Profile:
Cheap Trick's Producer
Tom Werman

MODERN RECORDING

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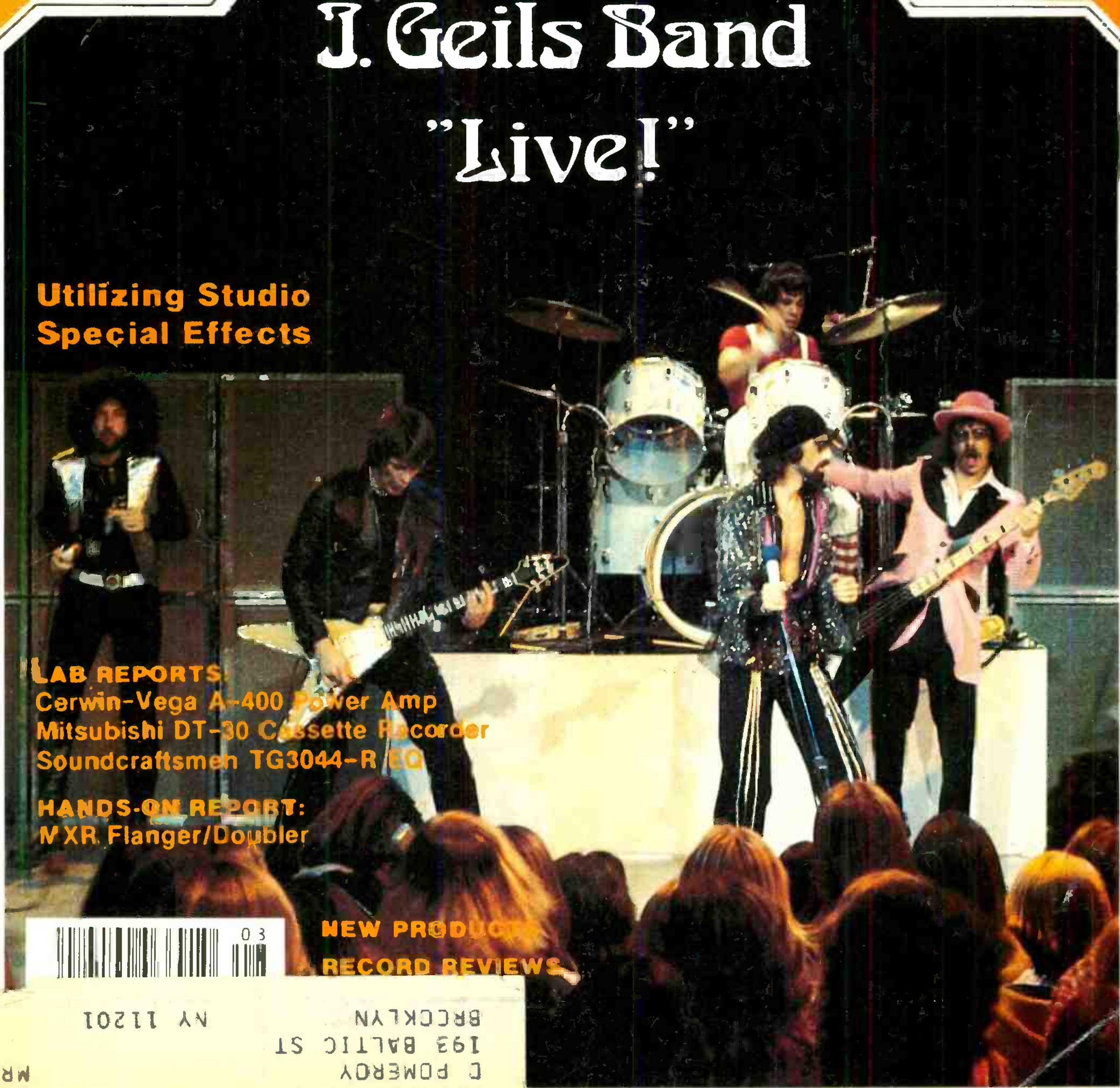
VOL.4 NO.11
AUGUST 1979

The J. Geils Band "Live!"



Utilizing Studio Special Effects

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HANDS-ON REPORT:
MXR Flanger/Doubler

NEW PRODUCTS
RECORD REVIEWS

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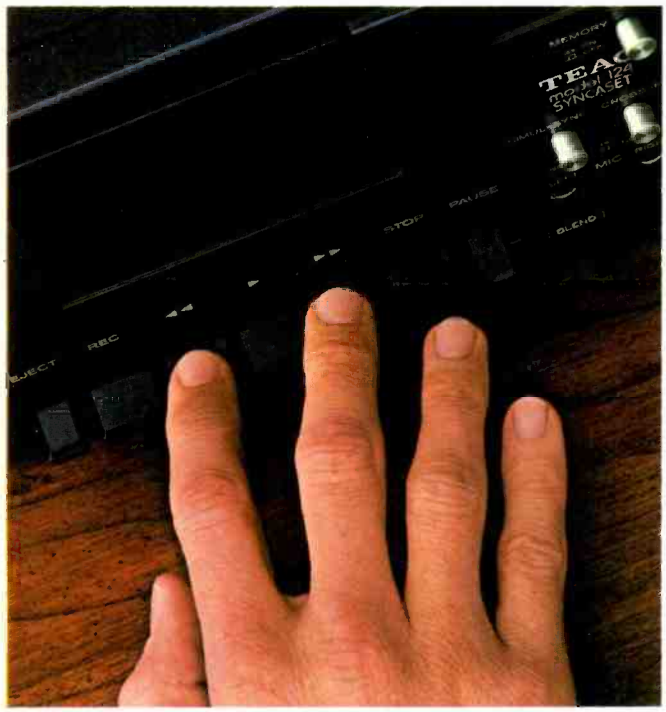

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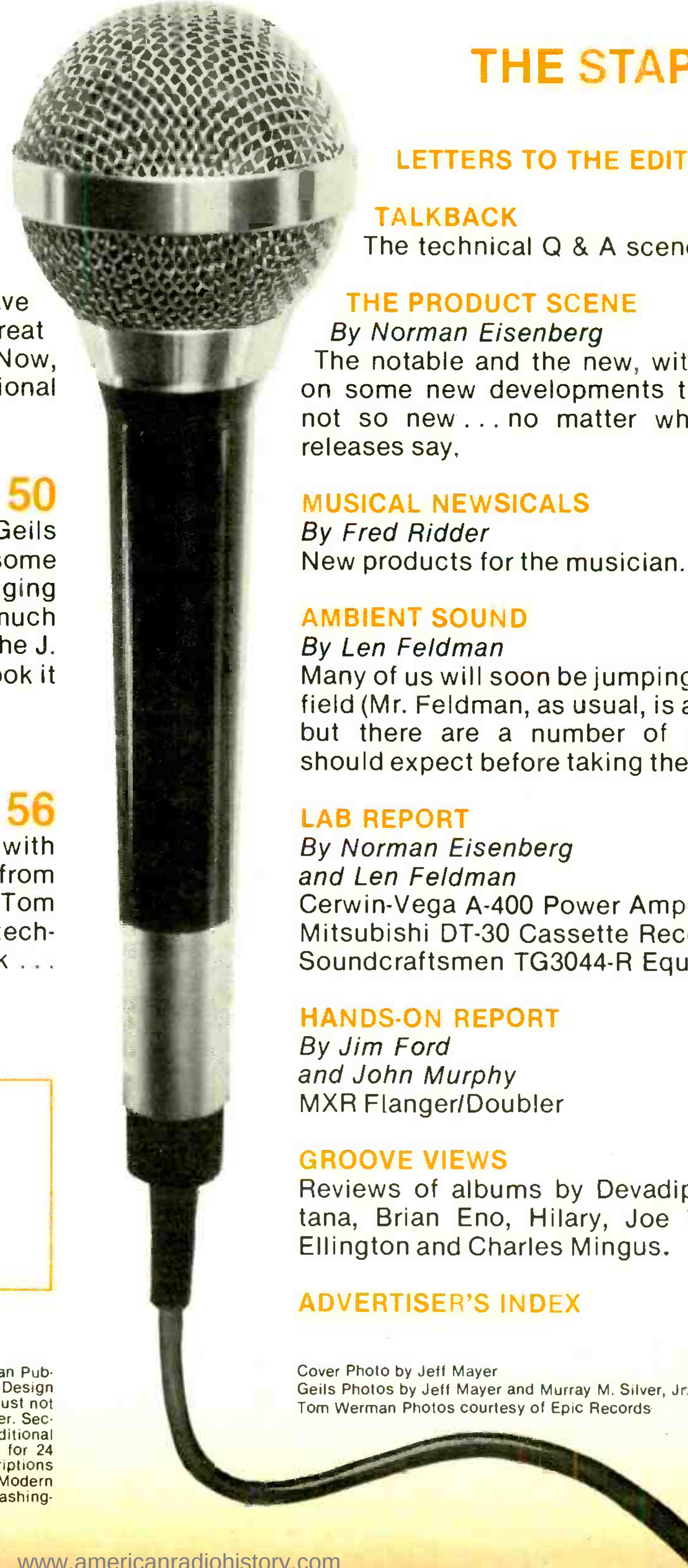

THE FEATURES

COMING NEXT ISSUE!
A Session with Firefall
The Electric Primer —Part I
Studio Designer John Storyk

THE STAPLES

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Letters to the Editor

Great and Noisy Debate

After reading the January 1979 "Letters to the Editor," I was amazed to see that anyone really believes that noise reduction doesn't affect or color the sound quality. *Of course* noise reduction colors the sound. Coloration is an unfortunate byproduct of the processes, and cannot be eliminated by present technologies. Let me explain why:

Dolby A (according to their 361 manual), starts by restricting the audio bandwidth, splits the sound into four frequency bands, limits one or all of them to remove your transients, separately compresses each band at varying levels, then puts them back together again, and onto the tape. Your signal goes through around 70 amplifier or processing stages, and that's for each of 16 or 24 channels. On playback it does it all again, in reverse! If the levels aren't matched perfectly, then the return signal doesn't get decoded properly. Dolby uses FET attenuators to control the levels, which are known for limited dynamic range, poor matching and distortion. Dolby also inserts masking noise to cover operational sounds. It is a real tribute to Dr. Dolby that the output sounds anything like the original at all! It is well known among the engineers in L.A. that it is best to mix on the same Dolbies that one records on, to eliminate unit-to-unit differences, or the result just doesn't sound like they expect. Many of these same engineers use the Aphex Aural Exciter to restore a lot of the "realism" and sharpness that seems to disappear when using the Dolby Laboratories system (remember the bandwidth and transient limiting).

dbx, on the other hand, uses a completely different approach to avoid Dolby's problems, but creates some of its own. To avoid band splitting, dbx uses a really great RMS level detector, the best in the business, and compresses the whole band at once, 2:1 for greater noise reduction. But greater compression also means greater problems on expansion. Tape machines can be aligned to a typical accuracy of ±2 dB so that record and playback, especially on two different machines, can be 4 dB off, magnified by 2X expansion to 8 dB! dbx also uses separate high frequency pre- and de-emphasis, adding another source of error. Why do I keep talking error? Because of the tolerances that exist in electronic components. No manufacturer can afford to build *perfect* equipment, and if they did, we couldn't afford to buy them. So when we add all these electronic errors to other variables such as different brands of tape (machines are usually aligned for just one type) and the tape machines themselves, the problems with encode/decode processes become much clearer. But dbx has other factors which color the sound, too. Their VCA, which varies the levels, has various distortion products which vary with both signal level and attenuation, so it distorts differently on playback than on record. It suffers from crossover notch distortion, modulation noise and log-antilog non-linearities, as do most of the other VCAs on the market (Burr-Brown devotes a whole chapter in their book on function circuits to the problems with log-antilog converters). Only the new VCA from B&B Audio is free from these problems that absolutely alter sound quality.

All right, let me summarize quickly: Noise reduction colors the sound due to physical limitations of its circuits; en-

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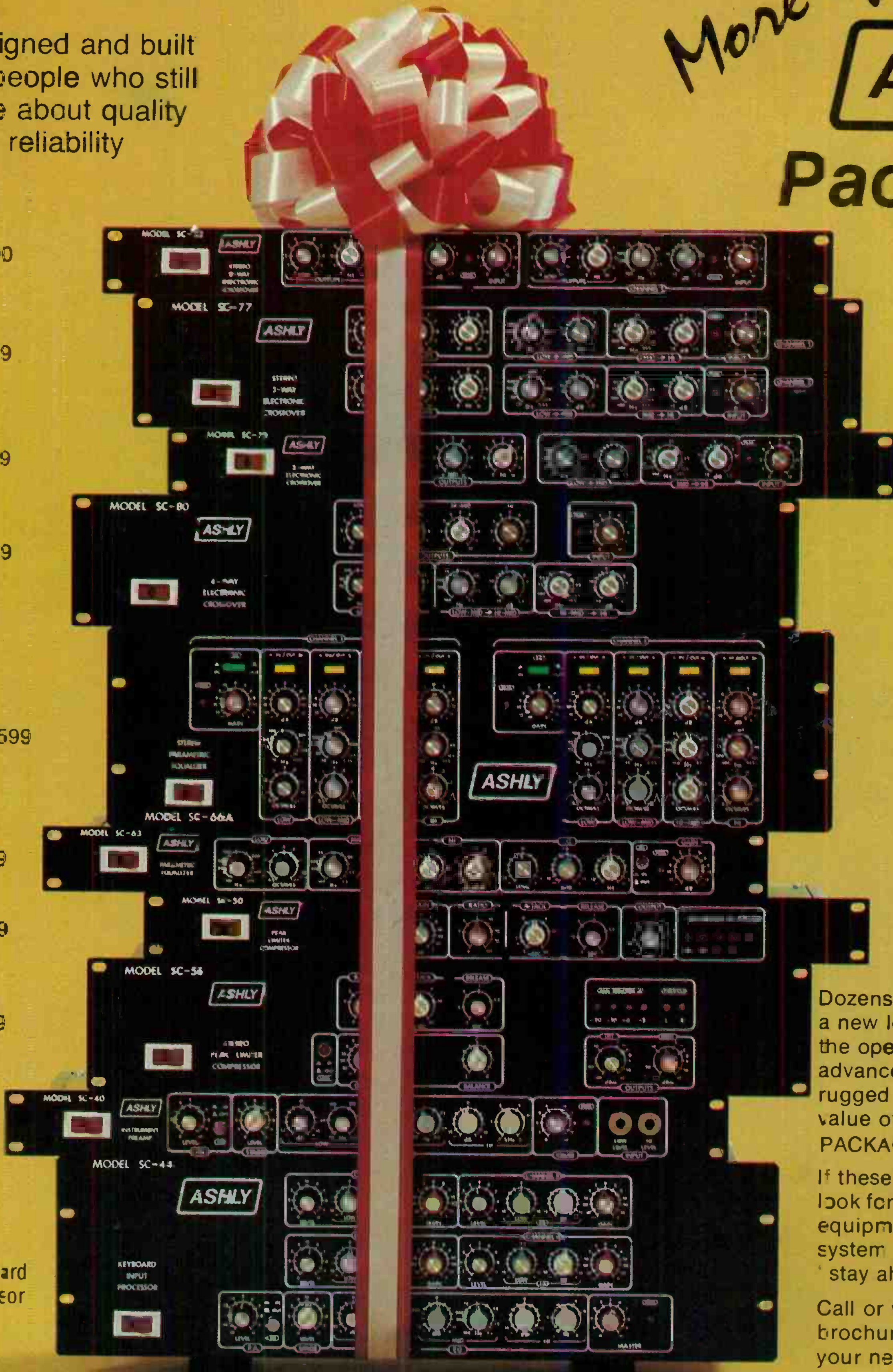

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code/decode errors, bandwidth and transient limiting and VCA imperfections.

Mr. Kinzel and the Marshall Tucker Band are certainly using the wisest approach to noise reduction, choosing the lesser of the evils between noise and noise reduction according to the situation (according to *Pro Sound News*, West Coast clients now prefer 30 ips to noise reduction). I strongly suggest to any user of noise reduction equipment that he compare in vs. around, using some live, noncolored material and learn his system's limitations. If anyone cannot then hear the difference, they should have their monitors and/or their ears checked. There is no reason to settle for second-rate sound.

—Jon J. Sanserino
Director, OEM Products
Aphex Systems Ltd.
Los Angeles, Ca.

We offered both Dolby and dbx the opportunity to respond to the above letter. The response from Dolby Labs follows; dbx declined to respond in writing, but its representative did comment on the topic, expressing dismay that Mr. Sanserino seemed to be using this space to promote the Aphex product largely by denigrating Dolby and dbx systems.

We reserve judgment. —*Ed.*

We believe that both the letter from Mr. Kinzel in the January issue and the one from Mr. Sanserino raise the same philosophical question of the purpose of the tape recorder and its immediately associated equipment (such as noise reduction). Is the objective to have the input signal and that coming off the tape sound as close as possible, or should the tape somehow sound subjectively "better" or "more real?"

Mr. Kinzel, for example, states that when recording drums or percussion, "the non-Dolby (tape) will sound more *real*, as if you were hearing the drums 'live.'" He does *not* say "the non-Dolby tape sounds closer to line-in." While that may be what he meant, we suspect otherwise, because we have heard objections before to the Dolby system's effects on high-level percussion. In each case we investigated, the result has been the same: the recording engineer was accustomed to saturating the tape on such program material, and subjectively preferred the coloration added by the resulting distortion. The Dolby system reduces that distortion and so the subjectively preferred coloration. However, in each case, we were able to show by means of blind listening tests that the Dolby tape was *closer to the sound of the input signal.*

As to Mr. Sanserino's comments, there have always been and always will be those who claim that the Dolby system colors the sound. Dolby Laboratories of course claims otherwise, and we and others have over the years repeatedly demonstrated, to our and their satisfaction, that the Dolby system does not alter real-life musical input signals. In addition, Mr. Sanserino makes at least one statement we believe to be misleading: "Dolby A . . . starts by restricting the audio bandwidth, splits the sound into four frequency bands, limits one or all of them to remove your transients . . ." Yes, bandwidth is limited—to 30 kHz. But the result is that transients are not affected by any more or less than the well-known effects of a 30 kHz band-limiting filter. Furthermore, the statement that "Dolby also inserts masking noise to cover operational sounds" is simply untrue. Finally, we point out

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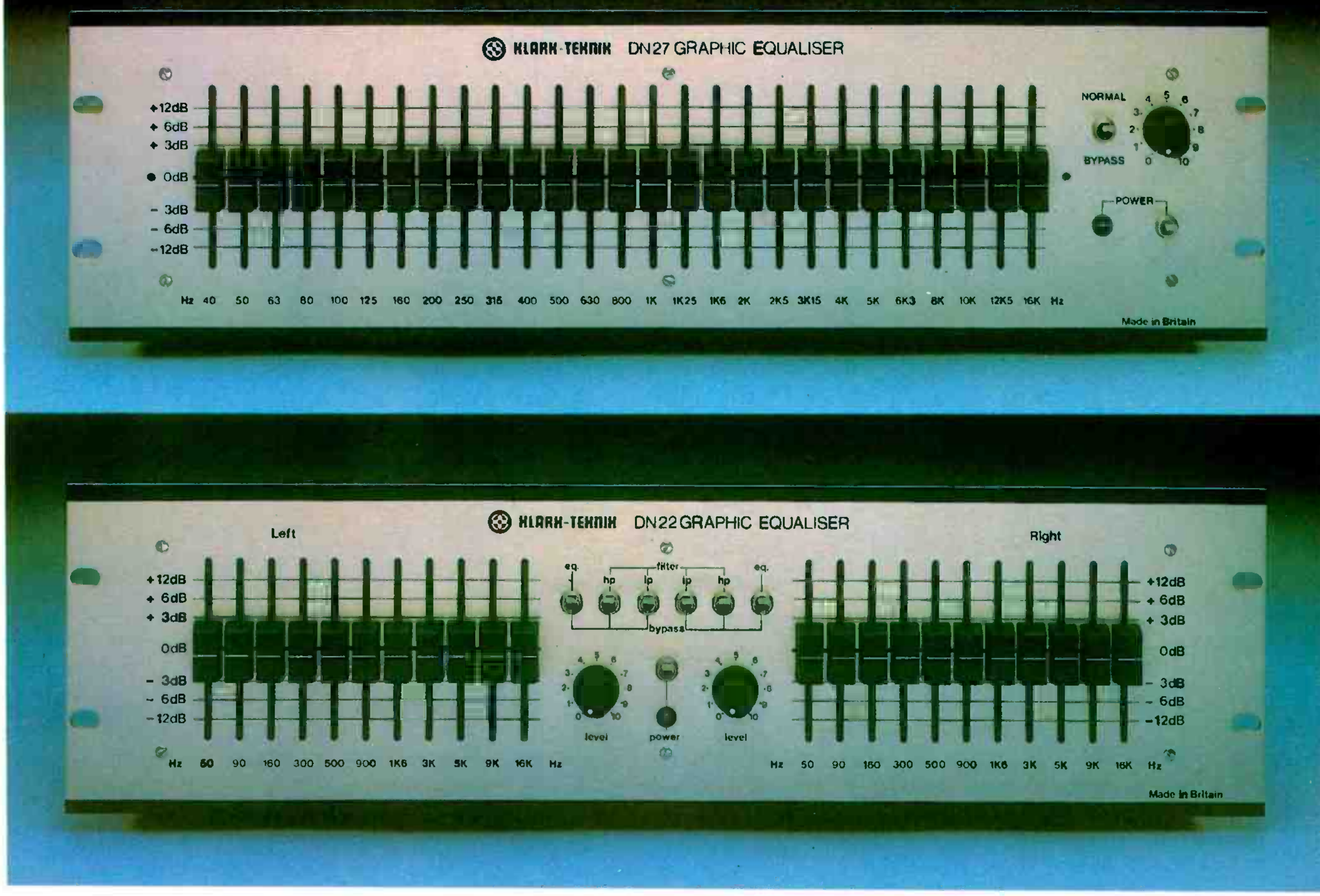

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that Mr. Sanserino's business, Aphex, is the deliberate alteration of audio signals for particular subjective effect, which may or may not be considered an improvement by any particular listener.

Dolby Laboratories' philosophy in this whole matter is highly conservative and equally straightforward. Which sounds more like the input, A or B? Is the tape copy with Dolby a more accurate replication of the input signal than the copy made without (or with another device)? What are the tradeoffs among amount of noise reduction, audible-side effects on what kind of program material, and cost? Do the listening test results hold up consistently with the widest variety of program material and operating conditions encountered in real use? And so on. The Dolby system has survived more than 13 years of the most searching and thorough testing and evaluation, by instruments and by ear, and has most importantly survived widespread day-to-day use throughout the world. We believe that the acceptance of the system is the ultimate judgment of its efficacy, and that the Dolby system will continue to survive, as it has in the past, the questions both real and imaginary which are from time to time raised about it.

Two final points. First, we do find ourselves in agreement with Mr. Sanserino on one issue: we too believe that the less the signal is manipulated, the better, and it is for that reason that the Dolby system treats only low-level signals, and limits compression/expansion to 10 dB. Second, we would like to emphasize that the Dolby system is remarkably tolerant of the calibration and frequency response errors encountered in studios around the world where a realistic amount of care is taken in setting up recorders.

—Joseph B. Hull, Jr.
Communications Manager
Dolby Laboratories Inc.
San Francisco, Ca.

All the News...

I would like to congratulate you on your fine publication and commend your integrity. In your article, "A Session with Leo Sayer," in the September 1978 issue, author Michael Gershman reports Studio 55's engineer Howard Steele as saying, "dbx does limiting and expanding to such an extent that you can hear it working." I had noticed this problem in my dbx 128 but was unsure as to the cause of the problem. Mr. Steele's observation has saved me the trial and error (and cost) learning of finding the same problem in a dbx interface for my 8-track. I was also pleased to see that this statement was not edited from the article as dbx is a consistent advertiser in your magazine. My compliments again.

—John Curran
Apple Valley, Ca.

Where's that Gizmo?

Can you please give me the address of the company that makes the Gizmotron (as mentioned in "Musical Newsicals" of your March '79 issue, p. 34)?

I would have filled out the reader service card, but it has expired by now.

Thank you, an avid subscriber,

—Wayne Kipp
Chagrin Falls, Ohio

Write to Gizmo Inc. at P.O. Box 139, Dept. G, Rosemont, N.J. 08556, or call (609) 397-2000.

Review Up Our Sleeves

I am waiting very patiently for your comments and a review about the best band to come out of Canada. You reviewed their album a while ago, Max Webster's *High Class in Borrowed Shoes*, but I've seen no mention of the album *Mutiny Up My Sleeve*, and I was wondering why. I'm hoping you will check it out—especially a couple of tunes called "The Party" and "Beyond the Moon," which are amazing recordings.

—J. O'Brien
Toronto, Ontario

You'll be pleased to know that we've contacted the label and asked for a review copy of Mutiny Up My Sleeve *(as we go to press, the album is not yet available south of your border). Expect the review within a few issues.*

Plugged In

My problem—delayed reaction. I am interested in Edcor's AP-10 headphone amplifier, which was shown way back in the June 1978 "Product Scene" (Page 34). Could you please supply me with an address or further information?

Also, do you have any features coming up regarding headphones and recording?

You've had me plugged in since your first issue. My library only lacks Vol. 1, Nos. 2 and 3. Thanks much indeed.

—Kenneth Sturm
Franklin Park, Ill.

We list Edcor as located at 16782 Hale Ave., Irvine, California 92714, phone (714) 556-2740. As regards headphones, we have no features coming up, but we'll see what we can do.

Your library is something many of our readers envy. Some of those issues, we think, are priceless.

Noise Reduction

I hope this letter reaches your pages, because I have an issue that concerns your readers and also myself.

That issue is—noise reduction.

I would like to see an article on this subject soon. I am a faithful subscriber for three years, and have enjoyed your articles on mixers, echoes and delays, but noise reduction seems obscure, although I understand the basic principle. Can you help?

—Robert Rasmussen
Tucker, Ga.

A two-part article by George Klabin that we published in Modern Recording's *May 1977 and June 1977 issues delved deeply into all aspects of noise reduction. Hopefully, you are one of those subscribers who hoards all reading matter and can check back. We have no plans at present to be running another article along these lines—Klabin's work is still valid—so those interested in back issues should act swiftly: we do have a small stack of both these mags in the stock room.*

Tape Tolerance

I believe that a great number of readers would like to see an article on "Acceptable Professional Recording Tolerances for Studio Tapes," encompassing a broad range of equipment performance necessities and minimum operating qualities for the production of a fair master tape. Of course, everyone has their own standards, but a typical set of guidelines would truly be quite a treat! How about it?

—R.R. Creech
Vancouver, British Columbia

Okay, we'll look into an article of this nature. We're a bit unsure, though, on what you mean by a "typical set" of guidelines. You could mean "average," but that is less likely. . . we'll investigate the best angle.

Our Fan with Banana

I am a sound engineer here in Argentina with a band called Banana—also, I'm an *MR* fan. Reading very carefully all the articles written in your magazine, I really learn a lot—and from people that are very experienced. As job experience in this field is something like, let's say, 99% of what makes for good work, well, there's nothing better than learning from them.

Unfortunately, we're living in a country in which "live" sound is still a nightmare—I think that only about two years ago did people start buying home stereo amplifiers with turntables and a pair of speakers, so we can say that they are just beginning to listen to music the way they should be. So think how long it will take for people like musicians, club-owners, theatre managers, etc. to come around.

Unfortunately—or fortunately (I really don't know which, yet), our band is one of the pioneers in this case. Maybe you don't know what it's like to be a pioneer in a place where people are very afraid to try new things: like going to a place to play where the maximum AC current outlet is 6 amps, and having to say to the local electrican that you've got to have a continuous 12 amps . . . he replies, "That's impossible—yesterday

MikeLowVolume was playing here and that AC outlet worked perfect." Or when the club owner asks, "Why do you come so early? Last week, MLV came half an hour before his show started and everybody enjoyed it."

Well, the thing is that we've got a 16 channel board, a Biamp system with Crown and Phase Linear, Altec woofers and HF drivers plus snakes, 16 mics, echoes, etc., plus all the stage equipment including individual Community monitors, etc., and I would like to know how we can set that system up (working) in half an hour.

Another thing working against us is that here, it is very difficult to buy good quality stuff. Probably, you can get stage equipment like Fender Rhodes, ARP, etc., but if you go to any store and ask for a digital delay or a parametric equalizer, they'll probably think you're from the moon or something like that—and here is the important point about your magazine: it helps me (I'm the one who goes to the U.S.A. to buy all the equipment) to be very well informed, to know what's new and to know what's the price for this and for that. Also, I write a lot of letters to the manufacturers and they send me all the information I need, but would I know the addresses or what each firm manufactures without a magazine like *MR?* I am a subscriber to a lot of magazines, like *Studio Sound*, *db*, etc., and they are really good; but none is so well-written and so interesting as *Modern Recording.* Honestly.

One last thing I'd like you to know is that we (the band) always need "five cents for a dollar," which is an expression we use to mean "how nice it would be if we had 1,000 more watts, or a monitor speaker there, etc." What I want to know is, is it always like this even if you're sound man with Pink Floyd or Chicago?

I don't want to make a bible of this letter, so the only thing I can say is that I would like to have an answer and that no matter what happens, mine is definitely the most wonderful job in the universe. Thank you for listening and thank you for doing *MR* and thank you for giving us every month a little piece of that fabulous Sound Wonderful country, the U.S.A.

Your friend and fan No. 1,

—Toro Martinez
Buenos Aires, Argentina

What can we say? We're delighted, we're flattered and we're glad to be such a positive influence in your work.

As to your question, we think that most truly creative performing artists and their support groups (engineers among them) feel as you do, forever speculating on the constantly changing creative ideal. Large or small scale. Happy monitoring!

Kansas Writer

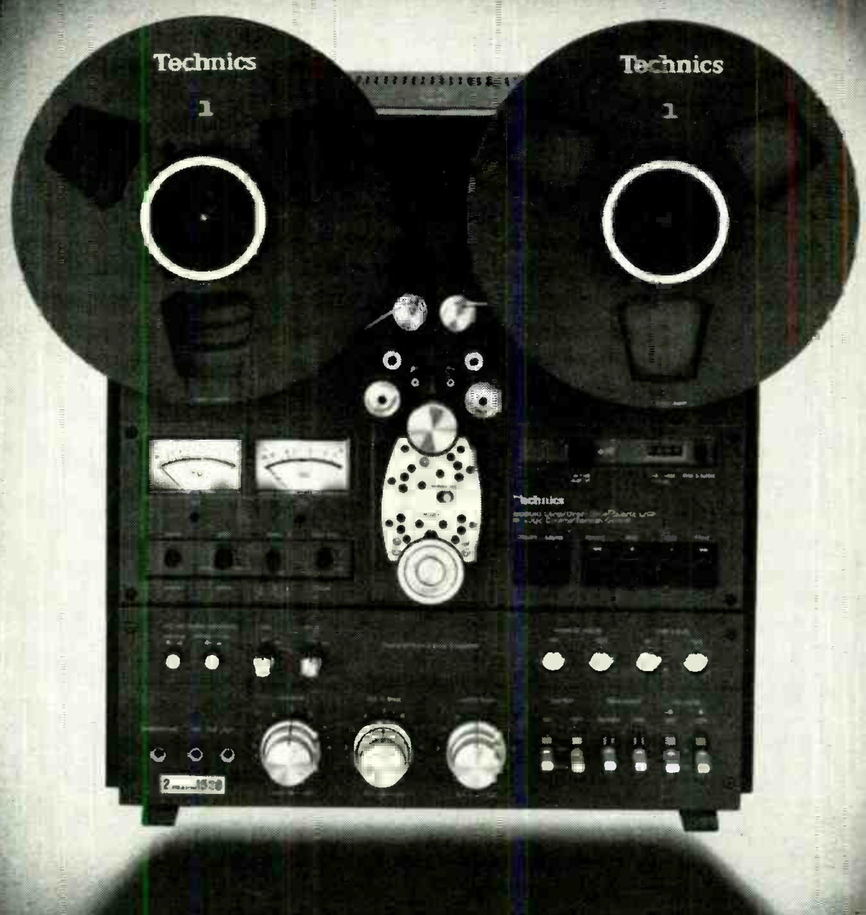
"A Session with Kansas," Modern Recording's *July 1979 cover story, was penned by Murray M. Silver, Jr. Our apologies for failing to credit Mr. Silver on the article's opening page.*

Another Bowl of Spaghetti

Brian Roth's article, "Spaghetti Sonata" was sound advice (I couldn't resist pun). Having experienced the joy/frustration of a few remotes, perhaps I can add a few tips.

If a "snake" is used with a raft of connectors on each end, go *now* & purchase a cordless soldering iron! It's bad enough to lose a vital run at the last minute, without threading a power cord thru dancers/tables (our remotes were

If Technics RS-1500 meets the high standards of A&M Records, why did we improve it?



After the music is recorded, and before it becomes a record, how do the top executives of A&M Records listen to Peter Frampton, Chuck Mangione, and their other stars? On the Technics RS-1500. Why? Because of its outstanding frequency response, constant tape speed and low wow and flutter. In fact they were so impressed, A&M Records bought seven more.

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out of a bar, in motion). Our best approach was a box on one end & connectors in the reel *hub* (no slip rings or twisting) for the other. Short cords, easily swapped, connected to truck audio equipment—with the ends thus rigidly held, the iron and track shoes could and did gather dust.

Crimpon terminals are great—*but.* On low level runs in long-term installations, *solder* the wires in. Oxidation can cause "opens" which will get bumped into conduction by an ohmmeter's voltage (embarrassing story behind this one). Also—flanged spades are my favorites: the bent tips help them stay put during installation. Ring shaped terms will never fall off, but one round of chasing 5-40 screws under equipment convinced me that it isn't worth it (Yes, barrier block screws are often 5-40 thread. Just try to find one in a junk box.).

Also- if you're going to put in a *lot* of crimpon terms, or just do a lot of connectors, consider spending $10 or so on a stripper. My Ideal stripmaster is *sacred.* I bought a similar but different brand one and gave it away after a week: the Ideal is still kerchunking after 14 years of abuse, without nicking like the $2 cheapies. Plus, its spring-loaded jaws are great for holding an XLR body during surgery.

Thanks for an informative & entertaining magazine. I enjoyed the crossword puzzles in the early issues (probably a writer's nightmare), and have survived a war with the subscription computer—though I still look on each issue as unexpected. Being in TV, perhaps *MR* isn't *totally* directed to my vocation, but it's just fine, thanks.

You might be amused/amazed to hear that while 2-inch quad machines can put out great video (requiring fairly wide bandwidth), one track on the tape is done with saturation (unbiased) recording. Shades of nostalgia! Needless to say, the fidelity requirements for it aren't severe.

Hope the "Spaghetti" comments are of use. . . .

—James Tolson
Chicago, Illinois

Who's Who

Mr. Robert Henschen: In *Modern Recording* magazine, Vol. 4 No. 4, January 1979, "Groove Views," page 86 . . . . The Who: *Who Are You*—"905" is John Entwhistle. Seems you went Townshend crazy.

—Cindy Wagner
Kutztown, Pa.

What Mr. Henschen had written was: "Several cuts on Who Are You *deal directly with Townshend's creative struggle for a new identity: in "905" he resists the mass production of men, minds and music. . . ." Bob's reply to Cindy is below.* —*Ed.*

In my zeal to impart the passion behind all Who albums, I glossed over Entwhistle's usual solid contribution to the group. Good point, Cindy.

—Robert Henschen
Record Reviewer
Modern Recording

Belated Info

*The "Lab Report" on Harman Kardon's hk3500 cassette recorder (*MR, *June 1979) did not include all the information in our usual format;* General Info *on the deck is as follows: Dimensions are 7 3/8 inches high; 16 3/4 inches wide; 10 1/4 inches deep. Weight is 20 pounds, and price is $549.*

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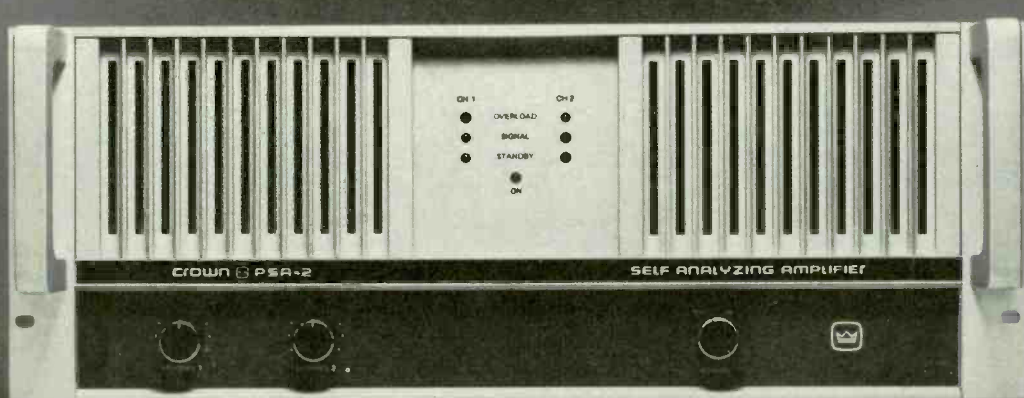

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TALK BACK

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

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

Traynor Tinkering Tips

I've been reading your magazine for almost two years now and have gained a lot of valuable information from it. I greatly enjoy the Talkback column as it provides answers to problems such as mine, which I shall explain.

The band that I work with is using a Traynor MX-12 mixer and I would like to know if it is possible to add break jacks to the inputs so that we could use limiters or compressors on individual channels. Our singers are always overdriving the inputs even with the attenuators set at −20 dB and our soundman is going crazy riding the faders to keep the output amps from clipping.

I would also like to know if we can add a mic input to the monitor section to make it easier to communicate between the stage and the technical crew as they are sometimes fifty or more feet away and yelling back and forth doesn't work very well.

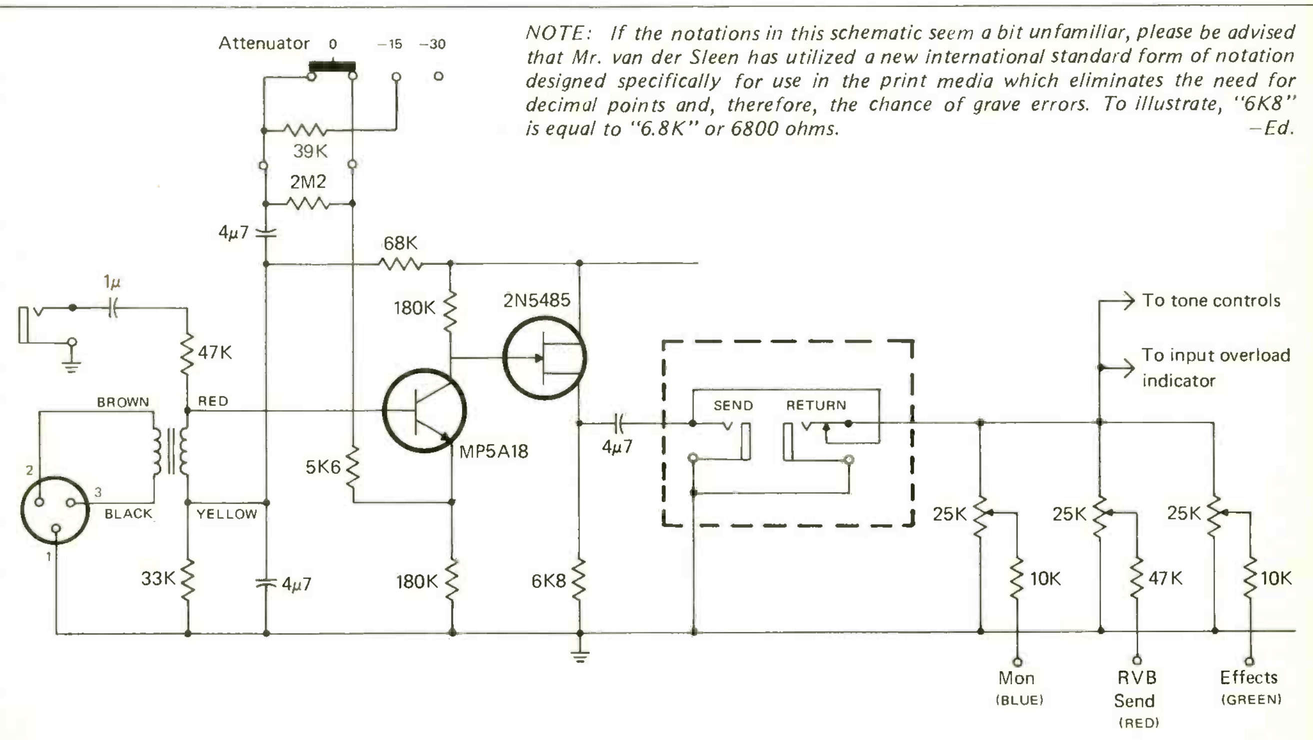
Thank you for whatever help you can give and thank you for a highly informative magazine written in a manner that even novices can understand.

—Mel Moore
Elkhart, In.

There are several possible reasons for the problem you describe.

The master might be set too low, requiring abnormally high channel fader settings. This could create clipping in the mixer stage.

Another reason might be that too many anti-feedback filter slide pots are set too low. This is equivalent to having the master faders set at a low setting, again requiring high settings of the channel faders. The normal setting for the anti-feedback filter controls is up full. Normal use of these controls for feedback reduction or room equalization requires only moderate adjustment of two or three of the slide pots.



NOTE: If the notations in this schematic seem a bit unfamiliar, please be advised that Mr. van der Sleen has utilized a new international standard form of notation designed specifically for use in the print media which eliminates the need for decimal points and, therefore, the chance of grave errors. To illustrate, "6K8" is equal to "6.8K" or 6800 ohms.
—Ed.

Circuit showing suggested locations of break jacks

Or, the bass and treble settings on individual channels are set too high, causing clipping in tone control stage. Adding limiters to each channel could be a bit expensive, but adding the break jacks for this or some other purpose should be relatively simple. Possibly the best spot is just after the input preamp stage, and before the feeds for monitor and effects. (See circuit diagram for illustration.)

The jacks used must be insulated from chassis ground and have the common connected to the "floating" circuit ground. (I suggest a Switchcraft N-112A or equivalent.)

To speak to the technical crew from the stage would require only that the crew shut down one channel fader, and then use that channel's monitor send control as a volume for the appropriate stage microphone. For the technical crew to communicate with the stage, it would require a repeat of the above on a second channel with a microphone plugged in and used at the mixer. A small amp with speaker or headphones could be plugged in to one of the monitor out jacks on the mixer to make it easier for the crew to hear "from stage" communications.

I hope this will help clear up some of your problems. Feel free to call or write if further information is needed.

—Dirk van der Sleen
Engineering Dept.
Yorkville Sound Ltd.
Scarborough, Ontario, Canada

Explaining the Echoplex

I would appreciate some information on how Echoplex cartridges are wound.

During my limited experience with winding my own blank tape, I find that every time I thread the newly-wound cartridge into the Echoplex and turn the device on, the tape bunches up inside due to what I believe must be uneven tension somewhere. Does the Echoplex require special tape or machines to wind the cartridges? Can I do this sort of thing myself at home?

I hope you can help me with this problem for I'm sure there are other Echoplex owners wondering about the same kind of things.

—Bob Stark
Bellevue, Wa.

Uneven tension is not the cause of your problem—the problem lies in the type of tape you're using. The Echoplex—which was conceived and designed in 1963 by Julius Kovats of Market Electronics and *has been marketed by Norlin Music of Chicago since 1964—requires a special, lubricated tape, similar to that used in 8-track cartridges. Standard tape loops will invariably stick.*

If the idea of buying 8-track cartridges only to unwind them doesn't excite you, Kovats assures us that blank lubricated tape is available from Market Electronics in bulk form. 3M also markets a lubricated tape (their stock number 178).

Once past the tape problem, Kovats warns us that winding Echoplex cartridges can be tricky to do at home. He reminds you to allow 37½ feet of tape for every minute of time desired.

For more complete information on the exigencies of the Echoplex, write directly to Julius Kovats, Market Electronics, 3867 Apollo Parkway, Willoughby, Ohio 44094, telephone 216-942-6969. He'll be happy to answer your questions and/or fill your tape orders.

Calculating Amplifier Power

While reading Jim Ford and John Murphy's Hands-On Report in the March 1979 issue on the Audio Perceptions Model 201 Active Crossover (page 74), a question was aroused.

How do you calculate the amount of power you're going to give to each component? How do you choose these components accordingly? Nothing difficult, I'm sure, but when you don't know . . . you don't know!

—Roger Guerin
Perfection
Montreal, Quebec

In any loudspeaker/amplifier system the amplifier should have sufficient power output to drive the loudspeaker to the required loudness level before the output waveform is "clipped" (overdriven). Unfortunately, there is no simple way to calculate the "right" amount of power to allocate to each loudspeaker driver in a multi-amped loudspeaker system. However, I'll try to give an overview of a scientific approach to the problem. Then I'll discuss a real (although not precise) approach to the problem.

The scientific approach requires the knowledge of several system parameters. It is necessary to specify the total volume of the room where the multi-amped system is to be used, along with the total surface area of the room and an average acoustic absorption coefficient. Given this information about the room, it is possible to determine the total amount of acoustic power that the loudspeakers must radiate into the room for a given sound pressure level (SPL) at a particular location in the room. Next, we must know the energy conversion efficiency of the loudspeaker system. The required acoustic power can then be divided by the efficiency factor to find the required electrical (amplifier) power for the specified SPL. The required number of loudspeaker drivers can then be determined if the power handling capability of each driver is known. The total required amplifier power is then distributed among the drivers for the various frequency bands based on the following considerations:

The efficiency of the different drivers in the frequency range over which they will be used. The more efficient drivers will require proportionately less power.

The frequency distribution of the energy in the program material that the loudspeaker system is to be used with. Most program material (pop music, for example) tends to have more

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energy in the lower half of the spectrum. (But clipping is most audible in the high frequency bands, so you can't skimp on power there.)

The frequency range covered by each driver. That is, if the audio spectrum is divided into octave wide bands (about 10 bands total) then the power should be divided according to the number of octaves covered by each driver. The frequency coverage of the drivers is of course determined by the crossover frequency (or frequencies) selected.

Although the above outlines a rigorous approach to the problem, it does not provide an answer to the question of how much amplifier power to allocate to the different drivers in a multi-amped system. The real world answer to the question "how much power for each driver" is "as much as you can afford," up to a limit. It's not unreasonable to use a power amp with as much as ten times the continuous power rating of the associated driver (continuous sine wave power as opposed to "program" power). This is because musical peaks can reach as high as ten times the average program power. For certain kinds of program material (e.g., individual studio tracks) this ratio can be even higher! Except for extreme situations, no amount of amplifier power is too much. This is especially true of the low frequency drivers.

So as it turns out, *calculating* the amount of amplifier power required by each driver in a multi-amped loudspeaker system is difficult at best and impossible at worst. On the other hand, providing enough amplifier power to such a system is not at all difficult, just expensive!

—John Murphy
Physicist/Audio Engineer
Ford Audio & Acoustics, Inc.
Oklahoma City, Ok.

Avoiding Tape Duplication Deterioration

Assuming that a tape deck is properly cleaned and demagnetized, can repeated playing of a master tape (as in real time duplication) adversely affect the frequency response of the recording? If so, does the use of dbx or Dolby noise reduction have any effect on the process?

—Timothy R. Hale
President
Horizon Recording Studios
Ripon, Wisc.

Dirt and residual magnetism are the two major contributors to the deterioration of signal on recorded tapes. Once these have been eliminated, the only real problems remaining are friction occurring at points along the tape path, and the consequent polishing (wearing away) of the tape oxide surface. To varying degrees, depending mostly on tape type and machine condition, this polishing occurs at all times during movement of the tape. The long-term effects of abrasion can be minimized for master tapes used for "real time" duplicating by using high-quality, professional-grade tape with low coefficient of friction, and by performing regular close-up inspection of all surfaces that come in contact with the tape oxide coating. These surfaces should be clean and non-magnetic. In addition, the surfaces should be visually checked for marring, pitting or burrs. Any of these defects will rapidly abrade even high-grade tapes and cause considerable deterioration of recorded signal quality (i.e., lower level degradation of signal-to-noise ratio and frequen-

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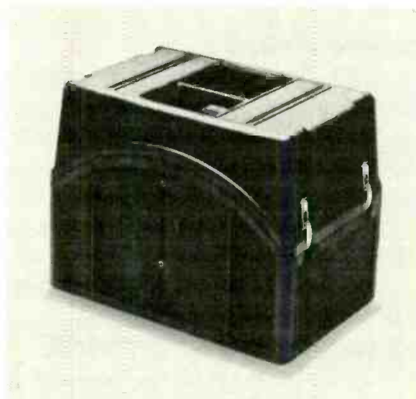

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cy response, drop-outs, etc.).

Using high-quality tape on a well-maintained machine will prolong the usable life span of a master tape, but not forever. Master tapes that serve as duplicating masters should themselves be visually inspected for signs of oxide wear. If highly polished stripes appear running lengthwise along the oxide surface, the master tape should be given an aural check. Rather than have an original master tape be subject to such abuse, it might be more practical to make a few on-speed, noise-reduction encoded copies of the original master, and to use these copies as duplicating masters.

The noise-reduction encoding (and subsequent decoding during the duplication process) will insure that the copies being used as duplicating masters will not have significantly higher levels of tape noise than the original master. This procedure employing noise-reduction encoded intermediate master tapes is used extensively throughout the professional mass-duplicating industry, and is equally well-suited to real-time duplication. Double ended noise-reduction systems such as Dolby can expand the signal-to-noise ratio of a properly encoded and decoded tape. However, such double-ended systems are sensitive to level for proper decoding. If the level on a noise-reduction encoded tape is lowered because of oxide abrasion or other effects of repeated plays, then the decoding will be imperfect, and signal-to-noise ratio will suffer as a result. Noise-reduction encoding cannot prevent or overcome the effects of tape wear.

—Peter Weiss
Contributing Editor
Modern Recording

TEAC—The Source

I own a TEAC Tascam Model 10 mixing console which seems to be producing more noise than I care to live with at this point. Compared with many of the newer (not to say more expensive or less flexible) consoles, the Model 10's signal-to-noise and headroom specifications seem to be a great deal less than desirable next to the 75 and 80 dB s/n claimed for the new units. I am also in the position of not being able to afford a whole new mixing console at the present time (not an uncommon problem among owners of small studios). Are there any modifications that can be performed on the 101B input modules to increase the s/n ratio as well as the headroom at the microphone preamp?

Also, it is not unusual for me to have to attenuate the microphone inputs 35 dB to prevent overload while recording music of medium loudness. Can this problem be overcome with a modification to improve signal to noise? Any assistance you can provide will be greatly appreciated.

—Jim Berg
The Barge Recording Studio
Wayne, N.J.

We suspect that there may be a miscalibration somewhere in the system, perhaps in the board itself. There is no modification available from TEAC that specifically attacks board electrical noise. Today's TEAC/Tascam boards are quieter than those of the past, but the Model 10 series didn't have a problem in that regard.

We invite you to contact us directly and perhaps with more information, we can help to pinpoint the source of the problem with your board.

—Roy Kamin
National Training Manager
TEAC Corp. of America
Montebello, Ca.

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ALIGNABILITY: Any tape recorder must be aligned to achieve maximum performance. With the MX-5050-B, all primary alignments are on the front panel. So is a 1-kHz test oscillator. Secondary alignments are inside the bottom panel. You or your maintenance people can align it fast and easy. This saves you time, money, and enhances your reputation.

INTERFACEABILITY: With a flick of the output switch you can plug-in to any system: +4 dBm 600 ohm or −10 dB high impedance. No line amps or pads to mess with. A perfect match everytime.

ADDITIONAL BENEFITS: Three speeds, dc servo ±7%, ¼ track reproduce, full edit capability, over-dubbing, noise free inserts, XLR connectors, NAB/CCIR switching, unique three-position alignment level switch.

PRICE: Suggested retail price $1,945 (USA).

MX-5050-B: THE CHOICE IS OBVIOUS

OTARI®

Call Ruth Fruett on 415/593-1648 for the name of your nearest Otari professional dealer.
Otari Corporation, 981 Industrial Road, San Carlos, CA 94070 TWX 910-376-4890
In Canada: BSR (Canada, Ltd.), P.O. 7003 Station B, Rexdale, Ontario M9V 4B3 416/675-2425

Don't let your speakers control your sound...

Electro-Voice stage systems put you in control.

Why should a musician allow his creativity to be limited by his speaker system? With amplifiers, pre-amps and the myriad of other state-of-the-art electronic devices available, why should a musician limit his sound by playing his equipment through a speaker system that hasn't changed in design since 1957? The answer is he shouldn't, and with Electro-Voice Stage Systems he doesn't have to. These three new instrument speaker systems let you hear all the sound your instrument is capable of producing, the way you want it to be heard, by choice not by chance.

The S18-3 lets you hear all the notes you never heard before from your keyboard or synthesizer from below 40 Hz to above 16,000 Hz. The famous EVM-18B delivers the bass. The Electro-Voice exclusive vented midrange driver delivers the midranges as efficiently as a horn, but without the typical "honky" small horn sound. The ST350A tweeter gives you clean highs over a solid 120° angle, eliminating the high-frequency "beaminess" that limits the enjoyment of your performance to the few people sitting directly in front of the speaker.

The two new bass guitar systems also incorporate the VMR™ vented midrange. It can be controlled from the front panel, thus giving the bass player total control over the midrange harmonics missing from "standard" bass enclosures. The B115-M uses a single EVM-15B in an optimally vented enclosure. The result is the tight sound preferred by many jazz bassists and studio musicians. The B215-M has two EVM-15B's for a bass sound with more "carry"; perfect for the larger venue or for the rock musician who wants more low-frequency "punch." In both cases, the VMR brings out sounds you may have heard before only on studio recordings.

All systems have identical styling. Black vinyl covered ¾" plywood construction for durability, further protected by extruded aluminum trim on all edges. A metal mesh grille screen protects the drivers from accidental abuse.

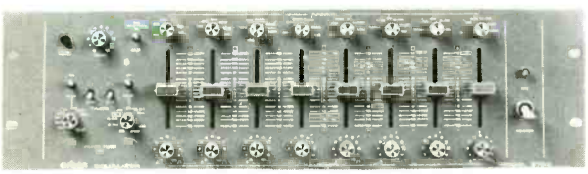
If you want to have control over your sound, these are the speakers for you. See and hear these superb instrument speaker systems at your Electro-Voice dealer.

EV Electro-Voice®
a gulton company
600 Cecil Street, Buchanan, Michigan 49107

THE PRODUCT SCENE

By Norman Eisenberg

ORBAN PARAMETRIC EQ



A single-channel, eight-band parametric equalizer, the model 672A, has been announced by Orban Associates of San Francisco. Graphic style EQ sliders, plus knobs, are featured. EQ sections are reciprocal and have a ±16 dB range. Also included are independent high- and low-pass filters. These may be used to limit the band in the usual way or, by means of the auxiliary low-pass output, to provide a two-way 12 dB/octave continuously tunable crossover. Special "tic" markings allow setting the controls to simulate an 8-band graphic equalizer.

CIRCLE 1 ON READER SERVICE CARD

TASCAM SPEED CONTROL

From TEAC comes word of the VSK-88 variable speed control for use with Tascam 40-4 and 80-8 tape machines to change either speed or pitch. The unit retails for $350 and can be installed at authorized Tascam service centers at no additional cost. According to TEAC, the control adjusts speed at 15 ips by plus or minus twenty percent. In musical terms the VSK-88 can adjust pitch a tone and a half, e.g., from A up to C, or from A down to F-sharp. Says TEAC, the new device can serve as a production aid to reduce over-long tapes to a desired time limit. They see added applications in cueing and timing audio-video soundtracks and in slide or filmstrip audio tracks.

CIRCLE 2 ON READER SERVICE CARD

CASSETTE AND CARTRIDGE STORAGE

What strikes us as an excellent storage unit for cassettes or 8-track cartridges is the "Music Box" from BASF, the tape manufacturer. Originally offered as a promotional item for BASF tapes, the Box is now available on its own, thanks to a rising demand—which, after handling the unit, we can well understand.

Measuring 19$\frac{1}{16}$ inches wide, 10$\frac{5}{8}$ inches high, and 4 inches deep, the Box weighs only two pounds, 10½ ounces. It is made of black high-impact plastic

and is quite sturdy. It has grooves in four dividers that can hold up to forty cassettes or cartridges, in or out of their original containers. The arrangement permits titles to be visible, with ample finger space for easy access in removing or inserting. The box may be placed wherever convenient, and more than one can be stacked. They also may be fastened, via screw holes provided, to a wall. List price is $15.

CIRCLE 3 ON READER SERVICE CARD

AMP AND PROCESSOR FROM GLI

GLI Integrated Sound Systems (a subsidiary of VSC Corp. of Long Island City, N.Y.) has announced a new amplifier and an audio processor. The amp is the model SA-2125, described as a rugged, lightweight and reliable power amp suitable for rack-mounting and rated by FTC standards for output power of 120 watts per channel (continuous, sine-wave) into 8 ohms with less than 0.1 harmonic distortion over the 20 Hz to 20 kHz range, both channels driven. Input signal for rated output is 1.5 V. Input impedance is a minimum of 30 K ohms, unbalanced. Slew rate is 20 V μsec; damping factor for 8-ohm loads is 150. Input connectors are ¼-inch phone jacks; outputs are standard ¾-inch spaced 5-way binding posts. Input level is adjustable via calibrated attenuators located at the rear of the unit. The amp weighs 27.5 pounds.

GLI's processor is the model 1010, offered as a companion piece to an earlier GLI product, the 3990 preamp/mixer. The model 1010 features complete tape facilities, including two sets of tape monitors with dubbing. A low-noise, three-band active equalizer is included and is switchable into line or tape paths. The device contains electronic logic patching for stereo reverse, mono and other combinations. Also provided are blend and balance controls. The 1010's meters can be calibrated to power amplifier overload, and also may be switched to show level between inputs. The unit may be rack-mounted. Among its specs: S/N of 90 dB; frequency response of 20 Hz to 20 kHz ± 0.25 dB; IM and harmonic distortions of less than 0.01 percent.

CIRCLE 4 ON READER SERVICE CARD

NEW OMNI MICROPHONE

From Electro-Voice there's news of a new shock-mounted omnidirectional microphone, the DO56. Primarily intended for hand-held broadcast and sound-reinforcement applications, the DO56 achieves isolation of handling noises and mic cable vibration by having the main acoustic cavity and the diaphragm/voice coil assembly isolated as an integral unit from the case. This technique, explains E-V, makes a capsule/case collision virtually impossible even under the most demanding circumstances. Response of the DO56 extends to 18 kHz; a slight emphasis in the 2-kHz to 12-kHz range is claimed to enhance the mic's vocal qualities, while a slow rolloff below 200 Hz reduces low-frequency noise interference. The mic also is protected against "P-popping" by a high-density Acoustifoam blast filter. Suggested retail price is $100.

CIRCLE 5 ON READER SERVICE CARD

AUDICON REVERB SYSTEM

Audicon Marketing Group of Nashville, Tennessee has announced "The Plate" which is described as "a new standard for reverberation systems for use in demanding studios and broadcast facilities." Says Audicon, there are no limiter or compressing units in the "extremely low-noise" amplifier; circuitry offers a wide dynamic range "with headroom to spare." The plate system in the device is pretensioned prior to shipment so that no adjustments or set-up is required. Price, including shipping within the continental U.S., is $4500.

CIRCLE 6 ON READER SERVICE CARD

EIA BOOKLET

The 1979 edition of Consumer Electronics Annual Review has been released by the Electronic Industries Association. Its 46 pages are crammed with general and statistical sales info on a broad range of electronic products. Up to five copies are complimentary. For six to 24 copies, the price is .50 cents each; for 24 to 99 copies, the price is .25 cents each; for 100 or more copies, the price is .15 cents each. Write to Consumer Electronics Group, Electronic Industries Association, 2001 "I" Street, NW, Washington, DC 20006.

CIRCLE 7 ON READER SERVICE CARD

AKG EXPANDS STUDIO LINE

Different types of condenser microphones, and a successor to its well-known model D 202 dynamic mic have been announced by AKG whose products are handled in the U.S. by Philips Audio Video Systems Corp. There are two new capsules for the model C 451 condenser modular-microphone system, which is based on a phantom-powered FET preamp and consists of various capsules plus a wide assortment of accessory parts for pro applications. The CK 22 capsule is an omnidirectional type. The CK 4 capsule has a figure-8 pattern.

A recently introduced microphone is the C 414 EB, a double-diaphragm single-channel condenser mic whose twin diaphragm capsule permits selecting several polar patterns.

New in AKG's line of stereo microphones is the C 422, a direct successor to the C 24. Again, a double diaphragm arrangement permits varying the response pattern, with quick change between the "MS" and "XY" stereo recording techniques. More compact and said to be ideal for mobile recording use is the model C 34. The new model C 424 is a four-channel microphone, while the model C 303 is intended primarily for speech recording outdoors.

The tiny model C 567 is a new lavalier microphone with an omni pattern and built-in FET-IC preamp. Described as a rugged cardioid intended mainly for in-studio pro musical recording is the model C 565.

In dynamic mics, AKG has announced the D 222, for studio applications. Successor to the model D 202, the new mic uses two transducers.

CIRCLE 8 ON READER SERVICE CARD

TDK TEST CASSETTES

A comprehensive series of laboratory-grade test cassettes has been announced by TDK. The thirteen cassettes are designed to cover recording characteristics alignment, playback level calibration, Dolby playback level calibration, head azimuth alignment, frequency characteristics and wow-flutter and tape speed measurements. The cassettes come in a sturdy self-contained container that may be perched wherever convenient. Cards in each cassette packet provide spaces for noting test results, and a booklet contains detailed specifications on each cassette. Price is "about $340."

CIRCLE 9 ON READER SERVICE CARD

PIONEER PRODUCTS

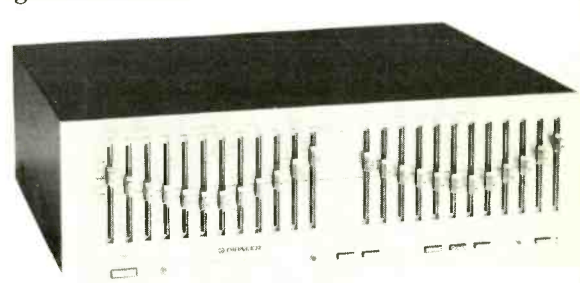
Of special interest from U.S. Pioneer Electronics are four new items. The model RT-909 is a reel-to-reel tape deck with 10-inch reel size capability, three motors and four heads. The fourth head handles reverse playback. Feather-touch logic controls permit fast buttoning, and the unit has a closed-loop dual capstan transport. A four-track, two-channel configuration, the RT-909 features separate mic/line and right/left input level controls, and a pitch control. Price is $895.

Also new from Pioneer is the SR-303 "bucket brigade" all-electronic reverb and three-dimensional

time/depth display unit. In addition to providing reverb, this device also displays it visually on a small front-panel screen. Price is $195.

The company also has announced the SG-9800 stereo graphic equalizer (12 sliders on each of two channels). EQ range is ±10 dB, and nominal center frequencies run from 16 Hz to 32 kHz. Excessive inputs are handled by input attenuators, with an overload LED. Price is $395.

The model RG-2 from Pioneer is a dynamic range expander and noise reducer. Control of its action is aided by a fluorescent meter plus indicators from 4 dB to 16 dB expansion. Upward gain is rated from +2 dB to +10 dB. Downward gain is rated from −2 dB to −6 dB. Attack time is 0.3 millisecond; S/N is given as 100 dB.



CIRCLE 10 ON READER SERVICE CARD

CERWIN-VEGA OFFERINGS

Among recently introduced products from Cerwin-Vega of Arleta, Calif. is the Metron FD-1 power amplifier, rated to produce 800 watts into 8 ohms in a bridged configuration. Featured is a fluorescent bar display with a 100-dB range that indicates power levels in single decibel steps from clipping down to 1 microwatt.

Other C-V amplifiers include the A-200, the A-400 and A-600, said to be aimed at the pro audio market. The 8-ohm continuous power output ratings, respectively, are 125, 225 and 350 watts. With 4-ohm loads, the ratings are, respectively, 240, 350 and 600 watts.

In microphones, C-V is offering the UD-1, a cardioid dynamic for vocal or keyboard applications. The model UE-1, an electret condenser design, is a basic instrumental mic, also with a cardioid pattern. For omnidirectional pattern use, there's the model OE-1 which is said to offer flat bass response down to 20 Hz.

Cerwin-Vega also has a new passive crossover. The model CX-2 is available in fixed frequency designs of 100, 150 and 250 Hz, has a 12 dB/octave slope, and requires no power supply.

CIRCLE 11 ON READER SERVICE CARD

SO WHAT ELSE IS NEW?

The French have a saying: "The more things change, the more they stay the same."

Audio people have a similar, though less elegant, way of putting it: "There's nothing goes down better than reinventing the wheel."

These homilies occurred to me recently as I contemplated some of the "innovations" and "breakthroughs" all over the audio scene . . . from direct-to-disc recording to vented speaker cabinets. I'm not saying that these things are not good, or in some cases even great. I would just prefer a little less hype and more acknowledgement of their technical antecedents—if only in the interest of historical continuity, which is something that I feel our culture needs.

For instance, consider the recent "mini components." If you take an ordinarily-good, medium-powered receiver and separate it into its basic elements of preamplifier, power amplifier and tuner, and repackage them accordingly, you will have effectively evolved three mini components. Indeed, many of the very first solid-state units that appeared some years ago were just such units. The difference between then and now is of course a vital one: those early efforts did not always stand up to expectations for too long a period, whereas today's models—thanks to the ongoing mastery of technology in the audio field—can be expected to work as claimed for relatively long periods of time. Who knows? The mini-component idea could induce a lot of buyers, who otherwise might shy away from quality sound, to take the plunge.

Rack-mounting is another "new" idea. Studio personnel of course know what is meant by a rack mount. It has open sides and back, and it holds equipment that is 19 inches wide by means of bolts through overlapping ends on the front panels. The "home" variations on this idea run the gamut from modified bookcases to units that are suited for only one brand of equipment. Whatever, I'd be cautious about stuffing a lot of high-powered gear into one unless you can be sure of adequate ventilation, not to mention relative heights for normal access to controls and meters.

Vented speaker cabinets are older than a lot of us are. What supposedly distinguishes the current crop is the use of the Thiele equations to get the correct dimensions for a given size driver. A further refinement is to run the data through a computer instead of relying on someone with a pad and pencil to come up with the right numbers.

Want to talk about metal tape? You know what the very first recording tape, invented by Poulsen in 1899, was? That's right—it was all metal. To be sure, the new metal tapes are particle-metal on plastic carriers, and they are far better. But someone must have been reading his basic textbooks.

Direct-to-disc recording also is not new. It was the first, and at one time the only, method of recording at all. Again, however, today's direct-to-disc recordings are uncommonly good-sounding, although the effort expended during a session by both the performers and the recording team must be uncommonly strenuous. Digital recording *is* new of course. In fact, it is so different that it probably will be quite some time before we have anything like a completely digital sound medium from performer-input to playback system output.

In the meantime you can make and enjoy a lot of first-rate sound with the hardware on hand, including those items that unblushingly owe their existence to some basic developments that took place years ago.

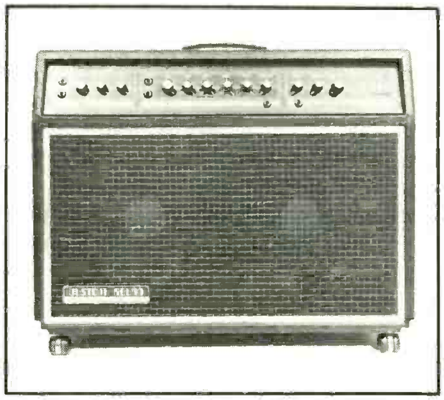
MUSICAL NEWSICALS

MUSICAL INSTRUMENT AMPLIFIERS

Yorkville Sound of Toronto, Ontario and Batavia, New York has introduced a new 200 watt, twin-cabinet guitar amplifier system, the TS-200. The head has high and low sensitivity inputs, input gain control and master volume control for full control of overdrive and distortion characteristics. Equalization is quite comprehensive with bass and treble controls with a +30 dB range, a midrange control which is a variable 10 dB notch at 400 Hz, and a four-band equalizer providing ±15 dB at 400, 800, 1600 and 3200 Hz. Output power is rated at 200 watts RMS into a 4-ohm load. The speaker cabinet designed to go with the TS-200 is the TS-98, a 200-watt, 8-ohm cabinet. The TS-98 contains nine 8-inch speakers in a vented reflex enclosure. Of the nine speakers in each cabinet, five are full-range models and the remaining four are special midrange drivers for a bright, punchy sound. An additional interesting feature of the TS-200 amplifier is the inclusion of a small monitor speaker in one end of the amp head to allow low-volume practice or tuning without the acoustic power available from the big speaker cabinets.

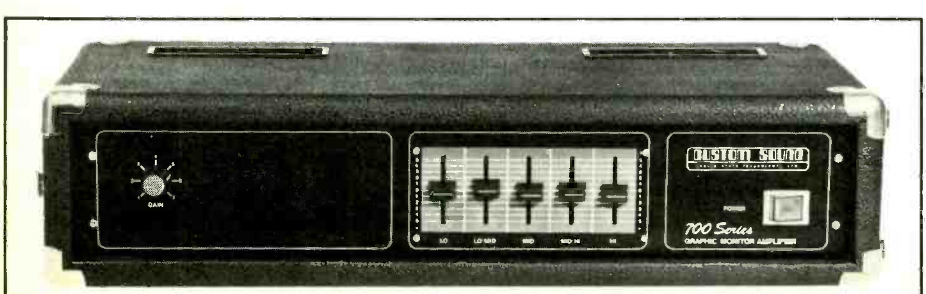
CIRCLE 12 ON READER SERVICE CARD

Custom Sound Solid State Technology, Ltd. of Oswestry, Shropshire, England, has announced its intention to enter the American amplification



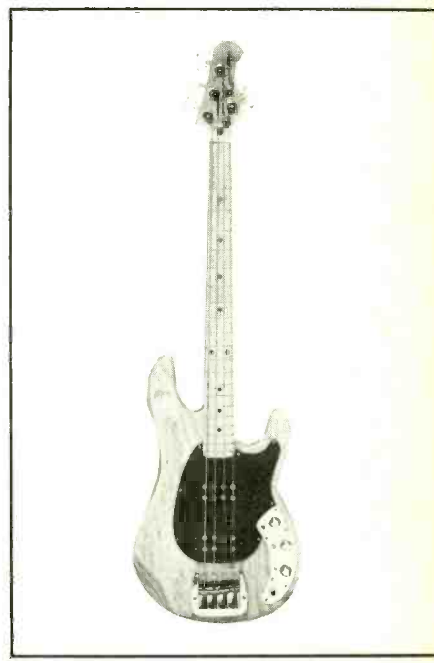
market in a big way this year. Custom Sound S.S.T is quite well known in England and Europe as manufacturers of a full range of professional quality amplification and P.A. equipment. Among the most interesting products in their line-up is the Trucker Tube amplifier, which will initially be available as a 100 watt/single 12-inch combo amp or as a separate amp head with a matching 4×12″ cabinet. Other noteworthy products include the Model 705, a 2×12″ combo model with a 100/150 watt amplifier section, and the 707 Bass Combo and the 706 bass amp with Delta Bass Bin.

CIRCLE 13 ON READER SERVICE CARD



MUSICAL INSTRUMENTS

The Sabre is a new model electric bass from Music Man Inc. which adds the versatility of twin pickups and more sophisticated electronics to the features of the company's popular Sting Ray bass. From its introduction three years ago, the Sting Ray has been adopted by many leading bass players for its built-in preamplifier and unique tuning head design which contributes to a well-balanced instrument. The new Sabre bass retains the preamplifier with its separate bass and treble controls and low impedance output, and adds a second pickup and a selector switch to give the musician the choice of a conventional bass guitar sound or a crisp, well-defined lead bass sound. The versatility of the instrument is further enhanced by a

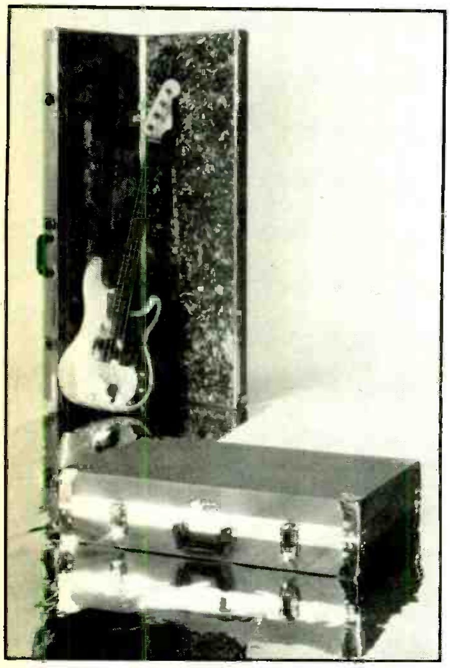


phase reversal switch for the two pickups' outputs and a "bright" switch. A new cast bridge assembly has been used on the Sabre to produce a full, rich sound with unusually long sustain. The body of the Sabre is a fully contoured design resulting in a lighter instrument and greater playing comfort when combined with the familiar Music Man neck.

CIRCLE 14 ON READER SERVICE CARD

ROAD CASES

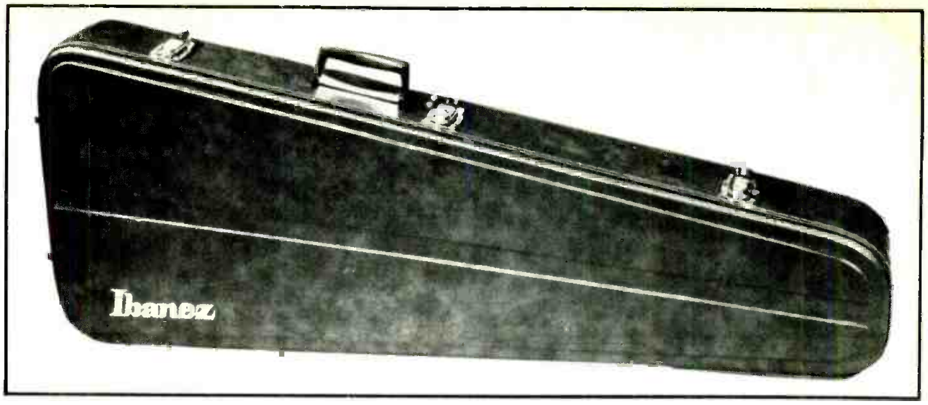
A new line of medium-duty road cases has been announced by ATS Cases. The new line is designed to fill the gap between cheapo cases which offer inadequate protection for instruments, and the various top-quality, heavy-duty cases which provide plenty of protection to the instrument but which also are likely to bankrupt most musicians. An additional benefit of ATS's new line is their light weight in addition to the attractive price. Construction details are comparable to most manufacturer's heavy-duty cases, and includes ABS plastic



exterior laminated to a plywood core, foam-lined interiors, tongue-and-groove valence closure, full-length piano hinges, and steel corner bumpers. ATS cases are available in blue, silver, gray and black.

CIRCLE 15 ON READER SERVICE CARD

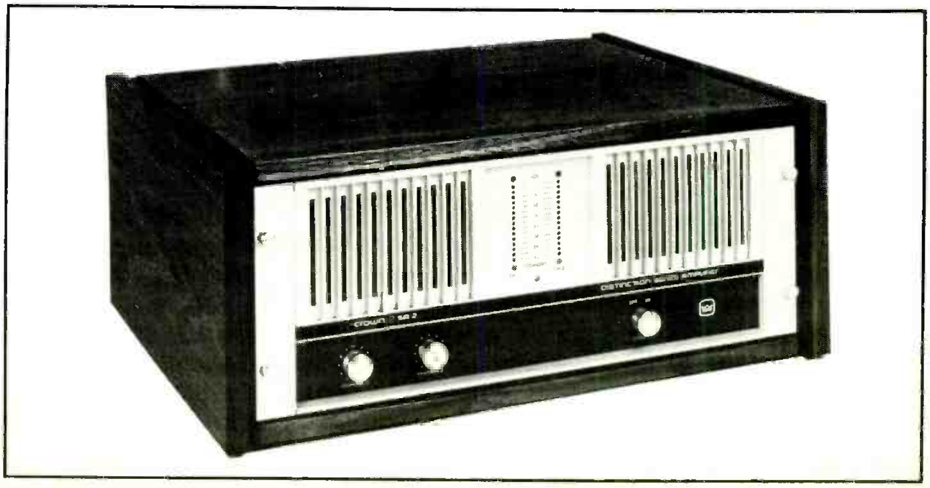
A new case for electric guitars called "The Vault" is the latest product from Ibanez. The case is built from 3/16-inch hardwood, lined with plush-covered foam. The overall shape of The Vault is a wedge shape, which offers plenty of protection and room inside the case without excessive bulk. In fact, it's in the category of interior room that The Vault really shines, because it was designed to accommodate everything a guitarist is likely to carry except an amplifier. Inside the top of the case is extra padding and a sheet music compartment that's big enough to accommodate standard size sheet music or the latest issue of *Modern Recording* with ease. The oversize equipment compartment has plenty of room to fit cables, straps and even an electronic effect or two. Furthermore, the lid to the compartment has an elastic pouch which is perfect for strings and picks and such, plus the lid serves as a guitar holder which supports the instrument at just the right angle to provide easy access for set-up, restringing or repair. The Vault is currently supplied as standard equipment with the Ibanez Studio Series of guitars.



CIRCLE 16 ON READER SERVICE CARD

SOUND REINFORCEMENT EQUIPMENT

The latest news from Crown International is the introduction of the newest high-powered amplifier, the SA2. The new model is rated at 220 watts RMS per channel into an 8-ohm load, or 350 watts RMS per channel into 4 ohms; both ratings are for both channels driven, 20 Hz to 20 kHz, total harmonic distortion less than 0.05%. The real breakthrough in the amplifier's design, however, is in the protection circuitry; the amplifier is referred to as a self-analyzing design which allows the output transistors to function at optimum levels under any conditions when a conventional design would have gone into output-limiting conditions. The SA2 builds on Crown's extensive research into the behavior of high-powered transistors, and includes an on-board analog computer which analyzes both the current operating conditions of the output devices and what they have been called on to do in the last period of time and from that analysis decides whether the devices are still operating within their safe limits. This is in contrast to conventional protection circuits which are triggered into output limiting whenever ar arbitrary voltage or current rating is reached. The net result of this is that the Crown should be capable of much higher peak power outputs without sonically annoying (and potentially speaker-damaging) output limiting. The SA2 is virtually two separate amplifiers sharing a common chassis since the channels have individual power supplies to reduce interchannel interaction. The unit features cooling via a two-speed fan built into the rear of the chassis which draws cool air in through a filter and blows it over specially designed heat sinks inside. In addition to a power on/off switch and level controls for the two



channels, the front panel of the SA2 features LED output level indicators for each channel. The indication range of these displays is 42 dB, and they incorporate Crown's IOC system which indicates all forms of output signal distortion, including clipping and output protection circuit activity, by lighting up a red LED when the output signal does not exactly resemble the input signal.

CIRCLE 17 ON READER SERVICE CARD

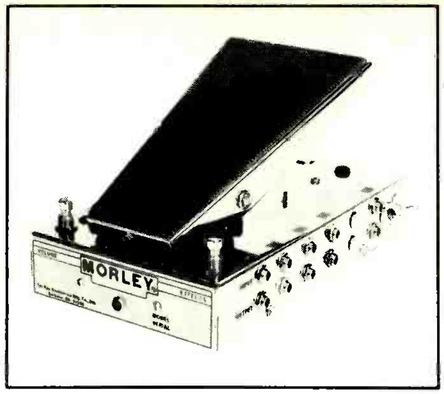
MUSICAL INSTRUMENT ACCESSORIES

RolandCorp US has announced two new effects accessories. First is the DC-30 analog chorus/echo for use in musical instrument amplification systems, P.A. systems or in home stereos. The unit uses a new analog delay chip and a Frequency Control Filter noise reduction system to provide a clean, quiet delay and chorus effect. The unit

has two inputs, allowing a "live" mic to be mixed with another program, or two instruments to be mixed and used simultaneously. The DC-30 has provision for use of an optional footswitch for effect on/off, or an optional foot pedal for control of echo repeat. The other new unit is the RV-100 reverb, a low-cost two-channel stereo or monaural unit. The unit uses a new limiter circuit to provide clean, "nonboingy" reverberation, and has a pan pot for control of spatial perspective.

CIRCLE 18 ON READER SERVICE CARD

Morley recently introduced its eighteenth effects pedal, called the Select-Effect. The Select-Effect is basically a control center for the musician who uses multiple effects devices; up to five separate effects devices are plugged into the Select-Effect which is then used to select each particular effects device and control the volume of the effect. The foot pedal of the Select-Effect moves in the horizontal plane to select one of the five



effects units plugged into the Select-Effect, and in the vertical plane to vary the volume of the effect selected. Two footswitches are provided, one of which bypasses the selected effect turning the Select-Effect into a straight volume control pedal, and the other of which deactivates the volume control action. Like all Morley products, the Select-Effect uses an opto-electronic system rather than a potentiometer as the control element for quiet operation and reliability, and is constructed of durable chrome-plated, heavy-gauge steel.

CIRCLE 19 ON READER SERVICE CARD

The makers of the Nasty Cordless system, who recently changed their name from Nasty Cordless, Inc. to Nady Systems, Inc., have announced a new addition to their product line, the Nady VHF System. This new system adds the advantages of a crystal-controlled VHF transmitter and receiver to the performance of the familiar Nasty Cordless system. The VHF system should eliminate tuning and interference problems even in metropolitan areas with combined airwaves.

CIRCLE 20 ON READER SERVICE CARD

Intersound, Inc. makes products which will be of interest to many musicians. Two are The Notch and the Bass Notch, which are compact, battery-powered matching preamplifiers with equalization and feedback-elimination capabilities. Both models feature a volume control, a wide-range treble control, and a sharp, variable-depth notch filter for elimination of feedback. The frequency of the notch is 370 Hz in the case of The Notch and 110 Hz in the Bass Notch; these correspond to common feedback frequencies for electric guitar and bass. The input of both devices uses a low-noise FET for high input impedance and quiet operation; nominal gain in both cases is 6 dB with the EQ set for flat frequency response.

CIRCLE 21 ON READER SERVICE CARD

A new Moog 12-stage Phaser has been announced by Norlin Music. Twelve stages of phasing are provided along with controls to vary the phasing stages and resonant stages from two to twelve. Phasing sweep is derived either from the internal oscillator or an external control voltage; a "motor start" feature engages and disengages the phasing effect gradually to simulate the effect of a rotating speaker device. Positive and negative outputs allow stereo panning effects, and a phase-shift-only signal is available at the output to produce pitch change effects in addition to the effect produced by mixing the phase-shifted signal with a non-phased signal.

CIRCLE 22 ON READER SERVICE CARD

Ross Musical Products has announced the availability of a new Distortion/Phaser combination. The phaser section has a sweep rate which is variable from .1 to 8 Hz, a sweep width which is selectable to either 4 octaves or 1.6 octaves, and a 70% regeneration loop which is defeatable. The distortion section has 40 dB of available gain resulting in a distortion threshold of 1.5 mV. Input impedance of the unit is 500 K ohms, output impedance is 50 K ohms, and the unit can handle a maximum input level of +5 dBV (about 1.6 V).

CIRCLE 23 ON READER SERVICE CARD

Jaco Pastorius | Acoustic Amps

Together for over a decade

phcto: Gorm Valentin

acoustic

professional sound equipment

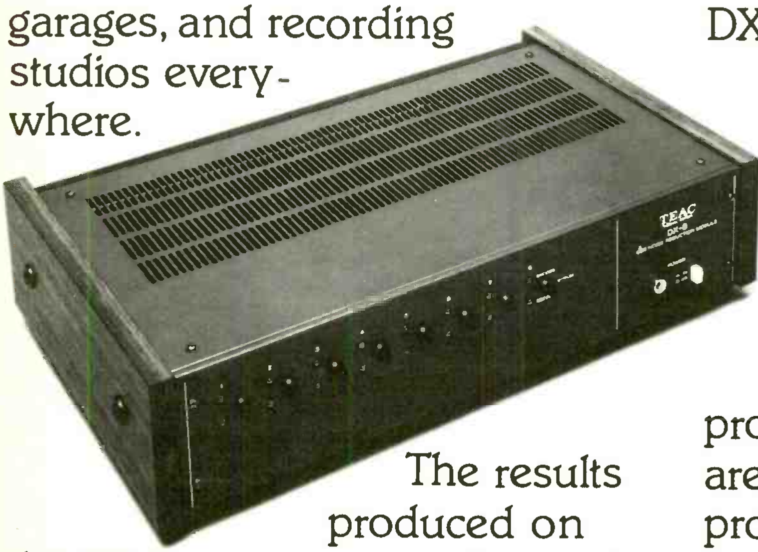
On stage: 330 top/408 enclosure/361 bass system
Off stage: 727 stereo amp/648 & 625 studio monitors

7949 Woodley Avenue
Van Nuys, California 91406

CIRCLE 54 ON READER SERVICE CARD

THE MACHINE THAT HOLDS THE WORLD TRACK RECORD.

The Tascam Series 80-8 has become the most popular 8-track multichannel recorder in the world. Its reliability has been proven in basements, garages, and recording studios everywhere.



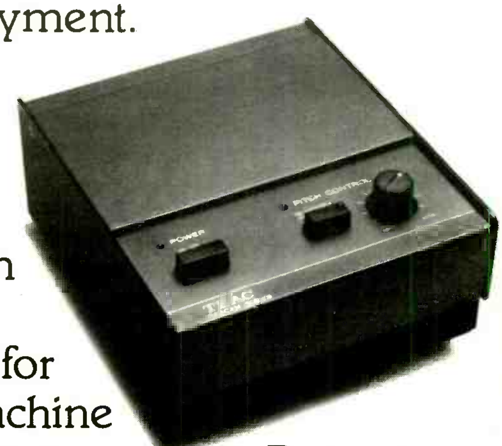
The results produced on the 80-8 are a matter of record. Sometimes gold.

The 80-8 proved a new standard was needed. Eight tracks on half-inch tape. 15 ips only. This new format allowed us to create a combined record/reproduce head, with full frequency response in the sync mode.

The 80-8 proved multichannel recorders could be relatively easy to operate. Our Function Select buttons determine the record, monitoring and dbx* status. One button for each track.

The 80-8 proved that performance and versatility could be affordable. Signal-to-noise is better than 95 dB (weighted) with our integral dbx unit (Model DX-8). Once installed, it's totally automatic. And our new Variable Speed Control** lets you adjust 15 ips ±20% to solve tough cueing and timing problems or add creative effects.

The 80-8 is proving that in professional recording, results are all that count. Because to us, pro means results. On demand. For payment.



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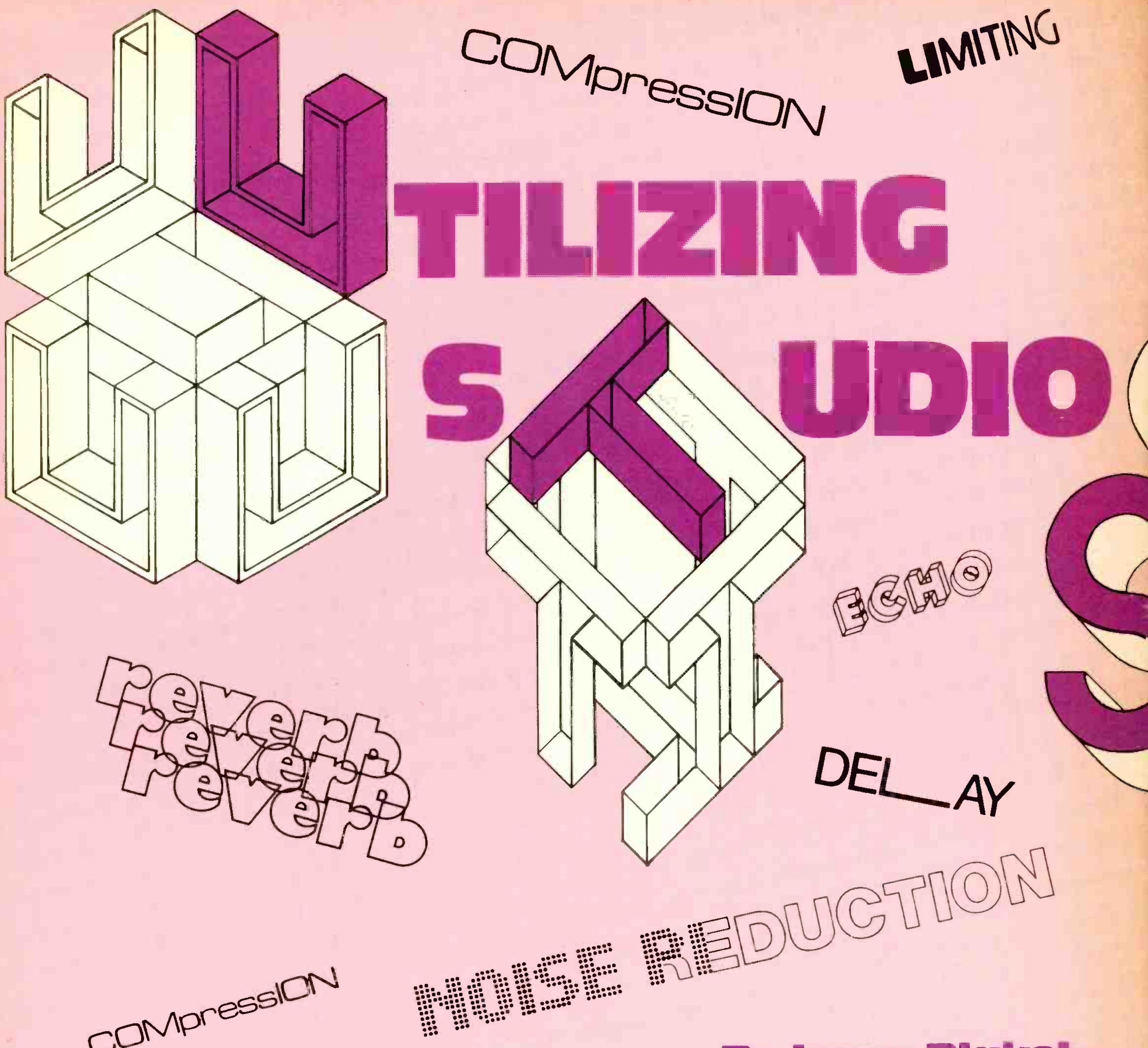
Utilizing Studio



By Larry Blakely

A frequently asked question is: "Do I need a compressor/limiter (or outboard equalizer, noise gate, flanger, phaser, echo unit, etc?)" Often the question broadens to: "What do I need each of these for and where do I connect each if I need to use it?" This last is a real loaded question.

In an attempt to answer such a question we will first explain a few of the common uses for each of the above-mentioned devices. This will enable the user to determine which of these devices will best fit his requirements and pocketbook.

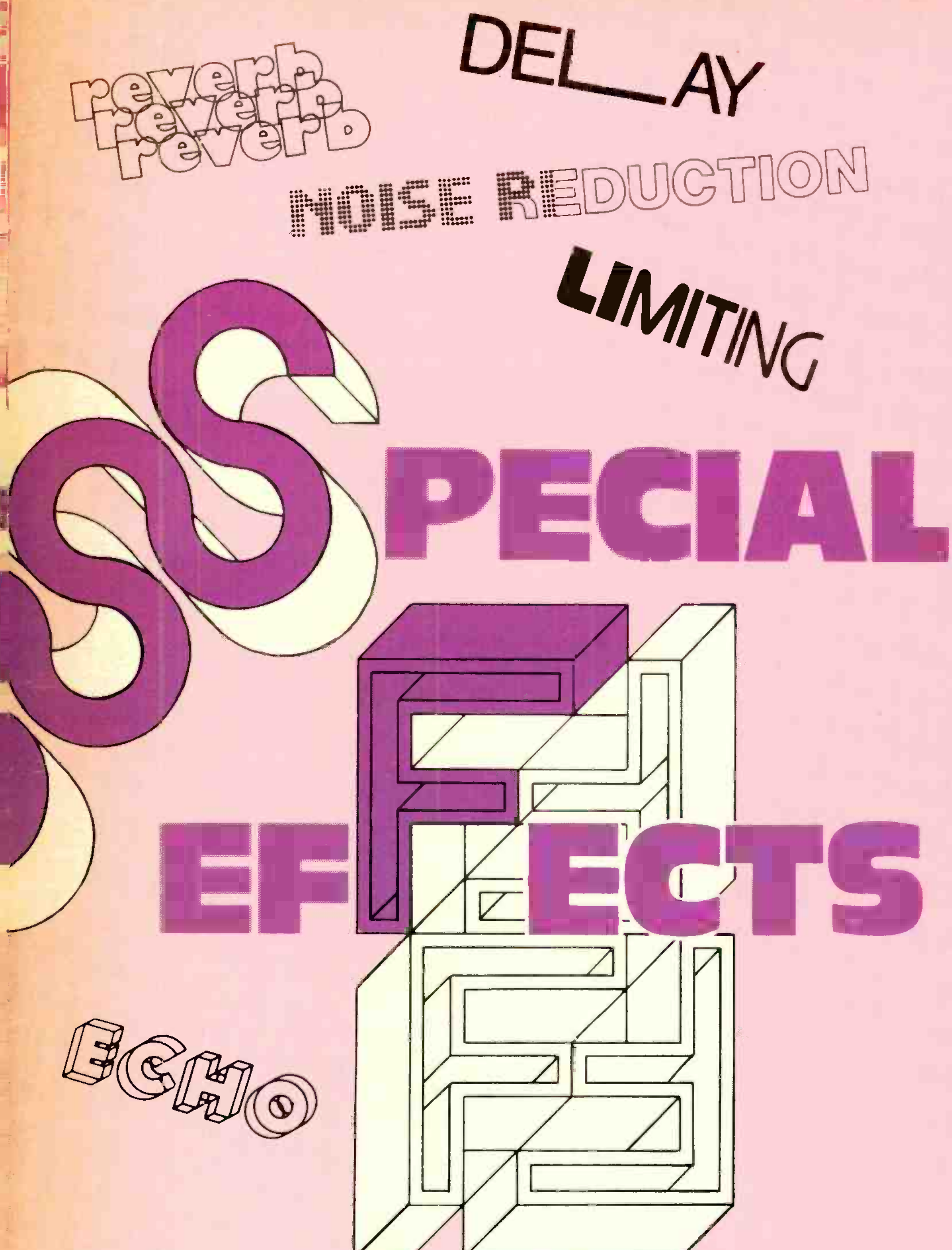
In the balance of this article we will explain some additional basic uses and effects that can be obtained with each type of auxiliary equipment, along with where it can be connected to obtain that particular effect.

In the April and May [1979] issues of *Modern Recording* there was a two-part article regarding the interfacing of auxiliary equipment in a recording system. The main thrust of the article was to help one to understand signal flow, block diagrams, patch points, and what could be accomplished by patching-in at various points in the recording system. If you have not read this two-part article, you should do so as it will increase greatly your understanding of the interfacing of auxiliary equipment and prove to be an invaluable asset to your recording career.

Effective Explanations

Let's now identify each of the major types of auxiliary equipment, explain its most common uses (not *all* of its uses), and where they can be connected for different effects and purposes.

Compressor Limiters have been used in recording and sound reinforcement for many years. A compressor/limiter is used to decrease the amplitude



(level) of an audio signal. It is an electronic volume control that will automatically reduce the level of a signal above an operator pre-set level. This operator pre-set level is called a "threshold." When the threshold level is exceeded by the audio signal it is automatically reduced in level by an amount proportional to the original audio signal level. If the signal level that exceeds the threshold were to be reduced by half (50%) it would be compressed by a factor of two to one (2:1). If the signal level that exceeded the threshold were reduced to one-fourth of its original level, it would be compressed by a factor of four to one (4:1). If a signal level that exceeded the threshold were reduced to one tenth of its original level, it would be compressed by a factor of ten to one (10:1).

Now, after having read the above you might ask, "What is the difference between a compressor and a limiter?" A compressor is described above in detail. A limiter is a compressor that works at high compression factors (also referred to as ratios). A limiter is usually referred to as a compressor with a compression ratio (factor) of 10:1 or higher. For example, a compression factor of 20:1 would also be considered limiting. The term compressor normally refers to a compressor with lower compression ratios (factors)—lower than 10:1. So in short, a compressor is a compressor that compresses at ratios less than 10:1, and a limiter is a compressor that compresses at ratios of more than 10:1. Got it?

Now that we know what a limiter and a compressor are, what are they used for? A compressor is used when you want a mild form of gain (level) reduction. A limiter is used when you want a drastic amount of gain (level) reduction. You would use a limiter when you do not want the signal level to exceed beyond a certain level. A limiter can be used to act as a sort of signal level safety valve.

For the purposes of recording, compressors and limiters are used to obtain a "certain sound" or to reduce the dynamic range of a signal to aid the mixing engineer in the mixing process. Since a compressor/limiter will alter the dynamics of an audio signal it will also in many cases alter the sound. For example: when drums are compressed or limited the dynamics can be drastically reduced while making the drums appear to sound fuller or "fatter." The extent to which compression or limiting is used to obtain such a drum sound is at the option of the recording engineer. The use of compressor/limiters on drums has been a standard practice in the industry for pop recording for many years.

Compression or limiting can be used for brass instruments to keep them from exceeding a certain level. Compression is handy to use on vocals as it will allow good presence and level from quiet portions of the vocal and prevent overloading when loud portions come along. Compression can be used on bass tracks to keep the level of the bass more even and up-front when you are mixing. This is a real help in keeping a "tight" mix (balance) between the bass, drums and other rhythm instruments. Good mixes can be done without the use of compressors or limiters but it requires far more work on the part of the engineer.

If you wish to compress or limit the signal from one microphone (usually a single voice or instrument) the compressor/limiter should be patched between the microphone preamplifier output and the line input. If you wish to compress or limit the signal from a group of microphones that is assigned to the same output bus, the compressor/limiter should be patched between the combining amplifier output and the line amplifier input (if these patch points are available) or between the console bus output (channel output) and the tape recorder input. These connections are commonly used for the recording of tracks in multi-track recording.

It is common practice to use compressor/limiters on individual tracks during the mixdown process. This allows the recording engineer the option of compressing or limiting a track during the recording of the track or to compress or limit the track during the mixdown process. Essentially this allows the engineer (you) the opportunity to do twice as much with a single compressor/limiter—by processing some tracks while they are being recorded and yet other tracks during the mixdown process.

To compress or limit during the mixdown process the compressor/limiter should be connected between the tape recorder output and the console (mixer) line input. The compressor/limiter can also be connected at the console fader output and equalizer input, if these patch points are available on your recording console.

Compressor/limiters can also be used on the stereo output bus of a recording console. A stereo compressor/limiter would be used for this purpose, or two mono (single channel) compressor/limiters that can be wired or strapped for stereo operation. The use of a stereo unit or the strapping of two mono units is mandatory. Failure to do this will cause image shifts of vocals, instruments or any signals that are center or near center in the stereo perspective. When stereo signals are compressed or limited the identical amount of gain reduction must be done in each channel or stereo image shifts will result.

A stereo compressor/limiter may be connected between the combining amplifier outputs and the line amplifier inputs on channels "A" and "B" (if these patch points are available) or between the console stereo outputs and the tape recorder inputs.

Noise Gates are also devices often used in recording and sound reinforcement. Noise gates can be used to remove noise on recorded tape tracks, or to remove unwanted leakage from other instruments or voices that are being picked up by a microphone or group of microphones. A noise gate will remove this noise or leakage by attenuating (reducing) the level when the signal level is below an operator adjustable threshold level. A noise gate is kind of an on/off switch for the signal. If you are trying to keep leakage *out* of a vocal microphone, it is therefore reasonable to assume that the microphone is picking up *both* the vocal and leakage. Obviously, the level of the vocal should be louder than that of the leakage. The noise gate has a threshold which the operator can adjust for a level at which the vocal will open the gate. When the vocalist is *not* singing, the level of the leakage will not be sufficient to open the gate—the leakage will be attenuated.

The same principle applies when using the noise gate on a recorded track during mixing of a multi-track tape. The threshold control is adjusted to a level that will allow the desired signal through the gate while rejecting the undesired signal. It is important to note that a noise gate is a "dumb" sort of electronic on/off switch and cannot distinguish between signal from leakage or signal from noise. Any signal above the threshold level will pass and any signal below the threshold level will be attenuated.

When a noise gate is used for a single microphone it is connected between the microphone preamplifier output and the line input. If a noise gate is to be used for a group of microphones that is assigned to the same output bus, it can be connected between the combining amplifier output and the line amplifier input, or between the console bus output and the tape recorder input. For mixdown purposes the noise gate can be used on an individual recorded track. The noise gate can also be connected between the tape recorder output and the console (mixer) line input, or between the console fader output and equalizer input (or equivalent patch points, if any).

Compander-Type Noise Reduction Systems are used in many recording systems. Common types of compander noise reduction systems are Dolby "A," Dolby "B," dbx, dbx II, MXR, Telcom and Burwen. These systems achieve noise reduction by first compressing the signal (a special compression process) prior to recording the signal, and then expanding the signal during playback (a complementary special expansion process). These special compression and expansion processes tend to keep the majority of the signal above the noise of the tape recorder and therefore reduce the audible noise. A reduction of noise from 7 dB to approximately 30 dB can be achieved in tape recording with the use of a compander type noise reduction system. Incidentally, it is important to understand that a compander system will only reduce noise (tape hiss) added by the tape recording process. A compander will not remove any noise in the signal source. For example: if you had a tape with noise on it that you wanted to remove, it could not be removed with a compander. Companders are of greatest value in "live" recording, or in making tape copies with no tape noise build up.

A compander is connected between the console bus output and the tape recorder input (compressor section of the compander). The expansion section of the compander is connected between the tape recorder output and the console input. Just remember that the compander is connected at the input and the output of the tape recorder. A compander process is a two-part process—special compression prior to recording and special expansion during playback.

Additional Units

Other special units are available beyond the compander-type noise reduction system, and the noise gate. *Expanders* are a type of noise reduction system that will make the quiet (low level) portions of music quieter. When the level of the quiet passages is made even quieter the noise level is also reduced. Other types of noise reduction devices are the *multiband noise gate*, and the *horizontal filter*. Noise reduction systems such as the expander, multiband noise gate, horizontal filter and noise gate are called "after the fact" noise reduction systems. These noise reduction systems will reduce noise that already exists on recorded tapes, etc. If you have a tape with noise you desire to remove, use one of these devices. A compander *will not* remove noise already existing on a tape, but will only prevent noise that would be added in the taping process.

Since these "after the fact" noise reduction systems are used mainly to remove noise from previously recorded tapes they are to be connected between the tape recorder output and the console input. These noise reduction systems can also be connected between

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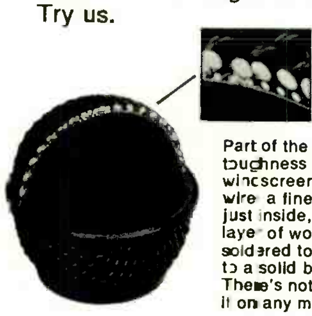

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the console fader output and the equalizer input (if these patch points are available).

Reverberating Thoughts

Delay Lines have become very popular in recording. Delay lines can be used for a number of different applications, but for the purpose of this article I will mention only two. One of the very popular uses of delay is to "fatten" out a sound such as a vocal or drums. This is accomplished by adding a small amount of delay to the original signal. The amount of delay can usually be varied or changed via the unit's controls to the engineer's taste, depending on the delay unit that is being used. If longer amounts of delay are used, one will perceive two distinct signals: first the original signal and second the delayed signal. This effect produces a sound that sounds like two sounds or double the original sound.

To connect a delay line for these purposes the unit would be connected to one of the echo send or cue (phone) send outputs and to an echo return or bus mult (parallel) to which that input is assigned. The signal is sent to the delay line with a console echo send or cue send pot. The delayed signal is added to the original signal by returning it through the echo return portion of the console or through a bus mult (a direct access to the console output signal).

You can also use a delay line to delay the send to a reverberation chamber or device. (Longer amounts of delay can be used—typically around 250 milliseconds—to delay the echo send signal to the chamber and produce a delayed echo. This procedure is most popular and has been practiced for many years.) To connect a delay line for this purpose, connect the unit between the console echo send output and the reverberation chamber (device) input.

Echo Units

Since echo units are available in various shapes, sizes and prices, let's first define the terminology. For our purposes a device that will "change" a signal such that it will sound as if it were being (or had been) produced in a large hall or a hard tile bathroom we will call "reverberation." A device that will produce discrete delays (multiple examples of the original sound) we will term "echo." The discrete echo effect is sometimes called "slap-back."

The most common of these recording studio devices is the reverberation chamber (still commonly called the echo chamber). The reverberation chamber is connected between the echo send output and the echo return input on the recording console. Such a reverberation device can also be connected between the headphone or effects send output and returned to the output buses or routed to additional inputs on the recording console. The dry (pre-echo) signals from each input must be combined via a mixing bus (echo send, cue send, phone send or effects send) and sent to the reverberation chamber and then the output of the chamber returned to the console via the echo return input or directly to the output bus (channel) to which the input signals are assigned.

If an echo device is to be used for a special slap-back type of effect, it can be connected in place of the reverberation unit should your recording console only accommodate one such device. If your recording console will accommodate two echo or reverberation devices, simply connect the echo device to the second echo send bus on the recording console and use the extra set of echo return inputs. Also, as mentioned earlier, the phone send, cue send and effects send portions of the mixing console can be used. The return signal can be connected to a line input (assigned to the appropriate output bus) with a "Y" connector to the output bus, or to the echo return input.

Okay, but . . .

"In what order do I connect all this stuff?" Another common question, and again not a simple one to answer. Let's take some of the units we have covered one at a time and outline potential problems and their possible solutions individually.

Compander Noise Reduction System: Perhaps the most critical unit of all. Nothing can be connected between the special compressor and the tape recorder input. Likewise nothing can be connected between the tape recorder output and the special expander input. All signal processing must be done prior to the compander's compressor input or after the compander's expander output. Failure to observe this will cause mistracking of the compander system and will possibly give you a tape that you can't use.

Other Types of Noise Reduction Units: Other noise reduction units can generally be connected nearly anywhere as described earlier in this article. It is good practice to allow the signal to be noise reduced *prior* to compression and/or limiting, and prior to entering a noise gate. It is generally not a good idea to use an equalizer prior to noise reducing a signal; however, for the knowledgeable user sometimes better results can be achieved by utilizing an equalizer before a noise reduction unit, but only in rare cases.

Echo Units and Reverberation Units: All signal processing should be done prior to the input of echo and reverberation devices, with exception of equalization unless some form of unusual effect is desired. Limiting can sometimes be handy at the input of a delay line to prevent overload problems should you encounter them.

Compressor/Limiters: These can be used in most places, the general exception being the echo or reverberation outputs.

It has been our purpose in this piece to give *general* guidelines for connecting and using auxiliary equipment. The subject is complex and the number of various situations that can arise could certainly number in the thousands. It would take many, many pages to cover all of the many combinations that one could encounter. The information contained herein is what this writer considers good and safe practice based upon personal recording experiences. The information contained in this article should be of great benefit to those who are starting their careers in recording.

As pointed out earlier, it is important for those who have not read the two-part article on the interfacing and connection of auxiliary equipment that appeared in the April and May 1979 issues of *MR* to do so at the earliest convenience. The information contained in those articles will add greatly to your understanding and will increase your ability to make excellent recordings in the future.

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MXR on the use of multiple effects.

The use of special effects has become a major part of today's music. Contemporary musicians are able to reach new levels of self-expression by combining instrument and effect in the development of their technique. If multiple effects are used, not only is this expression enhanced, but a new world of totally unique sounds is provided.

Although many of today's knowledgeable musicians use multiple effects, this technique has not been exploited to its fullest. One reason for this is that the artist usually has a musical rather than technical background, and is more familiar with artistic expression than the concepts of signal processing. At MXR we feel that our knowledge as designers may be very useful to the artist. The information here has been gathered by our engineering and marketing staffs through customer feedback, research, and personal use.

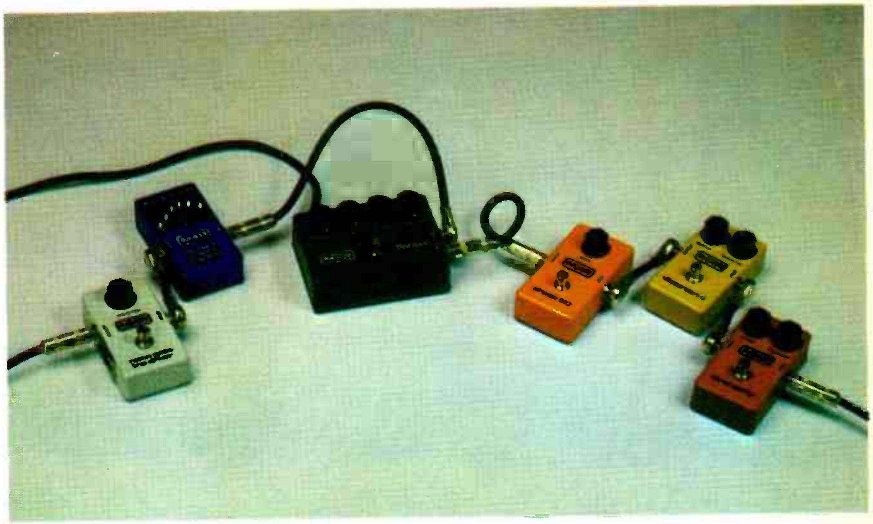
The purpose of this paper is to be a comprehensive yet understandable guide for the musician who is using or planning to use effects devices. We feel that MXR products provide well designed, legitimate effects that easily interface between your musical instrument and amplifier, whether employed singularly or in multiples. The following information, however, applies to any device of the same generic type. These devices are organized into groups according to their effect on the signal.

In a hypothetical situation, where all types of devices are used simultaneously, the following groups may be distinguished. Note that the order given is for the general case. Later we will discuss some alternatives and specific applications which you can try. The first group is composed of level boosting devices and includes preamps and distortion units. They are generally placed in the front of the effects chain because they are capable of providing a higher signal level for the successive devices to work with. The second group, the dynamics dependent devices, vary their effect in relation to the level dynamics of the incoming signal. This group includes envelope filters and envelope generators. Next are compressors and limiting devices which provide a more uniform signal level. Next are phase shifters and flangers which alter different frequencies in differing amounts. Delay lines, including tape delays, analog delays, digital delays, and doublers, are the next group. Noise control devices such as noise gates and noise filters would be next and would help to decrease the overall system noise. Equalizers belong in the last group and help to tailor the entire tonal character of the signal.

Let's examine a typical situation in more detail. It is beneficial to use a preamp or compression device such as our Dyna Comp first. The compression device sets up a more constant level for the next device, which might be a distortion device. Many additional harmonics are provided by the distortion device for later stages. Next a phase shifter would produce its characteristic frequency cancellation notches. If desired, a flanger would be used next to generate many more notches, especially at higher frequencies. Use a wah-wah pedal towards the end of the effects chain since it has the capabilities (in certain frequency ranges) of overdriving devices placed after it. If you are using a delay line, it should also appear near the end, regardless of what type it is (tape, analog, or digital), since it could potentially add noise to the signal. Use a volume pedal next. It can serve as a manual noise gate, a dynamics controller, or an envelope control. A Noise Gate/Line Driver, used at the end of the chain, can create a threshold between unwanted noise and the musical signal. It can also serve as a signal splitting device for remote recording or P.A. amplification. The last device is usually an equalizer. This contours the sound or may be used to compensate for the acoustical response problems at a club, auditorium, or studio.

The goal is to achieve a desired effect, impression, or sound, without losing control and without introducing unnecessary noise. An important idea to keep in mind is to make connections as short as practical, in the interest of a cleaner signal as well as neater physical organization. For instance, if you are using four different devices simultaneously, chances are it would be to your advantage to have them relatively close together. More noise can be introduced by the cables than by the circuitry within the devices. Most MXR effects devices have been designed so that when they are switched "out" the input and output jacks are effectively tied together to help maintain a clean signal when bypassed. The exceptions are our graphic equalizer pedals which are always "in" and our Noise Gate/Line Driver whose line driver function is always "in."

Buffering devices (preamps, compressors, etc.) establish a strong or uniform driving signal. In the beginning of a long chain (more than four devices), they can prevent the effects of "loading." The symptoms of loading are loss of highs and lows and the lowering of the "signal to noise ratio." The resulting sound is one of lost crispness. All MXR products have a relatively high input impedance so that more of them can be used with less loading effects.



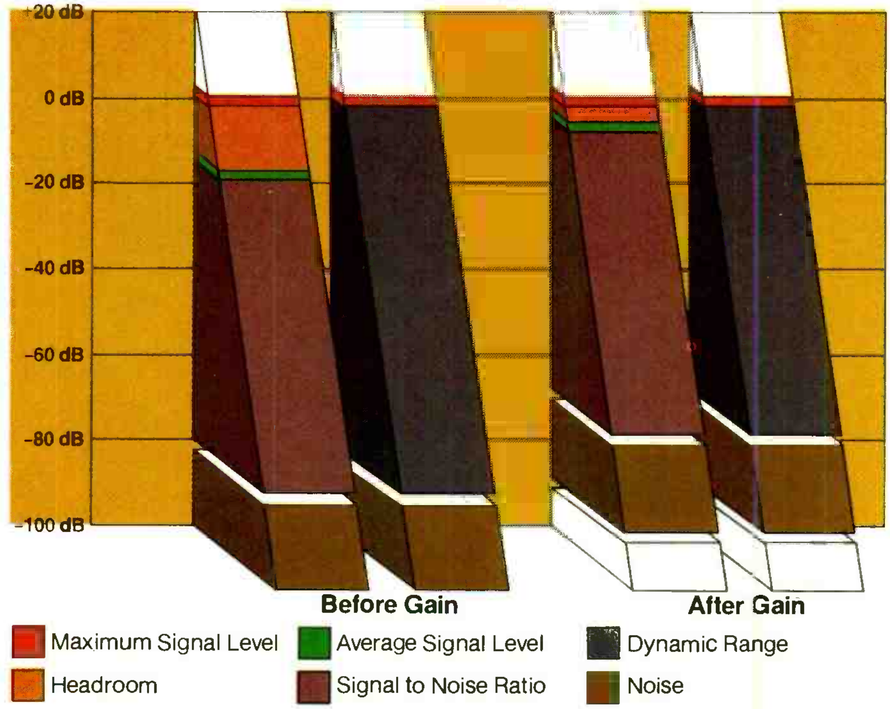
Whether on stage or in a recording studio, a proper arrangement of signal levels must be set at all places along the signal path. In order for any electronic device to operate at its optimum it must be used within the specific level range for which it was designed. MXR devices have been designed to accept a wide variety of input levels. These levels must be considered, however, since they can vary greatly, depending on where the signal is coming from. Keep the signal, at all points, below the clipping level and above the noise floor. Maintaining adequate "headroom" refers to having enough room at the signal peaks to ensure that they are not clipped or distorted by the unit. Maintaining a high "signal to noise ratio" refers to having enough signal level to prevent unwanted noise from being heard.

Until now we have been primarily concerned with dynamics and level, but frequency response is also important. It is generally not advised to limit the frequency response in the beginning of the effects chain. An exception to this is the use of an equalizer at the beginning in order to purposely alter the apparent tonality of a particular instrument. An example of this is a guitarist who wants to make his guitar with humbucking pickups sound as if it had single coil pickups. Equalizers can also be used at the end of the chain to filter out unwanted high and low frequency noise.

Something to consider when using several devices is the level difference which can result from the device being switched in and out. Usually there will be a gain or output level control on the unit which enables adjustment of the modified signal's level in relation to the original or dry signal. A recommended method for achieving uniformity between signals is to play a note or chord with the device "out" then switch it "in" and play the same note again. Rotate the output control to the place where the signals are in their proper relationship according to your needs. Remember that the first devices have the potential of overloading all the rest, so be aware.

Another factor to consider is the harmonic character of the material and how it affects subsequent signal processing. To illustrate a point, consider a situation in which we can use either a compressor or a distortion device to feed a flanger or phase shifter. Theoretically, a compressor should not change the harmonic character of the signal, while the distortion unit adds many harmonics. The thickened frequency spectrum of the distortion unit offers more signal for the flanger or phase shifter to act on. The result is a more intense sound.

A compressor or limiter should generally not be used at the end of a signal chain. By definition, a



compressor brings up the lower level signal to a predetermined level. If that signal is the noise floor, then that is what will be amplified. What will result is a high concentration of circuit noise. Even though most of the higher quality units have been designed for low noise, it is still in your best interest to keep this in mind. By using a compressor or limiter before a volume pedal an interesting technique can be developed. This technique is particularly appropriate for steel guitar and electric piano. The compressor boosts the softer notes, giving the instrument greater sustain, and the volume pedal permits the artist to fade the music in and out smoothly for a fluid and gliding sound. The artist is also afforded the benefit of always knowing what level the signal is at before going into the volume pedal.

If you are using a dynamics dependent device you would not want to preceed it with a limiter. These devices are dependent upon the unaltered level dynamics of the incoming signal to control their effect and therefore will seem more subdued if preceeded by any device which alters the dynamics of the signal.

One interesting thing to try with multiple effects would be splitting the signal and processing it through different effects chains and monitoring amplifiers. Another thing to try, which is especially effective with keyboards, is to have a phase shifter sweeping with a narrow width and fast speed, running into a flanger, sweeping with a wide width and slow speed.

Once again the general rule is to run gain devices ahead of effects devices. However, always consider what the desired effect is to be. Combining knowledge and imagination can certainly help musicians (and engineers) to get more out of their art and the most from their electronics.

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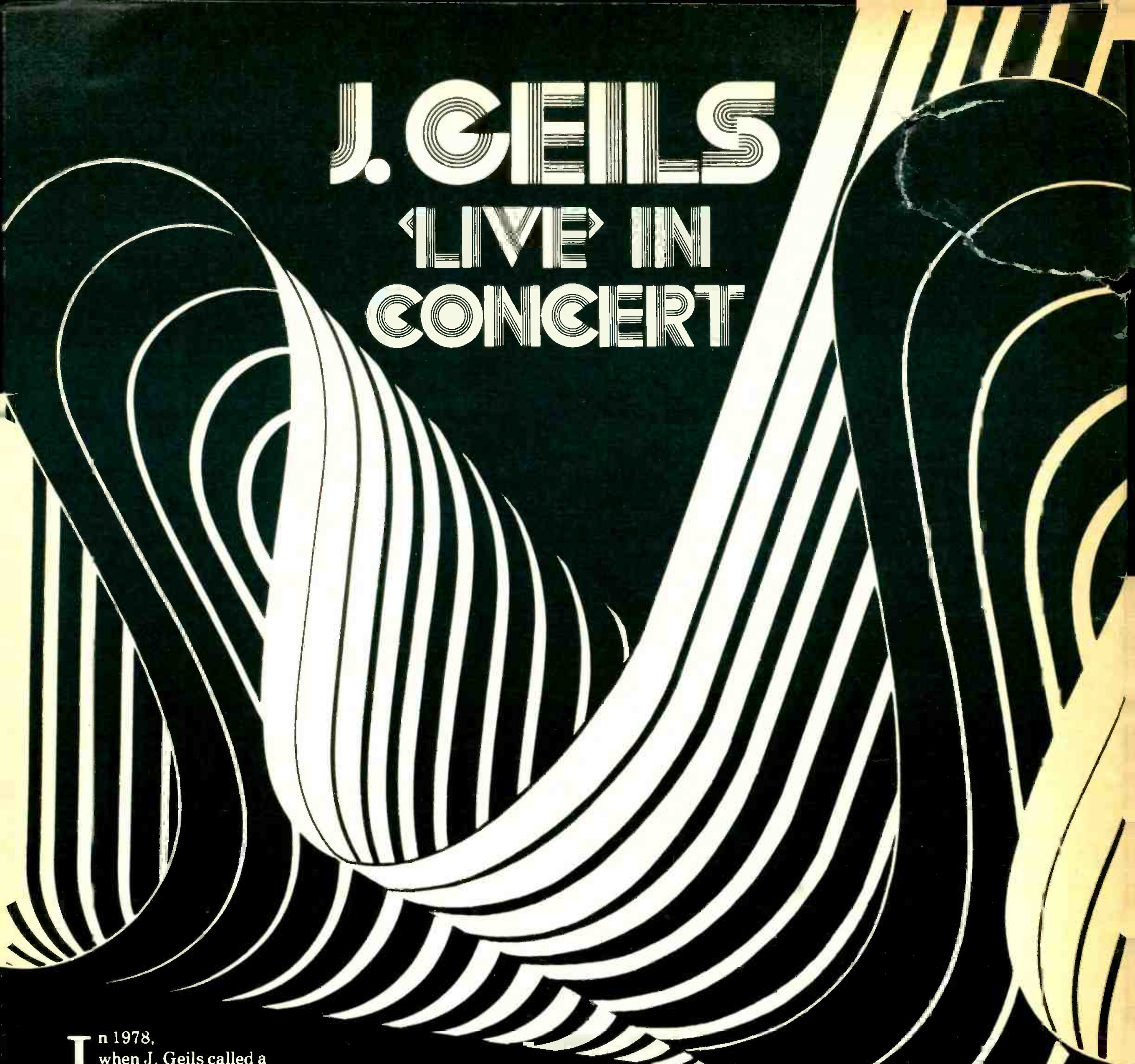


In 1978, when J. Geils called a band meeting to discuss the packaging and presentation of a tour, planning was divided into two stages and the problem was methodically conquered. The first consideration, how to rock the socks off guests at their moveable house party, was carefully produced and staged. It is a show which the band's chief engineer Frank Pavlich appraises as "evidence that Geils' [stage] approach is as surgical as producing an album and reveals a stage presence bordering on theater."

The second consideration, and that which is the subject of this article, was the design and operation of a quality sound reinforcement system. "We want something ominous," Magic Dick told Pavlich. "A wall of amps eight feet high and forty feet across."

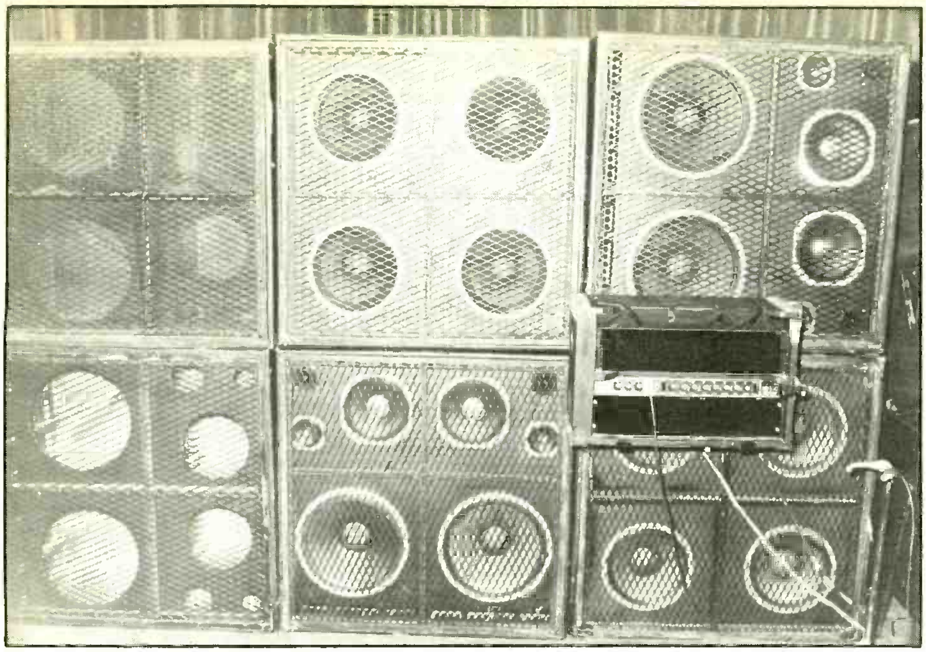
"A clean stage appearance," said keyboardist Seth, "perhaps with speaker stacks capable of being suspended from the ceiling."

"A custom-built system," added Wolf, "tailor-made to our specs. And make it uniform and interchangeable in case of emergencies."

"But first," concluded J., "I look at the purse."

The task of filling the order was offered to Audiofreqs Sound Services of Cincinnati. Design engineer Harold Blumberg spent days toying with the novel concept of providing an ominous wall of sound on a budget. To further complicate the dilemma was the factor that the Geils band in its youth had had a rather spotty regional following. The sound system would have to meet the demands of 18,000 seat hockey rinks on the band's home turf in the Northeast as well as function optimally before 250 students in Ivy League school gymnasiums. Basically, Blumberg would have to struggle with a design which could transform roller rinks into living rooms and preserve aesthetic plea-

By Murray M. Silver, Jr.



Magic Dick's and J. Geils' drum and guitar stage monitor stacks.

sures in smaller venues without blowing out the walls. The whole concept drastically reversed the procedures of Harold's last project—mixing monitors for Audio Analyst of Montreal for the Emerson, Lake and Palmer world tour which included several dates performed with symphonies.

As Pavlich explains, Blumberg and his two-man crew are the sole reason this particular system works. Blumberg's aim is to allow the band to play on stage in a manner befitting their lifestyle—on the edge. It's a street band playing mostly for street people. 110 dBs in the shade. And it has to run like a Swiss watch because, "Hey," says Geils, "we don't wanna have to spend suppertime jerking around in a sound check. . . ."

The first thing that observers ask Harold about his command post is, "Where's your real time analyzer?"

"I'm wearing it," Harold replies jokingly, pointing to his ears. "For J. Geils we don't need to do sound checks," he explains. "Before the tour started we spent four days in rehearsal conducting the original sound checks; it doesn't make any difference where we are because the only variables I have are volume and EQ. Prior to the show I am only concerned that the system works, and I can ascertain that by playing a cassette. Once the band begins, I start with the system tuned basically flat and just go from there."

The sound system functions at 200 amps in each of three phases and the lights are geared at 600 amps per stage. The bare minimum level of AC power can be half of the normal supply before some of the 126 lights have to be cut off.

The mixing consoles from which Blumberg operates are two modified Interface Series 100s by Stevenson. From the factory, Blumberg modifies the tone and gain controls and substitutes a low noise, high slew variety of I.C.s. When finished, the Stevenson board provides a great mix for a tailored budget.

Compressors are inserted between submasters, and mixdown and the entire production is creatively compressed. The main board handles vocals, guitars and keyboards and receives a pre-mixed signal from the second Stevenson board carrying the rhythm section into its submaster 4. The bass passes through a UREI 1176 LN Peak Limiter; the drums through two Allison Gain Brains (for kick drum and snare) and an Ashly Audio SC-50 Peak Limiter/Compressor (toms-toms) before returning to mixdown in the main board. Two mixing boards are required—just the toms alone rate nine inputs on the rhythm console.

From the main board, Peter Wolf's vocals are treated by another UREI 1176 LN Peak Limiter and a DeltaLab Research DL-2 Acousticomputer which is a unit for doubling, reverb, echo and chorusing. A Delta-Lab Digital Delay Module DL-1 is also used in conjunction. J. Geil's guitar, Magic Dick's harmonica and Seth Justman's keyboards are each processed through individual Ashly Audio SC-50s (parametric peak limiter-compressors), and on occasion the drums are treated by an MXR Digital Delay.

Once each submaster is properly channeled into its individual compressor, the entire mix is filtered through the system's Gain Brain limiter. The system is equalized by the UREI Model 527 1/3-octave equalizer. The normally used crossover is a hybrid piece incorporating a Spectra Sonics unit and Automated Processes line amps. This custom piece is backed up by a Biamp M2/V electronic crossover.

Blumberg's Babies

The speaker stacks are the pride and joy of Blumberg's production. The design showcases the Community Light and Sound Leviathan—a 48-cycle flared fiberglass unit loaded with JBL 2205As. There are two drivers per channel and each speaker receives 210 watts of power. Depending on the arena, the system can be suspended from the ceiling, especially for the dome-shaped, high-ceilinged halls. A small stack of four-way speaker cabinets custom-made by Audiofreqs are utilized for front fill.

The system is housed in galvanized steel "Uni-struts" which are lightweight and built on rollers for easy mobilization and storage. The design allows for each unit to be aimed in any direction within its mount as provided by Roy Bickel of Arena Rig Tech, Ltd. For the half-house configurations or more intimate venues, the system remains grounded minus two Leviathans. For large halls, there are six Leviathans per side.

Completing the stack are from four to as many as eight mid "Zoids"—a 90 degree horn—each containing a JBL 2202A midrange cone transducer. These are normally situated between the stacks of Leviathans and under eight Community Light and Sound RH-60 Zoids, each containing six JBL 2482 drivers. The highs are projected by three groups of four Community Light and Sound LRH horns, each with a JBL 2420 driver. The horns are powered by Uni-Sync model 100 amps.

The stacks are powered by Phase Linear 400 amps, one rack for low-frequency, one for mids and highs. All bass, low-mids, and highs of each amp have their inputs assigned through a thumbwheel switch to a nine-channel snake, resulting in a completely interchangeable system which can be regulated either from the stage or by Harold from out front.

The Ears Have It

As mentioned earlier, Blumberg and Geils have dispensed with the sound check procedures with which most of us are familiar. When the system was originally assembled, however, Harold used every 1/3-octave analyzer he could find on it.

"Most analyzers aren't as reliable as our ears," says Blumberg. "When we tuned the cabinets, of the alternatives between a logarithmic or linear sweep, we used a log sweep generator on each cabinet set up outdoors. By using nature's own anechoic chamber, we fed the speakers with a log sweep where the output is picked up by a calibration mic synched to a scope. Instead of getting a 1/3-octave readout, we got an infinite readout."

"For example," Harold continues, "the vocal monitors with JBL K120 speakers on a 1/3-octave analyzer will show a 3 dB bump in three places; log sweep will show you that in fact there are narrow 10 to 15 dB spikes, so the log sweep is much more accurate."

Sound checks were conducted on the system before the tour began and for the first two shows. Experience has carried Blumberg the rest of the way. Now, Harold will settle for a brief listen to an Oscar Peterson tape and a sound check in the latter part of the tour would only be necessary should there be a change in the show or a new cue from the band.

Preventing feedback was a problem that was taken into account in the system's design by allowing for headroom. Harold firmly believes that the right way to solve a problem is acoustically, not electronically, and therefore all the cabinets are first tuned as flat as possible and then EQed to remedy feedback.

The reason 1/3-octave analyzers have not been used thus far is because Harold claims that most of them are too "slow" to be used in "live" situations. However, he does favor some of the newer units (i.e., White and Ivie), and eventually plans on purchasing one to make mixing eight individual monitors easier.

Of Monitors and Mixes

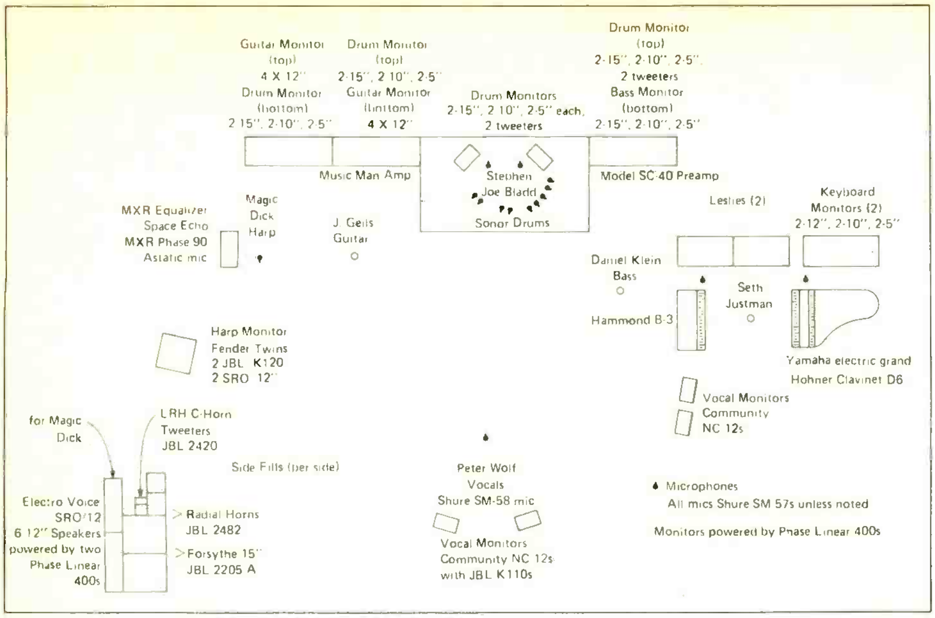
When it comes to monitors, Blumberg takes a pragmatic approach. A 1/3-octave analyzer will not always show where a monitor is going to feedback during a show. Consequently, the Geils band, being hard rock and rollers, is best served by *making* the system feedback and then stopping it. Adjustments are then made for tone.

Signal sources from the stage are processed through two Stevenson consoles (one 16×8 and one 16×4). Eight Phase Linear 400s power the stage and EQ is provided by six UREI Model 527 1/3-octave equalizers. There are eight stage monitor mixes.

The first mix of the stage monitors is a three-way wash. Each stack contains two Forsythe SR 115 horns with JBL 2205A speakers. From the board, the signal is processed through an Ashly Audio SC-66 parametric equalizer into the UREI units before passing into an Ashly Audio SC-50 limiter compressor and the Biamp crossover. It crosses over at 600 cycles to Community RH60 Zoid horns and JBL 2482 drivers. At 5000 cycles, it crosses over to LRH horns with JBL 2420 drivers. The bass and mids are powered by Phase Linear 400s; the highs by Uni-Sync 100s. The second mix is Peter Wolf's slant monitors—four Community Light and Sound NC12s with JBL K120s.

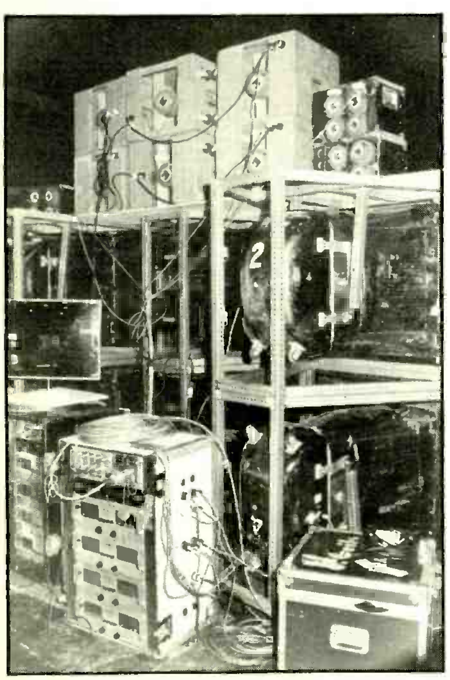
The next mix is for the drums. As Stephen Joe Bladd sits behind his Sonor drum kit, he is being monitored by two cabinets containing two 15-inch, two 10-inch, two 5-inch speakers and two tweeters on either side. They are housed in a short-throw cabinet that Blumberg designed. The whole stage serves as a drum wash because J. Geils is a drum-driven band. The bass end is contained in a Thiele-aligned cabinet containing its own JBL 2205s, two JBL K110 loudspeakers and two JBL 2105 5-inch

Harold Blumberg (left) of Audiofreqs at the Stevenson stage monitor console.



speakers. The system is capable of being tri-amped with a 6 dB-per-octave passive crossover designed in conjunction with the log sweep.

At stage left, harpist Magic Dick stands before a guitar monitor stacked atop a drum monitor each containing four 12-inch speakers. Dick plays harp into an old Astatic mic preamped by a 25-year-old Fisher mono tube preamp. (This preamp contains "photo-equalization" allowing a tonal quality unobtainable with modern solid-state equipment.) His signal then goes through a MXR graphic EQ and a Roland Space Echo. At a 45-degree angle to his right, Dick listens through two Fender Twin



Galvanized Uni-struts holding Community Light and Sound Leviathan horns.

Reverb amplifiers, one with two JBL K120s and one with two Electro-Voice SRO 12-inch speakers. Dick has his own side-wash monitor consisting of six SRO 12s per column powered by Phase Linear 400 amplifiers.

Axeman J. Geils stands near Magic Dick before his own drum monitor stacked over his guitar monitor. Both are similarly equipped with four 12-inch speakers. This system constantly is played on the verge of feedback. Geils plays through four speakers, one of which is behind the stage backdrop miked for the P.A. system. Bassist Daniel Klein stands at stage right before his own drum monitor and his bass monitor preamped with an Ashly Audio SC-40 instrument preamp.

Projected into the center of stage right stands the keyboard complex over which Seth Justman presides. His setup includes a Hammond B3 organ, a Yamaha Electric grand and a Hohner D6 Clavinet. Two Leslie speakers —with JBL 2482 drivers in place of the standard units—stand at service alongside his two keyboard monitors which are equipped with two 15-inch, two 10-inch, two 5-inch speakers and two tweeters.

The miking of the drums and keyboards is designed to prevent, not create, problems. There is a Shure SM57 mounted inside each Sonor drum three inches from the heads. Each mic is shock mounted to resist rumble. This position is the only one which has been found to minimize bleed, and the separation that is achieved is far more valuable than any resulting rumble. The entire sound system is tuned to the SM57s because Harold considers them to be a tight rock and roll drum microphone. An AKG D12 is used on the kick drum, however.

Gadget feedback on stage is a restricted problem because aside from Magic Dick's toys, only J. Geils uses special effects. Geils uses a Magic Man amplifier and a box he built which contains a distortion unit, a phaser and a Harmonizer.

In mixing sound for J. Geils, Blumberg cautions that it is easy to over-compress and squash everything into a lifeless, dull and muddy mix. Equally tragic is the possibility of over-compressing to create too much volume.

"The problem is that many soundmen don't know how to mix down," says Harold. "They don't hear one part against another."

"Many soundmen mix up to higher levels. If you need more guitar volume, don't turn everything else *up* to meet it, turn the organ *down*. I don't always mix by volume, I mix by EQ."

Although the key to the design of the J. Geils stage is simplicity, it still requires four good roadies in addition to the crew of eleven to have the show ready three and one-half hours after the trucks drive into the gate.

"Perhaps the only fault with the entire production as a whole," concludes Harold, "is the packaging. The only area that could stand improvement is the amount of time and work it takes to set up and tear down the system. However, with our Uni-struts we've cut the time in half."

. . .And They're On the Road Again

As a post script to this article, during a subsequent date at the Warehouse in New Orleans, a promoter's employee attempted to help Harold and his crew tear down the system in record speed. However, the technician failed to shut off the AC power before attempting to disconnect the system and after a near brush with death, the young man recovered to discover the total destruction of all of Harold's Phase Linear 400s. For the first time in ten years, the cancellation of a date was forced while Harold called around to find power for the Atlanta date.

"Yes, I would have liked some new equipment," says Harold, "but I would have preferred to have to make the replacements in a more conventional manner."

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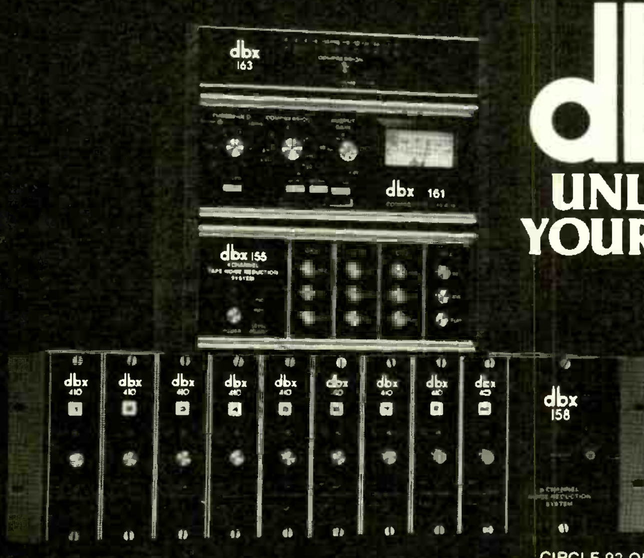

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UNLOCK YOUR EARS

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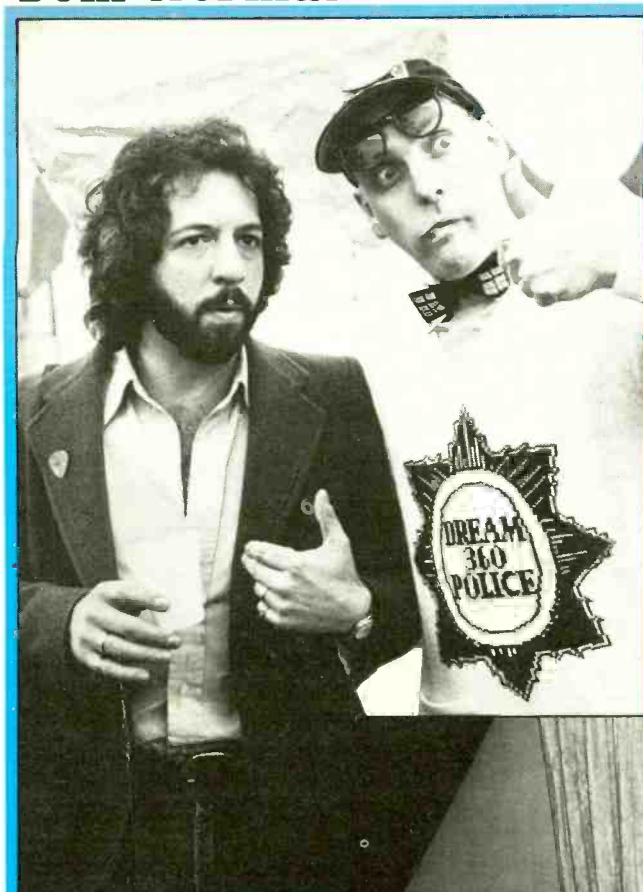
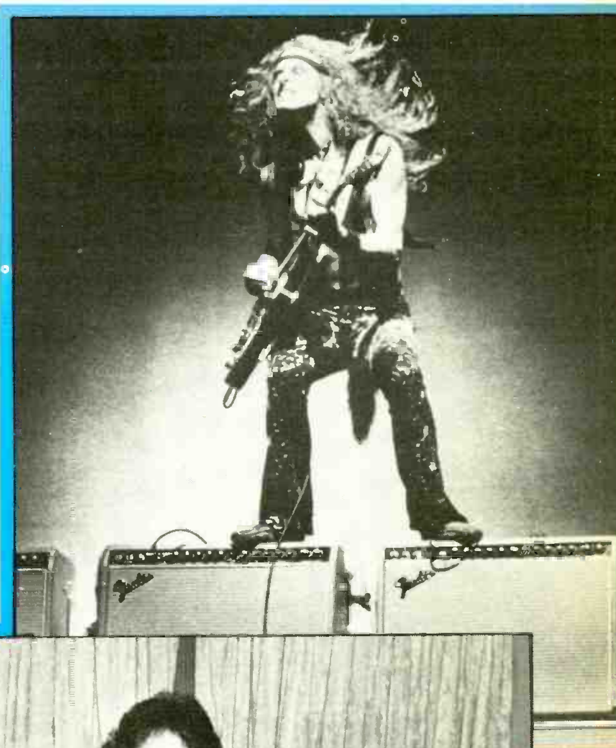
Profile of Cheap Trick's Producer

Tom Werman

By Nina Stern





Producer Tom Werman has been equally successful in acquiring talent as he has been in recording it. Having joined CBS subsidiary Epic Records in the early 1970s, Werman has been involved in bringing such notable acts as Ted Nugent, Cheap Trick and Boston to the label. He served as co-producer on Nugent's first five albums on Epic but recently has left the team currently working on Nugent's sixth LP. Werman has solo-produced the last three studio albums recorded by Cheap Trick, "discovered" by him in a bar in Quincy, Illinois. He has also produced albums by Mother's Finest, the Jeff Beck/Jan Hammer group [executive producer], Brownsville and Molly Hatchet. At the time of this interview, Werman was in the studio working with Blue Oyster Cult on a new album.

MR: As an A&R man at Epic, you were involved in signing some of the biggest acts in recording today. How for instance did the involvement with Boston occur?

TW: Well, Lenny Petze, my boss (then vice president of A&R for Epic Records) had just joined the A&R department. He had an appointment with Paul Ahern, Boston's manager, whom he had known from when they were both promoting records in Boston. He heard a tape and about half an hour later he brought it down to my office and said, "Tom, I'd like you to hear something."

We then both went down to Steve Popovich's office, Steve was our boss, and we played the demo—the Boston demo. After the third song, I said, "We really don't need to hear more."

MR: Was Ahern shopping for labels at the time? Was yours the first he went to?

TW: *No way*, a lot of people passed on Boston, which is hard for me to believe [For additional information see *MR's* July 1979 issue: "Profile: Boston's Tom Scholz.]. We told Ahern that he had a deal for sure . . . that we wanted the band . . . and I expressed an interest in producing the band.

MR: What an incredible decision to have been a part of.

TW: Well, I guess, but actually that's where my involvement ended. Lenny and I did go up Thanksgiving vacation that year and see them at the Warehouse, which was Aerosmith's rehearsal hall in Waltham, Massachusetts. I remember, I had just done Ted's (Nugent) first album, but after that, Boston became *his* [Petze's] band. But it means much more to me to have signed Ted and Cheap Trick.

MR: How did you become involved with Ted?

TW: His manager at the time, Lew Futterman, started badgering me. Every few days he'd call up and say, "Ted Nugent is available," and I'd say, "Okay, great, what else is new? How are the kids?"

MR: Was it Nugent *and* the Amboy Dukes at the time?

TW: Yes. Finally, I gave in and said, "okay." I'd never heard anything by this guy, but I had heard his name. That was it. I went to Chicago and heard him play at the Illinois Institute of Technology . . . to about 400 people in a hall that could seat 1000. And I went berserk! It was ridiculous that that guy wasn't making records. He was *wonderful*, extremely charismatic. All he had to do was walk on the stage and you said, "Whoops—this is without a doubt the most important man in at least a ten block area."

Anyway, he was a wonderful guitar player that night and I loved the way he presented himself. I wanted to sign him, so we talked it over after the show, and I said, "I'm going to recommend that we make a deal right away." Then it turned into a production deal which I wanted to have a hand in and Lew said "fine."

MR: Was that your first album?

TW: Yes, and also the first Epic Ted Nugent album. Which I still think is the best of his albums.

MR: Did his association with the Amboy Dukes end when you signed him?

TW: We decided to drop the name because he was too important. It was like a break with the past. So I started to make records with Lew and occasionally with Cliff Davies, Ted's drummer, who serves as musical director and rehearser for the band.

Ted is really not interested in spending long hours doing anything but pursuing the finer things in life, among which he does not happen to include rehearsing.

MR: Do you feel there has been a progression in Ted Nugent's music from *Ted Nugent* in 1975 to *Weekend Warriors* (the last one Werman co-produced)? How do you see his music progressing, or, how would you like to see it progress?

TW: I think—I'm not sure it's been progressing—there has been a change. There's been some kind of evolution. The sound on the first album was rather naked and basic, but it was aggressive and tough—and raunchy. Not dirty, but raunchy. That's how I wanted to record the band at the time. Cliff's approach to recording differs significantly from mine where rock and roll is concerned. Ted's sound now is more polished, it's more "glassy"; it's not as brutal. I think that Ted Nugent's music should retain a more offensive stance.

MR: Are you talking about it conceptually or technically? How has it become more polished?

TW: Concept and technicality go hand in hand. Cliff is against the idea of doubling guitars, or layering. He doesn't like it, just as a rule. I don't have any rules. In Ted's case, I felt it was always very effective to layer or double guitars. Double or sometimes triple guitars, because he really doesn't give you much of an opportunity to fine tune his instruments in the studio. He goes in and throws everything on ten and plays. Ted doesn't bring a huge array of amps, and plays one guitar and one guitar only . . . and it's the "wrong" one. I think he used a Les Paul on one or two songs on the last album.

MR: What [type of guitar] does he usually play?

TW: A [Gibson] Byrdland, forever and always. He can get what he wants on stage from that and he can get certain feedback things on solos, but I have always maintained that it's the wrong guitar for him to be using on rhythm tracks.

MR: So he plays the same guitar for lead, rhythm . . .

TW: Yes. He would like to and he does wherever he can. I will walk into the studio and say, "Will you please try this amp and please try this guitar," and he'll say, "All right," and pluck a few perfunctory notes and then say, "Oh, that sucks."

MR: I read that he plays all the guitars on his albums. Does that also include bass guitar?

TW: Yes, on certain songs.

MR: Has there been an album where he has done *all* the guitar work?

TW: Yes. Most of the guitars on *Weekend Warriors* and actually most of the guitars on every album.

There's a song called "Workin' Hard, Playin' Hard" on *Cat Scratch Fever*, and hardly anyone I know could keep up a rhythm like that for an entire song. *And*, he *doubled* it, really well, too. I'd like to see him do more stuff like that—that's a weird song. It's got a very strange sound.

I like to treat each sound differently, to serve the material. A producer should serve the material, not get a drum sound and say, "Well, there's the drum sound for the entire album. Boy, it's great, isn't it?" No, it's not great—it's great for *this* song. The same holds true with every instrument in every project. I'm always suspicious when I start to do things the same as I did on the last album, or on somebody else's album.

MR: How long does it take to record Ted's albums? Does he go in well prepared so you just lay down the tracks and spend most of your time mixing?

TW: He goes in well prepared, but it just doesn't go that fast. His albums took an average of four to five weeks to record because a lot of tracks are done over again; it was not a very well-organized recording process, not what I would do alone. The way I record other groups differs totally from the way Ted Nugent albums are made.

MR: Is his studio band his road band, as well?

TW: Yes.

MR: Don't they spend time rehearsing before [recording]?

TW: They do rehearse. Their sound checks are rehearsals on the road. But Ted can only play one guitar on a sound check. He's never had a chance to try out all the other parts that he has in his head in the studio. He can never take a day off really and go into a demo studio.

MR: Does he ever write anything down or is it all in his head?

TW: He only writes lyrics down. The rest is in his head. He has an infinite number of guitar lines and parts and screams and wows and things like that in his head. He has an excellent mind. In the beginning, I would say, "Why don't you do this," and it took me two albums to learn to wait till he was done. I'd make a suggestion and he'd say, "Yes, okay Tom, but wait a minute and let me show you what's coming, what else is happening" and it would always work pretty well.

The way his records are made has little bearing on how I prefer to make records. In fact, when we made the first record I was shocked and dismayed. What we wound up doing very often was have Ted go into the studio and play the rhythm guitar to a "click track" (an electric metronome in his earphones), just for time, as if he were playing with a drummer. Then, we'd overdub the drums and the bass to the rhythm guitar and then we'd re-do the rhythm guitar, because it was a guide. We'd re-do that and then we'd start getting down. And that would sometimes take two whole days, to get one track. Whereas with a band like Cheap Trick, you might take just a few hours setting up sound.

MR: Why the choice then to have him do all the guitar parts on the album, even though he has a complete touring band?

TW: 'Cause they can't play it the way he wants it.

MR: They only learn the songs once they've been recorded?

TW: Yes. Ted Nugent's guitar lines *should* be played by Ted Nugent. They're not just lines and notes and charts, they sound different from other people's guitar parts. They have twists and bends and tails, they're not just "E string, 4th fret." Ted ignores frets. He's really a marvelous guitar player. . . .

MR: But if you were sole producer, you would take him where now?

TW: Oh boy . . . I would make him spend more time in the studio. The records would sound different. The drums, for instance, would sound completely different, "meatier," probably louder. The whole mix would be a completely different kind of mix. Cliff never cared for any of the mixes Lew and I did. So I finally said, "Well, gee, be my guest." Absolutely no hard feelings. I love Cliff. We just don't agree in all ways on how to make a record. He's a very capable producer, but I don't want to make records with him anymore. It just got to the point where I was asked to do somebody else and I said, "How can I give up Ted Nugent? He's my artist, my baby. It would be the end of an era." Then I realized if I worked with another band, *I* would be the sole producer.

I now feel I would never do a co-production again and I am wasted sitting in the studio with Ted at this point. Whereas with Blue Oyster Cult (Werman's current project) it's a consuming challenge every single day. This is a hard album for me, because these are guys who've made seven records with the same production team for six years. I'm working with them for the first time and I hardly ever saw them before "live." I admire them and I respect them and I've grown to love them in the last few weeks. They're very nice, extremely intelligent people. But, it takes a few albums for a producer to learn how to exploit the strengths and tone down the weaknesses in a band.

MR: Where have you been working on the Blue Oyster Cult project?

"... I'm just beginning to feel I can call myself a 'Producer' ..."

TW: We did the tracks at Kendun Recorders, and that turned out to be a good experience. I wanted to do them at the Record Plant [L.A.], 'cause I really like that room there. . . .

MR: Which room?

TW: Room C. It's very large and you can use it for the drums, while doing a "live" track. You just put the bass in one isolation booth and the guitar in another isolation booth. And you save whatever else till later. At least you can get a three-piece track down with exactly the kind of sound you want. Then you can replace the guitar later if you want but you end up getting exactly what you wanted out of the drums and the bass. We moved to the Record Plant [L.A.] for overdubs this week, and next week we go to New York because they want to go home for a while. So we'll do two weeks in N.Y. If we're not done by then, I have the option to drag whoever has to work back here (Los Angeles), whoever is not finished.

MR: When did you move to L.A. from New York? Just recently?

TW: Last summer.

MR: I noticed in going over past albums you've produced that you worked a lot at the Sound Pit in Atlanta. Tell me something about that.

TW: We did the first three Nugent albums there, I guess, and I did Molly Hatchet there and mixed it at the Record Plant. And I did two Mother's Finest albums there, too. I guess I did about five or six albums in part or wholly there. Mainly because I love the engineer who was in charge.

MR: Who was that?

TW: Tony Reale. He did the Cheap Trick *In Color* album also. When I moved out here, I got in touch with Gary Ladinsky, who has become my crutch, so to speak.

MR: Is he staff at the Record Plant?

TW: He's not staff. He's attached to the Record Plant, but he can work anywhere. He and I have an almost perfect working relationship in the studio. We get along. In fact, we do a considerable amount of laughing—which is very important. We understand each other's neuroses. I'm not an easy person to get along with. I'm not one of these "closed session" producers—I like people to come in. Anyway, I don't want to make any more records *without* Gary. I think he is an absolutely wonderful engineer.

MR: He was mixing engineer on *Cheap Trick at Budokan*, wasn't he?

TW: Gary *re*-mixed it. The original mix was overall very good, but it did not at all reflect what was going on on stage. The guitar balance was wrong and things like that. Because of deadlines, Gary came in and re-mixed the better part of the album in three days.

MR: *Cheap Trick at Budokan* has an incredible history, doesn't it?

TW: Yes, it has. Originally, it was scheduled for release in Japan only. But the demand for the import became so great here, that the record company had to protect itself by releasing it in the U.S. too. It's nice to make an album that obviously so many people want; it looks like it will be a platinum seller. It has already surpassed *Heaven Tonight* (the previous studio album) in record sales, and it hasn't been out very long. There were a couple of weeks where it was selling over 100,000 per week. Meanwhile, I'm considering making new parts for *Dream Police* (their latest Werman-produced studio album, as yet unreleased) because I don't want them sitting around for four months. I'm not sure if lacquers can survive four months of

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CIRCLE 101 ON READER SERVICE CARD

sitting. Because now *Dream Police* is supposed to be released in about a month, but who knows?

MR: What is your title now? You are no longer A&R, you're strictly a staff producer at Epic, right?

TW: Right.

MR: Did you ever think you'd end up as a producer?

TW: Yes. . . .

MR: Even without the technical background, or did you just pick it up along the way?

TW: I have very little technical knowledge. I just thought I could make better records than the ones I was hearing when I was growing up. The longer I spent in the music business, the more obvious it became to me that I *could* make records. It was just a question of waiting for the right time. And now—I have made thirteen records—I'm just beginning to feel I can call myself a "Producer."

MR: Do you have a philosophy from group to group about recording or producing, or does it change depending on whom you're working with? Like, "live versus recorded" sound.

TW: There are very few rules. The only philosophy I have is that a good producer should never be heard. If Phil Spector were a producer today, I think he would be going about it the wrong way . . . he should be an artist.

MR: You said that Gary Ladinsky is your crutch. How does that relate when you're in the studio?

TW: Well, I'm not interested in being able to take apart and re-wire a recording studio. I'm not interested in sine waves, and I'm really not interested in why a ribbon mic is different from a condenser mic. I'm interested in the music—the arrangements, the rhythm, of course, the artist, anything that has to do with the music. I do feel it's necessary for a producer to have a full understanding of what tools in the studio can do, how to use a Marshall Time Modulator or any Eventide instrument or a phaser but that doesn't mean I'm going to use them myself. If I tell Gary, "Listen, can we open this up; can we space it out, just kind of crispen it up, with a little space underneath?" You know, things that other people wouldn't understand, he understands what I want really well. Sometimes, he'll argue with me or he'll just come up on his own with something. I prefer to be able to get up and walk around the room, have a drink, go lie down in the corner and shut my eyes. You can't do that if you're engineering, you just can't. I used to think it would be advantageous for me to go to engineering school and know all about engineering and how everything worked—everything. But I can edit as well as any engineer, except for Gary. And I know what all the equipment is.

MR: What are some of your equipment preferences?

TW: My main preference is the EMT 250, it's the ultimate echo unit. There are many in town now. It's the cleanest echo available. I guess it's called a "reverb unit." It's got a variety of settings at a variety of lengths and it can also act as a delay line. It does different things to different instruments—one thing to the snare, another to the bass drum. You can use EMT delay on a bass drum, which sounds silly but it's not.

"... a good producer should never be heard ..."

MR: Where have you used it on *Heaven Tonight* or *Dream Police*?

TW: On the drums, on some keyboards, certain guitars that I would want a lot of even sustain on . . . powerful, long sustains.

I also like distant miking of guitars. I don't always like to rely on echo or delay. I like real-time delay, so the sound actually reaches the far mike a little later than it reaches the near mike. Which gives you a sound on the guitar that you cannot get from using a digital delay line. You just can't. If you sit in the control room while somebody in the studio is playing the guitar really loud, there are times when someone will come into the control room and open the door and you'll hear that perfect blend between the distant roaring guitar and the monitor in the control room. And you'll say, "Wow, that's it! How can I get that sound?" I've come close once or twice. I'm always throwing mics around. I'll say, "Gary, would you please open that hall door and put a mic way at the end of it?" He makes a face, but one out of five times we'll get something special.

MR: What about the new [3M] digital tape machine? They have one at the Record Plant now.

TW: Yes, they do. I haven't had a demonstration yet. Somebody told me that apparently you can't edit the tape, and if you can't do that it would be pretty difficult for me to use it all the time. I really have to go in and hear it. I also think it's not going to be enough just to digitally record and then play back on conventional systems. I think the sound of radio has been fine-tuned to take magnetically recorded tape—analog tape—and I think radio's equipment wouldn't have any idea how to deal with a digital recording, you know. That's a problem, because I want my records to sound good on the radio *first*. Often my records will sound worse, considerably worse, in someone's house than they do on the radio, and that's a problem. Some of them sound really good at home and on the radio. Cheap Trick records tend to sound good anywhere, others of my records tend to sound really great on the radio.

MR: Is that because of the kind of monitors you're using?

TW: Yes, I think so. I like to use "real-world" monitors. I use Auratones, universally, right from the beginning. Not because I think they're necessarily the best on the market, but because they represent the average. They're a very good small speaker. I can understand the bottom on Auratones, and I've heard others where I can't feel the bottom end.

MR: Getting back to your artists. How did your relationship with Cheap Trick begin?

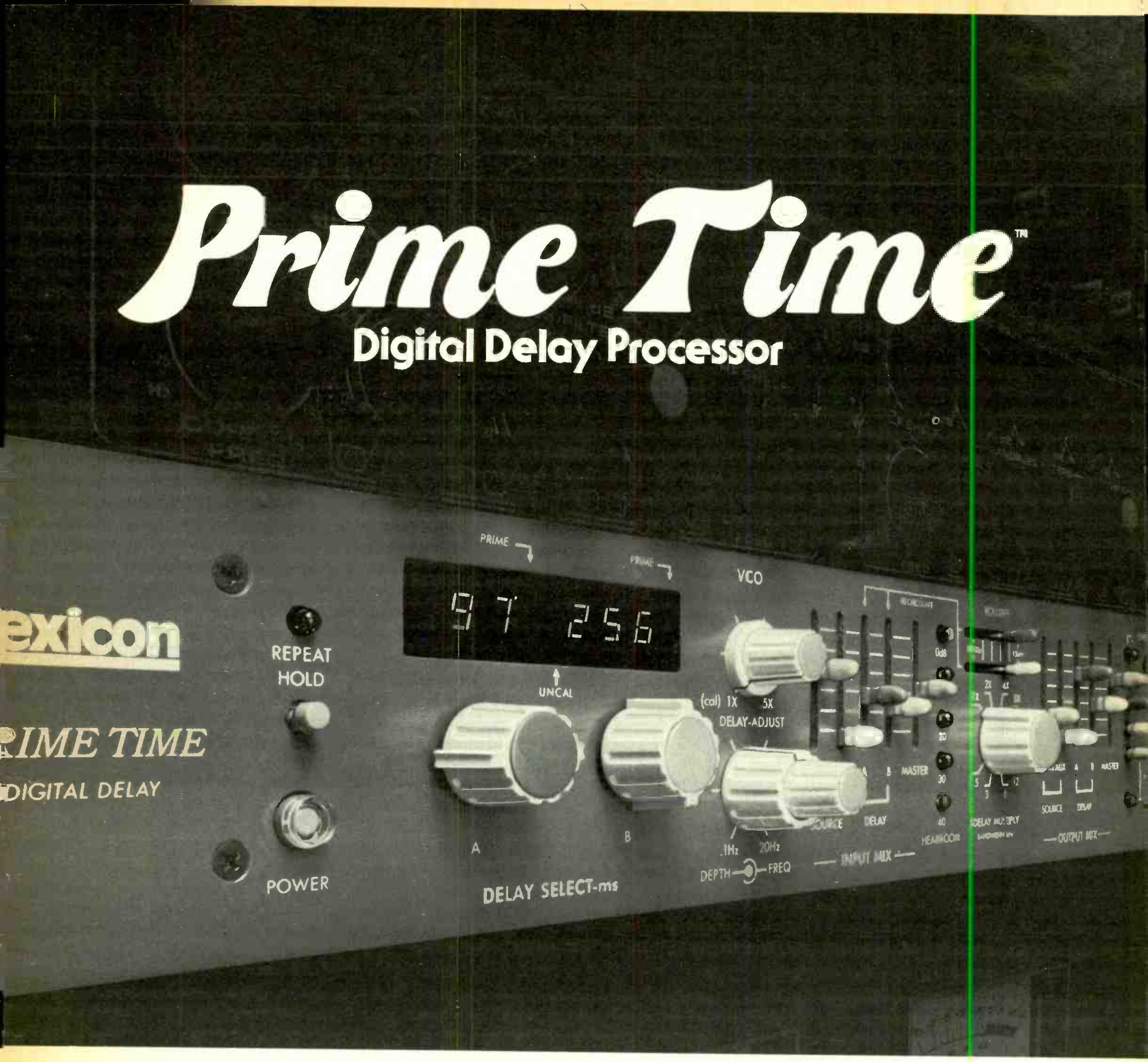
TW: Producer Jack Douglas (Aerosmith, Frankie Miller) called me up and said, "I'd like you to see a group." I went because he was Jack Douglas. I thought he was wonderful. Right at that moment, I was very sensitive to Jack because he had made the *Rocks* album, which was, as far as I'm concerned, Aerosmith's best. One of the best rock and roll albums of that year. So I went out to see Cheap Trick in one of those shopping mall clubs which seated about 100.

MR: Where was this?

TW: Quincy, Illinois. They were dreadful. Actually, I should say, *they* were great, but the *sound* was horrendous. Very loud. I couldn't judge the material, but they looked good and there was obviously something special about the band. I kind of begged off; I didn't make a decision that night.

MR: Were they already signed to a label at that time?

TW: No. I went back to New York, but a couple of days later, I said, "I've



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got to see that band again. I really miss that band!" It was an emotional decision, not a professional one. So, I went back with Steve Popovich to another club in Illinois—on the shopping mall circuit. That was a better situation for them, the sound was better. It was a better showcase place and the audience went completely berserk.

MR: It was a club?

TW: A little club. And we signed them then. Jack made the first record, *Cheap Trick*. Then Jack wasn't available for the second album. For some reason, he was busy, which happens all too much in the producing business. So, they asked me to do the album. I was about to come out to California to do the Eddie Money album. At the last minute, Epic said, "We're sorry, we need you to do Cheap Trick." I had a couple of days to listen to the material. As always, they never give me more than a couple of days. That album was *In Color*.

MR: I've been listening to "Budokan" for the last couple of months, and then I got *In Color* just a few days ago. I love the version of "I Want You To Want Me" on *In Color*. It's different. The timing is so good in the version on the studio album.

TW: Yes, that's a very interesting thing. A lot of people are going to get introduced to Cheap Trick via their "live" album and they're going to go out and buy their catalogue and I think be very pleasantly surprised.

MR: Oh, it's nothing but improvements. The music is good, so the "live" album is good, but the studio albums are so crisp and so clear—there's so much more on them.

TW: That's nice to hear, because I think that *In Color* is a pretty primitive album as far as production goes. And *Heaven Tonight* represents a leap towards the polished, which is as far as I want to go. *Dream Police* is not more polished than *Heaven Tonight*.

MR: How does Cheap Trick approach recording? You said they are the optimum situation for you.

TW: Well, their songs are not written when they get into the studio. They have ideas and fragments, hooks, lines, and verses, and they're mostly [lead guitarist] Rick's (Nielsen) things that he'll play. He'll say, "I have this line and this line and we have to do something with it." We go into a rehearsal studio for an intense two or three days, and that's it. No more pre-production than two or three days. And I run a cassette of them as they run through everything and we talk about some of the songs, and then I go home and listen to the cassette about ten times in a row. Making notes all the time. Then I go back and say, "This is my top ten or eleven. You should record these songs and pick the best nine or ten. This song should be arranged like this, and I think this verse should come twice. I don't want two guitar solos there. We'll discuss that and work it out."

We'll run it down and get the basic arrangement of the track down, so that when we walk into the studio, [drummer] Bun E. (Carlos), who among other parts of his brain has a computer built-in, will just say "All right, we decided to do this on this song. It goes verse, chorus, verse, chorus, B section instrumental, verse, chorus, fade." And we'll try it out in the studio and somebody will say "fine," or "I don't like having another verse come back then," and they'll play it like that and we'll talk about it some more. We'll arrive at a basic track and get the track down. Things go fast with Cheap Trick, very fast. With "Need Your Love," which is also on *Dream Police* (a studio version), it took us four hours to set up for the right sound, for the drums and bass and guitar. Then they played it once, and that was it.

MR: Are there any "live" vocals?

TW: No, they're all dubbed. Robin Zander (lead vocalist) sings along. Especially on the cuts that are being done for the first time. He'll do them again, but he doesn't take very long with the vocals.

MR: My feeling was, listening to *In Color* and *Heaven Tonight*, that *In Color* is more of a "live" sounding album and *Heaven Tonight* is more "recorded." Is that true in how you approached them?

TW: Yes, and there's a reason for that. I don't envision Cheap Trick as a polished group, but I definitely and very deliberately have set out to make them more appealing to more listeners. I don't care how many critics say "Werman took the cutting edge off," or, "Werman glossed over the aggressive attack." There's plenty of aggressive attack waiting in the wings, as soon as they get the audience, which is what is happening right now. When the audience is there and they have got a million or more people waiting for their next album, *then* they can do what they want and we can get brutal or aggressive or whatever we want. But the increased polish from the first album to the second album to the third album—yes, it was very deliberately designed to get them airplay. That's what "Surrender" is all about. "Surrender" is a pop song, it is not a rock and roll song. I think it's a great pop song. I wouldn't call it rock and roll.

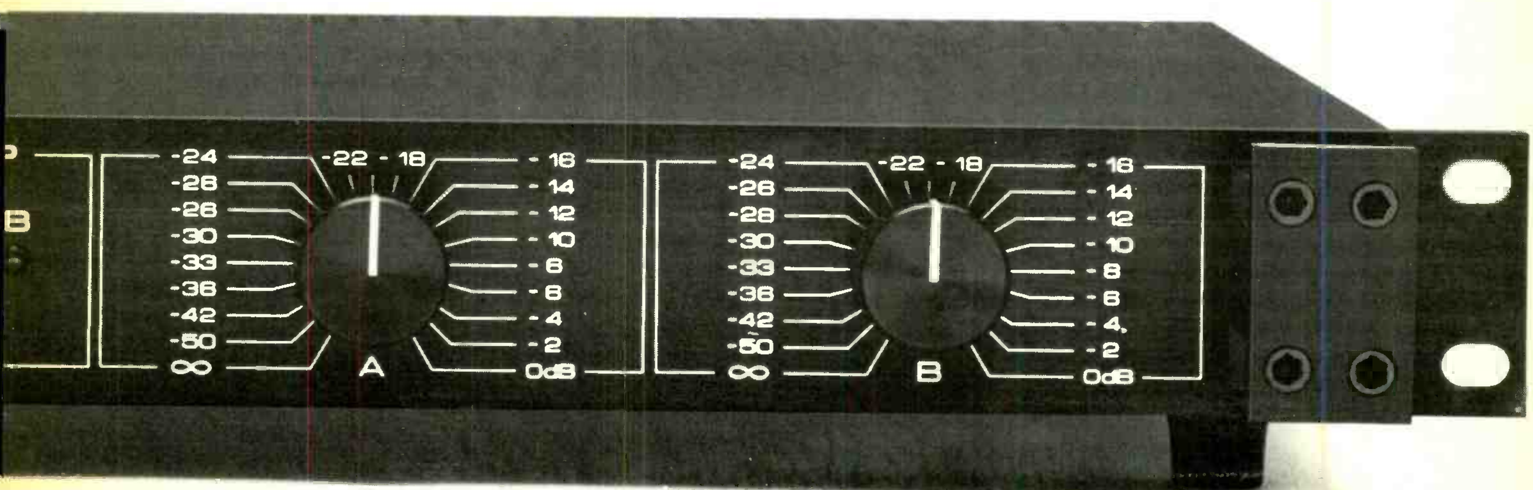
On the new album we have a song that's a throwback. It's a song that [bassist] Tom Petersson sings, it's his vocal debut and it's a real raunchy song. It's got no keyboards, and a little percussion on it (handclaps and shakers). I think every song needs percussion of some kind. The song belongs on their first album, it's a throwback, a fine sounding thing. It's not polished and it's really going to irritate parents. And the kids are really going to like it a lot.

MR: What's it called?

TW: It's called "I Know What I Want." It's a screamer and it's an anthem, you know, "I know what I want and I know how to get it"—over and over and over. That's the kind of thing that whatever the group wants to do . . . if they're all really excited about a song, then the luxury we will have in the future is that we all made compromises to make albums that would get the group a name. They're not foolish, they know that they can't be uncompromising artists and just say, "To Hell with them. We're going to play our music, the man can't bust our music." They have to sell records and they have to get fans.

MR: Are you surprised at the success of the "Budokan" album, or was it always what you were heading for?

TW: Ever since I made the first album, I've been pretty sure that Cheap Trick would be very big some day. I was beginning to get a little impatient when *Heaven Tonight* didn't do it for them, but I think the combination of those first three albums set up "Budokan" and "Budokan" will set up *Dream Police*. I'm never surprised if my artists become successful, because I wouldn't sign an act unless I want to work with them. In fact, Blue Oyster Cult is the first act I've produced that I haven't signed.



MR: You talk about how you polished the sound of Cheap Trick album by album because you're sensitive to what people want to hear. You're saying you are sensitive to what the commercial demands are, and with Cheap Trick, accomodating those demands doesn't hurt their sound, but rather enhances it. Do you think as a producer you could work with a group where making them more commercial would hurt them? Are there situations like that as well?

TW: Yes, I think that's what's happening with Ted. I think that Cliff and Lew have always been striving for a commercial sound. Get him the hit single, stuff like that. Whereas, *Ted Nugent* shouldn't have to become commercial, *commercial* should have to become Ted Nugent. Like, "Cat Scratch Fever" wasn't a natural single, but Ted was so big at the time that radio actively was seeking to play whatever it could. I think in order to grow and survive, Ted Nugent should intensify his musical attack or approach, not soften it—and I think it's been softened.

With Cheap Trick, Rick Nielsen writes such catchy, wonderful songs that I don't feel any conflict in the fact that I occasionally try to make them commercial by throwing in a synthesizer line or some strings or handclaps—things that you can hear on other commercial records. Because they are commercial songs and if you are going to serve the song, then you have to embellish it in a way that is most appropriate to the lyric line and melody. A beautiful woman is a beautiful woman when she gets out of bed in the morning, but she looks even better to most people when she walks into a party that evening with the right clothes on and the right jewelry and make-up. It's like being a hairdresser or designer. If you don't serve the woman right, then she doesn't look good. You have to understand the material to approach it right.

MR: Do you pick your singles? How does that work?

TW: No, I'm not really good at picking singles. I've always tried to ignore that, although it's getting more and more difficult. Rather than picking singles, I'm very insistent about control of sequence. I have a pattern that I like to use. I usually put the song I feel worst about in the next to last position on side 2. Just so it's not the last thing that people hear. And I like to load side 1. I like to make side 1 an absolute killer. And make side 2, cut 1 real strong. Side 2, cut 1 has to be even stronger than side 1, cuts 3, 4 and 5. The first two songs on the album are the most important. If they're great, then somebody is going to want to take the time to listen to the whole album.

MR: Do you consider your albums concept albums?

TW: No, they're just collections of songs, really. Nugent's albums, in a way, are concept albums. You know, they deal with sex and having a good time and rock and roll. A little politics now and then—sexual rock and roll in violent terms.

MR: One last question. You have always been a staff producer at Epic. In terms of your own development, is there ever a time in your future when you foresee yourself becoming an independent? Or do you have enough freedom to do both?

TW: Well, it's a pretty sensitive subject. I do, naturally, look forward to being an independent producer, because you can make your own deals, there is no set royalty, there are not restrictions on which acts you can work with. I haven't wanted to do everyone on the [Epic] label, that's why I've had to go out and sign groups that I've wanted to do. But I'm getting tired of traveling around to night clubs, I don't want to have to keep looking for new bands. I don't mind it, but I don't want to be dependent on it.

Since moving to L.A. last summer, I've probably turned down twenty offers. I don't think it was moving to L.A. that did it, but rather the general success and Cheap Trick's success. Two or three of those I turned down, I really wanted to do. I think I'm going to try to work towards amending my situation with Epic to be semi-exclusive to CBS, meaning I'd be able to give CBS a minimum number of albums per year. Whatever time I had left would be mine.

I have enjoyed being a CBS employee immensely, they take very good care of us, but they may tend to specialize in certain kinds of acts, whereas I want to work with others that are not [presently] available [on the CBS roster]. If I have a burning, consuming passion to do a particular act, I'd like to be able to step forward and say, "I would like to produce you." I can't do that now, where I am. So I'm going to try . . . and I think I will be able to do it.

Ambient Sound

BY LEN FELDMAN

Audio Recording on VCRs

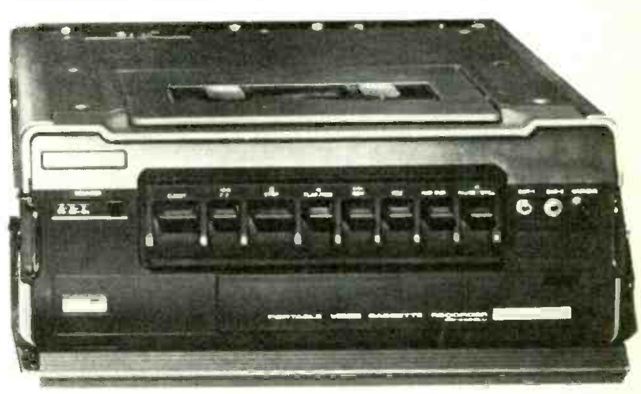
I have just purchased my first video cassette recorder. Actually, that's not quite true. I purchased a VCR that employs the popular VHS system a few months ago, and, after learning how to use it to record off-the-air color TV programs and employing this fantastic machine for a few months for that purpose, I decided that I wanted to get more deeply involved in video recording. The situation is not unlike what happens to the audio hobbyist who starts out buying a stereo cassette deck or open-reel machine with the sole intention of transcribing discs or FM programs, and, after a while, goes out and buys a microphone or two. That leads to the purchase of a small mixer, some equalization equipment, perhaps a reverb or time delay unit and on and on and on. Before you know it you have the makings of a home recording studio.

Well, in order to do the equivalent thing in video recording you have to buy a video camera. And, while there are several low-cost black-and-white video cameras around, I wasn't about to settle for b&w when my recorder had full-color video capabilities. So, I opted for a newly developed color video camera. The camera, while lower in cost than such gadgets were just a year or two ago, nevertheless cost well over $1000 and, since I wanted to be able to record outdoors or "on location" (note how quickly I adopted the vernacular of the movie makers), I decided I needed another VCR, this time a portable one. That set me back another thousand or so to which I had to add the cost of a couple of battery packs (it was either that or a very long line cord . . .), the necessary AC adaptors and chargers, etc. My additional investment over and above that first VCR came to around $3000 but I was ready at last to go into the video recording business. I had given little thought to the fact that movies became "talkies" way back in 1927 and television has also had an audio channel (however poor its quality) since way before Uncle Miltie started broadcasting his tv variety show in the late 1940s.

It turns out that the designers of all my fancy video gear apparently gave as little thought to the audio requirements of the system as I did. To begin with, all presently available VCRs intended for home use provide only a mono audio track on the half-inch wide video tape. That, in and of itself was no surprise since after all, TV programming in this country is still mono and is likely to be so for some time to come. (Japan, on the other hand, initiated dual-channel audio in their TV broadcast operations last October and are using the multiplexing technique for either bi-lingual audio or for stereo audio in the case of musical concerts which would benefit from such transmission.)

In the case of the VHS system, tape speed is only 1.31 inches per second—considerably slower than the standard 1.785 ips used in audio cassette recorders. Video signals are impressed upon the tape by means of a rapidly revolving dual-tape head arrangement which lays down spiral tracks, one below the other, on each segment of tape. So, the *effective* tape speed as far as the video signal is concerned is many, many times the actual longitudinal tape speed; enough to be able to record the short wavelengths which constitute the color video signal.

But, as far as I can tell, the *audio* track is recorded in the more conventional or usual manner and, given the slow tape speed, you can guess what kind of quality you can obtain. As a matter of fact, the published specifications for the audio portion of both my portable and AC operated VCRs pretty much tell the story. The frequency response is listed as extending from 70 Hz



A popular portable video cassette recorder, representative of those available on the market today.

to 10,000 Hz (no plus or minus dB tolerance is provided, which sort of tells you something, doesn't it?). Worse than that, the audio signal-to-noise ratio is quoted as being better than 40 dB. Interestingly, the video signal-to-noise capability is better than 45 dB. I made a quick check of audio frequency response on my particular machine and found that the −3 dB point actually occurred at around 8,000 Hz.

As far as the control facilities related to the audio track of a VCR, they are about as non-existent as those on an inexpensive portable cassette recorder. You guessed it! The audio record amplifier is equipped with ALC, as the purveyors of those cheapie cassette recorders like to call it, or automatic level control circuitry. That means that no matter what you do ahead of the microphone input (use fancy mixers so you can multi-mike the sound you hope to record, equalize, sweeten, you name it), that idiot-like ALC circuit is going to shift everything up as close to one common level as it can. Of course, for the uninitiated home video cinematographer who has got to be his or her own sound man, script girl, director, producer and grip I suppose this has some advantages. You know that up to a considerable distance from the mic (which, by the way, is cleverly hidden in the handle of the color camera) you are going to pick up the sound of your actors and actresses. When the performers get closer to the mike, for a close-up shot, their voices will become intimate—and the background noise will drop down—but they won't be much *louder* in playback than when they were standing across the room several feet away. Such is the miracle of ALC. *Yeech!*

Now, surprisingly, the microphone itself is not bad at all. It's a low impedance omnidirectional (naturally) electret job whose response goes way out to 15 kHz or so and is reasonably flat all the way down to the low bass range. But it is irrevocably tied in to that blasted ALC record-electronics. You can disconnect the built-in microphone and feed audio to your recorder through an additional microphone input, or even through a high-level audio input using a microphone mixer with gain, as opposed to a passive mixer. It was while trying this that I discovered that there *is* a way to beat the ALC monster inside my recorder. It seems that using the high-level audio input, there is a threshold level below which the automatic level action ceases to function and a change of input level of, say 10 dB actually comes out during playback as a change of output level of "almost" 10 dB. Spurred on by this discovery, I found that I could manage a dynamic range of audio of almost 40 dB from the lowest levels to the point at which that ol' devil ALC takes over.

The only trouble is when I tried to lower all input levels from my mic mixer so as to take advantage of that dynamic range, the overall signal-to-noise ratio in my recorded audio results was nothing to be proud of. To get any reasonably low level of tape hiss you darn well better feed in enough signal to wake up the ALC circuits *or else*! Perhaps my separate outboard Dolby B units are the answer. I'll have to think about that!

Meeting the Challenge

So here I am, dear readers, with around three grand worth of superb video equipment (the picture fidelity is absolutely superb, the color is great, there is not a hint of jitter or vertical picture roll; horizontal resolution is better than on many TV sets picking up actual broadcasts) and I can't record a decent audio accompaniment for my extensive cinematic efforts. It sort of makes you wonder.

Come to think of it, this has been the way with TV ever since Uncle Miltie first appeared on the small screen back in the late '40s. Everyone worried about the picture and no one paid much attention to the sound. But perhaps things are changing. Now, I am told, the broadcasters are using satellites and a system called diplexing that allows full 15 kHz response on the audio channels of TV broadcasts. Audio equipment makers are meeting the challenge by coming up with separate TV-audio tuners that pick up the sound-only of TV transmissions in full fidelity, bypassing those 3-inch speakers and one-IC miniwatt amplifiers TV set makers grudgingly provide. Committees are already being formed to study systems for stereo TV sound in this country (it should probably take us only three or four years to catch up with Japan) and there is hope for better TV audio on the horizon.

A video camera similar to the one discussed here by Mr. Feldman, this one by JVC (shown with its adaptor).

I suspect that if TV sound does get better, the makers of those neat home video recorders will come up with better record capability for the audio tracks of the VCRs as well. Already there is talk that next year's home VCRs will feature two audio tracks. If there's really to be the marriage between audio and video that everyone has been talking about, the audio partner in that marriage will have to play an equal quality role. Oh well, maybe I took the plunge into video/audio a couple of years too early. It will not have been the first time!

MODERN RECORDING LAB REPORT

NORMAN EISENBERG AND LEN FELDMAN

Mitsubishi DT-30 Cassette Recorder

General Description: The Mitsubishi DT-30 cassette recorder uses three separate heads for erase, record and play. Provisions for adjusting the azimuth of the record head include a recessed screwdriver adjustment in the head cover and a built-in test signal generator; these are used to obtain proper readings on the deck's signal meters.

The transport uses a closed loop, dual capstan system and two motors. For capstan drive there is a phase-locked loop DC motor, and for the fast modes there is a separate DC motor. Complete fast-buttoning via the "feather-touch" keys is possible, including the option of "run in" or "flying start" recording.

The meters are peak-reading, with a peak-hold option. Input mixing of line and microphone signals is possible. In addition, the deck has a mixed-playback feature whereby the recorded signal on a tape being played may be mixed with the line or microphone inputs for special effects in playback.

The deck also has an automatic spacing pause system (ASPS) which enables the recordist to mute any input signals while recording for about three seconds in order to introduce silent sections between recorded signals more smoothly than may be possible by using the conventional pause control.

A timer switch permits unattended recording or playback in conjunction with an external timing device. A special socket and cable (supplied) permits synchronizing the recorder with an external turntable (the same manufacturer's model DP-EC20). The recorder's memory switch may be used to start the deck or stop it. Another switch permits automatic rewind or repeat when the tape reaches its end.

A front-loader, the DT-30 has no cassette well or compartment in the usual sense. Instead of a recessed area, the cassette fits onto an actual continuous part of the front panel itself through which the spindles project. The cassette is held in place by an overhanging top section and by a hinged member along the bottom which also serves as a dust cover. The largest portion of the cassette thus is not covered during operation. The heads, capstans and related parts are housed in this lower raised section which also contains the buttons for the functions of stop, rewind, play, fast-forward, record, pause and the ASPS.

In a vertical row left of the cassette area are the switches for: power off/on; timer operation; memory; automatic rewind or repeat; turntable synchro. To the right of the cassette area, near the top of the panel, are the tape index counter and its reset button. Below them is the peak-hold switch for use with the signal meters at the right. The meters are calibrated from −40 to +7, light up during use and show playback as well as recording levels.

To the right of the transport control array are two switches and an associated LED indicator for the built-in test signals. One switch activates the generator; the other selects the frequency of either 440 Hz or 8 kHz, used in making azimuth and bias current adjustments. Left- and right-channel bias trim knobs are directly under the generator switches.

The monitor selector switch has three positions. In addition to the usual source and tape settings, the third setting of "play mix" engages the optional feature mentioned earlier whereby already recorded material may be mixed with new material (for playback, not for "add-on" recording). There are two regular tape selectors, for bias and for equalization. Each has three positions for selecting various tapes. The owner's manual lists a large number of tape brands and types with recommended switch settings. The Dolby NR switch includes a position with FM multiplex filter, if needed.

The deck's output level control is a single knob that adjusts playback volume through the line outputs (at the rear) and through the stereo headphone output (located at the lower left corner of the front panel). Dual-concentric knobs are used for line and microphone input level controls, permitting adjustment on both channels simultaneously or separately. In addition there is a larger "recording master" control that is used for changing recording level without using the line or mic controls. Associated with the master control is a rotatable reference tab that introduces a detent position at whatever position it is moved to. This may be used to establish a "pre-set" level that may be used for fade-in and fade-out recording. The left- and right-channel microphone jacks are below this recording master control.

The rear panel of the DT-30 contains the line in and out jacks, the optional socket for use with the turntable-synchro cable and the AC line cord. The DT-30 is supplied in a metal case with front handles and mounting feet. Included with the deck are a TDK AC-511 alignment tape; the turntable synchro cord (which mates only with Mitsubishi's model DP-EC20 turntable); two pairs of stereo signal cables; and a small screwdriver for azimuth adjustment.

Test Results: In *MR's* tests, response of the DT-30 with both normal (ferric-oxide, low-noise) tape and with high-bias (CrO_2 or equivalent) tape exceeded published specs. Response with ferrichrome tape confirmed the claimed high end at 18 kHz, but the low end rolled off sooner than as per spec. Signal-to-noise, using the Dolby NR, was excellent with any tape. Distortion was lowest with normal tape.

The frequency reponse plots shown here were made using our spectrum analyzer on a real-time swept basis, with bias adjust controls set at their midpoints. For the tabulated results stated in our Vital Statistics summary, bias was optimized to suit each individual tape sample. Once optimized, however, other parameters such as S/N and distortion also were measured for that same bias setting in each case. As is always true, somewhat lower distortion figures could have been obtained had we been willing to sacrifice high-end response by changing to a little higher bias In any event, since the deck does provide this kind of flexibility, the user can make these choices as desired.

Transport action of this deck was flawless, and it is perfectly safe to go from any mode to any other without jamming or damaging the tape. Whle the method of cassette insertion may, at first, seem crude to the user accustomed to damped swinging doors and the like, the direct method used in the DT-30 is in fact completely satisfactory. If anything, it probably results in more accurate alignment of the cassette relative to the heads than do the more conventional systems.

The deck's built-in test tones do help with the vernier bias setting chore; they are a must for anyone who doesn't own an audio generator and VTVM (the deck's own meters guide the user during bias adjustments).

Very likely it will take the owner of this deck a few practice sessions to fully understand and master the use of all its features, but once this is done the deck may prove quite enjoyable to use. The fact that Mitsu-

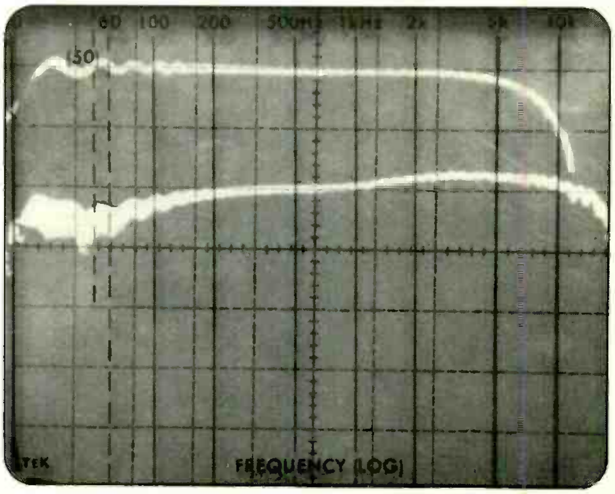


Fig. 1: Mitsubishi DT-30: Record/play response, at 0 dB and −20 dB, using TDK AD tape.

bishi has used peak-reading meters, and has set "zero dB" at 200 nWb/M, makes it very important that the user not go above 0 dB in setting record level. This is especially true when using high-bias (chromium-dioxide or ferric-cobalt) tapes which will produce 3% harmonic distortion when levels are as little as 1 dB above the nominal zero on the meters. This need not be regarded as a drawback, however, since the total dynamic range down to the noise level from the 3% THD point proved good with all three tape samples we used for the purposes of this report.

It should be noted that in order to meet record/play response specs, it was necessary to use not only the front-panel bias controls, but also to adjust the azimuth of the record head for each new tape sample used in the tests. Apparently, this is the price one must pay for three discrete heads in a cassette deck as opposed to either two heads or to a three-head configuration in which record and play heads share a common housing.

General Info: Dimensions are 16¾ inches wide; 6¾ inches high; 14⅞ inches deep. Weight is 23 pounds. Price is $650.

Individual Comment by L.F.: There seems to be no end to the features that manufacturers of stereo cassette decks can come up with for their more expensive models. In the case of the Mitsubishi DT-30, some of these "extras" strike me as definitely worthwhile, but there are others that add to the cost but do not—to my mind—really provide important user benefits.

Let's examine the latter features first. I can't see why anyone would want to repeat a given tape (play it over and over again) ad infinitum, nor am I especially taken with the auto-rewind feature, since I usually record on both the "A" and "B" sides of a cassette. Of course, it could be argued that some folks only record on, or want to listen to, one side of a cassette and that there is a certain elegance in having the tape whizz back to its start, but I could live without this feature.

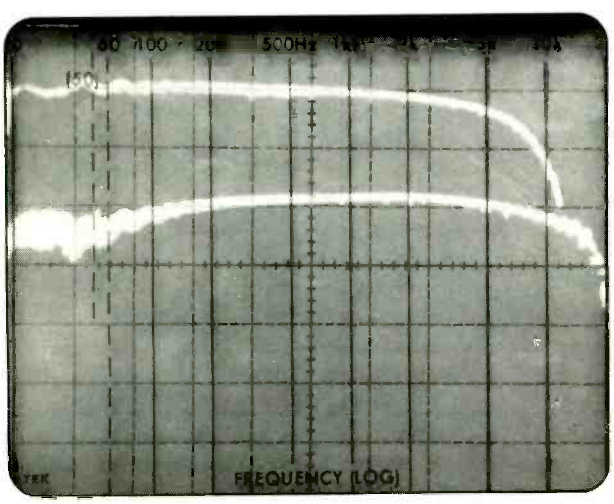


Fig. 2: Mitsubishi DT-30: Record/play response, at 0 dB and – 20 dB using TDK SA tape.

Nor can I see the need of the third playback position which allows you to mix recorded signals with whatever is then being fed into the line or mic inputs. This would be a great feature if we were able to record one track at a time, since it would constitute a kind of multi-track facility with simul-sync capability—but in this case it does not. All it enables you to do is to accompany the music on your tape with other program material *while listening*.

Among the features I did feel were worthwhile, first mention must be made of the three-head configuration, although—as stated in the lab test report results— we did find it necessary to readjust both bias and azimuth for optimum response with each of the three tapes.

The peak-reading meters worked well, and I appreciated the peak-hold feature, though many other manufacturers seem to be abandoning conventional mechanical meter movements in favor of LED or fluorescent type indicators which can respond more quickly to transients than any meter needle.

While we could not check out the "turntable synchro" feature of the DT-30, it seems like an interesting idea. With the special cable linking the DT-30 and Mitsubishi's DP-EC20 turntable, the deck can be made to start recording automatically when the tone-arm of that turntable touches down on the lead-in groove of a disc being played. In the single-play mode of the turntable, recording will stop when the record has been played completely. In the multi-play mode, the deck will go into "pause" whenever the tone-arm lifts up from the record.

I liked the transport action, and the ASPS option which allows you to insert fixed 3-second pauses between recorded selections. When this button is pressed, the recording stops while the tape moves forward for three seconds. At this point the transport shifts itself into the pause mode (no tape motion). To resume recording, you simply touch the pause button.

The owner's manual covers everything clearly and in logical order. Popular brands and types of tape are listed with their correct bias and EQ switch settings. Although two switches are provided (one for bias and another for EQ), they are always used in a "tracking mode" if you follow the manual. Of course, one could ignore these instructions and experiment with different combinations for bias and EQ if desired, but from a practical point of view a single combination switch would have served equally as well.

A definite plus in this deck is the built-in twin-tone test generator. As to whether the DT-30 is worth the asking price of $650, I would have to say that it is—not because of its many gadgets and switches but rather for its three-head configuration, its excellent two-motor transport and its full-logic operation.

Individual Comment by N.E.: It's getting to be a guessing game with cassette recorders—each time we receive a new model for testing we ask the question: How will this model differ from previous ones? In the case of the Mitsubishi DT-30, as with most recent

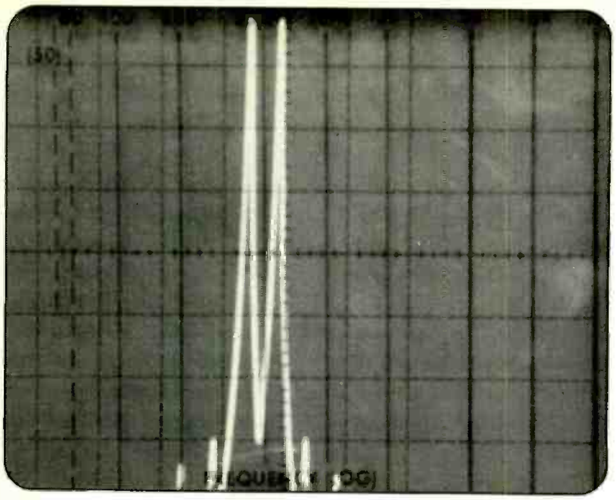


Fig. 3: Mitsubishi DT-30: Record/play response at 0 dB and – 20 dB using Sony Duad (FeCr) tape.

units, the answer is not in terms of audio or mechanical performance so much as it is in terms of "features"—those special design flourishes a manufacturer chooses to build into a particular recorder in order to generate interest on the part of prospective buyers.

The DT-30 is a mixed bag of such features, from its unusual (but satisfactory) method of loading the cassette to its option for adding program material to whatever is being played back. This latter feature might be of use in P.A. or disco applications.

The turntable synchro feature is of value only to those who also use the Mitsubishi turntable. For all others, it is just there, of no use.

The ASPS feature may or may not appeal to you. I for one prefer to time my own silent sections. Three seconds is not always what I might need, especially if I'm dubbing off the air and want to eliminate commercials whose timing does not match the ASPS interval.

The peak-hold feature for the signal meters becomes useful only when peak recording levels exceed the 0 dB point. In fact, as the manual states, as long as the peak level remains less than 0 dB, the peak-hold button has no effect on the pointers. In my view, a peak-hold feature should involve some kind of indication other than the normal meter pointer (as it does on alternate signal readout systems such as LEDs or fluorescent displays), and it should be operative regardless of peak levels reached.

The separate heads for record and play are, by definition, to be regarded as a plus vis-a-vis the combined r/p head (assuming, that is, that they have been optimized for their respective functions). However, the system that combines them in a common housing which obviates the need for periodic azimuth adjustment seems more convenient and less subject to misadjustment.

The automatic return to the start of a tape after it reaches its end (on one side) could be helpful as long as one is not planning to flip the cassette to continue recording or playback on the other side.

Features aside, the DT-30 does have very high performance. Judging all in all I would say that its best overall results are to be obtained with normal low-noise ferric-oxide tape. This tape produced excellent r/p response along with the lowest distortion and the best signal headroom. S/N with the other tapes was a little better, but normal tape's S/N was still in the "excellent" class, especially with the Dolby NR activated. Other tested parameters all were well within the ballpark for this class of cassette machine. Headphone output was adequate for comfortable listening with the output gain control turned just about full up. The transport seemed especially good, with very smooth fast-buttoning.

Additional Tests for Cassette Decks

As the state of the art in cassette recorders advances, *MR* is adopting more sophisticated techniques and tools for testing these products, to provide more detailed information for its readers. The table of "Vital Statistics" accompanying this report reflects this enhanced program.

Note that we are now presenting key performance data for three different kinds of tape. One is so-called normal or standard tape, a low-noise ferric-oxide. The second is tape requiring high bias, and this applies to chromium-dioxide as well as to cobalt-tested varieties. The third kind of tape is whatever a given machine is spec'd for—be it metal-particle or, as in the case of the Mitsubishi DT-30, a ferrichrome.

By studying the results of different tests using all three kinds of tape, the prospective user can judge which tape to choose for a particular recording chore. Also, some kind of general feeling as to which is the best tape for general use with a given machine can be obtained, although admittedly this is often a matter of opinion which will be discussed in either the "Test Results" section, or the "Individual Comments" section, as applicable.

Speed accuracy is a test we have recently added, as well as a standard playback curve (thanks to the new TDK test tapes). Note that for the playback curve, the actual deviation from "0 dB" is more important than the span of frequencies—it is unlikely these days that any normal component cassette deck would not be able to span, on playback, the range from 40 Hz to 12.5 kHz that is provided on the test tape. What is germane here is the degree of flatness of that response across the band, which can be shown only in terms of a curve as presented in this report.

More work for us, but more info for you.

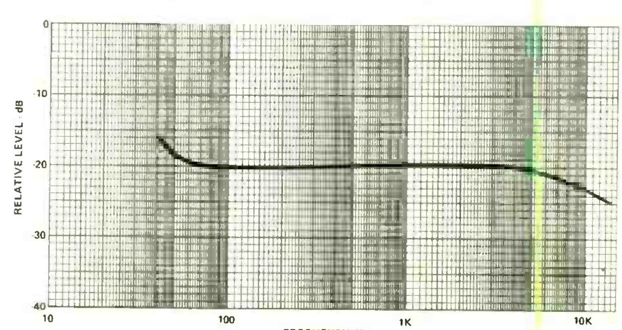


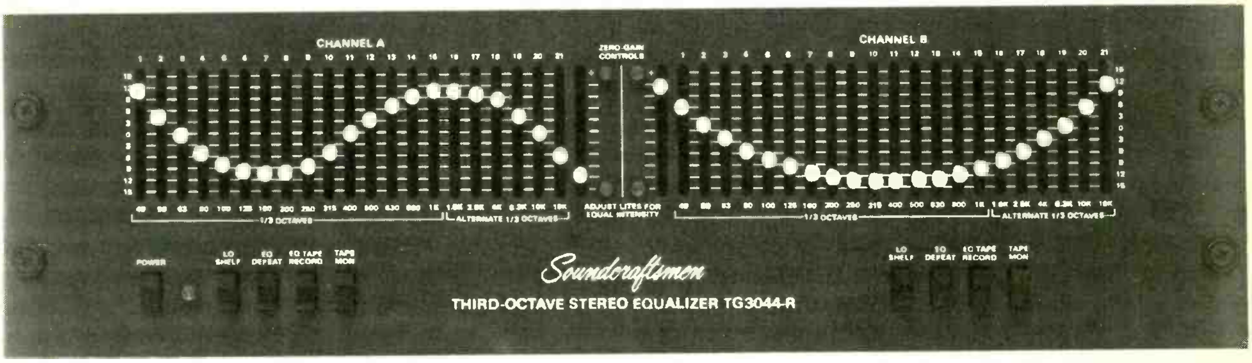
Fig. 4: Mitsubishi DT-30: Playback response using TDK test tape AC-337.

MITSUBISHI DT-30 CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response		
normal tape	± 3 dB, 40 Hz to 15 kHz	± 3 dB, 32 Hz to 18.5 kHz
high-bias tape	± 3 dB, 40 Hz to 17 kHz	± dB, 31 Hz to 17 kHz
other tape (FeCr)	± 3 dB, 40 Hz to 18 kHz	± 3 dB, 60 Hz to 18 kHz
Signal-to-noise (re: 3% THD record level), Dolby out		
normal tape	NA	56.5 dB
high-bias tape	58 dB	57 dB
other tape (FeCr)	NA	59 dB
S/N, as above, Dolby in		
normal tape	NA	64.5
high-bias tape	66 dB	65 dB
other tape (FeCr)	NA	67.5 dB
THD at 0 dB record level		
normal tape	1.0% (ref 160 pWb)	1.6% (ref 200 pWb)
high-bias tape	NA	2.8%
other tape (FeCr)	NA	2.8%
Record level for 3% THD (0 dB = 200 pWb/m)		
normal tape	NA	+ 3.8 dB
high-bias tape	NA	+ 0.7 dB
other tape (FeCr)	NA	+ 1.0 dB
Line output at 0 dB	440 mV	500 mV
Headphone output at 0 dB	90 mV (8 ohms)	110 mV (8 ohms)
Mic input sensitivity for 0 dB	0.3 mV	0.2 mV
Line input sensitivity for 0 dB	100 mV	90 mV
Wow-and-flutter, WRMS	0.045%	0.045%
Speed accuracy	NA	− 0.3%
Fast-wind time, C-60	NA	87 seconds
Bias frequency	85 kHz	85 kHz
Power consumption	28 watts	30 watts

CIRCLE 25 ON READER SERVICE CARD

Soundcraftsmen Model TG3044-R Third-Octave Stereo Equalizer



General Description: The "44" in the model number of this new Soundcraftsmen device refers to the no less than forty-four slider controls on its panel. There are twenty-one sliders for equalization plus a slider for "zero gain" on each of two independent channels. The unit thus functions as a stereo equalizer or as two separate mono equalizers if desired.

The first fifteen sliders have nominal ISO center frequencies of 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800 and 1 kHz. The remaining six frequency sliders have alternate (two-third) octave center frequencies of 1.6 K, 2.5 K, 4 K, 6.3 K, 10 K and 16 K. The last (gain) slider may be used to restore unity gain regardless of the settings of the frequency sliders.

Unity gain is achieved when the gain slider is adjusted so that two LEDs (one above and the other below the slider) glow with equal intensity. In this way, average unity gain is maintained, and comparisons between equalized and unequalized signals (using a front-panel button provided for making such comparisons) become more meaningful.

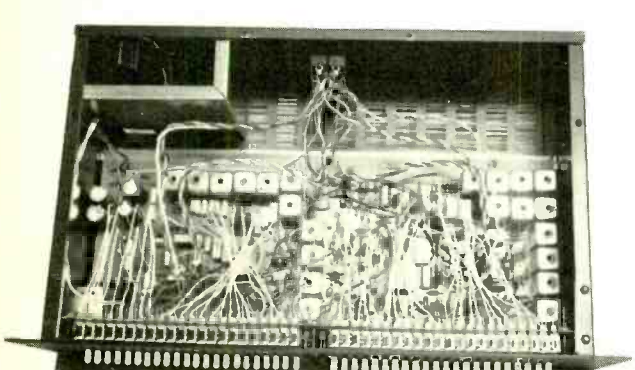
The control range for each frequency is marked in steps of 3 dB between the two extremes of ±15. Under the sliders, in addition to the EQ defeat switch, are switches for a subsonic filter, to a tape recorder and for tape monitor. The righthand layout is a mirror image of the left-channel layout. The device's AC power switch and associated LED are at the left. Front-panel styling is neat and functional: crisp, legible white lettering against a black background. Supplied in a metal case with walnut side panels, the TG3044-R may be rack-mounted if desired.

Inputs and outputs at the rear are standard ¼-inch phone jacks, eight in all. One stereo pair handles line in; another, line out; another, tape in; another, tape out. Also at the rear are explanations and diagrams for balanced and unbalanced operation, and suggested system hookups for stereo in the unbalanced mode. The AC power cord is fitted with a three-prong (grounding) plug, and there is an AC convenience outlet that also is a three-prong type.

The choice of third and alternate-third octave equalization is based on the manufacturer's desire to provide a unit that is aimed at the studio engineer and performer as well as the advanced audiophile. Providing one-third octaves up to the 1-kHz area is designed to correct for acoustical problems that require relatively narrow frequency segments for correction. Room problems above 1 kHz are said to be of a broader nature and lend themselves to correction via relatively wider frequency segments.

The jack options and controls permit the operator to pre- or post-equalize during or after tape recording.

Test Results: All of the published specifications for the Soundcraftsmen TG3044-R were met or bettered in *MR's* tests. Especially notable was the device's extremely low distortion at rated output of 2 volts. S/N ratio for this output also was commendable at 100 dB



Soundcraftsmen TG3044-R: Internal view.

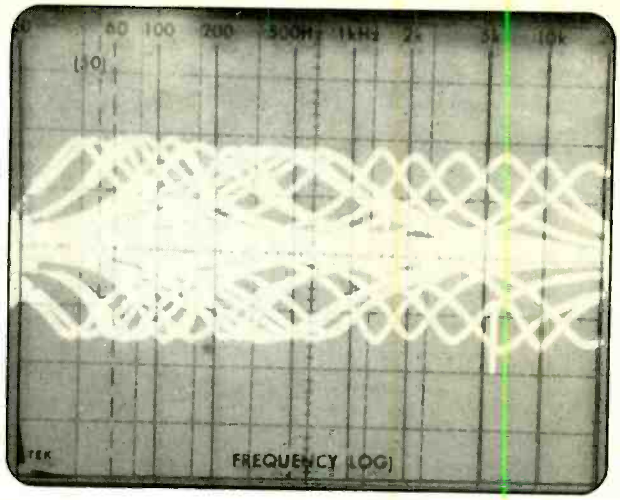


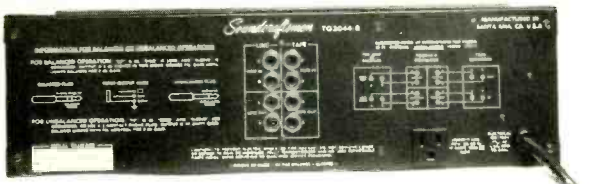
Fig. 1: Soundcraftsmen TG3044-R: Composite plot of control range of 1/3 octave and 2/3 octave controls.

"A" weighted.

A composite plot of the boost/cut characteristics of each of twenty-one slider controls on each channel is shown in the 'scope photo. While the amount of boost and cut was virtually the same for each control, we did note that the center frequencies, particularly at the low end of the audio band, were not as precisely adjusted as they are on some other equalizers (including some made by the same manufacturer) that we have checked in the past. While this is not a very serious matter (there are indeed enough sliders down there, at frequencies close enough to each other, to take care of virtually any low-frequency EQ problems), it is somewhat surprising, in view of the fact that it is at these low frequencies that the device changes over to op-amp "gyrator" inductors whose filter characteristics are determined by feedback capacitors rather than by passive coils.

The device is not supplied with an instruction manual as such. Instead it comes with a test record in a foldover jacket which contains detailed printed and illustrated instructions for installing and using the equalizer. For room equalization, three methods are explained. One is called "simplified" equalization which is done by ear, using the pink-noise signals on the test record. Another is called "more accurate" equalization which also is done by ear but which involves moving the speaker systems. The third "professional" method requires the use of a calibrated microphone and sound-pressure level meter. A note enclosed with the record points out that the record was designed for octave-wide EQ and is not usable for third-octave balancing. It still may be of some use as a pink-noise source if used with the SPL meter.

Similarly, the frequency plotting charts supplied are octave-band charts and do not correspond to the actual third-octave and alternate-third octave frequencies on the device itself. But by referencing these charts to the tones on the record, a frequency curve can be obtained that will represent overall response.



Soundcraftsmen TG3044-R: Rear panel view.

General Info: Dimensions are: 19 inches wide; 5¼ inches high; 11 inches deep. Weight is 23 pounds, and price of the device is $550.

Individual Comment by L.F.: For Soundcraftsmen to have called this unit a third-octave equalizer constitutes what I would have to call a two-thirds truth. While the unit does provide individual controls for the standard ISO center frequencies at one-third octave intervals from 40 Hz to 1 kHz, it provides alternate third-octave controls above that point. Apparently, the objective here is to give the demanding audiophile and the semi-pro user as much EQ facility as possible for the targeted price point. This, of course, the manufacturer has succeeded in doing. Its explanation of "less detailed" EQ needed above 1 kHz is reasonable in context of the unit's price, and this design is not a bad compromise at that.

Another compromise is evident in the device's use of discrete passive coils for a majority of the bands, while using so-called gyrator inductors for the lower frequency bands where coils would have had to be rather large. In view of the excellent signal-to-noise we measured for this unit, I have no particular quarrel with this money-saving approach either.

I am not altogether certain that I go along with the omission of an owner's manual as such. The well-known Soundcraftsmen test-record is supplied (the same one they offer with their octave equalizer, and which cannot be used for setting up your listening room unless you own a sound-pressure level meter and can work with the wideband pink-noise signals of 20 seconds duration). The record liner does carry useful information on the uses of an equalizer, but it seems to me that some helpful words somewhere regarding the hookup and interface options possible with this unit ought to be included.

The front panel sliders happily have center detents so that one can set any slider to its no boost/no cut position without having to squint at the markings.

The front panel photo was supplied by Soundcraftsmen, and for a while I wondered what kind of a strange room would require the unusual settings they show of the left and right channel controls. Then I rotated the photo 90 degrees (with the power switch at the top) and stopped wondering. Get it?

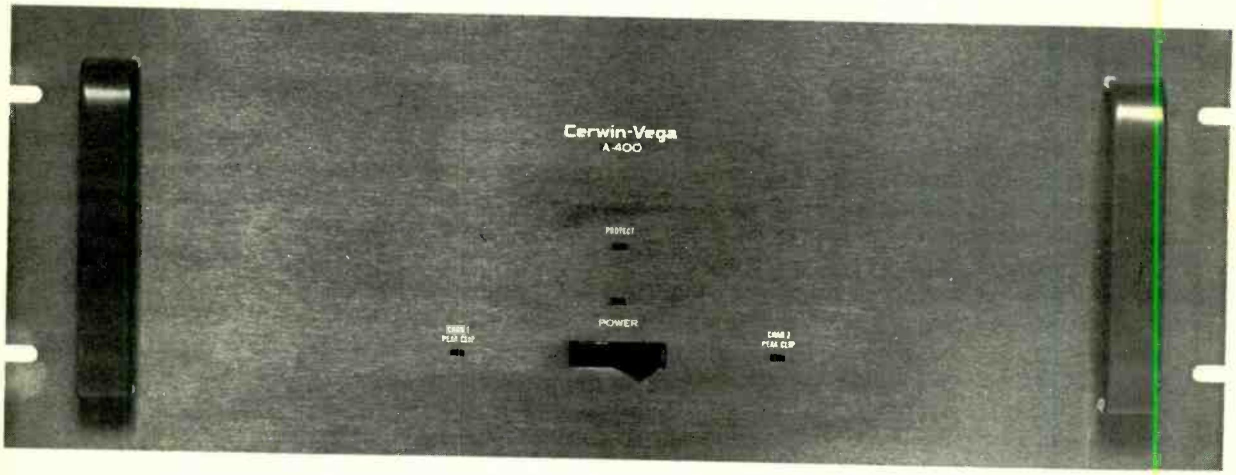
Individual Comment by N.E.: The TG3044-R is obviously a more versatile, and somewhat better performing unit than an earlier Soundcraftsmen equalizer (the model SG2205-600) tested some time ago. At that, it reflects—as do so many recent audio products—the "overlap" that has developed in terms of product appeal and design between what once was regarded as the domain of the "serious audiophile" and that of the "semi pro" or even the economy-minded full pro. At its price, and in view of its actual performance, the design compromises noted in our report are certainly quite acceptable.

At that, this device could be more of an "all things to all users" product if there were more explicit instructions furnished for using it in all its possible applications which include, of course, tailoring frequency response when making a tape recording in addition to room equalization. The former function is one that, according to letters and comments we receive, interests a good many recordists and it could be better spelled out for a device of this capability. The latter function is not fully realized in terms of the test record and instructional material supplied. It would seem that Soundcraftsmen's information department needs to do some catching up with its strong and solid engineering department.

SOUNDCRAFTSMEN MODEL TG3044-R THIRD-OCTAVE EQUALIZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response (controls set for flat, EQ on)	± 0.5 dB, 20 Hz to 20 kHz	± 0.3 dB, 20 Hz to 20 kHz
Harmonic distortion, 2 volts out	0.01%	0.003% at 1 kHz 0.002% at 20 Hz 0.006% at 20 kHz
S/N ratio, re: 2 volts out	100 dB	100 dB "A' wtd
Input capability, maximum	10 V (22 dBm)	11 V (23 dBm)
Output capability	20 V (28 dBm) balanced; 10 V (22 dBm) unbalanced	11 V (23 dBm)
ISO center frequencies	1/3 octave, 40 Hz to 1 kHz; 2/3 octave, 1.6 kHz to 16 kHz	confirmed
Boost/cut range	± 22 dB with other controls at max.; ± 15 dB, other controls flat	confirmed
Subsonic filter	– 3 dB at 15 Hz, 12 dB/octave	confirmed

Cerwin-Vega Model A-400 Power Amplifier



General Description: Cerwin-Vega's model A-400 stereo power amplifier is rated to deliver an output of better than 200 watts per channel into 8-ohm loads, or over 350 watts per channel into 4-ohm loads. The front panel, in addition to the handles, contains the power switch and four LEDs—a power indicator, a protection indicator and peak clip indicators for each channel.

The rear contains both phono (hi-fi pin type) jacks and ¼-inch phone jacks for inputs on each channel, with separate level controls handling each channel. Outputs are standard color-coded binding posts. Also at the rear are a switch to select between 120 and 240 VAC line power, a fuse-holder and a grounding terminal. The AC line cord is fitted with a three-prong (grounding) plug. A cooling fan, mounted just behind the rear panel, is designed to operate when temperature at the output heat sinks exceeds 70 degrees C.

An internal timing circuit is designed to delay signal operation for 2 to 4 seconds after the power switch is turned on, in order to prevent possible damage to the speakers from turn-on transients. The "protect" LED on the front panel lights up whenever the built-in protection circuits sense a potentially damaging condition and go into operation to prevent the damage from occurring. Since these circuits become active whenever AC power is lost or restored, this LED will automatically come on briefly when the AC power switch is turned on or off. According to the manufacturer, the amp will fully protect itself and all ancillary equipment for the duration of any unsafe operating condition.

Test Results: Power output and distortion of the Cerwin-Vega A-400 amplifier were measured for both 8-ohm and 4-ohm loads and found to exceed published specifications. In fact, subjected to both "conventional" and "more sophisticated" lab tests (see accompanying explanation), the A-400 came through in top form, and impressed *MR* as offering very ample power output combined with ruggedness and reliability.

The total performance picture, taking together such parameters as IM and harmonic distortion, TIM, slew rate, transient response, turned out to be superb—as did the sound of the amplifier.

Happily too, the built-in cooling fan is not designed to run all of the time. Rather, it is designed to come on only when heat-sink temperatures reach 70 degrees C. This did not occur during our bench tests, even though a good deal of the time that the amplifier spent on the test bench was devoted to delivering rated output or higher, and in the usual pre-conditioning at one-third rated output.

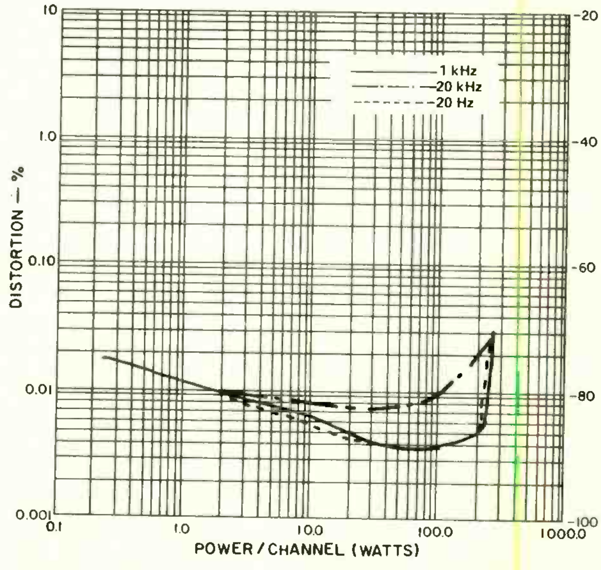


Fig. 1: Cerwin-Vega A-400: Power output/channel vs. distortion, 8-ohm loads.

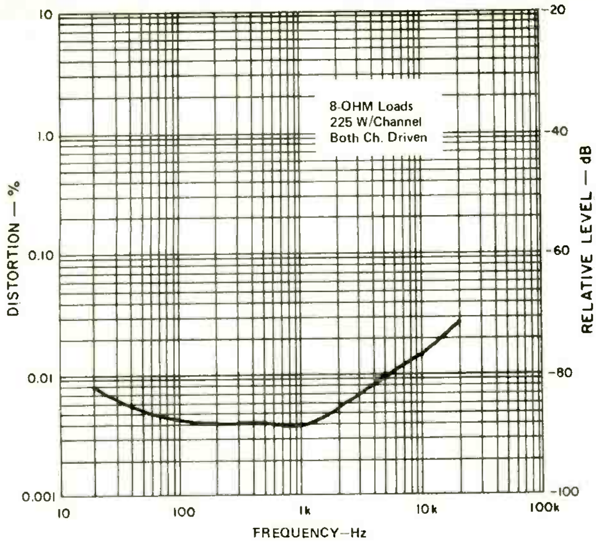


Fig. 2: Cerwin-Vega A-400: Distortion vs. frequency.

General Info: Dimensions are 19 inches wide (rack mount); 7.4 inches high; 14 inches deep. Weight is 45 pounds. Price is: $900.

Individual Comment by L.F.: Frankly, I had always thought of Cerwin-Vega as being synonymous with lots of sound power—the kind of ear-deafening and body-massaging sound that I was subjected to when attending one of the "Sensurround" films for which this company supplied the electronics and loudspeakers. I never thought of the company as a maker of highly accurate, state-of-the-art audio amplifiers such as those by some of the more esoteric smaller companies which audiophiles delight in discovering.

Well, these preconceptions have been shattered. Of course, the brute force is there—all 225 watts plus per channel of it. But there also is ample proof that a professional amplifier maker need not sacrifice accuracy of signal waveform reproduction in return for ruggedness, longevity or high-power-handling capacity with absolute safety. Here is an amplifier (and I hesitate to suggest this) that would be very much at home in a top-grade audiophile hi-fi component system as well as in professional studio and sound reinforcement applications for which it is primarily intended. I suspect that Cerwin-Vega must have realized that some members of the hi-fi fraternity would discover its virtues soon enough, or else why would they have paralleled their phone-type input jacks on the rear panel with home-style phono-tip jacks?

What makes this amplifier a "professional" unit instead of a high fidelity component? I'm not at all sure, unless it's the rugged construction, the carrying handles affixed to the standard-width 19-inch front panel or that ventilation fan which never had to turn on during our tests. Call it a pro amp, or a hi-fi amp—whatever you like—Cerwin-Vega has done one hell of a good design job on it.

Additional Tests for Amplifiers

In keeping with *MR's* desire to provide second-to-none reliable information on audio equipment, its testing facilities have been recently augmented by new equipment that enables us to measure CCIF IM distortion as well as the new IHF IM distortion. These new figures, together with the older SMPTE IM distortion, are given in the "Vital Statistics" table for this report.

The IHF IM distortion test employs two-tone test signals, 1-kHz apart, which are used to drive the amplifier under test to rated output.

In the CCIF IM test, we measure any "beat frequency" 1-kHz signal produced by intermodulation effects.

The IHF IM method is the more stringent of the two methods, since it takes into account *all* newly generated IM products in the audio band up to 20 kHz. Actually, we have found that the IHF IM measurement offers very good correlation between the percentage measured and the way an amplifier "sounds" when reproducing program material containing steep transient waveforms. (The Cerwin-Vega A-400 turned out to be an excellent first subject for using these new measurements).

The method of calculating IHF IM distortion can best be understood by examining Fig. 3. The two tall spikes at center-screen represent outputs at 19 kHz and at 20 kHz, whose combined peak amplitude is equivalent to rated output at 8 ohms. Frequency sweep is *linear* (the frequency notations should be ignored in examining this photo), and sweep rate is 2 kHz per horizontal division. Vertical sensitivity is 10-dB per division.

Just to the left of the tall 19-kHz and 20-kHz spikes we see two lower-frequency spikes (at 17 kHz and at 16 kHz). No additional components were noted in the audio range within the 80-dB dynamic range of the spectrum analyzer. By combining the two IM products (−70 dB below peak equivalent desired signal, and −75 dB below desired signal level), we arrive at an equivalent IHF IM distortion percentage of 0.0326 percent.

To readers who have had no experience yet with this type of measurement, that percentage may not mean very much. We must assure you that it is indeed an excellent figure. So, for that matter, is the CCIF figure of 0.0014 percent. Note that the CCIF figure uses only the 1-kHz component produced by the two tone combinations and, since 0.0014 percent is equivalent to better than −97 dB, that explains why we did not see any of this component when using the spectrum analyzer in testing the amplifier.

Readers also will note THD measurements are now given for 20 Hz and 20 kHz in addition to the data for 1 kHz.

S/N ratio re: rated output is still given, as well as the newer (and preferred) method of S/N ratio for a 1-watt output with an input of 0.5 volt.

Dynamic headroom is now given, and damping factor is measured at 50 Hz.

The older form of input sensitivity is given, as well as the more recent IHF input sensitivity which is referenced to a 1-watt output.

Slew rate is another measurement we now plan to include on all amplifier tests.

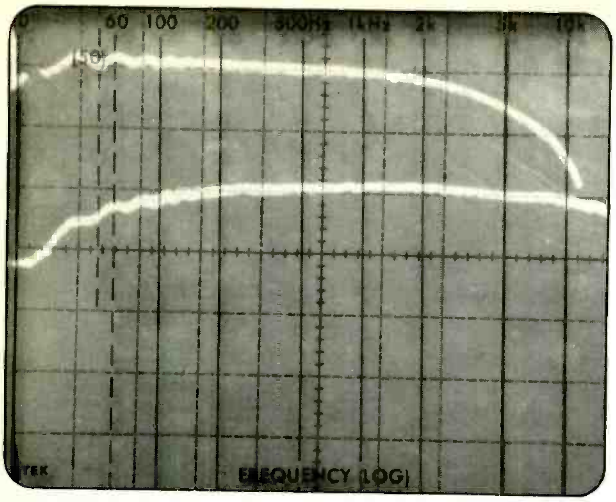


Fig. 3: Cerwin-Vega A-400: Spectrum analyzer display of two tone (19 & 20 kHz) test signal and IHF IM distortion components at 17 kHz and 16 kHz. Components above 20 kHz may be ignored in calculating IHF-IMD.

Individual Comment by N.E.: More and more—judging from our continuing tests of recent amplifiers—the dividing line between "professional" and "home audiophile" in this product category is becoming fuzzy indeed. The performance of the Cerwin-Vega A-400 certainly puts it in the top class of amplifiers now available, regardless of what "application category" is nominally intended. While the actual performance of the A-400 leaves nothing to be desired for a high-powered audiophile system, the unit obviously has been built with an eye to professional standards—the kind of circuit boards found inside, for instance; or the extra pair of output devices that are designed to permit efficient operation under various loads at high output levels: the elaborate self-protection system built into the unit. Of course, the manufacturer cannot foresee every possible chance for misusing speakers with this amplifier that may be incapable of absorbing all the power that the A-400 is capable of delivering. For this reason, they wisely counsel the owner to fuse speaker lines, or to use some form of peak limiting unit.

CERWIN-VEGA A-400 AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Continuous power for rated THD, (W)(8-ohms, 1 kHz)	225	263.5
Continuous power for rated THD, (W)(4-ohms, 1 kHz)	350	390.0
FTC rated power (20 Hz to 20 kHz) (W)(8 ohms)	225	235
THD at rated output, 1 kHz (8 ohms)(%)	0.03	0.0055
THD at rated output, 1 kHz (4 ohms)(%)	0.03	0.015
THD at rated output, 20 Hz (8 ohms)(%)	0.03	0.008
THD at rated output, 20 kHz (8 ohms)(%)	0.03	0.025
IM Distortion, rated output, SMPTE (%)	0.03	0.015
IM Distortion, rated output, CCIF (%)	N/A	0.0014
IM Distortion, rated output, IHF (%)	N/A	0.0326
Frequency Response @ 1 W, Hz-KHz (for −1 dB)	7-100	7-100
S/N ratio, re: 1W out, 0.5 V IN, "A" weighted, IHF (dB)	90	86
S/N ratio re: rated output, "A" weighted (dB)	113	114
Dynamic headroom, IHF (dB)	0.82	1.0
Damping factor, @ 50 Hz	250	250
IHF input sensitivity re: 1 watt out, (volts)	0.095	0.1
Input sensitivity re: rated output (volts)	1.4	1.5
Slew rate (volts/microsecond)	More than 80	Confirmed
Power consumption, idling (watts)	115	120
Power consumption, max (watts)	1200	1220

MXR Flanger / Doubler

By Jim Ford and John Murphy

The Flanger/Doubler from the MXR Professional Products Group is a new product which will process audio signals through a wide variety of time delay effects. In addition to a full range of flanging and doubling effects the unit will create vibrato and chorus sounds as well as hard reverberation. It is rack mountable and has both line and instrument in/out connections making it easy to use both on stage and in the studio. The Flanger/Doubler employs analog time delay technology (as opposed to digital technology) to achieve its purely electronic time delay. The list price of the unit is $425.

Flanging and Doubling

Before proceeding with the product review, let's discuss flanging and doubling effects. These two effects sound quite different even though the electronic methods used to produce each of them are similar.

The flanging effect was originally created using two open-reel tape recorders. The signal to be flanged was input to each machine and the tape outputs of the machines were then added together to produce the flanged signal. The flanging effect was then created when, with both machines running, the operator used his thumb to apply pressure to the flange of the supply reel on one of the tape recorders. This would introduce a varying time delay through that machine while the speed of the second machine remained constant, and when the outputs of the two machines were summed phase cancellations would result. It is easy to show mathematically that the two signals when summed produce a series of sharp notches in the frequency response of the system. It also can be shown that the series of notches results in a corresponding series of pass bands with center frequencies which form a harmonic series. (This harmonic series of the filter pass bands is much like the harmonics of an individual musical tone, such as a single note from a piano. This may account for the musical character of the flanging effect.) The frequency of the first pass band in the series depends directly on the amount of time delay. To make the first pass band fall at 1 K Hertz a delay of 1 millisecond (1/1000 of a second) is required. Increasing the delay time moves the peaks proportionately lower in frequency, shortening the delay raises the frequencies of the peaks. Thus, by varying the amount of pressure applied to the flange of the

tape reel (and therefore the time delay) the operator can sweep the series of notches and harmonic pass bands through the audio frequency spectrum. This imparts a unique sound effect on the signal passing through the tape machines, something like a gentle swishing sound.

To achieve the flanging effect electronically requires two signal paths, just as with the tape recorders. The first path is direct (undelayed), whereas the second path is through a variable time delay. Adding the direct and delayed signals results in the series of notches and pass bands (also known as a "comb filter" because a frequency response plot of the filter resembles the teeth of a comb). If the amount of time delay is controlled by an oscillator then the notches can be made to periodically sweep up and down the audio spectrum. The width and speed of the sweep can be put under user control.

Doubling refers to an effect where a single instrument (or voice) is made to sound like two instruments playing in unison. As with flanging, doubling was originally performed using a tape machine. The process is known as "double tracking" and is accomplished by recording an instrument playing the same part on another track. On playback this sounds like two separate instruments playing in unison. Until electronic doublers became available this effect was restricted to studio use on recorded material.

Electronic doublers provide an effect similar to double tracking and can be easily used on stage. In these units the signal passes through direct and delayed paths which are then summed as with the flanger. The main difference is the amount of time delay used. The doubler uses time delays of about 35 milliseconds whereas the flanger uses delays more like 1 millisecond. Simply delaying the signal does not however, provide a very convincing doubling effect. Slowly sweeping the delay time over a small range introduces pitch and timing errors which more accurately model two separate instruments playing in unison and improves the realism of the electronic doubling effect.

General Description: The Flanger/Doubler is basically a time delay unit with provisions for sweeping the amount of time delay over two distinct ranges. For flanging the unit uses time delays over the range .25 to 5 milliseconds. As a doubler the time delay ranges from 17.5 to 70 milliseconds. The device also has a regeneration control which directs some or all of the delayed signal back to the input to produce hard reverberation effects along with the flanging or doubling effect.

The front panel contains all the functional controls as well as the instrument level in/out connections (¼-inch phone jacks). To the far left is a power on/off pushbutton switch with a red LED pilot indicator just above it. To the right of the power switch is another pushbutton switch for selecting either flanger or doubler delay ranges. Next there is a rotary control labeled "Manual" with two LEDs above and to either side of the control. This control determines the delay time in conjunction with the sweep control. The LEDs provide visual indication of the status and speed of the sweep. At the right of the "Manual" control are two controls for sweep width and sweep speed. These controls are associated with an oscillator which directly varies the amount of time delay employed. When the sweep width control is at 100% then the manual control is disabled and the delay time is varied over the entire range for the mode selected. With the sweep width control at 0% there is no sweep action and the time delay depends solely on the setting of the manual control. With the width control at intermediate settings the delay time depends partly on the sweep oscillator and partly on the setting of the manual control. This allows for very subtle as well as very dramatic flanging effects. The sweep speed control determines the oscillator's sweep rate and is labeled .03 and 20 Hz at its slow and fast extremes, respectively.

Continuing to the right across the front panel, the next control is a continuously variable rotary labeled "mix." The left extreme is labeled "Dry" and the right extreme is labeled "Delay." Through its use the output of the unit is panned between the undelayed (dry) signal and the delayed signal. It allows the user to mix in the amount of effect desired. In the flanger mode it determines the depth of the comb filter notches whereas in the doubler mode it determines the balance between the original and the second ("double") signal.

Next is the regeneration control labeled "0% to 100%." As this control is turned clockwise more and more of the delayed output is fed back to the input and a hard reverberation results.

Just to the right of the regeneration control is a pushbutton labeled "Invert." Pushing this button causes the phase of the delayed signal to be inverted (shifted 180°) before it is sent to the mix control. This feature is primarily used in the flanging mode where it interchanges the peaks and notches and cancels the frequencies below the first notch. It provides a variation on the flanger effect.

The final control on the unit is a pushbutton labeled "In/Out" which allows the device to be completely bypassed when the button is out. That is, the output jacks are tied directly to the input jacks when the button is out. This assures that when the unit is bypassed, it contributes no noise or distortion to the signal. A LED above the button indicates when the unit is "in."

At the far right of the front panel are a pair of ¼-inch phone jacks for musical instrument input and output. These are intended for inserting the Flanger/Doubler in-line between a musical instrument (electric guitar for example) and an amplifier. On the left side of the rear panel are another pair of phone jacks for the line in/out connections. These are intended to be used with higher level signals (such as studio line-level signals at +4 dBm) and will handle higher level signals before clipping. Beside the line in/out jacks on the rear panel is another ¼-inch phone jack marked Bypass. The unit can be bypassed remotely by connecting a switch to this jack. A typical application would be to plug in a foot switch so that the effect could be switched in and out by a musician performing on stage. The remote bypass is only operative when the delay is switched "in" on the front panel.

There are also phone jack connections on the back panel for external voltage control of the delay sweep oscillator. This allows the voltage control signal from one flanger to drive one or more slave flangers. It also allows the delay time to be varied remotely, such as from a foot pedal on stage.

The unit is attractively and ruggedly packaged, and to make the Flanger/Doubler even more road-worthy MXR offers an optional road case.

Field Test: We used the Flanger/Doubler in a recording session on several different instruments and voices. We especially liked the flanging effect on electric piano and both acoustic and electric guitars. It adds a richness to the sound that is very appealing. We used the Doubler on voice and found the effect appealing, but it didn't exactly convince our ears that we were hearing two separate performances in unison. We still prefer double tracking on the multi-track tape machine to electronic doubling. However, for stage use (where you can't do double tracking) the Doubler would probably provide a very satisfactory effect.

The controls were found to be quite satisfactory in their action and range of control. The "Mix" control for example, makes it easy to add just the degree of effect desired.

Adding a bit of "regeneration" while in the Flanger mode introduces an interesting "soaring" effect, while using regeneration in the Doubler mode adds hard reverberation which we judged to be less useful.

When we inserted the Flanger/Doubler in our reference system for listening tests we found the audio quality through the "dry" channel to be very good. There was no audible noise or distortion and the only difference we could perceive when the unit was alternately switched in and out of the system was a slight loss of sharpness on some transient sounds. Listening through the delay channel things were a little different. With the "mix" control set all the way to "Delay" and the sweep width set to 0% we were able to listen through the delay channel with no effects other than a short time delay. When we alternately switched the unit in and out of the system we noticed a moderate loss of highs with the unit switched in. The delay channel also seemed to introduce some degradation of the audio quality when listening to good disc recordings with the unit switched in.

Lab Test: When we took the unit to the lab, we ran the usual tests for both the direct and delayed signal paths. The Lab Summary section contains the test results for the Flanger/Doubler.

The overload levels for both line and instrument inputs indicate adequate headroom for most signal levels. However we were a bit surprised that there were no level controls or overload indicators. We would suggest that even a single LED overload indicator would be of great value to the user in avoiding overdriving the unit.

Noise levels were quite low, even through the delay path. For most signals the dynamic range of the unit is about 90 dB for the direct path and about 80 dB for the delayed path. This is very good performance for an analog time delay unit.

Distortion through the direct path was very good, but distortion through the delay path was on the order of 1% or so. In the delay path the distortion increased at higher frequencies and reached as much as 42% for the Doubler at the maximum delay using 10 kHz test tone. The distortion products in this case were not harmonic but rather, consisted of a low-frequency tone which varied in frequency as the amount of time delay was varied. With less than maximum delay time settings the distortion was more like 1%.

Frequency response through the direct path was very good. However, high-frequency response through the delay path is compromised to some extent. We should remember that the Flanger/Doubler is usually used with a *mix* of direct and delayed sound. This tends to make the compromises in the quality of the delayed signal less significant than if the delayed signal were used by itself.

MXR provides an excellent operation manual with

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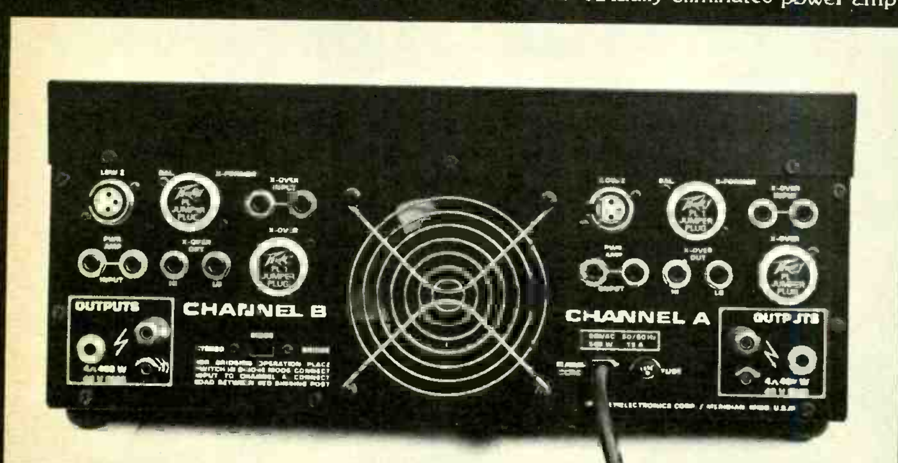
Our new CS-800 has been refined to offer new features such as DDT compression, more advanced circuitry, and expanded patching facilities, at only a very slight increase in price, making it an even greater value than before.

Take, for example, Peavey's unique Distortion Detection Technique® (DDT) compression circuitry. DDT® electronically senses the onset of amplifier clipping and engages a specially designed circuit that virtually eliminates power amp clipping, thus greatly decreasing the possibility of square waves reaching the speakers. Not only does this feature offer maximum protection for your speakers, DDT® enables the total system to enjoy freedom from most of the commonly encountered headroom problems with power amplifiers. This compression feature may be easily defeated from the front panel by built-in switches on each channel.

The CS-800 features an improved patching system with provision for electronic crossovers and transformer balanced inputs for each channel. The convenience of the rear patch panel combined with the optional, low cost plug-in accessories give the CS-800 versatility that is unmatched by other professional quality power amps.

Our competitors will be advertising and displaying their new units soon. While you're looking at theirs, we'd like to invite you to compare specs and features with ours. You'll see why the Peavey CS-800 is still one step ahead.



Performance Specs:

Frequency Response:
+0, -1 dB 5 Hz to 50 kHz (1W, 8 Ohms)

Power @ Clipping:
400 W RMS per channel into 4 Ohms
800 W RMS bridge mode into 8 Ohms

IM Distortion:
Less than 0.1%, typically below .04%

Total Harmonic Distortion:
Less than .05%, typically below .02%

Damping Factor:
Greater than 200

Input Sensitivity:
1.3 V for 400 W into 4 Ohms

Load Protection:
Short, mismatch, open circuit proof voltage/current limiting instantaneous with no thumps or cut out.

Speaker Protection:
Instantaneous crowbar circuit clamps the output upon advent of amplifier failure.

Cooling System:
2 speed forced air cooling over massive aluminum heatsinks

Output Transistors:
20, 4 drivers (TO-3)

PEAVEY ELECTRONICS CORP.
711 A Street/Meridian, MS. 39301

CIRCLE 89 ON READER SERVICE CARD

the Flanger/Doubler. The manual supplies detailed information on installation and operation, along with a recommended control setting for various effects in an applications section. The manual also provides an informative description of the circuitry and discusses the theory of operation.

Conclusion: The MXR Flanger/Doubler can be used to produce a wide range of effects, either in the studio or on stage. The controls are easy to use and they are good visual indicators as to the status of the delay sweep and bypass. If you're looking for a new sound toy we recommend you give it a listen.

LAB TEST SUMMARY

(Note: All signal levels are referenced to 0 dBV=.775 Vrms)

Line Input and Output Levels at Clipping

Direct Path:	7.3 Vrms (+19.5 dBV)
Delayed Path:	2.1 Vrms (+8.6 dBV)—Flanger
	3.5 Vrms (+13.1 dBV)—Doubler

Instrument Input and Output Levels at Clipping

Direct Path:	1.84 Vrms (+7.5 dBV)
Delayed Path:	.53 Vrms (−3.3 dBV)—Flanger
	.89 Vrms (+1.2 dBV)—Doubler

Noise Levels at the Line Output (20 kHz Bandwidth)

Direct Path:	−78 dBV
Delayed Path:	−74 dBV—Flanger Min Delay
	−68 dBV—Flanger Max Delay
	−71 dBV—Doubler Min Delay
	−66 dBV—Doubler Max Delay

Noise Levels at the Instrument Output (20 kHz Bandwidth)

Direct Path:	−86 dBV
Delayed Path:	−85 dBV—Flanger Min Delay
	−79 dBV—Flanger Max Delay
	−83 dBV—Doubler Min Delay
	−77 dBV—Doubler Max Delay

Distortion (THD) through Line IN/OUT at 0 dBV

	100 Hz	1 kHz	10 kHz
Direct:	.018%	.018%	.018%
Delayed:	.46%	.79%	4.4%—Flanger Min Delay
	.22%	.18%	1.74%—Flanger Max Delay
	.26%	.46%	2.75%—Doubler Min Delay
	.33%	1.13%	39.5%—Doubler Max Delay

Distortion (THD) through Instrument IN/OUT at − 10 dBV

	100 Hz	1 kHz	10 kHz
Direct:	.014%	.015%	.029%
Delayed:	.60%	1.02%	4.4%—Flanger Min Delay
	.24%	.20%	2.3%—Flanger Max Delay
	.35%	.63%	3.1%—Doubler Min Delay
	.42%	1.55%	42.0%—Doubler Max Delay

Bandwidth (− 3 dB points)

Direct:	13 Hz	27 kHz	
Delayed:	14 Hz	7.5 kHz	—Flanger Min Delay
	13 Hz	11.5 kHz	—Flanger Max Delay
	17 Hz	11.5 kHz	—Doubler Min Delay
	17 Hz	11.8 kHz	—Doubler Max Delay

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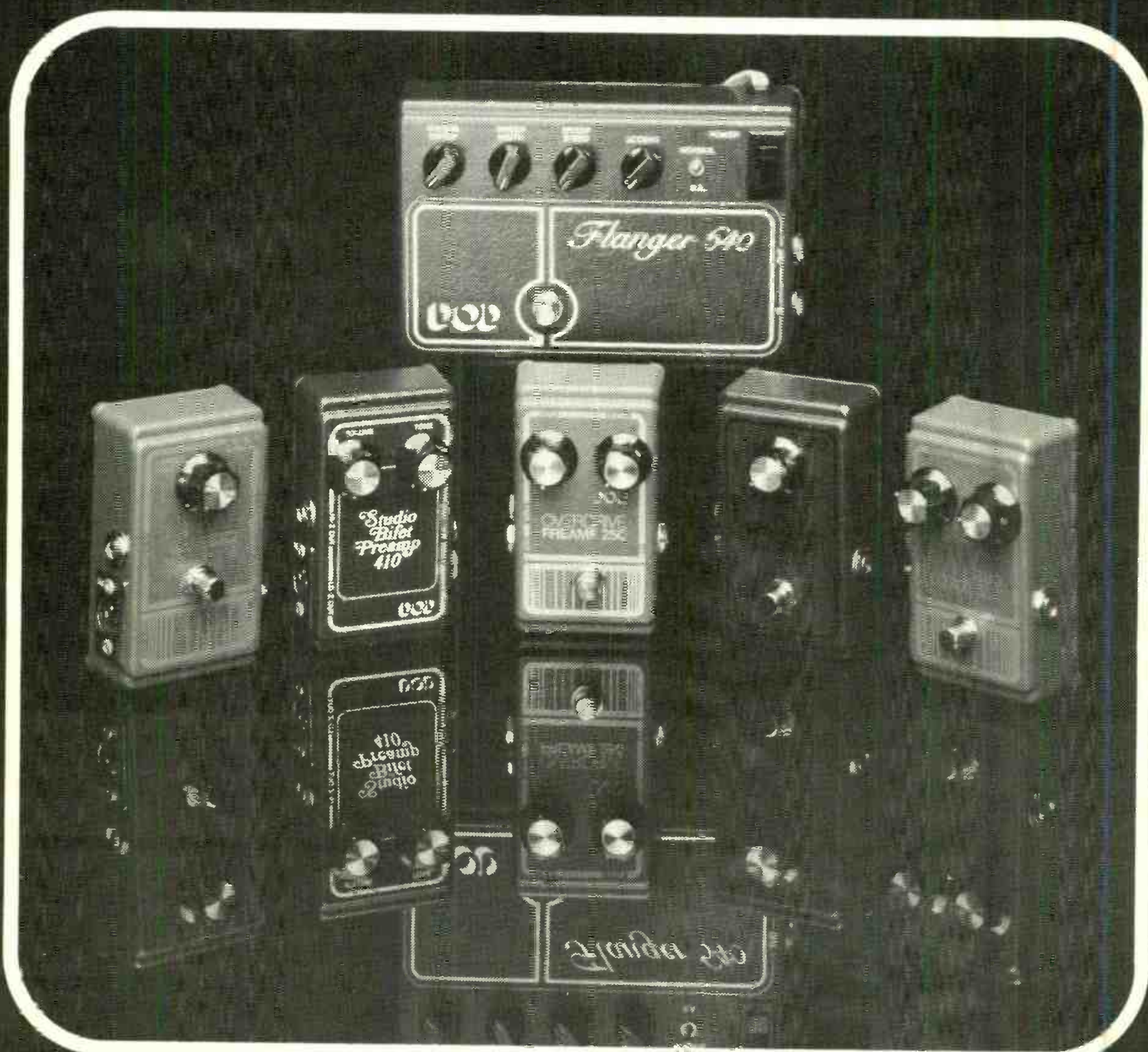

When we design a product we keep the musician in mind all the way. We demand a product that is functionally superior and to ensure that it will stay that way we use rugged name-brand components such as CTS, Carling Switch, National Semiconductor, Texas Instrument, Switchcraft, etc. We also feature solid Die-cast Zinc and Aluminum cases and FR-4, G-10 Glass epoxy circuit board. All our products are hand assembled and individually tested by qualified technicians and have a full one-year warranty on parts and labor.

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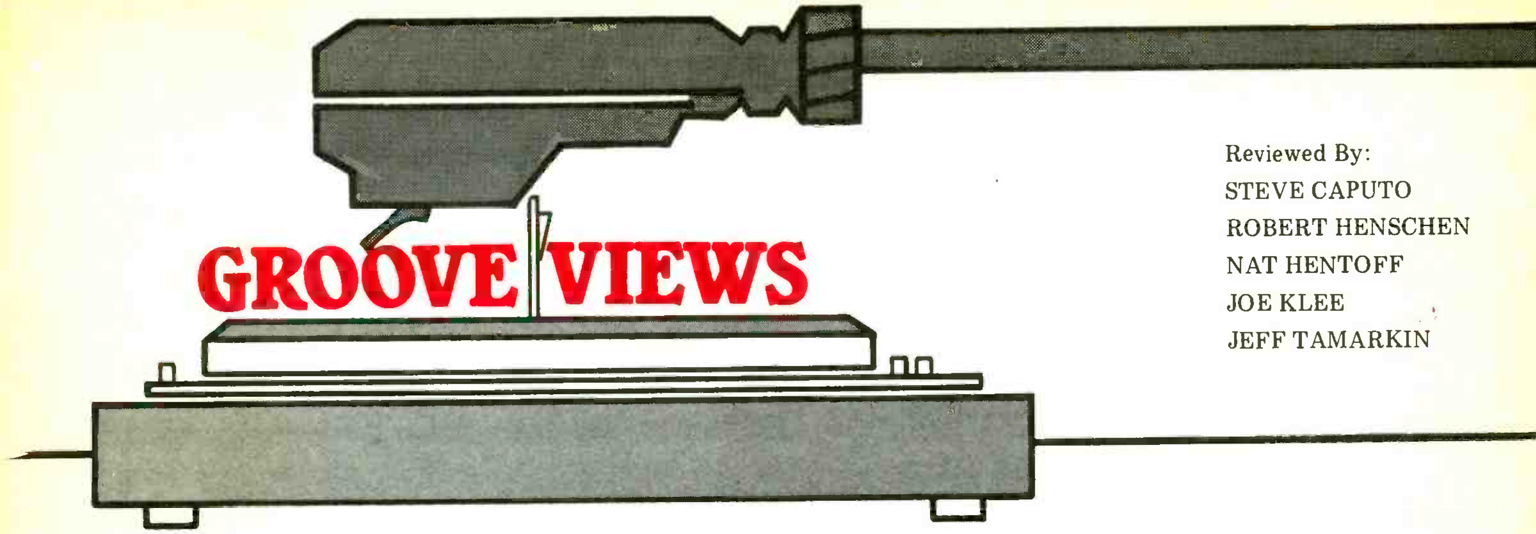

Reviewed By:
STEVE CAPUTO
ROBERT HENSCHEN
NAT HENTOFF
JOE KLEE
JEFF TAMARKIN

POPULAR

DEVADIP CARLOS SANTANA: *Oneness—Silver Dreams Golden Reality.* [Devadip Carlos Santana and Glen Kolotkin, producers; Glen Kolotkin, engineer; recorded "live" in Osaka, Japan and at The Automatt, San Francisco, Ca.] Columbia JC 35686.

Performance: **Devotional strengths, secular weaknesses**
Recording: **Diverse adequacy**

Prayer bell chimes, a softly chanted "Om," and serene gong open this album's "The Chosen Hour," and hymnlike organ/synthesizer chordings are added for "Arise Awake." From the Buddhistic album cover down to the pious themes of every *Oneness* song, the sole (soul?) purpose here is religious expression. But if that sounds like a limited scenario, then you haven't heard Santana when they're inspired.

For several years now, Devadip has been at his best when he departs on tangents from his salsa-funk-rock norm. Blatant commerciality isn't necessary for this man—he's almost always on fire—but the tendency to formulate and stagnate has been haunting albums since *Amigos* (Columbia, 1976). Just prior to that, Santana had gone through his most fertile creative period, getting into Eastern religion, experimenting with the jazz of John Coltrane, and turning out such excellent albums as *Welcome*, *Borboletta*, and *Love Devotion Surrender.*

That the spiritual connection is no absolute guarantee of musical perfection was evidenced by Devadip's collaboration with Turiya Alice Coltrane on *Illuminations* (Columbia, 1974), an ambitious project that alternately rose and fell to the heights of eternal bliss and the depths of near boredom. But *Oneness* is radically different and constantly changing. The opening theme, for instance, quickly adds rock drums and electric guitar, builds on a hefty bass figure, then cranks at high energy levels—almost the old Mahavishnu format. A jazzy remake of Alan Hovhaness' "Mysterious Mountain" climaxes in a dialogue between Santana's guitar and Tom Coster's piano. A high energy approach to Chico Hamilton's "Jim Jeannie" gives drummer Graham Lear a chance to sizzle and provides jazzlike solo room for the other musicians.


SANTANA: A prolific flow of powerful, substantial music

Side one momentarily hits the skids with indulgent strings and a sappy vocal by Saunders King, but the quality control returns with side two. "Oneness" is an excellent mixture of slow rock and tense mystery. "Life Is Just A Passing Parade" gets a funkier lead vocal out of Greg Walker, and "Golden Dawn" is one of Santana's rare acoustic guitar spots, an impressive combination of Spanish, classical and blues licks. "Free As The Morning Sun" starts off acoustic too but evolves into a very strong Latin jam. A weak poetry recitation interrupts the mood, then Narada Michael Walden's "Song For

Devadip" closes the proceedings on an upbeat rock note.

The flaws here may seem fairly serious to a general listnership. There's enough imagination in the instrumental sections of the album to sustain longterm interest, but in contrast to strong performances by keyboardist Coster, drummer Lear, and a percussion ensemble that includes Pete Escovedo and Armando Peraza, the vocal material comes off sounding amateurish and doctrinal. Ironically, Santana's intent to communicate through language interferes with a much more prolific flow of powerful, substantial music. R.H.

BRIAN ENO: *Music For Films.* [Brian Eno, producer; no engineer listed; no recording location listed.] Antilles AN 7070.

Performance: **Pleasant and uplifting**
Recording: **Unobtrusively hypnotic**

The first time I played *Music For Films*, I used it as background music while I read. I kept the volume on low because my girlfriend was sleeping nearby. When she awoke, she looked up, heard the ethereal waves of Eno's synthesizer floating through the room, and asked quietly, "Am I in heaven?" Although it was funny at the time, she may have extracted the essence of this album in her initial reaction to it.

BRIAN ENO: No clues here

This all-instrumental Eno recording of music composed for use in films could indeed be described as heavenly. On the other hand, it serves as perfect background music. Unlike Eno's solo vocal work, or his much-heralded production with Devo and Talking Heads (or the grating, progressive/new wave collection *No New York*), the pieces on *Music For Films*, which fall between light classical and muzak, do not demand attention. The single notes and tones, which beautifully connect together, often become so secondary that they blend into the decor of the listening room, and only become noticeable again when the shock of the record ending and the turntable shutting off put an abrupt end to Eno's mood music.

Although there are individual tracks listed, any titles would be insignificant for the purpose of reviewing this record. The segues from one track to the next are so smooth and controlled, that the record is more a symphony with several movements than a collection of individual songs of any sort. In fact, the only apparent reason for dividing this music into names would be to distinguish the points at which the various guest musicians appear. And there *are* some high priests of prog-rock putting in time on Eno's most understated recording to date. Guitar heroes Robert Fripp and Fred Frith, percussionist Phil Collins of Genesis, and John Cale are all listed as contributors, but none are dominant in any sense. Compared to the collaborations between Fripp and Eno in the past, this is hardly as monumental a meeting, as there is little to show that anyone new has entered the picture here. The recording is that low-key.

Basically, the music on this release consists of ripples of sound, primarily coming from Eno's synthesizers, building on top of, and then replacing one another. It is not mentioned which films these recordings were used for, but it would appear that they could have been placed in anything from "*Star Wars*" to a James Bond flick. Probably, they were used for less commercially-oriented experimental ventures, but that too is insignificant. Perhaps the best film that this LP can serve as the soundtrack for is the film that the listener is acting out at the time the disc is playing. It's perfect for just about any activity where intelligent muzak would aid the ambiance, and though the production here will not offer a clue as to the genius of Eno's work with rock'n'roll, it shows another side which demands, in the long run, an even greater sense of imagination and the application of recording techniques. J.T.

JAZZ

HILARY: ***Just Before After Hours.*** [Wayne Henderson, producer; Alan Sides, Reggie Dozier, Michael Evans, engineers; recorded at Oceanway Studio, Los Angeles, Ca.] Columbia JC35547.

Performance: **Outstanding**
Recording: **All out**

Hilary is only 28 years old, female, and has been playing flute and soprano saxophone for just nine years. Yet, in spite of these "handicaps," her album, *Just Before After Hours*, is as fine a collection of music as I've heard from any new artist in quite some time. Her sound is a powerfully infectious brand of funky jazz in which her self-acclaimed jazz, classical, Latin, R & B, and funk influences are clearly evident. Tunes like "Do it," "Sundancers," and "Amazona" have that lively Latin feel to them, and always remain lighter than air. They are more than adequately handled, not only by Hilary, but by every musician involved. Hilary slips into a funky sound with "Wanderer" and the title cut, which at times, brings Tom Scott or Chic Corea's *Romantic Warrior* to mind. And, as if to show that she and her instruments are capable of expressing the spectrum of emotions, Hilary brings the "Evening Essence" alive with strings, piano, and a haunting Tim Weisberg type flute arrangement. "So in Love with You," a Wayne Henderson, Richard Flowers composition, is a typical mellow Crusaders' piece, done up nicely here. Finally, Hilary looks to heaven with "Reach for the Stars," a dreamy number on soprano sax.

One reason Hilary sounds so strong on her debut album, is the solid work of Crusader Wayne Henderson, as producer. With instruments like the flute or soprano saxophone, that lend themselves very easily to production

Two Grand Masters: Duke Ellington and Charles Mingus

By Nat Hentoff

While practically no other jazz leader ever went into a recording studio unless the date had been commissioned by a record label, Duke Ellington quite often made—and paid for—his own sessions. He might have been between labels and expected to license the masters later. Or he simply wanted to hear and study some of his new compositions. Still another reason, as Stanley Dance, his friend and historian has noted, was to get some return from his expensively contracted musicians during a period when they had no other engagements.

Gradually, a number of these dates, unlisted so far in the discographies, have been turning up. Some on Norman Granz's Pablo label, and now, *Unknown Session* on Columbia. Made in 1960, the set is doubly rare because Duke leads a septet rather than the full band. (The voicings, however, are such that this is the biggest-sounding septet I've ever heard). The repertory consists largely of Ellington standards: "Black Beauty," "All Too Soon," etc. The playing is extraordinarily relaxed, yet abounding in vitality. Because it's a combo, everyone has the space to stretch out; and there are wondrously inventive solos by the leader (who usually limited his improvising on orchestral dates); Johnny Hodges; Ray Nance; Harry Carney; and particularly trombonist Lawrence Brown, the most flowingly romantic trombonist in jazz history.

The sound quality, in an unnamed Hollywood studio, is among the best ever accorded an Ellington date—vibrant presence and flawless balancing of solos and ensemble. Moreover, Ellington's piano is as incisive and full-bodied as on those "live" performances that for so long were among the most deeply satisfying pleasures in jazz.

Though himself an unmistakable original, Charles Mingus credited Duke as one of his key formative influences; and in time, Mingus produced a body of composition second only to Ellington's in range and imagination. A most valuable part of Mingus's canon was a series of 1959 sessions, most of which have been hard to find and four of which were never issued at all. In *Nostalgia in Times Square*, a two-LP Columbia album, all have been resurrected; and indeed those previously released are now heard for the first time in full-length unedited versions.

The pieces encompass much of the history of jazz, from "Jelly Roll" and "Boogie Stop Shuffle" to "Open Letter to Duke" and a number about Charlie Parker, "Gunslinging Bird." The impassioned interpreters include some of Mingus's bravest sidemen (Booker Ervin, Jimmy Knepper, Horace Parlan, and the indispensable Dannie Richmond, a drummer who could anticipate everyone of Mingus's surprises.)

Engineering on all these dates was sensitive to Mingus's broad scope of dynamics and the need to capture the insistent passion of the music. Mingus himself, by the way, is in towering, boldly venturesome form. A special nod to musician Sy Johnson's liner notes which are unfailingly informative to the point where they form one of the most illuminating essays on the essence of Mingus's leapingly high art.

DUKE ELLINGTON: *Unknown Session.* [Irving Townsend, original producer; Michael Brooks, reissue producer; no information on original engineer; Frank Abbey, reissue engineer.] Columbia JC 35342.

CHARLES MINGUS: *Nostalgia in Times Square.* [Teo Macero, original producer; Jim Fishel, Frank Abbey, reissue producers; no information on original or reissue engineers.] Columbia JG 35717.

techniques like echo, reverb, or overdubbing, there is a tendency to overproduce, burying any talent the artist might have. But a veteran jazzman like Henderson realizes this, and keeps the production simple, allowing the music to attain its own level of complexity. The sound of the album is tight, clean, and crisp. Henderson has achieved a good blend of the session musicians with Hilary's sound. Even the occasional vocals are used as instruments. But Henderson skillfully moves the spotlight from Hilary's flute or sax to, say, keyboards or percussion, in expanding the music, and this is what strengthens the sound.

An artist can only be as successful as the sound of his or her record, and Wayne Henderson has done a fine job bringing Hilary alive on this album. *Just Before After Hours* is a well-paced album from song to song, and this intensifies Hilary's ability to express a multitude of feelings. And for all her "handicaps," Hilary may just be the one to bring us back to what jazz is all about—feelings. Because, as jam-packed with talent as Hilary is, for a 28-year-old female with only nine years experience to have created as powerful an album as this one, it has to contain an incredible amount of feeling. S.C.

JOE VENUTI: *The Best Of Joe Venuti.* [Hank O'Neal, producer; Fred Miller, engineer; recorded at Downtown Sound, New York, 9/27/73, 5/20/74 and July, 1975.] Chiaroscuro CR 203.

Performance: **Certainly the best**
Recording: **Up and down**

JOE VENUTI: *Violin Jazz 1927 to 1934.* [Nick Perls, reissue producer and engineer; no recording locations given.] Yazoo 1062.

Performance: **Certainly the best**
Recording: **From crude to almost Hi-Fi**

Let's begin with the fact that Joe Venuti is the best violinist who ever played in the jazz idiom. I know that's just my opinion but it's an opinion that I've held most, if not quite all, of my life. I couldn't have been more than seven or eight years old when I heard my first Joe Venuti record and I've been collecting them ever since. There's something incredible about this

JOE VENUTI: Swings and plays with the strength and imagination of a genius

immigrant (don't believe those stories about him being born on the boat) from Lecco, near Milan, Italy and who played hot jazz like he was born in New Orleans. Actually, according to the best estimates, Joe was born one year before Louis Armstrong, so if he had been born in New Orleans nobody would have been too surprised that he was such a good jazz fiddler. That such a great improvisational jazz virtuoso can come from Italy is proof of the universality of the music.

These two reissues concentrate on totally different periods in Joe's career. The Yazoo LP reproduces sides from Venuti's period of highest popularity. During the '20s and '30s, it was not unusual for Joe to do two or three record dates on the same day—scurrying from one to another as energetically as he played. All but three of these cuts feature Venuti in the familiar company of his guitar-playing pal Eddie Lang all the way from duets like "Goin' Places" to the larger, more orchestrated cuts such as "Jig Saw Puzzle Blues." Whatever the setting Venuti swung mightily and played with the strength and imagination of a true giant.

The Chiaroscuro release covers a shorter timespan from 1973, shortly after Joe's re-emergence from obscurity, until 1975, only three years before his death. It was during this period that Joe took to making records with Zoot Sims and a rhythm section for Hank O'Neal's thriving Chiaroscuro label. The rhythm section nearly always had Cliff Leeman or Bobby Rosengarden on drums, Milt Hinton or George Duvivier on bass and, in this case Dick Hyman, Dick Wellstood or John Bunch on piano.

In the almost forty years between the latest of the recordings reissued on Yazoo and the earliest of the recordings on Chiaroscuro, it would sound as if Joe Venuti had undergone a rather drastic change. It was not a change in rhythmic intensity so much as it was a change in rhythmic feel. Joe still swung, but he swung in a different way. It wasn't even so much a difference in Joe's playing as it was a tribute to his flexibility. The '20s and '30s rhythm sections were a two-beat affair with Adrian Rollini supplying the "oomphasis" with his bass saxophone. When the string bass (especially when bassists like Milt Hinton were present) or drummers like Cliff Leeman took over, the time changed from two-four to four-four and the music took on a more laid-back attitude. Joe could cut it either way. If there's an advantage to the Yazoo over the Chiaroscuro, it's the inclusion of several lesser known alternate takes since the Chiaroscuro sides are made up of only issued takes from the albums that Joe and Zoot did for the label. The Chiaroscuro advantage is the presence of Zoot Sims, surely a giant on the tenor sax, and the way he and Venuti interact.

The recording quality is variable on both discs—being at its best in the 1934 "Satan's Holiday" recorded in England for Regal Zonophone and at its worst, surprisingly, on some of the Chiaroscuro sides, where it sounds like drummer Cliff Leeman is right on top of the mic.

J.K.

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