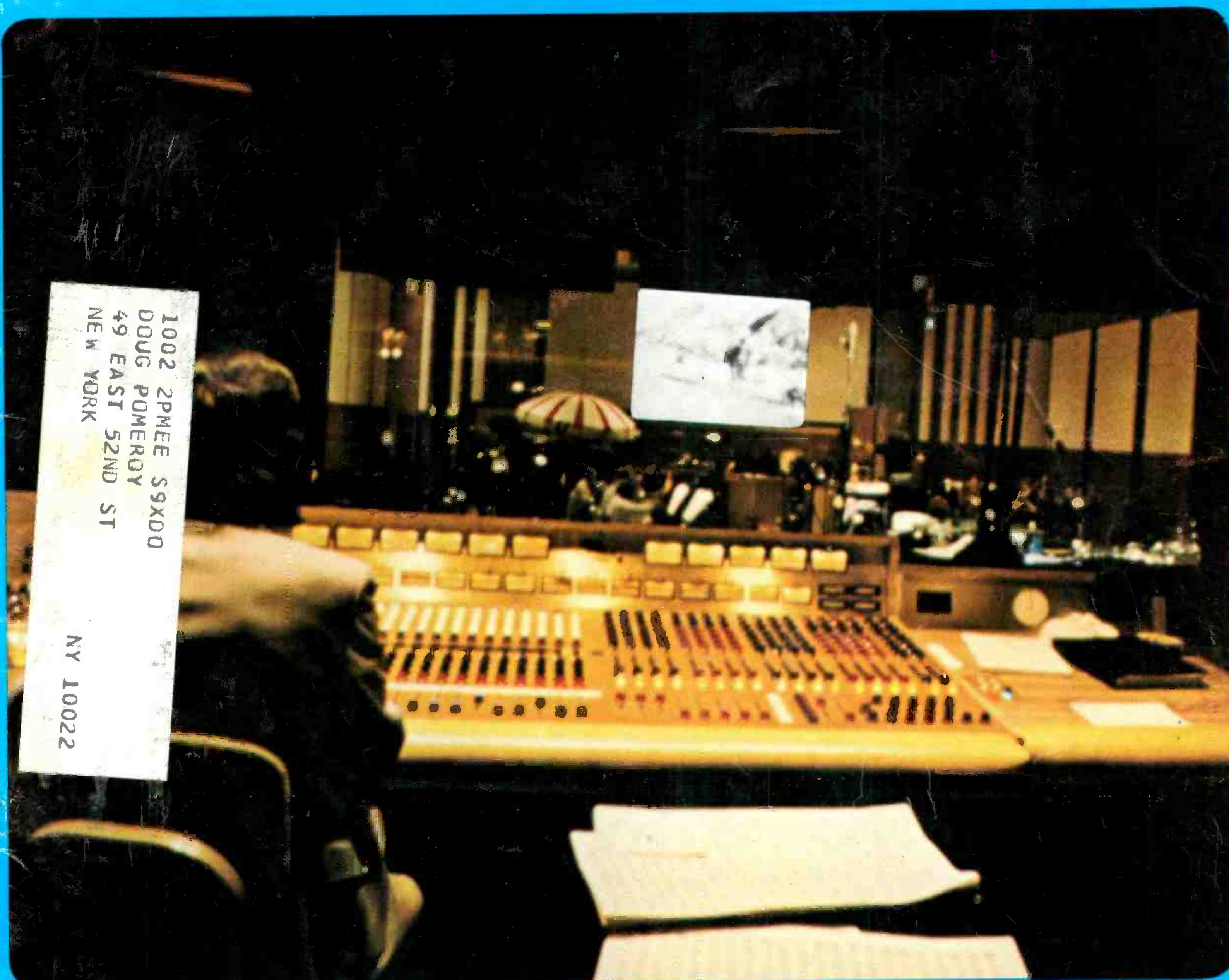
VOLUME 5 – NUMBER 1
JANUARY/FEBRUARY 1974

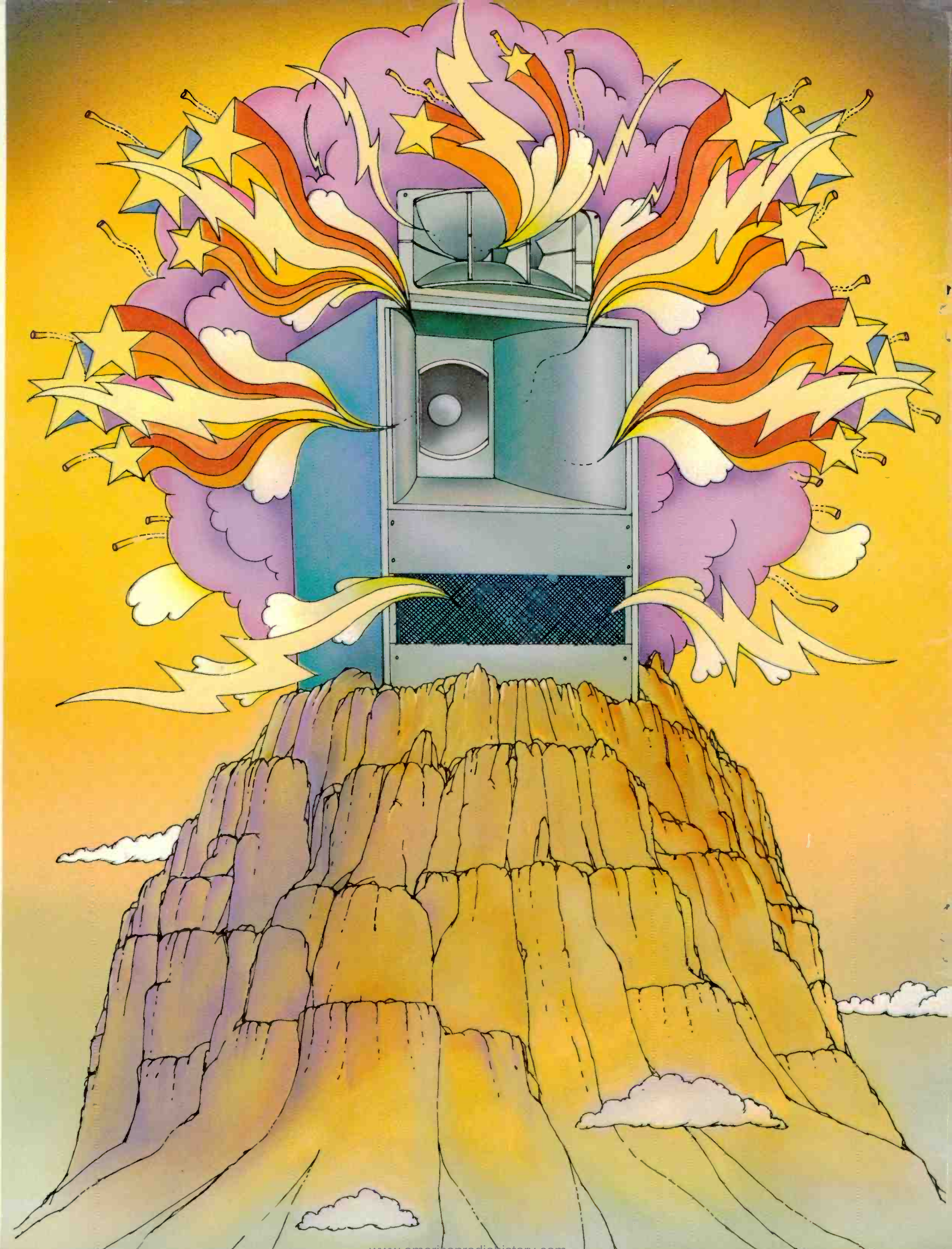
INCLUDING: CONCERT SOUND REINFORCEMENT

Recording engineer producer



The Burbank Studios . . .
'Film Score Mixing' – Page 18

RELATING RECORDING SCIENCE • TO RECORDING ART • TO RECORDING EQUIPMENT



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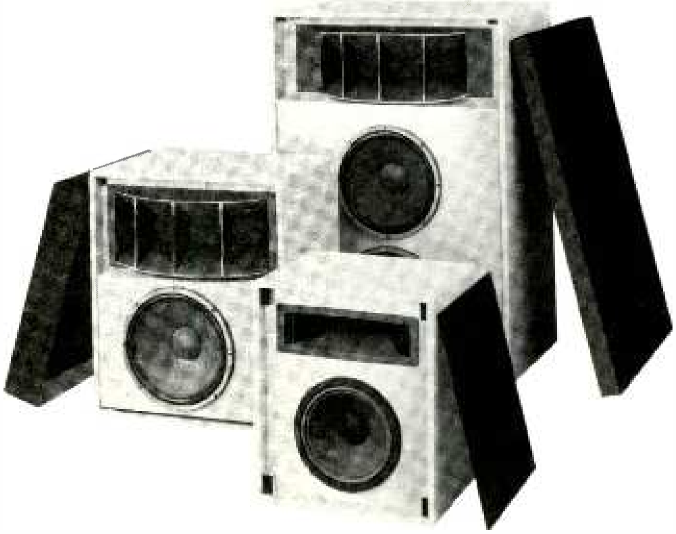

The challengers.

From front to back, it's the 9849A, the 9846-8A and the 9848A. If you listen for a living, you should know more about them. Write or call. We'll send you all the facts and figures.

Circle No. 101

RECORDING
engineer/producer

– the magazine to exclusively serve the recording studio market . . . all those whose work involves the recording of commercially marketable sound.

– the magazine produced to relate . . . RECORDING ART to RECORDING SCIENCE . . . to RECORDING EQUIPMENT.

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RECORDING engineer/producer is published bi-monthly and is sent free to qualified recipients in the United States. Subscriptions for other than qualified individuals or companies may be purchased at $5.00 per year. (All foreign subscriptions: $6.00 per year.) Material appearing in R-e/p may not be reproduced without written permission of the Publisher.

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Controlled circulation postage paid at Los Angeles, California.
RECORDING engineer/producer
6430 Sunset Boulevard
P.O. Box 2287
Hollywood, Ca. 90028
(213) 461-7907

Circle No. 102

JANUARY/FEBRUARY 1974
VOLUME 5 – NUMBER 1

bvd Visits - Allison Research

by Elas Tikshorts

Allison's founder, Paul Buff, greets the infamous author, Elas Tickshorts, at the main entrance to the Allison complex.

This month, we turned our cameras southward to Nashville, Tennessee. The trip down was fairly uneventful, however, the endless rows of white lightning stills did serve to break the monotony. There seemed to be no energy shortage here, what with the columns of chain gangs prodded along by red necked sheriffs.

When we arrived at our destination, we were given an unforgettable lesson in southern architecture. The Allison factory appeared no larger than a good sized station wagon, yet purportedly housed some 80,000 employees. Allison's engineering chief, Paul Buff, informed us that it was done with mirrors.

Once inside, we were given a warm welcome by Allison herself and by Chuck Dunlap, Allison's production wizard. Chuck immediately guided us to the production complex and showed us the secret to Allison's success, the prized company soldering iron. He explained that the community iron approach saved the company over $750,000 per year. He went on to explain that the problem of long lines of prospective solderers was solved by a former California Department of Motor Vehicles executive, who devised a "take a number" system.

At this point, Allison appeared with a delightful tray of mint juleps and invited us to tour the executive offices. As we relaxed in her 20,000 square foot luxury office, Allison proceeded to explain the workings of the interoffice communications system. We went to a large map on the wall, where Allison pointed out 11.6 miles of air tubes which rush messages throughout the plant. Powering for the complex system is provided by a souped up Chevy 409 engine which was salvaged from an old company car.

Just then the phone rang. It was Paul. We didn't pursue the subject, but when Allison answered, I would swear that she was speaking into an old soup can with a string on the end of it!

Paul was in trouble in the lab. As we rushed to his aid, Allison began to expound on his unique drafting system. It seems that Paul uses a modified "etch a sketch" to produce tape up drawings of his printed circuit boards. Special shaft encoders have been attached to the "etch a sketch" controls. The outputs of these encoders are fed to an IBM 360 series computer which in turn drives a large X-Y plotter which has been modified to apply drafting tape to a mylar surface.

As we approached the door to the lab complex, we could discern distinct cries of, "HELP". Upon entering the drafting area, our eyes beheld a very unique situation. There was Paul laying on the X-Y plotter table, completely bound with a myriad of black drafting tape. Overhead, the plotter was merrily chugging along, unmercifully pasting more tape and little black doughnuts over the farthest extremities of Paul's helpless body! Paul became silent as the relentless mechanism slapped a very large wad of tape over his mouth.

Chuck raced for his master power switch, which responded by relinquishing its tattered handle to Chuck's grip. Chuck began to cry as he held the broken lever and the plotter taped on.

Just at the moment we thought all was lost, the hulking monster stopped dead! It had run out of tape!

As we began to free Paul's bonds, Allison excused herself. It was lunch time and she had to prepare 80,000 portions of food for the employees. As she left, Allison muttered a sentence that I didn't understand, something like, "We're having tomato soup today cause we need some more phones".

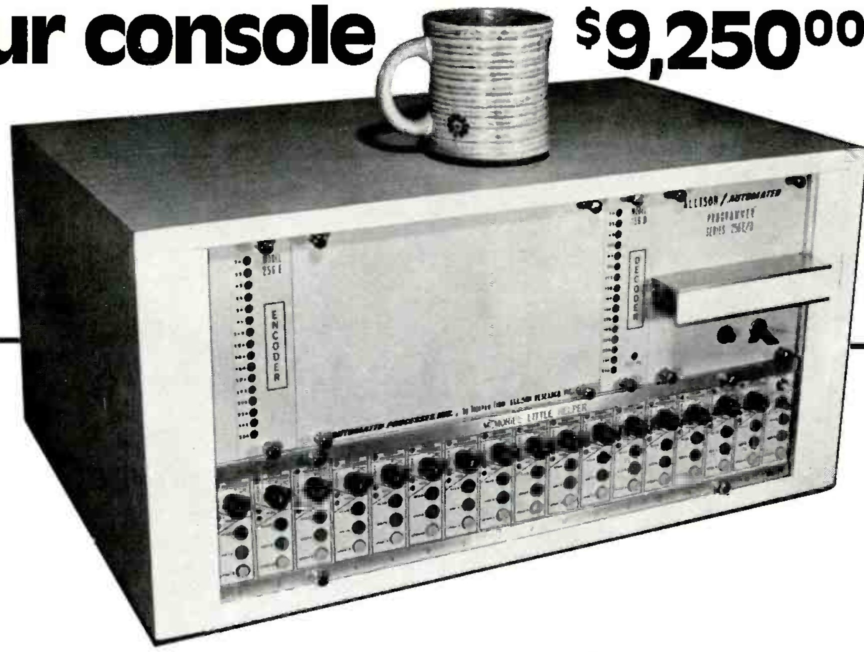
Here's the incomparable Allison company soldering iron. Note the special tip used for microcircuit work.

That's Allison herself, talking on the phone with another satisfied customer.

Chuck Dunlap is explaining the safety features of the new company vehicle to Allison. (She is about to buzz up to the post office for stamps.)

A well rounded design facility plays a major role in the creation of top level audio equipment.

automate your console $9,250^{00}



MEMORIES LITTLE HELPER, it's even "good-er" than it is cheap! $9,250 buys a *complete* 16 channel automation package for *your* board. You can hook it up in one day and start enjoying the benefits of automated mixing the next day. You do the mix right there on your good old console, same faders and all.

The heart of MEMORIES LITTLE HELPER is the now famous Allison/Automated Programmer. (You know, the one that works!)

Its highly unique 5 level digital/analog code offers capacity and proven performance way beyond conventional methods.

Your precious audio passes through our brand new VCA2-1's, whose job is to control gain and not add distortion or noise.

An awful lot of human engineering and recording engineer/producer experience has gone into MEMORIES LITTLE HELPER and is evidenced by the following features:

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- Remote, local and master mode controls (Read, Write & Update)
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- Over-range indicators
- Plug in portable applications
- So on and so forth

NEW PRODUCT ↑ ↓ **NEW WARRANTY**

P. S. If you're thinking "Machine Replaces Record Producer," you are dead wrong and we can prove it! There's a whole lot more to console automation than making mixing easier. Call us or circle us and we'll show you.

3 YEAR WARRANTY

All MEMORIES LITTLE HELPER equipment is warranted by the manufacturer, Allison Research, Inc., as follows:

Should any equipment malfunction occur within a period of 3 years from date of purchase, Allison Research, Inc. shall, at its option, repair or replace said defective equipment, at no charge to the customer.

Defects caused by customer misuse or abuse shall not be covered under this warranty. It is, however, Allison's policy to make repairs to accidentally damaged equipment on a minimal charge basis.

Equipment to be repaired should be sent, postage paid, to the factory, together with correspondence indicating the nature of the malfunction.

Return shipping charges will be paid by Allison Research, Inc. No other warranty, express or implied, is herein made.

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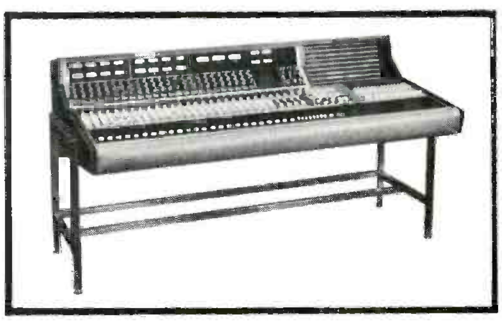
Circle No. 103

Creative producers, mixers, and musicians agree that Automated Processes' consoles "have what it takes."

Now the Model 2488 console is available, incorporating the quality, flexibility, versatility and sophisticated advantages of Automated's leading console technology.

Without compromising our rigorous standards, creative engineering has made possible its production at an attractively low price.

Features of the Model 2488 include: up to 24 inputs, each with 440 Fader, 550A Equalizer, direct output, LED peak indicator, panner, high pass filters, mic trim, phase reverse, and send to 2 cue lines and 4 echos; 8 mixing busses switchable to 16 tracks; simo stereo and mono; solo on inputs, echo, monitor, and cue; monitoring and metering for mono, stereo, quad, and up to 24 tracks with overdubbing facility; 4 cue busses; limiters; oscillator; modular power supplies; total plug-in installation; and much more. The Automated modular design permits other features and options to be added at any future time, including Mixdown Memory!



Choosing a console is a decision you must live with ... "underbuying" can be a mistake. Purchase the console that is appropriate to your needs with adequate provision for future expansion . . . Automated Processes' quality, reliability and state-of-the-art engineering is a combination to satisfy your most demanding session.

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Circle No. 105

Letters & Late News

from: **BRAD MILLER**
Mobile Fidelity Productions Inc.

Concerning the "3 WAY CONTEST FOR A 4 CHANNEL FORMAT" I believe that the article by Gary Davis was acceptable, as far as it went. In my opinion, the information that related to CD-4 was dated by at least six months. Now, this cannot be attributed to Gary, but due to the lack of communication and common sense exhibited by the source of Gary's information in most cases.

For example; many of the so called limitations in cutting CD-4 discs were self imposed by JVC because of the poor performing qualities of their consumer version demodulator. This does not speak poorly of CD-4 as a system; it does speak poorly of manufacturers who provide less than optimum quality.

Secondly, the prices quoted for the SQ decoders and the CD-4 demodulators showed a great price difference and I am not surprised. The SQ quotation was for the i.c.; the CD-4 quote was for the demodulator, in the box, with the retail (list price) figure given.

Thirdly, why not acquire disc cutting information from an individual who is responsible for his labels CD-4 efforts, such as Lee Herschberg (Chief honcho at Warner Records). Afterall, Lee has had to live with CD-4 from its inception, and I am positive that he could offer a much needed and objective opinion, rather than reading from the JVC manual.

Fourthly, a super CD-4 i.c. chip has just been released to all manufacturers by Lou Dorren. You all remember Lou, he holds the Patent for a discrete quad broadcast system, and performed the necessary surgery on many JVC demodulators, including mine, so that we could properly evaluate the system. And not one conversation was held with Lou, when this article was written.

Fifthly, yours is still the best, so keep it up, or else!

Reply from:
AUTHOR GARY DAVIS

Thank you for your comments on the quad article, and for the additional information you forwarded to us. We did make every attempt to use the latest information in our article, including interviews and phone calls that were made the week prior to publication. The news of the CD-4 chip was not yet released, to our knowledge, when the article was typeset late in November.

Insofar as the decoder-demodulator comparisons, you are right that com-

paring the OEM and consumer pricing might be misleading. However, we tried to extrapolate the consumer value of the OEM equipment because, at the time, we had no access to OEM prices for the CD-4 circuitry.

As we pointed out, the available quad hardware could, and probably would change; it did. Not only has the QSI/CD-4 chip subsequently come to our attention, but we understand that a CD-4 cartridge and stylus is now available from Grado for under $12 (FTR + 1).

We only hope that we have helped to clarify some of the aspects of quad which have often been ignored or overlooked. The response from knowledgeable readers, like yourself, will certainly help all of us learn more about where quad is at.

From: R. STEVEN MINTZ
Chief Engineer
Custom Sound Productions
New York, N.Y.

With regard to the recently published article by R.B. Annis "Notes on Demagnitization", I would like to point out that the residual magnetism on recording and playback heads and guides on professional recording equipment is of such low magnitude that an attempt to measure it with magnetometers of the type manufactured by Mr. Annis would certainly make the situation worse. Such magnetometers contain a small permanent magnet on a spring-loaded vane. Bringing the device into contact with the head being measured will induce an amount of residual magnetism onto the playback gap of such great degree that it will adversely affect any tape passed over it without prior demagnitization of the gap.

Reply from: R. B. ANNIS
R. B. Annis Co.
Indianapolis, Indiana

. . . With reference to Mr. Steven Mintz's objection to the use of Pocket Magnetometers in conjunction with professional tape Recording equipment, I feel he is possibly confusing our Magnetometers with some, that at one time were made by G.E. Company, which did contain a spring loaded magnet on a probe which was introduced into the field being measured. Like "Toledo Scales" our pocket Magnetometers have no springs.

There is a small moving magnet system at the staff of the instrument, not close to the part being checked, however. On the 5 gauss instrument, the optimum range for use with tape recorders, any induced field in the part being measured would be negligible.

It might be interesting to point out that the first use of these Pocket Magnetometers, on magnetic tape equipment, was by Ampex. They have been using them in quantity for several years now. In fact it was through their efforts that we "woke up" to the fact that they were useful in this area.

Of the thousands that are now being used for measuring residuals in magnetic tape and similar equipment, we have had no reference to any difficulty due to induced magnetism.

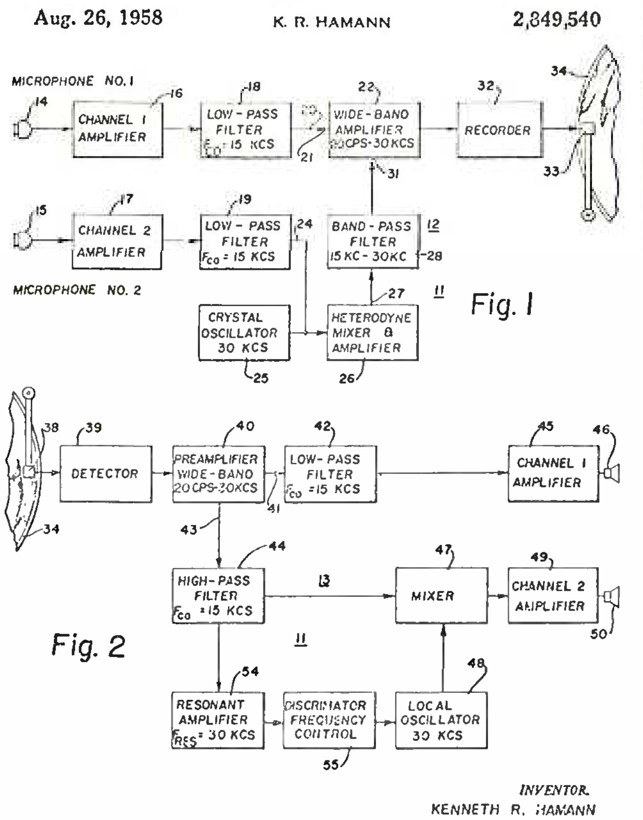
From: JOHN EARGLE
JME Associates

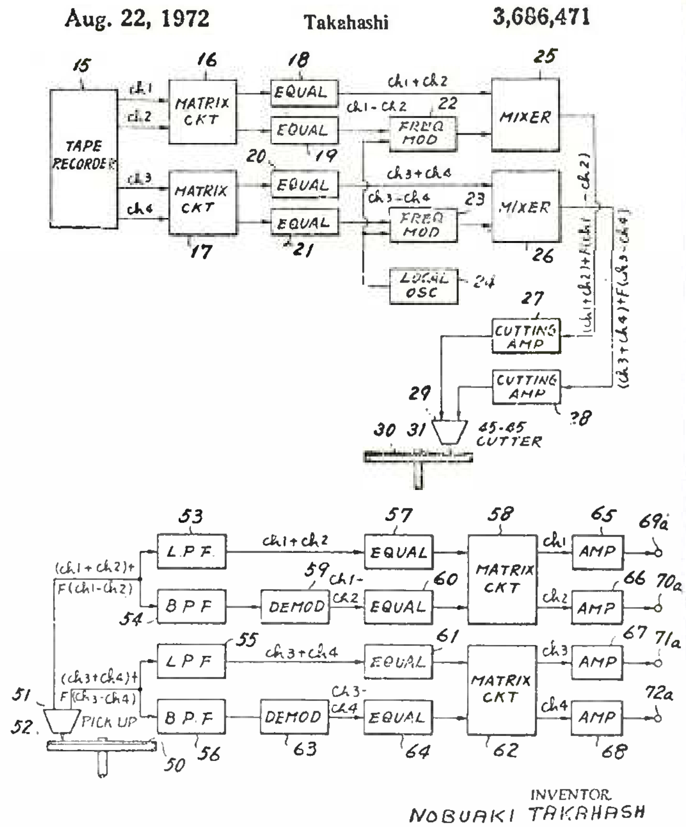
As a consultant to JVC, I have been asked to reply to a news item that appeared in your November/December edition.

The item in question was one which stated that the Cleveland Recording Company and the Victor Company of Japan were in a dispute over patents pertaining to the CD-4 disk. This is not the case. U.S. Patent No. 2,849,540, held by Mr. Kenneth Hamann, describes a recording system where one channel of a binaural or stereo program is assigned to the normal base band of a phonograph record. The second channel is heterodyned against a 30 kHz sine wave, band-pass filtered from 15 to 30 kHz, and then added to the base band signal. Upon playback, the base band signal is recovered through a 15 kHz low pass filter, while the other channel is recovered by beating the heterodyned components with a 30 kHz local oscillator, whose frequency is stablized by locking it in with the residual 30 kHz on the disk.

U.S. Patent No. 3,686,471, held by the Victor Company of Japan, clearly describes a system in which a pair of sum signals are allocated to the two base bands of a stereo disk, while a pair of difference signals are allocated, via a combination of phase and frequency modulation, to each wall of the stereo groove. The difference signals are recovered upon playback through standard FM demodulation techniques, while the base band signals are recovered in the normal fashion. These four signals are then fed to a simple sum and difference network which recovers all four quadraphonic inputs to the system. None of Mr. Hamann's twelve claims describes a technique of phase or frequency modulation.

The difference between a heterodyning approach and a frequency-phase modulation approach is a substantial one; the two techniques are as different as FM and AM. That both systems should use a 30 kHz carrier is not surprising; 30 kHz represents the lowest practical carrier





EXCERPTS FROM THE PATENTS INVOLVED

frequency if side band energy is to be accommodated to provide for 15 kHz overall system response.

Reply from: KENNETH R. HAMANN
Cleveland Recording Co.

Thank you for the opportunity to reply to John Eargle's letter.

First, I have never claimed to have invented the specific 4-Channel CD-4 recording system. The credit for this brilliant engineering achievement very properly belongs to Mr. Nobuski Takahashi and his colleagues at JVC in Japan.

What I do seek to establish is the claim that I did invent, in 1954, the very basic principle of utilizing a band of supersonic frequencies centered on a carrier signal of 30 kHz to convey multichannel stereophonic audio information on phonograph discs or other media. I am enclosing the block diagrams for the reproduction section of the JVC CD-4 system and the equivalent diagram from my system. The similarities between the two are readily apparent, even though, as pointed out by Mr. Eargle, there are significant differences in the recording method.

What is not noted, however, is the actual method of recovering the supersonic signal from the disc in the JVC system. In other literature published by

JVC, the use of a voltage controlled oscillator and a phase comparator is described as the "most important feature in the demodulator". It can be noted that the same technique is described in my system, and is indeed an integral part of all claims in the patent.

COLLEGE FOR RECORDING ARTS FOUNDED IN SAN FRANCISCO

A very unusual school has just begun operations in San Francisco. The facilities of Golden State Recorders, Inc., one of the city's major sound recording studios, will be used as home base by the COLLEGE FOR RECORDING ARTS. This school is completely dedicated to teaching all facets of the record and recording business.

The school will teach the music business literally from "beginning to end", offering courses which cover (but are not limited to) the following fields:

MUSIC: Basic Music Theory, Musical instruments and their use in recording, Types of Music, Methods of Music Creation and Control, Composing, Conducting, Songwriting etc.

MUSIC LAW: Basic Music Law, Contracts, Artist's Management, Music Publishing, Performance Rights and Performance Rights Societies, International Performance Rights, Artists Unions etc.

AUDIO CONTROL: Audio equipment and equipment applications, Mixing, Dubbing, Editing, Recording Techniques, Advanced Operational Techniques, Multitrack techniques, Equipment Maintenance etc.

BUSINESS and FINANCE: Basic Business Practices, Economics of the Record and Film business, Financing, Cost Control, International Operations, Product Packaging, Advertising, Sales Techniques, etc.

These courses require three Semesters (one full year) to complete. As a special offering for students wishing to prepare for a career as record company executives, there is a fourth semester devoted entirely to Record Company operations where participants are able to translate all previous learning into practical experience through the performance and transaction of actual record company business ranging from talent scouting all the way to the final sale of the finished product.

Instruction will be given by both a resident staff and by prominent guest lecturers from the industry. The majority of resident teachers have for many years been active instructors at various California Universities (UCLA, University of San Francisco, Columbia College), but they are also experienced professionals in their fields, i.e. practicing attorneys, Audio Engineers, Composers, Musicians, and Businessmen.

For more detailed information and a copy of the school's academic bulletin, write to:
COLLEGE FOR RECORDING ARTS, 665 HARRISON STREET, SAN FRANCISCO, CA. 94107

EARGLE LEAVES ALTEC TO FORM JME ASSOCIATES

John Eargle has left his position at Altec to form his own consulting firm. Eargle's new firm, JME Associates, will devote its full efforts to special interests in the quadraphonic hardware and software areas.

Eargle's background in the audio industry is extensive and diverse. Most recently, he spent 2½ years as Director of New Products for Altec Sound Products in Anaheim, California. Prior to joining Altec, Eargle had a long history with the recording business. In New York, he was employed by the RCA Record Division as manager of record facilities and recording quality, later becoming manager of quality manufacturing and recording for the industry giant. Eargle was also Chief Engineer for Mercury Records, a position he left to accept the post at Altec.

Eargle holds Masters Degrees in both Music and Electrical Engineering. He is currently a senior member of the Institute of Electrical and Electronic Engineers, the Acoustical Society of America and is

President-Elect of the Audio Engineering Society.

JME Associates is located in the RCA Building, Suite 533, 6363 Sunset Blvd., Hollywood, CA. 90028. (213) 461-4229.

"MIKE" AWARD TO GERMAN SCIENTIST

World-famous designer of microphones and disc recorders, Georg Neumann (center), shows his firm's managing director Guenter Luetzkendorf, the Maker Of The Microphone Award for 1973 just received from Peter Burkowitz (left), presenting the trophy in memory of Emile Berliner, inventor of the microphone and disc record. Burkowitz is engineering director of Deutsche Grammophon (Polydor), founded by Emile Berliner in 1898, and represented the Berliner family in this presentation, which is made annually for an outstanding contribution to the world of sound.

SLACK, AMES, IN NEW APPOINTMENTS AT CETEC

Donald W. Slack has been appointed as national marketing manager of Cetec Inc., manufacturer of professional audio equipment under the Cetec, Gauss and Langevin names. Cetec is a subsidiary of Computer Equipment Corp., El Monte, California.

Slack joins Cetec after 18 years in the audio industry, beginning his career with Stromberg-Carlson.

Prior to joining Cetec, Slack formed Environmental Music Inc., an originator of on-location background music systems and audio message devices.

Jack Ames has been named national sales manager for the tape duplicating and recording industry products.

Before joining Cetec, Ames was marketing director of Otari of America, Inglewood, Calif., and marketing vice president of TelePro Industries, Cherry Hill, N.J.

Ames was a co-founder of Liberty Records.

AUDIO PRODUCTS DIVISION ANNOUNCED BY BOUSE

Tom Bouse, President of Bouse Manu-

Circle No. 109

facturing Company, Newport Beach, California, announces the formation of a new division of the company to be responsible for the manufacture and marketing of their professional multi-track studio tape recorders.

The current model of the 16 track recorder manufactured by Bouse will also be given a new name. Formerly known and publicized as the PRO MASTER 16, it is now the APD 1600.

William F. (Bill) Jones, who was in charge of marketing the PRO MASTER 16 through Custom Fidelity Inc., has been appointed Marketing and Sales Manager for the newly formed Audio Products Division (APD).

STUDER AND WESTLAKE AUDIO SIGN SALES PACT

A sales agreement between Westlake Audio, Los Angeles, and Willi Studer, America, U. S. marketing division of Studer Franz AG, Swiss manufacturer of highly sophisticated tape recording systems, was signed last week, according to Tom Hidley, Westlake's president.

The Studer line will be available through Westlake and demonstrations can be arranged by contacting Westlake.

As a result of Studer's recent successes in building international distribution and sales, Hidley reports that significant reductions have been made in Studer pricing which should enlarge the U.S. demand.

Westlake will soon receive the most advanced Studer unit, their 24-track Model A80/VU-24-2". It will be the first shipped to the U.S. and, says Hidley, boasts extraordinary transport stability and driving reliability.

BOB BUSHNELL NAMED MANAGER OF MARKET DEVELOPMENT AND SALES ENGINEERING AT PACIFIC RECORDERS & ENGINEERING CORP.

Jack Williams, President of Pacific Recorders announced this month the appointment of Bob Bushnell.

Bushnell has been active in the recording industry since 1953. He had been until recently president of Bushnell Electronics, a manufacturer of custom audio consoles for the past six years. Previously he was in sales engineering for U.R.E.I., North Hollywood.

At Pacific Recorders Bushnell will be responsible for market development and sales engineering of the company's broadcasting and recording audio products.

PLAN NOW: to attend
48th AES CONVENTION
LOS ANGELES HILTON HOTEL
May 7 through May 10, 1974

Circle No. 111

16 track . . . 3 track, 35 mm . . . 6 track, 70 mm, stereo
. . . optical stripe . . . magnetic stripe . . .
stereo . . . quad . . . the challenges
of multi-format . . .

FILM SCORE MIXING

. . . interviews with KEVIN CLEARY, scoring mixer at the Burbank Studios . . . and engineer STAN POLINSKY.

KEVIN CLEARY, with a music degree from Sydney's Conservatory of Music, began his professional audio career during the 40's with Columbia Records in his native Australia. The years that followed saw Kevin working, with the Canadian Broadcasting Company in Toronto, and many other recording and orchestrating assignments, toward his status today as one of the world's few full time feature film and TV scoring mixers.

In addition to the current production of 'MAME' (Lucille Ball and Robert Preston) several of Cleary's more recent and memorable credits include the scoring of the Streisand musicals, 'Funny Girl' and 'On A Clear Day', as well as 'Paint Your Wagon', 'Rosemary's Baby', 'Hawaii' . . . .

by: WAYNE YENTIS
RAINBOW RECORDING
SANTA BARBARA, CA.

R-e/p: For this production of "Mame" what were some of the more significant problems you encountered?

KEVIN CLEARY: Well, Fred Werner, the composer, from the beginning had the idea that he wanted to have a "pit band" sound in Mame, rather than a normal movie type sound, you know the kind of underscoring where you see 5 people on the screen and all of a sudden you've got 40 violins and you wonder where that kind of orchestra came from. Fred wanted an authentic "pit band" sound because a lot of the numbers are stage type orchestra numbers. And I think he achieved it very well in the writing and the orchestrating. As the picture progresses, the music changes in character because so much time passes in the story that there's a significant change in the style of music. Fred orchestrated it that way so the music got more and more contemporary as it went along. The underscoring was more or less conventional movie type underscoring.

R-e/p: How did you go about achieving a "pit band" sound?

KEVIN CLEARY: I miked it looser than I normally would have. It wasn't as tight as I would normally have miked it on the drums and percussion, and I used quite a bit of overall miking. In fact, I always use a lot of overall miking in motion picture scoring, usually 3 channels of overall. In most of the scoring for motion pictures done on this stage I'd say 60% of the sound is overall. There may be 40 microphones out there but 60% of it is overall. The rest is what you might call sweetening, or reinforcement.

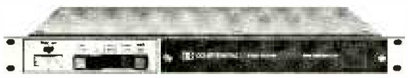
R-e/p: Is there anything in particular that you face as a film scoring mixer that is notably different from what an album mixer or recording engineer might encounter?

KEVIN CLEARY: I'd say the main problems are in the area of studio setup. In the record field, if you're doing an album with Frank Sinatra, for example, it's a set orchestra and vocal, you can set them up the way you want them and handle all your isolation problems, and that's it for the session. If they're going to do something that requires a different setup then that'll be another session. But in the motion picture field you can come in here at nine o'clock in the morning for a double session and for the first 2 hours you may have a 70 piece orchestra playing symphonic type music that's orchestrated-to-balance on the floor. The next thing you know you're doing a cue with a brass band within that orchestra, and then they're broken down to a bossa nova group, and then you're doing a rock group, for the various scenes in the picture. There isn't time to make separate sessions for these different setups. So that poses a problem; you have to find out ahead of time what types of music they plan to do at the session and then make a compromise setup. But we make the compromises as small as possible, and slanted toward the main score that's going to be done. For example, at a session you find that 19 of the cues are going to be symphonic; underscoring cues and main titles and

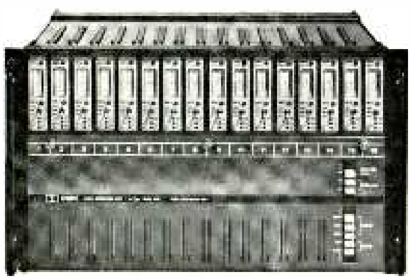
There is a Dolby noise reduction unit for every professional application

Professional Recording and Transmission Applications

360
The Dolby 360 is a basic single-channel A-type noise reduction unit for encoding or decoding. This unit is normally used in a fixed mode such as in disc cutting or landline sending or receiving; the operating mode is manually selected.

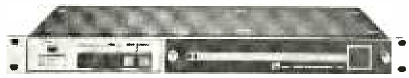


361
The Dolby 361 is similar to the 360, providing a single channel of A-type noise reduction, but with relay switching of operating mode and tape recorder connections. The changeover can be controlled automatically by the recorder.

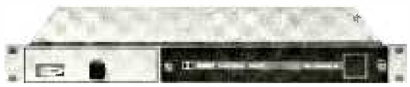


M-Series
The Dolby M16 A-type unit is designed specifically for professional multi-track recording, and incorporates 16 channels of noise reduction in a compact chassis only 10½ inches high. The similar M8 is an 8-track version, and the M8X allows simple extension of the M16 for 24-track use.

Motion Picture Industry

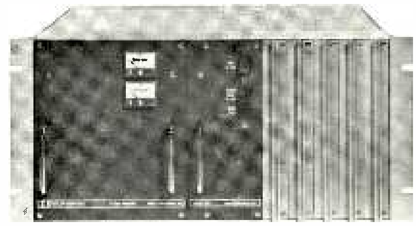


364
The Dolby 364 Cinema Noise Reduction Unit is intended primarily for use with Dolby A-type encoded optical sound-tracks. The 364 also includes a standard 'Academy' filter for conventional tracks, a clean-up circuit for old or worn prints, and provision for playback of magnetic sound-tracks with or without Dolby system encoding.

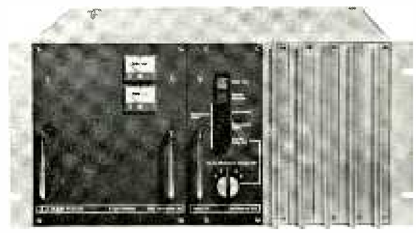


E2
The Dolby E2 Cinema Equalizer is a companion unit to the 364, and has been specifically designed to solve the response equalization problems of cinemas. Used with the 364 and Dolbyized optical sound-tracks, the E2 enables most cinemas to achieve modern sound reproduction standards without replacement of existing equipment.

Professional Encoders for Consumer Media

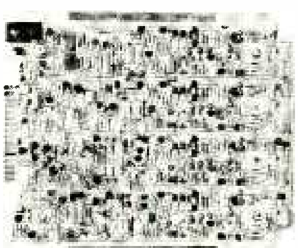


320
The Dolby 320 Duplication Processor is a professional quality unit with B-type (consumer) noise reduction characteristics. The unit is used for encoding duplication master tapes in the high-speed duplication of Dolbyized cassettes, cartridges, and open-reel tapes. The 320 is a two-channel unit.



324
The 324 Broadcast Encoder allows broadcast stations to encode stereo FM broadcasts with the Dolby B-type characteristic. The unit provides for an optional reduction of high frequency pre-emphasis, reducing the need for high frequency compression, and thus allowing a significant additional improvement of reception quality.

Noise Reduction Module



Cat 22
The Dolby noise reduction module, Cat 22, is the basic functional unit employed in all A-type equipment. The Cat 22 is available as a spare or in quantity to OEM users for factory installation. A half-speed version of the module (Cat 40) is also available.

Test Set (A-type)

Cat 35
The Dolby NRM Test Set, Cat 35, permits rapid verification of performance of Cat 22 Noise Reduction Modules without their removal or the need for additional test equipment.

Dolby™

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Circle No. 112

things, and then there's one bossa nova and one rock tune. We're not going to set up for those, we'll set up for the symphonic ones and let the others take care of themselves. You don't have time to change the set up. If there's 10 minutes left and we have a rock cue and a brass band cue and a dixieland cue to do, you've got to do them wherever they're (the players) sitting. And it's very difficult to get a sound on a rock group when they're spread out all over. You know, you've got a trumpet here, a guitar there and a piano over here and then someone says, "Oh yeah, and by the way we've got a French horn back there," and it's spilling all over the place. If I have the time I'll always try to move them in and screen them and make a different setup, another compromise setup, but usually there isn't time to do that.

R-e/p: Is there anything on the console you can do to bring it all together?

KC: Certain things, yes, but I find the secret in getting a sound from any particular group is in the setup. If you have a bad setup there's nothing you can do in the booth.

R-e/p: Well, just what do you mean by a good setup?

KC: A good setup is where the musicians feel comfortable playing, where they can relate to each other. So that the guitar player isn't too far away from the piano player, and so forth, and also a setup that gives me the isolation that I need for the various mikes. I don't want problems with brass leaking on the string mikes or bass leaking into the piano. It's a juggling process of setting up so they feel comfortable and I know I can get a sound out of it. Sometimes it's hard to visualize ahead of time. I've made setups on paper and the stageman set them up, then I've come in and had to tear them half apart because when all the chairs, stands, and screens were in I just knew it wasn't going to work.

R-e/p: You always work with a score, don't you. Doesn't that make things easier?

KC: I always use a score when I'm working, always. I don't know how you can mix without using a score, really.

R-e/p: But record mixers rarely use scores.

KC: But they usually have an 'A and R' man in the booth, and in our business you seldom have anyone in there. Very seldom. Once in a while if the composer has something in mind that is really out of the ordinary, and he's worked closely with the orchestrator in achieving a certain effect, he'll come in the booth. But normally there's nobody in the booth except me. So if you can't read a score, you're kind of flying blind.

R-e/p: On Mame you recorded the score on 16 track. Is this out of the ordinary for recording motion picture sound?

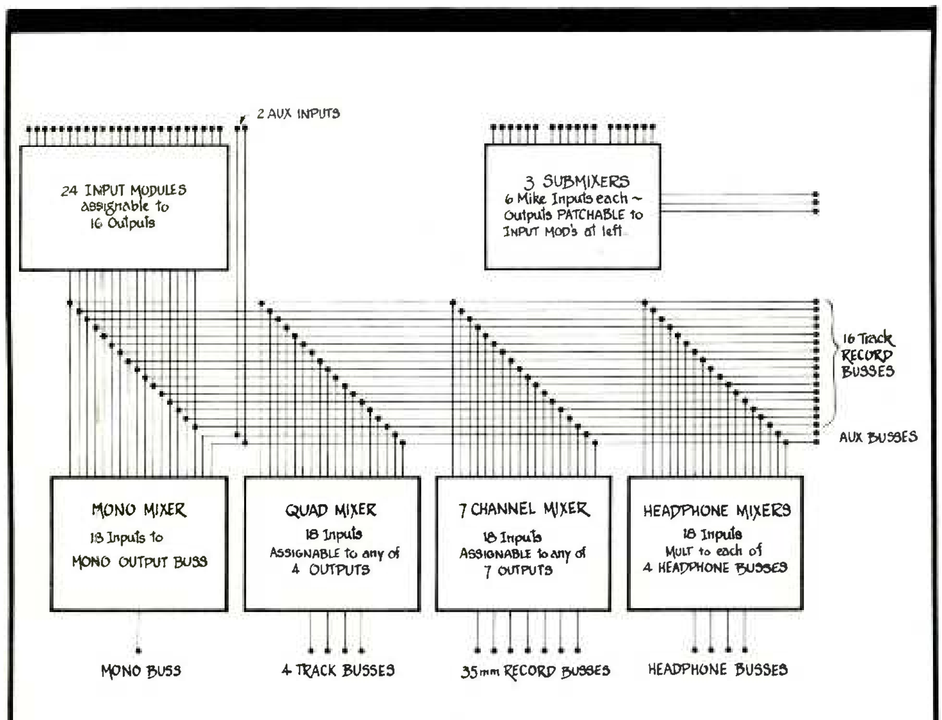
KC: On Mame, and on most of our motion picture scoring, we record simultaneously on 3 track 35mm film stock and 16 track, and that poses balance problems. We're working two mediums. In record recording when you're recording on 16 with an eye to mixing down later you want to put as much level on each track as you can. But at the same time we're recording a 3 track stereo master which is ready for cutting by the music editor and then put into the dubbing units for the picture. So this 3 track master has to be in perfect balance. We compromise again, usually we try to get the 3 track as good as possible and let the 16 track ride along, keeping a general eye on the levels. Consequently some tracks are going to be light on level. Particularly percussion tracks, or harp, or something that normally you don't want sticking out in your mix. It'll be fairly low level, but then we Dolby it, all the 16 track is Dolbied.

R-e/p: How do you feed your 3 track machine, from a mix taken from the 16 track machine?

KC: Actually there are two completely separate feeds. The 16 track is getting various instruments or sections designated to a particular track. And at the same time, groups of those tracks are being assigned to the left, center, and right tracks of the 3 track 35. For example, on the 16 track machine the violins may be going to track 1, the harp to track 2, piano to track 3, and guitar on track 4. But on the 35 all of those may go to track 1, which is a matter of assignment to an output buss on the board. We actually use the monitor buss to feed the 35. The monitor buss can feed 7 channels, so we could even record on 6 track 35 and be recording a separate vocal track on another 35mm machine, as well as recording on the 16 track, all at the same time.

R-e/p: Is the 3 track 35mm format the standard recording medium in motion pictures?

KC: 3 track 35 is only one of the standard formats. There are also 4 and 6 track formats on 35mm film. But most commonly we record on 3 tracks. It produces a recording that the dubbing mixer in the dubbing room later can rebalance to some degree. Normally he will take those 3 tracks and set them up straight across and ride the overall level against the dialog and sound effects. But strange things happen. A cue that sounded marvelous on the stage and everybody loved, you'll get in the dubbing room and some sound effect will completely change the balance. Suppose



MIXER INPUT & OUTPUT DIAGRAM
Quad Eight Console at the Burbank Studios, Scoring Stage 1

Figure 3

whole thing going, and you get in the dubbing room and you find that the idle of the car, or the motor sound, or a siren, or some other sound effect is competely wiping out those frequencies so all you hear is the strings and brass, and the rhythm section is entirely gone. Well, then it's a car chase scene, for example, and there are guitars and a marimba playing that makes an excellent rhythmic effect, and really keeps the whole thing going, and you get in the dubbing room and you find that the idle of the car, or the motor sound, or a siren, or some other sound effect is completely wiping out those frequencies so all you hear is the strings and brass, and the rhythm section is entirely gone. Well, then with the 3 track the dubbing mixer has the ability to raise the rhythm track. Along the same line there are certain instruments that don't play well under dialog. An english horn, or oboe, for example, may sound fine on the stage, they loved the cue and it's marvelous, but you get it in the dubbing room and it's pulled down in level. All of a sudden Fletcher-Munson comes into play and all you hear under the dialog is an oboe solo. The low strings and low woodwinds and tuba, the trombones, all the bottom end has disappeared. There again, with the 3 track, the mixer has some ability to cool off the woodwind track and raise and lower the others, either by adjusting the balance of the 3 tracks, or by equalization, which is another way.

R-e/p: Is there a standard track layout for the 3 track format?

KC: No, every mixer has his own idea of the best way to lay it out. I've spent many years in dubbing rooms, dubbing pictures as well as scoring them, and I've found a combination that works best for me, and that's the way I usually lay it out when I score. Usually it's just 3 track stereo recording, with left, center and right channels. If it's a rock type score, I probably would break it down into a rhythm track, a solo or lead track, and maybe a string sweetening track. If it's symphonic music, which most motion picture scoring is, or dance band type, then I would lay it down in my conventional manner. That's subject to change, sometimes the conductor will suggest a layout. For example, with a score where you're doing a lot of Moog synthesizer. There I would change, I would have 2 tracks for the orchestra and keep the Moog on a separate track. That can become very controversial in the dubbing room, not only from a balance point of view, but in the dubbing room you have other people involved that weren't at the scoring session and all of a sudden somebody says, "I hate that sound, what is it?" and you say "Well, it's a Moog synthesizer," and they say "Oh no, not in here!" or "at least pull it down, let me hear more of the orchestra." And if the Moog is tied in with the violins, the dubbing mixer is dead, he's got no place to go. A lot of the time you're guessing, because you don't know what the sound effects are doing, and you don't know what the dialog is doing. There again the 16 track comes into play. There have been times where the track layout that's been laid down on the scoring stage becomes impossible when it gets into dubbing, because of some other element that conflicts with the music. So we'll put up the 16 track and make another dub down to 3 track with another balance.

R-e/p: So aside from using the 16 track for producing an album release, you use it also as a safety for the dubbing.

KC: It's a safety for the dubbing, but it's primarily for the album release, because for many years we've made pictures with 3 track.

R-e/p: Your new facilities here at Burbank have received quite a bit of publicity. Have the recent advances in recording technology made a great difference in motion picture sound?

KC: Yes, definitely. And not just technological advances, but the whole attitude at this studio. Bob Hagel, the studio president, is very hot on sound, he's very into sound. And not all studios are. The sound in a lot of studios is still the last thing that gets updated. The money is spent everywhere else and the scoring stage comes last. But here it's been just the opposite. They're very up on sound; with this console, the redesigned stage, and the new smaller scoring stage which has Compumix capabilities. This is probably the best recording facility you can find anywhere in the world today, to my knowledge.

continued on page 25

Interview with Engineer STAN POLINSKY concerning several of the more technical aspects of Film Recording . . .

R-e/p: A little technical background would be helpful in understanding the requirements and capabilities in film recording. What kinds of sound tracks are used for theater releases?

STAN POLINSKY: Well, conventional mono release prints have photographic sound tracks. All stereo sound tracks are magnetic. The magnetic release formats are called Cinemascope, or "C-Scope" stereo, and 70 mm 6 track stereo. "C-Scope" release prints are sent to a lab where 4 magnetic stripes are laid on, and the 3 track stereo sound track is recorded on 3 of the stripes. The fourth stripe, for the surround channel, is usually recorded after the 3 track dub. The 70 mm format is exactly the same except the release print is 70 mm. Four magnetic stripes are again laid on each print, with 2 tracks recorded on each of 2 stripes and one track each for the other two stripes for a total of 6 tracks. 5 across the front and again a surround channel on the 6th track. Each C-Scope release print is recorded individually, it's a custom operation. We don't monitor with the Academy rolloff on magnetic release prints, the theatres have a flat playback response.

RCA's photophone division has proposed a photographic (optical) release format whereas the dual bilateral sound track on the release print is split and fed a stereo, dolbyized signal to be reproduced in two channel stereo in the theatres. Because all information would still be stored within the same area, this same release print would be compatible with those theatres whose sound systems are monophonic. Use of the cinema type dolby system would maintain (and even improve) a good signal to noise ratio as well as opening up the bandwidth.

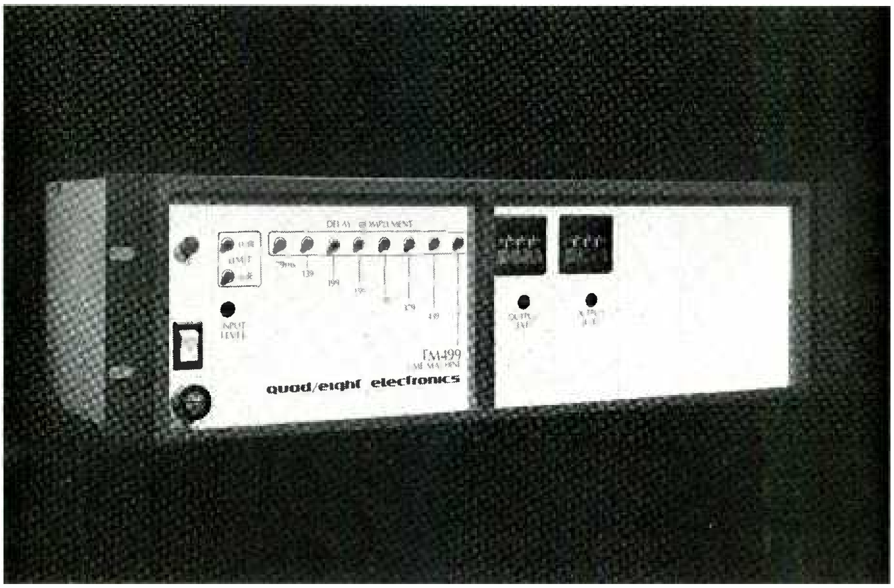
Budgetary considerations prevent a lot of 6 track dubbing now, but we have the 3 into 5 electronic spread which is a reasonable way to get into it. That's done by monitoring with 5 horns at the dubbing sessions, but recording in 3 tracks. But if it's a real 6 track picture, the original score will be recorded in 6 track, and it's truly 6 channels of stereo. That's why you get the added presence and realism. The best sound you'll ever hear in a theater is a 6 track magnetic release print. It really is fantastic. Also, the C-Scope speed is .90 FPM (18 ips) as is the mono release print – but the 70 mm 6 track speed is from 112.5 FPM (21.6 ips) to 146.25 FPM – according to which wide screen system is used.

R-e/p: Is Mame a 6 track stereo picture?

STAN POLINSKY: No, it's going to be released in mono. They canceled the stereo version, which is a shame. The mono release print has an optical sound track, where stereo releases are always magnetic sound. Optical tracks impose their own limitations on the frequency response and dynamic range. Optical slit width and film grain structure among other things limit the high end. And the Academy rolloff is always applied to optical tracks. When the Academy standards were set up many years ago the amplification systems of the time were very limited and in order to improve the signal to noise and achieve a more or less consistent sound in theaters around the country the Academy rolloff was introduced. Now there's some interest in dropping the Academy rolloff in favor of a Dolbyized system. The results are supposed to be about as good as magnetic sound tracks. In addition the Dolby theatre system has facilities for switching Dolby in or out, and a new mode called enhancement whereby non-Dolbyized release prints can be audibly enhanced.

continued on page 25

the time



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Polinsky (continued)

R-e/p: With wide recording tracks you must have a comfortable signal to noise ratio. But don't you sacrifice something else? Where are the tradeoffs? As you know all recording systems are set up and based upon a reference of playback level.

SP: We call O VU playback level the point where the recorded signal has 1% distortion. It corresponds to a certain flux level on the magnetic emulsion. We also refer to it as 100% modulation. We use 3M type 341, a high level, low noise emulsion on a mylar backed film. With this stock and the way we align our machines, the recorded noise due to bias (with all input signal removed) is from 74 to 76 dB below the 100% modulation level. We try to keep our average level below the 1% distortion level by about 8 dB, so we put 8 dB of lead in the meters in the dubbing console, so that the ballistics of the meter are compensated for in the peaks of dialog. We also use neon VI's which are peak reading devices. We calibrate those for true 100% modulation, which also corresponds to 100% galvonometer deflection at the photographic recorder. So the dubbing mixer has a dual indication of level and the neon volumume indicater also looks almost identical to the galvo action.

R-e/p: Why 8 dB of lead?

SP: That's just here in our dubbing rooms on the Columbia lot. Over in the dubbing rooms at Burbank they work with 6 dB, and I believe at Goldwyn they work with 10 dB. It varies somewhat, it's simply what practice has been set up; what mixers have become accustomed to. You can learn to work with anything, really. I've always believed that the true judgment in what you're doing lies in how it sounds. I don't care if the meter is not even moving, or even it it's pinned and not released, although you have to stay within certain boundaries. That's what meters are for, they're not to be used as the Bible, they're to be used to tell you when you're in the ballpark, after which you have to judge aesthetically. How does it sound? It it in perspective, is it distorted, is it hissy? That's something nobody can tell you except the computer between the eyes and ears. That's what tells you if you're accomplishing what you're paid to do. This is of course based upon a technical viewpoint — what's really important is whether or not you made the producer happy!!

R-e/p: So it's the extra wide track width that allows such a high signal to noise ratio?

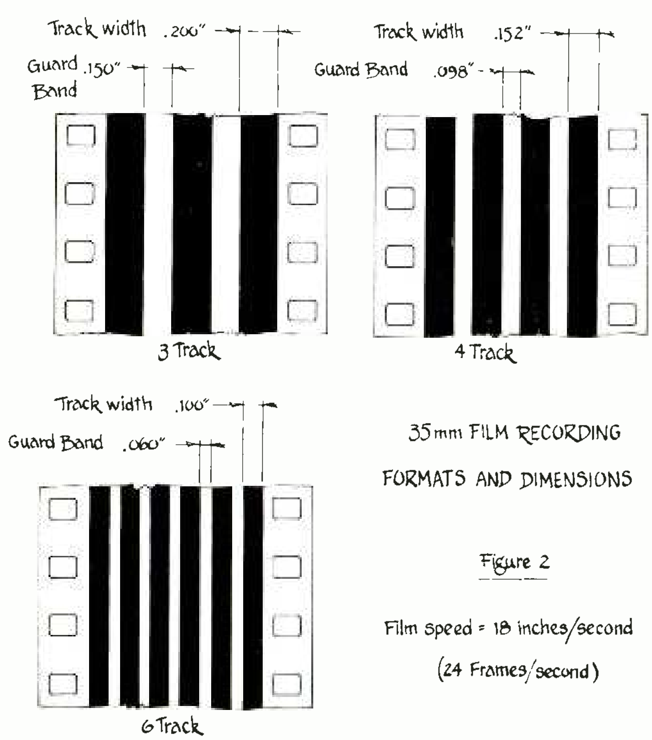
SP: Yes. Well, 16 track widths are 70 mils wide, with 20 mil guard bands, where our recording tracks are 200 mils wide on 3 track 35. As you know as you double the track area you get an increase in SNR of about 3 dB.

R-e/p: How about other differences in performance, compared to tape formats?

SP: Print through. Due to the thick film backing there is very little print through at all.

R-e/p: Is the emulsion used in magnetic film recording different than in tape?

SP: Our stock emulsion is called high level low noise emulsion, and the research that went into it is the same as that for tape. The emulsion is thicker on magnetic film, and the output increases proportionately. That's another reason for the high SNR.



35mm FILM RECORDING FORMATS AND DIMENSIONS

Figure 2

Film speed = 18 inches/second (24 Frames/second)

Cleary (continued)

R-e/p: What are the unique features in this console you're using, what does it enable you to do that perhaps you couldn't do before?

KC: Well, it has complete equalization facilities, on every position.

R-e/p: And more generally how does it differ from, say, a complete record recording console?

KC: The only way it differs from a record recording console is in the flexibility in being able to record in several mediums at the same time. Most record recording consoles are usually a straight 8 track console. Or maybe these days 16 or 24 track. But this console is unique in that you can record on at least four different mediums at once. You can record 35mm 3, 4, or 6 track stereo, 16 track 2 inch tape, 2 track ¼ inch, and mono at the same time. Or quad. On this console you can get 4 different mixes out at the same time, four completely different balances and outputs at once. That's unique to my knowledge.

R-e/p: Why so many sets of outputs?

KC: Well, it meets the needs of film scoring, which is the business we're in primarily here, making movies. And it meets the needs of making record albums too. I think the concept of this whole facility is unique in that respect. There aren't too many studios that meet the needs of both.

R-e/p: So you never really had the availability of equalization on so many separate channels as you now have?

KC: No, because the 35mm and film scoring consoles in most Hollywood studios were just simple basic recording consoles. They had a number of microphone inputs, not much in the way of equalization, and track selection, and that's about it. There wasn't too much you could do. The equalization was mainly left to the dubbing process, and that was limited too.

R-e/p: You mentioned automation equipment. Can you tie that in with the 35mm 3 track system?

KC: I don't know. I've read articles about a system of that type for dubbing, but I've never seen one in this country. Our console on the other stage is wired and has all the facilities in it for Compumix. All the buttons are in, all the switching, everything in the console. They haven't purchased the computer yet. But of course most of the work is in the console itself. The computer, like a 16 track machine, you just plug in. It's a very useful tool, Compumix. When you've done a mixdown on 16 track and everything's perfect except one spot on one track, to be able to go back and update just that particular section is marvelous!

Is your present monitor system too good to be true?

If your present monitor system is too good, that's too bad. Chances are, with your system, you're mixing to a sound that doesn't exist under normal broadcast or listening conditions. The result is a mix that sounds flat. And dull. The sound isn't true.

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R-e/p: How about problems of a general nature. Did Mame present any performance editing problems, for example?

KC: Well, a musical is the most difficult recording challenge there is for a scoring mixer, I guess because of the various types of music you find within a musical picture. You've got the underscoring, the big production numbers, the small stage numbers, the main title which is always a big piece of music and normally occupies an entire session. Plus all the incidental effects that have to be achieved to match the picture. You've got to look at every scene and decide what environment the orchestra is in pictorially and try to duplicate that on the stage. You have the vocals, the chorus overdubs, we had about a 50 piece chorus on Mame, and the incidental vocal parts, particularly in plot integrated numbers where characters are singing various lines. All those people have to be recorded separately and the whole thing overdubbed and put together . . . taps, footsteps, handclaps, whatever you see on the screen has to be reproduced.

R-e/p: But you do most of that on 16 track, don't you? You can't do much overdubbing on 3 track machines.

KC: For many years we did, before 16 track came along. What we would do is make a basic 3 track orchestra recording, and play that back as a guide while recording perhaps a 3 track stereo choir on another machine. Then we play those two back on dummies, which is what we call the reproducers, play them back and record a solo vocal on another machine. Then maybe make a quick dubdown of them onto a mono 35 and play that back to the dancers. Actually everything that is done these days with 16 track we've been able to do for years with synchronized 35mm machines. You see, everything is sprocket driven in sync. You talk about 16 or 24 track sessions. I've worked on motion pictures where we've had 48 tracks, all first generation, all running in sync. In the dubbing process of a picture, that's normal procedure. You may have 15 to 20 sound effects units, you've got 3 or 4 dialog tracks, you've got 5 or 6 music tracks, and you're getting up to 35, 40 tracks, but they're all running on separate reproducers locked together on one distributor.

R-e/p: So where the record industry is just beginning to use synced machines, the movie industry has had it for some time?

KC: We've had it for years. When you think of the complexity of some of the musicals that have been done on 35, some of them would have been difficult to do on 16 track. And they couldn't have been done without losing generations; you eventually have to start ping-ponging.

R-e/p: Why then, do you record on 16 track?

KC: Mainly because it's in demand by composers, because it's become popular in the record field, and these musical directors work in the record field as well. Don't get me wrong, I'm not knocking 16 or 24 track, it has capabilities that we don't have in 35. It's faster, it's definitely faster. You get to the end of a 5 minute take on 35 and you have to wait for the recordist to take it off the machine, rewind it, and thread it up again for playback. With 16 you zing back and you're there. Right away. And 35 doesn't have the punch-in capabilities. Overdubbing and sweetening are so easily accomplished on 16 track. It's a very flexible tool. The other system was slower, and more complex. With 16 track you've got the complete number on one piece of tape, but you get into the complex musical production number with 35 and you've got film all over the room. And you've got to have a film editor, a music editor, it has to be assembled, leadered out physically on a Movieola so that it comes in at the right place, all the countdowns have to be cut off, any extraneous stage noises . . . it's a time consuming process. 16 is so much faster, although no better quality wise. So 16 track is a definite advantage. All the numbers in Mame were done on 16 track and in many cases we came in later, Fred Werner and I, to rebalance some of the dubdowns from the 16 track. It also gives the composer more flexibility. With 3 track you get as good a balance as you can, and everybody's watching the clock and they say "Yeah, that's . . . great." The orchestra's played it five times, maybe, if it's a difficult piece of music. And so there's a compromise there. It's a good take, but you're locked down to the 3 tracks. If you suddenly decide you want to dig out a second guitar part or something and it's locked together with the horns, you're limited. But with 16 track, of course, when everybody goes home you can really finesse it, and you end up with a better product.

R-e/p: Was there anything specific pertaining to Mame that required special attention?

KC: Nothing really specific to this production other than matching the orchestral sound to the visual. For example if (on the screen) you're looking at a long shot of a pit orchestra in a theater then you don't want to put your mikes in too tight because it's not going to match. In Mame there were a lot of these shots that I miked a little looser and with more

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Polinsky (continued)

R-e/p: The Film Lock system that synchronizes the 16 track machine with the 35 mm machines, does that work on the SMPTE time code?

SP: No. A 2400Hz square wave is prerecorded on a spare channel of the 16 track tape (before the session). That channel is fed into a comparator which compares it to the output of a tach which is running with the film distributor system. This results in an interfaced system with dead sync at all times between film and tape. The system (called Film Lock), available from Ampex Corporation, is so reliable that it actually starts film and tape from a dead stop and still maintains sync even in reverse, and up to 4 times normal speed.

R-e/p: From your viewpoint, what would you say were some of the major differences between recording for records and recording for film?

SP: Well, in a recording studio where music for records is recorded, you can really control the setup and environment. Especially in the last few years the quality of record releases, technically speaking, has really improved. Right along with improvements in the state of the art. New and better microphones, console designs, recording tapes, and so on, which were largely developed through demand by the record industry, anyway. The finished product is the record, with all the clarity and guts we've grown to expect. Now in film sound, the finished product is a film, and a good sound track is one that plays well with the picture. And that includes all the elements of the sound track; the dialog and effects as well as the music. Now in getting dialog and effects we're not always blessed with control of the setup and environment, and in spite of recent advances in technology (if we're lucky enough to have them available) we're still in a hostile acoustical recording environment. Record people are sometimes disappointed when they hear their studio produced score played under dialog and effects. It can lose punch. But a successful track achieves a total balance with the picture. It shouldn't break the continuity of the total motion picture experience. Really, the best compliment a film soundman can have is, "I didn't notice the sound in the picture!"

In retrospect, the last question might be revised to ask what are the major simularities rather than the major differences between recording for records and film. Also, looking ahead further, not only are the cinema and record recording techniques becoming more similar, but soon we are going to see if not a three way marriage, then at least a "living together" between video production, film production and record production. We of the recording industry are really in for some exciting and (pro) creative challenges ahead.

overall sound.

R-e/p: So you're translating the camera's view to an audio perspective?

KC: Yes, that's the basic thing to keep in mind in motion picture scoring. It must match the picture. Underscoring and the main title, of course, are different. They're normal orchestral sound, but if the music is visual the most important thing to consider is what you're seeing on the screen. That's why we score to a work print on the screen in the studio. In a 6 track stereo picture you have to physically assign the instruments to the tracks to match what you see on the screen, and that can change with picture. For example, as I remember in "Paint Your Wagon" there was a guy playing accordion on extreme left, a violin towards center and an old barroom piano over here and a guy with a banjo over on the right hand side. You have to look at the picture and physically assign those instruments to those tracks. In 6 track stereo pictures you've got 5 horns (speakers) behind the screen and the surround channel, so whatever you place on track one will come up on the left side of the screen. Once in a while the camera angle will change and now the accordion is someplace else. All you can do in a case like that is compromise or later in dubbing go into a process we call "swinging," which is the same as panning. You've got to be careful here as you're watching the scene that you don't tie it together with something that doesn't move. So you really have to get a road map of that particular scene and where everybody goes and what they're doing. If the accordion walks from screen right to screen left you make sure you have him on a track by himself so later on you can pre-dub it with the picture, panning across the speakers to match the action.

R-e/p: What about quick cuts from long shots to closeups?

KC: There are standard procedures which have been accepted by the motion picture audience over the years, and making big sound perspective changes for short cuts is not the thing to do. It's more acceptable to the ear to hear the sound remain in the same perspective than to hear big changes. Unless, of course, if it's a really major change, then it has to be made. And sometimes they can be short cuts. For example, if you're cutting from a scene of an orchestra playing on stage to a scene of dialog in the alley outside the theater, then you've got on and off stage perspectives, and sometimes the cuts can be very fast back and forth. They have to be made. You won't accept the orchestra playing just as loud in the alley as you heard it inside.

R-e/p: When you record it, would you assign a track or two to distant microphones and cut those to perspective changes?

KC: No, it's strictly done with level in the dubbing process. The music would be recorded here on the scoring stage, ignoring those on and off stage perspectives. What we would do is cue the picture. We write cues right on the workprint, plusses and minuses, 6 or 8 frames before the cut, with a squiggly line in front of the plus or minus. When you're dubbing you see the squiggly line and then the plus, and with most people's reaction times, they'll boost it right on the cut.

R-e/p: Sprocket drive makes it easier, doesn't it?

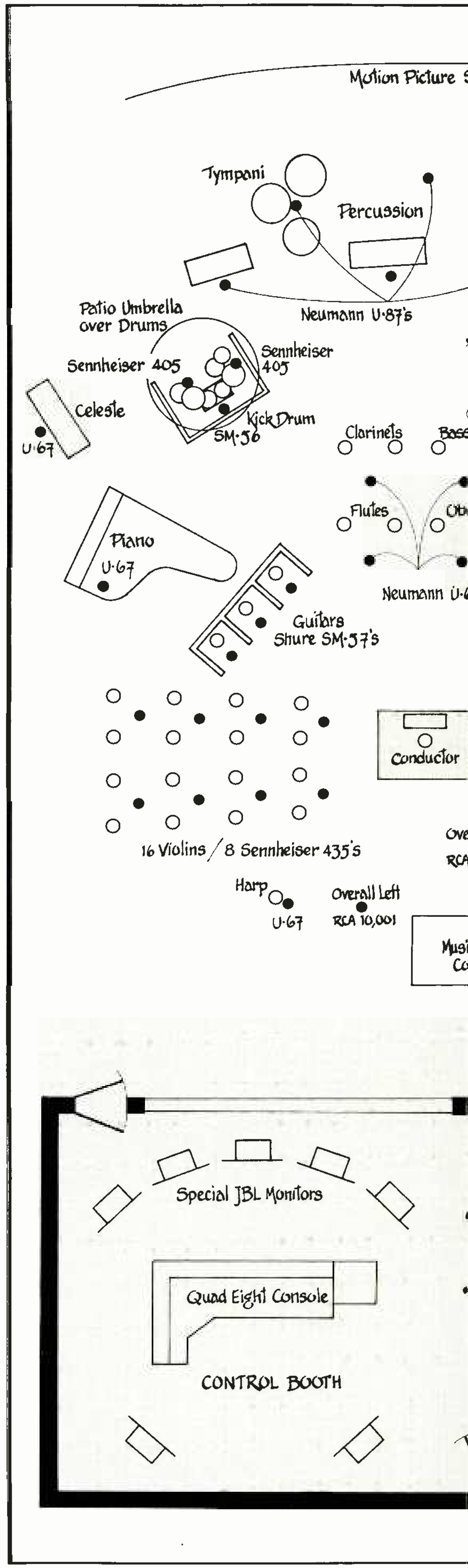
KC: Yes, everything's always in exactly the same place at the same time. Well, of course, the 16 track with the Film Lock system that we have is virtually sprocket driven, because the 16 track runs off the distributer along with the sprocket driven machines. It's absolutely synchronized. We often play 16 track back with the picture.

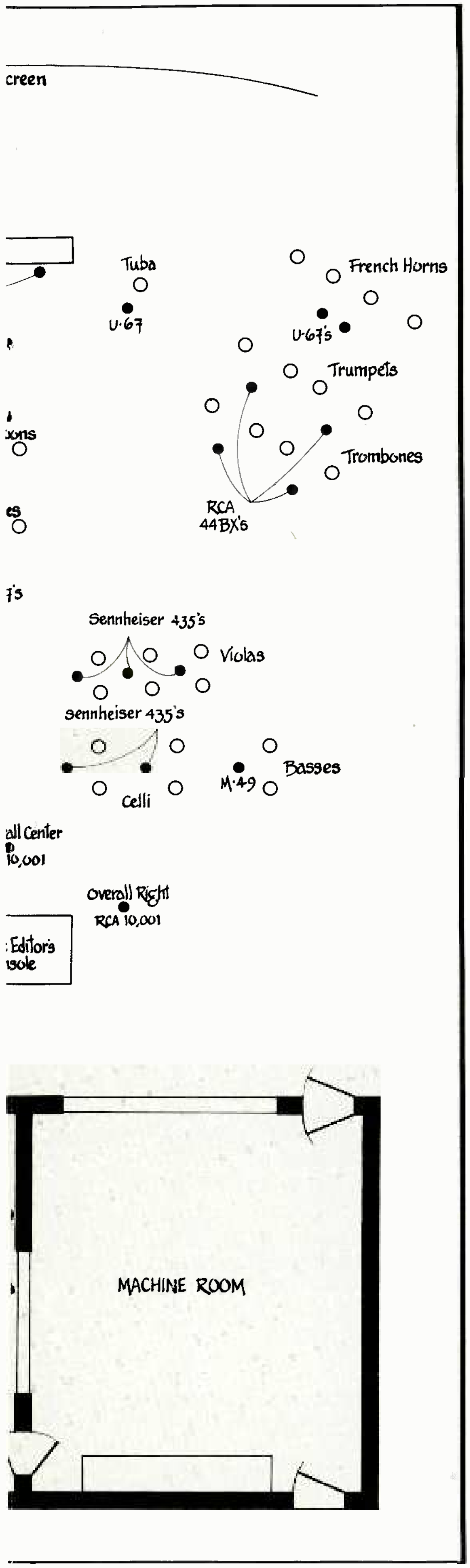
R-e/p: Where does the scoring mixer leave off and the dubbing mixer take over?

KC: In some cases the composer will insist that the music be recorded or at least dubbed down to monaural so that the balances can't be changed later. I personally don't think that's a good idea. I don't want to belabor the point, but we mentioned before that when you get into dubbing strange things can happen because you're adding other elements that you don't hear on the scoring stage. Sometimes we play it back with the dialog, which is a good idea, but the sound effects are usually not available. And the sound effects can destroy the music completely, particular frequencies can cause certain instruments to disappear.

R-e/p: Are there any particular guidelines to that?

KC: Not really, because there's a wide range of effects available. Sometimes it'll be a motor or piece of machinery, an air conditioner or a printing press that'll interfere with the music. Or conversely, the music can interfere with the effect. It's a matter of finding a balance again. But that's the dubbing process, a whole other process. There you're concerned with balancing the music, sound effects, and the dialog for the finished product. Here in the scoring end we're concerned with getting as good a music recording as we can. In 99.9% of the cases that means a straight orchestral balance, a good musical quality. Here's where we find differences between the balance you want to get for the picture and the balance you want for the record. Particularly in underscoring cues, I put on a very full low end and subdue any solo instruments that





may stick out. I watch the screen as we're making rehearsals to see if there's anything under dialog. We're not playing the dialog track but I can see if they're talking, and if they are I'm very careful to keep the low end up, keeping a full round bottom end and mid range: anything that may stick out, particularly woodwinds or percussion, I subdue so that it's down a little below where you would want to hear it musically, because Fletcher-Munson's going to get you there when that comes down under the dialog. The bottom end's going to disappear. Here, again, we have the best of two worlds in being able to record 16 track and 35mm at the same time with different mixes. Primarily here we're trying to get the music recorded and into the music editor's hands as soon as possible. So you don't have time for dubbing down. But you also have a 16 track recording which you can balance differently for an album.

R-e/p: In mixing the score for the album, from 16 track, were there any special problems that related to the album?

KC: Well, not specifically because it was done on 16 track, but one thing that was out of the ordinary was keeping a consistent quality of the orchestra sound. We recorded over a long period of time and very often smaller orchestras were integrated with bigger orchestras. It was quite a job of matching the tracks, getting them to sound like it was all recorded at the same time. This took place over the period of a year, I guess, from the original pre-recording sessions to the final sessions. We did sweeteners here on the final sessions which were inserted in sections that had been recorded a year ago.

R-e/p: So you made extensive use of overdubbing techniques?

KC: Overdubbing, equalizing, especially equalizing to match the quality. The same was true of the vocals because they also were recorded over a long period of time and at different locations, different rooms here at the studio. Some were recorded at the ADR room, the looping stage, some were recorded here on the scoring stage, some were recorded after the orchestra had gone home, with the vocalist outside the booth. There was a variance in quality even though the same mike was used.

R-e/p: What did you do to achieve similar sounds in these different conditions?

KC: Equalizing, and pre-dubbing. Some of the vocals I had to take into the dubbing room so that I could use the back and forth machines. We have the latest dubbing facilities here, with back and forth dubbing. You can put up X number of units, maybe 40 units if you want, or more, along with the picture if necessary, and you can run them from the console by pressing forward and reverse buttons. All those tracks, back and forth, just like on an Ampex MM1000. And on the 35mm you can also punch in on any track just like on 16 track. That is, you can in the dubbing room. We did a lot of work on the vocals, working with words or phrases which were intercut from different takes. The easiest way to do it was to take them up in the dubbing room where we could back up and punch in a word at a time. In some cases I'd put them on different tracks. I'd get one section on track one up to a certain word that was OK, and then so there was no danger of punching in and ruining any of that I'd put the next section on the next track, punching in and equalizing a piece at a time, using as many tracks as needed. Finally when the whole number's done you can play back all the tracks and you don't even know you're switching from one to another.

R-e/p: When you're mixing a score for 6 track stereo, do you monitor on 6 loudspeakers while recording?

KC: Yes.

R-e/p: Well, what about those times when you have to match position of an instrument visible on the screen, and it moves from one side to the other?

KC: The panning would be done later in the dubbing room. Matching the action on the screen is accomplished by track placement. Here on the scoring stage we try to get the best music recording we can get. For instance, we may record a beautiful piece of music that will go in the album, and it will be a 3 minute bossa nova type number with strings, brass and everything, really a nice piece of music, but in the picture it may be *futz* coming out of a radio. Here we get the best music recording. What they do with it in dubbing to match the picture is done strictly for the picture. We still have the original 16 track or 3 track, or whatever it was recorded on, untouched.

R-e/p: Is there ever a situation where you're filming a scene of musicians playing, and you use that performance for the sound track?

KC: Very rarely. The only time I could think of would be some kind of source music where there's a guy in a bar plunking on a guitar, or a guy sitting on a porch playing a harmonica or something. The sound man will try to get it, the production mixers always do try to get that kind of recording. They have to get it anyway, for sync. They have to make a guide track. Most of the time it's redone on the scoring stage. Also you've got intercutting problems on the stage, where you're filming from several angles. How are you going to have the guy on the porch play it the same way for all the camera angles so they can be cut together. It's almost never done, but there is the exception. Maybe the guy on the porch just plays a few bars and there are no intercuts. If the production mixer gets it, that's fine.

R-e/p: You also have to contend with special equalization for film sound. How does that affect you as a scoring mixer?
KC: Well, we're not so concerned with that here, I'm more concerned with the orchestral balance, knowing that it's going eventually to optical, and being familiar with how the music will sound in the dubbing room with the Academy roll-off on it. I compensate for it in the scoring booth in the balance rather than in equalization. Once in a while in the equalization I'll peak something up in the midrange because I know it'll sound dull in the dubbing room. Maybe the high end too. But basically it's a matter of balance. This console has a monitor mono test position, so even though you're recording in stereo you can punch up and listen to it monaurally. In the mono mode you can also punch in the Academy rolloff so you can hear exactly how it's going to sound. That's a great facility I haven't seen anyplace else. Particularly on TV scores where you're going really fast and you know there's not going to be any time for any after thoughts, once it's made that's it, and it's probably going to be dubbed in an hour. I often listen to it in the mono mode with the rolloff in and sometimes mix several cues that way. It gets tiring on your ears, though, I don't do it too long; it's kind of depressing.

R-e/p: So you're just headed for the best possible sound that you can get. As a rule, are there any particular instruments that you trim with equalization?
KC: Well, I have a set pattern of equalization which, again, is tied to the setup, the microphones, the stage, and to my experience with the instruments on this stage. Guitars, for example, always need brightening with a big orchestra. Invariably they're written in a register where they're playing with the woodwinds or the low strings, and they need quite a bit of brightening in the midrange and highs. Sometimes I rolloff the low end. Rolling off the low end on various sections of the orchestra is very important when you don't use screens (baffles). I don't use baffles as a general rule, so rolling off the low end in sections where there's no low end involved can cut down a lot of the extraneous "bouncing around the room" sound, the rumbly stuff. Particularly in the violins, woodwinds, the higher percussion, the tree-bells and vibra-slaps — those instruments I automatically roll-off the low end. Then I get just those frequencies that instrument is going to give me. Anything I don't want, extraneous stuff, I want to roll-off.

R-e/p: Do you ever roll-off the high end on instruments that are predominately low-end, like bass drum?
KC: No, the only thing I roll high-end off is Fender base. Every Fender bass has a problem of its own. Everybody has a different amplifier and every one of them seems to have hum and buzz to some degree. With some extreme cases I really roll off the high end, and maybe even dip it to take out the 60 and 120, and sometimes the 240 or whatever's in there that's a problem.

R-e/p: I notice that in your microphone setup for the main title session for Mame you used an M-49 on the bass, and nowhere else. Is that your pet microphone for bass?
KC: For string bass, in that setup, yes. These are all pet microphones of mine. All mixers have mikes that work in their type of setup for any particular instrument. That doesn't necessarily say that he's endorsing it as being the only microphone for that instrument, but for his type of setup he's found it works best for him. And if you were to interchange microphones here, for example, and put the RCA ribbons on the strings and the Sennheisers on the brass, this setup would be a disaster. There'd be no possible way to mix it. It'd be an absolute disaster.

R-e/p: Well, what is it about the M-49 that made you choose it for bass?
KC: It's got good side rejection, and it gives a nice clean, round bass sound. It's an earlier Neumann condenser model. The RCA's for example, on the trumpets and trombones, are the only mikes that have ever worked for me on trumpets and trombones. They're bi-directional and they work with the setup. If you were to move the setup around and put the brass in another location, and you had a bi-directional mike on it it would also be a disaster. So the choice of microphone and the setup go together.

R-e/p: Well, in this case, the 44's on the trumpets will be picking up both trumpets and trombones, and the 44's on the trombones will be picking up from this space in front.
KC: Yes, but there's nothing in that space in front, and that's significant. You've got a trombone playing here 2-1/2 feet from the microphone, and 12 feet away you've got some violas and basses that are pointed in the other direction. Your main signal is the trombones, the leakage from the strings in this direction is insignificant. The same is more or less true here on the trumpets, too; the trombones are blowing away from the trumpet mikes.

R-e/p: What about the horns. They're in the brass section too. Why not 44's for them too?
KC: Well, the French horns are the most difficult instruments in the orchestra to record. Over the years I've tried many different combinations of microphones. I've had baffles in back with 44BX's between the horns and the baffles so you get the direct sound plus some reflected sound. I've tried miking them from the side, the back, and the front, and I've found the U-67 works best for me and my setup, hanging pretty much over the center of a close group of 4 horns, angled out toward the outside horns. With 4 horns sitting like that there's a tremendous buildup of sound in the center. If you were to mike in the center all you would get would be the two inner parts. So these two mikes are close together and pointing out toward the first and fourth horns . . . which balances the four parts well. The outward angle is critical. Sometimes I have to go out and move it an inch or two. In rehearsal I ask the conductor to have the horns play a four part chord and hold it, the first horn plays the bottom note and holds it, then the second horn, and so on to get a four part chord, and I listen to see that all four parts are equal. It depends on the players, and whether it's a hot day, or humid day; the horns are so critical, it's unreal. The same guys may have been there before lunch, and it sounded beautiful, they come back after lunch and all I'm getting is the second and third parts. I have to go out and move something a little bit one way or the other, until the chord sounds evenly balanced. The next most difficult instrument is the tympani. On this setup I used U-87's. I like them down fairly tight over the tympani, maybe 2 or 3 feet above. Just so the player has enough room for his mallets. If there are four tympani I'll put 2 or sometimes 3 mikes on them.

R-e/p: In this setup you are pretty liberal with the mikes in the percussion section.
KC: Yes, I usually put out at least 4 mikes in percussion, although here we have 5. In percussion you have no indication of what they're going to play, the only indication you have is the number of players. On your instrumentation you know what the trumpets, trombones, horns and violins will be doing, but in percussion it'll say, 2 men, 3 men, or even 4 men if it's a big score. If it's one man you know he can't play too much so 3 mikes will do, maybe; but I've had occasions where there's been one man and

continued on page 35

continued from page 30

I've needed 6 mikes. He was surrounded with stuff, a bell here, a gong there, a tree bell there, vibra-slaps somewhere else and a tympani behind him . . . he was running around like a squirrel in a cage.

R-e/p: With a lot of drums in the percussion section there must be some leakage problems, particularly with the low frequencies.

KC: Well, the only real low frequencies are in the lower conga drums, the bass drum and the tympani, and bass marimba. By miking the tympani close, I get the sharp sound of the mallet hitting the head. Any spillage into other mikes is desireable in this case, it rounds it out. It makes it sound like it's in a studio and not in a closet. A lot of people are afraid to mike a tympani close; well, if you've got just one microphone and a set of tympani in a room and you mike it close, it's going to sound dead and flat. But if you've got a lot of other microphones around, it rounds out.

R-e/p: What are the characteristics of these RCA 10,001's you use for overall mikes?

KC: They've been around for many a long year. They used to be a standard boom mike in the studios. They're a ribbon mike, you could compare them to the 44BX's really. They have a good smooth response. You can get an entire string section on one of those. In fact you can use it for main pickup. At times I almost close down the tight mikes and use mostly the 10,001's.

R-e/p: Talking about RCA ribbon mikes, I don't see any 77's in your setup.

KC: 77's are unidirectional, and I find a bidirectional mike on the brass to be an advantage. It gives a more rounded sound. Some of these things you just go by ear on. You can get somebody to come in and make various tests and say, "Well, it's bouncing off the floor and it's coming back in here," well, great, you move something two inches and it doesn't work anymore. It's just a matter of what you hear and what you want to hear.

R-e/p: Your JBL monitors here in the booth are obviously not like the speakers to be found in the theaters. Does that present a problem in your mixing for theater release?

KC: First of all, the only resemblance between these JBL's and ones that you could go out and buy is the box. They've been custom made with a different combination of standard JBL components. There are also supertweeters in the monitors which are switchable. You can have them flat all the way out, you can punch out the supertweeter, which is the way I mix in pictures. It gives you a good overall idea what you're putting on film without it sounding too dull.

R-e/p: When you're mixing a score do you monitor at high level?

KC: I have never been able to mix music at a low monitor level and I don't know any mixer that has. You're listening for inner voices in the orchestra. You're listening for parts that if you're listening at a low monitor level you just can't analyze. So you have to mix at a higher monitor level. I don't know of any other way to do it. Everything can sound marvelous at a high monitor level and if you're making a record you have no problem. Most people are going to play it back at a relatively high level, too. But a lot of the music you listen to, in scoring, you're monitoring at high level, and you know it's going to be played at low level under dialog. So you have to be conscious all the time of what's going to happen to it. The first thing that's going to happen is the low end and the high end are going to roll off. Fletcher-Munson again. It's different with a main title, or a chase cue, or a love scene cue that's out in the open with no dialog, or a big western cue . . . and we're panning the prairies . . . and we know it's going to be played close to the level you're monitoring it at.

Another thing, you've got to remember that you can listen or mix just so many hours and your hearing becomes completely unreliable. In scoring and in dubbing I've had experiences where everybody's been tired, they loved the take at the time, but you play it back the next morning and you can't believe that you made it. Ya gotta be kidding. Usually it's too loud. The longer you sit here, the less you seem to hear, you want to hear it louder and louder. I've dubbed reels they played back nine o'clock the morning after and you can't stay in the room with it, and yet the night before it sounded beautiful.

And here's another thing. There are a lot of mixers starting in this business that never go out and listen to the orchestra out on the stage. They'll set up their mikes, the conductor says, "Let's go," and they'll walk in the booth and start mixing. I always make it a point to listen to at least one rehearsal on the stage. Then my ears get attuned to the orchestral sound rather than the speaker sound, which is always going to be a little different. Plus, I want to hear how he's orchestrated it, how it sounds out there, if it's balanced out there on the floor. So a lot of mixers set up their mikes and they go in the booth, sit down and move everything on the board, and it's a disaster. They know the setup's all right, it's worked for them before, there's no reason why it won't sound right. The only thing they haven't thought of is that it probably doesn't sound right out there, on the floor. Then there's nothing you can do in here. An example is overblowing. I'm not knocking Hollywood musicians, but they probably play more every day than any other musicians in the world. The symphonic musician has a couple of rehearsals during the week, and a concert or two on the weekend and that's it. But a lot of Hollywood musicians are playing literally 15, 16 hours a day, almost around the clock. They come in, sit down and start playing . . . the session's over, they whip out, and sit down and start playing somewhere else. If the conductor isn't cognizant of the fact that they're not paying attention to the dynamics that are written into the score, they will be playing forte all the time. And they also have a tendency to get used to playing loud, loud, loud. I've been out on the stage where the brass section would deafen you, but the melody's in the strings.

R-e/p: What can you do then?

KC: Nothing, only go out to the composer and quietly take him aside, or call him in the booth, and make him aware of the fact. I'm not saying the composer doesn't know what he's doing, but he's got his problems too. He's written the score, now the producer's sitting behind him, he's never heard it, he's a little on edge . . . is the guy going to like it, is he going to hate it, will it fit the picture, he's concerned with timing, he's watching streamers (legends on the workprint) go by, he's conducting and varying his tempo to hit cues, sometimes the last thing he hears is the orchestral balance. And I've never had a case yet where the guy didn't say, "you're right, they're overblowing." And he'll go out and have a talk with them, "Boys, if you don't have the melody, play down, if you've got the melody, play up . . . in bar so and so there's a diminuendo in the strings, make sure you do it." All of a sudden you've got a whole new thing. There it is. But that's something I've noticed a lot of mixers have ignored. They've never gone out on the stage and listened and said, "My God, how could I record that? No way!" Finally, there's that old syndrome of what I like to call "Mixer's Twitch!" Some mixers can't resist constantly playing with a mix, during a take. They just won't leave things alone. When an orchestra's balanced, and you're happy with the balance, you may know that 6 bars in you have to raise something or lower it a little, a slight change in things, but apart from that, leave it alone. I've seen guys sit there, and they can't resist fiddling. The violins are going up and down, they're moving this and that. They must feel if they're not doing something it's not going to go on the film. And they're absolutely destroying what they had. And then they have no way to repeat it. If you're satisfied with the sound you have, sit back, fold your arms, and let it play.

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An old technique with a new twist: DIRECT - TO - DISC RECORDING

by ANDREW P. TETON

The development of tape recording twenty-five years ago eliminated the need to record directly to master laquer discs. While many people would think a return to direct disc recording is a step backward, Douglas Sax and Lincoln Mayorga think it is the key to the ultimate in high fidelity. Together Sax and Mayorga founded Sheffield records, a label dedicated to the release of direct disc recordings. Judging by the recently produced "Lincoln Mayorga and Distinguished Colleagues – Volume III," Sheffield appears to have achieved its goal of producing records with genuine musical merit and at the highest possible fidelity.

WHAT INSPIRED THE DIRECT-DISC REVIVAL

Doug Sax and Linc Mayorga have shared a personal and professional interest in music and high fidelity recording since the late 1950's. At that time, Sax was a symphony trumpet player, and Mayorga was a pianist and arranger for "The Four Preps," a popular soft-rock vocal quartet. One phenomenon that continuously intrigued them was the sonic impact of records that were produced before 1950. This impact could alternately be described as a combination of presence, clarity, and dynamics. It was most obvious on solo piano recordings, where limited frequency response was not as detrimental. Recordings after 1950 did not seem to them to have the same impact, and Sax and Mayorga believed that tape recording might be responsible for the apparent loss.

In 1959, Mayorga and Sax made a test record, cutting a lacquer directly, bypassing the master tape stage, during a piano session. The recording was made at Electro-Vox studios (Hollywood), and was primarily done to test their theory about tape recording and sound quality. As they had suspected it would, when they played the lacquer, it had a phenomenally live, authentic sound that far eclipsed any available commercial recording. They had, they believed, isolated the *magic* element.

PREPARATIONS FOR A COMMERCIAL DIRECT-DISC RELEASE

In the years following that first Electro-Vox experiment, Sax and Mayorga made several attempts to record a solo classical piano directly to disc. Problems with studios, lathes, transmission line, cutter heads, and a variety of equipment breakdowns repeatedly frustrated their efforts. By the mid- 1960's, they realized that they required a state of the art disc-mastering facility if they were to succeed with the direct-to-disc project. Initially they undertook a feasibility study to determine if they should open their own mastering facility, which could also be used for the direct-to-disc work. The decision was made to proceed with the project and Doug's brother, Sherwood Sax, became a co-partner. Sherwood also designed the cutting room. Their Hollywood facility opened its doors in early 1968, was and is known as The Mastering Lab.

MAKING THE FIRST ALBUM

In late 1968, just one month after installing the first operational Neumann SX-68 cutter head in this country, Sax and Mayorga recorded, directly-to-disc, "Lincoln Mayorga and Distinguished Colleagues – Vol. I." The album featured instrumental versions of pop tunes, with Mayorga playing piano and doing the arrangements. The album also utilized many of the West Coast's finest studio musicians.

It became immediately apparent that direct-to-disc recording was going to be tremendously taxing to the talents of the musicians to do *perfect*, straight through 17 minute takes. This kind of rigorous perfection was obviously necessary because the music program was fed directly to the cutting lathe, and because the lathe must obviously cut an entire side without stopping. The musicians felt additional pressure because they were aware that their best performances might not wind up on takes that would be usable for pressing. At the same time, engineer Larry Brown felt similarly pressured, having to do all mixing and equalization in real time. There just aren't any second chances with a direct-disc recording.

In producing "Colleagues," Sax and Mayorga encountered a number of other unanticipated difficulties. In the middle of one cut, for example, a microphone began detecting the signal of a local radio station. When the problem was located and corrected, the side had to be started from the top. Then, too, several times they found flaws in the lacquer after cutting a side; flaws which ruined otherwise good takes. These kinds of problems, while minor during multi-track tape-recorded sessions, are both frustrating and expensive in a direct-disc situation. In doing direct-to-disc recording there is not the security blanket of a multi-track master tape. The content of the final product is *locked* as soon as the take is completed.

Despite the difficulties, "Volume I" turned out well, containing many of the qualities that Sax and Mayorga had hoped it would. Sheffield sold the album by mail and through high fidelity stores. The response to the album was so favorable that the entire limited edition was sold without any advertising. Metal mothers of the album were also sold to Trio Electronics of Japan, who pressed their own discs. The Japanese albums sold extremely well under the apocryphal but enticing title, "An Evening in Manhattan."

MAKING THE SECOND ALBUM

The success of their first direct-to-disc album enabled Sheffield to undertake "The Missing Linc Mayorga and Distinguished Colleagues – Vol. II," beginning in late 1971. While the format was similar to "Volume I," this second album was more ambitious, including a string quartet on side two. With the experience gained from the first album, Sax and Mayorga were better able to anticipate and deal with production difficulties. Consequently, they were able to cut more lacquers, allowing a greater quantity of "Volume II" to be pressed.

Aside from sales, "Volume II" was suc-

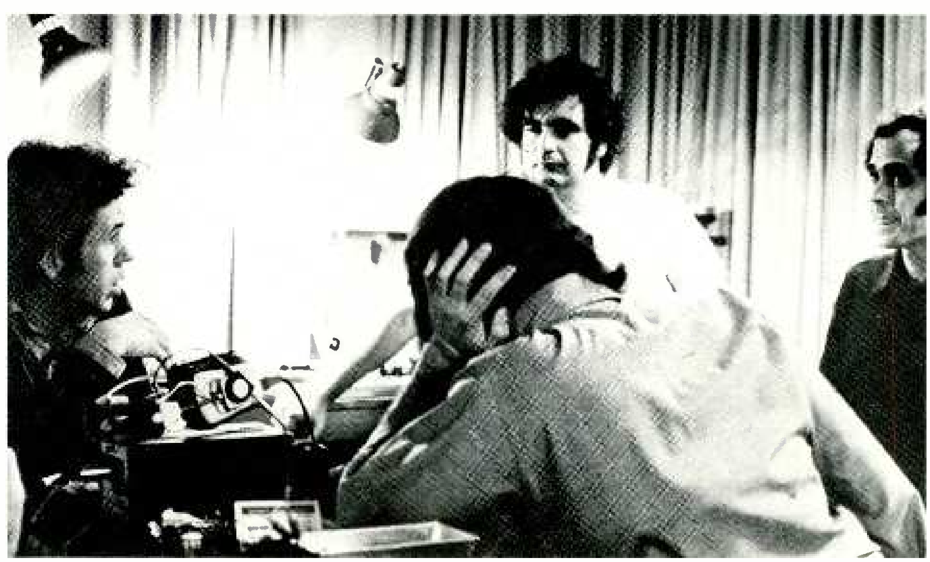


Photo by Nurit Wilde

Linc Mayorga *John Schnee* *Doug Sax*

cessful in another way. It forced the recording industry to reconsider its skepticism about direct-to-disc work, and to realize that the Sheffield venture was both serious and valid. The album has sound quality that has made it a classic, as well as a popular demonstration record for quality audio dealers. Moreover, "Volume II" is frequently used by audio magazines and laboratories for hardware evaluation.

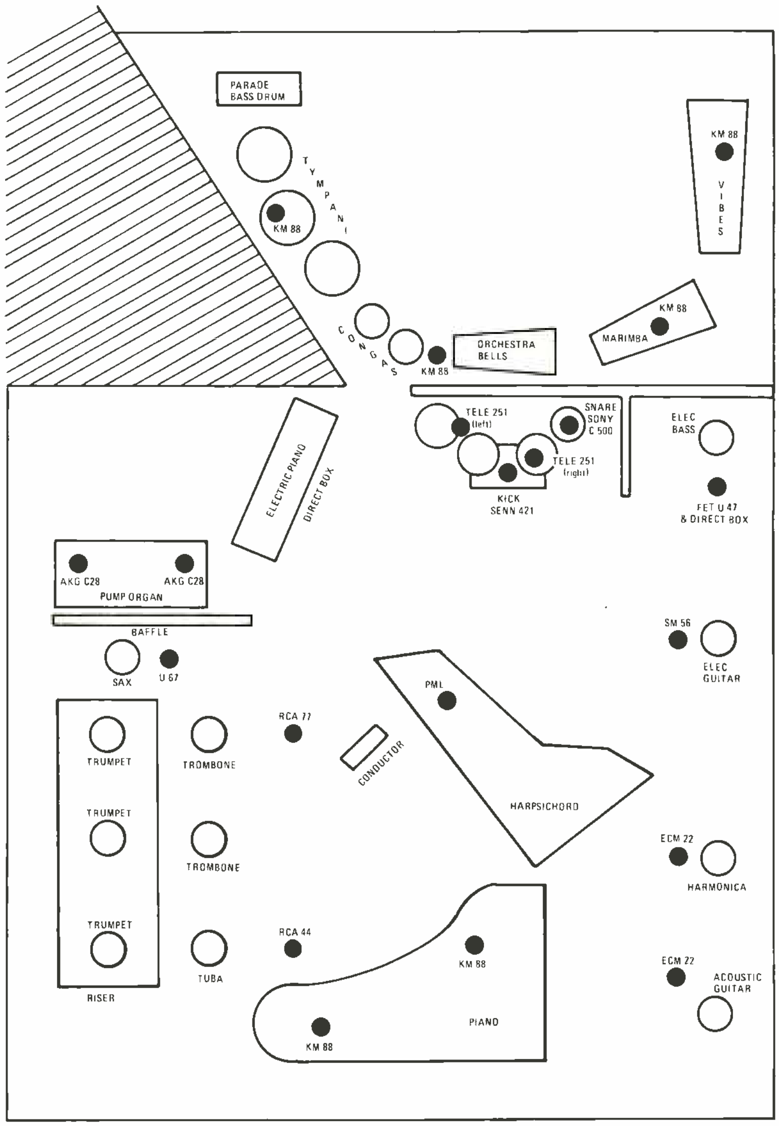
ONE STEP BEYOND

The success of their second direct-to-disc album re-affirmed Sax's and Mayorga's determination to record an album with a degree of high fidelity so advanced that it would even surpass their own previous albums. With this goal in mind, they recorded "Lincoln Mayorga and Distinguished Colleagues – Vol. III," late in 1973. Again, Mayorga did the arrangements and piano work. Doug Sax was technical overseer, co-producing the album with Lincoln. Even prior to its release, "Volume III" has stirred a good deal of excitement. Musicians and engineers who listened to test pressings felt the album had a vibrant realism that only live recording can capture. Furthermore, critical listeners felt that the fidelity of Volume III exceeded anything that has ever been recorded on a tape or a disc.

WHY DIRECT-DISC RECORDINGS ARE SUPERIOR

There are two major aspects to the superior sound of direct-to-disc records. One of the foremost reasons for the founding of Sheffield was to capture live music. The Sheffield concept is to take a roomful of top notch musicians, like Plas Johnson, Jim Keltner, Mike Deasy, and Larry Knechtel, and to present them with challenging arrangements. What happens after that is, according to Sax and Mayorga, *what musical recording should be; the creation of a performance where gifted musicians work together and off each other to produce lively and genuinely inventive music.*

The excitement of the performance is only one of two important elements in the Sheffield records. The second area involves both electronic and acoustic



studio set-up for 'COLLEAGUES VOLUME III' principles. According to Doug Sax, direct-to-disc records can outperform the best conventional discs because they are free of several limitations imposed by tape recording. Sax explains:

"The truth of the matter is that a well recorded and mastered standard record, played on good equipment, will very closely represent the best that could be obtained in a studio. That record doesn't sound a full generation removed from the master tape. It's right up there in quality with the master tape. But direct to master disc recording is something that's better than tape from the outset. That's what makes "Vol. III" a $10 record, worth it to an audiophile. An audiophile cannot buy anything else like "Vol. III" – at any price . . . I would say the two main factors that are the most instantly obvious when you play a Sheffield record are improvements in phase shift and headroom."

NEW GENERAL TECHNICAL ASPECTS

Sax claims that tape simply cannot accept the full peak energy of most instruments; particularly the percussion instruments, where transients may be several times more powerful than the average (VU) level. Excessive energy causes the tape to saturate, creating distortion. In sharp contrast to the tape, the ability of the disc itself to accept instantaneous peaks is immediately apparent to anyone who has listened to a direct-to-disc recording. In proof of this some of the groove excursions caused by the kick drum are so wide that they can easily be seen with the naked eye.

Recording level, in terms of headroom, is not the only technical advantage of direct-to-disc recordings, according to Sax:

"The headroom is not to be discounted. But even if tape had more headroom it would only be a marginal improvement. The phase shift is another significant limitation of tape. You could record on tape very conservatively and not get into its headroom as much. The headroom problems would be reduced but the phase shift would be the same, and that is possibly the worst single aspect of tape recording. That's why the bass you hear off of a tape sounds softer than the clear, distinct bass we get on the monitors. We have another advantage over tape with modern cutters when we record direct-to-disc. Disc cutters are wrapped in feedback from the amplifier. Anything the cutter does that's incorrect is corrected immediately by the amplifier."

Today's feedback-controlled disc cutters are largely responsible for the tremendous improvement in direct-to-disc recordings, compared to the discs of the 1930's and 1940's. The advances in microphones, studios, and mixing boards also allow a quality not available thirty years ago. While noise is reduced by the direct-to-disc method, Sax points out that there is a qualitative difference between low noise direct-to-disc recordings and noise-reduced tapes:

". . . A disc has less asperity noise than tape. We don't have that terrible sounding "grotzel" that you build up on tape. Disc also gives us an improvement in headroom. That is not the same as signal to noise because you can have a Dolby tape that is as quiet as a disc – that is, the Dolby tape could match our signal to noise, but it would have no increase in headroom whatsoever over a non-Dolbyed tape. Also, the harmonic distortion of tape is very high. The disc measures higher harmonic distortion, but only above the audio range. They'll tell you that they can put a full 15,000 cycle tone at full level on a tape, and that you can't put that on a disc. That's true, but there's no real music that generates full level at that frequency. This is one area where admittedly tape can outperform a disc, but it *adds nothing to the music.*"

While Sax thinks that the increased dynamic range of the direct-to-disc method is only a part of its superiority to tape, many people cite it as only one of the most impressive and immediate properties of Sheffield Records. Actually, as Sax says, the dynamic range increase is not that great – measured on an averaging VU meter. But the actual peaks are quite a bit higher than the peaks on conventional discs, and this gives the disc an immense psychoacoustic impact. The usual limiters, compressors, and noise reduction are omitted, and the original dynamics really come through. The bells are unusually clear and ring out in tune, the horns are crisp with unbelievable presence, and the kick drum really makes itself felt. Most impressive is the piano, which is the most live sounding of any recorded pianos we have ever encountered.

DIRECT-DISC LIMITATIONS & HOW TO WORK WITH THEM

Like any other recording situation, the recording of a direct-disc album has certain distinct physical limitations. Various trade-offs have had to be made to allow the best possible recording and stay within the physical limitations of the equipment. One of the primary areas that must be dealt with is the anticipation of musical activity by both the mixer and the lathe operator: There is no remixing for the engineer to rely upon, so his levels must be perfect, his cues must be right-on, and his equalization must all be done accurately in real time. And, since there is no tape machine with a preview-head, the lathe operator must manually adjust the pitch to obtain the closest groove spacing that will still permit peaks to be cut without risk of overcutting.

To assist the engineers in their gargantuan task, Wesley Lindskoog acted as recording supervisor. Once a brass player and teacher, Lindskoog was able to closely follow the score and to advise both engineer Bill Schnee and lathe operator Arnie Acosta of impending changes in the program. A full talkback system was installed so that Lindskoog could communicate with Schnee and Acosta through headphones.

For Schnee, the most immediate concern was logistical. He had to set up all 22 microphones with perfect placement. The microphones were all individually selected condenser mikes, gathered from every available local source. The idea was to get the cleanest and most natural sound possible. The mikes went through a custom-designed Bud Wyatt console, which was a 24-in, 8-mixing bus arrangement. Only two mixing buses were used, and this output went directly to the adjacent cutting room.

All 24 inputs to the board were utilized, although there were only 22 mikes*, because two channels were used for alternate panning. Also, only 16 channels of the board were equipped with equalizers, so Schnee placed the instruments which required the least EQ on the non-equalized inputs. Then a bit of brightening was added to the overall stereo mix in the Mastering Lab. Schnee commented on his first real experience with totally real-time mixing:

"In mixing, the main problem was trying to second-guess the dynamics of the musicians. That wouldn't be hard if they were going to play consistently, but, of course, they're going to do it differently each time, depending on how they feel. We lost one lacquer of a very good performance in that way. Plas Johnson really cut into a saxophone solo which he usually did very laid back. The other musicians picked up on it, and when Keltner his his drums, the lathe made a 90-degree turn and ruined that lacquer."

We asked Schnee why some sort of limiting was not used to help prevent the kind of sudden level problems he just described. He replied:

"We could have used limiting to help control the dynamics. We even tried it briefly. But we decided that for this type of completely live, high energy album, the limiting would hurt more than it would help. We thought it was better to ride the stuff and possibly lose a part here or there, and not have any of the compression of the limiter. We had worked very hard to get as much peak energy as we could, and were willing to take a certain amount of risk with an occasional overload to get as much impact as possible onto the disc. Limiting certainly would have made Arnie's job easier at the lathe, but we were after as much energy as we could get.

Arnie's job was made a bit easier by Lindskoog's running commentary, which typically sounded like this; "In – organ and bells, . . . out – tenor . . . alto solo . . . vibes – out," and so forth. Arnie actually ran two lathes simultaneously, although he only had to control one, and the other followed automatically. This feature, and the reason for it, will be described below. The lathes are both Scully units, fitted with Neumann SX-68 cutter heads that are attached with Sherwood Sax's own mounts. Sherwood also designed the cutting amplifiers, which are tube-type units.

The two lathes were mechanically linked with a rigid connecting rod joining the carriages. One lathe was used for pitch control and the other was slaved. The cutters, turntable motors, and cutting amplifiers were independent. In this way, two lacquer masters could be cut simultaneously, without essentially more effort than it takes to cut one master. Making simultaneous masters has twin advantages in a direct-disc situation. On one hand, if a disc has an imperfection or if there is a failure in the cutter, amp, or turntable, at least one copy will be usable. On the other hand, the total pressing capability is limited by the number of usable master lacquers, and the more lacquers that are cut, the more discs will be available for sale.

In terms of actual numbers, the total quantity of discs which can be pressed from one lacquer varies from 16,000 to 24,000, depending upon the specific program, as well as the quality of the pressing operation and the pressing compound. Four metal mother masters are made from one master lacquer. From each mother master, four metal stampers are made. Each stamper can turn out between 1000 and 1500 discs before it wears out. This is why the mastering must be as perfect as possible for every take: The difference between a profit and a loss on a given album can be determined by one master lacquer. The most critical step, and the riskiest involves the silvering and plating of the master lacquer. An error there destroys the entire production of that lacquer.

THE PRODUCTION SCHEDULE OF VOLUME III

By definition, direct-to-disc recording is "now-or-never." At the first session for "Vol. III," held on a Monday morning late last November, there was a kidding reference to "money players on a kamikaze record date." The date was recorded at The Producer's Workshop, a studio

* Actually 20 mikes and 2 direct boxes.

located immediately next door to The Mastering Lab. Instrument placement had been mapped out the previous night in order to minimize the set-up time and confusion. Schnee quickly obtained a satisfactory mike set-up. All the instruments were brought in to remain until the album was completed.

Monday afternoon was primarily spent rehearsing the "B" side, which would be recorded first. While several of the musicians were at ease because they had worked on the last Sheffield album, others had to adjust to the long seventeen minute takes. It was also a pretty large band for a small studio. When everyone was ready, the lathes were started, and the performance for side two began.

The music was pretty good, and there seemed to be no difficulties with the mixing. But in the middle of the third take, the musicians were told that none of the previous takes could be used because all the lacquers had been over-cut. The overcutting was attributed to a few factors. For one thing, Bill Schnee's mixing was very aggressive as he endeavored to record as much energy as possible. The hot mixing remained clean, with the Bud Wyatt board's substantial headroom, but the actual peak level into the cutter made a mockery of the VU meters' readings. Despite the fact that Acosta had already allowed for wide cutter excursions, even more space was needed.

The increased width meant that less program material would fit on the side, so Doug and Lincoln deleted one cut rather than sacrifice any level. The session was quickly back in motion, and it was soon clear that the shortened side was entirely manageable. All went well.

After the Monday session ended, Sax and Mayorga joined the rest of the technical staff in comparing some of the lacquers to previous Sheffield albums, as well as to other commercial releases. They specifically wanted to know whether they were meeting the criteria in terms of sound quality. One problem was detected.

There was an inordinate amount of vertical modulation in a few sections of the discs. This was traced to the percussion section that was miked in stereo with four microphones set up to allow any sound to be perceived with a true stereo perspective. The natural vertical resulting from the tympani and large bass drum caused lifts at the cutter head. The miking was changed to compensate for this problem.

Tuesday afternoon the session started a bit late, and ran into overtime. But the takes were good, and by the end of the session, several good lacquers had been cut. During the breaks, the musicians went next door and listened to playback of test lacquers, as well as to some finished performances from the 30 ips safety tape. This way they were able to evaluate their performance, and their comments to Schnee helped improve each subsequent take.

Tuesday evening rehearsals for side "A" began, and a new conductor and a whole brass section were brought in. The brass gave Schnee some more work to do. He later told us:

"I think the biggest problem we had was with the brass. In that room, because of the limited cubic volume, we couldn't move the mikes far enough away. As a result, any of the good quality condenser mikes we tried were just folding up against those charts. No matter what combination of input attenuation we tried, we just couldn't capture that incredible brass sound."

In order to solve the problem, Schnee switched to RCA 44 and 77 microphones. These ribbon mikes worked superbly, bringing out a full, rich brass sound. Schnee learned the arrangements, including the brass section, which were even more challenging than before. Nonetheless, by the end of the evening, everyone had learned his part, and the whole crew was exhilarated by both the session and the arrangements.

Wednesday morning everything went smoothly. When the lunch break came before the seventh and final session, everyone was happy with the knowledge that a good take of side one had already been completed. The afternoon session started a bit slowly. By the middle of the session, however, the performance was back to par, and at least one more successful take was cut. The session ended on schedule, thus completing the recording phase of "Lincoln Mayorga and Distinguished Colleagues – Vol. III."

Later in the week, a critical listening test from the tapes, and microscopic examination were used to determine which of the lacquers were musically and technically the best. These would be used for plating. When the album is released, Sheffield expects to be able to press enough copies to make this record profitable and not merely a labor of love.

Doug Sax and Lincoln Mayorga feel that "Vol. III" is a milestone, and will be a collector's item for both music lovers and audiophiles. This was Bill Schnee's first experience with direct-to-disc recording, and he summed up everyone's sentiments when he said: "I didn't expect to have as much fun as I did. I thoroughly enjoyed myself. It was more fun than I've had in quite a while."

We can only say that listening to "Vol. III" was more fun than we have had in quite a while.

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The New DC300A has *double* the number of output transistors, effectively twice the muscle of the old DC300 at the same price. Each channel has eight 150-watt devices for 1200 watts of transistor dissipation per channel. The DC300A is rated at 150 watts per channel continuous into 8 ohms or 300 w/ch continuous into 4 ohms (both channels driven) and 500 watts continuous into 2.5 ohms (single channel driven).

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As a dual-channel amplifier with separate level controls and circuitry for each channel, the DC300A is almost *two* amplifiers in one. This gives you additional flexibility in controlling your speaker load, as when driving separate front and back speaker systems in a large auditorium, or when bi-amping a system. For 600 watts continuous output at 8 ohms, the DC300A converts to a mono amp with two plug-in parts. This makes it possible to drive a 70-volt line directly without a matching transformer.

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The DC300A output protection circuitry is a radically new design which completely eliminates DC fuses and mode switches and further reduces service problems to the negligible level. It is superior in every way to the old VI-limiting circuit pioneered by Crown and now used by most other high power amplifiers, since it introduces *no* flyback pulses, spikes or thumps into the output signal, whether operating as a single-or dual-channel amp.

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BY PAUL STARK

"Multi-Various" is a phrase we have invented to describe the spectrum of challenges, problems and responsibilities the sound man is continually exposed to in working to achieve the best possible audio environment in club-type establishments. The obvious grammatical redundancy of the phrase is intended to emphasize what may well be the most demanding job in audio. "Multi-Various" equally describes the nature of the equipment. Many clubs have their own, built-in sound system, others rely on the musicians to bring their own sound gear, while still others have a favored sound service contractor who is familiar with the club's requirements.

The wide variety of club type operations has a great deal to do with the measure of effectiveness of various types of audio systems, permanent or portable systems, which are intended to be used in them. Some of these establishments might best be called theatre-clubs, while others are the traditional night-club types. Some are simply bars, and still others of the same sort, removing the alcoholic conotation, are coffee houses. The conveyance of entertainment in one form or another, the owners profit margin notwithstanding, is the sole purpose of any of these types of establishment. Everything should play in second place to this entertainment consideration.

Additionally, for the purposes of this article, all clubs have a few other things in common; small audience, with seating for everyone (10–400), intimate atmosphere, probably some kind of food and beverage service during the performance. Dance floors are common, and are often a special problem. But, without a doubt, regardless of the type of club, one thing a soundman can be absolutely certain of is that there will not be enough room or tolerance allowed for a proper audio set-up.

Specifically, the audio environment and equipment in the finer clubs often rivals that found in small, high quality recording studios. A few have a booth in the audience, or set aside, with multiple input and output mixing consoles that control as many as 30 microphones, and banks of bi- or tri-amplified speakers. Some of the better systems installed in larger rooms employ multiple banks of speakers distributed around the club, driven through delay lines to equalize the transit time so that the direct sound from the stage arrives at nearly the same time as the sound from a nearby speaker. At the other end of the spectrum, there are jukebox and muzak rejects, leftovers from someone's discarded hi-fi junk. There is also a broad range of equipment intended primarily for only moderate to high level reproduction of music, or for reinforcement of speech only. These systems rarely do justice to the cost or effort put into their installation.

ACOUSTIC CONSIDERATIONS

THE ROOM:

The acoustic of the room is at the same time the most important factor to be considered, and unfortunately, is the least likely to receive much forethought

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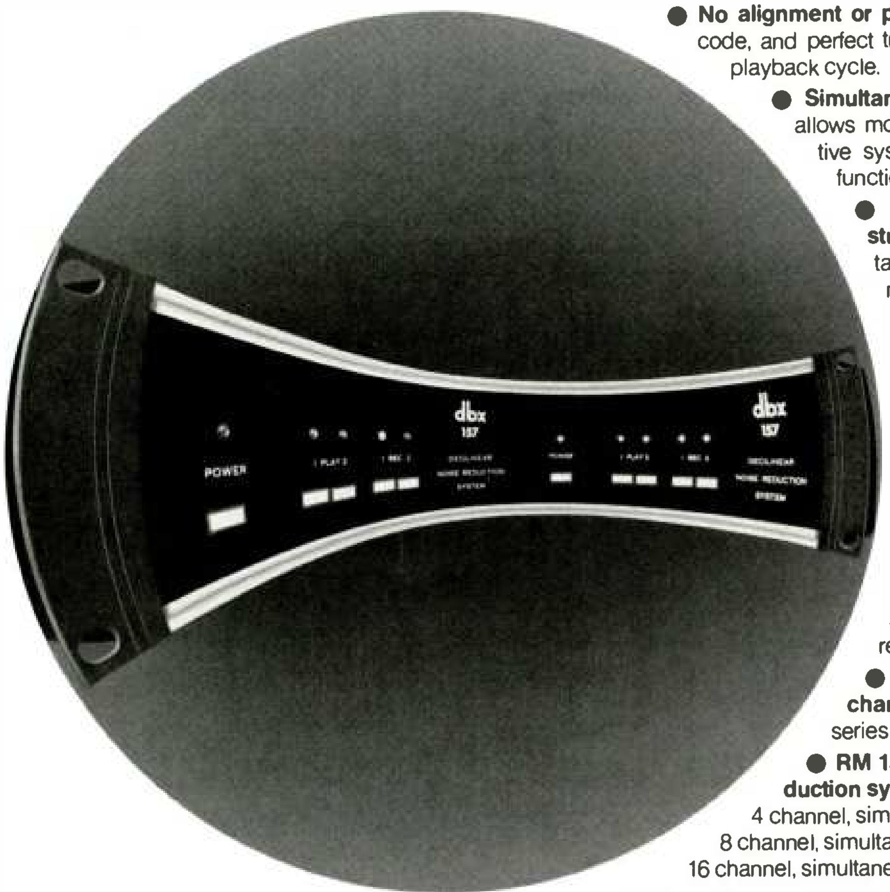

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and attention. Typically, to produce a feeling of warmth and intimacy, many clubs install quite low ceilings, especially, for some unknown reason, in the stage area. Rooms tend to be small enough and live enough to have serious resonance modes in the fundamental vowel range (250-1000Hz). Sound reflections off such objects as hard wall panels and large glass areas wreak havoc with the reverberant response of the room. Another acoustic problem area is the speaker-to-front-seat / speaker-to-rear-seat distance ratio of the room, which is often so imbalanced that it is extremely difficult to distribute direct sound evenly. Exposed low ceiling beams can compound the problem.

For aesthetic design reasons there may not be very much the soundman can do to alter the resonance and reverberation conditions that exist in the room. However, where the management is negotiable on these points, the emperical and tactical use of drapes, carpets and curtains will certainly improve these acoustic characteristics. Even though it is all too obvious, we should all be reminded that a full house (versus a room without an audience) also helps to reduce room resonances . . . in addition to its beneficial effect on the proprietor's financial problems.

In dealing with the uneven sound distribution problem it has been found that most high frequency speakers suitable for club applications are short-throw devices, quite different from those used in large concert halls or outdoors. In order to optimize the pattern of these short-throw speakers and overcome the direct sound distribution problem, the high frequency speakers should always be located as high up as possible, and aimed or angled downward toward the audience.

THE STAGE:

The stage acoustic seldom, if ever, gets any attention. It has however, the governing effect on the performer's sense of tonal balance, particularly in the middle and high frequency range with amplified rock bands. It has often been the case that the high frequency energy from the percussion will bounce around the stage area, causing a *ringing* effect. This multiple reflection, when uncontrolled, causes intense sound pressure levels in the higher end of the audio spectrum, which can and does cause hearing threshold shift. A cycle of events then develops where the soundman and musicians not hearing enough highs boosts the high frequency equalization, this in turn causes further acoustic trauma, with the resultant attempt to boost the highs even further.

Threshold shift is another way of saying that the ear and brain become desensitized to a sound. It is the result of fatigue, especially after exposure to high

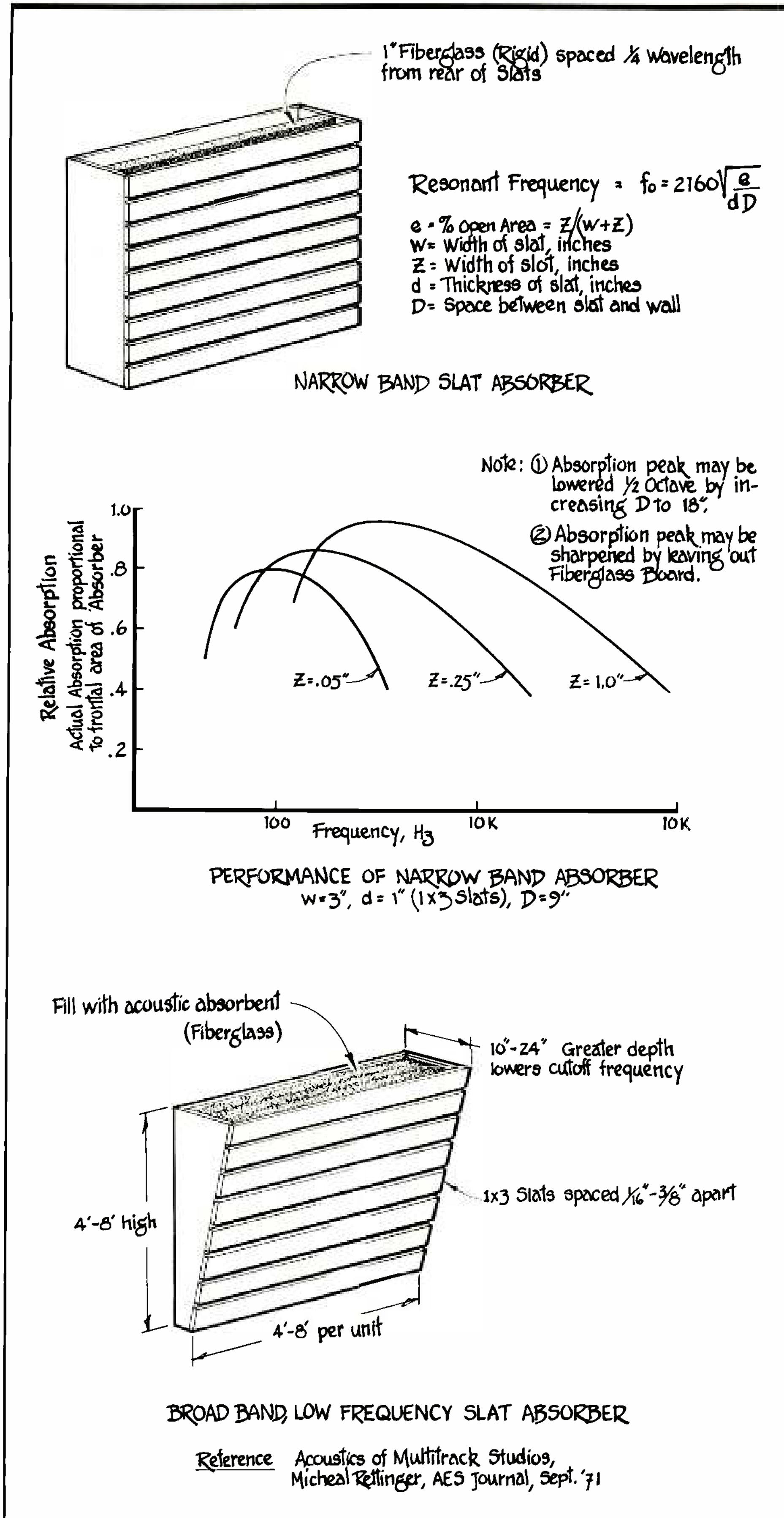


Figure 1 Slat Absorbers

frequency, high energy sound, where the shift causes a loss of perceived highs. Threshold shift is usually temporary, although with prolonged or extreme exposure it can be permanent. Tonal balance is judged from what you hear: as a sound mixer you may be in a club working a loud rock and roll act for some time and generally you detect that the bass is muddy and there aren't enough highs so you boost the EQ and the level until it sounds right again. Actually the problem may have been simply that your hearing over a period of time has lost the highs, due to threshold shift. When someone walks in from the street, he may perceive the sound as terribly shrill. His hearing hasn't been affected yet. But if he stays around for a couple of hours, his hearing may be just as affected as yours. After working one of these shows, as you drive home you sometimes can't hear the engine in your car quite right. The motor may sound *funny* because you can't hear the highs.

High frequency threshold shift can affect the audience as well as the musicians and the soundman. The instrument amplifier speakers are often mounted at the musicians' hip-height. This beams the highs straight out. So while the musicians don't get exposed to the highs, the sound is aimed right at the audience. However, the musicians do get the highs from the drums because the sound, unless confined somehow, is washing all around the stage area: Cymbals, for example, will reflect off the floor, walls, and ceiling. These reflections must be controlled in order to provide the optimum performing atmosphere on the stage.

A method for selective absorption of low frequencies and an extension of the principle to broad band absorption is the slat absorber shown in fig's 1a and 1b. These are practical renditions of the Helmholtz resonator. The absorber consists of resonant chambers (space between slats) connected to an air mass spring (the volume behind the slats). The absorption spectrum may be broadened by the addition of damping material between the two as shown. The use of acoustic rather than electronic "equalization" is necessary because the reinforcement system has no effect on the local acoustic response of the environment to the musicians and their instruments.

What can be done in most cases is to use combinations of reflection, diffusion, and absorption, arranged to focus and control the drum sound. Very helpful is a reflector panel behind the drums which can be fashioned from 3' x 6' x ¼" thick sections of plexiglass, joined with piano hinges, and fitted with edge trim. Placing the drums on a riser also helps, but practically speaking, you can't stuff the drummer in the rafters, and even if you can, most drummers won't stand for it.

Large instrument and equipment cases, instead of being the usual pain in the neck to store until moving on to the next gig, can be used strategically to break up any plane surfaces in the rear of the stage. This increases the low and middle frequency absorption coefficient. Artfully covering a stack of them with drapes or curtains will help break up the highs as well. Just as is done in the recording studio it is sometimes advantageous to place a large beach umbrella, or some similar semi-absorbent surface, above the percussion instruments. Normally much of the sound goes straight up into the ceiling and then returns within milliseconds, its spectrum altered by the complex attenuations of the transit and reflection.

Reflections of drum and percussion sounds contribute to poor sound quality as well as threshold shifts, not only among the musicians on stage, but also in the audience, since the mikes pick up the reflections. The umbrella or other semiabsorptive material above the percussion cures both problems. Otherwise in the vicinity of a microphone element,

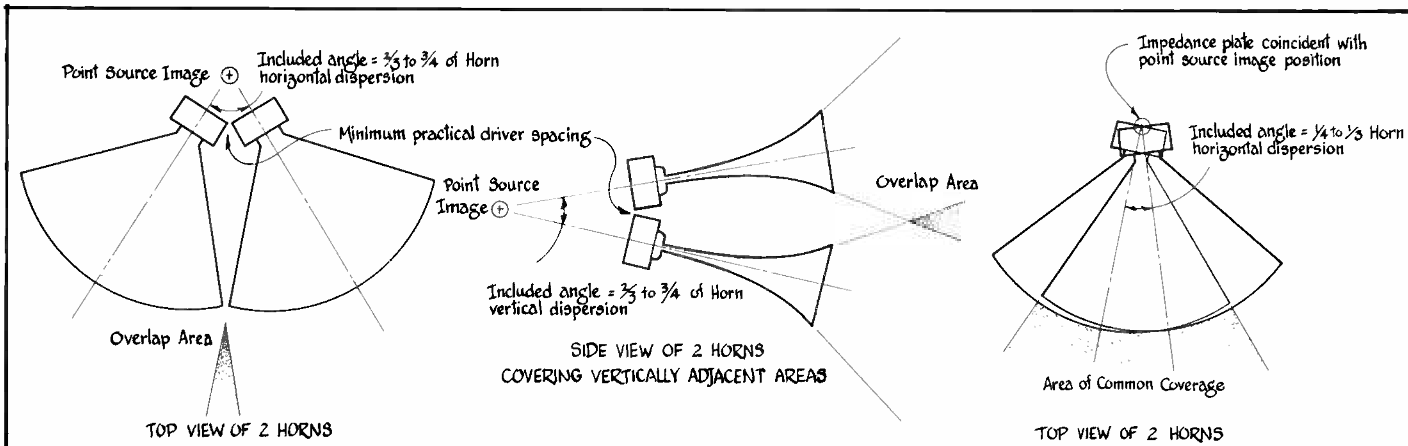


Figure 2 PROPER OVERLAP ANGLES FOR HORNS

the reflected ceiling sound will meet sound reflected from the floor. If the sound is approximately equal from both reflections, an addition or cancellation may occur at various frequencies, depending upon the relative phase relationships. The result will be unpredictable.

SPEAKERS

Speaker selection and placement is, singularly, the most distinctive feature of any given sound system. The two most popular styles are: combinations of horn-loaded direct radiators and compression drivers, and line arrays of direct radiator cone loudspeakers.

If one has decided to go the horn route, it is very important to remember the facts of life concerning horns: Any horn's capability to deliver *clean* sound at rated power levels decreases at least 18 dB per octave below the manufacturer's recommended crossover frequency. Horn choice should be limited to those made from composite materials, such as; combinations of fiberglass and plastic, fiberglass and plywood, or non-homogeneous fiberglass. Generally, horns should not be boxed, although this is sometimes done to facilitate carrying and stacking. However, if horns must be boxed, it is advisable to fill the box with foam or fiberglass to prevent any sound which leaks through the back of the horn from reflecting around the inside of the box causing extraneous noise and distortion. Dense foam or fiberglass will also protect the horns from breaking loose if a box is accidently dropped. If the horn is to stand free, it is good practice to coat the rear facing surfaces with auto undercoat, carpet, or even machinery rubber. This will reduce rear sound radiation as well as adding mass to the horn material. Mass will reduce the tendency for the horn to ring. It will also help to reduce the tendency of the horn material to propagate a wavefront faster than the air in the horn, which can cause a position smear in the output of the horn, and non-linear frequency response. Speed of sound in aluminum is 10 times the speed of sound in free air.

Always attempt to keep all the driver elements covering a given sector in close proximity to one another. The higher the frequency range, the closer they should be kept together. This same point is valid for high frequency direct-radiator arrays. Both systems have equal difficulty in achieving the desired effect, and compromises in distortion are involved with either system.

The difference in direct-radiator and horn loaded sound lies in the spectral distributions of the distortion products. In other words, everything has some distortion, but the frequencies involved may vary. Choice of speakers then becomes a matter of personal taste, at least partially. Some people prefer the "punch" they get with horns. Others feel that horns have a lot of distortion, and they prefer the "soft" sound a cone produces. Practically speaking, the difference in cost, size, weight, and efficiency, as well as maintenance, must be weighed against the requirements of the specific application. A permanent installation has it all over the road system in this category.

Speakers are usually located on both sides of the stage, at or slightly above the stage level. Ideally, such units should be kept well off the floor for two reasons. One, people get in the way. And two, the previously stated problem of distributing the sound equally to the front and the rear seats. Particularly where the high frequency units are too close to the audience in front, a vastly different tonal balance will exist in that area. Sometimes the speaker system can be divided into short throw and longer throw sectors, each radiating primarily into its own section of the audience. Projecting the highs to the rear is fairly easy. But throwing high bass (400-800Hz) to the rear, especially in a club where the floor slopes up sharply, as in theatre clubs, requires a horn that is angled upward. Accurate and carefully placed sound is very critical if there is a dance floor over which the performers must relate well to their audience: If a horn system is to be considered here, it should have an acoustic lens for the near-stage area so that the edges of the horn pattern fall off gradually. If more than one horn is to cover an area, they should be positioned as shown in figure 2. A suggestion for some effective direct-radiator arrays is illustrated in figure 3.

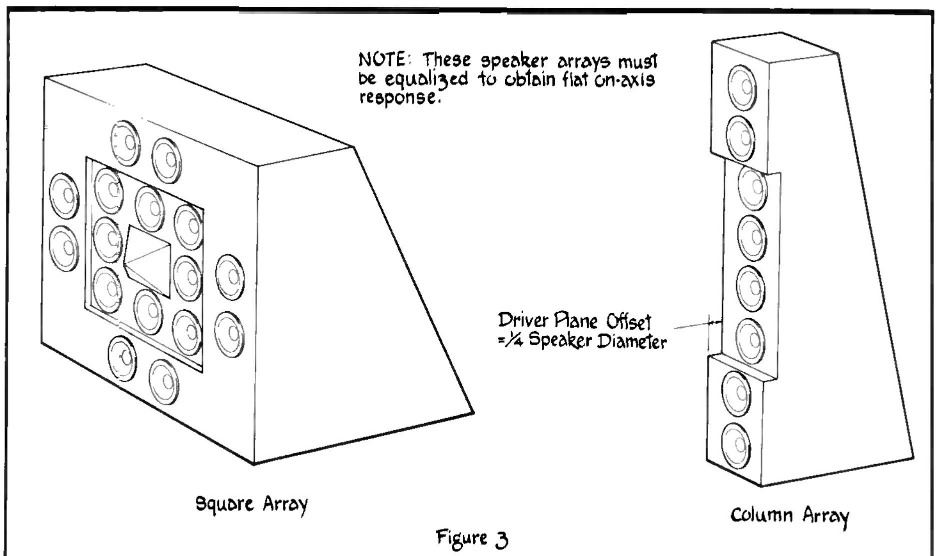


Figure 3

The configurations just described aid in obtaining phase coherence and better pattern control (no hot spots) throughout the driver's range. This results in better intelligibility and localization because there is smoother direct and reverberant response. Another way to enhance localization is to suspend a speaker over the performers, and to direct it at the audience.

The position of the mixing console within the sound field should provide the mixer with virtually the same blend of direct and reverberant sound as that part of the audience which he most wishes to please.

POWER

To make an estimate of the current which a particular set-up requires, we add the AC fuse capacities of the amplifiers, and then we throw in another 25% to 33% for a margin of overload protection. Relatively few amplifiers have good power line regulation. Unfortunately the same goes for mixers, as well as for other ancillary equipment. This sometimes causes problems which can be baffling if you don't realize what is happening: There are times when a high gain mixer will interact with the amplifiers through the power mains, causing a low frequency *motorboat* type oscillation. This problem can be solved by connecting the mixer to a separate circuit, preferably on the opposite side of the AC neutral when the power is split-neutral 220V AC, single phase. This method of delivering power to light and plug panels is common, although sometimes power will come only via a single 110V circuit. The soundman then might have to experiment to find an outlet in the house that sufficiently isolates the mixer from fluctuations in voltage, due to power amplifier current draw. By far, the recommended way of getting power to the mixer is from a well regulated split-neutral distribution system, (described in "The Concert Soundman" article which appeared in the November/December 1973 issue of this magazine).

Any such power system should obviously be constructed to comply with the local industrial electrical codes, and should include a built in tester. We have seen, on occasion, some clubs with three-wire "delta-connected" mains, (220-250V AC), to run refrigeration and air conditioning units. This voltage can cause instant total destruction of solid state equipment in the interval it takes for the line fuse to blow. These mains are easily identified because all three wires are fused or switched simultaneously: Split-neutral single phase 220V AC normally fuses the two "hot" legs with the neutral going directly to the service meter, in which case the neutral is often bonded to ground. This ground, while it can be used for an audio circuit *earth* ground, is not preferred to a separate, high capacity current path to earth, such as cold water or drain pipes. Beware of PVC plastic pipe, which is becoming more and more popular in buildings. PVC will not provide a ground. Watch for situations where metal pipes in the building to which you may have attached a ground may either be interrupted, or may terminate outside the building, above ground, in PVC.

MICROPHONE CONSIDERATIONS

Each performer or group has its own setup. Most tend to think that what they do in the recording studio will work in a reinforcement situation. This is only somewhat true. It must be remembered that live performances are the most difficult to record, and for the same reasons reinforcement is always difficult and frustrating to engineers. Some guidelines can be established with regard to staging and microphone technique in general. Most specific comments that can be made are along the lines of improving the working conditions and increasing the amount of confidence, as well as relieving a good deal of confusion.

Experiments should be conducted to determine the exact microphone and the technique to be used in each case. Using as few mikes as possible is desirable because each time the number of "open" mikes is doubled, the acoustic feedback threshold is lowered by about 6dB. A very real consideration here is that some performers really feel naked without a lot of mikes. If such is the case, as sneaky as it sounds, it may even be helpful to set up mikes that are not connected to the mixer.

The pattern and leakage for each microphone should be considered in terms of what is being done with the microphone. Wherever possible, omni directional microphones are advisable because of their generally superior amplitude and phase response. Amplitude response is the parameter commonly referred to as the frequency response. Amplitude is one kind of response relative to frequency. Phase is another kind of frequency response. Non linear phase response can be analagous to time delay or impulse response (risetime).

For vocal microphones, or other applications involving unusually high gain, a differential microphone pair is appropriate. When contemplating the purchase of omni microphones for this purpose, be wary. Peruse the manufacturer's data sheet to determine that the microphone is a pressure transducer with flat response for both near and far sound sources. This should not be confused with a pressure gradient microphone, which responds to the difference in pressure between two diaphragms, and which is more sensitive to proximity effects. However, it is important to remember, as has been discussed by Jim Coe in his article "Differential Microphones" in the May/June – 1973 issue of Re/p, that not all performances are able to accommodate themselves to the properties of differential microphone operation.

Some popular omnis have acoustic or electrical equalization in them to obtain a response that the manufacturer judges is popular with engineers.

Unidirectional microphones exhibit ragged off-axis response, often with harmonically related peaks and dips. Your ear may perceive this as being flat, but it doesn't help the feedback threshold. Since much more of inter-microphone isolation is due to proximity than to the directionality of the microphone, little is to be lost and much is to be gained with omnis. As a rule of thumb, if you get one mike 4 times closer to a sound source than another mike, the other mike essentially isn't there. An additional advantage is that omni microphones are less expensive and more rugged than comparable directional microphones. This is an important consideration in the real world of sound reinforcement.

Percussion instruments sometimes require a distant, directional microphone technique. Only the highest quality dynamic microphone should be considered here. Condensers tend to clip, and ribbons are too delicate. So dynamics do the best job, providing long life and good sound quality. In reinforcement work drum overhead microphones seem to be more trouble than they are worth. It can be assumed that there is excessive cymbal sound already reflecting off the heads of the drums, sound that overhead mikes will pick up disproportionately. The high-hat cymbals can be picked up by a non-directional or a bi-directional microphone. The snare drum can be assigned to the same mike by placing the mike somewhere between them in a position which gives the best and desired balance.

STAGING

Set changes are always traumatic experiences if they are not carefully planned. The results of a poor set change can foul up the sound of the entire following set. Disorganized microphone setups and cabling can trip up the performers, causing both distraction and anger. A valuable suggestion to begin with is to use one master multipair snake for routing all microphone lines from some central location on the stage to the mixer. (Refer to the diagram on page 46 of the November/December 1973 issue of R-e/p.) Smaller, multipair subsnakes for drums and for vocals will also reduce the effort and confusion.

Immediately after the set, all the microphones used for that set should be unplugged and gotten out of the way of stagehands and equipment men. The mikes can be removed to the side or front of the stage, and are then replaced when the next setup is complete: This job should be assigned to one man. Most club stages are small, and any increase in the freedom of movement will be appreciated. Also, it is remarkable how many microphones are saved from the need to be shipped back to the factory for complete repair as a result of having been knocked over, or trampled on during set changes. This little extra seems well worth the effort.

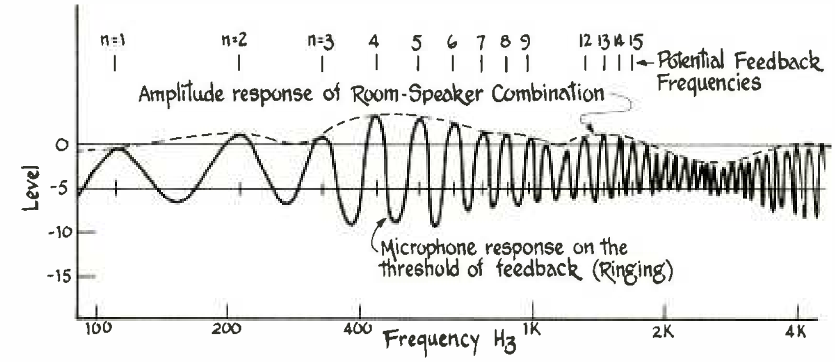
MONITORS

Until recently there was one, and only one, way to provide monitor foldback on stage. Floor mounted, slant-back speaker enclosures were placed on the floor in front of the performer, and were angled so as to aim in the vicinity of the head. However, since this locality was also covered by the rear lobes of the performer's unidirectional microphone, this meant that feedback could and often does become a problem. Newer, more effective monitoring involves far-field cancelling microphones (differential microphones) and arrays of speakers spaced about the stage near ear level. Far-field cancelling mikes are not always desirable, however, because some performers can't get accustomed to the associated close mike technique that is necessary, as commented upon previously.

In small enclosed spaces, such as a small club environment, the reverberant field of the audience speaker system contributes greatly to the sound in the stage area. This contribution, mainly in the vocal vowel range (250-1000Hz), combines with the monitor sound field and the cavity resonances of the stage. The interaction of the two systems can cause operator confusion concerning feedback. Monitor feedback must receive the "killer of the ear award." Monitors tend to feedback at high frequencies and high intensities. The resultant accelerations and loads placed on the ear structure are painfully dangerous. Normally, the system will start to ring and your ears detect a hollow sound before full feedback actually occurs. But operator threshold shift occurs, and the feedback then becomes fully developed before it is detected. To help avoid this severe, damaging feedback, a limiter on the monitor channels is recommended. It must be adjusted so that the gain reduction portion of its range falls slightly above the normal operating levels: This will not cause wholesale gain changes when program ceases, which can cause feedback.

One-third octave equalization can be quite helpful in "flattening" the response of a given setup, and should also be used to eliminate some of the serious "ring modes" or feedback modes after the room/system has been made usefully "flat". The equalization applies to the house system as well as to the monitors, although separate curves will be used. One thing to keep in mind is that over equalization will cause a hollow sound due to rapid frequency dependent phase shifts concomitant with extreme narrow band equalization. Also, the soundman shouldn't be overly eager to comply with requests to put almost everything in the monitors. This is often requested by less experienced performers. One slightly sneaky way to resist this is to have only a few inputs on the monitor mixer.

Figure 4 O = Level Required to Feed Back (Total System Loop Gain = Unity)



Potential Feedback Frequencies for Microphone to Speaker Distance = 10 feet

EQUALIZATION AND ANALYSIS

The purpose for narrow band equalization and analysis is to "flatten" the response in a given area for the room/speaker combination. After this first step, the equalization equipment can and should be used to eliminate the most severe ring modes that are present when the microphone gain approaches the feedback point. The stage should be configured as it will be for the performance during these adjustments. Depending on the amount of acoustic gain desired and on the amount of "naturalness" desired in the channel, this second step can be minimized or left out altogether.

Feedback is due to the combined effects of sound, from the speakers' direct and diffuse fields, impinging on the microphones (on or off axis) either directly or by single or multiple reflection. When the microphone to speaker distances are small, the off axis sound from the speaker may enter the microphone directly, causing feedback at unusually low frequencies. Although the low frequency feedback is sometimes difficult to hear, it does cause added distortion, use power unnecessarily, and it can obviously disturb people.

There are two criteria which, together, determine whether or not a system will feedback, and at what frequency the feedback will occur. The frequency is directly related to both the transit time of the sound wave through the air, and to any electrical delays due to irregular frequency and phase response. While the air delay is usually much larger than the electrical delay, excessive equalization can have an effect. The total phase shift at the frequency of feedback will be related to the feedback frequency in this manner: The total loop phase shift is equal to one complete period (360°) or some multiple of a period of the feedback frequency. While the time delay just described will determine the feedback frequency, it alone does not determine whether feedback will actually occur at that frequency or any other. The gain of the total system "loop" must be greater than unity at that same frequency before feedback commences.

A short discussion of the relationship between phase and time seems appropriate. Phase shift and time delay are two ways to say the same thing in different systems of measurement. Phase relates to percentage of one complete cycle for a wave of a given frequency. The units of phase measurements are degrees; 360 degrees = 1 cycle or period. Phase may also be thought of in terms of the period of the wave. For example; a phase shift of 270° at 1000 Hz, corresponds to a time delay of 3/4 of the period of that frequency, or .75 millisecond. Given the speed of sound as approximately 1100 feet per second this converts to a transit distance of about .6 feet. Figure 4 shows the potential feedback frequencies for an *idealized* feedback system based on microphone-to-speaker distance of 10 feet.

Changing the position of the speakers and/or the microphones will shift the frequencies of the potential feedbacks. If this shift moves the potential feedback frequency into a range where there is insufficient "loop" gain to allow feedback, then the shift in position will cure the feedback problem. Related ways to eliminate feedback include changing the pattern or direction of a speaker or microphone and changing the reflection of sound in the stage area. These changes can alter the gain spectrum sufficiently to move the potential feedback into a more manageable portion of the audio spectrum.

Ideally, 1/3-octave or other narrow band equalization for ring mode suppression should not be applied to the total program channel. It is best applied only to the group or microphones involved in the feedback threshold problem. Narrow band equalization can be applied in a manner that is less than grossly apparent to the ear and yet retains some beneficial effect on the feedback. It should be noted that reversing the phase of the system anywhere in the *loop* reduces any potential feedback frequency by one octave. This is because the phase inversion has added one half period to all the frequencies.[1] Phase reversal can be used, therefore, to change the nature of the feedback situation entirely.

It may also be noticed that in small rooms a reversal of phase has a profound effect on the low bass response of live music only. This is because the live music from the stage combines in the room with the sound from the reinforcement speakers, for better or for worse. It is, for this reason, desirable to control the phase of the high and low frequency portions of the reinforced sound from the mix position. Some severe effects may be noticed in the frequency response in the vicinity of the crossover frequency when the phase is reversed, but the advantage in overall response may outweigh these effects. A likely place to change the phase is in the system's electronic crossover (assuming the system has or needs such a unit). Ideally the crossover would have independent controls, located at the mixer, for the phase and level of each frequency band. There should also be some method for monitoring the power output levels of the amplifiers with respect to the optimum peak level. More detailed specifications would depend on the application and the specific speakers used, which is a subject worthy of a separate article.

MIXING

I feel that I should restrict most comments on mixing to areas that are peculiar to reinforcement. This is difficult because there is a great similarity to mixing for recording. Microphone mixing and equalization go hand in hand with microphone selection and placement. The person responsible for engineering/production should have direct communication with and control over the stage microphones and environment. However, often the person making *production* decisions will not be too familiar with the requirements and peculiarities of the audio system. For that matter, he may not know reinforcement technology in general. An understanding should therefore be reached, before the show, about who will do what, and about when and where it will be done. The system operator will usually be responsible for program continuity, including any recorded music or announcements during the intermission and set changes ("The group will be showing up any minute now, folks!").

The usual signal processing techniques are used, but exercising extra care in the area of compression, limiting and equalization. These all have severe effects on the amount and the nature of feedback. On the other hand, "hard" limiting is often used to protect the speakers. The best place for this is after the crossover, as this avoids modulation effects (specifically, the higher energy, low frequencies will not modulate the gain of the highs).

When there are problems obtaining the desired degree of intelligibility, which frequently occur on the vocal channels, it is recommended that the mixer resist the urge to raise the gain of the respective middle or high frequency equalization. Instead, try turning everything else down, and start to work from there. To do otherwise is an invitation to get caught up in the circle of increasing high frequency threshold shift, requiring further increase in HF boost. This results in listening fatigue, poor quality sound, and ultimately severe feedback. The combined effects of the hearing loss, due to threshold shift, and of the EQ allow the feedback to get out of control before it is noticed.

For every performance setup, some responsible person should walk throughout the audience areas to determine that the distribution of sound is even in most sections of the room. Any gross inadequacies in dispersion, reverberation, reflection, or phasing can be dealt with before the sound check. The sound system can then be adjusted to taste from the position of the mixing console.

PREPARATIONS PRIOR TO THE GIG

Some of the recommendations set forth in this article are idealistic, to say the least. Compromises will be made. Various types of equipment and differing complexities of staging, audience and artist requirements will demand compromises — not to mention money, or the lack of it. This author is of the opinion that sufficient advance notice of some unusual requirement will always help the situation. If need be, some special equipment can be rented for a single show. Artists' managers, their agents and soundmen should know what their performance requires in the realm of equipment and of service. They should make some effort to communicate these requirements to the sound system operator at least a week in advance so that the necessary provisions can be made.

1. Robert B. Schulein, "Microphone Considerations in Feedback Prone Environments," Audio Eng. Soc. preprint No.825 (K-1), Oct. 1971.

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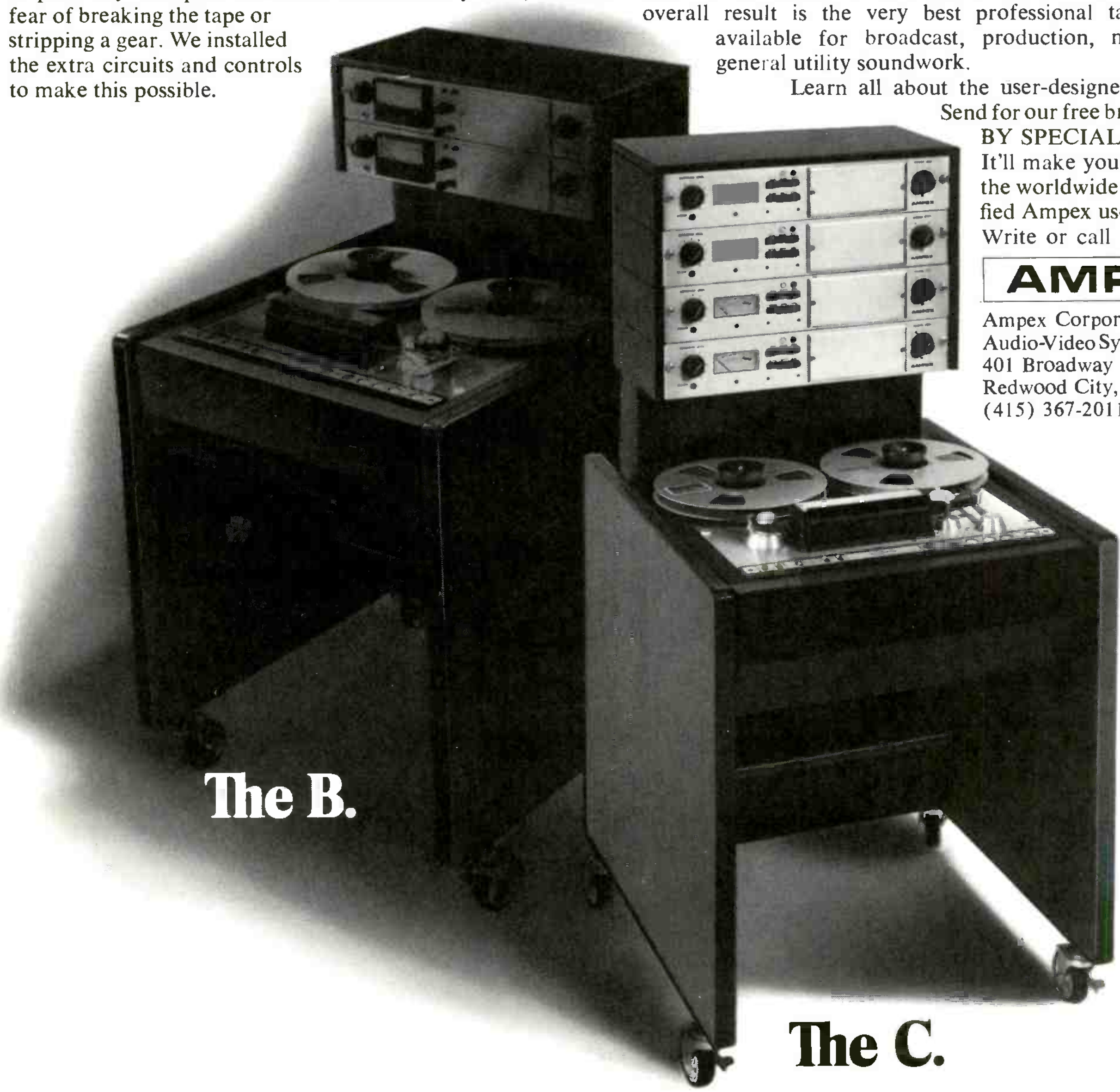

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TWO MICROPHONE SUBSTITUTION DEVICES

by Rick Hilburger
Courage Enterprises

A handy new gadget destined to take a respectable place amid the milieu of sound reinforcement equipment is the microphone substitution device, hereinafter referred to as an MSD.

The MSD is a resistive network or attenuation pad. It plugs into the "extra speaker" jack of a musical instrument amplifier to provide a microphone level output.

While on the road as a sound technician working for Seneca Sound Inc. of Tonawanda, N.Y., I came across several such crude custom-built devices that roadies in the employ of British rock groups religiously toted about with the rest of their regalia insisting they be used in place of our microphones. Most often, only one MSD was used and that for bass guitar. In other cases, notably with Manfred Mann, as many as three or more MSDs were pressed into service for any and all electric instruments.

The MSD'ed amps, I discovered, always seemed to sound better than the amps that were miked, just as I suppose a home recordist would expect to get better results taping directly from his pre amp outputs rather than from two mics set up in front of his stereo speakers. As an audio heavy might put it: two stages of transducer-contributed distortion are bypassed thus resulting in a less muddy sound.

On our tours, other roadies without MSDs often asked if we had them, and so it was decided to design an adjustable MSD to fill the demand. After several experimental models, I finally arrived at a circuit which worked out well in field tests in half a dozen such units (see Fig. 1).

While in use, many practical advantages came to light. No more mike stands to set up in front of amps, to be knocked over by guitar cords and exuberant musicians. No more microphones or loose mike holders that too often ended up pointing in the wrong direction during the middle of a set. And no more seepage from ambient noise. Other problems associated with microphones such as coloration, transient response, proximity effect, wind noise, and mechanical shock susceptability, just don't exist with MSDs.

For such advantages one might expect to pay more, but in this case not so. The MSD of Fig. 2 can be built for under fifteen dollars in about four hours with all the necessary parts and tools already at hand.

The only possible disadvantage that I have come across thus far is that MSDs, being flat, may require some equalization. Since an MSD takes it signal directly from the amplifier, the coloring effect of the speakers and cabinet are not encountered and may be compensated for at the mixing console usually by an appropriate amount of high frequency roll off. However, with some instruments, for example a Mini Moog, rich overtones normally lost in a speaker of limited bandwidth, may come to life with the use of an MSD. Electronic effects such as fuzz, tube distortion, and reverb, etc., will be picked up, of course. And also keep in mind that wherever an "extra speaker" jack is unavailable, a simple "Y" adaptor suffices.

Here is the aforementioned design that was first to be successfully field tested.

The single-pole five-position rotating switch is preferably the shorting type and in this circuit provides –30dB, –45dB, –60dB, –75dB and –90dB of attenuation in 15dB steps. Input and output shields may be continuous or switched to non-continuous. In spite of a ground loop possibility, best performance in my experience was most often attained with shields continuous. In either case, hum should almost always be imperceptible providing the guitar itself doesn't contribute hum. The transformer used is a UTC ouncer, type 0-8, and is a quality audio transformer specified by the manufacturer to have a ± 1dB 30-20,000Hz response. Its purpose here is to provide a low-impedance balanced output signal at mike levels. Its primary impedance is 15 K Ohms and its secondary impedances are 50, 200/250, 500/600 Ohms selectable.

A more recent design of mine, a patent for which is being applied for, eliminates the need for a transformer or a balanced output signal, yet provides the same advantages (see Fig. 2).

The output here still utilizes the standard XL-type 3-pin male connector. The cable is still two-conductor shielded, and the intended input is still a balanced low-Z microphone input. Any stray magnetic fields capable of inducing hum currents will induce currents of equal

continued on page 62

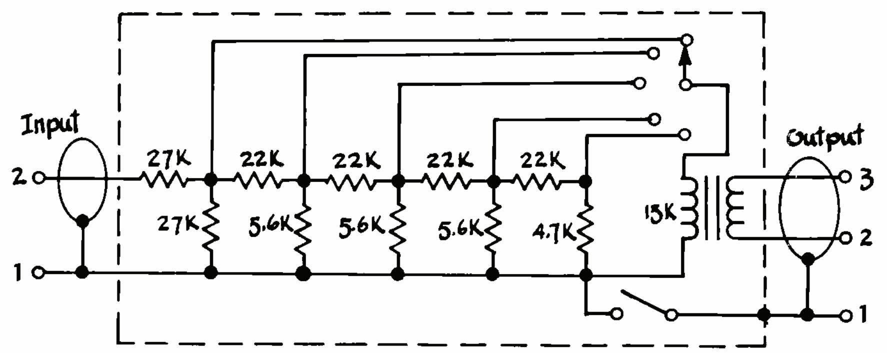


Figure 1

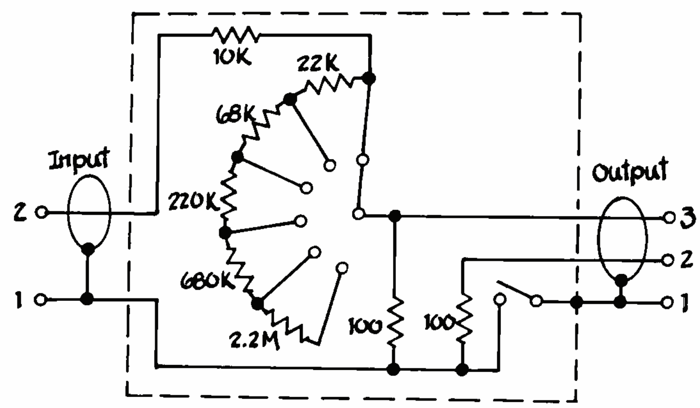


Figure 2

Scully Shows You How To Be Perfect

Without Paying The Price.

As a professional, you want the finest in a professional recorder. The best sound reproduction possible. Simplicity of operation. Reliability coupled with ease of maintenance. And, you don't want to pay a fortune to get it. In short, you want perfection at a perfect price. You want the new 280-B Recorder/Reproducer.

Unmatched Performance.

By designing the 280-B electronics around the new high-energy tapes. The S/N ratio is perhaps the best available in any recorder at a comparable price. Up to 72 dB on full track .25″ tape at mastering speed. A sharp 68 dB on two-track .25″ and four track .50.″

The 280-B also features more head room and an increased record level for maximum signal utilizing the high output tapes. And band widths are a very flat ± 2dB, 30Hz to 18 KHz. It all adds up to greater performance than you've ever been used to.

Quick, Simple Operation.

The more sophisticated we've made the 280-B, the simpler we've made it for you to operate. Our new OPTAC™ motion sensing system gets a new standard of efficiency in tape motion control. Now you can go from one transport mode to another without touching the STOP button. And enter and leave RECORD while the transports in PLAY. OPTAC™ and the 280-B's new logic circuitry make the exact moves for you at the right time.

Easy Maintenance.

New solid state circuitry and mother-daughter board architecture give the 280-B a greater reliability factor. They also make testing, repair and replacement easier. All signal electronics are in slide-out drawers. No more bending down and reaching around. Individual channel modules go in and out easily, too.

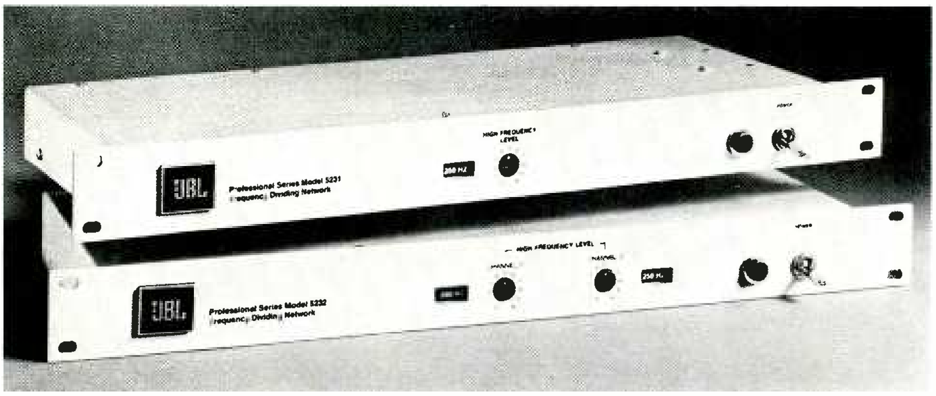
If the 280-B sounds too good to be true, wait till you hear it. And wait till you find out the price. We've made it very easy for you to get the best.

For more detailed information and prices on the 280-B, call or write: Scully/Metrotech, 475 Ellis Street, Mountain View, California 94040. (415) 968-8389. TLX 345524.

Scully/Metrotech
Recording Divisions of Dictaphone

Circle No. 132

NEW PRODUCTS



JBL PROFESSIONAL SERIES ELECTRONIC FREQUENCY DIVIDING NETWORKS

Two new electronic frequency dividing networks in their Professional Series have been announced by James B. Lansing Sound, Inc. (JBL). The models are 5231 for single channel and 5232 for dual channel applications. The networks are designed for use with studio monitor or sound reinforcement loudspeaker systems where bi-amplification or tri-amplification is desirable. The use of electronic frequency dividing networks and multiple amplifiers results in a cleaner signal being fed from the power source directly to the individual loudspeakers of the system. By dividing the audio spectrum before power amplification, treble tones are separated from, and are unaffected by, bass frequencies. The result is more efficient utilization of available amplifier power. For example, a system consisting of 100-Watt low frequency and 50-Watt high frequency amplifiers will provide the same low distortion performance as would a single 300-Watt amplifier driving the loudspeaker system through a conventional passive frequency dividing network. Direct coupling to the loudspeakers eliminates the insertion loss typical of most passive networks and also permits realization of the maximum damping factor available from a given amplifier.

The crossover frequency is determined by inserting the proper printed circuit card into each channel's circuitry. Inserts are available for the following crossover frequencies: 250 Hz, 500 Hz, 800 Hz, 1200 Hz, and 5 kHz. A blank card is also available for construction of circuitry to provide alternate crossover points. Each channel is provided with a level control for high frequency shelving.

Features: High frequency shelving control, Crossover frequency selected by plug in circuit board, 12 db per octave filter slope, THD less than 0.5%, Signal/Noise ratio greater than 90 db.

Size: one EIA standard rack space. Price: Model 5231 $165.00 List, Model 5232 $198.00 List, Crossover Card (one per channel) $15.00 List.

Complete specifications are available from the manufacturer.

JAMES B. LANSING SOUND, INC., 3249 CASITAS AVE., LOS ANGELES, CA. 90039

Circle No. 133

LATEST IN AMPEX 440 SERIES RECORDER/REPRODUCER, THE AG440 'C' MODELS ANNOUNCED

Improvements in the 440C manual controls, tape guidance, electronics, signal-to-noise ratio, tape editing and serviceability are features described in the introduction of the new AG440C.

The 440C transport eliminates the problem of switching from fast forward-rewind to play by adding motion sensing. Edit control now releases transport brakes eliminating handling of the tension arm. Sapphire guides and a flutter idler reduce skew, thereby improving tracking.

Improved electronics design of the 440C Series extends high frequency response, and Sel-Sync TM signal-to-noise ratio is equal to the reproduce signal-to-noise ratio. Automatic switching has been incorporated in Sel-Sync mode. Output now switches automatically from monitoring Sel-Sync to monitoring input when a channel being reproduced in Sel-Sync is put into record.

Switch-controlled two line output impedances, 600 ohms or 150 ohms, are now incorporated along with a plug-in etched board, and meter sensitivity for + 4 DBM or + 8 DBM line, selectable by a switch.

Head assemblies are easily replaced with full access for editing, cleaning or demagnetizing. Space for a fourth head is provided for a four-track stereo head, single track head, or any other special purpose record or reproduce head.

The AG440C is available in full-track, 1/2 track, two-track, 1/4-track configurations for 1/4-inch tape and a four-track configuration for 1/2-inch tape. Portable or console models or unmounted machines for rack installations are available.

Maintenance is simplified. Record and ready lamps can be replaced easily from the front, coin slot screws are used on PWB panel covers, and VU meters are flush mounted, minimizing breakage. Meter lamps are bayonet-type, and the stainless steel electronics front panel is easily removed for cleaning or replacing.

The AG-440C-8 is an eight-track capstan servoed version that handles 10-1/2 inch reels of one-inch tape.

The AG-440C is designed to take advantage of the capabilities of Ampex 406 and 407 Series professional audio tapes.

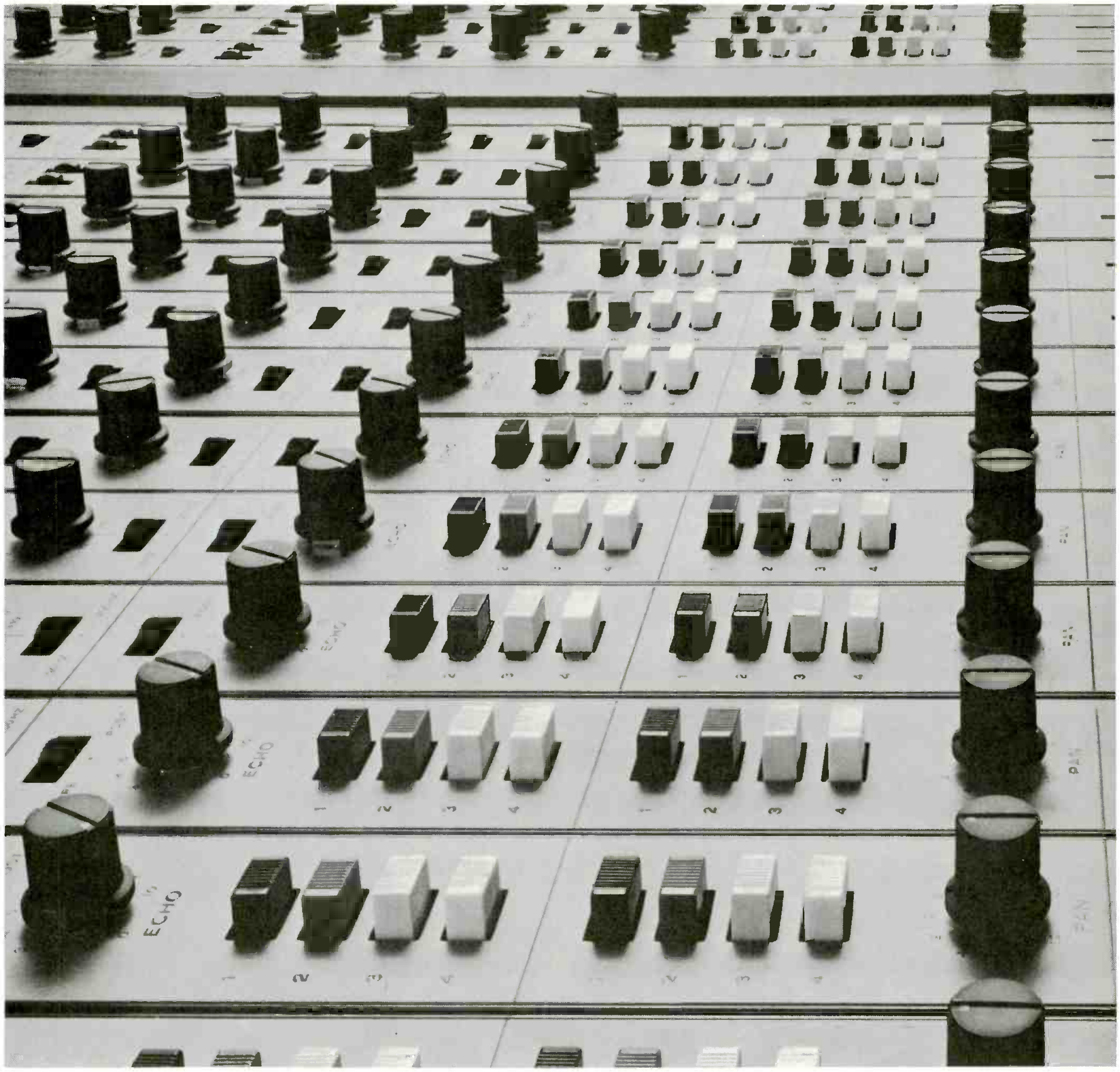
Prices for the 440C Series range from $2,585 to $9,950, depending on customer requirements. Deliveries begin in February.

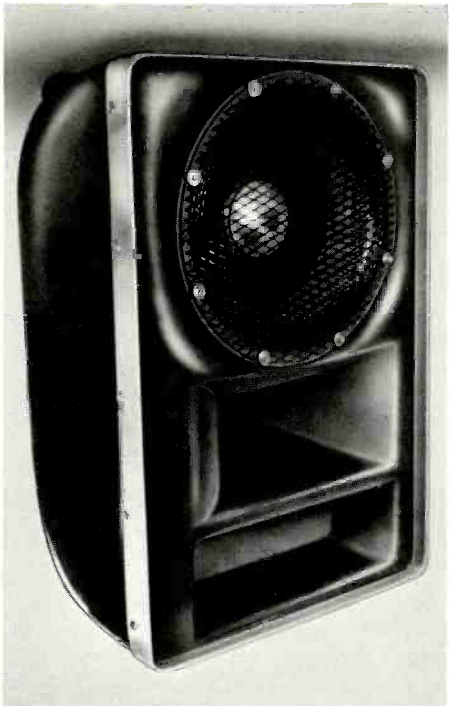
AMPEX CORPORATION, 401 BROADWAY, REDWOOD CITY, CA. 94063

Circle No. 134

'PBL' ADDED TO COMMUNITY LIGHT & SOUND REINFORCEMENT LINE

The PBL by Community Light and Sound is a recent addition to their line of fiberglas PA equipment. Ideally suited for instrument amplification or PA use, the "roundback" reflex design of the fiberglas cabinet enables the PBL to take inputs in excess of 200W RMS. Incorporated in the one piece molded


ECHO
PAN



faceplate is a 15" speaker opening, a molded tuned port, and an integral HF horn that accepts any 1-3/8" x 18 threaded driver. Total fiberglas construction and heavy-duty hardware make this compact, full range cabinet virtually immune to the rigors of touring and rental service.

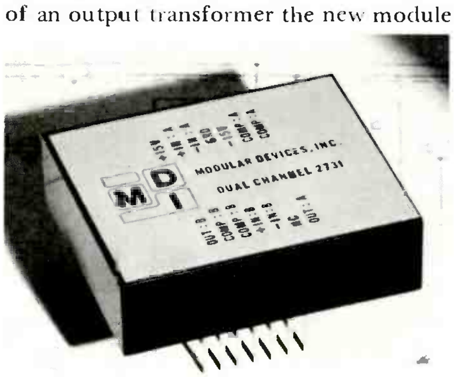
COMMUNITY LIGHT & SOUND, SIXTEENTH AND REED STREETS, PHILADELPHIA, PA. 19146

Circle No. 136

DUAL LOW NOISE AUDIO OPAMP FROM MODULAR DEVICES

A new dual, low noise operational amplifier designed for medium power level audio frequency applications is available from Modular Devices, Inc.

Known as Model 2731, the new dual operational amplifier contains two complete amplifiers in a single package. This feature greatly enhances the already wide flexibility inherent in the operational amplifier configuration. The user can now design two amplifier combinational circuits such as microphone preamplifier-line amplifier, phono preamplifier-line amplifier etc., with a minimum of additional components. With the addition of an output transformer the new module can drive a speaker to 4 watts continuous peak power.



Model 2731 features output power of 2 watts per channel, and therefore, medium power applications of 4 watts may be satisfied with a single module. Other features include - output short-circuit protection, small size of 1-1/2" x 1-3/4" x 1/2", dual in-line 14 pin configuration for PC board mounting and a simplified system wiring.

Price: $39 ea. small quantity. Delivery: Stock to 30 days.

MODULAR DEVICES, INC., 1385 LAKELAND AVENUE, AIRPORT INTERNATIONAL PLAZA, BOHEMIA, NEW YORK 11716.

Circle No. 137

NEW REVOX A700 TAPE RECORDER

The A700 is a completely new, "state of the art" stereo tape recorder using tomorrows technology to provide ease and simplicity of operation and to be capable of the most demanding performance requirements in recording, broadcasting and audiophile applications.

Some major features include the following: 3 motor, 3 speed (3.3/4, 7 1/2, 15 ips), computer type digital control logic with memory circuits, quartz crystal speed control reference, frequency and phase servo system for capstan speed control, two tape tension sensors governing servo-controlled reel motors, logic controlled tape tension, automatically maintained even with mixed reel sizes, electronic tape motion sensor, minutes and seconds readout on tape counter.

Plug in head assembly (1/4 or 1/2 track available), 3 heads (standard), with fourth control head (option). Fail-safe auto stop logic to eliminate possibility of tape breakage, electronic pause control operating on all functions. Instant repeat play control, continuous unattended record or play function, solid state switching of audio circuits, built in four input mixer. Switched selection of 12 input sources including 4 balanced hi/lo microphone inputs, built in magnetic phono pre amplifier.

Master record level slide fader, stereo echo, 5 independent stereo outputs. Standard zero-level line outputs and level & tone controlled outputs. VU meters with instantaneous over modulation indicators, variable speed (+ or - 7 half tones; with remote control accessory), variable speed (2.5 to 21.5 ips; with external oscillator), input or off tape metering.

Engineering features include: 1.638 Mega hertz speed control reference, 2 custom built large scale integrated circuits, 19 integrated circuits, 93 transistors in non critical circuits.

Price: $1,800.00.

REVOX CORP., 155 MICHAEL DRIVE, SYOSSET, NY 11791

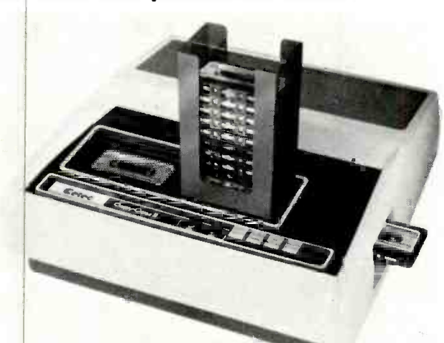
Circle No. 138

CETEC ANNOUNCES NEW AUTOMATIC CASSETTE DUPLICATOR

Cetec, Inc., has announced the expansion of its line of high speed duplicating systems with the introduction of the Copy-Cass II Automatic Cassette Duplicator.

Operating at 20 IPS, the Copy-Cass II duplicates a cassette master on up to fifteen blank cassettes automatically at one loading and signals the operator when the duplication job has been completed.

In normal use, stereo tracks are recorded simultaneously. Monophonic tracking also can be selected alternatively by the touch of a pushbutton switch.



The Copy-Cass II utilizes a unique silent-sense logic system which senses the end of master program, automatically causes both the master and the duplicated slave cassette to be rewound, then ejects the completed duplicate before beginning the next duplicating cycle.

The Copy-Cass II requires no special wiring or installation.

It is priced at $1,875.

CETEC INC., 13035 SATICOY STREET, NORTH HOLLYWOOD, CA. 91605

Circle No. 139

PLAN NOW: to attend
48th AES CONVENTION
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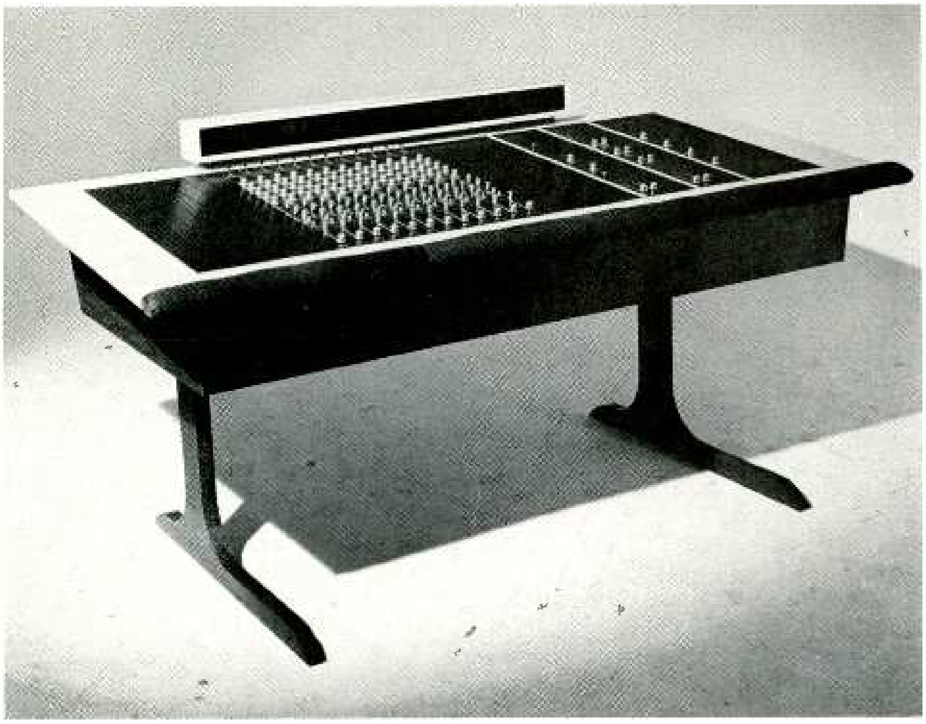
HOTEL
May 7 through May 10, 1974



INTERFACE ELECTRONICS SERIES 300, 24 X 8 PRO MIXER

The new Series 300 offers eight track 16 and 24 input fully wired mainframes with power and XLR type input and output connectors, plug-in input modules with nonexclusive pushbutton track selection, panpot, echo send, cue (which doubles as monitor-only solo), three octave-wide peaking boost or cut equalizers with a choice of three frequencies on each, adjustable input gain and input pad, line/mike switch, and a six inch conductive plastic slider. Each module is provided with balanced 200 ohm mike input and bridging single ended line input, as well as module output.

Using module outputs, more than eight tracks can be fed. The fully modular system also includes masters and setup oscillator on the output module, and up to three mixdown-monitor modules with automatic transfer of cue to monitor if desired, and mixer-playback switch; the talk-slate module includes slate track select and talkback / slate microphone.

All Interface Electronics mixers are capable of performance comparable to the finest professional equipment, and insure reliability through the use of plug-in integrated circuits, plug-in modules, conductive plastic sliders, gold plated card and IC connectors, tantalum or computergrade condensers.

Common specifications frequency response: $\pm$ 1 dB 20–20,000 Hz. Equalizing: $\pm$ 12 dB at specified frequencies. Distortion: less than 0.1% THD @ 400 Hz, + 3 VU. Noise: less than 0.6 microvolts equiv. input. Inputs: Mike: 200 ohms balanced, XLR type connector max. level 0.5 volts RMS max. level 5 volts with int. pad (100B, R). Line: 10K unbalanced phone plug. Outputs: Track: approx. 1 volt RMS at zero VU unbalanced, to not less than 600 ohms, XLR connector. Echo: same as track, but phone plug. Echo Returns: 1 volt RMS into 5K required, phone plug.

24 input eight track: $10,640.00 FOB Houston. 16 input Model 16 X 8, $7860.00 FOB Houston.

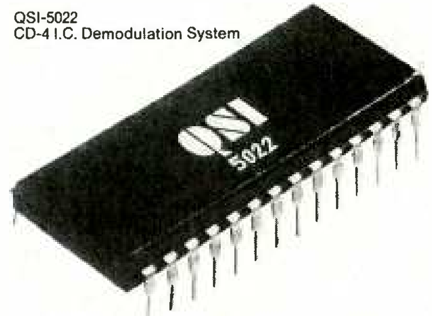
INTERFACE ELECTRONICS, 3810 WESTHEIMER, HOUSTON, TX. 77027

Circle No. 140

QUADRACAST CD-4 DEMODULATOR IC

Quadracast Systems, Inc. announces the 5022; an LSI linear integrated circuit for the purpose of CD-4 demodulation. This integrated circuit performs all functions in the CD-4 demodulator. These include a high quality, low noise preamplifier, Phase Lock Loop FM Detector, high speed carrier drop out cancellation circuit, and a 2-band active expander.

QSI-5022
CD-4 I.C. Demodulation System



Two of these integrated circuits together with a handful of discrete components make a complete HiFi 4-channel demodulator.

A 10-page Application Note and a Data Sheet are available.

The devices are being distributed by: **MATSUSHITA ELECTRIC CORP. OF AMERICA, 200 PARK AVE., N.Y. 10017. QUADRACAST SYSTEMS, INC., 107 NO. BAYSHORE BLVD., SAN MATEO, CA. 94401.**

Circle No. 141

BURWEN MODULAR PEAK VU DETECTOR PREVENTS OVERLOAD AND DISTORTION

New peak VU detector module Model VU306 from Burwen Laboratories permits sound engineers to monitor true sound levels, rather than average or RMS values, hence avoid tape recorder and transmitters distortions that occur when high signal peaks are concealed by modest average or RMS values.

The VU306 modules are intended for use in tape recording, reproducing, record cutting, and in F-M broadcasting applications. They also enhance the versatility and flexibility of sound mixing consoles, simplify microphone placement in concert halls and public address installations, and add to the reproduction quality of high performance Hi Fi installations. As a rule-of-thumb, Burwen Laboratories has found that virtually all consumer Hi Fi music suffers from sound distortion, typically as inadequate amplifier power, or limited preamplifier dynamic range, clip off signal peaks.

A precision full wave peak rectifier within the VU306 module measures individual signal peaks within 5 microseconds, develops a proportional DC output that remains steady for two seconds. The DC output activates any standard d'Arsonval VU meter. After each 2-second "hold" period, the module automatically takes a fresh sample and displays that for the ensuing two seconds. Should a higher peak value occur during any 2-second reading period the detector's DC output advances to the new peak level, and "holds" at that higher peak. Absence of rapid pointer swing vastly simplifies VU meter interpretation.

Frequency response of the VU306 modules is either flat, or has pre-emphasized high frequencies to simulate FM, RIAA, or slow speed tape recorder preemphasis. The modules are 1.5" square by 0.65" high, operate from ± 15 volt DC supplies, and develop 0 to + 2 volts DC output in series with 5,600 ohms for direct d'Arsonval VU indicators. 0 to + 10 volts DC is also available for other uses. The detectors develop full output on input peaks as short as 5 microseconds, provide 0.1 dB response from 10 Hz through 30 Hz, and handle an input signal range of 0.58 V (- 2.4 dBm) or higher.

Price & Delivery: Individual VU306 modules list for $85, less in OEM quantities, and are available from stock for evaluation purposes, with production volumes shipped in 4 – 6 weeks. The modules may be soldered directly onto PC cards, or plugged into standard 7-pin operational amplifier sockets.

BURWEN LABORATORIES, INC., 209 MIDDLESEX TURNPIKE, BURLINGTON, MASS. 01803

Circle No. 142

NEW SERIES 'B' RECORDING CONSOLES FROM MULTI-TRACK

The new version of the Series B recording console is a completely redesigned version of both the electronics and the aesthetics. Among the features are:

All switching done with M.O.S. analog switches, no troublesome relays. Advanced equalization section using inductorless circuits for minimum phase shift and ringing. Light Emitting Diode clipping level indicators on all output meters giving the engineer simultaneous indication of average and peak levels.

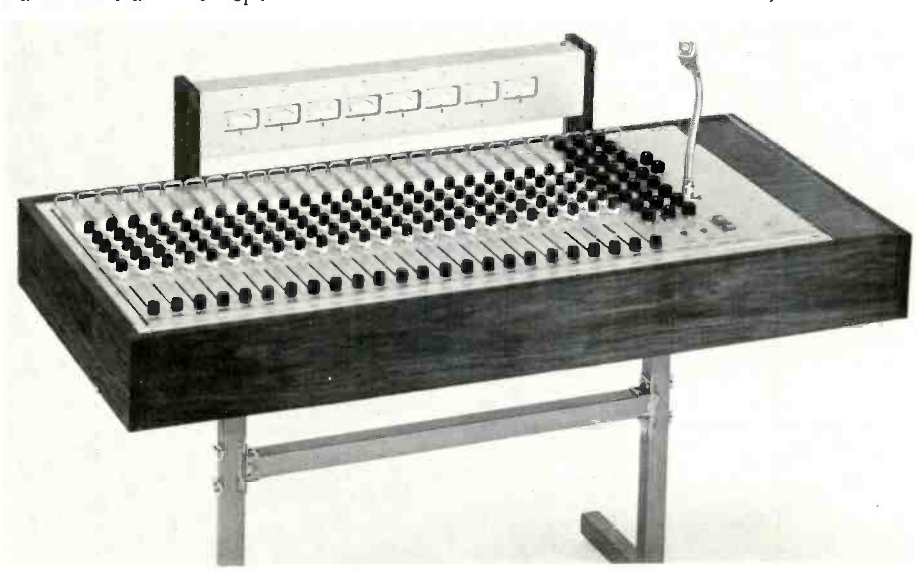
Module interconnections are accomplished using computer flat cable and connectors for maximum reliability and ease of field expandability. All coupling transformers have been eliminated for maximum transient response.

100% operational amplifier circuitry includes short circuit proof bus driver stages eliminating unstable push pull transsistor outputs. All modules are fully enclosed for maximum shielding providing minimum crosstalk and noise. Newly designed slide fader using sealed rotary cermet potentiometers eliminating noise and short life associated with conductive plastic faders. Custom designed, dual tracking power supply that features: regulation to .1%, ripple below 2MV RMS and SCR crowbar circuitry that protects the console in event of any malfunction. All panels are anodized and etched, not silk screened, the panel markings will not wear off.

These features are in addition to other standard features, such as the built in echo system.

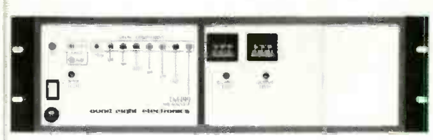
MULTI-TRACK, P.O. BOX 3187, HOLLYWOOD, CALIFORNIA 90028

Circle No. 144



QUAD/EIGHT TM499 TIME MACHINE

An all electronic audio signal delay unit, the Quad-Eight TM 499 Time Machine has many new features and has a new standard of specifications. With a single audio input, up to 499 milliseconds of 12 bit quantized audio delay may be routed into a maximum of five outputs.



A modular package, the TM 499 has these exclusive features: 12 bit digital encoding, resulting in lowest distortion and broadest input dynamic range is combined with a complementary pre and de-emphasis control, automatic muting function, and an integral compressor.

Typical specifications: Frequency response: 20 Hz to 16 kHz ± 1.0 dB, dynamic range: 82 dB, distortion: less than 0.2%, output noise: less than - 90 dBm.

Designed to rack mount in 5 1/4 inches of vertical space, the totally self-powered mainframe is a complete assembly which permits simple and trouble-free operational set-up.

QUAD/EIGHT ELECTRONICS, 11929 VOSE STREET, NO. HOLLYWOOD, CA. 91605

Circle No. 145

continued from page 53

proportions and of the same polarity in both leads of the mike cable's shielded pair. This is because both leads see 100 Ohms to ground at the device's output end. These induced currents will cancel each other out at the balanced input just as they would if a transformer with balanced output were used.

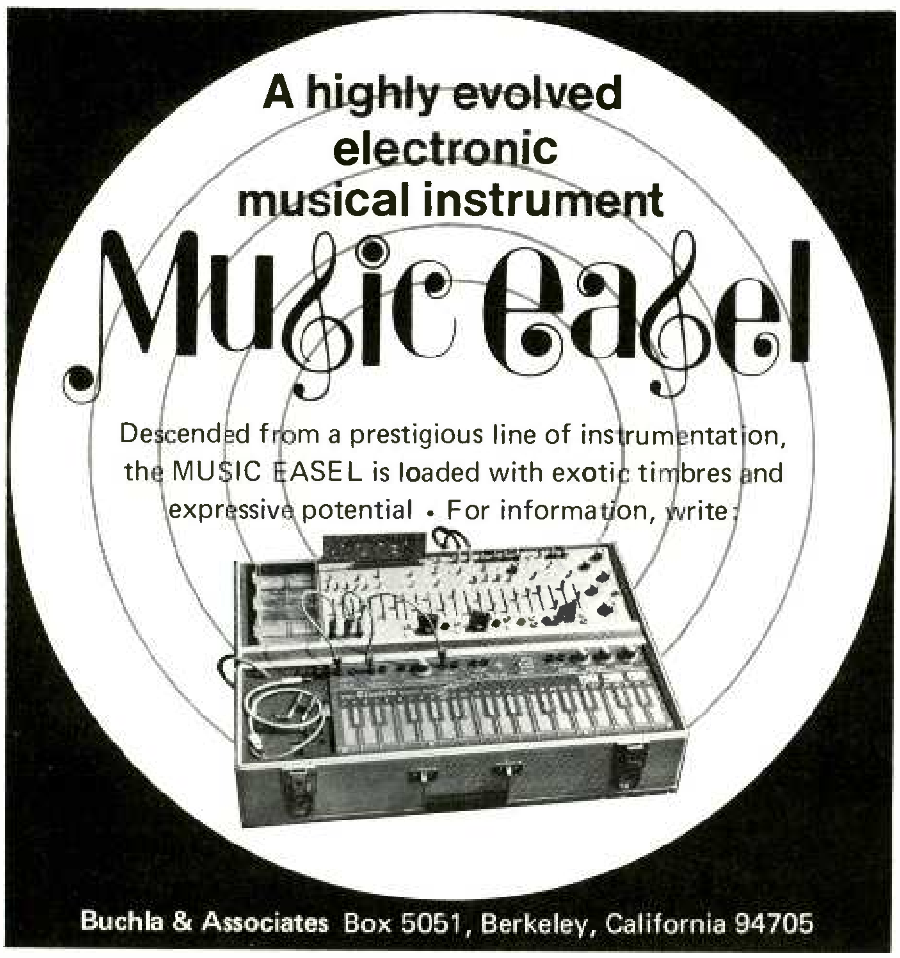
The single-pole six-position rotating switch is preferably the shorting type and in this circuit provides –40dB, –50dB, –60dB, –70dB, –80dB, and –90dB of attenuation. I have found 10 decibles to be the ideal compromising increment for switched pads. This wide range of levels permits both this and the model of Fig. 1 to be used with high-Z and line-level inputs as well as low-Z inputs.

Since loud rock musicians are often concerned about their potency, take care to note the following. This resistive network will yield a very small loss to a power amp. In fact, at the MSD's hottest output setting, a maximum load of not less than 10,000 Ohms is realized. For a 100 Watt amplifier into an 8 Ohm load, the MSD of Fig. 2 would at most draw only .08 Watts or less than one thousandth of the amplifier's power. In terms of decreased sound pressure level, such a tiny loss is by far imperceptible to the human ear.

Soundmen might want to keep in mind these two other uses for MSDs that have worked well for me.

Two Fender Bandmaster amplifiers were cascaded by interconnecting them via an MSD thus doubling the power available to a single guitarist. This method of coupling amps, as opposed to Y-ing a guitar output, enabled the volume and tone adjustments on the first amplifier to be 'automatically' entered into the second amp. Using the same set-up but separating the amps with a longer cable, provides a unique and simple solution to some tricky monitoring problems in which cross-stage musicians have a hard time hearing each other.

Anyone wishing to save themselves the time and trouble required to make an MSD and/or would like a handsome professional-looking unit, may purchase the type illustrated in Fig. 2 from a company formed to satisfy the need for a commercially available MSD. This unit may be recognized under the trade mark of Direct Box™. For a free brochure on the Direct Box™ including order blank, write to Courage Enterprises, 168 South Huxley Drive, Cheektowage, N.Y., 14225, or circle No. *146* on the Reader Service Card.



Circle No. 147

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Circle No. 152

Circle No. 153

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F. T. C. BREWER COMPANY
P. O. Box 8057, Pensacola, Florida. 32505

SPLICE FASTER, BETTER BY SHEARING REPLACES RAZOR. Has attached splicing tape dispenser. Professional quality. Specify 1/4 inch or cassette groove. Price: $24.95 plus $1.00 for handling. Distributors wanted. Details: NRPR, BOX 289, MCLEAN, VA. 22101

FOR SALE: Ampex AG-440 8 Track, $6,200. – super clean, available immediately. Crown CX-844 ¼" 4 Track, 4 channel, $1,500. 2 Bozak CMA-10 Mixers, $200. each. **BEE JAY RECORDING, P.O. BOX T, WINTER PARK, FL., 32789, PHONE (305) 293-1781.**

FOR SALE: Audio Designs Console. 16 in/16 out. 3 years old. $15,000.00 Call (312) 644-1666 for further information.

FOR SALE: One 8 track recorder Ampex 300 deck almost new heads. Custom designed and built electronics with selsink machine currently in use asking price $6,000 or make offer. Reply CONTINENTAL RECORDINGS INC., 12 IRVING STREET, FRAMINGHAM, MASS. 01701, TEL (617) 879-2430.

FOR SALE: AG 350-2 Ampex recorder. Please call for complete description Scott Kent, BKM Assoc., (617) 658-6565.

BUILD YOUR OWN highest quality microphone mixers, consoles, phono preamps, crossovers, equalizers, or voltage controlled devices, using modules. Free catalog. BURWEN LABS., 209 MIDDLESEX TURNPIKE, BURLINGTON, MASS. 01803.

SOLID STATE REPLACEMENTS FOR TUBES Direct plug-in replacement of low noise tubes with solid state FET components. Without circuit modifications, replace signal to noise ratio. For mixers, duplicators and pro. series tube type record electronics. Write for data information and application notes for your equipment. AUDIO APPLICATIONS, BOX 3691, HAYWARD, CA. 94545.

FOR SALE: Ampex 8 Track Recorder AG–300 deck with 1" motors, PR–10 electronics with sel/syncs – $4500. THOMAS GREENE, 1814 CRITTENDEN RD., ROCHESTER, NY 14623 CALL: (evenings) (716) 271-6307

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INVESTORS WANTED
For Quad 48 in/48 out dual 24-3M'79 studio, expandable to 96+ in/48 out. Building near completion. 40'x40'x14' studio located in Ann Arbor, Michigan Write or call RICHARD CURTIS COLORTRONICS RECORDING IND. 7887 JACKSON ROAD ANN ARBOR, MICHIGAN 48103 (313) 769-2815 PAGE NO. 193

Used MCI 16 track console & machine 16 in – 16 out console – $15,500. 16 track machine w/locator – $10,900 also 20 in – 16 out custom console – $12,500. FRONTIER AUDIO CORP., 3103 ROUTH, DALLAS, TX. 75201, (214) 651-0152

Allen & Heath mixers, Community Light & Sound horns and drivers, H/H amps. All your sound reinforcement needs, 3-SIXTY SOUND touring packages and consultation. BRANDY BROOK SOUND INC., 488 GAUVIN DR., WARWICK, R.I. 02886, PHONE (401) 821-9580.

FOR SALE: ORTOFON dynamic motional feedback mono disc cutting system. Complete amplifier system: Drive, feedback and feedback–playback monitor preamp. Rebuilt – original factory parts. Guaranteed. ALBERT B. GRUNDY, 64 UNIVERSITY PLACE, NEW YORK, N.Y. 10003, (212) 929-8364

FOR SALE: 1 brand new 3-M, M-79 4-track recorder/reproducer. Full factory warranty, less than 3 weeks old. $4950. CALL CHARLES SOLAK AT (215) 363-7855, (607) 797-3909.

FOR SALE: Quad Eight 16 in/8 out console. 3 years old. Complete 16 track monitoring, full patch bay and producers desk. $14,500. CREATIVE WORKSHOP, 2804 AZALEA PLACE, NASHVILLE, TN. 37204 PHONE: (615) 385-0670 or 383-8682

FOR SALE: (2) Pultec E.Q.P-1 Equalizers – good condition $325.00 each or best offer. CONTINENTAL RECORDINGS, INC., 12 IRVING ST., FRAMINGHAM, MA. 01701, PHONE: (617) 879-2430

FOR SALE: MCI JH-416 Consoles: 24 x 24 $19,000; 16 x 16 $13,500. JH–16 16–Track w/auto locator $13,000, 2 months old. SOUND 80, 2709 E 25th ST., MINNEAPOLIS, MN. 55406, (612) 721-6341

Studio design, acoustical, technical and operational consulting. 20 years experience in Chicago and the midwest. No job too small. BRUCE F. SWEDIEN, 705 HERMITAGE DR., DEERFIELD, IL. 60015, PHONE: (312) 945-8629

FOR SALE: 18 in x 4 out Nuemann self contained console. Danier faders, multi-range EQ, 3 position low-cut filter, cue and echo before-after on each input. 5 position oscilator – talkback – 4 channel and mono monitoring. Excellent condition. Original cost $35,000, – available March 1. Contact: PAMS, BRUCE COLLIER, 4141 OFFICE PARKWAY, DALLAS, TX. 75204, (214) 827-0901

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At a medium price range we can offer you a total turnkey installation from 4 to 24 tracks.
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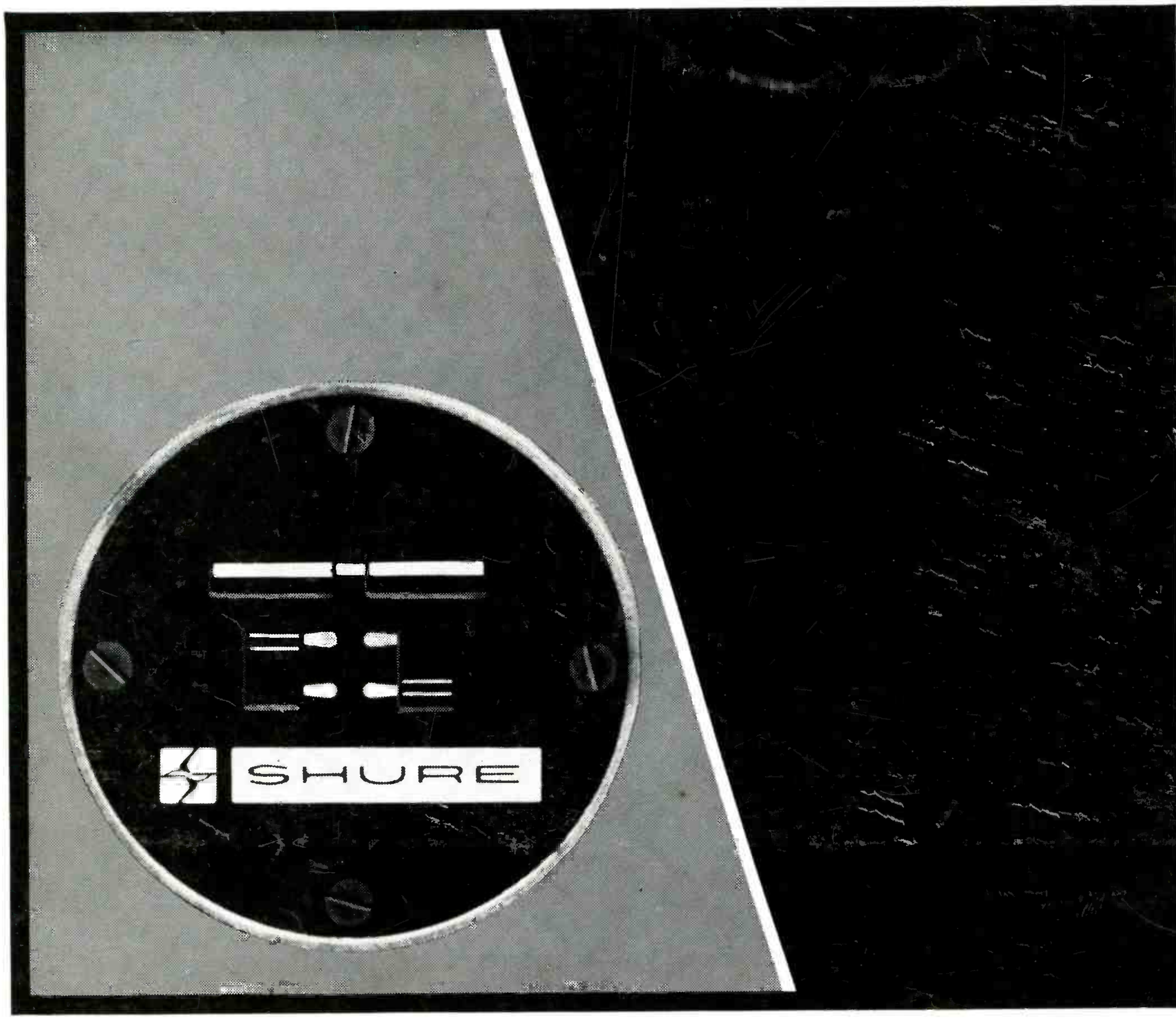
DALLAS, TX. 75201
(214) 651-0152

FOR SALE
COMPLETE DISC MASTERING SYSTEM
Neumann AM-32-B disc recording lathe with variable pitch and depth amplifiers, Westrex 3D cutter head and amplifiers. Two six foot racks of electronics with UREI 1176 limiters, McIntosh 75 watt monitor amplifiers, Pultec program equalizers, HP/LP filters, push-button program monitoring, compatible mono capability, low frequency crossover, JBL S7 monitor speakers, Ampex 445 stereo tape machine with preview heads. Complete system now in operating condition. Total price $22,500.00. For more details contact: BILL ROBINSON, SUNSET SOUND RECORDERS, 6650 SUNSET BLVD., HOLLYWOOD, CALIF. 90028, (213) 469-1186.

EMPLOYMENT

Dedicated audio man for Los Angeles FM Tape Syndication Company. Must be available for personal interviews in Los Angeles area. Salary open. Only top men in the field will be considered. Superb conditions and high pay for the right man. This is a career position with the Foremost Syndicator in the world. Send your resume and a current photo to JIM SOMICH, 6430 SUNSET BLVD., SUITE 1102, HOLLYWOOD, CA. 90028

HELP WANTED – MALE/FEMALE
National Representative - supporting dealers for Hi-Fi company. Not sales. 75% travel. HARMAN KARDON, 55 AMES COURT, PLAINVIEW, L.I. 11803, (516) OV 1-4000.



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The extraordinary Shure SM7 professional microphone features something you've never seen before: a *built-in Visual Indication Response Tailoring System* that offers you four different frequency response curves—and shows you the curve you've selected with a graphic readout (see above) at the back of the microphone! Choose: 1. flat response; 2. bass roll-off; 3. presence boost; 4. combination of roll-off and presence. And there's more: the SM7 delivers exceptional noise isolation with a revolutionary pneumatic suspension mount . . . an ultra-wide, ultra-smooth frequency response . . . an integral "pop" and wind filter . . . and a cardioid pickup pattern that looks "text-book perfect." The Shure SM7 Studio Microphone was extensively field-tested in recording studios and broadcasting stations! Write:

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