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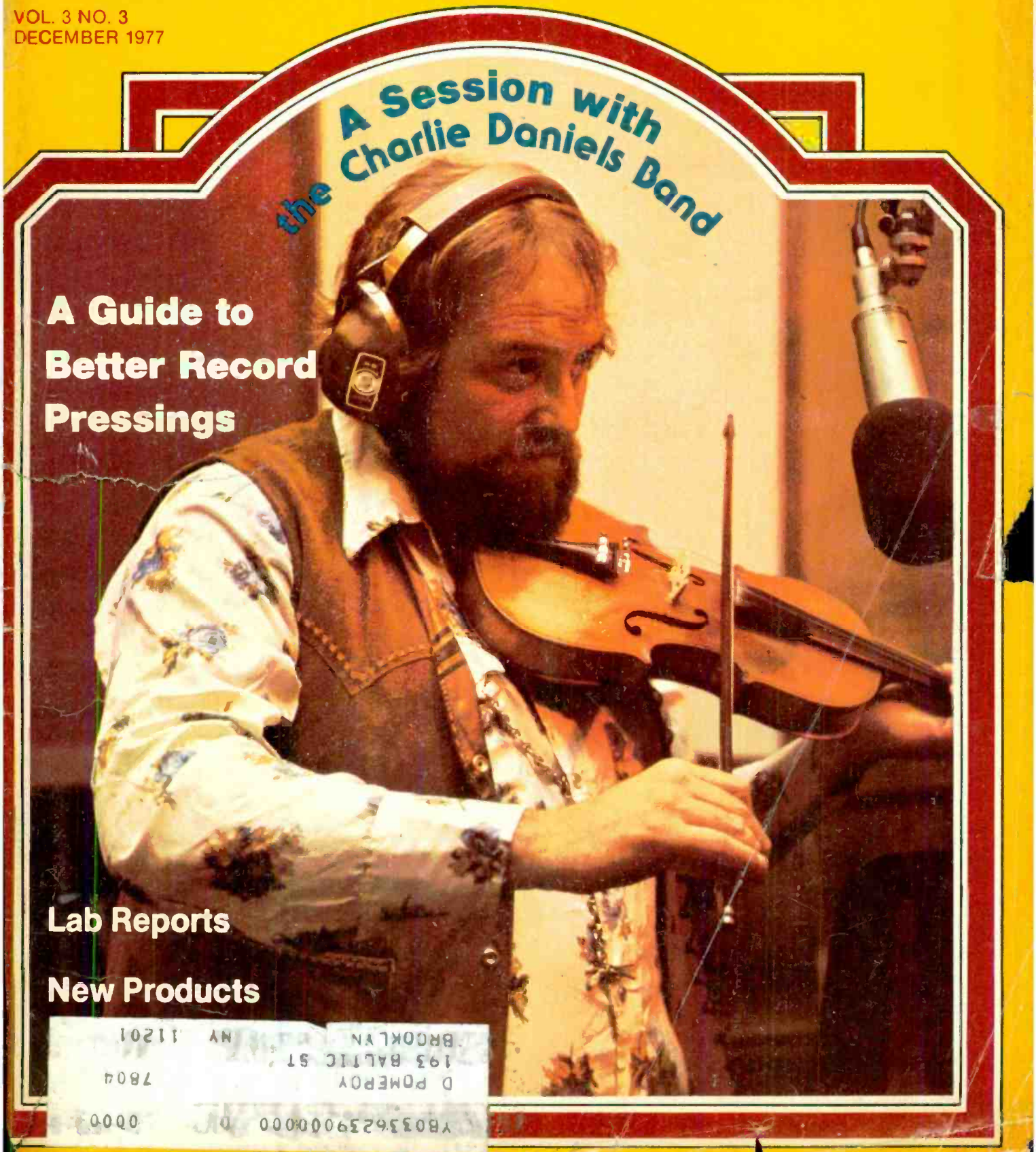
SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

VOL. 3 NO. 3
DECEMBER 1977



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DECEMBER 1977
VOL. 3 NO. 3

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

THE FEATURES

COMING NEXT ISSUE!
A Session with the Jefferson Starship
Profile: Engineer/producer Phil Ramone

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THE STAPLES

Cover photo by David Walton

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Letters to the Editor

An Edifying Article

John Gates' article on providing sound for the Newport Festival was a delight ("And All That Jazz, Sound Reinforcement at Newport, R.I. '77," October 1977, page 52). Having worked with Capron Lighting and Sound at an unmitigated mess known as President Carter's inauguration, I can appreciate the understatement running rampant throughout.

By the way, Mr. Gates mentions only in passing a wonderful device called the Capron MLT. This Mic Line Tester is undoubtedly the best we've seen, and we've given ours constant use in keeping our intercom cables in good repair. It's well laid-out, fits in a pocket, and saves time and temper by diagnosing sick cables without elaborate codes of LEDs or recourse to translations. New products editor, please note!

—Rosemary F. Heath
Senior Partner
Consolidated Edification
New York, N.Y.

Mr. Gates asked that we pass on the information that Capron worked with Consolidated Edification on sound for the official celebration of Jimmy Carter's inauguration—not on the actual inauguration itself.

Some Staple Assistance

We are building a studio in our home and we would like to obtain the reprints of any articles in your early issues on home studios. I know that these issues are out of print (I do, however, have all the issues from Vol. 1 No. 5 on). I would greatly appreciate it if you could help me get this information on home studios and any related topics.

We enjoy your magazine very much and eagerly await your reply.

—Robin Gordon
Outbound Train
Arlington, Ma.

Unfortunately, reprints of early articles are not available at this time. The article that would aid you the most is "Building Your Own Recording Studio (for under $500.)" by Jeff Cooper which appeared in the Dec/Jan 1976 issue (which is out of print). However, that article was an excerpt from his book which should be published by the end of the year, so don't give up hope, help is on the way. In the meantime, you can pick up some pointers on studio construction from the "staples" of M.R., most notably Len Feldman's Ambient Sound column and Talkback.

The Number's No Object

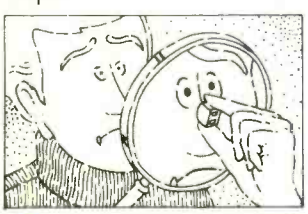
First, I must commend you on the quality of your magazine and service. I was exposed to *MR* two weeks ago and I subscribed immediately, ordering a Buyer's Guide as well. The Buyer's Guide came just *seven* days after I mailed the order!! I was *very* impressed.

ARE YOU BLAMING YOUR TAPE RECORDER FOR PROBLEMS CAUSED BY YOUR TAPES?

Every day people all over the country go into hi fi dealers with complaints about their tape recorders.

When in reality what they should be complaining about is their tapes.

Because the fact is, a lot of the problems that plague tape recorders can be attributed to bad tape.



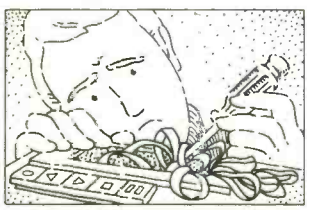
HEAD WEAR IS CAUSED BY YOUR RECORDER. OR IS IT?

If you have to clean your tape heads more than usual, for example it could be your tape doesn't have a special nonabrasive head cleaner.

Maxell has one.

If your recorder jams, it can be any number of things. Maxell does something to prevent all of them.

We make our cassette shells of high impact polystyrene. And then so they won't crack.



JAMMING IS CAUSED BY YOUR RECORDER. OR IS IT?

even after years of use, we finish them to tolerances as much as 60% higher than industry standards.

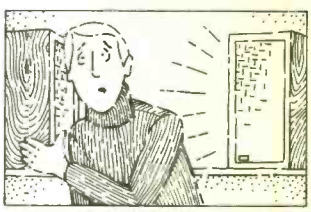
Inside, we use free rolling Delrin rollers so the tape doesn't stick.

And finally, we screw instead of weld everything together because screws make for stronger cassettes.

If your recorder frequently suffers lapses in sound, it could be the tape is of inferior quality. And nobody's bothered testing the tape for dropouts before it leaves the factory.

DROPOUTS ARE CAUSED BY YOUR RECORDER. OR ARE THEY?

Maxell tape is made of only the finest polyesters. And then every



POOR TRACKING IS CAUSED BY YOUR RECORDER. OR IS IT?

step of the way it's checked for even the slightest inconsistencies.

So if you're having problems with your recorder, try a Maxell cassette, 8-track or reel-to-reel tape.

You might find there's really nothing wrong with your tape recorder, just with your tape.

MAXELL. THE TAPE THAT'S TOO GOOD FOR MOST EQUIPMENT.

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CIRCLE 31 ON READER SERVICE CARD

Anyway, I would also like to find out where I can buy metal film resistors (1% and 5%) in small (i.e. five of each) quantities. Also, sliding faders and mic transformers in similar quantities.

Your reply will be greatly appreciated.

—Don R. Chin
Fort Lauderdale, Fl.

Russ Hamm at Gotham Audio Corporation in New York City told us that you can probably get metal film resistors in just about any industrial electronic supply house, but he specifically suggested Hamilton-Avnett in Hollywood, Florida (telephone number 305-925-5401). As for the sliding faders and mic transformers, write directly to Gotham (741 Washington Street, New York City, New York 10014) where you can purchase them through the mail in any number you wish.

Norman, Len and Some Friends

I wish to call your readers' attention to an error made by Mr. Len Feldman in his August 1977 Ambient Sound column.

In discussing the merits of bi- (and tri-) amplification, Mr. Feldman makes a basic error which leads to an incorrect conclusion. The point in question is whether the total equivalent power required by a bi-amplified system is less than or equal to that required in a full-range system.

The parameter of a power amplifier which determines its output power is its maximum output voltage. If this voltage is exceeded, the amplifier clips. (This voltage will vary with different loading conditions, etc., but will be very nearly equal to the square root of 2PZ where P is the rated continuous sinewave power into load impedance Z.

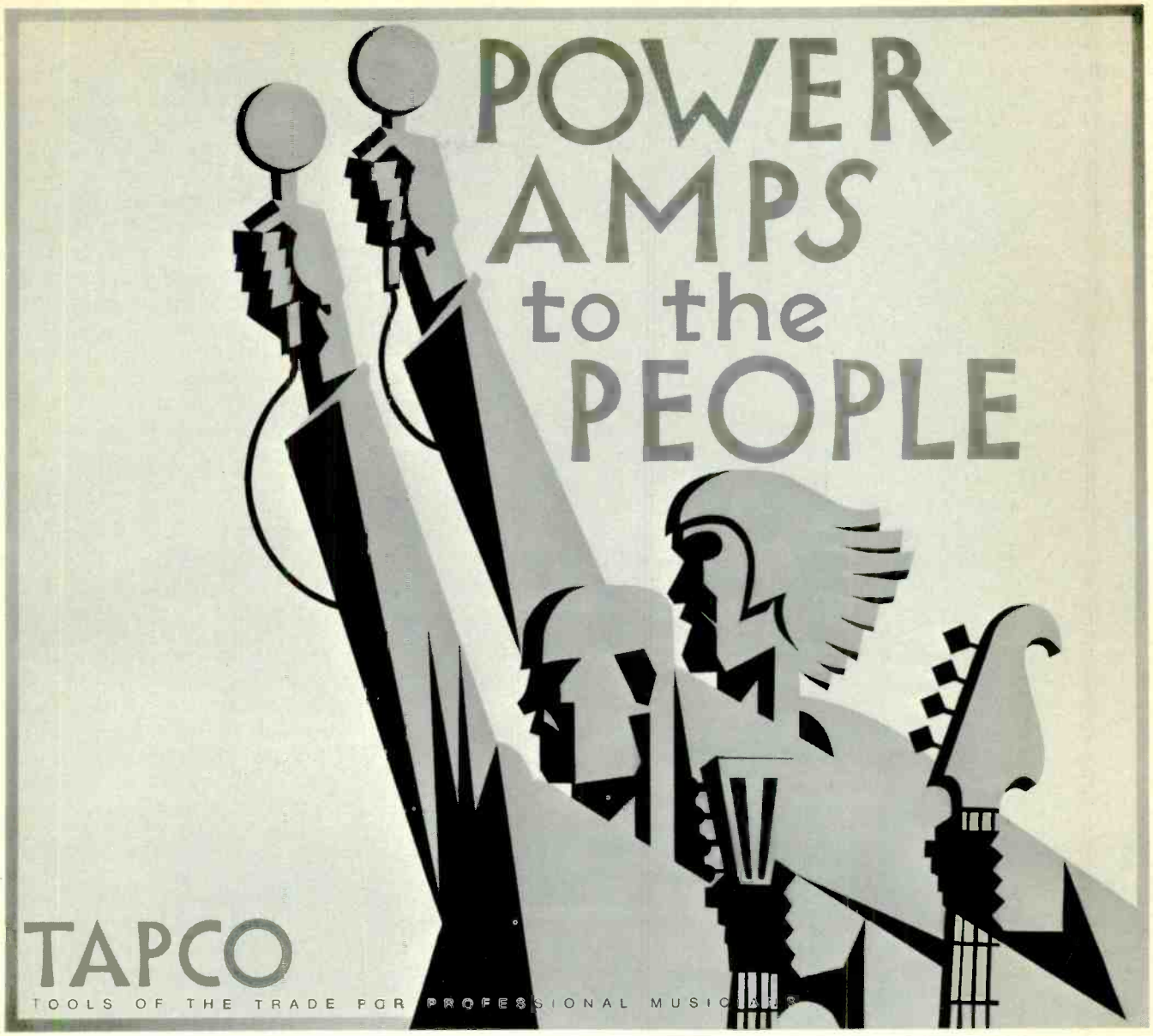
If we suppose, as did Mr. Feldman, that we are reproducing a 100 Hz, 100 watt sinewave and a 3 kHz, 50 watt sinewave, we can see immediately that the total rated amplifier power must be 150 watts RMS (continuous sinewave) in a bi-amped system. If we are using a full-range system and neglect any losses in the crossover network, the peak amplifier voltage required is equal to the sum of the peak voltages required by the two drivers, since, as Mr. Feldman states, "... instantaneously, the two voltages may be additive." The peak voltages are

$$\sqrt{2(100)8} = 40\text{V}$$

and

$$\sqrt{2(50)8} = 28.3\text{ V}$$

The peak amplifier voltage is then 68.3



Power amplifiers have been the stepchildren of the electronic revolution for too long. Many so-called "power amps" are nothing more than redesigned hi-fi amplifiers. Others sacrifice sound quality to attain high volume levels. Still others risk blowing out expensive speakers every time the volume is turned up.

Tapco meets the challenge with two new stereo power amplifiers. Both are designed from the ground up. Both employ Tapco's exclusive Power Sentry® to control clipping distortion and help protect expensive horn drivers. Both are designed to reproduce distortion-free sound at the volume levels demanded by today's professional musicians. And both are engineered and built by the most respected name in sound reproduction.

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CP 120

Rated at 61 watts per channel in stereo, 122 watts mono. Each output is capable of 500 watts total stage dissipation for only 61 watts of rated output power. Sophisticated protection circuitry guards against gross fault conditions without ever affecting sound quality. Power Sentry® protects against clipping distortion, and advanced-design convection cooled heat sinks keep the CP-120 running cool even at full power. $339.00

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CIRCLE 74 ON READER SERVICE CARD

V and the rated amplifier power must be E^2 divided by 2Z, equalling 292 watts RMS.

The actual average power delivered to the load is still only 150 watts, but the amplifier is now delivering a more complex waveform with a higher peak-to-average ratio. Actual musical waveforms behave in a similar manner, although the extent of the effect varies with the nature of the program material. The basic, underlying reason for this is that a full-range music signal has instantaneous peaks which represent a power 10 to 20 times greater than the average power (I'm talking here about short-term, not long-term averages).

When the signal is split into two or more bands, the complexity of the signals is reduced, which reduces the peak-to-average ratio and consequently the required amplifier power. In general terms, the narrower the bands, the lower will be the peak-to-average ratio.

We may see from this that since music very rarely takes the form of a sine-wave, the RMS or continuous sinewave power ratings of amplifiers are only a convenient method of comparison and do not necessarily have anything to do with the actual effective power delivered by an amplifier with a music program.

One of the biggest advantages of multi-amplification, not mentioned by Mr. Feldman, is the protection which it affords the drivers. If we take the previous example, we can easily see that if the amplifier is driven to its maximum continuous output power at one frequency, say 60 Hz, the power delivered to (in this case) the woofer is almost 300 watts. If it is rated at 100 watts, its fate is obvious. Since this type of condition most often is the result of an improper hookup, it is accompanied by clipping, which generates a square-topped output having an average power as high as 500-600 watts!

Bi-amplification reduces the so-called "fault power" to the drivers to a more reasonable level. The drivers' lives will very likely still be in danger under such conditions, but there is a better chance that the operator can react in time to save them.

—William J. Dickerson
Audio Consultant
Midland, Mi.

Len Feldman has just (mistakenly) argued himself out of one of the main benefits of biamping (Ambient Sound, August, 1977, page 44-45)—the savings in total amplifier power required.

In "De-Bunking an Old Tri-Amp Myth," (page 45) he gives the example of a 100-watt bass note plus a 50-watt 3 kHz tone, noting that the total RMS power requirement in a biamped system is 150 watts (*n.b. RMS power*).

But, of course, RMS power doesn't tell the whole story—what garbles up a program is clipped *peaks*, not clipped "RMSs." And for a single amplifier to reproduce both notes unclipped, it would have to be capable of a peak output voltage of 68.28 volts (40 volts for the 100-watt bass peak, and 28.8 volts for the 50-watt midrange peak). (The actual peak instantaneous powers are 200 and 100 watts respectively.) At 8 ohms, this corresponds to an instantaneous power level of 582.84 watts. An amp able to develop such a peak would have an RMS output half as great, or 291.42 watts RMS (since for sine waves, peak power is always twice RMS power).

Thus, the biamped system, with a total RMS requirement of 150 watts, gets by with *over 48% less* power than the single amplifier system at 291+ watts.

Naturally, choosing different relative power levels for the example will change

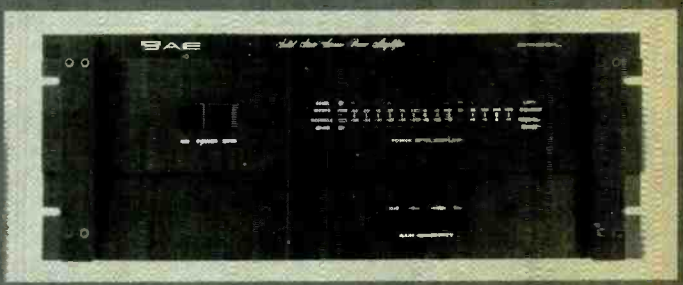
SAE Power

SAE's goal today, just as it has been for over 12 years, is the design and production of fine audio components which offer the best value in both sonic performance and quality construction. Our line of amplifiers stand as a testament to this goal.

First, their design – all SAE amplifiers have fully complimentary circuitry. In this unique design approach, not only the output (as in conventional amplifiers), but the drive and input stages, are completely complimentary. This ensures low transient and steady-state distortion, plus full stability and fast overload recovery. Combine this with our high slew-rate for accurate transient response, feedback gain controls which will not degridate the input signal (2600, 2400L), and monocoque construction with its low weight and high reliability (2200, 2400L).

The result is state-of-the-art performance, but to realize this performance we must have the second part of our goal – production. In order to ensure optimum performance from these unique design concepts, SAE retains total control over the manufacture, selection, and assembly processes. We maintain 40,000 sq. ft. of production area where the latest techniques in metal and circuit board fabrication, component selection and product assembly are employed. The result of these efforts is the line of high quality amplifiers pictured here, each an outstanding value in its power range and each a true SAE component where performance and value come together – that's SAE Power!

2600 - 450 Watts* – our most powerful amplifier, designed for high power home environments. The 2600 ensures clean, dynamic reproduction at the highest power levels.



2400L - 200 Watts* – combining performance and reliability in a surprisingly compact package. This amplifier can reproduce the most demanding program material without strain.

2200 - 100 Watts* – incorporating our advanced circuitry and technology, the 2200 offers high levels of clarity and definition at a popular power level.

*All power ratings are per FTC requirements and are stated with the following parameters: 20Hz to 20kHz, from 250 mW to rated power with less than 0.05% Total Harmonic or Intermodulation distortion.

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the results (changing the high frequency power from 50 to 25 watts results in 44% power saving). The principle doesn't change, however—on a pure watts basis, biamp is better.

—Rob Lewis
West Hollywood, Ca.

I read Len Feldman's Ambient Noise column, "Bi-Amping, Tri-Amping and Such" on pages 44-45 of the August issue of *Modern Recording* and I thought, "hey, that's wonderful. The hi-fi world has finally gotten the message straight on bi-amping." Then I turn the page and read Norm Eisenberg and Len Feldman's Lab Report on equalizers; it seems the "hi-fi boys" need an education on that subject now.

Writing on the Delta-Graph EQ-10, "The results are impressive in that they show the lack of interaction of adjacent controls, a characteristic not always managed by less sophisticated equalizers in which opposite settings of adjacent controls tend to neutralize or cancel each other out. What this indicates, simply, is that the EQ-10 can provide a fine degree of control to permit tailoring overall response to specific needs of program material and/or acoustic environment."

And, writing on the Spectro-Acoustics Model 210, "Finally, to illustrate the precision with which the model 210 can create a given response curve, we photographed (Fig. 4) the actual response curve obtained when the ten controls are adjusted as shown in Fig. 1."

I would like to ask Mr. E and Mr. F one question, "Why do all the expensive and sophisticated equalizers (Altec, UREI, etc.) interact? When they have the answer to this question, they will know a lot more about equalization than they do now.

I don't really mind that they don't know much about it now, but such reviews encourage manufacturers, who often aren't applications oriented, to continue to go through the costly process of designing and building a product that doesn't properly meet the needs in the field.

—Carolyn Davis
Synergetic Audio Concepts
Tustin, Ca.

[The following is Norman Eisenberg's response to Ms. Davis' letter.]

Apparently, pet concepts about graphic equalization are becoming as fashionable as pet notions in other audio areas. But what works for one audio person in a given acoustic situation may not apply to all others, and so there always is the danger of elevating a pragmatic solution to a particular problem to the dignified status of a "scientific truth." Be that as it may, it would seem self-evident that one of the criteria of good circuit design in which several frequency-critical segments are strung together is the capability of that circuit to produce its intended results as a complete device, and also in terms of the scientific action of each of its integral segments. So, the extent to which segment A degrades the action of segment B could be taken as a "limitation" on the whole.

If such a device satisfies some audio need, fine. But other devices—designed to a somewhat alternate philosophy—should not be penalized as a result. As for "sophisticated" equalizers," the UREI is unfamiliar to us, but we have worked with and tested many others, including the Altec which has been around for at least eight years. In MR's *tests of recent equalizers we have shown response characteristics that looked fairly like those reported on the Delta-Graph. If we are to believe the writer of this letter, then in addition to Delta-Graph, such companies as Crown, bi-amp, Soundcraftsmen, Klark-Teknik, et al also do not understand equalization. This seems to us a rather untenable and unprovable position.*

—Norman Eisenberg
Audio Editorial Board
Modern Recording

[The following is Len Feldman's response to letters from Carolyn Davis, William J. Dickerson and Rob Lewis.]

Evidently both Mr. Dickerson and Mr. Lewis have "fallen into the same trap" that I fell into some years ago when I first started analyzing and discussing bi-amping and tri-amping. At that time, the simple addition of sinewave signals seemed to imply that a considerable savings of power requirements was gained by going to multiple amplifiers and electronic crossovers. Coincidentally, it was Mr. Don Davis (the husband of Carolyn Davis, the third correspondent in this group) who straightened me out. It is for this reason that Ms. Davis, in her opening paragraph of her letter, exclaims that we "finally got the message straight on bi-amping."

Since Norman Eisenberg has already voiced my opinion regarding our comments concerning interaction between adjacent filters on home graphic equalizers, I will not dwell on that point

HIGH BIAS.

These cassette deck manufacturers use SA as their reference for the High (CrO_2) bias/EQ setting:

AIWA · AKAI · DOKORDER · JVC
KENWOOD · MERITON · NAKAMICHI
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SHARP · TOSHIBA · TEAC
UHER · YAMAHA

And are joined by these in recommending SA for use in their decks:

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AND MANY OTHERS.

There's been a quiet revolution going on in the cassette world. □ Leading makers of quality cassette decks have adopted TDK SA as their reference standard tape for high (CrO_2) bias and equalization settings. Why TDK SA? Because TDK SA's advanced tape formulation and super precision cassette mechanism let them (and you) take full advantage of today's advanced cassette deck technology. □ In addition, a growing number of other companies are recommending SA for use with their machines. □ So for the ultimate in cassette sound and performance, load your deck with SA and switch to the "High" or "CrO_2" bias/EQ settings. You'll consistently get less noise, highest saturation and output levels, lowest distortion and the widest dynamic range to let you get the best performance from any quality machine. □ But you needn't believe all this just because we say so. All you have to do is check our references.

TDK

The machine for your machine.

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other than to say that the best of all possible graphic equalizer worlds would be one in which the user is able to vary both the bandwidth of each filter as well as its amplitude. Recently, we have seen an increasing number of such equalizers which some manufacturers are calling "parametric." My own experimentation with equalizers suggests that there are times when narrow band boost or attenuation is demanded while at other times, a broad, gradual action is what's called for. If I can't have both, I'd still rather have an equalizer where, when I push one lever up three dB and the next one up by some other amount, I don't have to allow for the interaction between the adjacent bands and can be guided by the front panel nomenclature.

Now, returning to the matter of bi-amp and tri-amp power requirements: Mr. Dickerson hit upon the essence of the matter when he said that most audio amplifiers are "voltage limited" at clipping. In other words, the voltage swing that a signal can undergo when being amplified by a conventional audio amplifier is determined by the power supply voltages. Mr. Dickerson goes on to agree with me that there is no inherent power savings in going from a full-band amplifier to two or three separate amps in a bi-amp or tri-amp arrangement. So, Ms. Davis and Mr. Dickerson are not disputing the only contention I made in the article, specifically, that bi-amping or tri-amping reduces the power *requirements in reproducing a given complex signal. That leaves Mr. Lewis all alone to argue that power savings do accrue— a point which he unfortunately is mixing up with voltage peaks—as I once regrettably did before the Davises straightened me out.*

—Leonard Feldman
Audio Editorial Board
Modern Recording

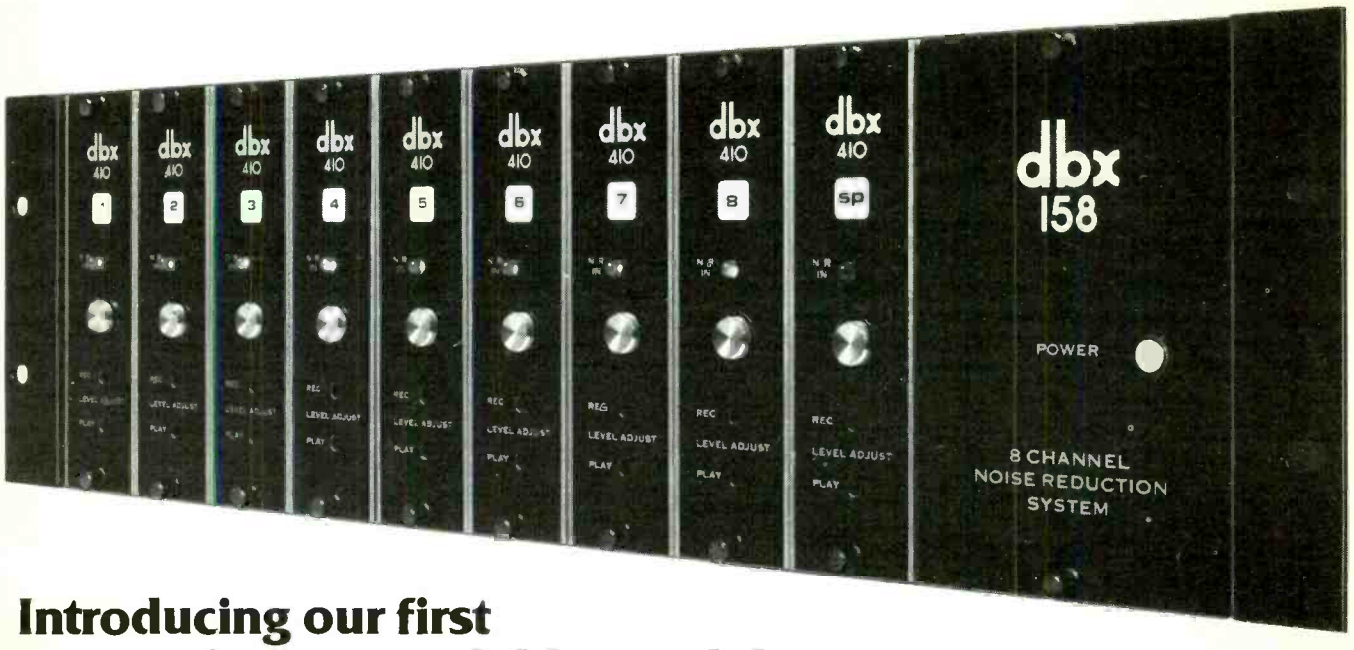
Delta-Graph Detractors?

As the designer of both the Delta-Graph EQ-10 and the Spectro Acoustics Model 210 Graphic Equalizers (and the author of both those products Owner's Manuals) I read the August, 1977, Lab Report (pg. 46) with great interest.

Unfortunately, I feel very strongly that the two journalists responsible for reporting their opinions of the products (and I am referring specifically now to their comments on the Delta-Graph Instruction Manual) must not have a very accurate "feel" for the interests of your readership.

It is amazing to me that after having

dbx 158.
IT'LL GROW ALONG WITH YOU.



Introducing our first economical, expandable, modular, simultaneous tape noise reduction system.

Now you can have a tape noise reduction system that will stay with you from high-end audiophile, through semi-pro and into full professional equipment.

Our new dbx 158 system can start life in your place with the 158 main frame and as few as two modules or as many as eight modules for its full eight channel capacity. It also has storage space for a ninth spare module in its compact chassis. The rear panel has phono and multi-pin connectors that will interface directly to your cables. Additional 158's can be used for 16 or 24 track recording.

The dbx 158 offers the semi-pro recordist or small studio all the advantages of dbx professional systems, including 30 dB of noise reduction, and 10 dB additional recorder headroom. It's a classic 2:1 mirror image compander which preserves the full dynamic range of program material without audible tape hiss. Each module contains separate record and playback noise reduction electronics. Its simultaneous record/playback capability permits the noise reduced, decoded tape to be monitored while recording without manual switching or remote control.

Requiring only 5¼" of rack space, the 158's light weight (17 lbs.) makes it easily portable for location dates. And naturally, tapes recorded with this system are compatible with any other dbx professional tape noise reduction system as well as on board dbx tape noise reduction in TEAC/TASCAM recorders. We'll be happy to send you further information and the name of your nearest dbx dealer. Just write us.

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Here's a generous offer: buy all 8 channels up front, and we'll throw in the ninth module free.

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sold over 2,000 Model EQ-10 kits in the past 11 months I have not had one negative comment from any paying customer about the ease of assembly or clarity and straightforwardness of the Instruction Booklet if what Mr. Feldman and Mr. Eisenberg say about the booklet is true.

On the contrary, we have received literally dozens of unsolicited letters from customers who appreciated our kit's ease of assembly and the very complete, simple and thorough instructions with over 75 illustrations. I have taken the liberty of reproducing excerpts from typical customer responses. (Xerox copies of the complete letters are enclosed for your verification.)

If you want more of the same, I have plenty! I really cannot understand why these two reporters found such greivous fault with a booklet that up until reading the Lab Report had received nothing but the highest praise. And this from paying customers!

Maybe the fact that these two gentlemen did not build the kits (even though I supplied kit-form units to them along with wired kits) had something to do with it. Or maybe they are (as journalists inherently are) more concerned with the type style and size and the English usage aspects of the manual.

Had any customer *ever* complained about the instruction booklet, I most certainly would investigate and make changes or additions with diligence. I consider it quite unfair to leave the impression that was left by these two writers when the evidence of the facts indicates precisely the opposite feeling!

I would hope that, in the future, such unfounded negativity should be kept in check, or at least that a qualified individual whose concerns ran reasonably parallel with those of your readers be consulted for an opinion more closely attuned to the potential buyer's needs!

—Bryan T. Morrison
Owner
Delta-Graph Electronics Co.
Seattle, Wa.

It is somewhat surprising that, in the context of some very favorable, thorough and accurate product test-reporting, Mr. Morrison has seized on the one critical note in the report on the Delta-Graph EQ-10 (see Lab Report, August, 1977, page 46)—especially in view of the fact that our criticism was offered in a reasonable and constructive manner—to react to as he did in his letter.

It also is interesting that his criticism of our criticism does not really challenge the validity of it in its own terms, but rather takes exception to the fact that we included it in the report—and this is based on his assertion that over 2,000 paying customers have bought the kit and have not complained about the instruction booklet.

Well, it just may be that there are at least 2,000 hardy experienced souls who can cope with that instruction booklet. In my comment (page 48), I said that, "The instructions and general presentation of this product indicate it is aimed primarily at the seasoned technician and not intended for the first-time or novice kit builder." Nothing in Mr. Morrison's letter contradicts this viewpoint. All his letter says, really, is that he has actually found a market of 2,000 buyers who are able to hack it with the help of that instruction manual. Okay, but what about the next 2,000? And the next? What about the vastly greater number (than 2,000) of our readers? It would be gross irresponsibility to our readers if we had failed to mention this point about the manual. And, in any case, we did not say any-

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thing like, "do not buy this;" rather we advised that the potential buyer look through the manual beforehand to see what he or she may be getting involved with. Is that unreasonable? Is that negative? Is that poor journalism?

In Len's individual comments, too, as published, the tone is one of constructive (rather than negative) criticism in that he (as I did) suggested a broader market for this product. In other words, our joint feeling was something that can be summed up as, "Do this thing a little better and you probably will sell lots more units."

Our motivation in this attitude was based on the fact that the EQ-10 is a good product. In our "Test Results" section of the report we used phrases like "excellent performance" and "the EQ-10 met or exceeded its published specifications." It was this recognition, on our part, of the device's capability and merit that prompted us to make the point about its difficult instruction manual.

And make no mistake about it—despite Mr. Morrison's 2,000 happy customers—that manual is difficult. It could easily turn off 20,000 other potential buyers.

—Norman Eisenberg
Leonard Feldman
Audio Editorial Board
Modern Recording

A Correction

In the Hands-On Report of the August, 1977, issue (page 54), an error apparently slipped our attention. In the discussion of the power supply of the Yamaha P-2200 power amplifier, the description of the filter capacitors was stated as being ". . . large 2,200 mfd electrolytic capacitors. . ." That figure should have been 22,000 mfd. —Ed.

News From The North

Concerning your article, "Building Your Own Recording Studio," by Jeff Cooper (Dec/Jan 1976): It said that the book was to be published by Tab books here in Toronto, Canada, yet the folks at Tab say they have never heard of it. Is the book available in Canada and, if so, where can I get a copy? I'm building my own 4-track basement studio and this book would be a great help to me.

I also see your magazine is distributed here in Toronto and has been selling like hotcakes. Also, how about an article on Toronto Sound Studios? There are some

Continued on Page 95

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TALK BACK

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

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

Folded Horn "How To"

Where can I get blueprints for a folded horn bass cabinet to be used in a PA system, as I saw in the August 1977 Talkback column? (See "A Clarification of Terms," page 10.) I have written to JBL Sound for this information and they were unable to help—perhaps you can.

—Lloyd V. Leeman, Jr.
Brewer, Me.

Due to our limited amount of space, it was not possible for us to publish the specs that you require here, but we were able to find several easily available sources for you. It had been our understanding that JBL offered a kit containing the blueprints that you need, but a call placed to Howard Colby of their engineering department informed us that this is no longer available, only factory built enclosures are. He did recommend that you refer to the Audio Cyclopedia *by Howard M. Tremaine, Howard W. Sams and Co., Inc., Indianapolis, Indiana for the information that you need. Christine Kofoed of Community Light and Sound in Philadelphia contributed the names of two volumes where you will find actual blueprints and specifications for folded horn enclosures. They are* Hi-Fi Loudspekaers and Enclosures *by Abraham B. Cohen, Hayden Book Co., New York, New York and* How To Build Speaker Enclosures *by Alexis Badmaieff and Don Davis, Howard W. Sams and Co., Inc.*

Coloring Sound—Why and How

Why does everybody put so much boost on their boards, amps, etc.? Any more than 5-8 dB really colors the sound. The correct method is to use less cut and more gain when using EQ. Why not 6 dB boost, 20 dB cut, for instance? If you build a board with a little boost and lots of cut and gain you can concern the design of the circuits for quiet gain instead of just being able to handle peaks.

Also, cut-only circuits are cheap, easy to make and can be simple torroid coil/resistor/cap/pot combinations that hook onto a normal Boxendahl boost/cut bass pot.

—Anthony Gennary
Brooklyn, N.Y.

Ever since Mitch Miller (many years ago) started using large amounts of reverb to achieve his "sound," processing equipment such as equalizers, filters, flangers, reverb, etc., have taken an increasingly important place in creating "sound." There is no relation between "live," unamplified sound and what is bought recorded. The trick in re-recording is to color the sound in such a way that it appeals to the buyer. Since this is very subjective, there are wide variations from artist to artist and producer to producer.

Technically it is very simple to build gain, boost and cut circuits, especially with operational amplifiers. It is easier to build a peaking cirucit to boost a narrow band than to build a circuit that will droop the balance.

The final deciding factor is the console manufacturer's customer. If he wants a console with 15 dB boost or more, that is what he gets. How he arrives at his specification is anybody's guess. But no console manufacturer will buck a sales trend, since there is always another who will follow it.

—Fred C. Roberts
Chief Engineer
Dallas Musical Instruments/USA/Ltd.
Mahwah, N.J.

Expanding Fidelity

I have some cassettes that were recorded on a $75.00 Panasonic recorder with hand-held mics. Is it possible through re-recording on a Nakamichi machine to expand the fidelity of the cassettes? Would a graphic or parametric equalizer (SAE 800) do the trick? If so, how would I go about it?

—Paul Beckwith
Las Vegas, Nv.

To begin this short discussion, let me say that it is difficult, at best, to make a bad recording good, but a reasonably good recording can be enhanced through processing and re-recording the signal. You mentioned "expanding the fidelity of the cassettes." I take this to mean trying to correct for poor frequency response in the microphones or the tape machine. While this is possible, you should be aware that while you're trying to recover those lost highs for instance, you will also be dredging up other demons—for instance, tape noise. Similar problems occur when trying to emphasize lows as any wind noise or mic handling noise or any other low frequency disturbances will also be emphasized. As far as what type of equalizer to use, this is somewhat dependant on what you are trying to fix. For you, a good graphic equalizer (preferably with ½ octave centers) and a good ear will probably do the trick.

It might be appropriate to say here that there are many other ways to doctor-up a recording. Hum and buzzes can be reduced (though rarely eliminat-

ed) through notch filtering (i.e. URIE's Little Dipper), reverb can be added to an otherwise dry recording (especially true for recordings made out-of-doors), and there are a few fairly good noise filters to reduce tape noise, crosstalk, etc. (i.e. Phase Linear's Auto Correlator or Burwen's Dynamic Noise Filter.) There are other such blackboxes, but these are possibly the most common. However, there are some things which are irrepairable such as distortion and speed variations (wow and flutter). As always, your best bet is to get it right the first time.

—Mike Farrow
Engineer
Tapemasters
Indianapolis, In.

Dimming The Buzz

Recently, I installed light dimmers in my studio. When I'm miking high-level instruments like electric guitars and pianos, I get a buzz from the instrument (through the amplifier itself). Turning the dimmer off completely eliminates the buzz—even though the lights and electric outlets are on separate circuits.

Friends have suggested that I look for higher quality dimmers because the ones I now have only cost about $8.00 each. Is this a valid assumption or will all incandescent dimmers cause this problem?

—Ward Archer
Memphis, Tn.

You'll get this problem with almost any dimmer short of the zero-voltage switching, super-high filtered professional dimmer pack used in small theaters. In dimmers at a price we can afford for normal room lighting, manufacturers' ideas differ radically from those of audio people as to what constitutes adequate filtering in the dimmers.

A 120-volt power line is 82 dB hotter than a 10-millivolt (nominal) mic line—*102* dB hotter than a 1-millivolt signal. From a practical viewpoint, look at it this way: A 120-volt AC line is +44 dBm. A triac-type dimmer operating at 50% brightness switches on midway into each half-cycle, where the instantaneous voltage is about 170 volts. This sharp peak, rich in harmonics, produces the radio-frequency transients that appear as a buzz, is then about +48 dBm. Suppose the manufacturer considers 100:1 (40 dB) as good enough filtering. This brings the peaks then down to +8 dBm. Now suppose that only one ten-thousandth of this signal gets into the unbal-

anced, high-impedance lines. I would suspect that even though the *sound* of the guitar amps and piano are quite high, the actual signal levels of the pickups are not. A high-impedance line, and pickup (especially the kind without hum-bucking coils, and marginally shielded) and unbalanced to boot, is most susceptible to this type of interference. Since 1/10,000 is -80 dB, the buzz signal in the guitar cord is -72 dBm, and the guitar amp increases the signal at least 50 dB, you have not at least -22 dBm coming out of the guitar amp speaker—very irritating unless you are recording a Suite for Bagpipes in B-Natural. (Note that these calculations are based on voltage rather than power level and therefore are independent of impedance variations.) To make matters worse, some incandescent bulbs make a noise themselves when used with a triac-type dimmer.

Now the remedy. First, check guitar cords by substitution. A cord that checks good on a cable checker may still have enough resistance to cause a problem. Also check grounds on amplifiers, lifting them if necessary, reversing the plug, and grounding the guitar body (pickups) separately. I'm sure you have found that the way a guitar is positioned in the studio will affect hum pickup, too. But RFI (radio-frequency interference) can be a combination of power line problems directly and radiation through the air from the power lines just like a legitimate radio signal from an antenna. So, separating circuits for lights and outlets, installing RFI filters in power lines and amplifiers, and even encasing the whole thing in conduit may or may not work. Before you decide to lay down a couple of grand on a theater dimmer pack, try autotransformers, and throw out the triac-type dimmers entirely. In fact, don't let a triac dimmer anywhere near your studio. In our situation, we had to remove all of ours from the building, even with massive RFI filters on all the studio AC lines, which were in conduit, conduit on all the other AC lines, including right up to the light fixture, all audio lines run at least 12 feet away from any AC lines, and the audio lines to the studio were low impedance (200 ohms), balanced, and transformer-coupled to the mixing board. The only point of common ground was at the board, also.

For about $40, you can get a totally quiet dimmer rated at a kilowatt which will give you superb control from a soft glow to full brightness, with no flicker, no sudden turn-on, and above all,

Their weight makes them portable. Their performance makes them professional.

Introducing Technics new professional portable cassette decks. Our top-of-the-line RS-686DS speaks for itself. Its 6 lbs., 3 oz. say it's portable. Its 3 heads say it's professional. And all the other features say it will give you recordings of professional caliber.

Features like a unique anti-rolling mechanism for unprecedented portable transport stability. A frequency generator servo motor that immediately counteracts any variation in rotational speed. Separate bias and equalization. Even Dolby.*

The RS-686DS also gives you controls you won't find on many non-portables. Like a tape/source monitor switch. Low cut filter. Mike attenuator. And a three-minute tape end alert eye.

A less expensive alternative is the RS-646DS. The portable deck with performance specifications usually found only in higher priced cassette decks.

The RS-686DS and RS-646DS. Professional specifications. Plus the flexibility of recording sound wherever it may take you.

TRACK SYSTEM: 4-track 2-channel record/playback. MOTOR: FG servo-controlled DC motor (RS-686DS). DC electronic speed control motor (RS-646DS). FREQ. RESP. (CrO_2, ± 3 dB): 50-16,000 Hz (686). 50-14,000 Hz (646). WOW AND FLUTTER (WRMS): 0.07% (686). 0.1% (646). S/N RATIO (Dolby): 66 dB (686). 65 dB (646). DIMENSIONS: 3"H x 9½"W x 7⅞"D (686). 4¼"H x 14½"W x 11"D (646). SUGGESTED RETAIL PRICE: $599.95† (686). $299.95† (646).

Technics RS-686DS and Technics RS-646DS. A rare combination of audio technology. A new standard of audio excellence.

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no noise. The only disadvantage is you will need some type of enclosure since an autotransformer is much larger than a switch box. Autotransformers come in various sizes and types. Some well-known manufacturers of reliable units are General Radio (Variac), Ohmite (Ohmitran), Superior Electric (Powerstat), Staco, Inc. (Adjust-A-Volt), and General Electric (Volt-Pac).

Speaking from experience, I know that this route will solve your problem.

—Wayne A. Pommer
Chief Engineer
Holden, Hamilton & Roberts Studios
Seattle, Wa.

Proper Use of Mic Pads

All the descriptions that I've read and heard of microphone input pads say that such pads can be used to cut down or attenuate an incoming mic signal which is overloading the preamp of a given mic input channel. What confuses me is that when such a pad is switched into the circuit the operator must turn up the channel volume control to compensate for the volume loss from leaving the signal cut down by the pad. Now, once volume compensation is made, is not the signal just as strong? And, therefore, just as likely to overload the preamp?

I should point out that I have found a practical use for input pads. I occasionally have clipped a mic to the bell of a saxophone (to allow the musician total freedom of movement without the signal variating). When doing so, with some mixers I only have to turn the channel volume pot up a hair before its too loud. By inserting a pad, I can move the slider up much further and not have this problem.

—Paul Lazzaro
Waterbury, Ct.

Questions such as this can often be answered by a quick look at a block diagram of the system in question. Block diagrams are an extremely functional shorthand for showing signal flow and system operation; and a few moments spent studying any you may chance upon can provide a large amount of useful information.

Figure 1 shows a simple microphone input section. Notice that the gain control *follows* the first stage of amplification. As such, it can only prevent overload of following stages, *not* the first stage—and this is where the problem occurs.

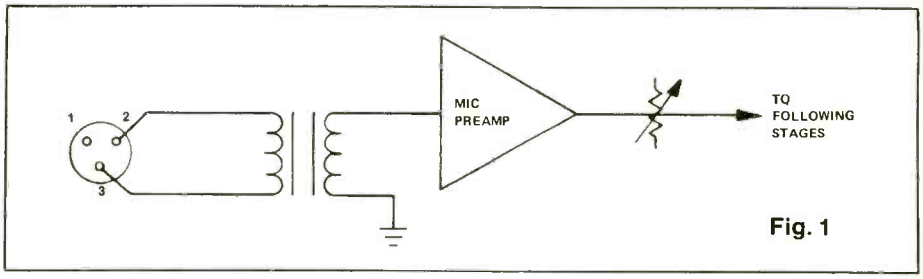


Fig. 1

Figure 2 shows an attenuator switch placed *ahead* of the first stage. This attenuator may be in the form of a simple resistive "pad" or a feedback loop around the amplification stage. In either case, it is able to reduce the signal so that it does not overload the *first* stage.

Another potential overload point is the input transformer. Good, high-level input transformers are fairly large, heavy, and not at all inexpensive. As a result, many economical mixers simply can't have such transformers.

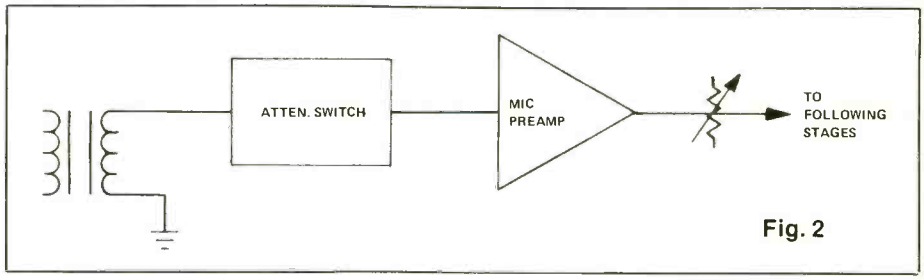


Fig. 2

Transformer overload can be avoided by placing the attenuator switch ahead of the transformer, or by using an external attenuator. The latter method should be used with caution if phantom powering is used, as certain in-line attenuator designs may short-out or imbalance the power feed. In this case,

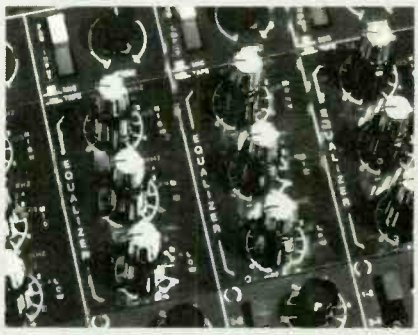
One year later.

Last year, Sound Workshop introduced the 1280; an 8-track recording console of compact functional design that would sound *significantly* better than the rest. Now, one year and thousands of sessions later, the 1280 has become the most respected board in the semi-pro field. Very simply...It sounds better.

The equalizer section of the 1280 provides ±15dB of shelving at 100Hz and 12kHz, and 15dB of peak and dip at 3.7kHz. **Super EQ** was developed for applications where 3 frequencies do not offer enough selectivity.

Super EQ provides 3 bands of equalization with a choice of 5 frequencies per band.

Super EQ



Low	Mid	High
20Hz	250Hz	4.8kHz
40Hz	500Hz	7.5kHz
90Hz	1.2kHz	9kHz
200Hz	2.4kHz	12kHz
400Hz	4.8kHz	12kHz (shelving)

The 1280B-8EQ shown with optional Meter Bridge, which works in conjunction with standard **Tri-Lite LEDs.**

1280B (balanced mic inputs)	$3200.
1280B-8EQ (8 inputs super eq, 4 standard)	4000.
1280B-12EQ (12 inputs with super eq)	4400.
1280B-EXP (12 input expander)	2500.
1280B-EXP-12EQ (expander with super eq)	3700.
1280A (unbalanced mic inputs)	2850.
1280A-EXP (12 input expander)	2150.
1280 METER BRIDGE (for 1280A and 1280B)	850.

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it is desirable to have a mic with an internal attenuator.

Any situation in which a specific input gain control can be opened up only a "hair" before the signal drives the internal meter (or other level indicator) into the "red" may indicate that *input* attenuation is needed; especially if the master gain control is also opened up only a "hair."

Unnecessary input attenuation will result in poor signal-to-noise ratio from the first stage. This is because the gain control will be turned "way up" and, as a result, will allow more noise from the first stage to be passed on to following stages.

—Geoffrey M. Langdon
Technical Manager
AKG Acoustics
Div. Philips Audio Video Systems Corp.
Mahwah, N.J.

A Reading List for Recordists

I have been reading your magazine for about two months and have found it to be by far the best of its type. I especially enjoy the Talkback and Ambient Sound columns.

I need to know if there are publications other than *Modern Recording Techniques* by Robert Runstein dealing with professional sound reinforcement. I am especially interested in learning about the different types of amps, mics, speakers (and drivers), speaker enclosures, mixers, compressors, limiters, equalizers and cable. I am also interested in learning about various types of system setups and line voltages, impedances and crossovers.

Any information you could supply would be most helpful.

—John Kauble
Gahanna, Oh.

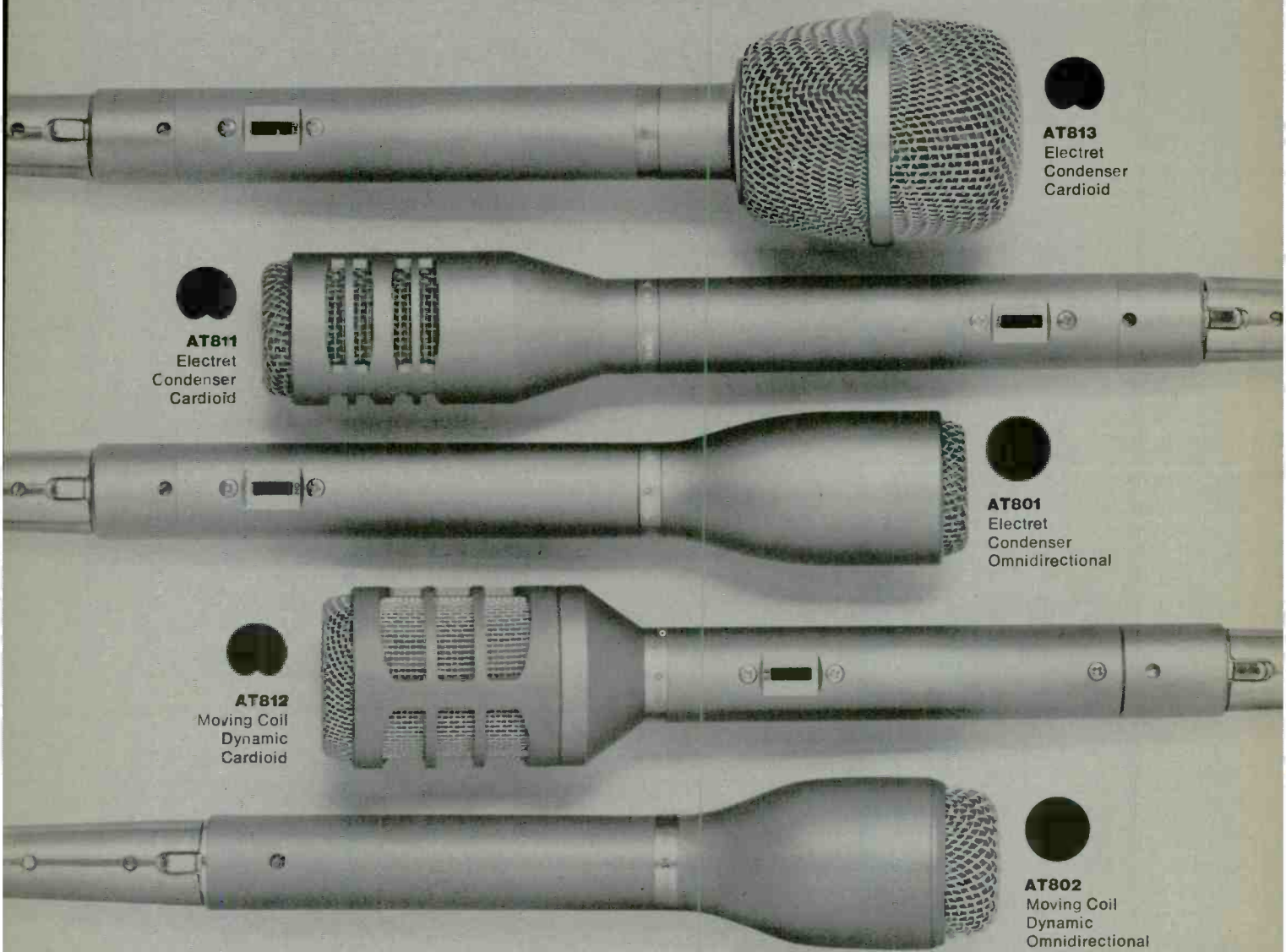
For a list of magazines and periodicals dealing with the items you mention, you might look up the Talkback question entitled "Your Own Studio," April 1977, page 13. We'd like to add to that list, however, the following books: The Audio Cyclopedia *by Howard M. Tremaine, Howard W. Sams & Co., Indianapolis, In.;* Modern Sound Reproduction *by Harry F. Olsen, Van Norstrand Reinhold Co., New York, N.Y.;* Manual of Sound Recording *by John Aldred, Fountain Press, London, England;* The Use of Microphones *by Alec Nesbitt, Hastings House, New York, N.Y.;* Microphones: Design and Application *by Lou Burroughs, Sagamore Publishing Co., Inc., Plainview, N.Y.;* Magnetic Recording *by Charles E. Lowman, McGraw-Hill Book Co., New York, N.Y.;* Four Channel Sound *by Leonard Feldman, Howard K. Sams & Co.; and* The True Sound of Music *by Hans Fantel, E.P. Dutton Co., New York, N.Y. We're sure you'll find the answers to your questions in any one of these fine publications.*

Locking Onto The Future

First of all, here's another round of congratulations on a great (and long needed) magazine. We've been selling *Modern Recording* in our store since your first issue and the customer feedback is terriffic.

Now that the age of basement studios is here, I have an observation—as well as a question. Half-inch 8-track is just getting worked in but I think everyone's realizing that one-inch 16-track is going to be very big in the next year or two. Obviously, any 16-track is going to be out of the price range of most people making tapes at home now. My question is: why isn't someone working up a sync-lock unit for two half-inch 8-track

Audio-Technica announces a creative new start toward better sound.



AT813
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Cardioid

AT801
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AT812
Moving Coil
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Every "live" recording has an excitement all its own. You must capture sound that will never be heard exactly the same again. And in some cases the recording is literally a once-in-a-lifetime chance. With no opportunity to remake the tape or disc.

Your creativity, your knowledge and experience are on the line. Now is when you appreciate the precision of Audio-Technica microphones. Performance is consistent. With results that help you extend your personal standards.

Audio-Technica gives you a choice of superb new electret condenser or moving coil dynamic microphones. A choice of omnidirectional or cardioid (unidirectional) pickup patterns. With smooth, extended response that complements the finest recorders. Audio-Technica microphones look, sound, and act very, very professional.

Add more than a little creative excitement in your life, with Audio-Technica microphones, today.

audio-technica. *Great sound, right from the start!*

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machines? It *can* be done. The pro companies like MCI and Scully, etc., have them. Why not invent something for the semi-pro home recordist who could be making 16-track masters rather than 8-track demos? I've spoken to several manufacturers about this; they say they've been throwing the idea around, but that's been the end of it.

We have sold three TEAC 80-8s from our store in the past two months and are now beginning to deal with Otari. The market is there and our customers are already beginning to talk about bigger set-ups.

As far as I can gather, two locked-up 8-tracks would be possible for as little as $8,000.00. When you're talking about a $6,500.00 difference between that and a $14,500.00 16-track (at the very least) the idea begins to seem worth it.

I don't know the technical implications of this project (whether it only works with servo motors, etc.,) but I really would like to know if someone's working on it—there's no better time than now.

—Paul Beckwith
Las Vegas, Nv.

The subject of synchronizing tape recorders, one to another of the same type, one to another of a different type or make, one to a film projector, to a camera, or to a videotape recorder is repeatedly raised by our creative friends.

You report that "pro companies like MCI and Scully, etc., have them." I do not know that this is entirely accurate; Scully does not have such an interlocking device that has been field proven. Ampex does make such a device that is intended to interlock and synchronize video and audio tape recorders, and several manufacturers of peripheral hardware do make such things which are intended for use in editing, primarily for videotape operations.

The very simple reason why few manufacturers offer such synchronizing or interlocking accessories is cost. There are variations, however minute, from machine to machine, even with the same model number and make, that present technical difficulties with respect to precise synchronization. There are variations from reel to reel of magnetic tape, the medium itself, that introduce errors in attempts to properly synchronize, that is to make two or more recorders operate precisely in synchronism, accessories are required that tend to cost more than the combined cost of two multitrack tape recorders. Without researching the exact commercial offerings, I daresay that the combined cost of two Otari (or Tascam) eight-track machines and the external synchronizing hardware could very well exceed the cost of a single 16-track machine. Mr. Beckwith might contact some manufacturers of videotape editing systems for more details.

I am reasonably certain that any progressive manufacturer of professional audio tape recording systems is examining such possibilities. And, because we are in a competitive commercial society, all such endeavors are considered company confidential.

Incidentally, it is not unusual for such interlocking systems as are used in videotape editing to require the dedication of one or more tracks to control/synchronizing signals; thus, interlocking two eight-track units does not provide a useful sum of 8-times-2 for music recording tracks.

—Leon A. Wortman
Marketing Manager
Otari Corp.
San Carlos, Ca.

The prospect of synchronizing two tape machines to achieve multi-track capability is certainly very appealing, and it is, of course, possible to design a device to achieve this. There are several areas in which problems must be resolved, however.

First, some problems might arise in adapting recorders which use line frequency related capstan motors, since the only way to vary the capstan speed on such units is with an audio oscillator feeding a line voltage amplifier which then drives the motor. This is a clumsy (and expensive) procedure. Servo motors offer more possibility for control, since it is usually easy to break into the servo loop and establish a new reference for the motor speed, which may then be precisely controlled or smoothly varied as required. The problem here is that every manufacturer uses a different system for servo capstan control, thus requiring a different interface from the sync controller for every machine.

Another difficulty is encountered in achieving absolute sync between two recorders while maintaining operational flexibility. We must be able to start, stop, and fast wind the tapes, yet achieve lockup within a few seconds at any point on the tape. The only means of implementing this would be to record a unique time code signal on one track of each recorder, which could

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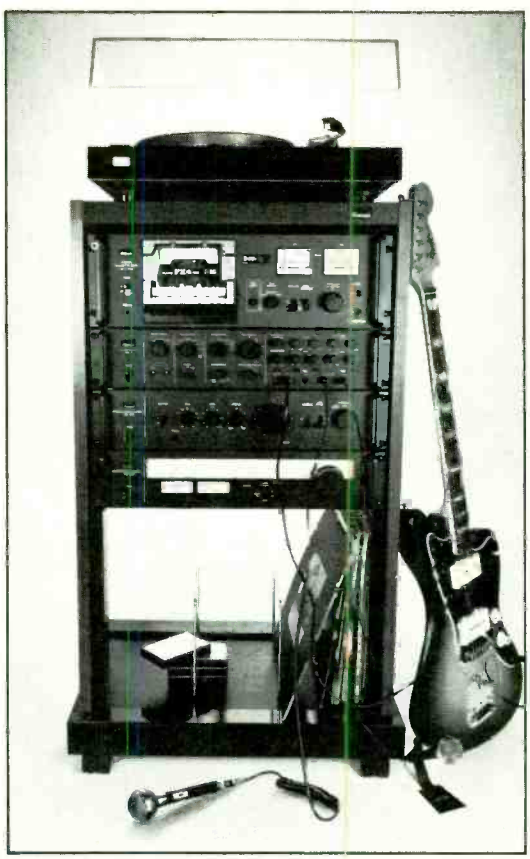

The Sansui AX-7 mixer with reverb is a unique new component that is ideal for every musician, audiophile and serious recordist. With an AX-7 you can create professional quality recordings at home, using your existing high fidelity components. You don't have to spend money to redesign your entire high fidelity system. You can experience a whole new world of recording for less than $240*.

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CIRCLE 47 ON READER SERVICE CARD

then be used as an absolute position reference by the control unit. We could now use resolving circuitry to analyze the relative locations of the tapes on the two recorders and produce the appropriate correction signals to bring the machines into sync.

Using discrete integrated circuit techniques, a system with all the features we have described would require quite a large volume of circuitry, with a consequent high cost. As a result, and since the demand for a sync controller would be relatively low, it would be difficult for a manufacturer to justify producing a general purpose device at the present time. The problem is compounded by the need to interface the device with the variety of multitrack recorders now available.

Despite this, there is some hope for the future. Work is now being done in the video field using microprocessors to control a multi-sync system for video recorders, and it is reasonable to expect that the same technology could be extended to our application. If so, it should certainly be feasible to produce a device to meet our needs at a (hopefully) reasonable price. Again, there would still have to be sufficient demand to encourage a manufacturer to take up the project—perhaps *Modern Recording* could run a survey to determine the interest in such a device.

—Bill Burns
Manager, Technical Services
Hammond Industries
Syosset, N.Y.

A Simple Explanation of Binaural Recording

Could you explain what binaural recording is? Does it involve any special equipment? I've been hearing rumors about it and I'm curious how it differs from stereo. Also, can a binaural recording be played back on present state-of-the-art stereophonic equipment?

—Elizabeth Herzog
Princeton, N.J.

Binaural recording is a special form of two-channel recording technique and was used as far back as the '30s during early experiments with stereo sound. The objective in binaural stereo is to capture and reproduce the exact audio information as it exists at a listener's ears. Since the sound field there is influenced by the acoustic properties of the listener, mainly his head, and differences between right and left ear are very subtle, it becomes apparent that

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CIRCLE 45 ON READER SERVICE CARD

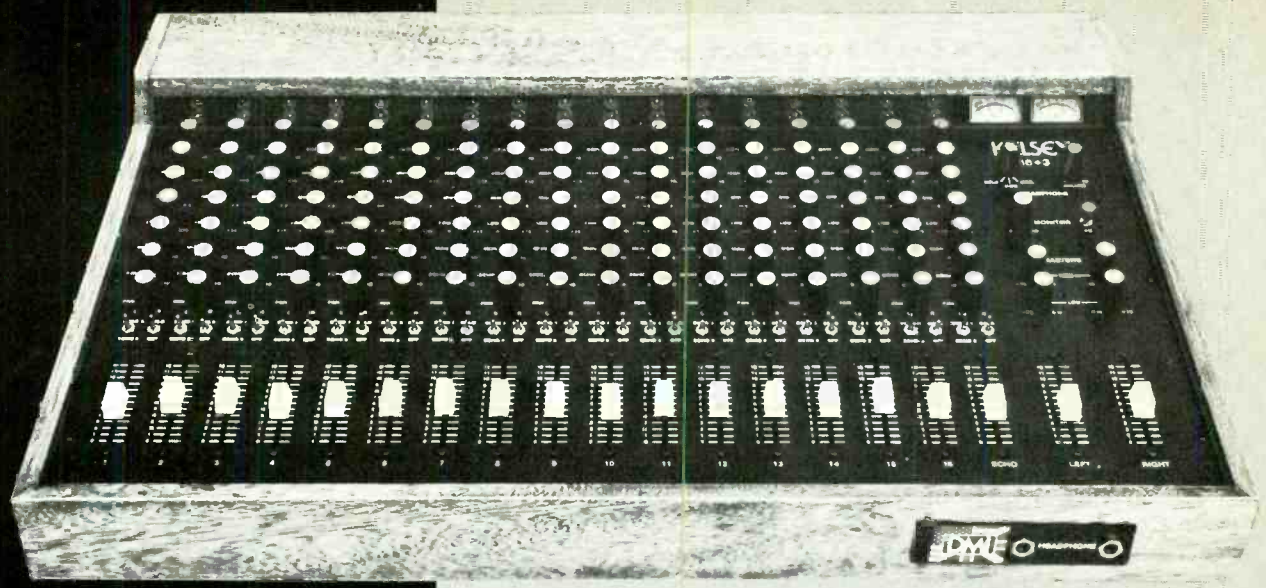
satisfactory results can only be achieved when employing closely matched high-quality microphones in combination with the recordist's real head or a carefully designed acoustical substitute "dummy head." Some such artificial heads are constructed with the microphone diaphragms taking the place of the eardrums while others have miniature microphones placed at the auricle. Complete binaural microphones are marketed by several manufacturers.

Optimum reproduction is achieved only when binaural recordings are played back through headphones, preferably of the supra-aural or Open Aire design. Playback through loudspeakers will result in a mediocre stereo reproduction unless the listener is positioned on the base-line halfway between them. In this uncommon situation, there will be some semblance of the typical spacious quality of binaural stereo.

Any existing quality stereo equipment can be used to process binaural signals. The only important requirement is the close match on all technical data for left and right channels. Where conventional two-channel stereo allows the location of a sound on an imaginary line between two speakers, and four-channel stereo at best brings the possibility of orientation on a plane, binaural stereo stands unsurpassed in realism. The exact preservation of the sound field around the head contains all information from which the auditory senses can derive the pertinent data for direction and distance of any sound. This technique is not only useful for recording musical performances, it is the only way to record groups of people (e.g. a conference) and enable the listener to clearly distinguish almost any individual voice at random. It allows an acoustical engineer or a performer to monitor how the sound is perceived in an auditorium. It can be used to enhance and complement visual events such as film and TV, etc.

Some binaural records have been made available and convinced most listeners of the advantages and simplicity when compared to standard or even 4-channel stereo. To achieve a vast improvement in fidelity, the only investment is for a binaural microphone on part of the recordist (studio). It is expected that this technique will gain increasing acceptance in the near future.

—Uwe Sattler
Technical Service
Sennheiser Electronic Corp.
New York, N.Y.



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SPECIFICATIONS:

INPUT

IMPEDANCES: Hi Z = 50 K unbalanced; Low Z = 200 Ohm transformer balanced. **MAX. INPUT LEVELS:** HI Z = +20dBm; Low Z = +8dBm. **GAIN:** Hi Z = 0-46dB, continuously variable; Low Z = 12-58dB, continuously variable. **EQ:** High ±15dB at 10K, shelving; Middle ±9dB at 2K, peaking; Low ±15dB at 100 Hz, shelving. **MONITOR:** Pre-EQ, unaffected by off switch. **ECHO:** Post-EQ, Post-fader. **LEDS:** Green lit from −10 to +21; Red lit from +15 to +21; 6dB headroom left when Red lit. **EQUIVALENT INPUT NOISE:** −110dBm from Hi Z input; −122dBm from Low Z input. **T.H.D.:** @ 1kHz, any level up to clipping typically less than 0.1 percent.

OUTPUT

IMPEDANCE: Nominal 600 Ohm unbalanced. **MAX. OUTPUT LEVEL:** 8.8V RMS @ 10K Ohm (+21dBV). **GAIN:** Mike in to line out +60dB. **EQ:** Hi ±15dB @ 3.5 kHz; Low ±15dB @ 35 Hz. **V.U. METERS:** "0 VU" = +4dBm at output of buss amp., switchable from stereo mix to monitor mix. **FREQUENCY RESPONSE:** Mike in to line out — ±1dB, 30 Hz - 20kHz. **SIGNAL TO NOISE:** Mike in to any output — typically 70dB. **T.H.D.:** Any output 1kHz any level up to clipping typically less than 0.1 percent. **POWER REQUIREMENTS:** ±15V DC @ ½ Amp.

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CIRCLE 71 ON READER SERVICE CARD

PRODUCT

By Norman Eisenberg

JVC INTRODUCES ELCASET DECK

From JVC comes word of its model LD-777, a stereo Elcaset deck that features "super ANRS" for noise reduction; three heads, of which two are SEN-Alloy and the third ferrite; six peak-reading LED indicators; automatic adjustment for bias and EQ through the unit's sensing system; and automatic switching for ANRS and other noise-reduction systems. The logic-controlled transport is activated by solenoids for "soft-touch" direct switching from one mode to another (bypassing "stop"). An automatic tape-standby mechanism, said to be a JVC exclusive, is designed to assure correct tape-to-head contact for instant recording and playback, and without damage to the tape. Independent controls provide for mic line or DIN/line input mixing. A built-in 400 Hz/15 kHz signal oscillator may be used to help compensate for sensitivity differences among various tapes.

Rated response with low noise/high output tape is ±3 dB from 25 Hz to 20 kHz. With ferrichrome or chromium dioxide tape, the high end extends to 22 kHz. S/N ratio is 58 dB; above 5 kHz it improves by an additional 10 dB with the ANRS switched in. Wow and flutter are spec'd at 0.04% WRMS; THD is listed as 1% with low-noise tape. Weighing 28.6 pounds, the JVC LD-777 Elcaset deck is priced at $799.95.

MARANTZ SHOWS NEW FRONT-LOADERS

From Marantz comes word of three new front-loading cassette recorders. Top-of-the-line unit is the model 5030, a three-head deck that uses a DC-servo transport and full-process Dolby noise-reduction system. Mic and line mixing is facilitated by an extra master level/fader control. The 5030 also has a three-button tape EQ and bias switch, extended-range VU meters and a defeatable peak-limiter.

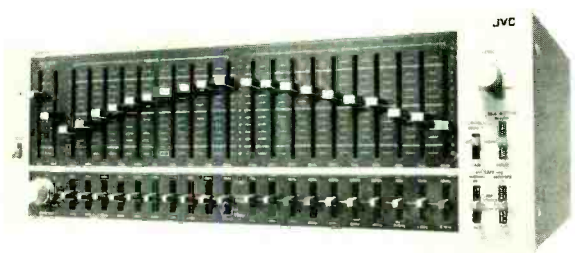
The model 5025 is a two-head recorder using the same DC-servo transport as the 5030 and sharing many other of its features. In the 5025 the Dolby system is single process. The model 5010 also

employs Dolby circuitry, extended range VU meters, peak limiter and EQ and bias settings for three types of tape.

CIRCLE 16 ON READER SERVICE CARD

JVC GRAPHIC EQUALIZER

The model SEA-7070 is a ten-band graphic equalizer from JVC which, says the company, can produce no less than 649,539 possible tonal combinations—thanks to its ten center frequencies each for left and right channels, plus two alternate, selectable frequencies for each zone that may be adjusted one-third octave above or below each center frequency. Additionally, there's an extra frequency control for each channel with eleven selectable positions that do not duplicate the center or the alternate frequencies. The thirty possible center frequencies are switchable for control ranges of ±6 dB or ±12 dB. A recording switch puts the equalizer section into the tape-recording circuit for simultaneous EQ while recording and a two-way tape-monitor recording/duplication option is provided, with EQ from either deck to the other deck. Input levels may



be set via a 3-position input attenuator; outputs include one variable and one fixed. Rated output is 2 volts at 0.005% THD with controls flat.

CIRCLE 1 ON READER SERVICE CARD

PASSIVE EQUALIZER

The model 4004 Passive Equalizer from White Instruments of Austin, Texas is a full-range, one-third octave equalizer designed for professional sound-reinforcement applications. The device employs twenty-four double-tuned constant-K notch filters on I.S.O. one-third octave centers from 63 Hz to 12.5 kHz. Each filter section provides from 0 to −15 dB of attenuation via a front-panel calibrated control. Responses of any two adjacent sections add smoothly without response curve "ripple." To control the ends of the audio spectrum, there are finishing filters which give varying degrees of roll-off. Designed to be inserted into a 600-ohm link circuit, the model 4004 is rated for less than 1 dB of insertion loss with all controls "flat." An accessory socket at the rear permits plugging in a low-level crossover for bi-amping. The device fits standard 3½-inch rack space. Suggested list price is $795.

CIRCLE 20 ON READER SERVICE CARD

AC TO DC CONVERTER

The Vista YS-300, smallest of the AC to DC converters from Clifford Industries of Camarillo, California, offers 300 mA at a selection of 4.5, 6, 7.5 or 9 volts DC output. Supplied with a four-way universal plug, reversing polarity capability, and switchable from an input of 117 volts AC to 220 volts AC, the YS-300 is encased in flame-proof plastic (to meet UL standards). Price is $6.95.

CIRCLE 9 ON READER SERVICE CARD

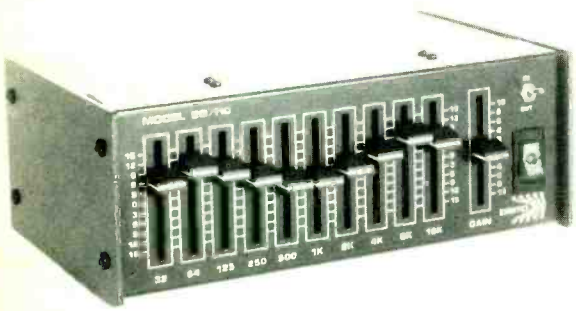
MARANTZ ANNOUNCES TOP-OF-THE LINE SPEAKERS

Marantz is offering its new top-of-the-line "Design Series" speaker systems which, claims the company, combine advanced technology of sophisticated speaker engineering with elegant contemporary styling. The new speaker system, according to Marantz (a subsidiary of Superscope, Inc., of Chatsworth, California), offers "the greatest bandwidth, smoothest response and minimum phase shift and distortion." Drivers used are a 12-inch woofer, 5-inch midrange, dome tweeter and dome super-tweeter. The model 940 is a floor-standing version; the model 930, a bookshelf style model. A novel feature of the system is a foam plug which can be removed from the cabinet to convert the system from infinite-baffle loading to a ported design, resulting in an increase of 3 to 4 dB in the 30-Hz to 75-Hz frequency region.

CIRCLE 15 ON READER SERVICE CARD

BIAMP SYSTEMS PRODUCTS

Biamp Systems of Tigaro, Oregon (whose model EQ/210 Graphic Equalizer was test-reported in our May 1977 issue) has issued various literature pieces describing its full line of audio products. In addition to the EQ/210, there are two other equalizers—the model EQ/110 (a mono version), and the model EQ/270 which offers 27 bands on standard 1/3-octave centers.



The company also has two electronic crossovers. The M2/V is a mono 2-way device; the model SM /23 is a stereo 2-way version that also may be used as a mono 3-way crossover.

Finally, Biamp Systems offers three stereo power amplifiers: the models TC/60, TC/120 and TC/225—rated respectively for 60 watts, 120 watts and 225 watts RMS per channel into 8 ohms. All amps have a mono bridging output option.

CIRCLE 19 ON READER SERVICE CARD

BOOSTER AMPLIFIER

Sonab of Foster City, California has introduced its model A4000, a dual-channel booster amplifier designed to double the output power of a stereo amplifier (up to 60 watts RMS). Maximum power obtainable with the A4000 is 120 watts RMS per channel. Strictly a stereo power amp with no additional controls, the A4000 may be switched on when extra power is needed. When not in use, a bypass circuit connects the original driving amplifier to the speakers. Weighing 22.5 pounds, the A4000 retails for $400.

CIRCLE 6 ON READER SERVICE CARD

PORTABLE SOUND BAFFLES

Interlocking studio "gobos" recently announced by Sugarloaf View of New York City are said to offer separation better than .85 NRC. Three models, in widths of 3 feet and 3½ feet, are available. The smaller models are designed for drum platforms and seated instrument positions. Plastic top portions provide vision and variable sound reflection. All models use high-quality brass casters, and they feature interlocking hinging for system connection and support without "legs." Durable washable fabrics are available in many colors, and are fireproofed if required.

CIRCLE 13 ON READER SERVICE CARD

NEW AUDIO RACK SYSTEM

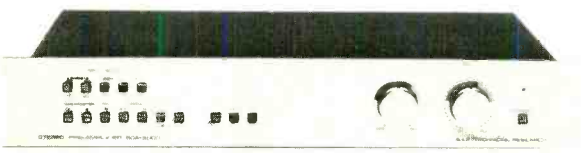
The "Audio-File" from Hammond Industries of Syosset, N.Y. is a new 19-inch rack mount system designed to be functional, attractive and flexible to suit a broad range of user needs in professional and consumer applications. The 48-inch high unit has a main frame with brushed aluminum finish, and it can be fitted with black plexiglass panels. Rack-

mounted and free-standing units can be combined in the overall housing. Priced at $350, the Audio-File is packed in kit form. For home users there's a decor-matching kit for about $210 which gives the buyer the option of selecting panels and shelves to match individual decor plans.

CIRCLE 2 ON READER SERVICE CARD

NEW PREAMP

Said to be a preamplifier for audio purists, the A & E SCA-2000, announced by Osawa, is designed to ensure maximum waveform transmission accuracy by its minimizing of phase-shift nonlinearities. Phase shift is held to 0-3 degrees across the 20-Hz to 20-kHz range. Basically a DC amplifier design,



the SCA-2000 uses a low-noise dual FET differential input amplifier and IC computer amplifier in the EQ stage, with separate negative feedback (for low-frequency), and RC (for high-frequency) networks, with rated RIAA accuracy to within ±0.2 dB. The preamp has two phono inputs with switchable impedances and a 2-mV sensitivity rating. Front-panel push buttons handle most of the control functions including program selection, filters, and a two-deck monitor facility with dubbing from either to either. Rated response of the preamp is from 1 Hz to 500 kHz through the phono input, with response down to DC on the aux inputs. Maximum output level is 10 volts, with 75-ohm/75-ohm matching or conventional bridging connections at the output terminals Suggested retail price is $950.

CIRCLE 11 ON READER SERVICE CARD

STOKIE—SOUND MAN'S MUSICIAN

From a UPI dispatch September 14, 1977: "He was one of the world's foremost experts on acoustics and his careful placing of instruments in a concert hall lent exceptional volume and color to his music."

Referring of course to Leopold Stokowski whose death at the age of 95 marked not only a loss to the world of music but also to the whole field of sound and of recording. For Stokie—as he came to be called affectionately by studio personnel and indeed by all who had the good fortune to work with him—more than any other major conductor actively and vigorously *and knowledgeably* pushed the basic idea that sound conveyed the musical message. He literally put his money where his mouth was by boldly innovating, both at "live" performances and in recording sessions. It once was said of him that he spent as much time in the engineer's booth as he did on the podium—not as a pest to bug the recording team, but rather as a talented member of that team who knew what had to be done (on both sides of the glass panel) to create a modern kind of sound that did justice to the music while taking full advantage of the potential of modern recording techniques. In the early days of electrical recording there was no "producer" as we know this role today, and in a real sense Stokowski antedated and foretold that important function even while welding the awesome sonic and musical forces of some one hundred players who made up the Philadelphia Orchestra. Often his efforts were "in violation" of convention, as for instance when he experimented in the 1920s with the orchestra's seating plan and finally evolved the arrangement whereby the descending frequency range for the strings is fanned out from left to right, with the cellos up front—a seating plan that has become virtually the standard for all major orchestras. In general he was constantly preoccupied with how instruments sounded and how to best convey the sound to the audience.

In the mid-1920s when electrical recording came in, it really turned him on, and his enthusiasm and insights in turn stimulated the new process and its practitioners. Early on he employed electrical amplification for some instruments; he brought the Theremin into the concert hall; once he used oscillators to generate noise by way of educating an audience; in 1933 he staged a sound show at Philadelphia's Academy of Music in which the orchestra—playing remotely—was heard over three loudspeakers behind an empty stage while Stokie ran a control panel for the transmitted signals. Probably his greatest contribution involved multi-channel sound: a major event in this area occurred on April 27, 1933 when, to demonstrate a new wired sound-transmission system developed by Bell Laboratories, the orchestra piped a three-channel stereo concert (under the engineering direction of Dr. Harvey Fletcher) from Philadelphia to Washington, D.C. The multi-channel idea climaxed in Stokowski's collaboration with Disney to produce the sound-track for the 1940 film "Fantasia" which the maestro recorded in 18 tracks, something unheard of at that time. Hindsight today tells us that this pioneering effort paved the way for wide-screen sound, mixing-down, ambience enhancement, stereo and of course surround-sound generally.

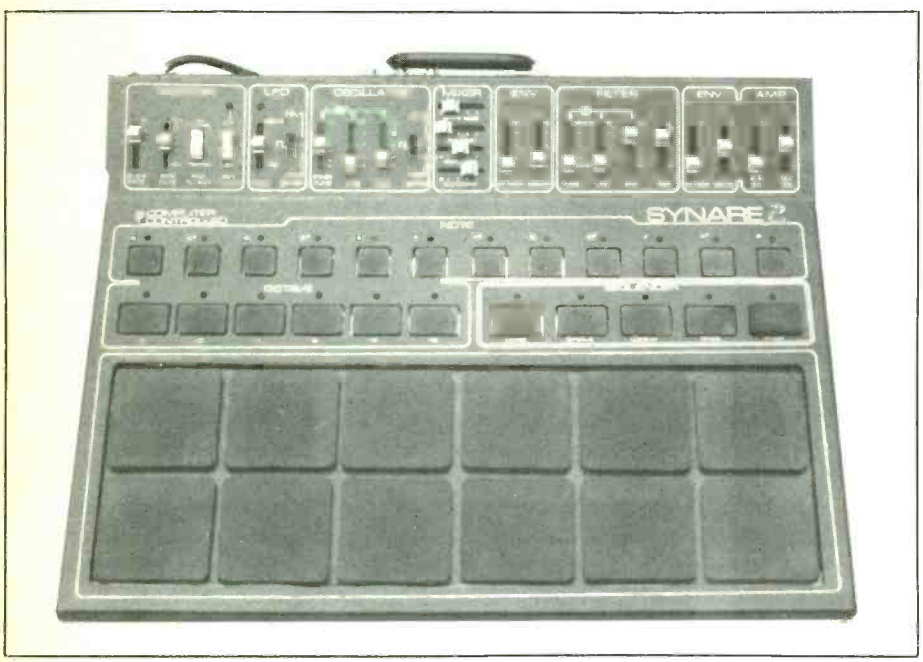
Stokie, then, not only helped advance "the state of the art," he also set an example for subsequent performers and recordists to follow.

MUSICAL NEWSICALS

SYNTHESIZER EQUIPMENT

Several months ago we covered several of the percussion synthesizer systems available on the musical instrument market at that time. The units were fairly similar in their basic design and function; they, and most of the other percussion synthesizer systems available, were fairly simple devices which basically generated one or more varying control voltages when struck. At the Atlanta NAMM show two manufacturers were showing sophisticated new systems which overcome many of the limitations of previous designs.

The first of these two new percussion synthesizer systems is the Synare 2 from Star Instruments Inc. (P.O. Box 71, Stafford Springs, Ct. 06076) which replaces their earlier Synare model. The



Synare 2 uses a digital microprocessor to allow pre-programming of control functions and to provide a 125-step sequencer. The sequencer incorporates an exclusive "Accent Add" function which allows the musician to play "live" on top of a pre-programmed sequence which is playing back. The Synare 2 embodies a unique approach to the problem of playing and controlling the device simultaneously. The instrument is played via a panel of thirty-five rubber pads which are touch-actuated by hand or drumstick. The twelve largest pads are the actual playing pads. When the Synare 2 is first turned on, these twelve pads are tuned to a tempered chromatic scale beginning at middle C, but they may be individually retuned at will by first touching one of the twelve Note Pads to set the musical note, then one of the six Octave Pads to select the octave for the note (from four octaves below middle C to two octaves above), then touching the Playing Pad to which this note/octave combination is being assigned. This retuning procedure may be used at any time to change the pitch of the pads, even while playing. The remaining five of the thirty-five pads control various sequencer functions along with a slider control for sequencer playback speed. In addition to the pad panel is the voice module, which contains the microprocessor and the associated circuitry which digitally generates all the source waveforms. Among the more unusual features of the voice module are digital glissando, dual envelope generators, sub-octave oscillator, a digital low-pass filter with 24 dB/octave slope which tracks the frequency of the note played and provision for foot pedal control of various functions.

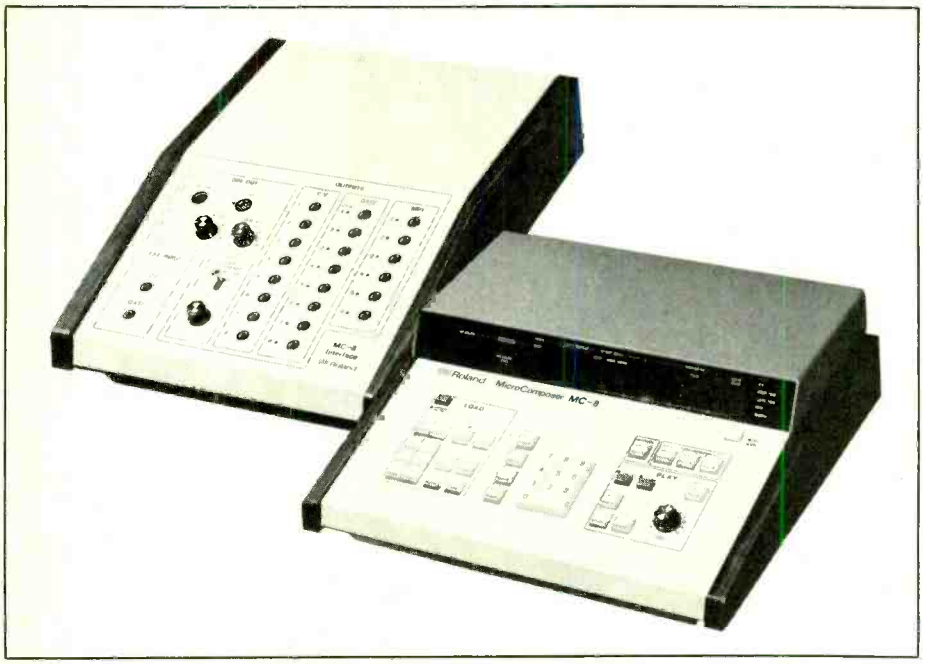
CIRCLE 4 ON READER SERVICE CARD

A very different approach to electronic percussion is seen in the Syndrum from Pollard Industries, Inc. (9014 Lindblade Street, Culver City, Ca. 90230). Syndrum's designers felt that the most serious limitations of previous attempts at synthetic drums were a result of monophonic design. Not only was the drummer limited to one note at a time, he was also limited to one basic set of attack and decay characteristics and timbres, the only functions he could vary from note to note were pitch (by playing on more than one pad tuned to different pitches) and volume (by playing harder or softer). In the Syndrum system, each drum unit has its own tone generators and control panel so that each one is actually an independent synthetic drum which can be set for its own individual sound. Each control module has controls for volume, sustain, filter range, vibrato rate, vibrato spread, coarse tuning, fine tuning, and switches to select one of three signal waveforms (sine, triangle or square), one of three vibrato waveforms (ramp, triangle or square), upward, downward or stationary filter sweep and three degrees of snare sound. In addition, each syndrum unit has an associated sensitivity control for precise setting of

the individual drum's dynamics and "feel." Syndrum systems of up to eight units are available complete with mixer modules having high and low impedance outputs and a headphone output.

CIRCLE 14 ON READER SERVICE CARD

RolandCorp US (P.O. Box 22289, East Los Angeles, Ca. 90022) has unveiled their MC-8 Microcomposer, which is a sophisticated digital sequencer utilizing a microprocessor for greatly increased capacity and versatility. The



MC-8 is loaded directly from a standard keyboard synthesizer or from its own 10-digit adding-machine type keyboard. A unique feature is the multi-channel output which allows simultaneous control of up to eight independent voice lines. The contents of the MC-8's memory can be loaded onto tape for storage and later re-used using a cassette tape recorder. The MC-8 is compatible with all patchable synthesizers including Roland's own 100 and 700 series synthesizers.

CIRCLE 5 ON READER SERVICE CARD

MUSICAL INSTRUMENT ACCESSORIES

Rowe-DeArmond, for many years a manufacturer of a wide variety of magnetic and contact dynamic instrument pickups, has introduced a new, high-output electric guitar pickup called the Superbucker. The Superbucker features the same high output level as todays other "hot" pickups, but is said to have a warmer sound than competitors' models. The pickup is a humbucking type with individually adjustable pole pieces and is an exact physical replacement for the large-size Gibson humbuckers as found on Les Paul and SG models and their copies.

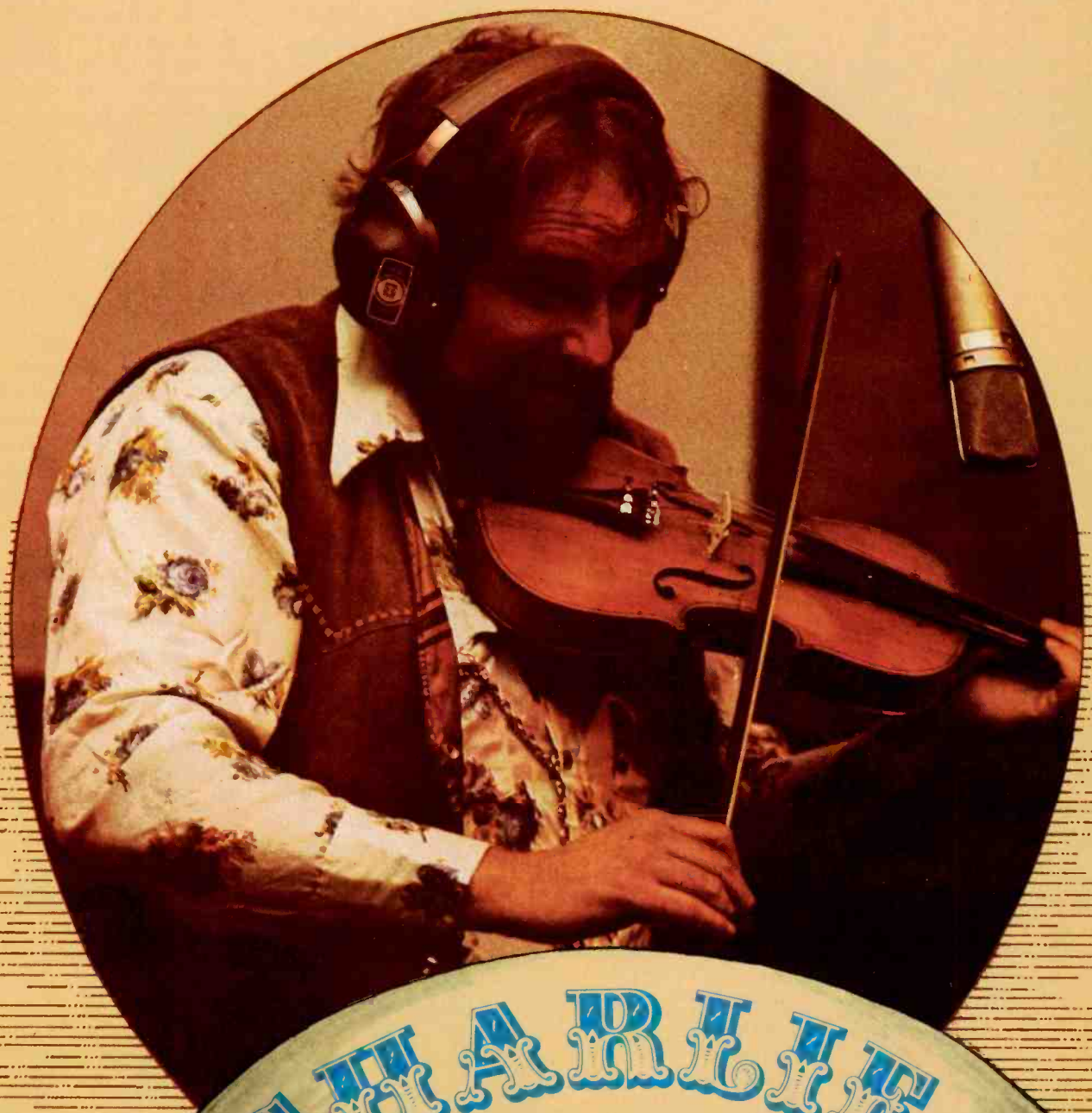
CIRCLE 18 ON READER SERVICE CARD

Among the newer group of pickup manufacturers, one of the best-known names is DiMarzio Pickups (643 Bay Street, Staten Island, N.Y. 10304), who have just introduced three new models. First is a replacement pickup for Fender Stratocasters which for the first time gives Strat players individually adjustable pole pieces. The model is called the SDS-1 and features a tailored frequency response and higher output than other single-coil pickups, yet is said to exert 50% less magnetic pull on the strings than the stock Fender pickups for greater sustain. The second new model is the P-Bass, a high-output replacement pickup for the Fender Precision Bass which also features adjustable pole pieces. The third new product from DiMarzio is the Key Mix System, a two transducer plus mixer set-up designed for acoustic pianos but also applicable to one or two acoustic guitars.

CIRCLE 8 ON READER SERVICE CARD

Altair Corporation (P.O. Box 34, Ann Arbor, Mich. 48107) has announced the introduction of one of those products that starts us all wondering why nobody has thought of it before. In this case the product is Altair's Model PW-5 Power Attenuator ($119.50). The most basic means an electric guitarist has for changing the tonality of his instrument is by inducing amplifier distortion. Most professional amplifiers nowadays have a master volume control which allows the player to use the input volume control to vary the amount of distortion while controlling the playing volume with the master control. With many higher-power tube amplifiers (Marshall being perhaps the prime example) a significant portion of the distortion at high volume comes from the output stages of the amp. Unfortunately, the only way to achieve this kind of distortion is to crank up the master volume —which of course means playing very loud. This is where the PW-5 comes in; the PW-5 is connected between the amplifiers output and the speakers and does nothing more nor less than attenuate a portion of the amplifier's power before it reaches the speaker. This allows the musician to set the amplifier for the distortion he likes and then control the speaker volume with the attenuator. The PW-5 has an in/out switch and a large attentuation control knob on its front panel, and also boasts a high level line output which can be used to drive a slave power amp or feed a P.A. system with the fully distorted signal, adding even more versatility to an already useful unit.

CIRCLE 12 ON READER SERVICE CARD



CHARLIE DANIELS

A Session With

By S

During the hazy, lazy days of July and August, in Macon, Georgia, the Charlie Daniels Band is busy recording and mixing its latest album, *Midnight Wind*, at Capricorn Studios because, according to Charlie, "There's a lot of good studios around, but Capricorn is built with the idea of a band in mind."

Midnight Wind is the fifth studio album the Charlie Daniels Band has recorded in Macon (a "live" *Volunteer Jam* album was mixed here). Daniels Band producer Paul Hornsby says, "The secret to this studio is knowing how to harness and contain all that musical energy the band puts out and get it down properly on a small enough format so folks can enjoy it in their living room."

Along with groups like the Marshall Tucker Band, Sea Level and Grinderswitch, the Charlie Daniels Band depends on Capricorn Studio for the vinyl character of its "Southern Rock" with its origins in country, rock, jazz, soul and rockabilly. Originally, the Capricorn room was built in 1969 for the Allman Brothers. It has since gone through several design changes ranging from early nouveau funk through its *Brothers and Sisters* [the title of an Allman Brothers album] stage to the current Tom Hidley-designed "nosecone" studio control room.

Midnight Wind engineer and Capricorn staffer Kurt Kinzel explains, "The design of this room plays a very important part in all the stuff we do. The way the instruments sound when played in the studio, if they are captured on tape right, will give a final recording that is very close to a live performance."

Inside the Capricorn control room there are: a 24-track Studer tape machine, one 4-track and two 2-track Studers, an Automated Process console with 30-in and 24-out and Dolby noise reduction units. The monitors (Hidley) are two JBL 15-inch speakers powered by Studer amplifiers. Outboard gear includes two Eventide digital delay units, and an Eventide Harmonizer. The Memory Plus system by Allison Research will soon be added for computerized mixing.

Engineer Kinzel stresses that the abundance of Studer tape machines stems from the fact that "they are a lot gentler on tape than most other machines." For the *Midnight Wind* album Kurt is recording with Agfa 550 tape at 15 ips. "They stopped making the 550 about two years ago," Kurt says, "but I have a stash of it; the tape sparkles when you play it back so you get more of what you put into it."

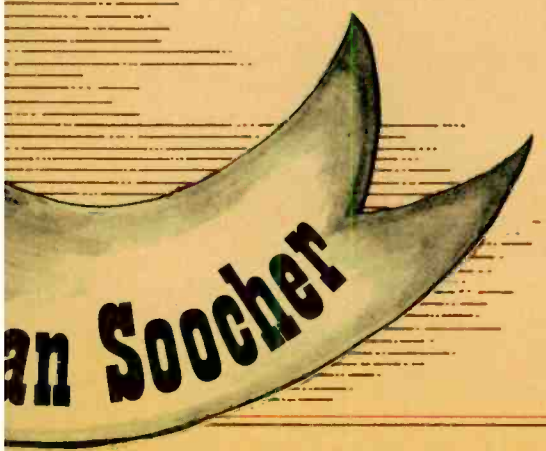
Kurt, who is working on his fourth studio album with the CDB, came to Macon from the Record Plant in Sausalito, Ca. where he started as an assistant engineer when that studio was still in its construction stages.

"I like the control room sound here as much as in Sausalito. The rooms at the Sausalito Record Plant are smaller. The speakers at Capricorn are further apart, so there's more of a stereo spread."

For the Nashville-based Charlie Daniels Band, Macon, Georgia is a home away from home. "Hell, even our road crew pampers us when we're here," boasts the garrulous leader of the band. Even so, *Midnight Wind* is taking longer to complete than previous albums. In all, the band devotes nearly ten weeks to rehearsing, recording and mixing *Midnight Wind*.

Charlie Daniels' own studio experience goes back to his days as a session player in Nashville, which were highlighted by his work on Bob Dylan's *Nashville Skyline, New Morning* and *Self-Portrait* albums.

"There were times during those sessions I got so into the music I actually



forgot Dylan was there," Charlie grins. "Bob was just one of the boys. And that was a *good* atmosphere."

Later, Charlie produced two albums for the Youngbloods and two solo albums for Youngblood member, Jerry Corbitt. Daniels admits that being on both sides of the glass has helped him out. "I prefer playing, but I've enjoyed producing too. I was never into the technical aspect of producing, though. I really don't know very much about the board. I don't want to have to tell an engineer or producer how to get there. I figure that's their job. My job in most cases is making music."

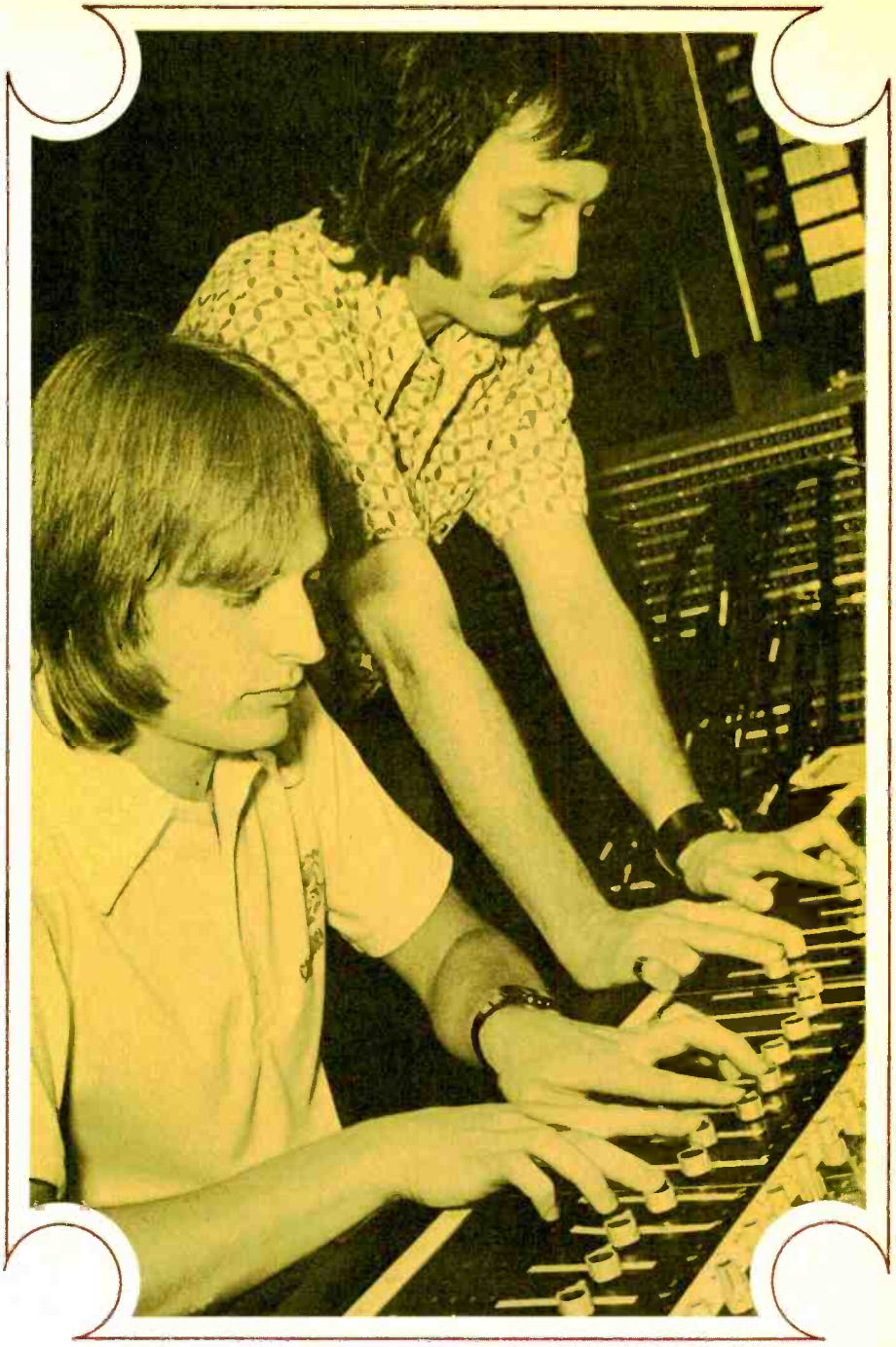
Relaxing in the studio lounge, Charlie stops speaking for a moment, empties a wad of chewing tobacco into a gold spittoon and continues, "I like the sound of our music hot and clean. *Put the dirt in the music, not in the studio.* As far as I'm concerned Paul Hornsby and Kurt Kinzel are the best team around for what we need. They save a lot of time and they make superclean records. They hear blips and buzzes that I never hear."

Paul and Kurt are the same team which has engineered and produced a string of hit albums for the Marshall Tucker Band (Kurt has engineered three, Paul has produced six). What are the differences between recording the Marshall Tucker Band and recording the Charlie Daniels Band?

"The Marshall Tucker Band is more experimental," answers Paul. "Their songs are not as well-rehearsed and laid-out as the Daniels Band. A lot of Tucker songs are created in the studio. The end result, in either case, is usually very good."

For Paul, the producer's most important function is knowing when the artists have reached their finest performance. The artists are usually too close to the music to know, and the engineer is too busy watching levels. The producer is that final inspiration for musical perfection. "Then again, it all depends on the artists and the producer. With the Charlie Daniels Band, that's my function."

Paul Hornsby began his professional career as a musician playing bars and concerts before joining Capricorn as a session player in 1969, when Capricorn Records was signing single artists like Livingston Taylor. (The Daniels Band is on the Epic label.) As the record company became group-oriented, Paul evolved to engineering, then producing, and he still relies on his knowledge of the board to suggest EQ settings or when to ride the faders.



Photos by Eliot Kamenitz & David Walton

When the Charlie Daniels Band is ready to record its basic tracks, all six players assume positions in the studio so they can see one another. Charlie Daniels is on guitar or fiddle, Tommy Crain on rhythm or lead guitar, Joel "Taz" DiGregorio on keyboards, Charlie Hayward on bass and Don Murray and Fred Edwards on drums.

Basic tracks are often recorded without follow [reference] vocals; they are added later. Primarily, this method of recording six-piece rhythm tracks assures a minimum of overdubbing in the long run. As the songs are recorded, false starts occur if one of the musicians is tripped up by a snag in the arrangement, but most of the time, few complete takes are neccessary. Nineteen basic tracks are cut in approximately eight days.

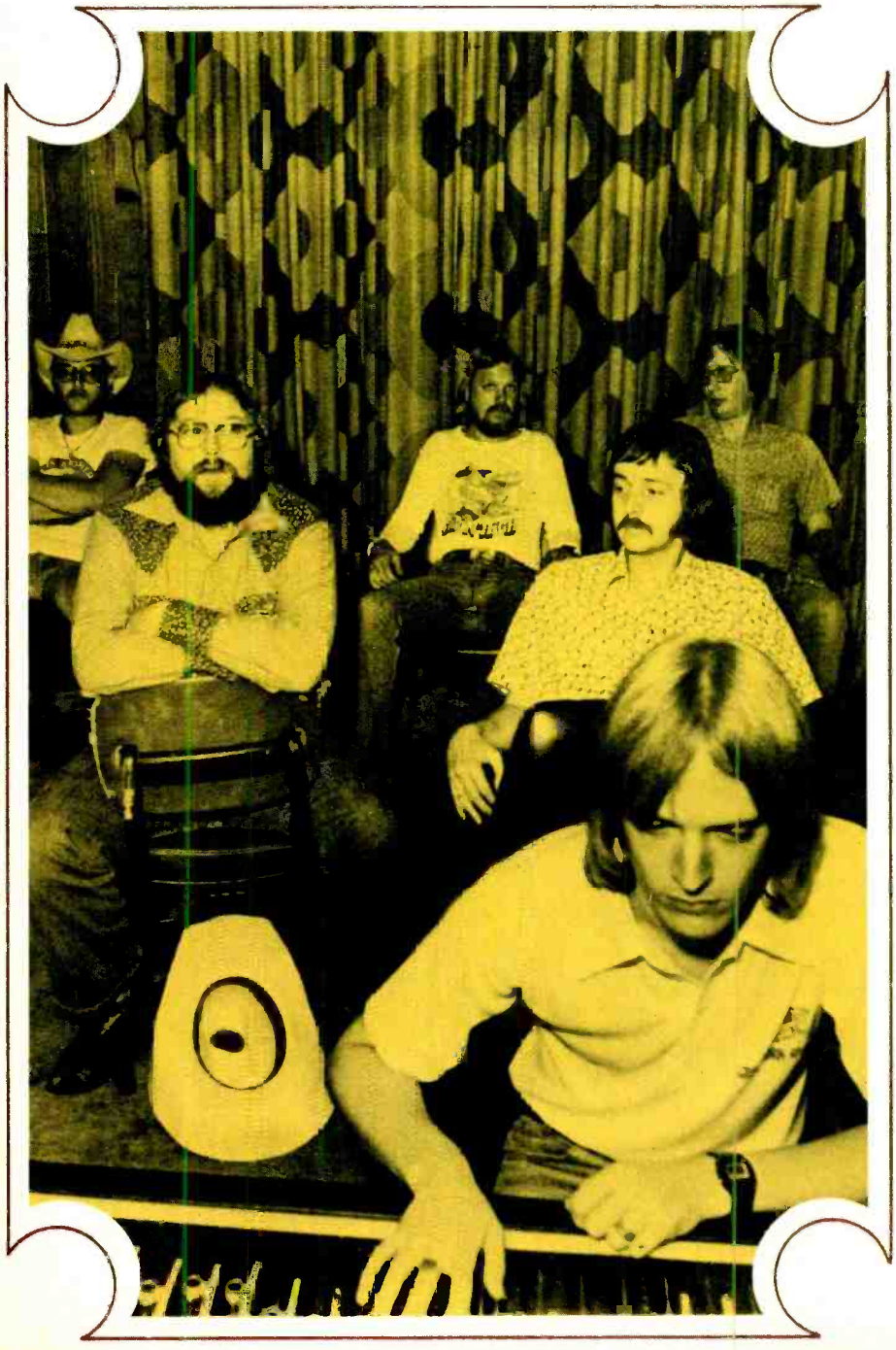
For miking techniques, Kurt is employing much the same methods used on the last few Daniels albums—with an interesting switch. "I did use the Sennheiser MD211U for the bass this time after reading about it in an issue of *Modern Recording*. The mic is an omni, but the bass amp is isolated [to prevent leakage]." Onstage, eight cabinets, each containing two 15-inch

JBL speakers, are provided for the bass. In the studio, two cabinets are placed in a booth with blankets draped over the top.

Several of the guitar and piano solos are also recorded during the basic tracks. Occasionally, a sloppy solo is recut, but more often than not a composite is made where rough spots are touched up by punching-in. Paul Hornsby says he wants all solos as precise as possible. "They're like lead vocals for me." On overdubbed guitar solos, amps are set up the same as during the basic tracks, and the same mics are utilized. Charlie Daniels and Tommy Crain adjust their equipment to obtain the sound they want for a song.

The acoustic fiddle is cut "live" on "Ode to Sweet Smokey" and "Redneck Fiddlin' Man," so Charlie stands in the side room to avoid having to have the instrument close-miked. Electric fiddle is cut in the studio on the hardwood floor as are all acoustic overdubs, including the viola and four violins on "Heaven Can Be Anywhere," the album's showcase ballad.

On acoustic overdubs, Kurt Kinzel utilizes his favorite mic, the AKG-C24. "It's a very hot stereo omni, and it requires very little EQing except for bringing up the very bottom and adding to the very tip-top."



Most of the EQing is done during recording rather than during the mixing. While Kurt usually adds 2-4 dB on most instruments, he adds 6 dB to the tom-toms in the drum kits. "But I've been known to roll off as much as 12 dB on some tracks."

The middle of the room is somewhat open, but dead, so only the vocals are recorded there. Although Tommy and Taz also sing, Charlie Daniels sings most of the lead vocals. "I don't consider myself a great vocalist. I try to sing on key and make it sound natural —the way I talk. I try to put as much feeling into it as possible, but I don't scream or holler very often."

When all the songs are recorded and all solos cleaned up, the time for mixing has arrived. Paul Hornsby is instrumental in deciding which tunes will compose the new album, but everyone in the Daniels Band has a voice in choosing the final ten.

For the first mix, Paul, Charlie Daniels and members of the band leave the control room to engineer Kinzel and assistant Steve Tellisch, although keyboard player Taz stays behind to pick up pointers for his production work with the Winter Brothers Band. Paul Hornsby says, "I can't sit there for hours and listen to the songs being set up and still be objective in my opinion. I like to go in towards the end of the original mix and fine tune it, more or less, because I trust Kurt and I know it's all there from the recording end of it. Besides, I know what the final results should be before we ever start mixing."

Kurt sets up each song building outwards from the drums. Bass, rhythm guitars and keyboards are added next. Then solo instruments and vocals. Echo chambers are mostly EMT 140s at 30 ips. For recording purposes, the upstairs storage room with its natural ambience is used as an echo chamber, but only early in the morning "before the trucks start rolling in."

With Paul commenting on Kurt's work, the songs are mixed one by one. Charlie Daniels and band station themselves in the control room for the final mixes. On "Sugarhill Saturday Night" Tommy Crain thinks the bass needs more presence. Charlie Daniels wants to hear more fiddle in "Redneck Fiddlin' Man." On "Grapes of Wrath," Paul, Kurt and Steve Tellisch work and rework the faders during a particularly stubborn section of music.

Mixing continues hour after hour, breaking only when maintenance engineer Dennis Darling steps in to service the equipment.

Finally, the album is finished and congratulations are in order. While everyone is gathered in the control room, Charlie Daniels figures it's time for a story. "Years ago, I was recording in a small studio in Washington, D.C. called Edgewood. We'd been recording there all day, and it was getting on towards evening and the studio was getting ready to close down. I was overdubbing a guitar part with my earphones on and all of a sudden I heard, 'Wheeeeooo!' (Charlie whistles). They heard it in the control room too, and one of the engineers jumped up and took off runnin'. Seems there was a guy up there painting next to where they had their echo chambers and he was whistling, and it was leaking into my earphones."

Charlie Daniels gets up to leave the studio, and contemplating a fresh piece of chewing tobacco, he chuckles, "I was sort of hopin' that on this album we could find an appropriate place where we could overdub the sound of my spittoon."

MIC SELECTION

Fred's Drums

Kick	C-500
Snare	AKG-452EB
Hi Hat	Sony C-377
First Tom	Neumann U-87 w/Kepex
Second Tom	U-87 w/Kepex
Floor Tom	U-87 w/Kepex
Floor Tom	U-87
Overheads	U-87

Don's Drums

Kick	U-47
Snare	KM-84
Hi Hat	Sony C-377
Toms	(same as Fred's)
Overheads	AKG-452EB

Bass
Direct
Sennheiser MD211U

Electric Guitars

Charlie	Sony C-377
Tommy	Sony C-377

Keyboards

Piano Bottom	Sony C-38
Piano Top	Sony C-38
Rhodes	Direct
Organ Bottom	AKG-C24
Organ Top	Electro-Voice Triple 6

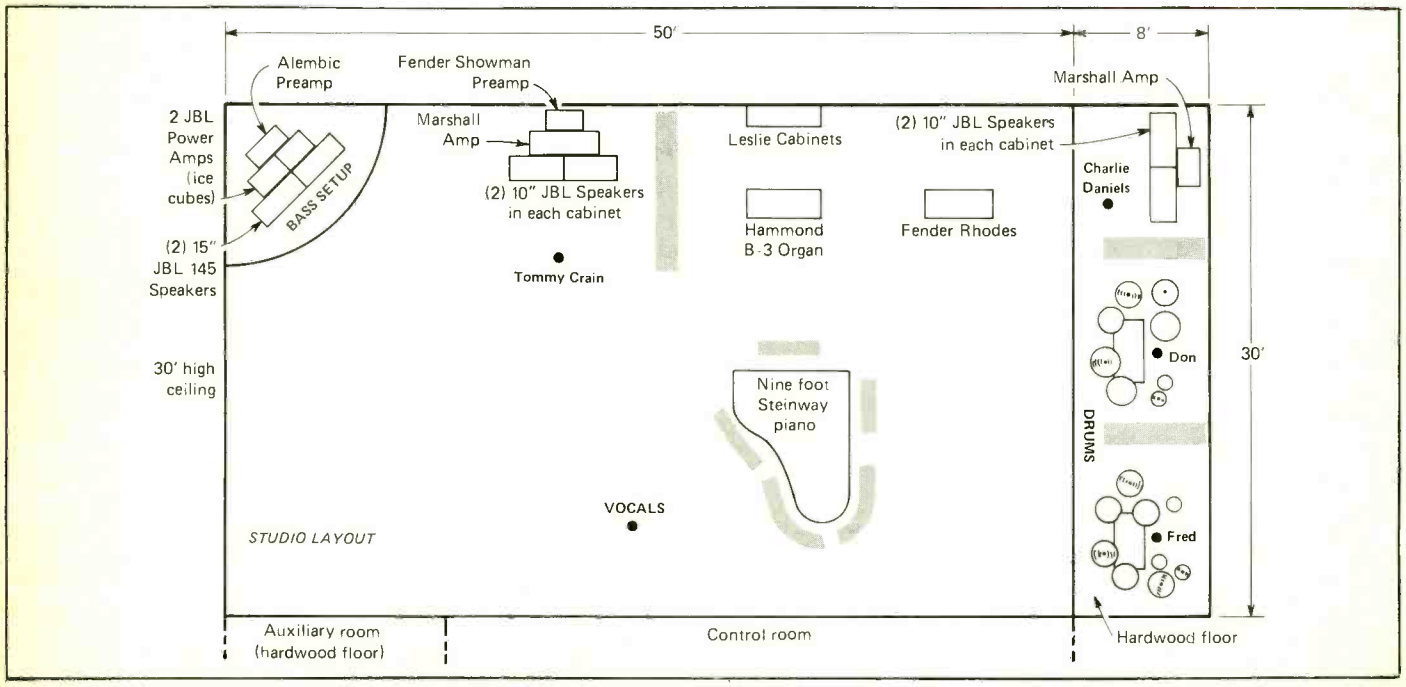
Vocals
AKG-C24 w/one capsule

Fiddle
AKG -C24

Room Mic for cueing
SM-57

Strings
U-87
Overhead AKG-C24 w/both capsules

All Other Acoustic Instruments
AKG-C24 w/one capsule



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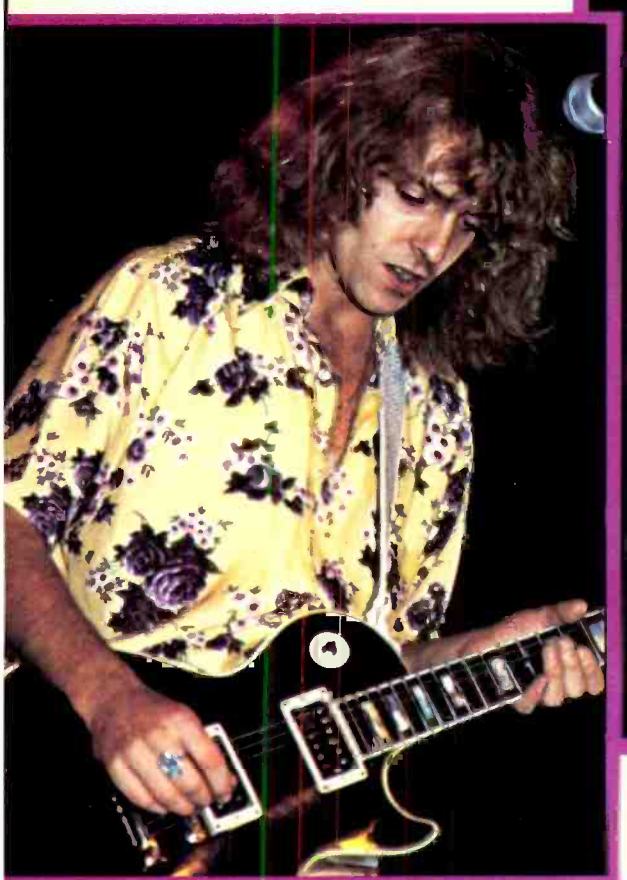
CIRCLE 92 ON READER SERVICE CARD

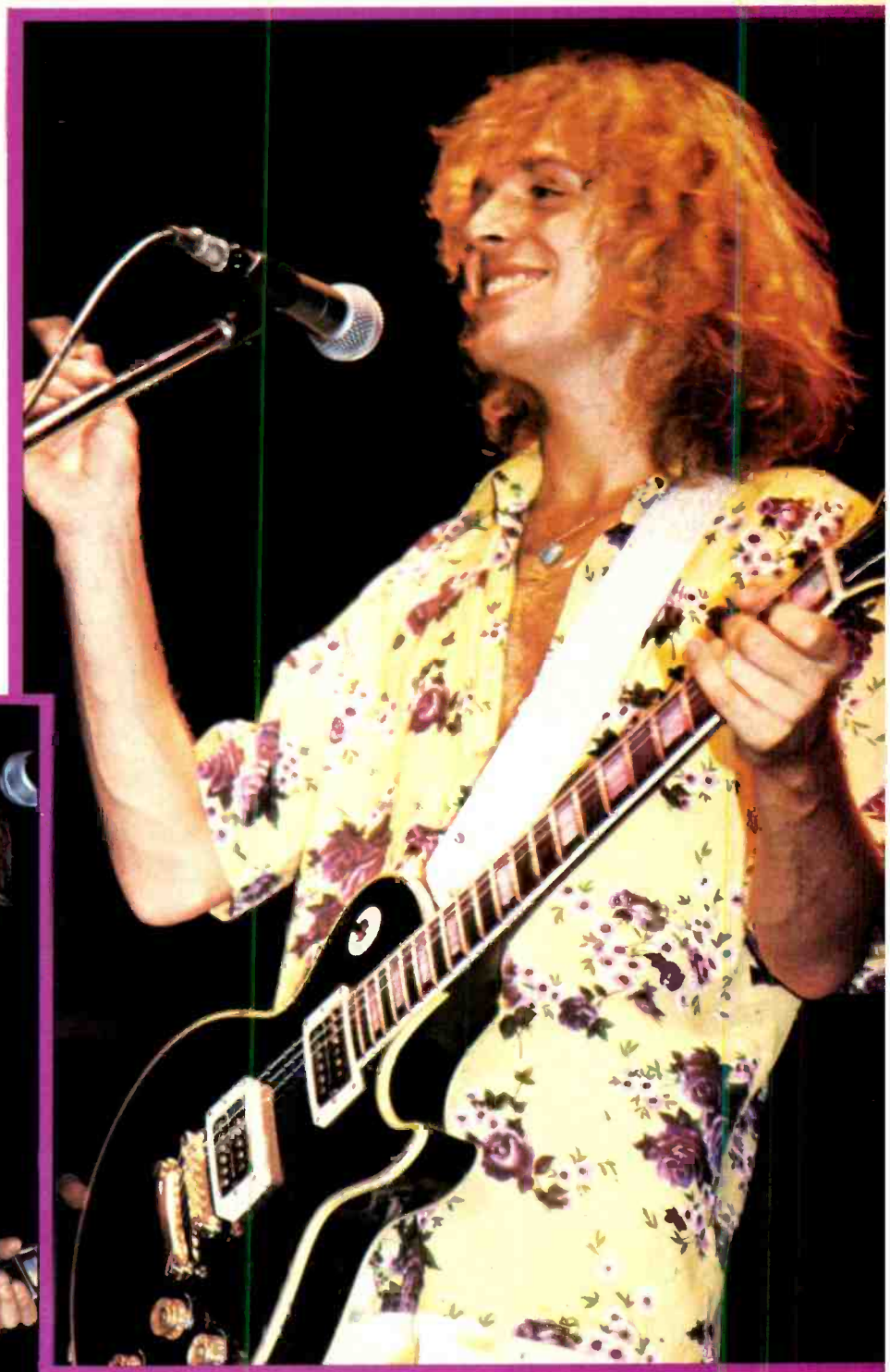
FRAMPTON LIVE

SOUND REINFORCEMENT INDOORS

Seattle's King County Domed Stadium (Kingdome) has become the latest center of controversy in regards to the sound reproduction of music. Several major acts such as The Eagles and Aerosmith have had sound problems when performing there, and the local press has been quick to write off the Dome as an acoustically poor hall for "live" music. Well known for inconsistency in their numerous inaccuracies due to lack of story preparation the press's edict didn't phase concert going fans, although the stories were picked up by others in the business and aided acts like Fleetwood Mac, after flying in their own sound people to inspect the premises, in deciding not to play the Kingdome.

While true that the sound is not comparable to that of the 17,000 plus capacity Coliseum across town, in some areas the problems are unavoidable. As a structure, the immediate problem is that it is totally enclosed and therefore has a long decay rate. Being almost perfectly circular inside (the stage is erected in front of the scoreboard where it is only double-





Color photos by Lynn Goldsmith

By Gil Padalinsky

decked, whereas the opposite end of the hall is tri-level, with seating to the dome) sound can roll around the back of the hall. By being totally enclosed The Kingdome presents a dilemma to a sound company: Do you use an outdoor system, which is created with volume and large numbers in mind, but not designed to contend with acoustics or returning sound; or do you go with an indoor system that is geared to a hall's acoustics but doesn't deliver anything near the necessary wattage?

Worse yet, what do you do if your sound company isn't big enough to afford two different types of systems, thereby eliminating any choice whatsoever? Before you ask how such a sound company would dare call itself professional, or could afford to be in the business, remember that the artist in question rents the system and pays for, the transportation of same, union help to mount it, room and board for those touring sound people associated with the sound company, etc. With a daily overhead well into the thousands (a group playing a tour of 3,000 seat venues, with an entourage of twenty-four people has a daily overhead of $5,000) on that score alone, it's no surprise that bands go shopping for sound companies just like they do for guitars. A sound company that is willing to undercut another—and possibly sacrifice certain frequencies and expertise as a result—can save a band a bundle over a thirty-to-forty date tour. With less than half-a-dozen concerts held thus far in the Kingdome, the "book" on the hall—sound-wise—is still being written. So, a critic who expects any domed stadium to reproduce sound as though it were designed with that intent in mind is being naive. The Kingdome was designed with the *visual* in mind to accommodate roughly 65,000 people. On that score it is highly successful, for whether viewing a sporting or rock event, you don't feel as though you're miles removed from the scene. However, as you might expect, the more people you have attending, the more extraneous sound absorbed. Attending a typical baseball game with 22,000 people scattered all over the seating configuration results in a very reverberant and oftentimes lost PA announcement, and every hit sounds like the result of a broken bat.

Some may blame the sound company if disappointed with the sound, but the blame may not rest entirely on the sound company. It doesn't make much sense for instance to set up a very elaborate sound system where the owners and operators know what to expect from every piece of equipment only to turn the running of the system over to a soundman selected by the band. Sometimes he knows the system and the hall, other times not. Regardless, it's the sound company that's remembered. If each act sounds terrible, not only that evening but *throughout* the tour, then the sound company *is* at fault. Any sound company that has a major group should be able to do a good job on any given night because they're not there through luck, but rather because of expertise and years of practice. But then too, anyone can have an off night. So what do you do?

Detailed Findings

If you're the John Bauer Concert Company, a promoter who is well respected for paying attention to minute details, and are promoting a concert with Peter Frampton, Foghat, Geils (formerly the J. Geils Band) and Blondie Chaplin, you learn who's doing the sound on Frampton's tour, in this case, Clair Bros., and fly out co-owner Roy Clair to look at the hall. Three weeks before the concert, Roy arrives. By this time, Kingdome officials also had become concerned about the poor sound reputation the hall was getting and hired a consulting architect to see what could be done. When Roy Clair arrived, the architect's report was in. He had come up with a plan to make the Kingdome sound perfect. The findings were that the best thing would be to put up acoustical suspension devices; or to put speakers very close to the audience through the use of digital delay lines. The time required—or rather the lack of time— however, made it impossible to institute the corrections. It might take two months to mount the acoustical devices, to say nothing of the cost. Clair Bros. wouldn't even have two days to set up. Going with a digital delay would make everything more complex. The more pieces involved, the more that could go wrong, plus it would impair visibility. Again, it would mean more time and money, to which no band would agree.

This was the first time a Clair Bros. system would be used in the Kingdome, so Roy flew back from his home base in Lititz, Pa. to supervise the actual show. Having been in practically every hall in the U.S., Roy feels no need for advance work in general, with the exception of situations such as this. His findings were that, obviously, it is a reverberant hall with a long decay rate at all frequencies, but that several other halls regularly frequented by a Clair Bros. system have equally as long a decay rate. Roy remarked several times over the two-day excursion that the hall was not nearly as bad as stories had led you to believe.

Since Frampton's tour included several huge outdoor concerts nothing was added equipment-wise to the sound system used in the Kingdome, for after two years of doing outdoor shows, Clair Brothers has determined that there is no optimum solution to any one problem since every hall is different. What, then, would be the concept of proper sound distribution? "The most important thing [to remember] when you have a reverberant hall is that you can't kill the reverberance," Roy began. "The more people you have, the better it is, but at no time can you expect to totally kill the reverb. So, we have to point a speaker at everyone in the hall, thus enabling them to hear direct sound. Reverberance won't bother them if they hear direct [sound] over the reverberant sound. But, if they are in an area where there is no direct sound, they're going to hear the reverb and it's not going to be clear." The conclusion is that the most important thing a sound company can do to rectify the situation is to place speakers high enough and wide enough to cover all the seats.

The Building Begins

At 5 a.m. Sunday morning, the day before the show, the stage hands begin constructing what eight hours later will be a stage. Before that nears completion Clair Bros. Sound Company goes to work erecting their own scaffolding for the speaker system. The placement of speakers is a problem common to all sound companies. In the past, Clair Bros. relied upon the promoter to supply the scaffolding. This led to problems because American scaffolding has centers of five and seven feet, while a Clair Bros. speaker system has eight and four foot centers. Since you can't put eight feet into seven, the scaffolding never worked. There were cross braces everywhere, eliminating the ability to stack speakers in order to couple them, which is very important. As a result, the scaffolding forced you to place the speakers at random, and prevented the

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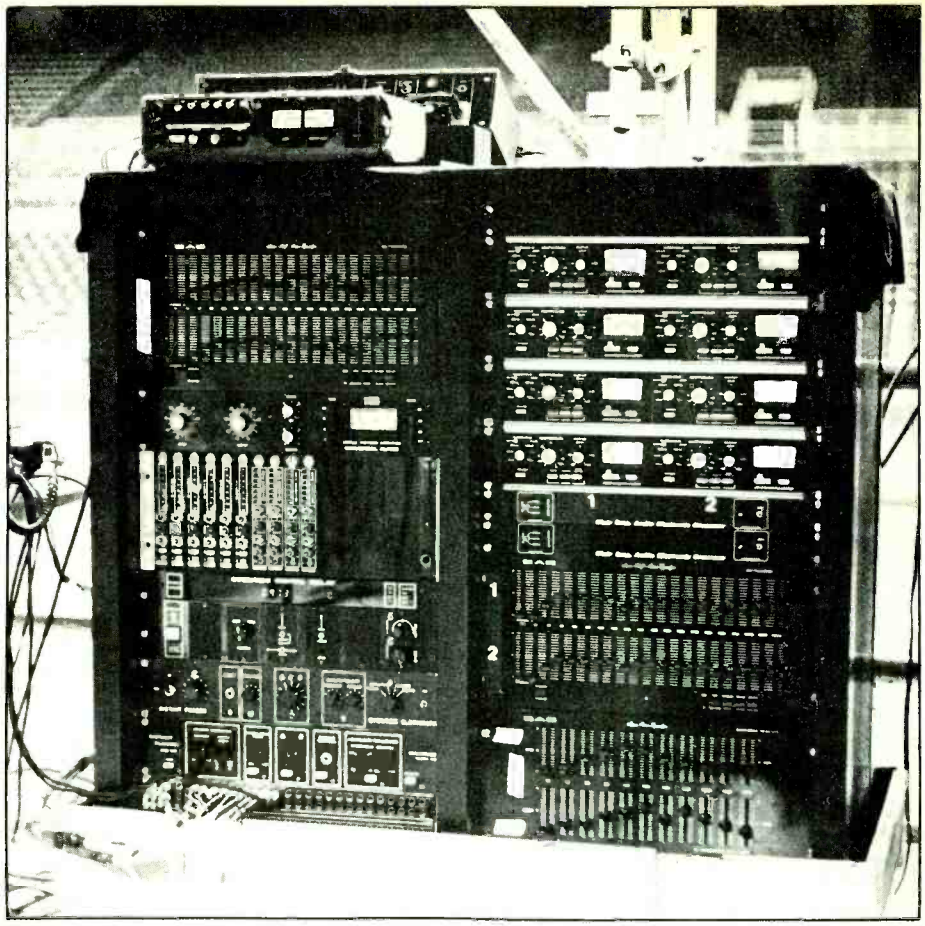
CIRCLE 81 ON READER SERVICE CARD

system from covering all the areas. Further, the bulk of the system required twenty-four hours to mount for an outdoor show, which left you with one tired crew. To alleviate the problem, Clair Bros. decided to go with English scaffolding, based on the metric system, so that the centers are 8.2 feet. There are franchises in the larger American cities so that this scaffolding length is now readily available. Time-wise, this enables a speaker system for an outdoor show to be set up in four hours, which pleases Clair, for it takes most people that long to set up an indoor show. By incorporating elevators, lifts and hoists, the 400-pound cabinets are lifted and placed three at a time. This has eliminated the fork lift problem, for prior to this innovation, the height of the tiers was limited to the reach of the fork lift. Also, the fork lifts couldn't handle three cabinets at a time.

On the average it takes eight hours to erect the forty-four foot, tri-tiered scaffolding, plus the four mentioned to add the speakers. Today, the scaffolding went up in only five, a fact which is going to cost Roy Clair extra money to his crew chiefs, for added incentives taking the form of bets add a slight competitive edge to the proceedings. He doesn't mind losing under the circumstances, and with pride he is the first to admit that without the high caliber of personnel both back at the warehouses and his crews on the road, Clair Bros. would not produce the quality product and earn the respect that it does. Everything was ready to go in nine hours time, including breaks. All in all, there are eight people from Clair Bros. working this show: four for the overall Frampton tour; two for the scaffolding; one for the outdoor system; and, of course, Roy.

To overcome the long decay rate and reverb problem, Clair Bros. solution is to arrange the speakers in a semi-circular arrangement in an attempt to point a speaker at everyone in the hall. To achieve this, the forty-four foot tri-tiered scaffolding concept is the result of what experience has taught them—it's too difficult to try and point speakers up from twenty feet to hit the top seats, so what they basically have is a tier for each level of seating. Normally the Clair Bros'. S-4, four-way bass-reflex system would be sufficient to do a hall like the Kingdome, but when you're an active sound company with up to eight shows on the road at once, you occasionally can find yourself short on gear. When Frampton played J.F.K. Stadium in Philadelphia, a 100,000 seat facility, Clair Bros. was able to utilize eighty S-4s because Philadelphia is close to their home base, and because two clients who normally use the gear were off the road at the time. Where the Kingdome could benefit from an optimum eighty S-4s, they will only have 60. "But," as Roy says with a grin, "we used to do outdoor shows with forty." It should be stressed that although power-wise there is 600 KVA available from the house, it's not a power factor for the hall that requires the vast number of sound devices, but Clair Bros'. theory of pointing a speaker at everyone in the hall. As a result, a second system, the four-way indoor horn system which is normally a hung system, was implemented.



B&W photos by Gil Podolinsky

Cabinets containing much of the signal-processing equipment. (Note the steel construction of the cabinets.)

The S-4 system was used because of the minimal time needed to mount the system, an important factor when touring. It is comprised of speakers and electronics, with the impedance varying due to each speaker configuration within the cabinet. The uniqueness of the cabinet is the electronic compensation. It has a high Q factor on the top end, which is most important when playing a reverberant hall for the result in sound is similar to pointing a speaker at everyone. So, while the rest of the system operates on a bass-reflex design and performance, the high end is not due to the directivity factor. The throw of the S-4 is straight out covering approximately 140°.

The horn system is comprised of eight, seven-foot W boxes (compared with the 7'6" W boxes used for Yes), which are placed to cover the seats that are questionable. Since the show is not sold out and seating is open, it's difficult to know whether people will be sitting, say, parallel to the stage. Therefore, the seats are covered, but with a less expensive system. The horn system is less expensive than the S-4 in comparison because it utilizes fewer speakers. Although both systems are equally efficient, the horn system is much bulkier and cumbersome and takes longer to set up, so Clair Bros. tries to limit its use on outdoor shows.

The actual horns themselves, not to be confused with speakers placed in a horn designed enclosure, are built into boxes, three to a box, which is a direct

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result of eliminating the dangerous practice of using 2'x4"s splayed to give the proper dispersion. Through this design each box is able to cover approximately 120° vertically and 90° horizontally. Each box has three different impedances due to the fact that two of the three horns utilize double drivers, paralleled down to 4 ohms, while the bottom horn has a single driver of 16 ohms. Overall, thirty-six drivers are used in the horn clusters, eighteen a side, pushing six JBL 075s, only two of which are used in conjunction with a Piezo cluster, and are placed three to a side.

Since two different systems are used simultaneously, different power levels must also be used. You don't want as much volume on the sides or on the horn system, as you do up front with the S-4 since the distance is greater in front. To be able to control two systems simultaneously, different crossovers, volume and gain must be employed from the board. To safeguard from someone making a mistake and blowing up the system, which happens occasionally, one central dbx system comprised of sixteen compressor/limiters is used, placed adjacent to the console. When you realize that 800 devices (speakers, horns, etc.) capable of delivering 100 watts each, or 80,000 watts, are being driven by 84,000 amp watts, you can visualize the potential danger. Clair Bros. carries two brands of amps—the new SAE 2600 and the Phase Linear 700B, which present no compatibility problems. The amps are placed behind the speakers on each tier to eliminate the problem of running cable up to the speakers from the floor. Three amps per tier cover the S-4 system, two per for the horn system. Outdoor shows present a different problem than indoors in terms of cable, requiring a special, very costly 300-foot cable running from the system to the board.

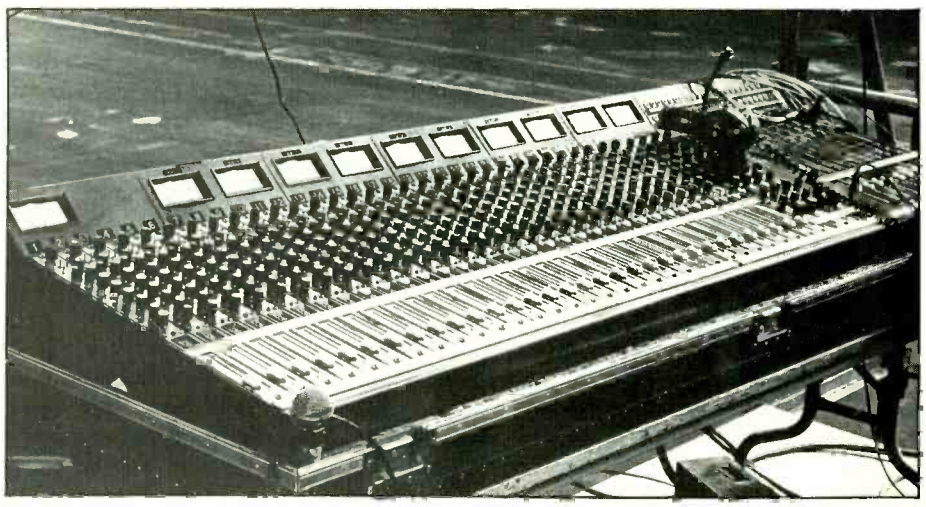
Is the S-4 system cleaner when used with one specific type of amp? "No," says Roy, "each amp has its own specifications. I still believe that if you take two good amps and a person with good ears, and put them in a room, he's not going to be able to tell the difference. That's debatable, but in a room like this (Kingdome), there's no way anyone's going to be able to tell the difference. In a real dead recording environment one *may* be able to hear subtleties, but in any of the halls we work in the U.S., *no one* can hear the difference (emphasis his). There's too much coloration, and we have to deal with it."

Reliability Factor

Aside from the necessity of getting maximum wattage per square inch of cabinet, the main factor is reliability. Hardware is one of the most important things associated with any equipment. After years of sound reinforcement work, Roy feels that anyone can make an amp, or a speaker, but to put them all together to tour day after day is possibly one of the most difficult things. "You may be able to get a speaker system to sound great in a local hall or in your garage, but to 'troup' it . . . the next night the system might not work because the speakers might have been jarred out of position, rubbing the coils as a result. There are about 3,000 things to consider when trouping a system and that's why not everyone can do it; *because* they haven't had the experience to put it together. I must say that packaging is a real big part of any organization. Consistency in packaging—putting your equipment in cabinets and mounting them correctly so they don't fall apart—is as important a consideration as sound itself."

A simple illustration is the manufacturer. An amplifier company doesn't necessarily build their product to troup, and they'll be the first to admit it. The metal casings are not thick enough for the wear and tear of road demands. In most instances, for example, there's a transformer sitting off of the back of the amp which, if left alone over a period of a few weeks, will tear the metal away from the front due to stress caused by weight. To remedy this, Clair Bros. has implemented a steel rack that supports each amp, rendering them immovable. Other pieces of equipment suffer from the same problem of unintentional fragility and are given the same encased treatment. Did Clair Bros. discover this through trial and error? "No, through common sense, but most people are in too big a hurry to consider it as a factor. Their attitude is, 'Well, let's screw it in a cabinet and get it working and worry about the rest later.' That's not the right way. If you secure it to begin with, then you have a two-fold thing going for you—it's going to work, and it's going to work the next day."

The associated equipment in a system usually depends upon the particular group's needs, but for the most part all of them are using Harmonizers, digital delay lines, phasers and flangers, echo units, etc., or, as Roy simplifies it: "All touring groups are using the same gimmicks." In terms of equipment, Clair Bros. feels that the most important are the instruments used to check out the acoustics of a hall. While it is possible to point a speaker at everyone, if that speaker isn't full-frequency, then obviously that speaker will have a peak that will be annoying and cause a problem as far as the acoustics are concerned. Add to that the fact that the hall, any hall, has a peak of its own, and instruments like the White Acoustic Analyzer become invaluable. It is used to flatten out the response of the speaker system so that there are no peaks in the hall. If the system is flat but the hall has peaks, then you cut that frequency down where the hall peaks. Besides the Acoustic Analyzer, which is employed



The English-built Midas console (30 inputs).

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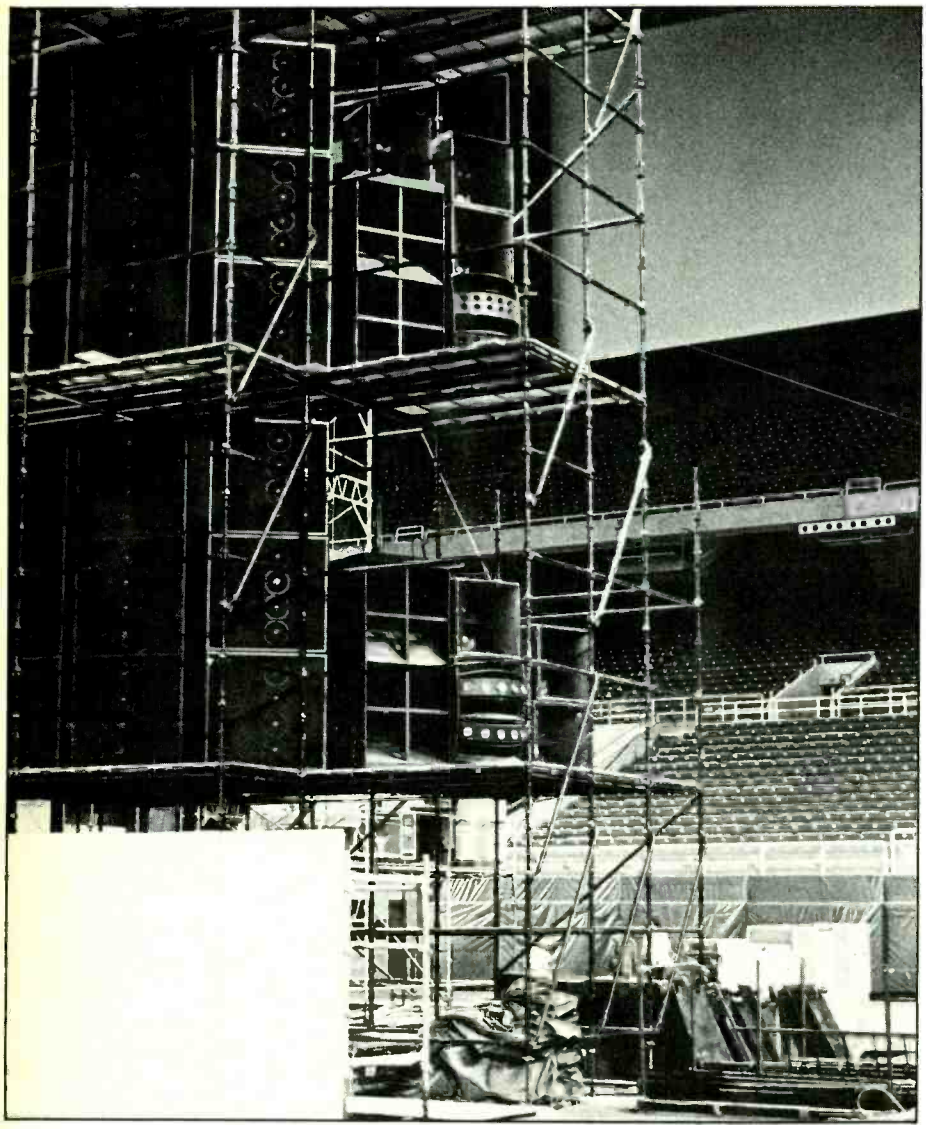
throughout the performance, other equipment used in conjunction with the thirty input English Midas board are: sixteen Clair Bros. crossovers, which are "very rugged and too expensive to market"; twelve SAE 2700Bs, eight of which are used for the monitors; the Eventide phaser, digital delay and Instant Flanger; and twelve dbx 160 and 162s. The 162 is the updated version of the 160. It is a stereo compressor/limiter where one master button controls all in a series; the 160 has a stereo configuration, but each one must be set manually.



Speaker stacks featuring some of the sixty S-4s used and the famous W box.

The board, lighting desk and related equipment are placed on about a twelve-foot scaffolding located 100 yards directly in front of the right side of the PA. While most sound companies have a trick of putting the board at the best spot in the hall so that the mixer is pleased at the expense of others, Clair Bros. has gotten away from that and places the board where the mixer hears approximately what the other people in the hall hear, enabling him to compensate if he's so able. "We don't care where he is," Roy remarks, "unless it's under a balcony or something, but it's not that highly critical. Granted, the better he can hear the better the mix will be, but he may not compensate EQ-wise for the other areas. If you put the board, then, in an area where the sound is in between, rather than optimum, it will sometimes work out better."

Look for the Sound

This begs the question as to how much configuration adjustment is possible during the show. Clair Bros. feels that adjustments can be made to handle holes in the hall during the concert since it's predetermined through the use of the acoustic analyzer where those holes will be. Since the show is *not* sold out, the system will be in operation to cover the top deck even if there are only three people sitting there. Again, Clair Bros. doesn't feel that it should affect the overall sound unless there isn't direct sound pointed at everyone else. *Plus,* with open seating, people can move if the sound is bad. The theory is basic: If you look and see where the speakers are pointing, you know there'll be sound, and therefore it would be foolish to go where there'll be no sound, like behind the stage. In each hall there are optimum places where the sound will be good. Generally speaking, if you have one tier, the sound is in the back, off the floor, middle of the back section; from the second tier you're in the same area in the hall, one deck up. The third is the same, so that essentially it's all covered through this semicircular dispersion pattern.

"The architect acknowledged that there *could* be a problem on the back floor. We tried to compensate for it, but you can't eliminate the fact that bass couples with the floor, making it boomier than in the other sections. If you add horns, it gets too shrill, so it's a very touchy situation. However, everyone should be able to hear Frampton, voice and music, tonight."

As to what exactly Peter will be hearing will be coming through a monitor with a bi-amped woofer and horn configuration. The monitor board itself has 24-in, 8-out, so that there can be up to eight separate mixes on stage. This board, which is always operated by Clair Bros. personnel, was designed and specially built for Clair Bros. by Midas, although now that it's been seen across the U.S., there are other people having Midas build them the same board. The latest of whom is the group Heart, who has copied the Clair Bros'. system practically to a T.

The system used for Frampton's tour differs very little from one sold to a Canadian company—Audio Analysts of Montreal—for use as the stadium house system at the Montreal Summer Olympics in 1976. That same system is now touring exclusively with Emerson, Lake and Palmer; it is *not* merely "a system based on the Clair Bros. design," as was incorrectly stated in a recent issue of Musicians News Magazine. "The cost to rent such a system, says Roy Clair, "is expensive, what with three semis: one for scaffolding,

The Luxman 5T50

The Laboratory Reference Series tuner.

A very remarkable component—by itself or as part of a complete LRS system.

Although the LUX Laboratory Reference Series was conceived—and introduced—as a completely new concept in systems, we believe you'll find the LRS tuner alone merits special consideration.

At a glance, the 5T50 is strikingly clean and elegant. The tuning display itself is digital, supplemented by what might be considered an advanced touch of nostalgia for the dial of old. One linear dial is calibrated in 1 MHz increments, and another provides for 200 kHz indication. However, instead of a pointer, LED's traverse the dials, giving linear readout of tuned stations.

Electronic tuning and memory.

Tuning is accomplished electronically, by a touch on either of two buttons. Station selection can be switched to either manual or automatic scanning mode, and tuning speed can be adjusted from slow to rapid. A further tuning option is provided through the tuner's C-MOS IC memory system, which stores seven stations in digital code. A touch of the appropriate button provides instant reception, with the station exactly center-tuned.

Tuning accuracy within 0.003 per cent.

Tuning is always exact, no matter how you select the FM station. Using a quartz crystal frequency synthesizer instead of a tuning capacitor and coil assembly, the 5T50 gives a degree of accuracy (within 0.003 per cent) and stability that cannot be found in conventional tuners. The station is received center-tuned and it remains that way—just as accurate and as stable as the FM transmitter frequency itself—which, for reasons of precision, is also referenced to a crystal.

But the utmost in design and performance doesn't stop here. From front end to output terminals, the 5T50 utilizes the best circuitry. For example, dual-gate MOSFET's in the RF amp and mixer. And a buffer circuit—which helps achieve outstanding image, IF and spurious response rejection—follows the crystal oscillator. In the IF section, a ceramic filter plus two pairs of 4-pole block filters provide for excellent selectivity without sacrificing low distortion characteristics. A double-tuned quadrature detector also keeps distortion low, at the same time protecting against signal overload. And to assure excellent stereo separation (45 dB at 1kHz and at least 40 dB at high and low frequencies), the multiplex section employs a Phase-Locked-Loop circuit.

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To help you take advantage of this advanced internal design—which includes a Dolby* decoding circuit for Dolbyized FM broadcasts—we've included a number of advanced external features. So tuner operation is just as precise as performance. For example, a multipath check button lets you detect multipath distortion audibly—no need for an oscilloscope. There's also an antenna attenuator for adjustment if signals are too strong in your reception area. For quick level adjustment when recording Dolbyized broadcasts, a 400-Hz test tone button is provided. The tuner output level is adjustable, along with interstation muting threshold. A digital display provides easy-to-read signal strength indication. And if incoming signals are weak, there's a high blend noise filter to assure low-noise stereo FM.

Other LRS components.

Of course, only you know if you're considering a new tuner at this time, or an entirely new system. If the latter, we'll simply advise you that the LRS system has carried the separates concept to "an extreme." That is, the preamplifiers have no tone controls—these are provided by the LRS graphic equalizer or separate tone control unit. The power amplifiers are available with or without meters—supplemented by a separate LED peak indicator. And if low distortion is important to you, the total harmonic and IM distortion of the LRS power amplifiers at rated power is no more than 0.008 per cent. That's double-0 eight.

However, if a superb tuner is really all that interests you at this time, that's fine. Chances are you'll see and hear it as part of the complete LRS system—where it will look and sound the most impressive. Suggested price: $1,495.

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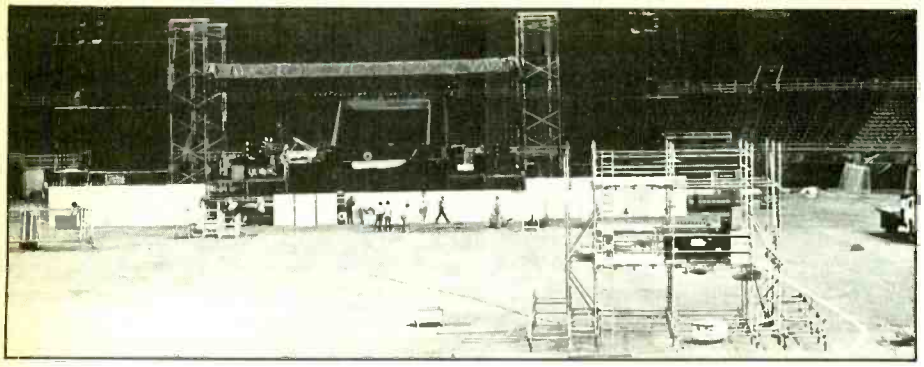

View inside Kingdome taken from second deck.

one for outdoor and one for the indoor system." According to People Magazine, ELP's system is worth a cool $1,000,000.

After taking numerous photos of the lay-out the morning of the show, I left and returned in the early evening for the actual test. Not knowing that the show had been moved up from 7:30 to 7 p.m., I arrived in time to completely miss Blondie Chaplin, the opening act. Outside, there were still several thousand people milling around in the sunshine; inside, rock tapes were running through the system in an attempt to passify the crowd during set changes.

As I made my way (eventually) to the board, I wandered through various parts of the hall amid firecrackers and occasional victims of alcohol being towed away by friends to a convenient pick-up point. When I caught Roy's attention, he beckoned me up the ladder. He was quite relaxed, not harried as when we had talked that morning, for at present, his job was done. Everything was running fine and he just sat back and watched, turning the board over to a succession of four soundmen.

Sound & Fireworks

Amid more firecrackers, intercom calls and the hall continuing to fill up to the eventual 45,000 or so, 20,000 shy of capacity, the second act, The Geils Band, prepared to get under way. The first few tunes sounded muddy, with the vocals somewhat garbled. As the band got more into the set, the sound became cleaner, although there were occasional traces of echo at the end of a song, but that must be attributed to insufficient crowd numbers at this point. The crowd response was enthusiastic throughout, which resulted in Geils being brought back for an encore.

During the break, I had an opportunity to speak with Dave Thoener, who had just mixed the set. Having engineered twenty-three albums at the Record Plant, New York, including all of J. Geils' albums, and not scheduled to do another project until fall, he was curious what the other side to engineering—the "live" aspect—was like. Having never mixed "live," Dave had already had enough of the experience less than half-way into the tour. It did serve the two-fold purpose of answering his curiosity as to what happened once the album was finished and the band left the studio, as well as giving him an insight as to what happened "live," giving him ideas on how to improve "live" studio sound. Unlike some sound engineers you'll meet on the road, Dave was quick to mention to the Clair Bros. personnel that if they heard or didn't hear something, to please help him out. Putting the considerations of the band and the audience above personal ego is truly the sign of a professional sound engineer.

Next up was Foghat, doing a nice hard rock, guitar duo set that took up the flow of energy where Geils left off. The hall now contained the 45,000 expected and Roy spent most of the set walking around the Kingdome checking for holes in the sound. The set seemed to go without a hitch, with the system projecting a clean, loud sound. I was specifically curious as to the back of the hall, both from what had been written as well as the natural tendency for bass to couple with the floor. At the set change Roy and I compared notes. To my ear, allowing for the size differences of a domed stadium and a coliseum, the rear floor, first and second decks sounded no worse than a 17,000 seat coliseum not filled to capacity, and obviously the periphery outside the seating arena always is boomy in any building. Roy was quick to agree, stating again that the hall is nowhere nearly as bad as first reports made it appear. He made no adjustments to the equipment nor recommendations to the engineers, so the system must have sounded good in the other parts of the hall as well.

As we awaited the headliner, we discussed various acts that use Clair Bros., how the expectations and demands of sound companies have altered greatly in a relatively short time, as well as how some groups that sound great on record end up sounding lame in performance due to different approaches—i.e., mics, and so on. It's become the norm today to hear or read about a sound company crediting Clair Bros. with a design or an innovation, but achieving parental status in the sound industry does have its drawbacks, like competitors copying, though unsuccessfully, the S-4 system, or stealing away clients by offering to do the tour at cost.

Suddenly the lights go out and there is a simultaneous eruption of shrieks, match-light and fireworks. Frampton walks out and spends the next five minutes running from one side of the stage to the other waving, jumping and generally working the crowd up even further. As the roar dies down Peter picks up an acoustic guitar, sits on a stool and begins his one and a half hour, twenty-song set, most of which is taken from his *Frampton Comes Alive* and *I'm In You* albums. The video screen which has been in use all night is perfectly synced with the audio, as well as with Frampton's stage movements. The sound is overpoweringly clear. Being 100 yards away from the right bank of the system, we're directly in the path of 115 dBs. Without a sign of hum, feedback or poorly mixed instruments, the show eventually came to an end.

No one was more pleased with the success of the show than Roy Clair, who was by now joking and accepting compliments for the outcome of the sound, again crediting his staff for the success. A few minutes of joviality, drinks and last good-byes and they'd be back dismantling their two day effort. As I walked towards the exit amid the debris that signified the presence past of 45,000 Framptonites, I couldn't help but be amazed at the fact that the tonnage of equipment, the towering scaffolding and stage, and the mountains of garbage yet to be made, would all give way in less than two days time to a soccer match, and in another city the entire process would be repeated.

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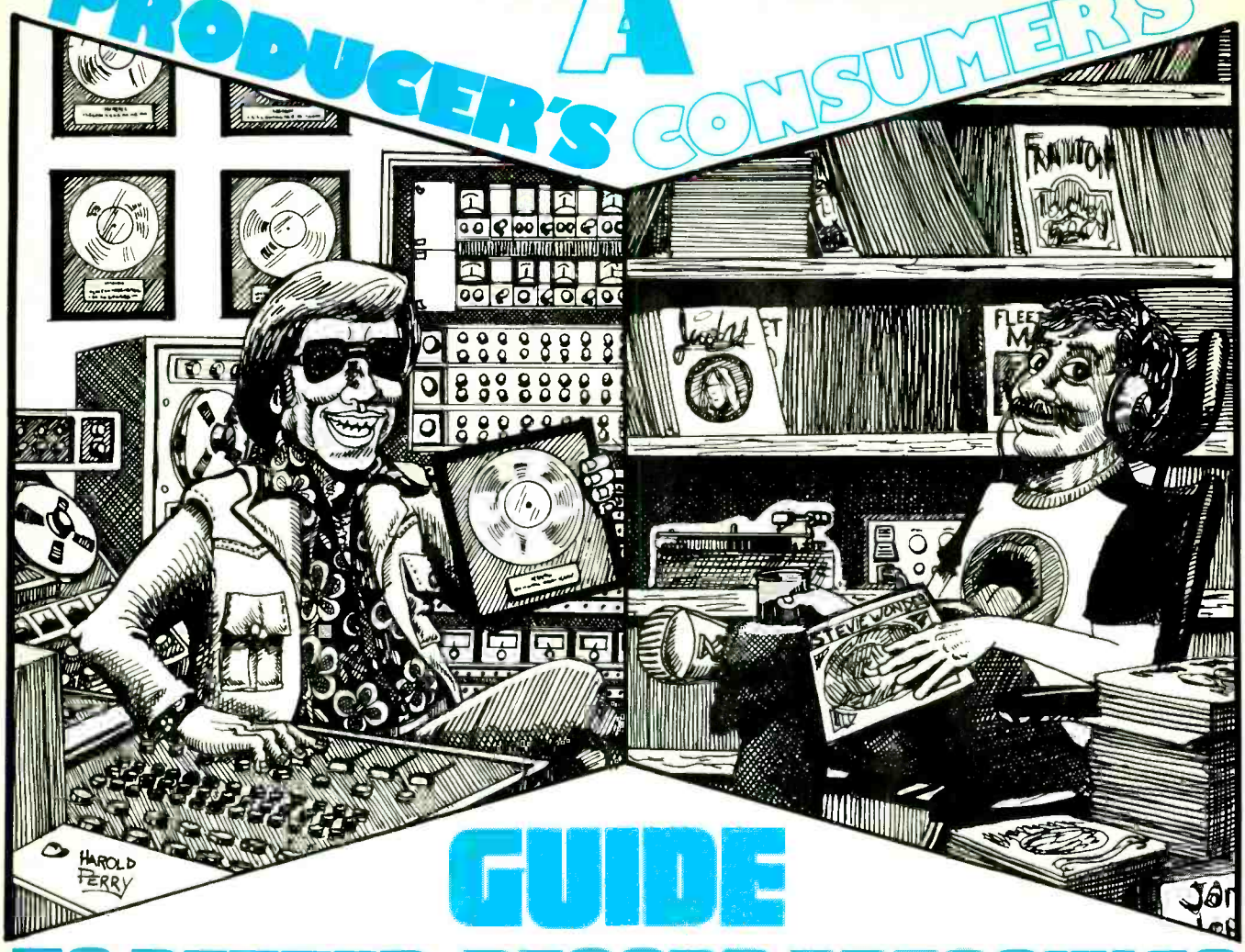
B·I·C VENTURI SPEAKER SYSTEMS

TOMORROW'S TECHNOLOGY TODAY

A PRODUCER'S CONSUMER'S GUIDE TO BETTER RECORD PRESSINGS



Although in previous articles for *MR* I have made somewhat hostile and unsavory remarks—out of the corner of my typewriter—aimed at the pressing plants of this country, it *is* entirely possible, in all fairness, to have fairly good records made. You must be equipped with two things though. First, a *little* more money to spend on pressings (but not that much); and second, a certain amount of knowledge about pressings in general. These two things will help you to get close to the best possible pressings obtainable in this country.

Sorry, I can't help you with the first part (why do you think I'm sitting at this typewriter writing a magazine article?),but there is a good deal of information to be imparted. Getting a good product from a manufacturer (records or anything else) requires common sense more than anything.

As a record producer you have a rather large stake in the balance . . . which will determine whether you eat steak or beans for the next few years. As a consumer you only have a few bucks and a few thousand hours of listening to surface noise in the balance.

Before we go out to do battle with the various plant managers, suppose we reconnoiter the situation. Let our reconnaissance begin with the mastering facility. Here is where the process for manufacturing the actual record begins. Your tape is transferred from the magnetic tape format to the mechanical disc format. This is the last point at which any change in sound level or equalization can be made. Distortion *can* be introduced at this stage, but that is nearly impossible using state-of-the-art equipment. For all practical purposes (assuming a good mastering facility is used), the distortion is caused by the record player and not the recording machine (if the tape itself is clean to begin with). The best way to find a reputable mastering plant is either by expensive trial and error or by asking a reputable recording studio for a recommendation. Finding that reputable recording studio is—as they say— beyond the scope of this article.

Anyway, the master when cut is shipped to a plating plant. When the plating plant receives the master lacquers they inspect them again for surface imperfections. (It is possible for a harried mastering engineer swamped by hordes of producers demanding instant service—every order is a rush, by the way, every order was needed two days before the tape was provided—to miss a small blister once in a while.)

Reasons for Rushing

Let me go off on a tangent here a minute. The best way I know of to get a lousy product is to demand that your job be done right away. You are no dif-

ferent from anyone else; *everybody* is behind schedule. Believe me, it will make no difference if your record is presented to the public thirty-three days from now or thirty-four days from now, unless you have airplay already. If you get airplay before you have a product to sell that's *your* problem, not the record industry's. There is no such thing as an emergency as far as entertainment goes. No one will die if they have to wait a day or two longer to hear a record, assuming they want to hear it in the first place. If you try to force your record through the system in faster than record time (forgive me for that one) you will run a very high risk of losing the record peoples' respect and you will force them to disregard the care that normally goes into processing your important product.

It goes like this—either you be patient and let the guy do his thing with all the care and expertise he can and will give it, and come out with the best product he is capable of, or make a nuisance of yourself and he will have no other choice but to give you less than his best. As a matter of fact, I just returned from a vacation to find two problems. Two different clients just could not wait for me to return so they took their business elsewhere. I can reasonably assume that they demanded instant service from my competitor, and what happened? One guy actually had the nerve to call *me* and ask why his record got screwed up at the place down the street! Not only that, but I think he was mad that I had the nerve to go on a vacation in the first place. Now I have both jobs to do anyway. So where did all that rushing get those two guys? Well, it made them poorer by some five thousand records and instead of getting records a week later than they wanted them, they are going to get them three weeks later than they *needed* them.

In my book there is only one valid reason for a rush. That is if somewhere in the process a part gets damaged *after* the record promotion machine has been started up. In other words, if the mastering place tells you you have a master on its way and you start promoting, should the gremlins get your parts, then and only then should the mastering engineer be hasty in pushing your job through ahead of someone else's and re-cutting it as quickly as possible. (This goes for all the processes in the chain.) Besides, you wouldn't want someone to jump line on you, would you? Now that I got that off my turntable. . . .

A Fine Process

The plating plant has this master lacquer which they inspect as soon as they receive it. It is possible not only to miss a blemish, but sometimes there are blisters or air bubbles below the surface which even after cutting are not exposed but which will erupt as a result of sitting in the sun in a delivery truck. If such an imperfection is found, the plant will ask the mastering house to replace the part. This is done at no cost to the client and usually without his knowledge. The plant will take the good part and clean it. It is put in something akin to a dishwasher so that all the grease, oil and gunk from our polluted air and greasy fingers are removed. Then it is placed on a machine and sprayed with a silver paint because in order to be plated the part must be made metallic. The cellulose nitrate is non-metallic and by itself cannot be plated. Thus the metal coating.

This is a fine process. The silver coating is only two or three *molecules* thick. After this is done, the metalized part is prepared for plating. The electrodes are attached, the part is fixed to the holing device and the whole thing is immersed in the plating tank. In the tank are nickel pellets which will be deposited onto the master lacquer in the process. Again, this phase has its hazards. The electrolytic solution must be kept at a certain ideal temperature and the proper rate of current must be maintained in order for the nickel to be properly plated onto the master. Sometimes the lacquer falls off its mount and drops to the bottom of the tank. This is one of the many ways a plating plant can lose a master.

If the lacquer has gotten this far safely, the next step is probably the most dangerous. This is where the nickel plate is separated from the master lacquer. A well-trained and experienced worker has this exacting job. It is something like opening a clam that was planning on being an introvert. An experienced man can perform this task in about thirty to sixty seconds. This is where most of the accidents happen. And often a plant will call up for another replacement with a terse, "We blew it."

When the part is finally separated it is dried, the center hole is punched out and the edge is crimped to specification. This depends on which pressing plant the part is going to, because each plant has different size center hole specs (the center hole of a stamper may be two inches in diameter, depending on the plant). So the plating plant has to know where the records will be pressed before it can punch the center hole and crimp the edge.

A Mother

So far we have spoken only of a simple "strike off" of the one step process. This is for a limited run where less than a thousand records at the very most will ever be pressed. There is also a two-step process and a three-step process. Let's back up to the part where the stamper is separated from the master lacquer. This is the first step—"strike off," or "metal master." It can be used as a stamper as we described, or it can be used to make a "mother." In the latter case the metal part generated from the master is not punched and crimped, but returned to the plating bath again to be plated itself. When these two parts are separated you have the mother which is a metallic replica of the master lacquer in that it is a positive part—i.e., there are grooves instead of ridges as in the stamper or metal master. *Then* this mother is plated to generate more stampers which are then used to press the records. The reason for these different steps is as follows.

A stamper is only good for so many pressings before it wears out. If you call a number of pressing plants and ask how long a stamper lasts you will get answers ranging from five hundred to five thousand pressings on one stamper, depending on the plant. I suppose the answer is a function of the plastic compound the particular plants use for the records. A mother is only good for generating more stampers and to give an intermediate check on quality. Unlike a master lacquer it can be played without damage. So actually we can get up to thirty-six stampers from one master lacquer. Here's how.

Most plants agree that you can generate only one metal master from the master lacquer, although there are a few who will say they can get two or three platings from a master. But each metal part can generate six more parts. So you can get six mothers from the metal master and six stampers each from the six mothers. Each generation will become a bit noisier as with anything else. Also, it takes

about three or four hours to plate each part, and thus it is at least a two-day project to go through a three-step process. Some of the major companies will order a half-dozen sets of master lacquers cut at one time in order to get six one-step stampers (thus saving the time and the quality, but at about five times the cost).

Anyway, we have gotten as far as the stamper. Incidentally, all of this is done in duplicate; each side is treated separately. Two masters are cut—one for each side—and two stampers for each record are made. This is because a record is actually a plastic mold just like a toy. It is made by squeezing a bunch of plastic between two complementary halves of a mold, and each side of the record is one half of a mold. By the way, the two-step process is where you go only as far as the mother and use the strike-off as the stamper. This is done in a situation where you only plan on pressing a few copies of a record but want to be ready in case the record becomes unexpectedly popular. The three-step process—or the cutting of several identical master lacquers —is used if you are sure you will be pressing many thousands of records.

On to the pressing plant. When the pressing plant gets the stamper they should, of course, inspect it to be sure it is all right (i.e., not damaged in shipment, etc.). A stamper is a very vulnerable thing. It has ridges that stick out and can easily be knocked off. If a metal mother is dropped the only damage would be cosmetic, since the grooves are naturally recessed and only the surface is scratched. Usually the record would play as quietly as any other. Not that anyone would use a scratched mother. But a stamper can be damaged and not show any signs of this damage until the first pressings are played. This is because you only have to damage the delicate and exposed groove bottom—which is now the peak of the ridge—by a ten-thousandth of an inch to ruin it, not difficult by any means to do. So it is very much possible for some clumsy lunkhead to drop a stamper on the floor and not see the damage, and rather than risk getting a reprimand from the boss simply not say anything. That is certainly inexcusable, but it happens.

After the stampers are mounted in the press it is time to begin making the records. The plastic is inserted by the operator and the press is put into operation. It is a hydraulic press and really squishes the plastic blob into every little microscopic part of the groove. Steam circulates throughout the press plates to aid the flow of plastic and then cool water circulates to make the record stiff enough to be handled. The press plates are made so that the area that the label is on is thicker than the groove area. The outer edge of LPs are also as thick as the label area. This is called a groove guard and prevents the groove area of one record from touching the groove area of the record stacked above it and below it. You see, the record is first trimmed as it comes out of the press so that the flash is removed and the record is 11-⅞ inches in diameter. The labels are inserted in the press before the plastic blob (called a biscuit). They are not glued on afterwards as many people think; they are actually embedded in the plastic.

The above explanation describes the functions of a manual press. Today many plants have automatic presses which perform all the above tasks automatically. It's really odd to see a plant full of machines doing this pressing by themselves like little robots, completely unattended. There is also another kind of press, and that is an injection mold type of press as opposed to the compression mold that we have described. The injection presses do not press, they stay closed and the plastic is forced in under great pressure. These also are automated, but are not in general use.

Plastic Formulas

The plastic—assuming (notice how many assumptions we have to make) the stampers are quiet—is the determining factor for noise. Different plants use different formulas for plastic. The good plants used to use pure vinyl made from polyvinylchloride, but then came the alleged oil shortage and the very real energy crisis, and since petrochemicals are an important part of PVC all kinds of other junk was and still is used to make records. Now noisy styrene compounds are being used in their stead. But normally vinyl is compounded with carbon black and antistatic stuff and several other things to make records black, quiet, removable from the press, etc. Records don't have to be black. They can be any color since PVC is clear. The reason the carbon black is used is to hide air bubbles and to prevent you from seeing clear through to the other side of the record, both of which for some odd reason tend to disturb people.

Some manufacturers also claim that the carbon makes the pressing quieter, others say it makes it noisier, and still a few others say it has a negligible effect. Take your pick. The only way to find out is to have a master record made with quiet grooves and then have a few hundred pressings made using regular vinyl. Then, using the same press and stamper clean out the press of all the compound with carbon black and start using vinyl without the carbon black. I don't know of anyone who has been able to get a pressing plant to go through that hassle and to go through that expense of mastering and plating at his own expense. I'm sure Columbia and/or RCA have done this themselves and probably know which is quieter, but I have no first- or even second-hand knowledge of it. But CBS and RCA both make black records.

Anyway, there are some plants which will use re-ground records. That is records that have been rejected and chopped up— labels and all—and melted down to be reused. This is usually the noisiest record you can find. Few plants will admit to this practice, but the fact is that too many do it. So it's worth paying a few cents more at a plant that is known for good quiet pressings. If you shop for the bargain basement price . . . you'll surely get records from a dungeon.

Now that we've had the 25¢ tour, suppose we get to the problems one is likely to encounter in pressing and buying records. If you have had a decent master lacquer cut, next you must find a pressing plant by whatever method you choose. A good way is to ask the reputable mastering house you used for a recommendation. Their reasons for pointing you in the direction of a good plant are obvious. They want you to get the best product available (to you), and if you are happy, they are happy since you will probably come back to them. Also, they don't want their own work mangled at some dump either. Of course, if you use a poor mastering facility, they obviously don't care about their own work so they certainly won't care about yours. Thus, they don't expect you to come back and will send you any place and maybe even to some pit where they might get a kickback.

Before we get into the heavy stuff, a word about package deals. These are offered by some mastering houses and

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some pressing plants. So if you came down with yesterday's rain and have had it up to here with just getting your record recorded on tape and figure it's either your record or your sanity—take the record, we cuckoos have lots of fun. Seriously (?), there is an alternative which will cost very little more if at all and you will get moderate-to-good quality. It's simple. You just provide the master tape, label information and a non-bounce type of check and the next thing you know *voila*! . . . hundreds of pressings. Of course, if someone else is doing it for you he won't be as painstaking as he would be with his own stuff, but still, it's a relatively good compromise.

Clicks and Pops

The problems that occur in pressing records are the same ones you would run into if you were to go out and buy a record in the store. You can get all kinds of clicks, pops, distortion and surface noise, plus warps, wows and skips. The most common problem by far is that of clicks and pops. The origin of such noise is varied and numerous. In fact, a perfectly quiet record can issue forth with a loud pop once in a while. This is the simple easy kind of pop made by static. It is due to the fact that a diamond spike is being dragged through a plastic groove under tons of pressure. Thus you've got to expect a little static to build up. And when it does, *POW*! . . . amplified by about a million times yet. To reduce or eliminate this manufacturers add a little antistatic ingredient to their compound, and audio buffs gunk up their records with antistatic fluid. You can make that decision for yourself, but if you use it do so sparingly. Static is a fact of life; and it's nobody's fault.

Now suppose the clicks and pops are not a cause of static and are due to something tangible in the groove. This can be either foreign matter which found its way into the groove, or, in rare cases, a recorded click or pop. To distinguish between static and a physical deformity, play the record. If the pop is always in the same place—no matter where you put the tone arm down—then it is a physical deformity and not static. It takes a certain amount of time for static to build up.

Now if it is a physical damage you must determine at what stage of the process it occurred so you know whom to complain to and to keep some unscrupulous plant manager from giving you the run around. You must have at least three copies of the record to do this, if you have only one stamper from which your records were made. Play one of the records and note carefully, *exactly* where the pop occurs. Now play the next record and see if the pop is in the exact same place. If it is not then your pressing is at fault. If it is in the same spot the defect is on the stamper. (Just to reduce the possibility of a coincidence play the third record.) The pop will be in that same spot on all three records if it is the stamper. If you did a one-step process you have no other choice but to go back to the mastering house and get another master. If you have a two- or three-step process you can ask to play the mother. If the mother has the pop, again it is back to the master maker. If the mother is clean just get another stamper made—free, of course. Just because the mother may have had the noise on it doesn't mean that the master lacquer was at fault. The metal master may have the defect, and from there everything else is for naught.

Take the record back to the mastering facility and have it checked under the microscope. This way the engineer can tell if it was a recorded defect or a defect in the later process. Needless to say, you have to trust him to be honest. But if he is reputable, he has a reputation to protect and will gladly replace the master if it was his fault.

One thing must be kept in mind when making these checks. You have to have *new* pressings. If junior has played it 4000 times on his record player (the one with the picture of Donald Duck on it) and then used it to practice for the international Frisbee championship, all bets are off. You should treat the records to be tested as if they were evidence in a murder—you wouldn't want your fingerprints on them, nor would you want to be accused of obscuring or damaging evidence. So keep it clean—immaculate. Handle records only by the edges. Try putting your four clean fingers on the label area and your clean thumb on the edge. This is a surprisingly good one hand grip and it keeps you from touching the groove area. Once you get it dirty or scratched, any complaints you make will be quickly and easily shot down. Also, ticks and pops can occur very easily and a few are to be tolerated if they are not too loud. But if you are going to go bananas over one small pop that can only be heard with earphones, you may as well sell your record player. On the other hand if the recording sounds as if it were made "live" in Viet Nam around 1971 you definitely have a legitimate complaint (unless it *was* recorded "live" in Viet Nam in 1971).

Suppose you have had two or three stampers made, then you have to carry the detective bit one magnitude further. You will need several records this time, three from each stamper. You will have to look for some common but distinguishing mark that is on one stamper but not on the others to sort out which records came from which stampers. To find these marks look around the label area and particularly at the matrix number (see Fig. 1). This shows the detail of a record and its matrix number. HR-4003-A means Hit Record #4003 side A. F/W stands for Frankford/Wayne (the mastering house). The figure ⏦ stands for me or the mastering engineer's initials and ϕ stands for Super Plater the plating plant.

Possibly the stampers were given distinguishing marks by the plating plant and that makes it easy. If not, you have to scrutinize every little mark or abrasion (if any) to find a pattern common to several but not all copies. This will distinguish the different stampers. If you had several masters made and did a one-step process on each, the master lacquers were probably given their own identifying marks such as 1, 2, 3 . . . A1, A2, A3 . . .1A, 1B, 1C . . . 1A, 2A, 3A . . . you get the idea.

Then you must listen to each and see if a pop or pattern of pops is on all the records (and stampers) or particular to only one of them. If it is on one stamper only, that one needs to be replaced. If it is on all of them, the mother is bad and you have to go back to the mastering house for a thorough evaluation.

Surface noise is generally a fault of the pressing so you need not look further. If it is a rare case and the noise was caused in some other step, you need that expert help again. A once around gravelly sound is something called "non-fill." This is where the plastic does not flow into every part of the groove. In extreme cases it may sound as if the record is very distorted. This is a result of the heating cycle not being long enough to warm up the plastic sufficiently. The flow of the plastic is inhibited.

A harsh but constant sound is probably due to cheaper compound or

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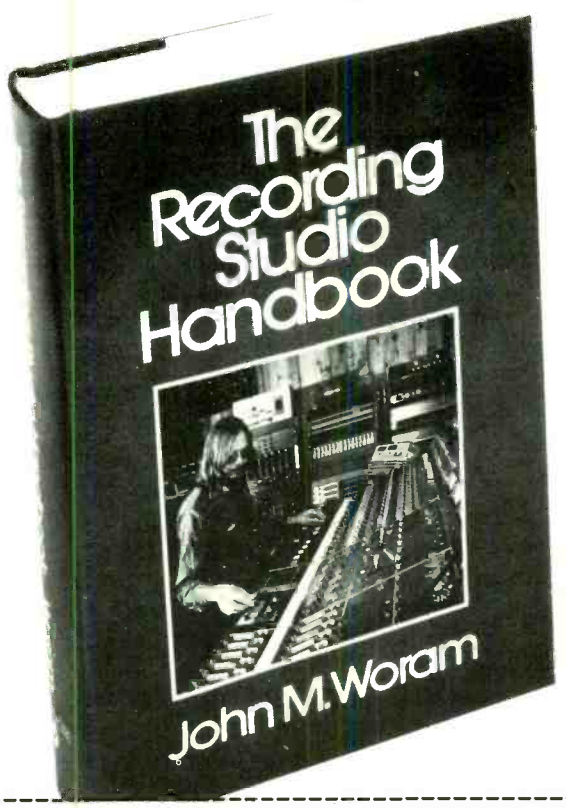

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Fig. 1

regrounds. This kind of noise can also come from a stamper that the gremlins got. Plating plants have a remedy for noisy stampers—they polish them. But if it is not done correctly, what you get is a very quiet record that sounds as if the treble control was turned all the way down. The delicate high end gets polished off too. This is a rare occurrence though.

Slow-Playing Noise

There is another way in which noise can be generated and that is by paying your mastering bill slowly . . . really. In the music business a studio owner is forked between courteous service to hyper music people and being hard nosed enough to collect bills in order to be able to pay the rent and the utilities. (Often when a first-time client is trusted and his record obviously is going to bomb he feels no need to pay any studio service bills incurred.) In mastering, a producer will make a panicky demand that his work be done quickly with an indignant promise he will return tomorrow for his product and at that time present payment. Sometimes the mastering house will cut the parts and take a trusting chance, then find that this guy does not come back or doesn't come back for a month. All the time his lacquers are sitting in storage and drying out. This kind of lacquer is difficult to plate and can produce a noisy stamper. It is the impatient slow-paying clients own fault. He gets exactly what he asks for. That's how you literally can get a noisy record by paying your bills slowly.

Wow and Warp

Next to concern us are the visible physical abnormalities. Everyone has experienced an off-center record. This causes a wow which is most annoying and apparent on sustains, legato, etc. An off-center record results because the stamper was put in the press off center or the stamper's center hole was carelessly punched. There are certain tolerances within which the center must fall. These tolerances are microscopic so if you can see your tone arm moving from side to side the record is off center. If you are *not* one of those poor souls cursed with perfect pitch and can hear wow, the record is off center. And you should complain, because the record is definitely out of spec.

The next problem—and this is definitely in the pressing—is warp. There is "dish" warp, "pinch" warp and plain old warp. Dish warp is where you can play one side of the record pretty well and you can use the other side to serve clam dip in. It is caused by the record coming out of the press too warm so that when it is stacked on the spindle the edges sag. Pinch warp is a result of a too short cooling cycle in the press. Consequently, the operator pinches the record trying to get it out. Other kinds of warp are caused by the record being too warm at the wrong time and can take any form, the most common of these is a ripple around the edge. If severe

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enough it will cause the tone arm to leap into the air.

Finding Fault

As a producer you will have to decide who is at fault when you encounter any of the problems discussed. If you are unable to make this determination you can go the pressing plant and be directed back through the chain all the way to the mastering facility. Or start at the mastering facility and be directed all the way to the pressing plant. Actually, going to the mastering facility is the quickest way to find the cause. But by all means do *not* go running into the place screaming like a maniac and making an ass of yourself. In the mastering house you may be given an embarrassing surprise. At the pressing plant your face may run straight into the fist of a burly 6'6" 250 lb. plant manager. Mistakes do happen even at the most conscientious of places and with the best people. Sometimes under the pressure of too little time and too much work, less care is a necessity. Sometimes the gremlins declare war and when that happens they *always* win and there's nothing man can do about it. Be polite and you will most likely be treated politely and fairly. If you act like a lunatic you will not be given any respect and no one will be willing to help you. Remember, don't demand immediate service when placing the initial order. By doing this you quickly brand yourself as a scatterbrained producer who doesn't have his act together and now is pleading for everyone else to bail him out. A picture that does not command much respect.

Now back to the poor consumer who has just blown a wad on his latest record purchase. A lot of people accidently (or deliberately) scratch a record and return it. The store knows you did it. When records were inserted in the jackets manually they were handled with cotton gloves. But most records are handled by machines. Machines don't drop records. They mangle them, but they don't drop them. So they all know you are a phony when you say, "It was scratched." The other problems, noise, warp, skipping, are legitimate. If you do take a record back, exchange it for another record by someone else and buy another copy by the same artist in some other store. The reason for this is that you will probably get a record that came from the same box, hence the same stamper and hence the same problem. By going to another store you lessen the odds of getting the same problem. It's as simple as that. If the record skips, sometimes it's better to just wait a while, the record people will undoubtedly discover this and have new ones out very soon.

That last problem is one that went unmentioned. Skip. This *is* usually a mastering fault unless caused by a very big chunk of plastic in the groove. If you're using a moderate to good record player no record should skip. If Mr. Duck's picture is on it, don't use it to make a point. A skip is also possible to miss in the mastering. With a half-mile to examine centimeter by centimeter that possibility is a definite reality that all mastering guys live in fear of. If he misses the skip and no reference dub is cut, it probably will be caught in QC at the pressing plant. But those things are not guaranteed. So once in a great while a record will skip, and when it does a lot of people make a lot of noise.

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Ambient Sound

BY LEN FELDMAN

Dolby FM—Is It Really Beneficial?

No reader of *Modern Recording* would argue against the merits of the Dolby B system as it is almost universally incorporated into any decent cassette tape deck these days. But, when it comes to the use of Dolby in FM and Stereo FM broadcasting, I have heard many otherwise knowledgeable audiophiles and professionals question the validity of Dr. Ray Dolby's approach to FM broadcasting. The argument runs something like this. True, Dolby B can add up to 10 dB of signal-to-noise ratio (at frequencies above 5 kHz, where most of the "hiss" is). In the case of cassette decks, where the best obtainable S/N even using good tape and a good machine hovers around the 55 dB mark (referenced to the 3% total harmonic distortion record level), pushing that figure up to 65 dB or so makes the difference between an acceptable recording and one that is too noisy to enjoy. However, latest state-of-the- art tuners and receivers boast signal-to-noise ratios (even in stereo) approaching or even exceeding the 70 dB mark! Furthermore, it doesn't take thousands of microvolts of antenna input signal to reach those wonderful quieting levels. Usually, ultimate quieting in an FM tuner is reached with less than 100 microvolts in mono, and perhaps a bit more signal strength in stereo. With stations popping up all over the country, there are very few listening locations that aren't close enough to FM stations to receive that much signal (providing, of course, a decent antenna is used). So, if we can attain a 70 dB or better S/N figure under most FM listening situations, why should we bother about Dolby which might theoretically "improve" the number to 80 dB.

The argument goes on even further to state that just about all program sources heard over FM these days are either disc recordings or tape recordings, neither of which has as good an inherent S/N as does the closed-loop system consisting of the station transmitter and your own super-tuner or receiver. Since, by now, everyone knows that Dolby cannot reduce noise (or improve S/N) that is already part of the program material itself (you'd be amazed how long it took audio journalists to get that message across to an eager audiophile public that was willing to believe anything if it meant quieter tape recordings), what good does it do to provide an FM system with additional signal-to-noise potential that can't really be used or appreciated.

The *Real* Reason For Dolby FM

The fact of the matter is that Dolby FM's greatest benefit, when it comes to FM broadcasting, is not its ability to further improve signal-to-noise ratio or to reduce received noise. The real benefit has to do with increased dynamic range capability of a station using Dolby FM. Here's why that is so.

When FM standards were first promulgated shortly after World War II there was, of course, no such thing as Dolby. In an effort to provide as quiet an FM service as possible, the FCC settled upon a process known as "pre-emphasis" and "de-emphasis." Broadcasters were instructed to *boost* high frequencies at the studios by a prescribed amount—known as 75 microsecond pre-emphasis. To give you some idea of what that means, any tones in music occurring at a frequency of 10 kHz are boosted by nearly 14 dB, while 15 kHz tones are boosted a bit more than 17 dB compared with a reference 400 Hz tone. At the receiving end, a de-emphasis circuit does just the converse, attenuating highs much as would a treble tone control if it is turned fully counterclockwise. Musical response, of course, comes out flat, since the de-emphasis restores previously pre-emphasized tones to their correct relative levels. But, in the process, background hiss and noise are reduced just as they would be if you turned down your treble control (but without losing musical fidelity).

So far it sounds just like Dolby, doesn't it? Except that the boosting and cutting are done to the same

degree, regardless of loudness level of the given musical moment. And that's where the problems start. In those early days of FM, recording technology was not nearly so advanced as it is today. (Remember, the LP was still a few years away!) So, most records and tapes had very little energy content up at the high frequencies. Today, studies show that while high-frequency energy of music is still much lower than mid-band energy, it is *not* 14 dB lower (at 10 kHz) or 17 dB lower (at 15 kHz) than mid-range energy content.

So now, if a station wants to broadcast in full fidelity, and still uses the 75 millisecond pre-emphasis formula, the studio engineer has two choices. He or she can either back off on average modulation levels, so that when those high-energy highs come along and get boosted they won't overmodulate the FM transmitter beyond the legal ±75 kHz permitted, or, the engineer can install all manner of compressors, peak limiters and other signal-processing (or signal squashing) devices that take the life out of program sources and make them all sound as monotonous as Muzak. Since the commercial realities of FM station operation prompt management to want to sound as loud (on average) as the competition, you can bet that most stations choose the latter course. If poor Major Armstrong could hear what has happened to his brainchild, he'd probably turn over in his grave.

Interestingly, in Europe and some other foreign countries where FM arrived a few years later, this problem was already recognized, and many countries chose a less extreme degree of pre-emphasis and de-emphasis just so that not quite as much compression and limiting would be needed to stay within modulation limits and still broadcast reasonably loud average levels. Serves us right, I suppose, for being "ahead of our time."

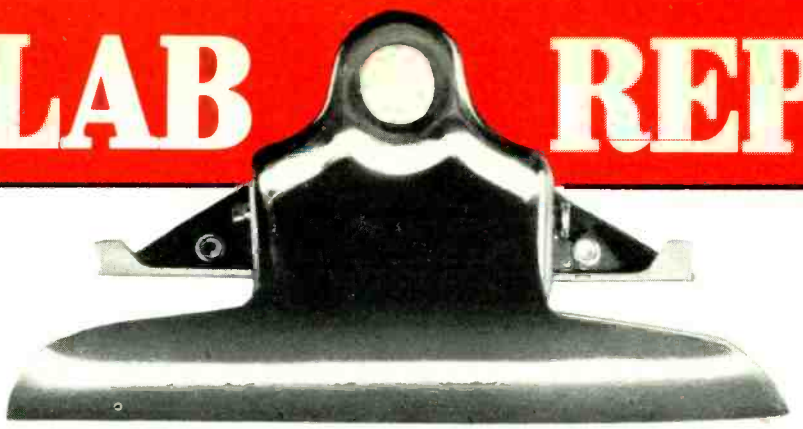
Well, along came Dr. Dolby, who said that if you use his system, you are going to gain noise reduction in a *dynamic* way (only the low-level highs are boosted during transmission and cut during reception, just as in the case of Dolby B applied to tape decks). So, since the noise problem is solved dynamically, why not go to a much lower amount of pre-emphasis and de-emphasis —say 25 microseconds, which means a boost of only around 5 dB at 10 kHz (and a similar amount of attenuation at the receiving end). That lower degree of pre-emphasis is consistent with the needs of modern program sources, whose high-frequency energy content will no longer "bump into" the modulation limits imposed by the FCC. Now, a station engineer should be able to by-pass those pesky limiters and compressors and let the music roll—naturally, as it was intended to be heard.

At the receiving end, of course, you've got to change some of the insides of your FM tuner or receiver. The de-emphasis circuits have to be modified to 25 microseconds, rolling off the highs by an appropriately lesser amount; that's even before you add a Dolby decoder. Many new tuners and receivers are being equipped with external switches which change the de-emphasis from the usual 75 microseconds (still needed when listening to non-Dolby FM broadcasts) to the new 25 microsecond value required when tuning in a Dolby FM program. Switchcraft offers a little outboard passive adaptor that makes the necessary conversion but also introduces a 10 dB insertion loss (usually not serious enough to require an additional amplifier stage between your tuner output and preamp input) in case your FM set is not equipped with the extra circuitry. Some tuners and receivers have the Dolby decoding circuitry (as well as the needed de-emphasis alternative) built right in.

You recordists who own any kind of tape deck equipped with Dolby are in a fortunate position, since you really don't have to buy a decoder at all to enjoy Dolby FM. All you do when you tune to a program that's being Dolby-ized if you want to record it is to tape it with your Dolby switch in the OFF position (yes, I said OFF, since the program is already coming in Dolby encoded). You turn the Dolby circuit on when *playing back* the tape, and normal proper decoding takes place in the usual fashion. Some tape decks allow you to listen to a decoded version of the incoming program even while you are recording it in encoded form, but even if yours does not you can still enjoy the benefits of Dolby FM during playback.

Abuse and Misuse

Wouldn't you know it—crass commercial competition has a way of rearing its ugly head even in the case of a technological advance such as Dolby FM. It didn't take long for some broadcasters to realize that with the new 25 microsecond pre-emphasis—which can legally be used when broadcasting Dolby FM—if they *still* interpose those nasty compressors and limiters, they can beef up average modulation *still more* and can still become the loudest guy on the FM band (albeit the least dynamically pleasing). So, of course, some Dolby FM stations have done just that instead of taking advantage of the real benefits of the Dolby FM broadcasting idea. Dolby Laboratories is as unhappy about this as you and I should be and has promised to take a closer look at station practices before signing up additional broadcast licensees to use the system. Hopefully, this will lead to the point where we are blessed with more and more stations that will consistently give us the kind of FM signals that we all hoped to listen to and to be able to record on tape. If you have any influence with your favorite local station and if they are not now considering the use of Dolby, you might let them in on the facts as I have attempted to present them here. Maybe, in a few years, Major Armstrong will finally be able to rest in peace.



NORMAN EISENBERG AND LEN FELDMAN

Soundcraftsmen MA5002 Power Amplifier

Soundcraftsmen MA5002: Front panel view.

General Description: For a power amplifier, the Soundcraftsmen model MA5002 has a fairly busy-looking front panel. On it are a pair of VU meters, one for each channel, and calibrated from −20 to +3 dB as well as in percentages (of power output). Between the meters are three buttons to vary the scale: "× 1," "× 10" and "× 100." Various controls and indicators are arranged across the lower portion of the panel. From left to right, these are: the power off/on switch and LED indicator for power on; a red LED to show clipping on channel A; a gain control for channel A; a green LED indicator for the "vari-portional" system; a major and a minor overload LED; another "vari-portional" indicator for channel B; the gain control for channel B; the clipping indicator for channel B; and two buttons for selecting either, both, or none of two pairs of stereo speakers that may be connected at the rear. Of rack-mount dimensions, the MA5002 is fitted with front handles. It is supplied with walnut side panels which may be removed for rack-mounting.

At the rear are the stereo input jacks (standard hi-fi pin jacks) and the speaker terminals (knurled-nut binding posts). Also at the rear are the amplifier's AC line cord which is fitted with a three-prong (grounding) plug, and a fuse holder.

The "vari-portional" phrase relates to the "class H" operation of this amplifier, described in detail in an engineering paper by the manufacturer but which may be summarized briefly as a circuit system that senses and calculates the amount of power-supply voltage required for the amplifier's power output in accordance with the signal, and then "directs" the power supply to furnish that amount of voltage precisely so that no energy is wasted. There actually are two power supplies in the amplifier, one designated as variable and the other as continuous. The former (for higher voltages) comes into play only when required, and this action is shown by the green LEDs.

Protection circuitry in the MA5002 uses no relays or circuit-breakers. Instead, the amplifier has what Soundcraftsmen calls an "Auto-crowbar" circuit that handles overloads, according to whether they are "ma-

jor" or "minor," by causing a brief interruption under conditions of overload or of overheating. Although the power cord is intended primarily for plugging into a grounded wall-socket, it may be used with a socket-adapter in the AC convenience outlet of a preamplifier provided that the latter outlet is rated to handle 1,000 watts or higher. The MA5002 has a built-in delay turn-on circuit to prevent initial high surges of operating voltage, thereby protecting the preamp power switch and avoiding turn-on "thumps" from the speakers. Internal circuitry also blocks any DC from possible damage to speaker cones.

The MA5002 is rated for 250 watts power output per channel, minimum RMS into 8 ohms, both channels driven from 20 Hz to 20 kHz, with total harmonic distortion less than 0.1 percent. IM is rated at less than 0.05 percent; signal-to-noise ratio, as better than 100 dB. Input sensitivity for rated output is listed as 1.3 volts.

Test Results: Our lab tests easily confirmed or exceeded the published specs for the MA5002. With both channels driven into 8-ohm loads, the amplifier furnished 250 watts per channel from 20 Hz to 20 kHz at well below its rated distortion. In fact, across most of the band, THD remained below half that amount, under 0.05 percent, rising to the rated 0.1% only at the 20-kHz mark. For rated distortion of 0.1 percent at mid-frequencies we were able to obtain 290 watts power output from this amplifier. IM distortion also remained extremely low, well under spec'd amount, up to the rated power output of the unit. S/N was measured at 105 dB, which is 5 dB better than claimed; damping factor was above the 100 figure listed. Slew rate was found to be in excess of 50 V per microsecond. Low-level frequency response was measured as within ±0.25 dB from 10 Hz to beyond 25 kHz.

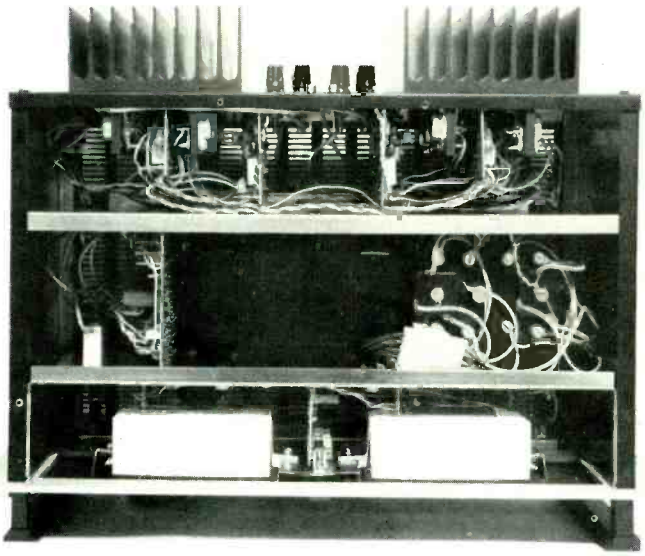
During MR's tests it was found that the MA5002 was capable of being operated at high power output levels without ever becoming warm enough to require the use of a ventilating fan. Internal examination indicated high quality construction and the use of rugged components, suggesting the suitability of this amplifier for long-term uninterrupted service in studio monitoring and sound-reinforcement applications or, conceivably, for some audiophiles, too, although there was some disagreement over its sound.

General Info: Dimensions are 19 inches wide; 7 inches high; 15 inches deep. Weight is 55 pounds. Price: $699. (Note: another version of this amplifier, the model PA5001, which has the same circuitry and specifications but lacks many of the front-panel features of the MA5002, also is available from Soundcraftsmen at a price of $549.)

Soundcraftsmen MA5002: Rear panel view.

Individual Comment by L.F.: Seems as though we are going to run out of alphabet when it comes to the new "classes" of audio amplifiers which have been proliferating lately. First we learned about Infinity's "class D" switching amplifier which in recent months

has been followed by a Sony amplifier that also uses pulse-width modulation for increased efficiency. Then Hitachi brought out its class G amplifier which utilizes different pairs of transistors in the output stage for different levels of signal handling, and thereby gains added efficiency over conventional class B output designs. Now Soundcraftsmen engineers have come up with a circuit they call "vari-portional" and have tentatively assigned a "class H" designation to it.



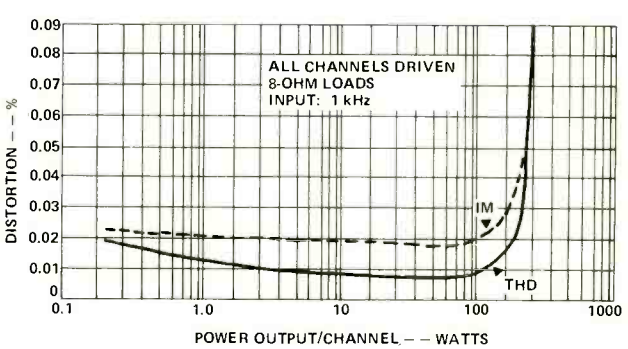
Soundcraftsmen MA5002: Internal view.

In a way, Soundcraftsmen's emphasis of its energy-conserving feature tends to obscure the fact that the MA5002 is an outstanding amplifier, output-class design notwithstanding. Our lab measurements produced really fine results—especially noteworthy is the slew rate measured and found to be better than 50 volts-per-microsecond, quite an achievement in an amplifier costing less than $700 while delivering nearly 300 watts of clean audio power into 4-ohm loads.

I also was impressed with the low, low THD and IM readings observed at low output levels, both of which are indicative of the virtual absence of notch or crossover distortion. And as our lab data shows, the 250-watt per-channel power output rating is conservative, to say the least. I also appreciated the three forms of built-in protective circuitry (crowbar DC-supply disabling, the TRIAC disconnect for AC overload, and the speaker protection in the form of automatic blocking of DC inputs to the amp). These techniques make this amplifier virtually impervious to any catastrophic failure and yet all this protection circuitry did not "get in the way" of transparent, uncolored sound realized when auditioning the amp with a variety of program material.

The idea behind the "vari-portional" system is that as long as the amplifier delivers up to about 50 percent of rated output power, supply voltages are maintained at a lower-than-maximum value. When that level is about to be exceeded, an analog logic circuit activates a second power supply (a higher voltage and variable), which furnishes the added p.s. voltage to meet rising power demands. The front-panel green LEDs light up when this extra high voltage supply comes into play. The purpose of the design is to make for a more efficient use of input power for most operating levels at which the amplifier is likely to perform. It is, in other words, more efficient than a class-B amplifier— according to its designers you are likely to save about 1 kilowatt of electricity for every five hours of use as compared to a conventional (class B or class AB) amplifier, both operating at 1/3 rated power output.

The meters are not, by the way, VU meters in terms of ballistics, but rather peak-reading meters that read approximately "0 dB" when subjected to music signals having peak-power values of that rated output. Obviously, the meter readings will depend on the sort of music reproduced, but the manufacturer chose this peak-reading approach so that it would not be in con-

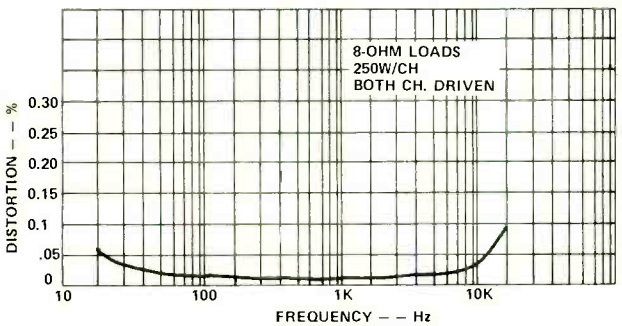


Soundcraftsmen MA5002: IM and THD distortion characteristics.

flict with the peak-LED indicators. Under typical music-listening conditions, both "0 dB" and initial LED peak indications occur simultaneously, showing peak powers of 250 watts or so. I appreciated also the 10-dB and 20-dB attenuators for these meters which

enable them to be useful at much lower power levels.

Individual Comment by N.E.: The MA5002 is a highly sophisticated amplifier capable of very high power output levels which would seem to recommend



Soundcraftsmen MA5002: Distortion vs. frequency.

it for applications in sound-reinforcement or other commercial/professional situations. I am not so sure of it from the standpoint of the critical music/audiophile because of a certain hardness in the sound that was noted by a listening panel. Various reasons for this have been suggested (e.g., underbiasing of output transistors, or the protection circuits "getting in the way" of the signal, etc.) but none is conclusive. For that matter, neither is the report of the "hard" feeling which admittedly is a subjective reaction that may not be shared by other listeners, or that may not even be germane to many possible applications of the amplifier.

The higher efficiency of the MA5002 vis-a-vis conventional amplifiers in terms of its output power available for input operating line voltage has been estimated to save approximately 1 kilowatt every five hours of use over a conventional amplifier, both units operating at 1/3 rated power. In terms of electricity costs, this is not terribly significant, although the actual savings will vary with hours of use and depending on local utility rates. It is of course a plus factor but a very small one overall.

SOUNDCRAFTSMEN MA5002 STEREO POWER AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Continuous power per channel (at 1 kHz)	290 watts
Distortion vs. frequency (250 watts per channel; both channels driven, 8-ohm loads)	under 0.1%, 20 Hz to 20 kHz
Rated THD	0.018% (250 watts at 1 kHz)
IM distortion	0.028% (250 watts)
Signal-to-noise ratio	105 dB
Damping factor	100
Frequency response	±0.25 dB, 10 Hz to beyond 25 kHz
Slew rate	50 V per usec.
Input sensitivity	1.25 V

CIRCLE 10 ON READER SERVICE CARD

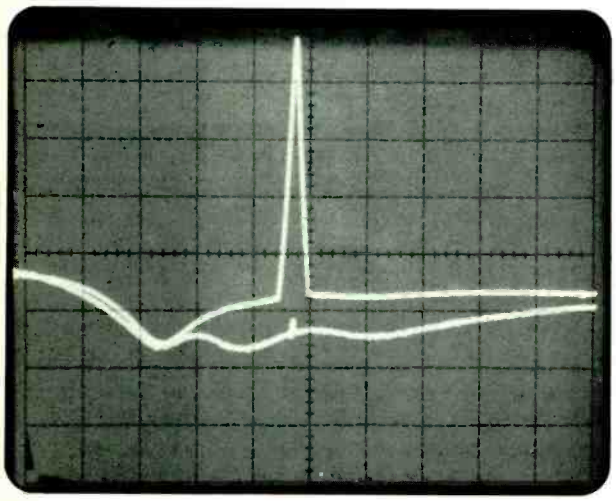
Heathkit Model AD-1304 Active Audio Processor

Heath AD-1304: Front panel view.

General Description: The Heathkit AD-1304 is what might be called a "combination" device in that it includes an active noise-reduction system, a volume expander and a high-frequency noise filter. The device's purpose is generally to remove a fair portion of noise from program material, and also to restore some of the dynamic range limited by the recording and/or broadcast processes. The AD-1304 may be patched into a sound system via the tape output and tape monitor jacks, and should not be connected between volume-controlled devices such as between a preamplifier and power amplifier.

The AD-1304 comes in a metal case with walnut side panels and a brushed aluminum front dress panel that

contains several operating controls and LED indicators. These include: the power off/on button and pilot lamp; four toggle switches (tape monitor; high filter; expander; NR); the expander control knob and two LEDs indicating +4 dB and −3 dB; five LEDs for NR level running from "out" to "max"; and sensitivity control on quiet musical passages until the "max"



(A)

Heath AD-1304: Successive sweeps of noise-only and noise-plus-program tones at (A)5 kHz; (B) 9 kHz; and (C) 13 kHz illustrate action of the three dynamic noise reduction notch filters.

LED flickers off occasionally; the expander control is adjusted so that the +4 dB LED comes on, and the −3 dB LED remains off during loud passages of music. Volume expansion may be engaged whether the NR action is on or off. Similarly, the high filter may be engaged independently of the other two processes.

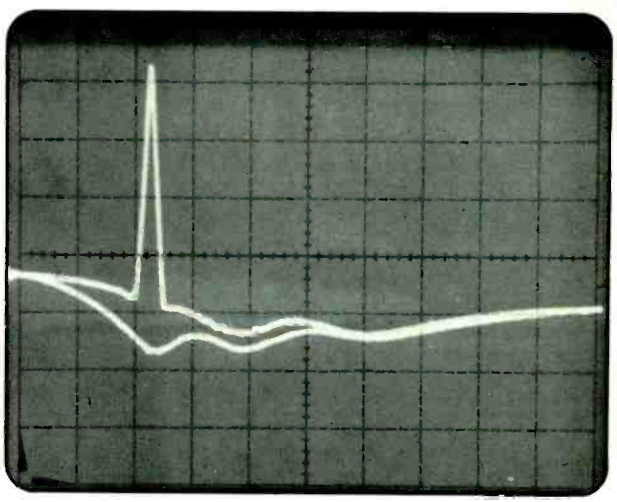
The rear panel of the device contains signal inputs and outputs—these are standard hi-fi pin jacks in stereo pairs for input, tape out, tape monitor, and output. There also are screwdriver adjustments for input and output levels on each channel; two convenience AC outlets, one switched and each rated to handle 300 watts; the unit's AC line cord; a fuse holder.

Between the 7 dB of dynamic range expansion, and the up-to-10 dB of apparent noise reduction, the AD-1304 is claimed to provide a maximum increase in the dynamic range of source material of 17 dB. The device is sold in kit form only. Kit-building work involves mechanical assembly, power supply wiring, and mounting and soldering circuit parts to three circuit boards.

Test Results: Although the Heathkit AD-1304 is sold in kit form, our test sample was supplied prewired, and so we can offer no comment on the ease (or difficulty) in building it. We did, however, study the manual that comes with the device and, like other Heathkit manuals, this one seems complete, clearly presented and amply illustrated. We would judge that anyone with some previous kit-building experience should have no trouble in successfully completing this project, following the instructions furnished.

The test sample met or exceeded published specifications in MR's tests. In our tests we made a series of 'scope photos to help understand the device's action which essentially involves three separate filter systems, each covering a specific portion of the upper two audio octaves (where noise is most obtrusive). These are shown in Figs. A, B and C. With controls and levels set as per instructions, and with the noise filter switched on, the audio band is divided into three processor circuits, or three frequency bands with center frequencies at 5 kHz, 9 kHz and 13 kHz. If, during a given moment of music, there are no frequencies in the signal within the range of these three notch filters, they remain "open" or operative within their respective ranges. When program frequencies fall within the range of any or all of the three filters, the filters are electronically switched out of the circuit. At such moments, although residual noise may be present in the output, it is effectively masked by the high-frequency content of the program material.

Such an arrangement is not too easy to measure on the lab bench. But we did come up with a method that illustrates the effect. We mixed a low level of pink



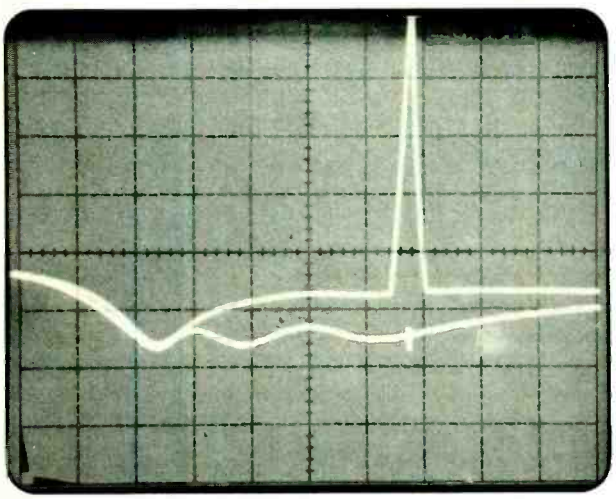
(B)

noise with specific higher-level discrete tones, and swept our spectrum analyzer *linearly* from 0 Hz to 20 kHz. Note that each horizontal division represents 2 kHz of sweep. For each experiment we first swept through the audio spectrum with only the pink noise applied. In Fig. A the results are shown in the lower of the two traces. The three areas of frequency-response attenuation are shown clearly, with the lowest response points being very close to the specified frequencies (5 kHz, 9 kHz and 13 kHz).

In the second sweep (upper trace of Fig. A), we added a 5-kHz tone to the pink-noise signal. This time,

the filter notch is smoothed out and flattened, which indicates that the filter centered at 5 kHz has been "switched out" electronically, and so the 5-kHz tone comes through as it should, while the base-line of noise is raised by some 10 dB.

Similar results were obtained using a 9-kHz tone (Fig. B) and a 13-kHz tone (Fig. C). In each case the presence of the program-tone caused the notch filters to be switched out of the circuit so that those specific



(C)

tones came through. Yet, the lower traces (plotted with only the pink noise applied) show how much noise attenuation would be gained during moments when there are no musical frequencies present in those particular frequency ranges.

The net result of these tests confirms that the AD-1304 does operate as intended, and rather well too.

General Info: Dimensions are 17½ inches wide; 4½ inches high; 8⅛ inches deep. Weight is 10 pounds. Price (in kit form) is $199.95.

Individual Comment by L.F.: Heath has managed to cram a great deal of sophisticated signal-processing circuitry signal-processing into a fairly innocent looking package. One-sided expanders of course are nothing new, and while the Heathkit device works well, it does so within a rather limited expansion range of 7 dB. From my point of view, however, the most innovative part of the AD-1304 is its active noise-reduction system which works extremely well. To be sure, an expander—in and of itself—can provide some apparent noise reduction, but this "indirect" noise-reduction effect should not be confused with the AD-1304's completely separate noise-reduction circuitry which can be used independently of the expansion circuits and is actually a one-sided dynamic noise filter. The noise filtering can be applied to any program material and requires no special encoding or pre-processing of that material. When both the expander section and the noise-reduction section are used together to improve audio reproduction the effects are quite amazing, even though I feel that the device's noise-reduction capability is superior to its expansion action (subjectively, I would judge it to be about 80 to 90 percent as effective as the Dolby-B presently used in tape recorders).

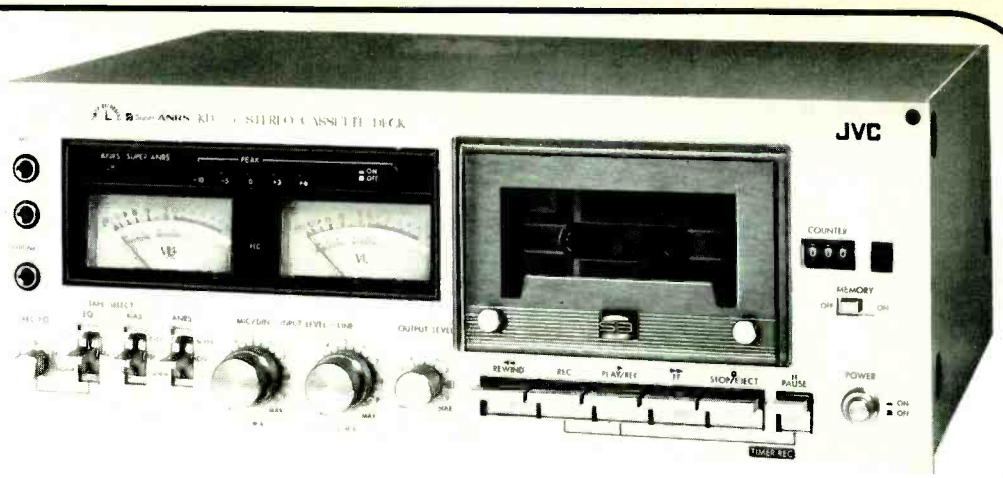
Heath decided to include a passive high-cut filter in the AD-1304 (in addition to the dynamic filtering) for those instances where program material is so noisy that the more subtle dynamic filtering proves to be inadequate. I cannot argue with that reasoning (the passive 12 dB/octave filter adds relatively little cost) except to say that anyone playing that kind of noisy program material in the first place is probably not a prospect for a device as sophisticated as the AD-1304, and probably—in any event—already has a high-cut filter on his or her amplifier or receiver.

Individual Comment by N.E.: While our lab tests confirm the action of the AD-1304, the device did not seem to provide "that much" audible improvement all the time and on all program material. On really noisy records, it had virtually no effect at all. On better signals, the expander worked fairly well most of the time, but with an occasional hint of "breathing" and—at least to my ears—with attenuating during very soft passages, some of the deepest low-frequency information such as bass-drum beats. The manual states correctly that it may be necessary to change the sensitivity control setting for different source material. I would underscore this advice and add that it also may be necessary to change that control setting for different cuts or portions of the same recording. The need to connect the device only into the tape-monitor loop seems to me a limitation from the installation standpoint—I would prefer that any device of this type could be patched between preamp output and power amp input. I do not agree that the device stands comparison with Dolby-B circuitry.

HEATHKIT AD-1304 ACTIVE AUDIO PROCESSOR: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Rated input	200 mV (used for all tests)
Input impedance	100 K ohms
Gain, expander off	0 dB, ±0.2 dB
expander on	−3 dB to +4 dB (depends on program level and control setting)
Frequency response	±1 dB, 20 Hz to 20 kHz
High filter cutoff	−3 dB at 7 kHz (12 dB/octave)
Sensitivity (for notch to open)	520 uV
Hum and noise (re: 200 mV in)	−70.5 dB
Output impedance	500 ohms
Input overload	6.7 V at 1 kHz
Total harmonic distortion	0.1% at 20 Hz 0.045% at 1 kHz 0.1% at 20 kHz
Power requirements	120 V 50/60 Hz, 15 watts (wireable for 220 V-240 V).

JVC KD-75 Cassette Recorder



General Description: The most newsworthy feature of the JVC-KD 75 front-loading cassette tape recorder is its inclusion of JVC's recently developed "super ANRS" signal- processing circuitry. By way of understanding this, the earlier ANRS (the letters stand for automatic noise reduction system) has been by now made almost completely compatible with Dolby-B, so that recordings made via Dolby-B can be played and "decoded" correctly (for the usual improvement in S/N) using ANRS.

Super ANRS is not merely "more" of ANRS. Indeed, it comes into play at high signal levels (at high frequencies) rather than at low signal levels. It compresses such high-level high-frequency signals during the recording process, and expands them on playback (just opposite of what ANRS or Dolby does for low-level high-frequency signals). The net result is that when recording at or near the 0-dB level, the high-frequency high-level signals, which normally would saturate the tape, are kept within bounds and therefore are recorded at progressively lower magnetization levels. The "super" part of Super ANRS then has nothing to do with noise reduction, but rather is intended to improve linearity (and to reduce playback distortion) for high-frequency signals that have been recorded at high levels. Simply put, it means a gain of recording headroom (explained more fully below in the "Test Results" section).

Aside from the added switch position and LED indicator for Super ANRS, the KD-75 looks generally like any modern front-loader. The lefthand portion of the front panel is dominated by a pair of VU meters, each calibrated from −20 to +5. The meters are mounted on a recessed panel behind a transparent cover; across the top of this panel are two LEDs for ANRS and Super ANRS, plus five more LEDs showing peak levels of −10, −5, 0, +3 and +6, with a button to activate them or turn them off. Between the meters is another LED to indicate the recording mode.

Left of the meters are three standard phone jacks for left and right microphone inputs and stereo headphone output. Across the lower portion of the lefthand side of the panel are the electronic controls. These include: a recording EQ switch with five positions marked −2, −1, 0, +1, +2; a tape EQ selector with three positions marked CrO_2, SF and NORM; a tape bias selector with three positions marked similarly; the ANRS switch with its three positions (Super, on, off); dual-concentric knobs for mic/DIN input (inner knob, left channel; outer knob, right channel); a similar pair of knobs for line input level; and a single knob for output or playback level. The tape-selector switches can be set for FeCR cassettes by moving both to indicated spots. The input level controls permit input mixing of mic and line or of DIN sources and line.

The cassette compartment dominates the righthand portion of the panel and is overlaid with a tinted plastic cover that may be removed if desired. To the right are a three-digit tape index counter and its reset button. Just below them is the memory off/on switch.

Below the cassette compartment are the transport controls— mechanical levers for rewind, record, play/record, fast- forward, stop/eject and pause. Fast-buttoning is possible in the play mode. At the lower right corner is the unit's AC power off/on switch. The transport uses a single DC servo-controlled motor.

Rear panel connections include two stereo pairs of pin-jacks for line-in and line-out signals, plus the optional DIN (multi-socket) connector. The KD-75 is supplied in a metal case with a brushed-aluminum front panel. As a front-loader it is designed for horizontal installation in the same manner as an amplifier or receiver, etc. The owner's manual is printed in three languages (English, German and French) with the respective texts running side-by-side.

Test Results: Performance of the JVC KD-75 in our tests was well within published spec's, although the overall spec picture was not quite as auspicious as that found on some of the higher-priced "all out" cassette decks. Be that as it may, with normal tape response ran within ±3 dB from 40 Hz to 15 kHz; with TDK SA tape (CrO_2 switch settings) it improved at both low and high ends to cover within ±3 dB range from 30 Hz to 16 kHz, which can be considered an excellent mark for any cassette deck.

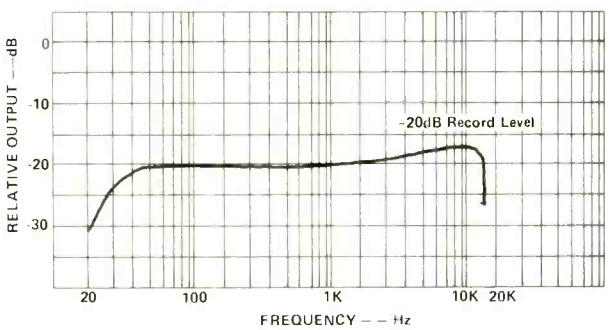
To document the improved signal headroom contributed by the use of Super ANRS, we recorded a response plot at the 0 dB level deliberately (normally, we would measure R/P response for a cassette deck at the −20 dB level). First we recorded without Super ANRS. Then we recorded with Super ANRS. The improvement is apparent, as shown on the accompanying

response graph which indicates an extended high end at 0 VU for Super ANRS vis-a-vis recording without it. At the more customary −20 dB recording level, of course, the response extends into the highs comfortably without the need for Super ANRS.

A novel feature of the KD-75 is its record-EQ switch, offered in addition to the more usual EQ tape selector. The former switch permits vernier adjustment of the equalization curve provided by the recording amplifier of 1.5-dB in five steps, and is intended to take care of minor differences among different tape formulations within a given category.

General Info: Dimensions are 16½ inches wide; 6⅜ inches high; 13¼ inches deep. Weight is 17.2 pounds. Price: $380.

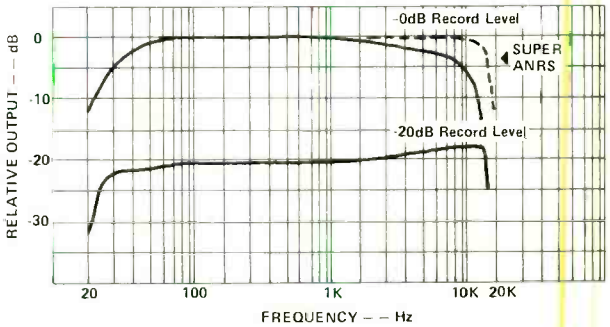
Individual Comment by L.F.: Several things about the KD-75 impressed me favorably. The most important one (and the one least understood by the general public because JVC simply has not publicized it very much) is the Super ANRS which our tests show does add significant headroom and improves linearity



JVC KD-75: Record/playback response using TDK "Audua" tape.

for recording high-frequency high-level signals. Another feature I liked in this deck is the incorporation of five peak-indicating LEDs, although I would have preferred that they were supplied on a "per channel" basis rather than as a combined reading of the peaks of both channels. In my view, this would have been particularly useful since the KD-75 does offer separate (dual-concentric) controls for each channel's level adjustment in the recording mode. The versatile recording-EQ facility (a three-position switch plus five more positions on another switch for each of the basic three) also seems to me a useful feature.

Individual Comment by N.E.: While Super ANRS is demonstrably an aid in improving cassette tape sound, I cannot detect that it in itself makes cassette recordings sound better than they already can sound at normal recording levels with ANRS or with Dolby-B. The ability to record at higher levels is, theoretically at least, a plus factor, although in recordings I made with the KD-75 the meter needles rarely moved up much off the −20 VU mark, and the LED peak indicators barely got off the −10 mark. Be all that as it may, the KD-75 is capable of making



JVC KD-75: Record/playback response using TDK SA C60 tape.

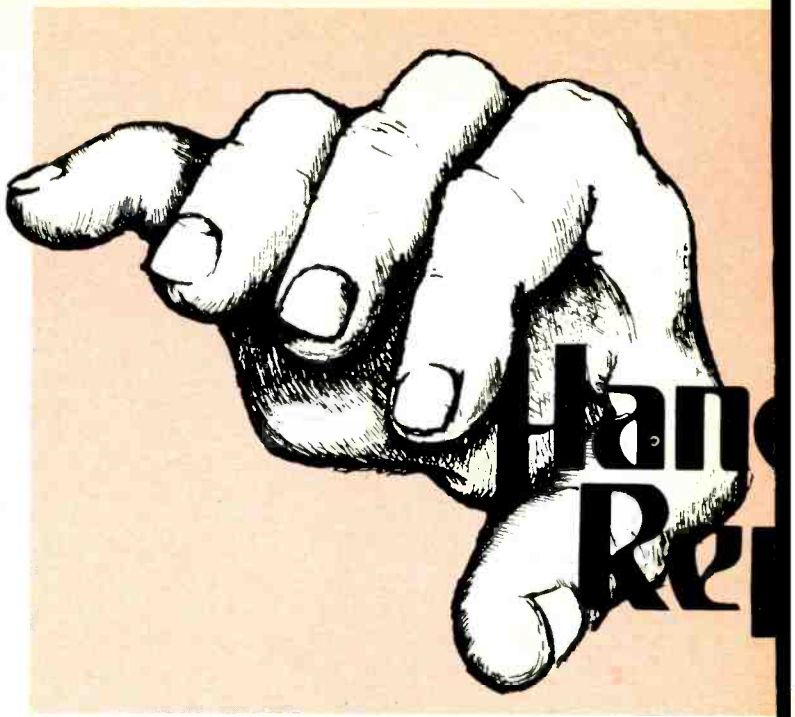
excellent-sounding cassette recordings and of playing back cassettes made elsewhere, Dolbyized or not. The Super ANRS, whatever, does function satisfactorily on record and play although the difference between it and regular ANRS strikes me as subtle indeed. The KD-75, incidentally, afforded me another opportunity to demonstrate the compatibility between ANRS and Dolby-B, interchanging as I did cassettes made with one system and played via the other.

The transport controls on the KD-75 functioned well enough although they are not the easiest-to-activate I have experienced. The owner's manual for this deck should be scrapped in favor of a single-language version which would permit better grouping of the text and the illustrations, especially for the all-important front panel controls and features picture which has the descriptive text for it running several pages after the illustration—which itself is too small for easy studying. The list of tapes by brand and type, with recommended switch settings for each, is especially good in this manual.

JVC KD-75 CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Frequency response, normal tape	±3 dB, 40 Hz to 15 kHz
TDK-SA tape	±3 dB, 30 Hz to 16 kHz (with Super ANRS switched in, +0, −3 dB, 30 Hz to 15kHz at 0 dB level).
S/N ratio, SA tape, ANRS off	57 dB
S/N ratio, SA tape, ANRS on	65.5 dB
THD at 0 VU, Audua/SA	1.2%/1.1%
THD at +3 dB, Audua/SA	1.9%/1.3%
Record level for 3% THD, (without Super Audua/SA ANRS)	+6 dB/+7 dB
Line input sensitivity	80 mV
Mic input sensitivity	0.2 mV
Line output level	0.45 V
Headphone output level (8 ohms)	37 mV
Wow and flutter	0.06% WRMS
Fast-wind time, C-60	80 seconds
Bias frequency	95 kHz
Power consumption	25 watts

Soundcraft Series II 24-in/8-out Mixing Console



By Brian Roth and Jim Ford

General Description: The British made Soundcraft Series II mixing consoles were designed to fill the void between the large, expensive studio recording consoles (some costing $100,000 or more) and the more limited mixers designed for home recording and semi-pro applications. True, the price of the Series II is probably too steep for many budgets, but it will buy a wide variety of useful control functions for 8- or 16-track recording (and even for a very sophisticated PA system) that are unavailable on most mixers except for the high-dollar studio units.

Each of the 24 inputs provides:

(A) Straight-line fader with conductive plastic element and 4-⅛" travel.

(B) Separate microphone input (via cannon connector) and line input (via quarter-inch phone jack).

(C) Microphone/line input selector switch.

(D) 20 dB switchable microphone pad.

(E) Input level gain adjustment control (it would vary both microphone and line input sensitivity).

(F) Four section equalizer with:

(1) Bass shelving (boost or cut) control.

(2) Low midrange peaking (boost or cut) control with continuously variable center frequency adjustment.

(3) High midrange peaking (boost or cut) control with variable center frequency adjustment.

(G) Treble (high cut) filter with variable cut-off frequency adjustment.

(H) Bass (low cut) filter with variable cut-off frequency adjustment.

(J) Equalizer in/out switch.

(K) Four auxiliary "send" controls (for echo or cue mixes) with pre- or post-fader selection switches.

(L) Input overload "LED."

(M) Pan pot and five assignment switches that operate in conjunction with the pan pot to route the input signal into the 8 output buses or to the stereo mixdown buses.

(N) Input on/off switch with solo (called PFL for "Pre-Fader Listen" on this console) position.

On the rear panel, each input provides a pair of jacks (quarter-inch phone jacks, as are all other jacks on the mixer except for the microphone inputs) for patching outboard accessories (limiters, equalizers, etc.) into the signal path. Each input also has a direct output jack on the rear. Each input provides a switch for sending 48 volt "phantom power" to condensor microphones.

The input faders, eight output master faders, stereo mixdown master fader and the two echo return faders are high-quality units made by Penny and Giles (of England) which are used in many very expensive studio consoles due to their smooth action, ruggedness and low noise.

At the righthand side of the console (above the eight master faders) is located the control room and headphone mixdown panel. Eight volume pots and eight pan pots allow monitoring various combinations of the console outputs or multi-track tape playback. Each of the eight monitor channels also has four auxiliary "send" pots that are wired into the same mixing buses as the auxiliary send controls on the inputs (see item "K" above).

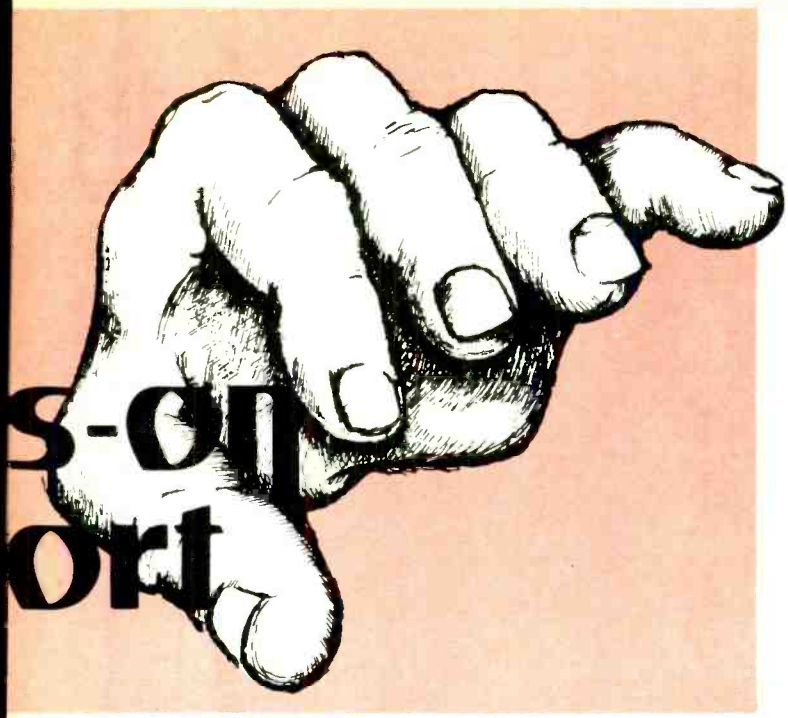
Since the auxiliary send controls have pre- or post-fader selection switches, they can serve multiple functions. For example, it is possible to have stereo reverb in the monitor mix (without actually recording reverb on the multi-track master tape) while also sending two independent headphone mixes to the musicians. Or, on mixdown, all four auxiliary "send" buses can be used for echo or effects "sends." It just depends on how the controls and switches are set.

Two echo (or effects) return inputs are provided with level controls, four auxiliary send pots (so that reverb can be sent to the headphone cue mix), a straight line fader and an on/off PFL (solo) switch. Also, each return has a pan pot. One return feeds only the stereo mixdown bus, while the other return can be routed to the eight output buses as well as the stereo mixdown buses on the unit.

Other features include control room source switches, control room volume control, talkback facilities and a built in oscillator.

The eight VU meters can be used for multiple functions. They can monitor the output level of the eight program lines, or playback from the multi-track recorder. They also can be switched to read the stereo mixdown outputs or the auxiliary "send" outputs.

Switching facilities are provided to allow operation



with a 16-track recorder, although only eight tracks can be monitored at one time on the monitor mixdown panel. The signal routing allows input channels to serve as an additional part of the monitor mixdown section. Naturally, using inputs for the monitor mix would mean less inputs available for microphones. However, with twenty-four inputs, this should not prove to be a great problem if eight inputs are used for monitor mixdown.

There are numerous other features, but that covers the important ones. It should be obvious that the Soundcraft Series II is truly a full-capability studio console for a legitimate 8- or 16-track operation.

Field Test: It took a while for us to acquaint ourselves with the wide variety of functions on the console. However, the control layout is very logical, although a bit crowded in areas due to the overall compactness of the unit.

After setting up an experimental 8-track studio in our shop, we proceeded to do a small session.

We immediately liked the flexibility of the equalizers. The "quasi parametric" type equalizers were very helpful in solving sound problems we encountered. Having the ability to "tune in" the equalizers was a fantastic feature. The sound of the equalizers was excellent. The midrange sections did not introduce any "peaky" sound when set for moderately large amounts of boost. Since the equalizer didn't have a bandwidth adjustment control (such as would be found on a true parametric equalizer) it is good that Soundcraft chose a broad, pleasing sounding equalization response for the console's midrange sections.

We noticed that switching the equalizer in and out slightly altered the sound quality, even with the equalizers carefully adjusted for a "flat" setting.

All of the controls in the mixer were very smooth in operation (particularly those P&G faders!) and, for the most part, quiet although the input trim pots sounded a bit "scratchy." We also noticed that the output bus assignment switches would introduce a small "pop" when actuated.

The console facilities were very extensive, making the normal operation of a complex multi-track recording session quite straightforward. We did miss a couple of features, however.

As it stands, both the control room monitor amp and a studio monitor amp (if used) must be controlled by a single set of controls; this could pose a problem in many installations.

The monitors did not automatically switch to the solo bus when a PFL (solo) switch was activated. It was necessary to push an additional switch to monitor the solo bus.

The use of a small 3-position toggle switch for the input on/off function as well as PFL proved to be a frustration occasionally. It was easy to overshoot the "on" position and to mute the input channel if the switch was being returned from the "PFL" position to the normal "on" position.

We enjoyed the great flexibility of the four auxiliary "send" buses. The arrangement proved to be capable of handling any situation we could encounter.

The mixer was very quiet in operation, both through the main program outputs and the stereo mixdown outputs. Under nearly all circumstances the tape recorders introduced more extraneous noise than the console. We did detect some low-level radio pick-up in the auxiliary send buses at times.

The upper portion of the range of the midrange frequency adjustment controls was a bit "compressed" making careful adjustment necessary when "tuning" the midrange equalization. This "compressed" scale was also noticed on the input sensitivity trim controls.

The continuously adjustable roll-off points for the high and low filters was very handy for eliminating various types of noise without greatly affecting the sound quality of an input signal.

A phase reversal switch would have been a welcome addition on the inputs, particularly for adjusting the sound on our drum mix.

We were quite pleased with the separate "phantom power" switches on each input. We have been in situations where the presence of "phantom power" would affect the performance of dynamic microphones, even though only a condensor microphone is supposed to "see" the phantom power voltage. We could program the input channels for phantom power for the condensor microphones while applying no power to the non-condensor microphones.

The microphone preamplifier was capable of handling any signal that we encountered without any audible signs of distortion. We did notice that activating the microphone pad would alter the sound character somewhat.

The input overload LEDs were positive in action. Once triggered on, they remained illuminated for a sufficiently long time to be noticed. We decided that it was best to operate the master faders slightly higher

than recommended to ensure adequate output level without the overload indicators flashing constantly.

We examined the interior of the console and were very pleased with what we saw. Construction standards were high, and the parts were good quality components of European manufacture. The console is semi-modular in that four inputs are grouped together on one module. Each set of four inputs plugged into a set of "mother-boards." Servicing the unit should be relatively easy except for the inconvenience of having to turn the console on its side and remove the bottom panel to gain access to the plug-in module connections.

Overall, the Soundcraft Series II mixer was a very professional unit. We think it would be quite suitable for a multi-track studio, or in many cases for a super flexible PA mixer.

SOUNDCRAFT MIXER PERFORMANCE

Noise
20 Hz - 20 kHz unweighted
dB figures are below 0 VU
(+4 dBm or 1.25 volts)

1 microphone input to line output, equalization out, normal fader settings	
Trim adjusted for 20 dB gain	−95 dB
Trim adjusted for 40 dB gain	−89 dB
Trim adjusted for 60 dB gain	−72 dB
Equivalent input noise	−128 dBm
16 line inputs to remix output, equalization in, input trims set for unity gain, normal fader settings	−81 dB
24 line inputs to line output, input trims set for unity gain, normal fader settings	
Equalizers in	−77 dB
Equalizers out	−79 dB

Total Harmonic Distortion
Line input to line output

Frequency	0 VU (+4 dBm) output	+20 dB (7.75 volts) output
20 Hz	.008%	.015%
1 kHz	.0065%	.012%
20 kHz	.035%	.14%

Note: +20 dB measurements made with master fader set at +5 setting (see text)

Microphone input to line output

As above except:	
Input trim min, faders normal, 0 VU output — 20Hz	.18%
Input trim midscale, faders normal 0 VU output	.028%

Intermodulation Distortion
(60 Hz and 7000 Hz mixed 4:1)

Line input to line output

	0 VU	+20 dB output
Equalization in	.008%	.05%
Equalization out	.006%	.04%

Note: +20 dB measurements made with master fader set at +5 (see text)

Maximum Output Level
Note: All measurements made with master fader set at +5 (see text)

Output Load

Frequency	High Impedance	600 Ohm
20 Hz	+22.25 dB (10 volts RMS)	+20 dBm (7.75 volts)
1 kHz	+22.25 dB (10 volts RMS)	+21 dBm (8.75 volts)
20 kHz	+22.25 dB (10 volts RMS)	+21 dBm (8.75 volts)

Remix maximum output levels about 1.5 dB greater.

Maximum Input Level, Microphone Input

Frequency	Without pad	With pad
20 Hz	−1 dB (.7 volts RMS)	+15 dB (4.4 volts RMS)
1000 Hz	+8.5 dB (2 volts RMS)	+25 dB (13.75 volts RMS)

Crosstalk, Worst Case
(output bus #2 into bus #1)

20 Hz	−60 dB
1 kHz	−65 dB
20 kHz	−56 dB

Lab Tests: Nearly all measured characteristics of the console were first rate. Noise levels were very low. We did not include a "typical mix" measurement as we usually do because in a console of this type there is no single typical mode of operation. Instead, we measured noise levels under several different modes of operation.

The overall frequency response was adequate, although not the best we have encountered. We found that switching the equalizer in would cause additional high-end and low-end roll-off, probably due to the introduction of the bass and treble cut filters. Perhaps Soundcraft should extend the "tuning" range of the filters so that they are further out of the audible range when set to their maximum positions.

We also noticed that the setting of the input sensitivity controls would affect the high-frequency response. The lower the setting of the input trim controls, the better the high-frequency response was.

Crosstalk was generally acceptable, although it increased at high frequencies. We found that the low-frequency response of the console's VU meter circuits was lacking. The meters' response was 1 dB down at 100 Hz and 3 dB down at 60 Hz.

The maximum output capability of the console was very good. We noted that with the master faders set at the recommended point that the input stages would clip before the output stages, thus lowering the maximum unclipped output level by about 5 dB. Consequently, we strongly suggest that the master faders be increased to the "+5" setting as opposed to the recommended "0" position. This will ensure maximum headroom throughout the signal path. This will also increase output noise levels by about 5 dB, but the console is quiet to start with so this probably won't matter very much.

Distortion measurements were all good. Distortion components were mainly second and third harmonics.

In summary, a respectable level of performance was measured at the bench.

Conclusion: The Soundcraft Series II console is a very flexible, very clean performing mixer. It should be right at home in a professional recording studio.

The operating manual was very thorough. The servicing section was complete with schematic diagrams, parts lists and parts location diagrams for all circuit boards. This is in keeping with the "pro" characteristics of this well-made console.

If your needs are beyond the capabilities of a small mixer, but you cannot afford a large studio console, the Soundcraft Series II consoles are definitely worth considering.

If You

create sounds or songs

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is Your Next Creative Step!

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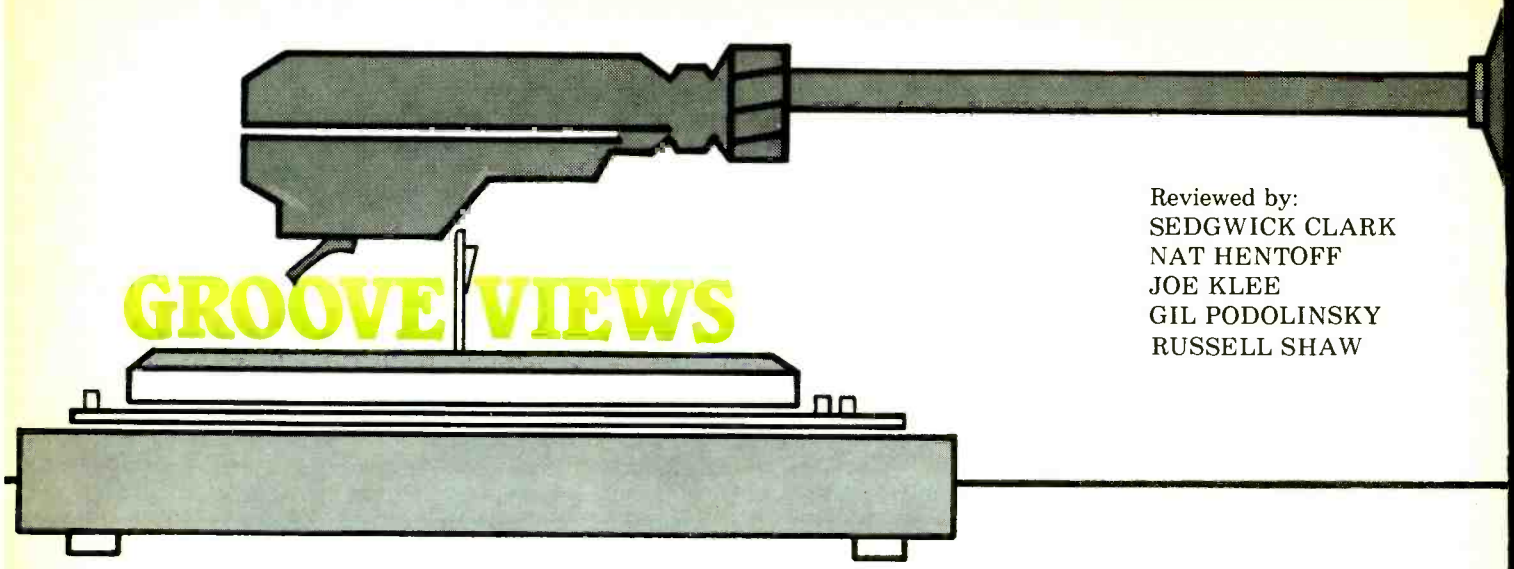


Reviewed by:
SEDGWICK CLARK
NAT HENTOFF
JOE KLEE
GIL PODOLINSKY
RUSSELL SHAW

POPULAR

BE BOP DELUXE: *Live! In The Airage.* [Bill Nelson, John Leckie, producers; John Leckie, engineer; recorded "live" on the Stones Mobile, P.A. systems by Brittania Row and Tasco. "Live" concert mix by Neil Levine. Mixed at Abbey Rd. and Advision Studios.] Harvest Records SKB 11666.

Performance: **Up to date**
Recording: **Representative of their "live" performance**

As Bill Nelson, guitarist/singer/songwriter of Be Bop Deluxe explains in the liner notes, the band's approach to "live" performances is that they are a separate entity from studio recording—a common belief that many groups have come to regret. In many ways, this is an interesting "live" concept, for three never-before-recorded songs are included. At the same time, the second record of this two record set is actually a 12" EP (extra play) containing only three songs, which prompts the question why with four previously released studio albums, they'd come up short on material.

After seeing Be Bop Deluxe twice in performance, I left disappointed each time for the exact separations achieved on the studio LPs were sorely missed.

BE BOP DELUXE: A new chapter

"Live," Nelson's guitar drowned the more subtle aspects of the music; "live" virtually all the keyboards tend to be very distortive and muddy. *Live! In The Airage* either represents, as Nelson alluded, a new chapter in the band's development, or it was given a shot in the arm in the studio mix. While Nelson's guitar is still over distorted and the overall mix muddy at times, the keyboards are more present and closer to the studio album counterparts than when I've heard them "live." The mix *is* Bill Nelson—guitar, vocal et al. Charles Tumahai's background vocals are mixed slightly under those of Nelson's, with his bass being overshadowed by Nelson's guitar. Drums, too, could have been brought up more in the mix. However, all this is *not* to say that this is a poor recording of no interest. Hardly. Be Bop Deluxe is a very talented progressive band featuring one of the best lyricists to emerge in the '70s. Plus, the addition of new material—as well as Nelson's incredible solo on "Adventures in a Yorkshire Landscape" from the first album, *Axe Victim*—are well worth the price alone. I still have high hopes for this band. G.P.

STEVE WINWOOD: *Steve Winwood.* [Steve Winwood, Chris Blackwell, Mark Miller, producers; Phil Brown, Robert Ash, Ray Doyle, engineers; recorded at Basing Street Studios (Island Studios), Chipping Norton Studios, and the Island Mobile Studio.] Island ILPS 9494.

Performance: **Rejuvenated Steve Winwood**
Recording: **Basic**

This is one of the more consistent recordings involving Steve Winwood as featured artist to come along in quite a while. The material, which is co-written with former Traffic drummer, Jim Capaldi, is better than much of what we've heard from Winwood recently. The album is centered around Winwood's voice with no exceptional solos from anyone else. What few solos there are sound forced, adding nothing and lending credence to the belief that they were included in keeping with the formula approach to songwriting ("ok, we'll put a solo here 'cos I have nothing more to say," unfortunately, neither do the solos).

Winwood, playing both keyboard and guitar, approaches everything as though it were a musical bed for a commercial. The mix is fairly unchanging, with bass,

drums and piano the staples, supplemented on several cuts by Hammond and occasional congas and guitar. The treatment given the Hammond is poor on side one, with it kept so far in the background that it serves no purpose. Studio gimmicks, such as quickly panning a Rhodes back and forth to give the illusion of motion, like on "Luck's In," are common. I'm also not too keen on varying an acoustical piano by scraping the low end and addint artificially created highs—the results are grating tinnyness. If you want a honky-tonk feel, go with a tack piano.

Generally, though, the material gives Winwood a shot in the arm—a shot of good which is quickly undone by production that tends to make it stale before its time. G.P.

STEVE WINWOOD: A shot in the arm

ALAN HOLDSWORTH: *Velvet Darkness.* [Creed Taylor, producer; Rudy Van Gelder, engineer; recorded at Van Gelder Studios.] CTI 6068.

Performance: **Hey!**
Recording: **Attempted change of image**

Surprises abound on this album. First, though I'm not surprised that it's a progressive performance of great stature, I *am* surprised that it's on CTI. Secondly, I'm surprised it's produced by Creed Taylor and, thirdly, that he and Van Gelder could understand the artist's statement well enough to bring it home. Since I've never heard them record anything that wasn't mainstream sickly pop jazz, I didn't think that they were interested in anything else. Alan Holdsworth is a guitarist well worth taking in. Cultists first became familiar with his work in Soft Machine and Gong. Able to go from one very electronics-dominated guitar number to a double-tracked acoustic guitar without pause, it's time he came to the attention of the masses for he is a very strong stylist.

The acoustic numbers feature his own accompaniment in the left channel and lead work in the right. While not the most imaginative approach, it works, which is all that matters. On electric numbers, the drums are recorded too dead for my tastes, tending to make the guitar sound very dry as a result. Also, quickly panned drum rolls were repeated far to often, like, when in doubt... On the title track the drums are so far removed from the rest of the instruments in the front of the mix it's as though they were in a different room with the door opening only occasionally. Why the cymbal work, which receives too much use in my opinion, gets special treatment in the mix over the rest of the kit is beyond me. Lastly, this is the first time I've ever heard the bass drown out the soloist—what gives? Regardless of the recording conditions, Holdsworth shines through, which is the mark of a great recording in the artistic sense. G.P.

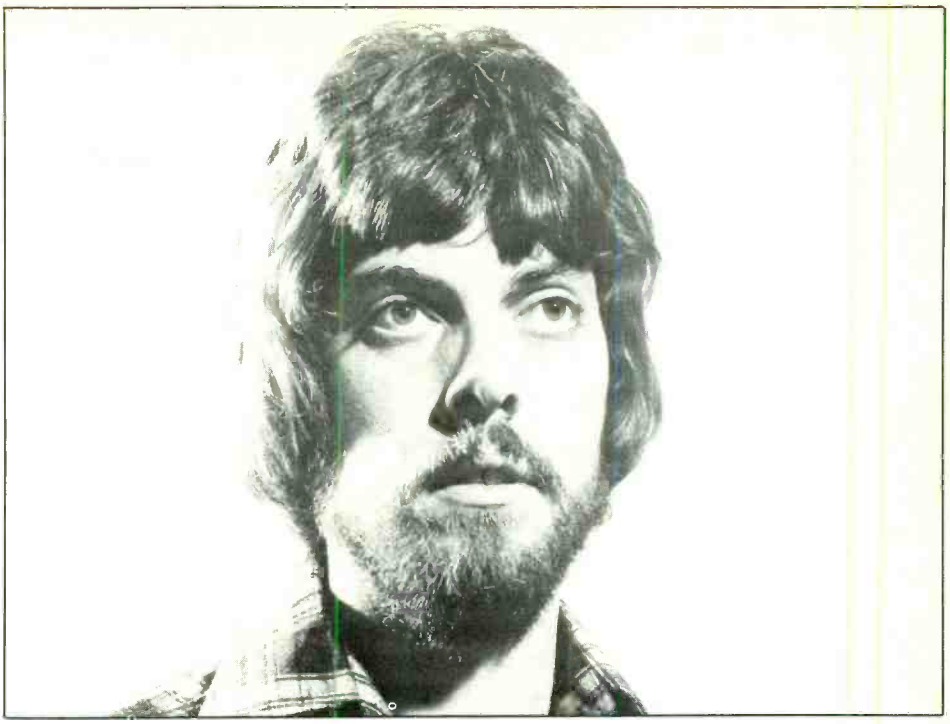
THE ALAN PARSONS PROJECT: *I Robot.* [Alan Parsons, producer; Alan Parsons, Pat Stapley, Chris Blair, engineers; recorded at Abbey Road Studios, London.] Arista 7002.

Performance: **Flawless**
Recording: **Sophisticated to standard setting—progressive engineering**

If you haven't heard by now, Alan Parsons, engineer, worked on such important works as *Abbey Road* (Beatles), *Dark Side of the Moon* (Pink Floyd), *Air That I Breathe* (Hollies), and *Year of the Cat* (Al Stewart) among others. After releasing last year's stunner, *Tales of Mystery and Imagination: Edgar Allen Poe*, The Alan Parsons Project, consisting of Parsons, executive producer Eric Woolfson and arranger/conductor Andrew Powell have again conceived another brilliant concept album—this one based on the rise of machine and the decline of man. The key to their success is two-fold: Having the technical expertise to do anything, thus allowing the imagination to create unabashedly; and taking a sessions-album approach, using both known and unknown musicians and utilizing their strengths. If anything, The Project knows how to create and evaluate.

Along the way they also invent instruments like the vocorder (a vocal synteisizer) and the projectron. In short, in two albums, these guys have advanced the recording industry further than anything done in this decade. Separations are amazing, and they must be, for there are many musican/technical subtleties constantly going on.

The most easily identifiable track style-wise is "The Show Must Go On," which definitely demonstrates his contributions to *Dark Side of the Moon*. His ability to layer tracks without loosing separations is an art in itself. Centered is a synthesizer that plays a repetitious line throughout; it's placed in the foreground just behind the vocal and even with the bass and snare. Behind that are the acoustic rhythm guitar and ride cymbal and toms. Underlying all this is an electric slide guitar. The vocal is so well done that the concept used is overlooked. The voice is separated with bass left, mids center and highs right, done with slight echo and probably a doubling device. The chorus somewhat resembles his work on *Dark Side*. Whereas there Parsons put the female chorus through a Leslie and allowed it to slow down to give that swirly-ending sustain effect, here he places the chorus center and a Leslied vibratoed electric guitar right, achieving a similar effect without resorting to copying himself, thus turning a fine technique into a gimmick. I could go on and on about the numerous lessons taught in record making here, but I'd rather put down the pen and enjoy the music G.P.



ALAN PARSONS: A brilliant creation

Two Uncommonly Graceful Swinging Pianists

By Nat Hentoff

When the demonic mood hit him, which was often, Charlie Parker would beat off so ferociously fast a tempo that his trumpeter at the time (in the late '40s) would often be moved to quit at the end of the set. Yet Bird would persuade Miles Davis to reconsider, and Miles would not quit again until later in the week. Yet another member of the quintet, pianist Duke Jordan, not only seemed to have no difficulties with any tempo but also served as a kind of reflective enclave in that volcanic combo. His solos never sounded rushed, and they were characterized by a lucidity and grace which were singular in much of early bop piano.

Through the years, other pianists have also talked admiringly of Duke Jordan's "touch"—his ability to combine crisp articulation with a singing, airborne feeling. There was also his conception, fresh ideas on a familiar base, along with so remarkable a structural sense that his improvisations sounded like unusually thoughtful compositions. All these qualities remain, and indeed have been strengthened by time, as is evident in one of the most thoroughly satisfying albums Jordan has ever made, *Flight to Denmark* (Inner City). Recorded in Copenhagen, the set joins Jordan with expatriate Ed Thigpen who, throughout his career in the States, acquired a reputation as what musicians used to call a "tasty" drummer. Bassist Mads Vinding is one of a line of big-toned, resilient young Danish masters of that instrument. The recording is clean, flawlessly balanced, with the sound/feel of a "live" session in an intimate room.

Another pianist who reflects apparently impermeable inner order, no matter what the tempo and surroundings, is Hank Jones. Like Jordan, he is a lyrical player although he too swings with easeful force. And, as Alan Goodman writes in the notes to Hank Jones' *Bop Redux* (Muse), like Duke Ellington, "he pays strict attention to the flow of melody and the ascension into song."

Although in no way a deliberately dazzling player, Hank is on wholly intimate terms with the instrument; and the infinite subtleties with which he reveals that command led the usually hypercritical John Lewis to say recently, after they had played a week as duo-pianists, "It was sheer pleasure just being able to hear him all night long."

In Hank Jones' *Bop Redux*, Hank plays a recital of compositions by Charlie Parker and Thelonious Monk. Adhering to the spirit of such distinctive originals as "Confirmation" and "'Round Midnight," Hank brings his own light to the pieces, transmuting them into newly personal dimensions of deftly updated bop. His drummer here is Ben Riley, who has often worked with Monk, and is very much a *listening* percussionist. On bass, the implacably dignified-looking, technically faultless, and deeply swinging George Duvivier. Like the Duke Jordan set, this is chamber jazz that transcends categories as well as time itself.

The engineering provides a spacious sense of full-bodied interplay, with every tub, to paraphrase Count Basie, securely on its own bottom but vibrating to the sound of the whole.

DUKE JORDAN: *Flight to Denmark.* [Nils Winther, producer; Ole Hansen, engineer.] Inner City 2011.

HANK JONES: *'Bop Redux.* [Fred Seibert and Dick Ables, producers; Chuck Irwin, engineer.] Muse MR 5123.

THE STRANGLERS: Why?

THE STRANGLERS: *Rattus Norvegicus.* [Martin Rushent, producer; Alan Winstanley, engineer; recorded at T.V. Studios, Fullham, England.] A&M SP 4648.

Performance: **Amusing**
Recording: **Nostalgic**

Well, the new wave of punk rock is definitely no asset to the advancement of recording innovations or technology. If anything, it's a throwback. On this particular attempt, the musicianship is terrible, with every tune the Stranglers do a nostalgic look back to the days of the Doors, Velvet Underground or Roxy Music—and those days weren't particularly sunny, either.

Lyrics here are a joke, and vocals are (as seems to be the punk rock way of doing things) merely a hoarse yell. Music has simply come too far for this to be acceptable. The mix is practically monophonic, with everything centered (which is, we assume, the way they wanted it). With so many talented new groups, and old ones, going by unheralded, one might question why the Stranglers have garnered so much attention to date. G.P.

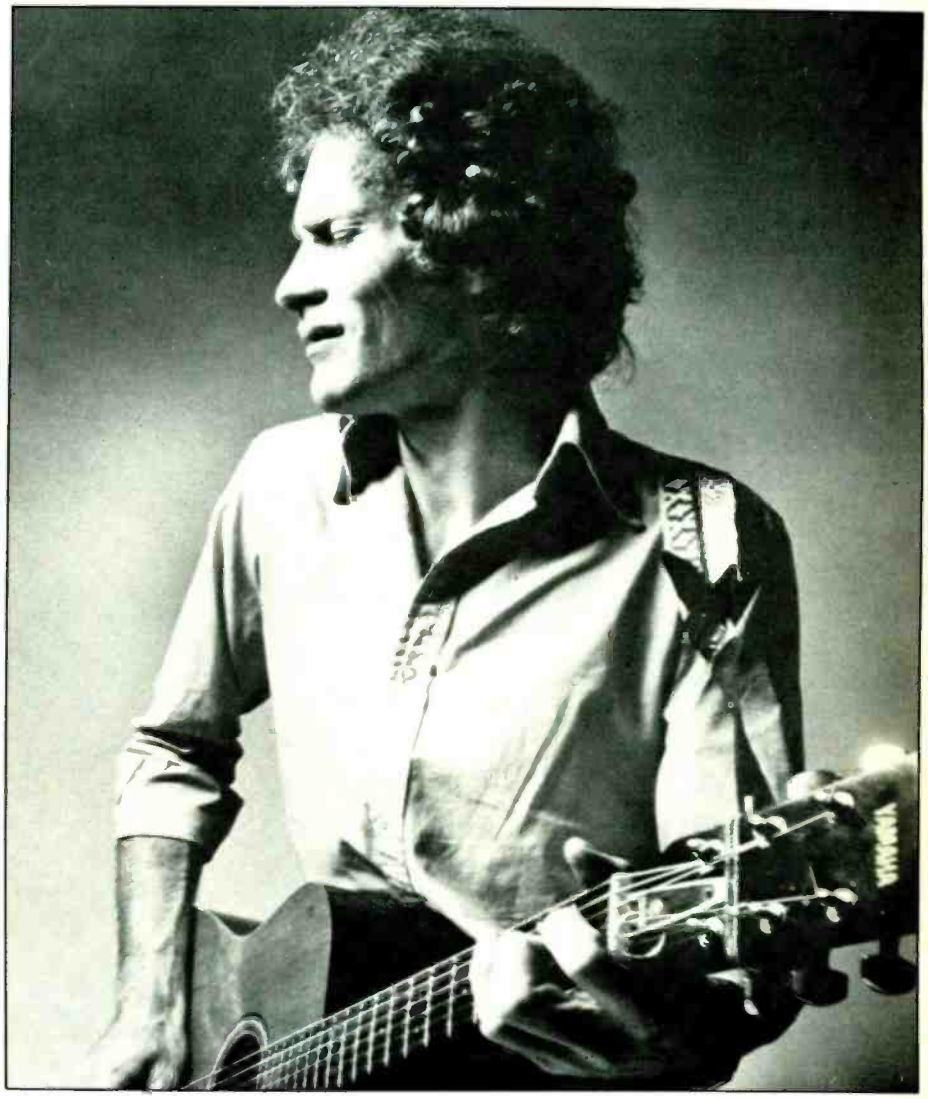
TONY BIRD: *Tony Bird.* [Tom Wilson, Larry Fallon, producers; Bernie O'Gorman, Damian Korner, engineers; record-

ed at CBS studios, London.] Columbia PC 34324.

Performance: **Taj in the Caribbean**
Recording: **Basic**

Well, there' nothing new going on here. Vocally, he's too close to Taj Mahal's style for comfort—or interest. Although judging by his accent it sounds as though he has claim to that sound by birth, compositionally, his material is of the South African folk song variety.

Mix-wise, Fender Rhodes is panned from left to right very quickly. The drums, composed of snare and hi-hat for the most part, are in the middle along with the bass and vocal. Electric guitar is on the right. Solos replace the voice in the mix. Maybe upon repeated listening you can get used to a voice that has a Jamaican inflection and delivery without the reggae beat, some blues lyrics, the basic three or four chord folk changes, and the vintage Clapton solos a la the *Disraeli Gears* album thrown in for effect. I have no doubt as to the authenticity of the music as can honestly be contested with many a rock album. It's just that it sounds like too many other stylists for me to get excited. G.P.



TONY BIRD: Nothing new or exciting

THE METERS: *New Directions.* [David Rubinson and Jeffrey Cohen, producers; David Rubinson, Fred Catero and Chris Minto, engineers; recorded at the Automatt Automated Recording, San Francisco, Ca.] Warner Bros. BS 3042.

Performance: **Very metric**
Recording: **Nearly perfect**

The Meters were playing funk long before the vernacular became associated with disco bands who can't say anything but "ow, yow" over a thudding bass and

THE METERS: Intensity and spunk

high hat. Who can forget "Cissy Strut," one of the great soul instrumentals of the sixties?

While the Meters, circa 1977, have been influenced a *bit* by the slime around them, happily they have managed to hold fast to previous definitions of drive and rhythm and not be dragged down by most of the idiom. In fact, the infusion of the peerless Tower of Power horns adds even extra punch to the Meters' long established intensity and spunk.

Several tracks reflect the Meters at their best. "No More Okey Doke" is (despite an overmixed drum channel) hypnotizing with its evil clavinet lick; the several reeds of the Tower plus additional guest Kurt McGettrick give "I'm Gone" a fifties' r&b charm, and the inspirational "Give It While You Can" is another example of the superb integration between the conscripted brass and the guitar/bass backbone of the protagonists.

Some other numbers exemplify other departures. Reggae is tackled with a loyal reading of Peter Tosh's "Stop That Train;" the rhythm section emulates reggae timings well. Yet that same resourceful subunit of bassist George Porter and drummer Zig Modiliste goof off on "Funkify Your Life," which, in its five minutes and 39 seconds, repeats "funk," or derivations thereof, fifty-six times. Oh, well, maybe it's a spoof, yet such a short aberration should not mar an enjoyable recording such as this. R.S.

JAZZ

EDDIE KIRKLAND: *Front and Center.* [Eddie Kirkland, Pete Lowry, producers; Pete Lowry, engineer; nc studio credited.] Trix 3301.

Performance: **Your basic blues**
Recording: **Your basic living room recording**

I am not a blues fan, and this album did nothing to change that. Other than being ethnic folk music and an early influence on rock, I find little redeeming value in it. Obviously, some people appreciate it, and are wondering, with justification, who the hell am I?

This is *the* basic stereo recording featuring just guitar and voice, occasional harmonica and one track with percussion and additional guitar. In terms of a recording attempting to capture authenticity, this blues record comes closest to the vrai for it is recorded in the most natural of settings—anywhere—for the blues originated on porches and sitting rooms and were played on the cheapest of instruments. Even a few mistakes were left in, going for feel rather than perfection. This recording offers the true blues sound, previously only captured on worn 78s, and I think that's as important a step in recording as was the introduction of multi-track recording. G.P.

CORYELL/MOUZON: *Back Together Again.* [Larry Coryell, Alphonse Mouzon, producers; Ron Johnson, Jay Krugman, engineers; no studio credited.] Atlantic SD-18220.

Performance: **Coryell at his finest**
Recording: **Teaching school on guitar mixes**

This is Larry Coryell's most crystalized playing, and I am not a Coryell fan. I am, however, a fan of European guitarist Philip Catherine who appears on this album and should be given a larger share of the credit for the success of this LP for he gives Coryell enough backing, room and challenge so that Coryell plays like he never before has. The real name of this record should be *Coryell/Catherine, For The First Time.*

If you like guitar and are interested in how many creative ways you can arrange multiple guitar tracks, this is the mix for you. On the average, there are three separate main guitar tracks, going as high as five on some cuts. There are alternating electric lead, acoustic lead, electric and acoustic rhythm, electric solo, etc. One track will have Coryell playing lead in the left of the mix, Catherine acoustic rhythm in the center and Coryell soloing in the right, with bass and drums in the center. Another track will have both guitarists in the center, playing electric lead, then move them around as they overdub other guitar parts. Space does not allow me to do justice here to the ten tracks, for each is both different and educational mix-wise. Mouzon's contributions I find wanting, although with the exception of one needless drum solo he does not get in the way. Granted, his "vocals" (one consisting of repeating only the song title, "Reconciliation," every four bars) are laughable.

The only blemish is accorded to Atlantic for their packaging concept, or lack thereof, to say nothing of not mentioning the studio. With all the interesting

CORYELL/MOUZON: Crystalized

mixes contained herein, the label should have either described them or enclosed a chart (a la A&M's Horizon jazz label) to enable the listener to discern who's playing what, when. Other than that, it's about time, Coryell. —G.P.

CHARLES MINGUS: *Three Or Four Shades Of Blue.* [Ilhan Mimaroglu, producer; Bobby Warner, Carmine Rubino and Don Puluse, engineers; recorded in New York City, 1977.] Atlantic SD 1700.

Performance: **One of those classic Mingus recordings**
Recording: **Slick, smooth and shiny**

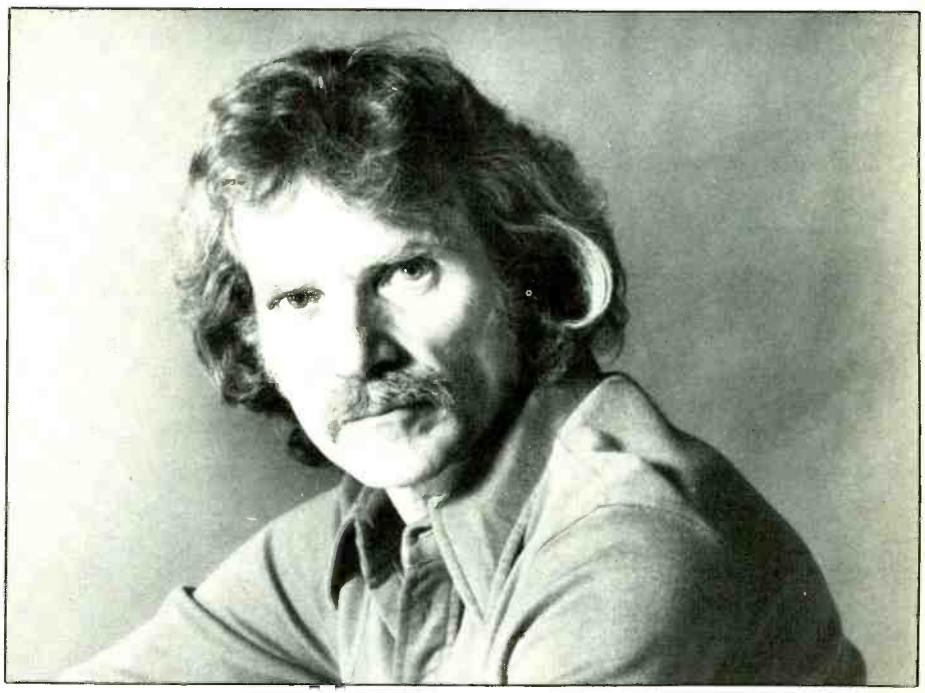
Every so often Charles Mingus makes a record. Much of the time it's simply a sound portrait of the current working Mingus band. The rest of the time it's something special. *This is something special*—one of those records they'll be writing about in the jazz history books.

In addition to the current working Mingus band there are three jazz/rock fusion guitarists (Larry Coryell, Philip Catherine and John Scofield); two avant garde saxophonists (George Coleman and Sonny Fortune) and former Woody Herman pianist Jimmy Rowles. In addition to the new Mingus Music there are redefinitions of "Better Get Hit In Your Soul" and "Goodbye Pork Pie Hat." But the gem of the group is the new major Mingus work, "Three Or Four Shades Of Blue." The work, which runs twelve minutes plus, is subdivided into eleven segments which Mingus has provided with thoughtful subtitles. It's not that Mingus has written program music in the sense that Richard Strauss wrote program music—the subtitles are simply signposts to remind the afficionado of familiar sounds or to help the neophyte to catch up on the homework he's missed.

The recording is typical of many jazz records these days. All the rough edges have been honed down to the kind of product that's just the opposite of some of the rough and tumble in-person performances you'll remember hear-

CHARLES MINGUS: Something special

ing from any jazz giant. Yet each has its place. The slick recording with every musical hair in place and the "live" excitement of musicians taking musical chances are both part of the same picture. They're both part of the same Charles Mingus. J.K.



DAVID AXELROD: Bigger than life

DAVID AXELROD: *Strange Ladies.* [David Axelrod and Earl C. Palmer, producers; Jay Ranellucci, associate producer and engineer; recorded at Capitol Studios, Hollywood, Ca.] MCA 2283.

Performance: **Personal confusion**
Recording: **Splashy and flashy**

Sometimes it seems to me that the idea of a fusion group is to mix so many elements together that it becomes a jumble. However, there are few who've managed to mingle as many styles of contemporary music as David Axelrod has here. A lot of it's rock but there's latin and disco, too. First and foremost it's a jazz album with fine jazz solos by Bobby Bryant on trumpet and Ernie Watts on flute and saxophone. But it's Jimmy Cleveland's magnificent trombone work that makes "Tony Poem" the standout track of the LP. Yet it is perplexing to find oneself listening to Cleveland's wailing bone solo and a rockin' rhythm section in such close proximity.

The recording, like everything else that comes out of Hollywood, is bigger than life. At a decent volume the sounds come out and stab you with their realism. Since David Axelrod came from the other side of the glass booth (where he served as producer for such diverse performers as Cannonball Adderly and the Electric Prunes) it's only natural that he should be well aware of the things that can be done with modern recording technology and it's only natural that he composes and arranges with all the possibilities in mind.

The only problem is that it's a highly personal album with six tunes written for and/or about six real ladies. Since the tunes are all originals there's no familiar thread to hang onto. It's very personal and the listener's enjoyment may depend on how far he is willing or able to put himself into the composer/arranger/performer's head and feel his feelings about these ladies. J.K.

THE NEW BLACK EAGLE JAZZ BAND: *In Concert.* [Nathaniel Johnson, Monte Drake and Walter Schwab, engineers; recorded between 10/29/73 and 9/26/74 in Boston, Ma. and New Orleans, La.] Black Eagle BE 2.

Performance: **True to tradition**
Recording: **Best in Boston... less so in Louisiana**

The best thing about a good revival band, and this *is* a good revival band, is that they keep the traditional repertoire alive. There aren't many bands around today that could cut an LP including "Wild Man Blues," "Skid-dat-de-dat"

THE NEW BLACK EAGLE JAZZ BAND: Keeping the jazz repertoire alive

and "I'm Travellin'." There are certainly few bands who play as excellent traditional jazz as you'll hear from the New Black Eagle Jazz Band, especially the crisp lead cornet of Tony Pringle, the New Orleans filigree clarinet work of Stan McDonald and the rock-bottom tuba playing of Eli Newberger. These guys all have regular day jobs varying from library director to pediatrician but they all manage to get out of the house on Thursday nights to play at the Sticky Wicket Pub in Hopkinton, Massachusetts. They also do concerts and broadcasts and it is from these performances that this LP eminates. The entire first side was recorded at WGBH in Boston and the quality is up to what one would expect of good current FM radio sound.

The second side was mostly recorded in concert at New England Life Hall in Boston and, while the sound is not up to those selections recorded at WGBH, it's considerably better than "Black Eagle Skuffle" which opens side two and was recorded at The Stage Door Canteen in New Orleans, Louisiana. It's a shame the sound is lacking because "Black Eagle Skuffle" (the New Black Eagle Jazz Band's reworking of "I'm Dancing With Tears In My Eyes Cause The Girl In My Arms Isn't You") is one of the most exciting traditional band performances I've heard in years—but maybe that's why it was included even though the sound wasn't up to par.

Don't expect to find this recording on sale at your local store that caters to the top twenty crowd. If there isn't a shop in your area which specializes in jazz records this LP can be ordered from The New Black Eagle Jazz Band, 128 Front Street, Marblehead, Ma. 01745.

J.K.

JO JONES: *The Main Man.* [Norman Granz, producer; Bob Simpson, engineer; recorded Nov. 29 and 30, 1976 at RCA Studios, New York.] Pablo 2310-799.

Performance: **The finest**
Recording: **Present and in focus**

Contrary to what Norman Granz says in the liner notes this is *not* the first recording under Jo Jones' leadership—just his first in too long a time. Papa Jo has gotten himself a fine band of all-stars for this LP, with giants like Harry Edison and Roy Eldridge on trumpets, Vic Dickenson on trombone, Lockjaw Davis on tenor sax and a rhythm section that includes Tommy Flanagan on piano, Sam Jones on bass and Jo's former partner from Count Basie's All American Rhythm Section, Freddie Green, on guitar. How could this band be anything but outstanding? It's a relaxed date and nobody gets in anybody's way and everybody gets a lot of space to blow. I was particularly impressed with Vic Dickenson's playing throughout the LP. I've never heard him in better form.

The engineering is straight and honest. The first thing to strike me is the presence, especially in the case of the piano and guitar. There's also a tight focus on the instruments when they solo. They seem firmly placed in space

rather than being panned out between the two speakers.

Except for Jo Jones' original composition "Adlib" (which is pure and simple blues) and Budd Johnson's tune, "Metrical Portions," the material is standard jam session fare ("Goin' To Chicago," "I Want To Be Happy," "Dark Eyes" and "Old Man River," to name a few) and this is the kind of band that can really jam on them. As for Jo Jones' drumming, he's still the driving Tiger that he was with Basie's band. His solos may not be the steadiest time-wise, but then solos never were Jo's strong point.

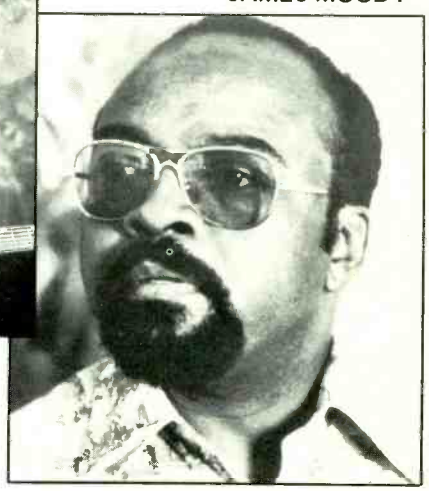
BUNKY GREEN

ELVIN JONES

CLARK TERRY

ROLAND PRINCE

JAMES MOODY



The important thing about the way Papa Jo Jones plays the drums is that he makes a band swing like a banner in a hurricane—and *that* he can still do. J.K.

ELVIN JONES, JAMES MOODY, CLARK TERRY, BUNKY GREEN AND ROLAND PRINCE: *Summit Meeting.* [Ed Bland, producer; Charlie Repka and Jeff Zaraya, engineers; recorded November, 1976, at Vanguard Studios, New York City.] Vanguard VSD 79390.

Performance: **Certainly a summit, but not quite a meeting**
Recording: **Clear horns but the rhythm's a bit lost**

Records which match players like Clark Terry on trumpet and flugelhorn, James Moody and Bunky Green on saxophone, Roland Prince on guitar and Elvin Jones on drums would have to be called a summit. There's no jazz fan in the world who needs to be reminded of the credentials of these players. Since they are all Vanguard recording artists, this LP is as much a sampler as it is a super-session. In that it will serve to introduce Bunky Green to Clark Terry fans and to introduce James Moody to Elvin Jones fans it's an admirable concept. In that there's very little that the avant-garde playing of Bunky Green and Roland Prince have in common with the mainstream horn work of Clark Terry, it's not such a good idea. There are simply too many places where the ins and outs of each player don't quite get together on what's going down.

Producer Ed Bland's idea to bring charts to the session accomplished just what he wanted it to. It sounds more like a well-rehearsed combo than a jam. There are tunes where Moody is featured or Elvin Jones gets to tear loose. These tunes are set to highlight that particular individual's way of blowing rather than lining them all up in the studio with no paper and giving them standard tunes to jam on. As for those who want to hear Clark Terry and James Moody blow on standards—there are plenty of records where they do just that. The horns come through crystal clear. I don't think Terry has ever been better represented on records and I find his playing, especially on his own "Tee Pee Music," the best I've ever heard him do. For some reason, the rhythm section doesn't come through with the same bite and clarity that the horns do—which is a particular shame since pianist Albert Dailey is one of the truly underrated musicians on the New York Jazz Scene today. J.K.

CLASSICAL

ELGAR: *Violin Concerto, Op. 61.* Pinchas Zukerman, violin; London Philharmonic Orchestra, Daniel Barenboim cond. [Paul Myers, producer; Robert Gooch, Mike Ross-Trevor, engineers.] Columbia M 34517.

Performance: **Emotional**
Recording: **Rich and well-balanced**

ELGAR: *Cello Concerto, Op. 85; Enigma Variations, Op. 36.* Jacqueline DuPre, cello; Philadelphia Orchestra (in the *Concerto*) and London Philharmonic Orchestra (in the *Enigma*), Daniel Barenboim cond. [Paul Myers, pro-

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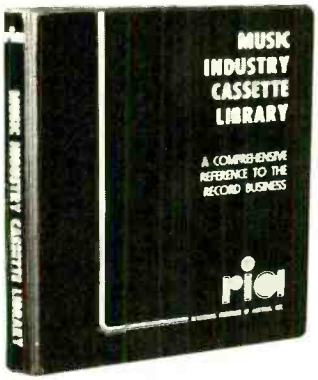

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ducer; Robert Gooch, Mike Ross-Trevor, engineers (in the *Enigma*).] Columbia M 34530.

Performances: **Romantic**
Recordings: ***Concerto*, adequate "live" pickup; *Enigma*, solid but monochromatic**

Daniel Barenboim has been recording much Elgar for Columbia lately, but not to my liking until these two discs. The *Violin Concerto*, in fact, is now my preferred version, surpassing the notable 1965 Menuhin/Boult account on Angel. Zukerman's playing is more technically secure than Menuhin's and Barenboim's expansive emotionalism really works to the benefit of the music this time. While Zukerman receives a closer balance than the ideal concert-hall perspective of Angel's engineering, the full, rich sonority of the LPO is never in the shade.

Some may still prefer the tighter, more structural approach of Menuhin and Boult (in 1932, the 16-year-old violinist had recorded the concerto with the composer himself). And a new London recording by Kyung-wha Chung and Solti with the LPO should be available by the time this review reaches print. But this new Columbia release should be heard by everyone interested in English music. Surfaces were fine.

Barenboim's "live" concert recording of Elgar's *Cello Concerto* with his wife, Jacqueline DuPre, and The Philadelphia Orchestra is another committed performance that never oversteps the bounds into mawkishness. DuPre indulges in considerable sliding from one note to another, to beautiful effect. The sound is only passable, however, and some may understandably prefer her studio recording with Barbirolli (the British EMI pressing, coupled with Delius' *Cello Concerto*—not the Angel, which contains encore pieces on side 2).

The *Enigma Variations* overside, Elgar's most popular work (except for the first *Pomp and Circumstance* march), is one of Barenboim's more straightforward readings, taken down in good, if somewhat monochromatic, sound. *Tuttis* are solid and impactive, but lightly scored passages are bland (a feature of the orchestra, perhaps?). Haitink's dynamic Philips recording with the same orchestra avoids this problem, though. Surfaces were bad. S.C.

BRITTEN: *Suite on English Folk Tunes; Four Sea Interludes, Op. 33a* **and** *Passacaglia, Op. 33b* **(from "Peter Grimes").** New York Philharmonic, Leonard Bernstein cond. [John McClure, producer.] Columbia M34529.

Performances: **Affectionate**
Recordings: **Satisfactory**

BUTTERWORTH: *A Shropshire Lad; The Banks of Green Willow; Two English Idylls.* **BRITTEN:** *Variations on a Theme of Frank Bridge, Op. 10.* Academy of St. Martins-in-the-Fields, Neville Marriner. [Chris Hazell, recording procedure; Stan Goodall, Simon Eden, engineers.] Argo ZRG 860.

Performances: **Spot-on**
Recordings: **Superb string sound**

Bernstein's "Tribute to Benjamin Britten" contains the first recording of the recently deceased composer's *Suite on English Folk Tunes*, a work similar to late-period Copland in its pleasant but trifling inspiration. The performance seems fine and the recording occasionally excessively spotlighted.

The other side contains orchestral excerpts from his first and most enduring opera on this side of the Atlantic, *Peter Grimes*. Bernstein introduced this opera to America in 1946 at Tanglewood, and unsurprisingly conducts the evocative *Four Sea Interludes* and *Passacaglia* with atmosphere and drama (although his doubling of tempo for section 10 in the final interlude, Storm, is a wrenching mood-breaker).

The recorded sound is more spacious, with less glare and more solid bass than most McClure/Philharmonic productions. Apart from some bad string ensemble in the faster sections, this performance is preferable to Previn's recent, rather bottom-heavy rendition, murkily mastered by Angel. However, the Guilini recording (also on Angel, but recently reissued in England in a remastering with much more impact—one *feels* the bass drum as well as hears it), coupled with a virtuosic *Young Person's Guide to the Orchestra*, remains a perfect intro to Britten's orchestral genius.

Most English composers in this century seem better able to work in small forms. Only Elgar, Vaughan Williams, Walton and, perhaps, Tippett have made lasting contributions to the symphony, while Britten's extensive output was primarily vocal and operatic. Such composers as Bax, Butterworth, Delius, Holst, Ireland and Moeran have composed miniatures which fully deserve to turn up more frequently in the concert hall.

The Argo disc is a sterling example, coupling one of Britten's rare non-vocal works and four short pieces by George Butterworth, a promising composer who was killed at age 31 in World War I. Marriner's Academy is up to its usual superb form and the engineers have provided one of the most astonishingly vivid recordings of strings in the catalogue. S.C.

INDIGENOUS

MAGMA: *Udu Wudu.* [Producer unnamed; Jean Paul Malek, Alain Francais, engineers; recorded at Studio de Milan, Paris.] Tomato Records TOM-6001.

Performance: **It's, uh, different**
Recording: **OK**

This is a *strange* record. French drummer Christian Vander, the main creative force behind Magma, discovered that as a language, French isn't adaptable to his needs, so he invented his own. Got the plot? It's like Star Trek meets Kraftwerk and together they loose you in the Ozone. Musically, there's a lot going on. Jazz, salsa, progressive synthesized rock, middle Eastern influences are all tastefully intertwined. With a varying supporting cast that has a core of percussion, keyboard and vocals, each song is recorded differently so that there is no common approach to the mix.

As the liner notes needlessly point out, this type of music hasn't created much of a following yet in this country. European tastes might be somewhat more attuned to futuristic works since Europe has been producing (both in terms of actual styles and material, as well as recording techniques) some of the most adventuresome records of the decade. That can only be attributed, I feel, to the fact that most American or English engineers cut their teeth on rock sessions and therefore all come at engineering from the same direction. European engineers with less rock available in their studios, work more varied projects and, therefore, bring other techniques to rock engineering. If you enjoy experimental music, something along the Germanic line, and can overlook the vocal content, check this out. G.P.

Continued From Page 16

top-name groups using them so something good must be happening up there.

Again—an excellent mag and keep up the great work.

—Dennis Brunet
Toronto, Canada

If you had caught the Cooper article, "How Acoustics Affect Recording," in the July, 1977, issue, you would have seen that his book is being published by the Recording Institute of America, 15 Columbus Circle, New York, New York 10023 and should be appearing very soon. Copies will be available by mail from them once it's out.

And thanks for the report of the good things happening up there in Toronto.

The End of A Beautiful Relationship?

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—Jim Lett
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Some Graphic Memories

I don't know if it was my general good mood that day or what, but Dave Moyssiadis' excellent "Disc Mastering" (August and September 1977) was very much enjoyed. That guy can write both a useful *and* entertaining article.

The advice in Part II relating how to produce a tape that's fit for the mastering room is also applicable to mixdowns to plain ole' cassettes. I found myself relating to many situations that were described, especially to the use of EQ.

The comments about low-end "building demolition" and about controlling sibilance on vocals and cymbals had me laughing out loud. Memories came back of trying to explain a distorted, unbalanced mix as too much EQ—the result of an artist not being satisfied with the mix and literally taking matters into his own hands (namely, the graphic equalizer). A few garish mixes later finds this individual convinced of the very fine line between natural-sounding EQ and that which is not so natural.

It's been a year and about a hundred mixes since I installed the equalizer in my studio. I've learned that finesse is the name of the game and that the smoother the curve portrayed on the unit, the more natural the results. I also use the equalizer to reduce noise. It is especially helpful here since all my productions are eight to twelve tracks "ping-ponged" on a four-track machine. My noise reduction method consists of recording somewhat "hot" and using the equalizer to add a treble boost which is later cut by the same amount. When I mix down to stereo, I use a gradual boost in the upper mid-range and a gradual cut in the 160-500 Hz region plus adding a little "kick" at 90 Hz to bring out the kick drum. Using this technique I can achieve clean, natural mixes with plenty of clarity and presence. The low end is tight and can be listened to at higher levels with a minimum of listening fatigue.

I learned a lot from Dave Moyssiadis' article and hope that we see more of him in future issues. And you guys at *MR* deserve some kind of award for your fine magazine. Please keep up the "in session" articles—they're *great!*

—Walter Kerr
Seattle, Wa.

Early Warrior Remembered

As a steady reader of *Modern Recording*, I find it a consistently informative, helpful and authoritative magazine. I think you people are doing a fine job.

However, in the Letters to the Editor column of the May, 1977, issue, you told Todd Nelson ("Hunting Warriors," page 8) that the Jade Warrior albums on Island Records included *Floating World*, *Waves* and *Kites*. Please let's not forget their first album, *Released*.

—Roy Peak
Aurora, Co.

Advertiser's Index

CLASSIFIED ADS

A 140-page comprehensive directory listing names, addresses, and phone numbers of every major record company, publishers, booking agents, managers and independent record producers. Also, sample contract forms for each. All for $4.95. R.I.A., 15 Columbus Circle, New York, N.Y. 10023.

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OTARI MX-5050 QXH 4-channel Professional Tape Deck. 7½/15 ips. Includes instruction manual, XLR-to-phono adaptor cables, hub adaptors. Excellent condition. S. Sarper, 64 West Broad St., Stamford, Ct. 06902.

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