MAY 1977

VOL. 2 NO. 5
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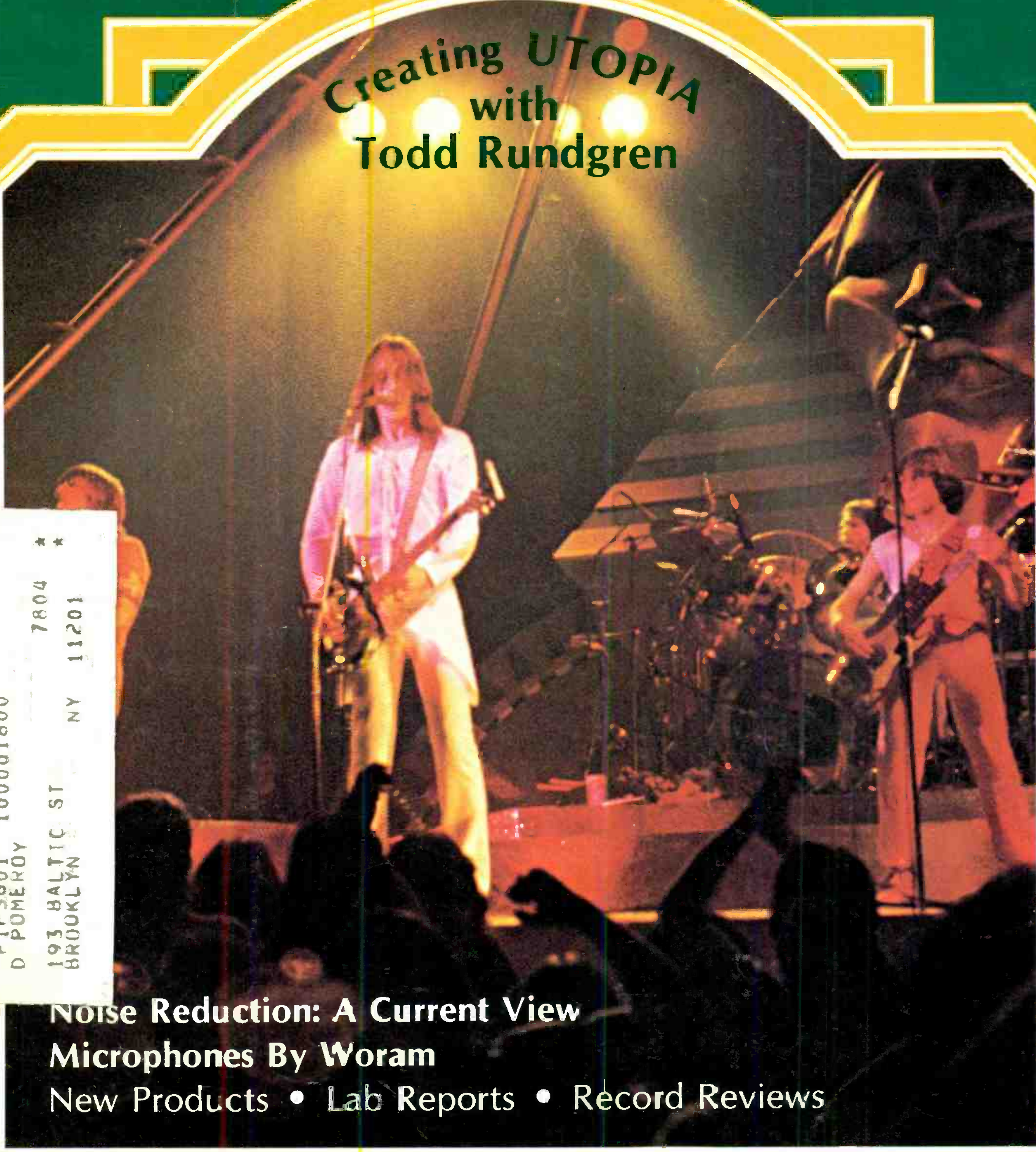
MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY



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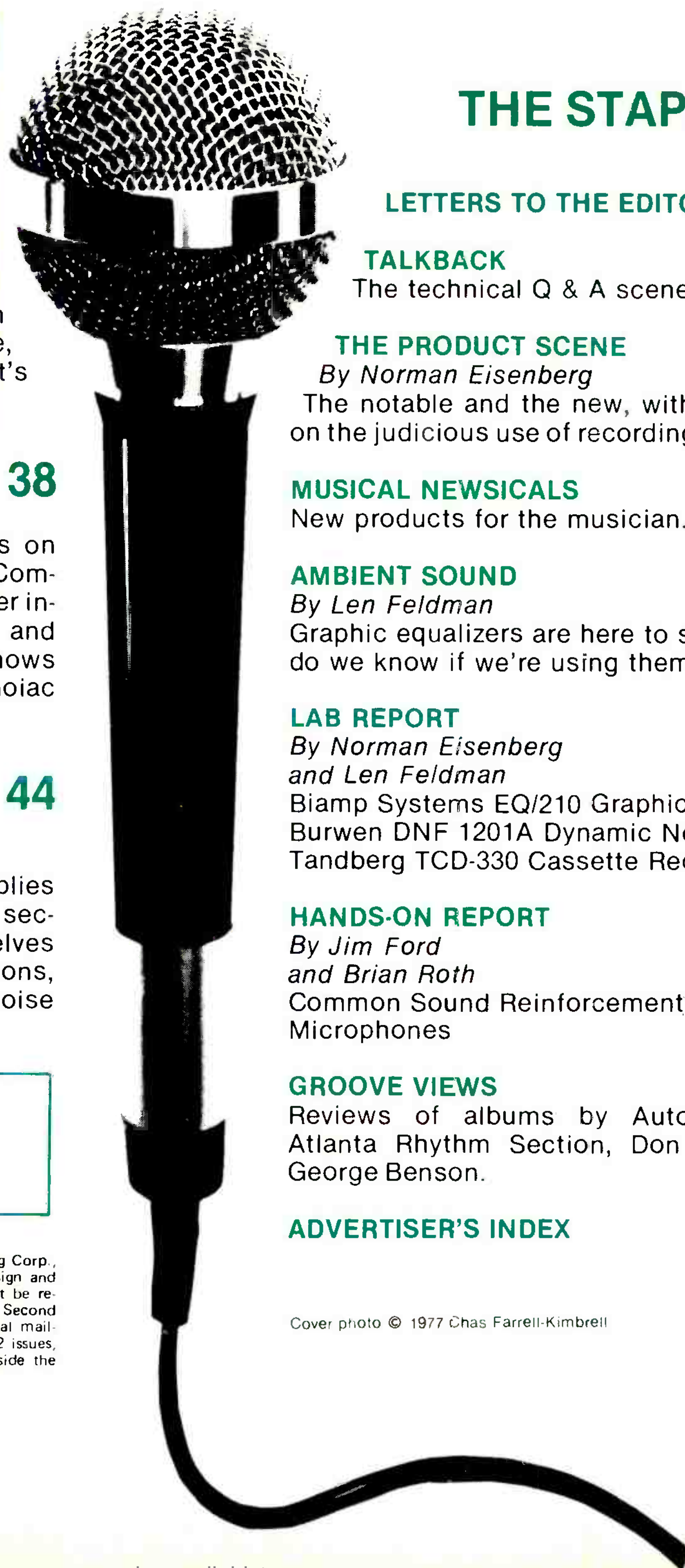
CIRCLE 31 ON READER SERVICE CARD

MAY 1977
VOL. 2 NO. 5

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY



THE FEATURES

COMING NEXT ISSUE!
Noise Reduction: Part II
Direct-to-disc Recording

MODERN RECORDING is published monthly by Cowan Publishing Corp., 14 Vanderventer Ave., Port Washington, New York 11050. Second class postage paid at Port Washington, New York and at additional mailing offices. Subscription rates: $7.50 for six issues, $13.50 for 12 issues, $18.00 for 18 issues. Add $3.00 per year for subscriptions outside the U.S. Subscriptions must be paid in American currency.

THE STAPLES

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MODERN RECORDING
SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

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Editorial contributions should be addressed to The Editor, Modern Recording, 14 Vanderventer Ave., Port Washington, N.Y. 11050. Unsolicited manuscripts will be treated with care and must be accompanied by return postage.

Letters to the Editor

Reading Amateur

I am interested in learning about the field of recording and sound reinforcement, mostly as an amateur, rather than as a possible career. I am writing to ask you to recommend some good books on the subject.

I know virtually nothing, so the book would have to begin at a non-technical level. From there, I hope to learn as much as I can on the different pieces of equipment, how they are used, etc. Thank you for your time & help.

L. Allen Ebert
Winchester, Va.

For a listing of good recording books, see MR issues Oct/Nov 1976 and April 1977 in the Talkback Sections.

Info on Turntables

Perhaps you could help me out. I'd like any information you might have on turntables lying around the office. Sure, I could go to a popular high fidelity magazine, but then I would get bombarded with information about equipment that is less than state of the art. Since we started receiving your publication, the quality of our P.A. work has taken an upward swing. Again, thanks for the information.

The boys are itching to read up on most any equipment they could possibly use in advancing their work so could you send that along also.

—C.A. Rynski
Technical Director
Student Union Board
University of Toledo
Toledo, Ohio

We don't do many reports on turntables so we have no information lying about. Our best suggestion is to write directly to the manufacturer of the turntable you are interested in. We suggest that you aid your boys with a subscription to Modern Recording *which will surely provide them with enough information for recording advancement.*

You Ask, We Help

Would you please send me a spec sheet of your commercial ad costs and rates? Also, could you please provide me with the correct mailing addresses to the music-trade publications listed: *Billboard*, *Cashbox*, and *Record World*.

I have seen you mention these magazines in your great publication, but I have been unable to obtain the correct mailing addresses to all three.

Also, I would like to request the name, address, costs, etc., and details of the RIA, Modern Recording Techniques course offered by the nearest affiliated studio. I have written to the address seen in MR to get all the details but to no avail!!

—Charles D. Burch, Sr.
Buffalo, N.Y.

THE PORTABLE SOUND SYSTEM COMES OF AGE.

1000S

200 Watts rms per channel
10 low & 10 high impedance inputs
20 Hz to 20 kHz response
Less than 0.1% THD

700S

120 Watts rms per channel
7 low & 7 high impedance inputs
20 Hz to 20 kHz response
Less than 0.1% THD

Peavey

Introducing the PA 700S and PA 1000S,...professional stereo mixer/amps from Peavey. Anything a soundman can do with a multi-channel stereo mixer, two 9-Band graphic equalizers and two power amps can be done with these super versatile systems.

Features such as variable input attenuation, monitor send, high and low EQ, effects send, stereo pan and output level slider on each channel, along with two 9-Band graphic equalizers, master section, VU meters, and a complete rear patch panel including low and high impedance inputs, put you in full command of the system.

All you do is add the speakers and mix!

For complete information, see your Peavey Dealer or write us and we'll send you a free catalog.
Peavey Electronics, Corp. / 711 A Street / Meridian, Mississippi 39301

CIRCLE 37 ON READER SERVICE CARD

An MR spec sheet of commercial ad costs and rates is on the way to you. Here are the music-trade publications mailing addresses you requested: Billboard, *One Astor Plaza, 1515 Broadway, New York, N.Y. 10036,* Cashbox, *6363 Sunset Blvd., Suite 930, Hollywood, Ca. 90028, and* Record World, *1700 Broadway, New York, N.Y. 10019. RIA informs us that they are very busy with requests similar to yours. They will be forwarding the information to you as soon as possible, but be prepared. . . there is no affiliated studio in Buffalo.*

Trade Journals

Please tell me also where I can get copies of "Studio Sound" magazine. (I already subscribe to *Modern Recording* magazine.) If there are professional trade journals for the industry, please forward a list of these.

—Steven D. Nijak
Rockville, Md.

For a list of trade journals, see Talkback, "Your Own Studio."

Reader's Suggestions

How about concentrating a little more on pro and semi-pro gear rather than hi-fi? Also, the suggestion of comparative tests (several different makes, price ranges of reverbs, ¼-track etc. at one time seems to make sense to me).

I'm enjoying *Modern Recording* nonetheless!!

—P.K. Almeter
Batavia, N.Y.

MR does not consider any of its reviews to be hi-fi oriented. Equipment like professional power amplifiers, equalizers and four-track tape recorders can not be considered basic hi-fi gear.

For price comparisons, please refer to the MR Buyer's Guide and check our new staple "Hands-On Report" in which we'll be lab and *field testing equipment.*

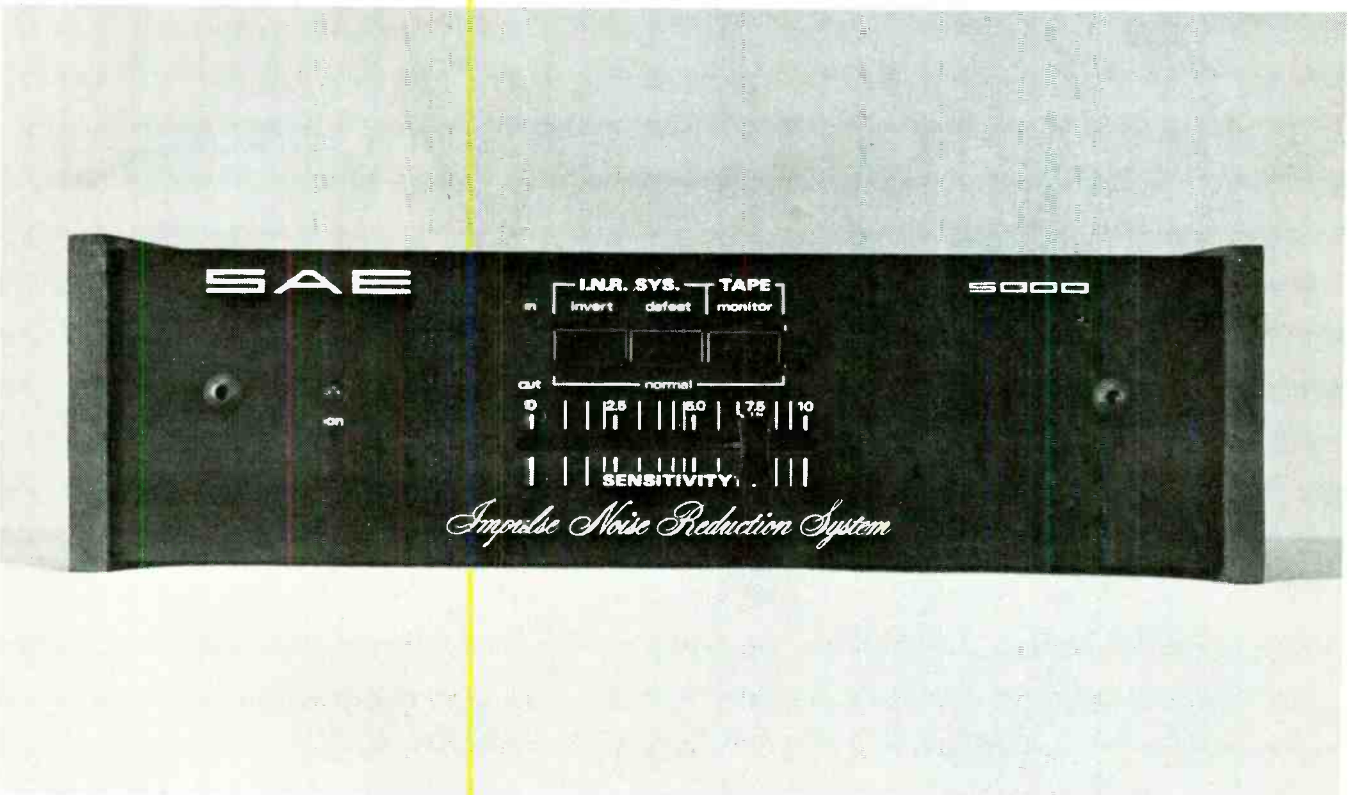
JBL Disagrees

Rob Lewis' "Monitors for the Recordist" contains many admirable points, but I strongly disagree with his positive assessment of narrow-band, peaked-response monitors.

Run-of-the-mill home equipment is far better today than it was ten years ago. One can plainly hear tape and/or cutter overload (resulting from excessive highs) on many home loudspeakers. One can also plainly hear air conditioning noises, turntable rumbles, room resonances and

The "Click and Pop" machine

only by SAE



Ever since the invention of the recorded disc annoying "clicks" and "pops" caused by scratches, static and imperfections have consistently disturbed the listening pleasure of music lovers.

Now, SAE introduces the unique model 5000, an Impulse Noise Reduction System which eliminates those unwanted sounds with no adverse effect on the quality of the recorded material.

This breakthrough in electronic circuitry is so demonstrably effective that the SAE 5000 is destined to become an essential part of any sound system.

The SAE 5000 is compact and sleek, built to SAE's exacting standards, and ready to enhance the performance of any system, from the standard receiver/turntable combination, to the most sophisticated audiophile components.

SAE is proud to add the 5000 to their broad line of *Components for the Connoisseur.*

SAE

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Please send more information on the 5000.

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CIRCLE 93 ON READER SERVICE CARD

the like on many current home units. These are inaudible on monitor designs which do not have extended response at both ends of the frequency spectrum.

Additionally, a monitor with elevated midrange response will result in tapes which are deficient in midrange vocal presence—the tape will exhibit the inverse of the speaker's response curve—unless the mixer can "guesstimate" a suitable correction factor.

Further, equalization is of limited help in restoring missing frequency extremes, particularly at the low end of a ported system's response. When a conventionally tuned reflex system operates below vent resonance, response rolls off at about 24 dB/octave and phase rotates rapidly. Bearing in mind that 10 dB represents a tenfold power increment, one can see that both woofer and amplifier power capacity will be quickly exceeded with little useful result if equalization below system resonance is attempted.

It seems odd that the most subtle differences in signal processing equipment are hotly debated and the last drops of distortion and noise are squeezed from electronics and tape machines, yet many people continue to depend on twenty-year-old monitor designs which are not capable of reproducing most of these electronic subtleties.

—Garry Margolis
Applications Engineer
Professional Division
JBL Sound, Inc.
Northridge, Ca.

Hunting Warriors

Could you please tell me which label Jade Warrior records on and possibly the name or names of their albums as seen in your Feb/Mar 1977 issue. I've been trying to find out for quite a long time.

You have a great magazine!

—Todd Nelson
Ventura, Ca.

Jade Warrior records on Island Records as listed in the Groove Views heading in the Feb/Mar 1977 MR issue which you have already noted. Jade Warrior has recorded three albums: Floating World, Waves, *and* Kites.

Upcoming Lab Reports

Could you do a Lab Report on the Tascam series model 80-8, half-inch, reel-to-reel recorder in an upcoming issue?

Great magazine, keep up the good work.

—Eric J. Brown
Schenectedy, N.Y.

We are seriously considering doing a Lab Report on the Tascam 80-8 in an upcoming issue. However, the machine has caused such a stir that TEAC is quite busy filling orders. When things at TEAC calm down, we will do our best to have a Lab Report as soon as possible on the 80-8.

Good Idea

Began receiving our subscription today with the Vol. 2, No. 3 issue that was long awaited. Thank you for getting us back on the list of subscribers. We will be looking forward to receiving back issues of MR since we've missed them dearly, and of course, all forthcoming issues. Glad to hear about the monthly format. We'll need more info about adding to our subscription.

It would be nice to see information geared to interpreting manufacturers specs and seemingly foreign abbreviations. I know most of this sort of thing can be found in Audio Cyclopedia and various handbooks, but it would be nice to have it documented for quick reference. This could also apply to consumer stereo type spec sheets on tuners and the like.

—Don Schott
Monitor Music Central
Cincinnati, Ohio

For a magazine with our format, it would be difficult to draw up and include a directory of terms. However, your idea is well taken and will be set on the drawing board.

If you subscribed when MR was bimonthly, your subscription will still cover 6 months of issues. With MR going monthly, those 6 months will only cover a half a year. So to cover an entire year, you will have to renew your subscription after 6 months.

Back Issue Request

Congratulations on a very useful magazine. I look forward to every issue. Thanks for going monthly!

How about making some back issues available?

—Richard C. Vonderlinn
Greenlawn, N.Y.

At present the only back issues available are Jun/Jul 1976, Dec/Jan 1977, Feb/Mar 1977, and the MR Buyer's Guide. *To receive back issues, simply write to our Subscription Department and request them at $1.75 per copy.*

JVC builds in what other receivers leave out. A graphic equalizer.

The only way you can equal the realistic sound capability of JVC's modestly priced S300 stereo receiver, is by adding an expensive, but highly versatile graphic equalizer, to another receiver.

For the price of a conventional receiver in its price range, the S300 has built-in JVC's exclusive graphic equalizer system. With five zone controls to cover the entire musical range. While most high priced receivers offer bass and treble controls, and some include a third for midrange, none approach the precision and flexibility of the SEA graphic equalizer system developed and patented by JVC.

371,293 ways to hear better sound.

By adjusting the five detent tone controls covering the frequency range at 40 Hz, 250 Hz, 1,000 Hz, 5,000 Hz and 15,000 Hz, you can create 371,293 different sounds. A feat never before achieved (with a stereo receiver) outside a professional recording studio. But, then, the S300 is a JVC professional.

Get better performance from your components and listening room.

Why do you need such tremendous variations in tone? Quite simply, they help you to overcome the shortcomings of the acoustics in your listening room; they also can help you to compensate for the deficiencies in old or poor recordings. Finally, they can do wonders for the frequency response of your speakers, and where you place them.

SEA is really quite easy to use. For example, the 40 Hz switch reduces record hum or rumble, and it can add greater clarity to the ultra low bass of an organ.

The problem of booming speakers is simply handled with the 250 Hz switch. And in the important midranges, the 1,000 Hz control adds new dimension to the vocals of your favorite rock performers, while the 5,000 Hz switch brings out the best in Jascha Heifetz. You can even reduce tape hiss and diminish the harsh sound of a phono cartridge at high frequencies, with the 15,000 Hz control.

SEA adjusts the sound of your system to the size of your room.

You see, small rooms tend to emphasize high frequencies, while large ones accentuate the lows. But the ingenious SEA allows you to compensate for room size and furnishings—so your system can perform the way it was meant to, wherever you are.

While most manufacturers reserve unique features for their top of the line model, JVC has included SEA in three of its receivers. The S300, the S400, and, of course, the top professional—the S600.

When you hear these receivers at your JVC dealer (call toll-free 800-221-7502 for his name), think of them as two components in one. In fact, it's like having all the benefits of a graphic equalizer . . . without buying one.

JVC America, Inc., 58-75 Queens Midtown Expressway, Maspeth, N.Y. 11378 (212) 476-8300

JVC

CIRCLE 85 ON READER SERVICE CARD

TALK BACK

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

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

Demagnetizer Info

I have a cheap-o head demagnetizer (the $4 kind). A. Are these units usually somewhat effective? B. How much distance should there be between a demagnetizer and the tape heads when releasing the demagnetizer button?

—Stephen Kayser
Stamford, Ct.

Most modern tape recorders are designed to be largely self-demagnetizing, at least as far as the erase and record heads are concerned.

A small amount of residual magnetization may eventually accumulate in the playback head and any other parts in the tape path made from magnetic material. This background magnetization may take many months to build up, and periodic use of a demagnetizer will minimize this accumulation. Particular care should be taken to demagnetize the tape path if the recorder is transported, since the varying fields encountered in transport can lead to high levels of magnetization.

Since the possible levels of magnetization are quite small, even the simplest demagnetizer will usually be quite effective. Ideally, the tip of the demagnetizer should be brought into direct contact with the parts to be demagnetized. (If the tip is bare metal, it should be covered with a thin layer of tape to avoid scratching the heads). The demagnetizer should be switched on and off at a distance of two to three feet from the heads.

These precautions should eliminate any possibility of residual magnetization inducing noise onto previously recorded tapes, or degrading new recordings.

—Bill Burns, Manager
Technical Services
Revox Corporation
Syosset, N.Y.

Parametric EQ's

What is a parametric equalizer? How does it work? For the past four or five years I've read nothing about EQ's except about graphic and octave.

—Bruce Jenkins
Olmsted Falls, Ohio

Several years ago an organization called International Telecommunications Incorporated began to market an equalizer for professional use and they called it a "parametric equalizer." I think it was around the same time that Dan Flickenger began to offer "parametric equalizers" as modifications to his consoles. Well, Flickenger is no longer making consoles and ITI is no longer making equalizers, but parametric equalizers are still with us.

The name "parametric" stems originally from the application to audio of a type of reactance amplifier which was, I think, originally developed for hi-sensitivity microwave applications. Whatever its origins, the name has come merely to mean that the individual parameters of the equalization function are induvidually and *independently* controlled. The parameters are of course: *Frequency*, the center of a boost or dip curve, or the roll-off point for a filter; *amplitude*, the amount of boost or cut desired at that frequency; *bandwidth* or Q, the shape of the curve, the number or fraction of octaves over which the boost or cut extends while having its center at the selected frequency.

A conventional equalizer doesn't allow for the change of any one of these parameters while keeping the others constant. As a matter of fact, parametric equalizers utilize only pots for the adjustment of all their parameters. Then the mark of a parametric equalizer is a continuously variable frequency setting, a continually variable boost or cut and the possibility exists of continuously variable Q or bandwidth.

The most common construction for this type of equalizer is to have three or four sections each offering the features listed above connected in series with the ranges of the three or four frequency controls overlapping considerably. Some of the models currently available limit the choice of bandwidth to two switch-selected settings or preset it for the various sections of the equalizer.

There are now some parametric equalizers configured to provide octave type control of signals. They don't offer Q control or variable frequency but they do allow for a somewhat safer method of tuning a signal than conventional closely set filter centers. Conventional equalizers tend to interact with each other; a problem than can be controlled with parametric sections.

I suspect that the new graphic like the Crown EQ-2 and the Technics SH 9090 are built around parametric equalizer sections because of the range of control of frequency they provide. In any case don't confuse format with circuit function. It is possible to utilize conventional or parametric technology to realize graphics either on octave , half octave or third octave frequency centers. You can also use the same equalizers with the controls arranged in rows so that no graphic display of the setting is seen when they are in use.

In use, parametrics are spoilers. Once you get used to being able to fully tune your EQ it's very hard to live with someone else's fixed, predetermined frequency points. It's also possible to set the Q high and "pan" through the frequencies resulting in an effect that soundssomethinglike phasing or flanging.

—Ed Rehm
The Nordine Group
Chicago, Ill.

Ribbon Mics

I have a question pertaining to a velocity type (ribbon) microphone.

I would like to know if there is any long-range harm caused by using ordinary screws, which are magnetic (made of ferrous material), to fasten the plates which hold the ribbon (in a velocity type micrphone).

The original screws (two for upper plate and two for lower plate) were non-magnetic but I had to replace one of them and I used an ordinary screw (with magnetic properties). The microphone appears to function normally, but I can't help wondering if this could distort the magnetic field and alter the specifications of the microphone, or possibly weaken the magnet over a long period of time. There must be some reason why non-magnetic screws were used originally.

—Robert Graf
Paterson, N.J.

Note: The following answers were made independently of each other to further answer the above question. —Ed.

We can see no harm in using ordinary screws to fasten the plates holding the ribbon. They should not effect the magnetic field in the ribbon gap, or cause long range difficulties. Because of the magnetic attraction, the only difficulty may be in trying to get the screws started.

—Kenneth R. Reichel
Sales Engineering Manager
Shure Brothers Inc.
Evanston, IL.

In ribbon microphone design the construction of the transducer assembly is most critical since an absolutely linear magnetic field is required. Brass blocks are therefore used to support the ribbon to provide an uninterrupted magnetic path betweeen the north and south poles of the microphone's magnet.

If the blocks were made from magnetic material, the lines of magnetic force would pass through the blocks rather than through the air gap. Similarly, screws made of ferrous material will create an interference path and distort the magnetic field.

Furthermore, there is a possibility of an electrochemical reaction between the dissimilar materials of the brass block and ferrous screw with the possibility of corrosion.

While the effects of using ferrous screws may be relatively small, we do feel that the extremely linear performance made possible by Beyer's special ribbon design should not be compromised in any way.

—Bill Burns
Revox Corporation
Care of: Beyer
West Germany

Four Channels Into Two

First off, let me tell you that your magazine is the finest, most interesting periodical I've ever read!

I have been playing the four channels on my TEAC 3340 with the use of "Y" cords between channels 1 and 3, and 2 and 4. I have been told that I might be causing some damage to the machine. Is this possible? Also, I notice that on the 2340 machine TEAC has thoughtfully included a switch to put all 4 channels into a normal 2 channel stereo output for the headphones. This is great but how can I do this on my deck? (I have made my own adapter but am afraid that running two channels simultaneously may cause some trouble).

Thank you and please keep up the good work. Great to see MR is monthly!

—Richard Pearce
Lafayette, N.J.

When you say you are playing the 4 channels of your 3340S by using a "Y" cord, I assume that you are using the "Y" cord to connect the "phone" outs on the front of the machine.

"Y" cords are actually designed to *split* signals apart not to *combine* signals together. The effect of combining signals with only a wire is that the impedance of the circuit changes (the impedance can be defined as the resistance in an A.C. circuit).

Look at it this way: Some of the electrons flowing down the wire on the left (electricity) will try to go back upstream on the wire on the right and vice versa. It sounds worse than it is. You won't actually hurt your machine but there is a consideration. Sitting in the circuit near the "phone" outs in the A3340S is the meter amplifier. By changing the characteristics of that circuit (changing the impedance) you can cause your meters to read inaccurately. The extent of this effect would depend a lot on the type of headphones you are using.

Check this by recording and playing something, checking the meters and plugging and unplugging your "Y" cord. If you aren't experiencing a radical change (like ± 1½ dB) you're probably okay.

If you are getting a bigger meter change than you want to work with, then get yourself a TEAC AX-20 (a low cost passive 4 x 2 signal combining network) and 4 RCA (pin jack) male/male into female "Y" cords. Use these to split the outputs on the back of the recorder feed; 1 set to the AX20, the left and right outputs of the AX20 to your receiver and monitor with headphones at the receiver.

—Theo Mayer
Manager, Training Dept.
TEAC Corporation of America
Montebello, Ca.

Converting Mic Lines

Can "unbalanced" mic lines be converted to "balanced" lines and if so, how? What are the advantages and disadvantages of running balanced and unbalanced mics?

—Howard Liggett
Studio City, Ca.

An unbalanced line uses one conductor surrounded by a shield which is grounded and acts as the second conductor, while a balanced line uses two conductors neither of which is grounded but is surrounded by a shield which is grounded. Balanced lines have the advantage that noise not screened out by the shield will be cancelled out at the input of the following stage, producing lower overall noise in the system than would be obtained with unbalanced lines. The noise cancellation is achieved by connecting the output side of the line to either a transformer or a differential input amplifier.

A microphone signal applied to the input of the balanced line produces a current flow towards the output end in one conductor and equal in amplitude but away from the output end in the other. Noise signals, however, induce equal amplitude currents in the same direction in both conductors. If the outputs of the two conductors are connected to the opposite ends of the primary winding of a transformer, the noise currents oppose each other as they try to flow either into or out of both ends of the winding at the same time. The noise currents are equal in amplitude in both conductors due to the conductors being twisted closely together within the cable, thus there will be no current flow in the primary and no output from the secondary. The microphone signal flows into one end of the primary and out the other, producing a signal output at the secondary of the transformer. Since the shield also carries noise currents, it is important that it is kept electrically separate from the two conductors at the primary of the transformer, for connecting it to either of the conductors would upset the equality of noise currents preventing complete noise cancellation. The secondary of the transformer is made to feed an unbalanced line by connecting one end of the winding to the conductor and the other end to ground. The shields of both the balanced and unbalanced lines are grounded.

A differential input, single-ended output amplifier can be used to achieve noise cancellation in the same way as a transformer. This type of amplifier has two inputs neither of which is grounded, and produces an output only in response to a difference in the signals fed to its inputs. Since the noise signal is the same in both conductors, the amplifier will not respond to it. The microphone signal is different at each input, for the current tries to flow into one and out of the other, and so it is amplified. The conductor of the unbalanced line is connected to the single output terminal of the amplifier, while the shield is connected to ground.

In addition to the cancellation of noise picked up by the conductors, balanced lines have the further advantage that they minimize ground loop problems. These occur when the ground connection at one piece of equipment is at a different electrical potential than the ground at another piece of equipment. Connecting the two grounds together with the shield of a cable will cause a current to flow in the shield, inducing noise into the equipment at the cable output. This noise can appear as hum or even supersonic oscillations. Since balanced lines do not use the shield to carry the signal, it is only necessary to connect the shield to ground at one end, usually at the input of the following piece of equipment. This is called a "telescoping shield." Since the shield is used as the second conductor in an unbalan-

Home Cookin'!

See that guy at the board? Once upon a time he was an engineer at the busiest studio in town. The place had everything big money could buy. And it cranked out super-slick albums at an absolutely psychopathic rate. But because its hourly rate matched its image, it wasn't only the busiest studio in town, it was also the most expensive. Which was alright if you had a fortune to spend– which the band you see here didn't.

After years of being a staff engineer he decided he'd been sitting behind somebody else's board long enough, thanks. So with the money he'd saved, he invested in a complete TASCAM Series recording studio by Teac– 80-8* eight track, 25-2* two track, mixing consoles– the works!

Two days later, he was making tracks like they'd never been made before– in his home! And at a fraction of the price charged by his former employer.

Well, the band cut a demo at his new studio and with it they got a record deal. And with the front money, they invested in their own Teac mini-studio. So with the band members taking turns at the board, they laid down the tracks for their album. And to make sure they got the most out of the tracks they made, they asked the old pro to do the final mix.

Could this story have happened without Teac recording equipment? Not on your life. But it's the sort of thing that's bound to happen whenever a second generation engineer and a second generation band team up with a new generation of recording instruments.

TASCAM SERIES BY TEAC

A new generation of recording instruments for a new generation of recording artists.

*Nationally advertised value. Model 80-8 tape recorder shown above, less than $3,000. Model 25-2 tape recorder also shown, less than $1900 (Rolling Consoles not included). Actual retail prices to be determined individually at the sole discretion of authorized Teac Tascam series dealers. Prices subject to dealer preparation charges where applicable.

TEAC Corporation of America, 7735 Telegraph Rd., Montebello, CA 90640 ©TEAC 1977 ced line, it must be connected at both ends, encouraging ground loops.

Unbalanced lines do have a cost advantage. Basic electronic devices are inherently unbalanced and transformers or differential stages must be added to the input and output of each circuit to make them compatible with balanced lines. This increases the equipment cost.

While the question did not mention impedance, it is important to note that unbalanced mic inputs are usually high impedance, while balanced output microphones are usually low impedance. By choosing the proper primary and secondary impedances, the transformer can optimize the impedance seen by the input stage of the mic preamp for lowest noise operation without loading down the mic output, while providing a voltage gain in the area of 20 dB. This means that the preamp requires 20 dB less gain, and noise generated by the preamp input stage and fed to the following stages will be 20 dB less. Differential input amplifiers can also be designed for optimum input impedance with low impedance microphones, and they have an advantage over transformers in the areas of extended frequency and transient response. However, preamps with differential inputs require 20 dB more gain than ones with transformer coupled inputs, so they are usually noisier. Differential amplifiers also require a power supply in order to function making them more difficult to add to an existing system than transformers.

The Shure A95 series of plug-in line matching transformers is a convenient means of producing balanced mic inputs in an existing unbalanced system as they have an XLR connector at the balanced low impedance side and either a phone or an Amphenol connector at the unbalanced high impedance side.

—Robert E. Runstein
Author, *Modern Recording Techniques*

Limit Before Or After

When using Kepex and a limiter, is it better to limit the signal before or after Kepex?

—George Klein
New York, N.Y.

First let us think about it. Before we can use a piece or pieces of equipment properly, we have to know why we are using it. If the answer to that is "because it's there," then don't use it! If you want to use the combination for an effect, you may want to put the Kepex in front of the limiter. This will

The Control of Power by Sansui.

For the audiophile who wants the finest matched stereo amplifiers and stereo control preamplifiers, Sansui offers the answer — its Definition Series. Look at two of these outstanding components. The Sansui BA 3000 is designed with plenty of power to handle those bursts of percussion and those dynamic fortissimos that give you concert hall presence. The CA 3000 controls and features are a true joy for the creative pro and audiophile who wants to tailor the music to his own personal preferences.

This extraordinary pair is designed for the most demanding tasks; recording studios, sound reinforcement and audiophile home listening. The Definition Series offers the clearest, cleanest fidelity available anywhere. Top of the line Sansui BA 5000, called the "Monster," is one of the most powerful amplifiers available today: 300 watts per channel min. RMS into 2, 4 and 8 ohms from 20 to 20,000 Hz with no more than 0.1% THD.

For complete information on the entire Definition Series visit your local Sansui dealer soon or write directly to us.

The CA 3000 phono preamplifier is within ±0.2 dB of RIAA equalization curve. Offers everything desirable in a preamplifier and more. Triple tone controls. Left and right input and output channel meters. Tape controls for copy and playback. Sensitivity controls. Phono input capacitance selector.

The BA 3000 stereo amplifier. 170 watts per channel, min. RMS, both channels driven into 8 ohms from 20 to 20,000 Hz with no more than 0.05% THD. Safety abounds: 4 jumbo heat sinks (Sansui patent pending); triple protection circuits for ample power protection; and front panel LED power/protection indicator on the front panel to show safe operation. Specially damped output power meter.

Each Definition Series component comes with a test data sheet, complete with all performance characteristics. And Sansui offers a limited 5 year warranty.

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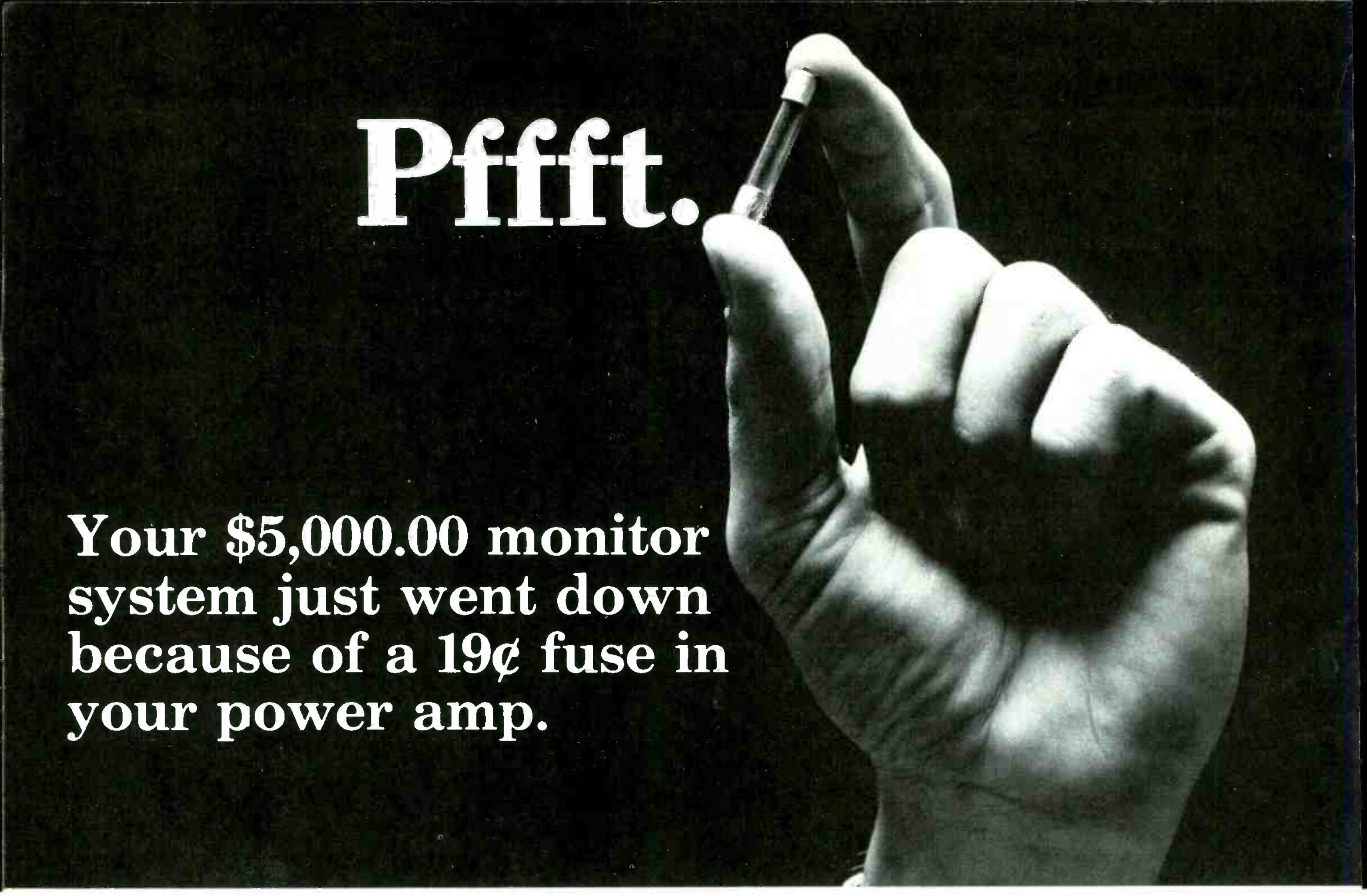
CIRCLE 77 ON READER SERVICE CARD



tend to drive the limiter up the wall, very subtly of course. If the limiter is a Gain Brain, which is rather speedy as compared to most "mortal" limiters, there would be more of that effect with a slower limiter.

Now that we have the special effects department out of the way, let us get serious. More than likely you are using the gate to mute a channel so that the undesirable spill is keyed off during quiet spots. If this is so, we would want the gate after the limiter, assuming that the noise we want to gate is somewhat below the signal level. There are two benefits here: first, all noise is gated including whatever sounds the limiter makes by itself and second, if the noise to be gated is loud enough to exceed the threshold of limiting, the limiter has a head-start when the desired signal comes back (not that the Allison needs it). A word of caution though. If the noise is really up there and drives the limiter much above the threshold, it may not key the noise gate, as the output differential may be too small for the gate to sense. That is, the level may remain too high with limiting to fall below the gate's threshold. But if your spill is that bad, you might be better off not using the gate since the change in room sound will become noticeable when the spill is keyed off. Keep an ear open for that problem when using any noise gate for that purpose.

—Dave Moyssiadis
Frankford/Wayne Recording Labs
Philadelphia, Pa.

Too Much Bass

I have an Akai GX-365 stereo reel recorder. Whenever I record from either the radio or a record, there is an enormous amount of bass content. I turn the treble up and the bass down on the receiver, both when I record and play the program, have cleaned and demagnetized the heads and guides, but still way too much bass. What can I do?

—Don Meade
Canton, Ohio

The analysis of an electonics problem given only a description is a rather difficult if not impossible task.

Your problem may be one of only a bias adjustment for your particular brand of tape.

However, your problem may also lie in a defectively worn record and/or playback head. If such is the case, you can have the GX head replaced at an authorized Akai Warranty Service Clinic. You are responsible for only the labor cost.

Whatever the case, I suggest that you have your Akai GX-365 inspected by your local Akai Service Clinic which is: Audio Warehouse, 719 E. Market, Akron, Ohio 44309, telephone (216) 376-6050.

—Linda Penny
Marketing Services
Akai America, Ltd.
Compton, Ca.

Quality or Safety?

My recording problem will seem rather primitive to you. I have been accustomed to recording music directly from a guitar amplifier by connecting the external speaker jack in the amplifier to the line input jack of my tape deck since I have no line output jack. The results were always satisfactory for me, however, I have been told that this process can be damaging to either or both the amplifier or the tape deck due to vast differences in both impedance and voltage. This is logical to me, yet I have never seen any adverse conditions arise due to this procedure. I would like to continue to record in this manner because external noise is eliminated as opposed to mic recording and results in a cleaner sound.

Is there a way to make such a procedure safe for the equipment? Is it all right to continue with this practice? I realize on the new tape decks that there is a warning against doing this but I have been doing this for years, even on inexpensive tape recorders which actually suggested taping the speakers of a source (such as an amplifier) and recording with the line input.

Is a combination transformer/impedance matcher that can take the 8 ohms from the external speaker jack on the guitar amplifier and change them to 50,000 ohms as well as lower the voltage to a constant 0.1 volts the answer? I would like to keep recording in the line input jack and also be able to line/mic mix onto one track.

Please help me with this very difficult problem!! Should I sacrifice quality for safety or can I achieve both??

—Henry Alletcha
Hartford, Ct.

Direct injection of an electric instrument's signal into a recording system is a common studio technique for the reasons you mention (lower noise and distortion) and because it eliminates the leakage of other instruments onto the track. The technique is usually used for bass guitar and keyboards, and the signal is picked off between the instrument and the amplifier to eliminate undesirable distortion or coloration from the amp and speaker. For guitar, most engineers prefer to mic the speakers because amplifier and speaker coloration is considered desirable. "Direct boxes" are available commercially, and allow splitting the signal between instrument and amp without degrading the signal, but these are not suitable for connection to the output of the amplifier.

There are three technical problems you face in connecting your amp and recorder directly. (1) There may be DC present at the output of the amp which can potentially cause distortion or even damage to your recorder; however, this problem is unlikely to occur. (2) You may create a ground loop and cause a buzz or hum; this problem is very annoying but seldom causes any damage to the equipment. (3) The signal voltage available at the speaker output of the amplifier (some 22 volts for a 120-watt amplifier with 4-ohm speakers, for example) is dangerously high to connect directly to a recorder; this is a serious problem since these voltages can easily burn out the transistors in the input stages of the recorder if the input level control were accidentally turned up too far.

Fortunately, a very simple device can be made to solve all three problems. In theory, a transformer to step the signal voltage down by an appropriate ratio at the appropriate impedances is all that's necessary, but the values required for this application are impractical. A better solution is to use a common one-to-one line transformer for isolation and a resistive pad to reduce the voltage. A side benefit of this type of design is that the transformer doesn't need to handle as high a signal level, which reduces the cost of the transformer. If you use a 600 ohm/600 ohm transformer, the pad can be as simple as a single resistor in series between the "hot" side of the input connector and the transformer's primary winding; a 12,000 ohm resistor will handle anything up to about a 120-watt amplifier, but a smaller value, say 5,000 ohms, may be necessary to get enough signal from a smaller amp. A device such as this will "bridge" the extra speaker output with 5600 to 12,600 ohms (impedance *matching* is only necessary and desirable if you are trying to transfer a significant amount of the amp's output power) and provide a 600 ohm output at about 1 volt RMS maximum to drive any line level input.

If you are building one of these boxes and want to use connectors such as ¼-inch phone jacks which connect one side of the jack to the chassis, be sure to insulate the input jack from the chassis of your box with a pair of fiber shoulder washers or else you can still create a ground loop and cause large amounts of hum. If you do insulate the jack, remember that you will have to run two wires between the jack and the transformer since the chassis is no longer the common.

—Fred Ridder
Record Plant Rec. Studios
N.Y.C., N.Y.

Rundgren's Ra

I was listening to an album by Todd Rundgren and Utopia called *Ra*. On the first side there is a song which, when listened to with headphones, has some parts in it which have perfect ambience and spatial quality. It sounds just like you were there. Were these parts recorded using binaural techniques or what? Also, could you explain about the advantages and disadvantages of binaural recordings?

—Gary Vogel
Oakdale, Mn.

On the *Ra* album during the song "Magic Dragon Theatre," there is an interlude where the music stops and there is some humorous dialogue. During this interlude, we do use a Sennheiser artificial head which is a binaural recording device. In actuality, it is a dummy head with two mics placed in the position of your ears such that the effect is recording in binaural stereo.

This is the only instance on the *Ra* album using a binaural technique. However, we *are* using digital delay, flanging, reverberation and a combination of techniques to produce more ambient sound.

Binaural recording from my experience is a very technical problem to apply to multi-track situations. It is a special technique used for sound effects or pieces to imply an ambient sound, but it is not a general recording technique. Because the recording mics are posi-

tioned where human ears are, the sound has to come directly to the ears. In other words, binaural recording should be played back in a reciprocal manner. To listen to a binaural recording with speakers, one not only has the ambience of the binaural recording but one must contend with the additional ambience of the room it is being played in. Therefore, to achieve optimum listening pleasure, binaural recordings need to be listened to on headphones.

—Roger Powell
Keyboard and Synthesizer
for Utopia

You can find more information on Binaural Recording in the Feb/Mar 1977 issue of MR in the Ambient Sound staple. —*Ed.*

Centering, Anyone?

In the Oct/Nov issue, there was a question in the Talkback section that received an answer that confuses me. The question referred to mic placement and signal feeding (p. 16). In part of the answer a phrase was used "the notorious 3 dB center build up."

When I'm miking down a band for the P.A. (usually at a club) I always put everyone at center. Usually, stereo separation isn't needed. Could you explain if there is a problem by "centering everyone?"

—Robert DeMoss
Fayville, Ma.

There is no problem centering everyone. Assuming you are using the mono output of the console to feed the amplifiers. You could pan each input completely left or right and not worry.

Don't be frightened by "The notorious 3 dB center build up." This situation occurs because of pan pot design and is not a serious problem. Pan pots (which is short for Panoramic Potentiometer) are actually two or three ganged level outputs that balance level between two output levels (or left and right channels in stereo monitoring). When you "pan" a signal source to the center, equal amounts of signal are sent to both channels and the image appears between the monitors. When you "pan" left, there is no output in the right channel, and when you "pan" right, there is no output in the left channel.

Due to the design of pan pots, the output in the center position is 3 dB less than the output in either left or right positions. This 3 dB loss is usually compensated by boosting the level with the fader. Incidentally, a 3 dB change is just about the smallest change that can be perceived by the human ear.

Now, assume that you are doing a stereo mix, and you have everything the way you want it. You then switch to mono output monitoring to check for phasing and lo and behold, everything panned to the center becomes louder with respect to signals panned left and right. This is center build-up.

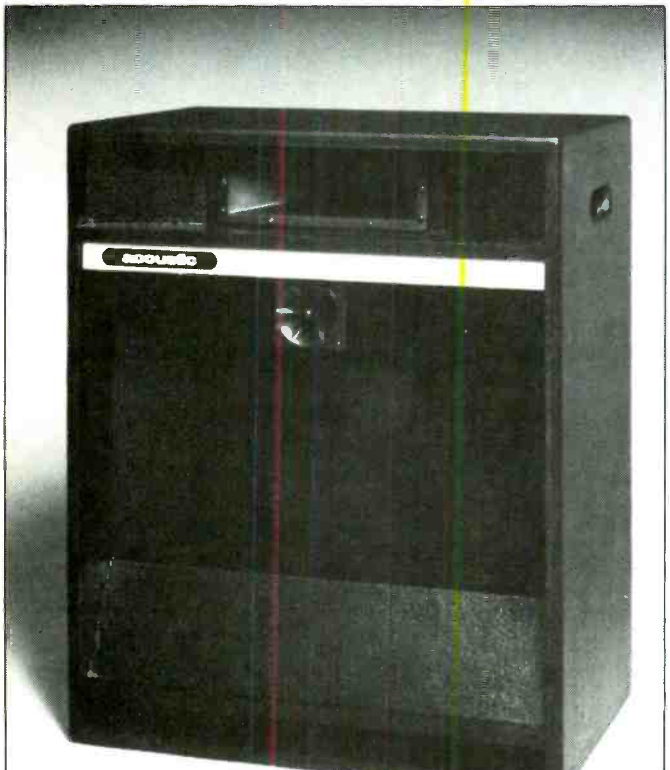
As I said, it is not a serious problem. Most producers do not compensate for this build-up unless they get a really bad mono mix by combining the left and right tracks of the stereo mix.

If you want a good source of information on audio problems, you might pick up a copy of "Audio Encyclopedia" by Howard M. Tremaine, published by Howard W. Sams Company. Good luck.

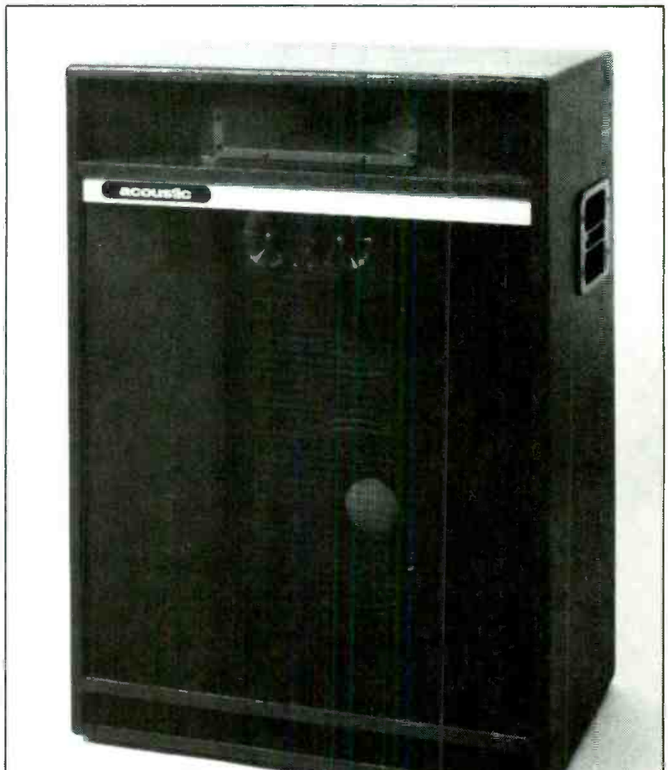
—Chip Allen
LeFevre Studios
Atlanta, Ga.

QUICK,

WHAT COMPANY MAKES THE MOST EFFICIENT P.A. AND DISCO SYSTEMS UNDER $500?



Acoustic Model 806



Acoustic Model 807

WRONG,

IT'S ACOUSTIC!

Introducing the Acoustic 806 and 807, two new speaker designs for maximum performance at a very modest price. Both are three way systems that cover the audio range with incredible efficiency. And efficiency means more dollars in your pocket since they don't require a monster amplifier to get the sound out. Visit your nearest Acoustic dealer, and compare the 806 and 807 to whatever you thought was the best. You'll be pleasantly surprised. Now check the price tags. When we said "Under $500," we meant "WAY UNDER $500." From Acoustic . . . the surprising company.

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CIRCLE 59 ON READER SERVICE CARD

THE PRODUCT SCENE

ESS EXPANDS SPEAKER LINE

Ten different speaker systems—all using some form of the Heil air-motion transformer as the upper-end sound-generating element—have been announced by ESS Inc. Systems span the price range from $566 down to $149 and include floor-standing as well as bookshelf models. Top of the line is the Heil AMT Monitor with a rated power capacity of 375 watts (clean music power), an input sensitivity of one watt to produce an output of 87 dB/SPL at a distance of one meter and rated response of 30 to 23,000 Hz within ±3 dB. The "Performance" series Model 8, at the other end of the line, can handle up to 100 watts; can produce from a 1-watt input an output of 94 dB/SPL at a distance of three feet, and rated response is 50 Hz to 30 kHz within ±3 dB.

CIRCLE 1 ON READER SERVICE CARD

ELECTRONIC CROSSOVER FROM CROWN

Internal circuit changes characterize the new Crown VFX-2A electronic crossover which replaces this firm's model VFX-2. Instead of the ten dual op-amps used previously, the new version contains six quad op-amps. As a result, says Crown, a better slew rate is achieved for improved handling of transients. One of the quad op-amps serves as an isolation amplifier to eliminate variable input impedance of the device, which removes the possibility of any impedance mismatch problems. The new op-amps also provide greater range on the level control.

The front panel now sports an amber power light, and rear panel changes provide for more sophisticated audio system interfacing. Two new output connectors have been added, each of which mixes two channels for separate mono bass and mono full-range outputs. For sound playback, these outputs permit easily adding center-fill speakers. For professional sound reinforcement, these outputs mean more convenience in single-channel applications, and provide for mixing of two different sound sources for a single transducer.

The VFX-2A's continuously variable filters can perform either crossover or band-pass functions. Rolloff is at a fixed 18 dB/octave rate. Setting both low and high pass filters in one channel to a common frequency provides a two-way crossover at that frequency. Cascading both channels together provides a combined band-pass filter and two-way crossover, or a three-way crossover. Specs include 300 ohms output impedance, with over 6 volts maximum output into 600 ohms. IM is rated at less than 0.01% at rated output; noise is down by more than 100 dB. Price of the VFX-2A remains the same as the former model, $329.

CIRCLE 2 ON READER SERVICE CARD

TEAC UNVEILS ITS ELCASET

Scheduled for delivery this Spring, according to a company spokesman, is Teac's model AL-700 Elcaset recorder, a three-head, three-motor unit utilizing a diecast transport with closed-loop, double-capstan system. Weighted wow and flutter are said to be 0.05%, and many of the unit's functions are automated. For instance, the electronics automatically match tape characteristics thus eliminating the need for bias and EQ switches. The AL-700—which will retail for "less than $1,000"—has Dolby noise reduction and also is equipped to accept an optional dbx unit and a remote control unit. A "rec mute" button kills any incoming signal at the end of a recording, thus easing the insertion of blank spaces between musical selections. A timer function automatically starts the transport in play or record mode as set by an external clock timer. The new deck weighs 45 pounds.

CIRCLE 3 ON READER SERVICE CARD

NEW HIGH-END AUDIO LINE

A new name on the U.S. audio scene is the Japanese brand Bohsei, whose products are aimed at top-quality-minded sound fans. Leading the line is a tape deck and two phono tone-arms. The deck, model IT-1000, has four heads, three motors, full-logic solenoid-actuated tape motion, 10-inch reel capacity, mic and line inputs, large VU meters and more. Also included is auto-reverse in playback.

The IT-1000 is a two-speed model (7½ and 3¾ ips) with claimed frequency response at the faster speed of 30 Hz to 20 kHz within ±3 dB. Wow and flutter are listed as 0.06% or less; signal-to-noise ratio is spec'd as better than 55 dB.

CIRCLE 4 ON READER SERVICE CARD

HEADPHONE/MICROPHONE COMBINATION

JVC has introduced a unique product—a combined headphone/microphone designed specifically for binaural recording and monitoring. Also useful as regular stereo headphones, the JVC device has the usual two earpieces plus an electret-condenser microphone in each earpiece. The mics are powered by size AA cells, also contained in the earpieces. Their physical spacing makes for true binaural sound pickup from "live" sources, and the operator can monitor the results while wearing the headset. Alternately, the headset can be placed on a dummy head, supplied with the unit, where it will still function as a binaural mic, with monitoring options up to the recordist.

Known as the model HM-200E, the new JVC product features 8-ohm dynamic headphones with a sensitivity of 96 dB. The mics are rated for 67 dB ±2 dB sensitivity, with an output impedance of 600 ohms. Overall frequency response of the system is 50 Hz to 10 kHz. Weight is 1.32 pounds. Cord length is about six feet. Phone plugs are provided for mic and headphone connections. A control handles headphone level for high, low and off and a tone selector for the mic provides for flat, low-cut, or off. Price is about $80.

CIRCLE 5 ON READER SERVICE CARD

OP AMPS, ET AL

From the firm known as Opamp Labs, Inc., comes word of a new d.c. operational amplifier for audio console and industrial applications. Known as the model 425, it features a pair of matched low-noise input transistors coupled to an Opamp model 4009 with a class AB power output stage. THD is rated at less than 0.05% at a gain of 100. The amplifier may be used at any supply voltage from ±6 V to ±25 V. Opamp Labs also has various devices which are described in company literature, including three mic preamps, other d.c. opamps, several power supplies, and audio and bias oscillators.

CIRCLE 6 ON READER SERVICE CARD

MIXING CONSOLE FOR RECORDING AND P.A.

From Sound Workshop there's news of the model 840 console whose eight inputs and four outputs can serve mixing needs for recording or P.A. applications. The four output buses are selectable as two stereo buses, with panning, for added versatility in P.A. and production work. All eight inputs can accommodate line or mic-level sources, and provide for a direct output too. Each input channel has two-band EQ, wide-range trim control, monitor and echo sends, pan pot, output bus select, rotary channel level control, and stereo bus mute switch. Master level controls are provided for echo send, monitor send, the four bus outputs, and the four echo returns. Fairly compact, the model 840, designed for four-track or two-track recording as well as to serve as the control center of a P.A. system, sells for $800.

CIRCLE 7 ON READER SERVICE CARD

MARANTZ DEBUTS EIGHT RECEIVERS

To meet what the manufacturer calls "the requirements of the broadest possible range of audio applications," Marantz has announced an all-new line of eight stereo receivers. Top model is the Marantz 2385, rated for 185 watts per channel, minimum RMS power into 8 ohms from 20 Hz to 20 kHz with no more than 0.05% THD. In addition, the 2385 contains what a spokesman claims is "the finest tuner section in any Marantz receiver ever manufactured."

The high-end units are equipped with a Bessel-derived 18-dB-per-octave high filter, and a convenient tape-copy facility that allows dubbing from one recorder to another while being able to listen to another source such as phono or FM.

CIRCLE 8 ON READER SERVICE CARD

DELTEK REVEALS PARAMETRIC SQ DECODER

Deltek, Inc. has begun production on what it describes as a super-performance parametric SQ decoder designed by Peter Scheiber and claimed to bring to SQ disc and broadcast reproduction a higher level of separation, and lower distortion, than any other quadraphonic disc system. Designated the Deltek Model One and priced at $2150, the new device—in addition to its SQ

decoding function—provides two functions for use with conventional stereo sources. One is an "ambience recovery mode," and the other is a "synthesis function" said to expand the stereo stage into a 270-degree panorama around the listening area. In its normal decoding mode, the device is expected to recover the "full 360-degree sonic environment" from SQ-coded discs and broadcasts.

Frequency response of the Deltek Model One is listed as 5 Hz to 50 kHz at the minus 3 dB points. Total IM distoriton is less than 0.05%, and clipping level is stated as 20 dB above zero level. Nominal input signal requirement is 350 mV rms. Weighing 19 pounds, the device is suitable for rack mounting; plain sides also are available.

CIRCLE 9 ON READER SERVICE CARD

NEW CASSETTE TAPE LINE

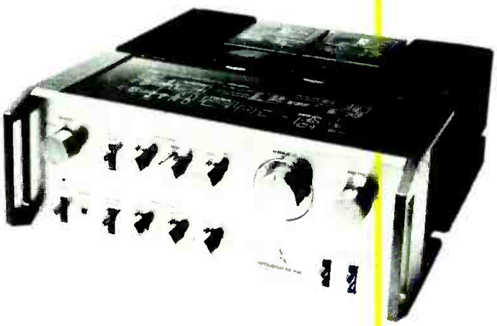
A new line of chromium-dioxide (CrO_2) cassette tapes in 60 and 90 minute lengths has been announced by Royal Sound Co., Inc. The chrome coating is on a Mylar base and the tape, says Royal, is highly calendered to minimize dropouts and to maximize high-frequency response. The resulting performance characteristics are claimed to be "unusual, even for a chromium dioxide tape." The precision shell is made of high impact plastic and is screw-assembled. The CRC-60 size will retail for under $3; the CDC-90, for under $4.50.

Royal Sound also offers a ULC series of cassettes using "pure crystal gamma ferric oxide" for normal bias settings; and a low-noise, extended-range series of cassettes. The ULC series are priced similarly to the CRC and CDC respectively. The low-noise series, APC, are lower-priced. Royal Sound also offers 5-inch and 7-inch open-reel tapes and cartridge tapes.

CIRCLE 10 ON READER SERVICE CARD

NEW POWER AMPLIFIERS AND PREAMP

A unique "component docking" design is featured in a new line of Mitsubishi amplifiers. Described as "dual-monaural," the units include: model DA-A10, a 100-watts per channel basic; model DA-A15, a 150-watts per channel basic; and the model P10, a preamp-control. Either power amp can be "docked" with the preamp, or—instead—with the firm's model M10 power-level meter. "Docking" consists of securing the preamp or the meter to die-cast handles on the amplifiers. Electrical interconnections are made via linking cables supplied. Signal and AC connectors are at the sides; trailing wires are routed in a channel below the amplifier heatsinks. According to the company, the "dual monaural" construction eliminates crosstalk and improves channel separation by as much as 30 dB



or more over conventional designs. Channel separation is given as −80 dB or better at 20 kHz. Other design features include high-capacity separate power supplies, a relay system for connecting speaker terminals to eliminate degradation of the damping factor associated with the contact

resistance of speaker selector switches, and a remote-control option for the preamp. Prices for the power amps are $390 for the DA-A10, and $590 for the DA-A15. Preamp and meter prices were not yet announced at presstime.

CIRCLE 11 ON READER SERVICE CARD

HOW DOES WHAT YOU RECORD SOUND WHEN PLAYED BACK?

I am not sure who actually did it first, but certainly the most auspicious and influential of the modern recordings ushered in by stereo was the all-stops-pulled production of Wagner's ring cycle launched at London Records by Producer John Culshaw over 15 years ago. Culshaw, probably more emphatically than any other single producer, once and for all broke with tradition by creating a "production" out of a recording session rather than merely "eavesdropping with a microphone" which was the rather passive technique favored for years.

In doing so, Culshaw and London Records not only opened up new creative vistas for the recording team but they also—perhaps not wholly aware of it at the time—made the consumer/record buyer a very important element in the whole recording/reproducing chain, from mic in the studio to speaker in the home.

How? Well, the end result of the kind of creative recording they ushered in was a program source more fittingly tailored for playback on a home music stereo system than anything done before. They captured movement and ambience which—even on two channels—lends the playback a touch of excitement and realism not otherwise enjoyed by home stereo listeners. This was in direct contrast to the dry, almost antiseptic sound that characterized most recording up to that time, including the most vaunted of monophonic efforts which were "clinically clean" but fairly unconvincing in subjective terms.

In many ways, the lessons of those early stereo breakthroughs have been well absorbed and even improved on by today's generation of recordists, both pro and semi-pro. But every now and then we get an example of someone's overdoing a good thing, of turning the knobs and injecting the juice for its own sake, or to serve one's own ego.

The important thing to keep in mind, I believe, is that the gadgetry and the devices exist to help you make a recording that will sound right when played back on good home stereo systems. For this reason, it behooves the recordist to listen, more than casually, to such systems and even to own one (or two) himself or at least have access to them for some serious auditioning. For it is what comes out of such systems, and how it sounds, that will in the long run be the benchmark of your efforts at the console in the studio. In this sense it may be as important, if not more so, for recording people to attend to hi-fi reproduction as it has customarily been for them to attend to "live" sound.

MUSICAL NEWSICALS

SOUND REINFORCEMENT. . . In professional sound reinforcement circles, Malatchi Electronic Systems (3731 E. Colfax, Denver, Colo. 80206) is becoming well known for their very versatile (though rather large physically) mixing consoles. With the introduction of their lower-priced and reasonably-sized Performer series, Malatchi is aiming to capture a portion of the semi-professional market as well. The Performer 6 ($699) is a 6-input mono mixer which can be expanded to 12-input or 18-input capability by adding on one or two Expander 6 units ($549 each). Malatchi also makes 12 and 18-input units in single mainframes rather than a two- or three-unit interconnection as the Performer 12 ($1199) and Performer 18 ($1799).

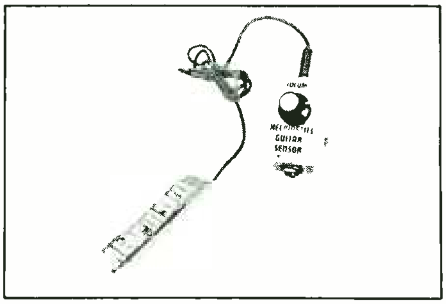
Several different mixing consoles are now available from Neptune Electronics (934 NE 25th Avenue, Portland, Oregon 97232). Most basic is the model 610 ($329) which is a six-input mono mixer. Each input channel has connections for unbalanced, low impedance mic input, line level input and preamp output, and controls for volume, bass and treble EQ, and post-fader reverb/effects send. If you want a pre-fader monitor send on each input in addition to the other features, the model 611 ($399) should suit your needs. The model 820 ($549) is an 8-input stereo mixer which adds a pre-fader monitor send and a stereo pan pot to the basic features of the model 610. Also of interest from Neptune are the Model 910 graphic equalizer ($149), a nine-band octave equalizer with 12 dB of boost or cut on each band, and the model 909 real-time analyzer ($549) which features a 9x9 LED display and a built-in pink-noise generator.

Peavey Electronics Corp. (Meridian, Miss. 39301) has introduced two new, high-power amplifiers to compliment their much talked-about super-amp, the model CS-800. The Peavey CS-400 is a stereo amplifier rated at 200 watts RMS per channel into 4 ohm loads with less than .05% THD and less than .1% IM distortion. Frequency response is claimed to be 20 Hz to 60 kHz +0, -1 dB. In addition to the inputs and outputs on the back panel, each channel has on the front panel a level control knob, an LED overload indicator and a line output jack to facilitate hooking up several amplifier channels in parallel on one

signal. Lighted pushbutton switches are provided for AC power and thermal cutout reset. The reset is unlikely ever to be used, however, as the amplifier is forced-air cooled. The front panel is steel-reinforced cast zinc, features integral handles and is designed for 19-inch rack mounting. Peavey's model CS-200 amplifier is basically identical except that it is a single-channel unit rated at 200 watts RMS with the same distortion specs. Also available from Peavey is the Monitor 260 amp, a 130 watt RMS (4 ohm) amplifier with built-in 9-band graphic equalizer. The Monitor 260 comes in a vinyl-covered wooden case rather than ready for rack mounting.

PICKUPS. . . Barcus-Berry has also introduced a most intriguing new pickup—a contact pickup for electric guitars. The Superducer ($69.50) transducer attaches to the bridge of any electric guitar to pick up the acoustical vibrations of the guitar body and bridge. A special, battery-operated Superducer preamp/mixer is provided which is designed to attach to the guitar body with Barcus-Berry's special adhesive, and which connects to both the transducer and the guitar's normal output jack. A single knob on the preamp then controls the mix of acoustic and electric signals from full acoustic to straight electric or anywhere in between.

Helpinstill Designs (5808 S. Rice Avenue, Houston, Tx. 77081) who make the well-known and much respected piano pickups, now makes a guitar pickup, the model 65 Acoustic Guitar Sensor ($79.50). The model 65 is a hybrid design which picks up the acoustic vibrations of the guitar body as well as picking up the vibrations of the strings, which is done electromagnetically as in an electric guitar. The pickup attaches to the guitar body under the strings near the bridge with a non-damaging pressure-sensitive adhesive strip. A passive con-



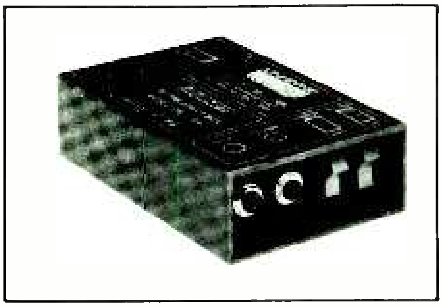
trol box with volume control and ¼-inch phone jack output is provided with a belt clip for convenient use of the pickup system.

DiMarzio Musical Instrument Pickups (643 Bay Street, Staten Island, N.Y. 10304) has announced two new additions to their line. One is a new high-efficiency replacement pickup for Fender Stratocaster guitars which may be mounted in any of the three pickup positions. This new model is the first pickup available for Strats which has adjustable pole pieces. The other new model from DiMarzio is a replacement pickup for Fender Precision basses. This model also features adjustable pole pieces and is a direct replacement requiring no modifications.

"Screamers" is the name of a line of replacement pickups from Mighty Mite (1707 Cloverfield Blvd., Santa Monica, Ca. 90404). Pickups to fit Gibson guitars are available in two basic versions, Vintage ($59.95) and Distortion ($69.95); either version is available with normal Humbucking coils or split coils at no extra cost. For Fender guitars, the Screamer line includes models for Stratocasters ($39.95 for conventional type, $44.95 for dual coil), Telecaster lead position (conventional—$39.95; dual coil —$44.95), Telecaster rhythm position ($39.95) and Precision Bass ($59.95).

ACCESSORIES... One of the most useful accessories for the electrified musician is the "direct box," which is basically an isolation and impedance matching device designed to allow direct connection of the instrument or the output of an amplifier to a mixing console or tape recorder. In its simplest form, a direct box need be no more complex than an audio transformer and perhaps a resistive pad, but many models have a variety of special features to add versatility or to make their use more convenient. This month we have news of five different models from three manufacturers:

Uni-Sync, Inc. (5559 Cahuenga Blvd., North Hollywood, Ca. 91601) who are the folks responsible for the Trouper Series sound reinforcement gear, offer the model 1104 Guitar Direct Box ($95.00). This model uses an advanced design transformer to provide high input impedance, low distortion, wide frequency response and excellent hum rejection. Unlike most direct boxes, the 1104 can be connected either be-



tween instrument and amp for the cleanest possible sound or after the amplifier to get the amp's contribution to the guitar's sound simply by flipping a switch. When connecting to an amplifier output, there is a filter which may be switched in to simulate the frequency response of a guitar speaker. A ground lift switch rounds out a versatile package.

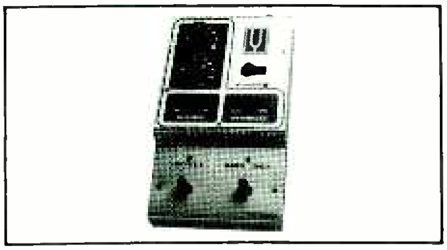
The Tycobrahe Direct Box ($59.95) is unique in that it combines direct box functions with an Automatic Noise Fader, which is basically a noise gate, to completely eliminate hum and noise at the output when the instrument is not being played. The noise gate's threshold is variable over a 50 dB range and the gating time constants have been selected to provide a gradual turn-off when playing is stopped. Power is supplied by a 9V battery or optional AC adapter. (Tycobrahe Engineering Co., 725 Cypress Ave., Hermosa Beach, Ca. 90254).

Whirlwind Music (Rochester, N.Y.) offers several products that serve as direct boxes. The IMP 1 ($19.90) converts a high-Z input (via ¼-inch jack) to a low-impedance output (via cable mounted male XLR) with 600, 250 or 50 ohm impedance selectable by a slide switch. IMP 2 ($23.90) is functionally the same but has two ¼-inch jacks in parallel and a panel mounted XLR. IMP 3 ($27.90) is designed to provide an isolated 250 ohm output from the speaker output of an amplifier.

JHD Audio (1370 Logan Avenue, Costa Mesa, Ca. 92626) has a very interesting little product called the Ice Cube ($19.95). The unit itself is simply a compressor which is said to increase sustain by a factor of 20. What is interesting is that the device is designed to plug into the back of a Fender amplifier in place of the connections to the reverb unit. Once connected in this way, the amp's reverb knob controls the amount of sustain and the reverb footswitch turns the effect on and off.

Two new electronic devices were shown by JXL (2314 Fourth Street, Berkeley, Ca. 94710). The first is the Final Phase ($99.00), a phase shifter with a few new twists. Final Phase has nine phase shift stages which give the device a different sound from most other phasers which have an even number of stages in the circuit. Other circuit innovations include a built-in noise gate to eliminate noise at the output when there is no input signal, and a "sweep modulation" circuit which produces a wide range of non-sinusoidal sweep patterns. The other device is the 2001 Octave Shifter & Delay Line ($299.00) which is a sophisticated and complex analog/digital device which can change the pitch or frequency of the input by any interval from one octave up to two octaves down in "real time." Controls are provided for octave selection, pitch interval within that selected octave and mix of normal and altered signal. In addition to the pitch shift, the 2001 functions as a variable delay line with 5 to 100 ms of delay available, selectable with the control provided. Three footswitches are used to give instant choice of no effect, delay only, perfect octave-interval effect or non-octave-interval effect.

Another new octave device is the Mu-tron Octave/Divider ($160.00) from Musitronics Corp. (Rosemont, N.J. 08556). The Mu-tron octave unit is an AC-operated device designed to produce a new note one octave below the input note. A mix control is provided to set



the balance between normal and sub-octave notes, and a tone control with specially-designed filters varies the tonal quality of the sub-octave note from a pure, deep tone to a very electronic tone. A special feature is the stabilization circuit which eliminates most of the false triggering that causes other octave dividers to "hunt" for the right note when the input signal has a complex waveform. Footswitches with LED indicators are provided for effect bypass and normal mix/bass note only switching.

The latest additions to the Morley product line (2301 W. Victory Blvd., Burbank, Ca. 91506) are the model SPV Power Panner ($99.95) and the model EDL Electrostatic Delay Line ($269.95). The Power Panner is a passive device that is inserted between amplifier and speakers to pan the amp's output between two sets of speakers, or to function as a volume control. As a volume control, the unit has the advantage of reducing amplifier noise along with signal level. The Electrostatic Delay Line produces a variable delay of from 10 to 150 ms by the same means Morley has used successfully in their echo units for several years. Controls are provided for delay length, input level, balance of normal and delayed signals and repeats, and outputs are provided for direct only, delay only and mixed signals.

Gallien and Krueger are busy building a professional reputation. They're doing it by building professional sound equipment like the Gallien-Krueger 1000-1S power amp. Take a look at the 1000-1S *guaranteed* specs and you'll begin to see what we're getting at.

Specs can tell you a lot about a particular piece of equipment, but they can't tell you everything.

For example, the 1000-1S is the only professional quality amp that is rugged enough to be *truly* portable. We can say it because we *build* it that way. In addition to assembly, we do our own metal fabrication, build our own transformers, *and* print our own circuit boards. The 1000-1S is built to be rugged and reliable, in the rack, or out.

And when it comes to sound quality, once you optimize slew rate and minimize distortion—and *all* amps have some, even the 1000-1S—it's up to you and *your* ears. So we're not going to spend time *talking* about our sound reproduction. We want you to decide about that for yourself, at your Gallien-Krueger dealer.

The Gallien-Krueger brand of quality doesn't exactly come cheap, but we think you'll like the sound of the 1000-1S price too.

Guaranteed Specifications

Power	200 watts	RMS per channel into 4 ohms with both channels driven from 20 to 20kHz
	125 watts	RMS per channel into 8 ohms with both channels driven from 20 to 20kHz
Harmonic Distortion	$<$ .05%	at 200 watts RMS per channel both channels driven from 20Hz to 20kHz into 4 ohms
	$<$ .025%	at 125 watts RMS per channel both channels driven from 20Hz to 20kHz into 8 ohms
Intermodulation	$<$ .025%	from .2 watt to 200 watts into 4 ohms
Distortion	$<$ .01%	from .1 watt to 125 watts into 8 ohms
Power Bandwidth	$\pm$ .25dB	from 20Hz to 20kHz at 200 watts RMS into 4 ohms
Frequency Response	$\pm$ 1dB	from 5Hz to 200kHz at 1 watt
Phase Shift	$\pm$ 15 deg.	from 20Hz to 20kHz
Damping	$>$ 400	into 8 ohms at 1kHz
	$>$ 200	into 4 ohms at 1kHz
Voltage Gain	30dB	
Sensitivity	.9v RMS	for 200 watts into 4 ohms
Slew Rate	40V/μ sec	
Input Impedance	25k ohms	
D.C. Offset	$<$ 10MV	adjustable to zero
Hum and Noise	110dB	below 200 watts RMS

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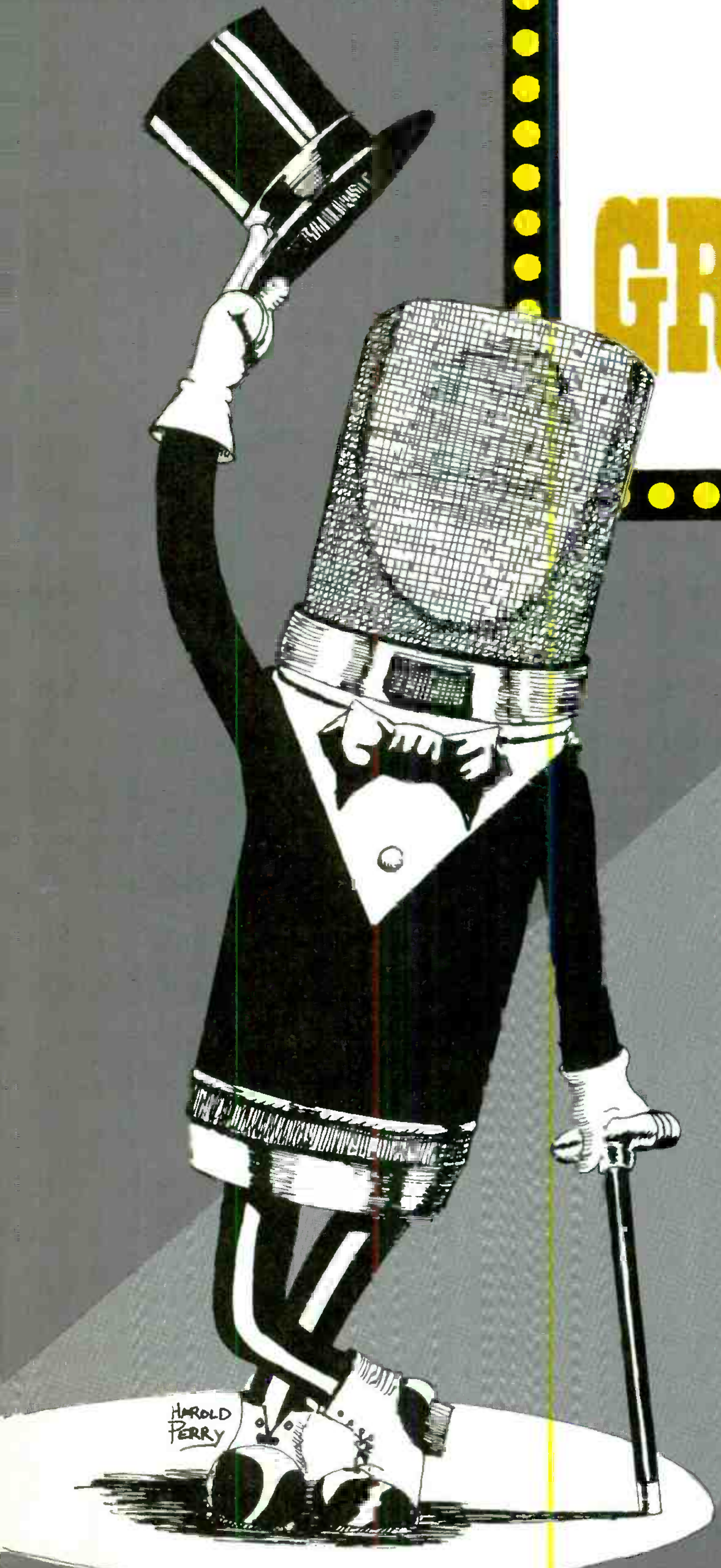


At last, right here in *Modern Recording* is the inside story, telling you just which microphones to buy to create tomorrow's hit records! On these very pages, you'll learn exactly where to place those microphones to get that guaranteed "top 40" sound!

As Ed McMahon might say, "Isn't it amazing—*everything* you need to know about microphones is right here before your very eyes!" To which Johnny Carson would reply, "Wrong condenser-breath! Everything you need to know is *not* to be found here." In fact, if you're looking for answers to questions like, "Which mic should I use?" or, "Where do I put it?"—keep looking, you'll find nothing of interest here.

It's not that this sort of information is top secret—although many beginners are convinced that professional recording engineers possess some sort of mystic rule book that tells them just what microphone to use, and where to put it. Well, there are no rule books containing this sort of data. Every engineer makes up his own rules as he goes along, and then breaks them (the rules, not the microphones) with regularity.

Microphones are pretty much like clothes—there is no universally "correct" standard or style. For example, when picking out a new coat, you probably have some definite ideas about fabric, style, length, weight and price. In many cases your impeccable tastes are not shared by your friends, and while you're wondering how they can appear in broad daylight looking as they do, they have long since given up on your strange-looking threads.

So, why should microphones be different? If you like it, it's right (for you). If you don't, try something else until you find the sound you like. But before rushing off to start your experiments, there are a few bits of information that may help steer you towards the microphone best suited to your individual application and taste.

Polar Patterns

Although all microphones pick up sounds directly in front of them (on-axis), some are less sensitive than others when it comes to hearing sounds arriving from the side or rear (off-axis). A polar pattern is simply a graph of a microphone's directional sensitivity. The graph itself is a series of concentric circles, with the outer circle labelled "0 dB." The inner circles are commonly drawn in 5 dB increments, with the smallest one labelled about -25 dB.

The Omni-directional Microphone: An omni-directional microphone is one that is sensitive to all sounds, regardless of the direction from which they arrive. Therefore, its polar pattern is simply a circle, drawn right around the 0 dB line of the graph.

The Uni-directional Microphone: On the other hand, a uni-directional microphone is not as sensitive to rear-originating sounds. For example, consider a microphone that is 20 dB less sensitive in the rear than in the front. Although the polar pattern starts off on the 0 dB circle, as it rotates around the microphone, it gradually moves in towards the inner circles. At 180 degrees (representing the rear of the microphone), the polar pattern touches the -20 dB circle, and then begins moving outward again as the rotation continues. By the time the pattern reaches 360 degrees (=0 degrees), it has returned to 0 dB. Try sketching this, and you'll discover a heart-shaped pattern; hence the popular designation, cardioid microphone.

With a super-cardioid microphone, the prefix "super" has nothing to do with the microphone's abilities. It's merely a modified cardioid pattern; slightly narrower (less sensitive) on the sides, and with a small lobe in the rear. The rear lobe shows that—compared to the standard cardioid pattern—the super-cardioid is somewhat more sensitive at 180 degrees off-axis.

The Bi-directional Microphone: A bi-directional microphone is equally sensitive in the front and in the rear, and is quite insensitive to sounds arriving from either side (90 or 270 degrees). In fact, the microphone is much less sensitive at its sides than a cardioid is in the rear—a point often overlooked by engineers trying to isolate the sound of one instrument from that of another. Sketching the polar pattern will reveal why this type of microphone is usually called a "figure-8" microphone.

Choosing the Right Polar Pattern for the Job

In planning and setting up the studio for a recording session, engineers and producers like to spend lots of time devising ingenious schemes for minimizing leakage. For it seems that one of the seven deadly sins of multi-track recording is to allow the sound of one instrument to be heard by a microphone placed near another's instrument. Although advanced cases of leakaphobia may lead to producer hysteria, there is indeed justification

Sole Control.

It's a new concept in sound effects pedals from Maestro. Much more than just on-again, off-again, these pedals you can really play. With the sole of your foot.

The new Maestro Stage Phaser is one of them. It gives you the wild, gyrating, phase shifting sound of a rotating speaker. But with an incredible amount of playability and control.

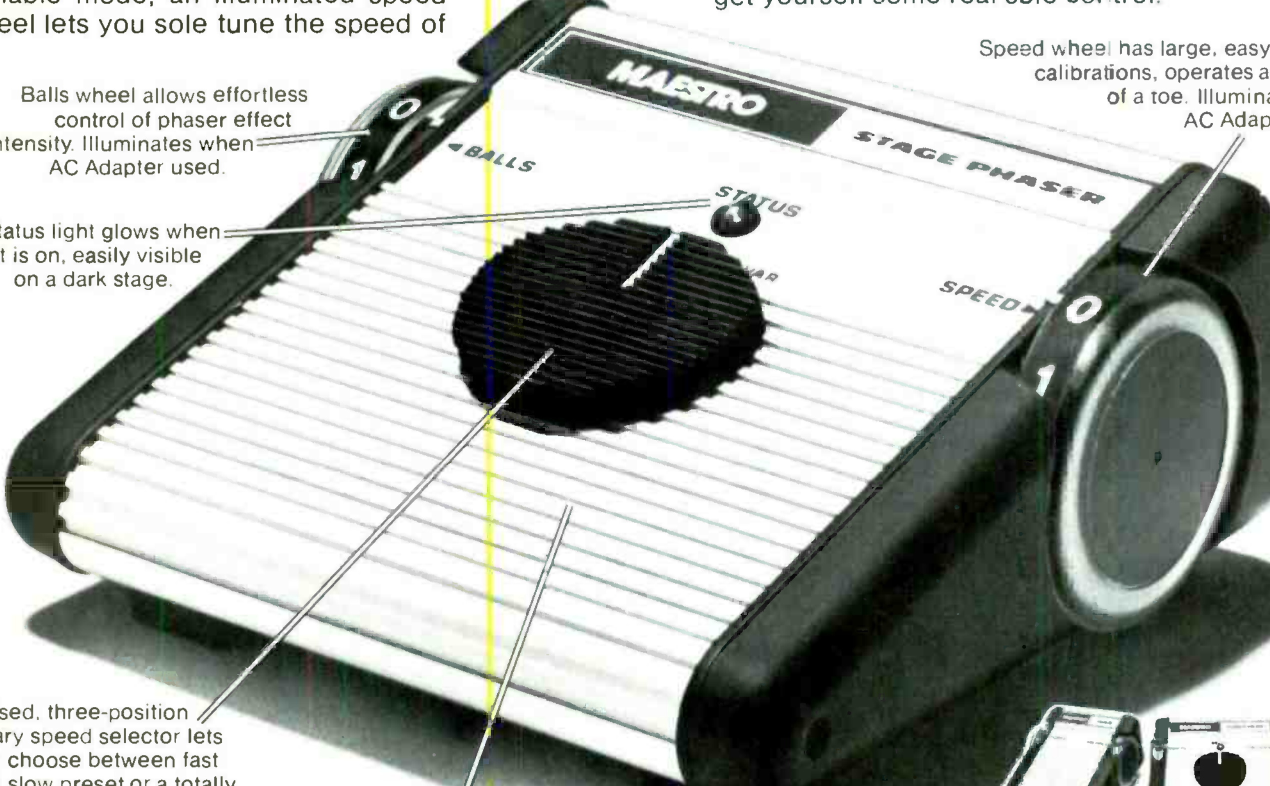
For starters, the Stage Phaser's got Balls. That's an illuminated foot control wheel that lets you effortlessly adjust the intensity of your phasing effect ...from a haunting whisper to a jet-like growl.

Stage Phaser also lets you toe instantly into three phaser speeds—two preset, one totally variable. In the variable mode, an illuminated speed wheel lets you sole tune the speed of your spin to the mood and tempo of your music.

With all these clever controls, where's the on/off switch? Well, there's no little button to grope for. The whole pedal is a switch. ..just step on it anywhere. Even on a dark stage, you can't miss it.

More about the sound. Stage Phaser duplicates the braking and accelerating effect of rotating speaker without a trace of noise or cut in response. And, if you plug it into a two channel setup, you get an eerie, spatial, stereo-like sound that turns your axe into a whole new instrument.

So why settle for some ordinary turn-on? Check out Stage Phaser soon at your local Maestro dealer. And get yourself some real sole control.



Balls wheel allows effortless control of phaser effect intensity. Illuminates when AC Adapter used.

Status light glows when unit is on, easily visible on a dark stage.

Speed wheel has large, easy to read calibrations, operates at the touch of a toe. Illuminates when AC Adapter used.

Raised, three-position rotary speed selector lets you choose between fast and slow preset or a totally variable mode.

Entire pedal acts as on/off control. Step on it anywhere —the massive cast aluminum base holds its ground.

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for trying to keep leakage down to a reasonable minimum.

The whole purpose of going to the effort of multi-track recording is to allow the engineer to re-balance the various tracks later on. In fact, "We'll fix it in the mix" has come to be standard operating procedure. Mix-fixing may mean anything from bringing up a vocal solo to dumping a sour guitar line; to say nothing of adding some sort of special effects to one or more instruments. But of course these changes must be applied only to the instrument in question, and not to the whole musical group. Hence the concern with leakage. If a track is completely free of extraneous information, it can be compressed, expanded, equalized, filtered and—when all else fails—removed entirely, without affecting any of the other sounds on the tape. And of course, if a certain instrument is removed during the mix, there must be no traces of it on other tracks due to leakage.

At first glance it may seem that a cardioid microphone is one's best defense against leakage. Since this microphone is less sensitive to sounds from the rear, it can be positioned so that unwanted signals are 180 degrees off-axis and therefore attenuated by 20 dB or so. However, before reaching for a cardioid microphone, there are a few potential trouble spots to consider.

Off-axis Coloration

Often, the actual shape of the cardioid pattern changes over the range of the audio bandwidth. In fact, at low frequencies the pattern may begin to resemble that of an omni-directional microphone. In practical terms, this means that the microphone efficiently attentuates high-frequency, off-axis signals but does little or nothing to reduce low frequency information. As a result, off-axis leakage will sound "muddy," and if you have several such microphones in use, overall clarity may suffer considerably.

Proximity Effect

Of course, the microphone may be placed very close to the instrument, in which case little or no off-axis signals will be heard, regardless of frequency. But now we encounter another problem. As one moves closer to a cardioid microphone, the bass response often rises significantly, and this is known as the *proximity effect.*

So, no matter which way we move, there's a potential problem. Move further back, and off-axis coloration muddies the overall sound. Move in closer, and the instrument in front of the microphone gets "boomy." And if the microphone is placed in front of a singer, the overall sound quality may keep changing as the artist moves slightly during the recording.

Most manufacturers have at least a few top-of-the-line cardioids in which these two characteristics have been "designed out." Needless to say, you'll pay a premium price for such microphones. Or, you can avoid the whole problem by buying omni-directional microphones in the first place. These don't have off-axis and proximity problems, and that means you can place them extremely close, and not get any increased bass response. And, at a very short microphone-to-musician distance, the microphone will

hear little or no leakage, despite its omni-directional characteristic.

The point to remember is that the cardioid microphone is not necessarily the engineer's best friend. In many cases, the omni-directional microphone will work just as well, or better. And don't overlook the possibilities of the figure-8 microphone. With musicians playing into both front and rear, the microphone is just about immune to leakage arriving at the sides.

Acoustic Baffles

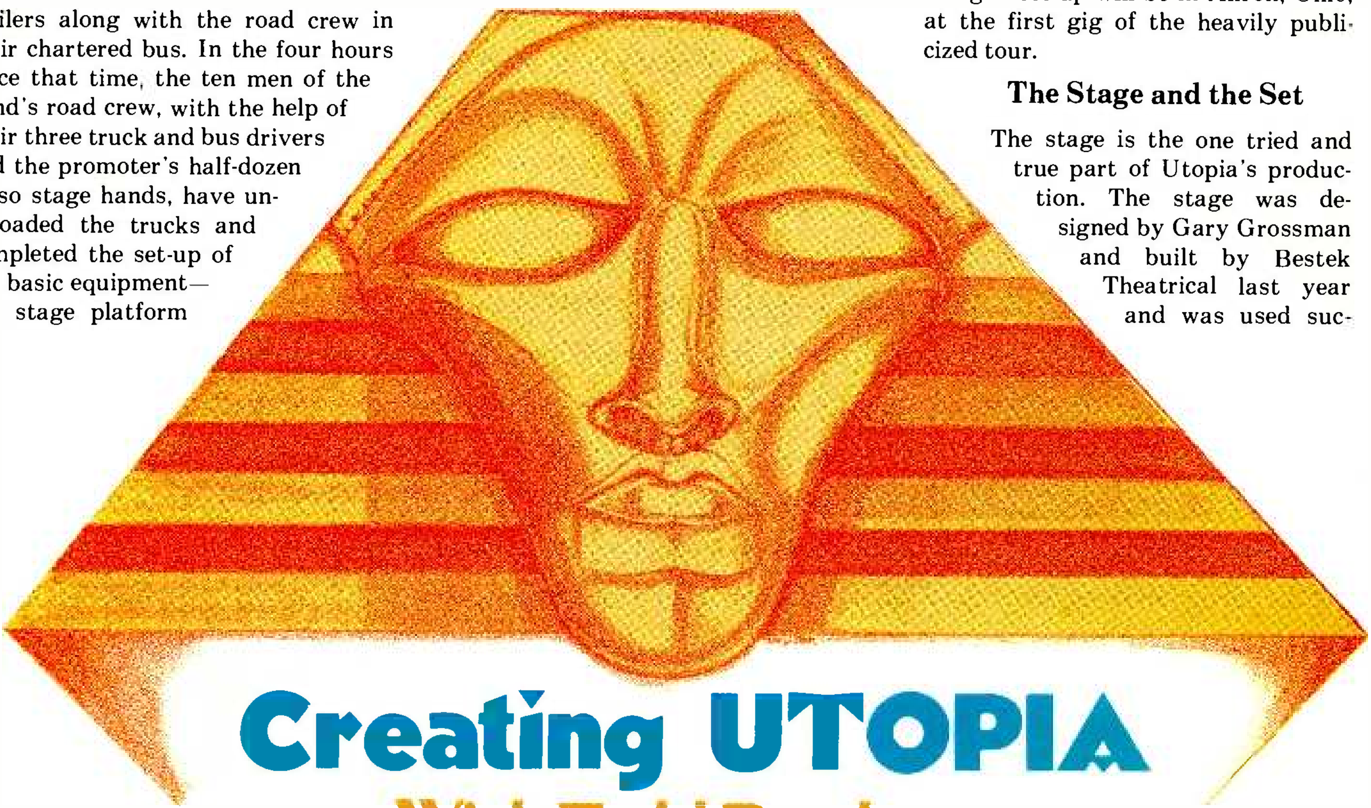
Acoustic baffles—popularly called "goboes"—are often placed between instruments in order to further reduce leakage. However (as with the cardioid microphone), there are a few important points to consider. The first is that goboes are subject to the laws of physics, just like the rest of us. That means that diffraction (the bending of a sound wave as it passes over an obstruction) may present some problems. To explain—as a complex wave passes over the gobo, low frequencies tend to bend around it, while high frequencies pass by relatively unaffected. That means that a microphone placed behind the gobo will be more susceptible to low frequency leakage. And if the microphone is an inexpensive cardioid, so much the worse. Which brings up another point: a cardioid microphone gets its uni-directional characteristic by allowing sounds from behind to enter it's rear- and side-entry ports. If you obstruct these ports—either with your hand, or by placing a gobo nearby—you'll be interfering with the microphone's performance. You can prove this by talking into a cardioid microphone, and while doing so, wrapping your hand around the side-entry ports. The sound quality will immediately go from good to wretched, and if you're on stage you'll probably get a horrendous feedback squeal. A gobo won't be quite as disastrous, but there's a good possibility that it may do more harm than good. And to add insult to injury, most goboes are not perfect absorbers either. So, at least some sound passes through them, while other sounds are reflected from the goboe's surface. So, listen to the microphone, with and without the gobo in place. If it's helping, fine. If it's not, get rid of it.

"Going Direct"

When recording amplified instruments, such as the electric guitar, its possible to by-pass the microphone completely, by "going direct." Many instrument amplifiers feature an auxiliary amplifier jack, originally intended for feeding a second power amplifier. However, a so-called line-matching transformer may be inserted in this jack. With a microphone cable plugged into the other end of the transformer, it directly connects the amplifier to the console (or tape recorder) input.

This all-electrical direct pickup eliminates leakage problems entirely, and produces a sound that many engineers and producers prefer, even when leakage is not a problem. In the case of a noisy amplifier, the transformer may instead be used, via a "Y" connector, at the input side of the amplifier. Although any effects created within the amplifier itself are therefore not recorded, this arrangement often helps get a better sound when the amplifier output is unsuitable for one reason or another.

Creating UTOPIA With Todd Rundgren



It's 2:00 in the afternoon and production coordinator David Rabb and two helpers are completing the erection of the pyramid. Meanwhile, three other workers are assembling the welded tubular framework that will support the sphinx. No, this scene isn't taking place somewhere on the banks of the Nile or in some museum display; it's happening on the stage of the Calderone Concert Hall in Hempstead, N.Y., where the road crew and stage hands are setting up for a concert appearance by Todd Rundgren's Utopia.

The equipment arrived at about 10:00 A.M. in two 40-foot tractor-trailers along with the road crew in their chartered bus. In the four hours since that time, the ten men of the band's road crew, with the help of their three truck and bus drivers and the promoter's half-dozen or so stage hands, have un-unloaded the trucks and completed the set-up of the basic equipment—the stage platform has been set up, the speaker stacks, amplifier racks and mixing consoles for both PA and monitor systems are hooked up and awaiting AC power, and the lighting system is wired up and in position ready to be hoisted into its final position and focused. Yet from this point it will still take most of the six remaining hours until show time to work out all the details and get everything working just right because the show includes an astonishing variety of special staging effects and makes use of several sophisticated technologies, some of which are being used for the first time on this Utopia tour. This is not to say that the road crew or the show itself are untried. In fact, the band and crew have just completed a full month of rehearsals in a rented aircraft hangar at Orange County Airport in Newburgh, N.Y. The hangar was chosen because it offered enough room to set up and use the stage, lights and sound systems without blowing down the walls, and still have plenty of room left over to work on the various parts of the touring system in comfort. According to Todd, "The rehearsals were primarily for the technical crew. We knew the songs from earlier rehearsals and the European tour." Much of the month's time was spent working out what effects to use, how to best implement them, and where in the show to use them, but there was also a lot of time spent solving the inevitable technical problems. The one thing everyone in the band agrees on is that there just wasn't enough time.

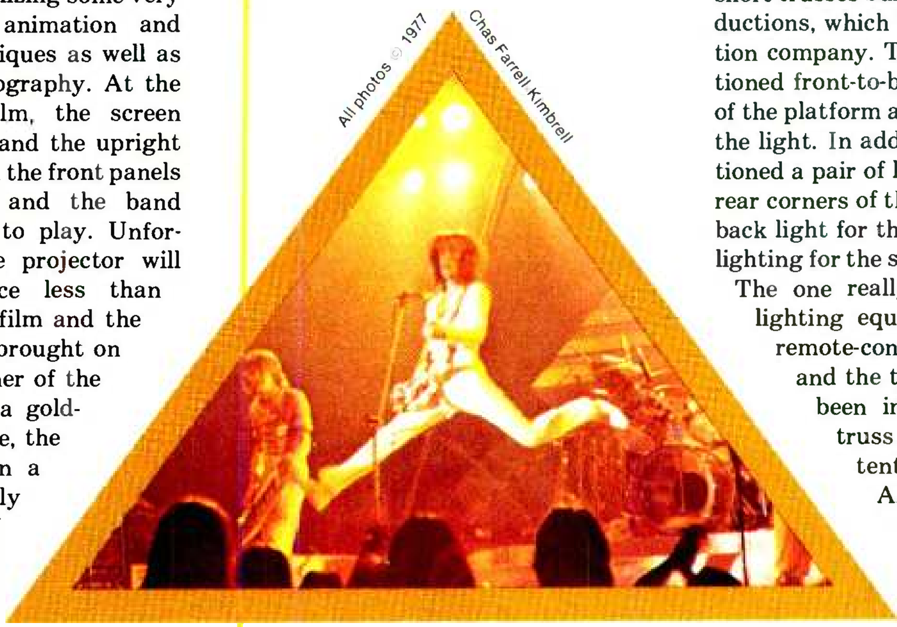
As anyone who has been involved in a performance situation knows, there is a world of difference between rehearsing and taking a show in front of an audience. Twice during the month of rehearsals the band and crew ventured forth from their hangar to make concert appearances and try out parts of the show before the ninety-show tour officially starts. Tonight's show is the third of these virtually unpublicized pre-tour shows, but there are two very significant differences between the earlier shows and the one tonight. First, since some of the effects were only made operational and tried out in the final days of rehearsal, this is the first time the total production will be presented to an audience. And second is the fact that this is the last opportunity the crew will have to fix up anything that doesn't work technically or esthetically; the next time everything is set up will be in Akron, Ohio, at the first gig of the heavily publicized tour.

The Stage and the Set

The stage is the one tried and true part of Utopia's production. The stage was designed by Gary Grossman and built by Bestek Theatrical last year and was used successfully on Utopia's European tour this winter, according to Eric Gardner who used to be production manager for Utopia and who now serves as the band's manager. The stage itself is basically a modular riser some 24 feet square and about two feet high with an additional riser for the drum kit in the center portion of the main platform. The deck surface is covered with a fiberglas-plastic material in a gold metalflake finish, and the underpinnings of the riser are concealed from the audience's view by a series of sloping panels finished in the same gold metalflake material. By using a full riser rather than just a deck covering such as many other groups use, monitor speakers, effects devices and cables can be kept below the level of the stage platform where they will be out of sight, and the result is a singularly uncluttered stage. Also concealed behind the panels at the front of the riser is a 25-foot long box which contains a clever contraption that raises and lowers a 15 by 20 foot front projection screen by means of two long folding arms and a cross-boom. When the audience enters, this screen is in the raised position, concealing the stage riser. The show begins with a 21-minute film featuring each of the four band members and utilizing some very sophisticated video animation and video synthesis techniques as well as conventional cinematography. At the conclusion of the film, the screen automatically lowers and the upright arms fold down behind the front panels revealing the stage and the band members who begin to play. Unfortunately, tonight the projector will lose its light source less than halfway through the film and the band will have to be brought on early. From each corner of the stage platform rises a gold-painted aluminum pole, the four of which join in a small platform directly above the center of the stage to form a 25-foot high pyramid. The bases of the four poles are connected together beneath the platform with aircraft cables for strength. Once he has bolted the poles to the platform, Rabb bounces on the top of the pyramid with his 200-plus pounds to test its sturdiness because at one point during the show Todd will climb the pyramid using the steplike projections on the front uprights. Satisfied with the rigidity, Rabb now attaches a specially built winch to the platform; this winch will be used by Todd in what he has referred to as, "an unusual means of dismounting" from the pyramid.

While the pyramid was being bolted together, other workers were assembling the frame for the sphinx at the back of the stage, and now Rabb supervises the mounting of the sphinx head itself, which is in several sections for ease in transporting and handling. According to Gardner, the sphinx is made of the same plastic material the Disney people developed for their super-lifelike robots, and is finished in gold and turquoise blue. Other than its stunning appearance, one is impressed by the precision with which the eighteen-foot head fits together.

The rationale for the Egyptian motif seems to be fairly complex. First is the obvious tie-in with the title of Utopia's current album, *RA* (who was the Egyptian sun-god) and the lyrics of two of the album's songs ("Communion with the Sun" and "Sunburst Finish"). Additional influences are Todd's long-standing interest in Eastern civilizations and philosophies (an interest he shares with Utopia synthesizer artist Roger Powell), the generally increased



By Fred Ridder

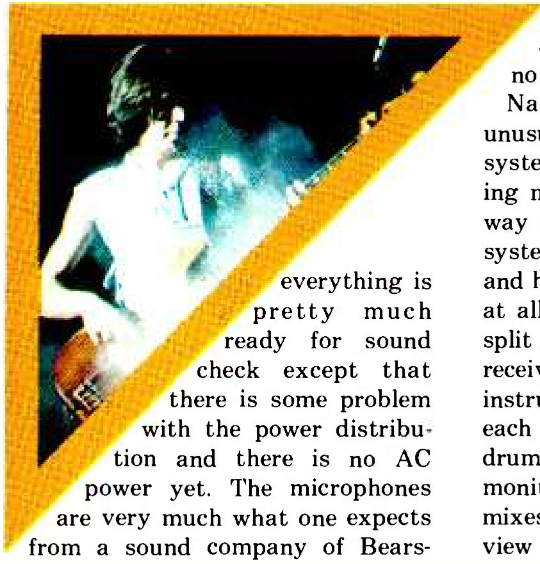
public interest in Egyptian things sparked by the recent exhibition of artifacts from Tutankhamun's tomb, and the pyramid fad. But beyond this and the inevitable bad jokes about guitars staying in tune longer if you play under a pyramid, Todd offered the explanation that "the original Utopia symbol was a triangle with an eye at the top like you find on a dollar bill, and the pyramid is an extension of the triangle into three dimensions." In fact, Utopia's pyramid is rigged with a pair of bright flood lights at the apex which could be considered an equivalent of the all-seeing eye. As for the sphinx, Todd claims that it was "mainly a device to contain some of our effects," and in fact the sphinx's mouth contains monitor speakers for drummer John Wilcox, the nostrils are outlets for the fog and smoke machines, the eyes house strobe lights and a laser beam comes out of the sphinx's third eye.

The Lighting System

While the pyramid and sphinx were going up, lighting designer Barry Cohen of West Road Theatrical Lighting has been working on getting his system ready to focus. Cohen has designed a somewhat unconventional system to meet the specific demands of the Utopia show. Since the usual cross-stage lighting trusses would interfere visually and physically with the set, Cohen had a pair of relatively short trusses built for Alchemedia Productions, which is Rundgren's production company. These trusses are positioned front-to-back just off each side of the platform and provide the bulk of the light. In addition, Cohen has positioned a pair of lighting "trees" at the rear corners of the platform to provide back light for the band and sculptural lighting for the sphinx.

The one really new aspect of the lighting equipment is the use of remote-control color changers, and the three units which have been installed on each side truss are getting lots of attention from Cohen and Alan Stillman, who designed and built them. Stillman, who at one time was master carpenter for Utopia, will be renting his instruments to Alchemedia for certain specific shows where Cohen thinks they will be of most value. Stillman's design, for which a patent is pending, uses pneumatic cylinders to move the gel frames in front of the light rather than the solenoids which previously have been used in color changers. Besides much greater reliability, Stillman's system allows infinite adjustment of changing speed. Tonight will be the first time Utopia has used the changers in a real show, and Cohen is concerned that the devices do not operate fast enough to suit the fast-paced lighting style he has developed for the Utopia show. By the time Stillman is through adjusting his units, the gels are changing in a fraction of a second accompanied by a slight "whoosh" and a not-so-gentle clack as the gel frames swing into place.

By 4:00 the drums and all the microphones are set up on stage and



everything is pretty much ready for sound check except that there is some problem with the power distribution and there is no AC power yet. The microphones are very much what one expects from a sound company of Bearsville's caliber. Shure SM-58s are used for vocals and the drum kit is covered by the usual assortment of Sennheiser 421s and 441s on bass drums and tom-toms, AKG D224s on the snare and hi-hat and AKGs or Sony condenser mics on the cymbals and percussion instruments.

The Monitor System

It is at this point that one might notice that there are no instrument amplifiers visible on stage. This is not because they are concealed as some bands have done for esthetic reasons, it is because *there are no instrument amps!* There has been a trend in recent years to use the smallest amps possible to give the sound man better control over the final sound, and Utopia has taken this idea to its logical conclusion by eliminating onstage amps entirely to give monitor mixer Rob Davis and PA mixer Chris Anderson *complete* control of the sound on stage and in the auditorium. The elimination of the amplifiers was "originally for esthetic reasons," according to Todd, "but ultimately the esthetic may be that we get better sound." Not only is the sound better for the audience due to Anderson having full control, but the musicians can hear one another better at a lower sound level since they are no longer sonically competing.

Naturally, a system of this type puts unusual demands on the monitor system. For this tour, Bearsville is using moderate-sized Cerwin-Vega two-way speaker systems in the monitor system. To reduce intermodulation and help assure adequate sound levels at all times, the monitors have been split into eight separate systems, each receiving a separate mix, basically an instrumental mix and a vocal mix for each of the band members. Todd and drummer John Wilcox each have two monitor speakers, one for each of their mixes, hidden from the audience's view below platform level and in the sphinx respectively. Off to either side of the platform near the front of the stage are twin stacks of four cabinets each which are monitors for Roger Powell and bassist Kasim Sulton, and which also provide side-fill for Todd when he moves away from his own downstage center speakers. Power for the monitor system comes from two racks each containing three Phase Linear 700B amplifiers.

The eight separate mixes required for Utopia's monitors are provided by Bearsville's brand new monitor mixing console, designed by Ted Rothstein partly with this tour in mind. The sophisticated console is physically quite large (over seven feet wide) since it accommodates 40 inputs and has 10 outputs. The input channels are split into two groups of twenty with the master section in the middle. Davis uses one side of the board for vocal inputs and the other for instruments for his own convenience since they are in all cases mixed separately. The console has eight mix buses feeding the eight main outputs for the eight separate amplifier and speaker systems. There are also two master outputs which are mixed from the eight mix buses, and Davis uses these to feed his own monitor speakers. Each input channel offers the flexibility of three-band selectable EQ plus switchable low-pass and high-pass filters in addition to the usual input attenuation and preamp gain controls and the eight color-coded rotary faders for the sends to the eight mix buses. The master section is quite unusual in that it uses Burroughs BarGraph displays instead of conventional VU meters or LED displays. The BarGraphs are interesting gaseous discharge devices which display increasing voltage by sequentially lighting a series of 100 segments. This means that they have much finer resolution than the typical ten- or twelve-LED displays while retaining the speed advantage that electronic displays enjoy over VU meter movements. In addition, Rothstein's design offers five different meter functions including peak metering and conventional VU-type ballistics.

The Laser

The band arrives at the Calderone at around 4:30 and since there is still no AC, they can't start their sound check. After saying hello to various crew members, Todd goes over to an area behind the stage-left speaker stacks where Gary Lemieux has set up the laser equipment which he will operate during the show. Lemieux, who normally works with Cohen's lighting company, is still learning how to run the system since the equipment only arrived in Newburgh from Laser Physics (who assembled the system), in time for the last few days of rehearsals. According to Todd, the equipment they are using now is only the basic part of an elaborate system they hope to be using by summer. What they have now is basically a four-watt blue-green Helium-Neon laser with a fiber optics "light pipe" and a remote control deflection system so that the laser unit itself—and the operator—can be out of sight off the stage platform while the laser beam exits from a third eye in the forehead of the sphinx. Eventually the system will have a digital programmer and a memory unit so that the particular effects may be pre-programmed and then exactly repeated at the appropriate point in the show. Tonight, however, Lemieux will be manually operating the remote controls, so Todd is working with him now to determine the control setting which will produce the effects Todd has in mind.

The laser beam points from the sphinx's head out into the audience, and its exact direction is controlled by the deflection controls which allow the beam to be moved up and down or side to side from its straight-ahead direction. By applying repetitive waveforms to the deflection circuits, the beam can be swept about in a wide variety of patterns much like an X-Y display on an oscilloscope. In fact, for safety reasons, the beam must be kept moving rapidly whenever it is pointing into an audience.

The first effect Todd works out with Lemieux is one in which the beam is swept back and forth very rapidly in the horizontal direction, creating a fan-shaped beam of laser light which is then slowly swept up and down from balcony to orchestra and back. Viewed from the audience the effect is reminiscent of the "time tunnel" sequence in the film *2001*, but is much more three-dimensional since it comes into the observer's space rather than being confined to a movie screen. The other two effects Todd works out are basically spiralling effects, one using an elliptical scan shape and the other a more complicated pattern achieved by using different sweep frequencies for horizontal and vertical deflection. When seen from the audience, both of these effects seem to draw the viewer up into the third eye of the sphinx. All three effects will be more impressive this evening when the air is smoky because the beam will seem to take on a kind of solidity.

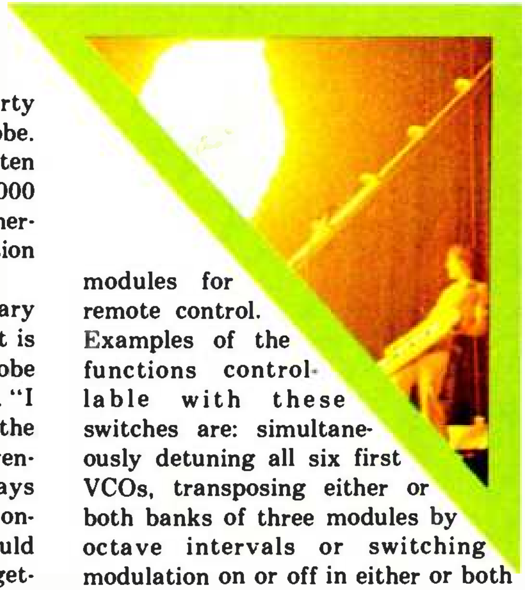
The Powell Probe

While Todd is toying with the laser, Roger Powell has been supervising the set-up of his synthesizer system, which is clearly the most sophisticated of the new technology used by Utopia. The essential differences between Roger's system, which he calls the Powell Probe, and the more common synthesizers are that the Probe is a full polyphonic system with six-note capability, and that a large number of control functions can be remote controlled from a hand-held keyboard unit, which frees Roger from the synthesizer control panel and lets him roam the stage. Physically, the system comprises two racks which house the synthesizer circuitry, the remote-control interface and some additional signal processors and a streamlined fiberglas keyboard unit which weighs only eight pounds. The two racks sit on the stage-right edge of the platform and are connected to the keyboard by a single aerospace-type 19-conductor cable that is only a quarter-inch in diameter.

The idea behind Powell's synthesizer is deceptively simple, but its implementation was bogglingly complex. "I was very naive when I started this project," says Roger, who has worked in product development and promotion for both ARP and Moog in the past. After several false starts, Roger found an engineer in Massachusetts named Jeremy Hill who took on the project. Ultimately it would take fourteen months of Hill's work and the talents of some thirty other people to produce the Probe. Powell plans to build and sell about ten units, if he can find buyers at $15,000 each, and use the money that generates to develop a second-generation all-digital Probe.

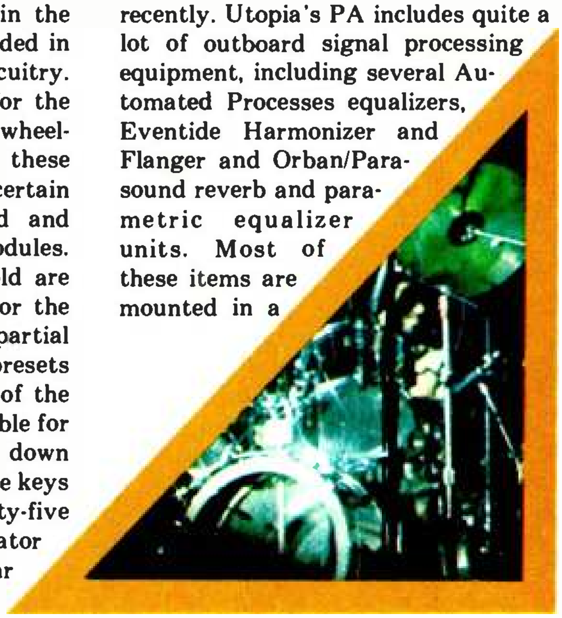
Mobility was not Roger's primary objective initially, but in the end it is what most distinguishes the Probe from other polyphonic synthesizers. "I was tired of having to deal with the mechanical idiosyncrasies of conventional keyboard instruments," says Powell. "So, first I wanted a polyphonic synthesizer with presets that would give me some of the sounds I was getting with other instruments. Then I figured I should take it a step further and separate the keyboard from the actual synthesizers since you can't possibly change all the controls on six synthesizers at once anyway." What eventually resulted was a system that combined partial presets with remote control of certain parameters simultaneously in either or both banks of three synthesizers.

At the heart of the current Probe is a six-voice Oberheim synthesizer with its own keyboard controller unit, both of which have been "extensively modified" to interface with the remote control functions of the Probe keyboard. The keyboard unit itself is an engineering marvel and is an all-new design even down to the keys themselves which use photo-optical isolators instead of switch closures. Information about what combination of keys is depressed is digitally and serially encoded to conserve conductors in the connecting cable and then decoded in the main rack's interface circuitry. There is a hand-hold opening for the left hand which has six thumbwheel-type knobs situated around it; these knobs simultaneously control certain functions such as pitch bend and volume in all six synthesizer modules. Also accessible at the hand-hold are switches and LED indicators for the selection of one of the fourteen partial presets. Each of these partial presets determines certain parameters of the modules and leaves others variable for maximum flexibility. Running down the length of the Probe above the keys themselves is a row of twenty-five rocker switches and LED indicator lights which access particular functions of groups of the modules for remote control. Examples of the functions controllable with these switches are: simultaneously detuning all six first VCOs, transposing either or both banks of three modules by octave intervals or switching modulation on or off in either or both banks of modules.



The PA System

By 5:30, the AC is finally connected, the Bearsville crew has checked out their equipment and everything is ready for the sound check. Todd personally takes charge of the sound check and presides over the PA console, setting EQ and levels for each instrument in turn, while Rob Davis sets the monitor levels according to the musicians' instructions. The house mix is being done on a relatively old 24-input Bearsville board until the new PA mixing console is completed. The old board is a very compact design in two units, and was originally designed and built by Ted Rothstein and Shimon Ron, who was maintenance chief at Electric Lady Studios until recently. Utopia's PA includes quite a lot of outboard signal processing equipment, including several Automated Processes equalizers, Eventide Harmonizer and Flanger and Orban/Parasound reverb and parametric equalizer units. Most of these items are mounted in a



rack to the right of the console and all of them except the Orban parametric, which is used in the PA feed, have their outputs returned to a Yamaha PM-1000 16-input mixer which is slaved to the main console, since most of its own inputs are taken up by microphone inputs.

The PA system uses a novel combination of speakers. Bass frequencies from sub-audibility to 250 Hz are reproduced by a pair of Cerwin-Vega Double-D "Earthquake" speaker systems per side. The band from 250 Hz to 2 kHz is covered by Bearsville's famous midrange cabinets, each of which contains 25 four-inch speakers in a sealed enclosure; there are six of these cabinets per side. Bearsville's treble speakers are the most unusual part of the system, because the drivers in the horn-loaded cabinets are Heil Air Motion Transducers. These drivers are the invention of Dr. Oscar Heil, and are virtually identical to the tweeters used by ESS in their AMT-1 hi-fi speaker systems. Eight treble boxes are used on each side of the stage. Power for each side's speakers comes from two triple Phase Linear 700B racks for a total PA power that approaches 9000 watts.

Todd's Guitar Set-up

Todd Rundgren has always used fairly extensive signal processing on his guitar, but he has recently forsaken the use of the usual guitar accessories in favor of synthesizer modules and studio devices such as the Eventide Clockworks products. "I now use synthesizer modules, E-mu modules to be specific, because they are much more reliable." Considerable original thought went into the actual form of Todd's system. His guitar signal is split and goes to an E-mu preamp, which is used only to generate trigger and gate voltages, and to a Sunn Concert Lead amplifier and 2x12" speaker cabinet which are located offstage. The speaker cabinet is miked and this mic signal is treated as if it were the actual guitar output. The signal from the mic is split and sent to the mixing consoles as a "dry" guitar signal, and to the input of Todd's effects rack. This rack is hidden in the sphinx's left foot, and contains an Eventide Harmonizer, an Eventide Flanger, and E-mu VCA and VCF, plus a patch bay and a set of relays which allows Todd to route his guitar through or past each of the devices by operating a footswitch panel mounted in the deck near his microphone position. The footswitch panel features neon indicator lights for each device (LEDs aren't bright enough on stage) and alongside it is a footpedal which varies the pitch ratio of the Harmonizer. The output of this effects rack is then split and sent to the mixing consoles as an "effects" guitar signal and is also sent to a tape-loop echo device. The echo's output is then sent to the mixing consoles as an "echo" guitar signal, and any mix of the three guitar signals can be dialed up immediately.

At the climax of the band's eighteen minute plus mini-opera, "Singring and the Glass Guitar," Todd climbs to the top of the pyramid, from where he somersaults into space only to be lowered back to the stage by the special winch. For this stunt, a trailing guitar cord presents an obvious hazard, so Todd uses a wireless guitar system designed by Ken Schaffer, who is, among other things, Utopia's PR man.

Schaffer's design is a modified wireless mic system which is being manufactured by Vega, the wireless industry leader. He calls his design a "diversity system" which means that it uses two receivers in different locations to minimize the fade problem. He also uses a mechanical resonator in the front end of the receivers to eliminate virtually all interference problems. The current Schaffer-Vega system works fine with bass guitars or amplified string instruments, but the dynamic range is marginal at best for use with a guitar, and it is for this reason that Todd only uses his wireless on "Singring." Schaffer indicates that Vega is hard at work on a vastly improved compressor/expander to replace the current compander which should make up the current deficiency in dynamic range. He says the new compander should be available in a few months, at which time Todd can go totally wireless, if he wants, without any sacrifice in sound quality.

Roger Powell and the Powell Probe

The Bottom Line

The question inevitably raised by viewing a spectacle such as the Utopia show is, "How much did it cost?" According to Gardner, the cost to Alchemedia Productions and a subsidiary company was somewhat in excess of $150,000, which includes the cost of designs, construction, purchases and rentals of special equipment, but does not include items like the Powell Probe. This figure is not unreasonable for a group that will be consistently playing 10,000 seat and larger venues, but how can Utopia justify such a cost when they will be playing mostly 3,000 to 7,000 seat halls? (In fact, the set is so perfectly scaled for that size hall that it will probably look too small and get lost in a hall much larger than eight or ten thousand seats.) "We should break even after a year," is Todd's comment. "I don't believe in doing anything just to make money." But why go to such lengths to *spend* money? "When you put out a record or give a concert, you're actually asking people to give you money, and that's a heavy karma. In return for their giving me their money, I want to give them a total experience, something special that they can only experience in this particular space and time."

Yet there seems to be another reason for putting together such an elaborate production. Says Todd, "Technical innovation helps keep me interested in performing the same show each night on tour. And danger is even better—the element of chance, the unpredictable. You know, the top of the pyramid doesn't get any closer to the stage as the tour goes on. I may break my neck some night." Whether or not the Utopia stage show succeeds in sustaining Todd Rundgren's interest, one thing is certain—Utopia will leave a lot of satisfied audiences in its wake.

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NOISE REDUCTION

Part I

By George Klabin

What do we mean by *noise reduction*? In simple terms, the noise reduction process is one which reduces or eliminates audible noise such as hiss or hum which can be caused by a variety of sources—most commonly thermal noise in transistors or AC hum components of signals.

The current usable dynamic range of multi-track tape recorders (without use of any noise reduction) is around 65 decibels, from the noise "floor" (residual noise level) to the maximum undistorted signal level, commonly referred to in tape recording as the point of 3% harmonic distortion.

For many years everyone lived with audible hiss on recordings, since 65 decibels is hardly sufficient to capture the full dynamics of most music, which requires around 100 dB minimum.

George Klabin has been a recording engineer for twelve years. He has worked with the Dolby, dbx, Burwen Noise Filter, Phase Linear Auto-Correlator, and Kepex noise reduction systems for many years, and has engineered countless recordings with them.

Since tape recorders could not contain all those dynamics, people learned to accept hiss as "part of the sound." In addition, limiters and substantial manual gain riding were necessary to "compress" the actual range into the allowable one of tape. Anyone who attempted to get just a bit more level on the tape risked greatly increased distortion, or, in order to accomplish that greater level, had to record at very high speeds. Thus it was a great and important challenge to invent a method for increasing the dynamic range of tape recording.

Reduction Introduction

As early as the beginning of the forties, Fairchild [Robins Inds. Corp.] developed a "compander," a device which attempted to restore dynamic range. It was placed between the power amplifier and loudspeaker and operated passively on the resistance characteristic of the filament of a pilot lamp. In a primitive sense, this was the first noise reduction unit, because it attempted to increase the dynamic range without increasing the noise level. The device did not work very well and was really intended for home use and not for tape recording.

Sometime in the 1950's Scott unveiled their "Dynaural Noise Suppressor," a device aimed more directly at reduction of hiss. It rolled off highs (and with them high-frequency noise) only when program levels fell below a preset threshold at which the music was no longer loud enough to "mask" the noise. While the device did work, it also introduced severe "breathing" effects, due to its slow release time which caused a hiss "trail" to be heard as the sound faded out. This effect was often more unpleasant and obvious than steady hiss which, provided it remains at the same level, becomes absorbed (psychologically) into the "background."

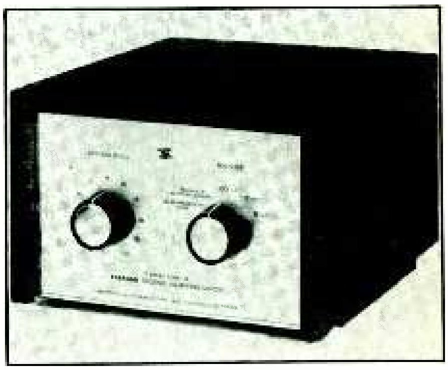
The main problem with noise reduction systems of the forties and fifties was the lack of sophisticated technology necessary to make them work properly. Later on, also in the fifties, another approach was developed. Fairchild brought out the "Auto-Ten," an automatic attenuator which was supposed to turn off a signal when it died down to a settable threshold. The idea was to aid the engineer when he had a multiple number of mics or inputs to mix, by adding a third "hand" to shut off those channels which had no usable information on them at any given moment. The unit, of course, would turn the channels back on immediately upon resumption of any signal, and do it fast enough to not clip (distort) any part of the sound. The device used a solid state, light dependent sensing cell in conjunction with a variable release time settable by the user to fade the signal down at various speeds—depending on the type of sound being processed. While Fairchild's intentions were good, the results from the device were not. Often the device would actually introduce a horrible type of distortion when the signal, fading down, got "caught" fluctuating around the threshold point. This would cause the Auto-Ten to start and stop fading so quickly that it sounded as though it were chopping up the signal. The device also failed in other situations. For instance, say it was across a saxophone's mic channel, and was supposed to turn off the channel when the sax was not playing. A transient, or continually loud leakage sound from a nearby instrument would turn the channel *on* again, reintroducing the leakage onto the recording. Setting the threshold higher to compensate for the loud leakage only caused the unit not to turn off at all. So the benefits of the device were limited, and the user had to be careful, for he could actually degrade the audio signals rather than improve them.

Noise vs. VCA

Several years later, Paul Buff of Allison Research developed the "Kepex"—a device which operated on the same principles as the Auto-Ten. However, it benefitted from the advanced technology of the early 1970's as well as from the invention of the VCA (voltage controlled amplifier). The attack time, variable release time and range settings were all more useful and functioned more smoothly than with the Fairchild unit. The Kepex is still used in many studios because it successfully reduces noise on certain types of material, but primarily because it has an additional feature from which it derives its name, that being a "keying input." The advent of voltage controlled amplifiers allowed the Kepex to be "keyed" on and off by an external DC voltage, or by the DC voltage component of audio signals. Thus one could create stunning and novel effects like turning a tack piano track on and off by the bass drum track's dynamics; or have one of three vocal tracks control the stopping and starting of the other two; or even creating automatic fades by the introduction of a remote DC voltage. Though the Kepex is a noise gate, it is most useful in effects generation, and has use limitation similar to the Auto-Ten.

Many imitators flooded the market with noise gates as technology advanced and VCAs became cheaper. By the mid seventies there were many companies making noise gates at a low price, among them Flickinger with its "Noise-X" (not to be confused with the EMT "Noisex" marketed in the fifties and sixties), Noramp and its "Noise Reduction Amplifier"; and even MCI with a noise gate on a plug-in card ready to fit into their tape decks! It was only when Ray Dolby, encouraged by Decca of Great Britian, developed his "Dolby Noise Reduction System," in the mid '60's, that noise reduction was to become truly acceptable and reliable for the purpose it was intended —reducing noise from recordings. Finally someone had found a way to remove a significant portion of the noise while minimally affecting the sound. A whole new approach was born—the *Before the Fact* system. In the earlier *After the Fact* types we are dealing with devices to eliminate or reduce noise from a pre-recorded source. With Dolby, as long as the device was used before the recording medium to encode the signal and then again upon playback after the medium to decode it, 10 to 12 decibels of noise were eliminated from the recording, effectively giving a usable signal-to-noise ratio of 75 to 77 dB! At the time it appeared, the Dolby's additional dynamic range was regarded as a great achievement and sufficiently useful so that, with judicious gain riding, very quiet clean recordings could be produced. The full dynamics of music still could not be reproduced, but it was a great improvement.

However, there is of course the one great disadvantage of *not* using noise reduction—hiss. With the current state of tape quality, it is possible to achieve a usable dynamic range of about 71-72 dB. This could be done without the use of noise reduction by recording on the latest Ampex "Grand Master" or 3M "250" series tapes at 30 ips, with a perfectly aligned, top quality tape recorder. This doesn't equal the 77 dB maximum signal-to-noise ratio obtainable with Dolby, but for purists it at least is a significant improvement in the overall noise level. Still, there are serious problems with these new tape formulations with regards to reel-to-reel quality control and print-through level, and these problems have yet to be overcome. And, of course, though reduced, there is still hiss. If in mixing such a tape one decides to make a significant mid-range or high-end frequency boost on one track, or to add moderate limiting, all the gains may be defeated and the signal-to-noise ratio will be lowered to the vicinity of the usual 65 dB range—thus defeating all the extra efforts to gain the 6-7 dB. Those "efforts" include recording at higher levels to obtain the benefits of the new tape formulas, and thereby risking saturation and greater distortion, doubling of tape cost to record at 30 ips, the cost of a machine with 30 ips capability, and much extra time to critically check alignment, bias, EQ and levels. If that isn't enough to discourage many people who do not have the money or time to have a maintainance man around at all times constantly checking, two additional evils which plague all tape recording are still present: asperity noise and modulation noise. Asperity nose is created by minute imperfections in the tape surface which are caused by variations in the size of the oxide particles of the coating. These imperfections increase or decrease the strength of the magnetic field passing



Scott Dynaural Noise Suppressor

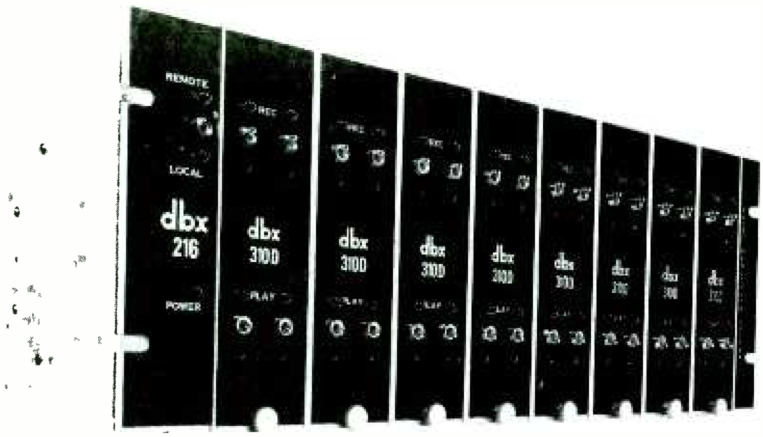
the head and result in audible noise, mostly in the mid- and high-frequency area. This noise may be present even when no program is recorded on tape. Modulation noise occurs when the asperity noise becomes superimposed on the signal; sounding like a swishing sort of background noise which is mostly audible when there are only low-frequency signals present. These lows tend to modulate or change the level of the higher frequency asperity noise. Fortunately, in many cases both asperity and modulation noises are masked by steady state hiss, which, of course, exists on recordings made without noise reduction.

Then we have the alternative—using noise reduction. What does one gain? In the most advanced systems, such as dbx or the new Telefunken, the usable dynamic range increases to 95-100 decibels, close to the full range necessary to record most music. There is no audible tape hiss, and recording may take place at lower levels, where tape distortion is lower and the danger of peaks saturating the tape is non-existent. Hum, crosstalk and print-through effects are also reduced to inaudibility.

AD (After Dolby)

The reign of Dolby as the system allowing the largest amount of usable noise reduction lasted several years, until in the early '70's the dbx system was developed. Since that time we also have seen introduced the Burwen model 2000 "Noise Eliminator," a terribly expensive system which delivered up to 110 dB usable dynamic range, but which was so overpriced it almost had to fail, as it did several years ago. People simply weren't willing to spend $6500 for two channels or over $50,000 for 16 channels. Recently in late 1976 Telefunken introduced their new C4 noise reduction unit, which they claim is a great improvement over all other systems

In the *After the Fact* field, advances continued to be made as well. The most significant was Burwen's model 1000 "Noise Filter," the early (and expensive) version of the more consumer-oriented models 1000 and 1201 which appeared later. This filter was a great improvement over the Scott "Dynaural Noise Suppressor," and I shall discuss it later in this article. In the past three years, with the incredible developments and lowered costs in technology, many newcomers have jumped on the *After the Fact* bandwagon, including Phillips, with its "DNL" (Dynamic Noise Limiting), and Phase-Linear, with its "Auto-Correlator."



dbx Noise Reduction System

Pros and Cons

Before I continue by exploring the *After the Fact* systems, let's consider whether there are any tradeoffs in quality when using noise reduction. The answer is yes. For those interested in recording "precisely" the signal coming into the mics and console on tape, with minimum changes in distortion and few side effects, noise reduction should not be used. It does cause slight changes in sound which may be audible to some people at some times. I speak of comparing a perfectly aligned and operating tape recorder with a perfectly aligned and operating noise reduction system.

When using noise reduction, the importance of aligning all recording equipment is greater than normal because another link in the recording chain has been added. Proper recorder frequency response and levels are critical in the various noise reduction systems, as errors can be magnified.

There is another tradeoff in using noise reduction, but it's purely psychological. It has been proven in tests that people feel that music recorded and played back without any hiss seems to have less highs than the same music with a steady hiss level. It also has been proven that, with proper use of noise reduction, there is absolutely no loss of highs or harmonics. People simply are still accustomed to mentally relating the sound of hiss with high frequency sounds in music. This is understandable since hiss contains a rich mixture of mid and high frequencies. On certain sounds such as a high-pitched cymbal ringing out, the hiss, which becomes increasingly apparent as the cymbal sound trails off, seems to be a continuation of that sound. Remove the hiss completely, play the same sound again, dying off into nothingness this time, and a person not familiar with noise-reduced recordings will probably feel that some of the harmonics of the cymbal have been removed or lost.

Fortunately, with the huge increase in noise reduction use and its growing popularity in the consumer market, many people are finally becoming accustomed to and even demanding noise-reduced recordings in their homes. Noise reduction is a reality; it must be dealt with, and I venture to stick my neck out and say that any studio owner who wants to do large-scale business in recording must at least learn to use and have available some sort of effective noise reduction equipment, whether he likes it or not, and whether or not he uses it on his own personal recordings. Recording has become the creation of an illusion, and in fact has never been able to precisely reproduce what our two ears hear when we are standing in a room. To achieve that effect one must record with a human headset binaural mic directly onto two tracks, perfectly aligned and with no additional equalization or processing of any kind, and monitor only through human headset type stereo headphones. This would rather limit the recording and listening experience. Thus recording reality is whatever we want it to be, and we can shape and change that reality almost infinitely. Noise reduction has a valid and useful place in that realm. Whatever minimal changes in the signal noise reduction introduces are offset by the tremendous gain in useful dynamic range.

After the Fact

Among the *After the Fact* devices which reduce noise on an already recorded source, there are three main systems which are most important: the Burwen Noise Filter, a dynamic filtering system; the Phase Linear Auto-Correlator, a different and more sophisticated approach to dynamic filtering; and the new SAE "Impulse Noise Reduction System," which is designed to remove sharp clicks and pops from disc sources.

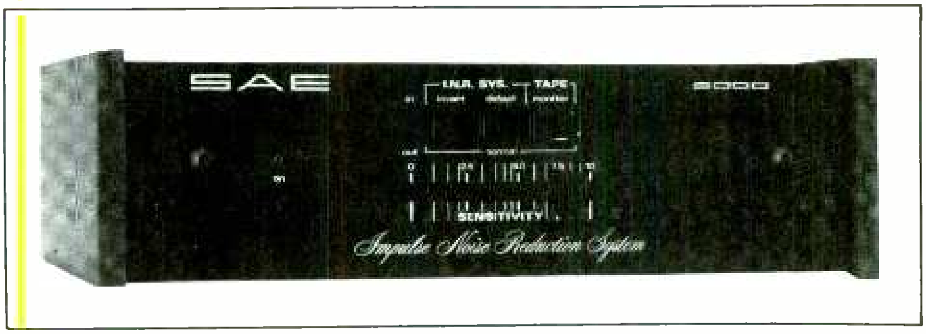
The first of these was the Burwen Noise Filter. In the early '70's Burwen brought out the model 1000 Noise Filter, a very expensive ($3300 for two channels) device which was a tremendous improvement on the old Scott Dynaural Noise Suppressor. In the past, the most effective way to reduce noise in any frequency range was to reduce the overall signal level in that range at a given moment, thus reducing the noise. This approach also reduced the signal content in the area affected and often took with it wanted musical energy. Burwen attempted a solution by designing a filter which attenuated low and high frequencies separately when no program material was present at a certain threshold. As the high frequency content of the signal increased, the upper cutoff frequency also increased with it from 1100 Hz to 32,000 Hz. Similarly, low frequency content caused an increase in low frequency bandwidth from 350 Hz to 13 Hz. In other words, with no material present above the threshold, the Burwen "closed down" to an effective bandwidth of only 350 Hz to 1100 Hz, allowing a tremendous reduction in hiss and hum—which occurs mainly outside that region. The most important aspect of this design is how quickly (or slowly) the filter responds (opens or closes) to musical energy at the preset threshold. To solve this problem, Burwen designed the threshold device so that the high frequency filtering began only at -35 dB (below +4 dBm) and the low frequency filtering began below -22 dB, though both these thresholds are somewhat variable by separate low and high sensitivity controls. Burwen decided on a frequency of 6.6 kHz for the high end and 85 Hz for the low end as determiners of threshold level, and energy around those frequencies really operated the filter's opening and closing functions. The circuit did respond to other frequencies as well, but at much lesser sensitivity. At levels above the thresholds selected, the unit had flat response and relied on the well-known "masking effect" previously described.

The transient response of the Burwen filter also was critical because if the attack time were not fast enough, part of the transients would be clipped, and if it were too fast it would respond to impulse noises such as clicks or pops. The attack time chosen for high frequencies was one millisecond, a compromise figure which seemed to work well. The attack time for the low frequency section was of necessity longer—ten milliseconds so as not to distort any part of a low frequency waveform. The decay time of the filter was equally critical—too fast and modulation of the signal might occur, too slow and hiss or breathing effects might be heard. A decay of 50 milliseconds was chosen for high frequencies and 500 milliseconds at low frequencies.

In his later and less expensive model 1100, Burwen added the choice of a faster attack time (400 microseconds) for tape processing and one millisecond for disc. He added a slower attack of two milliseconds for 78 rpm records, with their longer duration impulse noises. Decay time for the filtering is also slower at the 78 rpm setting to allow the filter to remove highs more gradually. An even more inexpensive consumer version, model 1201, had much the same function as the 1100, but with reduced input level capability, making it suitable only for home use. [The model 1201 has—since Burwen's recent merger with KLH—been replaced by a much improved model 1201A.]

The Burwen Noise Filter works well in many applications, because it does indeed remove hiss and hum without removing highs or lows. However, like any filter with fixed operating parameters, there are times when it can be "fooled." The sensitivity control has to be adjusted by the user and if not properly done, highs or lows will indeed be removed, or hiss will breathe along with the signal. The most severe tests for this dynamic filter design is solo piano, flute or cymbals. In some cases very audible "hiss breathing" can be heard as the piano or flute sound decays, and with cymbals especially it is possible to remove some of the actual harmonics if sensitivity is set too high. Remember though, when listening to a piano, that the sound of the dampers reseating on the strings can often be mistaken for "breathing."

To use the Burwen Noise Filter properly it should be connected between the output of the two channel mix from the console and the input to the tape recorder. Thus one mixes through the device. The easiest method by which to monitor the effect of the filter, is to listen to the two-track tape recorders' input. In home use the filter may be connected to the tape in and out jacks on an amplifier, or from preamp out to monitor amp in.



SAE Impulse Noise Reduction System

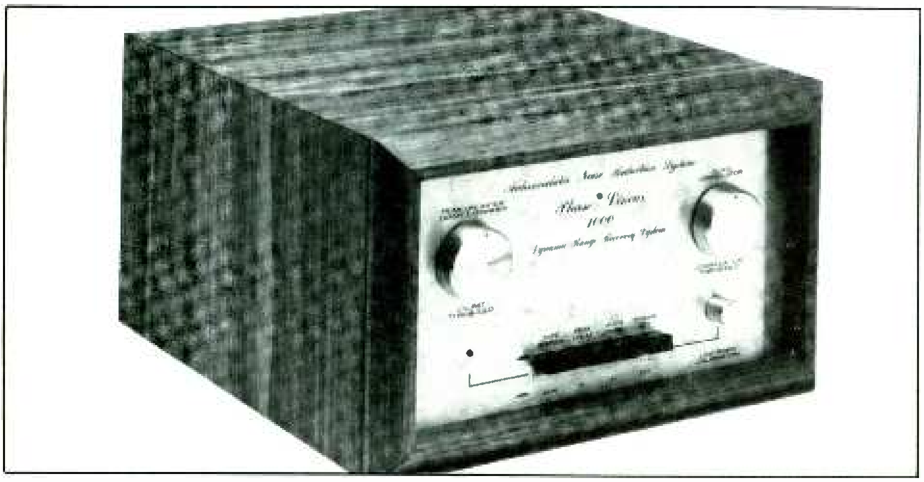
Energy Bundles

Just two years ago Phase Linear, a company known for its high power monitor amplifiers, developed a fascinating new approach to noise reduction, one which actually surpassed the usability of the Burwen filters, and at a very low price. The Auto Correlator is a development of the noise filter approach, but one in which the circuitry actually attempts to distinguish, electronically, between noise and signal in the same frequency bands and at or near the same levels. Bob Carver, president of Phase Linear, related the theory behind the device to

Bert Whyte in an article appearing in *Audio* magazine in 1975.

"Music energy appears in discrete energy 'bundles' throughout the audio band and is therefore not continuous. In addition, if some musical energy appears, for example, at a particular frequency, we know for certain that even and odd harmonics will exist simultaneously throughout the passband, and that energy will not exist between these harmonics. In other words, with music we are able to 'predict' where energy is likely to occur, if we know where the fundamental is, or even if we know where only one of the harmonics is. Also, and importantly, we know where the energy will not appear. In other words, music is coherent, or correlated."

Pure hiss is totally uncorrelated, thus distinguishing it from music in its randomness and unpredictable characteristics. The complex logic circuit in the Auto Correlator is designed as a series of "windows" or electronic gates, each of which controls a certain frequency range and each of which just overlaps its neighboring "window." The gates open or close, depending on the logic circuit which opens it for any correlated energy and keeps it closed for uncorrelated energy. The circuit is thus actually trying to predict where the overtone structure of a given sensed fundamental should be and open up the proper "gates" to let all the predicted harmonics through. Every little "bundle" of energy is being critically examined in real time and decisions are constantly being made whether to suppress it by 10 dB, or let it pass through. There is a threshold circuit control for the high and low frequencies and it allows the unit to become increasingly critical in deciding what to let through and what to attenuate by 10 dB. If the setting is too high, the noise will go through; if too low, some of the music will be attenuated with the noise. In use, the control is not too critical for most music, and the unit works amazingly well. It is most useful on program material containing moderate to low levels of hiss or rumble, because in these instances it effectively removes enough hiss or hum to eliminate it entirely. It reduces noise across the bandwidth of 20-20,000 Hz up to 10 dB—a sizeable chunk. On noisier material, the action of the device seems noticeable because the hiss level is moving up and down, and is still audible even at maximum attenuation, resulting in a "breathing" effect. This grainy coarseness in the sound can be objectionable, and limits the use of the unit on material with high noise content. However, for mixdown applications with relatively clean, well-recorded music with an existing signal-to-noise ratio of 55-65 dB, the Auto Correlator will add about 10 dB to that range.



Phase Linear Autocorrelator

The Auto Correlator has another function which makes it even more useful, and which allows an actual increase in dynamic range of up to 17.5 dB instead of the 1 dB for the noise reduction section alone. This benefit is derived from the "peak unlimiter" and "downward expander" section. In simple terms, the peak unlimiter can be adjusted with a variable threshold to actually add 1.5 dB of gain to program peaks. At levels just below these peaks the gain is unchanged, and during moderately soft passages the system gain is changed so that the overall output is altered by 11 dB for every 10 dB of change. The cumulative maximum change is 33 dB for a change of 30 dB, in a linear expansion mode. Then there is the downward expander which adds another 3 dB of downward level drop during very soft passages. By making all the expansion operations so small, no individual one causes noticeable alteration of the sound, and yet taken as a whole the improvement in signal to noise ratio is 7.5 dB. Combine this with the 10 dB from the auto correlator section and you have 17.5 dB, a significant amount. Audibly, the effect is a wider variation in levels than the original as it approaches the true dynamics of the music before it was subjected to the recording process.

There are some drawbacks to the Auto Correlator too. As previously stated, very noisy program material will not benefit from its noise reduction process. Also, the peak unlimiter, downward expander section can accentuate rather than suppress noise in noisy program material. The current model of the unit, available separately as model 1000, allows only 3V maximum input before clipping, too low a level for use in some recording applications. This limitation can be overcome by mixing through the device at lower levels, then boosting levels at the input of the tape recorder. This method may defeat some of the signal-to-noise gain of using the unit, but it is still useful. I have heard that Phase Linear plans to market a professional version with higher input and output levels. The current suggested price of $350.00 for the model 1000 makes it a very attractive and useful addition to any studio.

To gain the most from any dynamic filtering noise reduction system, one must be able to record the noise-reduced material on some medium, normally tape, unless one is simply listening to audio at home. Since the signal-to-noise ratio of tape is limited, much of the noise removed by the filter would reappear on the tape mixdown. Therefore one should use dynamic noise filters with some sort of *Before the Fact* noise reduction system which encodes the signal and preserves the wider dynamic range.

Keep in mind another useful aspect of *After the Fact* systems which is unique to their design. They can remove hiss and hum from any source. Thus they can be used while recording, across any instrument or channel which has objectionable hiss or hum. A

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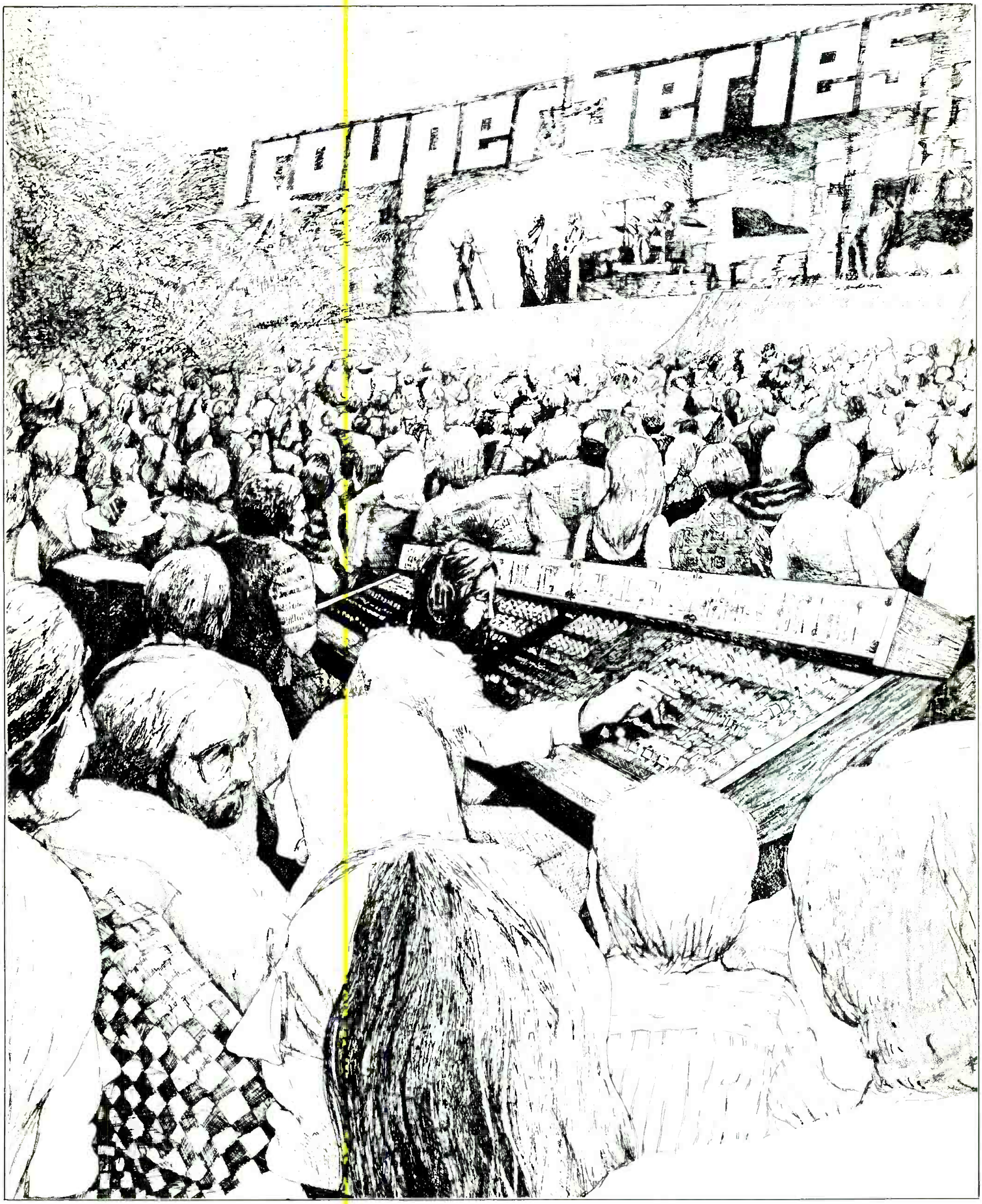

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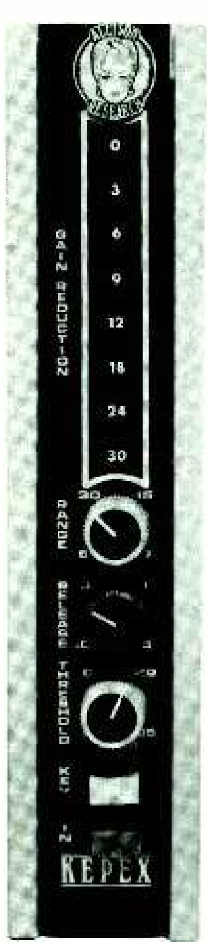
common use would be across a hissy guitar amp, perhaps one which has a fuzz pedal or phaser attached to it, because such units often increase the noise level. Unavoidable 60 cycle hum from transformers on a "live" recording, or room rumble from air conditioners can also be successfully



Allison Research Kepex

removed. Just remember to use the particular filter at the level which it is designed to work into, so that no overloading will occur.

Impulse Noise

The latest approach to noise reduction is a device aimed at removing clicks and pops from records without affecting the sound. SAE, a company with a reputation for superb quality products, has introduced their Impulse Noise Reduction System. Impulse noise is present on any record in the form of static electricity being discharged on the needle, minute gouges or imperfections in the vinyl surface, and scratches. By finding those properties of impulse noise which differ from music and designing a circuit to sense and eliminate them, SAE claims it can actually remove annoying pops and clicks and yet not affect the music. The main characteristics of impulse noise are its very fast attack (rise) and decay (fall) times. These tiny "spikes" of noise occur within a matter of microseconds (millionths of a second); and the SAE unit has a logic circuit which monitors the music, constantly searching for the required conditions of impulse noise as they have defined it. When an impulse noise is detected, a command signal is sent to the impulse removal circuit to remove the noise by turning off the signal for an instant. The maximum duration of any impulse noise that the unit will work on is one millisecond. During the instant that the music is turned off, a music restoration circuit operates by actually inserting preceding information into the "gap." This preceding information is taken from an analog delay circuit with a fixed delay of 100 milliseconds. The supposed result is a reduction or removal by as much as 60 dB of the noise "spike" without affecting the dynamics of the music.

I tried the device with several types of records—new and old. Ones with steady hiss from a poor vinyl pressing (accompanied by ticks and pops); ones with perfectly quiet surfaces where the ticks were mainly static discharges; and various ones in-between these conditions. The only control is a sensitivity slider pot, which controls the level of noise which is sent to the detection circuit. The maximum "trip" threshold is about 200 millivolts. There also is an important "invert" feature which allows one to hear only the ticks and pops being removed, so that sensitivity can be adjusted. The idea is to set the unit so that sharp transients in music, which are very similar to impulse noises, will not also be removed with the spikes. In the "invert" mode one can easily learn to distinguish the transients from the noise by their loud, longer "ripping" sounds. Backing down on the sensitivity level eliminates these transients from being processed. If the sensitivity is set too high the audible effect on the music is a slight loss of crispness, and dulling of the beginning of any loud sound, such as a trumpet or percussion attack. Setting the sensitivity even higher can introduce loud ripping sounds in the music. Lowering the sensitivity will allow all transients to pass through untouched, but may also allow many ticks and pops through as well. One other drawback is that the noise removal circuit takes a certain amount of time to "work"—about three-fourths of a second to one second. Thus any clicks and pops which are spaced closer than that duration (SAE says it compares to a one-fourth revolution on the outer edge of a 12-inch LP) will not be removed totally.

The nature of impulse noise is so varied, in the amounts of attack and decay time, the phase, amplitude and frequency relationships, and its confusing similarity with many music transients, that it seems an impossible task to take a unit with fixed operating parameters and ask it to totally remove only pops and ticks without affecting program material. The sensitivity control is critical, and has to be readjusted often. I found the unit did not remove many impulse noises unless it was set so that it also affected music transients, making it rather more of a noise "induction" system than noise "reduction."

An area where this device may prove useful is in removing tiny clicks on a tape track during mixing. It may also be able to remove distorted transients if you purposely set the sensitivity too high.

Overall, despite the good intentions and great reputation of SAE, and the low price of about $200.00, the Impulse Noise Reduction unit does not yet seem quite complete. I feel we will have to wait for further technological breakthroughs before being truly able to accomplish what SAE claims the unit does.

In conclusion, upon reviewing all the *After the Fact* systems of noise reduction, their usefulness seems to be significant in removing some hum and hiss from music without affecting sounds, provided they are of the dynamic filter design. Their relatively low cost shows that technology has advanced to the point where one can have the benefit of professional quality at consumer prices, and I expect that we shall soon see even more sophisticated "brains" developed to distinguish noise from music.

In Part Two, in the next issue, I will explore in depth the three main brands of *Before the Fact* noise reduction—Dolby, dbx and the new Telcom C4. I will also discuss the methods for getting the best performance from these systems.

Ambient Sound

BY LEN FELDMAN

Pre- and Post-Equalization

Now that the graphic equalizer has moved out of the professional sound reinforcement and recording studio arena and into the homes of audiophiles, semi-pro recordists and not-so-sophisticated hi-fi buffs, it might be a good idea to examine some of the benefits and pitfalls associated with the use of this latest "black box" add-on.

In its sound-reinforcement applications, equalization has been used in P.A. systems for many years to permit higher gain-settings on amplifiers in auditoriums where microphone/speaker feedback is a problem. Generally, as P.A. system gain is raised higher and higher, there appear one or more resonant frequencies which result in the familiar "squeal" or "howl" that quickly builds up to amplifier (and ear) overload. Third-octave equalizers can do wonders here, for even without the aid of expensive real-time analyzers, a sound contractor can usually identify the approximate offending frequency or frequencies. Then, a small reduction of gain at those frequencies, accomplished with the aid of those marvelous lever controls, stops the feedback and permits the system's gain to be raised another few dB. It is not uncommon for system gain to be improved by ten or more dB by careful adjustment—always *downward*—of a few of the appropriate lever switches. Overall sound system frequency response is not the issue here. The important thing is to get higher overall P.A. system amplification so that full coverage and intelligibility can be imparted to the hall or auditorium in which the sound system is installed. Nor is power output capability of the sound reinforcement amplifier a problem for, as we said, in most instances of P.A. system equalization, the levers are pulled *downward* and not adjusted to provide boost at any of the frequency bands available on the particular equalizer.

Now, let's consider what happens in the relatively new environment of home or recording studio type equalizers. I'd be willing to bet that if you own an equalizer (or know someone who does) and take a look at the present setting of its many slide controls you will find that if any of them are *not* set in their center or "flat" positions they are probably adjusted to some level of boost. Possibly, some of the band controls (usually the lowest bass octave or the highest treble lever) are even boosted all the way. In the case of most of the "home type" equalizers, that means a boost at those end-of-spectrum frequency ranges of 10 or even 15 dB! Intermediate levers (be they at the upper bass, mid-range or lower treble) will also occasionally be found set to boost positions—perhaps 4 to 6 dB worth. About the only levers that find themselves set to "cut" positions are the lower-mid lever or levers which are pushed downward to get rid of the typical "standing wave" humps found in smaller listening rooms and recording studio playback rooms. What effect does this have on the rest of your playback system?

Let's consider the case of an equalizer which has its first and second levers pushed up around 10 dB (the usual excuse is that the woofer of the system just doesn't have enough output at the lowermost octaves and you are missing that gut-bass). Suppose your amplifier has an output power capability of 100 watts per channel, and you are coasting along at a comfortable, at what you *think* is a comfortable, *average* music power level of around two watts per channel. Allowing for a 10 dB crest factor in music, when those loud peaks in the music occur, you'd expect that the amplifier will be called upon to deliver 20 watts or so. No problem—so long as that equalizer is out of the circuit. But suppose a peak in the music occurs which contains primarily those low frequencies which you've boosted by 10 dB. All of a sudden, the amp is asked to deliver 200 watts! Since it can't fill that demand, it goes into severe clipping and overload, producing something close to a square wave rather than the nice complex of musical waveforms you're trying to play back. This severely clipped waveform is fed to your speakers (which may not even have a 100-watt rating for the woofers to begin with, and certainly don't for the tweeters) and after a while, you find yourself with a damaged woofer or, more likely, even a damaged tweeter (since the harmonics produced by those "clipped" waveforms will find their way to the tweeter via the crossover network). All of which is hardly the end result you hoped for when you spent your hard-earned dollars on a graphic equalizer in the first place.

Pre-recording Equalization

Normally, most of the graphic equalizers used by hi-fi enthusiasts are inserted in the line for use during playback only. That is, if you are into tape recording, you would record "flat" and do any "sweetening" during playback. But users of these graphic equalizers didn't take long to discover that it is a simple matter to connect from the tape-out (or record-out) jacks of the amplifier to the input of the equalizer and from the *output* of the equalizer to the line inputs of the tape deck. After all, "That's the way the professionals use equalizers, isn't it?" And, as if to encourage that alternate use of a home graphic equalizer, some of the latest models even have a switch position which permits you to selectively choose pre-record or post-record equalization without having to disconnect cables. Well, yes, that *is* the way equalizers are used in recording studios, but remember that a professional tape deck (open-reel) is likely to have as much as 10 dB or more of "headroom," with reference to the 0 VU reading on its record level meters. As a result, you are not likely to get into tape saturation even if you apply fairly substantial amounts of EQ boosting at specific frequency ranges.

In the case of home machines, however, (and this applies especially if you are using a cassette deck) record level meters are already calibrated so that you can't go much above the "0 dB" reading without hitting maximum tape modulation level or, as it is more commonly called, tape saturation. This is especially true at the high frequencies, where saturation occurs first. So, again, trying to compensate in the extreme for the response deficiencies of a tape deck by means of pre-equalization can turn out to be a futile exercise and one that leads to high-distortion recordings. The solution, of course, is to back off on record-level meter readings whenever you apply substantial amounts of boost using such an equalizer.

Post-recording Equalization

Even assuming that you are content to use your new audio toy as a means of achieving reasonably flat overall system response during playback (and that includes making due allowance for room acoustics), the problems inherent in trying to do this correctly are substantial. To begin with, trying to do the job by ear is hopeless. Our hearing memory is just too fleeting for us to recall, with any accuracy, what "'live' sound" sounds like at various listening levels. A host of auditory and psychoacoustic factors cloud our judgment—from the familiar Fletcher-Munson loudness contour effect, to our own pre-conditioning from having listened to our own systems for so long (and having been convinced that they are "flat" when in fact they could be anything but).

If you seriously want to try to equalize your system, I have found that it takes *at least* a ten-band equalizer. That provides independent control on an octave-by-octave basis and while it does not give you the degree of control afforded by a one-third octave equalizer, you have a fighting chance of coming up with reasonably flat response assuming your system and room do not have any really serious "bumps" or "valleys" in their overall net response. But even if you own a twenty- or twenty-four-band equalizer, where do you start? The professional sound contractor doing an EQ job comes to his assignment with a real-time analyzer, calibrated microphones, pink noise generators and more. What does the poor hi-fi enthusiast, or recording fan have besides his or her own ears.

Recently, Shure Bros. Inc. (the microphone and phono cartridge producers) came up with a clever product that fits right in with the sudden interest in home and studio equalization. Everything you need to do a professional job of correct equalization comes in this package, known as the model M-615AS Equalization Analyzer System. This package is about the lowest priced kit of tricks we have ever seen for doing a first rate job of equalization. At $429.00, it is practical for a recording studio to include it in its inventory and, as Shure suggests, even the serious audiophile may soon find that Shure dealers are willing to arrange a rental arrangement for one-time equalization jobs.

The unit consists of a calibrated microphone and the newly developed Shure M-615 Analyzer box, which contains a built-in "pink noise" generator and two rows of LED indicators, each pair of which is responsive to each of ten octaves from 32 Hz to 16 kHz. The upper LEDs tell you when energy in that octave is too great (boosted response) while if a lower LED lights, that means that there is not enough energy in that octave reaching the calibrated microphone. The idea is to set your equalizer controls until all the lights go off—and when that has been done you are assured of ± 1 dB response on an octave-by-octave basis from 32 Hz to 16 kHz. The whole job can be done in just a few minutes, as I learned when I tried it on my own system in the lab. I also learned that no amount of boosting of the 32 Hz and 16 kHz levers on my equalizer would bring those frequencies into line, so rather than boost those levers to their fullest, I felt satisfied with the knowledge that at last the response in my listening room (with a given set of speakers) was flat from around 50 Hz to 15,000 Hz. That may not be too impressive if you're used to looking at amplifier and preamp response curves, but in terms of an input-to-output-to-listening chair location, I'm pretty proud of it. Interestingly, other than the end 32 Hz and 16 kHz levers (which I grudgingly backed off to a +6 dB setting in the interest of avoiding the pitfalls I discussed earlier), all the other octave sliders on my equalizer ended up requiring no more than ±4.0 dB of adjustment, and most of them even were closer to their mid positions than that. Yet, even with this moderate amount of adjustment, switching back and forth between equalized and unequalized listening, the suddenly-smooth and balanced response I heard was well worth the effort and the time spent.

MODERN RECORDING LAB REPORT

NORMAN EISENBERG AND LEN FELDMAN

Tandberg TCD-330 Cassette Recorder

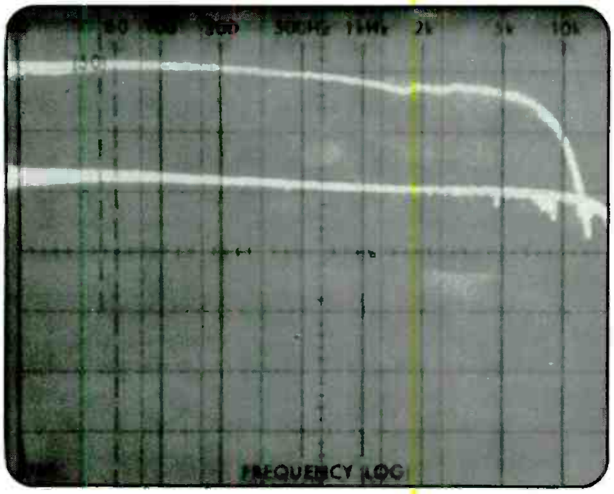
General Description: Superb performance combined with fairly unusual and highly useful design features characterize the new Tandberg model TCD-330 cassette tape recorder. A three-head (separate erase, record and play) machine, it provides for off-the-tape monitoring while recording. The transport is driven by three motors, and the tape path goes past two capstans—one before and one after the head assembly—for improved tape tension and steady movement. Tape speed is servo-regulated. Operational modes are controlled by electronic "logic circuits" and the device employs relatively sophisticated electronics in place of the mechanical parts formerly found in cassette decks. This helps things operate smoothly and reliably, and it minimizes the danger of functional breakdown. It also makes for, in this unit, fast-buttoning (you do not have to stop the transport first before going from one operating mode to another). Another nice touch is the option for "flying start" or "punch-in" recording by means of which it is possible to go directly from PLAY mode to RECORD mode without first stopping the tape. The TCD-330 has a built-in Dolby-B noise reduction system with an FM/Dolby-copy option, and a MPX-filter switch.

For optimum results when recording with any tape, the TCD-330 has a built-in azimuth alignment system complete with a 10-kHz signal generator and a handy screw adjustment. Access to these controls is obtained by lifting a small cover-plate just to the right of the cassette well.

The meters used here are peak-reading and very fast-acting. Moreover, they are connected into the circuit after the recording amplifier, which means they will read the high-frequency boost characteristics of the EQ networks, thereby showing true peaks and helping to avoid high-frequency saturation.

A single button selects both bias and EQ for tapes designated as "normal" (low-noise, high-output ferric tapes), or "special" (chromium-dioxide tapes and other newly developed formulations; among these the owner's manual lists as examples TDK-SA and Maxell UD-XL). Sizes C-60 and C-90 cassettes may be used, but C-120 is not recommended. In addition to this tape button, there are internal adjustments for bias and EQ that are not intended for regular use but which—together with several other special adjustments—are described in a separate service manual.

The TCD-330 does not provide for direct, controllable mic/line input mixing. However, mic and line inputs can be mixed by using the pair of recording-level sliders to set level for one of these input pairs while you control the other at some other point in the overall sound system. If one owns even a rudimentary mic mixer this arrangement should pose no problem since all mic mixing can be done on it, while overall master gain can be set on the TCD-330.



Tandberg TCD-330: Record/play response taken at 0 and then at −20 dB record level using Maxell UDXL-I tape. Upper trace (at 0 level) shows tape-saturation characteristic; lower trace (at −20 level) is more usual recording level.

Although the unit has no pause control as such, its absence does not hinder in any way cueing or editing ability, since there is a record pre-set button to show VU levels, and with this button "in," the machine goes into the record mode instantly when you press the record button.

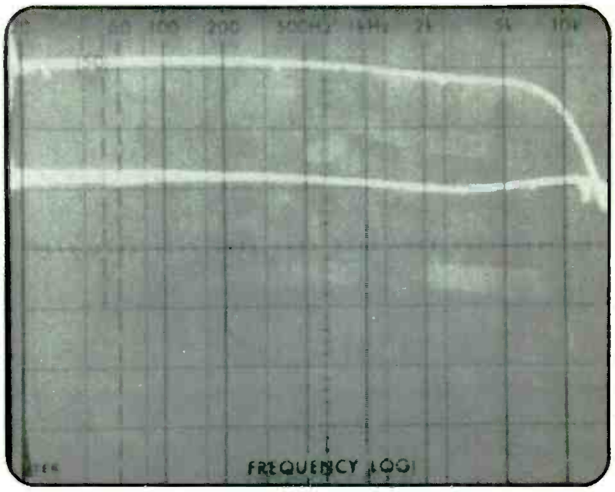
Styling is crisp and contemporary: light tinted metal framed in genuine rosewood side-panels. The TCD-330 may be installed horizontally or vertically with the aid of attachable little pedestals supplied with it. Pairs of sliders for separate-channel adjustments of output and input levels flank the two VU meters. Under the meters are the control buttons for power off/on; Dolby; tape selection; memory rewind option; tape/source monitor; record pre-set. When any of these buttons is pressed, a small lamp glows above it. To the right of the input level sliders is the index counter and its reset button.

Below these controls are the headphone output jack, the two mic input jacks, and the transport buttons for play, rewind, stop, wind (fast forward), record and eject. These are "feather-touch" controls which also have their own "on" lights. The eject button is part of the electronic action and so it operates only when the main power is turned on. However, there also is a mechanical override eject button at the rear, just in case you forget to eject a cassette before having turned power off.

The cassette well, under its hinged cover, is at the right and next to it is the azimuth alignment panel described earlier. The usual line in and out jacks, plus an optional DIN receptacle, a remote-control connector, the MPX filter switch, and the line cord are all found on the rear apron of the unit which would be the back in a horizontal position or the top if the deck is installed vertically. A removable hinged plate, supplied, may be fitted over this apron if desired.

Test Results: In terms of both mechanical and electrical performance, the Tandberg TCD-330 in sum noses out all other cassette decks we have tested so far, and has to be considered the finest yet encountered in these quarters. The logic-control transport system proved absolutely flawless, and indeed from an operational standpoint, the TCD-330 had the feel and alacrity of the finest of open-reel decks. The built-in azimuth alignment option, the highly useful metering system, and so on, all proved to be extremely effective and contributed, in MR's judgment, not only to ease of operation but also to superior cassette recordings.

Measurements of the usual performance areas either confirmed or went better than the manufacturer's specifications. It should be pointed out that the TCD-330's record level has been set at 250 nanowebers, or about 3 dB higher than the "0 dB" point of most other

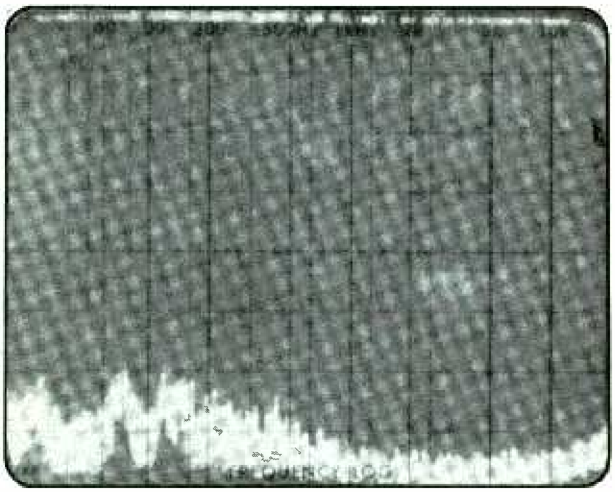


Tandberg TCD-330: Same response measurements were made using Maxell UDXL-II tape, with front-panel bias-and-EQ switch adjusted accordingly.

cassette decks. Dolby calibration therefore is at −3 dB on the TCD-330's meters, and there is still more than 6 dB of headroom above that level for achieving an excellent dynamic range when recording.

Our lab measurements speak for themselves, but we went a step beyond with this unit and did some spectrum-analyzer sweeping and recording to document frequency response on a real-time basis. Since this machine does have true monitoring facilities, we were able to record a continuous-sweep frequency response through the record/playback cycle instantly. With two kinds of tape, using alternate settings of the front-panel tape-selector switch as indicated, we got response that measured, within ±3 dB, better than 30 Hz to 20 kHz.

We also ran special curves to check out the deck's noise characteristics with the two kinds of tape and with the Dolby-B switched in and out for each tape. In each instance, the high-end noise content was really knocked down. Overall signal-to-noise ratios were just short of amazing, reaching an incredible 67 dB at best. The "worst-case" S/N was 55 dB, a figure that is considered a very good mark generally for tape decks.

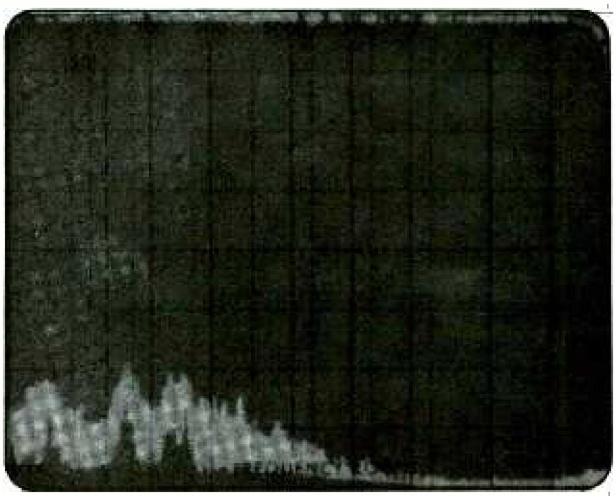


Tandberg TCD-330: Upper trace is nominal output level corresponding to record level of +4 dB (the 3% THD level for both tapes tested). Lower trace is noise spectrum. Each vertical division on 'scope graph represents 10 dB, proving that even without Dolby the Tandberg has a dynamic range of better than 60 dB.

General Info: Dimensions are $18\frac{1}{2}$ inches wide, $4\frac{1}{8}$ inches high, $9\frac{1}{8}$ inches deep. Weight is 15 lbs., 12 oz. Owner's manual is outstanding: clearly presented and amply illustrated. Price: $1,000.

Individual Comment by N.E.: I think the word "beautiful"—which I don't often use in describing equipment—does apply to the Tandberg TCD-330 very aptly. This includes everything about it, from the way it looks sitting installed on a shelf to the way it performs both in making its own recordings and in playing cassettes recorded elsewhere. I chose to install this unit vertically so that everything showed, and this is one styled piece of audio gear whose "cosmetics" have a subdued, classy look that somehow ties in with its functions and operational modes. Speaking of which, can you imagine the satisfaction experienced with a cassette deck that can be operated with the kind of ultra-light touch and deftness you might use on a top-grade open-reel deck? It's all here: fast-buttoning, flying-start recording, azimuth alignment, tape/source monitoring, pre-recording level setting, etc. As for audio response, our lab test results need only looking at to verify what this machine can do. I made several tapes, monitoring source against taped results while on the move, and no one around at any time during these dubs could tell the difference between the signal going into the TCD-330 and the taped results it was producing.

Individual Comment by L.F.: Regardless of price, the TCD-330 is the finest stereo cassette deck I have yet measured. True, there is a feature or two I would have liked to have seen incorporated, but this has nothing to do with its performance which is outstanding in all the important parameters. In my view, for instance, the built-in 10-kHz oscillator used in adjusting record-head azimuth alignment is the most effective way to accomplish this important adjustment, which should be done with each side of each cassette used. Not only does this feature give instant indication of perfect head alignment (and hence, optimum high-frequency recording response), but it



Tandberg TCD-330: Again, upper and lower traces represent +4 dB record level and accompanying noise spectrum respectively, this time with Dolby switched in. Dynamic range is about 10 dB better than before, and high-end noise content is greatly reduced.

provides instant quality control of tape for drop-outs (which can be clearly seen on the VU meters), and it even indicates when heads need cleaning (if they do, the 10-kHz recorded response drops from the norm, which again is easily detectable after you've used the machine for a couple of hours).

The logic-control transport system is a joy to use. The tape can be "rocked" between fast-forward and rewind in much the same manner as on a professional open-reel deck. The features that are not included—controllable mic/line input mixing, separate front-panel bias and EQ switches and a separate pause control—are all in one way or another made up for, as explained in the early portion of this report. In the context of Tandberg's design here, they are not really "missing items." Anyway, the splendid performance of the unit and its many positive attributes, is what matters here—and on this count, the Tandberg TCD-330 merits a "rave review."

TANDBERG TCD-330 STEREO CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Frequency response (UDXL-I tape)	27 Hz to 20 kHz, ±3 dB
Frequency response (UDXL-II tape)	28 Hz to 21 kHz, ±3 dB
Harmonic distortion at −10 VU (tape I/tape II)	1.7% / 1.0%
at −3 VU (I/II)	1.5% / 0.65%
at 0 VU (I/II)	2.0% / 1.9%
at +3 VU (I/II)	2.2% / 1.9%
Recording level for max 3% THD	+4, either tape
Signal-to-noise, unweighted	
w/o Dolby (tape I/tape II)	55 dB / 57 dB
with Dolby (tape I/tape II)	65 dB / 67 dB
Wow and flutter (WRMS)	0.07%
Fast wind time (C-90)	67 seconds
Mic input sensitivity	0.15 mV
Line input sensitivity	80 mV
Line output level	1.5 V
Phone output level	5 mW

CIRCLE 12 ON READER SERVICE CARD

Burwen Model DNF 1201A Dynamic Noise Filter

General Description: The model 1201A is an updated and improved verison of Burwen's former model 1201 Dynamic Noise Filter, an electronic device intended for patching into a sound system to remove audible noise from program sources: tape (open reel and cassette), turntable or FM tuner. It does so by "sampling" the high-frequency content from the source, and adjusts its bandwidth relative to both frequency and signal level. Dynamic filtering by the 1201A is accomplished by the device's generating of a DC control voltage that constantly regulates the filter's cutoff frequency.

The front panel has eight controls (seven buttons and a slider). At the left is the power off/on button. Centered on the panel are four signal-processing buttons marked "out," "max," "med" and "min." These select the degree of signal-processing offered by the device in accordance with the user's judgment of the quality of the program material in terms of its noise content. For instance, the "max" button introduces maximum signal-processing for relatively high levels of noise intermixed with the signal, such as old or worn records, very hissy tapes, noisy broadcasts. The "min" button provides minimum processing for high quality source material, and the "med" button introduces an intermediate degree of action. The "out" button defeats the signal processing entirely and so it may be used for "A-B" comparisons, as well as to keep the DNF 1201A in the sound system on a "stand-by" basis.

To the right of these buttons is the sensitivity slider which may be used to increase or decrease the filtering action with reference to the level selected by the buttons. This slider is flanked by two LEDs, a red one marked "suppression"; the other, green and labeled "wideband." The former area of the slider increases noise reduction; the green LED area decreases noise reduction. The device is said to be accurately "fine tuned" when the sensitivity control is in a position that permits both red and green LEDs to flash on and off alternately.

The final two buttons, labeled "monitor" and "pre/post," are designed to allow using the 1201A for tape recording: the monitor button permits monitoring from the tape; the pre/post button permits the user to connect (without rewiring) the 1201A before, or after, a tape recorder used in the system.

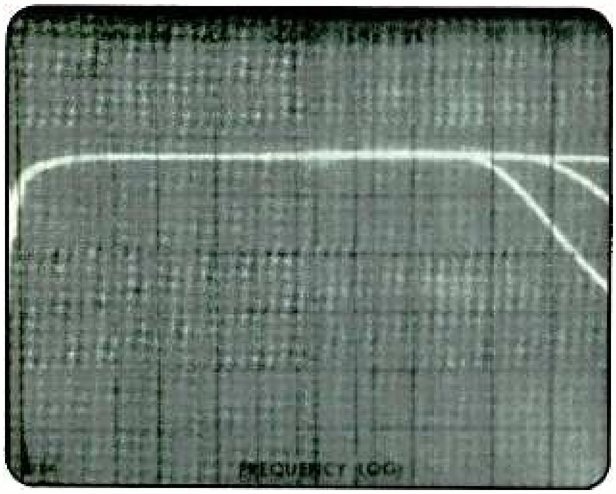
At the rear are four stereo pairs of phono signal jacks to permit patching the 1201A into a system that includes a tape recorder via the tape-monitor facility of preamp, amp or receiver so that any source fed into it can be handled by the 1201A with or without its also being taped. The output of a tape deck on playback also can be processed if desired, and tape monitoring is always available. Also at the rear are recessed input level screwdriver adjustments. These are factory-adjusted for unity gain, but may be changed by the owner (instructions indicating when this may be required are furnished with the unit). The device's AC line cord also is at the rear.

With its low silhouette and walnut-veneered case, the DNF 1201A noise filter presents a neat, contemporary appearance. It may be installed wherever convenient, although it does not have standard rack-mount dimensions.

Test Results: Vis-a-vis the original 1201, Burwen has improved time constants and also added control flexibility not found in their earlier unit. The present version offers three degrees of "attack time" as well as variations of cutoff frequency. For example, with the "min" button pushed in, cutoff frequency is set for 9 kHz, and attack time is fast. For the "medium" setting, the 9 kHz cutoff point is retained, but the attack time is slowed down somewhat to an average value. The minimum setting reduces the cutoff frequency to 5 kHz and provides for an even slower attack time.

"Static" testing of this device is somewhat iffy since it does not take into account the unit's attack-time variability which we regard as one of its advantages. Nevertheless, MR did devise a few tests whose results could be shown graphically.

Since the unit's sensing and control voltages are based upon the sum of signal-information from both stereo channels (the masking effect of noise works whether the loud masking signal is coming from one or from both speakers in a stereo system), MR fed a constant signal into one channel while varying the intensity of a 10-kHz signal applied to the other channel. We swept the channel to be photographed (on the 'scope face of our spectrum analyzer, from 20 Hz to 20 kHz), and varied the intensity of the 10-kHz channel in steps



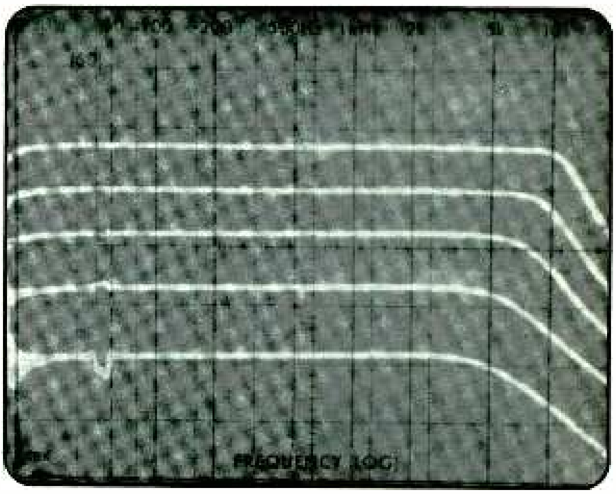
Burwen DNF 1201A: Fig. 1. Varying cutoff filter action due to changing high-frequency (noise) input on opposite channel. Note: "max" button pressed for this test.

of 10 dB. The results of three successive frequency-sweeps are shown in Fig. 1, which displays how the filtering action occurs, as per the level of the noise signal in the "opposite" channel, from full bandwidth through a cutoff point around 10 kHz, and then down to about 5 kHz.

Next we removed the fixed-frequency signal from the non-observed channel and simply varied the intensity of the sweeping frequencies applied to the observed channel. The data in Fig. 2 show that as signal level of the latter is reduced, the cutoff points of the filtering action are reduced from around 10 kHz to 5 kHz or even a bit lower.

Figs. 3 and 4 show the effect of the front-panel sensitivity control set to various positions. In Fig. 3 the "max" button was pressed; in Fig. 4, the "med" button was used. Note that with the sensitivity control set for most suppressions, cutoff occurs just below 500 Hz regardless of the settings of the signal-processing buttons, or of the level of the applied input signals. This of course is an extreme situation which we never had to deal with in actual listening tests. The more significant differences in filter action become obvious when we compare mid and wideband settings of the sensitivity control; note that for medium settings of the switch buttons, cutoff points vary, depending on the input signal level, and also on the settings of the sensitivity slider.

In addition to these tests, MR checked the unit against specs, all of which were confirmed or bettered except for a measured 5 volts, instead of the claimed 6 volts, for both maximum input level and output level. More important than this minor discrepancy was the



Burwen DNF 1201A: Fig. 2. Changing input level in observed channel results in varying frequency cutoff point. As signal level is reduced, filter cutoff frequencies also are reduced. Note: "max" button pressed for this test.

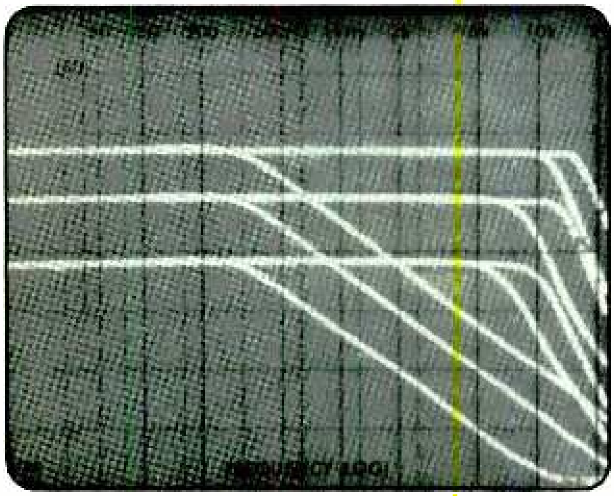
device's very low distortion (both THD and IM were less than specified), and its very good signal-to-noise ratio (82 dB as compared to the specified 80 dB).

Listening tests also were conducted; these are discussed under the "individual comments" below.

General Info: Dimensions (in case supplied) are 17 11/16 inches wide; 2 5/8 inches high (add about one-half inch more for rubber feet); 8 1/4 inches deep. Weight is 8 lbs. Owner's instruction manual is somewhat sketchy—could be more detailed, explicit and better illustrated. Price: $379.

Individual Comment by L.F.: The earlier DNF 1201, which I checked out some time ago, did remove background noise and hiss but in doing so, two effects also came into play. If the program material was of wide frequency response but also loaded with hiss, the single control on the older model had to be pushed to such extremes that there was a definite degradation of frequency response. And even at lesser settings I was aware of a "breathing" or "pumping" effect.

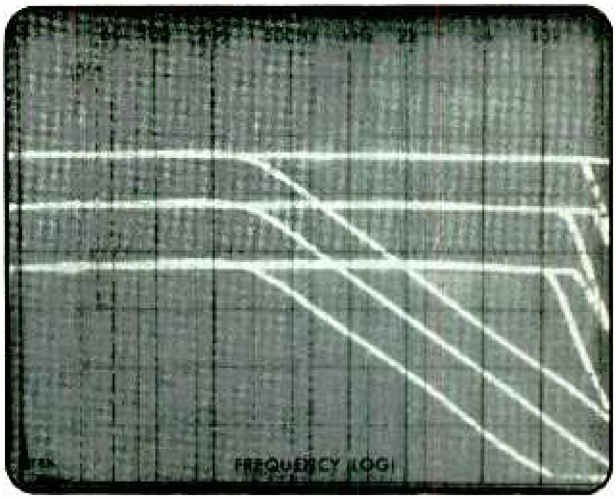
Happily, I find that the new 1201A, which bears little resemblance physically to the older unit, also has vastly improved performance by at least a whole order of magnitude.



Burwen DNF 1201A: Fig. 3. Traces show effect of sensitivity slider when set to extreme left, mid-position, and extreme right for different signal levels. Note: "max" button pressed for this test.

It is of course still possible to misadjust the device so that it slices into musical material and also so that pumping and breathing can be heard. But in my view, such misadjustment with the new unit would be the user's fault, not that of the DNF. Our test-result 'scope photos show some of the variations of effect that are possible with the 1201A. Actually, this very great variability might discourage the first-time user who fails to experiment sufficiently with the unit. In our listening tests we used everything from old, poor-quality 78-rpm discs and ancient hissy tapes to late-vintage discs that had hardly a trace of surface noise. For each source we were able to find a combination of pushbutton and slider settings that removed significant amounts of noise without audibly affecting musical content.

It would seem that the principle of dynamic filtering, as exemplified by the Burwen DNF 1201A, is a handy solution for audiophiles who want to listen to source material that has so much noise content that it



Burwen DNF 1201A: Fig. 4. Same test as shown in Fig. 3, but with "med" button pressed.

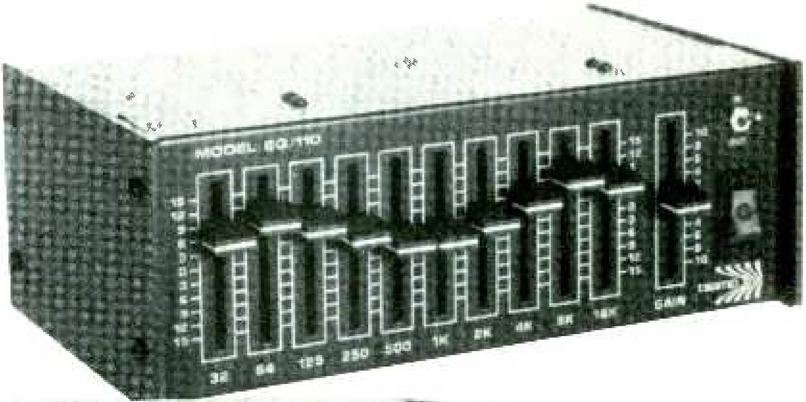
detracts from full enjoyment of the program. After all, you cannot Dolbyize or "dbx" a source that already is noisy, and the Burwen device does offer "one-sided" noise reduction that is far better than that obtained with passive filtering. Note too that the Burwen device does not alter the source's dynamic range, only its noise content.

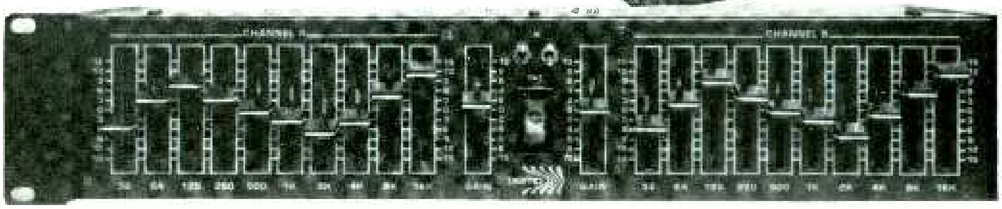
Individual Comment by N.E.: The DNF-1201A is audibly superior in its noise-filtering action than conventional filters typically found on amplifier control panels in that it can reduce a tremendous amount of "hash content" to allow the musical signal to emerge more clearly. However, some high-frequency signal content also is sacrificed in the process; just how much likely will depend on the original quality of a program source and the amount of noise-content that has been added to it. The three buttons ("max," "med" and "min"), and the sensitivity slider do provide options for adjusting the filtering action but finding just the right combination of settings may take a fair amount of fiddling with these controls. It is very difficult to say—perhaps more so than for a lot of audio devices—whether this unit will appeal to one or not. I suggest that anyone who is in the market for a dynamic noise filter check the Burwen device out for himself.

BURWEN DNF 1201A DYNAMIC NOISE FILTER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Frequency response	
Minimum bandwidth	down by −3 dB at 500 Hz; −10 dB at 1 kHz; −20 dB at 2.5 kHz.
Maximum bandwidth	±0.5 dB, 15 Hz to 25 kHz
Maximum input level	5.0 V
Maximum output level	5.0 V
Gain at 1 kHz	from 0 dB, adjustable to +10 dB
THD, 20 Hz to 10 kHz (at max. sens)	0.15%
IM distortion	0.035%
Internal noise ratio (below 1 V)	82 dB

biamp Systems EQ/210 Graphic Equalizer



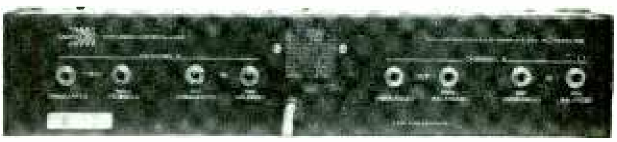


General Description: The model EQ/210 from biamp Systems, Inc. is a dual-channel graphic equalizer that may be used as a stereo unit or as two independent mono devices. Each channel has ten bands of equalization with center frequencies at 32 Hz, 64 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, 8 kHz and 16 kHz. Each band is adjustable via its own front-panel slider over a range of ±15 dB. In addition to the ten sliders each channel has its own gain slider, and an in-out switch to defeat or introduce the equalization. These all are arranged across the front panel in logical, symmetric fashion with a power switch centered between the sliders. Next to the gain slider on each channel is an LED overload indicator for that channel.

The equalizer band sliders are calibrated from 0 to 15 (each way) in markings of 3 dB. The gain sliders are marked (each way) in steps of 2 dB from 0 to 10. The front panel is flanked at either end by metal flanges for standard rack-mounting.

Signal connectors at the rear all are standard 1/4-inch phone jacks. There are eight in all—four per channel—to accommodate unbalanced and balanced lines for input and output. The balanced inputs and outputs are transformerless in the interest of low-noise operation, and permit the EQ/210 to be interfaced with standard recording studio and broadcast equipment, as well as with mixing consoles. The unbalanced (single-ended) inputs and outputs enable the device to be used with standard PA equipment, musical instruments, power amplifiers and normal home hi-fi components. The AC line cord, also at the rear, is a heavy duty cord terminated in a three-prong plug, the kind with a grounding prong. The EQ/210 draws 20 watts during operation.

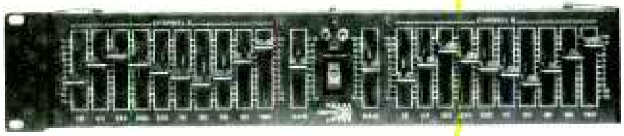
The EQ/210 is described as a fully professional graphic equalizer, offered for use in portable or fixed sound reinforcement, recording studio, with musical instruments or home hi-fi systems. Its applications include feedback suppression and room equalization, various options for modifying sound for greater clarity and frequency balancing, and—if desired—to create special effects.



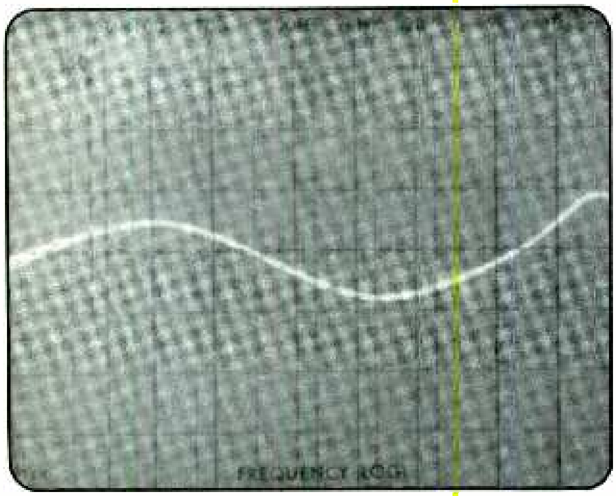
EQ/210: Rear Panel View

Test Results: The biamp EQ/210 is without doubt one of the superior ten-octave graphic equalizers presently available. It produces its claimed performance, with super-low distortion (THD measured was a mere 0.004%!), excellent frequency response, high signal-to-noise ratio and practically no interaction between adjacent octave controls. The peak LEDs lit up when output levels reached 6 volts, thus giving ample warning that line levels were incorrectly set, well before actual clipping at 8 volts rms took place. At all lower output levels, THD remained remarkably low regardless of whether individual sliders were boosted, attenuated, or left on "0 dB" position.

The settings of the octave sliders, shown on the front panel in Fig. 1, were used to obtain the frequency response characteristic shown in our 'scope photo, Fig. 2. Good correlation may be observed between the graphic configuration of the front-panel controls and the actual response obtained. Additional tests by MR confirmed the claimed range of each slider, as well as the accurate calibrations for boost and cut settings at the various specified center frequencies. MR also was



biamp EQ/210: Fig. 1. In one test, controls were set in random pattern on each channel.



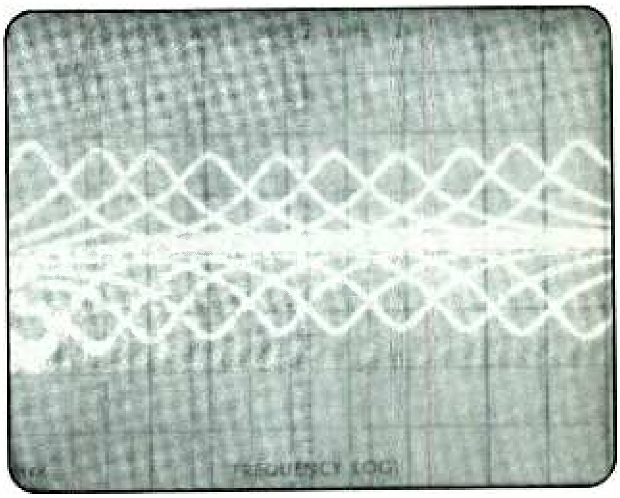
biamp EQ/210: Fig. 2. Photo of 'scope pattern shows actual frequency characteristic produced by settings of Fig. 1.

favorably impressed with the excellent isolation between the two channels of the unit, confirming another of the manufacturer's claims—that the device could be used, if desired, as two separate mono equalizers for separate single-channel systems and sound sources. The EQ/210, in short, is in every way a professional-grade equalizer.

General Info: Dimensions are 19 inches wide (standard rack-mount); 3½ inches high; 5¼ inches deep. Weight is 5 lbs. Price: $229. Owner's instructions: adequate but somewhat skimpy.

Individual Comment by L.F.: There is little to criticize when evaluating a graphic equalizer. It either does its frequency-tailoring job well, or it doesn't. The biamp EQ/210 assuredly does. What surprised me somewhat, however, was the rather amateurish owner's manual supplied with the unit. It contains no schematic diagram, and little discussion of circuitry other than the fact that the unit contains what biamp calls "gyrator" inductor circuitry (no physical coils are needed since an IC op-amp employing negative feedback presents the equivalent of the required inductance in each active filter). Instructions and suggestions for using the EQ/210 are limited to a few paragraphs about room equalization, special effects, and its use in recording to "bring up harmonics in instruments and subdue others." Hardly a definitive treatise on the use of an equalizer. An equalizer that performs as well as this one deserves a more comprehensive manual.

Individual Comment by N.E.: Without a doubt the EQ/210 is one of the best graphic equalizers now available. Equally without a doubt, its producers could use the services of a professional tech writer and illustrator to better present information and operating directives for the device. One who has had experience before with equalizers probably could make do with the skimpy instructions furnished, but a first-time user might find them less than completely helpful. Be that as it may, the EQ/210 is a first-rate, well-designed, and top-performing device in its class.



biamp EQ/210: Fig. 3. Control range, octave by octave, of the EQ/210.

BIAMP EQ/210 GRAPHIC EQUALIZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Number of octaves/channel	10
Center frequencies	32, 64, 125, 250, 500, 1 k, 2 k, 4 k, 8 k and 16 kHz
Gain, all controls set at 0 dB	−3 dB unbalanced; 0 dB balanced
Inputs and outputs, terminations	transformerless balanced, and conventional unbalanced
Output load impedance	600 ohms or higher
Input impedance	50 k ohms, balanced or unbalanced
Maximum boost or cut per octave	15 dB
Frequency response, controls set flat	+0, −1 dB, 6 Hz to 45 kHz
THD, unity gain, controls flat, 1.0 V in and out	0.004%
Maximum output level	+24 dBm (8 volts, unbalanced)
Signal-to-noise ratio	84 dB below 1 V output.

CIRCLE 14 ON READER SERVICE CARD

By Jim Ford and Brian Roth

Microphones for Sound Reinforcement

There are stacks of technical papers written about recording and sound equipment, and in the majority of instances the specifications given include all the information necessary for a sound engineer to make evaluations and decisions concerning the use of the equipment. However, in several areas it is very difficult for the young soundman to get a direct relationship between the written specifications and the resulting sound. One area that is particularly confusing is microphones. So, here is a short summary of the most common microphones used currently in the concert sound reinforcement field. This information should give the aspirant a good starting point and help him save time and money.

The Shure SM-57 and SM-58 microphones are today, probably the most widely used "live" performing vocal mics. There have been many large rock concerts at which 19 out of 20 microphones were SM-57s and 58s. (Usually the 20th microphone was a condenser mic suspended over the drums.) The success of this particular series of microphone is due to their performance on the "live" rock and roll stage—which can be a very difficult situation. The general characteristics are:

A. *Cardioid* or *unidirectional* pattern: These mics are more directional than usual and consequently they provide an increased amount of gain before feedback when compared to the average cardioid microphone.

B. *Proximity effect*: This is a characteristic of most cardioid microphones. As the sound source is moved closer to the mic, the bass response is increased in comparison with the other frequencies. What does that mean? When you sing closer to the mic, the sound is more bass heavy, and this is the effect desired by most vocalists because it makes them sound "bigger." The majority of all rock vocalists take advantage of this and use their microphones closer than two inches. One side effect due to this procedure and the microphone design is that these mics are excellent at rejecting bass from a distance. This helps minimize the leakage of sound from the stage instruments and vocal monitors into the vocal mics.

C. *High frequency boost*: A slightly rising high frequency response above 2,000 Hz adds brightness which helps to give the vocals a nice crisp sound.

D. *Dynamic*: Reasonably rugged for traveling sound systems.

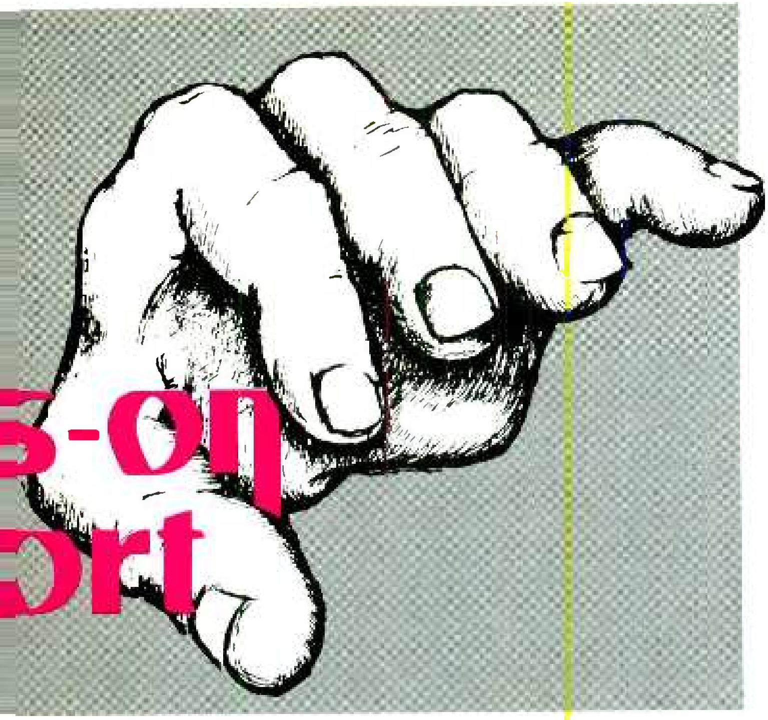
The conclusion is that the combination of the tight cardioid pattern to help with feedback and leakage, the bass boost when working the mic close, and the high frequency boost gives the sound that most pop vocalists want.

Now, the two Sennheiser mics and the Beyer ribbon show the same general characteristics and give the same type sound. The MD-441 differs in that it is a little brighter sounding and it has a five position bass cut filter.

The Sennheiser MD-441 is an excellent microphone

Most Common Sound Reinforcement Microphones

Manufacturer	Model	Type	Approx. Cost
1. Shure	SM-57, 58, 56	Dynamic, cardioid	90.00 to 118.00
2. Sennheiser	MD-421	Dynamic, cardioid	193.00
3. Sennheiser	MD-441	Dynamic, super-cardioid	275.00
4. Beyer	M-500	Ribbon, hypercardioid	147.50
5. AKG	C-451E	Condenser, cardioid	229.00
6. Sony	ECM-22P, ECM-33P	Condenser, cardioid	*No longer available*



and is also widely used in recording studios. It is a super-cardioid which means it is more directional than a regular cardioid microphone. A test in a "live" sound environment will show that more gain before feedback is possible with this mic than with the SM-57 and SM-58. It has a five position bass cut filter and a single position high frequency boost switch. One last point is that the MD-441 has an excellent extended high frequency response that makes it useful for many other applications besides P.A. vocals.

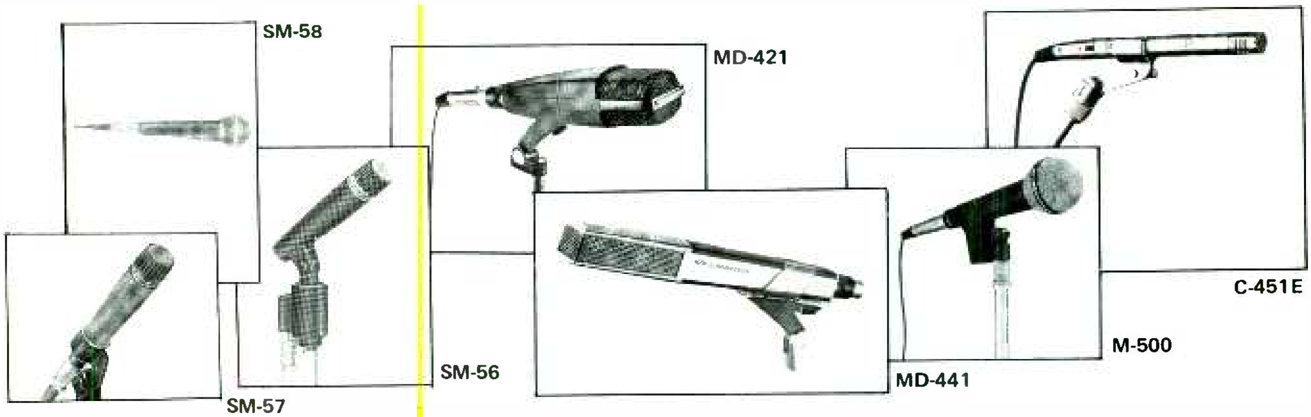
The Beyer M-500 ribbon is the most recently accepted mic in the sound reinforcement field and because of its exceptionally smooth sound, it is quickly becoming a top choice of soundmen and performers. This mic exhibits the same proximity effect and presence boost as the Shure mics but is slightly more directional and is smoother sounding.

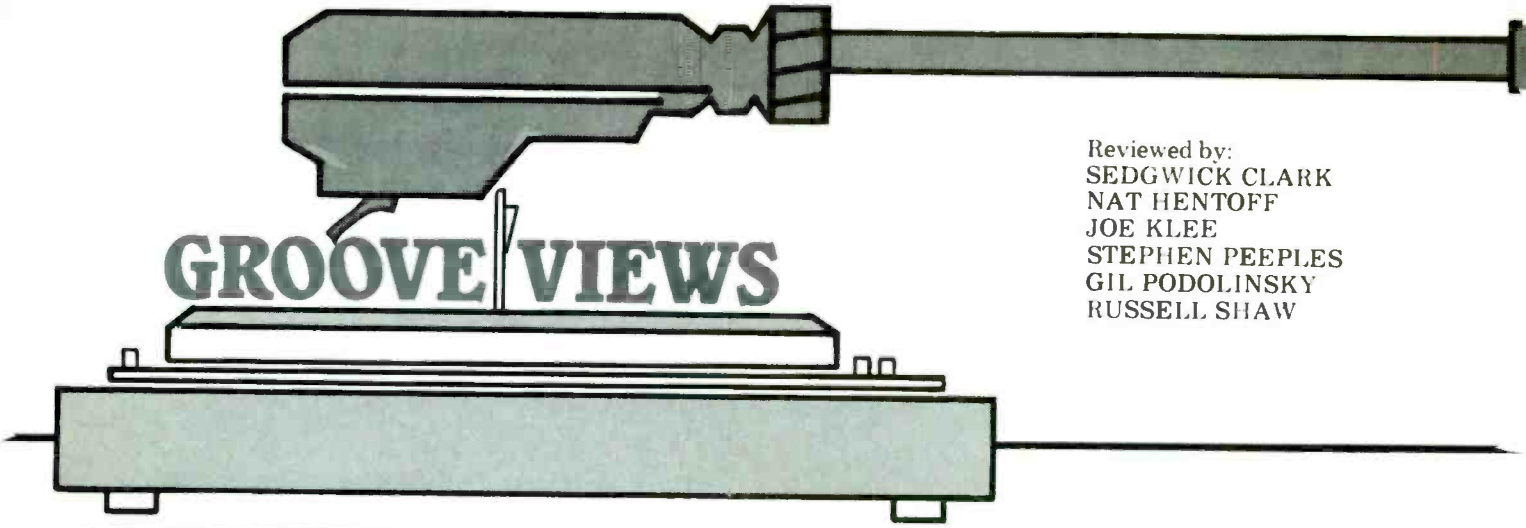
Soundmen are famous for carrying a box of "tricks" around with them and usually this includes several condenser microphones. Most often these mics will be used for drums, acoustic guitar, piano, strings, etc., because of their low distortion, extended high frequency response and ability to accurately respond to transients. What did that mean? They sound good when the soundman wants a lot of clean high frequencies! Probably the most used mic in this category was the Sony ECM-22P electret condenser which was superb for $99.50. Unfortunately Sony discontinued the mic about two years ago. The Sony ECM-22P was replaced with the Sony ECM-33P [which in turn was replaced by the ECM-33F] electret condenser, but this mic was also discontinued approximately six months ago. Three good options to the Sony mics are the AKG C-501 (C-505) electret condenser; Sennheiser MKE-402 electret condenser; and TEAC ME-80 (ME-120) electret condenser. All of these microphones are electret condensers and operate with batteries. They range in price from $80.00 to $157.00.

In the higher priced category of condenser microphones, the AKG C-451 has been a regular choice ($229.00). This mic is well accepted in the recording industry and as a result has been put to good use in sound reinforcement. The general characteristics are—low distortion, linear and extended frequency response and uniform cardioid pattern. A microphone of this type will most likely give good results on acoustic instruments. One small problem, however, is that this mic requires a D.C. power supply for operation. Some condenser mics will operate with batteries or a power supply, and this feature should be examined prior to purchase.

Now, as long as soundmen have opinions (did you ever know a soundman that didn't have an opinion?), there will be favorite microphones and techniques. Variety and experimentation is at the heart of the music industry, and it is recommended that the soundman feel free to use his imagination. Critical listening should be the final and best test.

There are numerous other microphones that will give excellent results. Try these: Electro-Voice RE-10, 11, 15, 16, DS-35; AKG D-202E, D-224E; Beyer M-160; Neumann KM-84, U-87, U-47; Sony C-37. Also, direct boxes for guitar and bass, and electrostatic pickups for piano are in wide use. Next time a rock concert is in town, look closely!





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

POPULAR

AUTOMATIC MAN: *Automatic Man.* [Lou Casabianca, Automatic Man, producers; Keith Harwood, Chris Kimsey, engineers; recorded at Island Studios, London, England.] Island ILPS 9397.

Performance: **Pretentious**
Recording: **Sloppy**

From the press copy on *Automatic Man:* "The engineering by Keith Harwood (Rolling Stones and Led Zep-

DONI HARVEY

pelin) and Chris Kimsey (Peter Frampton) is literally the state of the art. This is most dramatically demonstrated that albums engineered by those two engineers were the top number 1, 2 and 4 albums on the Billboard charts in recent weeks." Really. Atrocious English aside, the *equipment* used to record this album may be state of the art, but to presume that the state of the *engineer's* art can be dictated by chart success is ridiculous. I don't know any of the five million-plus people who bought the Frampton album solely for its mix.

PAT THRALL

MICHAEL SHRIEVE

The truth is that the engineering on *Automatic Man* is sloppy. The album opens with a noisy wall of sound that sounds like Jimi Hendrix as produced in mono by Phil Spector—flat, muddy and too much going on at once.

"My Pearl" almost makes it, but the vocals are too deeply buried. The emphasis throughout most of the album is on musicianship, which is certainly there, but the indistinct vocals and heavy-handed material leave the listener unfulfilled. With more economy, especially with master keyboardist

BAYETÉ

Bayete's synthesizers, the advanced musicianship could have been more fully realized. S.P.

ATLANTA RHYTHM SECTION: *A Rock and Roll Alternative.* [Buddy Buie, producer; Rodney Mills, engineer; recorded at Studio One, Doraville, Ga.] Polydor PD 1-6080.

Performance: **Flawless**
Recording: **Close to perfect**

The Atlanta Rhythm Section is an often-underrated sextet of former session musicians from Georgia whose individual histories intertwine with those of much of the region's music. Yet when the phrase "Southern Rock" is mentioned, ensembles with far less a grasp of harmonic basics are often thought of first. Admittedly, there are bands with more surface spunk and flash, but the Rhythm Section's forte seems to be in the opposite directions—lyrical and musical subtlety, an absence of braggadocio and a presence of intricate melodic ideas.

This release, sixth in a line of superb ARS issues, provides no departure from the band's high standards. Indeed, lingering imperfections have been expunged. There is a total lack here of *any* excess, be it instrumental or vocal. Each track is eminently hummable and memorable, while being a credible representation as well.

You'll never find these musicians getting in each other's way. The musical phrases are uncluttered, crisp and free of garage-band flash and rhinestone macho. Guitarist Barry Bailey, whose betters in the rock world could be counted on one hand and his peers on two, is the ultimate versatile picker. Raunchy and mean on a faithful resurrection of Robert Johnson's "Outside Woman Blues," but soft and sensuous on the alluring "So In To You." Bassist Paul Goddard, who also garners a good deal of acclaim, constantly affirms his unique trait of being creative within a precise framework.

Recorded at Studio One, a futuristic sound factory located in an Atlanta suburb, *A Rock and Roll Alternative* benefits greatly from the capacity of the plant to sketch passing moments in the minutest detail. Studio One recordings are known throughout the industry for their clarity and distinction—although low profile the work of drummer Robert Nix is audi-

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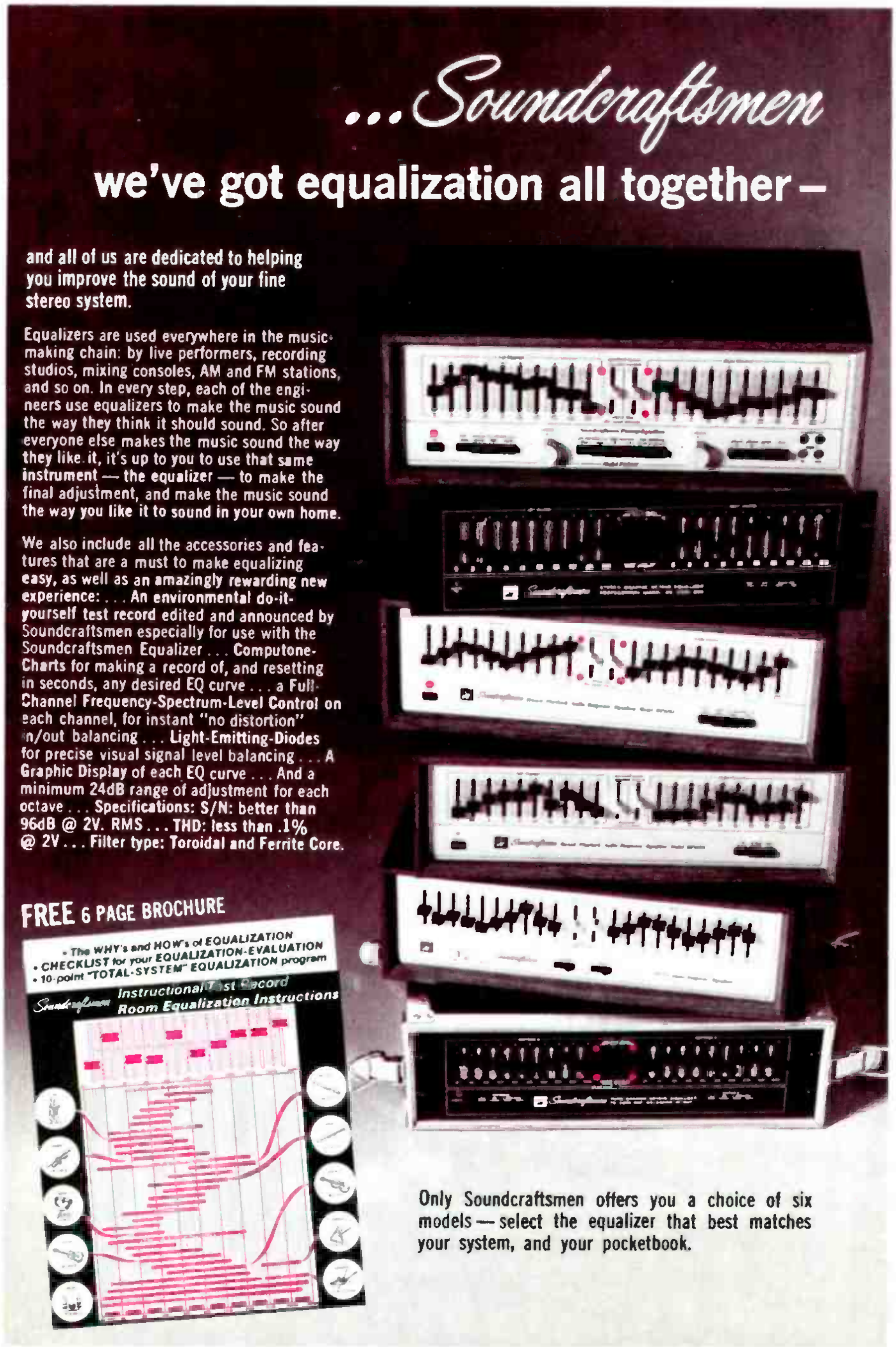

CIRCLE 89 ON READER SERVICE CARD

ble and crystal clear. Recorded at a less prestigious venue, his whispering flourishes would be muted, but here they are interwoven flawlessly, without the crutch of an overmix. The highs too, are uncommonly vivid, encapsulating a package which approaches absolute perfection. R.S.

JAZZ

DON CHERRY: ***Don Cherry.*** [Corrado Bacchelli, Beppe Muccioli, producers; Kurt Munkacsi, Michael Mantler, engineers; recorded at Basement Recording Studios, N.Y., Grog Kill, Woodstock, N.Y.] Horizon 18.

Performance: **In and out**
Recording: **Gimmicky but good**

Don Cherry is an enigma. One minute he's as melodic as anyone could wish and the next he can take his listeners out to freak city. He is a modern renaissance man coming out of so many bags that he doesn't stand still long enough to hang a category on him. This is to his credit because every listener can hear something in his music, but it can work to his detriment as well, since no two people will necessarily hear the same thing. Don Cherry's music is a well where all may come and drink but some will go away more satisfied than others.

The engineering is a bit on the rocky side using devices foreign to the average jazz recording such as heavy echo, digital time delay and tapes of altered speed. Still, I can't call it dishonest. What the producers and engineers set out to do, they accomplished. They made a recording of a jazz musician with a decidedly rock sound to it.

Interestingly enough, the two bass tracks by Charlie Haden were not overdubbed. Charlie played the part once. It was picked up acoustically by a studio microphone and electronically by a contact mic which was put through a wah-wah device giving the effect of a doubling of the bass line. It's a gimmick, but it's a clever gimmick, tastefully used. J.K.

MEL LEWIS: ***Mel Lewis and Friends.*** [John Snyder, producer; Thad Jones, musical supervisor; Tony May, engineer; recorded at Generation Sound, N.Y.] Horizon 17.

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Sonny Rollins and Other Long-Distance Improvisers

By Nat Hentoff

By 1956, when Sonny Rollins appeared alongside Clifford Brown in Max Roach's band, Sonny was already an extraordinary *presence* on the jazz scene. Although shaped by Coleman Hawkins, Sonny became an original early. There was the extraordinary cohesion of his brilliant, daring solos ("A master of 'thematic improvisation' " wrote Gunther Schuller about him in the early '50's). And the Rollins sound—instantly identifiable and yet containing so many glints and explosions of so many different colors. In addition, there was his sense of time. Sonny did so much more than swing. Implying such swirling pyramids of counter-rhythms that he didn't really need a rhythm section. And then the piercing intelligence of the man, often sardonic but also careeningly playful. Before he was thirty, Sonny was a jazz sovereign.

In 1959, he disappeared for two years in order to really explore himself and his music, for Sonny is chronically self-questioning. Some years later, he exiled himself a second time. But now, Sonny says, he's back for good. ("I don't have that much time left.") And in his newest Milestone release, *The Way I Feel*, Sonny is again magisterial. Bestriding a largely electrified rhythm section and, except for two selections, an assembly of nine horns, Sonny is in thoroughly resilient control of himself and his horn. (Or, as he once put it, "Sometimes I'm able to step outside myself and hear what I'm playing. The ideas just flow. The horn and I become one.")

On ballads and up-tempo drives, the Rollins *presence* is enveloping, pulling you into the vortex of his emotive ideas. His sense of time is even more resourceful than before. His sound is as powerfully authoritative as that of his onetime mentor, Coleman Hawkins; and his improvising continues to consist of leaps of imagination which, in retrospect, appear to have been part of an unerring prior design. And always, listening to him, I hear the jazz equivalent of what novelist Thomas Wolfe used to call "the goat cry"—an unabashed affirmation of life, sensual life.

The engineering focuses on Rollins, and he comes out of the speakers in the full spectrum of his sonic colors. (I do wish, however, that the brass had been brought up more.)

A superbly engineered session—its crisp, finely balanced clarity enhancing the lyrical subtlety of the players—is *Teddy Wilson and His All Stars* (Chiaroscuro). Like Rollins, Wilson, of course, is a long-lasting jazz *presence*—less overwhelming than Rollins but swiftly, clarifyingly compelling. Also present is the wittiest of all jazz trombonists, Vic Dickenson, who, like Rollins, cannot be narrowed into a category. Trumpeter Harry Edison plays with more fresh zest and invention here than in some time, and the increasingly impressive Bob Wilber is on soprano and clarinet. Major Holley and drummer Oliver Jackson complete the buoyant rhythm section.

This Wilson set is one of the most relaxed and yet authoritative sessions of classic jazz improvising to be released in some months. Like Rollins—but in different, more floating ways—it'll make you feel good. (The way the rhythm section has been recorded—light, full and clear—heightens that good feeling.)

SONNY ROLLINS: *The Way I Feel.* [Orrin Keepnews, producer; Eddie Bill Harris, engineer; recorded and mixed at Fantasy Studios, Berkeley, Ca. and Total Experience Recording Studios, Los Angeles, Ca.] Milestone M-9074.

TEDD WILSON: *Teddy Wilson and His All Stars.* [Hank O'Neal, producer; Fred Miller, engineer; recorded at Downtown Sound, New York City.] Chiaroscuro CR 150.


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Performance: **Straight ahead jazz**
Recording: **Honest but a bit fuzzy around the edges**

Basically jazz musicians get together to blow. This session just happened to be at Generation Sound and Tony May opened the mics so we could eavesdrop. There are heavyweights like poll winning trumpet and flugelhorn player Freddie Hubbard and lesser knowns like former Woody Herman saxophonist, Greg Herbert. The feel is free and loose and the recording communicates the relaxed and easy feeling especially when Hubbard turns his lyrical flugelhorn loose on Thad Jones' "A Child Is Born." This is Freddie's second go at the tune. The first was on a Richard Davis date for BSAF some years ago. Freddie mellows with age. So does Thad's tune.

Since it's Mel's date, he gets to do his tasty bit on drums. Nobody's brush work is more subtle and sensitive than Mel's and Tony May's engineering captures it crisply. It fairly bristles from the speakers. I wish the same presence had been brought to bear on Hank Jones' piano work. It's close but no bull's eye.

My main cause for complaint is something that is in danger of happening any time a record company feels generous enough to try to give the listener a bargain and put more than twenty minutes on one side of an LP. The closer you get to the label, the fuzzier the sound. It's too bad that the clarity deteriorates on the version of Charlie Parker's ode to his pusher, "Moose The Mooche," which closes side one because everything was going so well and Hubbard was playing like a man on fire. J.K.

GEORGE BENSON: *In Flight.* [Tommy LiPuma, producer; Al Schmitt, Don Henderson, engineers; recorded at Capitol Records, Hollywood, Ca.] Warner Bros. BSR 2983.

Performance: **Husbanded talent**
Recording: **Aborted in flight**

At approximately the same time in their respective careers, George Benson and Wes Montgomery went commercial. With the exception of Benson's singing, the changes in stylistic content went the same route. Although Benson's style is clearly influenced by Montgomery and Charlie

Christian, not until his move to Warner Bros. did he lack fire and zeal in his play. Apparently his guitar is now taking a back seat to his singing and the success of "This Masquerade" has sealed his fate. George sings on four of the six tunes featured, and they all have schlocky string arrangements. The arrangements are all the same—string intros giving way to a nondescript bottom, then guitar and funky Rhodes. The mix is bland and unchanging, not even worth mentioning. The intent is to create a pop album in the formula of the platinum selling "Breezin'." Don't expect the next album to be any different, either. I remember before his death, that Montgomery came to regret his commercial success, for he no longer could play what he wanted. I wonder if Benson will be content when it's all over to own a Mercedes and play Vegas. G.P.

GEORGE BENSON: Selling out for success

PAT MARTINO: *Joyous Lake.* [Paul Rothchild, producer; Steve Klein, engineer; recorded at Criteria Studios, Miami, Fla.; remixed at Sunset Sound Recorders, Hollywood, Ca.] Warner Bros. BS 2977

Performance: **Uninspired**
Recording: **Functional**

Past efforts have revealed Martino to be a guitarist who could move from funk to space with minimum effort. Finally, in 1976 with the release of *Starbright* on Warner Bros., his product was finally the beneficiary of a fully equipped promotional apparatus.

That last album was widely hailed for its unique twists and turns; generally wistful, surreal ambience and its flawless, clockwork production. That artistic success made one look forward with eager anticipation towards this new record, yet audit of *Joyous Lake* yields nothing but disappointment.

On *Starbright*, both Martino's guitar and the synthesizers of Delmar Brown wove some intricate and constantly changing imagery. Here, however, all is formula. Cutesy, trivial arabesques on Brown's EML 500 guitar are paralleled by Martino; Pat's rides all seem to follow a basic pattern of rapid cruising up and down the fretboard. Even Pat's awesome technique is a little suspect not to mention his conceptual abilities, which are cheapened by the "picking for his own amazement" deja vu's which turn most of the tracks into a boring exercise in self-indulgence.

As a result of these subterfuges, all the cuts sound virtually exactly like each other. Ironically however, the only concrete inspiration seems to come from the rhythm section. Bassist Mark Leonard's brief solo on "Mardi Gras" and drummer Kenwood Dennard's colorful cymbal work throughout furnishes momentary relief from boredom.

What is there to say for the technical quality of this disc? Stuck with constant regurgitation of bored, trite passages, Rothchild and Klein seem content to turn on the tape and let it roll. At times, Pat's guitar, the Rhodes, and synthesizers of Brown are virtually indistinguishable; little clarity and track-separation is evident. Yet, one gets the feeling that the goal of the artist is to produce an album full of space-age cliches, time-saving shortcuts, and worst of all, a marketable product.

I yearn for more of Pat Martino's crystalline, pure lead lines which sparkled throughout his earlier work on the Muse label. Those were the good old days. R.S.

CLASSICAL

BARTOK: ***Bluebeard's Castle.*** Tatiana Troyanos, soprano; Siegmund Nimsgern, baritone; BBC Symphony Orchestra, Pierre Boulez cond. [Paul Myers, producer; Robert Gooch, Mike Ross-Trevor, engineers.] Columbia M 34217.

Performance: **Intense**
Recording: **Bright and clean**

The excellent Kertesz recording on London with Christa Ludwig and Walter Berry is now joined by an equally noteworthy performance conducted by Pierre Boulez and sung by Tatiana Troyanos and Siegmund Nimsgern. Any attempt to choose a "best" performance of Bartok's brooding symbolist opera would be ridiculous in the face of such superb musicianship. Where Kertesz & Co., bathed in a warm, plush sonority, draw out the pathos and sadness of the score, Boulez and his younger soloists emphasize the bitterness of the outcome. The intensity when Bluebeard's new bride, Judith, accuses him of murdering his three previous wives and demands that the seventh door be opened is particularly urgent, and the brighter, more forward sound suits Boulez's more aggressive approach. Both views of the score are valid.

Both recordings "stage" the opera—which really amounts to little since the action is relatively static, taking place as it does in the imagination. The London producer, Erik Smith, writes that he gave "a good deal of thought and experiment to creating the right dreamlike sound for this opera." As the work begins, Bluebeard and Judith emerge from an upstage left door (right channel) and descend a flight of stairs. The London production attempts to follow this and further stage directions, and the soloists shift position in the two channels often. There is a definite illusion of a "live" staged performance with the orchestra on a lower pit level. In the Boulez recording, producer Paul Myers has the soloists enter as on the London record, but their subsequent movements are less specific; an excellent vocal/orchestral balance is attained, with all the forces appearing to be on the same level in a closely miked, reverberant setting—in this regard, producing less of a "live" performance feeling.

I cannot vouch for the accuracy of the soloists' Hungarian, but texts and translations are provided. Smith's notes for the Kertesz set are far more stimulating than Charles B. Yulish's stupid efforts at "relevance" ("Feminist listeners will find in this Bluebeard all that they detest.") on the Boulez set. S.C.

MAHLER: *Symphony No. 3.* Marilyn Horne, mezzo-soprana; Glen Ellyn Children's Chorus; Chicago Symphony Orchestra and Chorus, James Levine cond. [Thomas Z. Shepard and Jay David Saks, producers; Paul Goodman, engineer; recorded at Medinah Temple, Chicago, Ill.] RCA ARL2-1757 (two discs).

Performance: **Mixed**
Recording: **Massive**

Mahler's dictim that a symphony should encompass all of man's existence is no more successfully achieved than in his *Third Symphony.* The gargantuan first movement alone is longer than many of Sibelius' symphonies, and the range of mood is wider than in any other of Mahler's works.

This is 32-year-old James Levine's third and in many respects his most impressive recording in his Mahler cycle for RCA. The 32-minute opening movement has admirable dynamic thrust and cogency, the third movement posthorn solo by Adolph Herseth is the most beautifully played and evocatively balanced of any on record, and the fifth movement is the most "lively in tempo and jaunty in expression" of any performance I've ever heard. The second movement *Minuetto* is midway between Bernstein's Viennese *schmaltz* and the straightforward lyricism and simplicity of Haitink and Horenstein, both of whom I prefer above all. The fourth movement *Misterioso,* setting of the Nietzsche poem from "Also Sprach Zarathustra" is taken very slowly, and one's appreciation will probably depend upon reaction to Marilyn Horne's rather thick-textured vocalism. Levine's interpretation and the playing of the Chicago Symphony have so many excellent qualities that the miscalculation of the final *adagio* becomes even more regretful; in understandable love of this sublime movement, Levine caresses each note and phrases with so much rubato that at times one despairs of the music going anywhere. It's not a matter of tempo *per se* (although it is the slowest on record), but a lack of note-to-note continuity and flow which Bernstein, for one, has achieved in concert at an even longer timing.

The recorded sound is perplexing. The weight and size of the orchestra certainly comes across, but the unfocussed sound quality resembles many of the old Concertgebouw recordings on Philips which I discussed

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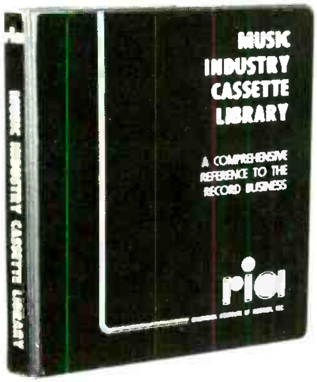

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in the April MR. Bass is muddy, timpani rarely make much effect, highs have no snap or bite, and winds are recessed into occasional inaudibility (especially the weak solo oboe). I would have liked to hear this score given the recorded transparency of RCA's Ormandy *Planets* reviewed above. London's spectacular recording of Mahler's Sixth with Solti and Chicago is a touchstone of what can be accomplished in Medinah Temple.

Surfaces on the new RCA discs were so-so—a problem since the cutting level is so low—and Jack Diether's notes are excellent, as always. Alternate recommendations for the Third are the unsurpassed 1960 Bernstein, the mid-sixties Haitink and the 1970 Horenstein, with the proviso in the latter recording of ludicrously underbalanced strings (but gut-pounding timpani and crystaline inner clarity of winds). S.C.

HOLST: *The Planets.* The Philadelphia Orchestra, Eugene Ormandy cond. [Jay David Saks, producer; Paul Goodman, engineer.] RCA CRL 1-1921.

Performance: **Good**
Recording: **Best available**

HOLST: *The Planets.* Patrick Gleeson, Eu Polyphonic Synthesizer. [Patrick Gleeson, producer; Neil Schwartz, Seth Dworken, Skip Shimmin, engineers; recorded and mixed at Different Fur Music, San Francisco, Ca.] Mercury SRI 80000.

Performance: **Straight electronic transcription**
Recording: **Monochromatic**

HOLST: *The Planets.* Isao Tomita. RCA ARL 1-1919.

Performance: **Augmented Holst**
Recording: **Impressive**

Also Sprach Zarathustra by Richard Strauss was not the only classic to achieve stardom due to Stanley Kubrick's *2001—A Space Odyssey.* English composer Gustav Holst's far more engaging work, *The Planets,* was popularized simply by dint of its coincidental astronomical theme—even though it wasn't used in the film. For awhile in the early seventies, it seemed that every major conductor had to record both works, but only now have Ormandy and Philadelphia made their

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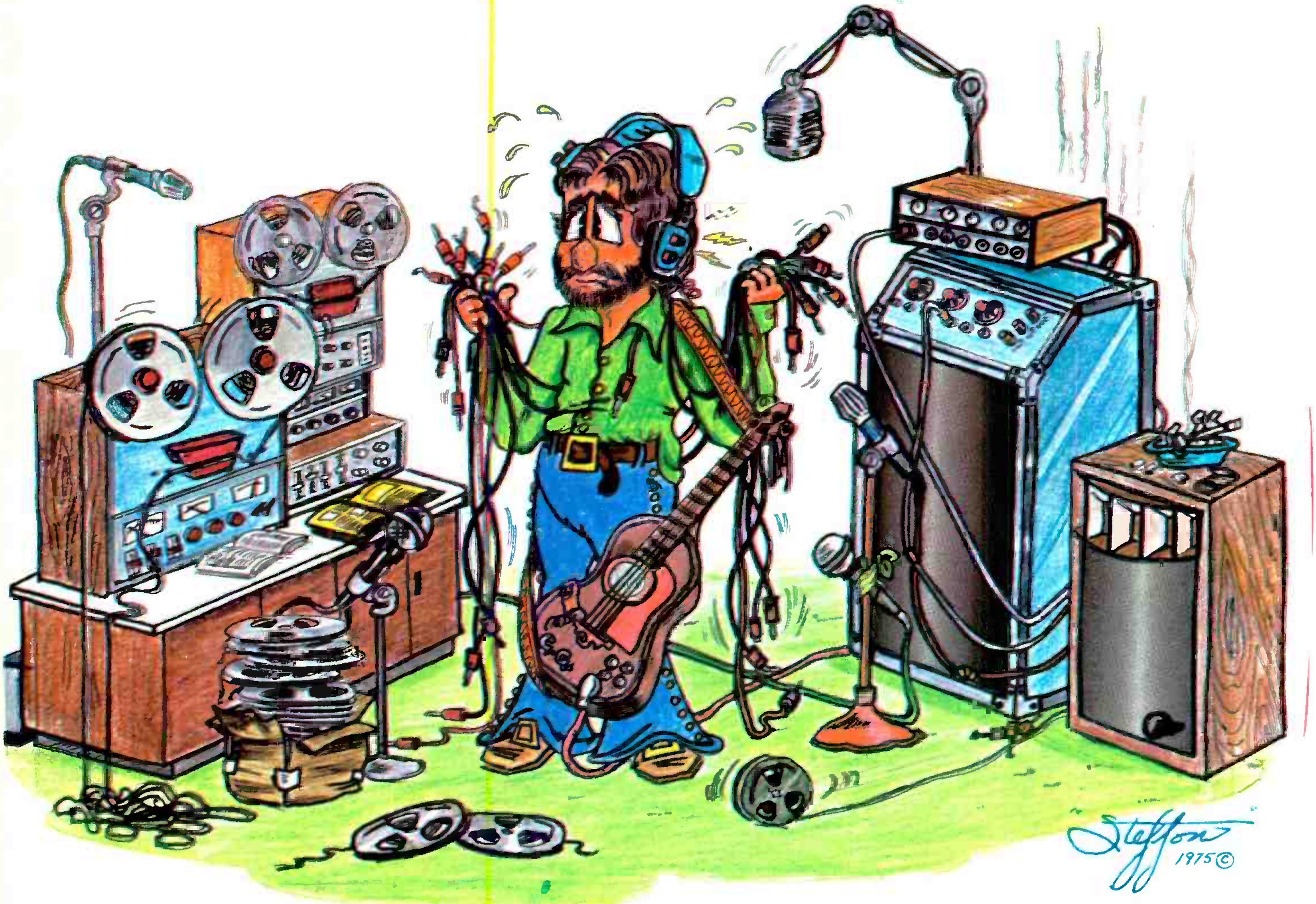

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first recording of *The Planets.* Concurrently, two electronic versions have appeared, and the sad fact (going by past sales figures) is that more listeners will probably hear these pale synthesized transcriptions than the real, full-blooded thing.

The bold colors and daring harmonies of *The Planets* (1917) set conservative English music, previously accustomed to the pomp and circumstantial stuffiness of Elgarian romanticism, back on its ears. One might say that Patrick Gleeson has retrogressed to pre-*Planets* England with his Eu Polyphonic transcription, for the thin, Hammond organ-like sonics will be laughable (or lamentable) to anyone familiar with the impact of the full Holstian orchestra. The only notable feature of this album is a striking jacket photo by one Paco North.

Gleeson appears to have followed Holst's score closely, but RCA's Tomita uses the music as a stepping-off point, beginning with his own interplanetary introduction and reshaping and abridging Holst's seven-movement suite to his own ends. I actually prefer his relaxed tempos for the *Venus* and *Saturn* sections over those set by the orchestral conductors. And the production *is* sonically impressive. But the dynamism, eloquence and humanity of this great work are completely missing.

The combination of a virtuoso orchestra and lucid, wide-range sound totally eclipses the electronic imposters, and Eugene Ormandy's usual no-nonsense interpretation is matched by the brilliant transparency and liveliness of the sound. None of the available recordings are ideal throughout, and it is possible to quibble over some of Ormandy's choices of tempo. His unwavering line in the slow movements (*Venus, Saturn* and *Neptune*) would beneficially support even broader tempos. The *scherzo, Mercury,* is a bit sluggish, but the resulting clarity (especially the timpani) is welcome. An unidiomatically sugary solo violin mars the *Venus* movement. Yet, overall, this reading compares well with any in the catalogue, and the recorded sound—so important in such a brilliantly orchestrated work—makes it a top recommendation. The organ pedal toward the end of *Saturn* is a particularly good test of a system's bass response. Surfaces were horrible, however. S.C.

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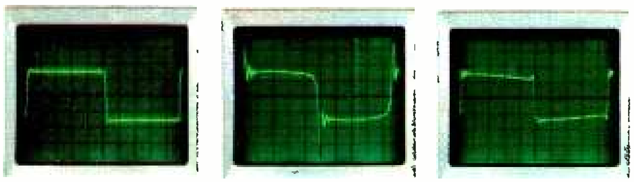

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