors and use a variety of "reference" speakers to check on consumer reality. In the studio itself were speakers ranging from JBL 4311 Bs (with the mids and tweeters turned down) to Auratones. Outside the studio were even more and equally diverse systems in the House of Music's own listening room and in the engineers' and musicians' homes—all in an attempt to

Cobham's elaborate drum kit which included a custom Tama metal rack for the bass and snare drums.

keep the reality of home listening in perspective.

"Most of the work on the record was done on the big speakers, very low," says Stephan. "As a studio owner, I could type the studio and the monitors by playing material at high levels—the intensity and operating parameters at high levels allow them to sound amazing. But, at the same time, I also have the responsibility of making sure it sounds good when they take it out of here and play it somewhere else. That end product is what I'm most concerned about, and, in any event, with this project we're not dealing with amateurs who would be susceptible to a lot of shuck and jive."



Jack Bruce and Peter Roulinavage observe as Stephan Galfas takes control of the console.

In terms of mixing and the questions of level and speakers, there were two approaches that Stephan and Friends used. Sometimes they mixed very loud on the big speakers and played it back on the small ones. But, more often they preferred to mix on the various small speakers to get into the material because of the music, rather than any hype, and establish basic relative levels. Then, they'd play it back on the big speakers to hear it more clearly, and again, always reference back to the small speakers to keep perspective.

Another check-and-balance precaution was to take a tune into New York the day after it was mixed, and master it to get a better idea of what the end product was going to sound like. This was to head off any unforeseen disc-cutting problems. Stephan elaborates, "We had a bit of a problem in that Billy's feet are virtual cannons. We have lots of stereo information, and, just with Billy, lots of bottom. For these reasons, a mastering check was very important." Stephan continues talking about the product after it's out of his hands. "I'm not a level fanatic insofar as the pressing is concerned because the radio station compressors are going to suck it up anyway. But, getting a decent record with the substandard vinyl they're pressing with these days, which is thinner than in the past and has more surface noise (which really defines how low you can press if you're going for length or limiting groove distortion), requires doing anything you can to check and protect yourself. I don't care about +3s, but I'd like to be in the 0 range. And I will use some compression and maybe speed up the record slightly when I master because I don't want the record remixed by the radio station. I feel like these moves will (commercially) help the sound of the record in the end, and if you're in this business you've got to be at least somewhat marketing- and business-oriented. That means realizing that your product must sound good on the radio in order to influence people to buy it."

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From these considerations on what to do with the end product, let's slip backwards somewhat and talk of the means to that end by getting back to Billy Cobham and recording the drums.

The toughest part of recording Billy, the story goes, was his dynamic range. At one point he'd be at −20 and then in the next phrase of music he'd be at +20. In an attempt to control the dynamics, Stephan started by putting noise gates on the bass drums to keep them quiet; but Stephan pulled them off because of Billy's ability to play so deliberately soft in certain sections that the gates became useless. The problem was solved by using careful mic positioning to do as much phase cancellation of unwanted mic leakage as possible, a little equalization to help, but there were no baffles or compressors used because a "live" and open sound was the objective.

"We used visual [TV] monitoring of the mic positioning and rechecked the mics after every take to make sure positioning remained correct. We also used a phase meter on the board and in- and out-of-phase channel checks. All to be sure that the mic placement didn't alter because it was so critical to the drum sounds as well as to the isolation."

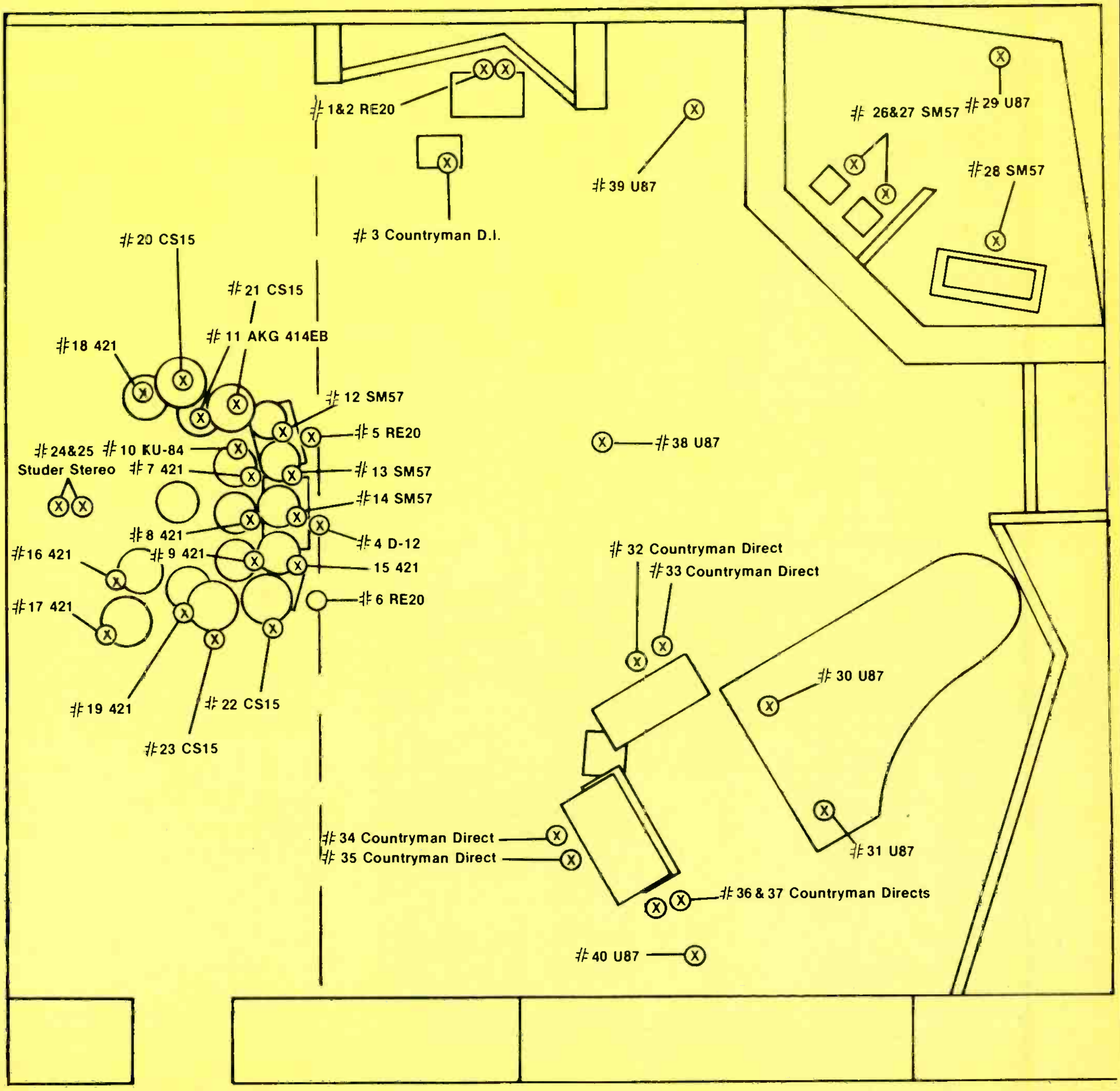
Stephan also places one stereo Studer microphone behind the drummer to get the drummer's perspective of the drums. "I do that so I can hear what the drummer hears when he plays. Hearing the drums from in front is quite different from sitting behind them like the musician. It also gives me the true stereo image of the kit, so that when I do my panning in the mix, I can fit the drums into the Studer's image such that they are exactly in line.

"Billy's miking problem (within his drum kit) was a simplistic one because it was of an acoustical nature; it was just tedious. Our biggest leakage problem, because of Billy's three "cannons" and Jack's bass rig being nearby, was the piano."

Low-end energy and leakage between the various instruments was cancelled to a great degree both through movement of mics and use of the bass traps in the room Jack was playing into, but, because they used no traditional gobos or baffles, they did build a special apron that fit underneath the piano to keep the sound from getting to the mics from the bottom.

This whole situation was complicated by having to turn the piano around into the room and towards the drums so all of David's keyboards could be set

1. Bass speaker cabinet (high end)
2. Bass speaker cabinet (low end)
3. Bass (direct)
4. 24" Bass drum
5. 22" Bass drum
6. 18" Bass drum
7. Hagar 14" expanding snare (top)
8. 14" Piccolo snare
9. Tama 14" rosewood snare
10. Hagar 14" expanding snare (side)
11. High-hat
12. 12" Tom
13. 13" Tom
14. 14" Tom
15. 15" Tom
16. 16" Tom
17. 18" Tom
18. 16" Roto-tympani
19. 18" Roto-tympani
20. 20" Zildjian pang cymbal
21. 18" Zildjian crash
22. 16" Zildjian crash
23. 18" Zildjian ride
24. Stereo
25. drum overhead
26. Roland
27. cube-40 guitar amps
28. 50-watt Marshall amp w/enclosed cabinet
29. Guitar room
30. Yamaha grand low end
31. Yamaha grand high end
32. Prophet
33. Polymoog
34. Yamaha
35. CS-80
36. Fender
37. rhodes
38. Room center - scratch vocal
39. Room left
40. Room right



up to be played together, as previously mentioned. So, the bottom line was to blanket off the top of the piano, build the special skirt/baffle and use both critical and close mic positioning to avoid leakage and keep the sound open for the instrumental tracks.

The guitar posed no problems in the midst of all this for, on basics, Clem and his Roland Cubes were in the isolation booth. Stephan continues, "As far as guitar equipment went, we had several setups: the Cubes, which Clem generally played through, a Marshall 50-watt combo amp (two 12s), and a Marshall 100-watt/4-12" cabinet which David generally liked (David doubled on guitar when a second guitar was called for).

"Half the guitar overdubs have been with Clem in the control room and mics about twenty-five feet away from his amps, and up in the air sixteen feet or so. We also kept the amp head in the control room and ran a speaker cable out to the studio rather than running a long guitar cord out to a head in the studio. We've found there is a significant difference having the long cable run be speaker/high-level, low-impedance stuff rather than unbalanced/ low-level hi-impedance.

"Also, conceptually with the guitars, as well as everything else, we were going for real "live" sound. The reality is that Marshalls fry, crackle, feedback and hum. Unless we were going for an effect, and purposely manipulating and/or changing reality, we decided not to dwell [alter] on these conditions. To me, that's real. Like at a concert, it's exciting, and we're trying to keep as much of what happens "live" into this record as possible."

★ ★ ★

So much for the peaks and valleys of basic tracks, let's see what happened with the vocals.

It depended on the circumstances as to just what mic was used for Jack's vocals. It seems that because of his "incredible" power, everything from [Shure] SM 57s to [Sennheiser] 421s to [E-V] RE 20s collapsed just in proximity; when he would move in close to the mic, the bottom end would become self-compressing. The answer was an [Neuman] U-87, without any compression or limiting. In order to properly record the vocals, Stephan had to learn each song and then "ride" sections. Everyone involved would talk about the music and lyrics and dynamics, and then they'd make a pass (of which there weren't many because Jack will only sing a song a few times and that's it). No punching-in between tracks. He just does a performance, and as the engineer, Stephan had to be ready to capture it from reference take one, on.

An aid in capturing the vocals was the video monitor (remember the drums earlier), which provided a profile perspective of Jack (he faced the control room), and allowed the way he worked the mic to be critically watched.

"Sometimes," says Stephan, "he'd get emotional and move into the mic such that I had to bring the fader down and use the cue balance to control his mic position. With the perspective advantage the monitor gave me, it was easy to see if he was too close (turn the cue up so he moves back) or too far (turn it down). It was also convenient that he didn't have any sort of pitch problems. That makes it possible to get him real low in a mix, and get tremendous power out of him without his wandering from the melody.

"The video monitor was also handy for cues from the band while they were playing. For example, although I could see David and all his keyboards through the window, I was better informed of just what he was doing watching through the monitor with the camera positioned up in the air, giving me an overhead view. From this angle I could better see which keyboard he was on and even better anticipate where he was headed via, if you will, body language. The camera, in fact, got moved for each new setup we were in."

★ ★ ★

The recording schedule for *I've Always Wanted To Do This* was like something out of a Buddy Holly movie. Four weeks of recording, one week of mixing and, give or take a few overdubs, it's done. Setups took almost as long as the recording of basic tracks, which was happening at the alarming rate of two or three per day. Re-takes were rarely needed because someone had made a mistake, but rather because there was a desire to change a subjective aspect of the take. The memory swirls through Stephan's mind:

"There were three engineers working on the project [Peter Roulinavage III, Julian Robertson, and Stephan], just to keep up the pace. We worked hard to get done in such a short time, but it was not because of schedule or budget, but because it was fun and the guys were so incredibly fast and tight. It's a great rock & roll record. Mature in most respects, but some is also just straight-ahead rocking stuff. It was a real pleasure and enjoyment to be a part of."

As keyboardist David Sancious so succinctly put it, "It's an amazing first album. As to the future and just how songwriting duties, style, approach, etc., will fall . . . the structure of the band is such that it allows for it to be fairly wide open. There's theoretically no limit to what might happen."

Jack Bruce takes a break from his bass at one of Sancious's seven keyboards.

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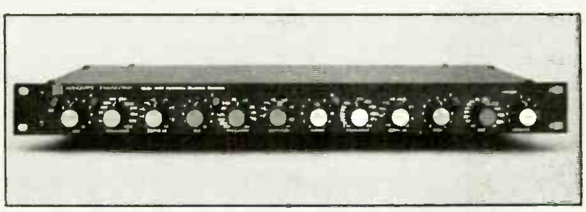
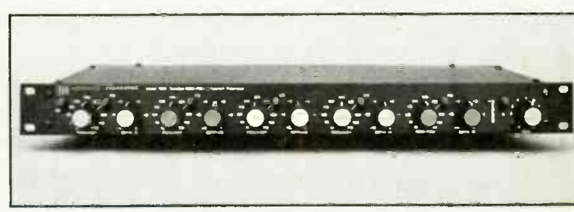


Model 1400 Parametric Electronic Crossover



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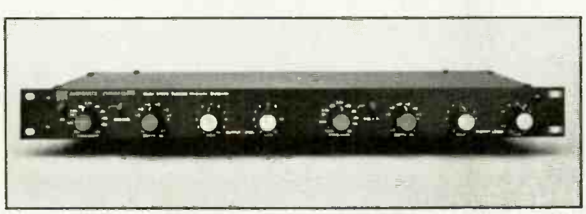
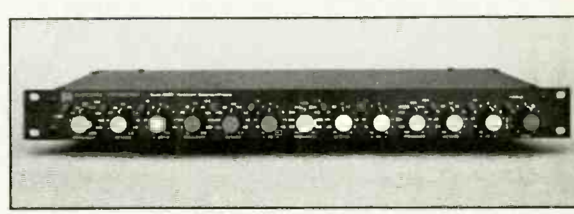
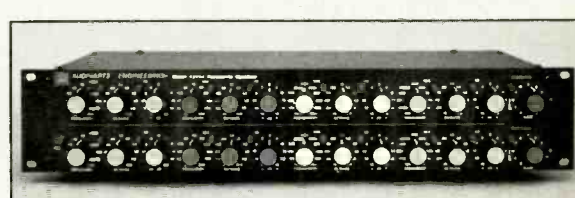


Model 2100A Tuneable Electronic Crossover



Model 4100 Parametric Equalizer – Preamp

Model 5200A Stereo Mixer/Preamplifier



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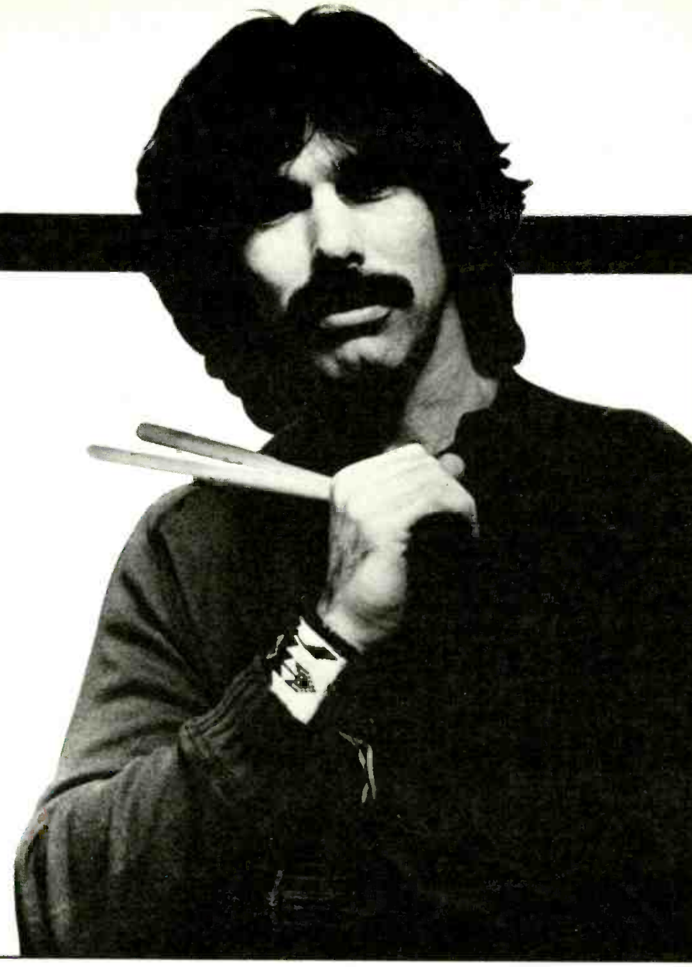
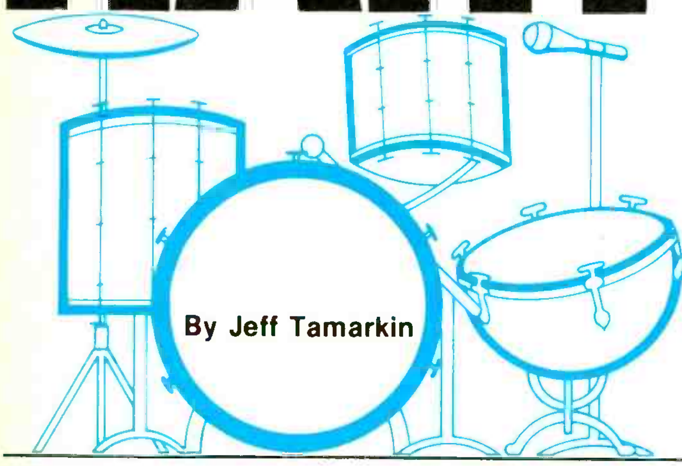
Profile: Drummer

MICKEY HART





By Jeff Tamarkin

Mickey Hart is perhaps best known as the more flamboyant half of the Grateful Dead's drum team. Behind his drums, Hart is never rigid or mechanical, but always flailing, a constantly active rhythm machine. And the rhythms he cranks out, usually in tandem with Bill Kreutzmann, the Dead's other drummer, are some of the most complex in rock music. The Dead drummers rarely rely on a traditional 4/4 rock and roll backbeat, but incorporate elements of drumming and percussion which they've learned both through active pursuit of ethnic drumming methods from around the world and over 15 years of playing together, getting to know how to work as a team. Unlike other drum duos in rock, Hart and Kreutzmann work in counterpart rather than playing the same licks at the same time. Their work together has always been one of the high points of Grateful Dead music.

However, Hart's interest in drumming and percussion goes far beyond the context of playing within a rock band. Since the late 1960s, Hart has sought out other ways of expressing his interest in drumming, from playing with the original formation of the country-rock New Riders of the Purple Sage to working with ten other Indian and American percussionists in a temporary ensemble called the Diga Rhythm Band, who released one album (Round Records) in 1976.

Most recently, Hart was asked by film director Francis Ford Coppola to compose and record the soundtrack for his epic, Apocalypse Now. *Hart, together with Kreutzmann, famed percussionist Airto Moreira, Dead bassist Phil Lesh, percussionists Mike Hinton, Jim Loveless, Greg Errico and Jordan Amarantha, and vocalist Flora Purim went ahead and created the Rhythm Devils, who then set out to record Coppola's jungle music.*

The project involved not the mere entering into a studio and playing of drums, but three months of research and trial-and-error, during which Hart and his cohorts set out to build *those instruments which were not available to them. Some of these were created from exotic materials imported from little-known lands. Not a single American trap drum was used, only those instruments which could have been used in the jungles where Coppola's war was fought. After three months of building, setting up and discussing their plans, the group of musicians entered Hart's studio in Marin County, California, and recorded their piece of basically spontaneous music in one take in an hour and a half.*

Ultimately, only a few bits of their recording were used in the finished film, so Hart decided to have a large chunk of the piece released. Passport Records picked up the project, and The Rhythm Devils Play River Music *was released this summer. While in New York, Hart spoke with* Modern Recording & Music's *Jeff Tamarkin about this project as well as his work with the Grateful Dead, who remain, after 15 years, one of the most rabidly loved and professionally respected bands in rock and roll.*

Modern Recording & Music: How long ago did you become interested in ethnic percussion music?

Mickey Hart: Ethnic music . . . what is that? *My* music, the sound of a drum, the sound of percussion, tuned percussion. That's where it's going, that's where I'm going, that's what I've always been into. Where percussion concussion meets melody and forms harmony. This latest thing (the *River Music* album) is just the latest thing. This is another crack at the famous twentieth century percussion ensemble. This is not drum music; this is music.

MR&M: As opposed to the more typical rock drum solo . . . ?

MH: Right. I'm not into that. I've done that and it's not appealing to me. The role that it plays in modern music is changing. I'm not into fads or styles. I try to keep that out of my life as much as possible.

MR&M: Did you have any formal training?

MH: I studied for years. I studied with Charles Perry. I read music and all that.

MR&M: What about training in the style that appears on the *River Music* album?

MH: No, you don't study that. That's something that's an accumulation of all your knowledge. Since '67 I've been studying North Indian classical music with Alla Rakha, and I've played with Zakir Hussain. When you play with people who are in that caliber of world percussionists, you start learning how to really hold time, how to use time, how to get friendly with time. Time is my friend. That's what I do with my whole life. I work with time. It's basically a matter of time and the instruments and the sound of the instruments and your affection towards them.

I love my instruments and I love my work. Or else I couldn't even attempt something like this. This is not entertainment by any means. Entertainment is not one of the things I had in mind. I didn't have beauty in mind. I didn't have simplicity in mind. This is pre-entertainment. This music has other emotional values. It was designed as an interpretation by the Rhythm Devils of *Apocalypse Now.* This is not a soundtrack album. This is music as it was played at the session. Some of it does appear in the movie, but most of it doesn't.

MR&M: I gave the soundtrack album away. I didn't care for it.

MH: It was horrible! It was one of the most horrible things I've ever seen done to a soundtrack. That is not a subject that is even worth talking about. Very quickly, it was more of a political situation than a musical one. It's not newsworthy.

MR&M: Who do you expect will buy the *River Music* album? Do you expect the average Grateful Dead fan to buy it?

MH: I think Dead Heads will buy it; I think jazz fans will buy it, percussion crazies. It's not meant as a commercial product. It's an offering. This is what we do. We didn't compromise it, and when you have uncompromising music, you're not necessarily appealing to the masses.

MR&M: Does Bill Kreutzmann (the other Grateful Dead drummer) share your enthusiasm for this music?

MH: Oh yeah. Kreutzmann takes one of the most amazing talking drum solos on the second side of the record on "Trenches." It was really inspiring. He's playing it abnormally well.

MR&M: What instruments were used on the record?

MH: Oh God, there're thousands of them. Mostly my collection and Airto's. We arranged all of them into a jungle of percussion, and we moved through it. There were no drum sets, no regular traps. We tried not to do anything we've done before. Traps don't live in the jungle so we didn't want to use them. We didn't want to be too geographic, because that identifies it—20th century western trap drums, you don't find that in the jungle. I'm not Vietnamese, so I can't play Vietnamese music. So it was the jungle of the mind. It wasn't specifically located anywhere geographically.

MR&M: Did you travel to the jungle to get a feeling for it?

MH: No, I didn't have to. I lived with that movie for three months. I had three screens: in my bedroom, my studio and my kitchen.

MR&M: Where did the Dead's concerts in Egypt fit in? Were they before or after you began work on the album?

MH: They were before, in '78.

MR&M: Did your trip to Egypt influence the sound on the record?

MH: Of course. All of my influences over the years came into play. Francis (Ford Coppola, director of *Apocalypse Now*) painted such a beautiful picture that it was easy to do.

Getting back to the instruments, we recreated the old beaded instruments, the wooden instruments, the instruments of stone, the lithophones, the early xylophones, if you will. We built "The Beast" (a series of large drums suspended from a long steel rod) and then we built this thing called the "Scritch," which is made of rods held together on a frame. You rub the rods with a white glove with rosin on it. It makes a piercing, eerie, high-end sound. Then there were gongs which we immersed in water, using hydrophones—underwater microphones.

MR&M: What was Phil Lesh (Grateful Dead bassist) doing? Was he playing bass?

MH: He was, but I processed it so it didn't sound like a bass. I processed it through gongs and stuff. I wasn't interested in a bass guitar playing with us because there's no bass guitar in the jungle. So I used his instrument to trigger gongs. I have my own studio where I live, so when I'm not in Grateful Dead land, I'm behind the board making mistakes, finding dead ends, so to speak. Then some day the mistakes turn into assets. The studio is our laboratory.

MR&M: How did the making of this record compare to your part in making the last Grateful Dead album *(Go To Heaven)*?

MH: Not at all, because I didn't have very much to do with the making of the last Dead album. I was composing for this.

MR&M: Were you satisfied with the last Dead album?

MH: No. Well, it wasn't easily put down. It wasn't what I would call ultimate Grateful Dead music by any stretch of the imagination.

MR&M: What kind of microphones did you use during the recording of *River Music*?

MH: I used Neumanns. I use dynamic microphones. For different reasons I use different microphones. I usually have them in stereo spreads, clusters. I usually delay them through a tube we call a "linear accelerator." That's like a 70-foot tube, with a microphone moving inside. For each foot the microphone moves, there's one millisecond of delay. That's a valuable delay if it's acoustic. The tube is small on one end and large on the other. We didn't use any reverb or echo in the making of the album.

MR&M: What about noise reduction?

MH: Nothing. All we used is acoustic delay through these tubes, which brought the sound outside. I've been working on this with Dan Healy (the Dead's sound engineer) for years. My whole house is filled with tubes. I live in the forest (in Marin County, California), and in the forest are these tubes. My place is like an acoustic garden. In my house, I even have tubes in the ceiling. Experimenting with them, I saw the nature of them. It doesn't sound like a tube. But you get this beautiful delay. And they're so inexpensive; it costs only $70 or $80 for a giant tube.

MR&M: Some of the jungle sounds on the album sound very authentic. It's hard to believe that was done in a studio with acoustic instruments.

MH: Isn't it amazing? I was trying to get the perspective of standing on a hill looking down into the next valley. How do you do that with presence? O.K., you take different images of those stereo pairs of microphones and you put them into delay factors. I put a piano in the other room and depressed the keys, tuned it into the open tuning and put a speaker into the piano. That was picked up through a magnetic pickup, brought back, delayed and that's where you hear Flora Purim's voice, on the piece called "Steps."

MR&M: Whose voices were used on the album? Some of them sound like they were recorded at a tribal meeting in Africa.

MH: Oh yeah, that's Flora, Airto and me. Airto is Mr. Jungle. Airto is *the* percussionist. I've always admired Airto's work. He was one of the people I always wanted to meet. We breathe in the same rhythm; we get along great. And we had Mike Hinton, who is a doctor of mallets. He plays broadway shows *(West Side Story).* And there was Greg Errico, Sly and the Family Stone's former drummer. Jordan Amarantha plays congas. Also, there's "Marimba Jim," (Jim Loveless) and Flora. What can you say about Flora? What a voice! I told her I didn't want any melodies or any harmonies, just the sound of a million years. It took three months to build the instruments and then we locked ourselves in for eleven days and nights. We looked at footage over and over.

MR&M: Was the movie playing while you recorded?

MH: Francis wanted us to perform the movie. He didn't want it scene by scene. So we looked at it and looked at it and talked about it. Then we shut the screens off and played it, because you can't play and look at the same time. Then we went back scene by scene, for the napalm, for instance, for all those explosions. Once you're rolling, you're rolling, and if you miss you miss. You can't go back. This was never done before so we didn't have any guidelines. But I learned how to mix a quad movie.

MR&M: Would you like to do more soundtrack work?

MH: Yes, I would. But my first love is the Grateful Dead; that's what I do. That's 24 hours a day.

MR&M: What did the studio look like? How was it set up?

MH: There were paths. We set them up in batteries and each battery contained certain kinds of instruments, which I pre-determined so they could co-exist with the instruments that were next to them and across from them. So, part of the composition was placing all those instruments and arranging them. That took me two days. See, nobody played one instrument. Everybody moved through the jungle and played what they came upon.

MR&M: So, it wasn't a typical studio setup, with instruments set up in booths?

MH: No, it was all wide open. I had to arrange it so you wouldn't step on my 300-year-old maraca. It was the middle of the night and we were on our bellies. War is hell, you know. It wasn't supposed to be pretty, and it isn't pretty. And it wasn't supposed to be fun. It's the truth and there's a certain beauty in the truth. We just tried to reflect it in the room, without going into massive overdub sessions. It's a real good representation of what the Rhythm Devils thought of that movie that day.

MR&M: Did you give the sound engineers any special sort of requirements?

MH: The only requirement was to record more than they erase. From working with Kreutzmann and me all these years, the Grateful Dead engineers are well-versed in the recording of percussion, which is so dangerous. You wouldn't want to do it for a living unless you are a stone-cold pro. By this time we're pros at percussion. There were a lot of new techniques used. I've told you about the tubes, and the pianos being in the other room. Every instrument was finely tuned. The wind chimes . . . every piece of brass was tuned. We did it with computers. They were amazingly accurate, harmonically speaking. We didn't use any EQ on this. This is straight open. And I mixed it the first time; there was no second time on the mix. Phil Lesh was sitting there and he asked, "Aren't you going to listen to that back?" and I said nope. We went right on to the next.

MR&M: Lesh was quoted in a *Relix* magazine interview as saying that he thought you got "burned" as far as making the soundtrack. What did happen to make Coppola cut your contribution out of the soundtrack?

MH: I didn't get burned. Francis and I are friends beyond this movie. What happened was that he got sick near the end of the making of the film and his lieutenants took over. If it was up to Francis, it would've been different. He got sick of it. His health was failing and he couldn't see the movie one more time. I'm like that, too. At the end of a project, I can't listen to it anymore.

I think the movie might have suffered, but I didn't get burned. I had a great time working with Francis. It was wonderful. Maybe if he hadn't used *any* of it, I would have been burned. But he came to dinner one night, and at one point he said, "You know, you're on the screen for six minutes straight. That's more than (Marlon) Brando is on the screen continuously."

MR&M: What were some of the problems that you had in the studio?

MH: Moving from one thing to another, and the setup time. When you don't have things set up and you don't know what you're going to do next, that has a redundancy to it. We went into an altered state. We went out in the zone. We weren't into sound-checking or anything like that. This was for real, and there was no way of organizing it. I figured if I organized it, it would turn out to be a production. I had to forget I was a producer and engineer and all that—take that hat off—and just become Micky Hart, drummer, percussionist. There was no leader. It was: *We're all in this together, now here we go, up the river.*

MR&M: Once the tape machines got rolling, you didn't want to know about what was going on in the control room?

MH: No. But the mics didn't move once we got rolling for the big performance. This was taken out of an hour-and-a-half performance. When we stopped, we looked at each other and clapped.

MR&M: How long did the recording take?

MH: One day. That was it, a real spontaneous performance.

MR&M: Three months to set up, and one day to record?

MH: That's right.

MR&M: Would you consider taking some of the instruments to the concert stage?

MH: You bet. The Rhythm Devils are going to play some day. Kreutzmann and I are going to take them out on a real tour. And more of it will be used in the Grateful Dead set.

MR&M: Was Jerry Garcia (Dead guitarist) asked to take part in the sessions?

MH: No, this was not a Grateful Dead-oriented project.

MR&M: Was the music actually composed, or was it totally spontaneous?

MH: The composition was the making of the instruments. This is not what you'd consider classically composed. We talked about a combination of instruments that could be used if we happen to meet in the night.

This was a risky, wild project. You wouldn't want to do this for a living. The chances of failure were 90%. I talked this project over with some of the quote, "heavies" of the L.A. studio world, and they said I was nuts: "You put that many people in the room, and they'll step all over themselves." But we didn't. It was truly an inspired work; one of the things in my life I'm truly proud of doing. This is profound music to me. We had to rely on total communication. At best, music should be like that, especially percussion. There's a limited amount of what you'd call melody.

MR&M: One of the basic tenets of Dead music has always been the dynamics of it.

MH: That's it. You spin a web. There's weaving, texture.

MR&M: As opposed to a basic rock 4/4.

MH: Right; that works, but it's not the whole meal.

MR&M: How has your drumming technique evolved since you began playing?

MH: I began with just the rudiments, and then went into jazz and rock and big band, then into hand drumming. That opened up a whole new world of drums. Stick drums—membrane drums, or membranophones—are only part of the world of percussion.

When the Grateful Dead go places, we learn things. In Egypt, I learned the "tar," the Nubian drum. So I expanded my vocabulary daily. I study; I read books. When we played in Alaska, I went up to the Arctic Circle for three days to look for Eskimo music. We recorded traditional Eskimo music, slept in one of their traditional sod houses on the water, had our parkas. I've studied tabla, and Brazilian drums, and keep adding to my vocabulary. In the last ten years, it's been possible to get records from all over the world. You can sit in your living room and hear anything from anyplace. You'd never be able to hear anything like we did unless it was on record. I travel around with the Dead and record native music.

MR&M: Where would you like to go that you haven't been yet?

MH: I'd like to go to meet a certain witch doctor in South America who takes peyote and doctors animals and does great things. She has music that goes with it. I'd also like to go to Japan. There's a troupe there that runs 18 miles a day and drums the rest of the day. They run to drum and they drum to run. I run every day myself and my running helps me to drum. My drumming helps me to run.

MR&M: How do you find out about these things?

MH: It comes to my door. San Francisco is a unique cultural crossroads. Musicians go there when they're not playing, or they pass through somehow. My antennae are always up for new and exciting musical things from other places. I'll go see a folk troupe or classical music from other countries any time.

MR&M: Do you listen to any current rock music?

MH: No, not much. I hear it on the radio and I bop to it, but it's just rock and roll. I love rock and roll and I've been doing it long enough, but you can't do it all the time. You can't just have meat and potatoes. I like fish and salads, too. I like to listen to music from around the world. Then you can find out who you are and what you are and why you are.

MR&M: Do the other members of the Dead share those same feelings?

MH: Well, we operate outside of the straight, or normal world. We have our own little Grateful Dead world. We're not affected much by trends.

I like to hear Ravi Shankar or Ali Akbar or Hamza El Din. I record them when they're in San Francisco. I give them a recording and I keep one. When I want to hear a fine recording of one of my favorite artists, I go and pull out one of my tapes.

MR&M: You're originally from New York. Were you in-

"... we recreated the old beaded instruments, the wooden instruments, the lithophones, the early xylophones, if you will."

terested in this music before you moved to San Francisco and joined the Dead?

MH: Oh no. All of that changed when I moved in 1965—Haight-Ashbury and the Dead and just living together.

MR&M: Can you recount how you joined the band?

MH: I had met Kreutzmann, and I was showing him some rudiments, when he said, "I have this new band. We're playing at the Straight Theater" [San Francisco]. So I went to hear them and I said, "What a band! I don't know what these guys are doing but this is great." Kreutzmann asked if I wanted to play, and for the second set I played and it was magnificent.

MR&M: Does it still feel that way?

MH: Well, it's more consistent. You never get the same feeling as you did the first time. Those days were wilder and everything was an experiment. You couldn't do that day in and day out. Not for 15 years. So, there are certain things you like and you say, *O.K., I like this; I'll repeat it.* Then we got songs that we sang. It will always be in a state of flux, but it's certainly not in the same state as it was in 1967.

MR&M: If you compare, say, the first five Dead albums, each one is totally different . . .

MH: Now our lives are more stable. We're not in as much of a state of change ourselves. I live on the edge and Jerry does. Bobby (Weir, guitarist) and Billy are really stable.

MR&M: How do the Dead audiences of today compare to the Haight audiences of '67?

MH: It's the same thing: people enjoying music. They don't wear as many feathers, but when we get it on and contact is made, that's the same. In our world, anything can change, and we hope that it does. We're too loose to structure it too much. We're not your average studio musicians whom you can tell: *you play this and you play that.* We usually feel what we want to play and play it. If it works, it works, and if it doesn't, it doesn't.

MR&M: If the music became regimented, what would that do to the group?

MH: Over. The Grateful Dead as we know it would be over. What I'm doing now only gives life to the group; my learning about new things and playing with Airto and making drums, turns up in new Grateful Dead music. This band will bend over backwards to help each other. I can't imagine Jerry saying, "No, you can't do that." If anything, it's, "How can I help you?"

MR&M: Where does commercial success fit in?

MH: I don't know what commercial success is when you talk about art, and this to me is art. Our project is a commercial success as far as it's paid for itself. It's not in the red. It's a percussion record and it's paid for itself. To me, this isn't drums; it's music. If people like it and happen to buy it, then to me, that's a commercial success.

MR&M: Do you consider *River Music* to be rock and roll?

MH: What is rock and roll? It's an attitude. It's music. The Grateful Dead don't play rock and roll. They play more jazz. It's more communication than it is straight 4/4. So, this record reflects my philosophy of communication more than that of rock and roll.

MR&M: You're very enthusiastic about your work. It's a pleasure to talk to a musician who is so totally involved with his music.

MH: I'm in love with my drums and my music. Without them, there would be nothing. I know who I am. I love to drum. I love to hit things. I love to hear them and I love to make a sound. I love the interaction you get when playing with other people. There's a high level of consciousness you get when you play ensemble music. There's a lot of hit and miss, a lot of failure.

In the old days I used to walk off stage dejected, feeling low, saying what are you doing with your life? I'd be saying: *You just went out there and threw up for five hours. How could you do that? You're charging people for that; that's ridiculous.* Then you come back the next night and you play out of your body. You work real hard at it and keep at it and eventually, maybe 50 years later, you become a musician. If I'm still alive and the Grateful Dead are still alive, you bet I'll still be doing it. Why not? Look at Basie and Ellington and those guys. They kept at it and their music changed, but they're still viable musically.

There's no reason in the world the Grateful Dead can't go on for years. And the audience will change. We've just turned a decade, turned a page. Now the kids of our original audience are coming, and so are their parents! These were kids that were had to Grateful Dead music. The Grateful Dead were for screwing. That's when we knew we had it. The music made you want to make love, touch people and be friendly. It's still like that, except that people don't sit down in the aisles and screw. It's like an atmosphere that's conducive to meeting friends and leaving the office behind, taking your head where you want it to go, and where peer pressure isn't that great. Certainly, being a Dead Head was not popular for many years. Being a Dead Head meant being a crumpled up little dopester somewhere on the street whom everybody made fun of and kicked at. Now, when the Grateful Dead come to town, he's the only one who's got the tickets and he's the big man on the block.

MR&M: Do you even feel that you owe certain things to the fans? For example, are there any songs you feel compelled to play?

MH: No way. And if we find ourselves getting into that, we snap out of it in a second. We played a gig with the Who a few years back, and after the set, Pete Townshend came up to us and said, "You guys didn't repeat one song! How do you do it? We've been playing the same set for six years!" We could probably play a week straight and not repeat a song, except that our singers don't have very good memories. All those words to remember. It's a lot for anybody.

MR&M: What will you do if and when the Dead hang it up?

MH: Everything I'm doing now. And more.

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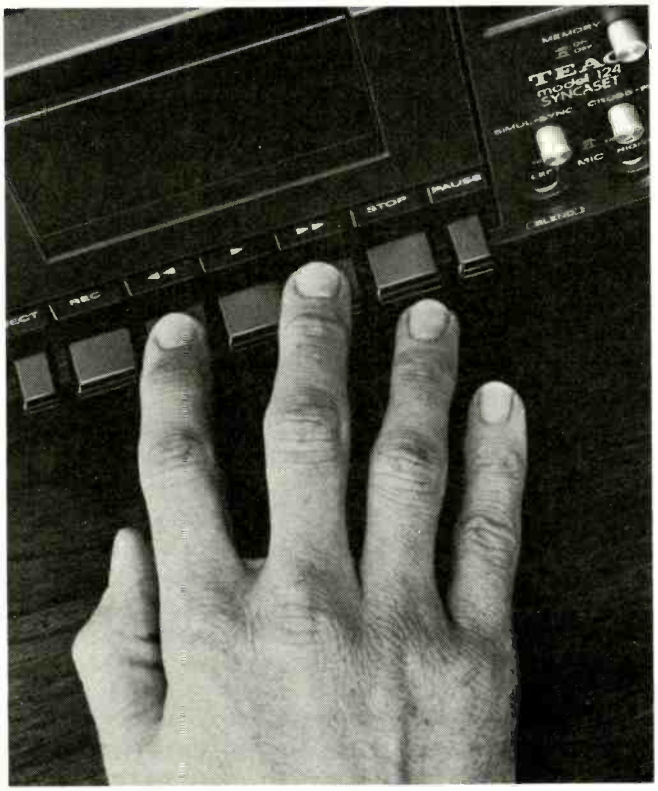

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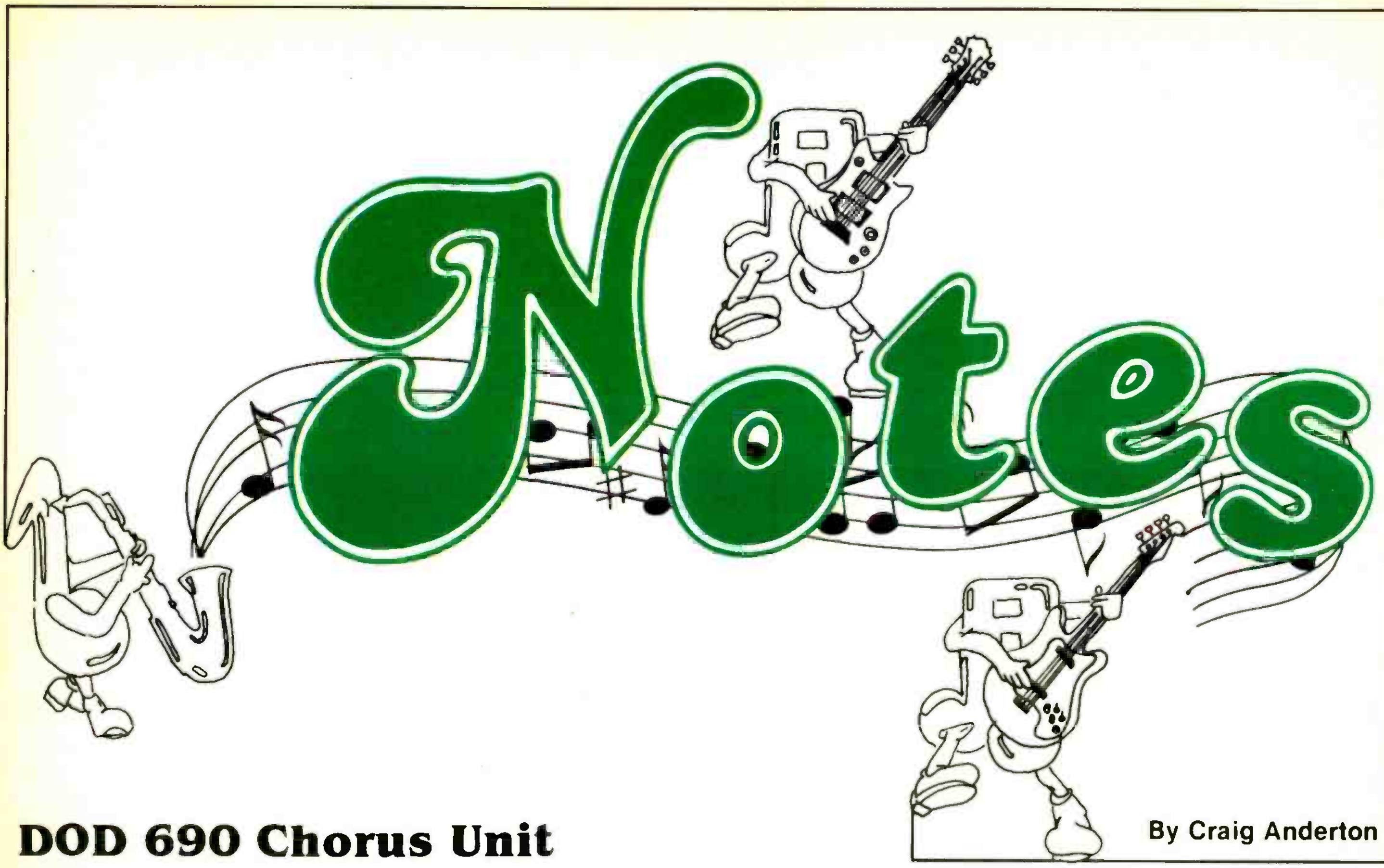

DOD 690 Chorus Unit

By Craig Anderton

The other day I visited a local music store, and after exchanging pleasantries, the manager said he had something to ask me. The question: "Are all analog delay units inherently noisy?"

I thought that over a little bit, and realized that the question did not lend itself to an easy answer. So, I asked him how much he knew about analog delay units, particularly chorus boxes. He said he really didn't understand much about the things, which made me think that there were also probably many musicians who don't understand the noise, performance and cost tradeoffs associated with analog delay devices. So before we get into discussing the DOD 690, which illustrates many facets of analog delay design, let's discuss a little background material. This material will not only make the following review easier to understand, but should also help you when evaluating any of the various analog delay-based devices available on the market.

Analog vs. Digital

For many years, the only economical way to delay audio signals was with tape recorder technology. In fact, in many applications tape echo units are still more cost-effective and give higher levels of performance than their all-electronic counterparts. Nonetheless, tape echo has severe limitations. For one, it is very hard to get an extremely short delay using tape. There are also the usual problems associated with recording any type of signal on tape, particularly hiss and reliability of the tape transport.

Digital technology attacked the fidelity and noise problems by coding an audio signal into a series of numbers, *then delaying this group of numbers instead of the signal itself.* The greatest advantage of this method is that unlike analog signals, digitally coded numbers are insensitive to hum or noise (a "3" with hum still looks like a "3" to the digital circuitry). As a result, as these numbers are delayed, there is very little signal degradation. Eventually this delayed group of numbers gets decoded back into an analog signal, resulting in a delayed analog signal. Unfortunately, while digital delay offers good fidelity, you pay a lot for that fidelity. People therefore started looking for an economical alternative to the digital delay of audio signals.

The answer arrived in the form of the *bucket brigade analog delay line IC;* these ICs are the heart of virtually all currently available analog delay units (*charge coupled devices* produce a similar result using different technology, but both charge coupled and bucket brigade devices are conceptually very similar). Analog delays have some nice features going for them: they can easily produce short delays; are inexpensive; require limited support circuitry; and draw very little supply current. But, there are also some disadvantages. Since we're processing the signal itself rather than a series of numbers, we open ourselves to possible signal degradation—particularly noise and distortion.

A bucket brigade device works by sampling an input audio signal at a very fast rate (several thousand samples per second). Each sample is like a "snapshot" of the signal at that instant in time; this sample is in the form of a voltage. This leads us to an explanation of why these parts are called "bucket brigade" devices. Bucket brigade ICs have hundreds, and sometimes thousands, of "buckets" arranged in one long line; each bucket can hold a voltage for a very brief period of time. Delay occurs by taking the sampled voltage and storing it in the first bucket, pausing for an extremely short period of time (a few microseconds), then passing this sample along to the second bucket. As the first sample passes down to the second bucket, another sample is being stored in the first bucket. When the voltage in the second bucket gets passed along to the third bucket, the voltage that was stored in the first bucket gets passed along to the second bucket, and a new sample arrives in the first bucket. These charges travel along from bucket to bucket and, eventually, hit the output of the device. As these voltages come out of the bucket brigade, the signal gets reconstructed. Note that since each bucket only causes a very slight delay, we would need to have lots of buckets in order to get any appreciable time change.

OK . . . if you've made it so far, you might be happy to hear that we have just completed the "technical jungle" part of this review. Now it's time to relate what we've learned to flanging, delay and, particularly, chorusing.

If you think about the above information a little bit, it becomes clear that there are two simple ways to create a longer delay: 1) you can add more buckets; or, 2) you can pass voltages from bucket to bucket at a slower rate. The problem with the first solution is cost; the more buckets, the more expensive the part. Also, additional buckets give additional potential for signal degradation.

The problem with the second approach is that if you pass the signal along at a slow rate, you start getting high distortion. This is because in order to get low distortion, you need to make as many samples of the input signal as possible, and stuff these samples into the buckets as fast as possible. As you take fewer samples, the signal becomes less defined (think of a connect-the-dots game—the more dots ("samples") the more obvious the picture; the less dots, the less defined the picture). A lack of definition produces distortion.

On to chorusing. Audio people like to classify time delays into a sort of "delay spectrum," which is analogous to an audio spectrum. At the short delay end of the spectrum (up to about 15 milliseconds), you get flanging effects. Increasing the delay to around 20 ms. produces chorusing. Delays greater than 20 ms. create the impression of two distinct signals, which gives a doubling effect. Delays longer than about 30 to 50 ms. create echoes.

As it so happens, chorusing falls in an awkward range from an electronic design standpoint. There are four basic analog delay chip configurations, even among different manufacturers: 512 buckets, 1024 buckets, 2048 and 4096 buckets. For an equal amount of delay, you can either cram lots of samples into a chip with lots of buckets, or slow down the sample transfer rate a bit and use a chip with fewer buckets.

It would then seem logical to try the first approach and use a chip with lots of buckets, since we could run it fast in order to get plenty of samples along with a relatively small time delay . . . but this also means highest cost. And I do mean high; a 4096 stage part can list for $50, while the shorter 1024 stage types still list for around $10—a pretty high price for an analog IC. However, while using a chip with fewer buckets keeps the cost down, this also decreases the fidelity and adds noise. Luckily, there are ways to help keep the noise under control. One process is called companding (short for compressing/expanding), and works in a manner that's very similar to the dbx noise reduction system for tape recorders.

THE DOD 690: WHAT is IT? Some of the more expensive chorus units use a chip with lots of buckets, and then transfer samples at a fast rate, in order to eliminate the need for companding. However, while the fidelity of this approach is generally quite good, there is still some audible residual noise that is particularly noticeable when you are not playing. In order to keep down costs and create an affordable chorus box (considering that the DOD 690 lists for under $190, it is indeed one of the least expensive "full function" chorus units on the market), DOD has taken the approach of using a shorter, 512 stage delay device. While attempting to squeeze the required amount of delay for chorusing out of a 512 stage device could lead to noise problems, thanks to companding any noise is held to a generally non-objectionable level.

The 690 has four controls, two footswitches, and AC on/off switch and three jacks. Two of the controls vary the depth and width of the chorusing effect. Increasing the depth varies the proportion of delayed signal present in the final mix, while varying the width injects greater or lesser amounts of the delay modulation that actually creates the chorusing effect. The other two controls vary the rate of this modulation; however, DOD has added an interesting twist here. Each control is a preset, and the rightmost footswitch switches between the two presets. The difference between the 690's presets and the other preset arrangements is that the speed changes smoothly as you switch from one preset to the other. In other words, suppose one of the presets is set for fast modulation, the other for slow modulation, and that you're currently set on the fast modulation preset. Step on the footswitch, and the modulation speed slows down until it reaches the slower speed. Step on the footswitch again, and the modulation speeds up until it reaches the faster speed. Sound familiar? Well, it should; this is the same type of effect that you get from electromechanical rotating speakers when you change from fast to slow settings. An indicator LED flashes at the modulation range whenever the chorusing effect is selected.

The other footswitch is your basic bypass switch, although it is not a true bypass (meaning that some of the 690's circuitry is still part of your signal, even when bypassed). There are no electronic clicks or pops when you switch the chorus effect in and out.

On to the three jacks. One is an input jack, while the other two are oppositely phased, dual mono output jacks for stereo chorusing effects (only the delayed portion of the output signals are out of phase with respect to each other; the normal signal is not). The extra output jack for stereo is definitely not a frill, since stereo chorusing gives a really lush and luxurious sound. You can restrict yourself to using one output jack for mono applications if need be.

So much for the controls, now for a few words about the construction. The case is extremely strong, being made of cast aluminum; once the bottom is screwed on—and it's held in place by six sturdy screws—you've got something that could probably be run over by a VW or thrown against the wall and still survive (although you'd probably chip the knobs and scar the nice blue finish). Inside the case, on first inspection the PC board looks like it's flimsily mounted, but further testing showed that it's not going anywhere. Most of the parts are of high quality, and there is an internal AC supply (with a three-wire, not two-wire, plug) so you don't have to replace any batteries. The complement of ICs includes an SAD512 delay line, 571 compander (both of these are socketed), LM324 (for the modulation circuitry, I assume), 4001 for the delay line clock and two 4558s for audio processing. Overall, it seems to me that the DOD 690 would have no trouble holding up to life on the road.

PRE-FLIGHT for the 690: First, find an outlet and plug in. Next, I would strongly suggest placing a compressor or other signal-boosting device in front of the 690. Here's why.

The 690's unpretentious instruction manual specifies the headroom as 4 V peak-to-peak. This assures that "hot" guitars won't overload the unit, but there is another more important ramification. Chorus units generally go at the end, or close to the end, of a chain of effects. At this point in the signal chain, the instrument's output has probably been boosted enough so that the 690's 4 V of headroom is necessary to avoid distortion.

However, not all stock guitars put out 4 V peak-to-peak (p-p) without some electronic help. In fact, I've played some guitars into an oscilloscope and noted that unless I was really banging away on the strings, the output seldom got much above 1 V p-p. What does this mean? Well, suppose we get a certain signal-to-noise ratio with a 4 V p-p input signal. Now, when we reduce the input to a 1 V p-p signal, we are lowering the signal going through the unit by 12 dB, which means that we have lowered the signal-to-noise ratio at the output by 12 dB as well! That's quite a difference. I remember one music store owner talking about a certain chorus unit (not the 690, by the way) as being a piece of crap because it was so noisy. By simply advising him to increase the input level somewhat, the signal-to-noise ratio improved to the point where the unit didn't sound noisy any more. The moral of the story: with *any* analog delay, put in the highest possible signal short of distortion.

Another excellent stereo application of the DOD 690 is in turning mono reverb into a rich sounding, synthesized stereo reverb. Patch the output of the reverb into the 690 input, then bring the two 690 outputs to a pair of channels panned to opposite sides of the stereo field. Set the speed control for a slow setting, and you'll hear some extremely rich and animated reverb effects. Using two chorus units with stereo reverb (again, pan both chorus outputs to opposite sides) gives an even richer effect.

EVALUATING the 690 SOUND: Start off with the width and depth controls on full, one modulation control set for the fastest possible speed and the other modulation control set for the slowest speed. Punch the left-hand footswitch to bring in the chorusing effect, then punch the right-hand footswitch until you've selected the slow modulation preset. This gives the traditional chorus sound, which is a swirling, luxurious type of effect that makes your instrument sound bigger and more animated. As you increase the modulation speed, you may want to pull back a bit on the depth and width controls to avoid an overly "gimmicky" sound.

If you play long, sustaining parts, chances are you won't hear much noise. However, should you play in a more subtle, gentle manner, you will start to hear noise superimposed on your signal. If you stop playing in order to hear the amount of noise, it will go away instantly due to the companding action. Noise is only present while you're playing, which is certainly a help since the louder you play, the more your signal will mask the noise. But if you play softly, the noise will no longer be masked as effectively.

As a matter of fact, my only real reservation concerning the 690 is the noise level. For most musicians, I'm sure that the 690's low cost is more than enough compensation for putting up with a little noise. For other musicians—particularly non-rock and rollers—the noise might be objectionable. So, when you check out this unit, listen carefully to the noise level. If it seems excessive, first make sure you're pumping enough signal into the input (short of distortion, of course). That will probably do the trick; if not, you're just going to have to reach a little further into your wallet to find a unit with better noise performance.

USING the DOD 690: While mono chorusing units are less expensive, I feel that chorusing really comes to life with stereo setups. One stereo application is to use two amps on stage. This gives a bigger, "live" guitar sound, although not all members of the audience will be seated so as to derive maximum benefit from the stereo effect. Recording is something else altogether—chorusing adds an amazing ambience that helps prevent some of the "flatness" associated with recording electric instruments (particularly those that are run directly into the board). One problem is that in order to record this stereo effect, you need to allocate two tracks to the instrument; however, there's no reason why you can't add chorusing in the mix to a previously recorded mono track. Or, suppose you have a 4-track recorder and have done a premix of several tracks into one track. You may process this premixed track through the 690 during mixdown for simulated stereo plus chorusing.

To hook up the output of the 690 with a mono setup, patch either output jack into your amp. For stereo setups, use two amps and both output jacks.

Incidentally, since the delayed parts of the stereo outputs are out of phase with respect to each other, you might wonder what happens when you play a stereo chorus track over a mono system (e.g., AM radio). All that happens is that the chorusing effect cancels; the basic signal remains unaffected. You also might wonder what stereo chorusing sounds like through headphones—wonder no more, it sounds great.

OVERALL EVALUATION: As with any device that uses an analog delay, how you feel about noise becomes a factor in the purchasing decision—even though the people at DOD have done their best to minimize the nasty effects of delay line noise. Personally, I feel that as long as you keep the input signal as hot as possible, noise is not a significant problem; but, the noise can be noticeable with low-level input signals or particular styles of music. Just remember that noise is a very, very subjective factor; what might be noisy to one musician may be perfectly acceptable to another, so listen carefully when you audition the 690.

My overall opinion is that the DOD 690 is a comparatively inexpensive, ruggedly built and simple to operate chorusing unit that should make it easier for musicians on a budget to enjoy the lush, full sound of stereo chorusing. If that was DOD's goal, they've succeeded.

You have enough to work on in building your act without having to work out to build your strength. Bulky, heavy PA cabinets can cause you a lot of grief. Like backache, arm strain, and crunched fingers. They can get you into heavy trucking costs, heavy roadie costs, and heavy airline costs. They take up too much stage space and distract the audience from your performance. Who needs it?

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Ambient Sound

BY LEN FELDMAN

Only A Few Years Too Late!

You are probably not going to believe this, but the Federal Communications Commission has issued a Notice of Proposed Rule Making concerning a matter that you have probably not given any thought to for the last three years or so. Can you guess what they are about to "rule" upon? It's FM Quadraphonic Broadcasting, folks! Yes, in case your memory of the quad history is a bit dim, let me remind you of the sequence of events.

Way back in 1972 (when 4-channel sound seemed as though it were going to outstrip stereo in popularity at any moment), the Electronic Industry Association sponsored the National Quadraphonic Radio Committee; this committee's objective was to report to the FCC its conclusions regarding quadraphonic FM broadcasting standards. In November 1975 (when quadraphonic records were still doing fairly well and when 4-channel hardware was still enjoying some sales) the NQRC submitted its report and conclusions. It took the FCC nearly two years from that date (June 22, 1977) to issue a Notice of Inquiry (the first step towards adopting rules or standards) whose objective was "to determine the degree of interest in quadraphonic broadcasting and, if sufficient interest was indicated, to develop a record to assist the Commission in formulating standards for service." I should point out that by this time (mid-1977) there was no longer any serious interest in 4-channel sounds, just about all manufacturers had given up making hardware for any of the many types of 4-channel systems and, to the best of my knowledge, recording companies were no longer producing any 4-channel records—at least of the discrete 4-channel type. (It has been rumored that CBS, who is the developer of the SQ matrix type quadraphonic discs, still does encode some present day stereo discs into SQ format, but it's hard to get anyone there to admit this.)

Taking the historical narrative a bit further, in September of 1978, the FCC determined that additional information was needed before standards could be proposed, and in early January of 1979 the FCC issued a further Notice of Inquiry to obtain information concerning the impact that the adoption of quad broadcast standards would have on the possibility of later reducing the channel spacing in the FM broadcast band to 150 kHz or even 100 kHz. Despite the fact that just about everyone involved in broadcasting (from set makers to station engineers) is dead set against narrowing FM channel bandwidth and has expressed that opinion many times in public and directly to the FCC, the Commission stated that it would "consider adoption of quadraphonic broadcasting standard(s) only for such system(s) as could be clearly demonstrated to not preclude future reductions in channel spacing.

Those two little "s" 's in parentheses after the words "standard" and "system" in the preceding paragraph need some heavy explaining too. You see, the latest thing that the FCC seems to be into is called "letting the marketplace decide." All of a sudden, after nearly a half century of setting specific standards, the FCC now believes that even in matters such as broadcasting standards, competition in the marketplace ought to govern which of many stations ultimately wins and predominates. So, it is entirely possible that if and when the FCC *does* finally approve quadraphonic broadcasting, they may do so in such a general way that several systems (each of which is incompatible with the others) may be allowed to go on the air. As an alternative to the adoption of specific standards (and at the moment, these are confined to the systems proposed by either RCA or QSI) the Commission is asking for comments on the desirability of adopting *general* standards. Such general standards would allow *any* type of quadraphonic signal to be broadcast providing it is compatible with existing frequency allocation structure and existing mono and stereo receivers. No minimum performance requirement would be established. For example, no minimum separation requirement would be specified between channels. The argument for such general standards, according to the FCC, is that the technology of quad broadcasting would be free to develop further in the future and broadcasters and listeners could choose which quadraphonic system is more appropriate for their needs. For example, the FCC says, a system with greater coverage but poorer frequency response may be better for one station while another station may favor a system which provides better frequency response with some loss of coverage area.

What amuses me about all this is that the very idea of general standards, in my view, makes about as little sense as having a national telephone system in which some subscribers have six-digit numbers, some have seven-digit numbers and some have ten-digit numbers. If such were the case, you could never reach a household with a ten-digit phone because as soon as you finished dialing the first six or seven digits you would be connected to the holder of that kind of service. If the FCC wants to bury quadraphonic sound for all time (it's already *dead*, for more reasons than I have space to mention, only one of which was FCC procrastination), the surest way to do so is to approve "general" standards which leave it up to individual stations to decide which discrete 4-channel system they would like to use.

If the FCC is now so much in favor of "marketplace" decisions, I cannot help but wonder why there was so little concern for the marketplace when, in 1961, they selected a very *specific* stereophonic FM broadcast system, thereby excluding for all time the possibility of an acknowledged superior system which would have yielded far better signal-to-noise ratios (in stereo FM reception) and very little of the horrendous multipath problem which we suffer with as a result of our FM/AM-subcarrier stereo FM system (the system generally preferred at the time by those who understood the matter technically utilized an all-FM subcarrier/main carrier system).

The sad part of all of this is that even if the FCC were to take what in my opinion is a more realistic approach and choose a specific system or even a specific combination of systems and options, the likelihood that quadraphonic sound will figure largely in the FM broadcast band, or in any aspect of the recording business in the near future is very doubtful indeed. We simply are too late. There are those who will say that it's just as well, but that's really not the point.

As long as I'm in a critical mood, let me bring you up to date regarding another couple of decisions that have come forth from the same FCC. You may recall that I previously reported to you that the FCC had made a preliminary decision regarding stereo AM broadcasting. Although AM is the kind of radio most predominantly listened to in cars (largely because of the poor stereo FM system I have already mentioned), broadcasters have, until now, been unable to play stereo discs in their full two-channel glory over the airwaves, and that, according to everyone who should know, has not only hurt the broadcasting business of AM stations but the recording business as well (not everyone, after all, listens to FM all the time).

Well, for a time there, it looked as though the FCC was going to throw this one out for public decision and "marketplace" popularity polls, too. At least two commissioners voted to let the public decide which of *five* different systems is best. Happily, the other commissioners outvoted these two, and word came down that a system proposed by Magnavox would be the chosen one. As soon as the announcement was made, the flack began. There were threats of injunctions, hollering on the part of the losers, and more screaming by the broadcasters who saw the chosen system as one that would force them to back off a bit on their modulation (and hence give up a bit of coverage). Based upon previous edicts and rule-making, you would think that the FCC would stick to its guns, right? Wrong!

The whole question is up in the air again, and a new notice of inquiry is out seeking additional data to help the FCC to arrive at a technically sound decision. In sum, the FCC is saying that it may have goofed with that preliminary announcement and that it may not have chosen the "best" system after all. Picture, if you will, the reactions at the board room of Magnavox or, for that matter, in the offices of at least one semiconductor manufacturer who is known to have already tooled up for an integrated circuit that would handle the Magnavox stereo AM system!

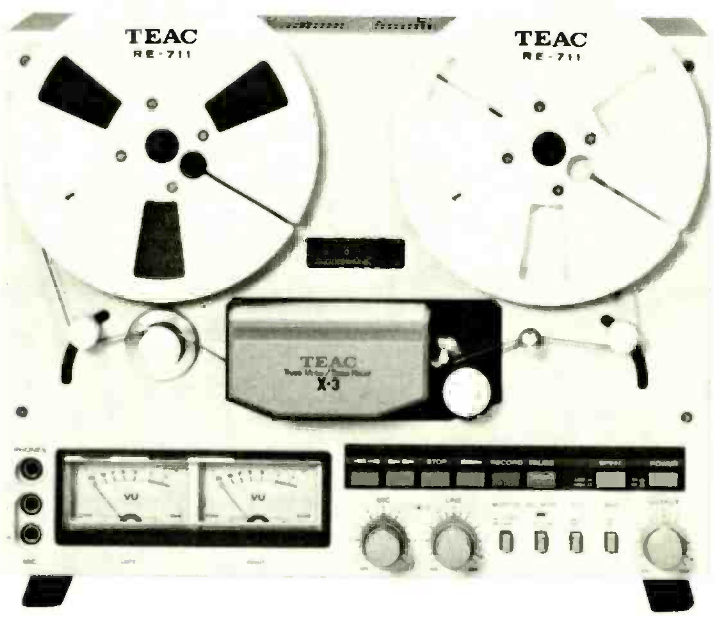
It's anybody's guess as to what the final outcome of all this may be. It is even possible that the "let the marketplace decide" faction may win after all. Or, that one of the other systems will be selected. In any event, don't expect to hear any stereo AM broadcasts of your favorite discs in 1980. We'll be lucky if the decision comes down from the FCC by the end of 1980 and if broadcasting can begin sometime in late 1981. As for myself, I just finished figuring out how the Magnavox stereo AM system is supposed to work, and now I'll have to wait till the FCC makes up its mind and then learn a whole different system. It's enough to drive an audio writer (not to mention a manufacturer, broadcaster or consumer) nuts!

MODERN RECORDING & MUSIC

LAB REPORT

NORMAN EISENBERG AND LEN FELDMAN

Teac X-3 Open-Reel Recorder



General Description: The Teac X-3 is an open-reel, two-speed (7½ and 3¾ inches-per-second) deck with 7-inch reel capacity. Three separate heads (erase, record and play) are used. The quarter-track configuration permits stereo or mono. The transport uses three motors: one DC servo motor drives the capstan, while two induction motors power the tape reels. Transport keys permit fast-button operations including direct recording from the playback mode (over-dubbing or punch-in recording). Front-panel controls include bias and EQ switches (each with two positions), and separate knobs for line and microphone input levels that permit input mixing. Unattended record and playback is possible with the use of an external clock timer.

The tape, coming off the supply reel, engages the left tension arm and a guide roller before entering the tape-head cover, between the capstan shaft and pinch-roller as it leaves the housing, and it engages the right guide post and tension/shut-off arm before winding onto the takeup reel. The four-digit tape counter and its reset button are located between the reels, just above the head assembly. The head cover may be readily slipped off to gain access to the heads.

At the lower left of the panel are three phone jacks—one for stereo headphones and two for left- and right-channel microphones. Next to these jacks are the deck's VU meters (−20 to a bit over +3) which show record or playback level as per the setting of the source/tape monitor switch. The transport buttons include rewind, fast-forward, stop, play, record and pause. In the same horizontal row with these controls are the tape speed selector and the deck's AC power off/on switch.

The bottom row of controls contains the microphone and line input level controls. Both are dual-concentric knobs that permit individual or simultaneous channel adjustment along with input mixing. To their right are the switches for tape monitor, record muting, EQ and bias. The final control at the lower right is another dual-concentric pair of knobs for output level; both line and headphone outputs are controlled at the same time, although the channels on each may be adjusted independently or simultaneously.

The rear of the deck contains phono jacks for stereo

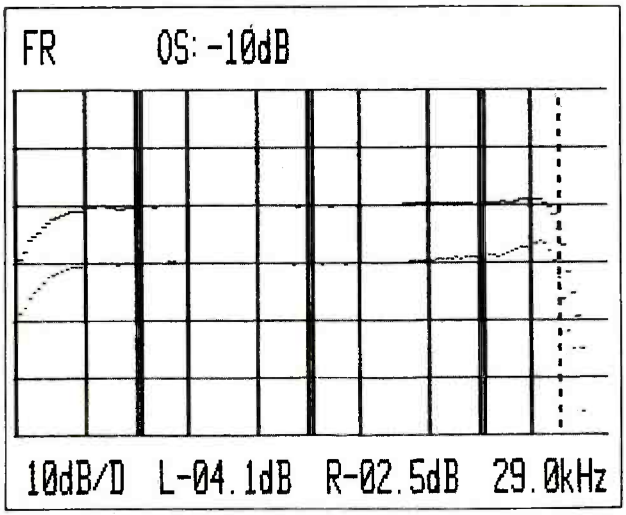


Fig. 1: Teac X-3: Frequency response, 7½ ips, using Maxell UD-XL tape at 0 and −10 dB.

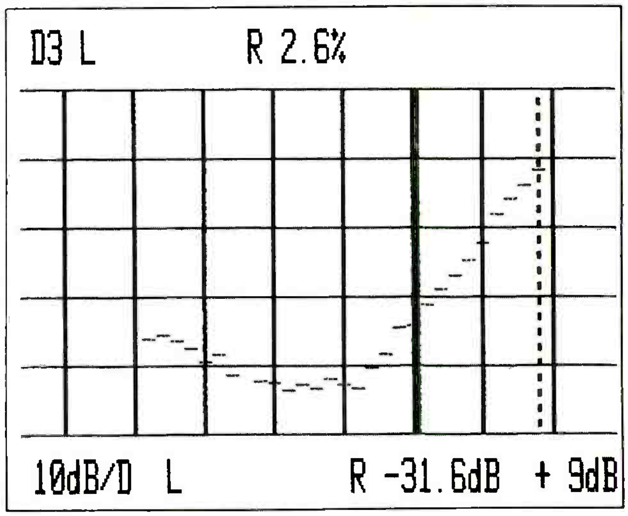


Fig. 3: Teac X-3: Third order distortion at various recording levels, 7½ ips.

line input and output and the machine's AC power cord. Although the deck is intended nominally for upright installation, it also could be placed on its back.

Test Results: Most of the specs for the Teac X-3 were confirmed in our lab tests and a few were bettered. Response at the 7½ ips speed (for the −3 dB points) slightly exceeded specs at both high and low ends; at the 3¾ ips speed, the −3 dB point occurred a little higher at the bass end than claimed, but went beyond the 20 kHz claimed for the high end. Since much of the data shown in our "Vital Statistics" table was derived from the new test setup we have been using, in reading the following material, the reader is advised to refer to that table and also to the various figures which were produced with the aid of a video printer associated with the test set.

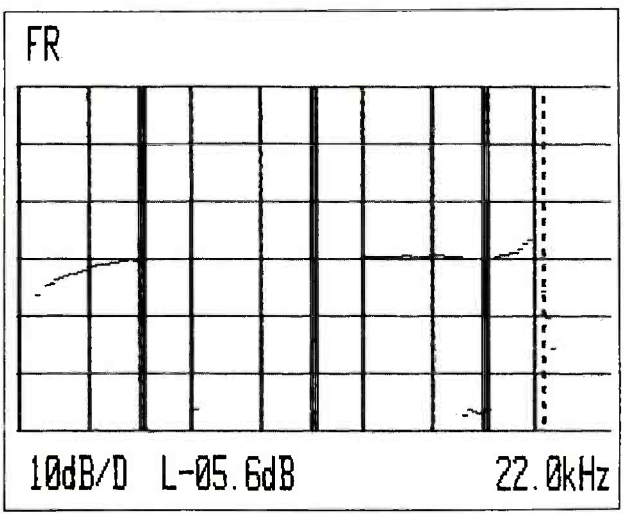


Fig. 2: Teac X-3: Frequency response, 3¾ ips, at −20 dB record level.

Note, for example, the response curves of *Fig. 1* (7½ ips) and *Fig. 2* (3¾ ips). *Fig. 1* shows response at a 0 dB level (upper curve) and at a −10 dB level. The latter curve is designated as "R" channel simply because the machine can only store one "L" and one "R" channel at a given time. In fact, both curves were taken for the same channel. The printout tells us that, with the cursor (vertical dotted line) set to 29 kHz, response is down 2.5 dB at a −10 dB record level, and is down 4.1 dB at the 0 dB record level. We list 29 kHz as the upper frequency for this deck, with the particular tape we used (Maxell UD-XL) and referenced to a −10 dB record level. Actually, the response extends slightly beyond 29 kHz since at that frequency it was down only 2.5 dB. For all practical purposes, however, 29 kHz is good enough. The low frequency limit (29 Hz) at this speed was determined by moving the cursor to the opposite end of the graph, but we did not bother to get a printout of the results since this curve is, of course, the same as in *Fig. 2.*

While the response at 3¾ ips was not as good as at 7½ ips, it still was better than expected, extending as it does to 21.5 kHz. Note that in *Fig. 2*, response is shown as being down −5.6 dB at 22 kHz. It was down only 2 dB at 21 kHz, so we had to interpolate for the −3 dB point which we estimated as being at 21.5 kHz.

Figs. 3 and *4* depict the 3rd-order distortion during playback of a tape which had been recorded at succes-

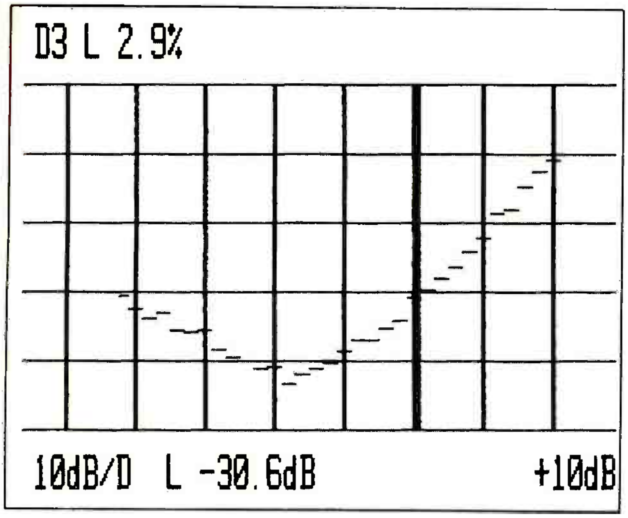


Fig. 4: Teac X-3: Third-order distortion level at various record levels, 3¾ ips.

sively lower and lower mid-frequency record levels. The double vertical line in each is the 0 dB record-reference level. We see here that at the higher tape speed, headroom for 3-percent third-order distortion is a bit better than +9 dB, while at the slower speed the headroom is actually 1 dB greater.

The signal-to-noise readouts in *Figs. 5* and *6* contain a seeming discrepancy between the data shown in the graphs and the figures listed in the "Vital Statistics" table. There is, however, no discrepancy. The test device reads everything with respect to 0 dB record-reference level. So, if you add the 9 dB of headroom (for maximum record output level) that was measured earlier (for the 3-percent distortion up to the 52.7 dB observed for the right channel—"worst case"), you come up with the value of 61.7 dB shown in the Vital Statistics table for S/N. Similarly, adding the 10 dB of headroom for the 3¾ ips speed to the 51.6 dB of S/N shown in *Fig. 6*, produces the figure of 61.7 dB. What is strange in this is the fact that both headroom and signal-to-noise are virtually identical for this machine at both of its operating speeds.

Interestingly, wow-and-flutter at both speeds also was almost the same.

General Info: Dimensions are 16⅛ inches wide; 23¹³⁄₁₆ inches high; 9⅛ inches deep. Weight is 30 lbs., 14 oz. Price: $550.

Individual Comment by N.E.: If nothing else, the Teac X-3 demonstrates the general superiority of the open-reel format—when conscientiously designed and built—that it is possible to obtain at a given price level vis-a-vis the cassette format. Of course, this is not the first instance of this sort we have observed recently (the Akai GX-625 reviewed in our last issue is an excellent case in point, as was an earlier Pioneer RT-707). What it means in the long run may be that a new trend is shaping up—the "revival" of open-reel for the mass market. Even if things don't reach such proportions, though, tape decks such as this fairly shout at the prospective buyer: "Here I am at a price very competitive with cassette decks, and with performance and facilities that really count in terms of where it's at in serious (pro and semi-pro) recording which is still, and likely to remain, an open-reel world." Viewed in this light the Teac X-3 could appeal to a number of persons just getting into tape-recording, or to those who are ready to "graduate" from cassettes to the open-reel format.

Individual Comment by L.F.: It is so nice to be able to talk about a lower-cost reel-to-reel machine that is designed for use with an audio system rather than as the foundation for a recording studio. There are, after all, thousands of our readers who want to record just

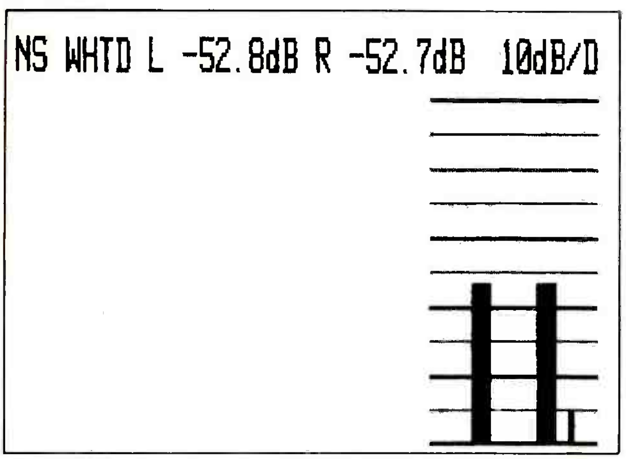


Fig. 5: Teac X-3: S/N (CCIR weighted) referenced to 0 dB record level, 7½ ips.

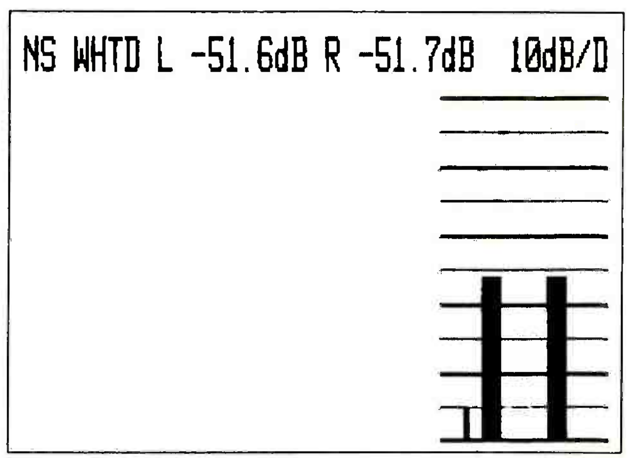


Fig. 6: Teac X-3: S/N (CCIR weighted) referenced to 0 dB record level, 3¾ ips.

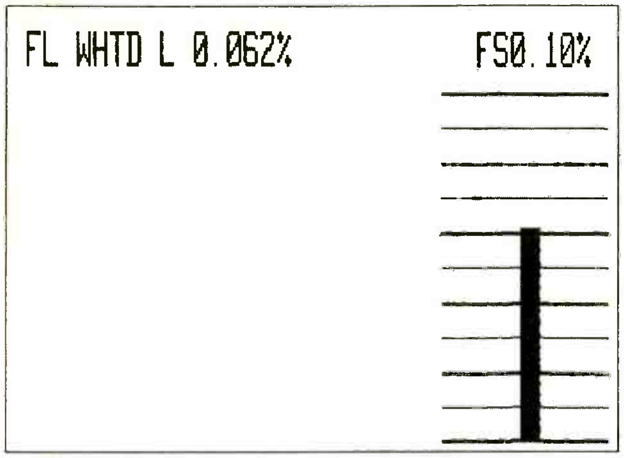


Fig. 7: Teac X-3: NAB weighted flutter at 7½ ips.

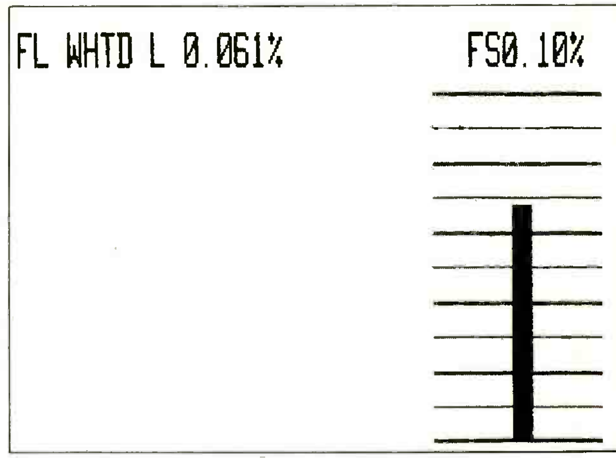


Fig. 8: Teac X-3: NAB weighted flutter at 3¾ ips.

for fun (with no ambitions for making and selling the next gold or platinum disc to hit the charts), who *don't* ever need to spend money for tape operating at the "pro" speeds of 15 or 30 ips but who would not be content with even the best stereo cassette deck. They may want to do some editing, for example. (Ever try editing a standard-sized cassette tape?) Or, perhaps they want to do some punch-in recording (to correct a section of a previous recording without having to splice in a new piece of tape and without running into switching thumps and clicks at the punch-in position).

These recording techniques are easily done with a machine such as the X-3, whereas they are almost impossible to do with even the finest-performing cassette deck that I know of. On the other hand, if you are looking for such fancy niceties as sel-sync track synchronization and track-by-track recording you will not find it in the Teac X-3.

I found the Teac X-3 to be an easy machine to operate. Tape threading is also quite simple and the transport operated flawlessly, with no tendency to stretch or tear tape no matter what buttons I pushed. VU meters seemed extremely accurately calibrated and I liked their ballistics, even if they did not correspond exactly to VU standards.

One added bonus that comes with this or any other Teac open-reel tape deck is the excellently written little "Information Supplement" concerning open-reel tape decks in general. This little pamphlet contains a wealth of information about reel-to-reel recording and the tape that is used with such recorders. If you have read it before, I suggest you read it again as a refresher course. If you've never seen it, your first encounter with it will teach you a great deal about this subject. Clearly, the pamphlet alone does not justify purchase of this new reel-to-reel deck from Teac, but the deck is well worth its asking price on its own merits and will be much appreciated by recording fans who thought they couldn't ever afford to go to this oldest—and still best—of all tape recording formats.

TEAC X-3 OPEN REEL TAPE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Tape Speeds	7½; 3¾ ips	Confirmed
Reel Capacity	7 inches	Confirmed
Wow/flutter, 7½; 3¾ (NAB wtd)	0.05%; 0.07%	0.062%; 0.061%
Third order dist. at 0 VU	0.9%	0.16%; 0.28% (7½; 3¾)
Frequency response, 7½	± 3 dB, 30 Hz to 28 kHz	± 3 dB, 29 Hz to 29 kHz
3¾	± 3 dB, 30 Hz to 20 kHz	± 3 dB, 37 Hz to 21.5 kHz
Best S/N ratio, std. tape	58 dB	61.7 dB (either speed)
Fast-wind time, 1800 ft.	100 seconds	100 seconds
Mic input sensitivity	0.25 mV	0.25 mV
Line input sensitivity	60 mV	58 mV
Line output level	0.45 V	0.90 V
Headphone output level	NA	96 mV
Erase ratio	NA	97 dB
Speed accuracy	± 0.5%	± 0.2%

Peavey Stereo Graphic Equalizer

General Description: The Peavey Stereo Graphic Equalizer (no model number, although to us it seems a model number would have made the unit easier to describe and identify) offers ten bands of EQ on each of two independent channels. In addition, each channel contains its own high and low filters plus separate level controls and EQ in/bypass switching. The EQ sliders for each channel have their center frequencies at 30, 60, 120, 250, 500, 1 K, 2 K, 4 K, 8 K and 16 K Hz. The EQ range of ±15 dB is clearly marked, with detents for the sliders at the 0 dB position.

The ±12 dB-per-octave filters are continuously variable, with markings on the low-cut knobs for 20, 50, 100, 250 and 500 Hz, and markings on the high-cut knobs for 5 K, 7.5 K, 10 K, 20 K and 30 K Hz. The level controls also are continuously variable, with markings on each from −15 through 0 to +15 in 3-dB increments. These controls, plus the unit's AC power off/on switch make up the front panel which is of standard rack-mount width, fitted with handles and slotted at the ends.

Connecting facilities are at the rear. For each channel there are a pair of 50 K ohm quarter-inch phone jack unbalanced inputs, an XLR 50 K ohm unbalanced input, a pair of 600-ohm phone-jack unbalanced outputs and a 600-ohm XLR balanced output. The rear also has a ¼-amp fuse. The unit's line cord is fitted with a 3-prong (grounding) plug.

Test Results: Lab tests of the Peavey Stereo Graphic Equalizer confirmed most, but not all, of its published specs. The overall impression, however, was one of an effective, professional-grade device. Frequency response was plotted automatically, using the new Sound Technology 1500A instrument which—although it is primarily a tape and recorder tester—easily performs many of the measurements required for amps, preamps and other electronic products. In *Fig. 1,* frequency response is seen to extend out to 29 kHz for an attenuation of 1.2 dB. By interpolation, we therefore listed the −1 dB rolloff point at 28 kHz, as opposed to the 30 kHz quoted by the manufacturer. We suppose if one wanted to be totally fair, one could say that since the manufacturer quotes a ±1 dB figure, and since there were no points in the plot that went to the "plus" side, the response actually could be described as extending to beyond 30 kHz. On the other hand, we feel that the intent of a frequency response measurement is to inform one of when the response begins to roll off or become attenuated. Thus, our reported figures. In-

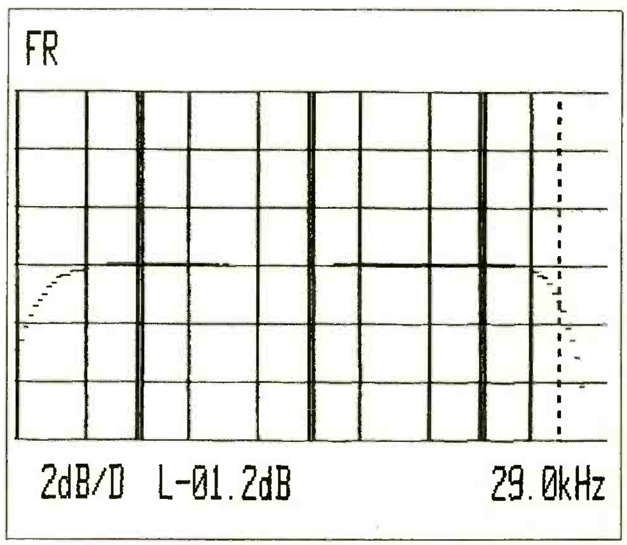


Fig. 1: Peavey Stereo EQ: Frequency response with controls set to flat.

cidentally, the scale in *Fig. 1* is 2 dB per division.

For the plots in *Fig. 2* we used the test set to illustrate the action of the Peavey's low-cut and high-cut filters. The highest bass curve and the lowest treble curve correspond to extreme filter settings (high-cut set fully counterclockwise; low-cut set to maximum clockwise). The other two curves correspond to mid-settings of the two filter controls. For these tests, the 0 dB reference, as indicated by the nomenclature on the panel, was set at 1 kHz. Note that with the controls set for extreme cut of bass and of treble, the bass cut begins at around 350 Hz (for the −3 dB point), while the treble cut's −3 dB point occurs at around 3.5 kHz. (The double-vertical lines in the displays of *Figs. 1, 2, 3* and *4* are at 100 Hz, 1 kHz and 10 kHz. The single vertical lines are at 20 Hz, 50 Hz, 200 Hz, 500 Hz, 2 kHz, 5 kHz and 20 kHz.)

Figs. 3 and *4* show typical boost and cut action of a single band control. For *Fig. 3* we alternately boosted and cut the 500-Hz slider maximally, while for *Fig. 4* we adjusted the 8-kHz control in similar fashion. Ignore the "L" and "R" notations in these displays; they are simply a convenient way of storing two independent curves on the face of the video screen of the test setup. However, the dB notation associated with "L" and "R" in each case are quite correct—that is to say, we obtained a cut of −11 dB and a boost of 10.4 dB at 500 Hz *(Fig. 3)*, while for the 8-kHz control we measured a boost of 9.7 dB and a cut of −11.5 dB.

In *Fig. 5* we plotted the response obtained when all of the band controls were moved to their maximum boost positions (upper curve), and when they all were moved downward for maximum cut. Under these additive conditions, for example, at 540 Hz there is maximum boost capability of 18.2 dB, and maximum cut of 20.2 dB, as indicated below the graph.

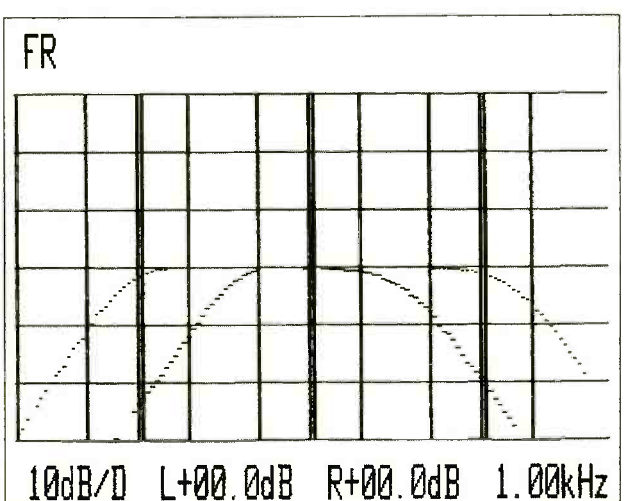


Fig. 2: Peavey Stereo EQ: Examples of high- and low-cut filter action.

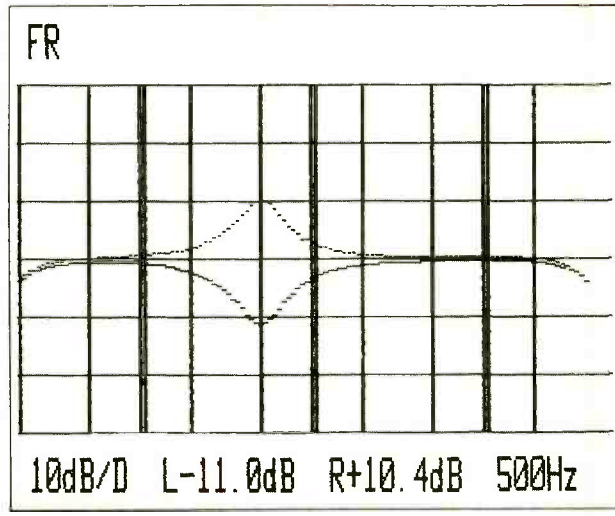


Fig. 3: Peavey Stereo EQ: Range of control, 500 Hz octave lever.

General Info: Dimensions are 19 inches wide; 5⅝ inches high; 7⅝ inches deep. Weight: 20 lbs. Price: $299.50.

Individual Comment by L.F.: Since most of today's graphic equalizers do essentially the same thing and do it fairly well, I looked for some differences between the typical 10-band equalizer and this entry from Peavey. One important difference I noted immediately was the presence of variable gain controls on the Peavey. These combination gain/attenuator controls (there is one for each channel so that they can be separately adjusted) provide 15 dB of boost or cut and thus allow level matching in a wide variety of operating situations.

Installation flexibility is enhanced by the incorporation of both low-impedance unbalanced XLR type and

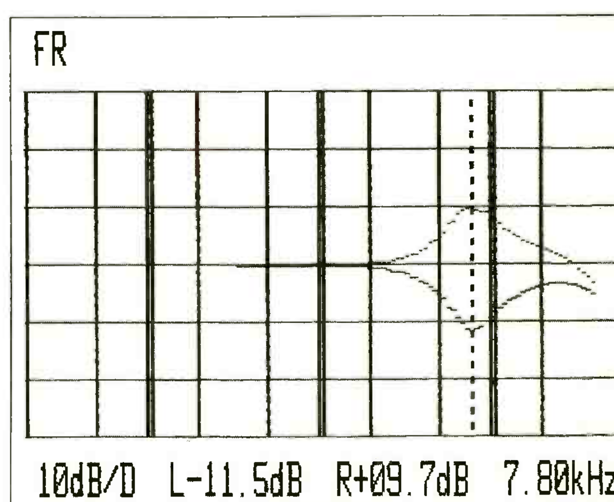


Fig. 4: Peavey Stereo EQ: Range of control, 8 kHz octave lever.

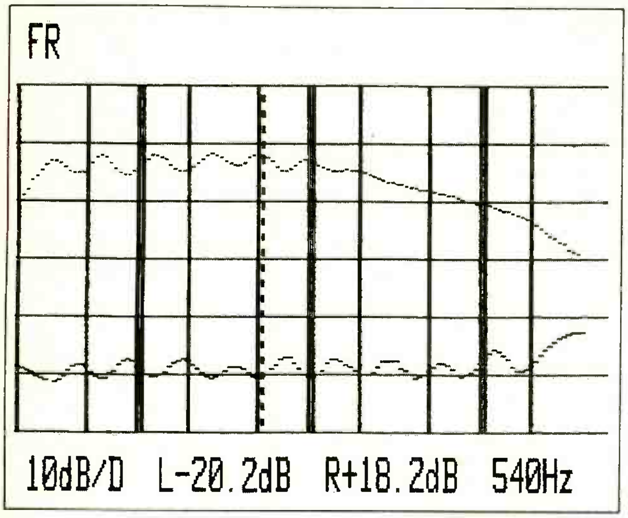


Fig. 5: Peavey Stereo EQ: Range and response when all controls are at full boost or cut.

high-impedance phone-plug (¼-inch) inputs. Outputs are low impedance (600 ohms or less), and balanced outputs are provided from a standard XLR male-type connector. I encountered no measurable signal degradation or level loss using a relatively long (in excess of 50 feet) twisted-pair cable from the equalizer to the measuring instruments.

I believe the Peavey equalizer is one of those rare devices that can prove equally effective in sound reinforcement, recording studio and even home-audio applications. In sound-reinforcement work, for instance, the availability of the low and high filters allows the operator to contour the sound of a vocal channel and still have a quality vocal system with higher output and without the typical interference from a nearby drum-set or bass guitar. In home system use, of course, the filters could help in removing turntable rumble and/or high-frequency hiss.

As with any quality graphic equalizer, a bit of experimentation is essential. My own experience has shown that most newcomers to equalizers misuse them at first. Only later, after they have had a chance to experiment with them at some length, has common sense prevailed and desired results been obtained. The Peavey equalizer offers the needed flexibility and controls that one would expect of a unit in its price category, and it contributes negligible noise and distortion to the system. It is professionally designed and built, and it should be able to withstand continuous and rugged use for many years.

Individual Comment by N.E.: I have only one quibble over this unit: A device that is as well-built and as versatile as this does merit an instruction manual that would be more detailed (and illustrated) than the single sheet with very small type that accompanied my test sample.

PEAVEY STEREO GRAPHIC EQUALIZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Gain:		
Unbalanced output	0 dB	0 dB
Balanced output	+6 dB	+6 dB
Input dynamic range:		
@ 0 dB level setting	5 V max	6.0 V
@ −15 dB level setting	7 V max	7.5 V
Output Level:		
Unbalanced, 600 ohms	5 V	5.5 V
Unbalanced, 10 K ohms	7 V	7.5 V
Balanced, 10 K ohms (bal. load)	12 V	12.0 V
Balanced, 600 ohms (bal. load)	8 V	9.0 V
Balanced, 10 K ohms (unbal. load)	7 V	8.0 V
Balanced, 600 ohms (unbal. load)	5 V	5.5 V
Equalization range @ 30, 60, 120, 250 500 Hz, 1, 2, 4, 8, 16 kHz	±15 dB	±11 dB *(See Fig. 3 & 4)*
Low-Cut filter range/slope	20-500 Hz/12 dB	20-350 Hz/12 dB *(See Fig. 2)*
High-Cut filter range/slope	5 kHz-30kHz/12 dB	3.5 kHz-30 kHz/12 dB *(See Fig. 2)*
Frequency response @ 1 V in. (all controls flattest)	20 Hz-30 kHz, ±1 dB	29 Hz-28 kHz, ±1 dB *(see Fig. 1)*
THD @ 1 V in. (controls flattest)		
20 Hz to 20 kHz	0.08%	0.25%
1 kHz	0.02%	0.014%
Hum and noise (controls flattest) re: 1.0 V out, unweighted	80 dB	74 dB

Fender SRA 400 Power Amplifier

General Description: The Fender SRA-400 is a stereo (two-channel) power amplifier rated for 100 watts output per channel into 8-ohm loads, or 200 watts per channel into 4-ohm loads. It also may be used in bridged monophonic mode for power output of 400 watts into an 8-ohm load. Mono operation into a 4-ohm load is not recommended.

A switch at the rear actually provides for three modes of operation: normal stereo, mono and bridged. In the mono mode, the SRA-400 remains a two-channel amplifier but the inputs of each channel are connected together. The resultant "A plus B" signal then is fed to both channels thus eliminating the need for a "Y" connector. In bridged mode, the SRA-400 becomes one 400-watt single-channel amplifier. The input is applied only to "channel 1" and gain is controlled by the channel 1 attenuator. In this mode, the speaker is connected across the two "positive" output terminals at the rear.

Built into the amplifier is a two-speed ventilating fan that runs in low-speed when power is switched on. The fan goes into high-speed only when heat-sink temperatures rise above 140 degrees Fahrenheit (60 degrees Centigrade)—which, by the way, it never did during our tests. The fan is located behind the rear panel and it vents through the front panel.

The front panel contains two identical stepped attenuator controls, one per channel; and two vertical LED meters that show output power for each channel. The attenuators are marked and detented for twenty-two steps. Each of the metering scales has ten LEDs calibrated downward from 0 dB to −27 dB. The LEDs are color-coded green, orange and red; the red LEDs indicate full power. The AC off/on switch is at the left. The panel is fitted with handles and is slotted for standard EIA rack-mounting.

The rear of the amplifier contains the three-position mode switch already mentioned, plus the inputs, outputs and protective fuses for each channel. Inputs are quarter-inch phone jacks (unbalanced). Two types of outputs are provided: quarter-inch jacks and banana jacks (color-coded binding posts) wired in parallel.

Test Results: In our tests, the Fender SRA-400 far exceeded its specifications for power and distortion, as a glance at our "Vital Statistics" table will show. It delivered actually 153 watts per channel into 8-ohm loads for its rated THD of 0.09 percent, and even at the frequency extremes of 20 Hz and 20 kHz, its power output was well in excess of rated levels, reaching 144 watts before rated distortion occurred. With a 4-ohm load, maximum power per channel with each channel driven was 232 watts as against the 200 watts claimed.

In addition to these available power levels under static test conditions, it is important to understand the dynamic headroom results we obtained in our tests. Precise measurement of this characteristic is a bit difficult since it depends upon observation of the start of clipping by studying the display on an oscilloscope and making a visual estimate of signal amplitudes. This could account for the fact that we measured a dynamic headroom of 2.5 dB while Fender claims a higher value of 2.9 dB. Even so, the 2.5 dB figure is awfully good since it means actually that under short-term musical signal conditions, this amplifier can deliver unclipped power peaks of nearly 180 watts. Our figure for dynamic headroom is referenced to 8-ohm loads. With 4-ohm loads (and using a 200-watt-per-channel reference level) the amp's dynamic headroom is a bit lower, but it still is high relative to that found on most power amplifiers of recent vintage.

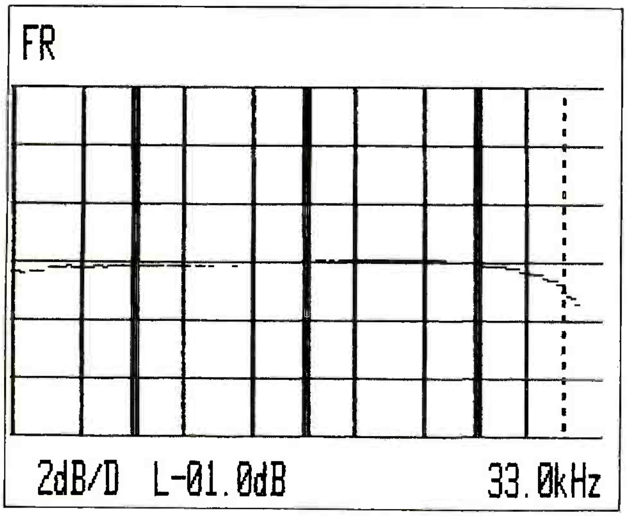


Fig. 1: Fender SRA-400: Frequency response (vertical sensitivity of display is 2 dB per division).

General Info: Dimensions are 19 inches wide; 5¼ inches high; 17.3 inches deep. Weight is 43 pounds. Price is $945.

Individual Comment by L.F.: The Fender SRA-400 is specifically described as an amplifier intended for professional sound-reinforcement applications. I never have been quite sure of what that means. Is an amplifier "professional" because it has a two-speed thermally switched fan that forces air from the rear through a tunnel and out the front? Is it professional because it uses sixteen 25-watt power transistors mounted on more than 6000 square centimeters of heat-sink area and employs fully complementary circuitry? Is it professional because it incorporates such protective measures as fact-action speaker fuses, electromagnetic circuit-breakers in the power-supply primary circuit, over-temperature sensing and output transistor commutating diodes which make it practically abuse-proof?

All of these features, of course, do contribute to an amplifier's professional quality, but over the years I have learned that one person's professional amplifier can just as easily serve as another person's high-fidelity amplifier. In the case of the Fender SRA-400, my own subjective listening tests were performed with this amplifier hooked up as a two-channel or stereo power amp. I can appreciate how simply the unit can be converted for mono sound or the bridged mode; that rear switch does it all with no need for internal wiring changes. I suppose when you get down to it, what distinguishes this professional amplifier from many "home" units are its ruggedness and its "conservative" specifications. Here, after all, is an amplifier that is rated for 100 watts per channel into 8 ohms and weighs 43 pounds net. (Compare this with the tiny home-type Carver amplifier—tested last month—which was rated for 201 watts per channel and weighed less than 10 pounds!) Both amps sounded fine to me, but as I reported concerning the Carver amp, it has what amounts to a warning in its instructions which clearly tells the user that it is *not* intended to take the abuse of "professional use" but is strictly a home music-system amplifier.

During my listening tests, the temperature inside the Fender never exceeded the 60° C. (140° F.) level which would have shifted the fan into higher speed. As a result, the sound of the fan was almost imperceptible, and with the amplifier positioned several feet from my listening chair, I was not aware of any fan noise at all. Bass sounded extremely tight and it seemed well-damped. Overall tonal balance was excellent with the amplifier driving my medium-efficiency reference speaker systems. I did not try bridged operation in the listening tests although the bridged mode was confirmed in our bench tests.

Certainly one can obtain this much power in an amplifier costing less than the Fender SRA-400, even in the professional amp category and certainly in a home music system amplifier. The question of price versus performance really boils down to the need (or lack of it) for uncompromised reliability and the unwillingness to tolerate "down time." If these are criteria that you must consider when choosing an amplifier for professional applications, then the Fender SRA-400's price begins to look less formidable.

Individual Comment by N.E.: Without a doubt, the Fender SRA-400 possesses the "three Rs" sought by the professional user: reliability, ruggedness and roadability. We also might add a fourth "R"—that of response to audio signals. It should be clear to the audio-minded by now that the term "professional" as applied to an amplifier need not imply inferior performance vis-a-vis "home" or "audiophile" type amplifiers. Just because an amplifier is built "like a battleship" and offers pro-type connecting and metering facilities need not mean that its "sound" has to yield anything in the way of accurate reproduction to other "less built" amplifiers. We have had several amplifiers through our testing mill that do combine the desirable attributes of both types of equipment, and the Fender SRA-400 certainly is one of them. It is definitely pro-grade equipment and it also performs like a superior music-system amplifier with wide-range, low-distortion, crystal-clear sound. The fan's noise is barely audible when listening up close with no signal going through the system, but it is completely masked at signal output levels of about 65 dB/SPL measured at a listening distance back about 4.5 feet. At a few feet farther back, the fan's noise level is lower than the ambient noise level of the room itself.

FENDER SRA-400 AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Continuous power per channel		
for rated THD, 8 ohms, 1 kHz	100 watts	152 watts
4 ohms, 1 kHz	200 watts	232 watts
FTC rated Power (20 Hz to 20 kHz)	NA	144 watts
THD at rated output, 8 ohms, 1 kHz	0.09%	0.0075%
4 ohms, 1 kHz	0.09%	0.012%
8 ohms, 20 Hz	NA	0.053%
8 ohms, 20 kHz	NA	0.02%
IM distortion, rated output, SMPTE	0.03%	0.013%
CCIF	NA	<0.03%
IHF	NA	<0.05%
Frequency response at 1 watt (for −1 dB)	NA	10 Hz to 33 kHz
S/N ratio re: 1 watt, "A" wtd, IHF	NA	83 dB
S/N ratio re: rated output, "A" wtd	NA	100 dB
Dynamic headroom, IHF (8 ohms)	2.9 dB	2.5 dB
Damping factor at 50 Hz (8 ohms)	130	130
IHF input sensitivity	NA	0.13 volt
Input sensitivity re: rated output	1.28 volts	1.30 volts
Slew rate (volts/microseconds)	40	40
Power consumption, idling; maximum	NA; 860 watts	175; 1060 watts (4 ohms)

CIRCLE 16 ON READER SERVICE CARD

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Tapco 7416R Mixing Console

By John Murphy and Jim Ford

Tapco has recently introduced two new series of sound reinforcement mixing consoles of which the 7416R is representative. The Series 74 mixers all feature 4 x 2 x 1 output sections with a choice of 8, 16, 24 or 32 input channels. The 7416R is the 16-input version from that series. The "R" suffix indicates that the mixer is equipped with Tapco's new optionally available "Adjustable Decay Reverb."

Featuring input sections identical to those of the Series 74 mixers, the Series 72 mixers provide stereo and mono outputs, lacking only the four channel submaster section of the Series 74 units. The reverberation option is also available for the Series 72 mixers for an additional charge of $220.

The packaging of the 7416 is rather compact as the entire console is only 31½ inches wide. Attractively stained solid oak end panels add a nice sense of warmth to the appearance of the mixer. When the unit is placed on a level surface, the entire top control area slopes downward toward the operator, providing excellent visual and manual access to the controls.

The mixer is absolutely loaded with control features such as: extensive soloing facilities, talkback system, special provisions for stereo tape input, a headphone monitoring system, separate effects return to monitor controls, a selectable pre/post fader auxiliary mix, as well as channel and sub-group patch points. With this kind of control flexibility, this could just be a soundman's "dream mixer!"

The 7416R (with optional reverb) is priced at $3,415.

General Description: Let's now look at the 7416 a bit closer. First we'll go over one of the sixteen (identical) input sections in detail, then we'll discuss the sub-group section and the output section.

Each input channel of the 7416 accepts either a mic input signal or a line input signal. The input connections are made at the rear of the unit directly in line with the associated input strip. The mic input connection is made by way of a non-locking 3-pin XLR connector; the line input connector is a ¼" phone jack. A pushbutton switch at the top of the input strip is used to select between the mic and line inputs.

At the very top of the input strip, just above the mic/line switch, are a pair of ¼" phone jacks labeled "Send" and "Return" which make up the channel patch point. These jacks are wired such that the send and return are normally connected (i.e., "normalled"). The normal connection is interrupted only when a plug is inserted into the "return" jack. In terms of the signal flow through the channel, the patch point is immediately after the EQ section and just before the channel fader. The patching connections might typically be used to route a channel's signal through an outboard signal processor (e.g., compressor, flanger, noise gate, etc.). Additionally, the "sends" can be used as direct outputs for the channels.

Continuing down the input strip, we see an input trim control and LED clip indicator just below the mic/line switch. The rotary trim control adjusts the gain of the first preamp stage in the channel for both the mic and line inputs. The clip LED illuminates whenever its circuitry detects an overload condition anywhere in the channel.

Below the trim control is a set of three rotary controls designated "Aux," "Monitor" and "Effects/Reverb." These controls are used to set the level of the channel signal in these various "side-mixes." The monitor control, for example, is used to set the level of the channel in the monitor mix that would normally be returned to the stage through a separate stage monitor signal chain for the performers to hear. The signal sent to the monitor mix is taken from a point after the channel EQ but before the channel fader. This allows the monitor mix to be established independent of the channel faders. The Effects/Reverb send, on the other hand,

is taken post-fader so that when a channel fader is pulled down, the effects send from that channel is likewise reduced in level. The Aux send from the channel can be either pre- or post-fader, depending on the direction the control is rotated from its center position detent. Clockwise rotation provides increasing level of post-fader send. Counterclockwise rotation from the detent position provides increasing level of pre-fader send. This control action provides an extra measure of flexibility in setting up the Aux side-mix by allowing the Aux mix to be used as either an extra monitor mix (pre-fader) or as an additional effects mix (post-fader) or for anything else the operator can dream up! Each of the three side-mix controls has an associated master level control in the output section. In the case of the 7416R, which is equipped with a built-in reverberation system, the master Effects/Reverb send is tied directly into the input of the reverb unit in addition to appearing at the rear panel effects send jack. The output of the reverb is tied into the Effects Return A in the output section but can be defeated (if desired) simply by inserting a plug in the Effects Return A jack on the rear panel.

The EQ control group is located below the side-mix controls. The high and low frequency EQ is of the shelving type with nominal shelving frequencies of 10 kHz and 100 Hz and a range of about ±18 dB. Mid-range frequency control of ±12 dB is switch selectable between 600 Hz and 3.5 kHz. EQ response curves are provided at *Figures 1* and *2*.

Assignment of the channel signal to the sub-group section is made by way of two push buttons working in

conjunction with the channel pan pot. The first assign switch is labeled "1-2" and assigns the signal to sub-group pair 1 and 2 when depressed. The channel pan pot is then used to pan between sub-groups 1 and 2. Similarly, the second assign switch directs the signal from the channel to sub-groups 3 and 4. By selecting the correct combination of assignment and pan-pot setting, the channel signal can be assigned to any individual sub-group or any combination of sub-groups.

Just below the pan pot and above the channel fader is the channel solo switch which when depressed allows (post-fader) solo monitoring of the channel signal without disturbing the main mixer outputs. (According to Tapco, a simple service center modification would allow pre-fader solo monitoring if desired.) When any solo button is depressed, the main stereo output that is normally fed to the headphone amplifier is temporarily replaced by the signal (or signals) on the solo bus.

Finally, at the bottom of the input channel is the channel fader. This is a slide-type level control that sets the output level of the channel signal being fed to the sub-groups. All the faders on the 7416 have a travel of 6 centimeters (a little less than 2½ inches).

The left-most two-thirds of the console face is occupied by the sixteen input channels. To the right of the input channels is the sub-group section, while the stereo/mono output section is located at the far right end of the mixer. Signals from the input channels are assigned to the sub-groups where the sub-group faders are used to adjust the relative group levels. The four sub-group signals can then be individually panned to any desired left/right position in the stereo mix.

At the top of the sub-group section are four fluorescent bargraph-type level displays (rather than conventional VU meters), one for each of the sub-groups. The displays are horizontally oriented and are calibrated from −20 dB to +8 dB. Below the level indicators are a pair of rotary controls labeled "Stereo Tape In." One control adjusts the level at which the stereo tape signal is added into the mixers main stereo output; the second control varies the left/right balance of the stereo tape signal.

The board's talkback system allows the operator to use the rear panel talkback mic input to communicate (talkback) through any of the main, monitor or aux mixes. Controls for the talkback system include a rotary level control and three momentary push buttons labeled "Mains," "Monitor" and "Aux." These are located in the sub-group section below the stereo tape in controls.

Sub-group controls are located at the bottom of the section and consist of a rotary pan pot, a solo switch and a fader for each of the four sub-groups.

Addressing our attention now to the output section, we see at the top another pair of fluorescent bargraph level indicators which normally monitor the levels at the left and right main stereo outputs. Depressing a

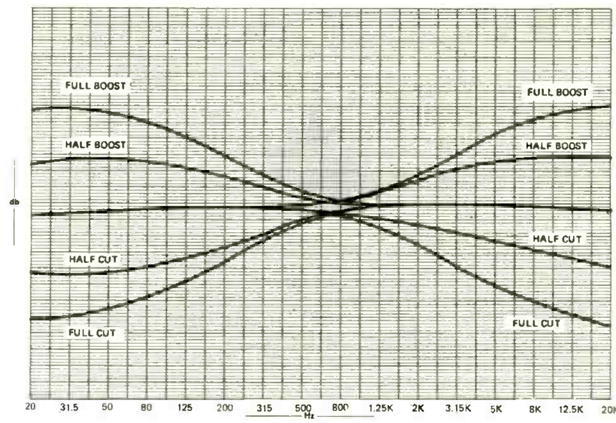

Fig. 1: Tapco 7416: High and low frequency EQ response curves for the mixer.

push button below the bargraphs switches them to indicate the output levels of the mono and solo signals. Located below the meter switch is an LED indicator labeled "Phantom Power 24V" which when illuminated, indicates that the built-in microphone phantom powering system is switched on. The Phantom Power on/off switch is located on the rear panel next to the main power on/off switch for the mixer. Tapco informs us that by December 1, 1980 the changeover from the powering of 24 volts to the more widely used standard of 48 volts will have been achieved.

Master level controls for the Aux, Monitor and Effects/Reverb outputs are neatly organized in a column above the left stereo master fader along with their associated solo switches. Across from these master controls and organized about the right stereo master are the controls for Effects/Reverb Return A and Effects Return B. For each return there is a level control and pan pot; additionally, there is a control labeled "To Monitor" which allows the effects to be returned to the monitor mix at any level desired.

Immediately above the left stereo master fader is a rotary-type control labeled "Mono Master" which serves as the master level control for the mixers' mono output. The mono signal is derived by summing the left and right stereo master signals. The Mono Master control, like the faders, has a "Nominal" setting identified by a heavy white line in the graphics. Tapco suggests that the faders be used near the nominal positions in order to obtain the best noise and distortion performance.

The headphone level control is located immediately above the right stereo master fader along with a push-button switch which selects between stereo and mono operation of the headphone amp. Also, associated with the headphone controls is a LED indicator labeled "Solo" which illuminates whenever any solo button is depressed to indicate that the solo system has interrupted the normal stereo headphone operation.

The left and right stereo master faders at the bottom of the output section control the overall level of the

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main left and right outputs.

Virtually all of the input and output connections to the 7416 are made at the rear panel by way of ¼" phone jacks. The exceptions to this are the mic inputs to the channels and for talkback which use 3-pin XLR connectors, and the stereo tape input connectors which are phono jacks. At the rear of the sub-group section are four pairs of normalled sends and returns which make up the sub-group patch points. There are also four sub-group stacking inputs which allow an external signal to be summed into the sub-group. One possible use of these connectors would be to add-in the sub-group outputs of another mixer for "master/slave" operation.

At the rear of the output section are the main output connectors along with "Solo Audio Out," effects returns A and B and seven different stacking inputs. The main outputs are from left to right: "Left"; "Right"; "Mono"; "Effects Low" (−16 dBV level); "Effects High" (+4 dBV level); "Monitor"; and "Aux." There are stacking inputs for: "Right"; "Left"; "Solo Audio"; "Solo Control" (for tying the solo systems of the two mixers together); "Effects"; "Monitor"; and "Aux." The rear panel also contains on/off switches for the mixer's main power as well as the microphone phantom power. There is also an AC line fuse on the rear panel; and for those mixers equipped with the optional reverb system, there is a rather large knob for adjusting the reverb decay time. The reverb, by the way, is a spring-type unit.

Field Test: In order to give the 7416 a thorough field test, we used it on a job where we had contracted to provide a mono mix as the audio portion of a group's video taped performance. We miked the group with a total of fourteen microphones and took direct sends from the bass and electric piano using direct boxes. This gave us a total of sixteen inputs thereby making use of the total input capability of the 7416.

Bass and drums were assigned to sub-group 1, leads

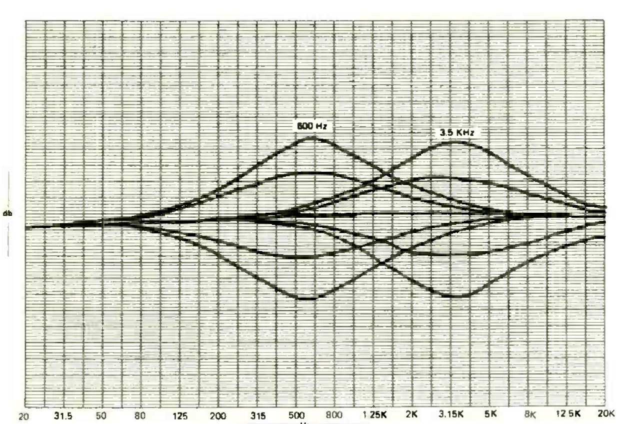

Fig. 2: Tapco 7416: Mid frequency EQ response curves for full boost/cut and half boost/cut.

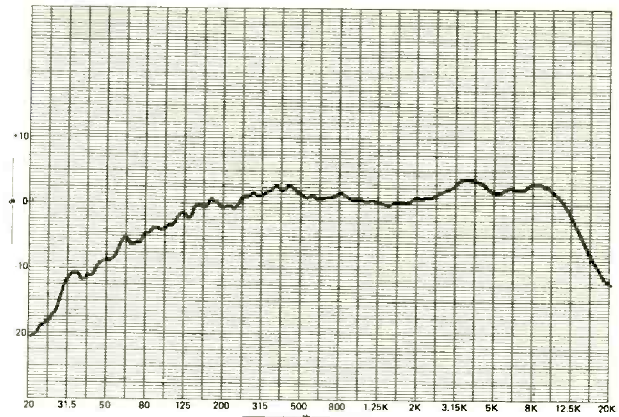

Fig. 3: Tapco 7416: Response of the reverberation section to pink noise.

to group 2, piano and rhythm guitar to group 3 and vocals to group 4. The patch point on channel 1 (bass) was used to process the bass through an outboard compressor. Similarly, the entire group 2 (leads) was processed through a compressor by way of the sub-group patch point; the vocal group was also compressed in this manner. A monitor signal was returned to the musicians in the studio so that they could hear vocals, electric piano and rhythm guitar. Reverberation was used primarily on the vocals but also on the fiddle and some guitar leads. The master mono output was sent to the video control room and also to a power amp driving our monitor loudspeaker. Solos were monitored over a headset driven from the 7416's headphone amp.

We were extremely pleased with not only the performance and operational convenience of the 7416 but also with the reduction in set-up and tear-down time it provided in comparison to the system we normally use. (We typically interconnect two smaller mixers and use an outboard reverb.) All in all, the day's taping session went quite smoothly and our job was easier than ever, thanks to the 7416.

After using the new Tapco mixer, we are highly enthusiastic over the convenience and control flexibility it brings to sound reinforcement mixing. About the only criticism we could come up with concerns the knobs. It was not quite as easy as we would have liked to visually determine the settings of the rotary controls, the EQ in particular, because the knobs are black. Zeroing out the EQ, or even making EQ adjustments in a dimly lighted environment could lead to errors. A white line or dot on the top of these knobs would greatly improve things.

We also put the unit through our usual listening test by patching it into our reference system and listening to a variety of high-quality discs. We made A/B comparisons by alternately switching the mixer in and out of the listening chain and are pleased to report that we observed no sonic problems, just good clean audio.

Lab Test: We performed a thorough lab evaluation of the 7416 and found nothing to detract from our enthusiasm over it. Detailed results of the tests can be found in the "Lab Test Summary" below. Both the mic and line inputs had more than adequate gain and headroom to accommodate any input signal they are likely to encounter. Likewise, the output levels (+4 dBV nominal, +21 dBV maximum) are sufficient to interface with any loads likely to be encountered.

The noise performance was good. Equivalent input noise was about −126 dBV and the output noise varied from about −66 dBV to −56 dBV depending on how many channels are assigned. The worst case noise (−56 dBV) is still 60 dB below the mixer's "0" level (+4 dBV).

Distortion performance was very good at about .006% THD across the audio spectrum. However, we did notice that the THD increases whenever a solo button is depressed or the headphone level control is turned past about a one o'clock setting. With the headphone level full up the THD was measured as about 0.4% across the spectrum. While Tapco doesn't consider the one o'clock setting one that would be reached under typical operating conditions, they are, as of December 1, 1980, improving the performance.

Because of the unit's good high-frequency design (the small signal bandwidth is less than the power bandwidth) we were unable to measure its slew rate limit. We could, however, observe output signal velocities as high as 10.5 volts per microsecond which indicates that the slew rate limit is at least that value. This excellent slewing performance is the result of Tapco's design engineers' choosing one of the new "BIFET"-type of integrated circuit operational amplifiers for use throughout both the Series 74 and the Series 72 mixers. The normalized slew rate limit of at least 0.81 (volts per microsecond per volt) is among the highest we've seen and is well above the minimum of 0.5 recommended[1] for freedom from slewing induced distortion. Indeed, the 7416 is virtually "slew-proof."

We found that the owner's manual supplied with the unit did a good job of explaining the operation of the mixer. An applications section provides hints for making good use of the sub-group facilities. The discussion of the built-in phantom powering system includes lists of those microphones which are and are not compatible with the 24-volt phantom power.

Conclusion: After a thorough evaluation we feel confident in recommending Tapco's new 7416R (and the entire Series 74/72 line) in almost any sound reinforcement or recording application. We are particularly impressed by the unit's extensive control features and its excellent performance on the test bench.

REFERENCE

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

LAB TEST SUMMARY

(Note: 0 dBV = .775 Vrms)

Input Levels

Mic Input:

Minimum input level for "0" level indication with input channel trim and fader at maximum (submasters and stereo masters at nominal settings): −72.0 dBV

Maximum mic input level before clipping: +7.19 dBV

Line Input:

Minimum line input level for "0" level indication with input channel trim and fader at maximum: −51.43 dBV

Maximum line input level before clipping: ±26.37 dBV

Output Levels

(at main stereo output)

For "0" level indication = +4.0 dBV

Maximum output level before clipping: +21.06 dBV

Noise Performance

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

Equivalent Input Noise: −125.9 dBV

With no channels assigned and the sub-masters and stereo masters at the nominal setting, noise at the output is: −71.8 dBV.

With one input channel assigned and set for "−40" input and the channel fader and all output faders at nominal, noise at the output is: −66.5 dBV.

With four channels assigned as above, noise at the output is: −63.4 dBV.

With eight channels assigned as above, noise at the output is: −59.9 dBV.

With sixteen channels assigned as above, noise at the output is: −56.3 dBV.

Distortion Performance

(THD plus noise at +10 dBV output level, solo off, headphone level at minimum)

Frequency	THD & Noise
100 Hz	.0088%
500 Hz	.0078%
2 kHz	.0059%
10 kHz	.0057%
20 kHz	.0059%

Frequency Response: ±1 dB 20-20 kHz

Bandwidth: (−3 dB points): 10 Hz to 51.5 kHz

Power Bandwidth: greater than 51.5 kHz

Slew Rate Limit: at least 10.5 volts per microsecond

Normalized slew rate limit (see text): at least 0.81 volts per microsecond per volt

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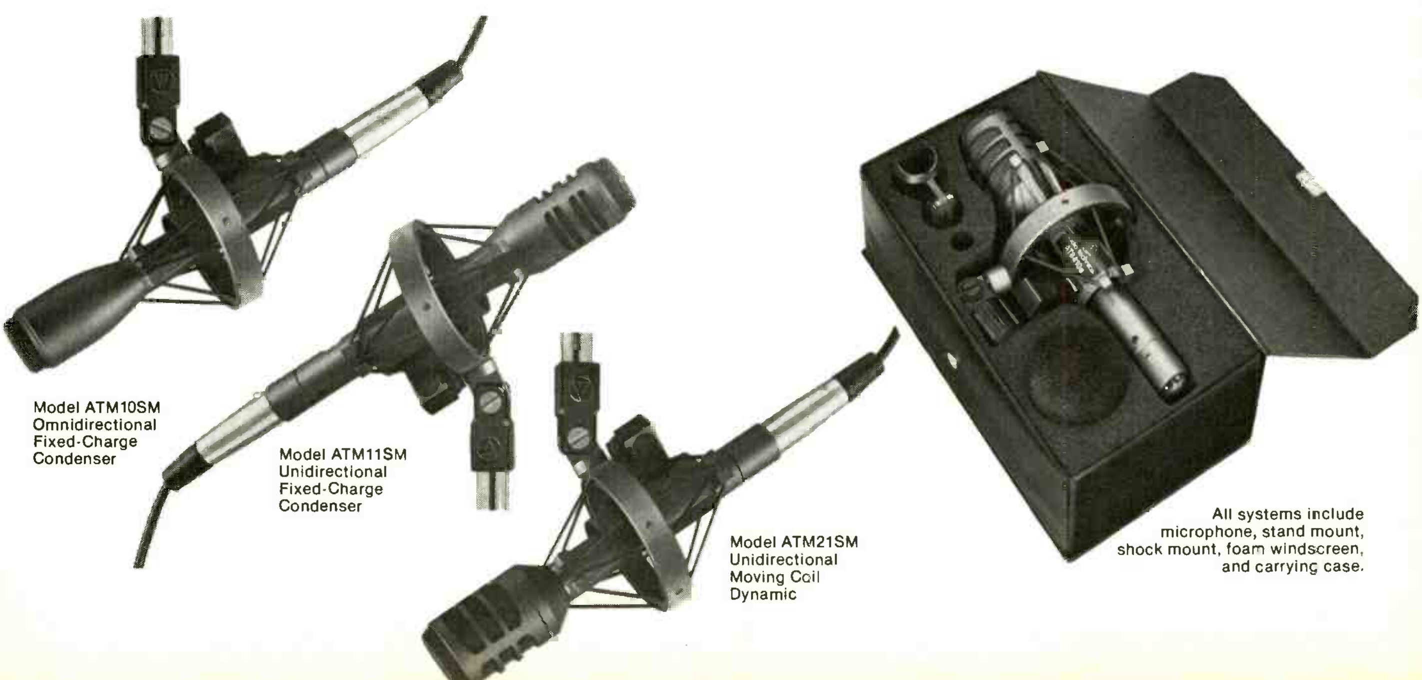

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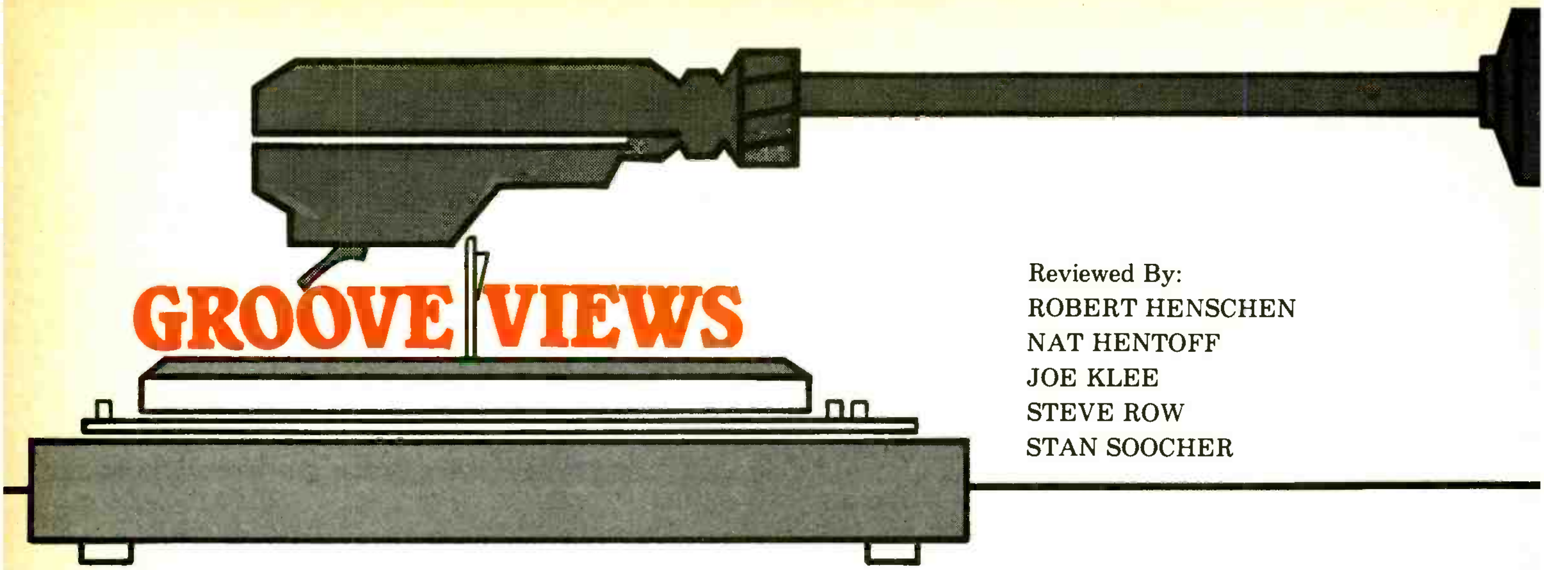

GROOVE VIEWS

Reviewed By:
ROBERT HENSCHEN
NAT HENTOFF
JOE KLEE
STEVE ROW
STAN SOOCHER

POPULAR

PAUL WARREN & EXPLORER: ***One Of The Kids.*** [Peter Coleman, producer; Doug Schwartz, engineer; recorded, mixed and mastered at MCA Whitney Studios, Glendale, Ca.] RSO Records RS-1-3076.

Performance: **Impressive Power-Pop**
Recording: **Captured simply**

It's going to get harder and harder to break from the pack of new New Wave groups. Paul Warren & Explorer may be just another anonymous name, and a band with much the same attitude and energy that supposedly distinguishes a lot of debuting rock bands of late. But do they really have that something extra?

Well, they may have a hit song possibility or two hidden in this package. The opening title track features Paul, now heading into his late 20's, setting himself up as "just one of the kids," defying convention and authority: "But I will stand upon my head/hold my breath till my face is turning red./ That's my answer for all of your intellect." Plausible? Maybe not, but the tune is a catchy "My Generation" type anthem that might rally singles buyers if this selection was released as a 45.

The very next cut, "Takin' Her Back," shows that Warren has listened to the Elvis Costello/Buddy Holly connection, and maybe some Cars along the way. It's a potent pop tune of simple teen values, but the thumping, neo-rockabilly hook is undeniably there. Warren comes close to this kind of appeal on side one's closing "Suzanne" and side two's opener, "You Can't Touch Her."

"The Others," and other cuts, maintain most of the electricity, but begin to sound more typically rock. "Kiss Me Chrissy" even has some chorus changes that remotely resemble the classic "Do You Wanna Dance." Elsewhere, the tunes are not always total grabbers, but the band is instrumentally tight and bursting with basic New Rock energy. Although all four of these guys hail from hard rockin' Detroit, they have been scuffling together on the L.A. scene for over two years now. As a local band without a label, they were impressive enough to land a spot on Midnight Special.

Peter Coleman and RSO picked up on Explorer, capturing their sound simply, without any special sparkle . . . like The Commander does with his troops. *One Of The Kids* attempts the increasingly difficult crossover between New Wave and Hard Rock . . . a plucky attitude copped on one hand, dual lead guitars on the other. Listeners without preconceptions may find themselves enjoying Paul Warren & Explorer; others may want them to play one role or the other.

Audiophiles will appreciate a little trivia about Warren, who has worked as a studio guitarist on over 30 albums, including "Papa Was A Rolling Stone" by The Temptations. He toured with acts like Rare Earth, The Funkadelics, and Pacific Gas & Electric, was a member of Ray Manzarek's Nite City band, and named his group after the 1958 Gibson Explorer guitar that he plays to this day.

Not willing to deny either his past or his future, Paul describes his current band's sound by saying that "it combines the passion of the 50's, the idealism of the 60's and the reality of the 70's to explore and define the sound of the 80's. If they can indeed accomplish all of that, they *deserve* a hit record.

R.H.

PAUL WARREN AND EXPLORER: Using yesterday's tricks to define the '80s.

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CIRCLE 141 ON READER SERVICE CARD

THE ROLLING STONES: ***Emotional Rescue.*** [The Glimmer Twins, producers; Chris Kimsey, associate producer and engineer; recorded at Pathe-Marconi Studio, Paris, France, Compass Point Studio, Nassau, Bahamas, and the Rolling Stones Mobile.] Rolling Stones Records COC16015.

Performance: **Sizzles**
Recording: **Calculated raunch**

"You've got all these idiots who review rock & roll—I can't read them. All these people who try to read so much into the music, read things into it that aren't there. It's totally phony, isn't it, because I know that the things they read into it *aren't* there."

When Mick Jagger recently told this to *Rolling Stone* reporter Chet Flippo, the idea of reviewing a Rolling Stones' record seemed a bit silly. (Don't it make you wanna go home, Mick?) But the Rolling Stones have been recording great rock & roll for nearly two decades now and it would be "idiotic" to ignore the fact that *Emotional Rescue,* their latest release, stands up easily to their best work. Two years ago, *Some Girls* represented an energetic resurgence for the band after a mid-seventies slump, and *Emotional Rescue* stubbornly retains the fresh vital impact of *Some Girls.*

At the heart of the Stones' sound is the commanding presence of Charlie Watts' gunshot snare drum. (It's the first sound we hear on the album.) Of all the drummers of the sixties' British Invasion (blok! blok!), Watts utilized his simple power most effectively to anchor the band (blok!). Throughout much of *Emotional Rescue* Watts' drumming is mixed louder than anything else. Rhythm and lead guitar parts quiver nervously, appear and disappear, but Watts churns on.

The Rolling Stones are masters of the frantic mix. "Summer Romance," for example, sounds quickly thrown together with the lyrics mixed too low and lead guitar notes jumping in where rhythm guitars should be—and vice versa. You believe that rats scurrying in and out of subway tracks could have done better. But, according to Keith Richards, the Stones are now spending long stretches of time perfecting (imperfecting?) the mix, creating a myriad of guitar tones, adding and subtracting reverb from Watts' snare, and preserving the sass in Jagger's vocals.

Perennial Stones' session players like Ian Stewart and Nicky Hopkins on piano, Bobby Keys on saxophone, and Sugar Blue on harmonica handle many of the instrumental passages on *Emotional Rescue*. Yet, Mick Jagger contributes guitar and piano fills, Keith Richards plays piano, and Ron Wood picks steel guitar. Keith's vocal on the bleary-eyed love/hate lament "All About You" provides the album's most touching moment. Someone somewhere may need the emotional help of Mssrs. Jagger, Richards, etc., but if the Stones are in need of any rescue, it certainly isn't musical. S.S.

THE BACKWOODS BAND: *Jes' Fine.* [Backwoods Band, producers; Barney Cole, engineer; recorded and mixed at Calf Audio, Ithaca, N.Y.] Rounder 0128.

Performance: **Jes' fine, but a little too slick sometimes**
Recording: **Nothing spectacular, but it's all there**

I admit I've always been a sucker for string band music from the old-timey bands like Gid Tanner's and Charlie Poole's right through to the Western Swing bands like Bob Wills'. I've heard a lot of attempts to revive string band music and I always felt the New Lost City Ramblers, on Folkways, did the best job—at least until they lost the services of Tom Paley—so I guess that's ancient history too. There's nothing like the sound of an unashamed country fiddler or two with a rock-solid guitar, banjo and mandolin background. Sometimes when these kids fresh out of Juilliard or the Boston Conservatory get hold of this music it comes out too clean. That's really the problem here: Everybody plays and sings correctly but a bit *too* correctly. The major exception is Eric Thompson, whose singing of "Rocks and Gravel" has the proper mountain flavor. Even though I miss the roughness of a Clarence Ashley or a Clayton McMitchen, there's still a lot to be said for a band that plays music like this and plays it well, if one isn't too turned off by the singing of Susie Rothfield who sings like a studio singer—doing the job expertly yet without much commitment or enthusiasm.

The recording is done well enough, perhaps it could have been mixed a little brighter, but then that's more a matter of personal taste.

J.K.

QUEEN: *The Game.* [Produced by Queen; engineered and co-produced by "Mack,"; recorded June & July, 1979, February and May, 1980 at Musicland Studios, Munich.] Elektra 5E-513.

Performance: **Anglican chorale, meet rock and roll**
Recording: **Approaching state of the art?**

I simply cannot conceive of Queen as a concert band, although the attendance figures and mass adulation certainly seems to attest to the fact that it numbers among a handful of universally popular rock groups. The reason I can't imagine the band as a concert band is because it is such an impressive studio band. Too, last year's double "live" album left me with a ringing in my eardrums that overpowered any true music the band was trying to play.

Some critics have lashed out at this group as holding its audience in utter contempt. I am not one of those, because I believe there is some merit in Queen's rock and roll. Self-indulgence and pretentiousness aside, this group can make some fine contemporary music, and Queen recordings generally are top-notch.

The Game shows Queen in fine form. From the appealing title track to the eminently hummable chorus in "Sail Away Sweet Sister" to the searing guitar licks throughout the album, this latest effort by the British quartet is a good example of English art-rock, with the slightest dash of pop and punk thrown in for good measure.

The usual hallmarks of a Queen record all are here—Brian May's guitar work is as angrily insistent, thick and hot as ever; Freddie Mercury's singing combines the qualities of an adolescent English choir boy and a preening dandy; and the rhythmic underpinning by bassist John Deacon and drummer Roger Taylor churns forward in the heavy selections, and offers proper backbeat for the lighter selections.

Something new has been added this time out, however. This album marks the first time that a synthesizer has become a part of the instrumental backing, and though the group doesn't go overboard on its use, the Oberheim OBX does pop up now and again to provide a different musical and sonic texture for the material.

The recording of this album is extremely well done. The opening track, "The Game," begins with white noise and features good blends of vocal and instrumental parts. Taylor's drumming on "Dragon Attack" moves fluidly from one channel to the other, while May's guitar work also switches back and forth between channels. The overdubbed a cappella chorus in this cut, and the use of feedback, also come across well.

Other techniques (some might call them tricks) include the well controlled swell of massed guitars and synthesizer ("Another One Bites the Dust," which has a rhythm line that sounds as if it had been borrowed from the Grateful Dead's "Shakedown Street"), and the evocative rockabilly sound, with echo effect, in "Crazy Little Thing Called Love."

The band manages to cover a number of musical bases with the cuts on the album, too. "Need Your Loving

Tonight" and the opening of "Rock It (Prime Jive)" both recall 1960's rock and roll, and yet "Prime Jive" also has a foot in the new wave or punk style of today. The Beatles come in for a dig in the lazily slow "Yeah, yeah, yeah" lines from "Coming Soon," and there is a rhythm-and-blues feel to the R-rated "Don't Try Suicide."

But it's the treatment of the vocal instrument(s) that comes across the best here. One almost suspects that somewhere, in the back of the group's collective consciousness, resides an English church choir waiting to break out. Or so it seems, when listening to the incredibly tight, soaring harmonies (presumably all by Mercury, overdubbed). The blend of vocals and guitars in Queen is unique, or close to it.

QUEEN: Breaking new ground without abandoning the old.

Lyrically, the material is a mixed bag of Meaningful and Playful and Important and Pretentious. Perhaps form over substance is the best description. But the lads could probably rhyme "moon" with "June," give it their patented approach, and make it almost palatable. This is one of Queen's better releases, one of its most interesting, and one of its best sounding. If no new ground is broken here, at least the ground that has been broken before hasn't dried up yet. S.R.

JAZZ

MINGUS DYNASTY: *Chair In The Sky.* [Ilhan Mimaroglu, producer; Mike Moran, engineer; recorded at RCA Recording Studio A, New York, N.Y., July 9-10, 1979.] Elektra 6E-248.

Performance: **Fitting tribute**
Recording: **Fitting quality**

The late, great Charles Mingus has been eulogized at length since his death nearly two years ago, but the memories are far from tiresome. This particular grouping, which gigs in various incarnations throughout the East, is comprised of musicians who lived and breathed their music fire alongside Mingus in some of the greatest Jazz Workshop bands: John Handy (alto sax); Joe Farrell (tenor sax); Jimmy Owens (trumpet, flugelhorn); Jimmy Knepper (trombone); Don Pullen (piano); Charlie Haden (bass); and Danny Richmond (drums).

An impressive lineup on paper, and on disc as well. Together they tackle three older Mingus standards and three of his more recent collaborations with Joni Mitchell. "Boogie Stop Shuffle," an upbeat oldie from the classic *Mingus Ah Um* LP, kicks the disc off with the hustling feel of rush hour, the combined horns honking out a hectic head arrangement that leads to soloing at length. The title track is a sad-beautiful ballad with more noteworthy solos. Then side one closes out with "My Jelly Roll Soul," an almost comical return to earlier jazz forms, with Owens using a beer bottle for a plunger.

"Sweet Sucker Dance" is a swinging new ballad with particularly excellent work by Farrell and Knepper. Haden's sneaky bass intro to "The Dry Cleaner From Des Moines" soon revs up into a blowing chart where everyone gets a turn to shine. Handy is loose, Pullen is Powellesque, and there are some exciting trades between sax and trumpet. The legendary "Goodbye Porkpie Hat" is the album's fitting finale, probably Mingus' best known composition, and one that takes on additional meaning since his passing. Farrell is again aggressively soulful here, and Pullen's piano playing profound.

Chair In The Sky is a nicely recorded album with some excellent musicianship on solid tunes. Perhaps the physical presence of a man like Mingus would have energized the session even more, and that's one reason to recommend some of the new reissues—like *Mingus At Antibes* (Atlantic) and *Nostalgia In Times Square* (Columbia)—over the Dynasty album. Then again, almost any Mingus offering will do.

R.H.

ERIC GALE: ***The Best Of Eric Gale.*** [Bob James, Ralph MacDonald, producers; Joe Jorgensen, Don Puluse, Richard Alderson, Ed Rakowicz, engineers; recorded at Columbia Recording Studios, Media Sound, Rosebud Recording, A&R Recording, and Sound Mixers, New York, N.Y.] Columbia JC 36363.

Performance: **Glossy but good**
Recording: **Polished**

Columbia is eking more mileage out of their 70's jazz favorites by putting out a new "best of" series featuring Return To Forever, Mahavishnu Orchestra, Tony Williams, Tom Scott, and Stan Getz . . . quality crossover product even if the selections sometimes seem limited in scope. You don't adequately cover Chick Corea or John McLaughlin in six or eight cuts.

One real sleeper in Columbia's catalog, though, has to be Eric Gale. Widely recognized as a session guitarist extraordinaire, and a member of the funky New York studio group Stuff, Gale has a unique background that enables him to write and play noteworthy music amidst so much "formula." His parents were from the Caribbean, Eric was raised in Brooklyn, and he broke in playing support for R&B groups like The Drifters, Marvin Gaye, Aretha Franklin, and dozens more. He's got that funky sound zinging through his strings, but he's heard Bird and Pres and Wes too. The Eric Gale guitar can definitely testify, and does so on almost every passage here.

For the most part, tunes like "Ginseng Woman" and "De Rabbit" are epitomes of the jazz studio "product," all-star casting for catchy—if less than spontaneous—compositions. The feel is hip, but improvisation is secondary and limited largely to the leader. Cuts as spiffy as "Ginseng Woman" and "De Rabbit" may initially come off as homogenized pop, but the grooves developed therein can be as irresistable as the melodies are commercial.

Slightly grittier are "Let-Me-Slip-It-latter stages of a haunting "Red Ground," with Grover Washington on tin whistle. Even less adorned is "Trio," a basic blues that Gale cut with Charles Earland on organ and Idris Muhammed on drums. The album-closing "Oh! Mary Don't You Weep" is pretty soulful too, with a full-blown gospel choir singing the spiritual while Gale sermonizes on electric guitar.

The songs Gale does on his solo efforts are more expansive and colorful than the tight R&B tunes his Stuff is

prone to jam on. If you're turned off by polished, accessible pop-jazz, you may lump Eric Gale in the Bob James School of Music simply by means of past associations. But beneath the cosmetics, this fine guitarist still plays with feeling and writes with an open mind. R.H.

THE JEFF LORBER FUSION: *Wizard Island.* [Jeff Lorber, producer; Rik Pekkonen, engineer; recorded and mixed at Hollywood Sound, February 5-22, 1980.] Arista AL 9516.

Performance: **Soul-jazz crusade**
Recording: **Feels "live"**

Jeff Lorber has a passion for capturing his band's "live" spontaneity in the studio. "Thanks to Rik Pekkonen," Lorber has said, "we wound up with a tremendous sound— full and fat on every track. The record has very few overdubs because we just concentrated on getting the "live," full dimension of the band on record." Pekkonen, known for his work with The Crusaders, was the perfect choice for a group that likes to get right down to it.

It may be a strange time to name an album after a volcanic island in the great Northwest, but that's precisely what Lorber has done. Actually, the jubilant "Wizard Island" was inspired by a summer visit to Oregon's Crater Lake in the pre-Mount St. Helens days. But even then, Lorber's Fusion ideas were anything but dormant and the band enjoyed a growing reputation for an explosive "live" act.

Like the quartet's previous albums, this one is dedicated to the almighty groove. Lorber writes all of the band's music by building arrangements around simple little melodic lines. Fusion's rhythm section serves up a consistently funky beat, and keyboardist Lorber solos at large on everything from piano to synthesizer. Also contributing to the soulful sound is Kenny Gorelick, a hot saxophonist from Seattle. The group as a whole is tight and enthusiastic; the music has an immediate familiarity and broad appeal.

Coming out of Philadelphia, Lorber was a natural for the popular R&B sound his Fusion so ably professes. But he also studied at Berklee and has taught jazz improvisation at the college level. Hence, his almost full-time preoccupation with the party music to be heard here is only a tip of the

TWO HORNS OF IMAGINATIVE PLENTY: STEVE LACY AND ARTHUR BLYTHE

By Nat Hentoff

He began as a Dixieland player, but then leapt far into the post-mainstream by working with Cecil Taylor and Thelonious Monk. Further proof that Steve Lacy is constantly seeking challenges is that his main instrument is the soprano saxophone—a difficult horn to master and to keep stretching. For much of the past two decades, Lacy has been a European expatriate, creating distinctive ensembles that, for the most part, play his own compositions.

A particularly arresting Lacy unit is heard in the two-volume *The Way*, produced by Hat Hut Records, a venturesome label with headquarters in Switzerland and an outpost in the United States. In addition to Lacy, the group includes Steve Potts, a strong and incisive alto and soprano saxophonist; bassist Kent Carter; drummer Oliver Johnson; and a strikingly clear-toned, intriguingly textured vocalist, Irene Aebi, who also plays authoritative cello and violin. The pieces, all Lacy's, are lucidly and freshly structured, spurring the soloists into ways of improvising that avoid familiar patterns (including those of the avant-garde). What is especially absorbing is the continuous interplay of colors with the vocal passages, for example, being an organic part of the rest of the proceedings.

Lacy himself has become the most accomplished soprano saxophonist in jazz, continually finding new dimensions of tonal expressiveness on the instrument as he builds solos that are both unpredictable and insistently coherent. The recorded sound is first-class, with flawless balance and much care for dynamics.

Another horn player sure to gain accelerating attention in the 1980's is alto saxophonist Arthur Blythe. With a powerful rhythmic drive, Blythe is also a soloist of penetrating lyrical sensitivity; and he is in command of the entire scope of jazz, from the traditional lineage to current approaches toward expanding the language. In addition, he has one of the most incisive, memorably reverberating sounds in the music.

For *Illusions*, Blythe uses two combos. One encompasses also, electric guitar (James "Blood" Ulmer), cello, tuba, and drums. The second involves alto, piano, bass, and drums. Blythe makes imaginative use of both contexts, providing bold original themes and then ample room for the mutually attentive improvisations of such resourceful sidemen as pianist John Hicks, cellist Abdul Wadud, and Ulmer.

The set can serve as an illuminating introduction to the newer jazz for those who sometimes find it hard to secure a center of listening gravity in the cries and whispers of other post-modern sessions. Blythe can roam far "outside" the jazz boundaries of the past but he never leaves the listener feeling directionless. Through the clarity of his beat (even when implicit) and the searing logic of his phrasing, Blythe makes vivid wholes of all these multi-layered parts. And there is a broad range of moods, for Blythe, above all, dislikes narrowness—of knowledge or of emotion.

The engineering might possibly be the best Blythe has yet received on record—there is a breadth of aural context that allows the warmth of this music to come through wholly unstrained.

STEVE LACY: *The Way.* [Pia and Werner Uehlinger, producers; Peter Pfister, engineer.] Hat Hut Records, Box 127, West Park, NY 12493.

ARTHUR BLYTHE: *Illusions.* [Arthur Blythe and Jim Fishel, producers; Don Puluse, engineer.] Columbia JC 36583.

creative iceberg. Lorber's tunes are so consistently effervescent that they begin to sound similar at times. The Latin "Rooftops" and a slower "Reflections" serve as excellent contrasts to the rest of the album, and leave us wanting more of this kind of variety from Jeff Lorber. Meanwhile, if it's soul-jazz you want, this band's got it.

R.H.

PATRICK GLEESON: *Rainbow Delta.* [Patrick Gleeson, producer; Steve Mantoani and Stacy Baird, engineers; recorded at Different Fur Recording, San Francisco, Ca.] PVC Records PVC 7914.

Performance: **Electronic art**
Recording: **Hardware mastery**

From the PVC label that brought you Brian Eno's *Music For Airports*, one of synthesized music's striking new experiments, comes *Rainbow Delta* by a man who has done much to advance the direction of contemporary sound. For one thing, Dr. Pat Gleeson runs the highly respected Different Fur facility at Frisco. For another, he has been instrumental in forwarding the careers of Herbie Hancock and other keyboard innovators.

Each side of this album offers a four-part suite, "Rainbow Delta" on the first, then "Draconian Measures." Gleeson writes in his liner notes that "it is somewhat out of the spirit of this music to think of it as having been *produced*." He goes on, however, to explain in some detail about the processes involved in developing the two suites from conception to realization.

"Rainbow Delta" begins stormily with "Frank Stella By Starlight" (Frank Stella is a minimal artist specializing in rainbows) and then locks into a propulsive electronic beat that is lashed by lightening whips and thunder cracks as the fury intensifies into "Unacceptable Dance Styles." The third section, "Take The 5:10 To Dreamland," is a beautiful and timely reprieve, with lush animal/jungle/rain noises of idyllic nativity. "La Grange Point Five," inspired by writer Gerald O'Neill's *High Frontier* and theories of artificial planets, concludes the first suite with an almost Latin syncopation, electronic pacing, and brasslike alarums. It's a fully satisfying piece of music.

"Draconian Measures" is more metallic but almost as varietal. "Arrival Music" contrasts high melodic themes with lower strains of foreboding, tension that drops off into "Ravel Goes To Germany," which begins with a simple radio bleep and builds dramatically. "Hobbits Are Dancing" achieves a carefree flute or recorder sound before finishing powerfully on "Clouds/Blue Skies." Another intriguing collage.

This album is recorded by means of layered motifs "laid down on a 24-track, a single line at a time, using E-Mu Systems and Sequential Circuits analog synthesizers." The sounds and ideas of *Rainbow Delta*, many of them improvised in the studio, are within the realm of prior electronic experimentation. But you don't have to make technological breakthroughs to create good art. With its mastery of existing hardware equalled by an artistic intent to explore, this disc is a noteworthy achievement for Patrick Gleeson.

R.H.

CLASSICAL

JOHANNES BRAHMS: *Four Symphonies; Academic Festival Overture; Tragic Overture.* The Chicago Symphony Orchestra, Sir Georg Solti, cond. [James Malinson, producer; Kenneth Wilkinson, Colin Moorfoot and Michael Mailes, engineers; recorded May, 1978 and January, 1979 at the Medinah Temple, Chicago, Ill.] London CSA 2406.

Performance: **A virtuoso conductor and a virtuoso orchestra play virtuoso music**
Recording: **London excellence, even in the Windy City**

As an old, one-time Chicagoan, I ought to know this orchestra cold by now but there have been so many changes since I left that only a handful of the players I knew remain. Sir Georg got to Chicago two years after I left and inherited an orchestra that, while still a fine instrument, had seen better days under the direction of titans like Fritz Reiner, Frederick Stock and (briefly) Artur Rodzinski. Whether or not the orchestral ensemble had seriously deteriorated during the six years between the death of Dr. Reiner and the coming of Sir Georg, the conducting of such as Jean Martinon was not guaranteed to bring out the best in any orchestra. Both Reiner and Solti were Hungarian-born and both were stern taskmasters. . .and it seems Chicago thrives under stern taskmasters from Hungary. The quality of the playing of the Chicagoans is once again certainly on a par with—if not miles ahead of—that of the other four orchestras that make up the Big Five American symphony orchestras.

Now to get to Brahms, who despite writing *Hungarian Dances*, is not Hungarian. To be sure, Brahms grew up in the shadow of and under the influence of Ludwig Von Beethoven. Yet it is as unfair for liner note writers to repeat Hans Von Bulow's reference to Brahms' First Symphony as Beethoven's 10th—as it was for Von Bulow to make the remark in the first place! He might just as truthfully have referred to Beethoven's First Symphony as Mozart's 42nd. Yet the obvious influences and the respect with which Brahms held Beethoven's Ninth Symphony did have the sobering effect of encouraging Brahms to withhold his first symphony for some time, constantly refining it to the state where it became the sort of mature statement that Beethoven was unable to bring forth until his Symphony No. 3, "Eroica." Consider simply the opus number of the first symphonies of each composer. Beethoven's was Opus 21. . . Brahms' was Opus 68. Yet as clearly as the influence of Beethoven was there so were the changes which Brahms had already begun to make in classical music. It's not really fair to consider Brahms as a classicist. He had one hand in the classics but the other was stretched forward to the era of the romantic. Just as surely as Beethoven handed Brahms the torch, the younger composer carried it on to places it hadn't been before.

All of which brings us squarely to Sir George Solti and the Chicago Symphony Orchestra. The symphonies of Brahms (let's forget the overtures which are simply filler because the symphonies themselves take up less than 4 LPs) respond well to the slow and brooding approach of a Wilhelm Furtwangler or an Otto Klemperer, as well as to the virtuoso technique of an Arturo Toscanini or a Herbert Von Karajan. Solti adheres to neither extreme, in fact, he can give the appearance of inconsistency with his tempi being generally slower paced than Toscanini's, except in the Symphony #3 in F Major where he outdoes even the Maestro in going for the big effect.

There's nothing wrong with this, at least with Brahms whose music responds to either technique, though if one overdoes the slowness one is in danger of losing the line of continuity that connects the music. Likewise if one speeds excessively there is no possibility of detail and even Brahms' autumnal colors can become blurred. Solti is in no danger of either extreme. I get the feeling that he sometimes stretches taste to the outer limits but even so he never goes quite beyond the boundary lines.

Medinah Temple has become the main recording studio for the Chicago Symphony Orchestra ever since the remodeling of Orchestra Hall left it less than ideal acoustically for recording. Orchestra Hall had been the home of some of the most stunning recordings of the early hi-fi era. One needs only to think back and remember the Victor recording of *Also Sprach Zarathustra* with Reiner and the C. S. O. and others from that period. Today the orchestra appears on many different labels under many different conductors and each record company seems to have their own way of dealing with the same situation. Rumor has it that some attempts are being made to record in Orchestra Hall again but, at least for the present, the Medinah Temple seems to be the best choice for symphonic recording in Chicago. The hall is spacious enough to allow the sort of separation which makes it possible to record each section on a different track allowing the producer considerable freedom in setting the sound of the orchestra by bringing either this or that channel up or down on the tape. If you feel, as I do that the strings should be more prominent than they are on these recordings, it's not due to any fault of the orchestra or the engineers—it's simply the way producer James Malinson wanted the orchestra to sound. There's nothing wrong with it. It's just a personalized judgment with which you may agree or disagree. Either way the recording is excellent, as London's recordings have been ever since I can recall and the music is performed as well as or better than any series of Brahms symphonies I can recall hearing on record.

If my figures are correct, Solti and Chicago are now celebrating their first decade together. A long association like that cannot help but strengthen the artistic ties and bonds that make for great music. I can only hope for a second decade and one as productive as the first has been. J.K.

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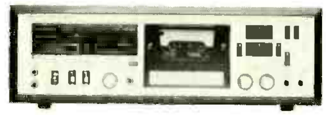

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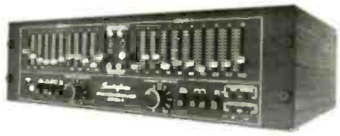

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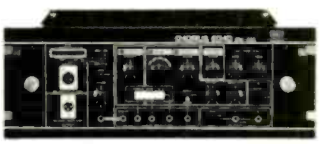

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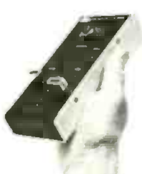

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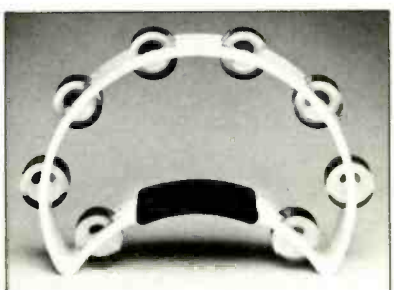

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