VOL. 2 NO. 7
JULY 1977

MODERN RECORDING

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SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

A Session with Patti Labelle

Crosby, Stills and Nash: Recording Again

An interview with their engineers—the Albert brothers

Acoustics
ct Recording

ile: Chick Corea

Reports
Products
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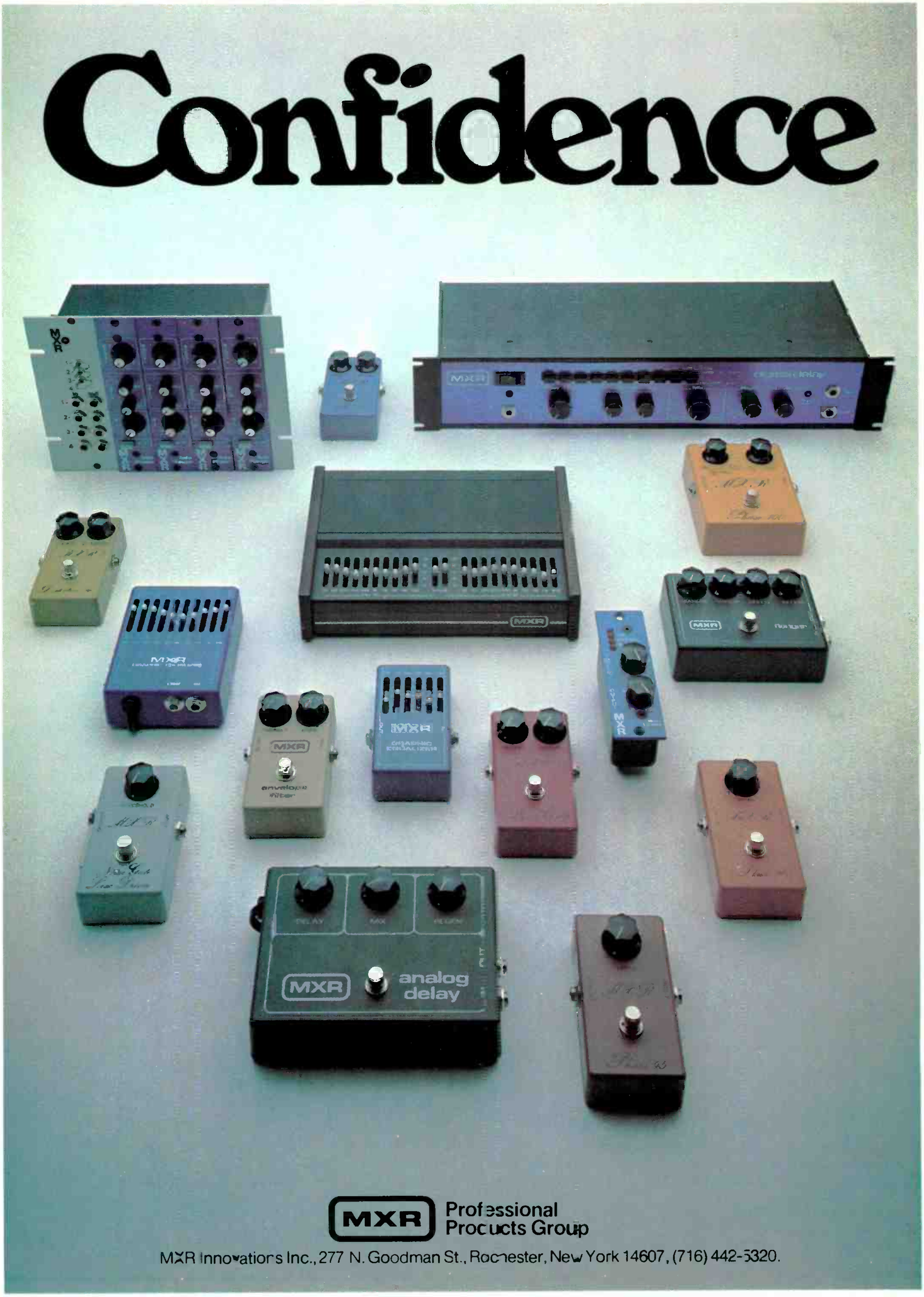
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MODERN RECORDING

JULY 1977
VOL. 2 NO. 7

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

THE FEATURES

COMING NEXT ISSUE!
Disc-mastering, Part I
Profile: Engineer Shelly Yakus

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THE STAPLES

Cover photo © 1977 Ed Perlstein

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SERVING TODAY'S MUSIC/RECORDING CONSCIOUS SOCIETY

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Letter From The Publisher

Welcome to a new industry. Although MODERN RECORDING has been around for nearly two years, for many of you this is probably the first time that you've seen a copy of MR. Commencing with this issue, we have increased our newsstand distribution by an additional 50,000 copies. So, for those of you who have just met us, we would like to tell you a little about ourselves.

The consumer area that MR serves is called "the semi-pro home recording studio market" by some and/or "the high-end audio electronic musical instrument market" by others. But, whatever individuals or manufacturers label MODERN RECORDING's audience, the effect is the same—an audience of active, creative and highly sophisticated equipment users. Consumers like yourself who have a more personal and maybe even professional interest in creating music through audio.

At first look the recording market itself might appear to be a very small and specialized interest area, but then so did photography fifteen years ago. Recording today is the most popular vehicle through which music, audio and electronics are united, explored and developed. Millions of creative musicians and performers aspiring to develop and enhance their art and talent are using recording as their vehicle of creative expression. Active audiophiles (different from the passive audio listener of the '60s) are the soundmen and home recording enthusiasts of the '70s. They are creative people who are using audio equipment just as a musician uses his instrument. So, if you see yourself in any of what you've just read, then the magazine you are holding is for you! Enjoy it, and again, welcome to a new, exciting and creative industry.

Vincent P. Testa, Publisher

The most exciting new recording combo in Music City

Willi Studer's Revox family of tape recorders and components for the audiophile has joined Willi Studer's family of professional audio equipment to form Studer Revox America, Inc., in Nashville, Tennessee USA.

This provides all Revox audiophile and institutional users the obvious benefits of factory direct sales and service enjoyed by professional studios and broadcasters.

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LETTERS TO THE EDITOR

Praise for "Live" Articles

I am a concert sound enthusiast and I have been very pleased with the recent attention you have been paying to this topic. Two impressive articles were "Position Filled by Claire Brothers," (Dec/Jan 1977) and "Billy Cobham/George Duke—On Tour in Europe," (April, 1977) because of their useful and interesting content, besides being written from a "Live" sound man's point of view.

I am looking forward to seeing future articles on other bands on the road and sound companies in your magazine. Having seen little material on rock and roll lighting elsewhere, an article on this might prove successful.

—Barry Popoff
Toronto, Canada

Building PA Speaker Enclosures

In the Dec/Jan 1977 issue (Vol.2, No.2) on page twenty-five in "The Product Scene" Audio Briefs, you mention Bozak speaker systems. Could you please send me the address of this company?

Where could I find out the correct way to build custom enclosures for PA speakers?

—Jon Poteet
Albuquerque, N.M.

Bozak, Inc. is located at 587 Connecticut Ave., South Norwalk, Conn.

As for your second question, the term "correct" is a relative thing. For each sound reinforcement company we might question, the answers would all be different because it is a personal "correctness." We suggest that you visit someone who does sound for concerts locally in Albuquerque or find out who does the sound for the concerts that are on tour nearby. Speak with the sound reinforcement people and they'll probably be able to help you out with some pointers on building your own PA speaker enclosures. For more hints, see the book, Practical Guide for Concert Sound *by Bob Heil and printed by Melco Publishing, P.O. Box 26, Marissa, Illinois 62257.*

Good Recording

Thought you might be interested in listening to a record which was recorded at home in the living room on a TEAC 3340-S. This recording was originally a demo tape. Many songs are one or two takes with as many as eight mics being mixed at once on one track. No noise reduction was used and some songs were recorded at very low VU meter levels. In spite of all this, the record still sounds pretty good. It is a good example of what a basic amateur can do with some nice equipment. I've noticed on most sound systems that the bass tone control must be turned down.

Anyway, I hope you will listen and any comments would be appreciated.

—Dave Shoffner
Steamboat Village, Colo.

Your recording sounds pretty good to us though the bass seems to be a bit overpowering. For any amateur, "when there's a will, there's a way!" If anyone is interested in how the sound is in a Colorado living room, The Steamboat Album is available by C.O.D. Mail Order from: Yampa River Records, P.O. Box 6132, Steamboat Village, Colorado 80499.

MODERN RECORDING Goes to Europe

I am a Polish musician, currently touring Scandinavia, and I own a small mobile recording studio. I would like to subscribe to *Modern Recording.* What will be the cost, including surface mail charges, to Warsaw, Poland?

—Andrzej Poniatowski
"Uncle Albin's Band"
Lillehammer, Norway

Subscription rates for areas outside the U.S. are as follows: $15.00 per year for twelve issues and $25.00 for twenty-four issues. Surface mail charges to Poland would be $2.04 per year for twelve issues ($ .17 per copy) for second class mail. If you want your copies sent via first class air mail, the cost would be $30.00 per year for twelve issues ($2.50 per copy).

The "Click and Pop" machine

only by SAE

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

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

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

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

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

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Please send more information on the 5000.

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CIRCLE 83 ON READER SERVICE CARD

Points on the "Primer"

We were quite pleased that Don Davis, one of the foremost names in the audio industry, would take time to read an article ("P.A. Primer," *Modern Recording*, Jun/Jul 1976) intended for the novice sound man. We appreciate Mr. Davis' comments (MR, Feb/Mar 1977). Following is a clarification of the points we are attempting to make.

On the matter of "power increase" of bi-amplification, Mr. Davis stated in his letter exactly what we had in mind. Mr. Davis said, "True, the voltage swing requires a 200-watt amplifier, if clipping is not to occur, to develop 50 average watts." Many sound systems are operated at (or past) the brink of power amp clipping (not to mention speaker blowing!). Consequently, any method of increasing the headroom of the system is crucial (that last batch of watts gets expensive). We were speaking strictly in terms of dynamic range, and not about any "magic" watts.

On the matter of the bass pedal tone with insufficient "oomph," we feel this problem is eliminated when the system is the proper size for the job. A number of factors, including loudspeaker efficiency and directionality, amplifier power, room size and shape, type of program material and desired sound levels must be considered if the particular P.A. job is to be a success.

The power advantages of bi-amplification become apparent in "cranked-up" situations with complex program material, *not* with a sine wave.

Mr. Davis makes a valid point when he mentioned that low frequency clipping can go undetected in a bi-amplified situation. The same point was most recently stated by John Lovda and Stephen Muchow ("Bi-amplification Power *vs.* Program Material *vs.* Crossover Frequency," *Audio* magazine, September, 1975). Since we were quite aware of this psychoacoustic phenomenon, we fully considered mentioning it. However, we felt that this would encourage novice users to overdrive their P.A. systems, thus increasing the probability of equipment malfunction. The fidelity also suffers under these conditions. This is obviously undesirable, and we very highly recommend that clipping of any sort should be avoided. It is definitely cheaper to power the system correctly than it is to replace overdriven transducers damaged by highly-clipped waveforms.

In closing, we feel a bi-amplified system generally sounds better than a comparable system with passive crossovers. Since there are a number of factors in-

Our new AD cassette takes the normal bias position to extremes.

We made a name for ourselves by creating the world's first non-chrome, "high" (CRO_2) bias/EQ cassette tape, TDK Super Avilyn (SA). The state-of-the-art tape that has quickly become the standard of reference for cassette tape performance.

Our latest innovation is called AD (ay-dee), and we predict it will soon become the standard of performance and economy in the "normal" bias/EQ position.

We produced the first high fidelity ferric oxide cassette tape some ten years ago, and we've been perfecting the formulation ever since. Our new AD delivers superior performance, especially at the critical high-frequency range (7kHz to 20kHz), where many mid-priced cassette decks and even premium-priced cassettes tend to fall off too quickly.

AD is our ultimate ferric oxide tape designed for the "normal" bias/EQ position. Overall, it provides the lowest noise, highest frequency response and widest dynamic range of any pure ferric oxide cassette tape. In 45, 60, 90 and 120 minute lengths, AD has the same super-precision cassette mechanism found in TDK SA, in a new blue-gray shell.

And AD brings its audible benefits to all cassette decks, with and without switchable bias/EQ, including those found in cars, portables and home stereo systems. So the music you love can travel with you, with all of the clear, crisp, brilliant sounds that make music so enjoyable.

AD is the finest pure ferric oxide cassette tape you can buy at any price. And it has TDK's full life time warranty. Give our new high-fidelity, moderately-priced AD a try–it's anything but normal.

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TDK®

The machine for your machine.

CIRCLE 86 ON READER SERVICE CARD

volved, we began a research project months ago to study *all* angles of bi-amplification based on different program waveforms. We hope to be publishing our findings soon.

Mssrs. Roth and Ford are contributing editors to Modern Recording *magazine.*

Say You Saw it Here First

I saw on page 15 of your April, 1977 issue that you had some information and data on studio construction.

Do you offer any books, or are back issues available? I hope to be a regular subscriber to your magazine soon.

Any information or sources of info you can recommend would be greatly appreciated.

—John Kondrk
Hopelawn, N.Y.

The items you saw in the April issue are all publications that we would recommend. Back issues of MR are available; just write to our Subscription Department and request the ones you want ($1.75 per issue).

Stevie's Dream Machine

On the recent Dick Clark special, "Fifty Years of American Bandstand," Stevie Wonder appeared in a brief pre-recorded segment from a studio. He played and sang a short tribute to American Bandstand. My question is this. What was the strange looking synthesizer (keyboard) in front of him?

As a second year (and this time for two years) subscriber, I applaud your efforts on this fine magazine.

—Tim Wilde
Dallas, Tx.

The strange keyboard synthesizer you saw Stevie playing was probably his made-to-order Yamaha Electron Olyphonic Synthesizer GX-1. Stevie has fondly named this one and only unique machine, "The Dream Machine." It also has built-in mics that Stevie sings through.

Oops! Mistaken Responsibility

Sorry! In the Dec/Jan 1977 issue of Modern Recording, *it was incorrectly stated in the article, "Position Filled by Clair Brothers" by Gil Podilinsky, that Clair Brothers were responsible for the sound reinforcement of Led Zeppelin. Clair Brothers have never worked on the reinforcement for Led Zeppelin.*

The Peavey CS Series

Last year when Peavey introduced the CS-800 Stereo Power Amp, professional sound men and engineers acclaimed it as the most versatile high performance power amp available for under $1,500.00.

Now, there are two superbly engineered additions to the Peavey CS series, the CS-200 and CS-400. These new high performance amplifiers are built with the same meticulous quality control and engineering standards that go into the CS-800.

We invite you to compare the features designed into the CS series. You'll see why no other power amp offers the value built into a Peavey.

CS-200 $324.50 *
Monaural power amplifier
- 200 Watts rms
- 20 Hz to 50 kHz response
- Less than 0.1% THD
- Less than 0.2% IMD
- LED overload indicator
- 19-inch rack mount
- Forced air cooling

CS-400 $424.50 *
Stereo power amplifier
- 200 Watts rms per channel
- 20 Hz to 50 kHz response
- Less than 0.1% THD
- Less than 0.2% IMD
- LED overload indicators
- 19-inch rack mount
- Forced air cooling

CS-800 $649.50 *
Stereo power amplifier
- 400 Watts rms per channel
- 5 Hz to 60 kHz response
- Less than .05% THD
- Less than 0.1% IMD
- LED overload indicators
- Loudspeaker protection system
- Balanced input and electronic crossover capabilities
- 19-inch rack mount
- Forced air cooling

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TALK BACK

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

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

Film Sound

I am planning to do a film soundtrack for an independent filmmaker next year and would appreciate information concerning magazines or books on the subject of film sound.

—Joe Ray
Coventry, R.I.

The best source of information pertaining to sound for film would be the professionals themselves who work in this field. Most post-production film mixing and transfer studios have personnel ready and willing to provide answers and explanations to your questions from "Where do I start?" to "How much will the final mix and optical sound track cost?" With or without the help of literature on the subject, I believe that you will still find most of your answers at such facilities.

Literature pertaining directly to the field of sound for film is quite limited. I am familiar with few current weekly or monthly publications which would help one looking for basics. The articles available are usually written for professionals and pertain to the special techniques used to solve problems peculiar to a given circumstance involving a particular production. However, one monthly publication which I do feel would be valuable to you is *American Cinematographer.* In the last several issues up to and including April 1977, Mr. Anton Wilson has been reviewing many facets of sound for film recording in the "Cinema Workshop" department of that publication. His reviews are helpful to professional and non-professional alike.

Other literature that I suggest to you for a basic understanding of "How to do it" and "What is needed" would be the following: *Audio Control Handbook* by Oringel; *Techniques of Magnetic Recording* by Tall; *Film and Its Techniques* by Spottiswoode; *The Audio Cyclopedia* by Tremaine; *American Cinematographers Manual* available through most professional motion picture equipment sales and rental houses.

—Valen Peters
Sound One Corp.
New York, N.Y.

Don't Smoke, Audio Equipment in Use

Is continual cigarette smoke harmful to recording and musical equipment?

—Gary Guthrie
Blue Goose Sound, Inc.
Fort Smith, Ark.

Recently, I had the extreme luck of receiving a handsome stereo FM tuner. The unit belonged to a relation who smokes very heavily. As a result, the unit's front plate appeared to be gold until careful cleaning uncovered the beautiful brushed aluminum panel. Also in need of cleaning were the slide selector switches and front glass tuner scale, all of which may not have been needed had the environment been cleaner.

In my days of television repair, it was quite evident that homes with smokers needed more frequent cleanings of their tuners and picture tube screens. The picture tube's high voltage and static attraction to all kinds of dust, especially before the bonding of glass directly to the tube, required taking the TV apart to clean. The build-up of nicotine was very evidently the cause of the problem. The tuner on the other hand, showed evidence of more oxidation which is also attributed to the smoky environment.

Modern recording studios use very highly sophisticated equipment with thousands of switch closures and contacts which are constantly exposed to the studio's environment. If this environment is continually smoky, the incidences of equipment failure increases. Thus, the frequency of required maintenance and routine cleaning of these contacts and faders will rise.

In conclusion, smoking and audio recording equipment should not be mixed. Otherwise, serious considerations of installing air conditioning or purifier filters to help eliminate the detrimental smoke in the environment should be made.

—Steve Friedman
Engineer
Robins Broadcast
and Sound Equipment Corp.
Commack, N.Y.

When to Use Delay

When recording an instrument using a delay effect, is it best to record the instrument "live," overdubbed, or to use studio outboard equipment to introduce the delay during mixdown?

—Harvey Schmidt
Little Silver, N.J.

The normal approach that I use and one I feel most engineers use, is to introduce the effect at some point during the mixdown process. If you introduce an effect of this type during the original recording session and the effect along with the original sound source is combined to one track at that time, it presents one or two problems. First of all, should the producer, the artist or yourself decide at a later time you did not like the effect, either in terms of amplitude or ratio of the effect to the original sound, it would be very difficult if not impossible to change it. This is analogous to not being able to remove distortion once it has been recorded on tape. Also, if you choose to add the effect during the original multi-track session, special care must be taken to see that the instrument to which the effect is to be added is very well isolated. Otherwise, any leakage to that instrument's mic will cause the effect to be added to the other instrument.

If it is desired on the part of the producer, yourself or the talent to add the effects during the original session, I recommend that you utilize one of the other tracks on the multi-track machine and record the effect separately from everything else. In that way you have the option of changing it later without re-recording the original instrument. If it is difficult to isolate the instrument in the original multi-track session, then I recommend overdubbing. This will give you the necessary isolation in order for you to treat the instrument as a separate entity when adding these delay effects. I still recommend, however, that unless you are certain of the desired effects, that it be added during the mixdown process, as opposed to the original session. The whole concept of multi-track recording is to allow the flexibility of changing balance, equalization, reverberation content, etc., at some time after the original session.

One final thought: often monitoring during an original multi-track session is at best a reference mix and not indicative of the final balance, equalization or content of the product. Therefore, deciding on the exact effect that you wish may be more difficult at that point in time.

I feel the above ideas and suggestions are relative not only to the use of an effect such as delay but are also true for phasing, flanging, reverberation and any other special effect you wish to use in the final product.

—Skip Frazee
Manager
Sound Techniques, Inc.
Dallas, Tx.

TEAC and Tape Headroom

What kinds of tapes, compatible with the TEAC's bias and EQ settings have the greatest headroom (remanence) and best signal-to-noise ratio?

How much better could those specs be by readjusting the deck for a different type of tape?

And finally, with the maximum output of the limiter being only +18 dBm, is there ample level to fill the extra headroom I would have by readjustment, considering the 10 dB extra headroom allowed by the dbx?

—Stephen Kayser
Stamford, Conn.

There are a number of tapes on the market now that provide an abundance of headroom and an excellent S/N ratio when used with the TEAC A-3340S. Among them are Maxell UD-35, Ampex 456, TDK-Audua, and Scotch Classic, to name a few.

No significant improvement in TEAC's specifications will be realized by calibrating the unit for one certain brand, as all of the tapes mentioned above have approximately the same bias levels and are within 2 dB of each other at the most with respect to output levels, frequency response, and S/N ratio.

The additional headroom created by using good tapes, and other factors is not cumulative. For example: 3 dB of headroom on the tape and 10 dB additional headroom using dbx will not give 13 dB of total headroom. The 3 dB of additional headroom on the tape can be taken advantage of by increasing the recording level of the deck. This will give an increased overall S/N ratio of 2 to 3 dB.

But, as far as this S/N improvement is concerned, no noticeable difference will be heard when using the deck with dbx. dbx presents an improvement of S/N of 30 dB—the final result on the TEAC being approximately 85 dB S/N! At this point, any small adjustments in recording level or S/N ratio just do not make any practical difference.

While you can increase your recording level 3 dB, we recommend no further adjustment and do not deem this adjustment necessary. Any further headroom can be left to insure against distortion during transient peaks. In either case, the +18 dBm of your limiter is plenty of output to operate the dbx, and your set-up should produce excellent results "as is" when following the instructions provided with all three units.

—Tom Spurney
Technical Correspondent
TEAC Corporation of America
Montebello, Ca.

Blinded by the Light

Peak level meters and other indicators employing LEDs suffer when used on an outdoor gig on a sunny day. You can't see if the lights are on or off due to the sunlight intensity. Even a temporary tarp, constructed for just that problem,

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

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

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

371,293 ways to hear better sound.

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

Get better performance from your components and listening room.

Why do you need such tremendous variations in tone? Quite simply, they help you to overcome the shortcomings of the acoustics in your listening room; they also can help you to compensate for the deficiencies in old or poor recordings.

Finally, they can do wonders for the frequency response of your speakers, and where you place them.

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

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

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

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

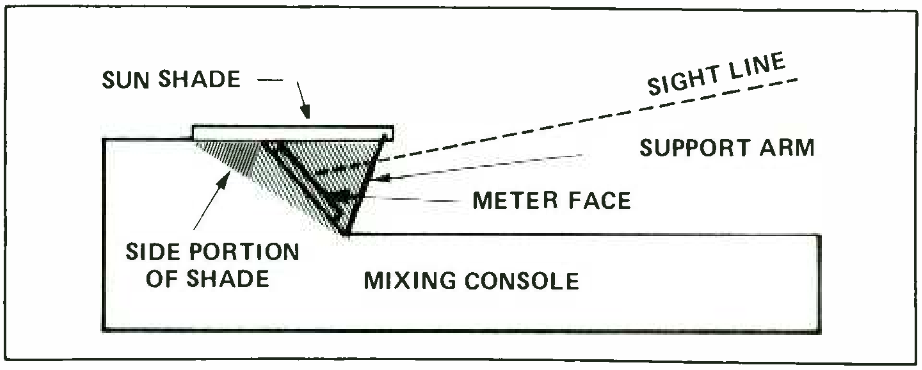
While most manufacturers reserve unique features for their top of the line model, JVC has included SEA in three of its receivers. The S300, the S400, and, of course, the top professional—the S600.

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

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

JVC

CIRCLE 53 ON READER SERVICE CARD



will not block reflected light enough in most cases. What to do?

—T. Young
NE Sound Services
Thomaston, Ct.

To solve this problem, we normally construct a sun shade visor of dull, black material and mount it as closely as possible across the top and down the sides of the meter face (see drawing). The tighter this is done, the better the ambient light will be kept out.

This shade, when combined with your overhead tarp, will normally cure problems caused by sunlight intensity. Should the LED indicators be mounted on the horizontal plane, the only cure would be relocating the indicators.

—Paul F. Bergetz
BSC
River Grove, Il.

How Does the Thing Work?

How do wireless mics work?

—Jeanne Malone
Valley Stream, N.Y.

Wireless microphones use a radio transmitter and receiver in place of the microphone cord. The microphone is connected to a small VHF or UHF transmitter and is carried by the performer. This enables more mobility, no tripping over the mic cord, and freedom from accidental hot ground shocks. The receiver can be located up to one-quarter of a mile away and uses an antenna to pick up the signal from the mic transmitter. The output of the receiver connects to the mic or line input of the audio mixer, mic amp, guitar amp, etc. More information can be obtained by writing for Vega's Technical Applications Bulletin No. 1 at Vega Div. of Cetec Corp., 9900 Baldwin Place, El Monte, Ca. 91731.

—Tim O'Hara
Director of Engineering
Vega Div. of Cetec Corp.
El Monte, Ca.

Dolby A and B Systems

What is the difference between the Dolby A and the Dolby B systems?

—John Blue
Wimberly, Tx.

Compression-expansion systems for audio noise reduction normally suffers from two side effects. Noise modulation ("breathing") is usually most audible on simple program material such as a quiet piano or solo flute, and is most apparent in equipment which treats the whole audio spectrum as one signal

("full-band companders"). Secondly, transient distortion can result from the compressor failing to follow rapid changes in input level and passing short-lived peaks which overload the noisy medium. If the response time is faster, audible modulation distortion may result.

The professional Dolby A-type system eliminates noise modulation by splitting the audio spectrum into four bands, with complementary compression and expansion operating independently in each band. The transient distortion problem is solved by a choice of parameters such that at critical high levels, where overshoots would cause distortion, the Dolby compressor behaves as a linear unity-gain amplifier, plus the use of attack and recovery times which adapt to the requirements of the program.

Dolby A-type noise reduction is used by the professional music studio, and all the equipment is manufactured by Dolby Laboratories, Inc.

The Dolby B-type system is a simplified system, eliminating noise modulation and distortion by techniques very similar to those in the A-type, and designed primarily for consumer applications in which the obtrusive noise is mainly high frequency hiss, e.g. low-speed tape and stereo FM broadcasting, as opposed to 15 ips recording where print-through and other mid-frequency noises are also audible. The B-type consumer products are made under license by over sixty companies throughout the world.

—Ken Grundy,
Senior Engineer
Dolby Laboratories Inc.
San Francisco, Ca.

Analog and Digital Delay

What is the difference between an analog delay unit and digital delay?

—Richard Hafner
Beverly Hills, Ca.

Delay is delay is delay is delay is delay is . . . How the delay is achieved is a different question. A couple of years ago digital delay devices began to appear on the market claiming to do all those things requiring delay that we'd worked so hard to achieve, only easier and more flexibly. The originals worked but the second generation of devices worked much better.

The digital devices must first convert the audio signal (an analog or continuously varying signal) to a string of pulsed DC voltages going from some low value to some high value and back again.

"...The Sansui TU-9900 qualifies as a true 'super-tuner'..." Stereo Review, March 1977

" 'Super' FM tuners are usually priced from $1000 up. Sansui's new model TU-9900 tuner, at (under) $450,* matches their performance..., at least in the most important respects".

Popular Electronics, January 1977

"...The TU-9900 is highly selective.....In the wide-band mode, the IHF sensitivity was...1.7uV in mono...indicative of very high performance.....S/N at 65dBf (1000uV) input was one of the best we have measured.....The distortion at 65dBf was also lower than we have ever measured on an FM tuner.....Mid-range separation was not only the greatest we have ever measured, but far exceeded the guaranteed rating of our Sound Technology 1000A FM generator.....Overall, we would consider the TU-9900 to rank with the other 'state of the art' FM tuners...." Stereo Review

"The Model TU-9900...is an ideal mate for the highest quality amplifiers and speaker systems.....Image rejection was unmeasurable, exceeding the 100dB range of our test equipment.... Stereo channel separation was almost as unbelievable as the distortion figures, exceeding 60dB from 60 to 600 Hz.....Clearly, the Sansui Model TU-9900 tuner is a very superior performer...[and] any untoward sounds heard via this tuner originate from the FM station.....It's a top value unit." Popular Electronics

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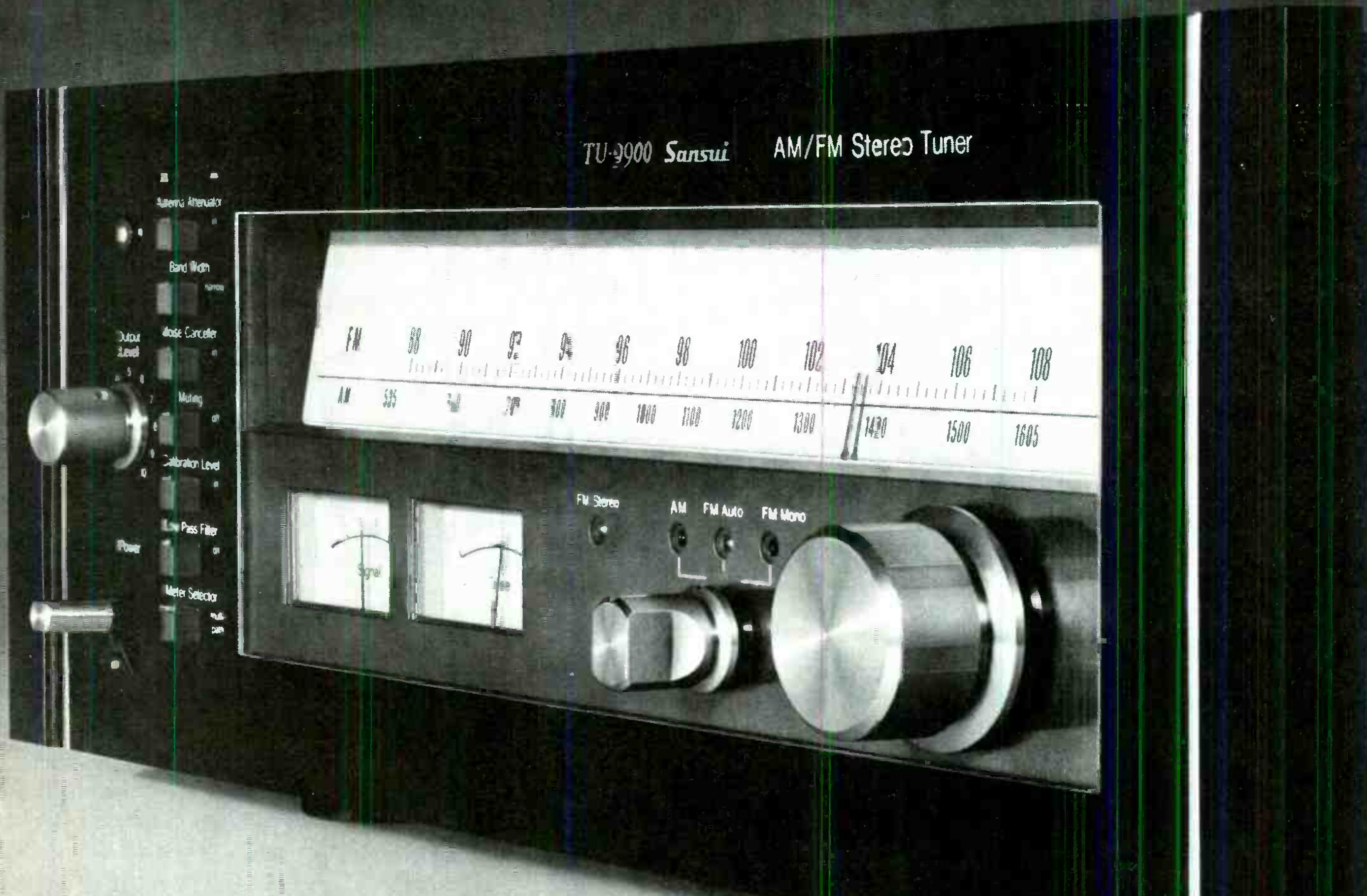

Sansui

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In many systems this is 0 volts to +5 volts. The digital signal only exists in the low or high state. This is usually equivalent to zero or 1 in a binary number system.

These pulse trains (coded digital information) are relatively easy to delay once you get them by simply passing them through a number of devices each of which takes a certain amount of time to flip from the "zero" to the "one" state. Add a switch or two to control how many of the delayers your signal will go through and you have a set of discreet delays that your signal will undergo. Then these pulse trains must be reconverted to audio again.

Two points should be stressed. First, the double conversion tends to produce much error (distortion) in the audio signal. This is true for the devices currently on the market. Secondly, only distinct steps of time delay are available. While these are on the order of one to two one-thousandths of a second, that still isn't a short enough time for some applications like flanging. More importantly, it's not variable except by increments of whatever the shortest delay is.

Now enter the analog delay units. These are more recent devices which utilize a different technology to achieve the delays. No conversion is required. The audio itself is stored in little pieces of charge which take a certain amount of time to move in and out of the devices. The result is delay without conversion. There are other types of distortion to watch out for.

Generally, these steps of delay are smaller, but more importantly, they can be hurried or slowed more conveniently so you don't get the stepping effect. These devices make the best flangers around. If you want long delays of large fractions of a second or longer, the digital devices still represent a better choice.

Let me stress, that with electronic devices relevant to both areas, a great deal of work is being done so it should be possible in the not too distant future to see newer generations of both digital and analog delay devices. By the way, all the phasing, flanging, slap, slap-echo-voice or instrument multiplication, comb filtering, and some Star Trek sounds, are time delay generated and so are phase errors.

—Ed Rehm
The Nordine Group
Chicago, Il.

Crossover Capacitors and Their Purpose

Would you please explain to me the process (formula) by which a protective capacitor's values are chosen when used on a HF horn to prevent over-driving and blowing out the diaphragm? Also what effect does it have on frequency response, etc.?

—Thomas Mifflin
Thomaston, Ct.

The capacitor you are speaking of is called a crossover capacitor. Its function is to attenuate the audio signals below a given frequency for the tweeter. These lower frequencies are unwanted at the tweeter input because of its design. Frequencies below its operating range could damage the voice coil and diaphragm due to its very limited excursion. It is designed this way to make it efficient and to minimize distortion within its operating range, but the crossover capacitor will still not protect it from over power. This is generally done with a fuse or other current limiting device.

Basically, all the crossover capacitor does is determine the frequency at which the tweeter starts to operate. Its value is determined by the desired crossover frequency, the impedance of the

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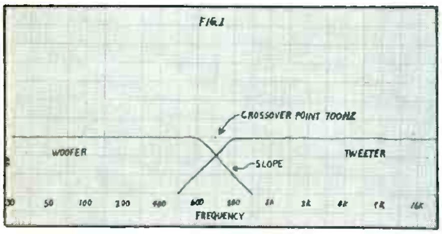
tweeter voice coil and the cutoff rate desired. You should find a crossover frequency that the woofer and the tweeter can reproduce properly, so the total frequency of the system can remain smooth and reproduce sound with little distortion. In most professional loudspeaker systems, this frequency is in the 500 to 800 Hz region where the woofer stops operating and the tweeter begins.

The higher frequencies going to the woofer have to be attenuated to keep distortion to a minimum. This is done



with a large coil or choke that allows only those frequencies below the tweeter crossover point to pass. What happens is the woofer reproduces only those frequencies below the crossover point and the tweeter reproduces only those frequencies above. The response of the crossover system should look like Figure 1.

The easiest way I have found to determine the crossover capacitor's value is through the use of the nomograph presented in the Howard Sams Publication No. 20520, "How To Build Speaker Enclosures" by Alexis Badmaieff and Don Davis on page 128 Figure 7-13. It carries a wealth of information on designing and building loudspeaker enclosures and how to make them work properly.

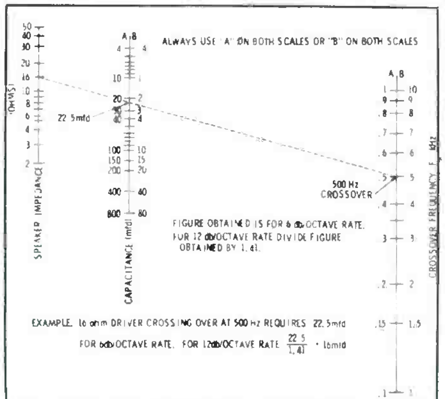


Fig. 7-13. Nomograph for determing proper capacitance value for crossover networks.

—Roland J. Solomon
Nashville Studio Systems
Nashville, Tn.

Real Advantages

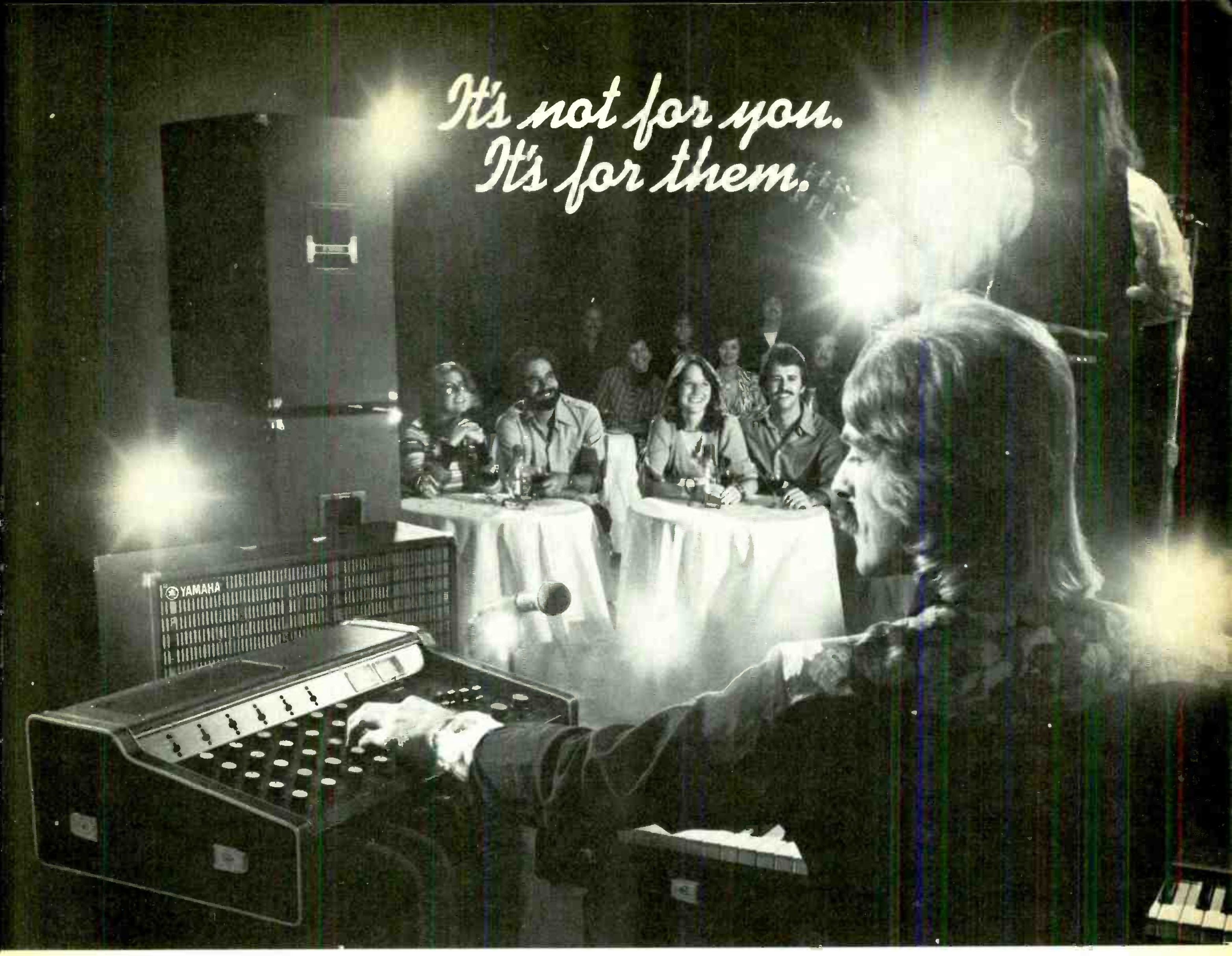
How do the professional Ampex and Scully tape machines differ from the TEAC or Tascam models? What are the real advantages in owning a Scully or Ampex for home and amateur use?

—Hugh West
Hurley, N.Y.

Let's broaden the question since Ampex and Scully are not the only alternatives in the professional field. Don't neglect AEG-Telefunken (the inventors of the AC biased recorder), Studer, or MCI to name a few. Since I have never been intimately acquainted with either the TEAC or Tascam, again, let me broaden my reply and make the comparison with the genre: expensive home recorders.

Generally speaking, the basic difference between the two genres is not to be found in the specifications but in the construction. The professional machine is much more rugged and is usually built to withstand continuous day-in/day-out use by individuals who do not own it (and therefore, do not cherish it). There is no reason for a home recordist to pay for this kind of reliability unless he does "live," irreplaceable recordings.

Home machines are capstan driven,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and if they have 3 motors, the torque of the take-up and supply motors is adjusted for the proper tension in the middle of the reel. When 10½-inch reels are used, this can lead to a difference in pitch of one semitone from the beginning to the end of the reel. Pro machines such as the Telefunken and Studer utilize sophisticated servo systems to keep the tension constant and thereby prevent the drift. Since most home users utilize seven-inch reels, this is not a dramatic problem. (They would rarely have to splice beginnings and ends of reels.)

Noise is one of the big problems that pro tape recorders must surmount because of the multiple generations required from studio to disc. Each generation adds noise, so the pro must have the lowest possible noise on his master tape. The home recordist does not have this problem for he is either copying from a radio program, record or catching his son's or daughter's piano performance. The first generation will be the last. Since the average living room has an ambient noise level of about 40 dB, a good home recorder with its 60 dB s/n will make most home recordists happy. For that little extra, they can always use the Dolby B system.

To sum up, the professional recorder is a must where reliable performance is absolutely necessary. The loss of one recording session can cost the professional the price of his recorder. The home recordist is not under this kind of financial pressure to get the take right in every detail.

If he has the extra money, he should put it into professional microphones where the difference in sound between such pro microphones and the usual microphones used by the home recordist is substantial.

—Eli Passin, Vice President
Gotham Audio Corp.
New York, N.Y.

An End to Buzzing

I have two questions that would definitely help my recording of live concerts. When taping remote concerts "live" with long lengths of mic cable, how do you prevent buzzing? Will I have to use a preamp at the mic end to drive the signal down the long length of cable?

In order to limit distortion, what kind of mics should be used for singers who scream into the microphones?

—Thomas N. Traks
Mineola, N.Y.

There are several important considera-

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tions to be made in attempting to minimize noise pickup on long microphone cables. The most important consideration is the preservation of balanced lines throughout the system. Balanced lines afford good protection from interference because noise is usually picked up in equal amounts on both of the signal-carrying conductors. This "common mode" noise signal cancels out at the microphone preamp input transformer or in a high-performance differential preamplifier. Of course, this is the result in an ideal case. Factors such as cable imperfections, connectors, and the source of interference can conspire to produce audible noise at the console.

Aside from using all balanced mic lines, you should be sure the cable you use has a good "shield coverage" as specified by the cable manufacturer. For remote use, only rubber-covered or highest-grade vinyl insulated cable, with a braided copper shield should be used. Such cable has a long life and will remain flexible in temperature extremes. Connectors should be kept clean, as contamination on the ground pin will affect the performance of the shielding.

Cable routing is also very important. Make sure that all mic lines are kept at least five feet from any stage lighting equipment or wiring, because many solid-state dimmers produce extreme interference, known as "SCR hash." Mic cables should cross lighting and power cables at right angles, to minimize noise pickup. In extreme cases, involving long runs over five hundred feet, preamps may be necessary at the microphone.

In answer to your second question, the nature of distortion you're encountering will tell you a lot about its source. If the vocal in question has a "hard" distortion, or is very abrupt, then possibly the mic is not at fault. The signal level from a microphone can be very high, depending on the sound source, and loud vocalists can produce a level high enough to overload the mic preamp in the console, in which case you need an attenuator or pad, on that input (try a 10 dB pad). To keep a singer from "swallowing" the mic, you can use a foam windscreen which will reduce blasting and popping as well. Keep in mind that in general, even the loudest vocalist cannot overload the microphone itself.

—Fred Ehrhardt
Technical Director
Fedco Audio Labs
Providence, R.I.

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CIRCLE 63 ON READER SERVICE CARD

THE PRODUCT SCENE

By Norman Eisenberg

TANDBERG ADDS TO LINE OPEN-REEL

A new Tandberg reel-to-reel tape recorder is the model 10X, joining the 10XD and the 3500X now on the market. The Tandberg 10X utilizes Cross-Field bias and is claimed by the manufacturer to offer quieter recording than most conventional recorders with Dolby. A 10½-inch reel model, the 10X has three speeds (15, 7½ and 3¾ ips), three motors, three heads plus the separate bias head. Speed regulation, selection and operating functions all are electronically controlled. The deck has balanced mic inputs, and mixing in stereo is possible. Included are facilities for echo, sound-on-sound, editing, A and B tests, peak-reading meters, photoelectric stop and more. Optional accessories include a pitch control, remote control, 19-inch rack mount kit, various carrying cases. Price is $1,099.

CIRCLE 18 ON READER SERVICE CARD

NEW APPROACH TO BELT-DRIVE TURNTABLE

From Elac, the West German manufacturer, comes news of its model PC-830 turntable, the first from this company that uses belt-drive. Unlike conventional belt-drive turntables, the PC-830 is said to

come up to speed almost instantly, by the use of an idler wheel which also indexes the arm over the lead-in groove of the disc. As soon as the arm descends and the stylus contacts the record, the idler disengages so that the platter runs only by belt-drive.

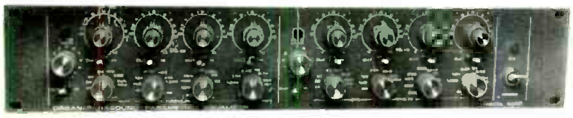
Operation is controlled by two pushbuttons for either single-play or multi-play mode. Speed monitoring is facilitated by an illuminated strobe. The platter weighs four pounds.

The arm supplied with the unit is a new low-mass tubular design said to be able to track as low as 1/2 gram. Antiskating is included, as well as silicon-damped cueing control. The PC-830 is supplied with manual and automatic spindles plus a low-profile base and dust-cover. A two-speed model (33 and 45 rpm), it is priced at $189.95.

CIRCLE 5 ON READER SERVICE CARD

NEW PARAMETRIC EQUALIZER

From Orban/Parasound comes word of a second generation parametric equalizer, its model 622. Encompassing all of the features of the model 621 series, the new 622 offers in/out switches on each of its four bands, standard balanced input with output



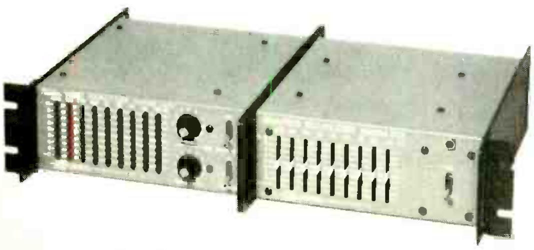
transformer option, extensive RF shielding, and a power supply capable of handling 115/230 volts 50-60 Hz AC. The manufacturer states that its "new proprietary parametric bandpass filter is virtually immune to the effects of control wear, and complements our unique 'constant Q' design by permitting -40 dB notches to be consistently obtained." THD is rated as less than 0.025% and response is given as 20 to 20,000 Hz at +18 dBm.

As in the previous 621, the 622 includes four cascaded sections, each with non-interacting, continuously variable center frequency, bandwidth and amount of boost or cut. Each section tunes over a 25:1 frequency range, with broadly overlapping coverage for maximum flexibility. An overload light monitors all potential overload points in the circuit, and overload can easily be corrected by use of the integral gain control. The 622 will be offered in single and dual-channel configurations on a 3½-inch by 19-inch rack mount.

CIRCLE 8 ON READER SERVICE CARD

LOW COST GRAPHIC EQUALIZER

From Neptune Electronics of Portland, Ore. comes word of its model 910 Graphic Equalizer, a monophonic unit offering ± 15 dB variation on nine octave bands. Two units may be attached side by side to serve as a stereo equalizer. The 910 is compatible with Neptune's model 909 Real Time Analyzer and may be attached to the model 909 to form a standard (19-inch by 3½-inch) rack unit. Solid walnut end panels also are optionally available. The model 910 lists for $149.



CIRCLE 10 ON READER SERVICE CARD

NEW ELECTRET CONDENSER MICROPHONE

The new model 1776 microphone from Electro-Voice Inc., Buchanan, Mich. is described as a ruggedly-built, cardioid-pattern condenser type suited for "live" performance sound reinforcement and superior-quality recording. It may be hand-held or stand-mounted, and is claimed to be capable of withstanding the most severe handling and use. The electret has a permanently charged element and so the mic needs no "bulky and troublesome electronics" for powering. Frequency response is given as 60 Hz to 18 kHz. Transient response is claimed to be excellent, and output "exceptionally high." The mic is a single-D cardioid with good off-axis rejection; close-up use will emphasize bass tones. It also is said to have excellent gain-before-feedback characteristics, making it good for difficult sound-reinforcement jobs. Furnished with a 15-foot cable and stand-clamp, the EV 1776 is priced at $99; with a 25-foot cable with 3-pin connectors on both ends, as model 1776P, it costs $105.

CIRCLE 19 ON READER SERVICE CARD

3M ANNOUNCES NEW TAPE

Designed for use on cassette recorders that have a ferrichrome (FeCr) switch position is "Scotch" Master III, an improved tape whose dual-layer oxide construction is said to provide a 3-dB improvement in output at low frequencies, and a 2-dB boost in high frequencies over existing chromium-dioxide and ferric-oxide tapes. At the same time a more durable tape binder system resists scratches and oxide rub-off, thereby contributing to a uniform level of output and a minimum of dropouts.

According to 3M, the new tape also can be used with switch positions in "normal" or "high" if an accentuated high-end response is desired. The Master III tape comes in a special shell designed for superior mechanical performance when used in a three-head cassette model. It features 3M's "water

wheel" flanged roller guides that permit easy observation of tape motion. Master III is available in 45, 60 and 90 minute lengths with prices respectively $3.69, $3.99 and $4.99 in album boxes.

CIRCLE 1 ON READER SERVICE CARD

BROADCAST/DISCO MIXER

Described as a practical, flexible mixing console of high performance designed for disco, production and broadcast use is the new model 421 from Sound Workshop, Roslyn, N.Y. Inputs include two stereo phono (magnetic; RIAA compensated); two stereo tape (or other high level); and one microphone (high or low impedance). Any or all of the stereo inputs may be assigned to the cue bus (active summing). A monitor select allows either the cue bus or the program bus to be fed to headphones or to an external monitor amplifier. Line jack patch points permit access for system EQ or other effects devices (such as reverb, delay, etc.) while maintaining line-drive capabilities.

The mic input has low-end EQ. For VCA controlled "talkover," the music level can be dropped by as much as 20 dB when the microphone is punched in. The device also includes a switchable rumble filter. LED readout provides indication of average and of peak output levels. The program output utilizes a "highly linear" booster which is said to be stable into any value of capacitive load and which will provide +20 dBm into 600 ohms or greater, and +26 dBm into 300 ohms. The Model 421 is priced at $500.

CIRCLE 7 ON READER SERVICE CARD

FOUR HEADPHONES CONTROL BOX

Westlake Audio has introduced its model 1200 Headphone Mult Box, a compact device that provides source selection (off, stereo, cue 1, cue 2) and level control for four separate headphones that may be jacked into it. Both phone and XLR connectors are provided. The HPM-1200 utilizes total printed-circuit construction and comes with an output multing connector for easy system expansion, and two-watt current-limiting resistors for protection of voltage sources. The unit, which measures 4½ by 2½ by 7½ inches and weighs under two pounds, can sit on its rubber feet on any flat surface, or be snapped onto a mic or music stand with optional clamps, or be wall-mounted using screw-slots on its rear. Price is $189.

CIRCLE 20 ON READER SERVICE CARD

PORTABLE CASSETTE HAS MIXING

A new top-of-the-line portable cassette recorder from Superscope—the model CR-3520—permits input mixing and has its own built-in speakers with separate bass and treble controls. A three-way VU meter provides readout of recording level, signal strength and battery strength. Also featured are: a tape selector for normal or chrome cassettes; a tape-speed control; built-in condenser microphone plus input for external mic. The unit also has a built-in FM and AM tuner which may be heard directly through the set as well as used as a source for recording (and mixing, if desired, with external vocal or instrumental inputs). The recorder also has a "cue and review" feature. Power for the CR-3520 may be batteries or AC or, via an adapter, vehicular power systems.

CIRCLE 14 ON READER SERVICE CARD

TURNTABLE ELIMINATES PICKUP LEADS

Instead of lead-out wires from tone arm to pillar, mercury-gold contacts make the signal connection and are claimed to remove one of the major problems of light tracking in a new turntable from the British firm, Environmental Sound. The turntable, model EST-4, is a direct-drive model featuring a slim base (just over an inch in depth), and a pickup arm made for it by Keith Monks Audio Ltd. The combination is claimed to "accept the top cartridges currently available and produce the best results from them."

CIRCLE 2 ON READER SERVICE CARD

MONITOR LOUDSPEAKERS— REVISITED

If the term and concept of a "monitor loudspeaker" implies a speaker that reveals more of the program material than some other model, then it may well be that there is no single speaker system as such that qualifies for "monitor" performance. And yet, by the same token, it is equally possible that a great many "ordinary" speaker systems do qualify as "monitors."

The key to this apparent contradiction lies in a technique for reproducing sound that has been, in one way or another, with us for several years but

which has been rather eloquently (from a music-listening standpoint) reestablished in the form of a new product which I have been fortunate enough to have had a chance to experiment with in recent weeks. The technique is the use of a sub-woofer via biamplification; the particular product that has started me off on this happens to be the Janis W-1 and its associated electronic crossover device, model B4SL-C. The general idea behind the Janis is to add it to an existing sound system via any of several interface options that include an individual W-1 for each speaker system, or a simpler and more economic (but surprisingly effective) hookup in which one W-1 can be connected so as to provide a "summed" low-frequency output in an existing stereo setup.

My acquaintance with, and listening to, just such a setup took place recently in the amply proportioned listening room of a local audio shop, The Sounds of Music, Lenox, Mass. The owner, Ross Tane, is a veteran sound engineer whose years of experience probing the innards of audio gear for repair and/or redesign plus "equal time" setting up some very fancy multichannel rigs for neighboring sound buffs, have bred a combined attitude of technical insight and a healthy skepticism for extravagant performance claims.

Ross and I went through the chores of unpacking the huge W-1 (a furniture-finished enclosure weighing 90 pounds), patching in the electronic crossover, adding the requisite power amplifier (we used the Dynaco Mark VI, test-reported for this issue of MR), balancing out the system, and then determining the optimum location for the W-1 which we finally placed about midway between the existing stereo speakers. The system utilized the Pioneer RT-2044 open-reel deck test-reported in our June issue, a Dynaco preamp and Stereo-70 basic amp feeding a pair of Celestion 66 full-range speakers. All during the work we made wisecracks about "How much sweat can an audiophile spare?" and "All this just to listen to music," and "Is this work really worth it or are we being parties to our own ripoff?"

The answers came when we turned on the tapes and stepped back to listen. The first thing I heard was a fullness and clarity of sound that I had not previously heard from either the tapes or the other equipment in the system. Ross must have been "hearing things" too because he just stood there with his mouth open. He finally said something like, "My God, what sound!" As we continued to listen we became aware of a sense of spaciousness combined with a revelation of inner musical detail and instrumental texture that we later agreed we never had heard before from two-channel stereo recordings. It wasn't only the bass that was better, the *entire* audio spectrum sounded more realistic.

It was hard tearing away from this glorious sound, but we finally reconnected the W-1 to other high-grade speakers and experienced similar results. We also confirmed our own exhilarated reaction by trying the setup on others, including one player from the Boston Symphony who started out by saying he didn't care about "reproduction quality, only the musical interpretation" and who also ended up listening in rapt amazement.

I think perhaps the idea of a "monitor loudspeaker" has finally taken on something tangible. I also think that a lot of sound enthusiasts may be going in for biamplification in the near future.

MUSICAL NEWSICALS

AMPLIFIERS... "Movin" amplifiers are a new line of semi-professional instrument amplifiers from Earth Sound Research Corp. (7 Grand Boulevard North, Brentwood, N.Y. 11717). "Movin" amps are moderate-power units delivering between 12 and 25 watts RMS at 5% THD (the customary total harmonic distortion level for guitar amplifiers). The amps feature tremelo and reverb and are said to use several new circuit techniques. Five different models are available, using either 10-inch or 12-inch speakers in the guitar models or a specially designed 15-inch speaker in the bass guitar version. A 52-page catalog of the entire line of Earth Amplifiers, including the "Movin" series, is available from the manufacturer for $.50.

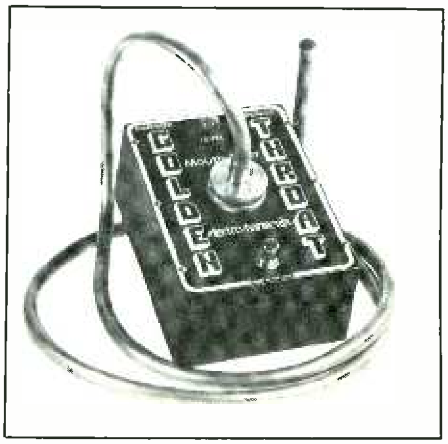
A self-contained, full-range amplifier/speaker system is a new addition to the Stage Amplifier line distributed by Unicord, Inc. (Westbury, N.Y. 11590). This new model was designed especially for keyboard amplification, but it

should be usable in some guitar or P.A. amplification applications. The system is powered by a 70-watt RMS solid-state amplifier with four input channels for mixing the outputs of up to four instruments or high impedance mics. Each input has controls for volume, reverb in/out, and bass and treble equalization, and a master section includes master volume and reverb level controls. The corner frequencies of the equalizer circuits were calculated to yield maximum effectiveness on keyboard instruments. The speaker compliment of the new system includes a special, extended-range 15-inch speaker in a horn-loaded bass reflex enclosure and a piezo-elec-

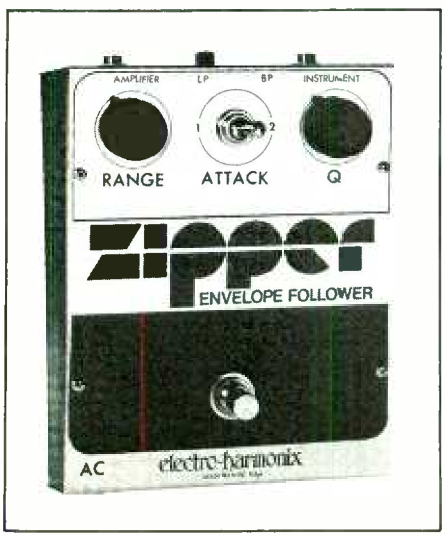
tric high-frequency horn. Usable frequency response is specified as 30 Hz to 18 kHz.

ACCESSORIES... Helpinstill Designs (5808 S. Rice Avenue, Houston, Tx. 77081) has announced an important new accessory for their line of musical instrument pickups. The Model 88 Mixer/Preamp ($285.00) is a low-noise FET preamp designed for use with Helpinstill's three-pickup Piano Sensors (such as the Model 75 upright piano pickup or the Model 110 grand piano pickup) in place of the passive mixer boxes furnished with the Sensor. Using the Model 88 Mixer/Preamp rather than the passive mixer gives the musician bass and treble equalization on each input in addition to the three individual level controls and master volume control, plus it also has a lighted VU meter and a choice of high-level outputs. The low-frequency and high-frequency equalizers in each input channel all have a ±15 dB range, and the corner frequencies of the filters have been designed to compliment the different frequency range being fed to each input. The VU meter indicates the output level from the Model 88, and is useful in precisely setting the balance of the three pickup units and for setting the output level for minimum noise and maximum undistorted output. Three high level outputs are provided, two unbalanced and one balanced. The balanced low impedance output is unaffected by the master volume control and is designed to feed a P.A. or recording system directly via a 3-pin XLR connector. The two unbalanced outputs are controlled by the master volume pot and are connected via ¼-inch phone plugs; one of these outputs is normal line level for feeding the input of an instrument amplifier or keyboard mixer, while the other is a special higher-level output designed to drive a power amplifier directly. The Model 88 is AC-powered, but to minimize the possibility of hum pickup from an AC field in the vicinity of the pickup or preamp, the power supply is remotely located and connected to the preamp via a low-voltage cable.

Electro-Harmonix (27 West 23rd Street, New York, N.Y. 10010) has an-



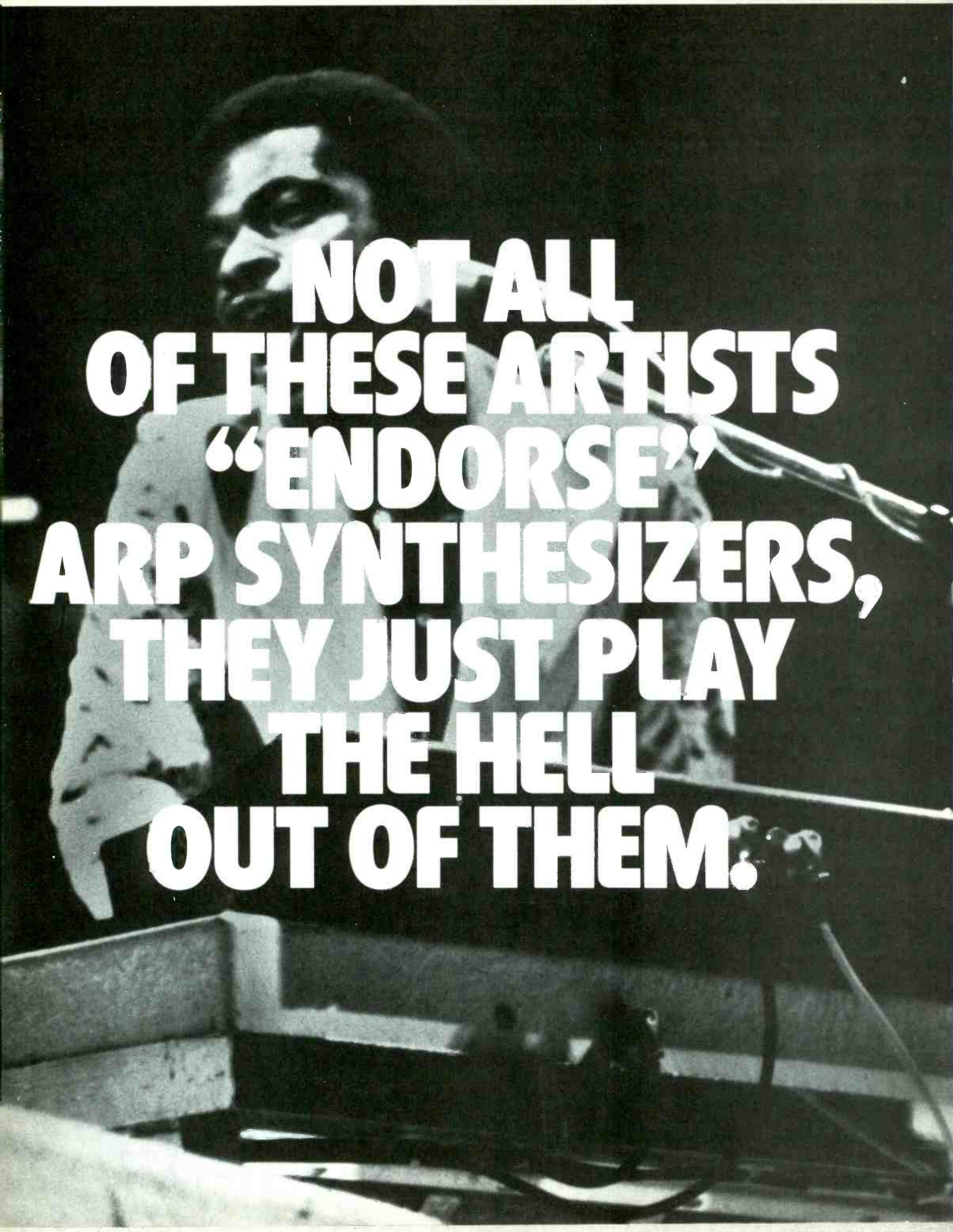
nounced the addition of two new products to their already extensive line of sound modification accessories. The first of these is the Golden Throat, which is a "talk box" unit. In this type of unit, the output of an instrument amplifier is fed to a compression driver rather than to its own speaker. The acoustic output of the driver is fed into a long tube, the end of which is placed in or near the musician's mouth. By changing the size and shape of his mouth opening, the musician has total control over an extraordinary range of resonant changes in the instrument sound which is now picked up via his vocal microphone. This type of effect has been around for years, but it has been popularized recently by the "talking guitar" effects used by Peter Frampton, Jeff Beck and Joe Walsh among others. The Electro-Harmonix Golden Throat has two significant features not available in other talk box units. First, the driver used is rated at 100 watts rather than the more usual 60 to 65 watt drivers used by other manufacturers, and second, the Golden Throat has a peak overload indicator for additional protection against burn-out. The other new model from Electro-Harmonix is the Zipper ($79.95), an envelope-controlled filter device designed for guitar, bass or keyboards. The Zipper also has several features not found in other envelope-controlled filter devices, including Electro-Harmonix's own Dr. Q and Y-Triggered Filter models. The Zipper has a low-pass/band-pass filter mode switch for a greater range of harmonic effects. There is also a Filter Form/Attack control which allows either a normal recovery sweep of the filter frequency after each note is played or an extra-fast return sweep which gives a very electronic-sounding synthesizer-type effect.



Unicord, Inc. (Westbury, N.Y. 11590) now has a four-channel micro-mixer, the Model FCM41 ($79.95). The unit has a new "automatic padding" volume control circuitry allowing use with virtually any high impedance microphone or instrument output. The mixer is powered by a 9-volt battery or the included AC adapter.

Design Engineering Labs, Inc. (4121 Redwood Ave., Los Angeles, Ca. 90066) is the manufacturer of the Frogg line of sound modifiers. The Frogg line includes a Ring Modulator ($95.00), a Fuzz-wah Pedal ($125.00), a Flanger ($195.00), the Frogg Micro-Fuzz ($89.00), the Frogg Fazor ($155.00) and a completely unique product, the Compu-Sound ($495.00). This latter is an extremely versatile effects device incorporating a digital computer microprocessor which allows any one of ninety-nine preset effects to be selected at will by means of a numerical keyboard. The effects are based around filtering (wah-wah type effects) or variable delay (flanging) circuitry, and control comes from an envelope follower, a low-frequency oscillator and a control pedal supplied with the Compu-Sound. A two-digit, seven-segment LED display is used to read out which of the ninety-nine operating modes has been selected, and LED status indicators are provided for pedal control mode, panel (oscillator) control mode, bypass mode, oscillator rate and input overload. A level control is provided for optimum results with any instrument.

Brand new from MXR Innovations (P.O. Box 722, Rochester, N.Y. 14603) is the MXR Analog Delay system ($299.95), which is designed to electronically duplicate the sound of tape or disc echo units without the frequent maintenance required by such units. Delay time is variable from 33 to 500 milliseconds, and special low-noise circuitry is employed to give a dynamic range of 80 dB, enough for even critical applications. A mix control sets the balance of "dry" and delayed signals, and a regeneration control varies the number of echoes from a single repeat through multiple echoes up to total regeneration (feedback) to give the user total control of the sound.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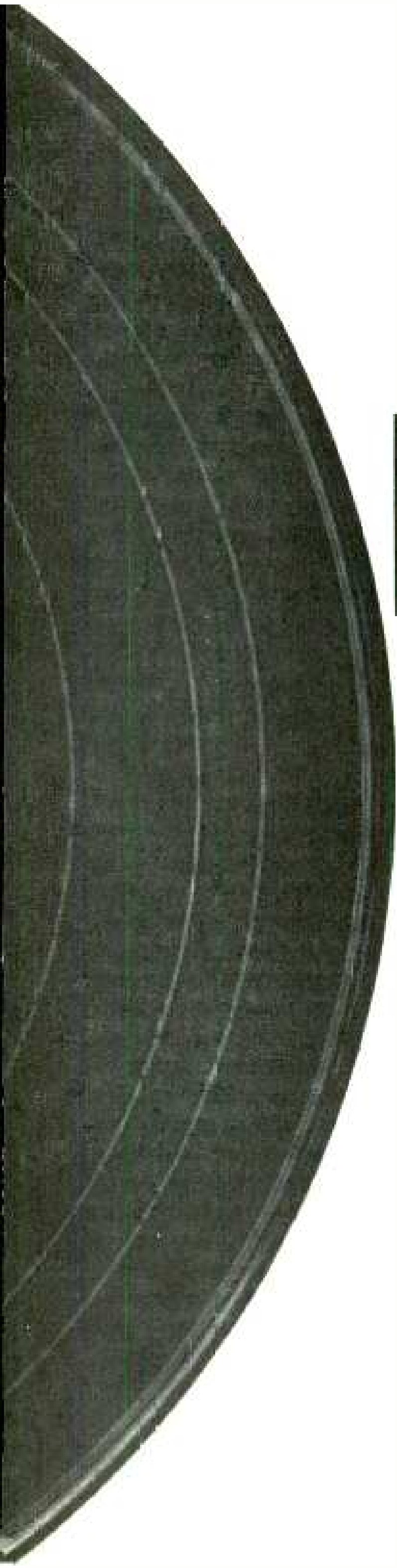
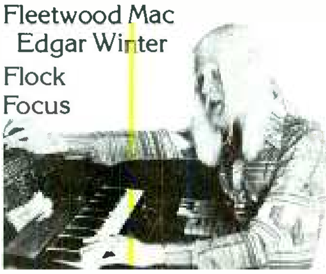


Paul Anka
America
Average White Band
The Band
Beach Boys
Bee Gees
George Benson
Blackbyrds
Black Sabbath
David Bowie
Brass Construction
Gary Brooker
Darius Brubeck
Dave Brubeck
Jack Bruce
Donald Byrd
Joseph Byrd
Camel
Glenn Campbell
The Captain & Tenille

Carpenters
Felix Cavaliere
Chicago
Commodores
Alice Cooper
Chick Corea
Floyd Cramer
Deodato
Rick Derringer
Neil Diamond
Doobie Brothers
George Duke
Electric Light Orchestra
Fabulous Rhinestones
Fleetwood Mac
Edgar Winter
Flock
Focus



Dan Fogelberg
Richard Furay
Jerry Garcia
Marvin Gaye
Genesis
Bobby Goldsboro
Jerry Goldsmith
Graham Central Station
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Isaac Hayes

Weather Report
David Hentschel
Janis Ian
Isley Brothers
Bob James
Jazz Crusaders
Jean-Luc Ponty Band
Jefferson Starship
Elton John
Quincy Jones
Michael Kamen
Kansas
K.C. & The Sunshine Band
John Keating
Kool and the Gang
Al Kooper
Kraft & Alexander
L.A. Express
Yusef Lateef
Ronnie Laws
John Lennon
Les McCann

Don Lewis
John Lewis
Ramsey Lewis
Lighthouse
Jon Lord
Lost Gonzo Band
Henry Mancini
Mike Mandell
Mandrill
Manfred Mann's Earth Band
Ray Manzarek
Curtis Mayfield
Sergio Mendes
Jimmy Messina
Ronnie Milsap
Joni Mitchell
Hugo Montenegro
Moody Blues

Patrick Moraz
Alphonse Mouzon
Don Muro
Michael Murphy
Johnny Nash
Randy Newman
Ohio Players
Nitty Gritty Dirt Band
Jimmy Page
Robert Palmer
Oscar Peterson
Pink Floyd
Linda McCartney
Paul McCartney
The Platters
Billy Preston
Renaissance
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B.J. Thomas
Three Dog Night
Tower of Power
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Tanya Tucker
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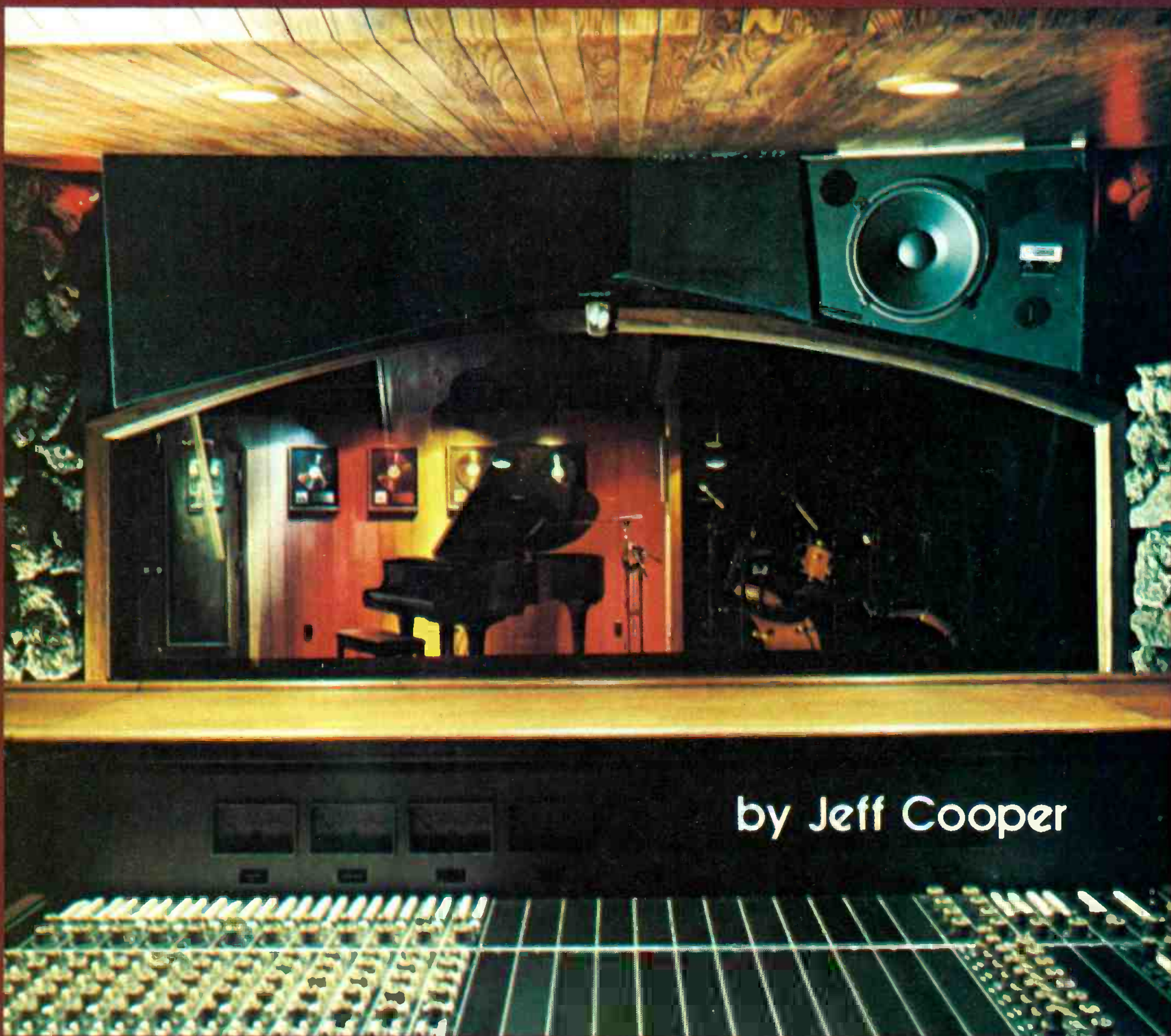
how acoustics affect recording



by Jeff Cooper

In small home studios, where music is often recorded and played back in the same room, room acoustics are doubly important because they affect both record and playback operations. For instance, if a room "colored" the natural sound of an instrument when it was recorded, the room will affect the "colored recording" in a similar fashion during playback. This makes the original problem sound twice as bad. Similarly, if a room were too reverberant and made the sound "muddy" during recording, the sound will be further muddied during playback.

To eliminate such composite problems, special attention must be paid to room acoustics in home studios.

Thus, the purpose of this article is to present the principles of room acoustics in order to demonstrate how these principles can be applied and controlled to produce better recordings.

Noise

Noise is unwanted sound. It is the first and most basic effect of room acoustics on recording. If a recording environment is noisy, chances are the recording itself will be.

Noise can be produced by an infinite variety of sources (from rush hour traffic to screaming little children) whose only common bond is their undesirability. Noise can never be completely eliminated—for its existence is intertwined with the very fabric of our existence. The random motion of molecular particles makes noise. Our living bodies are virtually factories of noise.

In fact, our very ears must make a degree of noise in order to hear.

Despite these impediments, in a recording environment we attempt to reduce extraneous noise to its lowest possible threshold. Here, this process is referred to as "soundproofing."

Ambient Noise: When talking about the penetration of sounds into the recording area, we are actually addressing the problem of ambient noise.

The ambient noise level may be thought of as the average amount of sound (from a conglomeration of sources) reaching the listener (be it the human ear or a microphone) at any given time. These ambient sound include street sounds, machinery sounds, distant conversations, as well as natural sounds such as wind and the patter of rain.

On a busy street the ambient noise level may be as high as 90 dBA (SPL). In most areas 70 dBA is more common. A quiet suburban home may have an ambient noise level of 45 dBA. In a recording studio, experience has shown that the ambient noise level must lie *well below 25 dBA* to be acceptable. The above values, as well as other common ambient noise levels are shown below.

Apparent Loudness	Source	Sound Pressure Level
Painful	Jet Aircraft at runway	140 dB
Extremely Loud	Rock Band on stage	135 dB
	Subway Train, track side	115 dB
	Heavy Truck Noise, roadside	90 dB
	Noisy Office, many typewriters, office machines, etc.	80 dB
Loud	Normal Traffic, roadside	70 dB
	Average Office Noise	60 dB
Moderate	Moderate Speech	55 dB
	Tranquil Residential Neighborhood, outdoors, nite	45 dB
Faint	Residential Neighborhood, indoors, nite	30 dB
	Rustling Leaves	20 dB
Very Faint	Normal Breathing	10 dB
	Threshold of Audibility	0 dB

From the chart we can infer that extreme care must be taken if we are to isolate the studio area from its noisy external environment.

Some Common Misconceptions About Ambient Noise

The first common misconception is that an absorptive room is a quiet room. Although abundant carpeting, curtains and acoustic tile do make rooms seem more quiet, by stopping the interior sound from reflecting off interior surfaces, these absorbers do not stop a *significant* amount of outside sound from penetrating into the room. They simply increase the ratio of direct to reflected sound heard in the room. Although this has an effect on the ambient noise level *perceived*, absorption does not make for low ambient noise. In fact, in recording studios we must *limit* the amount of absorption we apply, for an overabsorptive room is a terrible environment for the playing of music. Instruments sound "dead," and musicians sometimes wish they were.

We might add that conventional absorptive materials, such as carpeting and acoustical tile, absorb sound selectively in certain frequency bands. Thus excessive absorption in one part of the frequency spectrum can lead to tremendously imbalanced frequency characteristics in the studio. These points are emphasized to lay aside the common belief that a recording studio is simply a room with a lot of acoustical tile.

Secondly, we are often led to believe that a room which sounds relatively quiet to us has a low ambient noise level. This is not necessarily true. As humans, operating with a binaural listening system, we have a tremendous ability to reject certain noises. We are constantly hearing the din of traffic and wind, yet we very seldom notice these sounds. Our two-ear system enables us to *locate* the sounds we want to hear by "triangulation" (i.e., by subconsciously assessing the phase difference between sounds arriving at our left and right ears, and by using the information to deduce the probable location of the sound source). Thus we do not need very large sound pressure level differences to be able to distinguish between two sources. It is important to note that a microphone does not possess this ability. It differentiates between various sound by pressure-level gradients only. Thus, it will pick up background noise levels which we subconsciously eliminate. Since, in a recording studio, we are interested in the sound a microphone will pick up, simple listening tests to determine ambient noise levels are not adequate.

To truly ascertain the ambient noise level of the room, tests must be conducted with a microphone, sound level meter and filter set (octave or third octave). Readings are taken in the proposed studio area at various frequencies throughout the spectrum and a noise curve is plotted.

The height and shape of the curve indicates the severity of the noise problem and determines how much "soundproofing" will actually be required to reduce the ambient noise to an acceptable level for recording.

If you do not possess a sound level meter and are interested in ascertaining the approximate level of ambient noise in your studio environment, simply set up a microphone in the studio and record the ambient noise at normal operating level on one channel of a stereo recorder. On the other channel simultaneously record without any input device. (This will correct for inherent tape hiss added during the record operation.) Now compare the left and right channels at your normal listening level. The difference is the ambient noise of your recording studio environment.

Coloration: Most rooms have a number of resonant frequencies which they prefer to accommodate. These are related to the dimensions and surface characteristics of the room. These resonant frequencies are called "standing waves" or "room modes."

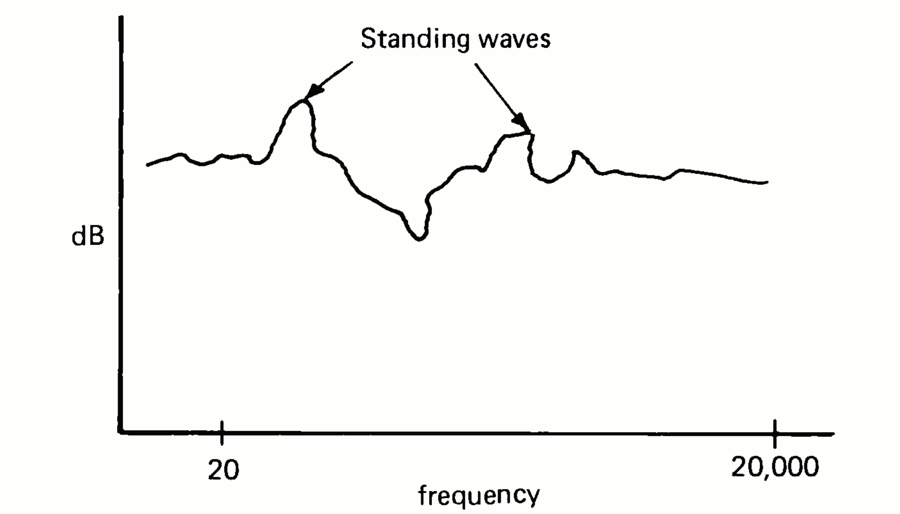
Standing waves can form in several ways. In the simplest case, a low frequency sound wave will resonate between two opposing surfaces of a room, continually reinforcing its amplitude by the method of constructive interference (i.e. each reflection adds to the previous one until all eventually die away).

This type of standing wave is called an "axial" mode and will occur at frequencies whose wavelengths are twice as long as the distance between the reflecting surfaces. Thus the lowest standing wave that occurs in a room has a wavelength twice as long as the room's largest dimension.

Axial standing waves can occur in *each* dimension of a rectangular room (length, width and height) and can be calculated using the following formula:

$$fo = \frac{v}{zd}$$

where fo = fundamental frequency of the standing wave
v = velocity of sound
d = room dimension (length, width or height)



Typical control room-speaker response, showing the presence of standing waves.

Other standing waves will form at the *harmonics* of the axial frequencies.

Thus, in a room whose length, width and height are 20, 16 and 8 feet respectively simple standing waves would form at the following frequencies:

	Harmonics (Hz) 1st	2nd	3rd	4th
$f(l) = \frac{v}{2(d)} = \frac{1130}{2(20)} =$	28	56	84	112
$f(w) = \frac{v}{2(d)} = \frac{1130}{2(16)} =$	35	70	105	140
$f(h) = \frac{v}{2(d)} = \frac{1130}{2(8)} =$	70	140	210	280

Simple standing waves for a room (20′ × 16′ × 8′)

The net effect of the presence of these standing waves would be a selective boosting or reinforcement of the room sound at these particular frequencies, resulting in a tremendous imbalance or coloration of the acoustical properties of the room.

Other types of standing waves, known as "tangential" and "oblique modes" also exist. Numbering in the hundreds, these modes have complex mathematical derivations. Fortunately, by the time a studio is filled with people, equipment, instruments and baffles, the path of sound is obstructed and most of these latter modes are eliminated. As a result, they play only a small part in determining the acoustical climate of the studio, and will not be dealt with in detail.

Most rooms have dozens of severe standing waves affecting the acoustical properties of the room. In the studio, the presence of these characteristic colorations poses two problems:

(1) The natural sound of the instruments being recorded is altered.

(2) The reverberation time at the standing wave frequencies (or at low frequencies in general) is exaggerated, causing excessive *leakage* between instruments.

In the control room, the presence of these standing waves causes the following problems:

(1) The sound of the recording is altered, causing constant guessing as to what balance is really on the tape and what is being manufactured by the room acoustics. This situation can cause engineers and producers to overcompensate or undercompensate with level and equalization, sometimes ruining a recording. (This effect is also responsible for the same type of speakers sounding drastically different in different rooms.)

(2) Recordings mixed in the control room will sound quite different when played back in another room, even when the same type of speakers are used.

(3) The reverberation time at the standing wave frequencies is lengthened, causing a lack of clarity at these frequencies, i.e. a "muddy" sound. This is by far the most common problem in listening and mixing rooms.

The remedy for these problems includes the application of absorbers specifically designed to eliminate colorations.

Reverberation

When sound waves leave a source they travel outward in a three-dimensional arc. Some of these waves travel directly to the listener and are called "direct" waves. The other waves (the greater proportion) strike the walls, floor, ceiling and room furnishings. Upon striking, three phenomena occur:

(1) A portion of the sound energy is *transmitted through* the wall, floor or ceiling to the other side.

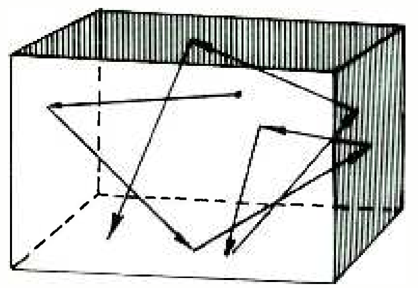
(2) A portion of the energy is *absorbed* by the barrier.

(3) The remaining portion of sound energy is *reflected* back into the room.

The reflected waves eventually strike other surfaces and the process repeats. Because the waves lose

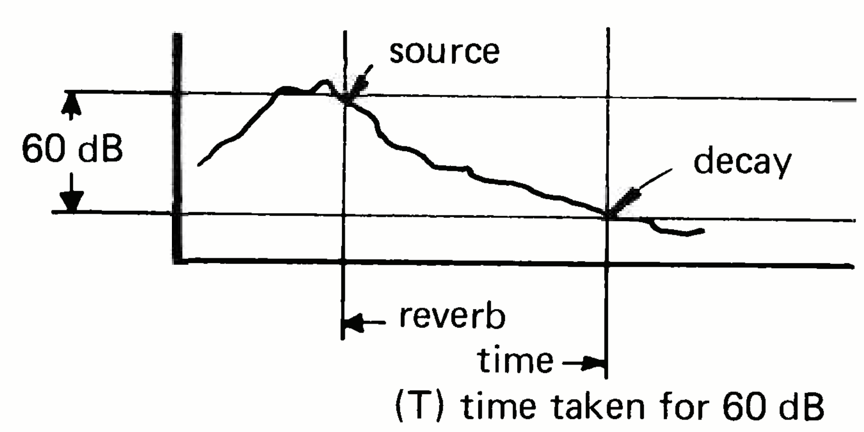
energy during travel, as well as each time they strike a surface, successive reflections become gradually weaker until they eventually become completely inaudible.

These reflections arrive at the listener in a continuous stream. They are so closely spaced that the ear is incapable of distinguishing them as individual sound waves. Instead, the ear perceives a gradual decay of the room sound, sometimes lasting several seconds after the source has stopped altogether.



Decaying reflections in a room

This gradual decay of sound in a room is called "reverberation." The time taken for a sound to decay to one millionth of its original sound pressure level (i.e. 60 dB of decay) is called the "reverberation time" (commonly



called "reverb time"), and denoted by the letter "T."

The reverberation time is a function of two main factors:

(1) The volume of a room.

(2) The absorptivity of a room.

In a large room a wave must travel long distances before reaching a reflecting surface. *Therefore, the reverberation time is generally longer for large rooms.* The reverb time for a huge cathedral may be six or seven seconds. For a small living room, the reverb time may be only half of one second.

If the surfaces of the room are very absorptive (i.e. carpet, drapes, thickly upholstered furniture) very little wave energy will be reflected back into the room at all. Thus, very absorptive rooms generally have shorter reverberation times. Reverb time, room size and absorptivity are related according to the following formula:

$$T = \frac{.049\,V}{A}$$

where T = reverb time
V = room volume (length × width × height)
A = total absorptivity of room (measured in sabins)

Because the formula does not take into account such variables as frequency of source, room geometry, temperature and humidity of air, presence of people and furniture and other random obstructions which increase the scattering of sound waves, the formula provides only an approximation.

Actual reverb time is usually measured using a sound level meter, a loud source (such as a pistol-blank) and an oscilloscope.

Rooms which have hard surfaces (tile, wood, plaster) and long reverberation times (over one second) are usually called "live" rooms. These types of rooms have the net effect of reinforcing the sound of music played in them, (especially at lower frequencies) and cause the music to sound relatively loud and resonant within the room. Many musicians are quite pleased to hear themselves loudly reinforced and tend to like to play in rooms that are slightly "live."

Reverberation tends to smooth out the inconsistencies in timbre and pitch, thus making a given performance sound more consistent and professional. This explains why singing in the shower is so popular. The hard tiled surfaces of the shower reflect the voice, emphasizing its lower frequency overtones, while smoothing inconsistencies of pitch and timbre, giving the illusion of power, control and strength.

Rooms with soft surfaces (curtains, carpet) and short reverb times (under .5 sec), on the other hand, are called "dead" rooms. They give little or no reinforcement to music played within, causing it to sound weak and lifeless. In addition, these types of rooms make small inconsistencies in timbre and pitch readily apparent to a listener (and a musician) and as a result are quite unpopular among musicians.

The fact that reverberation can help a voice or instrument to sound fuller and more "professional" does not mean a recording studio should have a lot of reverberation.

In multitrack recording, for example, instruments are isolated from one another and routed to separate channels of the multitrack tape for mixing and balancing at a later time. Too much reverberation in the studio would make isolation almost impossible to achieve; reflections from various instruments and amplifiers would bounce aimlessly, finding their way into microphones meant for other instruments. This effect is especially pronounced at lower frequencies, where wavelengths are quite long and difficult to absorb. In the extreme case, this misdirection of sound, called "leakage," would cause many instruments to appear on each track of the multitrack tape, thereby defeating the entire purpose of multitrack recording. It would mean that when the level of a particular instrument was to

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be raised or lowered, the sound of other instruments would be inadvertently raised and lowered as well. The overall recording would have a rather loose or muddied quality, reminiscent of recordings made with a single microphone in a basement or garage.

A second reason that excessive reverberation is discouraged in recording studios is that electronic means have been developed for adding controlled amounts of reverberation to a recorded sound after the initial recording has been made. Therefore, if a certain amount of reverberation is desired for a voice or instrument it can always be added later. Since any reverberation present on the initial recording cannot be subtracted, it is much safer to record an instrument slightly dry at first, and to add the desired amount of reverberation later, during mixdown.

A final reason overly reverberant rooms do not make good general-purpose studios, is that a recording will always seem to have more reverberation than is actually on the recording, due to the reverb added by the listening environment. If a room with a reverb time of approximately .5 seconds (average living room) is used as a listening room, recordings heard in that room will seem to have a reverberation time of approximately .5 seconds longer than that which actually exists on the recording.

This effect does not occur when monitoring through headphones, thus explaining why music heard through phones often sounds more crisp and direct than music through speakers.

These arguments all seem to suggest that a recording studio should have a short reverb time. However, if the reverb time is too short, the studio will become very anechoic or dead, and musicians will feel uncomfortable. So what is the optimal reverb time?

A partial answer is that optimal reverb time will depend on the type of music to be recorded. In a large studio for classical music, a certain amount of reverb will not only be tolerated, it will be desired. This will give the orchestra a group sound, blending inconsistencies between various players and sections. Classical music studios may have reverb times of several seconds or more. In a rock music studio, where levels are loud, space is tight and isolation is critical, leakage must be avoided and reverb time must be fairly short (¼-1 second).

If a studio is to accommodate as many types of music as possible, we must look into ways of making reverb time adjustable.

In general, reverberation must be reduced at low frequencies—the worst culprits for leakage—and adjusted at mid and high frequencies to suit the music played.

Absorption

When a sound wave strikes a surface three phenomena occur:

(1) A portion of the sound energy is transmitted through the barrier.

(2) A portion of the sound energy is reflectd back into the room.

(3) A portion of the sound energy is absorbed.

When we say that sound has been "absorbed" we simply mean that it has been converted into another form of energy, generally heat. Admittedly, the energy levels normally encountered in sound waves are so minute (i.e. a billionth of a watt per square centimeter) that the heat of absorption is scarcely detectable. However, this conversion phenomenon occurs to satisfy the laws of physics.

The fraction of incident sound energy absorbed by a material is called the "absorption co-efficient" or "absorptivity."

The absorptivity of a material depends on many internal characteristics (including porosity, modulus of elasticity, internal flow resistance and thickness). It also depends on external factors such as the frequency of the incident wave, the angle of the incident wave (absorptivity is greatest when a wave strikes perpendicular to the absorber), and the manner in which the absorber is mounted.

Almost without exception, conventional acoustic absorbers, such as acoustical tile, carpeting, curtains, mineral wool and fibreglas are porous absorbers. As such, they absorb more effectively at the high frequencies. This results from the fact that high frequencies have small wavelengths (from several inches down to only fractions of an inch long). The to and fro motion of these small waves gets easily trapped in the spaces of a porous material. Friction converts the sound energy into heat and the wave energy gets absorbed. Low frequencies, with large wavelengths (up to fifty feet long) do not get trapped in these tiny spaces and, as a result, are not absorbed.

The method of porous absorption is not the only means of acoustical absorption. A second type of absorber is the flexible panel or membrane absorber. This consists simply of a membrane, such as plywood, mounted over an airspace of several inches or more. A membrane absorber works in the following way: an approaching sound wave strikes the cover panel, setting it into motion. The panel resonates at a frequency determined by the size of the airspace behind it. The resonating panel dampens the approaching sound waves, in the same frequency range of the panel resonance. Thus, these particular frequencies are absorbed, while other frequencies are reflected. Membrane absorbers are absorptive at low frequencies, in contrast to porous absorbers, which are absorptive at high frequencies. Consequently, they are often called basstraps.

Echo

Echo is the repetition of a sound due to the delayed arrival of a reflected sound wave. In large rooms, we often hear distinct single echoes. In small rooms we usually hear a short continuous stream of echoes. These latter types are called "flutter echoes." Both single and flutter echoes interfere with the clarity of a recording and must be eliminated from the recording environment.

Diffusion

A sound field is diffuse if its intensity anywhere in a room is approximately equal. A room with a concave wall or a domed ceiling can never promote good diffusion. These shapes "focus" sound waves, instead of spreading them around the room. Standing waves and echoes are both indicators of a lack of diffusion.

The diffusion of sound fields is desirable in a recording environment because it allows instruments to be recorded in different room positions with a measure of acoustical consistency. Good diffusion helps take the guesswork out of microphone placement—and makes good recordings easier to achieve.

This article by Jeff Cooper was excerpted from his book How to Build Your Own Recording Studio. *The book will be published by the Recording Institute of America, 15 Columbus Circle, N.Y.C., N.Y., and should appear in the fall of this year.*

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A Session With Patti Labelle

By Steve Whiting

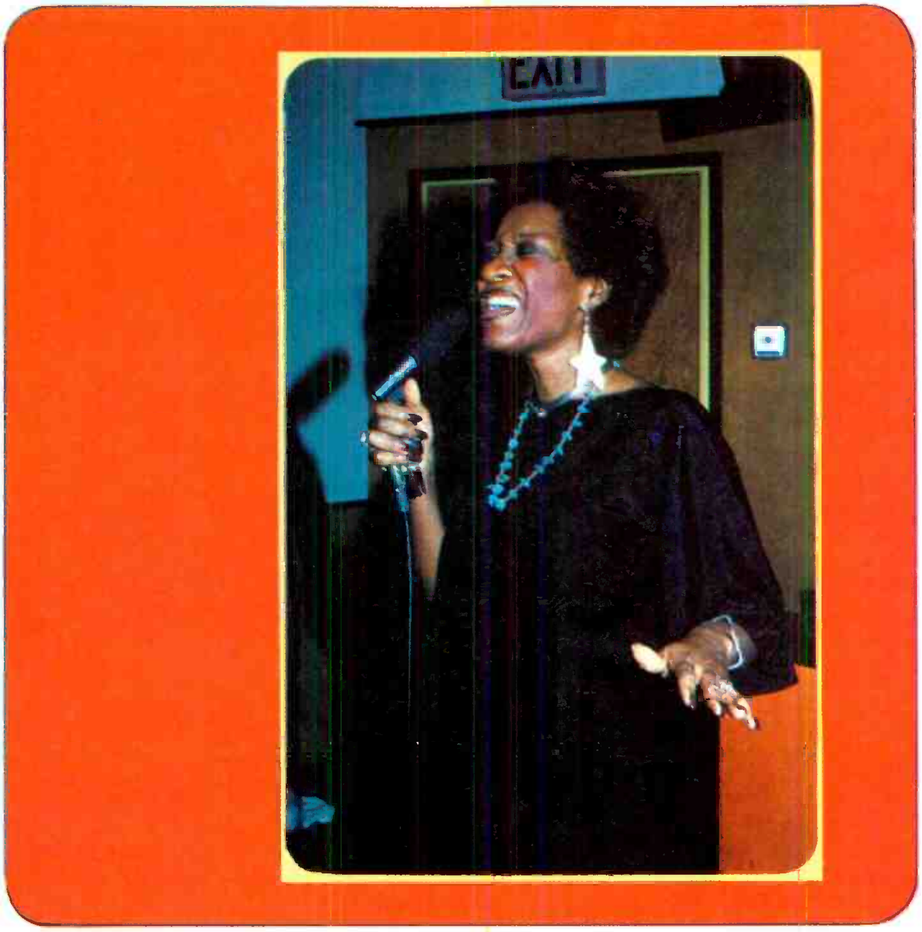
Anyone interested in modern recording techniques and equipment will readily attest that there is more involved in the production and engineering of a hit record than meets the ear. That which meets and pleases the public ear is primarily the end result of a symbiotic relationship among the artist, producer, engineer and all the machines and gadgetry which we collectively refer to as a studio. One such relationship certainly exists between artist, Patti Labelle; producer, David Rubinson; engineer, Fred Catero; and the new Automatt automated studio in San Francisco.

Collectively planned and designed by Catero, Rubinson and engineering consultant Michael Larner, The Automatt is the most modern and fully automated facility that has been built anywhere to date. Basically it consists of a custom Harrison 4032 automated console (which can handle forty microphone inputs and thirty two tracks of recorded information) interfaced with the new Allison Research 65 K Series Memory-Plus Automation System. The Automatt records on MCI 24-track and 2-track mastering machines, and can store additional recorded data or programs from the programmer on a Scully 8-track machine.

When the Harrison 4032 Console is interfaced with the Allison Memory Plus 65 K Automation System it is "programmable." The word "programmable" means that manually exercised functions may be entered into the programmer and subsequently these functions can be recalled or read exactly as entered. This gives the console a "memory" which can process as many as 65,536 separate functions or bits of data information simultaneously.

One function that makes The Automatt unique was developed by Michael Larner. It is a special 4-track earphone cue system. This enables each musician to mix his own cue balance. He or she can hear as much drums, vocals, bass or anything as is individually desired and need not be dependent upon the engineer for an appropriate balance. On one of the "live" tracks Patti got best results by using only one ear of the phones. She cranked up her vocal track in that ear, turned the other instruments off, and listened to the band "live" with her naked ear only. She was able to hear what she needed, while not changing anyone elses' cue balance, and still maintain the "live" feel of the musicians around her.

Another of the unique functions is



(to quote producer Rubinson), "programmable mute keys on every module. These have the same effect as zeroing the voltage control faders." In other words if you have a drum track and push the stock audio-mute button all it does is kill the monitored audio, not the voltage. If you switch The Automatt's special programmable mute toggle it is as if that fader had been completely brought down. This has the effect of bringing the voltage to zero as well, and that single fader is programmed to be off or "mute."

When a fader is used as a group master all the drum tracks or vocal tracks or any other tracks so selected can be controlled, muted and programmed for mute at once with a single fader or a single mute switch, thus freeing the four or five fingers it would normally take to push mute buttons, adjust faders for all the tracks or remember to bring them up or down at the same time on each run-through of the tape. The value of the group master capability becomes obvious during mix-down.

The console can also simultaneously perform duplicate functions. This means that the studio provides multiplicity of purpose. Rubinson: "We can have it all set up to do basic "live" tracks, but when the first group goes home we can be mixing an entirely different album on the other side of the board without disturbing any of what was set up for the "live" tracks. Simultaneously, it would be possible to be cutting tracks, overdubbing, mixing and updating another mix or whatever." There wouldn't be any disturbance of the program or the other functions set up at the same time because the storage capability of the programmer is essentially limitless and the console itself can be in more than one status at a time. The console actually serves as a signal path-controller.

A case in point of this multi-functional capability was on Herbie Hancock's "live" CBS album, V.S.O.P. After it was entirely mixed Herbie didn't like the amount of echo he had requested from Rubinson on the piano track. Herbie knew that even though he had every artistic right to demand a re-mix, to do so manually would have consumed a great deal of time and money, and heretofore, may have been impracticable. However, Rubinson was able to simply read or recall the entire program for the original mix and then update or change only the echo on the piano. (The Automatt uses EMTs exclusively.) He then simply "read" the program while the computer re-mixed the album exactly as before but with "updated" echo. This was done without interruption to the Heartsfield album whose tracks were being recorded at the same time.

In observing and talking to Patti Labelle, the studio musicians and the team of The Automatt, three words kept coming to mind in their description of the sessions: Professionalism, efficiency and smoothness. Mutual respect and trust ran rampant and private discussions were flowered with praise upon praise of one-another's techniques and abilities. This, of course, created a most congenial atmosphere for the sessions.

A large part of what makes a session run smoothly is the result of thorough pre-production work. According to Rubinson, "Pre-production is first conceiving the project, then finding material that fits the conception; or deriving the conception from the material that the artist provides. It's generally a combination of the two." In the case of this project, Rubinson and his a & r man, Jeff Cohen, selected forty or so songs which David then took to Patti in Philadelphia. Patti, during the same period, also had written and chosen a lot of material. Between them they decided on the songs to be used. However, the decisions are not always hard and fast. The song "You Can't Judge A Book By Looking At The Cover" was not arranged until they were actually in the studio. The groove just fell together and they recorded it "live" on the spot. Rubinson, "Then you select how you're going to be doing the material and with what components. Next you go into rehearsal on that material, writing down those things that need to be written in order to get the musicians to perform what it was you heard when you first conceived the project."

Rubinson starts with a chord sheet and then proceeds to a master rhythm chart for the whole band so everybody can see what everybody else is playing. Bass parts, drum figures, piano, voicings of the chords and the whole structure of the tune is written out. This includes the codas, repeats and dynamic markings. He uses both the chord name and the number symbol like $\frac{\text{IVm7}}{\text{Cm7}}$ because some players read numbers instead of names. He also labels the sections of the tune such as A, B, verse, chorus, etc. Therefore, everybody can relate to the song in his own terms. The chart is clearly a road map for the tune, and makes it very simple to find any section at any time. To further expedite locating any section Rubinson marks the MCI tape locator numbers of each section on his chart after the laying of tracks has begun. Confusion and/or being lost are problems which don't exist for the session players, engineer, producer or artist.

All of the work that goes into pre-production is well rewarded once the actual recording sessions commence. Patti Labelle was radiant, comfortable and obviously at ease with the surroundings, the studio personnel and her back-up musicians. The musicians included her musical director and pianist Bud Ellison, drummer James Gadson, guitarist Ray Parker (also on bass for a few cuts) and bassist George Porter and guitarist Leo Nocentelli from the Meters.

Nocentelli and Porter come from a different mold than the other musicians. It was their first time recording outside New Orleans. They are "groove" players who don't read music or charts. Yet, they fell right in with the likes of Ray Parker and James Gadson who are very professional L.A. studio sidemen and do as many as four or five sessions a day. The synergy was magic and each came up with grinning admiration for the styles, licks and proficiency of the other. The music bears witness and you can hear 'em smile.

The technical aspects of the Patti Labelle sessions at The Automatt were as interesting as the musical/creative process was captivating. Though perhaps not presented with the same dynamic intensity and impact of a Patti Labelle vocal, the experience, artistry and touch were there at the hands of engineer Fred Catero and his assistant Chris Minto.

On studio miking techniques Fred Catero had this to say, "The most important thing is to make sure the mics are in phase, not *which* mics you use, but whether they are in phase or not. That's the whole secret. It has nothing to do with phasing electrically. Electrically, we assume that every studio is going to have their mics in phase. It's acoustic phasing. Whenever you have more than one mic on anything you've got to be sure that they are cancelling the frequencies you don't want and building the ones you do want. Any good board will have a phase reverse switch. I listen to the drummer, for example, and then reverse the phases individually. I listen to the first two and then the next two, etc., and then I compare the overheads to the snare and then to the foot, then toms to foot and so forth. Sometimes you have to put a mic out of phase in order to get certain frequencies and avoid cancelling others. The highs you don't worry about because they're of such short wave length. The lows are more vital, being of longer wavelength, because if not properly phased instead of building toward a solid round, bottom sound you can actually be dropping out the low end." So it's the phasing, according to Mr. Catero, that is more important than which mics are used where.

Be that as it may, here are the mics that Fred used to record the band for Patti Labelle. He referred to this simply as "a standard, basic set-up." It involved using Neumann U-87 condenser microphones (padded of course) on the guitar amps. The basses were recorded direct from the instruments and direct from the effects, whether they be phasers, frequency shifters or wah-wahs. The line went from the bass itself to a direct box for one track to whatever effect was being employed, to another direct box for the "effects" bass track, and finally to the bass amplifier. The amp was only used so that it could be heard in the room by anyone not using head phones. The Fender Rhodes (Bud Ellison) was recorded direct with some limiting because of the nature of the instrument.

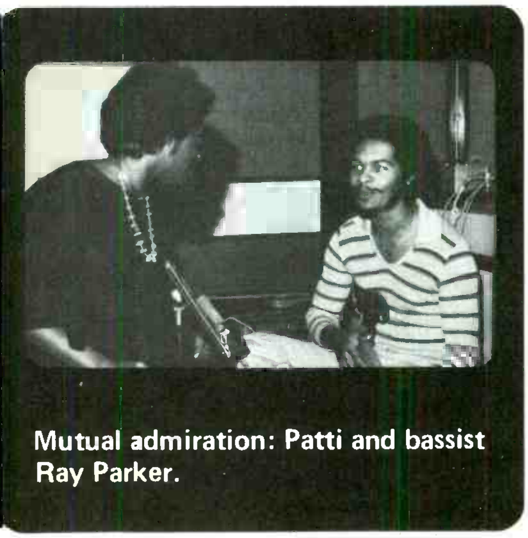
Patti checks arrangements with pianist/ arranger Bud Ellison.

Patti Labelle with producer David Rubinson.

With the tynes, if you hit it hard it will "splatter" and go to maybe plus 30 or 40 dB, or if it's played lightly it can be down to a minus 20 dB. Fred, therefore, compresses somewhat so that the overall sound is more even. On the acoustic piano Catero again used two U-87s. On the drums were U-87s overhead left and right; a Shure SM 56 (dynamic) on the bass drum and another to pick up the two floor toms and the snare; a C-22 Sony condenser was used to pick up the hi-hat. On a few cuts Rubinson had Gadson overdub an eight tom-tom set. Catero simply miked these with two U-87s overhead.

How does one mic a vocalist with the tonal, harmonic and dynamic range of Patti Labelle? On the cuts that were done "live" a Shure SM 56 was used. On other cuts like Ray Parker's "Dance With Me," where the lead vocal was overdubbed, a Neumann U-87 was used with Patti singing very close to the wind-screened mic. Being in the fat part of the cardioid pattern creates a very big, full sound but causes some of the highs to be lost. To compensate for this loss the highs were simply equalized up a few dB. Here, again, the miking apparently was not of vital importance. According to Catero, "Patti's voice is so strong and clean she could telephone in her part and it would come out sounding great."

As far as equalization goes, Patti Labelle again, according to producer David Rubinson, presents a unique situation. "It depends on the range she's singing in. The partials and overtone series in her voice are very strange. She has a tremendous series of upper partials where, as with all great singers, she completes an overtone series well beyond what a normal rock-and-roll singer would complete. The timbre of her voice is remarkable in that she completes a very long series of harmonic overtones going up." Normally one equalizes for upper midrange or upper register harmonics at 3000 or 5000 Hz which strengthens the presence and cutting edge of an adequate singer's voice. Patti Labelle's voice is so strong that you don't need to equalize that much. "You can't equalize Patti's voice unless you do it very *smoothly*. You can't peak equalize and you can't just knock up your standard rock-and-roll engineer's 3000 cycle EQ. The overtone is going far above that with her voice. You can't just EQ for say the fifth above the octave above the note. You simply have to EQ so that you don't have any sharp peaks or valleys." Catero accomplishes that by smoothly riding the fader instead of using any specifically formulated EQ for Patti's voice.



Mutual admiration: Patti and bassist Ray Parker.

Fred Catero has some illuminating thoughts on when and how to use equalization. He feels that the best time to equalize is when the tracks are being cut, not so much when they're being mixed. This enables one to maintain the balance of levels more consistently with that which actually comes out of the room. When dealing with professionals, the players generally know the sound they're going for and the object is to capture that sound as it is being played. Guitarist Ray Parker feels that this is what makes Catero a great engineer. "He gets the sound that's going down. If you play a part and then listen back and it sounds different from what you played, then that ain't what was *happening* in the room." Fred EQs the sound first as it is being played. "Then," he says, "if it isn't right you can always go back and try to change it later."

One must constantly remain aware when equalizing, limiting, etc., that the average person's reproduction system is in quality far below that to which one becomes accustomed in the studio. Fred Catero compensates in this way (*note*: Fred feels that the specific numbers of dB by which he EQs are not that significant to readers because no two boards, rooms or recording situations are the same. There are no hard-and-fast EQ settings which are universally correct, therefore, it is the concepts of when and why to EQ that matter, not the specific settings that Fred or anyone else uses in any given circumstance): "Guitar I usually don't limit since it's generally in the mid-range; the bass I limit a lot; on drums I only limit the kick to keep it even. Since the bass and drums are the two loudest things on David's records we want to limit them because they're always featured. We don't want any crazy dynamics because they contain low-frequency information. Lows are the hardest to deal with when you're working with the medium of discs. One little bass flash or peak can cause it to skip and therefore forces the mastering man to drop the level of the entire take by that amount. You want the bass and drums to be the most even in terms of intensity, bearing in mind that what is to be sold is not electro-magnetic tapes, but records, which are electro-mechanical." Catero would suggest that anyone involved in recording engineering learn as much as possible about the special problems of disc mastering. Otherwise he'll be surprised when he gets his test- pressing back and it doesn't sound like the tape.

In keeping with this same line of thought it is very interesting to note that there is no dbx or Dolby noise reduction equipment at The Automatt (except of course for the cassette-copy machines). "The point is," according to Catero, "that if you understand the medium you're working with (magnetic tape) you can turn out a product that is almost as quiet or quieter than the average engineer who uses noise reduction equipment. In order to do this, first you take advantage of the threshold level of the tape. We record 3 dB elevated. When you look at the specifications for tape, it's +3 dB at normal operating level for all frequencies, but there are some frequencies where you can go much higher than that and some where you can't. Most of the ones where you can't are obviously at the high end, but there are also some on the low end. In the mid-range (1000-3000 Hz) you can go as many as 6 dB above +3 and still not notice it. Knowing and taking advantage of this you can record very hot and saturate the tape. Then when you mix down and put everything back where it belongs, you have reduced a great deal of noise. The trick is to find the point where you're recording as hot as possible without deteriorating the sound." Generally guitars, for example, are mostly all mid-range and they can be recorded very hot without any worry of distorting the tape or saturating it to the point where it can no longer be erased.

Engineer Fred Catero: "Making sure the mics are in-phase is more important than what type of mic you use."

During the Patti Labelle sessions David Rubinson employed what may be considered the extreme techniques of production for the various tunes. At one end we have a tune being done completely "live" ("You Can't Judge a Book By Looking At The Cover") and at the other ("Dance With Me") we have a tune being built up from the ground floor. The third technique, which was most utilized at these sessions, is to do the basic tracks "live" and then to overdub the vocals, sweeteners and reinforcing tracks.

On "Dance With Me" Rubinson started with an electronic rhythm machine, (which will be muted in the final mix). He then added drums, bass, piano and guitar. Back-up vocals will be added still later. Patti would like to do many of her own back-up vocals and might very well do some parts. Rubinson feels however that for some things on a solo album her voice may be too strong to do back-up vocals without perhaps taking away some of the impetus of the lead vocal parts. Her voice may be too strong to do back-ups for anyone, including herself.

All involved in the Patti Labelle sessions at The Automatt have been very pleased with the progress and success of the sessions. The material, musicianship, vocals, engineering and production have all been employed with extreme efficiency and taste. Rubinson et al., seem to have surpassed the point where they need or depend on a hit record or the consequent monies earned thereby. It is the artistry that seems to interest them most. Of The Automatt Patti says, "It's great, but I don't really understand or care about all the technical things involved. David and Fred could be working in a garage and I'd still be here. I love them." Everyone seems to have had a great time working together and perhaps that, along with musical and technical proficiency, is the ingredient that goes into making a hit record.

Audiences and recording enthusiasts alike can very well sense when sessions have gone smoothly and all involved have felt good about a project. If this is true then it should be very easy for everyone to hear it on Patti Labelle's solo album.

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

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

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

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

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

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A blunt and totally biased viewpoint on electronic synthesizers from Bob Moog.

"If you're into synthesizers, you and I have a lot in common. A love of good music. A fascination with the magic that happens when a talented musician plays a well designed instrument. A desire to expand the boundaries of musical expression. And an intense interest in electronic equipment created for enduring musical value—with pleasing tone color, responsive action, functional versatility and rugged reliability. The people at Moog® have devoted themselves to these principles for thirteen years, and it shows: Moog is still the standard of excellence among the majority of professional synthesizer players today. Sure, most of our ideas have been copied. We're flattered. But I'm concerned, and maybe you are too, about some of the hype that's been going around on synthesizer design. It's the kind of hype that places too much importance on gimmicks that don't have any real musical purpose. Here's the way I look it.

Musicians like you help design Moog synthesizers.

Fads, gimmicks and basically unmusical electronic features don't make a synthesizer a musical instrument. Feel, sound and playability do. That's why Moog has always designed its instruments in close collaboration with working musicians—people like Chick Corea, Herbie Hancock, Jan Hammer, Walter Carlos—some of the most creative and talented performers of our time.

At Moog, literally thousands of hours are spent every year talking (and listening carefully) to our musician friends. We visit recording studios. Go on tour with major groups. Sometimes, even participate in live performances. All this teaches us a great deal about what you, the player, need and want in a synthesizer.

It all started with the Moog concept of voltage control.

Moog imitations abound in the marketplace, but the heart and guts of modern electronic music—the basic elements of synthesizer design—were developed right here at Moog, back in 1964. What started it all was the Moog innovation of voltage control—the use of electrical signals to impart texture, color and pattern to musical sound. First came the multiple waveform voltage controlled oscillator; the balanced, voltage controlled amplifier; a keyboard with memory and glide. Then the first four-part ADSR envelope generator was developed, along with wide-range voltage controlled filters and a multiplicity of control devices. Of course, Moog innovations don't stop there. We're reinventing the synthesizer every day. But even more important is how all this fancy technology translates when you sit down to play.

From Moog "musical engineering," the very best sound and performance.

Formulas and specs are critical, but without the taste and insight of Moog's "musical engineers," our synthesizers would just be sound generators and not the versatile musical instruments they are. For example. Part of the famous Moog "fat sound" comes from the patented Moog filter. But the total sound of the instrument is the result of dozens of individual design features meticulously proportioned to give just the right sound and feel.

The spirit of "musical engineering" is behind Moog's many exclusive performance controls, too. Take our Pitch and Modulation Wheels and our Ribbon Controller. They enable the player to easily control pitch bend, vibrato and other nuances so vital to brilliant solo performance. These are the features that make the difference between a "one note organ" sound and the soaring, fluid Moog lines you hear in the best contemporary electronic music.

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We're reinventing the synthesizer every day.

The truth about sliders versus rotary pots.

I've heard a lot about this one lately... the argument about whether slide controls are better than the round, dial type controls. From where I sit, they're both useful. Take a look at any recording console. You'll see that the faders are sliders, because the recording engineer has to manipulate many of them simultaneously, especially during a mix. But just about all the other controls on the console are rotaries, because they're so much easier to set precisely. Moog's answer? Use both... rotaries where precision is critical, sliders where speed is the thing (or where space is limited). But to say that there's only one right way, well again, that's just hype. Just try to accurately tune an all-slider instrument!

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The point of all this is that if you're into synthesizers, you really ought to know the difference between electronic gimmickry and solid musical engineering. I'm convinced that Moog offers equipment that gives you the most quality, playability and musical control over sound. Of course, until you try a Moog synthesizer, you might just say I'm biased."

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CROSBY STILLS

A first hand report by their

Based upon the influences the three artists brought with them from the Byrds, Buffalo Springfield and the Hollies, the first Crosby, Stills and Nash album, recorded in 1969, was a classic forerunner of the now popular California sound and served as a model for important seventies groups like America and the Eagles.

After a long series of solo and duet efforts, and a number of attempted reunions, Crosby, Stills and Nash have completed their long-awaited reunion album, engineered and co-produced by Ron and Howard Albert of Fat Albert Productions. In their careers the Alberts have engineered for such artists as Aretha Franklin, James Brown, the Rolling Stones, the Allman Brothers and Eric Clapton; and for the last two years they've been producing artists as diverse as Wishbone Ash, Sutherland Brothers and Quiver and Chris Hillman. In this interview, Ron and Howard discuss their recording techniques as well as the challenges they encountered as engineers, producers and close friends of Crosby, Stills and Nash.

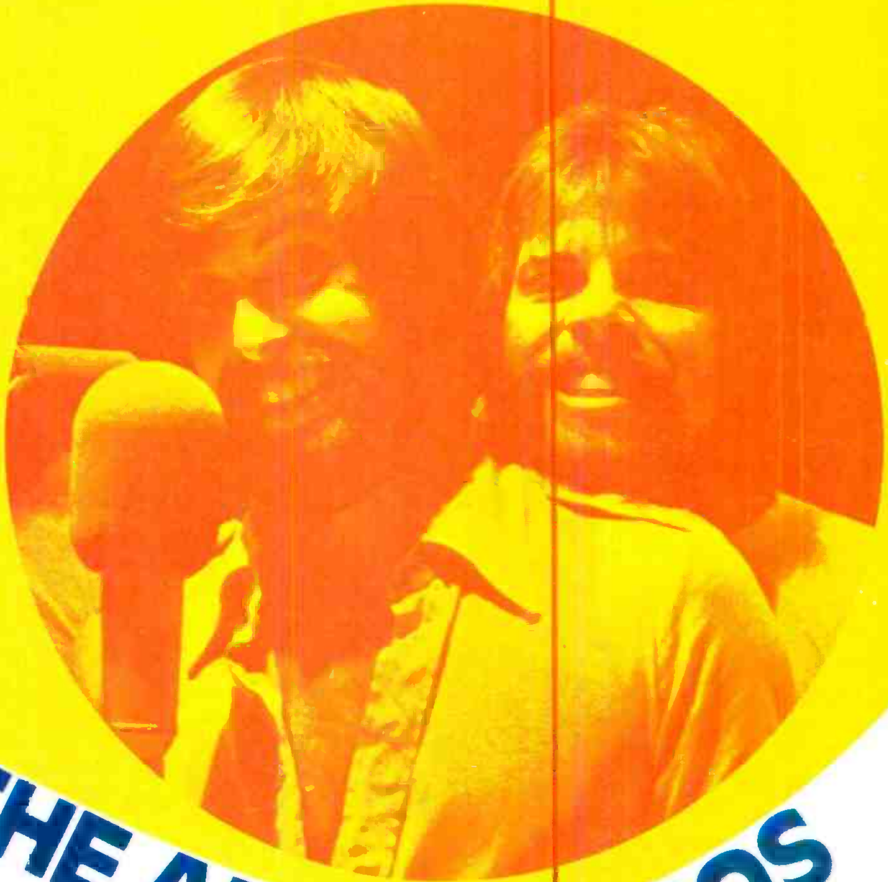
MR: This is the first time you've recorded an album with Crosby, Stills and Nash, but haven't you been recording Stephen Stills off and on for about six years?

RA: That's right. We first met Stephen when we were engineering the overdubs of the *Johnny Winter And* album at Criteria [a studio in Miami, Fla.]. Stephen walked in one evening around midnight, and, after introductions, Howard and I decided to set up in another room for the all-night recording of a song called "Word Game"—really a demo with Stephen

S & NASH

Recording Again!

By Stan Soocher



engineers THE ALBERT BROS

singing and playing acoustic guitar—which ended up on *Stephen Stills II.*

MR: Later you and Howard engineered the double *Manassas* album, also at Criteria.

RA: Yes, and that is one of our favorite albums. My favorite song of all time, "Johnny's Garden," is on that. The sound of the double *Manassas* album was actually a giant step for the "no rules" principle, which is a philosophy of recording Howard and I arrived at after years of experimenting with every possible combination of microphones and techniques—from the most logical to the most absurd. Basically, the "no rules" principle is: No matter what it takes to make something work, you do it to make the music sound as good as it possibly can, not stopping until it turns out better than you expected. The double *Manassas* album was definitely the beginning of that whole trip, and I think this new Crosby, Stills and Nash album is an extention of that.

Ron (left) and Howard Albert behind the console.

MIC SELECTION

VOCALS:	
	Shure 546
	AKG 414
	Sony C500
	Neumann U-87
GUITARS:	
Acoustic	Neumann U-87
Electric	Shure 546
	Sony 377
KEYBOARDS:	
Organ (top)	Shure 546
Organ (bottom)	Neumann U-87
Piano	AKG 414
	Sony 377
DRUMS:	
Snare	AKG 414
Kick	Altec 633
Hi-Hat	Sony ECM 50
Overheads	Sony 377
Toms	Neumann U-87
STRINGS:	
	Neumann U-87s

MR: When did you become involved with the Crosby, Stills and Nash album?

HA: David, Stephen and Graham were rehearsing some of the songs in David's house in California and Stephen called us up one night and said, "Hey, this stuff's really great. We're ready to record a new album. When can we start?" It was very close to Christmas, and out of necessity to get the project immediately underway and put the songs on tape, the sessions started at the Record Plant in Los Angeles. We broke for Christmas and after New Year's we all came to Miami to record; we completed the album almost three months later.

MR: Were the preparations for these sessions different from other sessions you've co-produced or produced—like Procol Harum, Wishbone Ash or Chris Hillman?

HA: Not really because we've tried to give everybody we've worked with the best treatment that we could. Usually, we listen to the artists' music for a few days and then consult with them before the sessions begin. This allows us to get an idea of what the music is before we go into the studio. Otherwise we wouldn't be producing anything; we'd just be sitting there pushing buttons. We want to hear the songs the way the group performs them in that raw stage and get the chance to refine them right along with the band.

MR: Were all the songs for the Crosby, Stills and Nash album written outside the studio or was there some writing done in the studio?

HA: Most of the songs were written prior to coming in, although Graham finished a couple of songs in the studio. He wrote "Cold Rain" in forty-five minutes. He played and sang it in one take that evening. That cut is on the album.

RA: You see, during the sessions, Howard and I tried to create an atmosphere where David, Stephen and Graham wouldn't be so conscious of the recordings being made and which would allow them to concentrate on the music. We did have to stand on our toes a lot to keep everybody happy. They might be rehearsing a song in the studio one minute and suddenly come up with new changes that could turn into a master recording. We had to be sure while working that fast that those "demo" tapes would later on be of a high enough quality to be used.

HA: Several times while we were making a twenty-four track edit, one of the artists would nonchalantly walk out into the studio and we instantly

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had to go from editing to recording. Switching back and forth did get nervewracking. We had to take everything into consideration, even the jokes and puns in between.

MR: Then how much creative freedom did you have with the studio equipment?

RA: Enough, I assure you, but remember creativity is fine only as long as you can record the music quickly and efficiently. These sessions were complicated by the fact that Crosby, Stills and Nash have their own individual ways of recording. For example, Stephen likes to use headphones for singing harmony background vocals while David and Graham like to use speakers, so we had to have these separate cue systems set up at the same time.

It *was* a harder album to record than most in that all the vocals with the exception of two or three were cut "live." There was very little overdubbing. Graham singing his ballads while playing piano in the room or Stephen singing and playing piano or electric guitar made it very hard to get a good, clear sound. At one point we were recording acoustic piano, electric guitar, bass, drums and a "live" vocal all at the same time—with a P.A. system. That's not normal recording technique.

MR: What is your definition of a "live" sound?

HA: To me, a "live" sound is very boomy, ambient, very open. We didn't necessarily try to get a "live" sound on this album although we'd gain "space" with that even if we'd lose some clarity. But we didn't try to make things sound unnatural either—where everything was close-miked and dry and we'd lose a lot of presence. We recorded the voices and instruments the way we heard them being sung and played in the studio.

RA: For a Crosby, Stills and Nash album we wanted a very special sound. So most of the effects are very subtle and most of the attention is placed on the vocals. You have to listen to the record several times before you begin to pick up some of the subtleties, and I think that's one of the reasons this record is so good.

MR: Did you use specific approaches to miking during the sessions?

RA: First of all, the type of microphone is not so important as the placement of the microphone and the way it's recorded, although to get distortion we did sometimes use a low-level microphone on a high-level instrument. Secondly, we varied our miking techniques drastically depending on the nature of the song.

HA: For an acoustic sound, we recorded the piano with the lid off, placing the microphone high above the piano so that transients had a chance to develop. Obviously, when we had electric guitars and a P.A. system in the room we then had to mic the piano close and put a cover on it.

RA: Most of the piano on the album was recorded on a seven-foot Steinway in Studio C—the small room at Criteria. We wanted the piano to sound like a nine-foot and about as still as an elephant so we had to embellish it with equalizers and limiters to make it sound bigger.

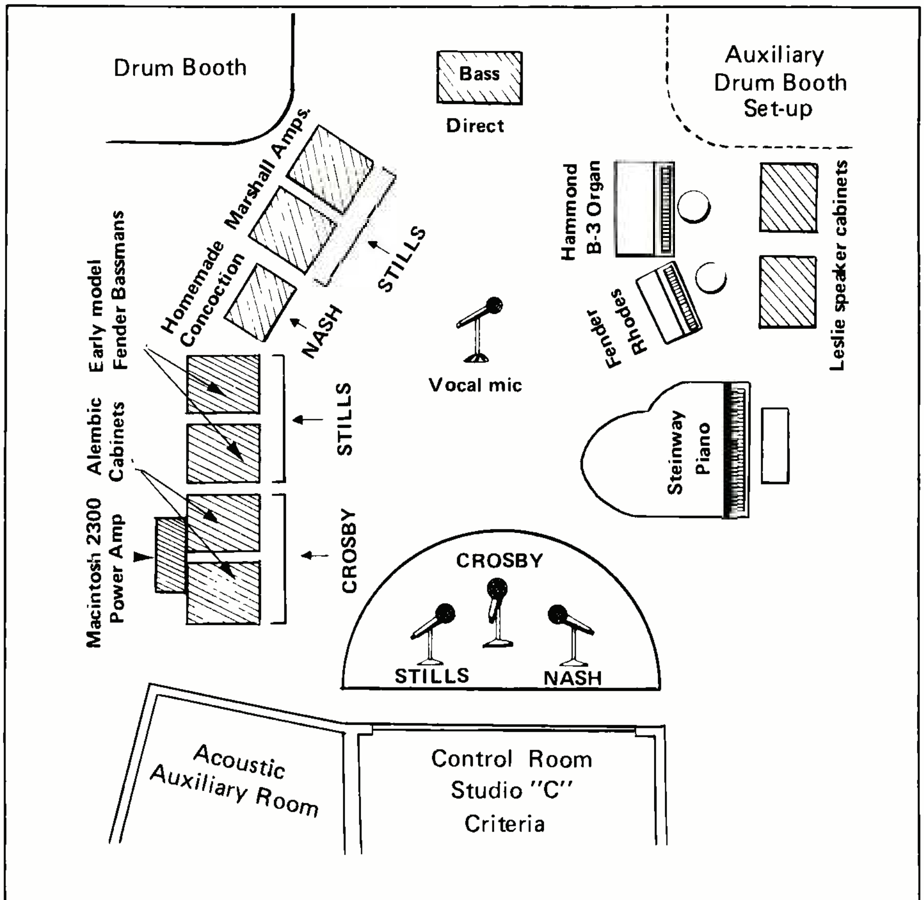
The technique for recording organ entailed a three-mic set-up instead of the normal two set-up. We miked the organ top in stereo and the bottom in mono and we spread that across two tracks to get a very deep perspective when we sank that back into the mix.

MR: What do you think highlights the electric guitar sounds on this particular album?

HA: On "I Give, You Give Blind" Stephen set up with a colossal P.A. system with Altec A7 speakers, and as he played we used a digital delay unit sending a very long delay—350 milliseconds—so he heard both the original signal and the delay. This caused him to phrase his guitar part in a very different place than he normally would, achieving a very exciting guitar feel. We used the delay just to record and once he had the solo down, we took the delay off and kept only the original signal. Overall though, there really isn't that much electric guitar on this album. Just a lot of piano and acoustic guitar. We didn't use many special effects for the guitars other than EQ.

But we did go for effect with the vocals through microphone types. For that hard rock 'n' rolly effect on Stephen's voice we used a Shure 546. Other songs ranged from a Neumann U-87 to a Sony C500 to an AKG 414.

RA: We recorded the bass by running it (in a very delicate fashion) direct through a limiter. If you don't get into a set way of using a limiter with, let's say, a 12:1 ratio, and instead vary to 20, 12, 8 and 4 for varying degrees of sound, it can really do something wonderful to the sound of the bass.



STUDIO LAYOUT (Crosby, Stills, Nash)

MR: Compared to the first Crosby, Stills and Nash album, there are a lot of sessions players on this album.

HA: Yes, they included Chocolate George Perry, Tim Drummond and Jim Haslipp on bass, Russ Kunkel and Joe Vitale on drums and Craig Doerge on piano.

RA: This is the first album where Crosby, Stills and Nash have strings. On Graham's songs, "Cold Rain" and "Cathedral," there is a small section with cellos and violas. On "I Give, You Give Blind" there are violins as well. It's a nice addition to the Crosby, Stills and Nash sound because it fits in so well with the acoustic guitars and piano. But the strings are not very obtrusive; they're very subtle.

MR: It seems like you recorded in a number of different rooms. How many did you mix in?

HA: We did record at the Record Plant and then in rooms A,B and C at Criteria, sometimes working in any two rooms at the same time. But we mixed in Studio C at Criteria. Occasionally though, while someone edited in one studio, someone else was doing a guitar or vocal overdub in another. Ron and I often split up to get the work done because we've worked so closely together over the years that we do almost everything alike.

MR: What type of monitor speakers did you mix on?

HA: We mixed on the regular Studio C monitor system which is, I believe, a bi-amped system—two fifteen-inch JBLs powered by McIntosh 2300 and 2100 amplifiers. It includes a crossover that Criteria has developed. Then we used flat, untouched JBL 4311s and small Orotones. We switched back and forth between the three to get a happy medium so we knew the music would sound good no matter what outside system it's played on.

MR: What did you spend the most time on during the mix?

RA: We took a great deal of time with echo. There is at no point on the record less than two separate chambers in use. At times there are four or five different systems—delay lines, "live" chambers, EMT plates, AKG chambers and tape loops all happening simultaneously. That gives the record its depth. Many people today will use a single chamber and put all the vocals and most of the instruments through that, which will tend to muddy the chamber and not give you a clear, bell-like sound.

MR: Who made the decisions on the final mix?

RA: Whenever you mix an album that includes more than one producer (this one had five), it becomes a question of who has final say on what. Usually the songwriter had a specific idea of how he would like his song to sound. But if the other four disagreed, the majority ruled regardless of who wrote the song. It's really a matter of taste. You can have an incredible mix that everybody thinks is great except for one person who has a different concept, and sometimes that can slow you down.

HA: Nevertheless, everybody wanted this album to be their best effort, and they felt that no one could do it any better. With artists like Crosby, Stills and Nash the ego thing is involved, so there was a lot of mental tension. But everyone bent over backwards to try and please everyone else because, after all, this is a reunion album and it's been many years past since the last one.

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ARTISTS' PROFILE

By Gil Podolinsky

Having been involved in recording for roughly sixteen years, Chick Corea came to the public's attention through the acceptance of his quartet version of Return To Forever—featuring Stanley Clark, Al DiMeola and Lenny White.

After doing three shows in two days—in two different cities, on three hours sleep—Chick armed himself with a cold pastrami sandwich and discussed his views on recording, the various transformations of RTF and his music.

MR: You've been involved in the recording process for some time now. Could you take us back to the beginning? Your first recording?

CC: Well, I made my first record with Mongo Santamaria in either 1961 or '62. I think it was on Columbia. It didn't sell that well. That era is kind of vague for me now. Then in 1963 or '64 I made two or three enjoyable records with the Blue Mitchell Trio—Blue, Junior Cook, Gene Taylor and Al Foster. It was a really nice Be-bop band, all Horace Silver alumni. We recorded it at Rudy Van Gelder's [studio in Englewood Cliffs, N.J.]. I'd learned solos from the records of other people and I was really excited because here I was on record, and on a *jazz* record at that! So, I'd take it home and listen to it over and over and when my solo came up I'd just listen and go "whew!"

MR: Were those early things two track, four track . . . ?

CC: For the longest time when I was a sideman I paid no attention as to what went on in the control room. I couldn't care less at that point, even when I started doing my own records. When I did *Tones for Jones Bones*, my one and only on Atlantic, I didn't even mix it or know whether we used two tracks or twenty tracks. I just wasn't into it in that way. We'd just do one or two takes, no inserting, overdubbing or anything. We just took the better of the two takes.

MR: When did you actually start paying closer attention to the studio recording process?

CC: When I started working with Manfred Eicher at ECM.

MR: Before we get into that period, I don't want to overlook that aspect of your career as a sideman.

CC: Right. I was doing sessions with people like Herbie Mann, Stan Getz, Willie Bobo, Elvin Jones, and of course, Miles Davis.

MR: You played with Miles in the late '60s, most notably on the albums *In A Silent Way* and *Bitches Brew*, both of which represented quite a change for jazz. Up to that point, jazz musicians were thumbs down on the electric influence of rock until, as the story goes, Miles went electric and the jazz community took it as the green light to go ahead and experiment.

CC: Yeah, it was like the next logical step for jazz to go. It was Miles who introduced me to the electric piano. I just showed up one day and there it was and Miles said "play that."

MR: Backtracking, what happened, in terms of a solo career, after the one Atlantic recording?

CC: Well, I signed with United Artists, where one of my records was released on the Blue Note label, and I did two or three for them. Then there was a period, just prior to ECM, where I had no record contract at all. It just sort of petered out; each release sold only three copies. So for about two years I just played without making a record. Then I met Manfred.

MR: How'd that happen?

CC: Well, he came to hear Circle, the group I had at the time which had Anthony Braxton, David Holland and Barry Altschul. He introduced himself and asked if I wanted to make a record. I said OK and made the trio album, *Arc*, followed by *Piano Improvisations 1 & 2*.

MR: What was it like working with Manfred?

CC: It was different, it really was. He was the first record producer [whom I worked with] whose sole intention was to record the *music*. He was into the overall product—making a beautiful piece of music come out on the record.

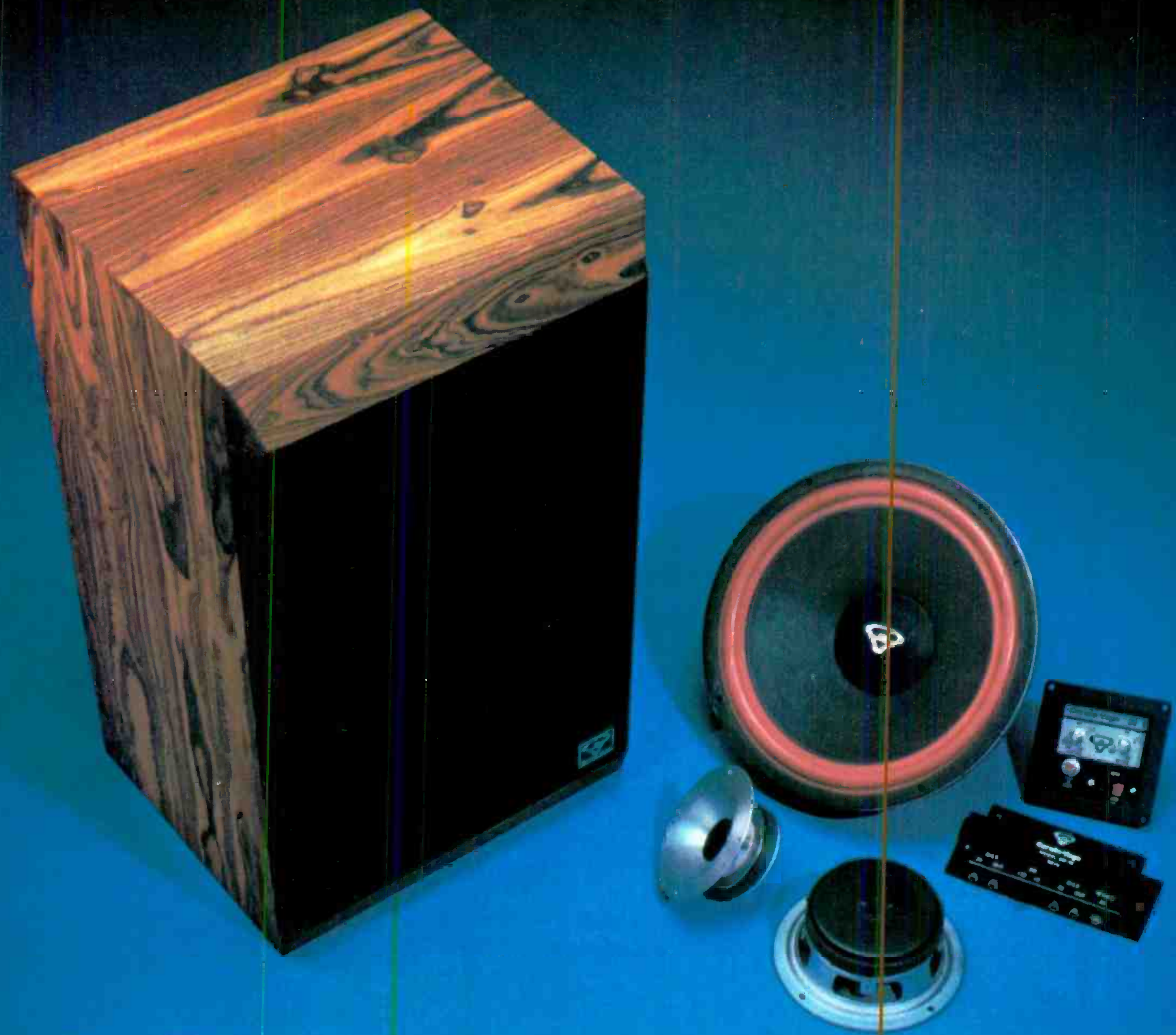
MR: At that time, [late '60s to early '70s] it seemed artists had a hard time relating to producers who, for the most part, were more sales oriented.

CC: They [the producers] were, kinda yeah—a bit detached. Plus, I was at a point in my life, before I met Manfred, where I wouldn't communicate with people. I'd make them stay away. I'd say, "I'm going in to do my album now, leave me alone, I'll give ya the record when it's done." But with Manfred I had some communication.

MR: Manfred produces all the records on his label, ECM. When did you start getting into producing?

CC: Essentially with the Piano Improvs. and the first RTF on ECM. I feel that I produced them. Manfred really handled the environment and the sound on the record. I did the mix, put the whole thing together, the album cover, etc. Those were the first. No, wait a minute, wait a minute. I did produce Tones. I wrote the music and put the album together. I didn't mix it, though, I didn't know about mixing then. I figured you just give it to someone and he did it. But after meeting Manfred I saw what it was to balance tracks and how different things can sound as a result.

MR: How do you record now? Do



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you prepare before hand or wait 'til you're in the studio?

CC: Well, everything starts with the obvious knowledge that it takes money to record. So, going from there it makes sense to be really prepared. Everything is written out, rehearsed and taken to the point where it's really popping before I go into the studio. When I finally do go in I want to be able to take tracks down immediately. Although there are arguments pro and con, I personally like first takes. I know it's possible to do something over and over and not lose the feel of it, but I like the feeling of doing a first take so I strive for a first, or maybe a second, take.

MR: You've changed your style often in your career, and apparently have again with *Musicmagic*. What brought about this latest change?

CC: I've just radically changed my style and I'm so pleased, really so pleased. It's the evolution of trying to attain a combination of keeping my music really straight, the way I want it, very esthetic and beautiful, and at the same time get across a really good feeling to a lot of people. I don't want this band's music to come across esoteric in any way. It's kind of a challenge. I fell prey in the last RTF to certain things that were being agreed upon by a lot of people around me, but I just felt that for the few years we were doing it, it wasn't right for me.

MR: What things in particular?

CC: One thing was playing at a volume that was insane, *absolutely insane*! It's like inhuman to do that. But there was this agreement around me to do that and that it was a good thing.

MR: Yet the rock aspect of your music is what brought you to the forefront and made you popular.

CC: Right, but the effect the volume had on people was that because we played so esthetically nice, they put the volume thing aside. However, by the end of a performance, the audience was *exhausted* from keeping the barrage of sound off them just to be able to hear the esthetics.

MR: Is that why you've reverted to more of an acoustic sound?

CC: Well, kind of. Now I have a brass section and a Steinway grand piano on stage, but I still play and utilize the synthesizers, electric pianos, etc. But to me, we're just playing music that's more music, the purpose *not* being to create hysteria among people, just the opposite—a kind of calm. It's OK for an audience *not* to stand up and go crazy, OK just to get caught up in the flow of it and go home fulfilled.

MR: One noticeable thing when comparing the two bands is that the current one allows the music to breathe more and the listener to follow the voice leading.

CC: That's a good way to put it. Turn the volume past a certain level and you can *forget* about voice leading, and, as a result, some of the beauty of the music is lost. I actually started thinking in terms of octaves, fifths, unison, etc., for at that volume level that's the only thing that would work. For a composer who wants to say something, that gets boring fast.

MR: I conclude then, that the records from the quartet period were not the most satisfying that you've ever made?

CC: Yeah. On the quartet records I can pick out choice moments from all the records, really nice stuff. However, it was sporadic, nothing continuous from first to last cut. On *Romantic Warrior*, for example, there's a piece called "Duel Between the Jester and the Tyrant," which was a really nice rendition in the studio with Al Di-Meola playing an incredible solo. That was the highlight of the album. The rest was not so consistent.

MR: How about your recordings since then?

CC: A number of my recent recordings achieved what I wanted. *Musicmagic* was an interesting album because we worked really hard at making it good. We didn't take a lighthearted approach, but actually got into wanting to get things clean and re-doing parts that at first didn't get the compositional intention across. Some of it was really hard work but *Musicmagic* really got across what I had intended it to say. The Piano Improvs. did that for me too, and were very easy to do. I had a very simple intention and design for that. I just went in and did it, heard it back and thought, "That's exactly how I want it to be." The first RTF was the same, really clean and simple. Some records after that were complex, and it was hard for my intentions to get through. The quartet period seemed like I was fighting energy rather than using energy. My first release from that period was *The Leprechaun*, and I really feel good about that record and what I wanted to get across. I really liked *My Spanish Heart* too. Man, I had a lot of fun with that record. Except for the quartet stuff. [Pause.] I'll tell ya, what was great about the quartet was the excitement that would happen in performance. People going nuts was new to me, I'd never experienced that before. I thought, "What is this, you know, you play a phrase and people love it?" Also, with the amount that we worked and toured we became familiar to a number of people. It got to the point where we built a following that would come back. I enjoyed it, but the thing I started not liking was that it wasn't what I was trying to do. I loved that they [the audiences] were loving it, but it wasn't what I was trying to do. *The Leprechaun* was a turning point, *Spanish Heart* a continuance, and with *Musicmagic* I feel [Laughing[that I've gotten back to square one again with RTF.

MR: Aside from already having written the material and rehearsed it, how do you approach going into the studio?

CC: Well, for me it takes a day, maybe two days to get all the mechanics rolling, to get the right EQ, the instruments to sound good, the right mics, etc. What's important for me in the way I record my music is to get the right kind of communication going in the studio among the musicians, which sometimes means using headphones.

MR: I take it then that you normally record "live" in the studio vs. track by track?

CC: I have done just the basics, bass and drums. Some parts of *Musicmagic* were recorded that way because the material called for it and it was necessary in order to achieve the best sound. So, for that reason, the rhythm section tracks will be recorded first and the other instruments added later with separations. I prefer to do it all 'live." The problem comes in separating the instruments. You can go one of two ways. If you want that distinctive separate effect, you end up having to go in for overdubbing. If you opt for the other where you get the group in the studio, set up a couple of ambient mics with nothing direct or close miked, you end up with the sound of the band. I'm going to use the latter approach when I record my orchestral music as well as my string quartet and piano music. I'm definitely not going to have everything miked right up to your face.

MR: Any other studio experimentation you're contemplating?

CC: Yes, I'd like to make a direct to disc recording. It sounds better obviously since you're saving one genera-

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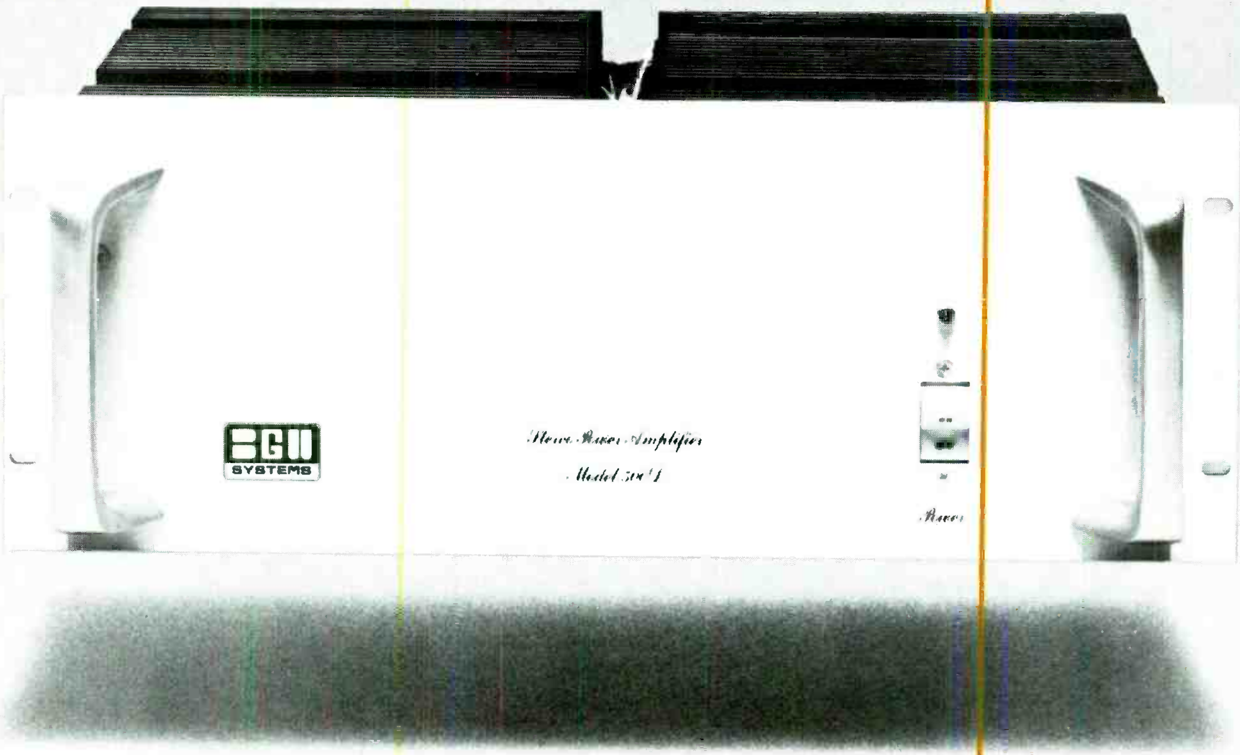

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tion, but they don't last as long. Maybe I should just get a disc in the studio and play into it without a mic. [Laughs.]

MR: Do you rehearse in the same studio/room in which you record?

CC: No. Unless you're doing a remote or own the studio the time/money factor doesn't permit it.

MR: Do you always use the same studio(s) *a la* Manfred Eicher?

CC: We move around. I have used Caribou Studios [a studio in Colorado] for the last two RTF albums, *Romantic Warrior* and *Musicmagic*, but I didn't for my solo records.

MR: What do you look for in a recording studio?

CC: I like Caribou because of the environment, i.e., the outdoors, and having the studio to ourselves everyday. In general, it's been my experience that all professional studios are the same in that they all have the same equipment. What makes or breaks a studio is the maintenance man. He's really the guy who runs the place. It's a drag for a guy like me who has got his stuff prepared and is ready to go, to stop for an hour and half to find out why there's a mysterious hum. There are a couple of guys who designed and maintain the equipment at Caribou and will assist our engineer on our projects. They're very service oriented and so on top of it. That makes a *world* of difference.

MR: Do you use the same engineer from *LP* to *LP*?

CC: Lately I have been. Bernie Kirsch did *The Leprechaun, My Spanish Heart* and *Musicmagic.* He's also on tour with us now mixing at the board, which is really a big plus. Bernie's not stuck in the typical engineer's mold. He's a very musical guy and the technology of engineering comes very naturally to him. He's very practical and knows how to apply what he knows to what we do. I have a good relationship with him.

MR: Do you have any standard/favorite mic set-ups in the studio?

CC: Miking techniques vary. Stanley's Alembic bass sounds *unbelievable* direct, so we always take it direct. The Fender Rhodes is a very hard instrument to record. Quite honestly, it's basically an unfinished instrument. The idea is great and the sound is very appealing, but the mechanics *is* [sic] *a mess!* I wish someone would do something about it, like Fender for instance.

MR: You mean CBS, who bought out Fender in the late '60s.

CC: Right, I have the new stage models. They're noisy and break down a lot. Rory Keplan, my equipment man, rigged up a nice little exterior pre-amp which keeps it from breaking up—makes the sound a little cleaner. I use it both in the studio as well as on stage. I'm using this Yamaha 16-channel mixer where I use eight or nine inputs, Gayle [Gayle Moran, lead vocalist and keyboardist in RTF] four or five, then I have a channel where I feed an MXR Digital Delay through all the keyboards. That MXR is an incredible little toy. It doubles, flanges—does all sorts of things. I use a couple of faders to bring it through. I use lots of toys in the show. I'm definitely going to use the MXR in the studio. I haven't gotten so excited about an electronics toy in a long time, like let's form an MXR band! I'm through with echoplexes. Rory has my sound the best it's ever been, basically because he's got great equipment. I have three Crown amps split bottom, mid and high, powering some small Kustom speakers. This is my set up both in the studio and on stage. I run the Rhodes both direct and miked, then mix the two. The synthesizers—Moogs, Arp Odysse—sound great direct. I just bought an Oberheim 8-voice that I'm sure will sound great direct because they're designed for that purpose. I used to use a Yamaha organ, but I replaced it with the Polymoog. The Polymoog is a great performing instrument—it has pre-sets, gets a nice string sound, etc., but, for me, is not a durable road instrument. It needs work to keep it consistent from night to night. I use a Steinway grand both on tour and in the studio. This is the first time I've ever rented one for a tour. When we would record in New York I would go to Steinway on 57th St. and choose a nice seven foot B or a nine foot D and have it taken to the studio. I like Steinway a lot, and I just bought a twenty year old Bosendorfer which I use at home. It's really quite different from a Steinway.

MR: What mics do you use on the Steinway?

CC: I'm afraid I haven't gotten into differentiating between mics yet.

MR: What about track assignment?

CC: What do you mean?

MR: Well, like putting the lowest frequency instruments like bass and bass drum on the outer tracks of the tape so as not to lose the highs.

CC: I just became aware of that kind of stuff in the last recording. I'm starting to spend more and more time in the control booth. Bernie really helped me out a lot; I learned a lot about recording from *Musicmagic*. I don't know how he assigned it, but I'm aware now that that kind of thinking exists.

MR: Was *Musicmagic* a twenty-four track recording?

CC: Well, they only have twenty-four tracks there [at Caribou], and it barely made it. We had to use my Yamaha 16-channel, and like that's in addition to a 36-fader board, and we actually did eight sub mixes. We just separated *everything*.

MR: Were you pleased with the overall sound of the record?

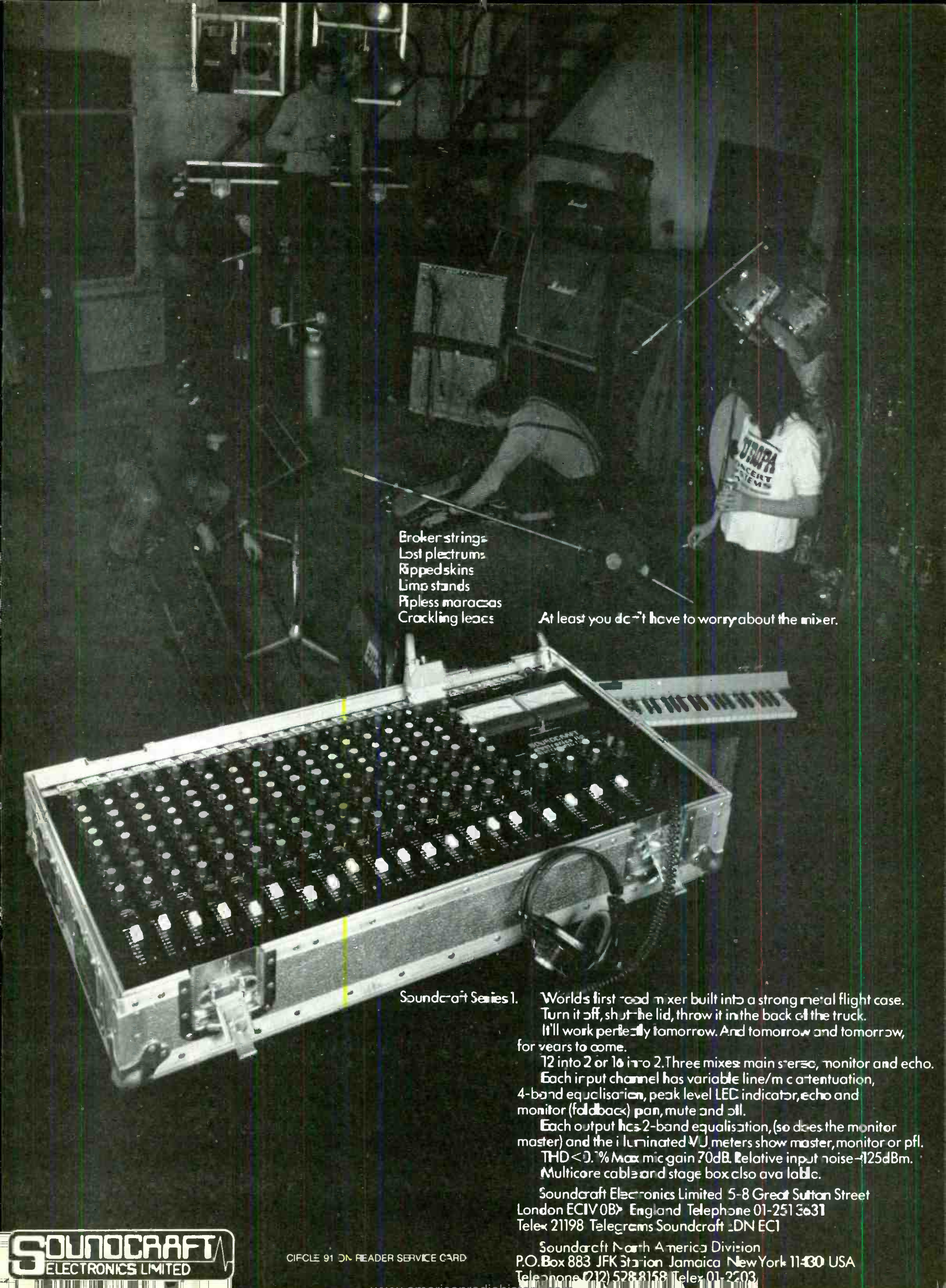
CC: With this stereo recording I actually got into the sensation of sound with timbre, melding and mixing. All the subtleties are just right. We spent so much time on it and it worked. The crazy thing was that these two guys from CBS wanted to do a quad mix of it and told me they'd do it in three or four days. I said, "What, are you kidding me man? Only three or four days when me, Stanley and Bernie spent two hard weeks all day long?" They did it and it didn't sound bad, but . . .

MR: How long does it take for you to do an album?

CC: On *Musicmagic*, I spent two weeks composing, three weeks rehearsing, two weeks doing basic tracks, one and one-half weeks overdubbing and cleaning, and finally two weeks mixing. That's about average, or maybe a little longer than the norm. The thing is, every time I work on an album, I always say that next time I'm going to allow for more time for each phase, because each phase is an entity unto itself. I love to compose, but I always feel that I don't have enough time.

MR: In closing, your colleague Gary Burton differentiates between recording and performing, calling them two separate mediums. In short, he feels that recording is like a painting, so perfection is desirous; whereas mistakes, etc., "live" often enhance the performance. Do you agree?

CC: Well, that's a simple view to understand, but no, I don't make the same differentiation. I try to make my records sound like a performance and my performances like a record. So, the record should sound both perfect and "live" and the performance should sound good, with the music perfected. We made a few steps in that direction with this tour of the "new" RTF.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Ambient Sound

BY LEN FELDMAN

In Search of a New Distortion

If you have been following the equipment lab reports written jointly by Norman Eisenberg and myself for *Modern Recording,* you are no doubt aware, by this time, that our approach to product evaluation involves both lab measurement and extensive listening and use tests. We believe that audio equipment cannot be evaluated strictly on an ears-only basis, nor can its virtues or failings be completely defined by "bench measurement" alone. When our ears detect differences between two products that measure the same in every way, I have always maintained (and I think Norman Eisenberg will agree) that if our instrumentation and measuring techniques were fully adequate, such differences also should show up in one form or another on the test bench. For, while audio engineering has made great strides in the last decade or two, it is clear that there are many things we still do not know about how our hearing works and how our audio electronic equipment reproduces sounds.

A very good example of this is the recent attention that has been paid to a hitherto "unknown" form of distortion known as TIM (Transient Intermodulation Distortion). Unheard of as recently as five years ago, this elusive form of distortion is now occupying pages and pages of print in scholarly engineering papers as well as in audio consumer publications. Methods have been suggested for detecting the presence of TIM in an amplifier or preamplifier and work is now going on to both quantify this new sonic abberation and to reduce it in newer audio amplification products.

The Feedback Tradeoff

Perhaps the greatest contribution ever made to the science of high fidelity audio amplification was the introduction of negative feedback in audio amplifier circuits. Back in the early days of tube amplifiers it was discovered that, in an amplifier, if a portion of the amplifier's output signal was applied or "fed back" out-of-phase with the input signal, distortion components created by the amplifier itself could be cancelled. Included in this definition of "distortion" was non-uniform amplification of all audio frequencies, so that measured amounts of negative feedback, in addition to "cancelling" harmonic distortion and intermodulation distortion, also tended to improve overall frequency response of the system. At first it seemed that the more feedback one applied to an amplifier the better, and that the only sacrifice one had to make in return for increased feedback was increased gain (or an increase in the number of amplification stages) required to make up for the reduced amplitude of output signal caused by the increased negative feedback. It was soon discovered, however, that there are limitations in the amount of feedback that in practical terms can be applied to an amplifier. Since every amplifier exhibits "phase shifts" at some low and high frequency extreme, what starts out as negative feedback at one frequency can turn around and become *positive* feedback at a non-audible frequency and can cause the amplifier to become an oscillator instead. In vacuum tube amplifiers, feedback amounts in the order of 20 to 30 dB were typical. But feedback of 20 dB was enough to reduce harmonic distortion by a factor of ten to one, so that if a tube amplifier without feedback measured 3.0%, the application of 20 dB of feedback reduced that THD to a very respectable 0.3%.

When solid state amplifiers came into the picture, efforts were made to apply more and more feedback. Earliest transistors were fairly non-linear and so amplifiers in their open-loop (no feedback) state exhibited higher distortion levels and earlier roll-off of high frequencies than did tube counterparts. Even as solid state devices improved in linearity and bandwidth, the trend towards more and more feedback continued and accounts today for the incredibly low rated harmonic distortion levels we see in the spec sheets on modern-day transistorized equipment. (Feedback of 60 dB "around" an amplifier from output to input would reduce an open-loop distortion level of 3% to 0.003%!) Bear in mind, however, that if such feedback were, for some reason, momentarily removed, the gain or amplification of the given amplifier would "try" to increase by the same 60 dB—a factor of 1000 to 1 in voltage, or a factor of 1,000,000 to 1 in *power*!

Sine Waves and Music

Traditionally, amplifiers have been tested using steady-state signals such as sine waves or square waves. In the case of a sine-wave signal (or any symmetrical, repeating smooth waveform), the application of ever more feedback to an amplifier (barring any instability due to phase shifts) continues to reduce distortion and "flatten" frequency response. But most of us prefer to listen to *music* rather than sine-wave tones, and music waveforms are anything but smooth or repetitive. Music contains extremely fast transients —rapid rises or "attack times" of some musical instruments require that signal voltages in an amplifier rise in a matter of a couple of microseconds from "zero" to some large value. Further, at the same time that such fast rise-times are demanded of an amplifier because of the fast attack time of one or more instruments, other instruments playing at the same time may require smooth, cyclical or repetitive amplification of their sine-wave like signals.

Enter Transient Intermodulation Distortion

Everything would be fine if signals could be put through an amplifier with zero elapsed transit time. But the fact is that transistors have a finite response time or rise time of their own. In addition, the introduction of such components as capacitors in the amplifying chain or even in the feedback network of an amplifier also introduce phase or time delays. So, under certain conditions, the feedback from the output of an amplifier that's to be applied back to the input takes a certain amount of time to get there. Let's see what happens when a fast transient signal tries to undergo processing in such an amplifier. Suppose the gain of the amplifier without feedback is 50 dB, and that with 30 dB of feedback applied that gain should be reduced to a net of 20 dB. If we feed in a nice calm sine-wave signal at an "easy" mid-frequency of 1000 Hz and an amplitude of 1 volt, feedback has plenty of time to do its thing, and the output voltage will be 20 dB or 10 volts. That ten volt signal, driving an 8-ohm speaker load, will develop 12.5 watts (voltage squared, divided by the load resistance). Assuming an amplifier with a power output rating of 100 watts, no problem.

Now, suppose the signal to be amplified is a very fast transient musical spike (such as might be caused by the attack time of a plucked string instrument). The electrical amplitude of the signal is still only 1 volt, but by the time the signal travels down the circuit, reaches the output, and sends the 30 dB back around to the input, the transient has already come and gone! It's as if this particular signal were stripped of the "benefits" of feedback because the feedback arrives too late. So, the amplifier, to all intents and purposes, tries to amplify *this* 1 volt input signal by a full 50 dB! That means an output of over 316 volts which, if translated to power across an 8-ohm load means 12,500 watts! Obviously, our 100-watt amplifier can't cut it, so it goes violently (if briefly) into overload clipping. Worse, if during the time it clips other more tame signal elements are riding along for simultaneous amplification, they too are buried in the clipped or overdriven portion of the output signal and are altered so that they interact to create what we now call Transient Intermodulation Distortion.

Tube Sound vs. Transistor Sound

As many readers know, there is a dedicated group of audio enthusiasts who have maintained, for years, that tube amplifiers still sound "better" than solid-state amps. But remember, we said earlier that tube amps traditionally used *less* feedback than did transistorized equipment. And less loop feedback means less transient clipping distortion when feedback transit time is too great to do the good it was intended to do. So, perhaps the tube-sound freaks were not crazy after all, eh? Does this mean we must resign ourselves to returning to those heavy, heat producing tube amps of the past? Not at all. Rather, design approaches to modern solid-state amplifiers must be (and are being) re-thought in the light of new knowledge regarding Transient Intermodulation Distortion.

Among the corrective steps being taken in new designs are the application of small amounts of feedback to individual gain stages, rather than one massive amount of feedback applied around the entire amplifier from output to input. Since, as we said, coupling capacitors in the audio chain introduce phase or time delay, new designs are appearing which do away with such capacitors entirely. These new amplifiers are being called D.C. amplifiers—which stands for *both* direct-coupled *and* direct-current and means that such amplifiers, devoid of low-frequency blocking capacitors, can amplify frequencies all the way down to "d.c." or "0 Hertz."

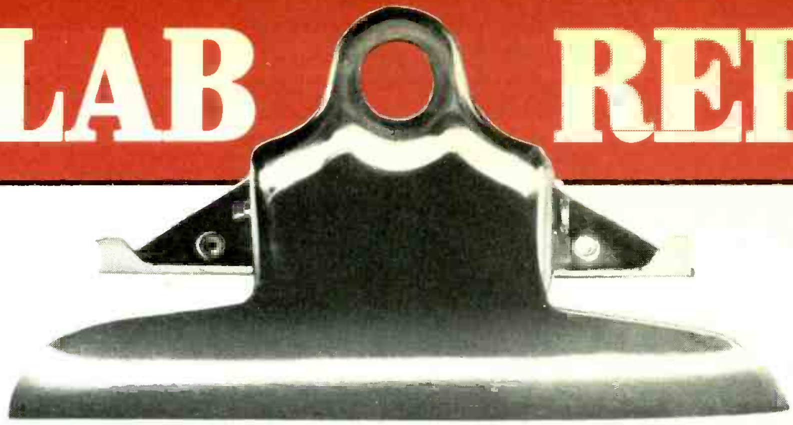
We also mentioned that the "speed" of the transistors used in audio amplifiers bears directly upon the time delay introduced from input to output, so higher speed transistors are being increasingly used in the design of better audio amplifiers and preamplifiers.

In terms of laboratory evaluation, as we mentioned earlier, ways are already devised whereby we can graphically and visually detect the presence or absences of TIM, using standardized test set-ups and procedures. While the industry has not yet decided upon how to "quantify" TIM (it would be nice if we could "spec" TIM as a percentage or a number, as we do with total harmonic distortion, intermodulation distortion, signal-to-noise, frequency response and other amplifier performance characteristics), work is going on right now to establish ways in which TIM can be specified by meaningful numbers and no doubt we will begin to see first mentions of this important form of distortion in manufacturer's literature in the coming months and years.

Towards Better Sound

To paraphrase the opening lines of an old popular radio serial, "Who knows what evil lurks" within our audio amplifiers? If, less than a decade ago, no one suspected the existence of TIM, what other forms of audio distortion still exist when we attempt to reproduce complex musical waveforms? Continued study in the field will surely uncover additional things we never knew about. The important thing is not to become too smug and conclude that we have reached perfection just because our equipment today sounds so much better than it did a few years ago. After all, the congress of the United States, in the mid-nineteenth century, actually seriously considered closing down the U.S. Patent Office because in that very enlightened age "everything had already been discovered or invented."

MODERN RECORDING **LAB REPORT**



NORMAN EISENBERG AND LEN FELDMAN

Dynaco Mark VI Power Amplifier

General Description: Dynaco's Mark VI is a single-channel (monophonic) power or basic amplifier utilizing vacuum tube circuitry and intended for professional applications or in any sound system in which there is a need for a high-powered, single-channel basic amp. It fits standard 19-inch rack mounts. As is generally true of high-powered tube-and-transformer amplifiers, the Mark VI runs relatively "hot," and adequate ventilation is essential, particularly when the unit is to be used for extended periods of time.

The front panel contains an amply sized VU meter calibrated from −20 to +3. Also provided are adjustments for bias, meter range and input level. These are all screwdriver adjustments. The range switch has four positions: three settings handle power output levels, and one setting may be used for observing bias, all on the VU meter. The front panel, additionally, contains a fuse-holder for the speaker connected to the amp, and the power off/on switch. At the rear are the input and output connectors plus a line fuse and the AC cord. The inputs offer a choice of flat response or a low-frequency rolloff (6 dB per octave, −3 dB at 70 Hz). The low-filter input is a standard pin-jack. For flat response the user has a choice of input jacks: one is also a pin-jack; the other, an XLR connector.

For connecting a loudspeaker there's a terminal strip with the conventional designations of 16, 8 and 4 ohms, and a common terminal.

Test Results: Performance specs for the Mark VI are just about what one would expect as a technical description of a professional-grade basic amplifier, and in MR's tests the Mark VI met or exceeded all of them. Rated for 120 watts continuous power, the amplifier produced in our lab 135 watts. Response and harmonic distortion were easily confirmed; IM distortion at 0.3 percent was significantly lower than the 1% claimed. Hum and noise were well down at the −95 dB mark; damping factor at 16 was a shade better than the 14 spec'd; input sensitivity for rated output was right on the nose at 1.6 volts.

Of special note was MR's observation of the manner in which the Mark VI handled overload ("clipping"). Instead of the abrupt "chopped" waveform tops, this amplifier clipped in a "gentler" manner, with gradually rounded waveform crests. This effect has been variously described as producing "softer" or "warmer" sound at extremely high power levels.

Circuitry in the Mark VI is essentially that used by Dynaco in past tube amplifiers, with the output transformer taps tied to the screen grids of the type 8417 (pentode) output tubes. Overall circuit design is straightforward, inherently linear, very stable under all load conditions and with a minimum of phase-shift. Full rated output power is available at any of three load impedances (4, 8 or 16 ohms).

MR's testing staff feels that the Dynaco Mark VI can be enthusiastically recommended for professional sound applications as well as to home audiophiles.

General Info: Dimensions are 19 inches wide; 8 ¾ inches high; 10½ inches deep, with the front-panel handles adding an additional 1½ inches to the depth. Weight is 55 pounds. Price is $649 in a factory-wired form; $425 as a kit.

Individual Comment by L.F.: In this day of solid-state circuitry, Dynaco explains its choice of vacuum tube circuitry thusly: "Vacuum tubes are most often favored (by the professional sound installer) in their ability to withstand enormous physical and electrical abuses—conditions frequently encountered in discotheques, public address and musical instrument applications. Its rugged construction . . . permits continuous duty at full power, while maintaining safe operating margins."

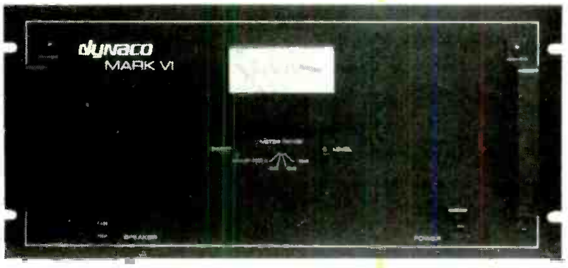


Fig. 1: Dynaco MK VI: Front panel view.

All true, I suppose, but my "return visit" to a high-powered tube amp caused me to reexamine certain often-overlooked truths about amplifiers in general. Consider, for instance, the unit's power response. Dynaco claims full power *within 1 dB* over a frequency range that extends "only" from 20 Hz to 15kHz. That "1 dB" qualifier means that the amp is guaranteed to produce only 95.32 watts at the frequency extremes. As many readers know by now, if this amplifier were sold for home consumer use, and *not* for "professional" applications, the Federal Trade Commission might be pounding on Dynaco's doors asking them to de-rate the amplifier. Yet, you and I know full well that a difference of 1 dB in power is of no consequence audibly!


Fig. 2: Dynaco MK VI: Overhead view with vented cover removed.

Or, consider the total harmonic distortion. Dynaco's rated THD for the Mark VI is a "full" 1.0 percent (actually, the amp does much better than that at most frequencies that really matter). But how many "specification hungry" audiophiles would be inclined to buy an amplifier rated at 1% THD in this day and age?

Well, I have news for you—news that won't come as much of a surprise to many audio-minded who have already succumbed to the lure of tube amplifiers. The Dynaco Mark VI is a damn good *sounding* amplifier! A few of the reasons why it is may be of interest to those solid-state adherents who think that any THD above 0.1 percent is "low fi."

For one thing, the circuit in the Mark VI employs only about 20 dB of overall loop feedback. Higher levels of feedback are now known to increase the chances of high transient intermodulation distortion—so, if you can reduce the feedback required, you do improve transient or musical response of an amplifier, all other things being equal.

Then too there's the way a tube amplifier of this type goes into clipping (explained in our Test Results and illustrated in the 'scope photo). To me the difference between this kind of clipping and that encountered with solid-state circuits is really audible, and I suspect that, the FTC notwithstanding, some "non-professionals" who can afford two of these amplifiers may yet be "caught" using them in home stereo systems.

We have plotted harmonic and IM distortion against power output (see Fig. 3), and harmonic distortion ver-

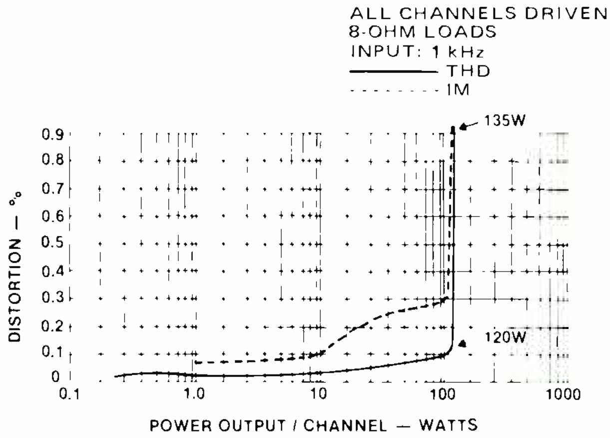


Fig. 3: Dynaco MK VI: Harmonic and intermodulation distortion characteristics.

sus frequency for 120 watts output (see Fig. 4). The latter curve shows why Dynaco had to "hedge" a bit on the 120-watt claim. Had they claimed 70 watts output, the amp could have met its maximum 1 percent THD figure right up to 20 kHz. That might have made the FTC happier, and enabled Dynaco to promote the amplifier as a "consumer audio component." But of

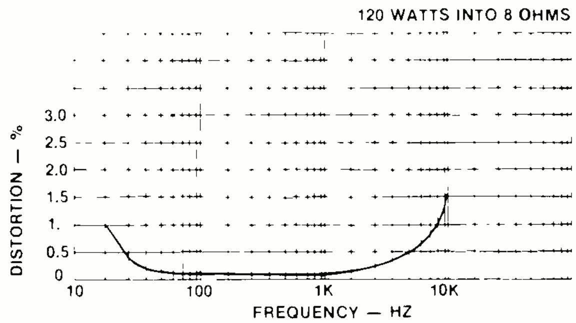


Fig. 4: Dynaco MK VI: Distortion vs. frequency.

course it would not have made the Mark VI sound any better, or last any longer!

So, dear reader, if you are a professional, have no fears about using Mark VI in all manner of applications while remaining confident about its ruggedness and reliability. And if you happen to be an audiophile who seeks a mono tube amp of this rating, perhaps you can get one of your "professional sound" friends to smuggle one into your home.

Individual Comment by N.E.: The Dynaco Mark VI raises some interesting questions about amplifier specs in general, and the possibility of an overemphasis in recent trade literature on certain performance areas while slighting others that may be at least as significant in terms of a unit's "listening" quality and its ultimate ruggedness and dependability. Regardless of what you think of the FTC regulations on amplifiers, if you have any kind of an ear for musical sound you probably will agree that the Dynaco Mark VI is an eminently clean, honest amplifier. It can handle any kind of load; it appears to be built like the proverbial "battleship"; and it responds to the multiple demands of complex program material with authority and ease. Vis-a-vis comparably powered solid-state amps, it is heavier and bulkier. It also does

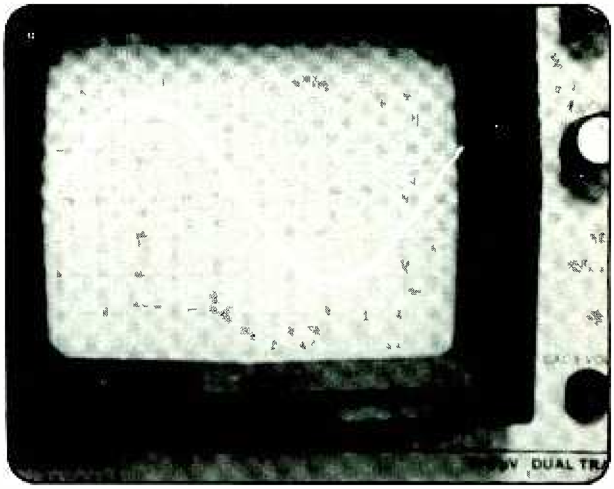


Fig. 5: Dynaco MK VI: Clipping in this tube amplifier is altogether different from that observed in solid state power amps (see text).

require some kind of air circulation; I would imagine for long-term use a small fan might be a good idea unless the unit is installed so as to be pretty much in "free air." In my view, a pair of these monsters would make an admirable stereo rig. One Mark VI could find various applications, apart from obvious professional uses, in such setups as a bi-amplified system or to run a "center-channel" speaker in a stereo system.

DYNACO MARK VI POWER AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTICS	LAB MEASUREMENT
Power output, continuous	135 watts at 1 kHz
Power response	within 1 dB of 120 watts. 20 Hz to 15kHz. less than 1% THD
Frequency response	below 10 Hz to 41 kHz. +0. −1 dB
IM distortion (60 Hz and 7 kHz. 4:1)	0.3% at 120 watts output
Hum and noise	95 dB below rated output
Input sensitivity for rated output	1.6 volts
Damping factor	16 at 8 ohms
Power consumption	275 watts. no signal: 420 watts. full output.

Sound Concepts SD-50 Electronic Time Delay

General Description: The model SD-50 from Sound Concepts, Inc. is described as a "hall simulation system" designed to add to a sound system specific acoustic qualities that are normally contributed by a hall: a certain amount of delay of sound, the addition of echo or reverberation and a contouring (rolloff) of high-frequency content. All three functions are offered as variables in the SD-50, under the control of the user. The device may be patched into a sound system in several different ways, depending on the kind of equipment presently owned, although generally it is assumed that the use of the SD-50 also involves the setting up of "rear channel" speakers powered by a suitable rear-channels amplifier in addition to the normal front-channel setup.

In addition to its applications for listening over loudspeakers or headphones, the SD-50 also can be used in tape-recording in several possible ways. For instance, patched into either channel input of a stereo recorder, the SD-50 can be used to reprocess mono recordings (or the monophonic audio portion of TV programs) for added "spatial enhancement" or "simulated stereo." Used with a multitrack recorder that has add-on dubbing facilities (sound-on-sound; sound-over-sound; simulsync, etc.) the SD-50 can be used for mixing delayed signals with the original material for re-recording. It can be patched into a "processor loop" at a mixer to add ambience while

Sound Concepts SD-50: Rear panel layout.

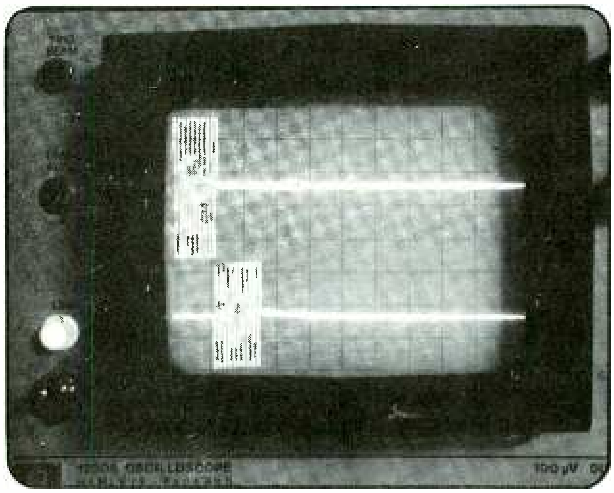
recording, and it also can be applied to four-channel recordings by processing their rear channels.

A sophisticated device, the SD-50 employs complex circuitry which, according to the manufacturer, relies on integrated circuit techniques to keep the size of the unit compact. "The tens of thousands of separate components needed to create an equivalent of the SD-50," says the company, "would cover the floor of a large living room." As presented, the SD-50 covers less than 100 square inches (it is available in two versions—one in a walnut housing, the other in a standard 19-inch rack mount).

The front panel contains five control knobs. One is a continuously variable control for delay, marked in increments of five from 5 up to 50 (milliseconds). Next to it is a continuously variable control for reverb (marked from 1 to 10). A "mode" switch has three positions: ext, stereo and mono. Another switch marked "rolloff" has four positions: flat, −3, −6 and −9 (dB). The level

Sound Concepts SD-50: Internal layout.

control, finally, is continuously variable, and is marked 1 to 10. Above the mode switch is a peak LED indicator. In the "ext" position of the mode switch, four-channel material passes through the SD-50 with no processing, and the level control is out of the circuit. In the "stereo" mode, left- and right-channel information is processed and sent to the "rear out" jacks (on the back of the SD-50). In this mode, there is no mixing of left- and right-channels other than that determined by the user with the reverb control. In the "mono" mode, left- and right-channel information is mixed and is passed successively through both of the device's delay chains. This mode thus doubles the delay introduced and the readings on the delay control knob become effectively 10 to 100 (milliseconds). The SD-50 has no off/on power switch of its own, and its AC line cord is recommended for connection to a "live" outlet (such as the unswitched outlet on other equipment) so that the unit is continually powered.



Sound Concepts SD-50: Five millisecond rear channel delay.

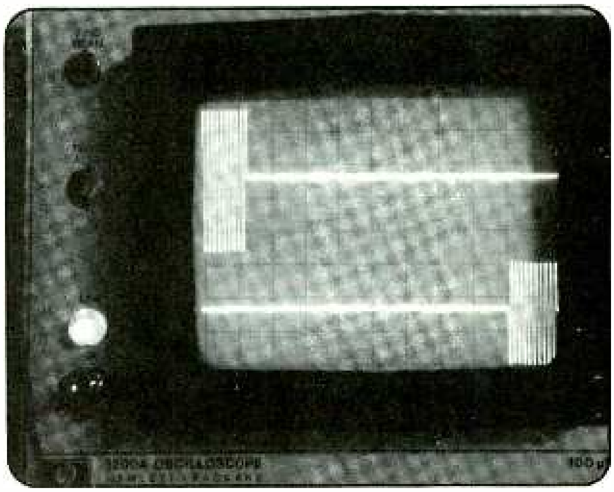
Signal jacks at the rear include stereo pairs of phono jacks for front channels in and out, rear channels in and out, and the two-channel mix output. The rear also contains two screwdriver adjustments for peak and balance; these are factory preset and do not normally require user readjustment.

The owner's manual is especially good, containing detailed instructions on the controls and their use, the various interface and hookup options for the device, and some discussion of its technical functions as related to music and acoustics.

Test Results: In addition to tests that confirmed the published specs for the SD-50, MR studied the action of its controls by means of oscilloscope patterns. Their accuracy and effectiveness are portrayed graphically in the accompanying 'scope photos. Two of these show delay for 5 and for 50 milliseconds respectively. In the 5-msec test, a tone-burst signal was fed into the front-channel inputs (upper trace) and the delay control was set for 5 msec. Scope-sweep rate was adjusted for 5 msec per division (horizontally). As the lower trace on this 'scope photo verifies, the device's output, as recovered from its rear-channel output jacks, is displaced in time (delayed) by exactly the 5 msec specified. In the next photo, a similar test is documented, this time with the delay control on the SD-50 set for 50 msec and the output (lower trace) delayed by 50 msec.

Another function of the SD-50, that of providing reverb, was tested and is shown in the 'scope photo whose lower trace has several signal bursts. The same tone burst as before (5 msec) was fed to the SD-50's front input jacks, but this time the reverb control was turned fully clockwise. As the lower trace shows, complete and diminishing-amplitude replicas or echoes of the first delayed output (at the rear-channel output jacks) appear neatly spaced to the right. What is not evident from this photo is the fact that the reverberance is "cross-fed" to the opposite channel when it is introduced, whereas the rear-channel outputs contain only delayed versions of the front inputs with no cross-feeding when the delay control alone is advanced.

The final 'scope photo shows the frequency response of the rear-channel outputs. Obviously a good deal of thought has been given by Sound Concepts to this parameter. The fact is, at a "live" concert in a large hall, the sound energy reaching you directly from an on-stage performance arrives with full-frequency response. However, the reverberant field energy reaching you by reflections from walls, ceiling, etc., does have a roll-off characteristic not unlike that built-into the SD-50. Since not all program material may require the same degree of rear-channel rolloff, the rolloff



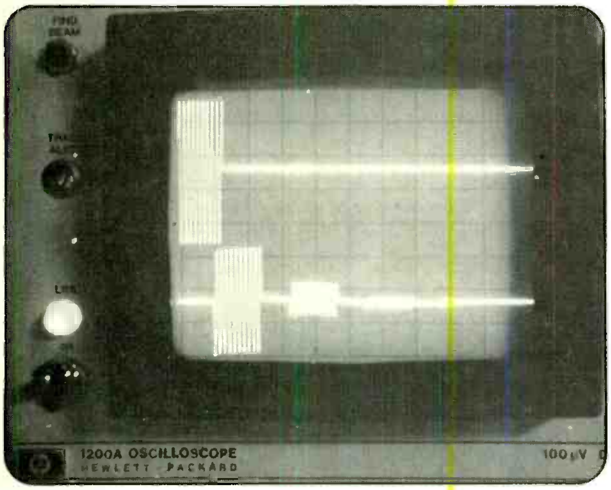
Sound Concepts SD-50: Fifty millisecond rear channel delay.

switch can be used to adjust the amount of rolloff. In the successive spectrum-analyzer sweeps shown in this 'scope photo, the frequency response characteristics of the rear-channel outputs are shown for all settings of the rolloff switch with the delay set for 5 msec (upper traces), and for 50 msec (lower traces). The rolloffs occur pretty much "as claimed." They are all smooth, and they do not degrade the responses above rolloff frequencies.

In listening tests, using the SD-50 with an extra pair of speakers and a "rear-channels" amplifier to drive them, the device certainly proved capable of effectively "enlarging" the aural perspectives of our listening room. We also were impressed with two important facts about time delay used in this manner. For one thing, it became apparent that speaker placement is not nearly so critical as it was in the case of early quadraphonic program sources which tried to accomplish the same thing. For another, since the response from the rear-channel speakers is *not* wide range, the demands made on the rear speakers are hardly as great as on the speakers used for primary, front-channel stereo playback. We also found that the most natural sonic effect could be realized by refraining from turning up rear-channel sounds to the point where listeners become aware of the rear speakers. If these sound too loud, the effect becomes too obvious, and unnatural.

General Info: Dimensions are 11¾ by 3⅝ by 7½ inches in walnut case; 19 by 3½ by 7½ inches in rack-mount version. Price is $600.

Individual Comment by N.E.: A strong point in favor of the SD-50 is its ability to "do its thing" with any kind of program material, regardless of how it was originally recorded, and this applies of course to the device's function for playback or for making your own recordings. As with all devices of this general type, you can expect to have to experiment somewhat with

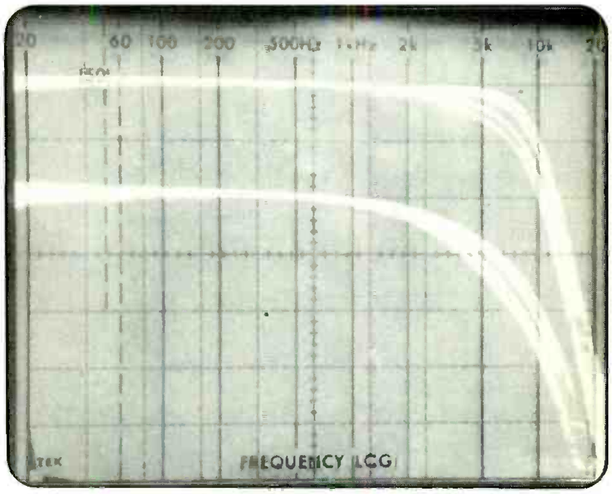


Sound Concepts SD-50: Five millisecond rear channel delay plus reverb.

control settings before getting the sound just the way you want it, but thanks to the accuracy and responsiveness of the SD-50's controls, this effort can produce some very pleasing results.

Individual Comment by L.F.: This is my second encounter with this amazing little electronic miracle from Sound Concepts, the earlier version having been configured for wood cabinet installation. Clearly, the introduction of this time delay unit as a signal processor for home high fidelity systems must have generated such interest amongst professional recordists and sound-reinforcement engineers that the company was prompted to produce a version suitable for standard 19-inch rack mounting—the version we currently are testing and enjoying.

While one may argue the virtues of either the bucket-brigade (analog) approach to electronic time delay or those of the digital approach, the fact is that *any* well-designed electronic time delay unit can add to the enjoyment of music listening in a way that a few short years ago was just impossible. Those of us who were involved in recording in those days can well remember some of the "Rube Goldberg" high-speed



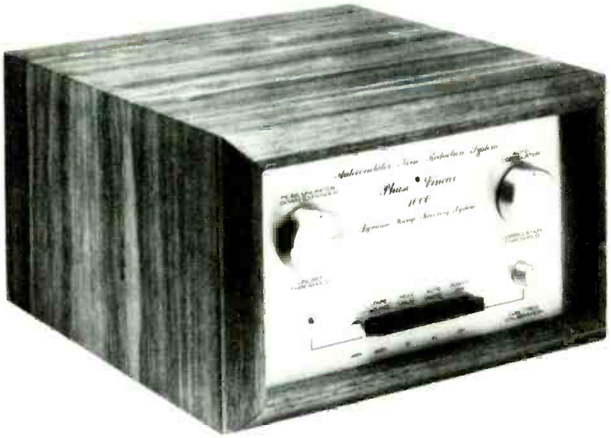
Sound Concepts SD-50: Frequency response, rear channel outputs, with minimum delay (upper traces), and maximum delay (lower traces) for each setting of rolloff controls.

tape loops that were used to achieve just a few milliseconds of time delay, reverb, or what have you. Not to mention the cavernous echo chambers that broadcasters and some recording studios used in attempting to achieve the "big hall sounds." Well, all of that is in the past with the coming of electronic time delay, and the Sound Concepts SD-50 represents an excellent example of that new approach.

SOUND CONCEPTS SD-50 ELECTRONIC TIME DELAY: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Distortion	0.09% at 100 Hz; 0.2% at 1 kHz; 1.0% at 5 kHz.
Input for full output	1.0 V (into front-ch inputs; adjustable from 0.2 to 20 V)
Maximum rear-ch output	10 V
Rear-ch gain	Set for 10 dB; adjustable
Rear-ch frequency response	30 Hz to 4 kHz ± 1 dB (at 5 msec delay, flat position)
Delay (rear chs)	5 to 50 msec (10 to 100 msec in mono mode)
Input impedance	100 K ohms nominal (60 K min.)
Output impedances	Front, connected directly to front input. Rear, 6 K ohms nominal. 2-ch mix, 300 ohms nominal.
Signal-to-noise ratio	92 dB below 1-V input; 85 dB at maximum control settings.

Phase Linear Model 1000 Autocorrelator Noise Reduction System



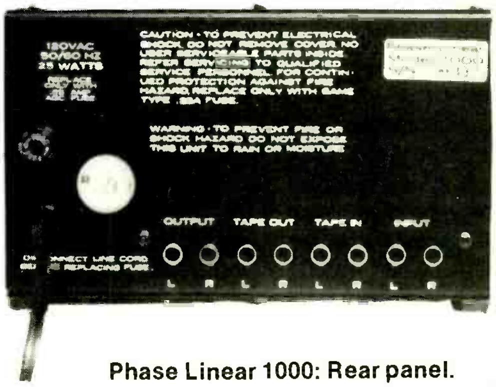
General Description: Phase Linear's Model 1000 is a noise-reduction and dynamic range recovery device designed to improve the signal-to-noise and dynamic characteristics of signal sources. It is intended for connection between the tape-output and tape-input jacks on a preamp (or integrated amp or receiver). The front panel contains seven controls. The large knob at the upper left adjusts both the peak-unlimiter and the downward expander and may be used to increase the dynamic range of program material. A lamp below it gives visual indication of peak unlimit operation. The large knob at the upper right is the high-frequency noise-reduction adjustment. Below it is a smaller knob for low-frequency calibration. Centered near the bottom of the panel are four pushbuttons. One is the tape source/monitor switch; another activates the peak unlimiter and downward expander; the third activates the autocorrelator; the last is the unit's power switch. Centered near the top of the panel is the system's power indicator lamp.

Four stereo pairs of phono jacks handle signal input and output connections at the rear, where there also are the unit's AC line cord and a fuse holder.

The general procedure when using the Model 1000 involves adjusting the correlator control for high-frequency noise-reduction; the low-frequency control; and finally the expander control. Of these adjustments, the low-frequency calibration need be repeated only if one's phono pickup is replaced with a model of significantly different sensitivity from the first. The correlator and expander adjustments likely would require "retrimming" for different program material.

Test Results: On the test bench, the Model 1000 did everything claimed for it, including the exactly prescribed downward expansion of up to 3 dB, linear expansion of another 3 dB, upward peak unlimiting of still another 1.5 dB, and overall single-ended noise-reduction of 10 dB. It was felt, though, by MR's testers that the audible improvement in general was not as significant as we may have expected it to be. Perhaps we have become too accustomed to such "two-sided" noise-reduction systems as Dolby, ANRS or dbx (all of which require prior encoding) and were expecting the same kind of dramatic improvement. On the other hand, it should be pointed out that by holding the total dynamic-range expansion to 7.5 dB, the Model 1000 succeeds in delivering that degree of additional dynamic range without at the same time introducing detectable "breathing" or "pumping" effects sometimes characteristic of other, less sophisticated expanders that have been appearing recently. In a sense, then, the 1000's circuitry might be thought of as the "purist's answer" to expansion: not nearly so dramatic as some others, but also free of "side effects" that might annoy the astute music listener.

The autocorrelator function (a circuit which can, ostensibly, sense the difference between uncorrelated noise and music-related harmonics and can "open up windows" for the latter while shutting down signal paths for the former), when correctly adjusted, is about as effective in reducing hiss in noisy program material as is the Dolby B in its two-part process—and that is no small accomplishment. We wonder, however, whether even such ingenuity is worth spending $350 for. The serious audiophile probably will have made recordings using at least a Dolby B system, or an open-reel deck capable of supplying at least a 60-dB signal-to-noise ratio in the first place. Careful treatment of disc recordings (and selective purchasing of them to begin with) can also yield about the same S/N. So the question really comes down to what audible benefit will be derived from the Model 1000's capability to improve that S/N (in terms of hiss) to, say, 70 dB.

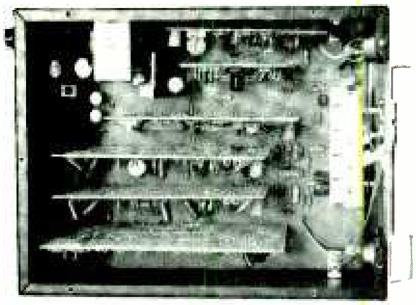


Phase Linear 1000: Rear panel.

One application that suggested itself to us had to do with improving the reception of weak FM signals, especially stereo FM. To test this application we connected an FM generator to a tuner, and reduced signal strength to about 10 microvolts. This naturally resulted, with this particular tuner, in a S/N ratio of about 60 dB. Using our spectrum analyzer, in its *linear* sweep mode, we then plotted everything from 0 to 20 kHz, as shown in the 'scope photo. Each horizontal division represents a *linear* span of 2 kHz (the superimposed log-frequency notations are to be disregarded). The spike at the left represents the 1-kHz test signal "received" by the tuner. The upper trace of noise extending to the right represents the noise level at frequencies from 2 kHz and upward. This trace was taken with the autocorrelator out of the circuit.

Next, the sweep was repeated, with the autocorrelator on the Model 1000 activated. Since the fundamental 1-kHz recovered audio tone is unaltered, it falls directly upon the earlier trace (which was stored by the storage 'scope facilities of the analyzer). The noise "floor," however, is seen as the lower of the two traces, about 10 dB lower than was the case without the autocorrelator in use. Notice that the frequency content of the noise is completely unaltered (thus, the frequency response of the music is unaffected)—only its overall level decreases.

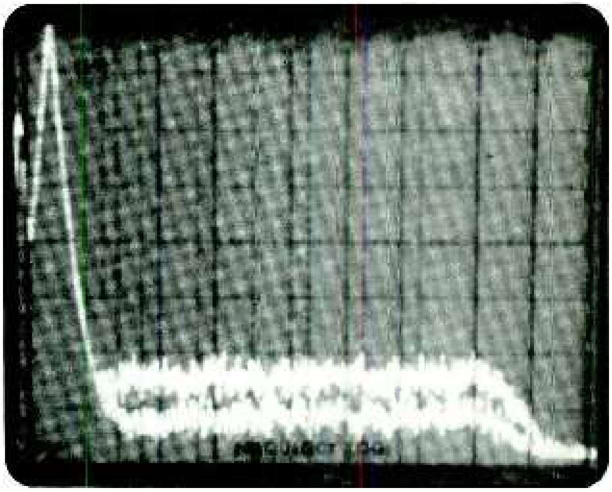
In general, we found that adjustment of the peak-unlimiter/downward expansion control was fairly non-critical, but we noted that settings of the autocorrelator control must be made carefully if noise elimination is not to be achieved at the expense of full frequency response.



Phase Linear 1000: Internal view.

General Info: Dimensions are 9½ inches wide; 5 inches high; 11⅝ inches deep. Weight is 6 lbs. Supplied with owner's manual and two sets of stereo signal cables. Price: $349.

Individual Comment by N.E.: My previous comments, expressive of a "qualified view" of this general class of audio device, remain unchanged. At that, the Model 1000 can be credited with offering a relatively less "overt" kind of processing with concomitantly less undesirable side-effects such as audible "breathing." I do not feel it is the answer to everyone's sound-system needs, although undeniably there will be some who are taken with it and will find a place for it in their setups. In any event, I feel that the owner's instructions, while offering a good deal of interesting and useful information, could be more explicit in details of hookup (especially in relating the Model 1000 to a system employing one or more tape recorders), and in explaining the sequence of operations and the use of all the controls.



Phase Linear 1000: Noise reducing capability (see text for explanation).

Individual Comment by L.F.: When I first tested Phase Linear's unusual Model 4000 preamplifier more than a year ago, I thought: "Why doesn't that company offer its innovative autocorrelator/downward expansion/peak-limiting circuitry (contained in that preamp) as a separate add-on device?" Well, now they have and obviously that circuitry represented the costliest portion of the previous $600 preamp-control unit. As a separate unit, the Model 1000 still retails for a hefty $349. In terms of the sophisticated circuitry it contains, that may not be a very high price but as all things in hi-fi sound, the real value is in the listening, and I must confess that despite its performing to specs, the audible improvement offered by the 1000 was—as explained in our "Test Results" section above—less than "dramatic." So while I can admit to some enthusiasm for the Model 1000, it is tempered by what the device seems to do vis-a-vis its cost. If future integration of circuitry permits a less expensive version, I might be more kindly disposed to it.

PHASE LINEAR MODEL 1000 AUTOCORRELATOR NOISE REDUCTION SYSTEM Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Total distortion	0.15% at 1-volt input; 0.30% at 2-volt input.
Input impedance	70 K ohms
Input level	3 V max. 0.55% THD at this level.
Output voltage	8 V rms max.; 3 V rms into 2000 ohms
Frequency response	±1 dB, 22 Hz to 27 kHz
Attack threshhold	0.2 V peak at input to peak unlimiter
Downward expansion	Begins at −35 dB
Autocorrelation characteristics	3 dB N/R at 2 kHz, increasing to 10 dB from 4 kHz to 20 kHz. Low-freq. N/R begins at 200 Hz, reaching 20 dB at 20 Hz
Power consumption	30 watts

TEAC/Tascam Model 5 Mixer

By Jim Ford and Brian Roth

General Description: One of the hottest selling mixers available today is the Tascam Model 5. An incredible amount of control is packed into this small, sixty-two pound unit, and it has an amazingly low price tag. The basic mixer is modular, and has eight inputs and four outputs; an expander accessory can increase the total number of inputs to twenty. A tabulation of features on each input module follows:

A. Slide fader for volume control with 2 ⅜-inch travel.

B. A two section equalizer with:

(1) Up to 15 dB low-frequency boost or cut at 75 Hz or 250 Hz.

(2) Up to 15 dB high-frequency boost or cut at 3 kHz or 10 kHz.

C. Pre-fader cue send "pot."

D. Post fader echo send "pot."

E. A switch to select line input or microphone.

F. A "pot" for varying the input gain (affecting both microphone and line).

G. A switchable pad switch with 0 dB, 20 dB or 40 dB of attenuation for microphone inputs.

H. Four switches and a pan-pot allow assignment of input signal into combinations of four outputs.

I. A solo pushbutton.

J. A pushbutton assigning the input signal to a "direct output" jack on mixer's rear panel.

K. A light emitting diode (LED) to indicate input overload.

While not mounted directly on the input module, it should be noted that each input in the mixer has an "accessory send" and "accessory receive" jack located at the mixer's rear. These jacks allow external equalizers, compressors, etc., to be patched in on each input.

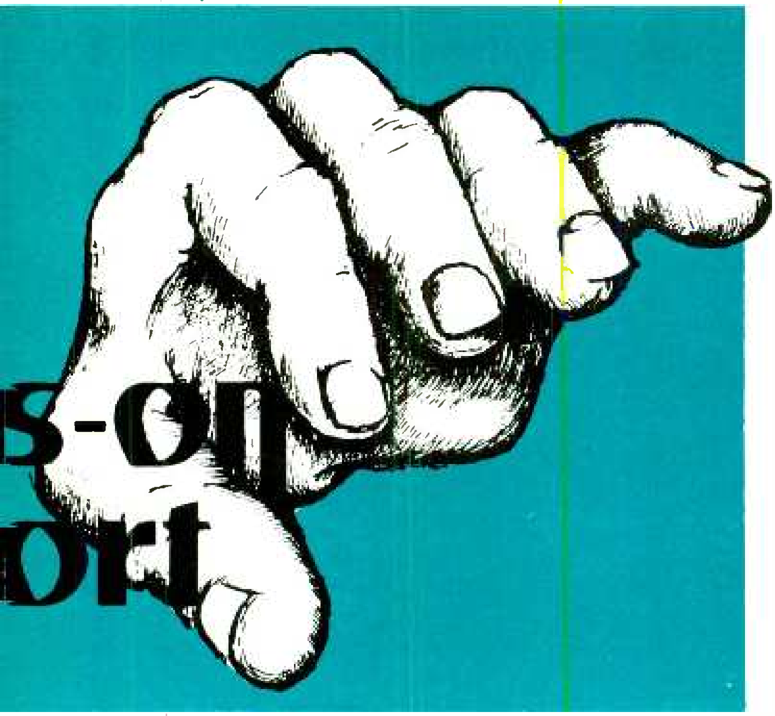
The four submaster modules include the usual slide fader as well as an echo return control for operation with an external reverb (or echo) unit.

The submasters also include a number of features that greatly increase the unit's flexibility, particularly when using the mixer with a four-channel multi-track tape recorder. This section is generally used for monitoring a combination of the four mixer outputs and the four tape outputs during multi-track recording (laying tracks, overdubbing, etc.). Four volume controls and pan pots generate a separate stereo mix for this purpose. Four switches select mixer output or tape recorder playback for use with this separate mix. A pot on each submaster allows playback of the four-track recorder into the cue (headphone) mixing bus.

The master module contains switching and volume control facilities for control room and studio loudspeakers (external power amplifiers are required to drive the speakers).

These features, along with a number of other user-oriented facilities, make this mixer a flexible and complete control center for a four-track recording facility. Additionally the Model 5 includes most all of the necessary functions for a P.A. system mixer. All microphone inputs are balanced and transformer isolated for use with low impedance microphones; "cannon" type 3-pin connectors are used for microphone connections. With very few exceptions, all of the other input and output connectors on the mixer are the RCA-type phone jack.

From our experience with the Model 5, just about all outputs (main program, cue echo, control room, etc.) are capable of driving high levels into 600 ohm loads. This eases problems when connecting professional equipment with 600 ohm input impedances. Naturally, these outputs are not balanced (that would be a trick



to do with an RCA phone jack!). But transformers could be added externally if they were really needed (very often they are not necessary).

Oh yes, metering facilities. All four outputs have an illuminated VU meter with very, very small overload indicator lights. A fifth meter can be switch selected to monitor the echo output level or cue level. As mentioned earlier, each input module includes a peak overload LED. We really wish that the overload lights in the four VU meters were the size of the larger ones on the input modules.

Field Test: We have had occasion to use the Model 5 in several small studios as well as with P.A. systems. In general, its performance was quite good.

In the studio applications, the mixer's extensive auxiliary facilities greatly simplified the operation of a recording session. As a P.A. mixer, the Model 5 supplied the basic necessary functions.

The mixer was reasonably quiet and quite clean sounding. The equalizers were effective, although the 3 kHz position of the high frequency control sounded a bit sharp. It was impossible to overload the mixer as long as the controls were properly adjusted (keeping the submaster and master faders positioned in the shaded zone marked on the front panel was important).

It was interesting to note that the input overload indicators on the input modules would flash only when the controls were severely maladjusted. This demonstrates that the first amplifier stages (microphone preamplifier) is normally operating well below clipping, thus ensuring plenty of "headroom" for sudden peaks in the input signal.

All controls did their job quite well. We do feel that the "studio" circuitry could be better; this auxiliary portion tended to distort easily and seemed prone to pick up RF (radio) signals. Sometimes, the use of the solo switches would introduce small clicks into the program outputs. Other than these two areas, the Model 5 seemed quite free from extraneous noises or interference pick-up.

Lab Test: Our measurements demonstrated that the Model 5 is an excellent performer. The noise figures are low under most conditions.

The harmonic distortion figures were quite low for the most part. Much of the time, the distortion component of a signal was lower than the residual noise of the mixer. The +20 dB (about 8.5 volts RMS) meaturement at 10 kHz was quite high due to an effect called *slew rate limiting* that is characteristic of the integrated "chips" used. At high frequencies, these chips cannot generate as much output level without distortion as at mid frequencies. About +19 dB (approx. 7 volts RMS) was the highest output level the unit could generate at 10 kHz; at this level distortion was near 0.5%. Of course at lower output levels the distortion figure drops. This peculiarity was not usually heard during actual usage. Nevertheless, perhaps Tascam could use a "faster" integrated circuit to completely eliminate the possibility of a problem. The higher distortion figure at 50 Hz through the microphone inputs is due to the input transformers; distortion contributed under these conditions was mostly low ordered products.

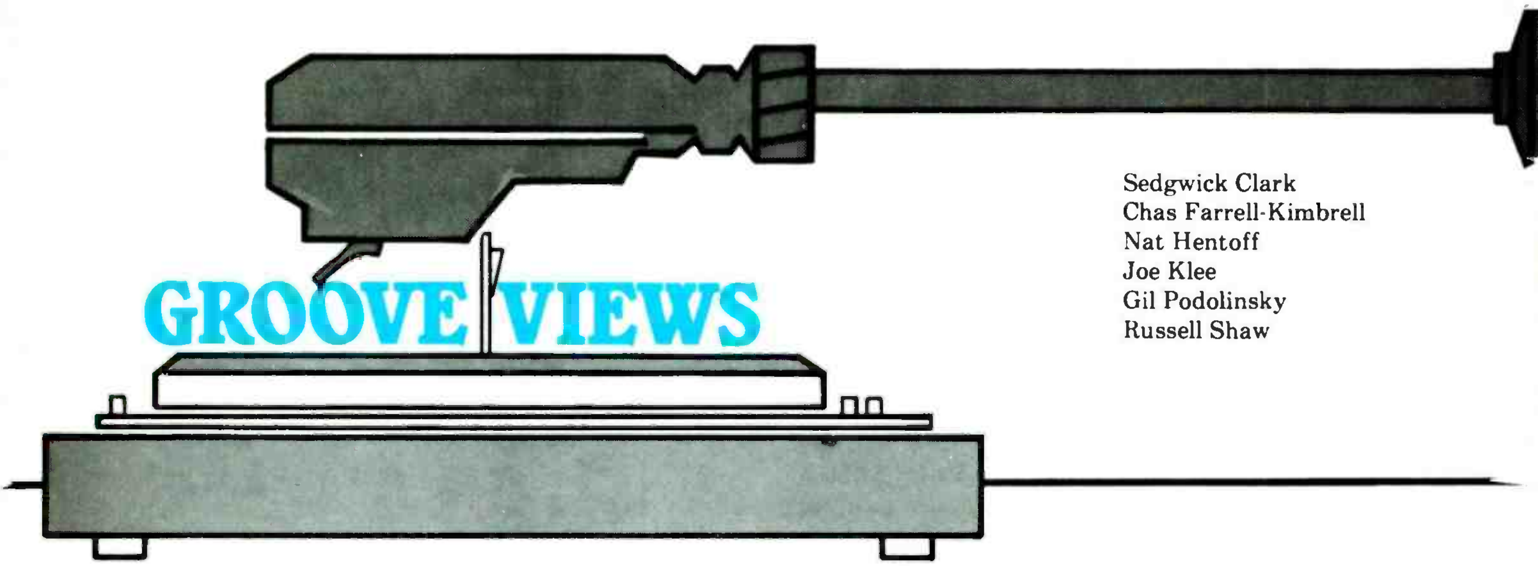
The main outputs were calibrated such that 0 VU = approx. +1 dBm (.86 volts RMS). This allows ample headroom in consideration of the fact that the unit can produce more than +18.5 dBm (6.5 volts RMS) into 600 ohm loads and better than +20.75 dB (8.5 volts) into higher impedance loads. Note that just about all outputs from the mixer are capable of this level of performance. The only outputs that fall short are the studio outputs; they can only produce about +13 dBm (3.5 volts RMS) before clipping. All of the output impedances are quite low (typically below 100 ohms) thus insuring a high degree of noise isolation when interconnected with external devices, even with long interconnecting lines.

The peak overload lights were checked, and it was found that the input overload LEDs would flash about 3 dB before preamp overload would occur. The output overload lights illuminated at about 10 dB above 0 VU.

In general, the Model 5 performed spledidly on the bench.

Conclusion: We have high praises for the Tascam Model 5. The mixer has astounding flexibility for its size and price, and excellent audible performance. Only a very few areas could be designed better than they presently are.

We would without hesitation recommend the use of this unit for a small recording studio or as a basis for a P.A. system, or *both* if you require a mixer for double duty. It has the features and the performance for excellent results.



GROOVE VIEWS

Sedgwick Clark
Chas Farrell-Kimbrell
Nat Hentoff
Joe Klee
Gil Podolinsky
Russell Shaw

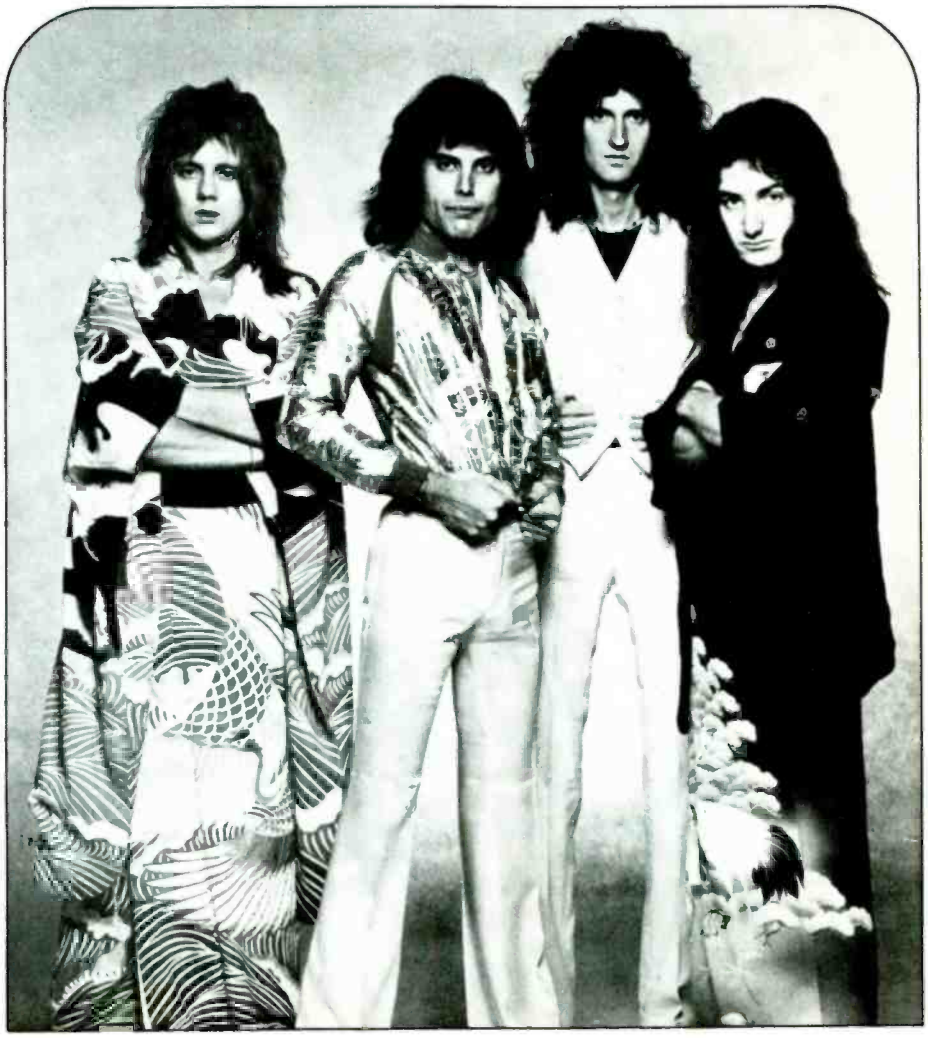
POPULAR

QUEEN: *A Day at the Races.* [Queen, producers; Mike Stone, engineer; no studio listed]. Elektra 6E 101.

Performance: **Quite impressive**
Recording: **Production oriented**

An album whose reception has been heralded more for being the first at the new $7.98 list price, resulting in an ineffective mini boycott, than for its content, well warrants constant listening to appreciate the subtleties of this inventive English quartet. This group is noteworthy for it has a distinctive and recognizable style. The vocal emphasis in its arranged choral sound is as important to rock as that of the Beach Boys. Brian May's home-made and sustain producing guitar succeeds in establishing a uniqueness in a world of clanging guitars. He may not be a master technician, but his play is refreshingly clever.

The success of the album, similar to that of its predecessor, *A Night at the Opera*, is that the production draws out and nurtures the talent, creating three characteristics of the sound in the process. The first and foremost is the vocal style: precisely arranged, multi-tracked, layered, stretched. "You Take My Breath Away" demonstrates the vocal abilities far beyond description. The sweeping vocals, flattening and fading out a la the group vocal style of the forties and early fifties, infrequently resurrected by The Carpenters of all people, is a powerful display. Backed only by acoustic piano and multi-tracked guitar solo, the voices consort to create the other instruments, with incredible crescendos and devastating results. Secondly, the drums play a far more important role with Queen than with most groups. The toms are tuned very low and deep, emitting a very fat snare sound with occasional use of echo. Cymbals are given the crisp, shimmering treatment and are quite effective when combined with rolls, which are frequent. As a result, there isn't much bottom on the bass, opting instead for the mid and treble ranges. Thirdly, May's sustaining guitar is the perfect counter-punch to the vocal chorus. Listen to his solo in "Somebody To Love." Playing through nine old Vox A.C. 30 amps, the sustain/feedback when combined



QUEEN: Mounted enthusiasm

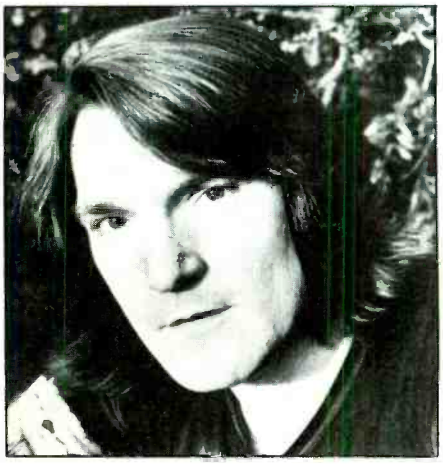
with the natural echo of the room create a biting edge that aids in putting across what is basically a soft ballad to a volume-hungry audience. In short, the excitement on the record is catchy, and I had no trouble mounting enthusiasm to review it. G.P.

BRIAN AUGER'S OBLIVION EXPRESS: *Happiness Heartaches* [Brian Auger, producer; Neil Schwartz, engineer; recorded at Different Fur Studios, San Francisco, Ca.] Warner Bros. BS 2981.

Performance: **Undermined by the engineering**
Recording: **Undesirably clean**

As another artist who suffered for years at the hands of RCA, this is Auger's first release in over a year and his first for Warner Bros. Further, this record marks important changes not so much so in style, as in recording. His first album to be recorded in the States, save two "live" albums, follows his change in residency to San Francisco.

In discussing the album prior to release, he was very satisfied with the reproduction of the rhythm section, a universally respected aspect of his sound. A long-time admirer of American R&B sound, he felt that the best he could attain previously on record was an English facsimile. Having first heard several of the selections "live" before the vinyl testimony, I am disgruntled with the recorded result. Lennox Langton's congas, for example, are no longer in the forefront of the percussive mix. Nor are they resound-



BRIAN AUGER: Lacking the "dirty" feel

ingly deep, sounding less active with lighter embellishments.

I may be misled, but the detracting aspect of this recording is that it's been given a heavy dose of Dolby, thus robbing the proceedings of that spontaneous "live" feel that so characterizes Auger's sound. Nowhere can this be heard better than in the recording quality of the keyboards. Both his Rhodes and Hammond B3 are unbelievably clean, forcing him to sound subdued. Both instruments have a characteristic tendency to break-up, or become "dirty" in the higher registers. When coupled with Auger's firey, hard-driving style, one associates the two after several albums and concerts.

I never thought I'd be complaining about clean recordings but can you take out the dirty feel and still have jazz-funk? Although the vocals come through fine, the rest sounds compressed. I trust that this isn't a sign of things to come.

G.P.

LEO KOTTKE: *Leo Kottke.* [Denny Bruce, producer; Ern Rose, Scott Rivard, Paul Martinson, Dave Hassinger, Douglas Decker, engineers; recorded at Armstrong Studios, Melbourne, Australia, Sound 80, Minneapolis, Minnesota, Sound Factory West and Western Recording, Los Angeles, Ca.] Chrysalis Records CHR 1106.

Performance: **Uninspiring**
Recording: **Unexpectedly consistent**

Recording Leo Kottke is usually quite a challenge, for by his own admission he is lazy. Also, one can immediately sense after spending any time at all with him that he must consume at least a fifth of gin a day. Faced with these acoustical problems it's no surprise to learn that Leo isn't usually pleased with his recordings, or that he'd prefer to reduce the problem by recording in mono. Obviously, there are insurmountable problems on this LP; however, the engineering team(s) did a commendable job of holding it together, for when you assembly-line a recording at four different studios with five different engineers, it usually tends to sound like it. In this instance

LEO KOTTKE: Another chicken-or-the-egg question?

it's easier to get away with, for it's not too difficult to set levels on an acoustic guitar and balance the EQ for strings, as is the case here.

Whether it was producer Bruce's original intent to add meat to these very short, skeleton-like pieces by using strings, or whether Kottke left him no choice, is rather another chicken-or-the-egg question. Regardless, the

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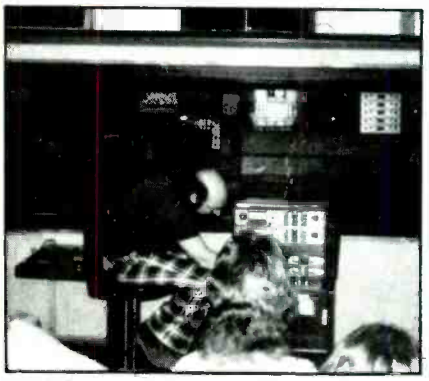

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the post-coltrane jazz pioneers

by Nat Hentoff

Throughout the history of jazz, certain places at certain times have become meccas—from New Orleans to Chicago and Kansas City to Minton's in Harlem and 52nd Street lower down. Now, still lower down in Manhattan, the newest mecca is a phenomenon called "Loft Jazz." In actual lofts (usually used as artists' studios) and in transformed basements and warehouses, many of the more insistently resourceful expanders of black music have converged in a geographic and psychic center where they are setting the principal jazz directions of the post-Coltrane era.

Some of this music is appearing on such invaluably daring labels as India Navigation and Black Saint. But the first wide-ranging survey of protean "Loft Jazz" is a five-volume set (each of which is available singly), *Wildflowers/ The New York Loft Jazz Sessions* (Douglas). All the performances were recorded during a 1976 Spring Festival at Studio Rivbea, the loft home and performing space of reedman-composer-teacher Sam Rivers.

I would recommend, to start with, Volume Four, because among the leaders at those sessions are two of the most forcefully original of all the players currently at the far end of the jazz frontier—tenor saxophonist David Murray (in his early twenties) and baritone saxophonist Hamiet Bluiett (in his mid-thirties and by far the most distinctively creative baritone player since at least the early '40s). In that same volume, there are also Julius Hemphill, an alto saxophonist of considerable lyric depth and trumpeter Olu Dara. The latter, from Mississippi, plays in conversational cadences with brilliant use of space (silences as integral parts of phrasing). And his work is further characterized by a diversity of subtle, incisive tone colors.

If you can, however, get all five sets, and then just absorb the music over periods of time. These are sounds (often complex, convoluted "cries") and continually intersecting layers of rhythms that create a continuum of powerful emotional testaments (just as intense when they nearly whisper as when they shout). Accordingly, it is much less valuable to try to analyze what's happening than to respond as openly and viscerally as you can. Or, as John Coltrane used to put it, open yourself to the *music*, not to any words about it.

Among the other high points are Sam Rivers on Volume One; alto saxophonist Marion Brown and trumpeter Leo Smith's New Delta Ahkri on Volume Two; Andrew Cyrille & Maono on Volume Three; and Sunny Murray & The Untouchable Factor on Volume Five. Cyrille and Murray are among the more influential percussionists in this advanced molten jazz scene, and both have served apprenticeships with Cecil Taylor. That volcanic pianist, by the way, has had a lot to do—along with Ornette Coleman and others—with this "new" language which, of course, has deep roots in the past.

The engineering, by and large, reflects both understanding of and attentiveness to the wide range of dynamics and the frequent thickets of highly diverse sounds in this music. There is clear, strong presence and generally excellent balance. These are surely historic recordings.

KALAPARUSHA / KEN McINTYRE / SUNNY MURRAY/SAM RIVERS/AIR: *Wildflowers 1.*
FLIGHT TO SANITY/KEN McINTYRE / ANTHONY BRAXTON / MARION BROWN/LEO SMITH: *Wildflowers 2.*
RANDY WESTON/MICHAEL JACKSON / DAVE BURRELL / ABDULLAH /ANDREW CYRILLE: *Wildflowers 3.*
HAMIET BLUIETT / JULIUS HEMPHILL/JIMMY LYONS/OLIVER LAKE/ DAVID MURRAY: *Wildflowers 4.*
SUNNY MURRAY/ROSCOE MITCHELL: *Wildflowers 5.* [Alan Douglas and Michael Cuscuna, in association with Sam Rivers, producers; Ron Saint Germain, engineer; recorded at Studio Rivbea, N.Y.] Douglas NBLP 7045-7049.

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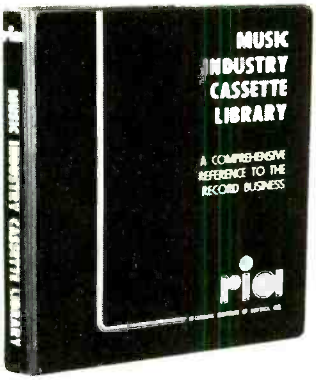

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end result is that this is a very bland pop album. Where Kottke was once original and complex, he is now redunant and simple. In its simplest setting this is an excellent example of engineering competency muddied by a series of dud performances. If Kottke has a secret admiration for Tony Mottola, he's done him proud. Or maybe it's just the latest George Benson album playing at 16 rpm. G.P.

SKYHOOKS: ***Living in the '70's.*** [Ross Wilson, producer; Bill Halverson, engineer; no studios listed.] Mercury Records SRM 1-1124.

Performance: **Deceivingly present**
Recording: **Distinctively clean**

This is the second release for the Australian quintet, and with it they have given notice that they are in the same artistic class as 10CC, Steely Dan or the Tubes. One of the few acts today virtually sans keyboards, the absence does not affect this progressive band. In a sentence, this is a straightforward recording. There is no panning, no gimmicks for the sake of gimmickery, and nothing to detract from the main point of focus. For me, this is a great sign of recording maturity; knowing when not to clutter the recording with tracks of self-indulgent, unnecessary filler recorded only for the purpose of establishing a producer or engineer's reputation at the expense of the artist's creativity. The production concept on this album was to record them as they are, and Halverson didn't miss a note.

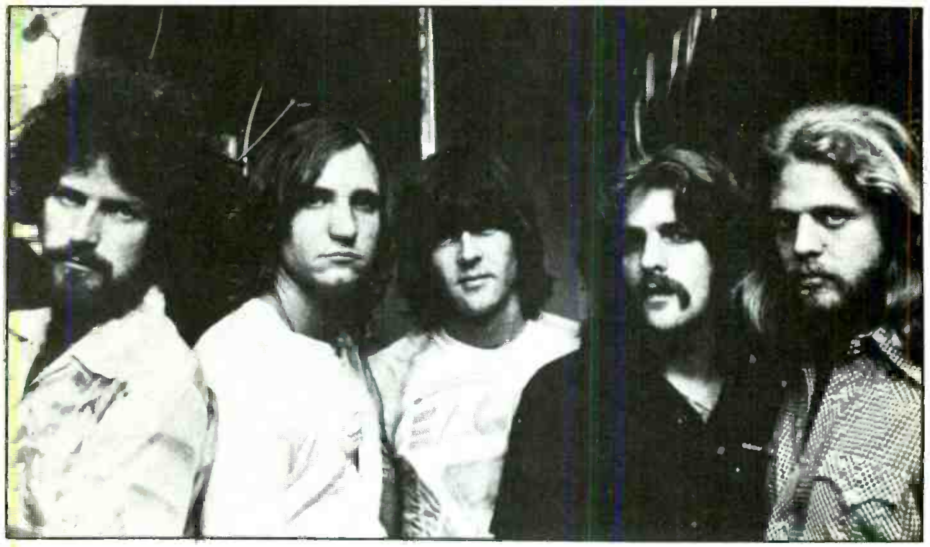
Vocals are the key point for this band, featuring a wide and sweeping range. Lead singer Graeme Strachen has a high and piercing range, but the listener is not subjected to twisting the volume knob to have the point put across. Instead, the engineering is heady enough to achieve the same result through clarity and balance, therefore making the listener pleasantly, rather than painfully, aware.

The balance is consistently smooth throughout the album, resulting in a seemingly effortless performance. The drums are exact—not too loud, too soft or flashy. The cymbals are exceptionally clear and the snare is solid, though the kick is lost for the most part. Further, it's nice to hear a guitar-oriented group without distortion and pointless solos. Everything is precise, well matched and superbly executed. G.P.

EAGLES: *Hotel California.* [Bill Szymczyk, producer; Bill Szymczyk, Allan Blazek, Ed Mashal and Bruce Hensal, engineers; recorded at Criteria Studios, Miami, Fla. and the Record Plant, L.A., Ca.] Asylum 7E 1804.

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

There is a sense of contract-fulfilling routine here. This impression is most unfortunate, for ever since *One of These Nights*, the Eagles have generally been granted membership in the pantheon of supergroups. This last release produced many tasty licks, musical surprises and three hit singles of uncommon quality, fueling the high esteem which many have held them in for nearly half a decade.



EAGLES: Formularized in anonymity.

Now, with the release of *Hotel California*, disappointment ensues. New Eagle recruit and former superstar guitarist Joe Walsh fails to contribute anything substantial, save for a decent slide-guitar passage on "Victims of Love," preferring to play meekly in the background. The four other Eagles join him in anonymity, playing formularized arrangements of tunes which are little more than inferior clones of previous work. "New Kid In Town" essays to be this album's "Lyin' Eyes," and the title track—a wordy, dense encyclical, presumably the platter's "heavy" offering—bogs down in its own obliqueness.

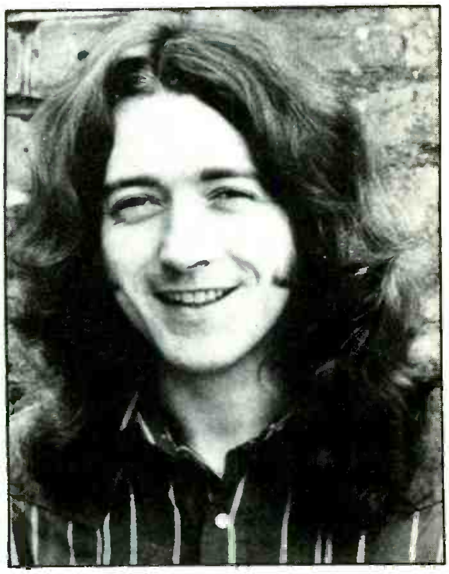
Saving graces? We must unwillingly admit that the production and mix are flawless. However trite, the instrumentation sounds unmuffled, fresh and clean, and the overdubbing of strings onto the uninspiring vocal track is, despite its pomposity, quite shrewdly timed. Indeed, maybe *that* is where the underlying problem lies; everything is *so* perfect and planned that boredom is inevitable. R.S.

RORY GALLAGHER: *Calling Card.* [Roger Glover, producer; MACK and Hans, engineers; recorded at Musicland Studios, Munich, Germany.] Chrysalis CHR 1124.

Performance: **Got the feel**
Recording: **Best track by track and "live"-in-the-studio LP**

For tax purposes, more and more English acts are recording abroad. Musicland Studios in Munich seems to be the favorite of hard rock and guitar blazing bands like Led Zeppelin and now including Rory Gallagher. In terms of the results, the studio produces some interesting testaments. They succeeded in giving Zeppelin's recorded sound a much needed face lift and through *Calling Card* they have expanded the appreciation for Gallagher. If it seems odd to credit a studio and not particular engineers, one must compare the recordings made there. The natural ambience of the room comes through strikingly, on each recording. In this instance, the mix is peculiar. Dealing only with guitar, voice, drums, bass, and piano, each respective track is, for the most part, produced differently leading one to the natural conclusion that it was done track by track. The guitar is recorded quite cleanly. The vocals are so up-front and alive that all the sibilancy and related sound hisses are not distracting but actually add character to the recording. The drums are mixed more with the crisp snare and clashy hi-hat cymbal sound in mind and are placed slightly behind the voice and guitar.

The amazing presence of the guitar, voice and drums is contrasted by the limp, dead bass, giving the impression that the mic was placed behind an over-



RORY GALLAGHER: Expanded appreciation.

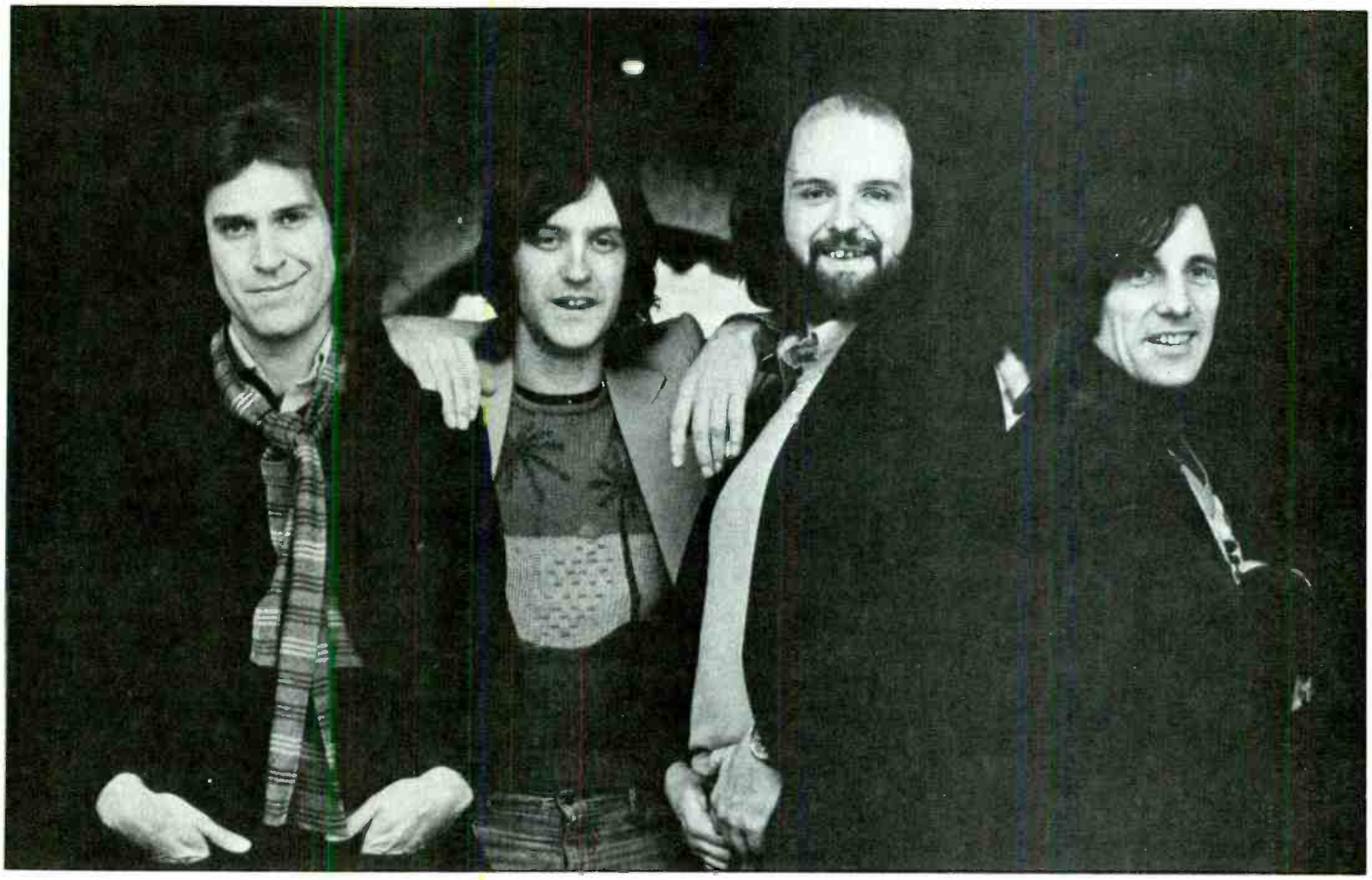
stuffed pillow. When the piano/organ is used it is placed to the rear left of the mix, usually adding nothing to the song. The piano itself was scraped clean of highs and sounded compressed.

Although I'm not crazy about Gallagher's music or guitar, this overall feeling of recording in a long narrow hallway with the door open at one end is quite compatible with Rory's style.
G.P.

THE KINKS: *Sleepwalker.* [Ray Davies, producer; Roger Wake, engineer; recorded at Konk Studios, London, England.] Arista 4106.

Performance: **Kinks of old**
Recording: **Worth the wait**

Kill the fatted calf. The Kinks have returned home. Home in this case is rock and roll. After umpteen uneven concept albums, Ray Davies has finally kept his word and delivered nine unrelated rock-



KINKS: Rock and roll is here to stay

ers. Better still, the recording highlights the positive aspect of a band with no soloists of note. The separations in "Jukebox Music" for example, are a textbook example of clean, clear recording. If you own a pair of headphones, they won't be wasted on this track.

I'm particularly impressed by the consistency in the recording of the drums throughout the album. Since the Kinks are basically one big, five-piece, rhythm section behind Ray Davies' voice, the drums play a more prominent role than on most releases today. Normally, we hear one of two extremes—too crisp or too deep. Mick Avory's drums on this recording are neither. They're not fried or loosely tuned, but active without stealing center stage. Engineer Wake succeeds in giving Avory all the room he needs to drive the tune without electronically narrowing the sound to make it fit. Even as small a thing as one cymbal crash sounds as natural as though you're standing next to it, and that's what recording is all about. It wouldn't surprise me to learn that the drum sound was achieved without the use of a dozen channels and three sub mixes.

The guitar work is competent, with Ray on Ovation acoustic and Fender Telecaster, and brother Dave on the Les Paul. The guitar in this band is primarily a rhythmic instrument. Organ and electric piano tend to be buried in the general bottom mix, with occasional short, non-complex synthesizer lines or organ chords emerging to create a bridge from one verse to another. Ray's voice isn't the best, but it is easier to take now that he's opted for rockers rather than the slower, lyrically profound show tunes of the recent past. This is why *Sleepwalker* is a success.

The Kinks are one of the best basic rock and roll bands and it's good to see them back in their element. Please Ray, no more concept albums. G.P.

GARLAND JEFFREYS: *Ghost Writer.* [David Spinozza, Garland Jeffreys, producers; Lew Hahn, engineer; recorded at Atlantic Studios, New York, N.Y.] A & M SP 4629.

Performance: **Consistent**
Recording: **No complaints**

Garland Jeffreys, somewhat of a cult hero, utilizes two musical styles on this his latest album—reggae and "street-smart" rock. In doing so, he reminds one of several people, including Springsteen, Jagger, Lou Reed and Bob Marley. That doesn't bother me, for Jeffreys writes good material. So what if his vehicle isn't original?

Engineering-wise this album is straight ahead, with nothing outside of the basics. There are no choirs panned from coast-to-coast, no solos, no screaming guitar, and as Queen emphasizes in a disclaimer on the back of each album, no synths! I left out another no—no monotony. The success here lies in the production. It's consistent; flowing from track to track and void of the usual album filler. Every song is needed and well done. It's a rhythmic album with poignant lyrics. With those two focal points, toying with them in an attempt to make engineering history would be ridiculous. Hahn supplies what's necessary; a clean recording with good separation.

Content-wise, one can't overlook Jeffreys' similarity to Springsteen. Whereas certain Springsteen songs encourage consistent listening, I find it difficult to listen to one of his albums in its entirety. His consistent "formula" sound and approach to writing about New Jersey life gets stale fast. Jeffreys is different and smart, too. He has discovered that one can write about the same subject, but

GARLAND JEFFREYS: Different and smart.

if one changes meter, style and instruments, it sounds new and increases interest. Although Jeffreys gets away with the "sounds - like - only - it's - different" definition, I heartily suggest a change of scene. G.P.

UTOPIA: *Ra.* [Todd Rundgren, producer; John Holbrook, engineer; Tom Mark, asst. engineer; recorded at Bearsville Sound, Bearsville, N.Y. and Utopia Sound, Lake Hill, N.Y. Except "Communion With the Sun" and "Jealousy" engineered by Eddie Offord and recorded at Turtle Creek Barn, New York.] Bearsville BR6965.

Performance: **Energized**
Recording: **Representative of the 'Wizard'**

This seems to be the album Todd has been trying to record. A lot of the things that were wrong on earlier attempts have been cleared up on this disc. One important change is that Utopia now seems to be a working unit consisting of four parts, each with equal input and output. Todd is no longer a one man show using Utopia as a vehicle (Todd's still in the driver's seat, mind you, it's just that now everyone is riding in the same car). Each member writes and sings lead vocals but, the equality doesn't stop there. The members even get a chance to display their virtuosity on other instruments. On this record, John

Wilcox, drummer, plays guitar; Todd plays guitar, piano and saxophone (a la the first Runt album) and in "live" performance of this album; Kasim Sultan, bassist, plays guitar; Roger Powell, keyboards, plays guitar and trumpet; Wilcox plays bass and Todd plays drums.

Kasim Sultan, Utopia's newest member, proves to be a valuable asset not only on bass and guitar but also as a strong vocal force. His lead vocal on "Eternal Love," the album's love ballad, is strong, melodic, and sensitive. The material reaches broader plateaus now that everyone is contributing. There's theatrical comedy, a love ballad, strong social comment, a touch of religious philosophy and a fairy tale obviously inspired by J.R.R. Tolkien's *Lord of the Rings*. Oh yes, and there's Rock & Roll.

The recording quality is technically as good as, if not better than, the other Utopia albums. But the real difference seems to be production. This album is far from overproduced. It is cleaner, less busy (in a production sense) than its predecessors. Todd seems more sure of himself and his band. The music is the real presentation here as opposed to earlier albums where the music seemed to take a back seat to Todd's self indulgence as a production whiz. Effects are well used on this album. They are mostly along the line of special effects of a theatrical nature used only to reinforce or compliment the musical statement (i.e. canned applause, water gurgling, etc.) Any other effects are mostly rather subdued synthesizer passages used to set a particular mood. The vocals are pretty much left clean which is good news because they've never sounded better. There's not much lead work on the synthesizers this time around either.

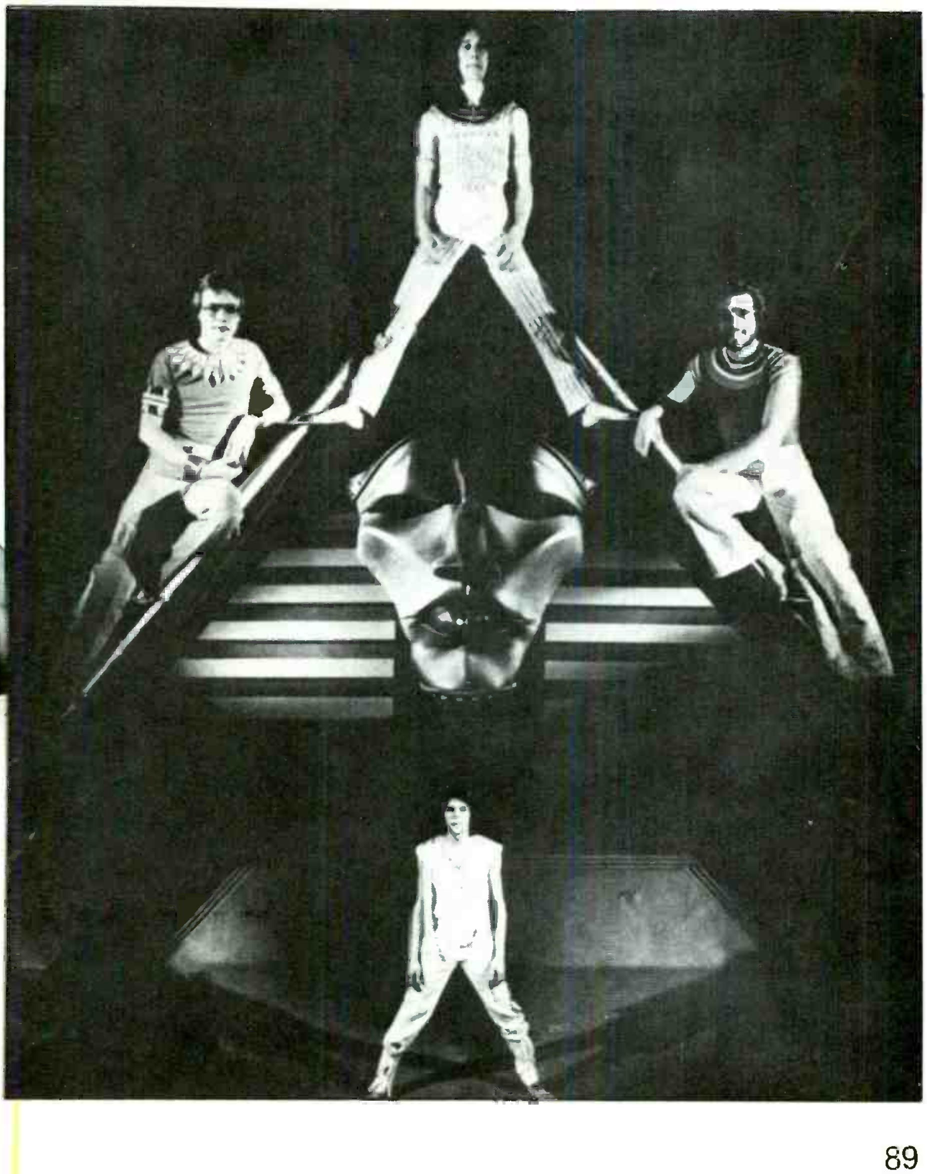
Utopia seems to have settled down to creating the one thing that this album is full of—good music. C.F.—K.

NILS LOFGREN: *I Came To Dance.* [Nils Lofgren, Andy Newmark, producers; Bob Dawson, engineer; recorded at Bias Recording, Falls Church, Va.] A & M SP 4628.

Performance: **Time to clock in**
Recording: **Dull**

Before I heard this album, I met Lofgren. During the course of the interview he repeatedly stated that he felt this was the best album he'd ever done. The reasons for his exuberance were 1) he didn't play piano at all on this recording; 2) this was truly his firt band (vs. solo) LP; and 3) rather than doing it track by track, they rehearsed as a unit and cut

UTOPIA: They've never sounded better



NILS LOFGREN: Better go West, young man

"live." Remembering his rock and roll punk image, and more importantly that great, "live"-in-the-studio-album pressed only for radio and press people, I was really expecting to be blown away. Ten seconds in to the first cut and I knew it was all over. As the saying goes, he's gone commersh. Strings, horn overdubs, female back-up, everything you'd never expect—or want—from Lofgren. He was only interesting before because of his tasty raunchiness. Now that he's toned down, I suffered listening to it several times, to the point where I had to take it off.

The production is at fault here, along with Lofgren's weak songwriting. As is often the case with studios that don't consistently see the traffic of major artists, Bias Recording shows signs of being inconsistent and generally out of date. The horn overdubs came out sounding so faint and compressed that they added nothing but nails to the coffin. Nowhere does the energy associated with Lofgren emerge. His guitar is subdued, almost tame, and is constantly overshadowed by the acoustic piano. There isn't one original production or engineering idea in evidence. To my ear, this is the farthest thing from a group effort possible, sounding more like the best reason why not to record track by track. If there is one overall reason why this record failed, it is that, and fault must lie primarily with the studio. Without the sizzling sound of instruments frying, Lofgren's lyrics come through without support and are forced to stand alone. Unfortunately, they're not up to the task. The mix is so average and lifeless that I needn't describe it—you've heard it so often it's second nature. This being the first time he's recorded on the East coast, I suggest he return to L.A. and the supervision of David Briggs and Al Kooper. G.P.

JAZZ

WEATHER REPORT: *Heavy Weather.* [Joe Zawinul, Jaco Pastorius, Wayne Shorter, producers; Ron Malo, Jerry Hudgins, Brian Risner, engineers; recorded at Devonshire Sound Studios, North Hollywood, Ca.] Columbia PC 34418.

Performance: **Exuberant**
Recording: **Quite enhancing**

One of the more melodic units recording in the jazz vein, Weather Report exhibits some fine compositions by keyboardist Joe Zawinul, saxophonist Wayne Shorter and new bassist Jaco Pastorius. With the addition of Latin percussionists Acunda and Bachena, the combination of the Latin influence and the composers' various use of space make this a beautiful recording musically. Without an excellent recording, this would never have come to pass.

The subtle use of echo and delay really make this recording for it suits the purpose of underlying that use of compositional space. Overall, the drums are given the fading echo treatment which adds a great deal to the feeling of motion in the pieces. In slight contrast, the Rhodes is echoed with quick delay, allowing for the distinctive entry of Shorter's sax or Zawinul's synthesizer

WEATHER REPORT: Excellent outing

to be emphasized by the contrasting still-like clarity.

The mix is deceptive for each instrument is brought to the forefront such that the usual sound of instruments dropped back or out of the mix is pleasantly absent here. An interesting example is "The Juggler." The piece is opened with Latin percussion establishing the rhythm, giving way to a synthesizer playing a repetitive theme. Contrapuntal motion appears through the addition of acoustic piano in a jazz vein. This section of the piece is superbly accented by echo fading crisp cymbals. With a building military march in the background played solely on snare drum, Shorter's alto joins in emphasizing the main theme along with a percussively echoed Rhodes.

The overall engineering consistency makes this album flow naturally from track to track, and before you know it, you're playing it again. G.P.

THE PHIL WOODS SIX: *Live from the Showboat.* [Norman Schwartz, producer; Keith Grant, Dale Ashby and Bill Ashby, engineers; recorded at the Showboat Lounge, Silver Springs, Md. November 18th and 19th 1976.] RCA BGL 2202.

Performance: **Beautiful playing filled with surprises**
Recording: **Like it says—"Live"**

Phil Woods has been labeled as a bebop

PHIL WOODS: The audience loved it

saxophonist and that's not really fair. Whatever Phil Woods has learned from others he has crafted into his own unmistakable style. But just as important as what Phil Woods plays on alto and soprano saxophone is the tight little band he's been working with for some time. The players, especially Mike Melillo on piano and Steve Gilmore on bass, are truly magnificent and they complement Phil's work expertly.

An example of what Phil Woods can do with a song is on side B of this double LP set. Phil plays some amazing out of the way changes on Irving Berlin's "Cheek to Cheek." Yet they sound so right that they sent me scurrying back to the 1935 Fred Astaire recording to check if that was how the song really went. This is only an example. The album's full of the sort of curves that Phil Woods throws at listener when he's not expecting them. Who, for example, but Phil Woods would conceive of "Django's Castle" as a bossa nova?

The recording is just like it happened on the stand, culled from four sets that the band played at the Showboat. The sound you hear came right from the house amplifying system with only a little extra miking required from time to time. The remote truck of Dale Ashby and father served as the master control room and got it all down. Also much credit goes to producer Norman Schwartz who managed to get close to half an hour of music on each of the four LP sides in this set with no loss of fidelity. It's all there... the music, the announcements, and the general ambience of a fine jazz musician meeting an audience which understands and appreciates him. J.K.

MAYNARD FURGUSON: *Conquistador.* [Jay Chattaway, producer; Joe Jorgensen, engineer; recorded at Media Sound in New York City and CBS Studios and Wally Heider Studios in San Francisco, Ca.] Columbia 34457.

Performance: **Predictable but excellent**
Recording: **Layer upon layer of jazz/rock**

When Maynard Ferguson first burst upon the U.S. jazz scene as a member of Charlie Barnet's band in 1949 his high register technique and musicality made him noticed as something extraordinary. In fact, his recording of "All The Things You Are" with Barnet's band was recalled because the publishers thought it too shocking a performance. Today, Maynard Ferguson doesn't seem shocking at all. He hasn't changed his act much in 30 years. We know what's coming and he doesn't want to disappoint us. What Maynard Ferguson *does* do is attempt to pick unusual vehicles for his highly personal identifiable style. The Ferguson recording before this included "Vesti La Giubba" from *Pagliacci.* This one has the hit tune from the film, *Rocky*, and the theme music from Gene Roddenberry's definitve science-fiction TV series, *Star Trek*, plus three originals by Ferguson and producer Jay Chattaway and one by Bob James. The players include the regular Ferguson band, a lot of guests, and a surprise appearance by Grammy winning guitarist, George Benson. Of the sidemen in

MAYNARD FERGUSON: **Still some surprises left**

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Maynard's band, I'd like to call special attention to the flute playing of Bobby Militello whose work on "Star Trek" is particularly excellent.

The recording was done in layers, rock style, track on track. Yet there's a truth and wholeness to the sound which was achieved by having Maynard Ferguson play into an isolated mic during the recording of the rhythm tracks. This playing was piped into the headphones of the rhythm players making the tracks so they had something to play to and against instead of just a sterile vacuum.

A lot of care went into this LP and early chart returns (especially the extracted *Rocky* single) would seem to justify the pains taken with the project.

—J.K.

CLASSICAL

BRAHMS: *Symphony No. 1.* Berlin Philharmonic Orchestra, Wilhelm Furtwaengler cond. ["Live" recording· Sender Freies Berlin (SFB); recorded on Feb. 10, 1952 at the Titania-Palast, Berlin.] Deutsche Grammophon 2530 744.

Performance: **None better**
Recording: **Excellent mono**

Although the major thrust of record reviews in *MR* is to discuss production values of the most important new releases, certain historical recordings cannot be ignored. There are many fine stereo recordings of the Brahms First Symphony: Haitink's dramatic Concertgebouw performance on Philips and Boult's autumnal reading on imported EMI are my favorites, with the Kertesz, Levine, Stokowski, Walter and Ansermet recordings only slightly less noteworthy.

WILHELM FURTWANGLER: Seeking philosophical depths

The Haitink and Boult recordings were in fact the first performances ever to convince me that this symphony was more than pompous, flatulent turgidity. Both of these conductors cut through the opaque orchestration with a taut grip on tempos, incisive string attacks, careful balancing of winds and brass, and insistence on clear-cut rhythms—yet never sacrificing the work's inherent weight of utterance.

Furtwaengler, on the other hand, approached the symphony from the German romantic tradition, with broad tempos, massive chordal attacks which give the orchestral textures a burly, sonorous spread, and a spontaneity rarely encountered in our day of prophylactic music-making. Rather than concentrating on technical perfection or symphonic structure, he sought the philosophical depths and elemental power behind the notes.

To be sure, Furtwaengler's musical solipsism of the present moment could yield extremely idiosyncratic performances. But at best, as in the Brahms First, his inspirational conducting had enormous sweep and a sense of inevitability that stilled all doubts. This newly released concert performance, recorded "live" in 1952, is played with extraordinary passion and commitment (if not perfect ensemble) by the Berlin Philharmonic. One easily accepts the occasional untidy *accelerando* as the product of enthusiasm over a score which all too often seems stolid and morose. Comparison with his fine 1947 Vienna Philharmonic studio recording on Unicorn reveals a very similar interpretation, but with palpably less excitement and abandon; it's worth owning, but Furtwaengler rarely operated at his peak when getting all the notes in place was a major consideration.

The sound is strikingly good for its day, with full, rich orchestral blend and an apparent absence of limiting. Some artificial resonance has been added to the final chord, presumably to mask the cut-off before applause at the end. D.G.'s surfaces are fine, reflecting this label's recent return after several years to high quality.

S.C.

Advertiser's Index

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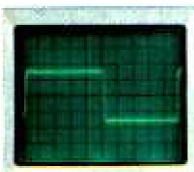
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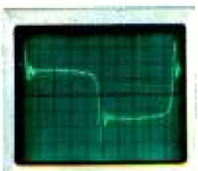
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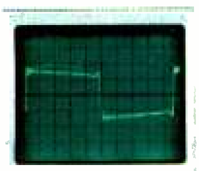
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