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SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

VOL. 5 NO. 3
DECEMBER 1979

ON TOUR WITH GEORGE BENSON

THE ELECTRIC PRIMER --PART III

Lab Reports:

Hafler DH-200 Power Amp

Phase Linear 7000 Cassette Recorder

Pioneer RT-909 Open-Reel Recorder

HANDS-ON REPORT:

H.H. Electronics 16;2 Mixer

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RECORD REVIEWS

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CIRCLE 71 ON READER SERVICE CARD

MODERN RECORDING

DECEMBER 1979
VOL. 5 NO. 3

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

THE FEATURES

COMING NEXT ISSUE!
A Session with Bob Welch
The Technology Behind
Apocalypse Now

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SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

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West Coast
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MYLES GROSSMAN
Advertising Director

VINCENT P. TESTA
Publisher

Editorial and Executive Offices
Modern Recording
14 Vanderventer Ave.
Port Washington, N.Y. 11050
516-883-5705

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Letters to the Editor

Slaying the Disco Dragon, or, Disco Gets a Beating

First and most important, keep up the good work on this unique magazine. I am writing in response to Mr. Rapoza's September Letter to the Editor on the subject of disco. Hopefully, my imitative format will prove effective.

Second, there's current trend you fortunately seem to be missing the beat on . . . Disco. . . . There, now that I've said the dirty word, I hope to never read it again in a quality magazine like yours.

Are you aware that:

1—35-40 percent of *Billboard's* top 100 songs are judged by 45-record sales to persons between the ages of 13-18 (not exactly music connoisseurs).

2—Almost all the songs are performed by musicians who couldn't hack the competitive market of jazz or rock.

3—Musicians between the ages of 13-18 have little trouble performing the primitive art of disco (especially boys whose voices haven't changed yet).

Yes friends, rock has finally evolved itself out of traditional 4/4 time and out of the "top 40" scene, somewhat. I am a professional drummer who appreciates a challenge in music, and disco (disgusto, I call it) comes far from it (with exception of the sexless vocals).

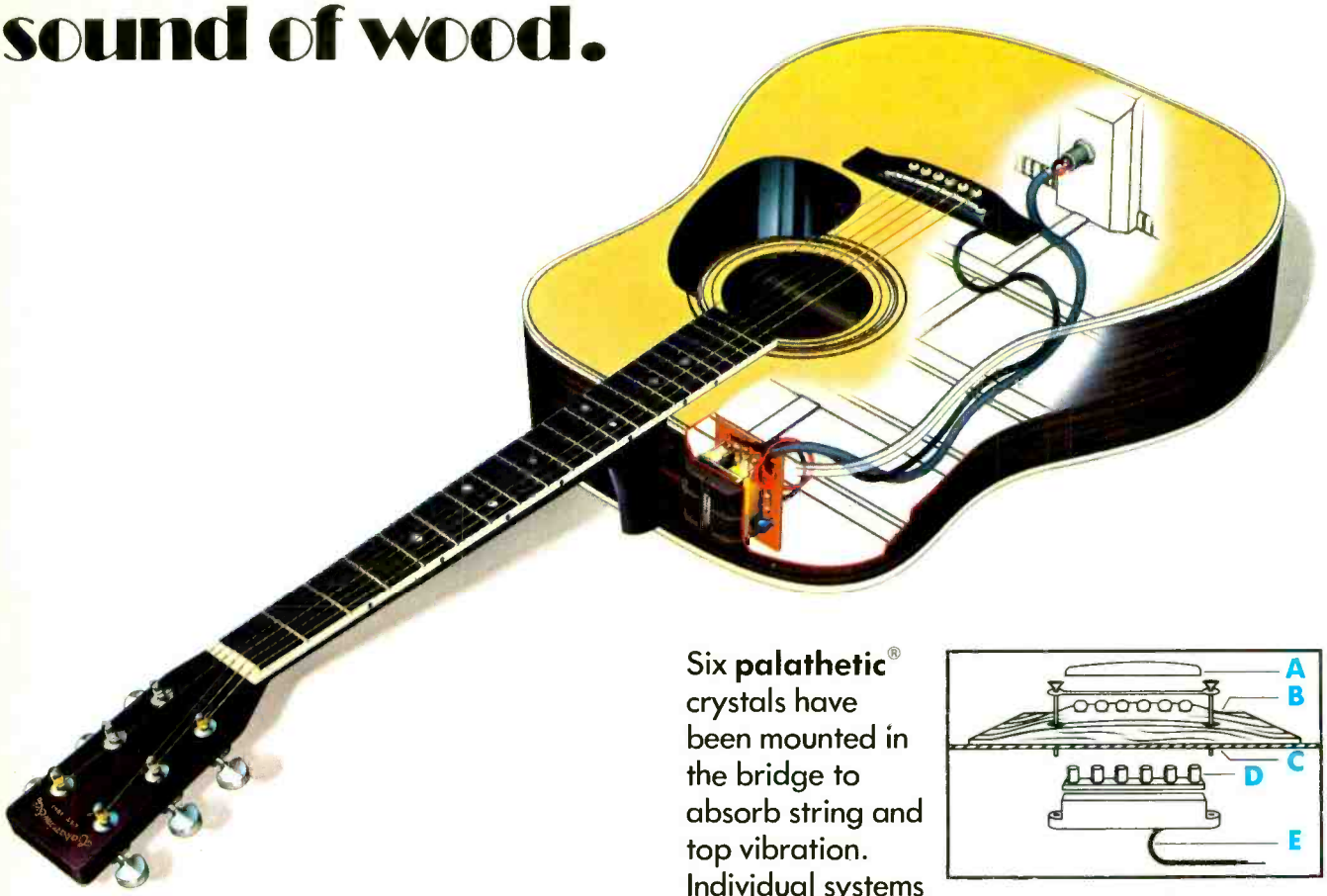
Just the thought that a major record company will give a 24-track tape to a disco DJ and leave him alone in the studio to do with it as he will indicates to me that the engineer wants nothing to do with it.

Why am I and so many other musicians so hostile towards the subject? First, half the clubs in the country that used to support "live" entertainment are now spinning records. The recession of '79 is hitting the entertainment business hard enough without multiplying the problem by closing the clubs to musicians. I am very distressed by the fewer number of young local bands forming, which is probably due to the lack of available clubs to perform in. How is the next generation of musicians going to be able to acquire stage presence and the business know-how to fight the androids behind the scenes? Perhaps they never will. It's a shame to watch in garages and basements the potential superstars of tomorrow who will probably never get a chance to perform for a large audience.

Secondly, as a drummer, I see a disco drummer's longevity as very short, since they will soon be replaced with electronic rhythm units. Electronics are more precise at making that obnoxious bass drum thump common to all disco releases. ("After all, a snare doesn't really have to sound like a snare.")

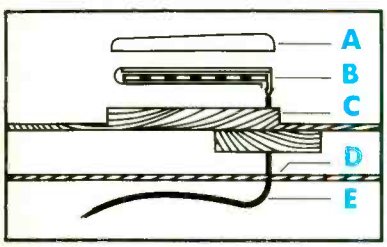
Special electronic effects and sounds in disco are generally reflected as out-of-proportional mixing to make the never-changing tempo stand out. To quote Mr. Rapoza (again), ". . . at the moment it's biggest thing happening in music . . . .". Wrong! The biggest thing happening in music today is what *Modern Recording* is all about, and *that* is the almost unbelievable technological breakthroughs in the art of recording and the major advancements in the field of electronics. Disco drummers beware, you'll be the first to go.

TAKAMINE amplifies the natural sound of wood.



For more than a decade, Takamine has created acoustically exacting instruments. The finest woods are carefully crafted into guitars which produce a pure, rich sound. The classic model features a cedar top and fan bracing pattern, while the steel stringed models employ a spruce top and X bracing pattern. Both designs project resounding highs, clear mid-range and full lows.

Now for the first time Takamine offers seven outstanding acoustic/electric models with a built-in **palathetic**® pickup system. This new design fully transmits the wood sound box's unique responsiveness and overall tone quality. Unlike other pickups that color a guitar's sound with their own properties, the **palathetic**® system projects true guitar tone.



Classic Model A. Saddle; B. Palathetic® Crystals; C. Bridge; D. Guitar Top; E. Pickup Wire

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CIRCLE 39 ON READER SERVICE CARD

Thirdly, I feel the amount of appreciation disco listeners direct toward the performers is epitomized by the lack of instrument soloing on the cuts. They listen for a beat to dance by and rely on plastic egos for entertainment.

I could go on, but your magazine space is too valuable to waste on such a void topic. People say disco won't last—I, for one (of many, I am sure), hope it doesn't, or that at least it changes into something of finer musical quality, for my current musical motto is: "Disco is to music as Etch-a-Sketch is to art!"

MR, thank you very much for the open invitation to comment.

—Randy Buchman
Musician/Music Lover
Defiance, Ohio

I was happy to receive the first issue of my subscription to your fine magazine. But I feel I must express my opinion.

I've never written a letter to any magazine before, but I feel so strongly about disco that I'm totally compelled to write. Not only do I say "no" to disco: I say "death" to disco.

Your magazine is superb in technical aspects as well as having taste when it comes to the artists you choose to examine and the ability of all your writers. So let's hope you won't waste all that on Muzak-sounding mechanical drivel. I'm not against dance music; just against the trash that money-hungry execs choose to put on DJs' playlists.

I feel you should not bend to this anti-musical trend, but that you should stimulate as well as educate newcomers to alternative music forms that they might not encounter in the wasteland of pop music.

—Danny Cowan
San Antonio, Tex.

Disco may be very exciting and creative for some—just as rock 'n' roll is for others.

What Mr. Rapoza neglected to offer in his letter is that special effects are created in rock music. It is true that sounds never heard before *are* showing up on disco records every day just as they are on discs cut by the rockers. But these sounds are credited mainly to the infamous synthesizers which Keith Emerson pioneered. It was mainly through rock also, that the inspiration of guitar, drum, and voice synthesizers as well as flangers, delays, drivers, etc. came about to alter the musician's sounds.

If I were a writer of music, I for one would surely not want anyone messing around with my ideas. If these disco DJs do what they please on the board, the end result will be entirely different in arrangement no matter how minor the alterations.

I feel the majority of readers enjoy reading the tips and equipment test articles as well as the in-depth reviews. I think everyone has a pretty good idea of how the DJs do it. Those readers or writers who want more on disco—why not try hitting the disco magazines that devote *all* their pages to your exciting and creative phenomenon for articles about yourselves and your fellow disco-lites?

—Steve Pawlowicz
member, Insane Coho Lips
Chicago, (City of Rock) Ill.

Mollusk in a T-shirt

Many thanks to you and Brad Aaron of "Kansas" for the free "plug" in the July 1979 issue of *Modern Recording*. With a name such as ours, many people wonder if Octopus Audio is a real company. Although a leader in the specification,

NOW SOUND AS GOOD AS YOU PLAY. JBL'S NEW CABARET SERIES.

Both you and your audience deserve to hear exactly what you're playing. Which is exactly what JBL's Cabaret Series is about.

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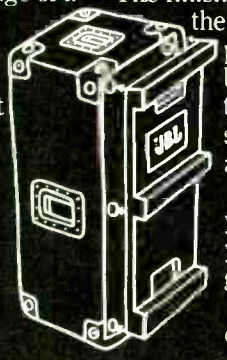

Their housings are made of the best birch ply available. The front covers fit flush to protect the drivers and grilles during travel. The finish is extra-rugged. Even the voice coils have unique protection: To minimize bounce during transit, they're automatically shorted when there isn't a plug in the input jack.

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CIRCLE 140 ON READER SERVICE CARD

supply, installation and service of professional audio in Canada, we still have the occasional identity crisis. Now, thanks to you, people will know we're legitimate (won't mother be pleased).

—R.M. Taylor
Operations Manager
Octopus Audio
Toronto, Ontario

On a Wing and a Buyer's Guide

I am once again flying the friendly skies with my copy of *Modern Recording's Buyer's Guide* (Winter, 1979) to create the illusion of a short flight.

I have been a musician for twenty years, an audio engineer for half that time, and now I am marketing American-made audio products domestically and overseas.

As you might understand, I both appreciate and enjoy the humor of "Inside a Soundman's Tool Kit," reprinted in the Guide. Much of the information you give sounds like you have been reading my mail, so to speak, but I hope some of those readers who are new to the field can easily benefit from what you and I learned from bruised and bloody knuckles.

Thank you again for a job well done, and keep up the wit and wisdom.

—Sam Goforth
Vice President
Woodel Int'l., Inc.
Temple, Tex.

No Complaints

Congrats on an excellent mag and all the rest of that stuff. You've heard it all before, and I mean it as much. Seems as good as you are, few complaints are written in, and justly so.

I'd like, though, to comment on the record review section of the magazine. Most particularly, negative reviews. At any time, there must be a hundred good records worth listening to (for whatever reasons). Let's hear about those. I do appreciate a fair warning concerning what may be a negative effort—maybe a paragraph—but I don't need close to a thousand words to tell me what I don't want to buy.

Give us something we can use. It's what we expect; it's what makes this mag the best in the music industry.

—Randal Timm
Encinitas, Ca.

In a fast flip through "Groove Views" in any batch of issues, we find a fairly even representation of both positive and negative reviews. In any art, though, negative evaluations, though not pleasant, serve an important purpose. Knowing what makes something a poor representation would ideally let us know what makes something a fine representation.

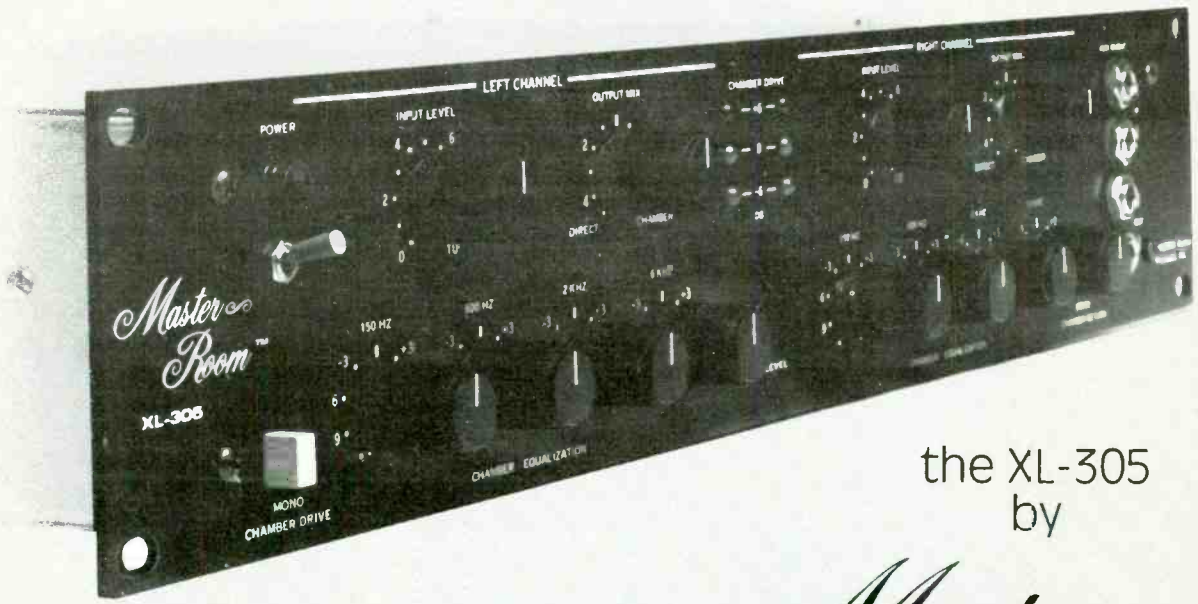
Our reviewers know they have to defend their arguments with examples—hence the multi-hundred word reviews.

Notches Withdrawn

We were surprised to find a note in the August, 1979 "Musical Newsicals" about Intersound's THE NOTCH and BASS NOTCH. There are a couple of comments we wish to make about the products. First, these devices were designed primarily for acoustic instruments fitted with piezo-ceramic transducers; while they may be used to advantage with electric guitar and bass, this is not their primary application. Second, though such devices were very desirable at their introduction a few years ago, recent advances in pick-up technology has reduced the need

Incredible...

the "Acoustic Chamber Synthesizer"™



the XL-305
by

Master Room™

- Totally new design approach
- The sound of a live acoustic chamber
- Natural sound, even on percussion
- Self-contained rack mount unit
- Full two-channel stereo

The Master Room XL-305 is a totally new design approach in reverberation technology. For the first time, the qualities and properties of a live acoustic chamber are available in a rack mount unit at an affordable price. There is a natural sound on percussion, as well as voices and all other musical instruments. This quality has not been obtainable from other compact reverberation devices. The XL-305 exhibits no unwanted side effects; it's as natural as a live chamber itself.

To hear this new advancement in reverberation, see your professional audio dealer and ask for a demonstration of this exciting new unit. Hear the XL-305 "Acoustic Chamber Synthesizer" for yourself, and you too will agree . . . It's INCREDIBLE.

MICMIX Audio Products, Inc.

CIRCLE 48 ON READER SERVICE CARD

2995 Ladybird Lane Dallas, Texas 75220 (214) 352-3811

for items such as THE NOTCH and BASS NOTCH. There has been a corresponding decrease in demand for battery operated preamplifiers; as a result, Intersound withdrew THE NOTCH and BASS NOTCH from production earlier this year. We hope this clears up any confusion about the availability of the products.

—James S. Johnson
President
Intersound, Inc.
Boulder, Co.

AES Publications

While reading the letter titled "Clarifying Impedance Check" in the "Talkback" column of your April 1979 issue, I noticed you mentioned (rather, Kenneth Berger of Eastern Acoustic Works mentioned) "the new anthology of papers on loudspeakers published by the Audio Engineering Society including work by A.N. Thiele and Richard Small." I am quite interested in obtaining these materials, so if you could notify me of how to go about it, I would be most appreciative. Thank you.

—Joseph Jurchak
Clark's Summit, Pa.

The Audio Engineering Society, which is located in New York City at 60 East 42nd St. (zip 10017), publishes its own journal. The anthology you seek is quite likely in one of its recent monthly issues. If was not published in the Journal, *the AES will be able to further further inform you. In any event, write to them for information on how to get what you're looking for and find other items of interest to you.*

Eastern Philosopher in the Job Market

My problem is heavy and light. I am interested in how and who in the professional recording industry has a maintenance training program or on-the-job training.

Could you please supply me with an address or further information on manufacturers or distributors on this subject? Can you help? Thank you!

—James Moten, Jr.
San Mateo, Ca.

Heavy and light, eh? You have posed a multi-dimensional question, we think. Are you looking for maintenance training for work in the recording studio? You really throw us off when you mention manufacturers and distributors.

Our best suggestion is for you to get a hold of Billboard's Int'l Recording Equipment & Studio Directory. *(Write to Billboard Publications, Inc., 9000 Sunset Blvd., Los Angeles, Ca. 90069.) Then contact studios in your area to find out if they'd be willing to let you in on some sort of apprenticeship arrangement. Check the Yellow Pages. Generally, just get out and pound the beat, as it were.*

If it is actually working on equipment that you are after, do all the above with respect to manufacturers.

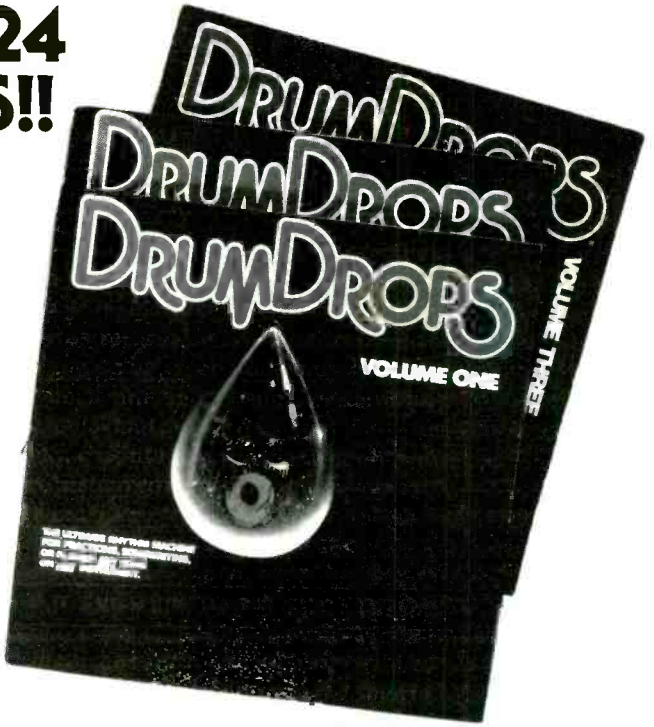
Sharing Ideas

Thanks for a truly great article on "live" sound in your August 1979 issue. I had the privilege of meeting Harold Blumberg last Christmas at the Spectrum in Philadelphia, through a mutual friend, Richard Feld (former Giles Soundman with Activated Air). Harold's system, reflects, in my opinion, one of the cleanest operations in the business. From racking to equipment choice, Harold shows what a great deal of thought can do for quality and budget. Harold is also a down to earth enough

Happy Holidays!

NOW. . .THE 1st TRACK OF YOUR 4 TRACK CAN BE 24 TRACKS OF DRUMS!!

And you can improve your demos 100% with a drop of a needle.



Introducing DrumDrops®, The Ultimate Rhythm Machine™ For Practicing, Songwriting, or Recording

Recorded at Sound City . . . one of Hollywood's hottest 24 track recording studios, where Fleetwood Mac, Foreigner, and many more have recorded. Played by one of the top session drummers on the West Coast . . . David Crigger, whose credits include: Don Ellis, Burt Bacharach, Brian Auger, Freda Payne and many more.

The DrumDrops masters were recorded on a Neve computerized 32 input console, to a Studer 24 track recorder at +3 -30 ips and Dolby A's on all tracks (modified a little by producer Joey D. Vieira.) Mastered on a Studer 1/2 track. The microphones were all Neumann and Sennheiser.

THE RESULTS: The cleanest, brightest, punchiest drums you've ever heard!

There is a total of 32 styles of rhythms to choose from on our three volumes of albums. There are all forms of Rock and Roll, Disco, Country and Jazz. All selections have been overdubbed with appropriate percussion like: electronic drums, tambourines, congas, timbales, cabasa, shakers, etc. There are no other instruments on the albums — just the drums and percussion.

Some of the biggest names in the music industry use DrumDrops as a songwriting aid. And we have on record numerous publishing and even label deals that come about for our buyers with the use of our professional drum tracks. Each selection can be transferred and edited for any arrangement or length you desire.

DRUMDROPS SELECTIONS

VOLUME ONE

Side 1	Side 2
1. Introduction	1. Funk Rock
2. Fast Disco	2. Straight Rock
3. Slow Disco	3. Medium Rock
4. Jazz	4. ¾ Country Rock
5. Bluegrass Country	5. Hard Rock Shuffle
6. Country Swing	6. Latin Rock
	7. Rock Ballad

VOLUME TWO

Side 1	Side 2
1. Introduction	1. Jazz Fusion
2. Samba Disco	2. Hard Rock
3.Pulse Disco	3. Country Rock
4. Medium Rock Ballad	4. Syncopated Funk
5. Medium Rock Shuffle	5. Country Rock Ballad
6. Raggae	

VOLUME THREE

"THE PROFESSIONAL ALBUM"

Side 1	Side 2
1. Intro	7. Jazz Fusion II
2. Straight Disco	8. Afro Cuban Jazz
3. Salsa Disco	9. Jazz Funk
4. Rhythm & Blues	10. 1/2 Time Rock
5. 5/4 Rock	11. R&B Ballad
6. 7/8 Funk	

CIRCLE 49 ON READER SERVICE CARD

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VOL. 1 ☐ Album ☐ Cassette ☐ $ _____

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VOL. 3 ☐ Album ☐ Cassette ☐ $ _____

T-Shirt ☐ $5.95

I am enclosing only $9.95 for each album or cassette and 75¢ for postage and handling (or $1.75 each for Airmail Rush!)

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Georgia Residents add 4% sales tax.

TOTAL $ _____

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Make check payable to DrumDrops

When you order the set of all three volumes together you will receive **A FREE** Musicians Drumdrops T-shirt a $5.95 value!

Circle Size

S M L X-L

person to give time during set up, to share ideas with up and coming companies such as ourselves. Such is the exception, not the rule, in this business.

Thinking back on my time spent with Harold, and my time spent bleary-eyed over a hot soldering iron, I would like to suggest articles on the following topics: racking and patching for tour systems; covering the areas of house electronics, monitor electrons, AMP channel assign, (Harold's remote design sounded good), methods for avoiding grounding problems, and design for quick patching away from problems.

I've been most pleased with *MR* and its articles on "live" sound. Sharing ideas is good for all of us. Our thanks to *MR* and Harold Blumberg of Audio Freqs for taking time out to give us that extra boost of knowledge that makes it all worthwhile.

—Lee Jennings
Senior Technician
Brandywine Sound
Hockessin, De.

Your topic suggestions for future articles sound provocative—we have covered the grounding ground in the past, but we'll be looking into your other ideas for sure.

The VTR Grind

I truly sympathize with Len Feldman regarding the ALC (automatic level control) problem he discussed in his "Ambient Sound" of August, 1979, in talking about VTRs. I encountered this two years ago when I decided to see if video equipment might make it in the small studio (it hasn't).

The system I bought cost about $6,000, and is made by Panasonic (WV-2200 camera and NV-3085 VTR). After many frustrating problems with the audio, I found the best combo was to feed my main mixer (Tascam Md. 10) into my Teac Md. 1 mixer, and then connect the variable headphone output on the front of the Md. 1 into the VTR mic input (the only audio input aside from the camera). By attaching an official Radio Shack Attenuating Patch Cord between the headphone out and the VTR, and carefully adjusting the volume, I could get pretty clean transfers. The one problem still occurs that when the audio program gets too quiet, I record VTR noise, a kind of grinding sound. A real problem at times.

I appreciate *MR's* articles about the future of video in the studio and the world. I feel the key will be when a lot more people by and get a feel for the home VTR.

(I also enjoy *MR* itself very much.)

—Terry W. Williams
Blue Light Recording Studio
San Diego, Ca.

Ha Ha Ha

I would like to know if you have a "Laugh Track" record or could tell me where I might get one from? Thank you for your time....

—Charles Williams
Lexington, Ky.

There are a number of records on the market that may include exactly what you're looking for. We know of no disc that consists entirely of laughs, but your retailer might. Check your Schwann-2 Record & Tape Guide (record dealers carry this publication) in its "Spoken and Miscellaneous" section, and look into the records listed under Sound Effects.

The standard bearers.

The high bias standard.

In the past few years, these fine deck manufacturers have helped to push the cassette medium ever closer to the ultimate boundaries of high fidelity. Today, their best decks can produce results that are virtually indistinguishable from those of the best reel-to-reel machines.

Through all of their technical breakthroughs, they've had one thing in common. They all use TDK SA as their reference tape for the high bias position. These manufacturers wanted a tape that could extract every last drop of performance from their decks and they chose SA.

And to make sure that kind of performance is duplicated by each and every deck that comes off the assembly line, these manufacturers use SA to align their decks before they leave the factory.

Which makes SA the logical choice for home use; the best way to be sure you get all the sound you've paid for.

But sound isn't the only reason SA is the high bias standard. Its super-precision mechanism is the most advanced and reliable TDK has ever made—and we've been backing our cassettes with a full lifetime warranty* longer than anyone else in hi fi—more than 10 years.

So if you would like to raise your own recording standards, simply switch to the tape that's become a recording legend—TDK SA. TDK Electronics Corp., Garden City, NY 11530.

TDK®

The machine for your machine.

*In the unlikely event that any TDK cassette ever fails to perform due to a defect in materials or workmanship, simply return it to your local dealer or to TDK for a free replacement.

CIRCLE 67 ON READER SERVICE CARD

PATCHWORK PAYS OFF.

When you start working with an 8-track recorder, there's a great temptation to rush out and buy a so-called "8-track" board. Forget it. You don't need the fat price tag. Because what you're really looking for is 8-buss operation.

That's why we've put together a systematic approach to 8-buss operation built around our new Model 5B. You get control over all your tracks. Total access to the signal flow. The ability to adapt quickly to each new session. And a rather large savings in cash outlay.

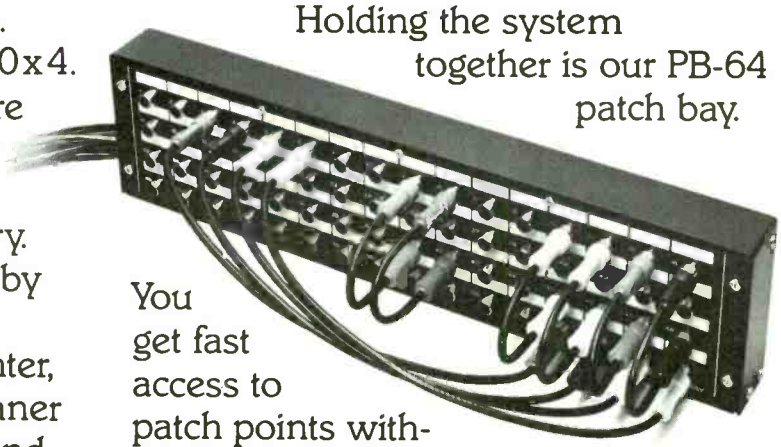
It starts with our new Model 5B. Eight-in/four-out. Expandable to 20x4. Color-coded controls get you where you're going faster, with fewer miscues. And new ICs have been incorporated throughout its circuitry. The slew rate has been improved by a factor of four. The result: better transient response. Tighter, sharper, cleaner sound.

Now add the Model 1. It gives you an independent 8x2 mix anywhere, anytime you need it. During basic tracks and overdubs, it's your monitor mixer. During final mixdown, it's ready for stereo echo. Any time you need another 8x2 mix, it's right there.

Our MB-20 lets you meter any line level signal you want. That's handy when you're using one mic or one instrument per track and driving your 8-track with the Model 5B's direct outputs.

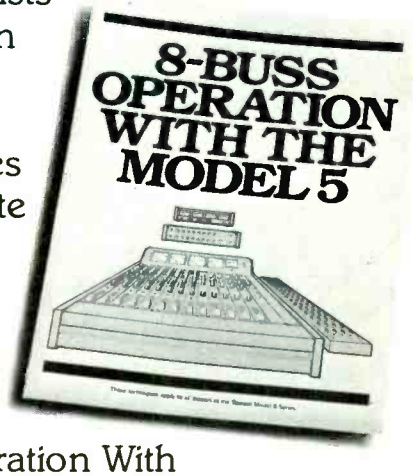
There's also some extra mixing flexibility in the MB-20. You get Buss/Tape selectors and an independent 4x2 monitor mixer.

Holding the system together is our PB-64 patch bay.



You get fast access to patch points without crawling behind and under your rig. You use jumper cables like push-buttons to route signals. And you don't get trapped into one hard-wired configuration. You can grow, change or completely re-arrange your set-up without throwing out the gear you've already invested in.

That's the Tascam concept of modularity. One that's paid off for recording artists for more than seven years.



See your Tascam Series dealer or write us for a free copy of our operations/hook-up bulletin, "8-Buss Operation With The Model 5." You'll see the functions you need. At a price you can afford.

Because patchwork does pay off.

TASCAM SERIES

TEAC Professional Products Group

TALKBACK

"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.*

Studio Headphone System With Plenty of Headroom

We are planning to open a demo recording studio sometime next winter. Up until now, everything has gone smoothly in our design except for one thing—the headphone system. We would like to have about 16 headphone outputs in the studio with a volume control for each one, but we just can't plug them in in parallel as this would overload the power amp. We've thought about a series of parallel 8 ohm pads interconnected, but dropped the idea as we felt it wouldn't be too stable. We also considered a 70.7 volt line, but it is too expensive for us (too many transformers involved). The last thing we considered was a distribution amplifier, but we have not been able to proceed with this idea, since no one we speak to seems to know if they exist or not!

Please keep in mind that we don't want to use headphones with an incorporated volume control, since we have found that most of these are not of a quality that we desire. We prefer an external fader on each headset.

We would appreciate it if you'd put an end to this madness and clue us in on what you feel is the best solution and what headphone system most professional studios utilize.

Keep up the good work!

—Denis Houle
Verdun, Quebec

The headphone system we use at Trod Nossel was developed over a period of time and has proven to be loud and reliable. We do not have an individual volume control but you could add one fairly easily.

Each of our cue systems uses a Crown D-60 amp, two 8 ohm-to-70.7 volt line autotransformers, and Beyer DT-100 headphones, with the standard impedance, which is 400 ohms.

You can connect your 16 phones to this system without any need for switching or pads. You will find that when plugging additional phones in, the volume of phones already plugged in will change very little if at all.

The purpose of the autotransformer (which, by the way, could just as well be a transformer) is to raise the voltage to the headphones. Without it, the amp would clip long before the maximum power handling capability of the Beyer phones was reached. With this system, I've found that the amps clip well beyond the volume level that most musicians can tolerate. There's a lot of headroom.

Your volume controls could be a 1,000 ohm audio taper type, mounted in a box with connectors, or you could build them into your distribution panel.

—Richard Robinson
Engineering Officer
Trod Nossel Recording Studios
Wallingford, Ct.

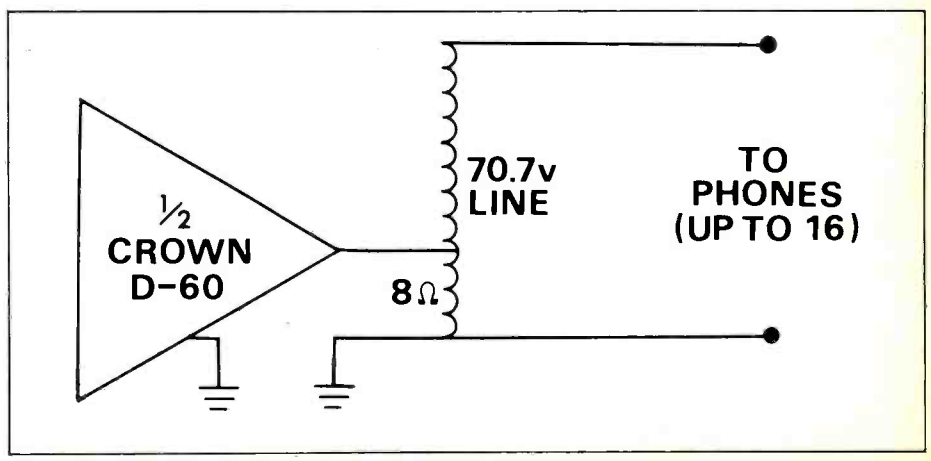
Help For RF Headaches

In order to use my low-impedance microphones with my TEAC Model 2 mixer, I had an input transformer box built for me using only the finest parts available. Even with the box, I'm still getting interference from the radio. Would lining the 12″ length of mic cord from the mixer to the box with lead suppress the interference?

—Dale Baglo
Kelowna, British Columbia

RF is always a drag and can be a headache for even professional equipment and studios. To answer your question simply—no, lining the mic cord probably won't help. Hopefully, the following will, however.

Insert a Phillip's ferrite chock (number VK20020/VB) in series with the microphone line at the TEAC side of the connection. Also, be sure that the



transformer box is connected to the input ground of the TEAC mixer.

The components mentioned are available in British Columbia at any good electronics outlet. If you have any problem finding them in Kelowna, however, contact electronic distributor L.A. Varah directly at 2077 Alberta St., Vancouver, British Columbia V5Y 1C4 or call them at 604-873-3211.

—Roger Monk
Recording Engineer
The Little Mountain Sound Co., Ltd.
Vancouver, British Columbia

Interfacing Ins and Outs

I have a P.A. system and I am confused as to how some of the interfacing should best be realized. The following is the sequence of signal flow: from the Uni-Sync Trouper I mixer (600-ohm unbalanced output) into a Shure 610 feedback controller EQ (RCA high-impedance input). This unit offers a choice of high-impedance in and out, or XLR balanced low-impedance in and out. Low-impedance out is the logical choice, but the signal flow is then fed into the Urei 521 crossover which has an unbalanced 600 ohm input. How do I wire this?

The same problem exists with my monitor system which is derived from the Trouper I monitor out (600 ohm unbalanced) and fed into the 600 ohm input of my Altec 9860A one-third octave equalizer.

I really appreciate any help you can offer on this subject.

—Paul Lazarro
Waterbury, Ct.

Many are easily confused by the issue of balanced and unbalanced inputs or outputs, as well as low and high impedance. I will give you some simple ground rules that will work in most all cases.

It is acceptable for unbalanced outputs to feed balanced inputs and/or balanced outputs to feed unbalanced inputs. The only problem you are likely to run into when mixing balanced with unbalanced is a hum problem. Such hum problems can most often be taken care of with additional grounding or by reversing the AC power plugs. Professional users are mixing today's balanced and unbalanced equipment all the time with little or no problem.

Regarding impedances: A general rule is that it is permissable for an output impedance to feed an input that is equal to or greater than its own impedance specified in ohms. For example,

a 600 ohm output can feed a 50,000 ohm input with no problems. Likewise, a 600 ohm output can feed a 600 ohm input. It is not good for a 600 ohm output to feed a 100 ohm input, however, as the 600 ohm output stage would be loaded down and would be required to work too hard driving a 100 ohm impedance. There may be an alteration of frequency response experienced, as well.

—Larry Blakely
Recording Consultant
Framingham, Ma.

For additional information on this subject, catch Mr. Blakely's articles on interfacing in the April and May, 1979, issues of Modern Recording *(back issues are still available as of this writing). A thorough reading of these articles will not only unravel your current problems, but give you a better understanding of the ins and outs (pun intended) of connecting it all together!* —*Ed.*

RFI Refresher Course

I was recently called in to clear up some problems in the recording set up in the apartment of a local producer.

His system consists of a TEAC Model 2 mixer and a TEAC 3340 recorder. The problem is pick up of a local radio station through the mic inputs on the mixer or the recorder.

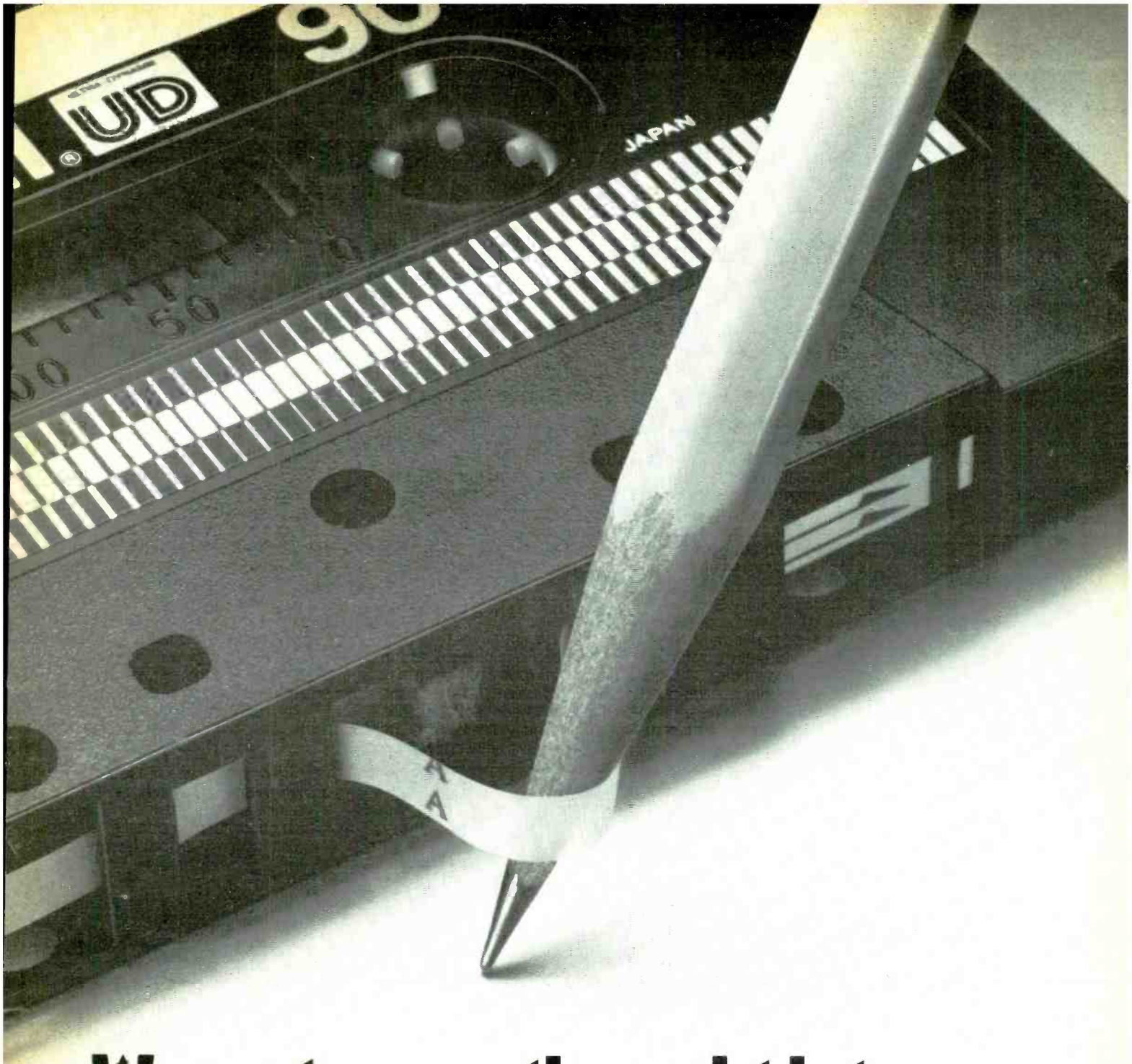
We are using a low impedance mic with a Shure A95U impedance-matching transformer at the input. The RFI only occurs when the mic cable is connected, so it is obviously acting as an antenna. We tried putting an additional transformer at the mic end as well, with no improvement. Please help us solve this problem before we modify the recorder to pick up FM as well.

—Jeff Ghitelman
Ghitelman Sound Services
New York, N.Y.

Eliminating RFI is an art which takes experience. Engineers spend a lot of time and effort in eliminating RFI problems. If your problem still persists after trying what I suggest, don't lose heart. Even experienced engineers have their failures.

There are three basic methods to help clear up RFI—grounding, shielding and/or filtering. Any one of these methods, or a combination of them, may prevent RFI.

By already knowing where the RFI is originating, you are steps ahead. For your problem, the grounding method



We put more thought into our leader than most manufacturers put into their tape.

One of the reasons Maxell has such a great following is because of our leader.

It has a built-in non-abrasive head cleaner designed to remove the oxide residue other tapes leave behind, without damaging your tape heads.

It also points out what side of the tape you're on (A or B) as well as which direction the tape is traveling. So it's almost impossible to make a mistake.

It even gives you a five second cueing mark, so you can set your recording levels without wasting tape. Or time.

Obviously, all the thought that went into our leader was designed to help you get more out of our tape.

So if you think our leader sounds impressive, wait till you hear what follows it.

maxell

CIRCLE 79 ON READER SERVICE CARD

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

should eliminate the RFI by sending the radio signal which is entering your Model 2 and A-3340 through the microphone cable to earth ground. This is done by attaching the chassis of the unit to an earth grounding point, such as a water pipe (not a plastic one).

Caution number one: Make certain that the shielding around the mic cable is electrically attached to the chassis of the unit in use.

Caution number two: In solving one problem you may create another: Ground loops may occur which will cause hum in output.

If grounding does not solve the problem, shielding or filtering are your alternatives. If more information is needed, please feel free to contact me at TEAC Corporation of America.

—Dale Dalke
Technical Correspondent
TEAC Corp. of America
Montebello, Ca.

Phasing Out Cancellations

Are there any phase inaccuracies or cancellations that occur between the console and the tape deck or when bouncing tracks from one deck to another?

You have a fine magazine, very informative. Keep those issues coming so I can continue growing!

—Jeff Robinson
Memphis, Tenn.

Total *cancellation* of combined signals due to phase inversion of one of the signals can happen as a result of wiring mistakes in a balanced system. Phase *inaccuracies* can occur in either balanced or unbalanced systems, and in a tape-to-tape transfer process can arise from either playback head azimuth misalignment, signal processing equalization or a combination of these effects.

As an example of how playback head azimuth misalignment can cause phasing problems, consider this tape-to-tape bouncing situation: On a 4-track original tape, tracks 1 and 4 contain elements of the drum set, and these elements are to be mixed together with the resulting combination recorded onto another tape. If the playback head azimuth is not correct, corresponding points on tracks 1 and 4 will reach the playback head gap at slightly different times. The difference in time shows up as a phase difference between the signals coming from the two tracks. When these signals are combined in a mixer, the high-frequency portions of these signals that are common to both tracks (and in the case of drum tracks there would be a lot of high frequency signal) would partially cancel and cause an overall loss of high-frequency signal. This effect is less troublesome when the two original tracks are adjacent to each other on the tape (e.g., tracks 1 and 2), but will still be present as long as there is some azimuth misalignment.

Signal processing equalization (as distinct from fixed equalization internal to the tape machine(s)) can be another source of phase inaccuracies when applied to one signal that is to be combined with another signal having similar or near-identical information. In this kind of situation the variables are under control, so if phasing innacuracies *do* occur they can be remedied by changing or eliminating the equalization.

Phase cancellation caused by wiring error will affect the low frequency portion of a complex signal most drastically; phase inaccuracies will be most apparent in the higher frequency portion.

—Peter Weiss
Contributing Editor
Modern Recording

Studio quality microphones that don't need a studio to survive.

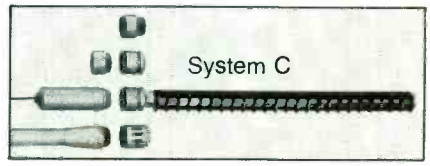
The CS15P condenser cardioid microphone is equally at home in a recording environment or broadcast studio. When hand-held it puts sex appeal in a voice with its bass-boosting proximity effect. With shaped high-frequency response and its ability to handle high sound pressure levels (140 dB with 1% THD at 1kHz), the CS15P is ideal for close-up vocal or solo instrument miking applications.

When boom mounted, the CS15P has better gain-before-feedback and a better signal-to-noise ratio than most shotguns. It's phantom powered and it's rugged.

The CO15P condenser omni extends frequency response to the very limits of audibility, 20 to 20,000 Hz. Unlike other "omni's," the CO15P maintains its omnidirectional polar pattern at the very highest frequencies. Perfect for the distant miking of an entire orchestra as well as up close on individual instruments. And like the CS15P, it's phantom powered and it's rugged.

The Electro-Voice warranty
Electro-Voice backs up these two microphones with the only unconditional warranty in the business: for two years we will replace or repair your CS15P or CO15P microphone, when returned to Electro-Voice for service, at no charge – no matter what caused the damage!

We can do this because we build these microphones to meet our standards for performance, ruggedness and durability. We accept nothing less, and if you're a professional, buying a professional quality microphone you shouldn't either.



Electro-Voice®
a gulton company

600 Cecil Street, Buchanan, Michigan 49107

CIRCLE 82 ON READER SERVICE CARD

THE PRODUCT SCENE

By Norman Eisenberg

NEW DBX PRO UNITS

From dbx there's word of three new devices aimed at the professional user. One is the model 164, a stereo compressor/limiter featuring dbx's "Over-Easy" compression curve said to be ideal for disco use. In a disco or stereo sound-reinforcement system, the model 164 may be set to regulate overall level and still "get every ounce of power out." The result, says the company, is full power with fail-safe amp/speaker protection. The unit also is offered for broadcast and recording studio work; in the latter application it serves as a compressor/limiter for stereo mixdown. Price is $379.

The dbx 501 is described as a single-band stereo dynamic range linear expander. Designed to bring out the detail in music, the model 501 makes the louds louder and the softs softer. In enhancing the dynamic range of discs, it also reduces surface noise. Price is $279.

The dbx 505 packages the model 501 with an earlier dbx product, the model 500 subharmonic synthesizer. The model 500 creates low frequencies one octave below the musical signal, and mixes them back into the program. Price of the combined package is $479.

CIRCLE 23 ON READER SERVICE CARD

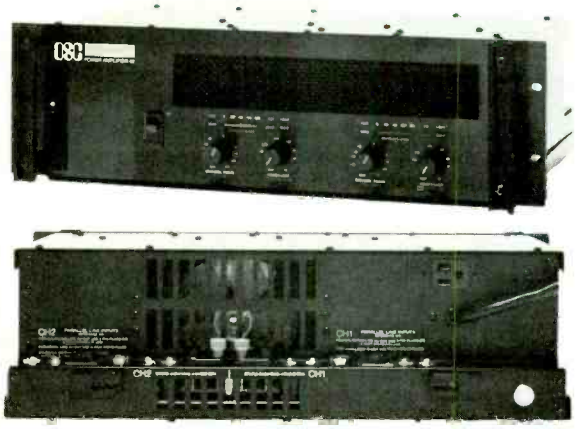
MODULAR CONSOLE

Soundcraft Electronics (of England) has announced its series 1624, a fully modular 16-bus console available in 16/16 and 24/16 frame sizes. The sixteen group/monitor channels may be split into two separate parts in mixdown, thus creating—within the 24/16 format—24 input channels, 16 submasters (or 8 stereo submasters), and 16 effects return channels with 3-band sweep-frequency equalizer. Other features include: "ultra low-noise" discrete mic preamps, three solo modes prefade (mono), post-fade (stereo), solo in place, two group mating buses, six auxiliary sends, 4-band sweep-frequency EQ with variable-frequency high-pass filter on all input channels, fully modular patchbay as standard and P&G faders as standard.

The larger frame is fully wired for 24-track operation; the addition of an optional module containing eight monitor channels further facilitates this operation. VU meters are fitted as standard, each one incorporating an LED peak indicator. The console is supplied complete with floorstand and is finished with solid hardwood end cheeks and a leather arm-rest. Price was not available at presstime.

CIRCLE 24 ON READER SERVICE CARD

QSC ANNOUNCES SIX POWER AMPS



Six new power amps with professional features have been announced by QSC Audio of Costa Mesa, Ca. Designated the A21, A22, A31, A32, A41 and A42, the units consist of two models in each of three power ranges. For 8-ohm operation, the per-channel power ratings are 80, 125 and 200 watts. With 4-ohm loads, the ratings per channel become 120, 200 and 325 watts. All power is rated with reference to 0.1% THD and 0.05% IM, 20 Hz to 20 kHz. Amps and loads are protected from shorts, reactive and mismatched loads, overheating, sub-audio output, turn-on thumps and ultrasonic RF burnouts.

Said to be among the smallest and lightest in their power ranges, the QSC amps employ flow-through ventilation, a lightweight high-turbulence heat-sink, direct mounted power transistors and a thermally-activated two-speed fan. "Ruggedness and roadworthiness" are aided by having the power transformers mounted in front, so stress is taken to the rack rails through $\frac{1}{10}$-inch steel struts.

Each amp has active balanced/unbalanced inputs with dual ¼-inch jacks and dual XLR connectors for easy cross-patching. Output connections can be made via 5-way binding posts or dual ¼-inch jacks. Calibrated gain controls and mono-bridging switches are standard. Models A22, A32, and A42 also include QSC's "PowerLimit" control which enables the operator to adjust the output power of each channel to avoid clipping and to protect loads. These models also offer LED level displays, distortion indicators and PowerLimit indicators. Prices range from $574 to $948.

CIRCLE 25 ON READER SERVICE CARD

SANSUI'S METAL TAPE CASSETTE DECKS

New heads for record/play and for erase are featured in two new metal-tape compatible cassette recorders from Sansui. Both models—the SC-3300 and the SC-3330—run on two motors and permit feather-touch transport operation with full-logic control. They also have 16-segment LED peak-level displays which create individual bar graph signal readouts for left and right channels. Both decks also have a "Roller Back" holdback tension mechanism that is claimed to help reduce wow and flutter to 0.04% WRMS and, by reducing FM modulation, is credited with improving tonal quality. The two new decks also include Sansui's "tape lead-in" feature which bypasses the leader portion of the tape. Both decks include memory rewind, auto play and auto repeat and operation from an external timer. With Dolby switched in, S/N is given as 69 dB. Response with metal tape is listed as 20 to 17,000 Hz; with CrO_2 tape, 20 to 16,000 Hz. The bias and EQ switches have settings for chrome and normal tapes in addition to the metal-tape position. Price of either deck is $420.

CIRCLE 26 ON READER SERVICE CARD

MULTIPLE OPTION PROCESSOR

Inovonics, Inc. of Campbell, California has announced its model 215 Audio Processor, offering broadcasters a building-block approach to audio processing through a number of equipment options. The model 215 can serve as an AM or FM peak controller. It also can be equipped to function as an AGC only, or as an AGC with compressor. The AGC option is gated to provide a slow, "gain-riding" function for wandering program levels. Set-up is facilitated by an integral pink-noise generator. The compressor option features a "soft-knee" transfer function for smooth, unobtrusive control over average program dynamics. Price ranges from $375 to $785, depending on the options selected.

CIRCLE 27 ON READER SERVICE CARD

"INTELLIGENT" REAL TIME ANALYZER

Described as a "programmable, intelligent, third-octave real time audio spectrum analyzer" is the "Specpet" from the Ken Schaffer Group. The system—priced at $1725—is a combination of the Eventide THS-224 RTSA board with a Commodore "PET" 2001-8K 9-inch (diagonal) video display computer. The analyzer divides the audio spectrum from 20 Hz to 20 kHz into thirty-one one-third octave bands and displays their relative amplitudes on the PET screen. Suggested applications include measuring sound and noise levels, optimizing equalization, checking frequency response, and other audio analysis chores frequently necessary in recording and P.A. systems.

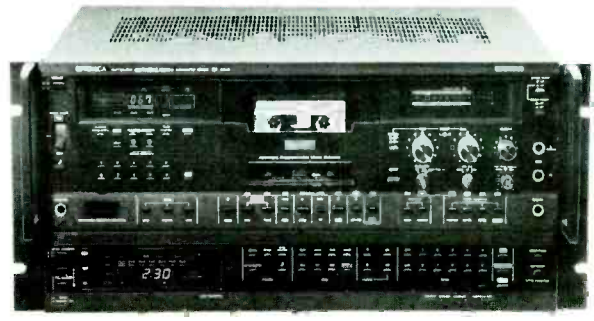
CIRCLE 28 ON READER SERVICE CARD

NEPTUNE'S NEW PARAMETRIC EQ

Four bands of equalization on each of two channels are offered in the new Model 342 parametric equalizer from Neptune Electronics of Portland, Oregon. Each band has frequency, band-width and gain controls, plus its own in/out switch, and an LED peak indicator. In addition, the device has complete channel in/out facilities, and balanced and unbalanced inputs and outputs. Compact size and rack mountability help make it a portable unit for the road. Price was not available at presstime.

CIRCLE 29 ON READER SERVICE CARD

PROGRAMMABLE CASSETTE DECK



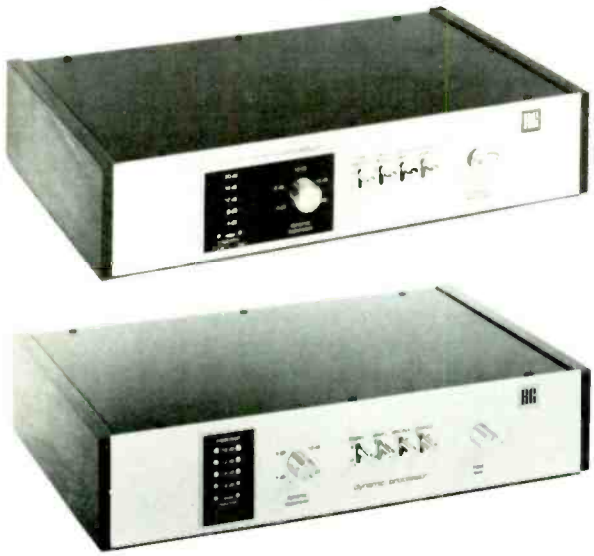
In addition to being metal-tape compatible, Optonica's new model RT-6905 cassette recorder incorporates this company's Automatic Program Music Selector (APMS) with infrared remote control and a timer that accepts up to forty-two different programming instructions. Priced at $1800, the RT-6905 consists of two units—the tape deck and the audio timer—in one self-contained system. The user can program for automatic play up to fifteen sections on a cassette in any desired sequence. The instructions can be automatically repeated up to five times. Automatic start, repeat, delayed programming, automatic search, skip, are some of the functions possible. Program instructions are displayed on a panel. All tape transport functions, including recording and automatic program search, can be activated via the deck's remote control.

Sensitivity and bias levels for normal, CrO_2, FeCr and metal tapes may be set on a "recording characteristics" control. In addition, a fine adjuster works in conjunction with a recording calibration selector to automatically fine-tune the sensitivity and bias currents for individual tapes. A two-color peak-level display has a peak-load feature. Tape tension is automatically optimized by a micro-computer-controlled adjustment. A two-motor model, the RT-6905 uses a dual-capstan closed-loop drive system. It has four heads (erase, record, play and a special sensing head for the automatic functions). Specs include response with metal tape from 20 Hz to 22 kHz (with metal tape), signal-to-noise ratio of 70 dB (over 5 kHz with Dolby on) and wow-and-flutter of 0.038 percent WRMS.

CIRCLE 30 ON READER SERVICE CARD

DYNAMIC PROCESSORS

RG Dynamics of Skokie, Illinois offers two sound processors designed to restore dynamic range and transients often restricted in commercially made discs and tapes. The RG PRO-16 at $299 offers up to 16 dB of program expansion, while the PRO-20 ($395) offers up to 20 dB expansion.



CIRCLE 31 ON READER SERVICE CARD

DO SUPER DISCS REQUIRE SUPER EQUIPMENT?

Undeniably some of the new "super-discs" contain unprecedented dynamic range. By super-disc, I mean the direct-cuts, the releases processed from digital tapes and the dbx coded discs. Some of these contain signal peaks that are 20 dB or more above what has been customarily on conventional discs.

All well and good—if your system can cope. But consider what is involved here. In a quiet room, the ambient or residual noise level might be 30 dB. The dynamic range of a conventional disc is, say, 50 dB. So, to reproduce the full dynamic range of that record in that room, your playback system must have at least a dynamic range of 80 dB.

Now, add 20 dB for the super disc. Your system must now be able to reproduce 100 dB. What's another 20 dB? In power terms, it is nothing less than 100 times greater.

To illustrate: Suppose you can reach 80 dB in your room with 1.5 watts driving your speakers. Another 10 dB requires upping the wattage to 15. And 10 dB over that would necessitate raising the power to 150 watts.

Interestingly enough, merely arming yourself with an amplifier that can furnish 150 watts per channel, and using it to drive speakers that can accept that power may not be all there is to it.

For one thing, there's the turntable. A unit that was perfectly lovely when spinning conventional discs may suddenly cop out with a super-disc played at high volume. This is no mere theory; it happened to me. My system, playing some of the new discs, developed the worst case of acoustic feedback I ever have heard in a "domestic environment." Investigation revealed that the inner suspension of the motor and platter was just too stiff to cope with 20 dB or so of increased sound, even with my speakers placed on the opposite side of the room. I got out an older, less glamorous turntable, fitted it with the same pickup used in the offending turntable and played the same record at the same level—and this time, no feedback. The older table, whatever else, does have a "springier" suspension that makes it behave more like a turntable and less like a mic.

As for amplifiers vis-a-vis this problem, in addition to ample power, they must also have good overload recovery *and* what is called "soft clipping" characteristics. That means the amplifier, when it is driven to clipping, will not go into wild distortion but will simply reduce the extreme peaks in the program while still producing reasonably clean—albeit reduced—waveforms.

As for speakers, it may be necessary to refocus attention—at the design end—on a lot of things we perhaps have taken for granted up to now but which may bear reexamination from "square one." Among these factors are the design of crossover networks and the actual circuit parts used for them; a new healthy respect for efficiency as such, though it must be accompanied by linearity of response; and better damping of the speaker mechanism directly through some kind of air-loading on the drivers instead of a fully-sealed enclosure which is relatively low-damped.

I know these thoughts are bound to stir some disagreement among my audio acquaintances. Whether you agree or not, we'd like to hear from you, especially if you've had any experience with the super-discs.

MUSICAL NEWSICALS

SYNTHESIZER EQUIPMENT

Parasound, Inc. of San Francisco, is now distributing products made by Synton Electronics of Holland. Highlights in the line are the Syntovox 221 and 222 Vocoders, formerly available only in Europe. The model 221 is a 20-channel analyzer, synthesizer, and control system which allows the user to create a wide range of vocal effects for recording, theatre use, film sound, commercials, and so on. The analyzer accepts an input signal and breaks it down by means of twenty bandpass filters. The levels at each filter frequency are converted to control voltages which impose the speech characteristics onto an input signal, either an internal pulse generator or an external source (any instrument with ample harmonics such as a synthesizer, organ or guitar. The model 222 is a simplified version of the device.

CIRCLE 4 ON READER SERVICE CARD

GUITARS

It's always news when Fender introduces a new model electric guitar, and that is exactly what the Lead I and Lead II are. Both models combine new circuitry, new body designs and familiar Fender design features in a new generation of Fender guitar. The bodies of both models have been reshaped and trimmed down in size to make them easier to play and are now fashioned from very dense ash wood. The famous Fender machine head shape has also been modified somewhat to complement the new body shape. Both models use a Stratocaster-type fully-adjustable, six-piece, die-cast steel bridge with through-the-body string mounting, and feature individually-mounted tuning machines and a rosewood or maple fingerboard. On the electronic side, the Lead I boasts a single high-output humbucking pickup with coil selector and series/parallel switches, while the Lead II has a pair of high-output high fidelity pickups with a phasing switch.

CIRCLE 5 ON READER SERVICE CARD

MODULAR SYSTEMS

Shure was one of the earliest manufacturers to offer a modular sound system with their famous VocalMaster systems. The latest offering from Shure is the Pro Master system which comprises the model 700 Power Console mixer/amplifier and the model 701 loudspeaker. The model 700 mixer/amp is a comprehensive unit featuring six input channels each of which can accommodate low or high impedance microphones or one of each simultaneously; each of these input channels features an input attenuator with clipping indicator LED, panpot, low and high frequency EQ, effects/reverb send and monitor send in addition to the main volume control. Additionally, two line-level inputs are provided, bringing the total number of inputs possible up to fourteen. The model 700 also includes two, ten-band, octave-band equalizers with Feedback Finder circuits, fast-attack LED level indicators with 42 dB display ranges, and two 200-watt solid-state power amplifiers. Add to this a Patch Block patch panel which gives access to the various signals at twelve appropriate points to allow a very wide range of setups for nearly any combination of P.A., stage monitor and/or taping functions. The speaker side of the Pro Master system is covered by the model 701 loudspeaker and the model 702 and 703 monitor speakers. The model

701 and 703 speaker systems feature Shure's exclusive variable dispersion treble horn which varies the horizontal dispersion from a wide 120° to a narrow 60° by inserting special foam wedges into the mouth of the treble horn. Additionally, the model 703 monitor can be laid on its long side to direct the sound upward for short-throw applications or on its short side to direct the sound more horizontally for long-throw situations, making it one of the most versatile monitor speaker designs.

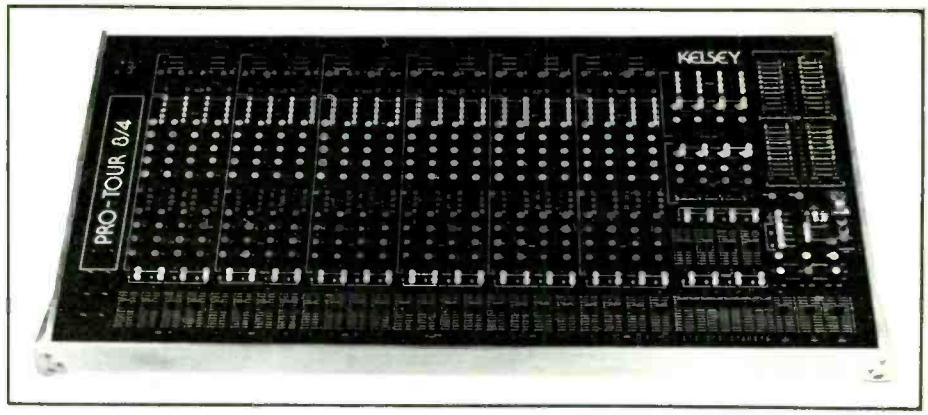
CIRCLE 6 ON READER SERVICE CARD

Tapco Sound, which is now a subsidiary of Electro-Voice, is now promoting three coordinated sound reinforcement systems which they refer to as "Rack 'n' Roll" systems. All three systems are based around the Tapco FR-3000 speaker system but differ in terms of their "front-end" electronics. The SRS 100 system is designed for the solo performer or small band who need more flexibility than most integrated mixer/amplifiers offer but without the bulk of conventional component P.A. systems. The SRS 100 features the 6001RB six-channel mixer, a model 2200 equalizer, and a CP500 power amplifier mounted in a convenient rack plus the pair of FR3000 speaker systems. The SRS 200 system provides more input channels with the 6100RB/EB mixing console which is backed up with a 2200 graphic equalizer and a CP500 power amp. The SRS 200 system is the most sophisticated offered by Tapco and features the 8201B stereo mixer which is backed up with a 2200 equalizer and the new 8201REB reverb which is equipped with an expander. On the power side, the SRS 300 system is stereo and biamped with a CP-X crossover feeding a CP 120/CP 500 power amp combination.

CIRCLE 7 ON READER SERVICE CARD

MIXING CONSOLES

Over the years, Kelsey has been one of the most popular mixing consoles in England, where it is made. The American importer, Dallas Music Industries USA Ltd. has recently announced a new model line from Kelsey is now available in America. The Pro-Tour 8/4 mixer is a versatile new unit which combines recording studio features and performance with the rugged construction needed for use on the road by



professional musicians. The Kelsey 8/4 series is an expandable design allowing configurations from 8 to 48 inputs in multiples of four. The input channels may be simultaneously mixed down to 8, 4, 2 and 1 outputs, plus there are four additional mix busses for effects and/or monitors. Other notable features of the Kelsey 8/4 series include complete patch facilities, multi-bus mixing, "zoom automation," gain status indication and eight peak level display arrays. Since it is designed for road use, the Kelsey 8/4 is sold complete with foam-lined SMF road case.

CIRCLE 8 ON READER SERVICE CARD

Peavey Electronics has announced the introduction of two new additions to their line of sound reinforcement mixers. The new models are designated the 801 and the 1201, and are 8-input and 12-input stereo mixers. Each input channel includes a low impedance mic input (XLR-type connector), high impedance mic/line input (¼-inch phone jack), insert point patching, input gain trim, three-band equalization, one post-fader effects send, two pre-fader monitor sends, a pre-fader listen (cue) pushbutton and a slide-type fader. Output channel facilities include the usual master functions and patch points plus several more innovative features such as reverb or effects returns to the monitor busses, pre-fader listen buttons on the main channel outs and a "sum" or L+R line output. Output levels for both the main outputs and the monitor outputs are displayed by means of ten-segment LED arrays.

CIRCLE 9 ON READER SERVICE CARD

P.A. AND MONITOR SPEAKER SYSTEMS

ESS has been known primarily for its hi-fi speaker systems using the Heil Air Motion Transformer high frequency driver, but now ESS's Professional Products Division promises to make its presence felt. The Heil AMT drivers are the most unique offering from ESS. Two models are offered, the HS400 which is a horn-loaded driver of high sensitivity and relatively high power handling capability, and the HS600 which is a bi-directional driver which can be uniquely useful in overhead disco or P.A. applications. The HS400 is available with two different diaphragm assemblies, a "studio" version rated at 30 watt power handling (RMS, pink noise signal, 1 hour) and

101 dB sensitivity (output at 1 meter on axis, 1 watt input) and a "disco" version rated at 50 watts power handling and 98 dB sensitivity. Frequency response is specified as 2500 Hz to 22 kHz ±2 dB, and the polar pattern is 45° vertical x 90° horizontal. The HS600 has approximately 3 dB greater sensitivity and similar power handling capability depending on choice of "studio" or "disco" diaphragm. Polar pattern for the HS600 is 40° x 150° out of each side resulting in total horizontal dispersion of 300°.

Also available exclusively from ESS is the RCF line of loudspeaker components, made in Italy by that country's largest audio manufacturer, a company with 30 years experience in loudspeaker manufacture. The line is extensive and includes twenty-three models of high efficiency professional loudspeakers in 10-inch, 12-inch, and 15-inch sizes and power ratings from 40 watts to 150 watts of continuous sine wave power applied for 100 hours, five compression driver models, four exponential horns, two horn/lens units and a variety of high fidelity loudspeaker components in various sizes.

CIRCLE 10 ON READER SERVICE CARD

A revised version of its SM-4 Stage Monitor speaker is the latest addition to the product line from Musimatic, Inc. The SM-4 system actually comprises two wedge-shaped speaker units which lock together into one rectangular unit with no exposed speakers for transporting. Each enclosure houses a 12-inch speaker and two piezo-electric tweeters. The cabinet design allows two different placements with different baffle angles for versatility in stage set-up. The new SM-4 has a new, non-tear finish to resist scratches, and retains the useful dual input/output jacks for convenient hook-up.

CIRCLE 11 ON READER SERVICE CARD

Electro-Voice has added a new, three-way, full-range P.A. speaker system they call the "Dominator." The Dominator uses an Electro-Voice EVM15L woofer in a folded-horn enclosure which makes the model the most efficient single-enclosure system Electro-Voice makes. Low-frequency cutoff for the system is 60 Hz which makes the system suitable for virtually all miked-instrument situations except the amplification of synthesizers which often have very low frequency components. Above 600 Hz

the midrange horn takes over; the horn used in the Dominator was specially designed to maintain 100° dispersion over its frequency range. Treble frequencies up to 16 kHz are handled by an ST350A tweeter with E-V's exclusive High Frequency Auto-Limiting circuit for protection. Maximum sound output from the Dominator is 123 dB SPL at 4 feet with 100 watts input.

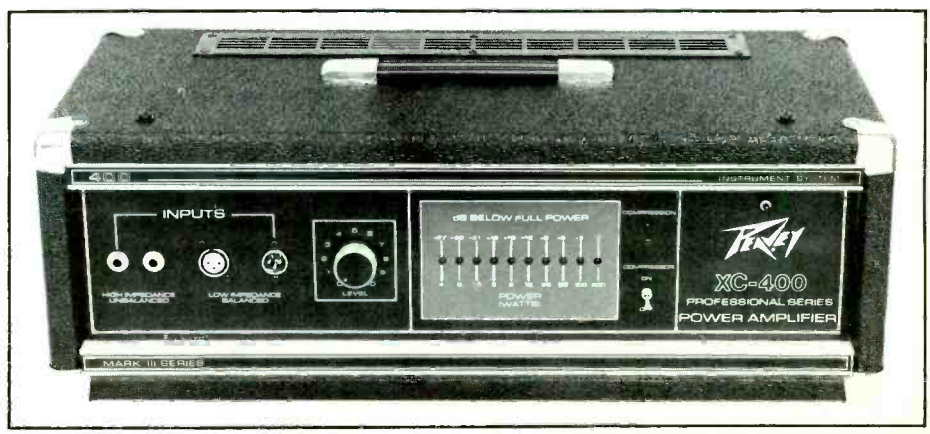
CIRCLE 12 ON READER SERVICE CARD

POWER AMPLIFIERS

New from Peavey is the XC-400 power amplifier, a two-channel unit rated at 200 watts RMS per channel into a 4-ohm load. The XC-400 is a "ruggedized" design using high-voltage, high-speed output transistors mounted on a massive aluminum heat sink, plus thermal compensation and short circuit and transient voltage protection at both input and output for reliability under adverse conditions. Unique to the XC-400 is Peavey's new "Distortion Detection Technique" compression circuitry which is said to virtually eliminate amplifier clipping due to signal overload by compressing the input signal as the amplifier approaches the clipping point. Output power is displayed with a wide-range LED array, and both low impedance XLR-type connectors and high impedance ¼-inch phone jacks are provided for connecting the input signals.

CIRCLE 13 ON READER SERVICE CARD

Scientific Audio Electronics, Inc., better known as SAE, has introduced a new Professional Products Group. Initially this new SAE Pro product line comprises three power amplifier models, the P400, the P300 and the P50, and the SAE Pro Termination Panel. The P300 and P400 amplifiers are virtually identical except for their rated output powers, 300 watts RMS per channel into 8 ohms for the P300 versus 400 watts per channel for the P400. As we have come to expect from SAE, both models feature fully complimentary circuit design from input through output rather than the less-ideal quasi-complimentary circuitry used by most other power amplifier manufacturers. For applications where even higher power levels are needed, the P300 and P400 feature a third input jack which automatically connects the two channels in a bridged mono mode, pushing the power specs to 1000 watts RMS into a single 8-ohm load for the P300 and about 35% higher for the P400; using a separate third jack is said to be both quicker and safer than the usual mono mode switch on other amplifiers. One shortcoming of most high-power amplifier designs is mechanical failure of the amplifier chassis due to the weight of the large power transformer; in the P300 and P400, SAE has bolted the transformer to the front, top *and* bottom of the chassis for improved strength. All three SAE Pro amps have stepped input level controls, but unlike the potentiometers with click stops used by



Double Your Bass.

Introducing a bowing device for your bass guitar.

If you could bow the strings of your electric bass guitar, you could turn it into a new instrument that combines the best of the guitar with the best of the classical double bass.

It could do wonders for both your rock and your jazz repertoire.

Unfortunately, your bass guitar doesn't have a bridge that permits bowing. Which makes it almost impossible to bow.

Until now.

The Bass Gizmotron™ allows you to bow the strings on your bass.

But it does more.

It permits infinite sustain, which is not possible with the double bass. You can hold one note, or several, at a time.

The sound is pure string sound, not electronic sound. And you don't need classical technique to play.

So don't be handicapped by a single bass. Double it, for the price of a few gigs.

The Bass Gizmotron

For dealer nearest you, write: Gizmo Inc., P.O. Box 139, Rosemont, N.J. 08556. or call 609-397-2000.

CIRCLE 40 ON READER SERVICE CARD

s, SAE uses
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put connectors, d includes LED peak indicators, cooling fans and 6 dB/octave subsonic and supersonic filters (defeatable for wideband response) with −3 dB points at 20 Hz and 25 kHz. The P50 is a 70 watt RMS per channel (into 8 ohms) amplifier of modest physical dimensions, making it ideal for use in bi-amped or tri-amped systems. A special feature of the P50, in addition to the standard Pro line features, is the use of a special toroidal power transformer which results in smaller size, improved electrical efficiency and reduced hum pickup within the amplifier. The SAE Pro Termination Panel is an accessory item which adds flexibility to any amplifier installation. It features XLR-type inputs with switchable polarity for either the U.S. standard (pin 2 "hot") or the European DIN standard (pin 3 "hot"), and a specially designed Dean Jensen transformer for improved RF rejection. This matching transformer will handle levels up to +24 dBm and is free of ringing even at 20 kHz.

CIRCLE 14 ON READER SERVICE CARD

CABLE AND CONNECTOR SYSTEMS

Distex Corporation manufactures standard and custom "snake" cables which have been carefully engineered for reliable and noise-free operation. Cables are available in either the conventional, single-jacket, multi-pair construction or else a spiral-wrapped construction made up from individually-jacketed cables for repairability and reduced inter-cable capacitance and crosstalk at the expense of size and weight. Low-noise Switchcraft connectors are used throughout. At the split-out end, special attention has been paid to strain relieving via extra layers of heat-shrink tubing; at the box end, the cable is strain-relieved with heat-shrink tubing outside the box, and securely anchored inside the enameled, galvanized steel box.

CIRCLE 15 ON READER SERVICE CARD

Walker Audio Visual Engineering has an interesting variation on snake-type cable: the Roll-Out Snake, Model 16-S. In its normal configuration, the Walker Roll-Out Snake is a 16-in, 3-out, low-impedance snake in a standard length of 100 feet. What makes the Walker Roll-Out Snake unique is its packaging. The stage box of the Walker is actually a cable reel contained within a genuine Anvil road case. The connectors (16 XLR-type females, and 3 XLR-type males paralleled with three insulated stereo ¼-inch connectors) are contained within the hub of the reel. To set up the Roll-Out Snake, the user merely puts the case down on the stage in an appropriate location, removes the top half of the Anvil case, and pulls out as much length as is needed to reach the mixer. After the gig, one unplugs the mic cables from the connectors in the reel, and simply reels in the cable. Walker claims a set-up of 30 seconds and a rewind time of 60 seconds.

CIRCLE 17 ON READER SERVICE CARD

PICKUPS

The Hocky Puck Pickup is a new design contact microphone from Freelance Music. The manufacturer asserts that for best fidelity a pickup or contact microphone must remain stationary while the instrument vibrates, a condition that subminiature pickups cannot meet because of their low mass. The Hocky Puck was designed to have the optimum mass for reproduction of frequencies from 1 Hz to 25 kHz, and has a relatively large diameter (about 2½ square inches) to insure good coupling of low frequencies. The unit may be mounted temporarily using the typical double-stick material (not included) or it may be permanently glued inside an instrument.

CIRCLE 20 ON READER SERVICE CARD

MUSICAL INSTRUMENT ACCESSORIES

Musicians who use their amplifier to contribute distortion to their sound usually shy away from direct boxes because they usually connect *before* the amp and thus don't pick up the player's real sound. The Hot Sink and Super Sink from Professional Music Products are designed to solve this problem because they are designed to connect to the *output* of the amplifier. The Hot Sink is designed to be used in place of the normal speaker and provides a line level output for P.A. or recording use and a headphone output allowing the musician to practice with his amp but without disturbing anyone. The Super Sink is a somewhat more sophisticated unit which allows speakers to be connected or not; it also features independent level controls for the line out and headphone outputs, and has an optional footswitch to bypass the device. Both units contain only passive circuitry and thus require no external power.

CIRCLE 21 ON READER SERVICE CARD

Ax Max is the name of a new, active direct box from Infinity Light and Sound. Direct boxes are normally based around a transformer which splits the output signal from a guitar, bass or other instrument into two isolated signals, one of which goes to the player's normal amplification setup while the other is fed to a mixing console for P.A. or recording use. The Ax Max, on the other hand, uses active amplifiers of the latest Bifet integrated circuit type to perform the same function but with a number of significant advantages, including much higher input impedance to eliminate loading down the instrument's output, lower output impedance to reduce treble loss and hum pickup in long cables, lower distortion and extended frequency response. Active direct boxes do, of course, require power, and in the case of the Ax Max the power comes either from the phantom power supply of a studio-type mixing console or an external 9V battery eliminator.

CIRCLE 22 ON READER SERVICE CARD

You're looking at three ways Technics pursues the one ideal. Waveform fidelity.

Waveform fidelity. It should be the objective of any professional component. Because perfect waveform fidelity would mean an output signal that's a mirror image of the input signal.

How do our engineers pursue this elusive goal? To begin with, they use two automatically switchable IF bands in the ST-9030 FM tuner. A narrow band for extra-sharp selectivity. And a wide band for extra-high S/N and extra-low distortion. But just as significant is a pilot-cancel circuit which Technics developed for high-frequency waveform fidelity. Even the basic tuning function in the ST-9030 is unique. Like an 8-ganged tuning capacitor for outstanding reception.

The engineering in the SU-9070 DC preamp is similarly impressive. There's a moving coil preamp with −157 dBV noise voltage. A moving magnet preamp with an extremely high S/N of 100 dB (10 mV input). Direct-coupled circuitry to keep distortion at a minimum of 0.003% (rated THD). What's more, the SU-9070 has inputs for three tape decks.

Finally there's Technics SE-9060 amp. It's DC like our preamp. Has a frequency response of 0-100 kHz (+0, −1 dB). And a "strapped" circuit for more than double the power in a multi-amp system. Compare specifications and prices. We think you'll agree. There's no comparison for these Technics components.

ST-9030. THD (stereo, 1 kHz): Wide—0.08%. Narrow—0.3%. S/N (mono): 80 dB. S/N (stereo): 73 dB. FREQUENCY RESPONSE: 20 Hz—18 kHz +0.1, −0.5 dB. SELECTIVITY: Narrow—90 dB. CAPTURE RATIO: Wide—0.8 dB. IF, IMAGE and SPURIOUS RESPONSE REJECTIONS (98 mHz): 135 dB. STEREO SEPARATION (1 kHz): Wide—50 dB.

SU-9070. PHONO MAX. INPUT VOLTAGE (1 kHz RMS): MM—380 mV. MC—9 mV. S/N MM—100 dB (10 mV input). MC—72 dB (60 μV). FREQUENCY RESPONSE: Phono 20 Hz—20 kHz (RIAA ±0.2 dB).

SE-9060. POWER OUTPUT: 70 watts per channel (stereo), 160 watts (mono) min. RMS into 8 ohms from 20 Hz to 20 kHz with no more than 0.02% total harmonic distortion. S/N: 120 dB.

Technics. A rare combination of audio technology. A rare standard of audio excellence.

Technics
Professional Series

CIRCLE 46 ON READER SERVICE CARD

The ELECTRIC Primer – Part III

By Peter Weiss

Here we are, after a one-month absence, at Part III, but the memory of Part II lingers on, like the faint after-aroma from a fish dinner. So, before we move on to the more exotic stuff we will tie up a few loose ends that are easily handled using information from Part II and the laws and rules of Herren Ohm und Kirchoff.

Loose end #1 is not really a loose end at all, but a continuation of the tail end of Part II, which dealt with the development of formulas for the effective total resistance of simple series and simple parallel combinations of resistances. What happens when the combinations are not so simple? *Fig. 1* is the schematic of a circuit that can be called "series-parallel" because it has portions that are in each type of configuration. Here are the known circuit values: $E_{tt} = 60$ volts; $R_1 = 8$ ohms; $R_2 = 7$ ohms; $R_3 = 5$ ohms; $R_4 = 60$ ohms; $R_5 = 20$ ohms; $R_6 = 17$ ohms; $R_7 = 13$ ohms.

The section of the circuit between points a and b is made up of R_1, R_2 and R_3 in series, and this section is itself in series with the section between points b and c. The section between points b and c is made up of R_4 and R_5 in parallel with each other and with the series combination of R_6 and R_7. We can assume that all of the resistances can be replaced by an effective total resistance, R_T. *(See Fig. 4.)* To make sense out of this circuit and find the value of R_T, we have to work backwards, step by step. At each step we will look at a portion of the circuit that can be handled easily using the resistance-combining formulas we already know. Okay, hip boots on? Here we go. Looking at R_6 and R_7, we see that these two resistances are in simple series combination with each other, and we can call the effective total resistance of this combination R_{T1}. To find the value of R_{T1}:

$$R_{T1} = R_6 + R_7$$

$$R_{T1} = 17 \text{ ohms} + 13 \text{ ohms}$$

$$R_{T1} = 30 \text{ ohms}$$

Fig. 2 shows R_{T1} replacing R_6 and R_7. The effective total resistance of the parallel combination of R_4, R_5 and R_{T1}, which we will call R_{T2}, can now be found using the formula for combining resistances in parallel:

$$\frac{1}{R_{T2}} = \frac{1}{R_4} + \frac{1}{R_5} + \frac{1}{R_{T1}}$$

$$\frac{1}{R_{T2}} = \frac{1}{60 \text{ ohms}} + \frac{1}{20 \text{ ohms}} + \frac{1}{30 \text{ ohms}}$$

Finding a common denominator *(See "Math Notes," Part II)*:

$$\frac{1}{R_{T2}} = \frac{1}{60 \text{ ohms}} + \frac{3}{20 \text{ ohms}} + \frac{2}{30 \text{ ohms}}$$

Now we have:

$$\frac{1}{R_{T2}} = \frac{6}{60 \text{ ohms}}$$

Taking the reciprocal of both sides *(See "Math Notes," Part II)*:

$$R_{T2} = \frac{60}{6} \text{ ohms} = 10 \text{ ohms}$$

Fig. 3 shows R_{T2} replacing the parallel combination which it represents. Now the original circuit has been reduced (*only* for the purpose of calculating R_T) to a simple series combination of R_1, R_2, R_3 and R_{T2}. R_T can now be found by applying the effective total resistance formula for resistances in series:

$$R_T = R_1 + R_2 + R_3 + R_{T2}$$

$$R_T = 8 \text{ ohms} + 7 \text{ ohms} + 5 \text{ ohms} + 10 \text{ ohms}$$

$$R_T = 30 \text{ ohms}$$

Like the classy textbooks do, we will leave it to the reader to find I_T, V_1, V_2, V_3, V_4, V_5, V_6, V_7, I_4, I_5, I_6, P_{tt}, P_T, P_1, P_2, P_3, P_4, P_5, P_6 and P_7. The writer is not simply trying to avoid work, and to prove this, will provide the correct

values and the methods for determining them along with the "Math Notes" for Part IV.

Loose end #2. So far, each example circuit has been a complete circuit in that there has always been a path for the current to follow in returning to the positive battery terminal. In order to maintain the high level of excitement generated thus far, we will now introduce some variety and look at circuit conditions when a complete path does not exist.

Fig. 5 is the schematic of a circuit consisting of a battery supplying a voltage (E_B = 10 volts) and two resistances (R_1 = 5 ohms, R_2 = 5 ohms). R_1 and R_2 are in series, so the effective total resistance of the combination, R_T, is:

$R_T = R_1 + R_2$

$R_T = 5\text{ ohms} + 5\text{ ohms}$

$R_T = 10\text{ ohms}$

From Ohm's Law we can find the current in the circuit I_T:

$$I_T = \frac{E_B}{R_T}$$

$$I_T = \frac{10\text{ volts}}{10\text{ ohms}}$$

$I_T = 1\text{ amp}$

Okay. Now the voltage drops across each resistance:

$V_1 = I_T \times R_1$

$V_1 = 1\text{ amp} \times 5\text{ ohms}$

$V_1 = 5\text{ volts}$

$V_2 = I_T \times R_2$

$V_2 = 1\text{ amp} \times 5\text{ ohms}$

$V_2 = 5\text{ volts}$

From Kirchoff's Voltage Rule:

$E_B = V_T = V_1 + V_2$

$V_T = 5\text{ volts} + 5\text{ volts}$

$V_T = 10\text{ volts}$

What happens if E_B and R_1 remain at 10 volts and 5 ohms, respectively, but R_2 is increased to 15 ohms? R_T becomes 20 ohms, and I_T becomes .5 amp. Readers can verify this by applying the series resistance formula and Ohm's Law as before. The voltage drops (also from Ohm's Law) are now V_1 = 2.5 volts and V_2 = 7.5 volts. V_T is still 10 volts, but now the greater voltage drop is across R_2.

Once again, holding E_V and R_1 at their original values, let us raise the value of R_2 to 95 ohms. Now R_T = 100 ohms, I_T = .1 amp, V_1 = .5 volts and

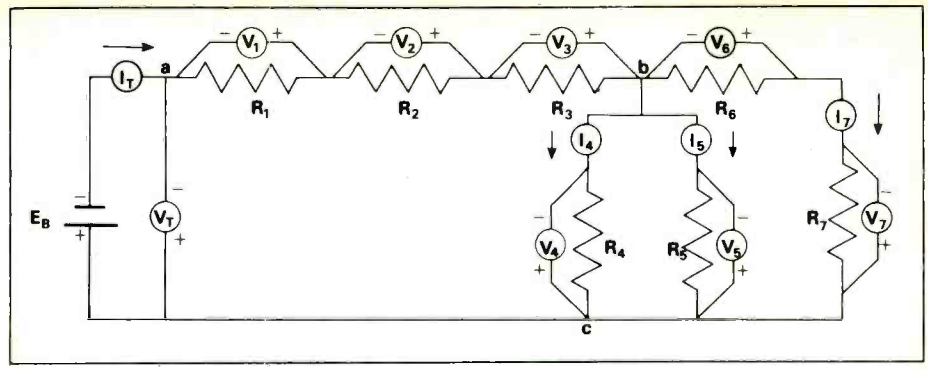

Figure 1

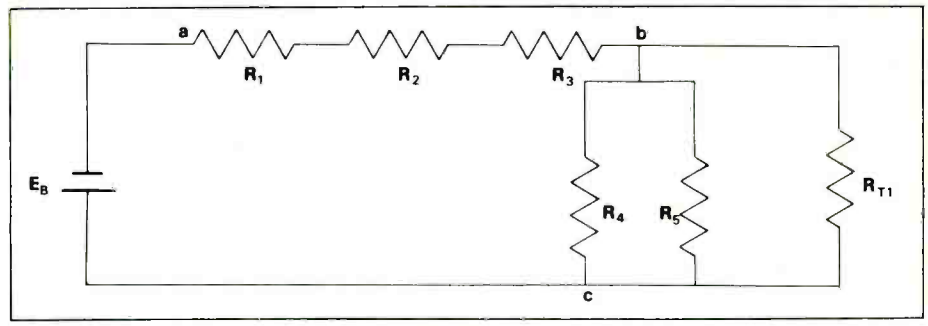

Figure 2

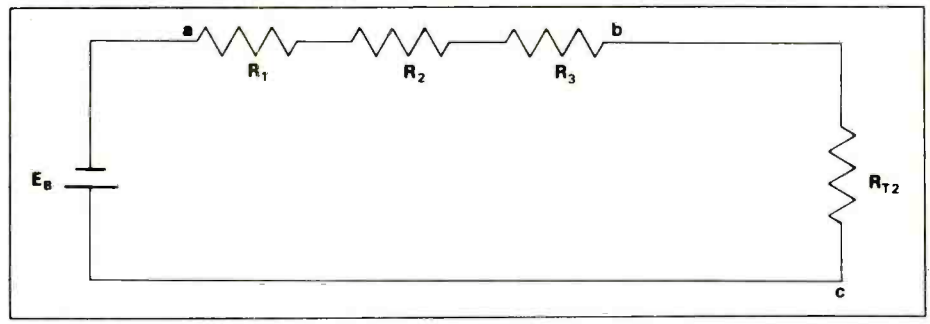

Figure 3

V_2 = 9.5 volts. If we continue this process of increasing R_2, I_T will become very small, *approaching* zero, and V_2 will *approach* 10 volts. We can go so far as to make R_2 "infinitely large," which under ordinary circumstances means replacing R_2 with an actual break in the circuit. Under these conditions I_T will be zero, since no complete path exists, V_1 will be zero as a result, and all 10 volts of E_B will appear as V_2 across the break in the circuit (across points **a** and **b**) where R_2 used to be. *Fig. 6* shows the circuit of *Fig. 5* with these new conditions in effect. Remember, these conditions exist *only* when there is *nothing* between points **a** and **b**, not even a voltmeter (more about meters later).

The circuit shown in *Fig. 6* may not seem to be of much use as it stands. However, this sort of arrangement can be used to represent, among other things, the workings of a "real" source of voltage and current. In general, sources of electrical energy have associated with them small, but non-negligible interval resistances. This statement is true concerning batteries, and any "real" battery is actually an "open" circuit like the one in *Fig. 6*, with the battery terminals at the locations of points **a** & **b**. We do not have to concern ourselves with the innards of batteries, but an examination of some of the facts concerning the connection of a battery to a load will be useful in the development of the next, very important, concept.

• • •

To begin, let's assume that the circuit of *Fig. 6* does represent a battery, with an "open circuit" voltage of 10 volts and an internal resistance of 5 ohms. We want to connect a light bulb between points **a** and **b**, and we want the light bulb to provide the maximum possible amount of light. In Part II,

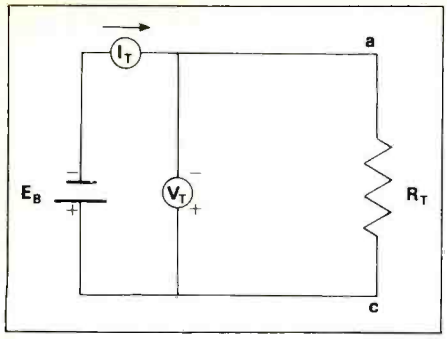

Figure 4

we saw that the amount of light (and heat) radiated by the bulb was dependent on the amount of power dissipated in the bulb. Therefore, for maximum light, we want maximum power dissipation in the bulb filament. In other words, and these are the words most often used to describe the desired condition, we want maximum power transferred from the source (E_B in *Fig. 5*) to the load (R_2 in *Fig. 5*).

Let's investigate the conditions necessary to achieve maximum power transfer. Since one version of the power formula, $P = I^2R$, involves the square of the current, a guess might be to try a bulb with a relatively low filament resistance, causing a considerable flow of current. Okay, let's try 1 ohm. This makes the total series resistance of R_1 and R_2 equal to 6 ohms. The current, I_T, will be:

$$I_T = \frac{10 \text{ volts}}{6 \text{ ohms}}$$

$I_T = 16.7$ amps

The power delivered by the source, P_T, is:

$P_T = I_T \times E_B$

$P_T = 1.67$ amps $\times$ 10 volts

$P_T = 16.7$ watts

For the power dissipated in R_1 and R_2:

$P_1 = (I_T)^2 \times R_1$

$P_1 = 1.67$ amps $\times$ 1.67 amps $\times$ 5 ohms

$P_1 = 13.9$ watts

$P_2 = (I_T)^2 \times R_2$

$P_2 = 1.67$ amps $\times$ 1.67 amps $\times$ 1 ohm

$P_2 = 2.78$ watts

This shows that of the total of 16.7 watts being delivered by the source, almost 14 watts are being dissipated in the internal resistance of the source, while just 2.78 watts are being dissipated in the bulb filament. Not exactly the conditions we were looking for.

Well, how about a filament resistance of 15 ohms? Now $R_T = 20$ ohms; $I_T = .5$ amp; $P_T = 5$ watts; $P_1 = 1.25$ watts; and $P_2 = 2.5$ watts. (Readers should verify these figures with their own calculations.) Not bad, but we can do better. How about a filament resistance of 5 ohms? This value is equal to the internal resistance of the source. Now $R_T=10$ ohms; $I_T=1$ amp; $P_T=10$ watts; $P_1=5$ watts; and $P_2=5$ watts. This value of P_2 is the highest we've achieved, and is in fact the highest possible in this example.

Fig. 7 is a graph of the relationship between R_2 and P_2, holding E_B and R_1 constant. We can express the conclusion drawn from the graph and the foregoing discussion as a general rule: *In order to transfer maximum power from a source to a load, the resistance of the load must equal the internal resistance of the source.* This rule applies not only to batteries and light bulbs, but to all sorts of source-load connections, and the maximum power transfer concept is particularly important in the interconnection of devices carrying audio signals. But more talk about audio signals later.

Now that we've explored an "open circuit" situation, let's take a look at what happens when, instead of placing an infinitely large resistance across the terminals of an electrical energy source, we put a "perfect conductor" in the same place. Once again, the circuit of *Fig. 5* will be the starting point for our discussion, using the circuit values from several paragraphs back

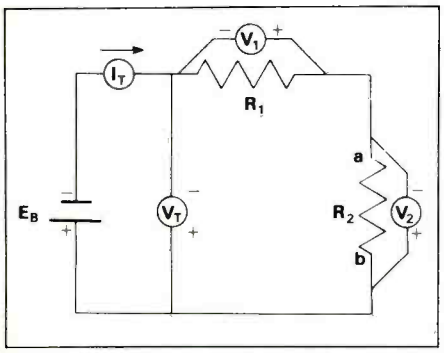

Figure 5

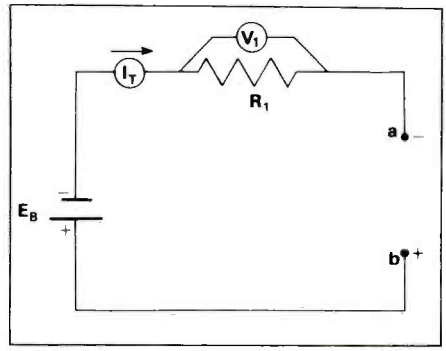

Figure 6

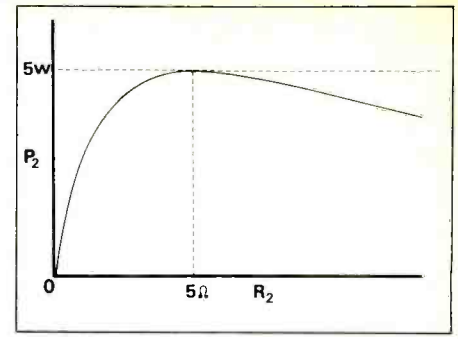

Figure 7

($E_B=10$ volts; $R_1=5$ ohms; $R_2=5$ ohms, $I_T=1$ amp, $P_T=10$ watts). If instead of a voltmeter across R_2, we were to connect a piece of copper wire, what would happen to the circuit values, specifically R_T, P_T and I_T? What is the effective total resistance of the parallel combination of 5 ohms and zero ohms? Let's see if the formula helps. We can call the effective total resistance of this combination R_{T1}.

$$\frac{1}{R_{T1}} = \frac{1}{R_2} + \frac{1}{0}$$

$$\frac{1}{R_{T1}} = \frac{1}{5 \text{ ohms}} + \text{"meaningless"}$$

Not much help at all, but we can get closer to the truth if we assume that the piece of wire has some very small resistance, say 1/10,000 ohm. Now:

$$\frac{1}{R_{T1}} = \frac{1}{5 \text{ ohms}} + \frac{1}{\frac{1}{10{,}000} \text{ ohm}}$$

We can make this a little more manageable by changing the second term on the right side according to the rules for fractions and reciprocals given in the "Math Notes" for Part II.

$$\frac{1}{R_{T1}} = \frac{1}{5 \text{ ohms}} + \frac{10{,}000}{1}$$

Finding a common denominator:

$$\frac{1}{R_{T1}} = \frac{1}{5 \text{ ohms}} + \frac{50{,}000}{5}$$

$$\frac{1}{R_{T1}} = \frac{50{,}001}{5}$$

Taking the reciprocal of both sides:

$$R_{T1} = \frac{5}{50{,}001} \text{ ohms}$$

which is a little smaller than 1/10,000 ohm. This agrees with what was said in Part II about the effective total resistance of a simple parallel combination being lower than the lowest resistance in the combination. Now, a resistance slightly smaller than 1/10,000 ohm can be considered as being effectively zero ohms, especially when compared to 5

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ohms. So, R_T is now effectively 5 ohms, I_T=2 amps and P_T is 20 watts, all of which is dissipated in R_1. If the original circuit represented an actual source-load connection like the battery/bulb arrangement discussed earlier, practically no current would flow through the load, a similarly negligible amount of power would be dissipated in the load, and practically all twenty watts of P_T would be dissipated in R_1, the internal resistance of the battery. Such a condition, that is paralleling any component or device with a near-zero resistance conductor, or placing such a conductor across the output terminal of an electrical energy source, is called a "short circuit." The device, component or output, is said to be "shorted" or "shorted out." If the short-circuited device or energy source is something like an electrical wall socket the results can be spectacular and quite dangerous. Inputs can be shorted out also, but the results are not much fun.

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One more loose end, and then we can move on. We will be discussing the interconnection of components and devices, so a few words about wire seem appropriate. Most wire used in electrical and electronic interconnections is made of copper, sometimes with a thin coating of tin to aid in soldering. Although wires are generally considered to have zero resistance, this is not actually the case. The resistance per unit length of copper wire is very, very low, and depends on the diameter of the wire. Thicker wires have less resistance per unit length than thin wires. This is easily understood if we consider a connection made with a thick wire as one made with many thin wires in parallel. Since the total resistance of a parallel combination is lower than that of the lowest resistance branch, the resistance of a bundle of thin wires will be lower than that of a single thin wire.

In actual practice, selection of wire size (diameter) is determined by how much current the wire must carry, and the physical length of wire necessary. A long, thin wire carrying large currents will heat up excessively. This is true, not only because of a non-negligible "I^2R" power loss through its length, but also because a thin wire can not dissipate the resulting heat to its surroundings very efficiently.

Well, that about wraps up our introduction to direct current circuits, although we will come back to this topic from time to time to support discussions of future topics. We can go on now to circuits in which, unlike the circuits considered so far, certain circuit values change with time.

Circuits Continued

Fig. 8 is the schematic of a simple circuit consisting of an "ideal" battery (no internal resistance; E_B = 10 volts), a resistance (R_1=5 ohms) and a switch. The switch provides a resistance-less means of completing or interrupting the circuit. When the switch is used to complete the circuit, I_T and V_1 instantaneously assume the values predicted by Ohm's Law: I_T = 2 amps, V_1 = 10 volts. If we were to graph one of these circuit values, say V_1, against time, the graph would look like the dashed line in *Fig. 9*. Note that the graph is a horizontal straight line at the 10 volt level. Using an old trick, let's interrupt the circuit by opening the switch, and replace R_2 with a break in the circuit.

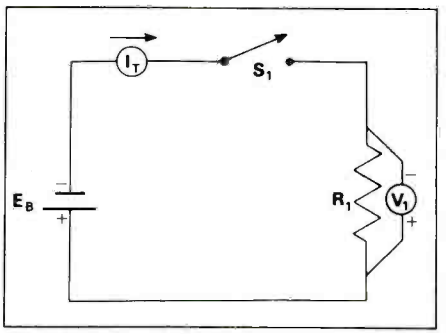

Figure 8

Now (following the open circuit discussion presented earlier) when the switch is closed, the circuit values instantaneously become I_T=0 and V_1=1 volt. Again, plotting V_1 against time produces a horizontal line like the dashed one in *Fig. 9*, indicating that V_1 does not change with time.

This last set of conditions will prevail as long as no other changes are made in the circuit. If, at points **a** and **b** we place two conducting metal plates parallel to each other (physically, not electrically, parallel) and a fraction of an inch apart. Each plate will instantaneously assume the charge polarity (positive or negative) of the battery terminal to which it is connected. [*Fig. 10* is a pictorial representation of such an arrangement.] The amount of charge on each plate is equal, but the polarities are opposite. The actual amount of charge that accumulates is determined by the area of the plates

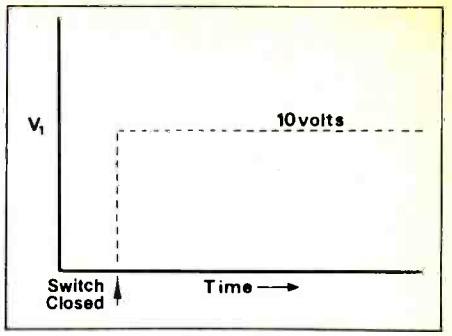

Figure 9

(we will assume that the two plates are equal in area), the distance between them and the nature of the material between them. For our discussion the intervening material will be dry air.

An important characteristic of a two-plate arrangement like the one we are considering is the amount of charge that can accumulate on the plates for a given applied voltage. Of course, since the amount of charge on each plate is the same but of opposite polarity, the total charge of the combination of plates is zero. But, for discussion purposes we will only consider the charge on one plate. The unit of charge is the "Coulomb," and we will use Q as the symbol for charge. Since we want to express the relationship between accumulated charge and applied voltage, we can form the fraction:

$$\frac{Q \text{ (in coulombs)}}{V \text{ (in volts)}}$$

This relationship describes a characteristic of our two-plate arrangement called "Capacitance" (C), and a device having capacitance is called a "Capacitor." The units of capacitance are (from the fraction formed above) *coulombs/volt* (read "coulombs per volt"), and the amount of capacitance equal to 1 coulomb per volt is taken as a unit called the "Farad" (in honor of Michael Faraday). In most situations the value of a capacitor is given in micro farads (written uf), or one millionth of a farad. Once all the possible charge is on the plates (an in-

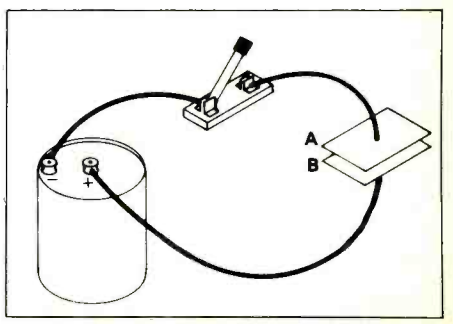

Figure 10

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stantaneous process), no more charge will reach the plates from the battery because of the electric repulsion of the "resident" charges. If the capacitor were suddenly removed from the circuit, with the charge still on the plates, what would happen? Remember, one plate is negatively charged, due to an overabundance of electrons, and the other plate is positively charged, due to the absence of electrons. Since opposite charges exert an attractive force on each other, the plates will tend to remain charged indefinitely (although in actuality the charge eventually leaks off). Thus a capacitor can be considered as an electrical energy storage device.

• • •

We will now take a look at conditions in a circuit in which both capacitance and resistance are present. *Fig. 11* is the schematic of such a circuit. The capacitor is located between points a and **b** and is represented by the standard schematic symbol.

Let us examine, step by step, the process that begins when the switch, S, is closed. [*Fig. 12* is a graph of the way current changes with time in the circuit of *Fig. 11*.] At all times, after the switch is closed, the sum of the voltage drops around the circuit must equal the applied voltage E_B. Written out, this is the equation:

$E_B = V_1 + V_C$

Substituting $I_T \times R_1$ for V_1:

$E_B = (I_T \times R_1) + V_C$

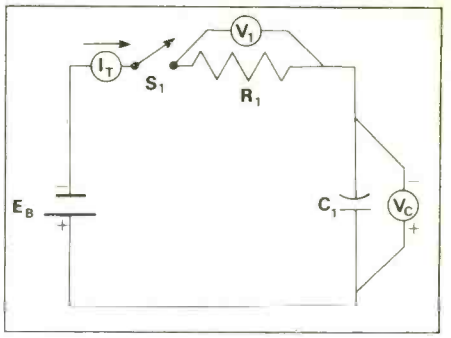

Figure 11

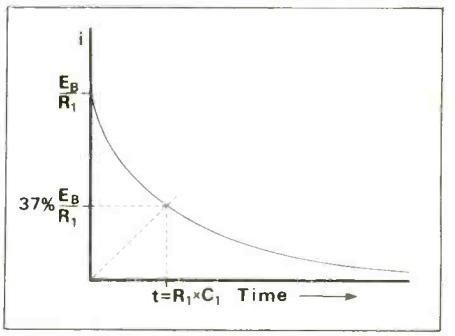

Figure 12

At the instant the switch is closed charge runs from the battery through to the capacitor plates. Since the same amount of negative charge leaves plate B as arrives at plate A, the circuit conditions for this first instant are as if the capacitor were short-circuited. What is the current in the circuit? Remember C is effectively shorted, so $V_C = 0$. Now take a look at the Kirchoff Voltage Rule equation written before:

$E_B = (I_T \times R) + V_C$

But $V_C = 0$, so

$E_B = I_T \times R$

or

$$I_T = \frac{E_B}{R}$$

This of course is just the total current from Ohm's Law, as if the capacitor were shorted. However, this current flows only at the instant that the switch is closed. After this instant, as

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So the more efficient the speaker, the less power and fewer amps you need.

Efficiency, Part II: Performance. The Fender 2-15 R delivers a sound pressure level of 107 dB at 1 watt/1 meter (pink noise) and handles 400 watts RMS continuous. The 1-15 HLR delivers 105 dB at 1 watt/1 meter (pink noise) and handles 200 watts RMS continuous. That's near 10% efficiency!

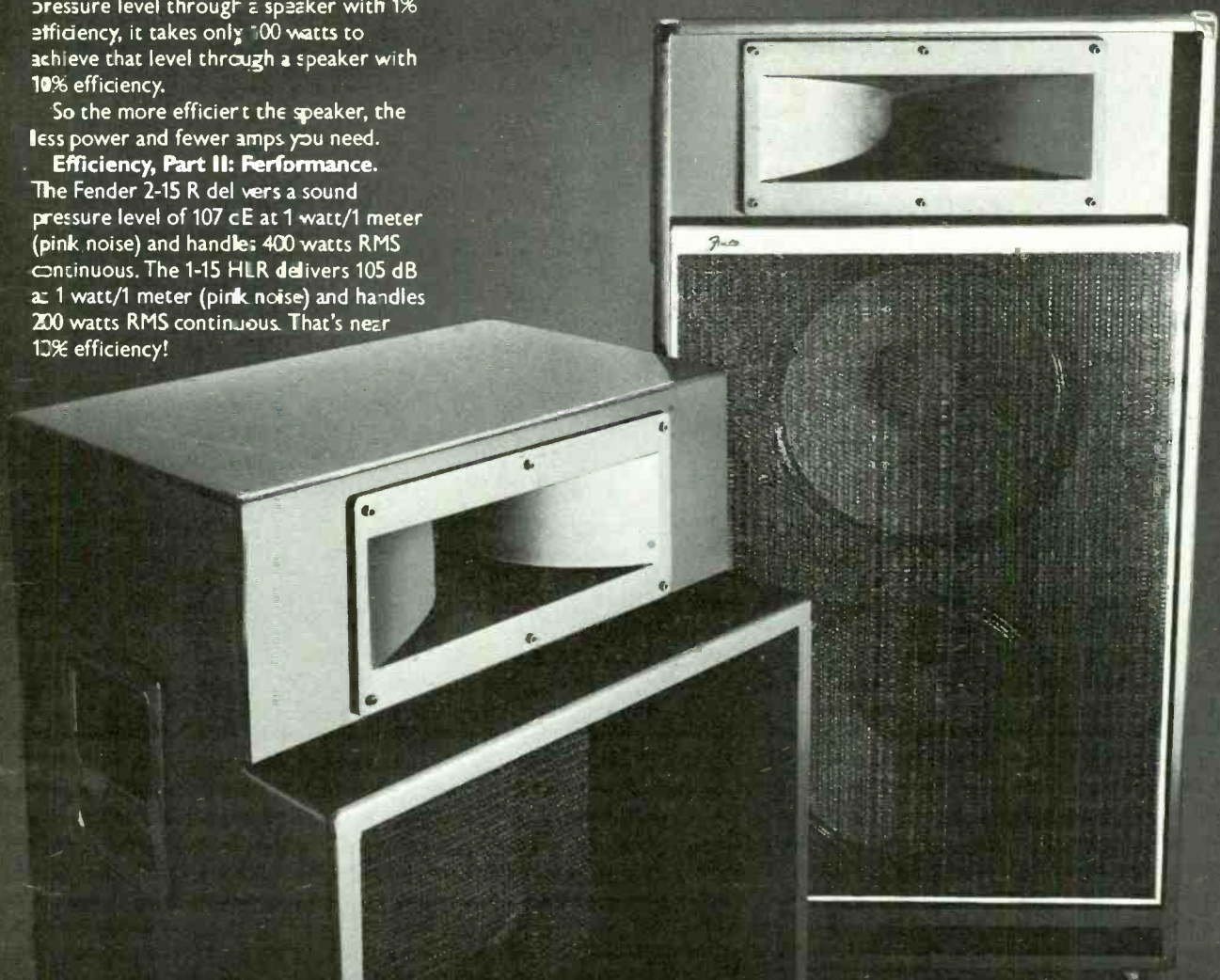
Both feature EV15L computer-optimized woofers with edge-wound aluminum voice coils that minimize heat failure. Flush-mounted compression horn/drivers that yield flat response without equalization.

Internal ground lift switches to keep power amps from grounding when the unit is biamped (which, incidentally, lets you separate horn and bass functions). Structural grade aluminum diaphragms that prevent fatigue. And sophisticated crossover using a maximally flat response 18 dB/octave filter.

Thiele-aligned cabinetry. Most important, both Fender cabinets are Thiele-aligned for maximum bandwidth and sound pressure level per cubic foot.

What have you got to lose? Ver fy the specs and the performance at your authorized Fender dealer.

The only thing you've got to lose is a lot of excess equipment.



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CIRCLE 56 ON READER SERVICE CARD

the capacitor is being charged up to its maximum capacity, the current begins to diminish. The rate at which it diminishes is determined by the value of R and C. Before we proceed, we must note that upper case symbols like E, V, Q and I are used for applied voltages, voltage drops, charges and currents (respectively) that do not change value, polarity or direction with time. For the time-variable counterparts of these values we will use lower case symbols, e, v, q, and i. Once the current begins to diminish the voltage rule equation becomes:

$E_B = iR + V_C$

Recalling that,

$$C = \frac{Q}{V} \text{ (or E)}$$

we can replace V_C,

$$V_C = \frac{Q}{C}$$

or, since the charge on the plates is changing with time,

$$v = \frac{q}{C}$$

Now,

$$E_B = iR + \frac{q}{C}$$

The value of the current i, or the charge q, at any time, is impossible to express in terms that are within the mathematical scope of these articles. However we can see that since current can not actually pass through a capacitor (a capacitor is actually a break in the circuit), i must eventually approach zero (so must iR), q must approach Q, the full charge capacity of the capacitor and, since iR approaches zero, V_C will approach E_B. The word "approach" is used because the final condition,

$E_B = V_C$,

according to the rather hairy math involved, is only achieved after waiting an infinite amount of time. It is unlikely that anyone wants to wait around for V_C to equal E_B. A good indication of how long it takes to charge a capacitor in a circuit containing resistance, can be gotten from determining how long it takes for q to equal 63% (approximately) of the final value of Q, or equivalently, when i drops to 37% of I. Not by magic, but through the application of mathematics to the laws of physics, these conditions occur at a time, t, that is equal to the product of R and C:

t (in seconds) = R (in ohms) × C (in farads)

The importance of the above quantity will be discussed in Part IV. But for now, how can the product of a resistance and a capacitance come out in units of time? Let's find out. From Ohm's Law and the formula for capacitance,

$$R = \frac{E}{I}$$

$$C = \frac{Q}{V}$$

Replacing E, I, Q and V with the units for these values, and remembering that

$$I = \frac{Q}{t}$$

the product of R × C becomes, in units:

$$\frac{\text{volts}}{\frac{\text{Coulombs}}{\text{seconds}}} \times \frac{\text{coulombs}}{\text{volts}} =$$

$$\frac{\frac{1}{1}}{\text{seconds}} = \text{seconds}$$

Seconds are units of time, and the product R×C is the time it takes for a capacitor to be charged to 67% of its full capacity charge. This time is called the *RC Time constant* of the circuit.

In Part IV we will examine capacitance further, and begin to look at magnetism and related topics.

Technics RS-M95. When price is no object. But performance is.

There are countless cassette decks to choose from when your object is price. But when your only object is performance, look at the RS-M95 with metal tape capability.

The heart of the RS-M95 lies in its quartz-locked direct-drive capstan motor and its computer-controlled tape tension system. Together, they team up to provide constant tape tension, remarkably low wow and flutter and complete immunity from speed inaccuracy. So when you record an A flat, you'll hear an A flat. No more, no less.

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You'll also get the advantage of the RS-M95's three-head configuration: Source/tape comparison. What you don't get is azimuth error, because our HPF record and playback heads are not only precisely gapped, they're also precisely enclosed in a single housing.

To add to the RS-M95's three-head configuration, we added double Dolby*. So you can monitor your tapes with the full effects of Dolby noise reduction.

Another glowing feature of the RS-M95 is its two-colored fluorescent (FL) bar-graph meters. They're fast, thanks to a device attack time of just 5 millionths of a second. And accurate, thanks to a Technics high-slew-factor operational amp for absolute wave and peak detection.

Also included are 4-position tape selectors with bias fine adjustments. A built-in 400 Hz/8 kHz test tone oscillator. A separate, coreless DC motor for reel drive. A microprocessor tape counter with triple memory functions and feather-touch controls.

Technics RS-M95. Its performance is its only reason for being.

FREQUENCY RESPONSE (Metal): 20-20,000 Hz (±3dB). WOW AND FLUTTER: 0.03% WRMS. S/N RATIO (Dolby in): 70dB. SPEED DEVIATION: ±0.1%.

*Dolby is a trademark of Dolby Laboratories.

Technics *Professional Series*

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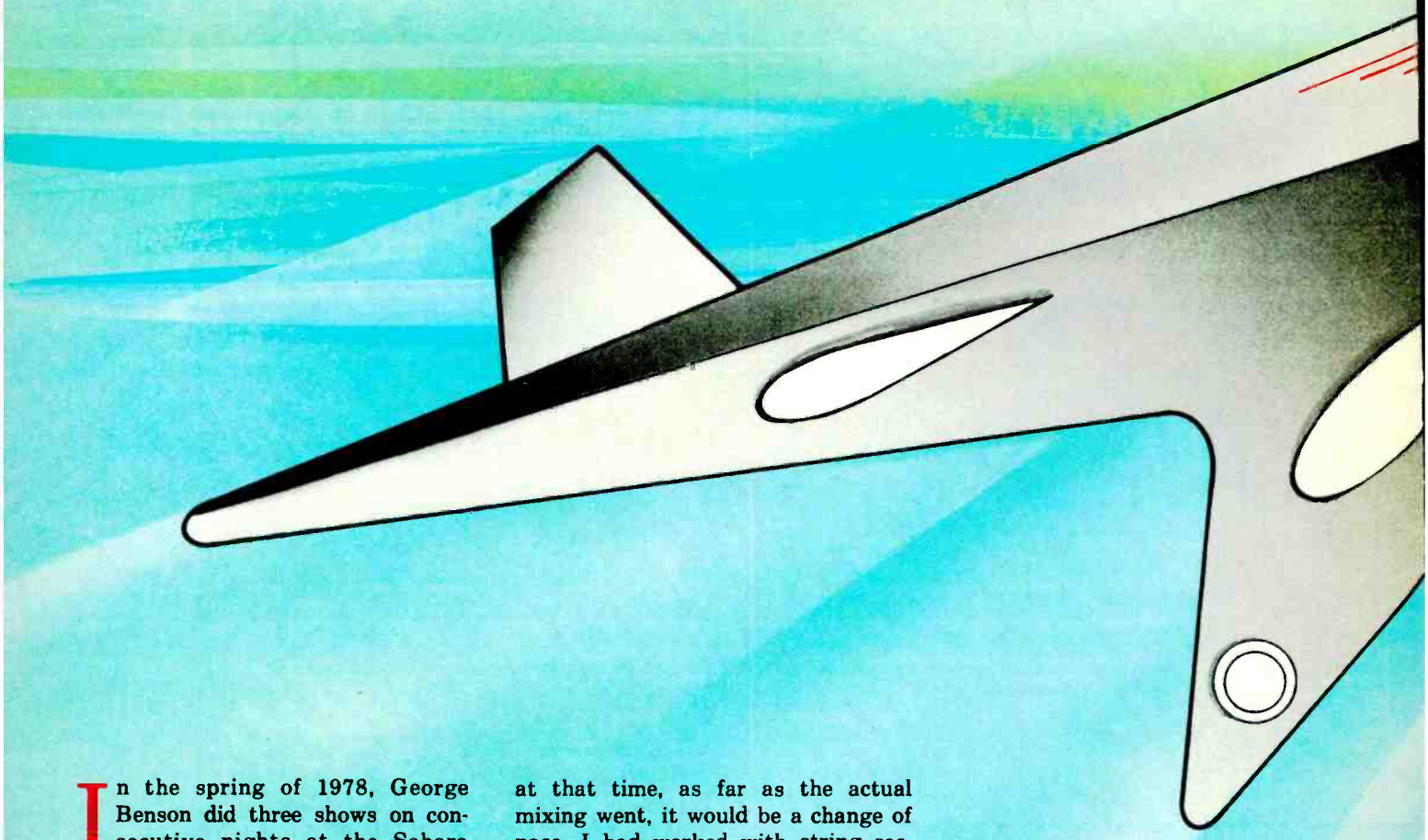
ON TOUR WITH GEORGE BENSON



In the spring of 1978, George Benson did three shows on consecutive nights at the Sahara Tahoe in South Lake Tahoe. These were the first shows we had done since our world tour that started in the fall of '77, and took us to Hawaii, Japan, Australia, New Zealand, France, England, Germany and Holland.

During the world tour we had used our regular band gear which is a fairly basic band set-up. On the way back from Europe, George and I talked at length about an idea he had. He wanted to add a stringed orchestra. For Tahoe he wanted twelve violins, four violas, four cellos, and three basses. I instantly accepted the challenge, if for no other reason than, at that time, as far as the actual mixing went, it would be a change of pace. I had worked with string sections before but had never mixed a show with one. It's a fact, I had no idea what I was getting into. This indeed was the beginning of the biggest challenge I'd accepted in a long time!

At first, I figured all the tried and tested ways of technically approaching this type of set-up. This approach was fruitless. The only way I'd ever seen this type of thing done was to put a Sony ECM-50 on the bridge of the violins and violas and close-mic the F holes on the cello and bass instruments, using the best mic available at that time. (Remember, this is a "live" situation and contrary to the popular belief that today's jazz-rock fusion groups work at relatively low volumes on stage, occasionally the volume on stage does reach a very "live" level. You can imagine how this affects the isolation of the string section.)

Tried and Tested, but . . .

Having never experimented with any other technique I went ahead with the tried and tested way. The Sahara Tahoe had plenty of the small ECM-50s. To tell you the truth, it worked, but that was about it. The sound of the violins was quite harsh, and the bleed from the rhythm section was more than what one might call negligible. The sound of the cello and bass instruments, however, was indeed what you

Terry Fountain is the sound engineer for the George Benson tours.

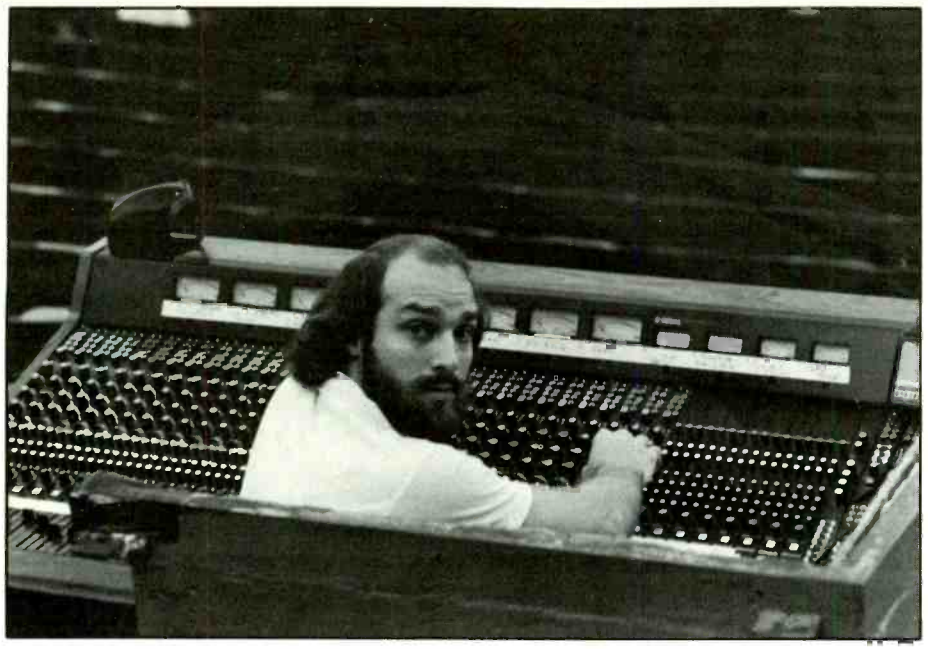
BY TERRY FOUNTAIN

might call negligible. After a long sound check and a lot of experimenting with graphic equalizers on the different string sections, an acceptable sound was achieved. Fortunately the audience was impressed and so was George. However, I knew then that I would have to keep trying. George decided to add string sections to about fifteen or twenty of the shows we were to do during the rest of our tour in '78. I did a great deal of experimenting on these shows.

We did a big show in Houston, Texas on September 30, 1978 at the Hoffines Pavilion. Since we were in Texas, we decided for a number of reasons to use Showco, Inc. for the sound and light production. One of the reasons was because my friends at Showco had told me about a Barcus-Berry pickup designed for violins, violas, cellos and contrabasses. They had been using these pickups on their current *Star Wars* laser tours.

I had very high hopes for the use of these pickups. The set-up for the Houston show was very smooth until it came time to install the forty pickups on the twenty-four violins, six violas, six cellos and four contrabasses. Besides the tedious task of installing forty of these very small pickups, I had to continuously promise these very rightfully concerned musicians that the grey putty-like adhesive, which Barcus-Berry supplies to attach the pickup to the bridge of the instrument, would in fact come off after the show. I told them they could have the remaining hair on my head if the adhesive didn't come off. Well, after the heat from the stage lights melted the adhesive all over the bridge of their instruments, the fate of my hair was doomed.

The pickups did, however, perform adequately. In this case, one could, in fact, say the amount of bleed into these pickups was negligible. The problem was, again, the sound of the instrument. Awful. This pickup needs more than a graphic equalizer. The signal from these pickups is a hi-Z pickup level. Barcus-Berry makes special 8-input mixers that are capable of being cascaded for the purpose of taking up fewer inputs at the main console. I found that if each musician didn't plug in correctly the whole string section made a terrible buzz until I found the musician that wasn't plugged in correctly. Unfortunately,

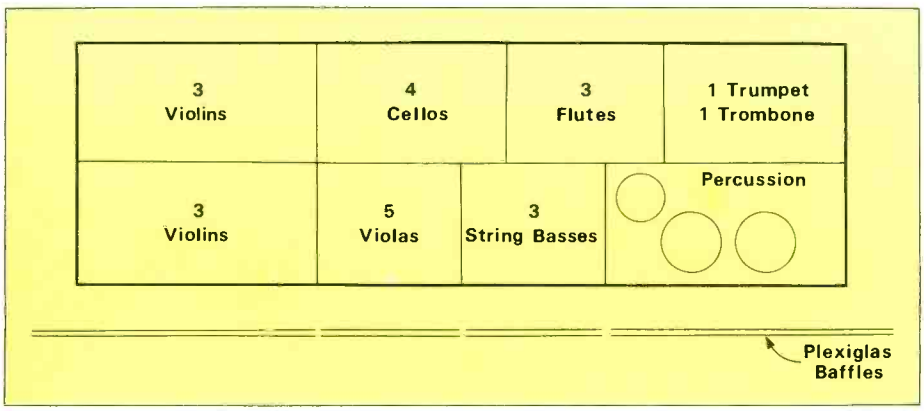


Sound engineer Terry Fountain at the main console.

due to the very tight touring schedule the orchestra only rehearsed the day of the show for about 1½ hours. Unless each musician played perfectly, and no one is perfect, any mistakes made by the string section were entirely too audible. So aside from the obvious hassles with this pickup, it picked up *too* good! But, I was not giving up.

There came a show in Toronto, Canada with a sound company that didn't have any Barcus-Berry pickups and also didn't have enough Sony ECM-50s for all the violins and violas. Instead I used overhead miking on the violins and violas, one AKG C-451 between each two musicians. Just opposite from what I'd always heard, I tried the Sony ECM-50s in the F holes of the cello and bass instruments. During sound check, I found the bleed from the rhythm section into the overhead string mics to be abominable! Alone, however, without the rhythm section, the string section actually sounded like a real string section. The key therefore was, of course, *isolating* the string section from the rhythm section. I dug around backstage and came up with the ticket—five pieces of 6 ft. by 4 ft. Plexiglas™ baffles. After putting this Plexiglas™ between the orchestra and the rhythm section, I heard success! I felt that, out of all the ways I'd tried, this way (accidently) sounded the best. It was also by far the easiest. By this time the 1978 tour was nearing an end. However, I was prepared for the 1979 orchestra tour.

In April of 1979 we began some very serious planning for the upcoming tour. Giving a lot of thought to gearing up the way we knew would work best from the beginning to the end of the tour. We wanted a system that we could set up



Orchestra Diagram for Greek Theatre (L.A.)

one way and make it work every night. Starting with the all-important choice of P.A. and lighting companies, we decided to use Showco. Certainly not an inexpensive choice, but as it turned out, we feel it was the best choice.

The tour was full of orchestra dates. Console equalization on a tour such as this is by far one of the most important factors in achieving success. A factor favoring the Showco "Superboard" mixing console. Three band fully parametric equalization on each of the thirty inputs is what I call adequate. Not to mention the eight quad submasters. We used a 4-way P.A. capable of covering seven to nine thousand people. For the larger shows we supplemented more speakers locally. The monitor system consisted of ten 3-way biamped floor monitors, two 4-way side monitors, one 30-in/8-out monitor matrix board and one very competent monitor mixer, Mark Hughes by name.

Rob Gleim, George Benson's lighting designer, produced the plan for the lighting system. It consisted of a front truss, a rear truss, two side trees and the Showco custom lighting console. Rob Gleim is also vice-president of Baltimore Stage Lighting. For this reason we called on Baltimore Stage Lighting to assemble a set of Plexiglas™ baffles. Nine pieces of 6′ x 4′ x 1/4″ Plexiglas™ along with the wooden supports to hold the glass during the show. That was the easy part. How about a road case for these nine pieces of glass? That wasn't so easy. With the glass in a case the weight was 550 lbs. Not exactly a case you would put anywhere in the truck pack. However, the glass was a necessity for overhead miking of the strings. Believe me, they worked.

As far as the mics for the strings go, after checking on all the string dates for the tour I found that most places we would be playing would have an assortment of proper condensor mics. To take up the slack just in case, we purchased some rather inexpensive Sony ECM-260P mics. They worked quite nicely when there wasn't anything better available. During that tour I tried a lot of different mics including, Neumann U87s and KM84s. Naturally these were the best I used. AKG C-451s and Sennheiser shotgun mics work quite nicely as well.

The Plexiglas™ baffles were probably the largest new piece of equipment we gathered up for the tour

this year. Scott Esterson, stage technician for George Benson, had his hands full, what with new cases to be built, stocking up on spare parts, guitar cords, guitar strings, etc. Lastly, adding a touch of elegance and aiding the sound of the stage monitors was our new carpeted stage.

The Band

From the group the Jacksons, came our new drummer Tony Lewis, certainly a valuable new addition to the group. Tony plays on Fiberglas™ Pearl Drums. His kit includes a set of six rack toms and one floor tom, all with pin strip heads. The pin strip heads eliminate the need for internal muffles. He uses a Ludwig snare mostly, but has a Rogers on hand for use as a spare. These and his Zildjan cymbals make for a killer set of drums. We mic the kit with an E-V RE20 on the bass drum, a Shure SM57 on the snare, an AKG C-451 on the hi-hat and Sennheiser 421s on the rack toms, with the Sony ECM-260Ps on the overhead cymbals. The sound produced from Tony's Fiberglas™ drums seemed to stand out significantly in almost every hall we played with very little effort in the equalization department.

The bass player, Stanley Banks, comes from another country—Brooklyn, New York. Stanley has been with George Benson since the *Breezin'* album [1976]. When I first started mixing "live" sound for this group Stan was playing a Fender Fretless bass guitar, but now has changed to a Fender G3 with Dean Markley strings. As far as the sound of the bass guitar goes, I believe Stanley has had better results with the latter. After trying many others, the Ampeg SVT bass amp and bottom [combination] has prevailed. Depending on the situation the bass direct is taken from the guitar pickup itself or from the SVT "preamp out." Stan is also the only person on stage that plays two instruments at once. While playing bass guitar with both hands, he adds a little percussion sound by playing tambourine with his foot. An interesting, creative touch.

On keyboards this year was Randy Waldman on grand piano, clavinet and Rhodes 73. Randy has played with such well-known performers as Lou Rawls, Frank Sinatra and the very great, late Minnie Ripperton. Such experience certainly qualified him to be the traveling Maestro on this orchestra tour, besides playing very excellent piano, which adds an unexpendable element to George's music. Randy succeeded in teaching to each and every different string section the music for the whole show in a couple of hours. Not bad, considering orchestra rehearsals were always conducted on the day of the show.

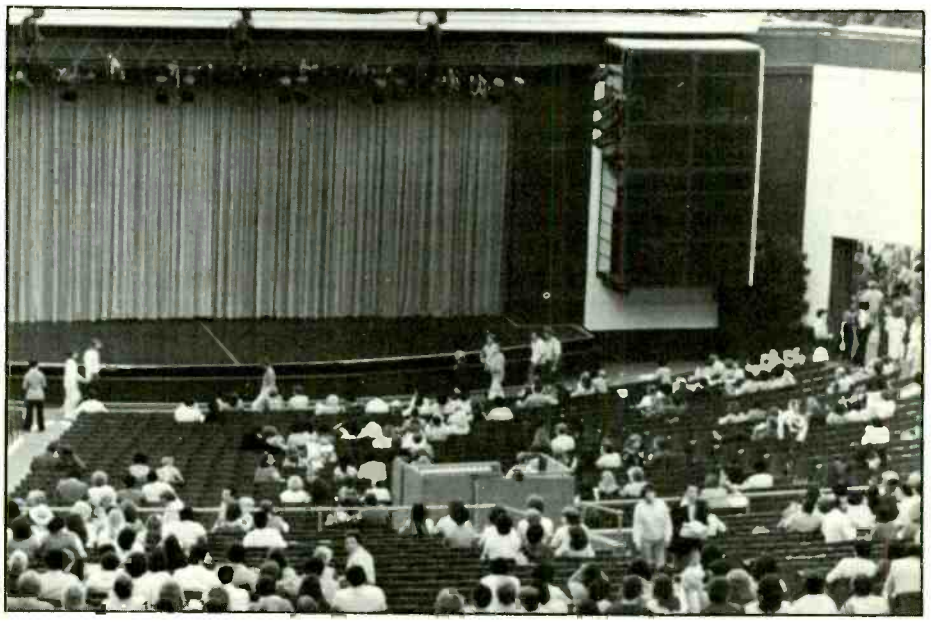
L.A. Orchestra Inputs & Mics

Instrument	Console Input	Mic
Violin	No. 1	KM84
Violin	No. 2	KM84
Violin	No. 3	KM84
Violin	No. 4	SM81
Violin	No. 5	SM81
Violin	No. 6	SM81
Viola	No. 7	KM84
Viola	No. 8	SM81
Viola	No. 9	SM81
Viola	No. 10	SM81
Viola	No. 11	SM81
Cello	No. 12	SM11
Cello	No. 13	SM11
Cello	No. 14	SM11
Cello	No. 15	SM11
String Bass	No. 16	SM11
String Bass	No. 17	SM11
String Bass	No. 18	SM11
Flute	No. 19	Beyer 69
Flute	No. 20	Beyer 69
Flute	No. 21	Beyer 69
Trumpet	No. 22	Beyer 69
Trombone	No. 23	Beyer 69
Tympani	No. 24	Beyer 260
Tympani	No. 25	Beyer 260
Tympani	No. 26	EV CS15
Toys (Perc.)	No. 27	EV CS15
Conga	No. 28	Senn. 421
Conga	No. 29	Senn. 421
Conga	No. 30	Senn. 421

For a natural sounding piano I always stay away from today's grab bag variety of piano pickups. AKG C-451s, Sony ECM-260s, or any comparable condensor mic does a far superior job for our application. Due to the fact that the stage volume did create a bleed factor for the piano mics, I found that especially on the larger pianos (preferably 9 ft. Steinways), miking the two highest holes on the sound board worked quite nicely. Bleed from the stage was nominal and the piano had that "tinkley" sound, creating the possibility for some very tasty solos.

From the Stevie Wonder group came a very wonderful addition to the group this year, Greg Phillinganes, on electronic keyboards—Yamaha CP-30 piano, Rhodes 73 piano, Arp String Ensemble and a MiniMoog. All these keyboards were taken direct with pre-pedal and post-pedal directs taken from the Arp and MiniMoog. I did this so the action of the volume pedal on these two instruments affected only the stage monitors and not the house mix. Greg's keyboard solos were an extra added attraction. There have been a few times when he has stolen the show by playing the MiniMoog while holding it over his head.

Last but not least, often called a legend in his own time, is Phil Upchurch. Phil was playing a sunburst Ibanez GB-10. In fact, the third GB-10 ever made. George gave Phil this guitar. However, Phil has recently acquired a new guitar from Peavey, a T-60. For Phil's type of guitar playing the T-60 has been working great. Phil uses a lot of special effects. The new guitar seems to bring out the sound of



Partial view of the Benson show P.A. at the Greek Theatre in Los Angeles.

Phil's effects, causing a more significant change in tone. Phil's sound ranges from a low level to a pretty high level many times during the show. The tone of the T-60 seems very "fat" at low levels as well as at high levels, depending upon his pedal action. Phil believes this guitar has tone and feedback controls precise for his style. These things, of course, make my job that much easier.

Phil's sound ranges from a low level to a pretty high level many times during the show. The tone of the T-60 seems very "fat" at low levels as well as at high levels, depending upon his pedal action. Phil believes this guitar has tone and feedback controls precise for his style. These things, of course, make my job much easier.

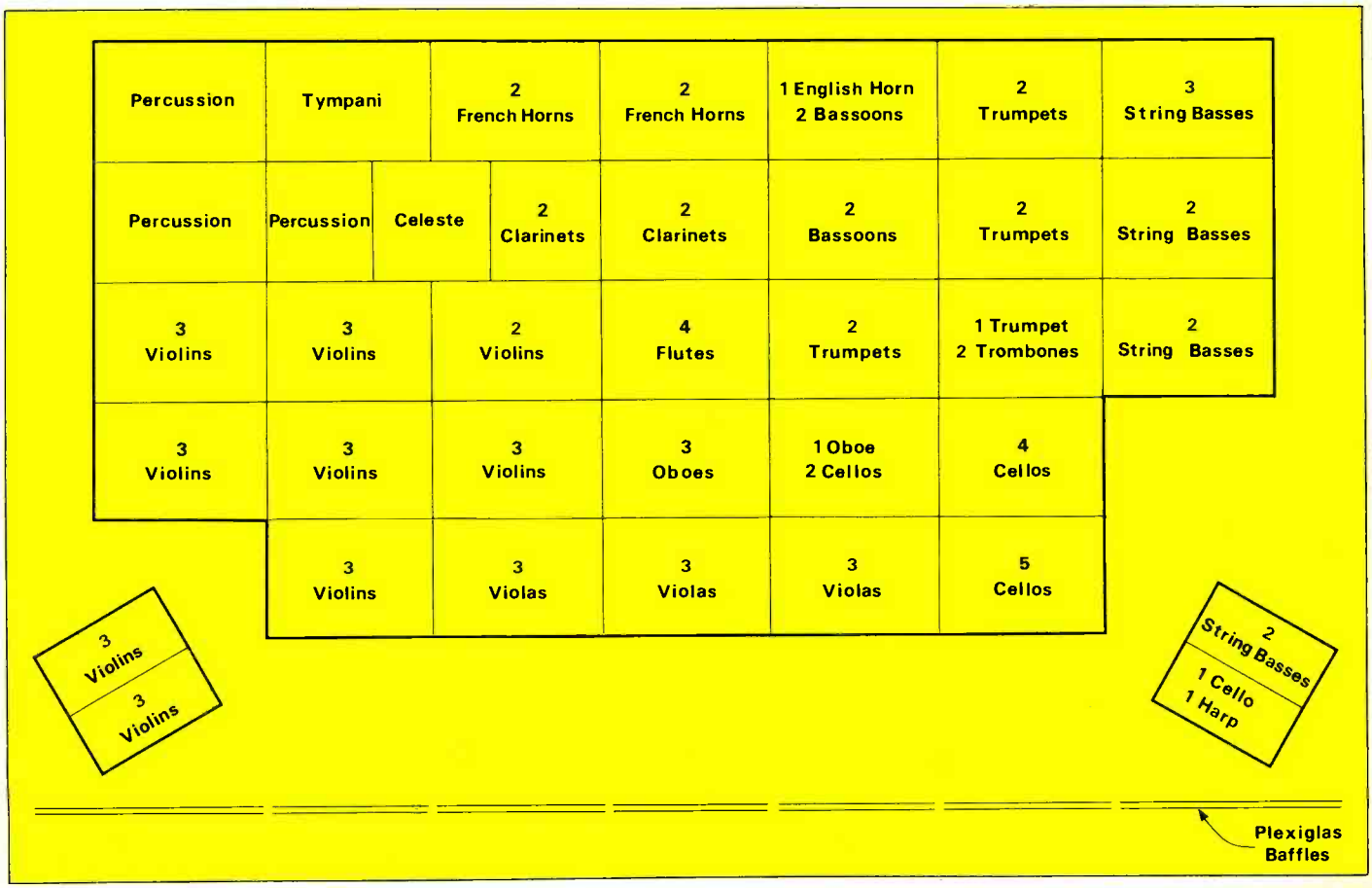
Phil's pedals and effects change from time to time. Now he is using a newly designed pedal board, from Paul Rivera Research. It has two built in AC power supplies, one for the effects and one for the power lamps that have been installed on each device. Some devices he is currently using include: a Boss Chorus; Mutron III; MXR graphic; MXR flanger; MXR Distortion +; Morley Power Wah; Ibanez Flying Pan; and an Echoplex. All this going into a vintage Peavey [amp], at least ten years old. All I can say about this amp is, "They don't make 'em like they used to." These factors combined with a talent like Phil Upchurch result in a sound that no one can match.

(Phil recently called on me to help solve a couple of mysterious noises in his home studio. Upon reaching his home I found a very clean slightly vintage 8-track Scully. The 8 tracks were mixed down through a modified AM4 Langevin, 12-in/8-out console. The noise problem happened to be some bad connectors and bad sliders in the board. Actually, with all of the auxiliary tape decks, special effects, a variety of guitars and other musical instruments, the studio's really not what one might call a regular run-of-the-mill "home" studio.)

There is another person [in the band], who stands out alone, and that is why I left him out of the lineup. George Benson, of course, stands steadfastly by his old blond Ibanez GB-10, and rightly so. For George it is the only guitar, and he feels the use of any kind ob pedals or special effects would be a crime. George believes in keeping a good thing the way it is. He uses Darco medium-gauge flat-wound electric strings. An interesting thing about the picks George uses—Fender medium size. The thing is, Fender makes two types of medium picks. One type has the word "Medium" in small letters, the other in larger letters. George plays with the latter, after Scott Esterson flattens the picks slightly by sanding the tip with emery cloth. This gives the pick the feel George is used to.

Depending on the situation I use an SM57 or an SM58 on both George and Phil's amps. For George's vocal mic I've discovered a new mic on the market. The Audio-Technica ATM41. George likes this mic because it gives his monitor sound a deep, rich feeling. This tends to make him feel more at ease while singing because he doesn't have to work as hard as he otherwise might to get the sound he wants.

Now that we've met the band and have familiarized ourselves with the equipment and its potential problems, let's discuss two of the more difficult shows we did this



St. Paul Minnesota Orchestra Diagram

St. Paul, Minnesota Orchestra Mic. List

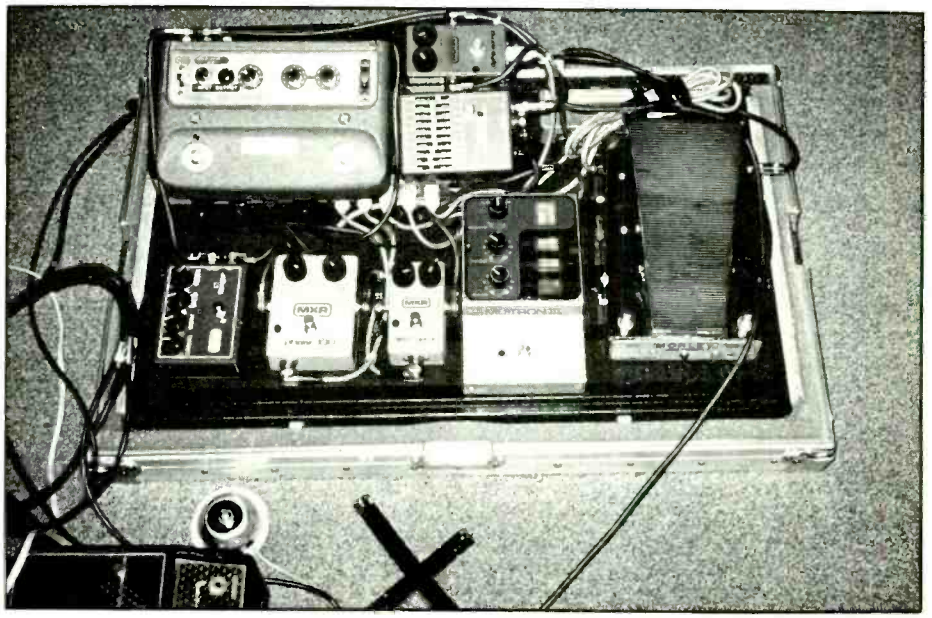
	Instrument	Mic
1	Violin (1 mic/2 musicians)	AKG 451
2	Violins "	AKG 451
3	Violins "	AKG 451
4	Violins "	AKG 451
5	Violins "	AKG 451
6	Violins "	AKG 451
7	Violins "	AKG 451
8	Violins "	AKG 451
9	Violins "	AKG 451
10	Violins "	Sony ECM260P
11	Violins "	Sony ECM260P
12	Violins "	Sony ECM260P
13	Violins "	Sony ECM260P
14	Violas (1 mic/2 musicians)	AKG 451
15	Violas "	AKG 451
16	Violas "	AKG 451
17	Violas "	Sony ECM260P
18	Violas (1 for 1)	Sony ECM260P
19	Cello (2 Y'd)	ECM 50
20	Cello "	ECM 50
21	Cello "	ECM 50
22	Cello "	ECM 50
23	Cello "	ECM 50
24	Cello "	ECM 50
25	Bass "	ECM 50
26	Bass "	ECM 50
27	Bass "	ECM 50
28	Bass "	ECM 50
29	Bass (single)	ECM 50
30	Flute	SM57
31	Flute	SM57
32	Flute	SM57
33	Flute	SM57
34	Trumpets (2 Y'd)	SM58
35	Trumpets "	SM58
36	Trumpets "	SM58
37	Trombone	SM58
38	Trombone	SM58
39	Bassoon	SM58
40	Bassoon	SM58
41	Bassoon	SM58
42	Bassoon	SM58
43	English horn	SM57
44	French horn	SM57
45	French horn	SM57
46	French horn	SM57
47	French horn	SM57
48	Clarinet	SM57
49	Clarinet	SM57
50	Clarinet	SM57
51	Clarinet	SM57
52	Celeste	SM57
53	Harp	ECM50 (soundhole)
54	Tympani	Senn. 421
55	Tympani	Senn. 421
56	Tympani	Senn. 421
57	Perc.	SM58
58	Perc.	SM58
59	Perc.	SM58

year, the first in St. Paul, Minnesota, on June 20, 1979. This one show stands out from the rest because we utilized a rather large orchestra. The usual string section we used numbered about twenty-three pieces. This event included the entire Minnesota Orchestra, some ninety pieces. Besides the larger than normal orchestra, the show was held in the Civic Center. You know, one of those 15,000 seaters that has a "multiminute decay time" after you turn the P.A. off. Fortunately, we knew about his gig far enough in advance to properly prepare for it. We therefore opted to load in and set everything up a day early.

I knew there would be no way to achieve the level needed to cover this hall with our standard 4-way Showco P.A., so I contacted a local (and dependable) P.A. company called Naked Zoo Lights and Sound. I chose them to hire extra hanging P.A. speakers, a Yamaha PM-1000 32-channel mixer for the orchestra and a special set of mics for the orchestra. Naked Zoo loaded-in early, the day before the show. Their hanging P.A. was going up in the air about the time the Benson and Showco crew arrived. The first thing I did was check out the AC situation. From many previous set-ups similar to this one, I've found that when mixing [hiring two or more companies for the same date] P.A. companies it pays to be careful early in the day. If the power distribution isn't from the same source for both P.A. companies, it can be very embarrassing when it comes to sound check time. (As you probably know, during the sound check is no time to be figuring out where the hum is coming from.) We were quite fortunate at this set-up. After both P.A. companies were finished interfacing their respective systems we had a nice quiet P.A. Even though some of the speaker cabinets were different in design there were no apparent phasing problems.

I ran the Showco P.A. as usual, out of the quad master on the Superboard. The Naked Zoo P.A., which was suspended from the ceiling, was run from a mono output on the Superboard. Together the different systems performed nicely—something not easily achieved in today's large hockey rinks. The basic plan was to use the Yamaha PM-1000 for mixing the band and the string section of the orchestra. Miking a stage set-up like this seems endless and with some 150 mics on stage it can get very confusing. Naturally we had to "Y" several mics together, some were resistively combined and some were mixed down through small mixers on stage. (Refer to the stage layout and mic reference charts.)

There were no rehearsals until the day of the show. The sound check would have to be done during the rehearsal. The show began with an overture of several songs that George has made famous. The 20-minute overture included only the rhythm section and the entire orchestra at first, with George coming out following the overture to do a set with only the string section and the flutes. Just before the rehearsal started I had each section of the orchestra play separately in order to set the equalization. Then, during the rehearsal we attempted to reach a basic balance for the entire orchestra. I'll save you the



Phil Upchurch's "Pedal Board," created by Paul Rivera.

aggravation of "doing" a sound check with an orchestra of this size (in a very short amount of time) by not elaborating on the details. It is something you just have to learn to love. If you don't learn to love it you'll probably lose your sanity. (You'll probably lose your sanity anyway, so you might as well enjoy it in the process.)

The first thing I noticed was a pretty bad feedback problem right around 100 Hz. With a third-octave EQ on the orchestra mix—brought from the PM-1000 to an input on the Showco Superboard—the problem was solved by rolling off a little low end.

The string section is hardest to EQ. To make the strings sound like strings, instead of a ukulele is not as easy as one might imagine. Even with the overhead miking you need all the EQ you can get your hands on. I usually "peak EQ" the strings with the parametrics. I do this one input at a time while dealing with each of the three bands of EQ separately. For example, the lows, with the gain cut completely on the mids and highs. Start at full gain on the lows with the narrowest bandwidth setting on the slope control, slowly bringing the input fader up while rotating the frequency select control. You will notice spurious resonant frequencies as you rotate the control. The trick is to figure out when you hear the resonant frequency of the actual instrument. Then you adjust the bandwidth control to include the correct amount of surrounding frequencies, depending on overtones produced by the hall. The gain control for lows is to be adjusted appropriately as well. If this procedure is properly done on the mid and hi bands of EQ as well, then you have, in a sense, tuned this input module to the sound of the instrument involved.

After going through this procedure for the violins, violas, cellos and string basses, I begin listening to a basic balance of the strings. Even after all this I usually have to roll a little off the very top end on the string's graphic. This is to get rid of the harsher tones sometimes caused by the actual bowing of the instruments.

If you're fortunate enough to get all this done during the rehearsal and you still have your sanity, then you're ahead of the game. Only the band instruments are left to check, and at this stage of the game, that's a cinch. Getting the EQ right for the band instruments is a piece of cake—after doing it every show for two years. It's when I have to do these "two-hour-ninety-piece-quasi-rehearsal-sound-check-miracles" that things start to get pretty tricky. Nevertheless, by showtime I felt pretty confident about the whole thing. It was certainly one of the biggest shows I ever had anything to do with.

George Benson and Phil Upchurch on stage.

Needless to say, the overture was a real rush. Fortunately the audience got off on it as much as I did. The only problem was some 100-cycle feedback from the orchestra mix. Anyone that has been in a situation where you're turning up a single submaster with 32 inputs going through it knows how hard it is to get rid of all feedback problems. It took a little more rolling off of the low end on the orchestra mix graphic EQ than I had anticipated. From then on the show went quite well. Believe me it was very satisfying to be controlling the sound for such an event. Even though my brains were scrambled by the time the show was over. George was equally satisfied with the show. And if George and the audience are satisfied, so am I.

The other show I wish to tell you about from this year's tour, is our five nights at the Greek Theater in L.A., July 25 thru 29. Again it was advantageous for us to set up a day early, what with union schedules, etc. We would never have made it in one day. This show not only had a thirty-piece orchestra, but as an "extra-added-attraction" the Los Angeles ballet was performing on stage with George. At first I visualized this whole thing as a three-ring circus. The ballet made a lot more difference in the stage set-up than I had thought it would. In order to see the orchestra, which was behind the drums and rhythm section, four-foot-high risers had to be constructed. The band equipment was set on two-foot risers in front of the four-foot risers. The risers created the effect of a three-level stage: the ballet was downstage, George Benson and Band centerstage on the two-foot risers and the orchestra on the highest level, upstage.

For this show we were going to be using only the Showco Superboard. The house sound in the Greek Theater was more than adequate. Stan Miller of Stanal Sound met with me several days prior to the set-up and promised me a brand new Yamaha PM-2000 32-channel mixer, but I politely told him I was going to bring in my Showco Superboard so he needn't worry about getting the new Yamaha. However, my plans were drastically changed when Showco informed me that their Superboard was needed for a KC and the Sunshine Band tour. Frantically I called on Stan Miller to produce the PM-2000. Stan, being one of the more professional sound people I've met since I left Showco, produced the board. It was in fact so new that it had never been plugged in.

I mixed the drums down on another brand new Yamaha: a prototype

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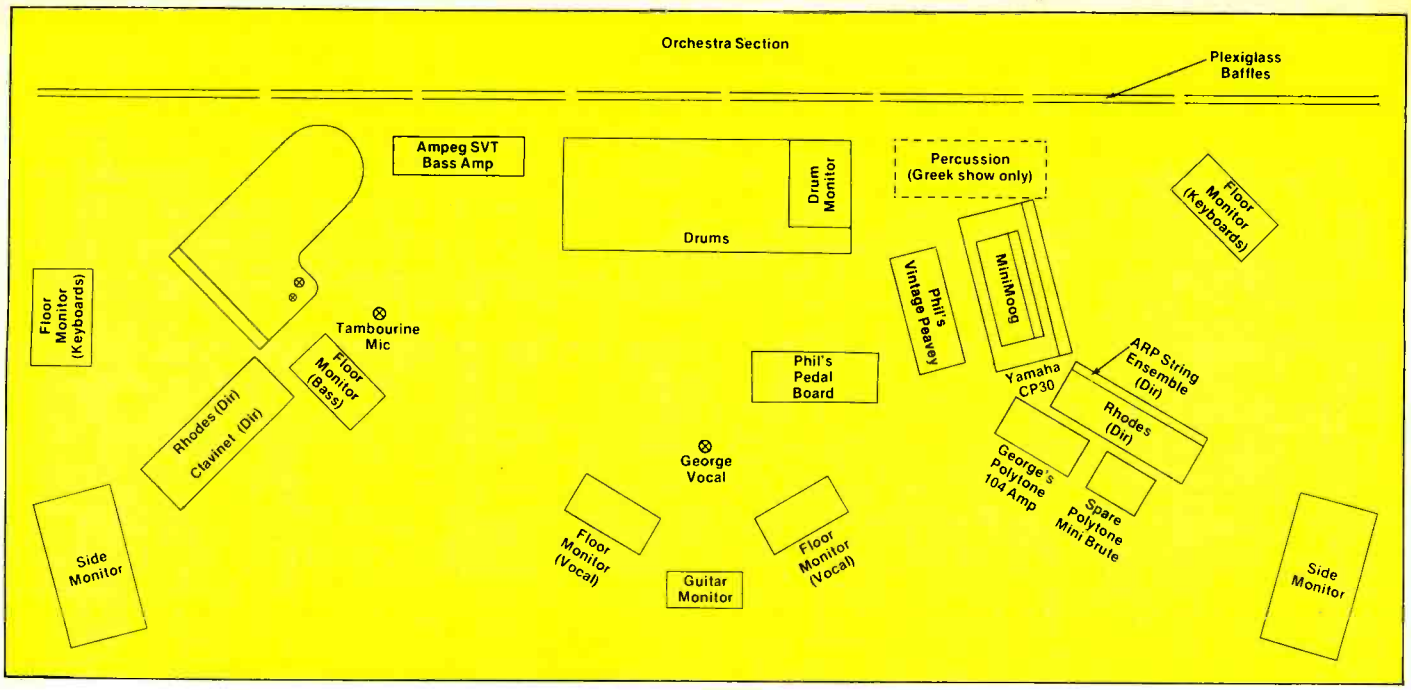
CIRCLE 81 ON READER SERVICE CARD



George Benson Group (Stage Diagram)

Yamaha 430 8-input console with 3-band EQ on each input. The flutes and congas were in a regular Yamaha 430 with 2-band EQ on each of the eight inputs. The eighth input of this mixer was used for a submix from the left output of yet another outboard six-input mixer, a PM-180, which contained the tympani drums. The right output of this third outboard PM-180, which contained the horns, went straight to the submaster input assigned horns on the PM-2000. The outputs for drums and percussion from the other two outboard mixers went straight to their assigned submaster inputs on the PM-2000. All this was done so that there would be enough remaining inputs in the PM-2000 for plenty of string mics and the rest of the regular band mics. Every input on every mixer was used. Can you imagine checking fifty-two inputs before every show?

Before mixing on the PM-2000 I had thought there was no other board for the road except a Showco Superboard. I have mixed on a lot of consoles, even ones in foreign countries that didn't have English calibrations. Finally I've found something that compared. I was delighted with the response and feel of this new PM-2000, 32 inputs each with 4-band EQ. Each band of EQ has + or − 15 dB gain and a five-position frequency select control. The sound of this EQ is what impressed me the most. I had been worried about not having parametric EQ for the string mics, but a proper sound was achieved with ease. Again the string mix needed a little graphic adjustment on the low end, but nothing drastic.

The monitor system consisted of a Yamaha PM-1000 modified with eight discrete sends. With the Stanal biamped floor monitors and three-way side fills, monitors simply weren't a problem. Of course, with the Plexiglas™ baffles between the orchestra and rhythm section, the orchestra couldn't hear anything, but a floor monitor with the grand piano and the conductor's voice running through it kept the section in touch with what was happening on stage. Incidentally, this is tricky because the monitor bleeds into the string mics, so extreme caution must be taken while mixing this monitor, especially during the more delicate string parts, when the string mics will be most susceptible to bleed from the monitor.

The Stanal Sound system and the PM-2000 were a very quiet sounding combination; not even so much as a little hiss. This really made the sound check a breeze. Just before show time, however, I thought my ears were playing tricks on me. Before each show I always listen very carefully to whatever P.A. system I'm using. This time I noticed the right side of the P.A. was lacking high end when compared with the left side. Stan Miller agreed to check into it for me. Sure enough, the ballet had been rehearsing—during our dinner break—downstairs near the amps for the P.A., and had unknowingly kicked out a couple of plugs for the high frequency amps. After figuring this out, I knew things were going to go good the rest of the way.

The show probably was one of the best I'll ever mix. The equipment was top-quality and so were the sixty-plus people performing on stage. The audience got its money's worth, that's for sure, and I think I can safely say "that's what counts." They were witness to the result of a lot of planning on everyone's part. It was a performance unlike any other on the road today. The audience knew it too!

George Benson Band

Instrument	Console input	Mic
Bass Drum	No. 1	RE20
Snare	No. 2	Beyer 201
Hi-Hat	No. 3	Beyer 201
Hi-rack toms	No. 4	Senn. 421
Mid-rack toms	No. 5	Senn. 421
Lo-rack toms	No. 6	Senn. 421
Floor tom	No. 7	Senn. 421
Ovhd Cymbals	No. 8	ECM 56
George Vocal	No. 9	ATM 41
George's Guitar	No. 10	SM57
Phil's Guitar	No. 11	SM57
Bass Guitar	No. 12	Direct
Tambourine	No. 13	SM56
Clavinet	No. 14	Direct
Rhodes 73 (SR)	No. 15	Direct
Grand Piano (Hi)	No. 16	EV CS15
Grand Piano (Lo)	No. 17	EV CS15
Yamaha CP30	No. 18	Direct
Rhodes 73 (SL)	No. 19	Direct
ARP String Ens.	No. 20	Direct
MiniMoog	No. 21	Direct

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CIRCLE 88 ON READER SERVICE CARD

By Allan Kozinn

For several generations of New Yorkers and visitors to the City, it seems a Christmas and/or Easter visit to Radio City Music Hall was a requirement of youth. Back in the thirties and forties, I'm told, going to Radio City was a special treat, something everyone in the family looked forward to. When I grew up, in the early Sixties, I thought that Radio City just wasn't all it was cracked up to be. As far as I could see, it meant being forced into a little suit and tie, standing on line for an hour or two and then having to sit through an awfully dull stage show just to see a wholesome Mary Poppins-type film. Like any kid of the time, I felt the whole routine was hopelessly old-fashioned, and would have preferred bicycling down to the local cinema to catch an unadorned showing of *Strategic Air Command*.

As a budding music enthusiast, something else impressed [depressed?] me about Radio City back then—an impression that was confirmed about a year ago when, believing that the place was to be torn down, I decided to see one last Radio City show, for old times' sake—and that was that the *sound* was absolutely dreadful. Even if you wanted to pay attention to the pre-film escapades, you couldn't understand anything that was said or sung on stage, and when the band played, the result was an acoustic mush. Chances are, of course, that the drop in attendance which recently threatened the existence of the hall can be more easily attributed to a lack of interest on old-time entertainment than to Radio City's airplane-hangar sound. Nevertheless, the hall's management, on examining the theater's assets and liabilities in an effort to discover the reasons for its declining appeal, realized that with so many sonically superior venues around town, Radio City had little chance of pulling in an audience of *any* kind—whether for rock concerts, stage shows or ballet—unless the antiquated sound system was revamped.

Enter Robert N. Estrin, president of Filmways Audio Services. Although Audio Services has been in existence for only three years, Estrin has established it as one of the premier suppliers of temporary sound systems for annual or one-time-only events and "spectacular" productions, providing both "live" sound reinforcement and monitoring and simultaneous television audio feed. If you've seen the half-time shows at the Orange and Super Bowls, of if you've watched the Boston Pops, the Academy Awards, the Grammys, the Country Music

Awards or the Miss America Pageant, you've heard Estrin's work.

(Estrin had also done some sound work at Disneyland and Disney World, so when Disney's Vice President in charge of Entertainment, Bob Jani, became President of Radio City, it was to Estrin he naturally turned for some sound advice.)

"What Radio City wants to do," Estrin explains, "is to turn the building as a whole into a total entertainment center, rather than just a film house with a stage show. At the moment, they are running one show twelve times a week, and showing classic films every morning. On the one day that the regular stage show isn't being shown, they're doing concert-type performances. They're also planning to bring in proscenium-type shows. For instance, Snow White and the Seven Dwarfs is set for October, and then there will be a Christmas spectacular in the old Radio City style. Next summer, they plan to open a dinner theater on the roof. And they would like to make sound connections between the hall and Plaza Sound [recording studios], which operates in the building, so that productions can be recorded right off the stage.

"The idea is to get enough things going to keep revenue coming in, so that Radio City is not the gigantic money losing situation it used to be. They've come to realize that because of the way film distribution works today, there is no way that films can support the hall as they once did. So that they want to do now is use the building as a sort of production facility for legitimate producers. They will not only present shows of all kinds right here, but package shows here to be sent on the road or for use at conventions and industrial gatherings. Of course, Filmways Audio Services is really only involved in installing the sound system to be used for productions that take place here. But because of the flexibility required by Radio City's overall conception, we've had to build the system in such a way that anything in the hall can be patched to anything else. There had to be an ability to interface with the studio upstairs as well as an ability to feed audio from the stage to "live" radio and television facilities without ultimately disturbing the audience."

If the ultra-flexible design criteria seem challenging in themselves, they became even more so when Estrin was made aware of the limitations within which he and his crew had to work.

One reason Radio City is still standing is that it has been designated as an historical landmark. Which means that absolutely nothing can be done to change the hall's appearance, inside or out. Things might have been somewhat simpler for Estrin if, for instance, he could have hung a speaker cluster from the ceiling and placed sound reinforcement columns along the side walls. However, since the original sound system had its speakers hidden in coves in the proscenium arch, that's where Estrin had to put his speakers.

View of one of Radio City Music Hall's sound rooms.

"We did insist on one modification," he admits. "The old mixer's position was on the projection level, all the way at the top of the hall, in a small booth set behind a wall. We were adamant that if the mixer was not out in the house and in a fairly reasonable position, it wasn't worth the trouble to do anything else with the system. You cannot do "live" mixing in [from] another room. So, they allowed us to displace about a dozen seats on the second mezzanine, right at the front."

Although the equipment in the mixer's position can be changed to fit the production, at present the sound engineer has at his disposal two Yamaha PM-430 consoles for submixing—one to premix the brass and one for the drums—and one 16-channel Yamaha PM-1000, into which the submixes are fed. A PM-1000 32-track mixer is also available, as are several smaller Yamaha consoles.

Most of the electronics in the system, it turns out, are made by Yamaha. The power amps, for instance, are P2050s (nine of them), P2100s (five) and P2200s (seven).

Asked how antiquated the hall was when he first entered it, Estrin, who appears to be in his mid thirties, smiled and asked his assistant, Tim Kerr, to bring out a few exhibits. The first was a large, heavy, rectangular RCA PB 31 microphone, dating from around 1933. According to Kerr, about two dozen of these were in active service until Filmways Audio Services stepped in, as were about half a dozen RCA PB 77s, which were about twice as big. A couple of years ago, the system had been minimally upgraded to include six BGW 250 amplifiers and a modified Kelsey console, but these were being fed signals from old RCA Rainbow mixers, which use 3-inch [rotary] knobs instead of the typical modern mixer's slide controls.

However, Estrin says, the film sound system—which is separate from the stage sound equipment—is fairly modern, having been upgraded over the years to accommodate such developments in film sound as Dolby.

"The hall itself has good natural acoustics, so long as you avoid hitting the walls, which, because they are curved, produce strange reflections. For the classic performer who does not use a microphone, though, the acoustics are amazing. We've found

working here that if two people are on the stage talking normally [at normal levels] the mixer up in the second balcony can hear every word without difficulty. So, one of the problems facing the hall technicians is convincing rock bands who play here that they don't need a lot of amplification. Do they listen? Well, bands are bands. Fortunately, Filmways is not involved in the operation of the system. The house has a sound crew, which we are training to run the system, and they're the ones who have to deal with people who come here to play."

The first change Estrin and his crew made was to replace all the hall's wiring. According to Estrin, the wiring he found was lead shielded, about an inch thick, and, when it was all piled on the junk heap, weighed about three tons. Most everything was hard wired, just the opposite of Estrin's current ultra-patchable system. About 35,000 feet of color-coded replacement wiring had to be installed, and Tim Kerr was put in charge of wiring the nearly 2,600 patch points. According to Kerr, that task took about nine days during which he worked approximately eighteen hours per day.

"You see," says Estrin, "the other complication was that we had only thirty days in which to get a working system into the hall. We scheduled it so that the initial phase—a week and a half—was spent mainly tearing things out. I had sixteen electricians removing old equipment and pulling out wiring, and while they were doing that I had a crew assembling cabinets, horns and drivers so that as soon as the area was clear we could hang the speakers. We couldn't work quite as scientifically as I like: between the time the speakers were hung and the time we were able to aim them was about a week, because although we had the speakers and amplifiers in place, the cables weren't ready. So at first, a lot of aiming was done by eye."

The primary speaker locations are at the very top of the proscenium arch, about ninety feet above pit level, and just over the front of the orchestra section. Aiming them was tricky, because if the speakers were aimed straight out, they'd be blaring directly into the third mezzanine. Therefore, they are aimed slightly downward, while the speakers in a pair of lower coves are pointed upward. A third pair of lower coves has been made available to the Filmways crew, and although Estrin was not going to use them at first, he now thinks he may have to, in order to reach the orchestra seats that are underneath the mezzanine—currently the worst seats in the house from a sonic standpoint.

The speakers in use at the moment are primarily Altecs with Yamaha F-1030 crossovers. There are ten Mantaray horns, ten 288 16G compression drivers and twelve 841 LF woofers. The 841s are housed in cabinets designed by Stan Miller and marketed under the name "Stanel Screamer Bass Bins."

Estrin has been having a problem with his horns, though. Apparently, the grills that shield the speakers from view also filter out too much of the high frequency signal for Estrin's taste, and he's planning on replacing them with either JBL 2441s or the equivalent model from Technical Audio "depending upon who can deliver first." (The change to alternate horns should have been made by the time this is published.)

When the hall opened for the preview of "A New York Summer," on May 28, 1979, about 60% of the system was finished and installed. Obviously, the priority functions were those used in the show. Beyond the basics of amplifiers, consoles and speaker systems, these priorities included a tape room, just off stage right, where pre-recorded music used in the production is fed to the speakers and stage monitors from two Ampex four-track machines and eight Spot-

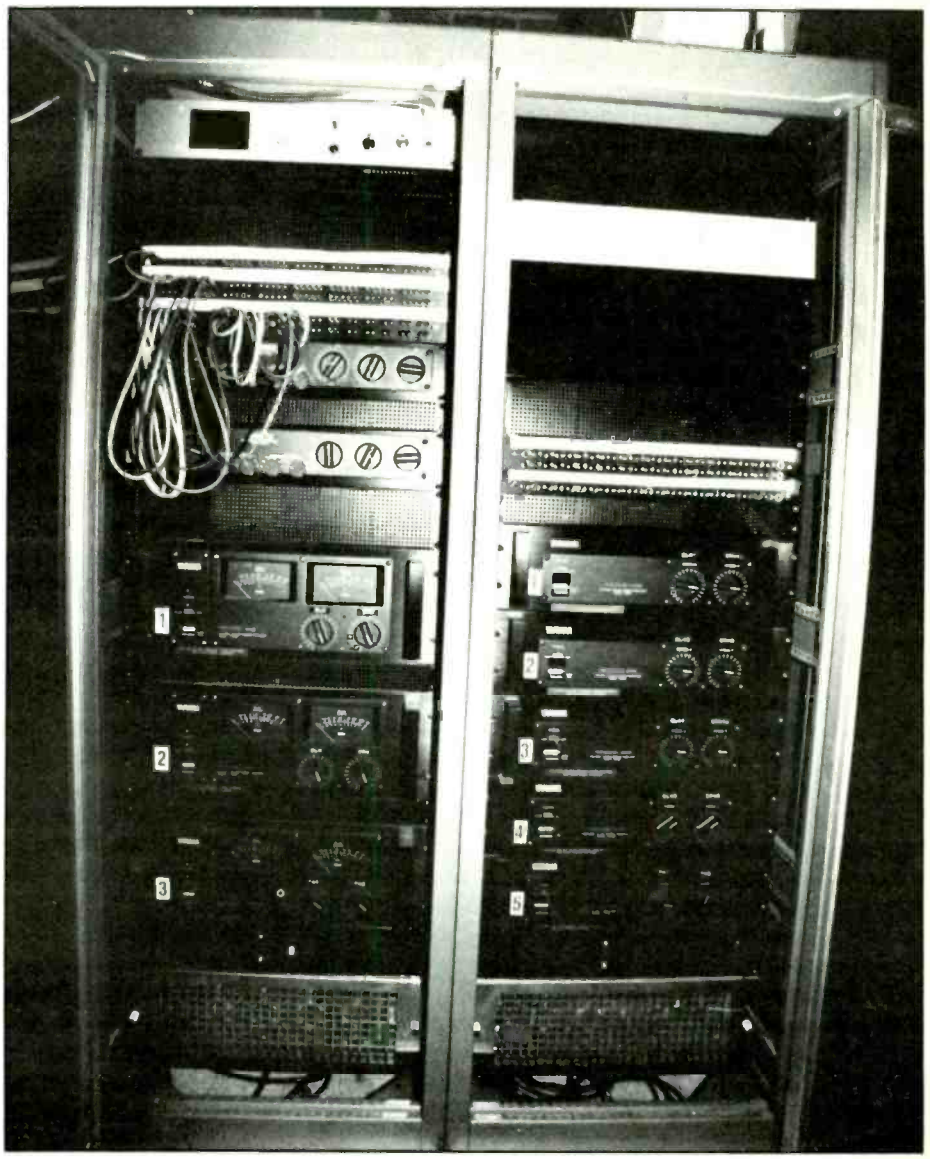


Photo of rack-mounted equipment utilized by the Radio City sound crew.

master cartridge players, and where the receivers for the show's eight HME compander/expander wireless microphones are located.

The orchestra pit also had to be wired for sound, and because the orchestra rides in a band cart that goes from two stories beneath the stage up to fourteen feet above it, and which is capable of making 360 degree turns, the wiring is rather complicated. Furthermore, the wiring of the band cart is not just a question of sending amplified signals *out*. The conductor must be able to hear incoming click tracks, and an in/out communications system must be working at all times. Estrin's crew is still perfecting the system for the band cart, but at present it requires some quick plug switching at a designated spot on stage. This feat is accomplished by a soundman who rides along with the orchestra (in costume) making the necessary connections and preparing a submix of the signal being sent out of the cart.

Since the opening of the "A New York Summer" show, Filmways has just about finished its installation. The cost was in excess of $300,000 for the equipment in the hall alone, and Estrin estimates that by the time Radio City has built its dinner theater, wired into Plaza Sound and acquired the equipment it needs for its various other enterprises, it will have spent more than a million dollars. "Which," he says, "is a lot of audio in a legitimate theater.

"Since we installed the system, we've used the IVIE IE 30 third-octave equalizer and the IE 17 microprocessor to take readings in the hall. Our engineering manager, in Los Angeles, has a whole stack of charts, and we're still in the process of compiling them so that we can see, graphically, what's really happening in the hall now. But you can *hear* the difference. The only complaint we've had since the show opened was that the sound was too loud—and that's not very difficult to correct. But we've noticed that when people have mentioned the sound, they've had good things to say. And of course, when the sound *isn't* mentioned, that's good too. You have to keep in mind that this was the first show to be mounted since the system went in, and that beyond the physical problems of continuing the installation during a schedule that runs two shows a day, six days a week, we have the task of educating the technicians who work here about the equipment we've installed. They're used to a very different way of doing things—some of the people here had never worked with patchbays before. So, yes, we still have problems, now and then. But on the whole, the hall sounds 1,000% better than it used to."

Ambient Sound

BY LEN FELDMAN

Standards and Standards

A press release from the IEC (International Electrotechnical Commission) just crossed my desk, and I must confess that I am still fuming over its contents. Mind you, I have nothing personal against this august international body which is headquartered in Geneva. My quarrel is with their announcement of yet another "standard," designated Publication 581 series, Part 6.

Now, when it comes to standards for the audio industry, I am all for them. In fact, in my capacity as technical director for the Institute of High Fidelity (IHF) I have the somewhat awesome responsibility of seeing to it that committees meet and produce measurement standards for the whole industry. Right now, active committees include those working on tape hardware measurement standards, turntable measurement standards and, heaven help us, loudspeaker measurement standards. (Though I like to think of myself as being on the bottom side of "middle age" I often wonder if the speaker committee will finish its job during my lifetime!) The IHF, as many of you know, has already completed and established measurement standards for FM Tuners (1975) and Amplifiers and Preamplifiers (1978).

So what's bothering me about this new IEC effort? Simply this: it is a document which attempts to provide *minimum* performance requirements for linear and equalizing preamplifiers, power amplifiers and integrated amplifiers. To make matters even worse, the first section of the new "Standard" is titled, "Minimum Requirements for Characteristics Directly Relating to the Reproducing Quality," and covers, for example, effective frequency range, gain alignment, total harmonic distortion and rated power output.

In my opinion, this is just the kind of "industry standard" that the industry *doesn't* need. Once you establish "minimum quality" standards for *any* product, you are in effect saying to a manufacturer that all he has to do to be able to label his product as being superb is to meet those minimum standards. Not only does such labeling mislead the buying public (which the standard is created to protect), but it discourages engineering research and improvements in performance.

Suppose, for example, that a few years ago such a standard had mandated that an amplifier need have power response only over the range of frequencies from 20 Hz to 20 kHz to be called "true high fidelity." Do you think you would have seen amplifiers with response down to DC and up to 100 kHz or so? And despite the arguments of a few die-hards, I think it is safe to say that these wide-band amplifiers of today *do* provide an audible improvement over limited-bandwidth amplifiers of just a few years ago. We haven't sorted out all the reasons why this is so just yet, and a lot of people are still engaged in promulgating theories about it (some of them bordering on the mystical, others a bit more scientific), but the fact is that more and more amplifier manufacturers are striving for "faster" rise times, better slew rates and lower TIM distortion, all of which are implicit in wideband amplifier design. Speaking of TIM, by the way, just a few years ago, no one had ever heard of the term, so if an outfit like IEC (or DIN, which I'll get to in a moment) had come up with their "minimum requirement" standards then, you could today produce an amplifier which was loaded with TIM distortion and could still say that it exceeded "minimum performance requirements" of any standard written back then.

Changing Standards

To my mind, there are three possible kinds of standards for an industry such as ours, and two of them are worth having while the third can do nothing but discourage engineering progress. The first desirable standard is, of course, the one which tells you *how* to measure a given performance parameter. It attempts to do this in such a way so that similar products can be compared by reading the brochures and specification sheets developed by their manufacturers. This kind of standard is often revised, as better and more meaningful methods of measurement are developed.

A good example is what happened recently with the Amplifier Standards of the IHF. The first version of this standard, last revised in 1966, called for signal-to-noise to be measured with reference to full rated output of an amplifier. In those days, most audio ampli-

fiers were in the low-power class, so a few dB of difference in the rated output of an amp didn't make that much of a difference in the signal-to-noise ratios measured. But today, when we use everything from 10-watt-per-channel amps to amps bordering on a kilowatt per channel, the rating of power output makes all the difference in the world if you use it as a reference level against which to measure signal-to-noise. A 100-watt amp and a 10-watt amp, both with identical residual noise levels, will show a difference of 10 dB in S/N if measured the old way. So, the new IHF Amp Measurement Standard calls for everything to be "normalized" to a reference level of 1 watt output. And, since amps have different degrees of *gain* (not to be confused with output power per se), the input level has been normalized to 0.5 volts as well. Now, when you read two S/N specs from two amps, (as reported in all of our amplifier test reviews), you are comparing apples and apples. Many other examples of the same kind could be cited, but I'm sure you get the idea.

The second kind of standard which I think is worthwhile is one which deals with the problems of interfacing between components. Unless we all plan to go back to all-in-one compacts or console radio-phonograph audio equipment (heaven forbid), we are all faced with the problem of matching one component's output to another component's input. This is all the more true if you are involved in pro audio equipment. It would be nice if when you plugged in a pair of phones for monitoring purposes, the *tip* of the plug always picked up the same stereo channel while the ring picked up the other. It would be nice, too, if the 0 dB mark on all record level meters corresponded to a fixed magnetization level applied to the tape. At the moment, on cassette recorders at least, you can find 0 dB corresponding to everything from around 145 nWb/M to 250 nWb/M, and more often than not, the manufacturer fails to tell you which level he has chosen for this important reference point. If you don't know any better you will naturally conclude that the machine which uses a lower "0 dB reference" magnetization point has more headroom than the one which chooses to set its 0 dB meter point higher on the magnetization scale. Not to mention the fact that you have to experiment for several hours of recording time to find out just where you should set the gain controls for optimum dynamic range and lowest residual tape hiss.

Microphones can certainly be built with a standard polarization, as can speaker systems and headphones, and it would be a good idea if all amplifiers in the world had the same input/output phase relationship. The same holds true for tuners, which should all be built so that a given direction of frequency modulation at the FM transmitter always resulted in a given polarity direction of output voltage (either positive or negative) at the output terminals of the device. I'm sure I don't have to convince readers of *MR* of the advantages that might be gained if we had more standardization of connectors, as well. All of these standards might be classified as "structural" or construction standards, and there is little in the way of gaining agreement on the part of an entire industry to follow such standards if indeed they existed.

Time for Trouble

When you come to the last of three kinds of standards, that's where we get into big trouble. I have a friend who claims he can hear 0.2% harmonic distortion in music. Experiments over the years have shown that most of the general population cannot detect this form of distortion until it reaches several percent, at least when they listen to actual musical reproduction. So, how can you set a "minimum requirement" for harmonic distortion—or for frequency response, or noise level, or, most ridiculous of all, power output of an amplifier? Hook up a 10-watt amplifier to a giant Klipschorn and turn it up to just below clipping and the sound will drive you up the wall. Hook the same amplifier up to a sealed acoustic suspension speaker monitor and turn it up and you'll be treated to background music levels.

DIN (Deutsche Industrie Normen) published such "minimum performance standards" for audio equipment years ago and they have become the joke of the high fidelity industry. No one pays much attention to them, other than a few European makers of compact audio systems who use them as an excuse to label their products as hi-fi equipment. To give you some idea of how some of these standards become frozen in concrete, early-on in the development of magnetic tape, the DIN organization selected a "standard" reference tape against which any manufacturer was to compare the performance of newer tape products. That tape (known as DIN Bezugsband 4.75/3.31) requires a lower bias than most modern tape formulations and is generally a poorer performer than any of the modern, more sensitive or "hotter" tapes we know today. Yet many manufacturers, for want of any other standard, continue to use it as a reference tape against which to publish their own product's characteristics. Needless to say, everyone doing so comes up smelling like a rose, and it is really difficult to compare results on the basis of this reference tape.

Standards of any kind, be they measurement standards, structural standards, or whatever, have a limited life, and once promulgated should not be thought of as unalterable. But, in the case of standards that attempt to set "minimum performance" criteria, they are obsolete at the moment of publication. With most of the world drowning in paper anyway, this new attempt by the IEC to set minimum performance standards for audio constitutes, in my mind, yet another waste of good paper.

NORMAN EISENBERG AND LEN FELDMAN

Pioneer RT-909 Tape Recorder

General Description: The new RT-909 from Pioneer is a reel-to-reel tape recorder with 10½-inch reel capability and separate heads for erase, record and play. There also is a fourth head for play in the reverse direction. A three-motor transport is used in combination with a closed-loop dual-capstan drive system. The head configuration is four-track (quarter-track), two-channel so that the unit may be used for stereo or for mono. Microphone and line input signals may be mixed by controls on the front panel. The RT-909 also permits sound-on-sound recording, and follow-up recording (taping a new program while playing an already recorded program). The unit also may be used for unattended recording or playback by means of an external timer. A pause control may be activated during play or record to permit manual reel-rocking to locate a specific passage. Tape speeds are 7½ and 3¾ inches-per-second. The tape index counter provides digital readout. The signal meters are horizontal "bar graph" types using fluorescent indicators and they may be switched to show peak and average levels. The recorder may be rackmounted.

The tape follows a symmetrical path from reel to reel, with tension arms and guides under each reel, and additional guides under the head cover along with left and right capstans and pinch rollers. The head cover contains several small holes through which are accessible screwdriver adjustments for recording-head azimuth, and for both playback-head azimuths. The cover is open on its underside to permit access for head and guide cleaning, and for demagnetizing as needed. The heads are pre-adjusted at the factory and are not intended for user adjustment except by qualified personnel using the recorder's service manual, which is not normally supplied with the machine.

Spring-loaded clampers hold the reels in place. A slit in each reel clamper permits adjusting the height of the reel bases if necessary. The four-digit tape counter and its reset button are centered between the reels and above the head cover. Below the head cover is the

signal display and below it are two buttons for dimming the illumination and for changing the readout from average to peak levels. The metering itself is calibrated from −30 to +8 dB.

To the left of the meters are the electronic controls; to the right, the transport controls. The former group includes: stereo headphone output jack; left- and right-channel microphone input jacks; the tape/source monitor switch; separate bias and EQ selectors, each with two positions; separate record off/on buttons per channel; the output level controls; the mic input level controls; and the line input level controls. All level controls are dual-concentric clutched types that permit adjustment on both channels at once, or on each channel separately.

Transport controls include a large and small size reel selector; the speed selector; a pitch variation (operates only during playback); a repeat button by means of which it is possible to repeat an entire tape or any portion thereof; the timer-start button for unattended operation; the pause button; and the regular buttons for fast-wind in either direction, stop, play in either direction and record. The transport movement buttons are feather-touch logic controls that permit going from any mode to any other without the need to press the stop button. Fast-wind in either direction is instantaneous, as is going into fast-wind from the play mode. When changing direction of play, or when going into record mode from fast-wind, there is a pause of about 2.5 seconds. To go into record from forward play, the action is instantaneous for "run-in" recording.

The recorder's power off/on switch is just above the pause button. The rear of the recorder contains its AC line cord, an optional grounding post and two pairs of pin-jacks for line in and out signals.

The RT-909 comes with "feet" that permit it to be installed on any surface. For rack-mounting, these feet are removed; a 19-inch blank panel (supplied) may be used to fill any gap above the panel. In addition to this piece, the RT-909 is supplied with an empty 10½-inch metal reel, two 10½-inch reel adapters, signal cables, three head-cleaning swabs, a small supply of splicing tape and some metal-foil sensing tape which is used for engaging the automatic-reverse function on playback. The owner's manual includes a chart of tape brands and types with recommended bias and EQ switch settings, and complete instructions for operating the new recorder in all its modes.

Test Results: The RT-909 in lab tests did better than its published specs in most areas, and came in a little under spec in others. The latter figures, however, were still indicative of excellent performance (e.g., frequency response at 7½ ips of ±3 dB from 17 Hz to 26 kHz). Distortion at the zero VU recording level never got above 1 percent, although for +3 VU, the THD rose to 2.3 percent. Input and output levels all were fine for interfacing with external equipment. Headroom was adequate; signal-to-noise better than claimed; erase ratio much better than claimed.

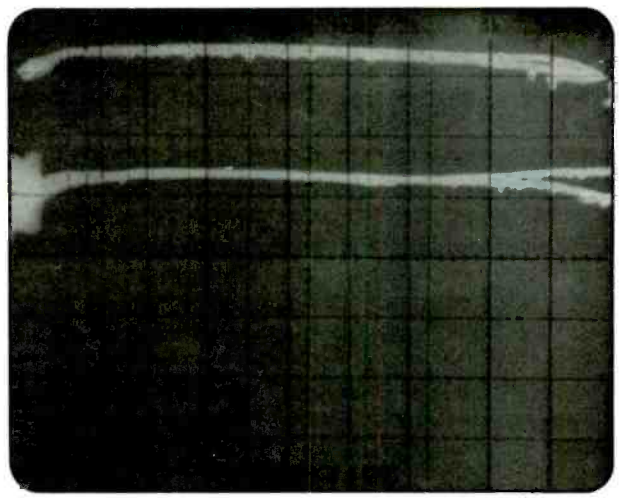


Fig. 1: Pioneer RT-909: Record/play frequency response at 0 dB (upper trace) and −20 dB record level, 7½ ips.

Measurements were made with a 3600-foot reel of Maxell 35 (UD) which, according to the owner's manual, requires that the EQ and bias switches be set to their number 2 positions, which are the same settings indicated for Scotch 206, which is the manufacturer's reference tape for this machine. With these settings we equalled Pioneer's claimed THD figures for the zero-dB recording level, but fell a bit short of reaching to 28 kHz (for −3 dB) in frequency response

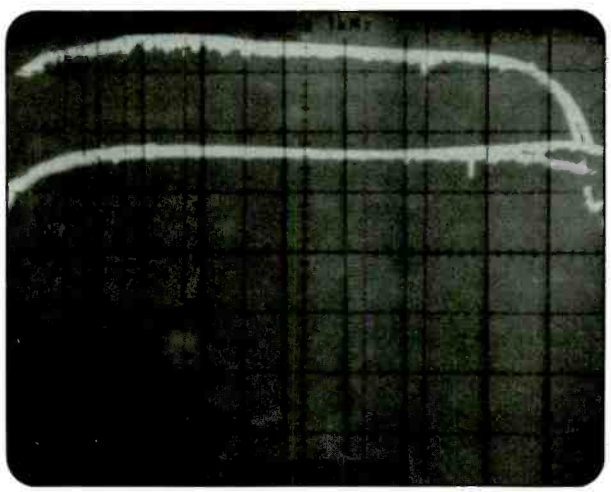


Fig. 2: Pioneer RT-909: Record/play response at 0 dB (upper trace) and −20 dB record level, 3¾ ips.

at the high speed, and got to 15 kHz (instead of the claimed 18 kHz) at the slow speed. Thinking that the tape might need the alternate EQ setting we tried that, and sure enough, the response picked up at the high end. Results of these measurements are shown in the spectrum analyzer photos of Figs. 1 and 2 (7½ and 3¾ ips, respectively, with the upper trace showing response at 0 dB record level, and the lower trace showing it at the recommended −20 dB record level. We then changed the bias switch setting; again, this extended the extreme high end but it also increased the distortion. In any event, with four possible combinations of bias and EQ from which to choose, the operator should be able to adjust accordingly.

The THD figures shown in the "Vital Statistics" table are really total harmonic distortion plus noise. Of greater interest is the third-order harmonic component which establishes the maximum recording level for a given tape. To ascertain this distortion level, we used the spectrum analyzer; the results for the 7½ ips speed (Fig. 3) indicate that the third-order component of a 1-kHz played-back signal (shown at the center of the screen) is about 45 dB below the reference peak, or equivalent to 0.56 percent. For the slow speed, distortion components are shown in Fig. 4; here the third-order distortion was about 41 dB below the reference level, or 0.89 percent. These are good average figures for an open-reel deck. However, the relatively high second-order component, immediately to the right of the centered reference peak in each case, suggests that the bias setting was less than perfect for the particular tape used in our tests.

Wow and flutter measurements were excellent, as indeed were all the functions relating to the RT-909's mechanical operation. Tape handling and movement were exemplary.

General Info: Dimensions are 18⅞ inches wide; 13⅜ inches high; 12½ inches deep. Weight is 46 lbs., 5 oz. Price of the recorder is $895.

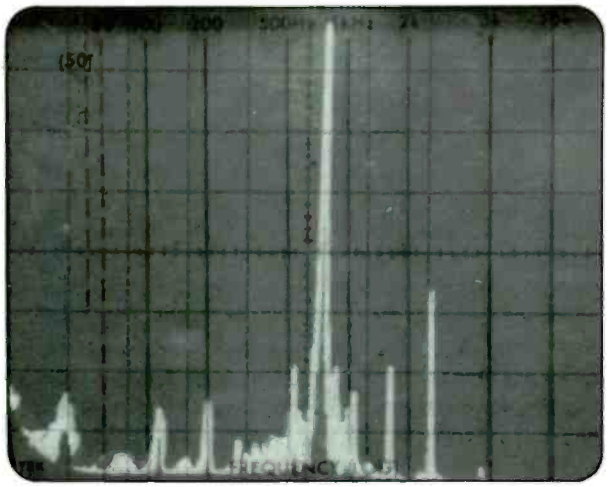


Fig. 3: Pioneer RT-909: Spectrum analysis of 1 kHz playback signal at 7½ ips.

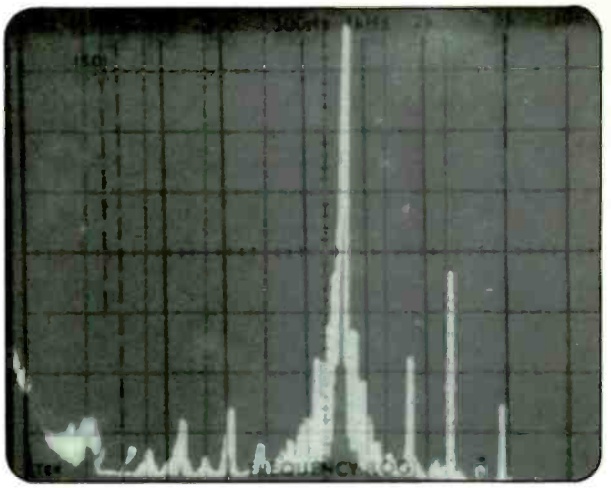


Fig. 4: Pioneer RT-909: Spectrum analysis of 1 kHz playback signal at 3¾ ips.

Individual Comment by L.F.: Having reviewed Pioneer's Model RT-707 some time ago in these pages [February 1978] and admired that unit very much, I expected to be even more enthused with the RT-909. After all, here was a machine that could accommodate professional sized 10½-inch tape reels and I had assumed that there would be other professional features on the RT-707's big brother. Well, there aren't. The unit is still very much a hi-fi consumer piece of gear, albeit one that can handle bigger reels of tape. All of its extra features cater to the audiophile, but not to the semi-pro or professional user. Thus, the ability to play back tape in the reverse mode is nice if you want to listen to tapes for a very long time without having to reverse reels, and I guess with a 3600-foot reel of tape playing in mono, one track at a time at 3¾ ips you could create your own unattended muzak program source, if you wanted to. Being awakened to taped music is also fun, though you'll need a form of external timer for this sort of amenity.

But considering the magnificent transport system with its touch-logic control buttons and fail-safe tape handling system, why oh why didn't Pioneer give us the ability to fast cue? What good is being able to rock the reels back and forth electronically, with feather touch ease, if you can't cue up a precise spot on a tape recording for editing or cueing? And while it is possible to record individual tracks one at a time and then use the tracks for sound-on-sound recording, that's not the same as having simul-sync (whereby the record head is converted into a monitoring head) for synchronizing a multi-channel recording, track after track.

Our tests and measurements of the RT-909 are amply discussed above. While there seems to be some question as to indicated EQ and bias settings for the tape we used (Maxell's UD takes the same bias and EQ as does the Scotch 206 used by Pioneer), the figures we measured are good averages for an open-reel deck.

Wow-and-flutter as well as all operating characteristics relating to the tape transport system were, in my

opinion, excellent and fully equal to those found in some professional decks. So, what we have is a machine that moves tape in an exemplary manner but doesn't let you do some of the things with the tape as a recording medium that you might be able to do with a machine more geared to semi-pro or professional applications. The price of the earlier RT-707 was so very appealing that one might be willing to overlook some of the desirable features required by a serious reel-to-reel recordist. But when you are talking about a deck with a suggested price of nearly $900, one is faced with the question of whether to spend a couple of hundred bucks more on a machine that can do all the tricks or settle for the RT-909 at the lower, but not that much lower, price. I guess it all depends upon what you want an open-reel machine for in the first place.

Individual Comment by N.E.: The Pioneer RT-909 suggests comparison with the previous RT-707. The RT-909 looks like an enlarged RT-707 and, in fact, that seems to be just about what it is. The main and most obvious difference between the two is the 10½-inch reel capacity of the new model.

There are, in addition, some performance differences which, according to our measurements, do give a slight edge to the RT-909. In the new model response at the faster speed is better, though at the slow speed the RT-707 made it out to 17.5 kHz (within ±3 dB) while the RT-909 was measured out to 15 kHz. Both models are spec'd for 1% THD; the RT-707 came in at less than that while the RT-909 just made spec. In any event, signal-to-noise is better in the RT-909, and so is the wow-and-flutter at 7½ ips and at 3¾ ips. The erase ratio is at least 15 dB better in the new unit.

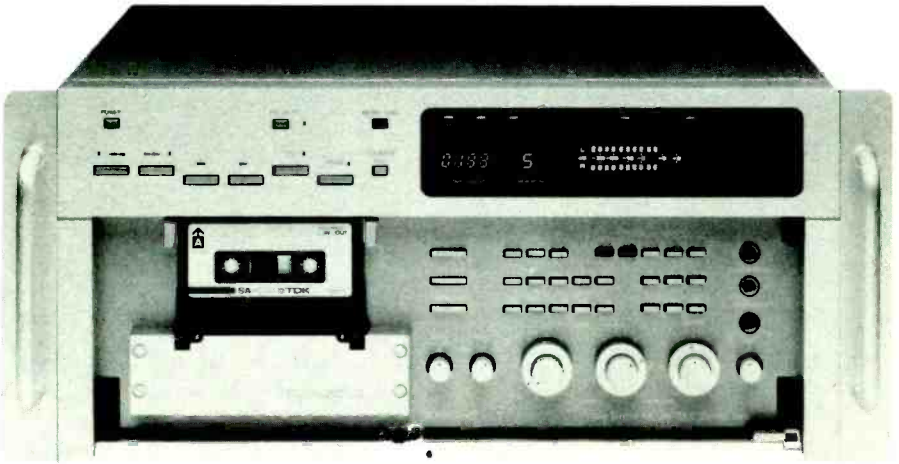
On final count, it appears that besides beefing up the mechanics of the recorder, some attention was paid to also improving its 7½ ips audio performance. The 3¾ ips performance still is very good, though not what might be called spectacular for the open-reel format. Even so, recording a lot of material at both speeds and comparing the results I feel that most of us would be hard-pressed to fault the 3¾ ips speed in the RT-909.

The former model—because of its relatively compact format for a high-performing reel-to-reel deck, and its relatively low price too—was thought of at the time as the kind of recorder that could woo a buyer away from the cassette format, especially one who prefers open-reel tapes not only for recording but for listening in terms of dipping into the available repertoire of commercially recorded stereo tapes in this format. In this regard, the automatic reverse feature for playback is a definite plus since it makes for long, continuous playback without the need to stop the machine and flip reels. Obviously, the new version, while still no physical monster as open-reel decks go, is larger and heavier than the previous deck. It also costs $300 more. Apparently, what the extra $300 gets you, essentially, is the option for using 10½-inch reels, somewhat improved performance at the faster speed and the new style fluorescent bar-graph metering. So, while it could be said (as I did) of the RT-707 that its price was a little hard to believe in view of its performance and features, I don't feel the same can be said of the RT-909. On the other hand, we are dealing these days with inflation and currency fluctuation, factors which are boosting prices for most products and services. This also must be considered in assessing the RT-909 for possible purchase. Whatever, it strikes me as a very well-built recorder that is as ruggedly made as it is easy to use. Operation is silky-smooth, and the sound—of tapes recorded on it and of commercially recorded tapes—is first-rate.

PIONEER RT-909 TAPE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Tape speeds	7½ and 3¾ ips	confirmed
Reel capacity	10½ inches	confirmed
Wow & flutter, 7½/3¾ (WRMS)	0.04%/0.08%	0.035%/0.05%
THD at 0 VU, 7½/3¾	less than 1%	1.0%/1.6%
THD at + 3 VU, 7½/3¾	NA	2.3%/3.5%
Level for 3% THD, 7½/3¾	NA	+ 4 dB/ + 2.5 dB
Frequency response, 7½ ips	± 3 dB, 20 Hz to 28 kHz	± 3 dB, 17 Hz to 26 kHz
Frequency response, 3¾ ips	± 3 dB, 20 Hz to 18 kHz	± 3 dB, 28 Hz to 15 kHz
Best S/N (standard tape) 7½, 3¾	better than 60 dB	67 dB/63 dB ("A" wtd)
Rewind time, 2400-ft tape	120 seconds	100 seconds
Mic input sensitivity	0.316 mV	0.260 mV
Line input sensitivity	50 mV	35 mV
Line output level	450 mV/900 mV	450 mV/900 mV
Headphone output level	50 mV/8 ohms	113 mV/8 ohms
Bias frequency	125 kHz	confirmed
Erase ratio	better than 60 dB	85 dB
Speed accuracy, 7½	± 0.6%	± 0.3%
Mic/line mixing	Yes	confirmed
Sound on sound	Yes	confirmed
Reverse playback	Yes	confirmed

Phase Linear 7000 Series Two Cassette Recorder



General Description: The Phase Linear 7000 Series Two is a high-performing cassette recorder of unusual design and novel features, of which the most striking is its built-in "Microscan" system. This system automatically detects the optimum bias, equalization and signal level sensitivity for any cassette loaded into the machine, and adjusts the recording circuitry accordingly. In addition, the microprocessor has a memory function to store this data (for up to nine different cassettes), and to recall it in the future. The memory system operates even when power is turned off thanks to a pair of penlight (AA) cells inserted at the rear. An ingenious lamp-indicator display shows the process from start to finish, and also indicates when the correct settings (or not) have been chosen on the tape-selector knob. To use this system takes a fair amount of study of the owner's manual, but when carried out correctly the system is foolproof.

Styling of the three-head model 7000 is unique. Most of the front panel is laid out on a recessed portion that may be hidden by a large door. What remains visible with the door closed is the top one-third which is flush with the door. This area contains several control buttons plus a display panel.

The control buttons include: power off/on; rewind; fast forward; stop; play; record; pause; record mute; tape-counter reset; and the Microscan activator.

Items on the display panel include: the four-digit tape counter (a digital type); three indicators for bias, level and EQ; another indicator that shows the memory number chosen; four indicators for the tape-type chosen on the tape-selector knob (metal, standard, CrO_2 and FeCr); and indicators for the Dolby system, the memory/repeat option and battery condition. The largest single portion of this display area contains the recorder's signal meters, a twin horizontal fluorescent bar display. Calibration runs from −30 to +8 dB. The metering may be switched to show peak, peak hold or average signal levels. The degree of illumination also may be switched.

The large cover swings down and slides into a space under the unit to reveal everything else. At the left on this portion of the panel is the cassette area. The cassette is loaded by inserting it onto the projecting spindles and fitting it under an overhanging piece that secures it in place. There is no "well" in the usual sense, and no eject mechanism or eject button. To remove the cassette (only when the transport has been stopped, of course) you simply move it forward.

The remainder of this section of the panel contains a great number of controls, some familiar and others relating to the unique Microscan and its associated memory system. There are switches for a multiplex filter, for the Dolby system and to monitor either the tape or the source being recorded. Another group of switches handles settings for unattended play or record via an external timer. A third group relates to the signal meters (peak, peak hold, average signal levels and lighting dim or bright).

There is a row of switches for the memory-rewind with options for stop or play as well as for repeating an entire tape or part of it.

There are nine buttons for storing the Microscan data concerning each of nine different cassettes. In addition there is a button to activate the Microscan memory, and another button to recall the data as needed for a given cassette. All told, there are twenty-seven buttons on this section of the panel.

Below them are several knobs. One adjusts the pitch on playback by varying tape speed over a ±6 percent range. It has a center detent position which, when engaged, lights up an LED under the cassette area. The tape selector knob has four positions: metal, standard, chromium-dioxide and ferrichrome. The appropriate indicator on the top section of the panel comes on as per the tape-type selected, and if this is not correct as per the Microscan system, the other indicators for bias, level and EQ will blink. Otherwise they will remain steadily lit.

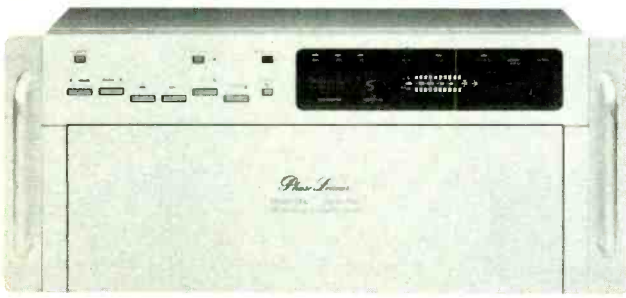
There are three pairs of level controls for microphone, line and output. Each pair is a dual-concentric type that permits channel adjustment individually or

simultaneously. Input mixing of line and mic signals is provided by this arrangement. The final control is a continuously variable bias trim with a center detent. This knob may be used to adjust bias manually without using the Microscan system. Above this knob are three signal jacks: left- and right-channel microphone in, and stereo headphones out.

At the rear are two pairs of line in and out pin-jacks. The "Play 1" and "Record 1" jacks are used in the usual way. The second pair of play jacks may be used to record on another deck whatever is being played on the model 7000 (tape duplicating). The second pair of record jacks may be used to record on another deck the same program being recorded on the model 7000 (simultaneous two-deck recording). The rear also contains an optional grounding terminal, the battery compartment for the Microscan memory bank, the unit's AC line cord and an unswitched AC outlet.

Access to the heads is gained by simply lifting out the cassette. No azimuth adjustment is provided since the record and play heads share a common housing. The closed-loop dual capstan drive is powered by a quartz phase-locked loop brushless DC servomotor. Reel drive is handled by a coreless DC motor.

The transport buttons permit fast-buttoning in all modes, including run-in recording from the play mode. The model 7000 is fitted with handles and it may be rack-mounted.



Phase Linear 7000: Front panel view of the unit with door panel closed.

Test Results: In *MR's* lab tests, the Model 7000 met or exceeded most, but not quite all, of its published specifications. The unit was tested with three different tapes, including a metal tape supplied with it (Scotch Metafine 46). Frequency response went a little better at the high end than specified for each tape. THD was lower than claimed for standard and for high-bias tape, and a shade over spec for the metal tape. Signal-to-noise was 5 dB under spec without Dolby for standard tape, and almost on the nose for the other two tapes. Things improved of course with the Dolby switched on, though not as much as spec'd. At that, the figures obtained were very much "in the ball park" for a high-quality cassette deck. Signal levels and sensitivities all were well suited for interfacing with external equipment.

Transport performance was excellent, with extremely low wow-and-flutter, very high speed accuracy and good handling of the tape in all modes, including when fast-buttoned from one to the other.

Understandably, an extra amount of time and effort was spent examining the Microscan feature and its associated memory system. It all works as claimed. Two main conclusions emerge from extensive tests of these features. One, we could not obtain better results by ignoring the Microscan and adjusting bias manually instead. After trying to outdo the computer by manually adjusting two cassettes we conceded that the Microscan could not be improved on, and all test measurements were made using the built-in "brain" for adjusting the deck to the tapes.

Two, try as we might, we could not confuse or fool the system. For instance, we loaded a ferric-oxide cassette (whose characteristics we had previously stored in the memory bank), but deliberately left the tape-selector knob in the "metal" position. When trying to call up the correct memory, the three lights indicating bias, level and EQ began to flash, telling us something was wrong. The same thing happened in any position of the selector except the correct one. Similarly, if you select on the knob the correct setting for a given type of cassette, but press the wrong memory button, the lights will flash instead of glowing steadily. (The memory number chosen for a given cassette should be written on the cassette itself to prevent this.) The system even operates for another brand of tape which has ostensibly the same coating formulation, but which has not been programmed into the recorder.

The Microscan detecting process takes about 40 seconds. When you press the Microscan button, the machine goes into the record mode and special test tones are applied to the tape. The three functions of bias, level and EQ are handled successively, and are monitored by the individual lights flashing. When the process is completed, the three lights stay on. The data then may be punched into the memory bank and recalled in the future for that particular cassette, without going through the 40-second process.

In addition to the record/play response, we also measured playback-only response using the TDK test tape AC-337 which runs from 50 Hz to 12.5 kHz. There was a slight rise in response at the low end (Fig. 1)

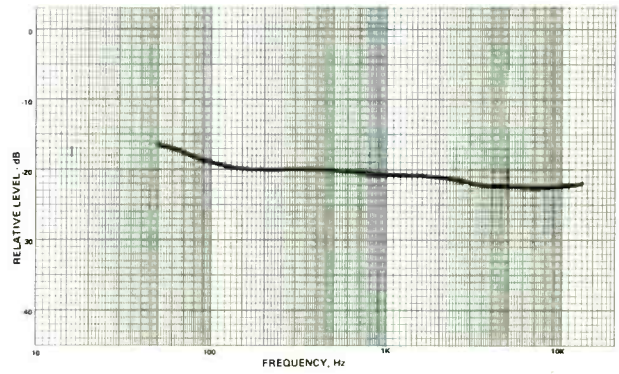


Fig. 1: Phase Linear 7000: Playback response using TDK AC-337 test tape.

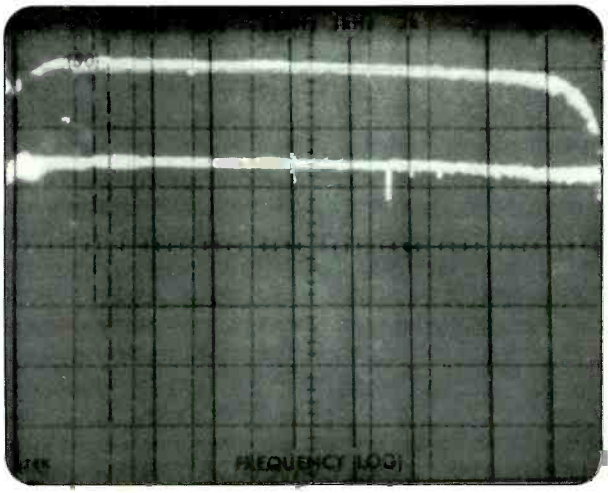


Fig. 2: Phase Linear 7000: Record/play response at 0 dB and −20 dB record level, using 3M "Scotch" Metafine tape.

which we cannot account for, but which did not prove a problem in playback of various prerecorded cassettes.

Fig. 2 shows the record/play response at 0 dB record level and at −20 dB record level for the metal-particle tape (Scotch Metafine). If you compare these results at the 0 dB level with those obtained for the cobalt-ferric tape (Fuji type FX-II, shown in Fig. 3), and with the results obtained with standard or ferric-oxide tape (TDK, shown in Fig. 4), you can see the primary advantage of metal-particle tape: that is, its high-frequency headroom which extended well beyond 10 kHz for the 0-dB record level.

General Info: Dimensions are 18⅞ inches wide; 8⁹⁄₁₆ inches high; 16¾ inches deep. Weight is 40 pounds,2 ounces. Price is $1350.

Individual Comment by L.F.: A single word can describe this incredible machine: WOW! And I definitely do not mean the kind that goes with flutter. When this deck arrived, I felt certain that someone had made a mistake. After all, how could a cassette deck weigh 40 pounds? This one does, and when you lift it out of its carton you might think it's a power amplifier, until you pop open the front door panel and swing it out of the way to disclose the cassette section and the numerous controls.

The big thing about this unit is its fantastic Microscan system, as explained above. The procedure takes about 40 seconds for any cassette. There are other less expensive decks that do the same thing (JVC has one, as does Hitachi), but there's an important difference in the Phase Linear unit. And that is, the model 7000 remembers what it has done, even when you turn off the power (the batteries take care of that). You can store the characteristics for up to nine different cassettes at the touch of a button, and recall them easily when needed. So, if you stay with your favorite tape brands and types, you don't have to wait even 40 seconds each time you start recording.

The table of "Vital Statistics" pretty much tells the story of how good the basic performance (aside from the Microscan feature) of this deck is. Actually, everything about it smacks of high quality and clever engineering. Nothing is left to chance. It should be noted too that of all the decks with metal-tape capability we have checked so far, this is the only one that retains three other possible tape-type choices. No less than forty specific tape types are listed in the owner's manual, and I'm sure there are many more that would work well with this machine. It goes without saying that you would not want to use any bargain-brand tape in this recorder if you expect to realize everything from it that its built-in computer can adjust for you.

One word of caution: You had better darn well read the owner's manual from cover to cover before you even plug in the machine. Not that you can do any harm to it if you don't, but the array of controls and buttons is so overwhelming on first sight that you might just give up in despair before making your first recording. Happily, the manual is carefully and logically written. Its twenty-five pages contain material that is useful and necessary, and they take you through the complex of controls in a manner that should make you fully competent to handle it all. There also is a large insert containing the unit's complete schematic (in print large enough to actually read), as well as a discussion of how the Microscan system works. For the technically curious this is fascinating reading. For the serious recordist, technically curious or not, the Phase Linear 7000 is an outstanding performer. It is, of course, hard to say whether or not it is worth its price of over $1300. For one thing, there is nothing on the market exactly like it. For another, it would be difficult for me to separate the cost of the computerized functions from the cost of the basic recorder. Anyway, I'm just enough of a fanatic about wanting optimum results from any recording equip-

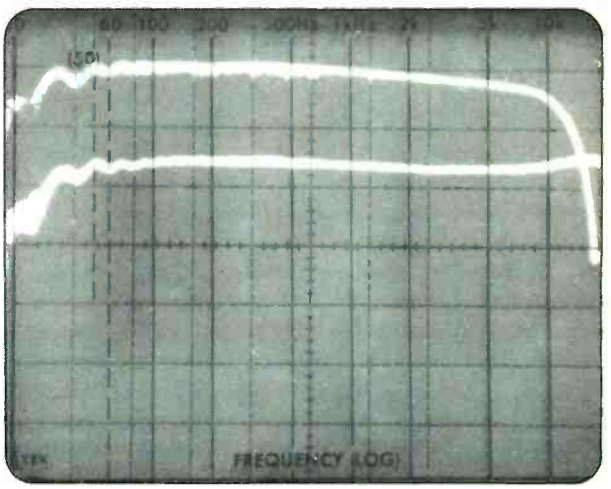


Fig. 3: Phase Linear 7000: Record/play response at 0 dB and −20 dB record level using Fuji FX-II tape (CrO_2 position).

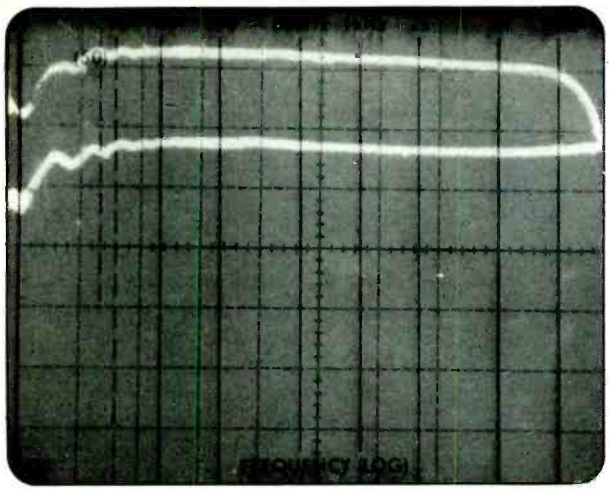


Fig. 4: Phase Linear 7000: Record/play response at 0 dB and −20 dB record level using TDK AD tape (standard setting).

ment I use to say that if I had that much money burning a hole in my pocket at the moment, I might just be tempted . . . .

Individual Comment by N.E.: If you forget momentarily the unique Microscan-and-memory system of the model 7000, you have a very good three-head, logic-controlled-transport, fine-performing cassette recorder with commendable audio response, ample headroom, low distortion, low wow-and-flutter and other desiderata—including fast-buttoning, off-the-tape monitoring while recording, on-the-panel mixing, versatile and easy-to-observe metering, very flexible no-compromise level controls and even a multi-purpose memory-rewind option.

When you add to this the Microscan-and-memory system you have a unique cassette recorder, and this uniqueness can be regarded in more than one light. To begin with, what is obvious from our bench measurements and use tests of the model 7000, the built-in "brain" is very competently engineered. It cannot be outguessed and it won't be fooled or confused by mishandling—deliberate or inadvertent. From the standpoint of equipment design and execution, it is a real engineering tour-de-force that can be admired on its own merits. It makes of the 7000 what may well be the most sophisticated cassette recorder yet offered.

Consider, though, that this ingenuity, and the circuitry involved in it, must add to the cost of the unit. The big question then is: Is it worth it? Only you can answer that, depending on your needs and inclinations. But whatever, there is another question too—and that has to do with just how far we can go with the cassette format before reaching the point of diminishing returns. In other words, given the realistic parameters of cassette performance—including even metal tape with its admittedly higher headroom for the upper-end of response—how much better are the results obtained with an automatic, foolproof system for adjusting bias, level and EQ as opposed to making those adjustments manually?

In great measure, how you answer that question—based on your knowledge of other units and relative performance differences among them—may determine how you react to the model 7000. Is it the greatest thing that ever came down the cassette road, or just another very good unit in a field of very good units in which it is necessary, in order to gain attention, to do something different?

PHASE LINEAR 7000 CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response, normal tape	± 3 dB, 25 Hz to 18 kHz	± 3 dB, 27 Hz to 19 kHz
high-bias tape	± 3 dB, 25 Hz to 18 kHz	± 3 dB, 27 Hz to 19 kHz
metal tape	± 3 dB, 25 Hz to 19 kHz	± 3 dB, 27 Hz to 20 kHz
Wow-and-flutter, WRMS	0.03%	0.025%
Speed accuracy	NA	−0.1%
S/N ratio (re: 3% THD record level)		
Dolby out,		
normal tape/high-bias/metal	All greater than 60 dB	55 dB/59 dB/59 dB
Dolby in,		
normal tape/high-bias/metal	All greater than 70 dB	64 dB/66 dB/67 dB
Record level for 3% THD		
(0 dB = 145 pWb/m)		
normal tape, high-bias, metal	NA	+ 6.0 dB/ + 5.5 dB/ + 5.5 dB
THD at 0 dB record level		
normal tape, high-bias, metal	1%/1%/1%	0.70%/0.85%/1.2%
Line output at 0 dB (ref/max)	450 mV/650 mV	450 mV/600 mV
Headphone output level at 0 dB	63 mV/90 mV (8 ohms)	63 mV/92 mV (8 ohms)
Mic input sensitivity for 0 dB	0.3 mV	0.32 mV
Line input sensitivity for 0 dB	60 mV	65 mV
Fast-wind time, C-60	75 seconds	75 seconds
Bias frequency	NA	80 kHz
Power consumption	45 watts	61 watts

CIRCLE 2 ON READER SERVICE CARD

Hafler DH-200 Power Amplifier

General Description: The Hafler DH-200 is a stereo power amplifier rated to deliver 100 watts (continuous average power) per channel into 8 ohms, 20 Hz to 20 kHz, both channels driven. Into 4-ohm loads, the amplifier is rated to deliver 150 watts per channel. For 2-ohm loads, the power rating is 100 watts per channel. The output also may be bridged to convert the DH-200 to a 300-watt monophonic amplifier.

The unit's active circuitry is laid out on two modules, either of which may be removed if necessary for servicing. Circuit design is claimed to achieve the benefits of Class A output stages (such as high speed, minimal crossover distortion, decreasing distortion at lower signal levels) but without what the manufacturer describes as the disadvantages that are a part of true Class A operation, such as high cost, high heat, low efficiency and thermal instability.

The front of the amplifier contains only the power off/on switch and a pilot lamp. The rear panel contains the unit's power cord and the left- and right-channel connectors. There are pin-plugs for the inputs, and binding posts for the outputs. Each channel has its own fuse. The inside of the amplifier is virtually a mirror-image of the two channels, with each going off in opposite directions to the large heat-sinks that flank the sides. The DH-200 is available in kit form, or factory-assembled.

Test Results: All of the Hafler amp's published specs were met or exceeded in our lab tests. Despite the unit's relatively small size, it ran remarkably cool during all bench and listening tests. It also was extremely stable under capacitive load conditions (such as may be encountered with certain speakers). The unit is shipped with 2-ampere fuses to insure against blowing out low-power-rated speakers. The instructions advise, however, that if one's speakers can take it, the 2-amp fuses should be replaced with 5-amp fuses (supplied with the amplifier). Tests confirmed the validity of this approach: the 2-amp fuses did pop when we subjected the amplifier to severe overloading into low ohm loads.

Additional fusing, by the way, may be found inside the amplifier, between each amplifier module and the common power supply. The unit uses, however, no special protective circuits. The manufacturer's design philosophy holds that such circuits, when inserted in the signal path, can introduce a subtle form of audible distortion during music reproduction. Hafler maintains that this amplifier circuitry is self-protecting, and that the output stage is designed to protect against thermal runaway without the need for signal-interrupting relays and similar devices. As far as we could determine, he is perfectly right.

Our only "frustration" with this amplifier was an inability to measure any verifiable distortion figures. Our Sound Technology distortion measurement system is good only down to 0.002 percent, which is exactly what was measured at every frequency at which the DH-200 was tested, and at every power level to beyond its 100-watt rating.

General Info: Dimensions are 16 inches wide; 6¼ inches high; 9½ inches deep. Weight is 30 pounds. Price: $299.95 in kit form; $399.95, factory-assembled.

Individual Comment by N.E.: The "DH" stands for, of course, David Hafler, who may be known by many readers as the man who founded the Dynaco Company years ago. Dynaco was known for offering excellent value for the dollar—the more so if you opted to spend your own time building the units from the kits of parts. In general, these products eschewed frills and features in favor of high performance within a de-

signated design goal—a sort of "no nonsense' approach to audio products. The new DH-200, now being produced by a new company that bears Hafler's own name, is the result of that same school of thought. The new amplifier lacks output meters and LED indicators for this and that. It has no level controls. It has no handles. It looks like a "black box."

But its performance is superb; clean, effortless, transparent reproduction. Testing it in the lab by the usual methods applied to all amplifiers is simply an exercise in confirming Hafler's specs. This amp merits "listening to" or rather "listening through" back to the source material. It may be relevant here that the manufacturer has indicated that this design does not depend heavily on large amounts of negative feedback. Nor does it try to get impressively low figures for THD and IM, while at the same time creating transient intermodulation distortion (TIM). This form of distortion—created by excessive use of negative feedback—is held to be more irritating. Thus, a design factor in the DH-200 was to use negative feedback as a refinement rather than a cure-all. Admittedly, this approach can be argued pro and con. What seems less debatable, in any case, is the fine "sound" of the DH-200.

Individual Comment by L.F.: I have known and respected David Hafler for many years, since he started the Dynaco Company in the early 1950s. Anyone who has been around in audio long enough to have used some of those early Dynakits knows that Dave Hafler's mission in life has been to bring high-quality audio products to the public at a price that doesn't empty its wallets. After selling his interest in Dynaco some years ago, Dave did a stint as an employee of other firms, both here and abroad, but like all dedicated audio enthusiasts and engineers, the quest for better sound at a better price prompted Hafler to get into his own design and manufacturing firm again.

His first entry, an unimpressive looking preamplifier-control (the DH-101) chassis has already achieved wide recognition and acclaim, and it was not unexpected that Hafler would follow through with a matching power amp of excellent performance. What I didn't expect is that he would be able to deliver that 100-watt plus per channel amp in kit form at the unbelievable price of $299.95 and in wired form for only $100 more. Although we tested the wired version (*MR* deadlines don't permit N.E. or myself the luxury of building kits to test), I did see the "parts" of the kit when I was first introduced to this neat little amplifier at the Consumer Electronic Show in Chicago last June. From that presentation I can conclude that while both the wired and kit prices offer a tremendous price advantage, the biggest advantage to be gained by a prospective user is with the kit version. The two amplifier modules (visible in Fig. 1), are supplied completely wired and tested, and about all that remains for the kit-builder to do is to wire and assemble a handful of parts, most of which relate to the power supply section. Unless you are a very high-paid person, I doubt if

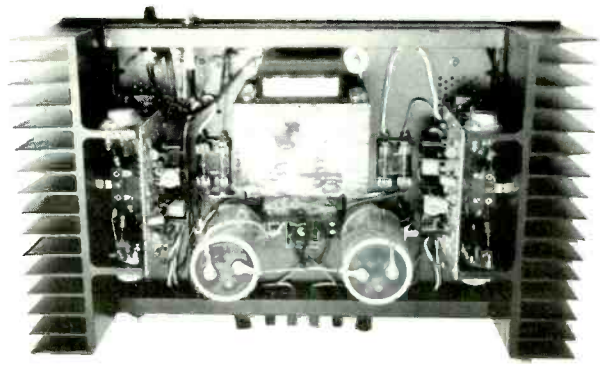


Fig. 1: Hafler DH-200: Internal view discloses modular construction of each channel and power supply parts centered in chassis.

you will expend $100 worth of labor, and, if you have any experience with kit building at all, you should be able to complete the job in a single evening. By buying the wired version, you will also miss the fun of building and assembling the unit yourself, turning it on for the first time and having it work. Just in case that doesn't happen, however, or if service is needed, it is possible to remove and return one of the amp modules (channels) as a complete assembly instead of worrying about having to ship a heavy amp back for service. You can even operate the amp monophonically while waiting for the repaired module.

As Hafler points out in his discussion of his new amplifier, the trick is not in being able to reduce static forms of distortion such as THD and SMPTE-IM but being able to do so with a minimum of required negative loop feedback so that other, more subtle forms of distortion such as TIM and IID (Interface Intermodulation Distortion, caused by speaker-generated signals getting back into the feedback loop) don't become audibly evident. From my listening tests of the Hafler DH-200 I would say that its designers have succeeded remarkably well in this respect. Interestingly, the slew rate of this amp is "only" 30 volts per microsecond, which kind of confirms what I and others have suspected for a long time—namely that you don't really need 100 or 200 volt per microsecond slew-rates for totally clean reproduction of musical transients that are available from any program source.

There is, in sum, nothing to nit-pick concerning this audio bargain of an amplifier.

Hafler DH-200: Rear panel houses input and output terminals.

HAFLER DH-200 POWER AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Continuous power for rated THD		
(8 ohms, 1 kHz)	100 watts	114 watts
(4 ohms, 1 kHz)	150 watts	161 watts
FTC rated power (20 Hz to 20 kHz)	100 watts	108 watts
THD at rated output		
(8 ohms, 1 kHz)	0.02%	<0.002%
(8 ohms, 20 Hz)	0.02%	<0.002%
(8 ohms, 20 kHz)	0.02%	<0.002%
IM distortion, rated output, SMPTE	NA	<0.002%
CCIF	NA	0.03%
IHF	NA	0.03%
Frequency response		
(at 1 watt)	for −3 dB, 1 Hz to 100 kHz for ±0.5 dB, 10 Hz to 40 kHz	for −1 dB, 1 Hz to 80 kHz
S/N ratio, re: 1 W, "A" wtd, IHF	NA	91 dB
re: rated output, "A" wtd	NA	111 dB
Dynamic headroom, IHF	NA	1.9 dB
Damping factor at 50 Hz	150	150
Input sensitivity, IHF	NA	0.15 volt
Input sensitivity re: rated output	1.5 volts	1.5 volts
Slew rate (volts/microsecond)	30	30
Power consumption (watts)		
idling	NA	118 watts
maximum	NA	540 watts

CIRCLE 3 ON READER SERVICE CARD

They said we couldn't do it!

For years Peavey (and everyone else) depended on the same two or three companies to supply high efficiency, high quality loudspeaker products for use in our equipment. These few companies have been around for years and are, for the most part, producing their loudspeakers in the same way and from the same materials they always have. As the market demanded better performance, Peavey and other manufacturers increased the electronic sophistication of their products far in excess of the capabilities of the available high efficiency transducers. We attempted to explain to the "speaker geniuses" the problems and shortcomings encountered with their "beloved" products. We tried to explain why paper voice coils were inadequate. We tried to explain the power handling requirements necessary with the new generation of power amps. We tried to explain the need for better cooling, for stronger and lighter cones and diaphragms. But they wouldn't listen. They said, "**We** are the experts and we know that most equipment manufacturers and soundmen don't understand our 'precision' transducers and how to use them."

In desperation, we agonized over what we might do to satisfy our customers and to match the increasing sophistication of our electronics. After examining all the alternatives, we decided that we must apply an old adage...."**If you want it done right, then do it yourself.**"

We did!

Over five years of research and development, millions of dollars, and many thousands of hours of engineering time have gone into what we believe is the finest series of transducers avilable,... at any price.

The Black Widow loudspeakers have been designed "from the ground up" to handle the power delivered by modern amplifiers and to fulfill contemporary music and reliability requirements. These speakers are not "rehashes" of units designed back in the 30's or 40's but are **all new**, utilizing the latest in computer aided design techniques and the most efficient computer and numerically controlled production equipment. We have discovered new and superior materials, instituted new production techniques and adhesives. The need for field-replaceability was solved by having a field-replaceable basket assembly,...(A Peavey exclusive!) New technology for forming huge, 4-inch aluminum dome/coil forms and ribbon wire processing techniques we perfected to allow maximum efficiency and power handling while maintaining transient response, structural rigidity, and resistance to many classic failure mechanisms prevalent in older designs. Special attention has been paid to increased cooling capabilities with larger venting holes featuring acoustic foam/stainless steel mesh filters to prevent entry of dust.

We recognized years ago the coming scarcity of **alnico** and we designed our loudspeaker around the new super-energy strontium ferrite magnets giving us additional efficiency and magnetic energy in the gap (12,500 gauss).

The "established manufacturers" of high efficiency loudspeakers have been very critical of our efforts and continue to emphasize the various features on which they have depended so long, while branding us and our products as "upstarts" and extolling the virtues of alnico and other venerable materials and techniques. Meanwhile, they have been **frantically** redesigning their dated products and you will soon see that their "fantastic new generation of loudspeakers," which will be introduced with great **hullabaloo**, will closely resemble our innovative Black Widow series. We would ask that you remember **where** you saw these features initially and also ask that you consider the amount of care and dedication we at Peavey are putting into offering you what we believe to be the finest series of loudspeakers ever introduced for sound reinforcement,...**First!**

PEAVEY

PEAVEY ELECTRONICS
711 A Street/Meridian, MS.

HH Electronic Stereo-16 Mixing Console

By John Murphy and Jim Ford

The HH Electronic Stereo-16 is a sixteen input, stereo output sound mixing console intended for sound reinforcement applications. It accepts inputs from either low impedance balanced or high impedance unbalanced microphones or from line-level sources. Since the inputs and outputs are electronically balanced there are no audio transformers used in the unit. Each input channel is provided with four bands of equalization and there is a seven-band graphic equalizer on the stereo output. There are also provisions for one effects bus (typically used with an outboard reverb unit) and for "foldback" or stage monitoring. A headphone monitoring system allows the console operator to listen to either the stereo outputs, individual channels, the effects return or the foldback mix. Optional accessories available for the Stereo-16 include carrying cases, a plug-in "Digital Effects Module" [D.E.M.] and a multicore cable/stagebox system which interfaces with the mixer through a single multi-pin connector. The price is $2,695 (U.S. dollars; duty prepaid).

General Description: The Stereo-16 is highly compact and lightweight thereby making it readily portable and suitable for use by musical groups requiring a sixteen into two mixer for "live" sound reinforcement. The controls for the sixteen input channels are neatly arranged in sixteen identical columns and occupy most of the face of the mixer. The other console controls (output levels, effects, etc.) are grouped to the right of the input channels. Before going further let's take a look at one of the sixteen input channels.

At the top of each channel (furthest from the operator) is an input gain control which is used to establish the signal level through the channel. The input gain would normally be set as high as possible (to keep the signal level well above the noise) without overdriving the channel. This is accomplished by increasing the input gain until the overload indicator lights and then reducing the input gain to a level where overload is avoided on the loudest peaks.

Below the input gain control is a group of four rotary controls for the channel EQ. The frequency response curves with these controls set for maximum boost/cut are shown in *Figure 1*. The bass and treble controls have more or less a shelving action whereas the 600 Hz control acts on about a three-octave-wide band of frequencies centered on 600 Hz. Similarly, the 2 kHz control acts on a band of frequencies about one and a half octaves wide. The channel EQ cannot be switched out.

Continuing down the input channel, the next control is a rotary labeled "Foldback." It is used to adjust the level of the signal from that channel that is mixed into the foldback signal returning to the stage monitors. In other words, it is a monitor mix level control for that channel. The foldback signal is normally taken pre-fader so that it is independent of the channel fader and this is how the Stereo-16 is initially configured. However, the input channels can be switched internally to provide a post-fader foldback signal.

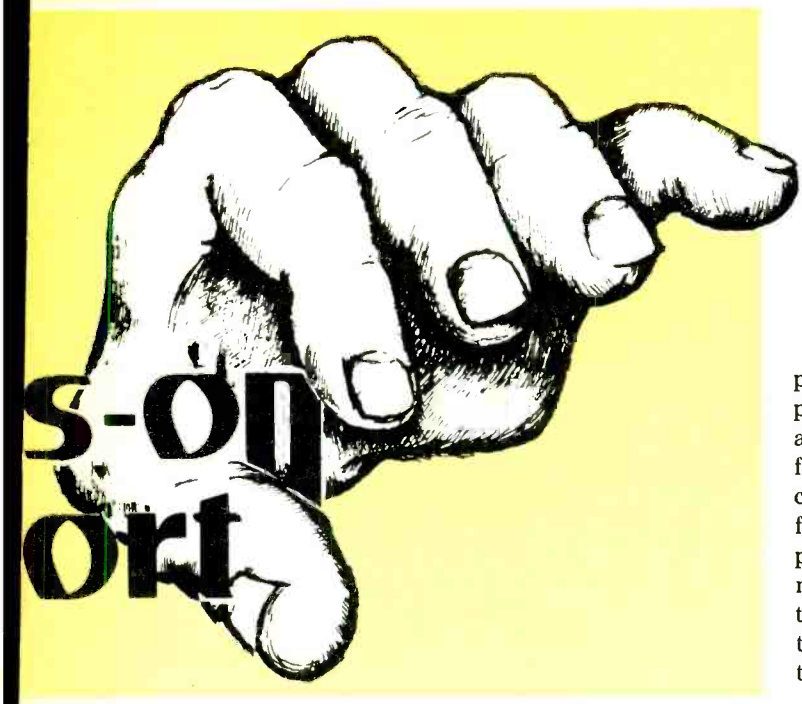
Next in line is a rotary control labeled "Echo." This control determines the level of the channel signal in the "echo send" mix. The echo send signal is taken "post fader" and is therefore dependent on the position of the channel fader (as it should be).

Below the echo send control is a pan pot for panning the channel signal between the left and right stereo outputs. When the control is centered the channel signal is fed equally to both outputs.

Between the pan pot and the channel fader there is a momentary contact pushbutton labeled "P.F.L." for "pre fade listen." When depressed the channel signal is available to the console operator for solo monitoring. However, this switch does not automatically interrupt the operator's headset monitor with the solo signal, instead the operator has to manually select the P.F.L. bus by switching the "monitor select" control to the "CH" position. Just to the right of the P.F.L. button is an LED channel overload indicator which illuminates at a signal level just below overload.

At the bottom of each input channel is the channel fader. This is a linear level control which adjusts the level of the channel signal in the stereo mix.

Now that we've examined the input channel in detail let's look at the output section. Just to the right of



channel 16 there is a linear fader with three rotary controls in line above it. These are the foldback master fader and the foldback equalization controls. The three tone controls are simply labeled "Treble," "Middle" and "Bass." These could be used to correct frequency response errors in the stage monitors and help provide more gain before feedback.

Near the top of the unit directly above the foldback controls are two slide switches for switching the graphic equalizer in and out of the two output channels. The graphic EQ controls are located just right of the in/out switches and span the remaining width of the board. The frequency response curves of this EQ are provided in *Figure 2*. Beneath the graphic EQ section there is a blank cover plate which is removed when the optional digital effects unit is installed.

Located below the cover plate are three master echo controls labeled, from left to right, "Echo Send," "Echo Pan" and "Echo Return." These serve as master level controls for the echo send output to control the signal level sent to an outboard effects unit (such as a reverb). The effects return then is used to adjust the overall level of effects signal mixed into the stereo outputs. The echo pan simply pans the returned echo signal between the stereo output channels.

The section just below the master echo controls is dedicated to the operator's headphone monitoring system. There is a ¼-inch stereo phone jack for connecting a pair of stereo headphones. Separate left and right volume controls are provided for the headphones and there is a four position monitor select switch for selecting the input to the headphone monitoring system. In the first position ("CH") the P.F.L. (pre fade listen) signals are supplied to the monitor. That is, only the channels for which the P.F.L. button is depressed will be heard in the headphones. The second position of the monitor select switch is labeled "Echo" and provides the echo return signal to the headphone for monitoring. The third switch position provides the console's stereo outputs to the headphones, while the fourth and last position of the monitor select switch provides the foldback signal to the headphone monitor. This monitoring flexibility allows the operator to monitor a variety of signals without disturbing the stereo output signal. The signal level of the monitored signal is indicated by the monitor VU meter.

The master stereo output faders are located at the bottom of the console just below the headphone monitoring section. This pair of linear faders sets the main output signal level from the board. The stereo output levels are indicated by a pair of lighted VU meters located just above the graphic equalizer. There is a slide switch between the meters labeled "Norm" and "−10 dB." In the −10 dB position the sensitivity of the meters is increased by 10 dB.

All of the input/output connections to the board are made at the rear panel. At the far left of the panel is a socket for the detachable line cord. Above this socket is a "Mains Voltage Selector" switch for selecting either 110 volt or 220 volt line operation. To the right of the voltage selector are two 3-pin XLR-type connectors for the echo send output and the echo return output. Although the echo send and return are set up for balanced lines they can also be used single sided (unbalanced). Directly below the echo connectors are the master outputs. These also use 3-pin connectors and provide a balanced output signal.

Continuing across the rear panel the next connector is for the foldback output. Again, this is a balanced output through a 3-pin connector. This signal is intended to return to the stage monitors to provide the performers with a monitor mix which in general is different from the master mix provided to the main (front) system. There is a DIN-type multipin connector located just above the foldback output. It is labeled "Tape Output" and makes the stereo output available to a tape machine so that the performance can be recorded if desired.

Most of the rear panel is taken up by the sixteen input connectors which are also 3-pin type. Below each input connector is a two position slide switch for selecting the input impedance of the channel (100 K ohms or 1.2 K ohms). In addition to selecting high/low input impedance, this switch bypasses a 20 dB gain

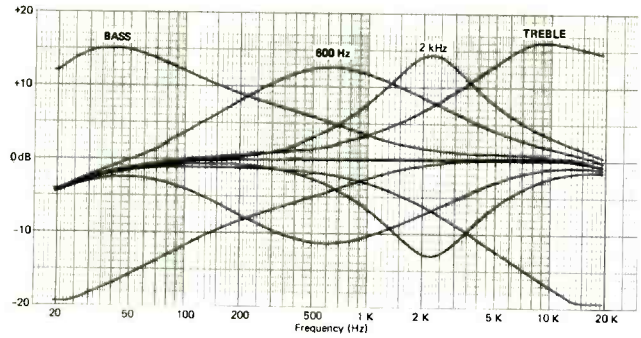


Fig. 1: H.H. Stereo-16: Input channel equalization curves (full boost/cut).

stage and changes the input from balanced to single sided when switched from the 1.2 K to the 100 K position. In the latter (100 k) position the input functions as a line input as well as a high impedance mic/instrument input. Low impedance, balanced output microphones should be used with the switch in the 1.2 K position. The last item on the rear panel is a multi-pin connector at the lower right-hand corner. This connector is used specifically to connect the optional multicore cable/stagebox system. This stagebox system provides sixteen balanced inputs from the stage to the console inputs, and returns the stereo outputs and the foldback output to the stage.

Listening/Handling Test: In order to evaluate the effect of the Stereo-16 on the audio quality of the signals processed through it we patched it into our reference listening system and listened to some direct disc music through it. We ran a line-level signal from our preamp into one input channel (with the input impedance switch in the 100 K position) and returned one of the stereo outputs to the tape input of our preamp. This allowed us to listen through the Stereo 16 or alternately bypass it by punching the tape monitor switch on the preamp. The levels were adjusted so that there was no loudness change when the console was punched in and out. The levels through the board were about 0 VU with peaks only occasionally going above that. After switching the Stereo-16 in and out of the system several times while listening carefully, we concluded that the unit introduced no sound of its own and was quite transparent to the music.

We were generally pleased with the action of the various console controls. However, a center detent on the channel EQ controls would be nice since the EQ can't be defeated. Zeroing that many (64) controls visually is not a simple task! The carrying handle on front of the unit makes it easy to carry and it seemed rather lightweight. A feature we would like to see in the monitor section is an automatic switching to the P.F.L. bus when a P.F.L. (solo) button is pressed. As it is, when a P.F.L. button is pressed the operator has to manually change the monitor select switch to get the signal in his headset.

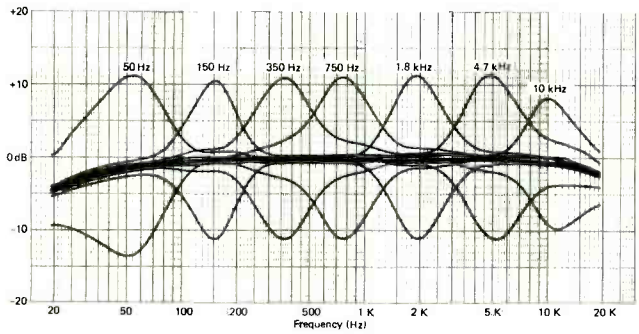


Fig. 2: H.H. Stereo-16: Seven-band graphic EQ response curves (full boost/cut).

Lab Test: The Stereo-16 was put through our test routine and the results are shown in the "Lab Test Summary." The input channel preamp has plenty of gain and adequate headroom for most applications. However, an external mic pad may occasionally be required. The unit has enough output level to drive just about any amplifier that might follow it.

The noise levels were fairly low but not quite as low as we've come to expect from some good P.A. mixers. With eight channels set for nominal levels the noise at the output would be about 62 dB below 0 VU.

We were surprised by our distortion measurements, especially considering that the console sounds very clean. We did note that the distortion products were almost all second harmonic components which probably explains the inaudibility of the distortion.

When the slewing performance of the unit was examined we first noted that the unit could be driven into "slewing" by a high frequency, high amplitude input signal. The slew rate limit through the console with nominal gain settings is 0.64 volts per microsecond. Considering the output signal level at 0 VU, this gives a slew rate ratio of 0.72 volts per microsecond per peak volt of output signal at 0 VU. The highest output level that keeps the slew rate ratio above the 0.5 recommended minimum is +3.15 VU. For higher output levels the slew rate ratio falls below 0.5. This Stereo-16 should be free from slewing induced distortion (SID) provided the output levels don't go much beyond 0 VU. A bit more slewing headroom would be nice though. The user handbook supplied with the mixer will allow a new user to properly interface the unit with a typical P.A. system.

Conclusion: The Stereo-16 sound reinforcement mixer by HH Electronic can be used to combine up to 16 inputs into a stereo output signal. The inputs can be individually equalized and there is a seven-band graphic EQ for the stereo output. The flexible headphone monitoring system, echo, and foldback (stage monitoring) facilities complete the package.

J.G. Jung; M.L. Stephens; C.C. Todd, "An Overview of SID and TIM Part II," *Audio*, LXIII (July 1979).

LAB TEST SUMMARY

Input Levels
(Note: 0 dBV = .775 Vrms)

Mic Input
(1.2 K ohms input setting)

Minimum input level for 0 VU indication with all level controls at maximum: – 65.6 dBV

With the input gain at minimum the input clips at: + 0.7 dBV

Line Input
(100 K ohms input setting)

Minimum input level for 0 VU indication with all levels at maximum: – 50.5 dBV

Input clips at: + 15.5 dBV

Output Levels
(at the stereo outputs)

For 0 VU indication with the stereo masters at nominal ("0") level: – 1.65 dBV

Maximum output level before clipping: + 20.5 dBV

Noise Levels
Note: 20 kHz bandwidth,
200 ohm mic input load, unweighted

Equivalent input noise: – 114.3 dBV

With all level controls for one input and one output at maximum, noise is: – 48.7 dBV

With all level controls at minimum, noise is: – 86.0 dBV

With input and output channel faders of one channel at nominal settings and the input gain set for a – 40 dBV input, the noise is: – 68.9 dBV

With eight channels set as above the noise is: – 63.1 dBV

Distortion
(THD at 0 VU with graphic EQ switched out)

Frequency	% THD
100 k	0.87
1 k	0.19
5 k	0.71
10 k	1.17
20 k	1.85

Note: The distortion is mostly second harmonic components

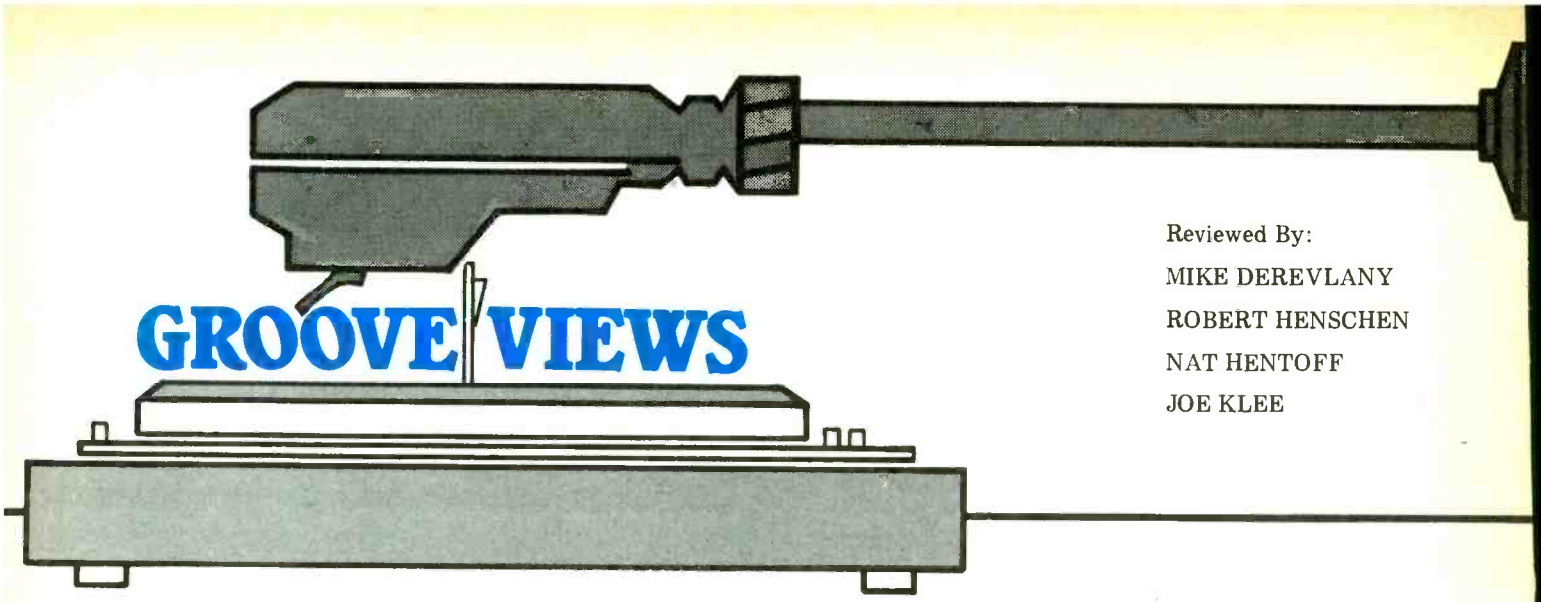
Bandwidth:
(– 3 dB points) 29 Hz to 24 kHz

Frequency Response:
± 1 dB from 64 Hz to 15 kHz

Slew Rate Limit
(nominal gain settings): 0.64 volts per microsecond

Slew Rate Ratio at 0 VU:
.72 volts per microsecond per peak volt output

CIRCLE 32 ON READER SERVICE CARD



Reviewed By:
MIKE DEREVLANY
ROBERT HENSCHEN
NAT HENTOFF
JOE KLEE

POPULAR

KANSAS: ***Monolith.*** [Kansas, producers; Davy Moire and Brad Aaron, engineers; recorded at Axis Studios and Apogee Studios, Atlanta, Georgia.] Kirshner FZ 36008.

Performance: **Seamless, distinctive fusion rock**
Recording: **Epic rock with subtleties**

Monolith is a concept album as good as anything Kansas has done. Powerful jacket artwork sets a scene of primitive American Indians as survivors in a demolished urban landscape. A liner quote from the Ghost Dance Chant presents an ideological perspective auguring rebirth from societal ashes.

While the Indian motif can be heard on "People Of The South Wind" and "How My Soul Cries Out For You," the real message here is ecclesiastically modern. Although man's future may be bleak and foreboding, possibly doomed, there is a time for action and that time is now. "Angels Have Fallen" represents lost values, but some hope can be gleaned from a final line: "It's you that must find them in living each day." Going farther still, "Stay Out Of Trouble" and "Reason To Be" approach the quest for salvation through spiritual improvement of the individual. Although none of the songs make specific mention of the Indian bit, and a romantic love theme works between the lines, there is a natural religiosity to the writings of Kerry Livgren and Steve Walsh in particular.

The music is a consistently strong vehicle for the message, starting with the emotional, directionless "On The Other Side." Surprisingly, no one musician stands out disproportionately and the cuts seem equally weighted, not always true of past Kansas efforts where the singles became increasingly easy to pick. Whereas the classical (keyboards), country (Robby Steinhardt's violin), jazz and rock elements were once possible to differentiate, Kansas has fused their influences into a more seamless, distinctive style.

The sound here is powerful, capturing the emotionalism of epic rock without compromising the subtle and creative nuances that still flourish. Kansas has found another potent vehicle for impressing their positive attitudes on the masses. R.H.

KANSAS: Ecclesiastically modern message, seamless, distinctive style

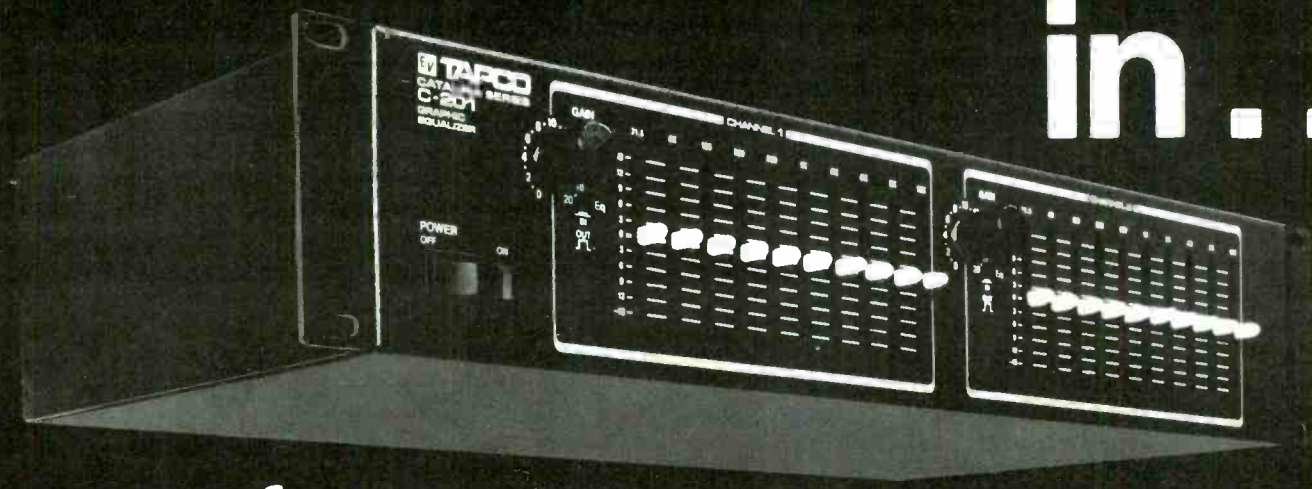
SPARKS: ***No. 1 In Heaven.*** [Giorgio Moroder, producer; Juergen Koppers and Giorgio Moroder, engineers; recorded at Musicland Studios in Munich, Germany, plus Sound Arts and Westlake Studio, Los Angeles, Ca.] Elektra 6E-186.

Performance: **Punk Disco**
Recording: **Perfect mass-production**

Pairing Ron and Russell Mael with producer Giorgio Moroder is like turning Chewbacca loose in a toy store. The heavily electronic music that results is a Saturday night disco social for reformed Hitler Youth – half tame, half crazy, and totally listenable.

"Tryouts For The Human Race" opens at a computerized pace that should turn Kraftwerk green with envy. Spark's former punkish penchant for manic beat and frenetic lyrics can still be heard behind cuts like

the Ultimate in...



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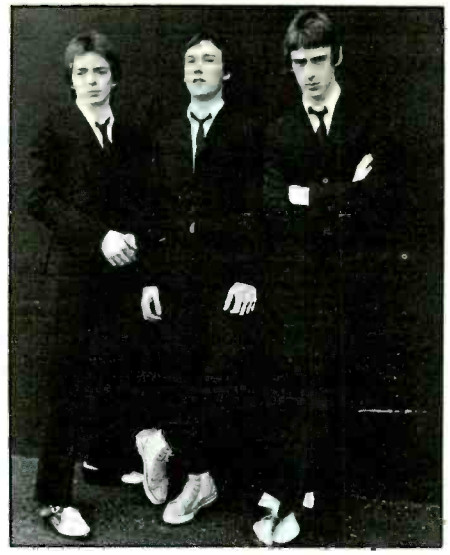
"Academy Award Performance" and "Beat The Clock," but you won't see Bud Collier dancing to these tunes.

Moroder, of course, is the machinized muscle behind Donna Summer's "Munich" sound, and what he does with Sparks is comparable. *No. 1 In Heaven* is full of propulsive disco rhythms, and the whole album owes a debt of gratitude to this decadent decade of dance.

The Mael brothers have correspondingly abandoned guitars in favor of a multi-synthesizer approach. In a recent *Melody Maker* interview, Ron and Russell called rock a joke and voiced approval for the mass-production sound of disco automation. Still, the two Sparks founders said that they wouldn't "be caught dead in a disco."

Despite some overtones of parody, the music here is good disco and state-of-the-art electronics. If the title track and others have pop potential and hit possibilities, there are heavier backdrops involved with instrumentation. "My Other Voice," for instance, employs an extraordinary effect by tape loop repetition of drummer Keith Forsey's cymbal figure—over which drums and keyboards eventually rise.

Lyrically, Sparks matches their robot music with words that are appropriately cryptic and ultimately laughable. Mock-serious deadpan antics are prerequisite to being *No. 1 In Heaven*, and if you see humor in machine age alienation, this one's probably meant for you. R.H.



THE JAM: **Rising above their origins**

THE JAM: *All Mod Cons.* [Vic Coppersmith-Heaven, Chris Parry, producers; Vic Coppersmith-Heaven, Greg Jackman, Roger Bechirian, engineers; Peter Schierwade, Philip Thornalley, assistant engineers; recorded at RAK (Upper London) and Eden Studios.] Polydor PD-1-6188

Performance: **Forceful but well-honed**
Recording: **Clean and effective**

When the label "new wave" is brought up, it immediately conjures images of rough sounds characterized by an almost violent simplicity. New wave is a term most often applied to groups such as The Jam and the now infamously defunct Sex Pistols, whose demise signalled a change in the whole new wave phenomenon. The Jam is representative of that change.

Gone is the harsh, grating quality of angry adolescent music and the thin, unflattering sound of second-rate recordings. In their stead is a relatively sophisticated trio with the ability and energy to entertain and communicate with unescapable impact. Distorted and unbalanced sound have given way to more standard and suitable methods.

The Jam has not become a new version of ELP or Steely Dan, despite their very noticeable changes. They're still a

heavily guitar-oriented group, with music and lyrics that are straightforward and uncompromising. The style has broadened, however, to include vibrantly mellow acoustic guitar, along with their previous hard, crashing, electric power. Either way, the guitar sound is bright and resonant.

While avoiding any fancy production tricks, the Jam uses some overdubbing on every track, which is really an inevitability with a three-piece group. It's done well on *All Mod Cons* and effectively rounds out the otherwise limited sound of The Jam. Superfluous sounds are present on only one track, "Down in a Tube Station at Midnight," which utilizes the sounds of a London subway.

With *All Mod Cons* The Jam shows the potential to become a major musical force. Their tunes are catchy and witty, with just enough complexity to be enjoyable. The songs are full of social commetary and human reality and contain a basic appeal. With a little more exposure, they could rise well above their tumultuous origins. M.D.

AXIS: *It's A Circus World.* [Andy Johns, producer and engineer; Ron Alvarez, Gray Russell, Rick Smith, assistant engineers; recorded at Record Plant Studios, New York and Los Angeles.] RCA AFL1-2950.

Performance: **Unenterprising**
Recording: **Par for the course**

There are always new and unknown groups coming out with albums. Sometimes these albums are masterpieces and proceed to make musical history; more often, they are bland and uninspired, filled with the music of their more popular, more original predecessors, whose shadow will inevitably eclipse any efforts of the unknown to encroach upon their success. Just as often, the group and their album are a not-unpalatable mixture of both, a curious fusion of limited vision and stale, over-duplicated formulas.

Axis is one such group. Some of their material is reminiscent of Rick Derringer, with whom two of the band's members recorded on three albums, yet the style is clearly their own. The trio does put out an appreciable product, but their sophisticated and richly textured heavy metal sound does little more than imply some superb, if not downright fancy, engineering.

Side one starts out with "Brown Eyes," a tune dominated by hard, driving guitar which, as it is on the rest of the album, is carefully balanced along with the total sound; instruments do not drown out other instruments or vocals, and nothing is emphasized to excess. "Busted Love" is more complex but still has that unflagging guitar rhythm; it is also a prime example of the lyrical void that pervades the album. Banal though they may be, the lyrics work well with many of the tunes and help reinforce their primal strengths, as shown in "Juggler" and the apparently Kiss-inspired "Soldiers of Love." A smidgeon of versatility is displayed in the western-flavored "Train," ending this side.

Side two takes off with a light sci-fi song, "Armageddon," that, despite its weaknesses, is the strongest cut on the side. Most of the other tunes, "Ray's Electric Farm," "Cats in the Alley," and "Bandits of Rock" are pretty much the same—sterile but enjoyable. "Stormy Weather" is a hybrid boogie-blues tune with excellent guitar-work that doesn't prevent it from becoming bogged down by its several flaws. "Circus World" is neither heavy nor metal but a crude combination of unimaginative strings and rejected parts of other songs.

Unfortunately for Axis, *It's A Circus World* is the kind of album that is quickly relegated to the background music pile since its defects, though not obnoxious, are large and readily detectable on less than careful scrutiny. Most of the tunes manage to start off brilliantly but very rapidly disintegrate. The one noticeably redeeming quality of the album is guitarist Danny Johnson, whose talents as a performer and his strengths as a wrtier outshine only slightly those of his capable cohorts, drummer Vinny Appice and bassist Jay Davis, but aren't enough to bring the band up to its full potential. As it presently stands, Axis is hardly a sideshow, let alone a ringside event. M.D.

JEFFREE: ***Jeffree.*** [Jeffree, producer; Jerry Hall, Barney Perkins, Brian Levi, Steve Smith, Lenard Jackson, Jim Nipar, and Arlene Ferandelli, engineers; recorded at Fort Knox Recording Studio, Star Track, ABC Recording Studio, Upside Down, Sound Factory, Los Angeles, Ca.] MCA-3072.

Performance: **In the finest Motown tradition**
Recording: **Adequate but inconsistent**

Motown was one of the great American record labels in the sixties, a prime mover in the development of contemporary soul music by The Temptations, Smokey Robinson and The Miracles, The Supremes, and many others. Motown seems pretty low key as we leave the seventies behind, but it's important to recall that the Motown organization was responsible for at least two pivotal albums in this decade: Stevie Wonder's *Music Of My Mind* and Marvin Gaye's *What's Goin' On.* Stevie has been a biggie throughout the last ten years, but some of us just never got enough vintage Marvin.

MCA's new solo artist Jeffree, however, is steeped in the Motown tradition and sounds very close to Marvin Gaye. That's because he *is.* Formerly a staff writer at Motown, for Diana Ross and Main Ingredient among others, Jeffree was personally encouraged by friend Gaye to enter the recording arena with his self-penned music. Jeffree's musical destiny has actually been shaped over a lifetime: all four of his brothers are writer-performers, and Uncle Robert Bateman was one of Motown's principal session

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engineers.

The "Marvelous Marvin" sound is obvious from the very first tune, "Mr. Fix-It," and crops up repeatedly on "One Last Chance," "Take My Love," and "Love's Gonna Last." Despite some orchestration (another Gaye trademark) Jeffree generates a loose atmosphere with his vocal ad libs and funky conga rhythms. His groovy band stretches out across lengthy cuts like "All My Loving (Was Made For You)."

Of course, nobody can quite duplicate the unique edge Marvin Gaye used to get on his vocals, and Jeffree cuts like "Better Wake Up Girl" and "I Can't Help It Baby" definitely lack any kind of lyrical clout. As a singer, Jeffree has plenty of potential in the laid back soul-jazz idiom, and he has done a pretty fair job of producing his own debut for MCA. Jeffree should continue in this direction, hang onto those important Motor City influences, and at the same time try to let his own uniqueness show through. R.H.

LEAH KUNKEL: ***Leah Kunkel.*** [Val Garay and Russ Kunkel, producers; Val Garay, engineer; recorded at Sound Factory and Studio 55 in Los Angeles, Ca.] Columbia JC 35778.

Performance: **Seasoned**
Recording: **Rich**

This lady's last name is well-known to fans of L.A. studio drummer Russ Kunkel, Leah's husband of several years. But the vocal quality is vaguely recognizable too, and when you see a close-up of Leah Kunkel's face, the loose ends of familiarity begin coming together—this is Mama Cass Elliott's little sister, finally stepping out for her first album.

And it's a promising debut, almost on the level of a Karla Bonoff or Valerie Carter session. West Coast stalwarts abound (Jackson Browne, Andrew Gold, Toto's Steve Lukather) and Russ Kunkel's colleagues from The Section provide the healthiest of support. Even original Papa John Phillips contributes a tune, "Step Out," to these entertaining, professional proceedings.

And Leah, who has been singing heretofore in the hinterlands of coastal studio sessions, is very much up to the challenge of playing alongside such heavies. Her voice has that mellow resonance that distinguished her sister's work in the sixties, and she's

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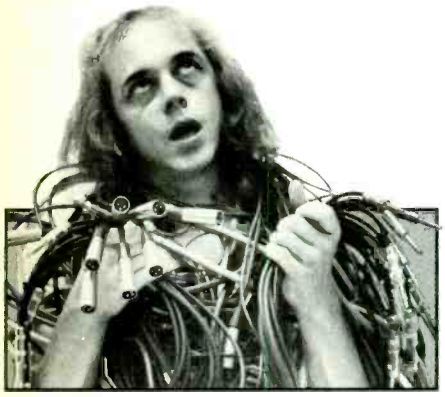

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LEAH KUNKEL: Promising

soulful when she wants to be. Kunkel covers two Motown oldies in "If I Could Build My Whole World Around You" and "Down The Backstairs Of My Life," and does a reproduction of the Bee Gee's "I've Gotta Get A Message To You," authentic right down to the reedy tremolo on backing vocals.

But these Ronstadtian remakes of bygone biggies are seldom as satisfying as new tunes that help shape an emerging artist's personal creative stance. Leah Kunkel's strong suit at this point is the torch ballad, and she does beautiful things with Stephen Bishop's "Fool at Heart" or her own "Souvenir of the Circus." Easy rockers like "Step Right Up" and "Under the Jamaican Moon" are equally strong, and recording standards are just first-rate throughout the entire album. It's a promising talent at work here—a singer strong enough to take more chances on her second effort. R.H.

JAZZ

OSCAR PETERSON: *The Paris Concert.* [Norman Granz, producer; recorded Oct. 5, 1978 at Salle Pleyel, Paris; engineer not listed.] Pablo Live 2620112.

OSCAR PETERSON: *The London Concert.* [Norman Granz, producer; recorded Oct. 21, 1978 at Royal Festival Hall, London, by Basing Street Studios; no engineer listed.] Pablo Live 2620111

Performances: **Oscar-winning pianist**
Recordings: **Just what went down in the halls**

Norman Granz may not have created the idea of releasing recordings of "live" jazz concerts but he certainly has been its primary practitioner. His "Jazz At the Philharmonic" tours were always taped and many of them showed up on the various labels with which he has been associated from time to time (Asch, Clef, Norgran, Mercury, Verve and finally Pablo).

As for Peterson, his fusion of Art Tatum's dexterity and bravura with Bud Powell's modernism and Teddy Wilson's melodiousness has led to a style that, if a bit overly ornamental, is moving and swinging and fits admirably with either a drummer or a drumless rhythm section. In London, he has Louis Bellson on drums and, in Paris, Joe Pass on guitar. I will admit to a preference for piano, bass, guitar over piano, bass, drum so my preference for Peterson with Pass must be taken as not totally unbiased. Yet hear the difference between "Sweet Georgea Brown," the only tune common to both sets, and realize that Peterson doesn't need any other timekeeper than his own good left hand. To be sure, Oscar changes his playing when he's working with Bellson and plays less percussively than when he's playing with Pass, but that's part of what makes me prefer the sessions with Pass. The drummer, as is also the case with Earl Hines, serves as much to inhibit Peterson's drive and percussive spirit as he does anything else for the recording.

Still both these sets, as anything Oscar plays, are worthwhile. If you're a Peterson fan, I'd recommend both. If you're new to Oscar's work, I might suggest starting in Paris and then crossing the channel once you've absorbed his style. Also, Pass (a marvelous solo guitarist) gets a few numbers to himself and some with Niels Pedersen on bass that break up the monotony of a double LP of piano dominated music. J.K.

HORACE SILVER: *Sterling Silver.* [Alfred Lion, producer; Michael Cuscuna, reissue producer; Rudy Van Gelder, engineer; recorded between November 10, 1956, and January 28, 1964, at the Village Gate, New York, N.Y. or Van Gelder Studio, New Jersey.] Blue Note BN LA 945H.

Performance: **On the Silver standard**
Recording: **Typical of Van Gelder's excellence**

In his liner notes, Mike Cuscuna points out that this is Horace Silver's twenty-eighth year with Blue Note and with the exception of one album on CBS to finish out a commitment that Art Blakey's Jazz Messengers had made to that label all the recordings of Horace Silver's group have been on Blue Note. As much as anybody that's a tribute to guys like Alfred Lion and Francis Wolff who spotted the talents early and signed them to the label first. That includes

HORACE SILVER: Consistent

not only Horace Silver but later day heavies such as Thelonius Monk and Jimmy Smith who, unlike Silver, later left for other labels but did much of their pioneering work for the Blue and White banner of Lion and Wolff.

This, apparently, is a set of leftovers. Things that either didn't make the LP they were intended for or were released as 45 RPM singles for the juke box trade. *If* these are the bottom of Horace Silver's barrel, so to speak, Horace's leftovers are better than most men's main course. Players like Junior Cook, Blue Mitchell, Donald Byrd and Hank Mobley dart in and out of this reissue like meteors and the music is as top-drawer as all these names would indicate. Particularly fetching are the alternate instrumental version of "Senor Blues", and the vocal version of the same tune. They were both originally issued as 45s and this is their first time on LP.

Special mention must go to Rudy Van Gelder for the uniform sound quality achieved throughout even though these recordings span eight years and include

A Legend Returns— and a New One Begins

By Nat Hentoff

Back in the 1940s, Joe Albany was one of the more fabled of early modern jazz pianists. He worked with Benny Carter and Lester Young, but became especially noted by other musicians for his time with Charlie Parker. Then, for the next two decades, Albany unaccountably slipped into limbo. He did gig around Los Angeles, but hardly ever recorded. Yet, as musicians and intent listeners heard air shots and other evidence of Albany's skills at the beginning of bop, Joe turned into a legend.

In the early 1970s, Albany was in Europe for several years, recording just enough to make listeners want more. Based in Los Angeles since 1975, he recently moved to New York where he has recorded his first full-scale American album as leader in years. *Bird Lives!* (Interplay Records, P.O. Box 93, Calabsas, California 91302). They're all Parker tunes (ranging from "Now's The Time" and 'Yardbird Suite" to "Confirmation" and "Barbados"). And the rest of the trio is compellingly inventive—drummer Roy Haynes and bassist Art Davis. Joe Albany himself is in stimulatingly regenerated form—playing with crisply swinging imagination, impressive sensitivity to dynamics, and an exultancy that gives his playing more vibrancy than it has ever had. The recorded sound is as vivid and clear as the playing.

Until a year or so ago, JoAnne Brackeen was almost entirely unknown to the general jazz audience. Yet now she too shows the makings of a legend. She was known to musicians, being the first woman player to rise through the ranks in such big league combos as those headed by Dexter Gordon, Teddy Edwards, Joe Henderson, Art Blakey, and Stan Getz. But, JoAnne insists, she was never hired as "a woman." "I'm a musician," she says, "and people have always treated me that way."

After working in Los Angeles, JoAnne came to New York in 1965; and in the last few years, she has finally ventured on her own—heading trios in this country and in Europe. Her playing is strikingly inventive, whether she's stretching out on densely textured, bristlingly swinging originals or transforming ballads with uncommon grace but not a touch of sentimentality. Already, in the 1979 *Down Beat* International Jazz Critics' Poll, JoAnne is listed sixth worldwide—preceded by Cecil Taylor, McCoy Tyner, Keith Jarrett, Bill Evans, and Oscar Peterson. And she has only begun to find her audience.

On *Keyed In* (Tappan Zee/Columbia), JoAnne works with two of her favorite improvisers, bassist Eddie Gomez and drummer Jack DeJohnette. The tunes are all Brackeen originals and demonstrate her consistent strength as a composer of sinewy, intriguing designs as well as an improviser of persistent freshness of conception and prodigious technique. The recording quality has a natural-sounding brilliance and depth of sound that wholly illuminate her wide-ranging musicianship. JoAnne has come this far, by the way, while also being married and the mother of four teenagers.

JOE ALBANY: ***Bird Lives!*** [Toshiya Taenaka, producer; David Baker, engineer] Interplay IP-7723.

JOANNE BRACKEEN: ***Keyed In.*** [Bob James, Joe Jorgensen, producers; Joe Jorgensen, engineer.] Tappan Zee/Columbia JC 36075).

both studio sessions and "live" remotes. It comes out a lot less of a hodge-podge than you'd expect but that's the name of the game—consistency. J.K.

DUKE ELLINGTON: *1938.* [Gunther Schuller and Martin Williams, reissue producers; master tapes prepared by Jerry Valburn; originally recorded between 1/13/38 and 12/22/38 for Brunswick Records, mostly in New York, N.Y.] Smithsonian R 003.

DUKE ELLINGTON: *1939.* [Martin Williams, reissue producer; master tapes prepared by Jack Towers and Jerry Valburn; records originally recorded between 4/11/38 and 2/14/40 for Brunswick and Columbia Records, mostly in New York, N.Y.] Smithsonian R 010.

DUKE ELLINGTON: *1940.* [Martin Williams, reissue producer; master tapes prepared by Jack Towers; records originally recorded between 3/6/40 and 12/28/40 for Victor Records, mostly in New York and Chicago.] Smithsonian R 013.

Performances: **1938, 1939, and 1940—good years for Ellingtonians**
Recordings: **Better for Brunswick and Columbia than Victor**

Duke Ellington's Orchestra made its first records for Blu Disk in 1924. The Duke Ellington Orchestra, under the direction of Duke's son and heir Mercer Ellington, is still making records today. It is important that those who like to say that the band isn't what it was before Duke Ellington's death in 1974 are reminded that the band Mercer took over immediately after Duke's death was virtually the same band that Duke had been leading just prior to that time. It is also important to admit that the Ellington Orchestra had its ups and downs. Just where the peaks and valleys were is something Ellington fans will argue about as long as there are Ellington fans. My personal opinion is that the band started out on a fairly high level of musical interest and capability from their recorded beginnings in 1924, increasing as more important soloists were added and Duke's writing and arranging matured until they reached their zenith with the addition of tenor saxophonist Ben Webster early in 1940. I suspect most Ellington fans would agree with me on the time period for the beginning of the golden age, but the time point for the end is less definite. Most accept the beginning of the end as the day in the final months of 1940 when, as Raymond Scott put it in a funeral ode he composed for his own orchestra, "Cootie Left The Duke." Others will go up to the recording ban of 1942 before writing off the band. Some will even extend the golden age up until the departures of Johnny Hodges (temporary) and Sonny Greer (permanent) from the band in 1951. The most die-hard Ellington fans liked the band right up to the time of Duke's death and some are still fans today. My personal theory is a gradual decline starting with Cootie's departure in 1940 which increased in 1941 as Jimmy Blanton left the band to die soon after and picked up in momentum as Barney Bigard, Ben Webster and Rex Stewart took their leave. By July 1946, when plunger trombone stylist Tricky Sam Nanton died, for me at least, the downward trend was irreversible.

For sure the years between 1938 and 1940 are vintage years of Ellingtonia. They include such well known gems as "I Let A Song Go Out Of My Heart," "Boy Meets Horn," "Serenade To Sweden," "Sophisticated Lady," "Concerto For Cootie" and "Never No Lament." There are also such lesser known but marvelous recordings as "A Gypsy Without A Song," "The Sargeant Was Shy" and "All Too Soon." These sides are generously sprinkled with spirited solos by the likes of Cootie Williams, Rex Stewart, Lawrence Brown, Tricky Sam Nanton, Johnny Hodges, Ben Webster, Barney Bigard and Harry Carney, plus a rhythm section that fairly bristled under the abrasive beat of Sonny Greer on drums. These sides should be required listening for any jazz fan of any age. Now they are available again with liner notes that vary from Albert Murray's intelligent essay (1939) to the pedantic and precious words of Gunther Schuller (1938) and for the notes of Larry Gushee (1940) you can add to my complaints about Schuller's notes a certain ugly complaining quality and a tendency to second guess the Duke.

When the 1938 set was issued there were so many complaints about Shuller's inclusion of items like "Lambeth Walk" and "A Blues Serenade" to the exclusion of "I'm Slappin' Seventh Avenue" and "Old King Dooji" that Martin Williams decided to include these two '38 items in the 1939 volume. The reason that the 1939 volume includes one cut from 1940 is simply that it brought the set up to the finish of Duke Ellington's Columbia/Brunswick contract so the 1940 set could begin with the Victor sides. The decision probably had more to do with record company policies and politics than anything else but it was a wise move anyway. Columbia/Brunswick had A&R men like John Hammond and Morty Palitz and a beautifully "live" studio in Liederkranz Hall. Victor had an A&R man named Eli Oberstein who despite giving Duke Ellington a plenteous opportunity to record his music must have had ears of lead not to notice the deadness of sound in Victor's studio. Having the Liederkranz Hall sides on one album and all those dead studio sides on the other saves the listener a lot of difficult ear-compensating.

I won't try to tell you that everything the Ellington band put down, even in their golden years, was a gem. I wouldn't even try to tell you that the sides of these albums are all gems. But what the Ellington band could do with trash like "Lambeth Walk" and "At A Dixie Roadside Diner" is worth considering also. I guess those of us who lived through the '30s and '40s don't need to be reminded of these sides but it's nice to be reminded anyway, and for those youngsters who missed the turn of the decade leading up to World War Two, here's your chance to do your homework. Hearing Duke Ellington at his zenith will help you to hear what came after in its proper perspective.

Smithsonian Records can be mail-ordered from Smithsonian Customer Service, P.O. Box 10230, Des Moines, Iowa 50336 at the price of $9.99 for each of these two LP sets plus $1.49 per order for mailing and handling. Or if you're going to Washington D.C. to see the cherry blossoms you'll find these blossoms on sale at the gift shop of the Smithsonian Institute. J.K.

JOHNNY GUARNIERI: *Gliss Me Again.* [No producer listed; Francis Miannay, engineer; recorded at Barclay Studios, Paris, France, March 10, 1975.] Classic Jazz CJ 105.

Performance: **The great Guarnieri returns with Slam still swinging**
Recording: **Good piano sound**

Once upon a time. back in the '40s, there were a million or so independent record companies each operating out of a shoe box somewhere putting out 78 RPM jazz recordings. Each and every one of these companies put out recordings with Johnny Guarnieri on piano and Slam Stewart on bass. They were the two favorites of the era. They backed Don Byas, Lester Young and just about everyone else you can think of. Slam even went on to record with Dizzy Gillespie and Charlie Parker and the boppers. I don't think Guarnieri got that far away from the mainstream but then I don't think I heard every one of those little labels that some fly by night record company put out. Whoever else they recorded for, Guarnieri and Slam did a lot of their most memorable work for Herman Lubinsky's Savoy Records in a trio setting with Sammy Weiss on drums. This was the group that first recorded "Gliss Me Again" on September 20, 1944. The trio did a whole album of four ten-inch 78s which, as I remember, had a rather wild cartoon of the three on the cover.

If it took this long to get the great reunion together, it was worth waiting for. Both the men are in as good form as they were even then. For some reason a guitar was added, Jimmy Shirley, and while he plays well I find him an unnecessary interruption to my memories. 1975's drummer was Jackie Williams but I won't complain about that because Jackie does a good job of swingin' away and that's really what was needed both in '75 and '44.

What is surprising is that this LP doesn't include more of Slam and Johnny's greatest hits from the '44 session. In fact the title tune is the only repeat. I'd love to have heard what Johnny and Slam would do with "That Old Black Magic" thirty-one years later. Of course there are nice tunes throughout the LP, ranging all the way from classics like "Indiana" and "Cherokee" to Guarnieri originals like "Walla Walla." It's just as easy to say that I'd like to have heard what the '40s trio would have done with those.

Slam and Williams and Jimmy Shirley are on a lot of this album but Johnny does get to do some unaccompanied solo work which is how he's been doing most of his playing lately. There is still a special magic when Johnny Guarnieri and Slam Stewart get together again—under any circumstances.

One thing is immediately apparent on comparing the Savoy sides with this new recording for Disques Black and Blue and issued here on Classic Jazz. If there is one category in which recording has improved markedly over 1944 it is piano recording. The new sides sound like a piano, thankfully not some thin, tinkling imitation of one. J.K.

CLASSICAL

ENRICO CARUSO: ***The Complete Caruso, Vol. 4.*** [John Pfeiffer, A&R Coordinator; digital restoration by Soundstream, Inc., Thomas G. Stockham, Jr., engineer; originally recorded between Feb. 11, 1906 and March 24, 1907 in New York, N.Y. or Camden N.J.] RCA ARM1-2766.

ENRICO CARUSO: ***The Complete Caruso, Vol. 5.*** [Same production credits as above, except recorded between Jan. 10, 1908 and Nov. 6, 1909.] RCA ARM1-2767.

ENRICO CARUSO: ***The Complete Caruso, Vol. 6.*** [Same production credits as above, except recorded between Nov. 6, 1909 and Jan. 12, 1910.] RCA ARM1-3373.

ENRICO CARUSO: ***The Complete Caruso, Vol. 7.*** [Same production credits as above, except recorded between Jan. 12, 1910 and Dec. 28, 1910.] RCA ARM1-3374.

Performances: **The greatest tenor of the ages**
Recordings: **From crude acoustic originals, Soundstream has created some fine likenesses**

Enrico Caruso, a name that is synonymous with the golden age of opera, was and is to opera what the Duesenberg was and is to automobiles. Each was the best in its time and each came to symbolize the entire milieu. Therefore, recent competitors have had to fight not only the genuine article but our romanticized memories as well.

Yet it was not always so. When Enrico Caruso burst upon the Metropolitan Opera House, there was those who longed for the days of yore and the voice of Jean De Reszke. Among those tenors who followed Caruso was one so formidable that Caruso is reported to have said

about him (Benjamino Gigli) that at least they could have waited until he was gone. Still others followed—Martinelli, Bjorling, Tucker, Domingo, Pavrotti—yet Caruso's majesty has not been diminished. It has grown with every generation that has listened to his recordings since their original issue on 78 RPM Victor Red Seal records. This latest reincarnation is part of what it is hoped will be the eventual issuance of the complete recordings of Enrico Caruso. The set begins at Volume Four because the previous recordings, made with piano accompaniment rather than orchestra, are less desirable to listen to and perhaps also more difficult to restore by the Soundstream process which involves the use of computer processing to compensate for the idiosyncracies of acoustic (horn) recording. I've heard from several people who actually heard Enrico Caruso at the Met that these recordings come surprisingly close to the "live" sound of his voice.

There can be no quarrel with these performances. Caruso's recordings of "Vesti la giubba" or "Celeste Aida" are too well known and respected to need justification. But they are all magnificent recordings—from the 1906 "Di Quella Pira" that opens Volume 4 to the tragic "No! Pagliaccio non son" from 1910 which closes Volume 7. The gems in between include such lesser known beauties as Tosti's "Ideale," a gorgeously understated "E lucevan le stelle" from *Tosca*, two recordings of "Miserere" from *La Traviata* with soprano Frances Alda (one with chorus, one without and a real gem), no less than nine excerpts from Gounod's *Faust* recorded in the company of artists such as Geraldine Farrar, Marcel Journet and Antonio Scotti. The liner notes are by a different writer for each volume—Francis Robinson, Harvey Phillips, Martin Sokol and George Jellinek, respectively.

Alternate masters do not seem to have been included in most cases. It may well be that they no longer exist. It was frequently the practice to destroy alternate masters which were not to be issued. When Caruso was asked what should be done with the wax master of one of his recordings which was not to be issued, Caruso reportedly replied "I think you better break him up." I wish we could have heard the version of "Miserere" which was recorded with Johanna Gadski but was not released. Yet we can be glad that both masters of "Miserere" with Alda were preserved

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and appear here. Likewise, two takes of "Deserto in terra" from Donizetti's *Don Sebastien.* Alternate masters are not as essential in operatic recordings as they are in improvised jazz performances which can vary significantly from take to take. Yet those we do have are of interest and sometimes give us a significant insight into the performer's modus operandi.

While a chronological reissue series is one way, a valid one at that, to handle such a reissue program I have long felt that a more significant means of organization would be to issue all the works from a given opera on an LP. RCA did this for a while on their Victrola series putting out a collectors' *Aida* highlights and highlights of the Met's first *Madame Butterfly.* Each has its advantages and disadvantages. One advantage would be to have all six versions of "Celeste Aida" following each other in chronological sequence from the 1902 version recorded for G&T (Gramophone and Typewriter Corp.) to the famous 1911 Victor version, the only one to include the recitative, "Se quel guerriero io fossi."

I have no idea how long this current Caruso project will take to complete. RCA will have to negotiate with EMI (the successors to G&T) and whoever owns the rights to Pathe Marconi and Zonophone for material prior to 1904 when the tenor signed with Victor. There are also many more recordings which Caruso made for Victor between 1910 when Volume 7 leaves off and his last recordings in 1920. What I do know is that by the time this project is finished, we will have an orderly and as complete as possible representation of the recorded career of Enrico Caruso restored to something resembling the original, "live" Caruso sound: not enhanced, not rechannelled for stereo, but a sound closer to the real thing than those who purchased the original 78s ever dreamed possible. J.K.

SHOWS and SOUNDTRACKS

ORIGINAL CAST: *Ziegfeld Follies of 1919.* [Carl Scheele, John M. Raymond and Martin Williams, reissue producers; selections recorded between Feb. 13, 1919 and Jan. 1920, mostly in New York, N.Y.] Smithsonian R 009.

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ORIGINAL CAST: ***Whoopee.*** [Brock Holmes, Stanley Green, Miles Krueger and Martin Williams, reissue producers; Jack Towers, reissue engineer; recorded between Oct. 2, 1928 and Jan. 28, 1929, mostly in New York, N.Y.] Smithsonian R 012.

Performances: **Slices of history**
Recordings: **I hope you aren't expecting stereo—hi-fi—quad!**

The Ziegfeld name was magic on Broadway. Whether it was one of his annual follies or a book show like *Show Boat*, *Kid Boots* or *Whoopee*, the Ziegfeld banner was enough to get a show off to a running start. So were stars like Bert Williams, John Steel, Ruth Etting and others who graced his many shows. But the star of stars had to be Edward Israel Iskowitz who, under his stage name of Eddie Cantor, made millions laugh and cry from his first Ziegfeld Follies in 1917 to his retirement from active show business in the 1950s. He was, along with Williams, Steel and Van and Schenck, one of the stars of the 1919 Follies with an Irving Berlin score which included "A Pretty Girl Is Like A Melody," "Mandy" and "I've Got My Captain Working For Me Now." By 1928 when *Whoopee* hit the boards Cantor was *the* star, even though artists like Ruth Etting and George Olsen's band were also in the show.

After the success they've been having with their jazz reissues, Smithsonian Records decided to do something about that peculiarly American commodity—the musical comedy. Never mind that the musical had its roots both in the operettas of Gilbert and Sullivan and of Franz Lehar and Johann Strauss. It was the Irving Berlins and the George Gershwins who honed the show tune and the musical show to its finest edge...and it took producers like Ziegfeld and George White to make them come to life in lavish style on the Broadway stage. It was a style so lavish that it did not transfer well to the strictly audio sensation of records. But people leaving the theatre who had heard Cantor sing "Makin' Whoopee" and "Hungry Women" wanted his records to take home and play at their leisure. It took a lot of effort to track down the recordings of music from these two vintage musicals, wherever possible by the same artists who sang them on stage or at least those who

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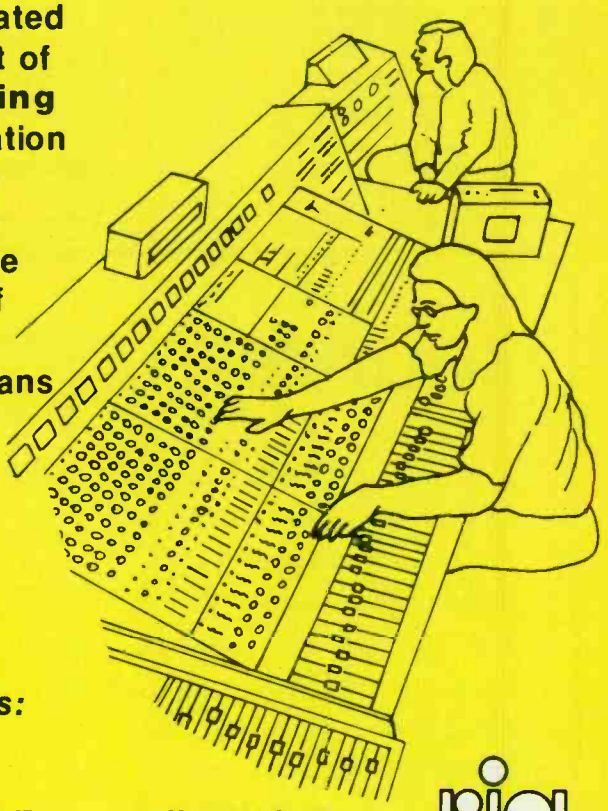

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were in the show. It was a task made more difficult by the practice of interpolating hit numbers that an artist had just recorded into the score of the show. This is why we find Bert Williams adding his specialities like "Somebody Else Not Me" and "The Moon Shines On The Moon Shine" into the 1919 Follies. Spots were reserved for stars like Bert Williams and Eddie Cantor and Van and Schenck to do just that. Listed in the program simply as "songs," they could change from night to night and the practice continued even into the late twenties. This can be shown by the inclusion of two Eddie Cantor specialities—"Eddie Cantor's Automobile Horn Song" and "I Faw Down And Go Boom" as sung by Eddie Cantor as the singing waiter in a score otherwise written by Gus Kahn and Walter Donaldson. Even "Hungry Women," though identified on the Victor 78 RPM record (no. 21831) as being from the Ziegfeld production, *Whoopee*, was not by the composers of the show's score, but by the popular songwriting team of Jack Yellen and Milton Ager. There were a lot of minor stars of these shows who never got recorded and a lot of major stars in the show recorded material which they did not sing on stage.

Using the available records by John Steel, Eddie Cantor, Bert Williams, Van and Schenck, Ruth Etting and the George Olsen Orchestra; Martin Williams and friends have managed to give us the shows tunes and a lot of the authentic flavor. There has been no echo added or sound enhancement but one way in which, I presume, Martin Williams did tamper with the evidence is in excising vocal choruses from recordings of George Olsen's Orchestra, Paul Whiteman's Orchestra and the Ambassadors. I object to this only in the case of Olsen where his featured vocalist who was excised both from "I'm Bringing A Red Red Rose" and "Makin' Whoopee" was Fran Frey—an actual member of the stage cast of *Whoopee*. It is true that Frey sang neither of these songs in the show. "Makin' Whoopee" was sung by Cantor and "I'm Bringing A Red Red Rose" was sung by Paul Gregory and Francess Upton, but then why give us "I'm Bringing A Red Red Rose" recorded by Ruth Etting, who not only did not sing it in the show but gives us only the women's version of the lyric (e.g. "I'm keeping this red red rose")? Etting's performance, regardless of whether

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you enjoy her vocal stylings (I do not), does give us the benefit of some nice guitar work by jazz virtuoso Eddie Lang. I, for one, would have gladly been willing to forego one of the interpolated numbers to have the full version of Olsen's "Red Red Rose" and "Makin' Whoopee" on this LP. Also, I can't quite see the point in including a second version of "Come West Little Girl Come West" by the Ritz Quartet and a third "Makin' Whoopee" by Paul Whiteman's Orchestra. True, Whiteman's band and the Ritz Quartet did replace George Olsen's Orchestra when he left the show early on, dissatisfied with the lack of speciality numbers for his band. I'm sure he would not have been happy at the treatment they receive here. And that is my *only* complaint about the *Whoopee* LP.

I have no complaints about the music on the *Ziegfeld Follies of 1919* LP. These are top stars of Broadway doing tunes from the show or that were interpolated in the show. It is interesting to concentrate on comparing Eddie Cantor's whiz-bang fast, high-keyed sense of comedy in such numbers as "When They're Old Enough To Know Better" with Bert Williams' laconic, laid-back, underplayed delivery on "Checkers." It's the difference between white (slapstick) humor of 1919 and black humor, which relied on subtlety even at that early date. It is even more revealing to find, in 1928, the more mature and assured of his own success Eddie Cantor taking on a more laid-back attitude resembling, to a great extent, Williams' technique.

As far as sound is concerned, what can you say? The records on the Ziegfeld Follies LP go back to the acoustic era of recording and even the 1928 and 1929 Orthophonic Victors and Viva Tonal Columbias used on *Whoopee* show their sonic age. It is a tribute to Jack Towers, a master of remastering, that on *Whoopee* and I suspect also, though unbilled, on *Ziegfeld Follies of 1919*, these historic gems were reproduced so you can understand the words—which is really what this kind of entertainment is about anyway.

While Smithsonian LPs may not be commonly available in all record stores, they can be mail ordered from Smithsonian Customer Service, P.O. Box 10230, Des Moines, Iowa 50336. The records sell for $6.99 each, plus $1.49 shipping charge per order.

J.K.

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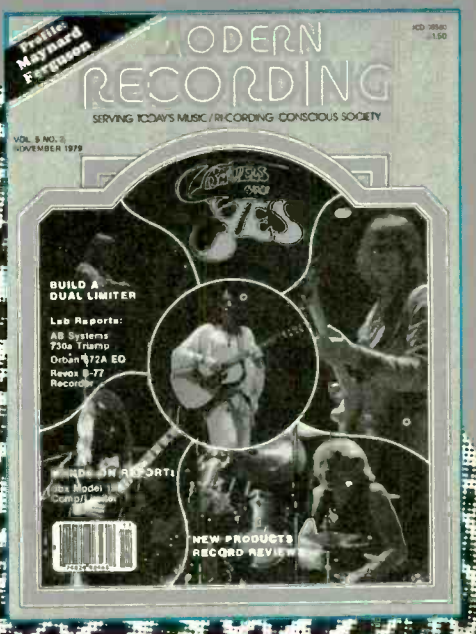

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