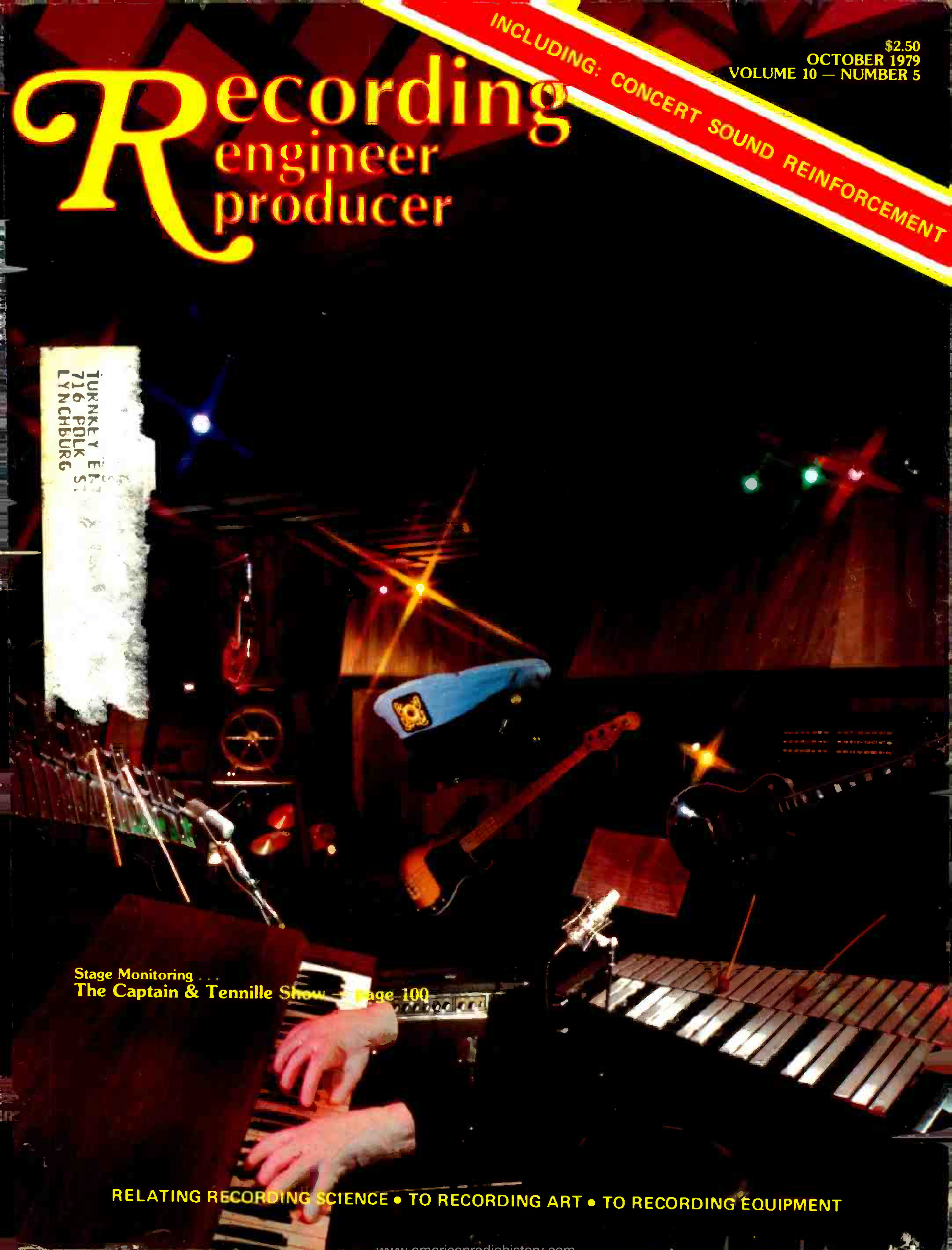
Recording engineer producer

$2.50
OCTOBER 1979
VOLUME 10 — NUMBER 5

INCLUDING: CONCERT SOUND REINFORCEMENT



Stage Monitoring . . .
The Captain & Tennille Show — page 100

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What do you know about the Amek 3000?

It's the top of the line from Amek. 36 in, 24 out. Full parametric equalization. VCA control on all inputs, grouping, echo returns, and mute matrix. And *exceptionally* versatile routing.

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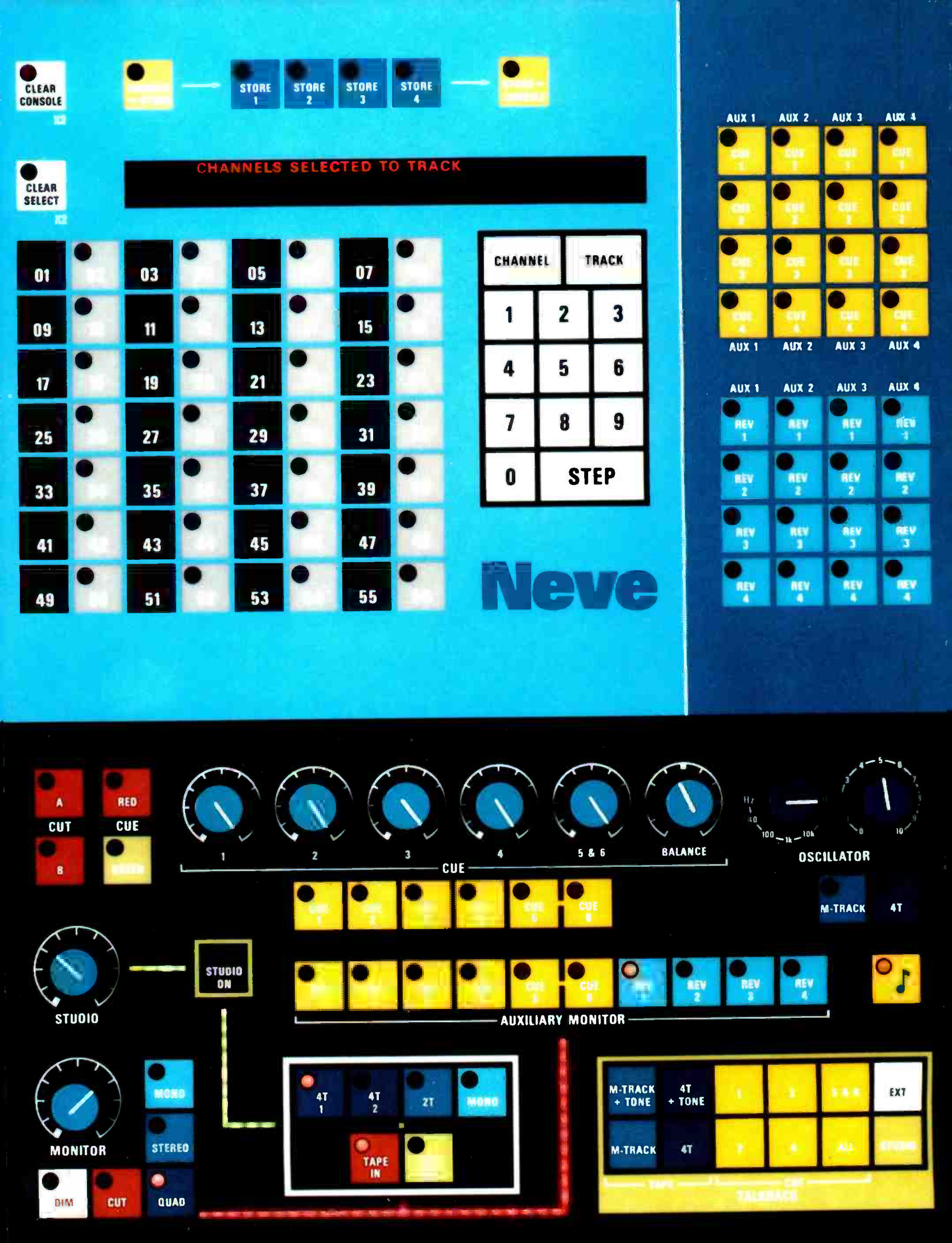
CLEAR CONSOLE
STORE 1
STORE 2
STORE 3
STORE 4
CLEAR SELECT
CHANNELS SELECTED TO TRACK
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BALANCE
CUE
OSCILLATOR
M-TRACK
4T
STUDIO
STUDIO ON
REV 2
REV 3
REV 4
AUXILIARY MONITOR
MONITOR
MONO
STEREO
DIM
CUT
QUAD
4T 1
4T 2
2T
MONO
TAPE IN
M-TRACK + TONE
4T + TONE
EXT
M-TRACK
4T

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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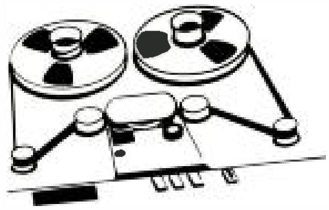


OCTOBER 1979
VOLUME 10 — NUMBER 5

Contents

The Cover:
"In The Captain's Absence" — photo by Art Rex. Taken at The Captain's Rumbo Recorders while the couple were away appearing in Las Vegas at the MGM Grand. Studio designed by Rudi Breuer; Neve, Studer equipped.

Departments:

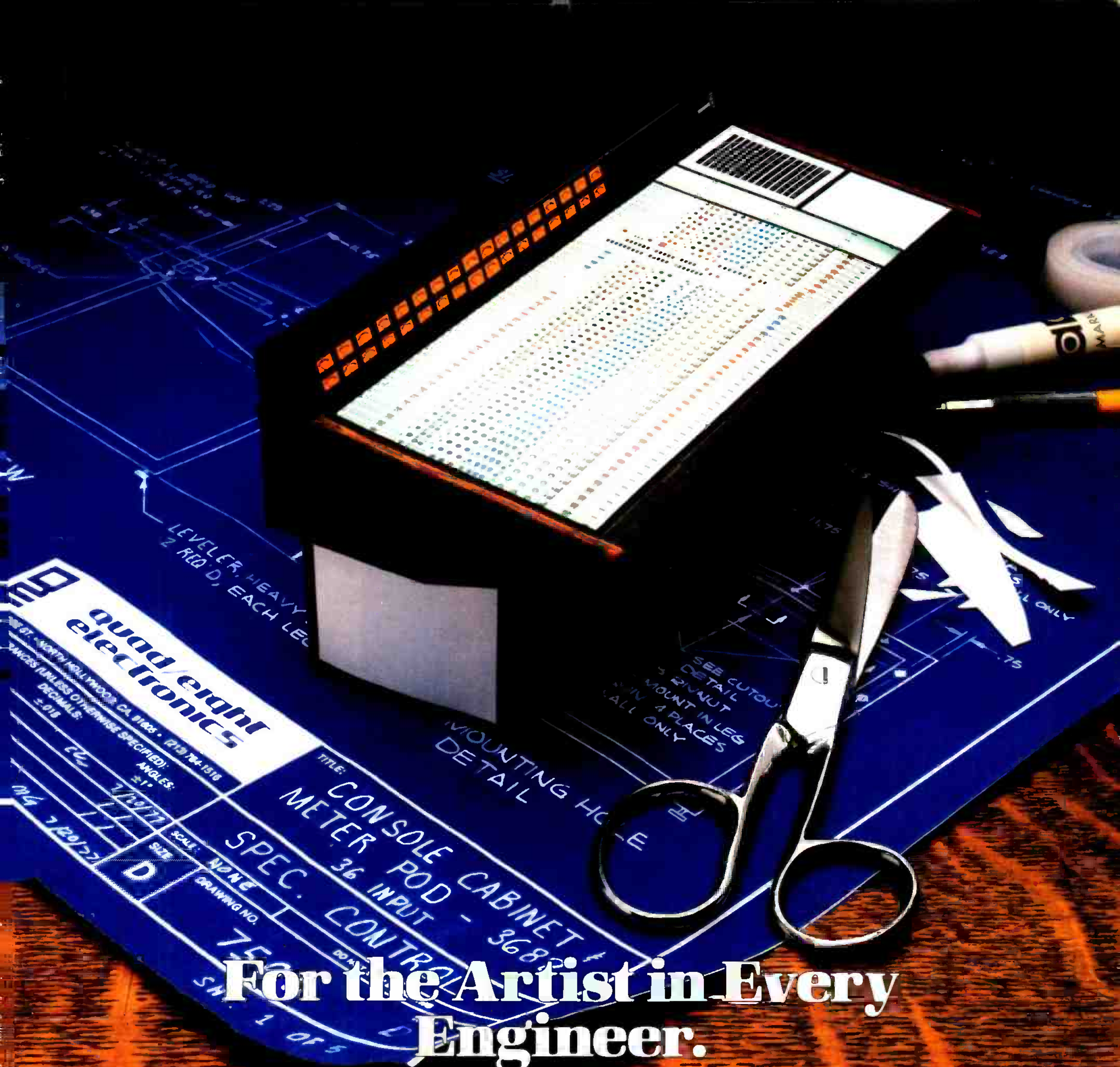
— LATE, LATE NEWS —

The Society of Professional Audio Recording Studios (SPARS) will hold its first annual national convention from Wednesday, October 31, through Sunday, November 4, headquartered at the Waldorf-Astoria. The SPARS activities include a meeting of the interium board of directors (Oct. 31), a general membership meeting (Nov. 1), and a full schedule of technical and non-technical seminars. The seminars will be presented in the Crystal Room of the Doral Inn (across the street from the Waldorf-Astoria). Topics to be discussed include, "The Clients' View of Recording Studios" (Nov. 2, morning), "Economic, Legal, and Financial Observations of Recording Studio Administration" (Nov. 3, morning), and "A Comprehensive Presentation on Multitrack Tape Machines for the 1980s" (Nov. 2 and 3, afternoon).

The multitrack program will feature eight major manufacturers who will have one hour each to address the following subtopics:

- Solid state logic control for transports and electronics.
- Interfaces so machines can be controlled by computers and interlocked with one another.
- Capstan motors and tension control by reel motors and tape guiding.
- Service-access to components.
- Electronics, signal-to-noise, distortion, headroom and how it is measured.
- Head stacks, i.e.: (frequency response), wear factors and adjustment, headroom for new tapes.

Only employees of SPARS members may attend these presentations. For further information regarding SPARS, contact the SPARS hospitality suite at the Waldorf. The room number will be at the hotel desk, (212) 355-3000. □□□



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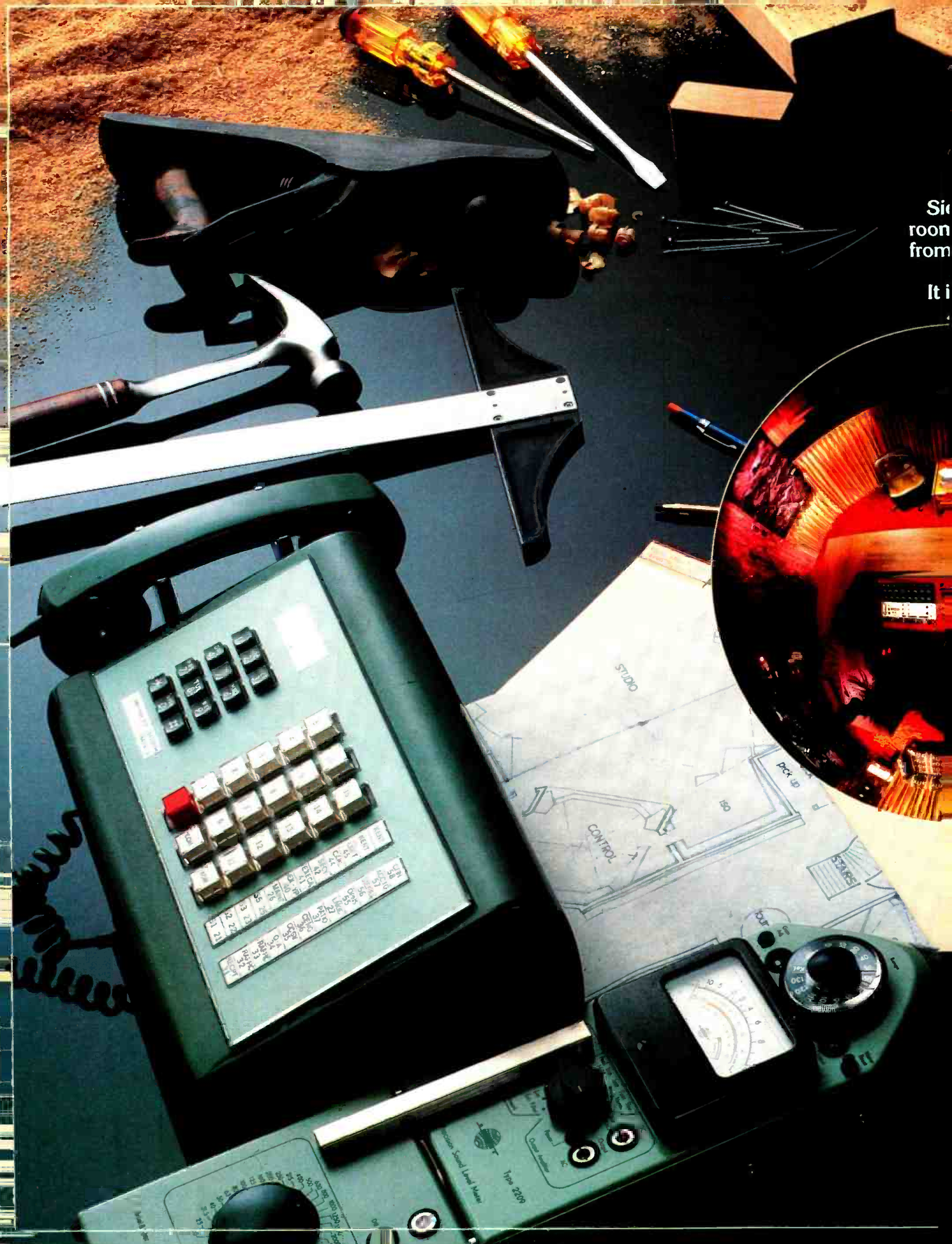
for additional information circle no. 4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FANTASY RECORDS — Berkeley (2 studios)
MR. "D's," KENDUN RECORDERS — Burbank
SOUNDMIXERS — New York City (Redesign and Rebuild)
RECORD PLANT — Los Angeles (studio C)
SOUNDS INTERCHANGE — Toronto
STUDIO SIX — Montreal
ARMSTRONG AUDIO/VIDEO — Melbourne
MANGO STUDIOS — Sydney
CINEMA AUDIO — Manila
JEM RECORDS — Manila
PIERCE ARROW STUDIOS — Chicago
KITTY RECORDS — Tokyo (2 studios)
MASTER SOUND — Atlanta
CBS/SONY — Roppongi, Tokyo (2 studios)
ALFA RECORDS — Tokyo
BIAS RECORDING — Virginia (3 studios)
SOUND PALACE — New York City
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Letters

from: **John R. Saul**
President
MICMIX Audio Products
Dallas, TX

James Cunningham presented a fine overview of reverberation in the August issue. One comment he made which cannot be overemphasized is that "the buyer of any reverb should listen to it with as many different types of music as possible, preferably some short brass passages . . . sustained music like strings . . . and some *short percussive sounds* . . . " (emphasis added). We would add to this statement that the prospective buyer should, if at all possible, listen to several units simultaneously and compare them on an A - B basis (alternately switching back and forth between units while listening to the same program material). Reverberation units which incorporate a mixing capability for the direct signal should, of course, be listened to in full reverb mode only, so as not to mask the reverberant channel's performance with the dry sounds.

Short percussive sounds as a test of reverberation performance are emphasized above because they are the most revealing single test for any undesirable characteristics such as what has commonly been referred to as boing, twang, or flutter echoes (periodicity). This applies to any type of reverberation unit, not just spring systems. If a drum track is not readily available, a sharp snap of the fingers on a microphone input to the chamber can be very revealing.

As manufacturers of the Master-Room series of reverberation chambers, we are familiar with the performance capabilitites of various types of reverberation systems. We must also note as comment on statements in the article, that Master-Room units *do* provide a series of early echoes at the onset of reverberation, followed by an increasing echo density as the amplitude decays. This is the characteristic of a natural chamber and avoids the 'slap' effect mentioned when additional pre-delay is used.

We should also note that our stereo units incorporate a differential timing pattern for the two channels which provides the cross-correlation he discussed and eliminates the need for synthesizing a stereo effect by using sum and difference signals.

Again, Jim Cunningham did a very good job on a difficult task of summarizing the typical characteristics of various types of artificial reverberation and must be complimented. His oversights regarding the capabilities of Master-Room design can be understood considering the need to generalize, but should be noted as an exception to the general rule.

reply from:
James Cunningham

I would like to point out a few errors that crept into my overview article on reverberation in the August 1979 issue. One page 52 it should have been pointed out that the 38.7' referred to is the average distance travelled between boundary surfaces. On page 54 the minimum for acoustic chambers should read 200 cubic feet. Also, on page 54 the frequency response plots for the live chamber and the plate reverb were reversed. On page 56 the room mode expression should be

$$(4\pi V/C^3)f^2.$$

Lastly, the echo density expression should be

$$(4\pi C^3/V)t^2.$$

Correction:

In the *August 1979* issue of *R-e/p*, "*Construction of Live Echo Chambers,*" by Scott Putnam and Tom Lubin, the equation on page 74 was not correctly reproduced from its reference, (*Acoustic Design and Noise Control*, by M. Rettinger). The formula should have read:

$$(5^{1/2}-1)/2$$

— ACOUSTIC DESIGN —
(continued)

from: **Bob Todrank**
President
Valley Audio
Nashville, TN

I've been reading with interest the recent exchanges regarding Control Room and Acoustical Design. The latest letter from Mr. Brian Cornfield nicely summed up the opinions of many people with whom I've spoken. I must congratulate him on his convictions and courage to speak out. I must join Mr. Cornfield in his overall evaluation that the latest advertisements (*excuse me, articles*) by Mr. Kent Duncan, of Sierra Audio Corporation, have concentrated more on PR work than on the dissemination of substantial information. Even Mr. Hidley's apology, concerning his unfinished article to have appeared in the June 1979 issue, read like a hyped-up press release.

As to Mr. Duncan's reply to Mr. Cornfield in the August 1979 issue, I must immediately set the record straight. I have ***never*** been an employee of Westlake Audio. I've never received a dime in payment for anything from Westlake Audio. The confusion may have arisen over two past projects in which Mr. Hidley's former firm and my firm were both involved. In 1974, Mr. Hidley provided control room design drawings for a studio in Nashville. Valley Audio was hired directly by the client for all construction supervision and electronics installation. Valley Audio has also done the electronics for another Westlake control room design in Nashville. Again, we were hired and paid by the client — not by Mr. Hidley.

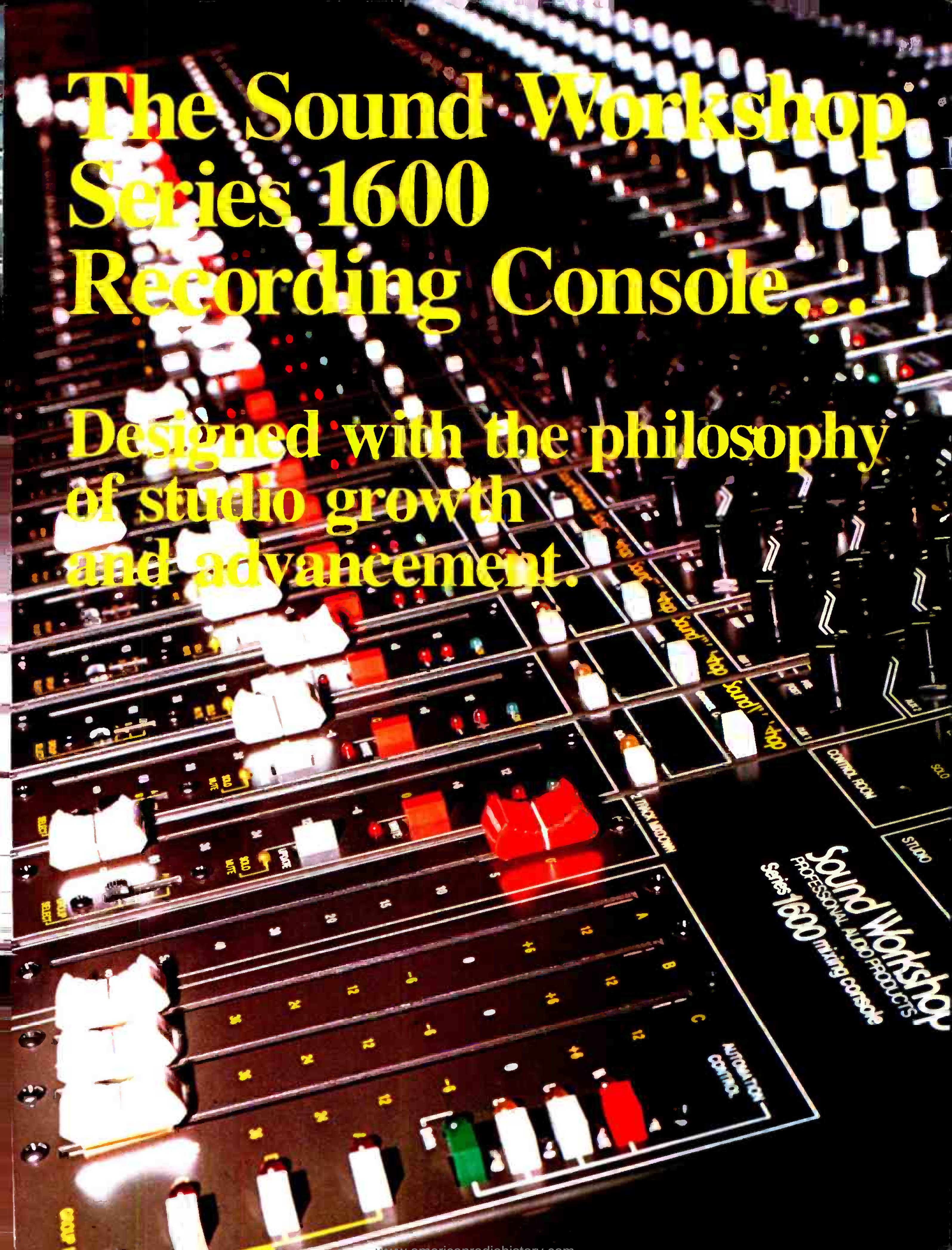
Since almost everyone has already done their best to confuse the public, I'll refrain from commenting on specific statements contained in any and all past articles and letters concerning the subject of studio and control room electronics and design.

A magazine such as *R-e/p* has a responsibility to publish a variety of articles, presenting differing points of view. It is exceedingly important for all concerned readers to survey this wealth of information with one eye on content and the eye on authorship. In the professional recording industry, we deal with a multitude of grays, no blacks or whites. There is hardly a right or wrong way to design a studio, only differing points of view, all dependent on the perspective of the design's originator.

Perhaps in the fervor of competition, we've lost sight of the true subjectiveness of our entire industry. We must remember that our final goal is the capturing and dissemination of an artistic expression. An analogy may be drawn between our line of work and the creation of a great painting. Rembrandt, Van Gogh, and Picasso all took radically different approaches to their work. Yet they all created artistic masterpieces. Similarly, the acoustical design of control rooms and studios may vary drastically. If a studio/control room creates a technically and artistically superior recording, then it must be said to have a valid design — no matter whose design philosophy has been followed.

Past and present articles and letters will serve a beneficial purpose, if we all emerge with the decision to study these controversies as part of a continuing educational experience. As a consultant in the field of music recording facilities, I feel it is my

. . . *continued on page 15* —


The Sound Workshop
Series 1600
Recording Console...
Designed with the philosophy
of studio growth
and advancement.
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PROFESSIONAL AUDIO PRODUCTS
Series 1600 mixing console
CONTROL ROOM
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In its first year

The **Sound Workshop Series 1600** has been chosen as the audio control center in over 50 studios around the world. Its modular design philosophy has allowed for a variety of installations ranging from 12 input/4 output production consoles, to 36 input/32 output multi-track recording boards complete with computerized mixdown.

Studio B—Boston, MA

As the most expensive single piece of equipment in the studio complex, the recording console needs not only to pass the highest quality audio with a tremendous amount of flexibility. It remains a major investment in the future growth of the studio.
The **Sound Workshop Series 1600** can adapt to the changing needs of the studio better than any other console available today. Thoughtful, innovative design allows a wide range of initial console options and configurations. But more importantly, these options can be added later with a minimum of extra expenditure and hassle.

Counterpart Creative Studios—Cincinnati, OH

The 19 Recording Studio—South Glastonbury, CT

The Sound Workshop Series 1600
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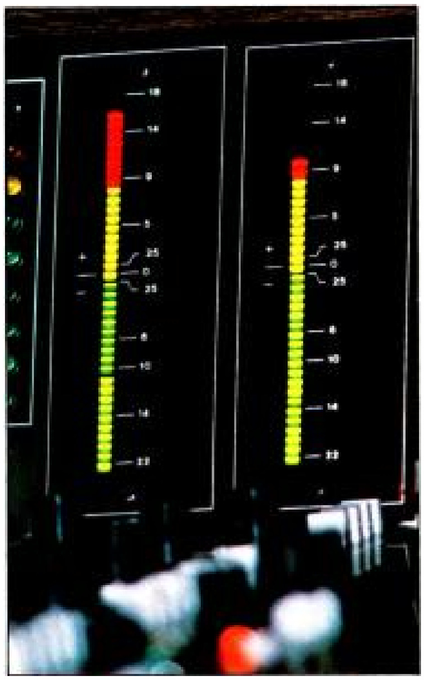
The diverse range of options available for the Series 1600 creates a console tailored to the individual needs of the studio: both today; and tomorrow, as need dictates and cash flow allows—

HIGH RESOLUTION METERING—a multi-colored wide range 40 segment meter with a dynamic range of 40dB and a resolution greater than .25dB at 0Vu. These meters offer 3 modes of operation including average, peak, and peak-hold. A built-in spectrum analyzer permits visual indication of the frequency distribution of any monitored signal. A spectrum analyzer can also be added to the standard 8 segment LED meters, which are available with Peak ballistics. Mechanical Vu meters with Peak indication are also available. Consoles may be configured with varied meter options.



EGC 101 VCA CELL—the state of the art in VCA technology (from Allison Research). Sound Workshop has incorporated the EGC 101 in its VCA grouping package. Studios adding the VCA package to their Series 1600 can take advantage of this sonic breakthrough. Studios with VCA equipped 1600s can retrofit, and offer their clients the current edge of VCA technology.

SUPER-GROUP—this new addition to Sound Workshop's **ARMS** Automation system sets new standards for grouping, flexibility, and ease of operation. No other console offers the visual status indication of group assignment available with Super-Group. "Negative Grouping" permits formally difficult or impossible group structures to be instantly available. In addition, the number of groups available is limited only by the number of input channels. Super-Group is now available for use with ARMS Automation, which features Independent Mute Write with FET-Mute switching, eliminating the sluggish punch-ins associated with ramped VCA muting systems.

Also available are **TRANS-AMP** (Valley People, Inc.) microphone preamplifiers, Sweepable and Parametric Equalization, and many other options. Most options can be added to existing consoles. It's part of the unique design philosophy exhibited in all Sound Workshop products.

The Sound Workshop Series 1600 is sold through a select group of Dealers. See one for assistance, or call Emil Handke at Sound Workshop.

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continued

responsibility to be aware of many points of view. A good consultant should be able to offer many alternatives, and explain the benefits and drawbacks of each.

If the readers can see through the PR work in this type of article, there is usually some interesting information to be gained. But I caution the buying public to be careful who they elect to the post of "Resident Expert." Once elected, or even self-appointed, such experts are hard to impeach. We should ask ourselves whether we listen to those who yell the loudest, or those who talk the smartest.

Many clients and fellow professionals have asked me about the great differences of opinion, even among the "experts." Mr. Ron Wickersham, of Alembic, Inc., and I discussed this very topic some months ago. We concluded that the basic reasons most "experts" disagree is the color of the axe they have to grind . . . be the axe ego, public recognition, peer support, or PR support for a sales business. Everyone has a reason, honorable or not, for saying what they say. Thus, it is critically important for the reader to concern himself as much with the underlying reason for making a statement, as with the statement itself. If a person's reasons are in harmony with yours, the reader, and your philosophies are the same, you have a basis for real communication with the expectation of positive results.

---

from: **Wayne Wadhams**
President
Studio B
Boston, MA

I hope *R-e/p* is not going to become a forum for the kind of argument and vendetta shown in the series of letters among Brian Cornfield, Kent Duncan, Michael Rettinger, et al., published in your August issue. These are all very important names in the recording industry, but let me remind you that

1 - Scientists and researchers tend to get overheated about their ideas and discoveries, and well they should. Yet this kind of debate, in which all parties are misquoting each other, using undefined technical terms, and in fact questioning the existence of allegedly-accepted theories in acoustics, convinces me that none of these experts has their clients' interest at heart, only their personal reputation.

2 - Acoustics and all the recording "sciences" are only quasi-sciences. From my own background in atomic physics, I know the value of specualtion. You observe something, play a hunch and spend some money to research it, and if you're lucky, you can come up with a handy mathematical relationship to describe what you're seen. However, in physics, the derived relationship is the product. Not so in acoustics! Despite all the contributions of Beranek, Duncan, Newman, and all the others, the real meat of acoustics and recording is making spaces and machines that make tapes which please ears. Pleasure and the human ear are very subjective things, and although the spaces and machines being designed today do an ever-better job, let's not get too pompous about whose designs are 'right.' Technically speaking, none are right; they just appeal to different sets of ears.

3 - It is possible, without a great deal of theory or expensive test equipment, to design studios and control rooms that achieve excellent results, i.e., that enhance musical sounds, produce tapes which satisfy producers and musicians, and moreover, produce tapes which still satisfy musicians and producers when played on other systems and in other rooms. As a long-time reader, I am more interested in hearing about the *variety* of approaches people are using in making tapes, rather than how the 'stars' of the industry are battling each other into theoretical extinction trying to design the *perfect* studio. Let's have more practical advice to the small studio owner and engineer. Let's have dollar value reviews of professional equipment. It's all very expensive, you can't see it in action, and aside from word of mouth, nobody really *reviews* products. I know that advertising pays for a lot of your publication costs, but if I am to continue reading, *R-e/p* has to help me improve my product and business in a very direct way. Otherwise the magazine will just become a heady and self-serving irrelevance.

Editor: In fact, the greatest majority of reader response has been very favorable to a continuation of R-e/p's presentation of debated concepts.

---

from: **Craig Anderton**
Clayton, CA

The Otari MX-5050-8SD 8-channel recorder is an excellent and versatile machine that has found a home in many semi-professional and professional recording studios. Among its convenience features is a variable speed control; however, there is an interlock circuit that defeats this control when in the record mode. In my opinion variable speed only realizes its full potential if you can vary the speed during recording as well as during playback, as this allows you to tune the track to out-of-tune instruments, change the timbre of voices, give a bigger sound to choral effects and string synthesizers, and so on.

Luckily, there is an extremely simple operation for defeating the internal interlock circuitry. I should stress, however, that very few (if any!) companies like you to mess around with the innards of their machines; so, while you're modifying your machine you should also be aware that you're modifying the terms of your warranty. With that consideration out of the way, let's proceeed to the actual modification itself.

1 - You will need to get at the back panel of the *transport* section of the recorder. Remove the 4 screws that hold the rear cover in place, then pull down the rear cover (as explained in the manual) to gain access to the inside of the transport box.

2 - Locate the fuse in the top center area of the transport. Centered behind the fuse is a circuit board, labelled #PM875A. This is the board we're going to be working on.

3 - Look towards the bottom of the component side of the board; you'll note a number of wires going into various holes, and each hole is assigned a number. Desolder the wire that connects to *hole 15* (on my machine, this was a purple/white wire).

4 - Tape the end of the wire so that it doesn't short out against any other components. Also, write a short note explaining that the taped wire has been removed from terminal 15 of circuit board PM875A, and tape this note to one of the inside walls of the transport box. In the event that you want to return your machine to "stock" at some future date, having the information right at hand will be helpful to you.

I should add that the variable speed control still works exactly the same as before: pull the pitch control out for variable speed enable, and push it back for variable speed disable. The only difference is that the variable speed control will now work in the record mode — and that your machine is considerably more versatile than it was before the modification.

News

LISTENER SUBJECTIVITY AFFIRMED AT "GREAT AMPLIFIER SHOOTOUT"

Responding to a great number of requests from buyers of their UREI Model 813 Time Aligned™ monitor sytems to recommend

Photo: C. J. Flynn, Westlake Audio

. . . continued on page 186 —

STUDIO UPDATE

Northeast:

☐ **SOUNDWAVE RECORDING STUDIOS** (New York City) announces the completion of a special project for **PHONOGRAM/MERCURY,** implementing a special process of disk cutting, resulting in two reverse groove LPs for in-store promotion. The albums, unlike standard LPs, start tracking from the inside of the disk to the outside where the groove is specially cut to prevent the stylus from going off the edge and being damaged. Standard equipment can handle the 12-inch pressings. The albums are entitled **COUNTERREVOLUTIONS IN ROCK** and **COUNTERREVOLUTIONS IN R&B,** and contain selections from recent Mercury and distributed label releases. *50 West 57th Street, New York, NY 10019. (212) 582-6320.*

☐ **SIGMA SOUND STUDIOS** (New York City) has **THE VILLAGE PEOPLE** in recording music for their upcoming **CASABLANCA** album and for their film, **DISCOLAND.** Both projects are being produced by **JACQUES MORALI.** *53rd Street and Broadway, New York, NY.*

☐ **RPM SOUND STUDIOS'** (New York City) recording activity has included the new project by the **NEW CHRISTY MINSTRELS** with **RON JOHNSEN** engineering, **BRIAN ENO'S** new album being engineered by **NEAL TEEMAN** and the new LP by **COME ON** with vocals by **KLAUS NOMI** with Johnsen again engineering. **HUGH DWYER** acted as assistant engineer on all three projects. *12 East 12th Street, New York, NY 10003. (212) 242-2100.*

☐ **BLUE ROCK STUDIO** (New York City) has been recording **BRIAN ENO** producing his own album, **RICHARD T. BEAR** producing German singer **INGA RUMPF'S** first U.S. album for **RCA,** and **JERRY LOVE** and **MICHAEL ZAGER** doing track for the upcoming **SPINNERS** LP. **EDDIE KORVIN** engineered all the sessions. **JOSEPH PAPP** is also producing **ELIZABETH SWADO'S** "The Runaways" at Blue Rock with **NIGEL NOBLE** engineering. *29 Greene Street, New York, NY 10013. (212) 925-2155.*

☐ **SELECT SOUND STUDIO** (Buffalo) has increased tracks with the addition of an MCI 24-track tape machine, a Syncon 28 x 28 console, and an MCI ½-track mixdown machine. Retained from the old studio are Neumann, Sennheiser, and AKG mikes, AKG reverberation, and Lexicon delay lines. Instruments include a 75-year-old Marshall and Wendall grand piano and a full compliment of Moog synthesizers. *1585 Kenmore Avenue, Kenmore, NY 14217. (716) 873-2717.*

☐ **SOUND PLUS TWO RECORDING FACILITY** (Philadelphia) presently an 8-track studio, is planning expansion to a 16-track format utilizing a Tascam 90-16 and adding UREI Time Aligned monitors, a PSA-2 power amp, and a compliment of studio instruments. Current outboard equipment includes a Cooper Time Cube, UREI limiters, and Orban Parametric Equalization. *1564 Temple Drive, Ambler, PA 19002. (215) 646-2026 or (215) 342-8093.*

Nineteen Recording

☐ **THE NINETEEN RECORDING STUDIO** (South Glastonbury, Connecticut) has recently added an MCI JH-114 24-track machine to its 16-track facilities and plans to link the two machines along with a disk drive system to provide 38 dbx tracks with automated mixdown. These upgrades are in addition to a Sound Workshop 1600, a variety of outboard, the completion of a string room, and Nineteen's 8/16-track remote truck. *19 Water Street, South Glastonbury, CT 06073. (203) 633-8634.*

☐ **QUEEN VILLAGE RECORDING STUDIOS** (Philadelphia) which recently installed a 40-track Neve recording console to their 24-track facility, has named **JOE CAMPELLONE** general manager for the studio. Campellone handled sales, marketing, and merchandising for **DOMINION MUSIC,** and was in charge of that company's east coast operation before joining Queen Village. Other personnel changes include the promotion of **WALLY HAYMAN** to studio manager overseeing all studio operations and acting as liaison for advertising agency business, and the addition to the staff of engineer **BILL OLSZEWSKI,** late of **MOTOWN, GOLDEN VOICE** and **HI RECORDS.** Queen Village was honored with seven **TRACK AWARDS** at this year's ceremony. *800 South Fourth Street, Philadelphia, PA 19147. (215) 463-2200.*

have you?
• increased track capacity — gone 24, 16, 8 •
• added key people • won awards •
• moved or expanded • added important equipment •
these are some of the interesting news items that can be announced in the next available issue. Write:
R-e/p STUDIO UPDATE
P. O. Box 2449 • Hollywood, CA 90028

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STUDIO UPDATE

☐ **SIGMA SOUND STUDIOS'** (Philadelphia) 12th Street facility was the taping site for a concert by **THE A'S** for an October BBC Television airing on the network's "Grey Whistle Test." Also at the studios, rhythm tracks are being recorded for **LINDA CLIFFORD'S** next LP, as well as the inception of new albums by **THE TRAMPS** and **JERRY BUTLER.** *212 North 12th Street and 309 Broad Street, Philadelphia, PA.*

☐ **MUSICOR RECORDING STUDIO** (Philadelphia) announced the addition of a Peterson Strobe Tuner, a Pioneer ten-band stereo graphic equalizer, and a new headphone cue system. The studio is also equipped with a Tascam 80-8 recorder with dbx noise reduction. Chief engineer is **ARNOLD TERRY** with **MICHAEL BROADNAX** as assistant engineer. *2539 West Columbia Avenue, Philadelphia, PA 19121. (215) 763-0741.*

☐ **SOUND SELLER PRODUCTIONS** (Pittsfield, Massachusetts) has added to its track capability with the installation of a Scully 16-track tape machine with additional dbx noise reduction. The installation now also features high speed cassette duplication. Present production includes guitarist **SID MARGOLIS,** formerly of the NBC and CBS orchestras. Sound Seller also announces the addition of **RIC CORIN** to its engineering staff. Sound Seller Productions is a division of AdCom, Incorporated. *P. O. Box 1303, Birch Grove Road, Pittsfield, MA 01201. (413) 499-3899.*

Long View Farm

☐ **LONG VIEW FARMS** (North Brookfield, Massachusetts) is recording the new **J. GEILS BAND** LP, the groups second release for **EMI/AMERICA.** *North Brookfield, MA 01535.*

☐ **DUNE RECORDING** (Eastham, Massachusetts) has announced the appointment of **CHRIS BLOOD** as chief engineer. Blood, a former computer technician and sound man for the rock group, **ANDROMEDIA,** will supervise the 16-track studio, which is equipped with a custom console, several synthesizers, and a Digital Group Z-80 based computer. *Between Provincetown and Hyannis, Eastham, MA (616) 255-4443*

Alpha Audio

Southeast:

☐ **ALPHA AUDIO** (Richmond, Virginia) announces the completion of the second phase of renovations of its Studio 1 with the newest construction designed for a 'live' sound for orchestra recording, choral work and film scoring. The orchestra shell occupies roughly one-third of the 2,200 square foot music studio. The structure provides built-in cue feeds and electrical outlets and employs an anechoic wedge foam, "Sonex," to eliminate flutter echoes and un-wanted stray reflections. Composer-conductor **PHIL COXON** was the first to use the completed facility recording a 19-piece orchestra for **MOBIL CHEMICAL.** *2049 East Broad Street, Richmond, VA 23220. (804) 358-3852.*

☐ **CRITERIA RECORDING STUDIOS** (Miami, Florida) was the recording site for **KENNY LOGGINS'** upcoming album, "Keep The Fire," his third LP for Columbia. **TOM DOWD** produced the project with Criteria's **STEVE GURSKY** engineering. Guests on the LP include **MICHAEL JACKSON** and **MICHAEL MC DONALD.** The **BEE GEES** were also in studio putting the finishing touches on the soundtrack for their upcoming TV special. *1755 North East 149th Street, Miami, FL 33181. (305) 947-5611.*

☐ At **THE MUSIC FACTORY** (Miami, Florida) producer/engineer **BOB ARCHIBALD** is putting the final touches on **SHARON ROBBIE'S** upcoming LP. Also in the works is an album with the **CORNELIUS BROTHERS** and **SISTER ROSE.** *567 North West 27th Street, Miami, FL 33127. (305) 576-2600.*

☐ **FANTASY SOUND STUDIOS** (Granite Falls, North Carolina) announces the completion of a new 8-track facility installed by **RELIABLE MUSIC** of Charlotte, North Carolina. The studio features an 80-8 with dbx noise reduction, a Tascam 24 x 8 mixing console, and JBL monitors with Crown amplification. Sideboard equipment includes dbx compressor/limiters, MXR digital delay, and Furman PEQ. Mikes include Neumann, Sennheiser and Electro-Voice. The studio has just finished a project with **JOHN ANTHONY,** formerly with **THE BROOKLYN BRIDGE** and **THE NEW CHRISTY MINSTRELS.** *14 Woods Drive, Granite Falls, NC 28630. (704) 396-1188.*

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STUDIO UPDATE

□ **CENTURI RECORDING STUDIOS** (Coral Springs, Florida) was the recording site for the soundtrack of "Toymaker," a film written and produced by **QUAY HAYS,** of Los Angeles. Engineering and mixing on the project were **PETER YIANILOS** of **ARTISAN RECORDING,** along with **FRANK GIARDINO** and **RICHARD LAVOIE.** Centuri is currently a 16-track studio with plans to update soon to 24-track. *11460 West Sample Road, Coral Springs, FL 33065. (305) 753-7440.*

□ **LIVE OAK SOUND RECORDERS** (Chesapeake, Virginia) has opened a new studio featuring a 24-track MCI master recorder and an MCI mixing console. Other equipment includes an Ampex 2-track recorder, a Tascam 4-track machine, dbx compressor/limiters, and an Ashley stereo parametric equalizer. AKG, Beyer, Shure and Electro-Voice microphones are employed. A Marshall Time Modulator is on order, with immediate plans to add VCAs to the console. An Allison computer will complete automation in the near future. The **CHRISTIAN BROADCASTING NETWORK** recently completed its fall campaign in the studio with **STEVE PEPPOS** engineering. *809 Live Oak Drive, Suite 14, Chesapeake, VA 23320. (804) 422-1646.*

South Central:

□ **BUTTERMILK RECORDS' BICKLEY STUDIOS** (Houston, Texas) has acquire a 16-track mobile mini-van for location broadcasting and recording. Artists having used the van include **TOD RUNDGREN, TANYA TUCKER, TALKING HEADS,** and **JOHNNY WINTER.** The equipment has Video-Sync capability. Buttermilk's regular installation is equipped with an MCI JH-114 16-track and an Interface console. Plans are to update to 24-track within a few months. Current projects include an album by **JOHN BELL** for **DECCA/LONDON** and the basic tracks for an LP by **CRAWLER** on **EPIC.** *1310 Tulane, Houston, TX 77008. (713) 864-0705.*

□ **POLLARO MULTI-MEDIA ADVERTISING AND PROMOTION** (Denison, Texas) has completely upgraded their control room with new equipment including a Tascam 90-16 recorder, Model 15 board, DeltaLab 06-2 Acousticomputer, Roland Jupiter 4, plus Sennheiser, Sony, Neumann and Electro-Voice microphones. PMAP is currently producing TV and radio spots for **SUBARU** dealers in 38 states. *Elkins Building, Suites 205, 206, 207, P. O. Box 668, Denison, TX 75020.*

□ **SOUNDSHOP STUDIOS** (Nashville, Tennessee) has just completed upgrading of its MCI 500 consoles, including the installation of Trans-Amps, and Allison VCAs, and a change to low-noise, low-distortion and high slew rate opamps. These conversions are coupled with automation in both studios. This upgrading program also extends to the outboard gear. *1514 South Street, Nashville, TN 37212. (615) 244-8872.*

□ **JACK CLEMENT RECORDING STUDIOS** (Nashville, Tennessee) is recording **MAC DAVIS'** first album for **CASABLANCA, LARRY BUTLER** producing and **BILLY SHERRILL** at the board. **ANDY WILLIAMS** is in the studio, with producer **DICK PIERCE** and Sherrill engineering tracking a series of pop-country songs, and **EMI** artist **SANDRA STEELE** was in Clement recording her first album with **RALPH MURPHY** producing and **HAROLD LEE** engineering. *3102 Belmont Boulevard, Nashville, TN 37212. (615) 383-1982.*


Pearl

Midwest:

□ **PEARL SOUND, LTD.** (Ann Arbor, Michigan) 8-track studio has been operational for six months. The studio features an Ampex AG-440 recorder, Quantum Audio Labs modified console, and dbx noise reduction. Outboard gear includes Lexicon Prime Time, URSA Major Digital Reverb, Orban Parametric EQ, and a Loft Flanger. *2075 Provincial Drive, Ann Arbor, MI 48104. (313) 971-2414.*

□ **COUNTERPART CREATIVE STUDIOS** (Cincinnati, Ohio) has reopened after extensive remodeling of the physical plant as well as the purchase of all-new electronics. The installation was designed by **SHAD O'SHEA,** president of Counterpart, and the control room was acoustically designed by **TOM IRBY** at **STUDIO SUPPLY.** Irby also installed the monitoring equipment, a Westlake Bi-Amp System. The main studio has dimensions of 50' x 31' with a 16' ceiling, and features a 32-track Sound Workshop console with an MCI computer-based automation mixdown. New tape machines include a 24-track MCI, and outboard gear includes dbx noise reduction, MXR digital delay, and an array of compressor/limiters. **GARRY JONES,** formerly with Appalachia Sound Studios, is the new chief engineer and studio manager. *3744 Applegate Avenue, Cincinnati, OH 45211. (513) 661-8810.*

□ **COUNTRYSIDE RECORDING STUDIO** (Crookston, Minnesota) recently completed the master tape for **THE INSPIRATIONS,** a 28-member gospel group. **GARY EMERSON** engineered for **SLADE RECORDS.** Equipment in the studio is by TAPCO, SAE and Tascam, with GTE Sylvania monitors and Eltec 414 mini-monitors. Keyboards include a Baldwin Electric Harpsichord, a Hammond B3 organ, and a Yamaha grand piano. *Rural Route 2, Crookston, MN 56716. (218) 281-6450.*

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STUDIO UPDATE

☐ **ACKERMAN & MC QUEEN ADVERTISING** (Oklahoma City, Oklahoma) has just completed a fully automated 24-track recording studio, which was designed by **MILAM AUDIO,** of Pekin, Illinois. The facility features an MCI JH-636 mixing console feeding an MCI 24-track tape machine with AutoLocator III. Other equipment includes two MCI JH-110 mastering machines, Dolby noise reduction, Klipschorn loudspeakers, and an array of Shure, Neumann, AKG and Sennheiser microphones. The announcement was made by **MARK KELLER,** vice president and associate creative director of A&McQ. *5708 Mosteller Drive, Oklahoma City, OK 73112. (405) 843-9451.*

Mountain:

☐ **SANBORN PRODUCTIONS** (Boulder, Colorado) provided its remote truck to **CURT GOWDY PRODUCTIONS** for live recording of the **PRO RODEO HALL OF FAME,** a CBS Sports Spectacular with music acts, including **LARRY MAHAN, RED STEAGAL,** and **TANYA TUCKER. Sanborn president CARL FROST** was at the console. The truck's mobile control room is 22' by 8' with silent air conditioning and a video link with the musicians. The format is two 24-track with dual 24-track machines and/or Dolby available upon request. A Sound Workshop 1600 series mixing console is used with a variety of outboard equipment. Monitors are JBLs and Auratones. The truck was designed by **GENE REYNOLDS, LARRY MARTIN** and Frost. It was recently used for live recording of the two **WAYLON JENNINGS** concerts in Omaha and Kearny, Nebraska. *The Boulder MusiComplex, 1865 33rd Street, Boulder, CO 80301. (303) 443-2372.*

☐ **NORTHSTAR STUDIOS** (Boulder, Colorado) has **DAN FOGELBERG** putting the finishing touches on his upcoming album, "Phoenix," for **FULL MOON/EPIC. MARTY LEWIS** is mixing. *P. O. Box D, Boulder, CO 80306. (303) 442-2001.*

Southern California:

☐ **CALIFORNIA RECORDING STUDIOS** (Los Angeles) has been recording **THE BRATS,** with **DICK MONDA** producing and **MICHAEL ZELLNER** engineering. Country singer **SONNY MARTIN** has been in the studio remixing his new live album with **JOHN BRADY** engineering. *5203 Sunset Boulevard, Hollywood, CA 90027. (213) 666-1244.*

☐ **DOCTOR SOUND RECORDING STUDIO** (San Diego) has just added a dbx 162 stereo limiter, a Furman parametric equalizer, and a Neumann U-87 mike to its 8-track facility which was recently the recording site for demos by **THE RADIATORS, MICHAEL EDWARDS,** and **SKYTRAIN.** Engineering was **RICK GORD.** *3191 Adams Avenue, San Diego, CA 92116. (714) 563-0164.*

☐ **OVERLAND RECORDING STUDIOS** (Irvine, California) supplied recording services for the two late **JOHN BILEZIKJIAN** albums, "Mirage" and "Saroyan Presents: An Oriental Bouquet." These are the second and third Bilezikjian projects to come out of Overland with **ALBERT LYON** engineering. *3176 Pullman Street, Suite 123, Costa Mesa, CA 92626. (714) 957-1544.*

☐ **THE PASHA MUSIC ORGANIZATION** (Los Angeles) has promoted **LARRY BROWN** to executive director of talent acquisition and studio operations, according to **SPENCER PROFFER,** president of the production company/recording studios. Brown did the basic acoustical design and planning for the Pasha Music House Studios, in Hollywood. Brown is currently producing **ARLAN GREENE'S** debut album, and is co-producing with **RONN PRICE BUCKEYE'S** second album for **POLYDOR.** Other engineering credits include **BILLY THORPE'S** "Children of the Sun" LP. Thorpe was recently in the studio doing promotional tapes for release in Australia. *5615 Melrose Avenue, Hollywood, CA 90038. (213) 466-3507.*

☐ **JOE GOTTFRIED'S SOUND CITY** (Van Nuys, California) recently took delivery of a custom Neve computer-assisted mixdown system to be fitted into one of the facility's Neve consoles. The Studio **"A"** console will provide mix memory on reverb returns as well as 24-track inputs. *15456 Cabrito Road, Van Nuys, CA 91406. (213) 873-2842.*

☐ **SALTY DOG RECORDING STUDIOS** (Van Nuys, California) is where **DOLLY PARTON** is working on a disco version of "Great Balls of Fire," for **RCA. DEAN PARKS** is producing with **ERIC PRESTIDGE** at the board. Also in the studio, **THE SANFORD TOWNSEND BAND** is mixing its new **WARNER BROTHERS** single, while **THE MARC TANNER BAND** with **NAT JEFFREY** producing is doing tracks and overdubs for its upcoming album on **ELEKTRA/ASYLUM. BOBBY THOMAS** is engineering on the latter two projects. *14511 Delano, Van Nuys, CA 91411. (213) 994-9973.*

☐ **GROUP IV RECORDING STUDIOS** (Los Angeles) has upgraded its installation in Hollywood with the addition of a Studer A800 24-track machine, an EECO MQS 100 Controller-Synchronizer, and Sierra/Tad Tri-Amp monitors in the control room. Allison/Fadex automation and a Sony 2850 ¾-inch video machine have also been added. **KENNY RANKIN** has been in the studio finishing his new album with **RON MALO** engineering. Several TV projects are scoring there as well. *1541 North Wilcox, Hollywood, CA 90028. (213) 466-6444.*


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STUDIO UPDATE

□ **MUSIC LAB STUDIOS** (Los Angeles) is recording **ALEX CIMA'S** second album featuring synthesizers, vocoders and special effects. Cima's first LP was released by **POLYDOR** in Germany. *P. O. Box 1594, Hollywood, CA 90028. (213) 662-8588.*

□ **EL DORADO RECORDING STUDIO** (Hollywood) opens its door for its first session in the new room to musician/producer **CARMINE APPICE.** He is currently working with his new wave group, "The Bank." The new room was recently completely by **NEW ERA WOOD WORKS,** who were carrying out the designs of **KEN FAUSE.** Pictured left to right is El Dorado's chief engineer **DAVE JERDEN,** studio manager **NADYA BELL,** and Carmine Appice. *1717 North Vine Street, Hollywood, CA 90028. (213) 466-6151.*

Eldorado

□ **HRS** (Granada Hills, California) announces the addition of **ROBERT BILES** as chief engineer of the studio. Biles background includes work as an engineer at Tewksbury Sound and Sound Genesis, and audio studios at the College of Recording Arts, in the San Francisco Bay Area. Working in the studio currently is **AL VIOLA** with **TOM LUBIN** engineering. New equipment added to the installation includes dbx noise reduction, an Ampex AG-440-B ½-track mastering machine, and a 3M-56 16-track recorder. *16052 Ludlow, Granada Hills, CA 91344. (213) 365-0709.*

□ **THE SOUND HOUSE** (Los Angeles). **DON PERRY ENTERPRISES'** 24-track studio is expanding with the construction of a new main room of 35' by 40' and an additional control room with an MCI automated console and a Stephens tape machine. The installation is designed by **WESTLAKE AUDIO** to handle larger orchestras, and the addition of a screen and projection booth will enable the studio to handle live film scoring sessions. The original control room and a smaller studio will remain in the facility, which is located at 1542 N. Cahuenga Boulevard, in Hollywood. Contact: *KTNT Productions, 13111 Ventura Boulevard, Studio City, CA 91604. (213) 995-3600.*

□ **RUSK SOUND STUDIOS** (Los Angeles) was the mixing site for **ELTON JOHN'S** new album on **ROCKET RECORDS,** with **TWIGGY** also in the studio recording an album with **JUERGEN KOPPERS** engineering and producing, assisted by **STEVEN SMITH** and **CAROLYN TAPP.** Other activity includes sessions with **THE VILLAGE PEOPLE** and **THE RITCHIE FAMILY.** Producing both groups were **JACQUES MORALI** and **HENRI BELOLO** with Koppers at the board assisted by Smith and **DAVID CLARK.** *1556 North La Brea Avenue, Hollywood, CA 90028. (213) 462-6477.*

□ **DAVLEN SOUND STUDIOS** (Universal City, California) recording activity includes **MELISSA MANCHESTER** in with producer **STEVE BUCKINGHAM** to record the studio's Bosendorfer piano. Producer **MICHAEL MASSER** working with **JANE OLIVER** and **DIANA ROSS,** and **CHRIS DESMOND** engineering and co-producing **AL STEWART'S** newest album for **KINETIC PRODUCTIONS.** *4162 Lankershim Boulevard, Universal City, CA 91602. (213) 980-8700.*

□ **MYSTIC SOUND STUDIOS** (Los Angeles) was the recording site of an afternoon demo session for the disco group, **DR. STRUT.** The session producer, **DAVE PELL,** sold the demo tape to **MOTOWN,** which used it as a master for the record's release. Mystic is a 16-track studio. *6277 Selma Avenue, Hollywood, CA 90028. (213) 464-9667.*

□ **KENDUN RECORDERS** (Burbank, California) activity includes **BRUCE BOTNICK** working with **JOHN GOLDEN** on **EDDIE MONEY'S** single from the soundtrack of "Americathon;" **JOHN MAYALL** finishing up his new project in Studio "D" with staff producer/engineer **JOHN STRONACH. GREY INGRAM** is mixing the music of the **DOOBIE BROTHERS** for a fall PBS broadcast; and **ED BARTON** mixing a new **JOHN DENVER AND THE MUPPETS** Christmas LP. *619 South Glenwood Place, Burbank, CA 91506. (213) 843-8096.*

□ **PAUL RATAJCZAK** announces the opening of **RATAJCZAK PRODUCTIONS STUDIO.** Used to produce new acts signed by Ratajczak Productions for such labels as **RCA,** the studio boasts the first west coast Amek M-3000 computer-controlled console as well as the first Lyrec 24-track tape recorder. The studio was designed by **EVERYTHING AUDIO,** of Los Angeles, and features a glass walk-through drum booth and vocal overdub booth. *601 Loray Avenue, Long Beach, California.*

□ **DAVID GATES,** of Los Angeles, has commissioned **EVERYTHING AUDIO,** of Los Angeles, to design and supply his new 24-track studio. The studio will be for personal use and features an Amek M-2000 console, an MCI 24-track as well as many other studio standards.

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STUDIO UPDATE

Northern California:

□ **FILMWAYS/HEIDER RECORDING** (San Francisco) has **THE BALL BAND** cutting tracks with **ANN FRY** engineering the project, **EDWIN HAWKINS** completing overdubs for his latest album with **ALLEN SUDDUTH** engineering and **CALVIN SETTLES** assisting. **MERLE HAGGARD** is recording and mixing his new LP with **ASHLEY BRIGDALE** at the console, and **THE TUBES** mixing a live recording from the **GREEK THEATER** in Los Angeles with **MIKE ABBOTT** and Ashley Brigdale engineering. *245 Hyde Street, San Francisco, CA 94102. (415) 771-5780.*

□ **AMERICAN ZOETROPE RECORDING'S** (San Franciscco) **RICHARD BEGGS** completed engineering, mixing, and music production for the film, "Apocalypse Now," at the company's Bay Area facility. *916 Kearny Street, San Francisco, CA 94133. (415) 788-8345.*

□ **RANCHO RIVERA RECORDING** (San Francisco) has completed upgrading to 16-track with equipment including an Ampex MM-1000 recorder, a custom modified Cetec/Electrodyne board, URSA Major digital reverb, UREI and Allison compressors, and a selection of mikes including Neumann, Sennheiser, Shure and AKG. The studio incorporates hardwood and stone, with a window to the outside. *1124 Rivera Street, San Francisco, CA 94116. (415) 661-6977.*

□ **TEWKSBURY SOUND RECORDERS** (Richmond, California) has taken delivery of a Teletronix LA-1 Limiter in addition to a new Eventide Digital Delay Line, and two AKG D-24 microphones. Recording at the studio are **JENNIFER MIRO'S** (formerly with the Nuns) new band with **CHRISTA CORVO** engineering, and **THE PSYCOTIC PINEAPPLE** laying background vocals with **RICHARD VAN DORN** at the board, and **THE CHARMERS** with **JOHN CUNIBERTI** producing and engineering. *6026 Bernhard, Richmond, CA 94805. (415) 232-7933.*

□ **FANE PRODUCTIONS** (Santa Cruz, California) recently installed a set of Tannoy Reference Monitors in their 16-track studios, along with a new URSA Major Digital Reverb System and a computer-controlled digital sequencer. Ex-Humble Pie and Small Faces leader, **STEVE MARRIOTT,** has recently finished sessions in the installation, and **LESLIE WEST** has been recording there of late with **FANE OPPERMAN** producing. **CORIE ANASTASION** is studio manager. *115-B Harvey West Boulevard, Santa Cruz, CA 95060. (408) 425-0512.*

□ **FREEWAY RECORDING STUDIOS** (Oakland, California) announces the addition of **BUD OSTERBERG** to their engineering staff. Prior to coming to Freeway, Osterberg was engineering freelance and teaching audio in the San Francisco Bay Area. *2248 East 14th Street, Oakland, CA. (415) 532-3700.*

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Solid State Logic

Master Studio Systems

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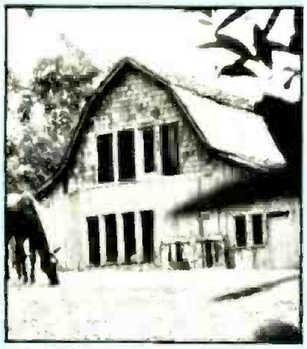
STUDIO UPDATE


Bear Creek

Northwest:

☐ **BEAR CREEK STUDIOS** (Woodinville, Washington) has updated to 24-track with Dolby and has added **BUZZ RICHMOND** to its staff as an engineer/producer. Richmond's credits as an engineer include **ELTON JOHN'S** release, "Mama Can't Buy You Love," and several platinum albums. *6313 Maltby Road, Woodinville, WA 98072. (206) 487-2533.*

Canada:

☐ **SOUNDSTAGE** (Toronto) studio manager **JIM FRANK** reports that the latest **NILS LOFGREN** album on the charts is one of the American and Canadian projects recently recorded or mixed at the studio. Veteran artists **SHIRLEY BASSEY** and **JOHN SEBASTIAN** have completed work at the Soundstage, while **PINK FLOYD** passed through for some overdubs. *39 Hazelton Avenue, Toronto, Ontario, Canada M5R 2E3. (416) 961-9688.*

☐ **STUDIO P.S.M.** (Quebec, P.Q., Canada) has taken delivery of a Trident 32-24-24 TSM console from **RADIO SERVICE, INC.,** of Montreal. Acoustic design at the renovated facility was by **SERGE MELANCON** of **ACOUSCIENCE,** of Montreal. The studio is a 24-track installation with Ampex tape machines. **JEAN-MARC PAYER** is the studio's engineer/director. *115 Saint Pierre, Quebec, P.Q., Canada G1K 4A6. (418) 691-1571.*

☐ **STUDIO TEMPO** (Montreal, P.Q., Canada) has purchased a Helios console featuring new EQs and automation on all 32 inputs, returns, and groups. The 24-track studio is equipped with Studer machines with new control room acoustics by **SERGE MELANCON.** Equipment was supplied by **RADIO SERVICES, INC.,** of Montreal. *0707 Charlevoix, Montreal, P.Q., Canada H3K 2Y1.*

☐ **STUDIO ST. CHARLES** (Longueuil, P.Q., Canada) installed during August a Trident 32-24-24 TSM to go with its Studer equipped studio. The console was supplied by **RADIO SERVICE, INC.,** Montreal. Acoustic design in the facility was by **SERGE MELANCON,** of **ACOUSCIENCE,** of Montreal. **PETE TESSIER** is chief engineer and director. *87 West Saint Charles, Longueuil, P.Q., Canada J4H 1C5. (514) 674-4927.*

Australia:

☐ **ATA STUDIOS** (Sydney) has just purchased the new MCI 600 automated console and is enlarging their control room and studio complex to include a second studio/mixdown suite in the near future, according to chief engineer, **DUNCAN MC GUIRE.** Other equipment at the studio includes an MCI 24-track tape machine with dbx noise reduction, Lexicon Prime Time, and an assortment of parametrics and compressor/limiters. *96 Glebe Road, Glebe, New South Wals, Australia.*

☐ **UNITED SOUND STUDIOS** (Sydney) has taken delivery of a Lyrec 24-track tape machine and has switched to Tannoys for monitoring, using two of the speakers for each side. Engineer/producer **SPENCER LEE** has just returned to United after mixing in England at **TRIDENT.** *21 Pier Street, Haymarket, Sydney, New South Wales, Australia.*

Guatemala:

☐ **DISCO DE CENTROAMERICA** has awared the design and equipment contract to **EVERYTHING AUDIO,** of Los Angeles, for their new studio and disk mastering facility in Guatemala City, Guatemala. Working with **JOHN MOORE,** the Everything Audio rep for Central America, **DIDECA** will be the first facility in their area to be involved in a total environment design project. The company's newly constructed buildings will house offices, studio, lab and disk mastering. Dideca is a major pressing and cutting facility and will look toward the U.S. for some of its clientele. *Apto. Postal 1792, Guatemala, Guatemala. Telephone: 537-137.*

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these are some of the interesting news items that can be announced in the next available issue. Write:

R-e/p STUDIO UPDATE

P. O. Box 2449 • Hollywood, CA 90028

MILAM AUDIO DUAL STUDIO DESIGN

Studio "A-2" Recording Connection, Cleveland, Ohio

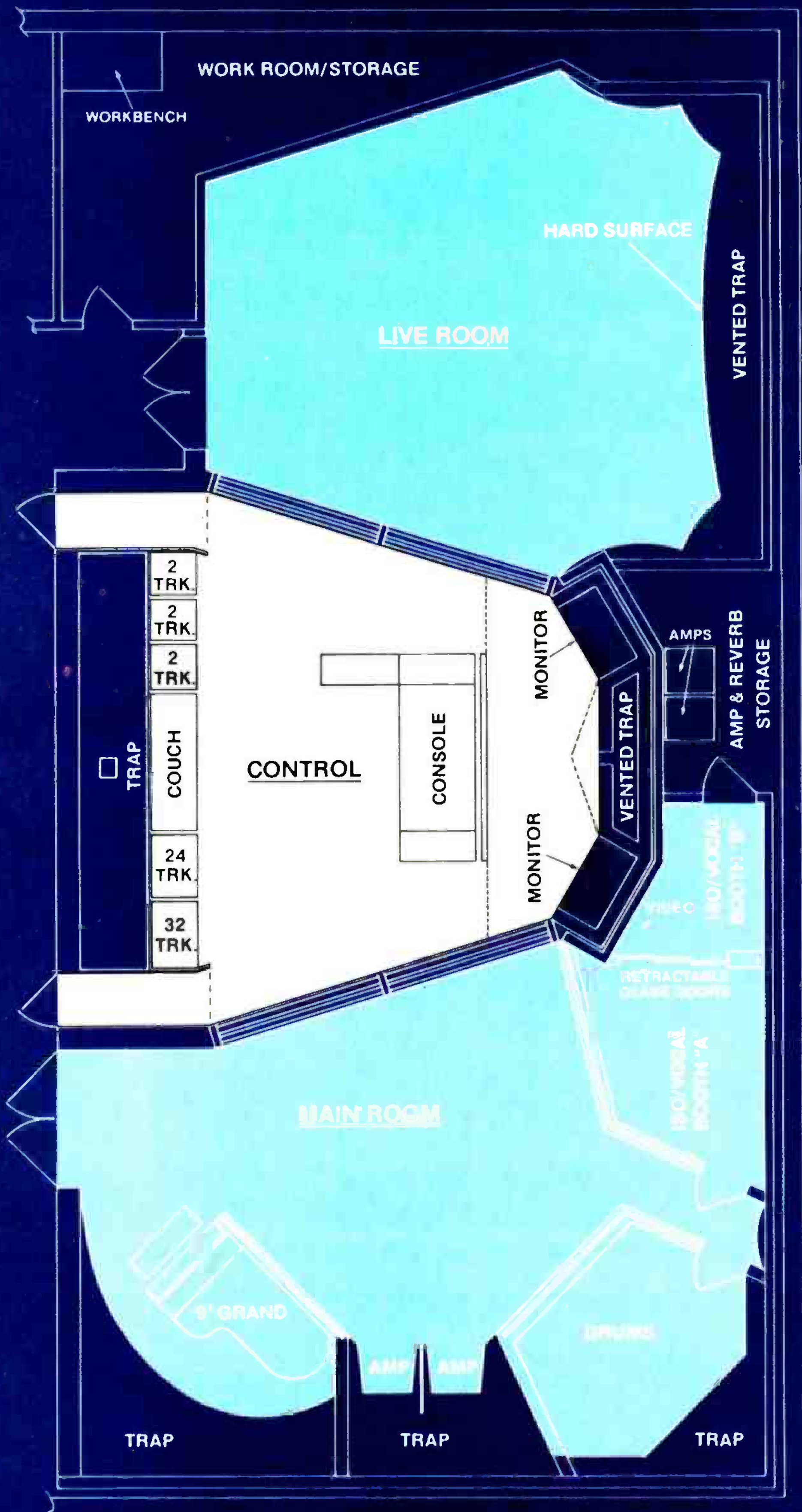

Photo by Joan Ruggles

SIGMA SOUND'S JOE TARSIA

by Tom Lubin

R-e/p (Tom Lubin): About 1971 "Backstabbers" became Gamble & Huff's first big hit on Philadelphia International. Were those the first records that you engineered for them?
Joe Tarsia: No, the first million-selling record I did with them was a record called "Cowboys To Girls" by the Intruders. That was in 1966 and was recorded at Cameo-Parkway Studios where I was engineering at the time. The "Backstabber" album was the first successful record they had signed to Columbia-Epic under the PIR banner. Previous to that album they had multiple successes with the Intruders, Soul Survivors, Archie Bell and the Drells, Wilson Pickett, Dusty Springfield, Joe Simon and Jerry Butler, among others.

R-e/p (Tom Lubin): When did you start at Cameo?
Joe Tarsia: I started there in 1962.

R-e/p (Tom Lubin): Was that the first studio that you worked at?
Joe Tarsia: Before that I had worked for a year in a studio I built for some people in South Philadelphia. I worked there for nothing. I learned a lot and then got involved with someone who was doing studio installation work in the Philadelphia area.

R-e/p (Tom Lubin): Who was that?
Joe Tarsia: His name is Norman Burke. He built Cameo studios and had built the Chancellor record studios. Chancellor at that time had Fabian and Frankie Avalon. There wasn't much of a recording community at that time, but what studios there were he built and maintained. I met Norman and did some service work for him.

— *continued overleaf*

Freddie started backup singing in his New Jersey junior high school. He earned a Bachelor of Music Degree from Howard University, and taught in Washington, D.C., while moonlighting as a producer. In 1969, his first Motown production, "I Want You Back" by the Jackson Five, went platinum. Since then, he has collected close to 30 gold or platinum records. Freddie now owns his own studio in L.A. and has recently produced disco hits for Yvonne Elliman, Tavares, David Naughton, Gloria Gaynor, and Peaches and Herb.

ON CREATIVE EXPRESSION

"I'm thinking charts. I'm thinking commercial. And I'm thinking hit, as opposed to creative expression. Because that's usually what I'm hired for. I mean, I hear the standard rap that I would get from a company person or a manager is that 'this group, live, is a knockout. I mean, they're killers. All they need is that hit record. When they get that hit record, man, you're gonna see the baddest group that ever existed in the history of recorded music.' So they want the charts. And that's why I approach it like that."

ON HEARING

"I only go by the ears, and I do hear very well. Musically and technically. I hear stuff all over the place. The guitar player—if he accidentally hits an open A string while he's fingering a chord, we could have thirty pieces on tape and I'll hear that and solo it out and bust him—say, 'Hey, could you keep that string quiet?' He says, 'You mean you actually heard that?' So my ears are really my fortune. That's where everything lies. Right in my ears."

ON RHYTHM SESSIONS

"I do my basic rundown on the rhythm date. The guys are really cookin' and the groove is there and everything. I come in and take a listen to what kinds of sounds I have. But if that sound is not there, then I don't record until the sound is right. There may be some other producers who would just go with the flow. 'If it's groovin', hey, you know, we'll save it in the mix.' But I've attempted to save things in the mix. It doesn't happen. It has to be on tape."

ON TAPE

"I do not know much about the characteristics, physically, of what tape is made of. I'm not too much into that—the chemistry involved. However, after spending six years at Motown—they had many, many rules and regulations. Now, one was that we always use Scotch Tape. When I ventured off into the world of independent producing, out of habit, and not wanting to change a good thing, I went right back to the same tape, which was 250. And I was then approached by other engineers telling me that if you switched, you could increase your performances here—you know, the bottom end, so forth and so on. And I did stray away and I did try cutting other projects on different types of tape. And the bottom line is that I came back to Scotch. I can't say that I noticed the difference of, you know, 3 dB and the low end with Scotch, and the other only gave me a dB-and-a-half. I can't say that. I only go with my ears, which tell me that my home is with Scotch Tape."

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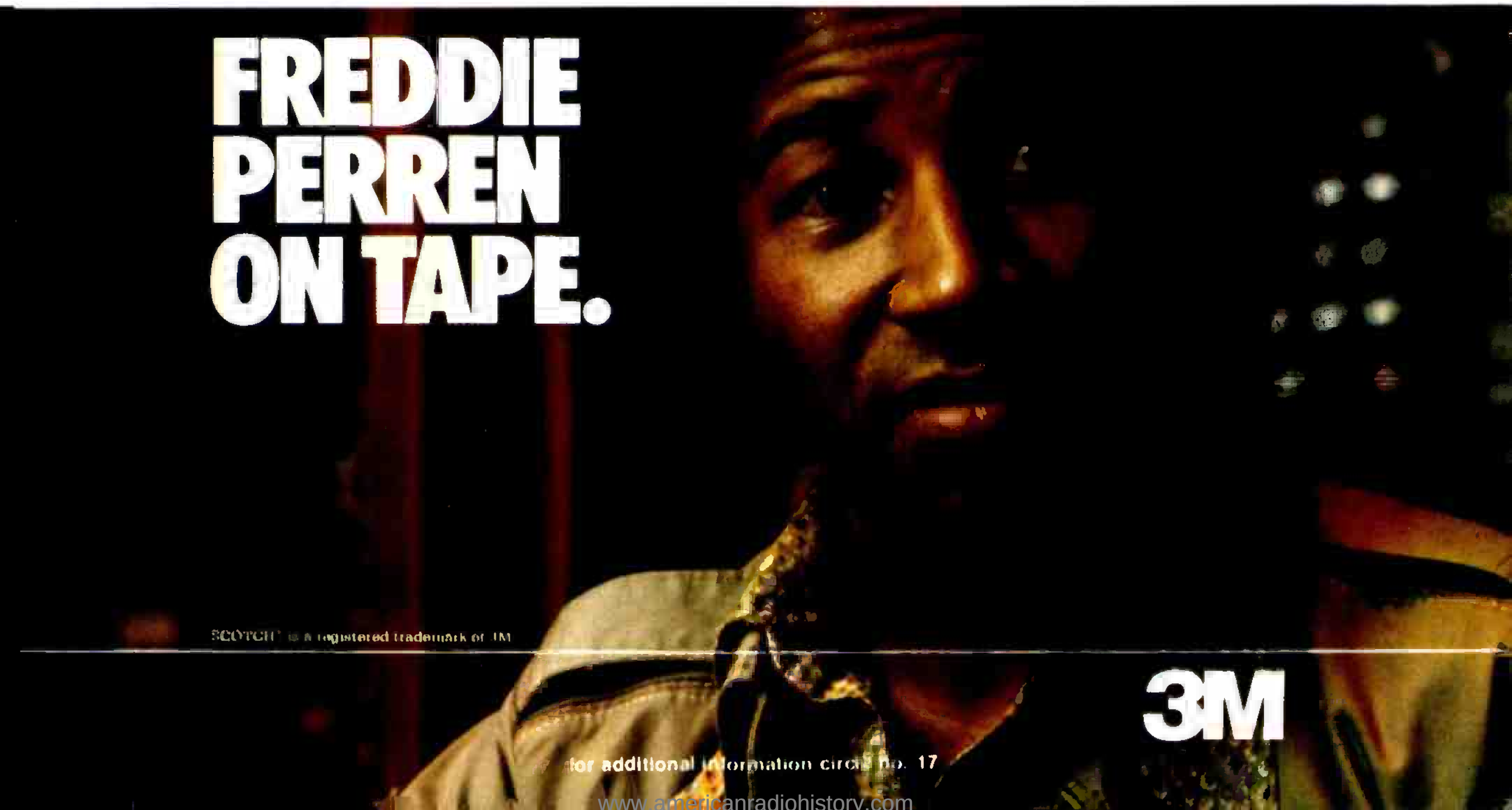

Since my background is in electronics, I started doing service work for Cameo records and had a chance to assist in the studio and began to get exposed to the record business. They threw me into doing some two-track sessions, since that's all we had in those days. I worked with Chubby Checker and Bobby Rydell, The Dovells, the Orlons, Dee Dee Sharp and the Tymes.

R-e/p (Tom Lubin): The "late fifties, early sixties" Philadelphia sound.

Joe Tarsia: Exactly, pre-Beatles pop, top forty-type records. Of course, when the Beatles came along they changed the whole industry in '63 and Cameo's fortune started to wane. We did have our own studios, but they were used exclusively for in-house work. About that time the original owner, Bernie Lowe, sold the company to two Texas people, Griffith and Bowen, and I saw the company start to change. Dick Clark was moving his national TV show from Philadelphia to the west coast, so a great vehicle for exposure was leaving town. As I watched Cameo's success start to decline, I found I had an extremely sensitive reaction to it. I couldn't play anything that I had done that I would be totally proud of. To this day when I listen to something I've done all I hear are mistakes. So, when Cameo's fortune started to decline I felt personally responsible. I felt that the sound could have been better and would have made a difference to their success, so I left the studio business. Norman Burke and I started a commercial audio sales and service company. I think it took about a month before I realized I made a very big mistake. Moving away from the business for a short time gave me a different perspective. I listened to other records and I began to be less critical of myself. Lo and behold, Cameo was in need of a chief engineer and they asked me if I wanted the job. They didn't have to ask twice. That was toward the end of 1965. I was out of the studio business about a year. It was then that Gamble & Huff started to form as a team and they had a couple of small productions that made money. They kept parlaying it and I started working with them. Previously I knew them both individually.

R-e/p: As players at the studio?

JT: No, as producer/writers. Kenny doesn't play an instrument. He was writing with Jerry Ross, a writer/producer. Leon was writing and producing with Madara and White, the writers of "At The Hop." Gamble & Huff got together in '66 and I happened to be there. Since then I've engineered almost everything that they've produced.

By 1967 I became concerned about the studio business in Philadelphia. Most of them were one-man operations, very small and were still two- or four-track rooms. I realized that if I was going to stay in the recording business, I would have to be in command of my own ship. The Philly studio owners at that time were not very progressive. So my only options were to own my own studio or move to a recording center like New York or Los Angeles. I was fortunate enough to find a bank that would lend me the necessary money to open a small studio. This meant that I went on the line for my house, my car and everything I owned. Sigma's first day of operation was August 5, 1968. At that time Gable & Huff had had a couple of successes and Tommy Bell was doing well with the Delphonics. It seemed to me there was a re-birth of the Philadelphia recording business.

I knew when Sigma opened I would get one shot at the various producers in town and I had to make the shot count. The studio had to be right. Fortunately, as things turned out, they came; they tried it; they liked it and they stayed, and that was it. They became very successful. They had six records in a row with Jerry Butler on Mercury: "Only The Strong Survive," "Moody Woman," "Western Union Man," and so forth. Bell continued to do well.

— *continued overleaf*

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LUBIN SIGMA SOUND'S

JOE TARSIA

R-e/p: And you were doing all of their records?
JT: Yes.

R-e/p: Did you do "Western Union Man?"
JT: I did part of it. It was cut at Cameo just before the opening of Sigma. In the beginning, Sigma was basically a one-man studio. I had a secretary which I shared with Frankford/Wayne. They did, and still do, all the Philly mastering. I worked one session a day on Wednesdays, Fridays and Saturdays and two on Mondays, Tuesdays and Thursdays. I tried to stay home Sundays. Most days I would come in at 10:00 in the morning for the day session and start the second sessions at 7:00 in the evening. We'd often work to 3:00 a.m. and I would be back in at 10:00 a.m. for the next day's session.

I did that for a couple of years and then the most traumatic experience I ever had as a studio owner occurred — letting somebody else sit down at my board and work with one of my clients. The hardest thing was being able to wean myself away and find competent people to help me run my business. Fortunately, I've developed an excellent staff. With just a couple of exceptions, everyone in my organization started in a menial capacity such as janitor or gopher and gradually worked themselves into the system and into the studio.

R-e/p: Do you still promote from the inside?
JT: Yes. We find our best people are the ones that grow up in the company. They learn our philosophies and our way of doing things. Some of our operating procedures are a little different, not because we set out to be different, but because we grew up outside of a major recording center and there were no sources of information. For instance, we used speakers instead of earphones when doing string overdubs. The string players are used to cue speakers now and prefer them to phones. It is a little oddball, but it works for us so we continue to do it. There are a few little idiosyncracies that we have evolved that are really different.

R-e/p: How big was that original room?
JT: Twenty-five feet by forty feet. It's still our largest studio.

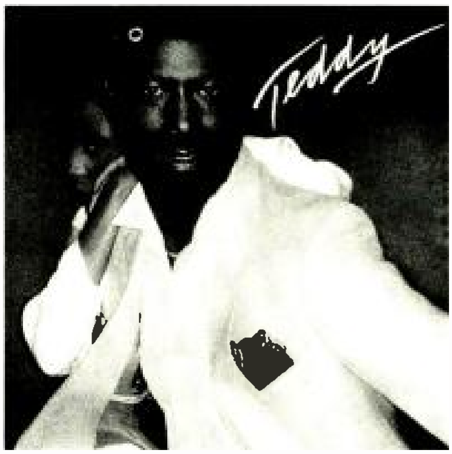
R-e/p: Does it look like it did in the beginning?
JT: Yes. But things are about to change. We have plans on the drawing board and have just purchased a building from NFL Films that was built by Warner Brothers in 1946. A nice, strong concrete building with high ceilings. We plan to go in there and build three new studios.

R-e/p: How many studios do you have now?
JT: We have five 24-track studios and hope to open a sixth in New York some time in November, that will be a complete 46-track mixing facility. The three rooms in Philadelphia also have 46-track capability.

R-e/p: Interlock machines?
JT: Yes.

R-e/p: Originally when Sigma opened what was it? Eight-track or sixteen?
JT: It was 8-track. We built the board ourselves, and two years later we built its replacement.

R-e/p: Who designed that first room?
JT: It was a studio before (Reco-Art). Emil Carson, who owned it, could do more with two mono machines than anyone I knew. I believe the sound of his recordings would stand up today. He was really a genius and an excellent engineer. The room had a great reputation. I have no idea why he left town, but it became available and I rented it. The acoustics of the room were pretty much intact. That gave me a little edge.



R-e/p: Did the mastering facility go in at that same time or were they already located there?
JT: Frankford/Wayne and Sigma moved in together. We split the rent and shared a secretary. Fortunately, hit records started to come and one thing lead to another. Within three years I had taken over the first as well as the second floors, and eventually I bought the building.

R-e/p: Since that original studio was on a second floor did you have any problems with acoustical flanking through the floor?
JT: Yes, we had a few problems. We made some structural changes which made the situation workable. The rooms work; there's no question about that. And let me say this . . . I'm not about to go out on a limb and cut it off behind me. We are planning to open one studio at a time in our new facility and then, depending on the market, we might possibly close down the old studios. We haven't really decided, but we won't dismantle anything in the old building until the new rooms prove themselves.

R-e/p: Has your operation there generally stimulated the Philadelphia area?
JT: Definitely. There was a time when Sigma was the only game in town. Now there are a number of good studios. Considering that it's not a recording center like Nashville, Los Angeles, or New York, Philly has its fair share. I would say there are ten-or-so 24-track rooms in the city. I couldn't attest to how much business they generate, but it is a healthy recording community with a lot of potential.

R-e/p: During those early days of Sigma do you recall your basic drum miking technique? How is it different today?
JT: In 1968 I think I was using an RCA BK5 on the bass drum, an Altec 633 saltshaker for snare drum, and one overhead mike that varied between a Neumann U47 and a 67. Of course, today, a lot more microphones are used. Since then, the drummers have changed, the room is different, and the microphone selection is more varied. Certainly the amount of flexibility is greater due to the 24-track machine.

R-e/p: Do you prefer to use more microphones as opposed to using less? What's your feeling on that?
JT: I think we have gone through an era of changing mike techniques. As the tape machine developed from 8-track to what it is now, and as the producers realized they had a greater ability to isolate the individual elements of a recording, the tendency was to use more and more microphones. With greater isolation, however, the result tended to lean towards sterile sounding recordings. I think that today the tendency in rock music is to go back to the ambience sounds of earlier recordings. We are going back to a bit more room sound.

I remember visiting sessions that were happening in large rooms with 18-foot ceilings and some 50-odd feet long, big rooms. When they soloed the snare drum mike, it would sound like the snare on any recording today, but that wasn't the sound you heard on the record. The sound of the snare drum you heard was coming from the string microphones across the room. It's that across-the-room sound that gives the big-as-a-house sound. I think mixers and producers are discovering this since the tight miked sound is gradually becoming a thing of the past.

We either have to develop ambience generators (that's my term for something other than an echo device that creates a room sound), or our studios have to have a legitimate room sound. With close miking techniques it doesn't really matter in what kind of room you record. If the microphone

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LUBIN

SIGMA SOUND'S JOE TARSIA

is very close to the instrument the room plays little or no part in the recorded sound.

But who hears a trumpet three inches from the bell, or the snare drum ½-inch from the head? They are not natural sounds. You have to let the sound of the instrument breath. This gives it a much fuller quality at the same given volume level.

R-e/p: When you mike a set of drums, how high up do you usually place your overheads?

JT: Well, the overheads vary, depending on what studio I'm in, but generally they're probably three feet above the highest cymbal. The nine or ten mikes that are used on a drum kit are used for accent. They are complimented by artificial ambience which is added during remix. What I do is feed some of the program through a Lexicon Prime Time into the studio monitors when I'm mixing and bring it back into the board through a pair of RCA 77s.

R-e/p: Do you use the Prime Time mostly for re-generation and a bit of delay?

JT: Well, since I do most of my mixing in a relatively small studio it's necessary to use delay to simulate the proper distance. By feeding the signal through the Prime Time I can make a 450-square foot studio sound similar to a 5,000-square foot ballroom. The re-generation is used to help disguise the discrete delays and simulate natural diffusion.

For a long while my approach to recording ambience was to set up spill microphones and that works great, except if the producer decides to punch out certain instruments in the mix, in which case the ambience tracks become unusable.

R-e/p: Right, because the room sound microphone would be picking up the unwanted instrument.

JT: Yes. Generation of artificial ambience is the way we solve the problem.

R-e/p: Microphone techniques in general, we talked specifically about the drums. The kind of recording that you do is pretty unique in that it's heavily orchestrated. How do you approach a full string section, or a brass section?

JT: Okay. The Gamble & Huff typical string section consists of six violins, two violas and one cello. Actually, that pattern has been changing but this was the compliment used for many of the older records. The violins and violas are miked in pairs using Neumann U87s. On cellos I've used a U47 FET.

R-e/p: How do you line them up, in two rows?

JT: Well, as opposed to the way string players would normally set up for performance with the heavier strings in the rear and the lighter strings in the front, I do just the reverse. The cellos and violas in the front and the violins in the back so that the front of the violin microphones are looking away from the cellos and violas and on some occasions the bowed bass. We record the strings by themselves in the live section of the room. The only thing we might cut with the strings is a harp. The brass and reeds are also recorded separately. I like to record the brass with ribbon microphones. For the reeds when I want a little bit more edge I usually use condensers.

R-e/p: You get a very crisp, clean piano sound.

JT: The player has a lot to do with the piano sound. It amazes me how different the piano can sound depending on the player. Leon Huff tends to play in the center, and hard. He is so intense that it's like he's got to let his energy explode through the keys. It's sheer emotion when he plays. The "great piano sound" is that emotion coming through on the tape. We also have really good, bright pianos. For the most part I use two 87s.

R-e/p: Above the hammers?

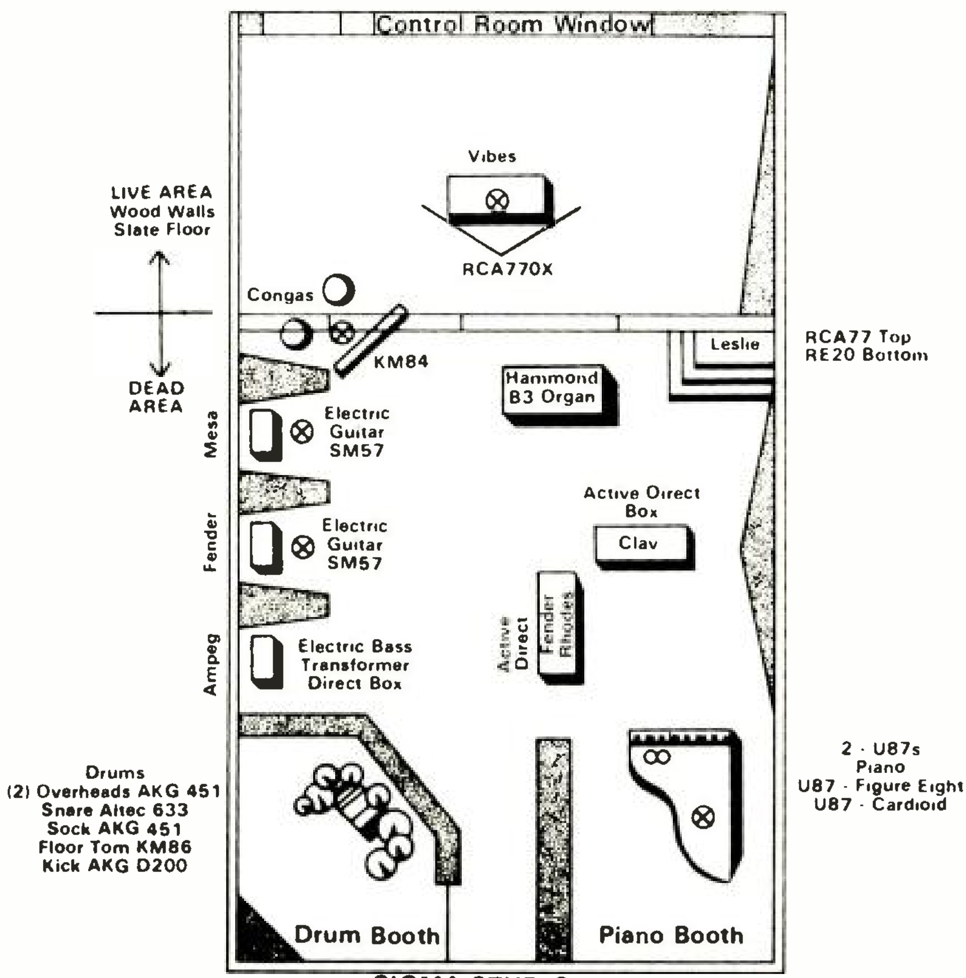
JT: It varies depending on the session. I've used one mike over the hammers, one along the low strings toward the back of the piano. I've placed them at a 90° angle over the hammers. A very full mono sound can be achieved by using a U87 in the figure eight pattern. About one third from the high end and perpendicular to the keys. It works very well. I would say, generally, on most of the recordings I use two 87s at 90 degrees. Recently I've gotten good reports from my people on the stereo AKG microphone and that's the next thing I'm going to try.

R-e/p: You work with great vocalists: Lou Rawls, Jerry Butler, the O'Jays, Teddy Pendergrass and Harold Melvin.

JT: Yes, it's fantastic working with such talents. I think though that they would all agree that even the best vocalist needs a vehicle of a great song to have a hit.

R-e/p: Do you think the significance of the recording studio has been somewhat blown out of proportion? Does it really come down to the song?

JT: A great song with a great arrangement can be successful with just an adequate



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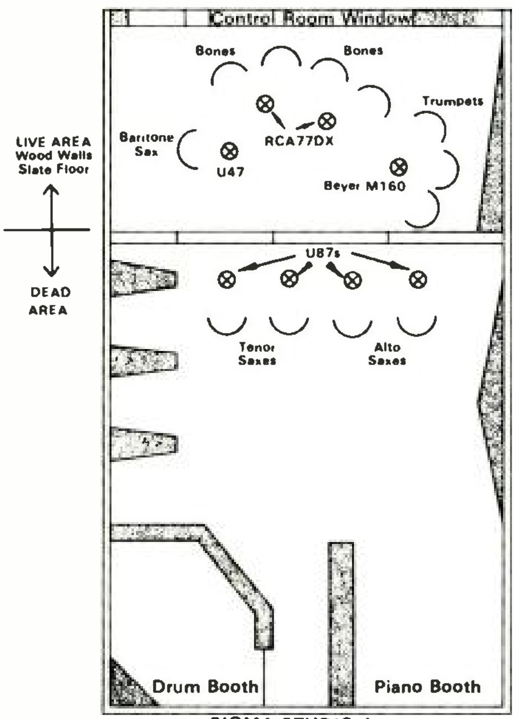
artist and recorded sound. I don't think you'll find too many cases of a mediocre tune being a big hit because of the sound. On the other hand, may I suggest, that the modern recording studio's role has changed since the day I started. On today's recordings we no longer try in all cases to faithfully record the sound produced in the studio but rather we bend and shape them into new sounds. Thus the studio not only functions in the mechanics of recording, but also becomes part of the creative production. If I were a producer I would want as many things going for my production as possible. And one of those things would be to record in the best studio I could find. There are too many intangibles in music to give anything away.



SIGMA STUDIO 4

R-e/p: You've worked with the same two producers for many years. Do they pretty much leave you alone at the board?
JT: Yes . . . that comes from being together for a long time. For the most part I mix without a producer being there. They give me that freedom. I can go in and experiment with an effect and no one gets impatient.

R-e/p: After doing the tracks and the overdubs and this and that, how do you then sit down and start a mix?
JT: I start mixing when I start recording. On every playback I try to evaluate what I've recorded. I might try a little EQ on the bass drum or I may add some flanging to a piano. While I'm doing vocals I'll fiddle around with an echo effect here or some digital delay there, so that when it comes to mix I have a pretty good idea of exactly what I'm going to do. Of course, that's all subject to the approval of the producers. But as I said, I am fortunate to have built a rapport with Gamble & Huff, where they for the most part give me a free hand.

R-e/p: Well, your taste and theirs coincide.
JT: Probably because I appreciate what they do. I have learned what they expect to hear and that's the direction I go. I enjoy the music and the people I work with and therefore it just flows.

R-e/p: Do you work very fast? With the kind of sessions that you do with a lot of studio players, there must be very little chance to be leisurely.
JT: The most hectic part of recording for me is dubbing, because strings and horns are done in the same day.

R-e/p: How many tunes?
JT: It depends, but generally three or four tunes with two or three arrangers in an eight-hour day. We set up for strings in the morning and do four tunes; the room is torn down and re-set up for horns and the same tunes are worked on again. The last thing we do is specifics like solo instruments. That tends to be a full day for us.

R-e/p: Are most of the horn and string overdubs doubled or tripled?
JT: We never double horns. Most of the time we double the strings. We never really like the effect of doubled horns. There's a certain unpleasant phasing characteristic when the same horn plays the same note that doesn't really work, at least not for me.

R-e/p: How many pieces in the horns?
JT: Depending on the tune, typically three trumpets, four bones, occasionally five reeds or any combination thereof. A full house for us would be maybe five reeds, two French horns, three trumpets and four trombones.

R-e/p: How do you usually set them up?
JT: In separate arcs, and I strive for the touch of ambience. I mike three trumpets with one microphone, four bones with two microphones. The reeds I mike individually.

R-e/p: How do you deal with the French horns? French horns are a very difficult instrument to mike as they tend to bleed into everything else.
JT: I've been using the AKG 414.

R-e/p: The title tune on Teddy Pendergrasses' " Life Is A Song Worth Singing" album sounds like it was originally cut for a Johnny Mathis record some years ago.
JT: You're right. Tom Bell produced that record for Johnny Mathis a couple of years back and produced it for Teddy in Teddy's first solo album.

R-e/p: Are they the same tracks?
JT: No, they were re-cut.

R-e/p: Did you do them?
JT: Well, I did part of it. Remember one thing about all the albums that you're speaking of right now. Gamble & Huff produce ten or twelve artists in a year, and if you figure they write, produce and record most of this material . . . it's an awesome job. They have staff producers and there are other mixers. Fortunately, it's all at Sigma, but other producers and arrangers do co-produce on the various albums. Sometimes you'll find a different texture or a different approach on part of an album because they have been produced by different people. It's listed on the back.

R-e/p: They put out an enormous amount of product and have for many years.
JT: I think their longevity is phenomenal. They have the ability to go in their office two weeks before an artist comes to town and sit down at the piano and write the material and have it ready to record. It's just a talent that comes to a very few people.

R-e/p: Are you usually booked way in advance with them? Or, have you gotten to where you simply prefer to work with them exclusively.
JT: I do work with them exclusively (mainly because with running a business with over fifty employees, I just don't have time to work with anyone else). But we should remember that Sigma is in operation with five 24-track rooms and we are just about to open our sixth. We have a varied roster of clients who are extremely important to us and who have made us what we are. And I would be remiss if I didn't acknowledge a few who are our backbone. Such as Tom Moulton, Norman Harris, Bobby Eli, Michael Henderson, John Davis, Vince Montana, Bunny Sigler, Jimmy Simpson, Jacques Morali, Ashford and Simpson, Mitume and Reggie Lucas and Jim Burgess, just to name a few. I am keenly aware of the directions of music and the importance of rock and roll. We have just finished an album with Edgar Winter, and have recorded or mixed other rock artists such as Doobie Brothers, Rod Stewart, Robert Palmer, David Bowie and Steeley Dan.

R-e/p: You mentioned that when you were overdubbing strings you used a lot of little speakers. Is there a problem with leakage?
JT: No. When I double them I reverse the phase of the microphones so that it makes the cue speaker spill almost nonexistent.

— continued overleaf

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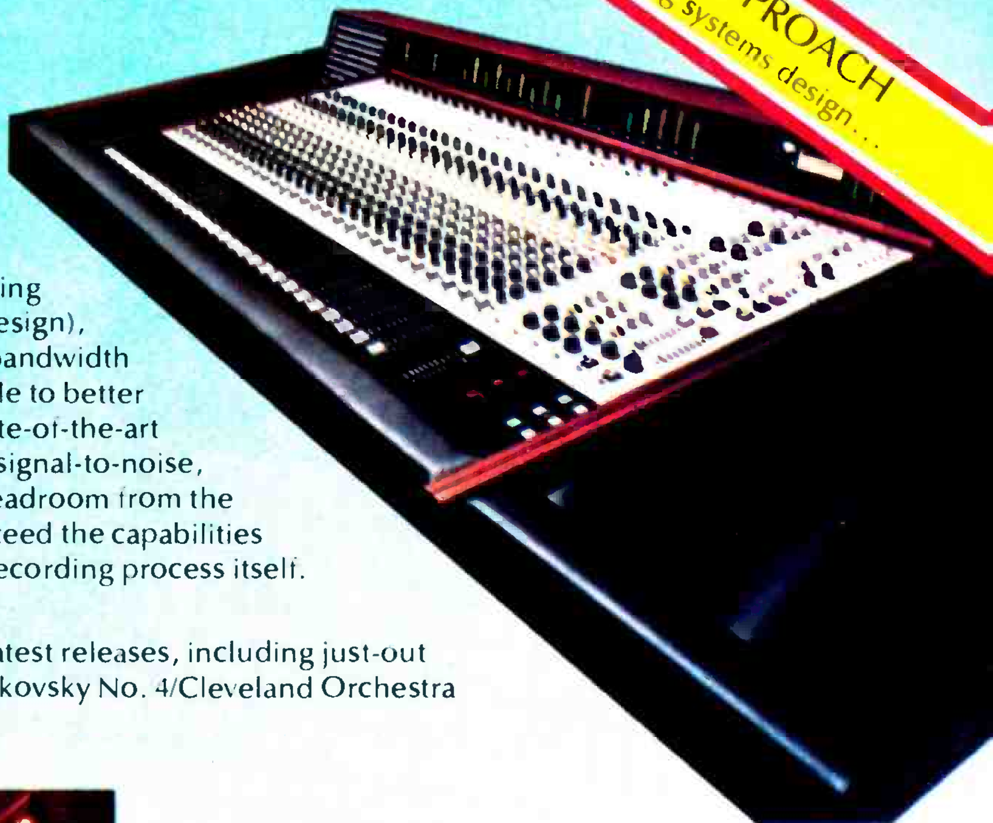

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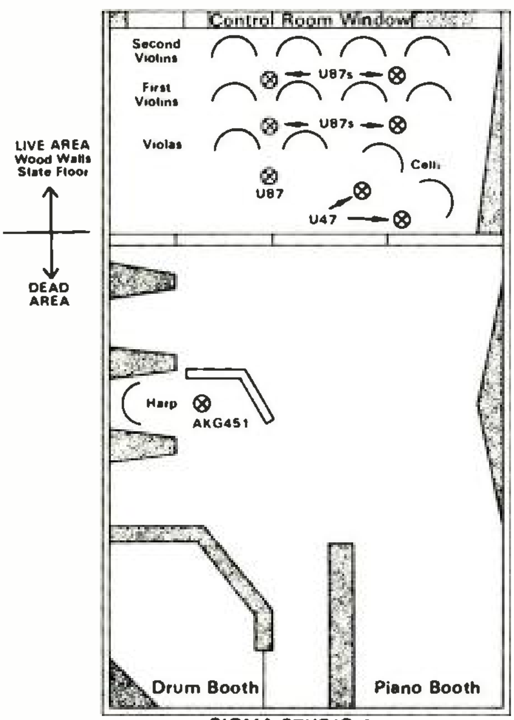
JOE TARSIA

R-e/p: How much does the phase reversal affect the ambience of the room?

JT: It doesn't. The only element with a phase relationship is the spill from the cue speaker.

R-e/p: When you do brass do you use speakers?

JT: In the early days Sigma started on a shoestring so it was a lot cheaper to buy a couple of inexpensive speakers and mount them on mikestands than to have fifteen or twenty pairs of earphones. This is going back to the sixties. The horn players now record with phones.



SIGMA STUDIO 4

R-e/p: Jones Girls', "You're Gonna Make Me Love Somebody Else." It sounds like there's some backwards echo on the claps?

JT: You know that's an interesting story. We had completed the Jones Girls' album and I had gotten a sample acetate. That's the way that PIR and Sigma work. An engineer who mixes a particular project, receives for evaluation a reference lacquer of the album. He makes any-necessary last minute changes and gives the necessary instructions to the cutter. I received, checked, and approved the Jones Girls' reference album and gave Frankford/Wayne final mastering instructions. The next day Kenny called me and told me he watned to re-sweeten and mix "You're Gonna Make Me Love Somebody Else." We went back to the studio, we overdubbed synthesizer, handclaps, and brought the Girls back in to double one of the vocal parts. The strings and horns on the original multi-track were erased. The new simple version of the same song was mixed for the second time with a completely different approach. Yes, we did try to be a little different with the handclaps.

R-e/p: Is that echo . . . is it processed?

JT: I made a tape of a continuous shaker sound which I use from time-to-time (I think the last time I used it was on the O'Jays' "I Love Music"). I gated the shaker tape and keyed it with a handclap track. I then ran the multi-track backwards and added echo.

R-e/p: It's a very good record. When you are mixing, especially in some of those long vamps, you'll hear things start to come up and go off and then jump out at you. When you start doing a mix, at what point do you start locking into muting sequences and what-have-you.

JT: Everybody has a different approach to it. Tom Bell is very specific and very precise. He writes what he wants the musicians to do and how he wants them to do it. In the case of Gamble & Huff, they allow a lot of freedom to the arranger and will have strings and horns from the downbeat to the very end of the song. Then the artist may do ad-lib overdub tracks from the beginning to the end. There will also be other effects all the way through the tune. You said that the recordings are full and heavily orchestrated. You should hear what we start out with.

The multi-track is played over and over and different combinations are tried and considered for their dramatic impact. This may mean that the strings and horns are used in the intro and then punched out until the first chorus. When we reach the vamp we may alternate between a horn figure and the vocal ad-lib or punch background vocals in for a few bars and then out again. This was definitely the case in the Jones Girls single. There is much more information on the multi-track than was every used on the released single. So as you can see it isn't until the mixing stage that the actual arrangement is decided upon.

R-e/p: Do they tell you what to mute or do you work it out for them to approve?

JT: I'll do it subject to Kenny's approval. We have been using automation for a long time, so sometimes Kenny will come into the studio. He will run an arrangement down and decide on the punches. We will write the mutes and then he leaves me to do the mix.

R-e/p: I would imagine you did a hell of a lot of editing before automation came along in order to get the same results. Did you do a great many mixes and then do a lot of editing?

JT: No, however we do now edit quite a bit. The seven or ten minute rhythm track is the reason for that. When it comes time to put a tune of this length in an album we usually cut it down to the best five minutes, and then for the single we edit it down to the best three minutes. It was never our practice to mix in pieces. Before automation there were just more hands required in the mix. Kenny is a capable mixer. With the use of the assistant, Kenny and myself, we usually got the job done. There was a time in the early days when I would play assistant. I would set up the EQ, the echo, the panning, etc., and he would ride the levels. Even now it is not uncommon for Kenny to reach over the board and change something while I am at work.

R-e/p: In your approach to EQ, do you try to do as little of it as possible and get what you want by microphone selection?

JT: Having done what I do for as long as I've done it, I usually have a good idea of what mike to use and where to place it to get as close as possible to the sound I'm looking for. Then I try to do the rest with EQ. In other words, in the quest for a particular sound the last alternative is EQ. But don't get me wrong, I'll use all the EQ I think I need as often as I think I need it. The case in point is the kick drum. I sometimes have so much kick EQ it's embarrassing.

R-e/p: What about the kick and snare drum?

JT: On the bass drum I use an AKG D200. On the snare I use an Altec 633. It is a very inexpensive discontinued PA microphone. It just has a certain characteristic which I happen to like. If I found something better I would use it. I've tried various mikes but that's my favorite. When it comes to microphones I'm always looking and I'm always open to change.

R-e/p: One of the most interesting O'Jays records is the "Ship Ahoy" album. The title cut was a very intricate mix of music and sound effects.

JT: That was a monster. We used a loop with wind sounds, one with ocean waves, another with the creaking sounds of a wooden ship and a loop of the sound of whip lashes. It was necessary to use the machines and aux equipment of two studios patched into the board to do the mix. It was definitely a three-man mix — the assistant, myself and Kenny.

I haven't mentioned Huff too much, when it comes to mixing that's one area he chooses not to directly be involved in. Huff's biggest contribution to the Gamble & Huff team is the writing of a song and the recording of the rhythm track. He directs the rhythm section, plays keyboards and sets the groove.

— *continued overleaf*

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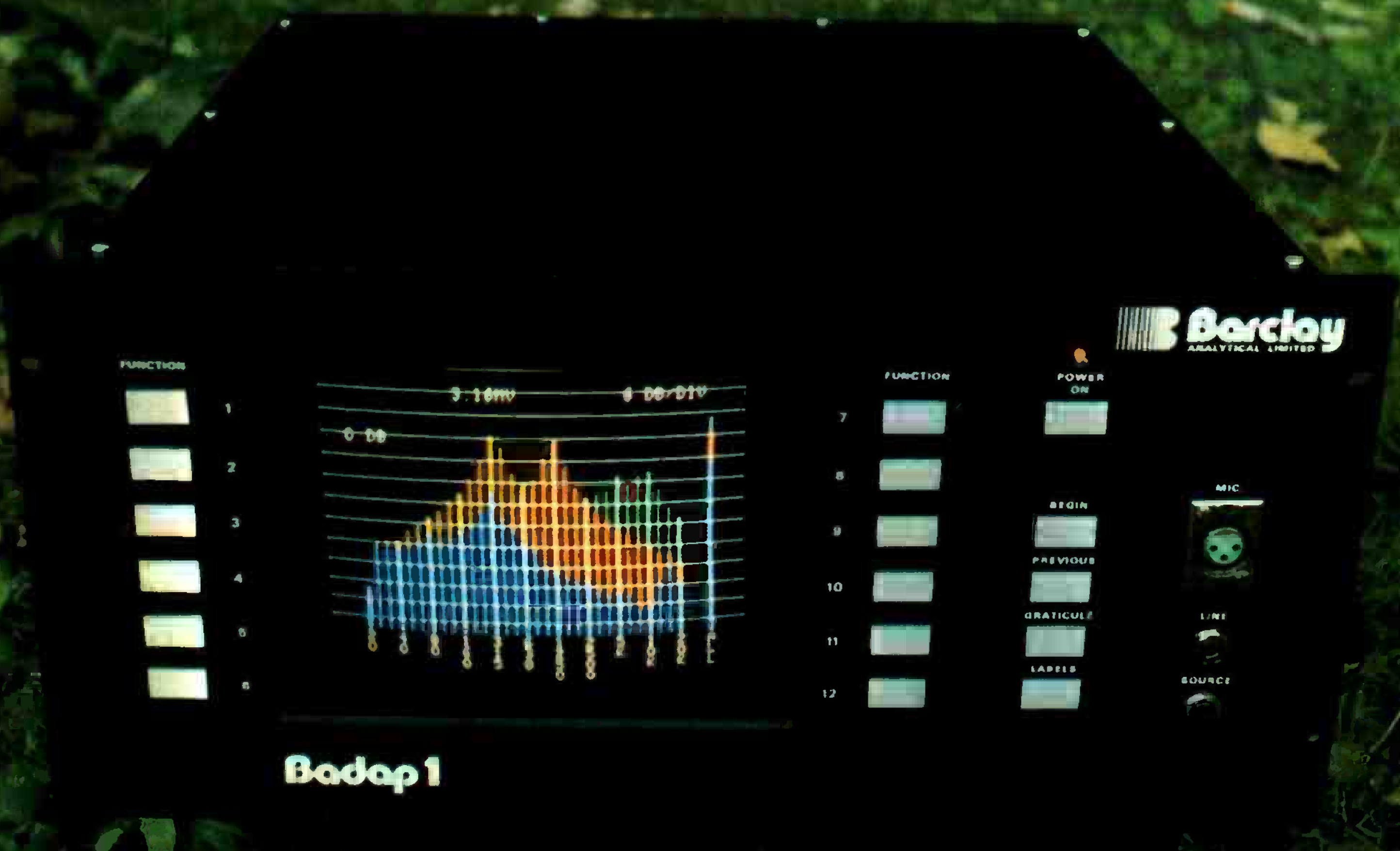

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JOE TARSIA

R-e/p: How long do you have to get your rhythm section happening?

JT: Fortunately, Gamble & Huff take a good bit of time and are very relaxed in the studio. They usually work with a cassette of piano and voice. Using the cassette as a guide they run the tune down over and over again until the musicians get the exact feel they are looking for. It's really a mixers' delight because I get an opportunity to change microphones and adjust and move things around while they are perfecting the chart.

R-e/p: Do yo try to mike them fairly open without a great deal of baffling so that they can hear one another?

JT: Yes. And their sessions are loud. Philadelphia musicians, with the exception of rock musicians, play louder than most sessions players I know. I remember years ago attending a New York City recording session. The electric guitar players played just loud enough to be picked up by the microphones. Huff likes his Fender-Rhodes loud. The musicians want to sit close together and feel the music. If you cover everything with baffles, they just play louder. There is no point in fighting it, there's a lot of feeling in black music and the engineer must allow the musicians to work the way they feel the most creative. It is the engineer's job to work around these obstacles. The engineer should not force the musicians into what he considers an ideal recording situation at the expense of their creative comfort.

R-e/p: Do you ever want to go back in and fix something?

JT: Yes, all the time.

R-e/p: The O'Jays' "Best Things Since Candy" — the bass sound in that record has a certain warm, wooliness to it that is very distinct. Do you use a direct on the bass, a mike, or a combination of both? Do you find that players like to play in the control room?

JT: Sometimes during overdubbing they play in the control room. Basically, however, the rhythm tracks are cut in total and are not piecemeal. With the exception of maybe little accents or a double guitar, everything is done simultaneously. The bass

is generally recorded through a transformer direct box. From my own experiments I prefer the transformer loading effect of the Fender bass pick-up as opposed to the very high impedance active direct box sound.

R-e/p: Is it something you guys have built yourselves?

JT: No, it's just a standard high impedance bridging coil. Originally, I used an Electro-Voice 666 on the B-15 Ampeg amplifier and direct injection from the bass pick-up. Later, I moved the direct injection point from the bass pick-up to the output of the Ampeg pre-amp. I have eliminated the microphone altogether and just use a transformer direct box right off the Fender pick-up. I limit the signal with an Allison Gain Brain. That's pretty much how I get what I want. The quality of the bass sound is highly dependent on the player. The only way you can get a good bass sound from a Fender Precision (they have that very gutty sound) is to play it close to the bridge and pull the strings. You can't tickle a Fender Precision.

R-e/p: Well, a couple of technical things. Have you found any signal processing equipment that you've been particularly impressed with?

JT: I like the Marshall Time Modulator. But their ergo-nomics leaves something to be desired and I think it scares a lot of people. You have to read the manual a couple of times before it becomes clear what the unit does. However, it's a real nice, clean piece of gear. I also like everything Lexicon makes, including their new 224 digital echo chamber.

R-e/p: What sort of monitoring do you use?

JT: In all of our studios, save one, we use the UREI 813 Time-Aligned system. Studio four, which is in Gamble & Huff's office building in Phiadelphia, has a very small control room and we use Big-Reds in there. So, basically, we are a 604 facility.

— continued on page 55 . . .

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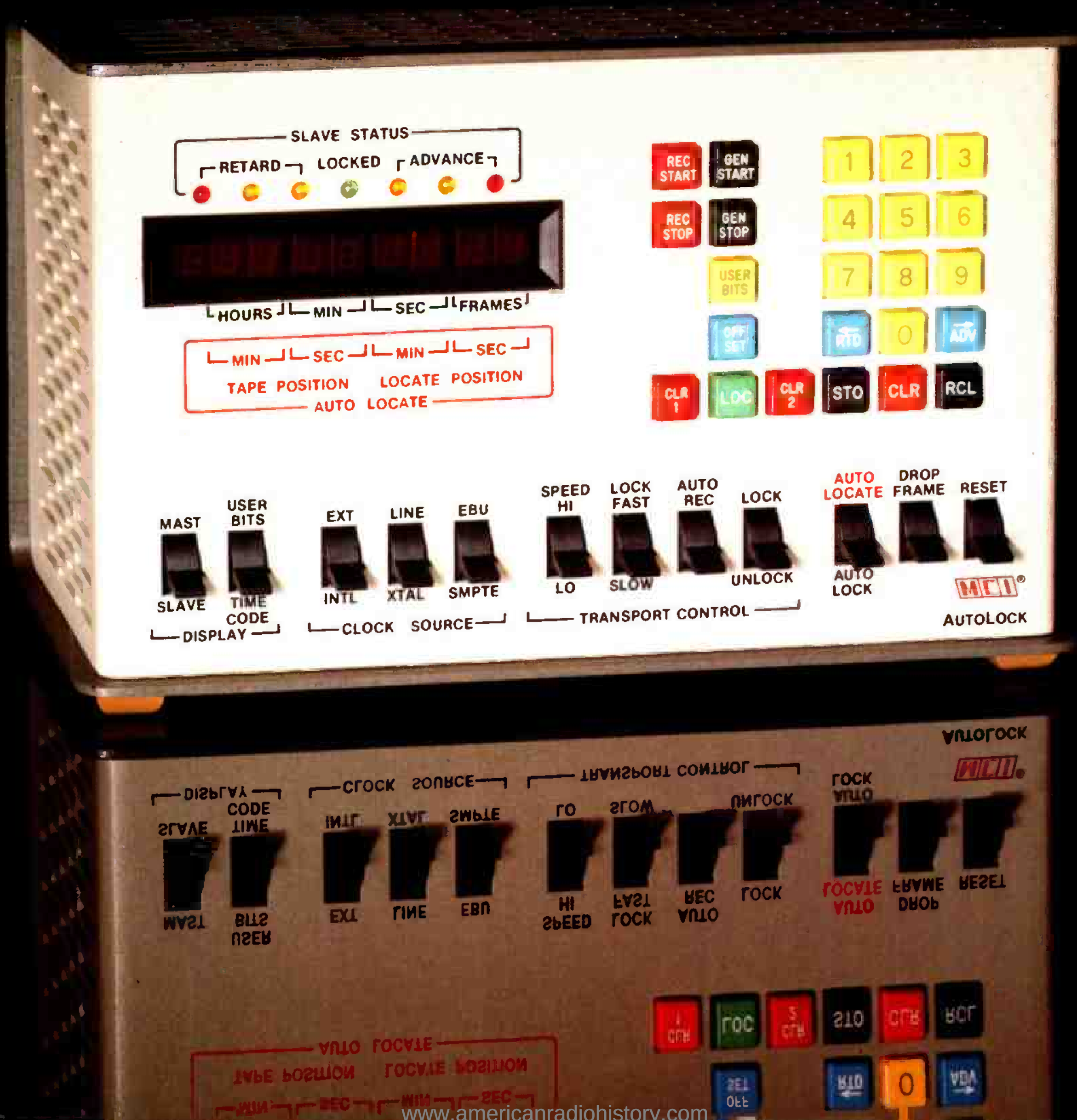

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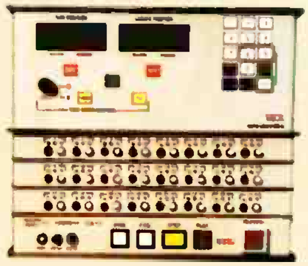
AutoLocator III



The optional AutoLocator III, with microprocessor based electronics, is the newest in the MCI AutoLocator line. New features include increased autolocate speed, increased functional capability and improved reliability. The AutoLocator III includes a tape velocity indicator reading in ips and 1/4 semitones, 10 memory locations, presettable up/down counter, real time display in minutes and seconds, flashing decimal point for a negative domain indicator and "shuttle" or repeat function. It is available in the same mechanical wrap with electronic and transport motion remote controls and includes a 35 foot cable.

RTZ III

An optional microprocessor based locator for the JH-110B transport series. Features include the return to zero function, four memory locations, presettable up/down real time counter reading in minutes and seconds, tape speed indicator in ips, and the ability to locate from positive or negative domain. The RTZ III is standard on the JH-110BC.

A special version, the RTZ III/M is standard on the JH-110M mastering machine. Features include 20 memory positions for banding/spiraling, expand/echo "leadout" lathe functions, plus 4 addressable tape position memories, the RTZ function and tape velocity indication in ips.

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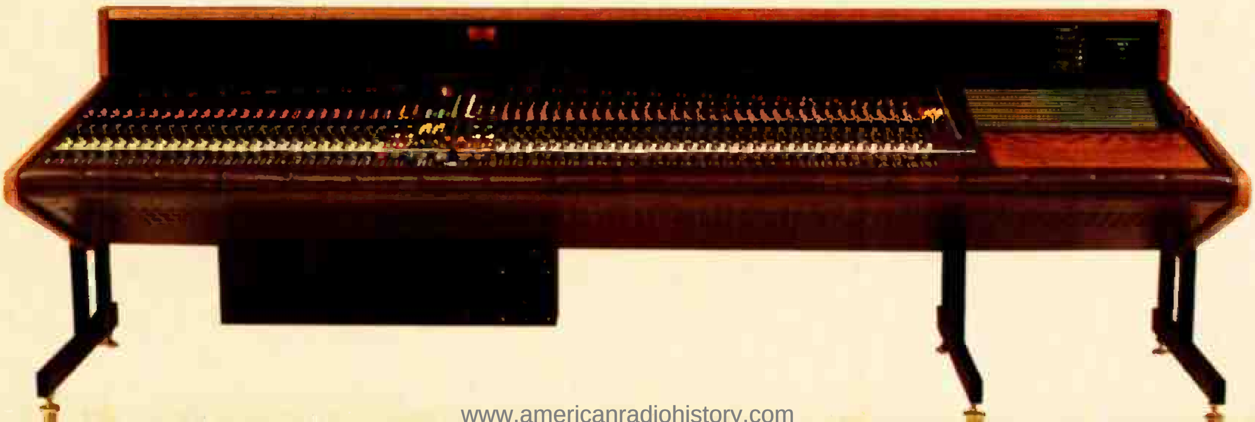

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LUBIN SIGMA SOUND'S

JOE TARSIA

. . . continued from page 50 —

R-e/p: Let's talk about being an owner of a studio business. You are running a very successful, busy, recording operation. There has been a great deal of talk about the softening of the record industry along with the general economy. A great deal of doom and gloom. How do you feel about the studio business right now? Is it going to taper off, or will new studios continue to open and existing ones expand and flourish?

JT: I can only wait and see. I think that the lull that we see at the moment is caused by a number of factors. I don't think it's a direct result of the inflation rate, although that has a bearing. I believe that it's caused by, 1) increase in record prices; and, 2) by the lack of material that people want to buy. I see the record business as being at a point similar to where it was in 1963 before the Beatles came on the scene and turned it completely around. I think the public is waiting for a new direction in music. We went from the highly produced Cameo type records in the fifties and sixties, to the self-contained group records of rock and roll during the sixties and early seventies, and now we're in the disco medium. I think we are at the crossroads of another change. The public is waiting for some creative producer to come along and hit us with a new direction. The last factor is how do you follow up a year-and-a-half of phenomenal sales from *Saturday Night Fever* and *Grease* with 27 million-or-so copies sold. That's a hard act to follow. I think all these factors contributed to the record business' current woes. I think that the record business and entertainment industry are here to stay. I would not alter any of my plans for expansion based on this temporary lull, because I think it's merely a pause to catch our breath. We will continue to go forward. I think the video recorder tied to high quality sound is going to open many possibilities in the market, and that the recording business will continue to grow.

R-e/p: Do you think it has much to do with the way record companies are being managed?

JT: I'm not familiar with that part of the business.

R-e/p: Have they been effected by the recent practice of shipping gold while knowing full well that there's going to be loads of returns?

JT: It's unfortunate that that is sometimes done. The record business is not like the shoe business. All the merchandising in the world will not bring people into the record store. If a store overstocks with shoes and runs a sale and, if you need a pair, you can go in and buy them. But no one needs a phonograph record. It's the songs and the artists that draw people. So I think it's a mistake to ship large quantities of records and think that merchandising is going to move them from the retail stores into the consumer's home.

R-e/p: Has disco to a certain degree dried up the record sales market?

JT: I think the twelve-inch single for consumer consumption was a mistake and has hurt album sales. If a person goes out and spends money for a disco version of a recording they are not apt to go out and buy the album.

R-e/p: The past few years have seen the emegence of the independent engineers and a decline of the staff mixer. It seems, however, with the increased sophistication and individual idiosyncracies of automated studios that this trend is reversing. Do you think this is occurring?

JT: Let me say that Sigma has always maintained a full mixing staff. We think we can serve our clients best with our own people because they are familiar with the room and have a certain knowledge and respect for the equipment. We have never gotten into a heavy outside engineer clientele. However, we would never refuse a client that wanted to bring in his own engineer. It would be pleasing to believe that we are going back to a staff situation. I think there will always be room for freelance engineers, but that's not Sigma's forte.

R-e/p: Have you any desire to move into production?

JT: If I thought I had enough imagination, I probably would have produced. I chose to work at making Sigma a better recording facility. Certainly anybody that's been in any part of the recording business, be it a lawyer or publicist, or whatever, at one time or another has contemplated producing. But I find myself as being most effective as a critic and helper in recording. I think I'm best offering suggestions to the creators. As you can see I'm an old gray-haired man. It's kind of strange. I don't think there's too many guys around in my position who stick exclusively to mixing. The Roy Halee's and Phil Ramone's are all involved in production. It's just that I really enjoy recording and for that reason I haven't had any great desire to get into production.

R-e/p: Would you care to share your experiences as a studio owner with someone who just opened a new facility? What suggestions would you make, for instance, in regard to spec deals where somebody says, "Hey, if I record here, you help me do a demo, and if it hits I'll do the master here."

JT: Well, one thing is, you have to be very selective in who you make that offer to. It requires an artistic judgement. This new studio has one thing that it can offer and that is time. The newcomer may have more time than he can sell. If I were in that position and someone offered me a deal, so to speak, on spec, the only thing I could lose is my time so I might entertain such an idea. Depending on who the deal is with or the owners knowledge of them, it probably would be best to run it through an attorney to protect the studio and the client's interest.

R-e/p: As an owner, what do you think is your biggest problem in operating a studio?

JT: I would say, first, keeping the studio state-of-the-art; purchasing today's expensive audio equipment wisely so as to avoid premature obsolescence. Just as important is keeping the client happy. Sigma puts a lot of time into seeing that the clients get what they're going after. Making my staff of engineers understand that they should be subservient to the producer is very difficult sometimes. A mixer is a servant. He does for the client, the producer, what the producer doesn't know how to do for himself. The mechanics. He must interpret what the producer says and translate it into the mechanical moves. Many times there is a tendency for the engineer to think he knows what's best for the producer. It's the biggest mistake an engineer can make. He should never try to make artistic judgements and jam them down the client's throat or to decide that the product isn't hit material and lose interest. I've been embarrassed many times by records that I thought were hits and weren't, and vice versa. You've got to maintain that excitement and interest. Obviously, we are all human beings and sometimes it's forced because you can't hear what the producer is going after. But you should try your best not to make these artistic judgements or say, "This piece of material is a waste of time and I can't put my heart and soul into it." When you do that you become ineffective as a mixer or engineer and you're going to hurt or lose the client.

R-e/p: Do you think with label cutbacks that the average album budget will also decline?

JT: I think we can all operate more efficiently. When times are good you tend to use a lot more limousine service and deluxe accommodations, showing up late for recording sessions, cancelling time.

One of the philosophies that Kenny Gamble has in the studio is that he will accept 95% because the other 5% is extremely expensive to attain. If you buy a stereo system for a thousand dollars and it gives you a certain level of performance, and then you turn around and buy one at twice that cost, you may improve the perform-

LUBIN SIGMA SOUND'S

JOE TARSIA

ance by 25%. If you then spend ten times that amount you may improve the performance by only a couple of per cent. Each time the investment is doubled the percentage of improvement becomes smaller and smaller. The point is one of practicalities. When you're striving for excellence in a studio there's a point where it becomes impractical to go further. You may be dealing with subtleties that only the most acute ear can discern. I'm not saying that we should strive for less than excellence, but the record business is a business. The idea of a business is to turn a profit. There is a practical end to which you should strive and not endlessly stay in the studio. There was a time in the good old days when a producer went into the studio for a three-hour session and cut three tunes and had a chart for the fourth, just in case. Today it's not uncommon to go into a rhythm session and spend the first three hours just rehearsing. That's expensive.

In 1974 there was an economic recession. During that period the recording business actually grew. We never felt that recession. In fact, it was a very good time for us. I believe the key to turning our industry around is simply producing product that people want to buy. Who knows, maybe the vehicle for this is already here in the form of the new wave rock and roll.

R-e/p: Something that is interesting as far as the kind of equipment that we are using is that the conventional money lenders don't seem to have a good concept as to what this equipment is worth at the end of five years. They tend to appraise it in the same light as computer equipment. Have you found that your financial people are starting to understand that this equipment maintains its value?

JT: In our particular banking area, they have a pretty long yardstick to go by. The record shows that because the United States has the leadership position in the recording field and our technology is far in advance of almost all other countries. We have a market for used recording equipment throughout the world. Up until recently you could sell an Ampex 351 for more money than you purchased it for. They are one of the favorites of the broadcast industry, and used recording consoles have a big market in South America.

R-e/p: We've talked a lot about studio acoustics over the past few years and not much about the decor. Part of what you say about giving the client what he wants comes down to the feel of the place. What do you think is important for a recording environment?

JT: It should be comfortable but it should not put you to sleep. The original studios that I worked in were basically factories. They were large, brightly lit rooms with acoustical tile on the walls. Today's modern studio tends to look more like a living room than a place of business, and that's fine if we can get the job done. The living room approach has some merits because many times the producer and musicians spend many continuous hours in these rooms. Nevertheless, we must never loose sight of what we all are here for. For that reason, I believe, that a studio should be a clean, comfortable place to work and conducive to creativity and productivity.

R-e/p: Air conditioning is a difficult problem. How have you dealt with that in your recording studio environment?

JT: You can have three people in the studio or control room and two of them will say it's too warm, and the third will feel it's too cold. It's a problem. We have found that it's necessary not only to control the temperature but also to maintain constant humidity. One of the main criteria is that the studio and the control room must be controlled by separate air conditioning devices. Anything less requires constant adjustment.

R-e/p: And you think a humidistat is essential?

JT: Sure, depending upon the climate of the particular area, the grand piano can absorb a lot of moisture. Humidity effects any acoustic environment. It also can change the timbre of a room, the absorption of high frequencies. The propagation of high frequencies in a humid environment is different from that of a dry one. We know that moisture is detrimental to a speaker cone.

R-e/p: Or a microphone.

JT: Of course. We have added dehumidification and humidification to our air conditioning system, plus electronic air filters so that we are able to . . .

R-e/p: Clean out smoke.

JT: The amount of film that builds up from tobacco is incredible.

R-e/p: How do you deal with studio phone calls. Everybody wants to make a phone call.

JT: The receptionist screens all the incoming calls. We try to ask the client before he goes into the studio if he wants his calls or if he wants messages taken. The client also decides before he enters the studio if he wants calls put through to the musicians or engineers.

R-e/p: What about calling out?

JT: Associated with our control rooms is a private lounge for the producer. It has a phone that can only make local calls. Long distance calls have to be made through the operator.

'SPARS'
The Society of Professional Audio Recording Studios

R-e/p: Tell me about the new studio organization, SPARS.

JT: In June of 1979 MCI held a studio workshop. At that workshop there were sixteen studios represented. In the course of our three days of discussion regarding the design of future recording consoles it was proposed that an organization of recording studios be formed. It was amazing. Everybody there felt that good things could come from a close dialogue between people in our business. The concensus was that we were at the crossroads of a lot of issues.

Concerns such as the present slump in record sales and new technical advances such as digital recording and automated consoles, etc., needed exploration and further discussions. There are also other things related to the recording business that further dialogue would help.

For example, establishing ideal recording levels and agreement as to what tones should be at the top of the reel. I'm sure that disk cutting houses would have plenty of suggestions for us concerning the preparation of masters. These are just simple items that would make it easier for a client to move from studio-to-studio and be confident of finding certain consistencies that would make his life easier. We are not looking for each studio to be the same. We want each studio to retain its individual methods and techniques. But if a producer uses a SPARS studio, he can be confident that that studio has and will provide a certain minimum quality of equipment and service and has pledged to adhere to association guidelines of good practice. Beyond that point it's up to the individual studio to say, "Hey, that isn't good enough for me. I'm going to supply even more service and equipment." I think you get the idea.

R-e/p: Will they do things like try to standardize legend sheets and the kind of information that goes on a tape box? The sort of things that are very different for every studio?

JT: There will be open discussion regarding those areas, such as take sheets and billing procedures. There are times when just a glossary of terms that the girl in the bookkeeping office could use would be invaluable. Now she deals with a number of terms, individual to each studio, that describes the same event. Through SPARS we will address our clients and say, "What

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do you find to be a problem in working with a number of different studios? Where do you find that description and types of services become hard to deal with?" That's one area. Another is through education. We plan in the future to devise a curriculum that we think could produce good mixers and maintenance people. We might make suggestions to various schools and supply the curriculum to them. We would hire people who have this type of qualification.

R-e/p: That would be excellent.
JT: It's our intention to ask manufacturers to hold additional seminars on the repair and care of the various types of equipment that they sell. We'll offer the opportunity for each manufacturer to come and speak to us about the benefits and care of a particular piece of equipment. Also, open up discussions with manufacturers as to where we think our craft is going and what kind of equipment we will need five years down the road. You know we are in a technological explosion; the advent of the IC and the microprocessor has resulted in a rush of new and expensive services. We can help lessen the possibility of a manufacturer or a studio being involved with equipment that a year or two down the road might prove to be obsolete.

R-e/p: What you are also saying is the organization will help the manufacturer more closely follow the equipment's practical use.
JT: Right. There are things we want understood. The studio business is a small one. The number of studios that record 95% of the record product is a very small group. We're faced with special kinds of problems and we sort of muddle through without the benefit of exchanging experience with others. SPARS, we believe, will help us streamline our craft through innovation, communication and education.

R-e/p: What are the qualifications for membership?
JT: We have a membership application with basic requirements. The applicant must be a high-quality mastering room or 24-track studio, all of which must be state-of-the-art. They must have established that their primary business has been sound recording for at least two years. Those are basically the requirements.

R-e/p: Is the organization going to have some sort of general office?
JT: Yes, we may hire an executive director who will deal with day-to-day activities. There will be a president, chairman of the board, and regional vice presidents. They will be the policy making group of the organization. We are open to any applicant who is willing to follow high standards. SPARS now has an office located in Philadelphia which is the center for information and activity. The address is: 7th Floor, 215 S. Broad Street, Philadelphia, Pennsylvania 19107. Telephone: (215) 735-9666.

R-e/p: How should a facility proceed that wishes to join SPARS and meets these requirements?
JT: Along with the application they have to send a check for two thousand dollars to the address I just mentioned. The admissions committee will then assess the applicant. If it is determined that the applicant's studio is state-of-the-art, that it has a good reputation, and that it conducts business ethically, the application would then be submitted to the board along with a recommendation to accept the applicant for membership. Certainly if a studio is not accepted the two thousand dollars would be

returned.

R-e/p: You are now the president?
JT: I am the interim president. Elections will be held this November.

R-e/p: How many studios are involved so far?
JT: There are now 26 studios. I hope, however, that before the November SPARS meeting we will have 30 to 35 member studios. I think eventually membership will be somewhere between 50 and 75 studios. The organization will benefit all studios, members and non-members alike. The dissemination of information will be to all. We believe that what's good for the SPARS members, who are the leaders, will be good for the smaller studios, and those studios who might be doing less than professional work.

R-e/p: Are you going to try and make it possible for the eight track and sixteen track studios to be "affiliates?"
JT: At this time there is no plan for affiliate members, but the seminars and educational conventions would be open to anyone who wishes to attend.

R-e/p: How will they be able to know of these functions?
JT: Through public announcement, or newsletter and through trade magazines.

R-e/p: Are you going to limit membership to studios or are you going to accept sponsorship from manufacturers?
JT: At this time we are going to limit membership to studios and mastering rooms, but we look forward to having manufacturers participate in our seminars and educational programs. We've also talked about having seminars between record companies and SPARS so that we can understand how we may serve our clients better. We are looking to open up avenues of communcation with both the people we serve and the people who serve us.

R-e/p: So you would like to get, for instance, CBS Records to participate?
JT: Yes. As a matter of fact, we will encourage the participation of record labels in our activities, not only with regard to recording but also to engage in dialogue with our member mastering rooms to enable them to better serve the needs of the various pressing plants. We think that this would extremely helpful.

R-e/p: What do you see as the future of the studio business in general?
JT: Technically, I see a storage medium that's not necessarily a tape machine. The recording console will be an extremely simple device which merely transfers the microphone information to the tape, to be treated totally in the remix. I see the mixing console changing form completely. It will be controlled by microprocessors with video displays and will be about the size of a coffee table.

R-e/p: Do you see video becoming a part of the record business?
JT: Yes, in part. I think the marriage can only help us.

R-e/p: What do you see in the future for you? You've been opening new studios for the last few years, one every now and then. . .
JT: Is there an end? I don't know. I want to be a part of the changes in the business. I want to have input into things which are happening. I look forward to the video cassette and laser recording. I would like to see Sigma move into the video field. As far as opening studios in other places, it requires only money. But to run studios successfully requires people — qualified, interested people. The people who work for Sigma have a strong desire to see it grow. The possibilities for Sigma's expansion are limitless. We are confined only by our ability to find qualified people who see the dream as we do. □□□

18 Foot Diameter Center Piece for Tiffany's Disco, San Jose, California. Designed by The Design Trust, Santa Monica, California.

DISCO SOUND SYSTEM DESIGN

by Kenneth R. Fause

The term Discotheque was coined in France in the late 1950s to refer to a place of entertainment where the music was derived from a record library. From those few, low budget clubs in France there arose what is now an international entertainment and social phenomenon. On the occasion of the New York Disco Show in February of this year, the staff of **Billboard Magazine** estimated the United States market volume of Disco clubs at $8 billion annually. Notice that this figure is for club operations only, and does not include the market volume of support industries such as record production, sound and lighting sales, food and beverage supplies, fashions, and the like.

Disco as an entertainment medium is now at least 20 years old. In this time appropriately programmed recorded music has proven to be cost effective entertainment compared to live musicians in the club environment. Where live entertainment remains a strong market asset, some clubs have experienced success using a format of alternate sets of live and recorded music; this permits continuous entertainment without incurring the expense of employing two live bands. Thus, for economic reasons, Disco in some form is likely to survive in the future club entertainment market mix. The present scale of the Disco marketplace and the cost-effective nature of Disco club operation has attracted the notice of the serious entrepreneur. The author proposes to address the situation wherein the club is designed, built and operated primarily as a **commercial investment venture**. In this case, the club and all its components are subject to the general investment decision process: each asset is evaluated on the basis of capital cost, useful asset life, salvage value at end of asset life, and probably rate of return on capital invested. The enlightened investor attempts to minimize risks while maximizing return on investment. Predictable results will, of course, minimize risk; in club operation, capturing a substantial share of the market activity should guarantee a reasonable rate of return if operating costs are held to appropriate levels. Referenced to the club sound system, we may observe that present design methods properly executed will yield entirely predictable results, and that a "good" sound system should be a strategic asset which will assist in capturing a market share.

It should be emphasized that the design of an investment club sound system is not merely a problem in engineering; it is as well an exercise in **engineering economics. The object is to engineer an optimum system which executes the functions required of it within the constraint of the investor's budget, and in support of the investment objective.**

The design method for the Disco sound system is derived from the well-known techniques for engineered design of sound **reinforcement** systems based on the work of the Boners, Don Davis, George Augspurger, and many other contributors to the literature. For the sake of this article, we shall attempt to restrain the discussion to those specific cases wherein the standard practice is modified to suit the Disco function or environment. Certain design problems are

—The Author—

After graduating from Cornell University with a B.S. in engineering, author KEN FAUSE joined the school's staff as a Research Engineer for the Social Psychology Laboratory where he developed data acquisition and analysis systems for human communication behavior experiments. More recently he earned an M.A. in Theater Arts at UCLA. Fause currently heads his own consulting firm, Fause & Associates, which is involved in performance and presentation technology. The consultant's work includes room acoustic design, entertainment sound systems and special-purpose audio/visual systems.

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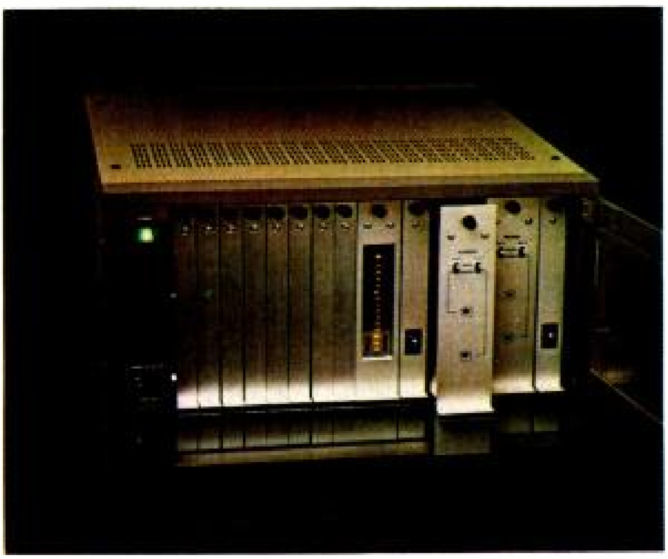

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analyzed with the aid of techniques drawn from the disciplines of Architectural Acoustics and Industrial Noise Control. The author hastens to mention that the design methods described are by no means unique to the practice of his firm; we are certainly aware of other system designers using similar and highly successful approaches.

In any engineering endeavor, the first step is to accurately

... built from the ground up for use as a Disco, VALENTINO'S on the coast at Mazatlan, Mexico, is surely among the world's most beautiful Disco venues.

The Spectra Sonics powered sound system was designed by Ray Kimber.

analyzed with the aid of techniques drawn from the disciplines of Architectural Acoustics and Industrial Noise Control. The author hastens to mention that the design methods described are by no means unique to the practice of his firm; we are certainly aware of other system designers using similar and highly successful approahces.

In any engineering endeavor, the first step is to accurately **define the function** of the proposed system. **Performance criteria** may then be selected which will detail the degree of nicety to which the function is to be executed. Consideration of the **environment** in which the chosen criteria must be met leads to **specification** of the specific elements of the engineering system.

Disco Sound System Function

The essential function of a Disco sound system is to play recorded music to an audience, usually of dancers, as one element which contributes, along with lighting, decor and special effects, to a total entertainment experience. In some cases, the entertainment experience is attained by deliberate sensory overload. We are in a sense describing a communication network consisting of Sources, a Transmission System, and Receptors (the audience). In Disco, the message communicated is **Boogie!**

Sources

By definition, the source music is pre-recorded; therefore a means of playback from the storage medium in use is required. At present, almost all commercial material is released on phonograph disk, so playback will be via a turntable system. For continuity, at least two turntables and a method of blending sources (a mixer with pre-hear function) are necessary; for special effects and redundancy, a third turntable system is desirable. Variable speed turntables are virtually essential to permit the DJ to vary tempo to suit mood and continuity. A microphone should be available to the DJ for occasional program announcements and for emergency crowd control; in some formats, a second microphone on a long lead will be useful to facilitate dance instruction of large groups. Installation of a tape recorder/player is urged in recognition of the finite size of the human bladder — the DJ will have to take a break sometime in the course of an evening.

System Criteria: No Boogie; No Bucks

Above all, it is essential that any technical systems for Disco service be **reliable** under conditions of 6 to 10 hours of continuous daily service. System downtime means lost customer revenue for the duration of the service outage, and possibly permanent loss of customers to the competition. A sound system failure when a club is full will almost certainly result in irate customers; in a large club there is a real risk of public disturbance. For these reasons, Disco sound systems should be designed to fail-soft; that is failure of a given system component should cause only a reasonably graceful deterioration of system sound quality, not a total service outage. Selection of conservative operating parameters and use of redundant equipment are obvious design techniques. The guiding rule is simple: No boogie; no bucks!

The nature of the Disco experience implies the need for high level, wide frequency range acoustic output to the dance floor. For a given club design, the sound level on the dance floor is a function of the expectation of the club's chosen market sector; Figure 1 is indicative of prevalent sound pressure levels for various markets in dB referenced to 20 micro Pascals, C weighted. Important to notice is the possibiltity of temporary or permanent hearing damage associated with exposure to sound levels common to all but the middle-of-the-road market (MOR). Designers, installers, operators and owners of high level Disco sound systems are urged to assess the risk of both short and long-term high-level sound exposure[1, 2] and to consult with qualified consul as to potential legal liability toward customers and employees. (See Polon, Martin, "dB's Can Be Hazardous To Your Health," page , this issue.)

Figure 1

ACOUSTIC LEVELS ON DANCE FLOOR, L_C

MARKET	AVERAGE	PEAK
FULL-ON**	110	125
MODERATE*	95	110
M.O.R.	80	95

** - Caution: Potential Permanent Threshold Shift

* - Caution: Potential Temporary Threshold Shift

It is the opinion of this author that Disco sound systems, especially those intended for high-level output should be designed for low listener fatigue: the implied requirement is for low distortion at all operating levels including maximum. The aim is to somewhat lessen the risk of acoustic trauma due to high level sound exposure. There is an unfortunate paradox: extremely clean systems may not "sound loud" even when delivering acoustic output that makes voice communication nearly impossible at one foot distance. By past experience, most owners and customers seem to have been conditioned to accept that "loud" and "distorted" are interchangeable descriptions for club sound systems.

A design method which contributes to a clean, low fatigue system is the providing of massive electrical and transducer headroom so that peak clipping is prevented. We generally design for a peak clipping level some 15 to 20 dB above the selected **maximum** average output level. Some courage and determination is required when specifying power amplifiers with such overhead. Of course, it is necessary to exercise extreme caution in selecting transducers to be so powered: an operator error in signal duty cycle could easily result in the expensive smell of roasted loudspeakers. This would violate our prime requirement for system reliability.

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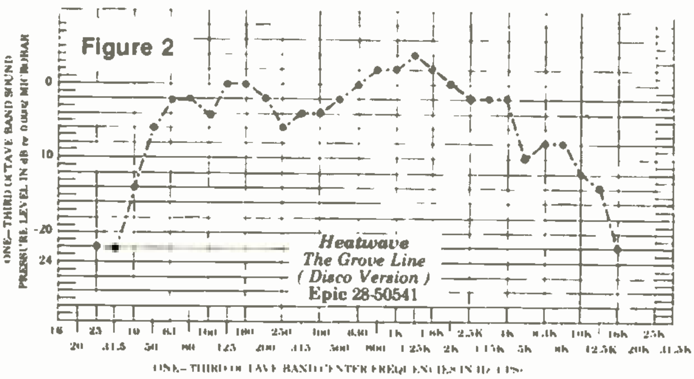
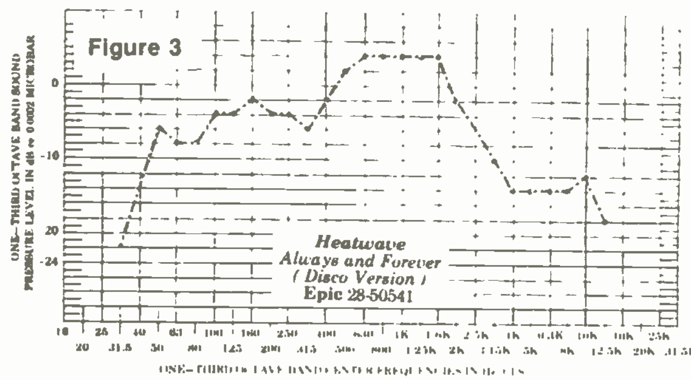
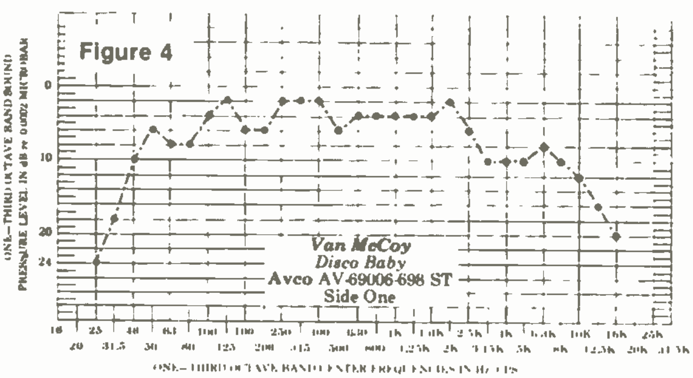
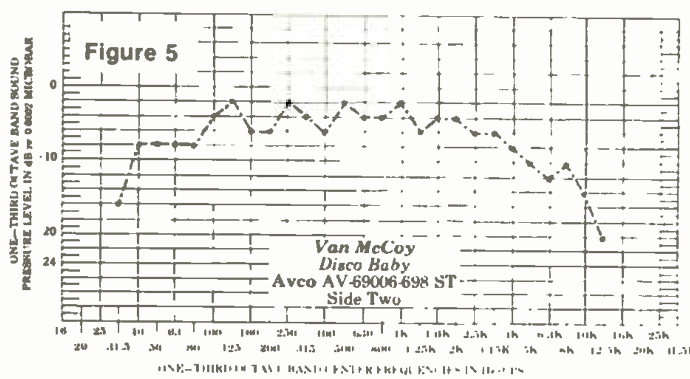
Dynamic range manipulation in the form of compressors and/or limiters may be used to reduce the design peak-to-average ratio to a more economical figure. Unfortunately, decreasing the energy density of the signal in this manner may also increase the potential hazard of acoustic trauma; each case should be investigated for its specific conditions. We generally prefer to design high-headroom systems with the gain structure manipulated such that attempts to drive the system to hearing-damage levels will result in flagrant hard clipping. The sudden, radical deterioration of sound quality is intended to serve as clear notice that the system is being driven beyond its design limits.

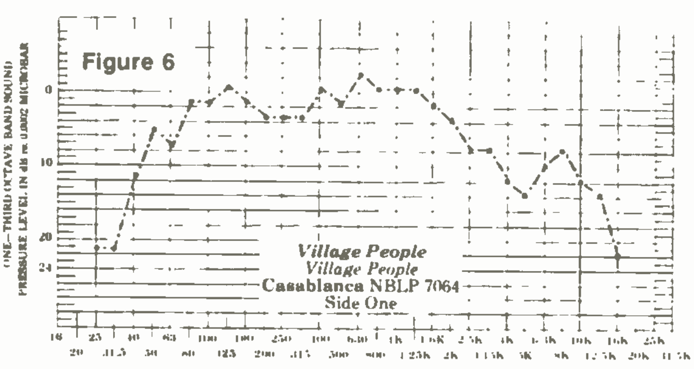
The sound spectra to be delivered to the dance floor may be derived from spectrum analysis of a range of typical program material. Cumulative spectra for 8 sides of Disco titles of varying style are shown as Figures 2 through 9. Data were

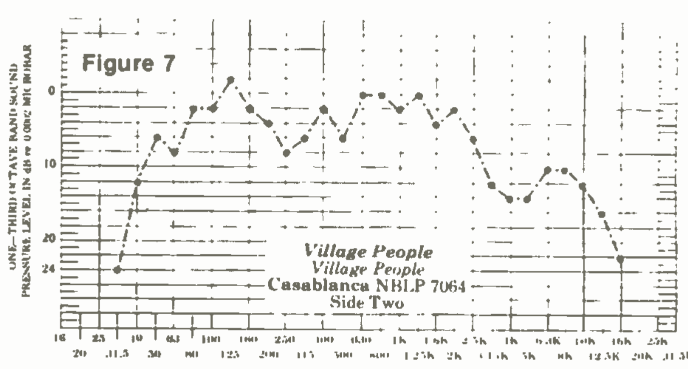


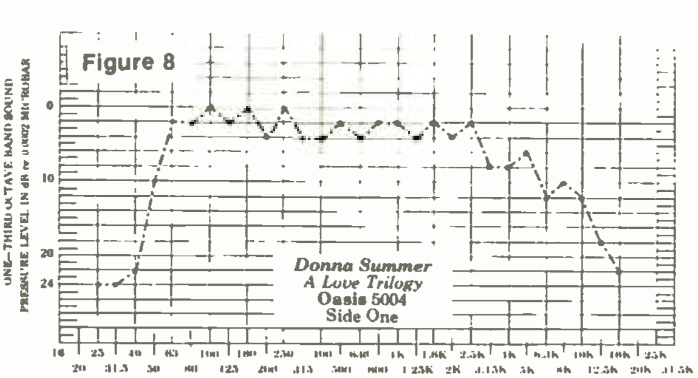


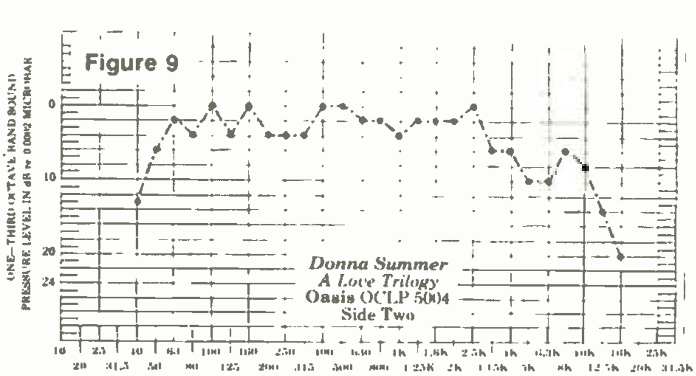












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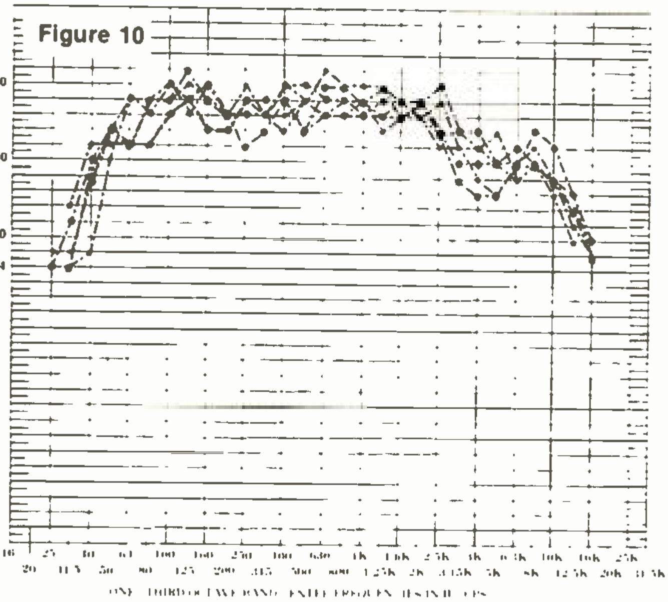
Ken Fause: DISCO SOUND SYSTEM DESIGN

taken using a Thorens TD-165 turntable/arm combination, Shure V15 Type IV cartridge, Bill Isenberg phono preamp (IC version) and Ivie 30A spectrum analyzer. The analyzer was set in accumulate mode at 2 dB per step with the memory initialized for each side. Gain was held constant for all sides with the 0 dB reference baseline at 110 dB μV (.316V); from this one may observe the variation in cutting levels among the sides. Detector decay rate was set at the "D1" position for a time constant of approximately 10 ms at the 1 kHz band center and corresponding approximately constant confidence limits in all other bands.

Of course, this sample is quite limited in a statistical sense, but useful information is gained. Of special interest is Figure 10, which is a graphical derivation of the maxima of six sides

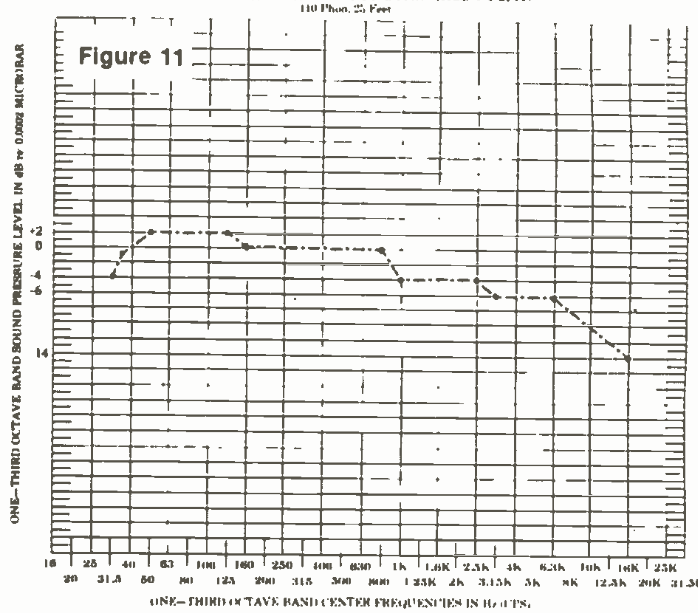


— the somewhat abberant "Heatwave" sides having been deleted from this data. A fairly smooth "haystack" shape appears which should refute the often-heard conjecture that Disco music is all bass and high with no midrange. Clearly untrue.

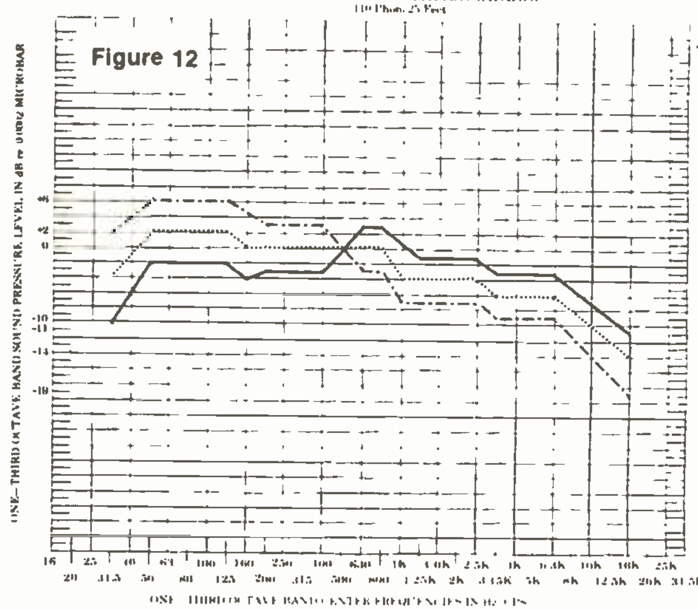
Experience in equalizing a number of Disco sound systems of various scales indicates that a listener preference function may be developed for the amplitude versus frequency characteristic of the system acoustic response. (This is analogous to the "Preferred House Curve" familiar in sound reinforcement practice.) It should be no surprise that the specific shape of the Disco Listener Preference Function will vary with the average system loudness level and the average acoustic source-to-listener distance. A sample curve is shown as Figure 11; this has proven suitable for a 100 Phon average loudness level[3] and 25 foot listener distance. In this case, the particular shelved characteristic resulted from manipulation of driver levels in large four-way systems to obtain best listener satisfaction on Disco music playback. (The music samples had source spectral characteristics similar to those previously described.) Numerous subsequent systems of this type adjusted to the characteristic have met with gratifying acceptance in the marketplace.

In practice it is necessary to define some realistic tolerance limits about the chosen preference curve when issuing a given system specification. For the curve discussed above, we have successfully employed the tolerance "window" shown as Figure 12. With pink noise excitation of the system, sound

LISTENER PREFERENCE FUNCTION (House Curve)
110 Phon, 25 Feet



"A" GRADE SPECIFICATION WINDOW
110 Phon, 25 Feet



pressure level in one-third octave bands is required to fall within the specification envelope when measured at standing ear height (5 foot, 6 inches) at any point on the dance floor. In this manner allowable errors in both frequency response and spatial variation are defined under a single specification. If reasonable care is taken in selecting appropriate transducers and most of the curve shape is derived by setting relative driver levels; then very little narrow band EQ (one-third octave) should be required to meet the specification if the room characteristics are at all sensible.

For reference, we have reproduced here as Figure 13 a portion of ISO Recommendation R226-1961 "Normal Equal-Loudness Contours for Pure Tones and Normal Threshold of Hearing Under Free Field Listening Conditions" (copyrighted by the American National Standards Institute, 1430 Broadway, New York, NY 10018; reproduced with permission). Although the title of the standard defines the

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. . . It means you can send all the guitars and keyboards from the stage to the mixer on one line of your existing microphone snake. MAINLINE eliminates costly balancing transformers and tons of additional cable. You save money, time, and hassle. It's simpler, more reliable, and most important, the music sounds much better!

. . . It means simple, quick system expansion for club, church or studio without costly new cable installation. Each existing built-in microphone cable can now carry either eight microphones or eight instruments.

. . . It means an engineer can now run a stereo tri-amp sound system with the crossovers located at the mixer allowing balance control during performance. The six separate line sends can be carried by any one channel of a microphone snake, with two sends to spare!

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DOWN THE LINE . . .

All this is possible right now. Imagine however, what this breakthrough means for the future! MAINLINE sets new standards for audio performance (live or in studio). It has eliminated one of the most common deterrents to audio excellence . . . loss of signal quality due to too much cable.

MAINLINE also eliminates the need for massive quantities of transformers and 3-pin connectors . . . greatly reduces set-up time . . . requires no maintenance . . . could cut the cost of a typical 16 channel system by 40% while improving reliability, flexibility and sound quality.

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With the exception of a few select audio dealers and commercial sound contractors, MAINLINE hasn't been available to the public. General distribution is 12 to 18 months away.

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Ken Fause: Disco Design

conditions under which the data presented are strictly valid, the same trends of course exist for music presentation at various loudness levels. It is interesting to compare the low-frequency lift specified in the Listener Preference Curve of Figure 11, derived empirically, with the 100 Phon contour of Figure 13: theory and practice do agree quite closely.

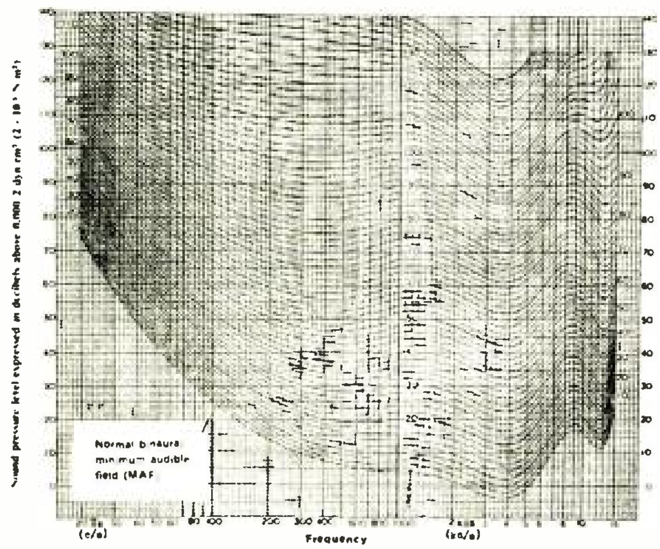

Figure 13: Normal equal-loudness contours for pure tones.
(Binaural free-field listening.)

Control Of Sound Distribution

Controlled sound distribution is usually another system design objective. The owner often requests the designer to produce thunderous levels on the dance floor . . . but please don't disturb the seductive conversations in the cocktail lounge immediately adjacent. Some compromise is inevitible.

When selecting sound coverage patterns it is essential to recall that the DJ must hear the mix presented to the dance floor: this is the only way the DJ can properly judge and manipulate the acoustic stimulus to the dancers. When conditions of geometry and decor prevent the DJ booth being in the direct field of the dance floor system, a "DJ monitor" system closely matched in spectrum and intensity to the main system is necessary. In remedial work, we frequently encounter the situation where the DJ booth is removed from the dance floor and located in a reverberant field with a predominantly bass spectrum content. To correct the apparent muddy sound at the control point, the DJ turns up the high end, making the system very shrill to the dancers and causing rapid tweeter failure. With several blown tweeters, the system sounds muddy again; the DJ then turns up the high end still more and soon blows the remaining tweeters. Club management will become annoyed at a continuous expense for tweeter replacement. It is kinder to both patrons and management to design so as to avoid the problem in the first place.

SPECIFIC DESIGN TECHNIQUES

It is convenient and appropriate in sense of philosophy, to discuss and design any sound system working from the listeners back to the sources.

Loudspeaker System Selection

Ideally, a Disco loudspeaker system would have the following attributes:

1. Rugged and reliable.
2. Acoustic output to meet market needs.
3. Low distortion (low fatigue).
4. Controlled coverage.
5. Efficient (power is expensive).
6. Compact (say 1 cubic foot).
7. Lightweight (20 pounds).

The last two items are the strong preference of the interior designer who usually does not want to see big, ugly loudspeakers intruding into the visual concept. Given the physics of planet earth and breathable atmosphere, it is apparent that 125 dB at 35 Hz will not issue forth from a 1 cubic foot box; so some intelligent compromise must be found.

Presume we have an existing room enclosing dancers on a dance floor. With the club market defined the first design task is to select suitable loudspeaker systems to meet the market sound level distributed over the required coverage area. We begin by selecting loudspeaker mounting locations which from prior experience we feel will meet the coverage demands while causing minimum detrimental impact on the decor. A typical single loudspeaker system location may then be analyzed to determine the acoustic output necessary to meet the selected market sound level on the dance floor and the electrical power required to achieve that acoustic output. The required source intensity is derived from the criteria level selected at the measuring location(s) on the dance floor with due consideration of the losses in the path from the source loudspeaker to the measuring location. A suitable adjustment is later made for the total number of sources operating to produce the total sound level on the dance floor.

The path loss calculation procedure follows the well known work of the Boners,[4] Don Davis,[5,6,7,8,9] George Augspurger,[10] Cecil Cable[11] and many others originally developed for analysis of sound reinforcement systems. A caution as to method is necessary: the basic equations for average sound energy density in a room are based on the assumptions of a non-directional sound source, $Q \equiv 1$, and boundary surfaces of the enclosure having uniformly distributed absorptive material with an average absorption coefficient, $\bar{\alpha}$. In clubs, the perimeter surfaces are generally acoustically "hard" while the occupied dance floor appears relatively "soft." To meet the previous "controlled distribution" requirement, directional loudspeaker arrays are then aimed at what may be the most absorptive surface in the room. Clearly, only a small portion of the incident energy is reflected to the boundary surfaces to create the reverberant field. A calculation method should be chosen which allows for the case where the absorption coefficient of the first reflection surface (the occupied dance floor) defined as α', differs from the average absorption coefficient, $\bar{\alpha}$.

In our own practice, we use the "adjusted room coefficient" method proposed by George Augspurger;[10] we find the scheme well suited to our method of tabulating the absorption coefficients and surface areas of various room surfaces. the "Q multiplier" or "architectural acoustic modifier" methods of Cecil Cable and Don Davis also yield appropriate results; the designer should choose whichever of the methods is convenient and familiar.

Thus, from Augspurger's analysis, in cases where "at least half of the energy from the sound source would seem to be initially reflected from surfaces having absorption coefficients differing substantially from $\bar{\alpha}$, the expression $(1 - \alpha')$ should be estimated as closely as possible and used to calculate R' "using the expression:

— *continued overleaf*

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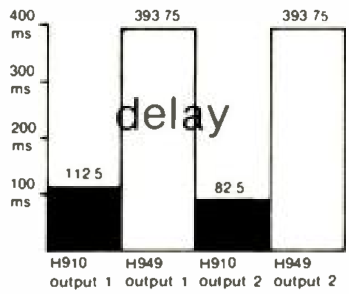

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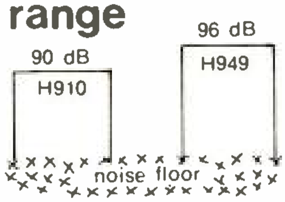
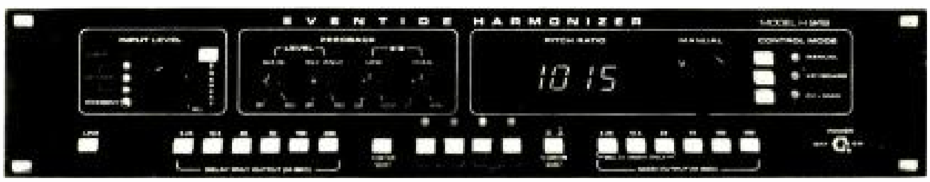


frequency response



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Ken Fause: DISCO SOUND SYSTEM DESIGN

$$R' = S\bar{\alpha}/(1 - \alpha') \qquad (1)$$

where:

- R' = adjusted room coefficient
- S = surface area of room
- $\bar{\alpha}$ = average absorption coefficient
- α' = absorption coefficient of first reflection surface.
- $1 - \alpha'$ = average first-order reflection coefficient

This expression is then used to modify the formulation of Boner and Boner.[4] In this case, the relative sound pressure level (SPL) in a room at any distance r from a source having a directivity factor Q in the direction of the observer is found by:

$$\text{relative SPL} = 10 \log [(Q/4\pi r^2) + (4/R')] \qquad (2)$$

where $R' = S\bar{\alpha}/(1 - \alpha')$

Returning to the path loss problem, we realize that a measurement of source level made directly at the source point, that is r = 0, will be meaningless — such a measurement will perforce be in the near field of any source device of finite dimensions. For this reason, it is appropriate to calculate path loss with respect to a reference distance D_{ref}, at which real loudspeakers may be measured and compared. Consider then the source-listener (dancer) geometry shown as Figure 14.

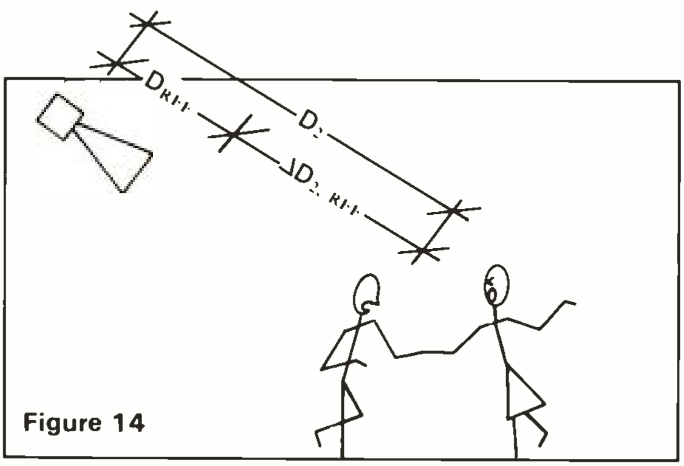

Figure 14

The path loss along the path from the reference distance to a distant listener, say at D_2, will be defined as delta D_{2ref}, where

$$\Delta D_{2ref} = \Delta D_2 - \Delta D_{ref} \qquad (3)$$

$$\Delta D_2 = 10 \text{ Log } [(Q/4\pi D_2^2) + (4/R')] \qquad (4)$$

$$\Delta D_{ref} = 10 \text{ Log } [(Q/4\pi D_{ref}^2) + (4/R')] \qquad (5)$$

$$R' = S\bar{\alpha}/(1 - \alpha') \qquad \text{(repeating 1)}$$

We must recall at this point that Q, $\bar{\alpha}$ and α' and thus path loss, are all functions of frequency. From experience, we know that most practical Disco sound systems will require three-way or four-way loudspeaker systems. To be rigorous, the preceding calculations should be performed for frequencies near the center of the bandpass of each transducer. Absorption coefficient data for architectural surface finish materials are generally published only at the ISO Preferred Octave Band Centers from 125 Hz to 4 kHz (i.e., 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz); calculations are thus limited to frequencies for which adequate base data are available. Unfortunately, when considering the behavior of sub-woofers; data at 125 Hz is usually at or near the top of the bandpass. In existing rooms, measurements of reverberation time in the octave bands at 63 Hz and 125 Hz will often provide sufficient trend information to complete the design. In new construction, the designer's experience and engineering estimates of acoustic material behavior at low frequencies must control.

Electrical Power Required (EPR)

Transposing equation (3), we know that:

(Desired Listener Level) + (Path Loss)
= (Required D ref level)

Desired listener level varies with frequency, for example as in Figure 11; path loss from equations (3, 4, and 5) also varies with frequency. Thus, in a final design; the required D ref level would be calculated for each band of a multi-way system. Recall also that:

$$\text{Power ratio in dB} \equiv 10 \text{ Log } (P_1/P_2)$$

Let us consider a major market Disco situation where we desire levels at the listener of 110 dB average and 120 dB peak. (This is a peak-to-average ratio of 10 dB; we would really prefer 15 to 20 dB, but the point is simpler to illustrate with round numbers). Let us also presume a path loss of 10 dB (not uncommon) and D ref sensitivity expressed in terms of 1 watt, 1 metre.

Consider a medium efficiency system with a D ref sensitivity of 100 dB (1 watt). From equation (6), the average EPR must then be 20 dB above reference power, and the peak EPR 30 dB above reference power.

Solving:

$$20 \text{ dB} = 10 \text{ Log [EPR avg/Ref Power (1 W)]}$$

which yields:

$$\text{EPR avg} = 10 \text{ Log}^{(20/10)} = 100 \text{ watts average}$$

for peak:

$$\text{EPR pk} = 10^{(30/10)} = 1{,}000 \text{ watts peak}$$

Figure 15
ELECTRICAL POWER REQUIRED (EPR)

LET: Path Loss = 10 dB
L_{REF} at 1 w., 1 meter
$L_{LISTENER}$: 110 dB Average, 120 dB Peak

L_{REF} (1 w.)	EPR, AVG. (110 dB)	EPR PEAK (120 dB)
100 dB	100 w. (+20)	1,000 w. (+30)
110 dB	10 w. (+10)	100 w. (+20)
90 dB	1,000 w. (+30)	10,000 w. (+40)

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Ken Fause: Disco Design

Clearly, these are not reasonable input powers for most practical loudspeakers. The calculations are tabulated in Figure 15 along with those for systems with 1 W outputs of 110 dB (high efficiency commercial horn) and 90 dB (typical home bookshelf system. Rapid inspection of the table shows why high efficiency systems are essential if high overhead ratios are attempted and why typical home loudspeakers are inadequate for nearly any Disco application.

Failure Modes

When operating loudspeakers at high power, it is necessary to consider failure modes of the loudspeaker. Thermal failure occurs when heating due to current flow in the voice coil is sufficient either to deform the coil by melting the binder material, or burn out the voice coil by melting the winding itself. It is pertinent to observe that heating is proportional to (1 – system efficiency). Suddently applied large currents may also destroy the loudspeaker by causing the moving system displacement to exceed its mechanical limits.

Loudspeaker protection for Disco application is similar to that for rock touring systems: Bandwidth restriction to the optimum operating region is most effective as practical loudspeaker output is limited by product of power and bandwidth. "Muddiness," voice coil heating and excess excursion are all reduced by removing input signals outside the useful bandpass of the transducer. Proper acoustic loading is also helpful in maintaining excursion within safe limits. In reference to an earlier discussion, it is essential to size power amplifiers large enough to avoid hard clipping on program peaks. Such clipping increases the transducer heating duty cycle in addition to sounding awful and contributing to listener fatigue.

If the designer was not already aware of the fact; the detailed design process would have led to a basic conclusion:

Loud + Low Frequency = HUGE

As Paul Klipsch has observed, there is no such thing as a miniature 32 foot wavelength. A common way to deal with this logistic and decorating problem begins with the realization that human hearing localization accuracy is much diminished at low frequencies. The signal information below say 100 or 125 Hz is summed from left and right channels and fed to a single large low frequency array (or a small number of arrays as architectural conditions require). A large array increases the radiation efficiency and floor mounting the array will double its effective Q — the floor is a large, stiff barrier compared to a wavelength. With frequencies limited to 100 or 125 Hz and up, reasonable size suspended arrays may be used for overhead, directional dance floor coverage. There is little artistic loss in using summed bass for Disco music: to achieve common Disco disc cutting levels the low frequency information is almost universally panned to center in mixing or summed to mono in the disc cutting process.

The specific design of hanging systems for suspended loudspeaker arrays is properly the province of an engineer or architect duly registered to practice in the jurisdiction where construction will take place. For suspension over human occupied areas, a static safety factor of 10 is often used in absence of specific code requirements. In active seismic zones, lateral restraint of suspended arrays is required. Suspension schemes should allow for sensible service access.

Power Amplifiers

Power requirements have been previously discussed. Reliable performance under continuous duty operation is essential. In addition, amplifiers for Disco service should exhibit subjective sound quality which would make them suitable for state-of-the-art recording studio monitoring systems.

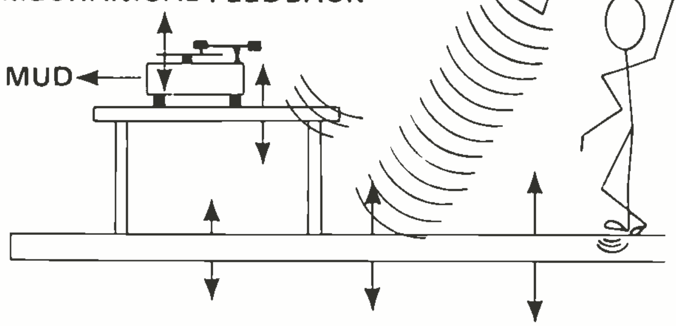
Turntable Feedback

When the airborne or structure borne radiated energy of the Disco sound system is coupled back into the active disc pickup cartridge, feedback occurs. As in sound reinforcement this establishes a limit on the system acoustic output level.

In the case of airborne excitation the most effective solution is simple: select a turntable/arm combination that is not susceptible to the problem!

A group of people dancing more or less on beat can induce considerable motion in a floor structure; high output sub-woofers may also cause reasonable amplitude diaphragmatic motion of floors, walls and ceilings. Such vibration induced into the turntable via its support structure constitutes mechanical feedback as shown in Figure 16. This should be

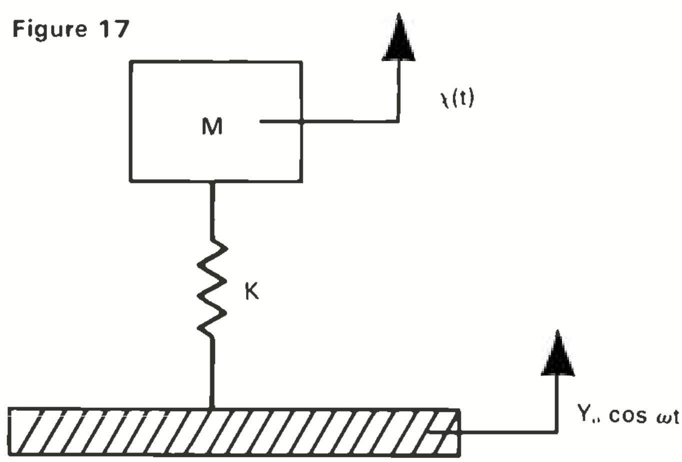
Figure 16
MECHANICAL FEEDBACK



considered even if system output level limiting is not a problem: low frequency excitation of the turntable will cause bass intermodulation distortion in its output[12] resulting in "muddy" sonic character.

A direction for solution lies in methods of long standing in Industrial and Mechanical Noise Control practice. Consider the situation shown in Figure 17. Here we have a mechanical

Figure 17



SYSTEM NATURAL FREQUENCY = ω_n

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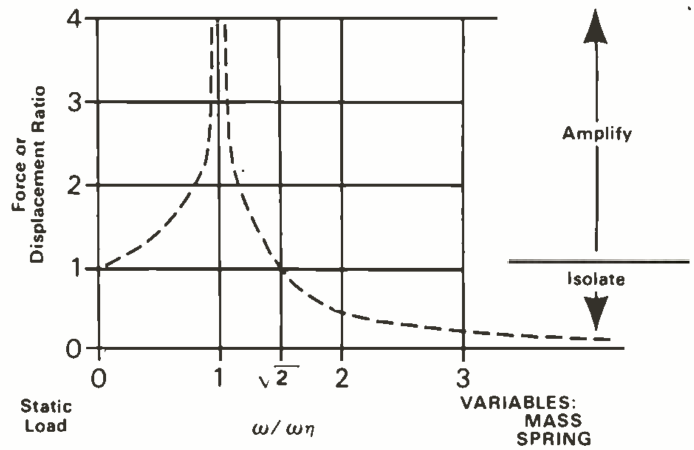
system with a single degree of freedom consisting of a mass m supported on a spring with a spring constant k. The base they are mounted on moves in simple harmonic motion at angular frequency ω with amplitude $y_0 \cos \omega t$. If the base is fixed, the spring-mass system will exhibit a natural frequency of ω_n. For this simple system, Figure 18 shows the transmissibility

Figure 18



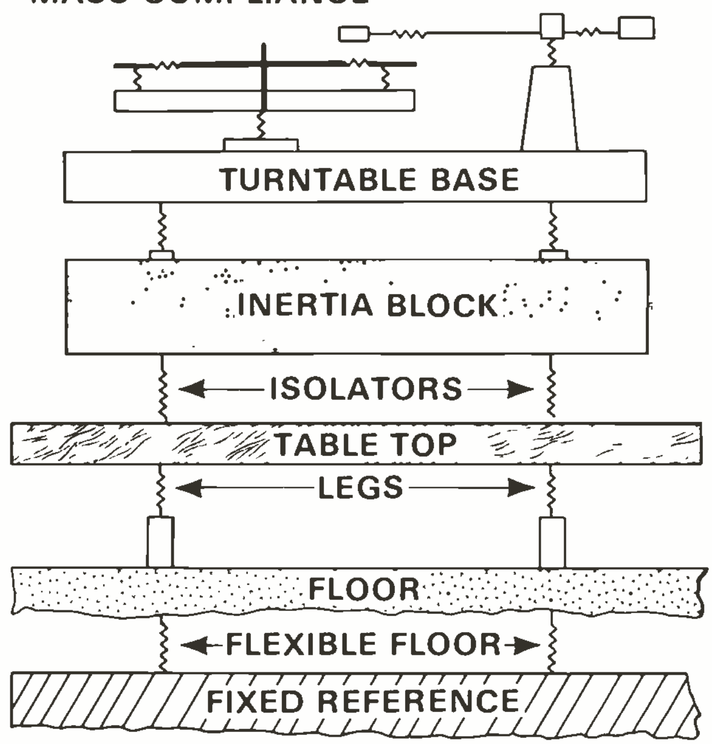
curve, the ratio of amplitude response of the system in steady state vibration to the excitation amplitude, shown as a function of the ratio of excitation frequency to natural frequency. If we manipulate the mass and spring constant such that the ratio ω/ω_n is greater than $\sqrt{2}$, then we may effectively isolate the mass from the motion of its support.

In a practical case we would have the turntable sitting on an inertia (mass) block which is mounted in turn on selected commercial vibration isolators.

We have designed specific isolation systems for a commercial turntable/arm combination to achieve a theoretical isolation efficiency of 98% at 20 Hz (approximately

Figure 19

MASS COMPLIANCE



17 dB insertion loss). We did not conduct instrumented measurements, but a brute-force test proved the point: we mounted a turntable/isolator combination directly on a system sub-woofer cabinet. The thermal limit of the woofer was reached before mechanical feedback could be achieved. We considered this performance to be adequate to the purpose.

Design of an isolator system is not a casual enterprise — it is quite easy for the unwary to get in trouble. The transmissibility curve shows that an error in tuning can cause amplification of the vibration. A practical system also exhibits multiple degrees of freedom of mass and compliance ("springiness") as Figure 19 shows; the isolators must be tuned to avoid driving any of the sub-systems into a resonance condition. Despite the cost of design and fabrication of the isolator bases, the scheme is usually more economical than the alternate of attempting to stiffen the DJ booth structure. In some cases, the structure simply cannot be made stiff enough. The isolator base is a truly effective and predictable approach to eliminating turntable mechanical feedback; we have yet to encounter a case where the scheme has not solved the problem.

Conclusion

Clearly, not every job will require use of all the detailed design methods outlined — in many cases, the system designer's experience with similar situations will provde all the information necessary. Where the job conditions are unusual or the arrangement lacks precedent in the designer's experience, then the engineering approach is extremely valuable: the designer, installers, owner and club customers are all rewarded with a Disco sound system that works properly the first time. □□□

Footnotes:

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3 - For a pure tone at 1 kHz, the loudness level expressed in Phons is defined as equal to the sound pressure level expressed in decibels. For other frequencies, the relation is defined by the Normal Equal Loudness Contours, Figure 13.

4 - C. P. Boner and R. E. Boner, **The Gain of a Sound System**, Journal Audio Engineering Soceity, Volume 17, No. 2 (April, 1969), pp. 147-150.

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8 - Don Davis, **Interpreting Field Measurements of Directivity Factor and Their Relation to the Proposed Standard Method of Measuring the Directivity Factor of Loudspeakers Used in Commercial Sound Work**, Pre-print #1031, Paper K-3 Presented at the 51st Convention of the Audio Engineering Society, May 13-16, 1975.

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IMPROVED INSTRUMENT TIMBRE THROUGH MICROPHONE PLACEMENT

by Wieslaw V. R. Woszczyk

Too often, choice of microphone placement is based on a personal preference with intuition presiding over factoral analysis. The subjective element cannot and should not be eliminated from the recording process. However, many microphone techniques unwittingly degrade the quality of sound, especially the natural timbre of an instrument. This article analyzes some of the factors which affect the timbral quality of recorded instrument sounds and practical data for the recording engineer using typical equipment in a studio environment. Experiments with a close multi-microphone system document an alternative miking technique that captures both the presence and full spectral content of a musical instrument.

- The Author -

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The author has worked at a number of commercial studios, including Big Apple Studios in New York City, where his credits include work with Harry Belafonte, Gloria Gaynor, Brian Eno, and George NeCree, among others.

Recently his classical recordings won the Grand Prix du Disque in Canada.

Directional Characteristics Of Instruments

Musical instruments are often regarded as loudspeakers that should have well defined directional response patterns with the optimum direction of their full spectral performance on axis. On the contrary, most musical instruments are very complex radiators which project sound energy multidirectionally, in different and constantly changing proportions of spectral density. Every sample of the direct sound picked up by a single receiver placed anywhere on a spherical surface around the instrument will differ. An individual sample does not have a spectrum of exactly the same nature as the total power spectrum of the instrument which is an average of the infinite number of samples creating complete field surrounding the instrument.

This total spectrum can be measured in reverberant rooms, which integrate acoustically the total output of the instrument. In anechoic rooms intensity readings of many samples would have to be taken on the sphere surrounding the instrument, then integrated, and a calculated average would produce the total spectrum.

Knowing that the method of extracting the instrument's total spectrum is quite different in reverberant and anechoic environments, we could presume that the environment plays an important part as integrator and carrier of the instrument's timbral information.

A short survey of the radiation properties of various instruments is provided on page

Function Of Acoustic Environment

A standard engineering practice involves finding a particular acoustic environment for recording a particular instrument. The acoustic character of certain areas in a room influence the acoustic output of various instruments. The timbre of an instrument in a reverberant environment is the result of the interaction between the

direct and reverberant sound fields. Reverberant sound contains a large proportion of lower frequency energy radiated by the instrument in directions other than the one of maximum output in the "presence" range. Most orchestral instruments are built to be more or less directional and have an axis of preferred radiation with a "presence boost" which enables them to project their sound to the back of the concert hall. This provides a desirable ratio of direct sound and reverberant sound to the audience in the concert hall. Almost all instruments radiate their low frequencies omni-directionally so the participation of low frequencies in the reverberant field is much greater than in the direct field. Thus, the reverberant processing by a room adds warmth and fullness to a sound. To achieve a fuller sound with more varied harmonics and substantial energy in the low end of the spectrum a microphone should be placed at some distance from the instrument to allow the room to collect and remix acoustically the multi-directional sound ingredients. If closer microphone placement is utilized the pick-up becomes selective and less adequately expresses the instrument's output.

In a concert hall, large reflective surfaces around the stage area have a strong influence on the definition and timbre of orchestral instruments. Side walls, front and back parts of the ceiling, and the back wall redirect large portions of sound energy into the audience and onto the stage area, presenting additional timbre and loudness information. Without these surfaces, as in 360 degree concert halls, there are problems in achieving a fullness of sound, flow of musical events, proper timbre balance, and loudness. In such halls the spectral balance of the orchestra is different in every listening position because the instruments are placed in a more or less free-field environment. Increased efficiency in the rear and side radiations through the work of reflective surfaces will result in a fuller and richer instrumental timbre quality perceived by the audience. The reflective surfaces are especially important for the woodwind instruments which have definite multi-directional patterns.

One strong reflection may substitute for reverberation if the instrument is placed next to a reflective surface to redirect the rear radiation of the instrument toward the microphone with a slight delay. (Figure 1) However, the rear radiation must have a substantially different spectral energy combination than the front radiation or audible phase cancellation may occur. This is not often a problem since the output energy spectrum of most instruments changes heavily with direction. The constantly fluctuating spectrum of most musical instruments reduces the probability of audible phase interference. The floor is the largest and closest reflective surface to the instrument and its effect on timbre quality is the strongest. Questioning the existence of this reflective surface so close to the source and calculating its degree of phase interference is impractical. Reflected sound and some phase interference are integral parts of the sound of a musical instrument played in a natural environment. If the reflective surface is located opposite the microphone on the other side of a complex radiator the direct and reflected signals at the microphone have less similar spectra.

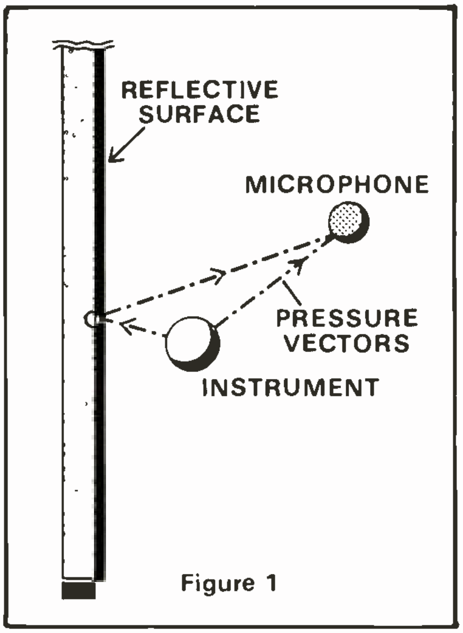

Figure 1

Few recording studio rooms provide useful acoustical assistance for directing sound radiation to control the timbre of musical instruments. Rare exceptions have hardwood floors, adjustable reflective panels below high ceilings, and adjustable absorption side walls. Most of the currently popular recording environments approach anechoic chamber rather than concert hall specifications in their average reverberation time and in their use of reflective surfaces. In an acoustically damped environment most of the energy of the instrument's sound is wasted because most of its radiation is absorbed, except for the small portion of the spherical radiation pattern occupied by the membrane of the microphone. The special characteristics of that wasted energy are not used to add to the spectral information of the instrument. A microphone placed in a dead room lacks the additional information which plays a vital role in the recognition of an instrument's timbral character, and can give a misleading interpretation of the instrument's total power spectrum. The absorptive environment most frequently found in a recording studio also has the tendency to process the energy information in the frequency domain, attenuating various frequencies in different proportions. This environment degrades the timbre information when distant microphone techniques are used.

Close Miking With A Single Microphone

It is common practice today to severely restrict the full spherical radiation of instruments with the extensive use of directional microphones. Usually one cardioid microphone is placed close to the instrument, in the direction of the instrument's most efficient sound production in the "presence" frequency range. A closely placed microphone picks up, to a great extent, only the partials of the sound which are directly radiated at the microphone and may completely miss sound components sent in other directions. A single microphone closely placed can only sample one part of the frequency dependent polar pattern of an instrument.

In close miking the placement is very critical for spectrum equalizing. A microphone placed close to an instrument samples the radiation, achieving a particular equalization of the instrument's power spectrum. In this case, electronic equalization is often used to compensate for defects, but it can only change the relative balance of energy in three or four places on the spectrum. Electronic equalization does not introduce any new harmonics which were missed by the microphone, or any other elements that would change the existing balance properly and increase the amount of information about the instrument. The complexity of sound is not matched by the correcting abilities of an equalizer.

A single microphone placed close and on the axis of an instrument generally produces a very hard, concrete sound which is especially pronounced on directionally radiated parts of the spectrum. It is necessary to have the clarity, immediacy, transient response, spectrum and dynamic range as received by the close miked sound received on axis pick-up. But it is also necessary to have the richness of all the harmonics produced by the instrument, the fullness from low frequency components, and the natural spectral balance. The ongoing "audio purist" campaign for a return to the simplicity that ostensibly captures the "live" sound of the concert hall substantiates the necessity for a more natural timbre in recordings.

Let us examine some typical close placements of microphones applied to various acoustical instruments and consider the extent of the timbre weighting the single-close-microphone technique achieves in each case.

The typical placement of a microphone used in recording a brass instrument is at the bell. Brass instruments extend their spectra toward high frequencies when they

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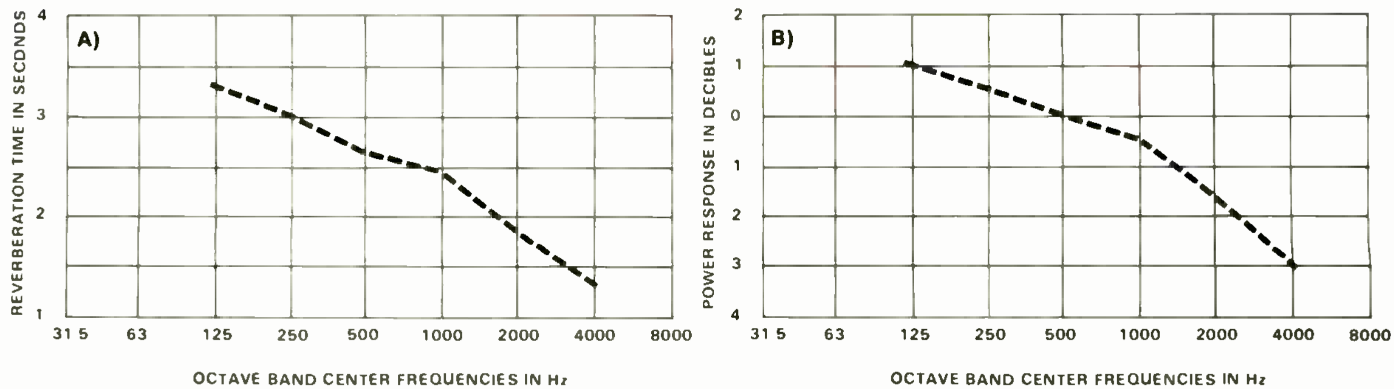

Figure 2: The empty concert hall: A) The measured reverberation time. B) The relative power response calculated from the measured reverberation time.

are played louder and narrow considerably when played softer, while the intensities in the main formant region remain at an almost equal level. This increase in amplitude of high frequency partials produces a large subjective treble boost. The very high frequencies are radiated in a narrow beam. Further, they do not reach the concert hall listener directly for a period of long time and are usually not heard after some reflections. (Figure 2) But these high frequencies certainly reach the microphone placed at the bell. In this position a microphone can over-estimate the relative level of the higher harmonics by as much as 25 dB, based on comparison with the results of acoustic averaging of power spectrum in reverberant rooms. A pressure-gradient single-diaphragm microphone placed close to the source produces a low frequency proximity boost that might counterbalance the exaggerated pick up in the upper parts of the spectrum. Unfortunately, the frequency of maximum boost caused by the proximity effect is often around 100 Hz (depending on the microphone type) and is too low to effectively increase the energy of the lowest partials of trumpet sounds in the loud register. Also, proximity remains when it is not needed, in softer passages where higher partials do not overbalance the lower partials. Ribbon microphones are appreciated for their high-cut, low-boost response which helps to recover the natural balance of the closely miked trumpet.

The close microphone placement for woodwinds, particularly the saxophone, is also at the bell. This placement picks up an exaggerated amount of high frequency components and lacks the fullness of a low frequency energy that would balance the spectrum. It totally misses the lower partials which radiate primarily through the open tone holes on the wall of the conical tube. The lowest partials of every tone on the instrument, except the lowest two in each register, are radiated through the first open hole below the mouthpiece. Instead of the full resonance of the vibrating modes of the air column the microphone placed at the bell gets the information describing the movement of the reed in the mouthpiece. The timbre of the saxophone on many current records is very electronic and shallow, and has mixed-in echo effects which aggravate this quality.

A single microphone placed over the bridge of a bowed string instrument does not collect the radiations of the instrument in proportions which correspond to its complete sound output. On the violin the resonant frequencies of the top and bottom plates should complement each other and should differ, not coincide. This shared radiation in the frequency domain insures maximum output for every harmonic of

— *continued overleaf*

DIRECTIONAL PROPERTIES OF INSTRUMENTS

Musical instruments are often regarded as loudspeakers that have well defined directional response patterns and optimum performance on-axis. On the contrary, musical instruments are very complicated multi-directional radiators.

The following survey of the radiation properties of various instruments is essential information for studies in microphone placement.

Bowed string instruments have exceedingly complex directivity patterns which are different for every partial and alter dramatically with any change in frequency. Numerous resonances of the top and bottom plates reinforce the harmonics of the string vibrations in a uniform and effective way throughout the usable frequency range. On the average, the violin radiates its low frequency components below 400 Hz omni-directionally. The preferred radiation of partials around 400 - 500 Hz is concentrated toward the back of the instrument. This is an important frequency range which gives the instrument the deep and dark singing quality of the vowel "o." Components around 700 Hz are better radiated toward the right side, over the top plate, and partials around 1,000 Hz toward the left. Partials around 1,500 Hz tend to radiate both right and left but less directly at the center above the top plate. All of the components above 1,000 Hz have larger amplitudes over the top plate of the instrument than below the bottom plate, therefore the frequency range which gives the timbre the open, lively quality of the vowel "a" (1,000 to 1,250 Hz) is found here. Sounds above 3,000 Hz exhibit many narrow angles of preferred radiation. The sound above the instrument is brilliant and lively and below has a deep dark quality. The radiation of celli and contrabasses is also of a multi-directional nature. Celli radiate the partials below 200 Hz omni-directionally. In the 200 Hz region and between 350 Hz and 500 Hz the frontal radiation is stronger. At 800 Hz a cello radiates more sound upwards in the direction of the neck, while components between 1,000 Hz and 1,250 Hz are best captured in front of the neck. Contrabasses radiate even the lowest frequency partials in a semi-circle around the front plate and upper frequency components are directed toward the front and back of the instrument.

Most of the sound from a woodwind instrument is radiated through the open tone holes. Only when all the tone holes are closed to obtain the lowest notes in each register is the radiation from the bell complete. The lowest partials of the woodwind sound are emitted from the first open hole below the mouthpiece. High frequency components are produced by a greater number of open tone holes and the highest components are best radiated by all of the holes and the bell of the instrument. The wavelength of any sound component radiated through the tone hole is much longer than the diameter of the hole and the sound is therefore diffracted. When measured by a distant mirophone all but the highest partials, which are radiated most effectively and directionally by the bell, appear to be radiated equally in all directions. However, a closely placed microphone will reveal small, sharp directional patterns which quickly alter with every pitch produced by the instrument. The proximity of the microphone to a certain tone hole will greatly emphasize a particular fundamental and the next few partials. The reedy quality of the woodwind instrument is produced by the higher partials radiated best through the bell. However, if the bell of the clarinet was stopped with a large cork the tone quality would not, except for a few notes, change substantially because the radiations through the tone holes sufficiently describe the instrument.

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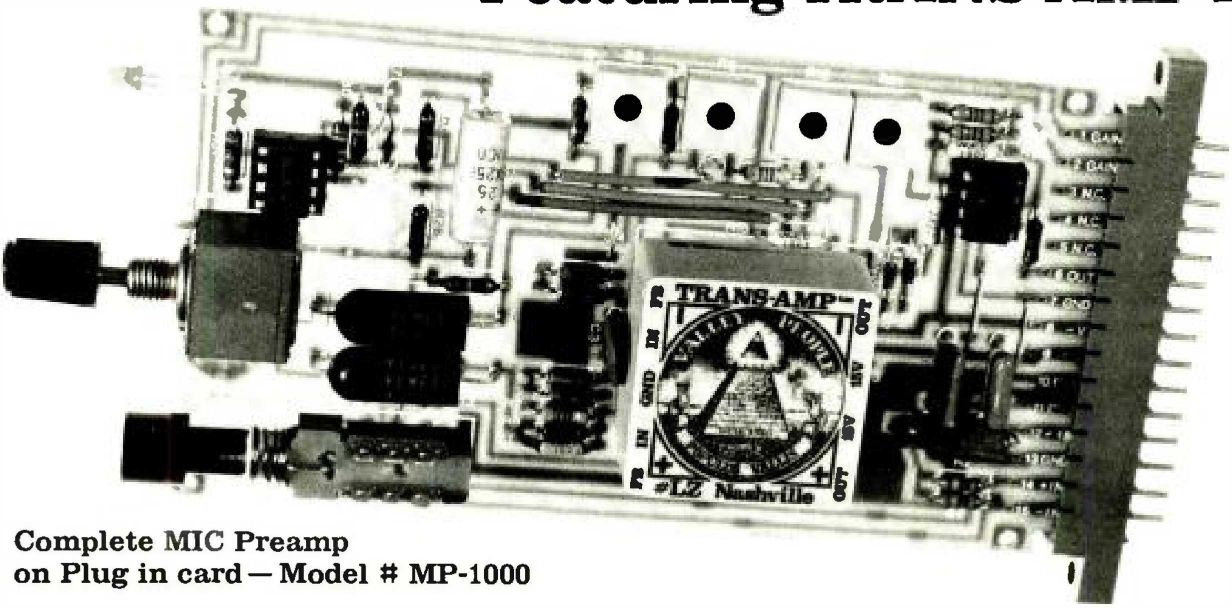

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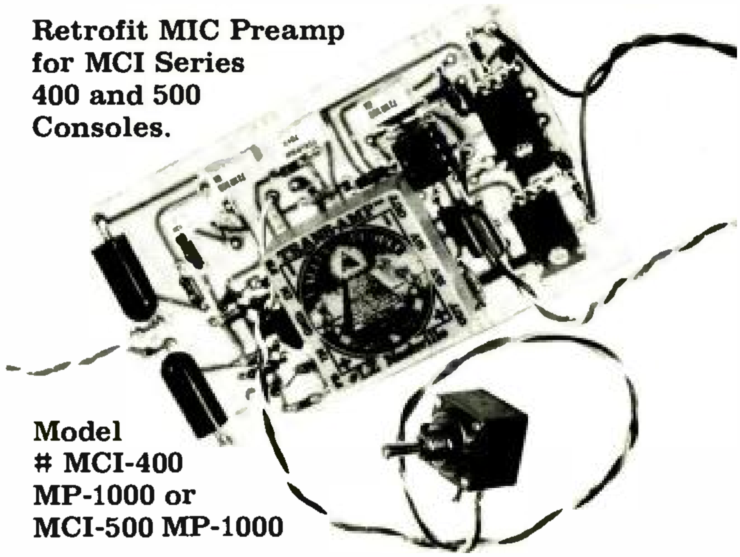

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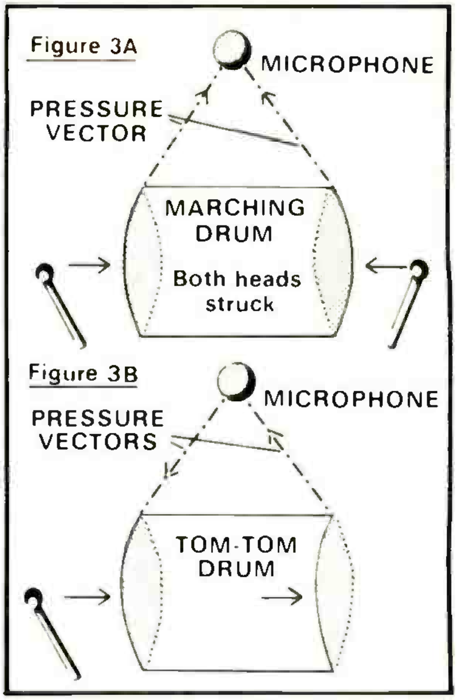
string vibration and indicates that very different sound qualities are produced by each side of the instrument. The bottom plate has a strong resonance around 500 Hz and the top plate has a better radiation of the higher partials so it sounds brighter and more nasal. The placement of a microphone above and close to the bridge emphasizes the tonal quality of the top plate. Some of the bottom plate radiation finds its way to the top through the "f" holes but is insufficient because of the baffle effect of the top plate. This microphone placement also emphasizes the sound of the sticking and slipping action of the bow on the string and the subsequent vibration of the bridge. As a result, the microphone will produce a vigorous, over-bright, and nasal sound quality lacking the full and dark timbre of the bottom plate.

A microphone placed close to the upper membrane of a tom tom with both membranes on will not pick up any of the partials of the lower membrane except the fundamental, which is coupled to the upper membrane through the air column. The pick up of the upper membrane's frequency components will be very detailed, but the balance is dependent on the placement of the microphone between the center and the edge of the membrane, or outside the rim. A microphone placed over the center of the membrane will pick up the fundamental more than the upper partials. A microphone placed slightly outside the membrane edge emphasizes the highest partials, decreases the fundamental, and increases the possibility of the lower membrane radiation reaching the microphone. If the membrane of the drum is of considerable size compared to the diaphragm of the microphone, as in the case of a bass drum, close placement of the microphone is extremely critical. Figure 3A shows acoustic summing while 3B illustrates acoustic cancellation

Place any of the above instruments in a reverberant environment and withdraw the microphone to the far field and the spectrum received would more closely resemble the total power spectrum of that instrument. The appropriate balance of many distinct sound fields could be achieved acoustically through diffusion in the air and reflections from the room boundaries. However, the positive qualities of the close microphone, the impact of the attack, the detail, and the dynamic range would be lost. The coloration by the particular room response and leakage from other instrument fields could impair the quality and the separation needed for the flexibility of independent timbre control. The particular sound quality and operational advantages of close microphone placement make the method attractive and easy to apply. The natural timbre distortions and simplifications caused by spot-sampling the acoustical field are largely overlooked. The lost information from the complete spherical radiation of the instrument is compensated for by an increase in transient information from the source near the microphone. Equalizers and microphone choice attempt to balance the spectrum, but do not add any information to it.



Close Miking With Multiple Microphones

The advantages of both close and distant microphone placements could be utilized by discovering where a minimum number of microphones could be located that would allow a reasonable averaging of the external spectrum for the most common tones of the instrument. A reasonable approximation of this average spectrum could be produced by combining the outputs of several microphones from the best samples.

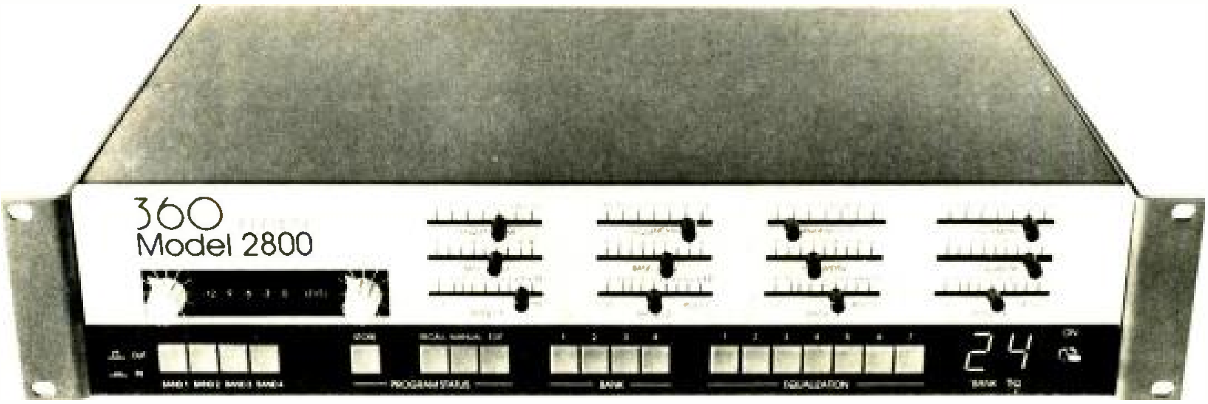
Through independent pick up and mixing of characteristic spectral qualities of the instrument, natural equalization of a sound spectrum could be accomplished. With this type of equalizing the spectrum does not merely change the relative magnitudes in one or two places on the spectrum. It uses resonant areas (formants) or reinforced groups of partials that are produced and radiated by the instrument in different directions, and which complement each other in reconstructing the desired total spectrum. If a number of single partials are weak in one microphone, they may be provided by another, hence a proper balance can be reached, a process not possible through electronic equalization. The changing spectral information produced during performance by the instrument preserves the characteristics of

— continued overleaf

DIRECTIONAL PROPERTIES OF INSTRUMENTS

Brass instruments have more unidirectional radiation characteristics than woodwinds. In their lowest frequency components the radiation pattern of brass instruments is omni-directional because the directional function of the bell is not effective. The omni-directional radiation characteristics become more directional with increasing pitch and develop side radiations which show adjacent peaks and dips on a polar diagram. The directional characteristics begin at different frequencies and depend on the bell diameter, the length of conical tubing, and the bore size. In trumpets side radiations begin above 600 Hz, in trombones above 450 Hz, and in tubas around 80 Hz. As the frequency rises the number of peaks increases but their relative intensity diminishes, and the radiated energy concentrates on axis into a still narrower angle. Side radiation between 2,000 and 5,000 Hz has 15 to 25 dB less energy than on axis radiation. The important partials of the trombone's distinctive timbre, around 700 Hz, are best radiated to the sides. Also, the partials of the trumpet around 1,200 Hz are side radiated and give this instrument its loudness and tonal character.

The sound and directional characteristics of drums are produced by two systems, the air-coupled vibration of the fundamental vibrations of the two membranes and the normal modes of vibration of the struck and passive membranes. The average directionality of the fundamental frequency, affected by the length of the body, corresponds roughly to a figure-eight pattern with the average sound intensity measured on the line that perpendicularly bisects the axis of the drum heads, equal to one-half the intensity measured on either of the two skin-heads of the drum. At any given instant the two membranes coupled through the air inside the drum can vibrate in phase with an omni-directional characteristic, out of phase in a figure-eight pattern, or can vibrate in any other phase relationship. Depending on the ratio of the fundamental frequencies of the two membranes, the changing phase relationship causes the intensity of the vibration to increase alternately in each membrane and exchange the energy of the vibration from one membrane to the other. This coupling is effective only at the fundamental frequency of vibration and does not affect the higher partials. The higher partials are radiated independently by the two membranes and are affected by the uniformity of the membrane thickness and its tension around the rim. The fundamental is radiated most efficiently at the center of the membrane while the higher partials are radiated off center and closer to the rim. Higher partials appear to be radiated omni-directionally when measured by a distant microphone since the vibrating parts of a membrane surface are small compared to the wavelengths they radiate. A closely directional microphone can pick up the different average balances of spectral energy separately on each membrane, or different balances of the upper and lower partials depending on the distance of the microphone from the membrane center, the rim, or outside the rim. □□□



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the instrument rather than that of the equalizer.

Some experiments were designed to test the ability of the close placement of multiple microphones to combine the advantages of close miking with the timbre achieved by distant miking. The objective of the experiments was to find a means of recovering the natural timbral balance which is lost in single close miking. The pick up of middle and low frequency energy radiated by the instrument off-axis was of primary interest, rather than the presence range energy which is radiated on-axis.

The experimental method used a multi-track recording of two instruments in three different acoustic environments. Microphone signals on the tape were mixed, spectral density analyses were taken, and several listening tests were conducted. The spectral density analysis emphasized the importance of the lowest harmonics that are largely responsible for creating an impression of fullness and the characteristic timbre of an instrument.

Spectrum averages of single notes as well as running scale passages were made to show the emphasis of particular microphone placements in certain harmonic regions. The captured samples of radiation have different characteristic spectral energy distributions. A graph comparison gives an idea of the polar radiation pattern of an instrument.

The baritone saxophone and the viola were selected to represent woodwinds and bowed stringed instruments, groups whose timbre is most affected by close microphone pick up. This is due to their complex, multi-directional radiation characteristics. The baritone saxophone is a large instrument with the bell and tone holes well separated from each other, permitting the use of more than one microphone to pick up distinctive sound fields which could demonstrate the influence of microphone placement on tonal character. It is also rich in partials and the lack or excess of important partials has a marked effect on the spectral energy balance and audible timbre. The viola, though smaller, is also multi-directional and the principles of multi-miking can be similarly applied to other instruments of the violin family.

Four microphones were arranged around the saxophone as shown in Figure 4. All microphones used in the experiments were Neumann U87s on cardioid position. The first microphone was placed in a common recording position about six inches from the bell of the instrument. The second microphone was placed near the largest open holes on the conical tube, about one foot below the bell. The third microphone faced the upper open holes close to the musician's left side, and the fourth was placed at the open holes, midway on the musician's right side and slightly behind the instrument. The signals were recorded simultaneously on separate tracks for the future spectral and aural analysis and flexibility in mixing. No electronic equalization was used, so all differences in spectrum and timbre were caused by the microphone placement on the polar pattern of the instrument.

Figure 4

The performer stood in the middle of a large room, about 20 feet from the nearest wall. The ceilings are high, the floor and walls carpeted. The average reverberation time is approximately half a second, corresponding to a typical studio environment. He was asked to play forte the most frequently used range consisting of Eb2, G2, Bb2, D3, F3, A3, C4 and Eb4. These are the fundamentals from 77 Hz to 311 Hz. Every note of the scale was played separately for five to seven seconds and then the scale was played legato, up and down a few times, at approximately two notes for every 1.5 seconds.

The same performer repeated the exercise in an empty 600 seat concert hall that is rectangular and has a T_{60} of 2.6 seconds at 500 Hz. The measured reverberation time and the relative power response determined direction from the reverberation time figures as presented in Figure 2. The musician stood in the center of the stage. An omni-directional Neumann U87 was placed in the reverberant far-field, the audience area about 65 feet from the performer, and its output was recorded on a separate track of the multi-track tape. Here the acoustical method of averaging the instrument's spectrum was employed.

In the third recording situation the musician continued the same exercise in a large, empty, reverberant room with concrete floor and walls, and an absorptive ceiling. The sound output of the instrument was averaged acoustically and picked up by a single omni-directional microphone placed asymetrically in the room. The musician moved around while playing to excite more room modes and increase the statistical smoothing effect of many simultaneously excited modes, producing a good average of the radiation behavior of the instrument. The output of the microphone was recorded on a separate track of the multi-track tape.

The experiments with the viola followed the same procedures but used a different microphone placement and different musical material. The microphones were arranged as seen in Figure 5. The first microphone was placed 12 inches above the top plate over the bridge, the second placed below the bottom plate at a similar distance and facing up, the third microphone was positioned above and behind the chin rest about 18 inches from the instrument, and the fourth was placed above and in front of the neck about two feet from the bridge. The viola scale consisted of D3, F#3, A3, C#4, D4, F#4, A4, C#5 and D5, the range of fundamentals from 146 Hz to 587 Hz.

Figure 5

A Hewlett-Packard Digital Signal Analyzer model 5420A was used with the plotter to measure and plot the spectrum density. (Figure 6) Signal averaging was used to reduce random components and emphasize coherent static components. Drawings of energy concentrations helped verify the extent of weighting that a particular mike placement had on the received spectrum of the instrument's total output. Twenty samples were taken in the time domain of a 3.6 second sustain of every individual note. Every time sample was 20 ms long and consisted of 256 data points. A new sample was taken every 160 ms in the "fast rate" for individual notes, and every 800 ms in the normal rate for complete scales. The frequency domain measurements were derived by the analyzer by taking the Fourier transform of time domain measurements. An ensemble of 256 uniformly spaced (every 50 Hz) frequency domain samples was created throughout the chosen bandwidth from 50 Hz to 12,800 Hz. The results of the power spectral density analysis were displayed and plotted on a logarithmic frequency scale and a linear magnitude scale. The linear magnitude scale shows more clearly the energy relationships between the lower partials which are the most critical for perceived timbre. It expands the differences in intensities between the meaningful half-dozen or so

— continued on page 91 . . .

Figure 6

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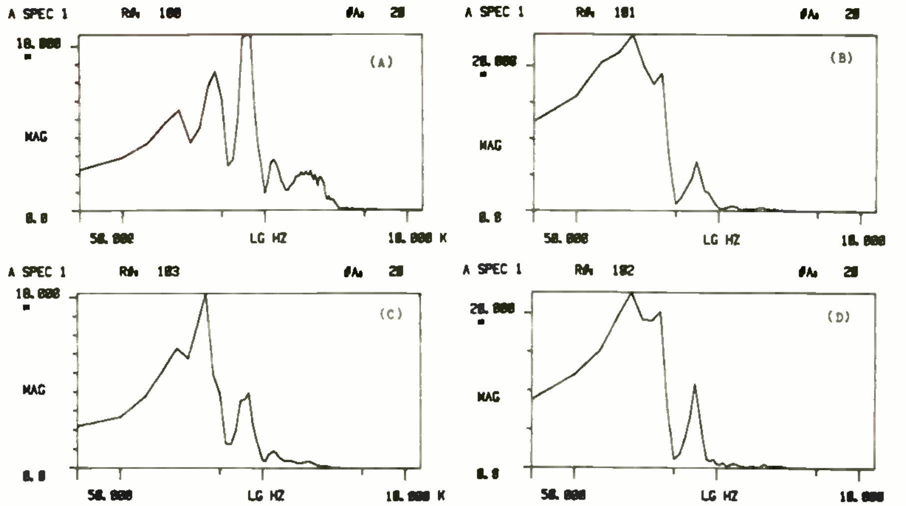

Figure 7. The Baritone Saxophone Eb2 (77 Hz): The averaged spectrum density analysis of the signal received from each microphone as seen in Figure 3. A) At the bell, B) at the open holes below the bell, C) at the open holes on the musician's left side, D) at the open holes on the musician's right side.

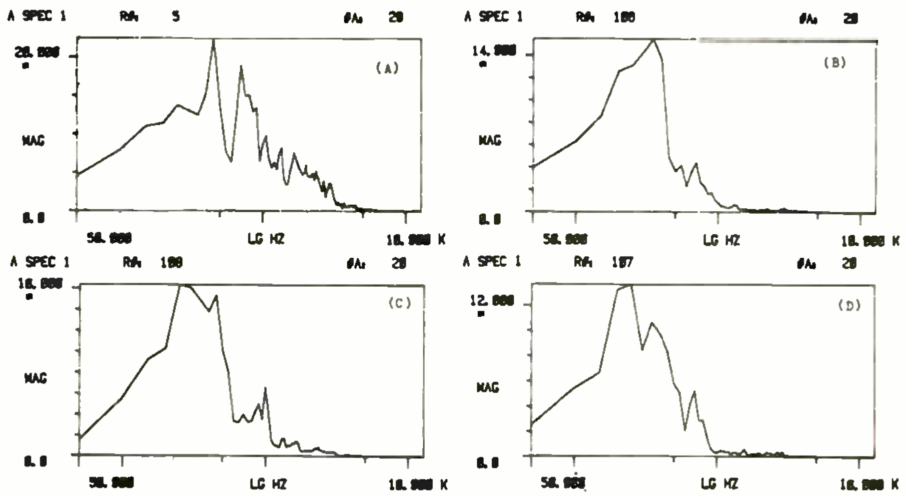

Figure 8. The Baritone Saxophone Scale: The long-time averaged spectrum density analysis of the signal received from each microphone as seen in Figure 4. A) At the bell, B) at the open holes below the bell, C) at the open holes on the musician's left side, D) at the open holes on the musician's right side.

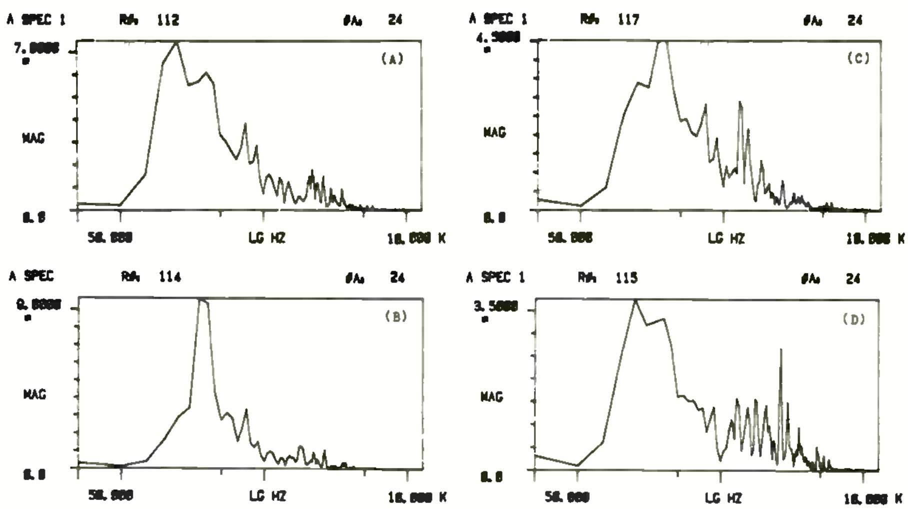

Figure 9. The Viola Scale: The long-time averaged spectral density analysis of the signal received from each microphone seen in Figure 5. A) Above the top plate, B) below the bottom plate, C) above and behind the chin rest, D) above and in front of the neck.

low partials. Each one of the lower partials belongs to a different critical band and is processed for loudness separately by the ear. The loudness of all the critical bands added together provides low-frequency perception of timbre. The lack of energy of a particular partial will prevent one of these critical bands from contributing its loudness to the timbre formation.

After the spectral density measurements of individual notes was made the analysis of the close microphones was compared to the total power spectrum averaged acoustically in the reverberant room and the concert hall. Long time averaging was used and twenty time samples were taken during 16 seconds of the running scale.

The averaged spectral density analysis of the four microphones placed around the saxophone (playing Eb2, 77 Hz) is presented in Figure 7. The complete eight-note scale analysis is shown in Figure 8. The microphone at the bell receives a sound spectrum with strong formant areas around 450 Hz, 800 Hz, and several resonant areas in the range of 1,000 to 3,000 Hz. In the subjective listening tests this microphone's sound was described as harsh, flat and extensively filtered. The listeners did not recognize in this sound the characteristic timbre of the baritone saxophone. The other microphones produced spectra which were very rich in low frequency partials. The two microphones placed at the open holes of the instrument reveal spectra containing the lowest partials below 500 Hz — with peaks at 250 and 300 Hz, and very small energy content above 1,000 Hz.

Figure 9 compares the long-time averaged spectral density analysis of the signal received from each of the four microphones around the viola. The sound picked up by the microphone placed below the bottom plate has a very distinctive color because of a high formant peak around 400 Hz, peaks at 550 Hz and 700 Hz, and a relatively small amount of energy above 1,000 Hz. The microphone behind and above the chin rest picked-up strong resonances around 1600 Hz that brighten the sound. The mike above and in front of the neck reproduced a series of wide resonances from 1,000 Hz to 4,000 Hz with a sharp peak at 3,000 Hz. This produced an over-bright timbre that was harsh and had a strong bowing quality. These varied timbres provided good material for balancing.

In conjunction with the four microphone arrangement a more practical mike set up that approximated the sound of the first set-up was tested. Two microphones were placed around the baritone saxophone as shown in Figure 10. One microphone was positioned at the bell but slightly above it to pick up some of the radiation from the open holes close to the mouthpiece. The second microphone was placed on the right side of the performer below the bell and close to the

Figure 10

open holes. The two microphone arrangement for the viola as shown in Figure 11 has one microphone above the top plate and toward the chin rest and the other below the bottom plate and toward the neck. The outputs of the two microphones were combined electrically in the desired proportions and spectrally analyzed for the single notes and the running scale. The output levels were first set to the optimum audible balance throughout the scale and remained in that position throughout the measurement procedure. These analyses were compared to the previous diagrams of independent microphones.

Phase Interference

To find the extent of phase interference between the common components of the combined microphone signals the averaged spectrum density analysis of the separate microphone signals were summed arithmetically and compared with the diagram of the electrically combined pair. Spectrum addition (summing) is done graphically and does not take into account the phase relationship of the signals the spectra represent. Electrical mixing produces phase interference which appears in the averaged spectrum density analysis as a stable component.

The slight differences between the spectra derived though arithmetic addition and those from electrical mixing indicate that there is some phase interference in the signals. However, if every microphone is placed over a different characteristic field of radiation, phase interference will be kept below the level of subjective detectability.

The multi-microphone method demands some precision. Linear distortion is audible as timbre coloration and can result from phase interference between the combined signals of individual microphones. The potential for distortion when using this method is dependent on the degree of similarity between the microphone signals in each band of frequencies. To test this, divide the individual microphone signals into contiguous frequency bands and measure the cross correlation of their corresponding outputs. As an alternative, a correlation meter comparing the outputs of the combined microphones can be used. This device indicates the average sum of in and out of phase components usually within the frequency band between 50 Hz to 15 kHz over the time of 1.0 second. An oscilloscope responding instantaneously to phase relationship is also useful.

If the signals contain few frequency components in common their phase relationship will be random and they will behave as independent signals that do not degrade each other when mixed. Microphone placements which pick up different timbres will not have many same frequencies with similar magnitudes, so the possibility of phase interference is decreased. Timbre differentiation is more likely if the microphones are placed close to sound radiators which are physically further apart and radiate directionally. Ideally, the ratio of the distance between two microphones and the distance of the closest microphone to the source should be more than 3:1. Interference is easily avoided on large instruments, especially those that vary in timbre in places around it. For example, it is easier to find unlike sound components on a bass drum of 26" diameter than one of 20." The fundamental frequency has its maximum amplitude in the center, while the higher partials are found nearer the rim. In the violin family the baffle effect of the instrument to itself contributes to different sound fields.

Instrument design prevents phase cancellations that would make it less efficient. Some frequencies are radiated most strongly in certain directions, and higher frequencies are radiated directionally. Some instruments, an open pipe organ or a drum, for example, have their sound sources separated by a distance which decreases the degree of effective phase cancellation from sources that vibrate out of phase.

Separation or random correlation between two signals can also be increased electronically. A low-pass filter applied to one signal and a high-pass to the other will decrease the number of common frequency components. Varied microphones will lessen the chances for critical partials to be of comparable magnitude. Phase interference is not more apparent in an electronic mix of near-field derived signals containing distinctive spectral characteristics than in an acoustic mix obtained in the far-field.

It is important to remember that phase interference is largely responsible for the particular timbre of many instruments. It plays an important part in shaping the sounds of instruments in a room or next to a single reflective surface, the floor. In practice, phase interference is always contributing to the sound we perceive.

Figure 11

The two microphone set up for the saxophone shown in Figure 10 and illustrated in Figure 12 proved to have

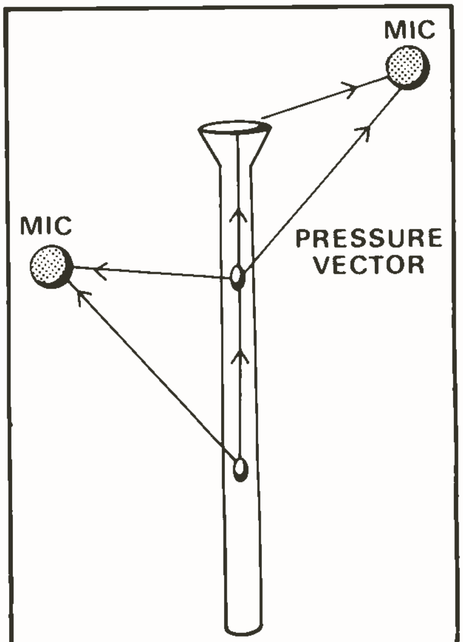

Figure 12: Examples of phase interference in musical acoustics: The interpath delay at the receiver between the open holes and the bell radiation of same frequency partials of a woodwind instrument.

enough flexibility for timbre control. The microphone at the bell showed a peak at 700 - 800 Hz that dominates the spectral balance of this arrangement. Diminished energy in the low frequencies and the relatively high energy of the partials between 1,000 Hz and 3,000 Hz gave the timbre presence but lacked fullness and warmth. The side microphone in this setup receives substantially more energy below 500 Hz. The averaged spectral density analyses derived from the two microphone arrangement are illustrated by Figures 13 (Eb2, 77 Hz), 14 (G2, 98 Hz), and 15 (F3, 175 Hz).

Figures 16, 17, and 18 show the averaged spectral analysis of the same sounds recorded in the concert hall and the reverberant room by the distant microphone. A comparison with the graphs of the mixed microphone signals reveals marked similarities.

The set-up shown in Figure 11 demonstrates a practical two mike setup for the viola. Compared in Figure 19 is the spectral analysis of the viola scale received in the reverberant environments with the results from the two mixed microphones shown. The similarity demonstrates that two close microphones can approximate the diversity of the timbre produced by an instrument in a concert hall.

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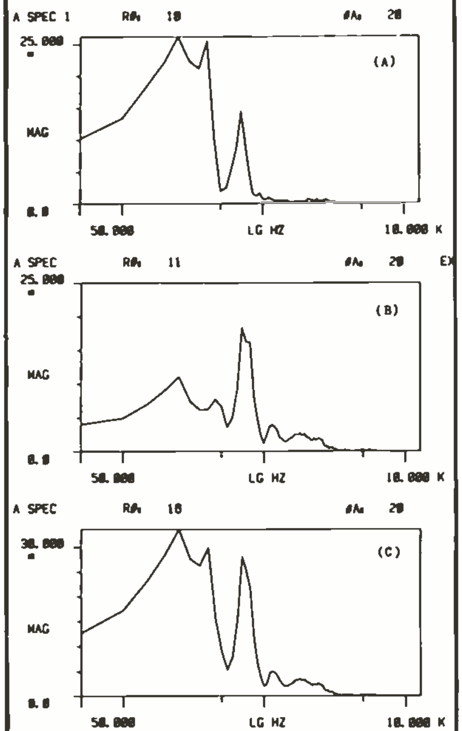

Figure 13. The Baritone Saxophone Eb2 (77 Hz): The averaged spectrum density analysis of the signal received from: A) the side microphone, B) the front microphone (both as in Figure 4), C) the spectrums of both microphones summed arithmetically.

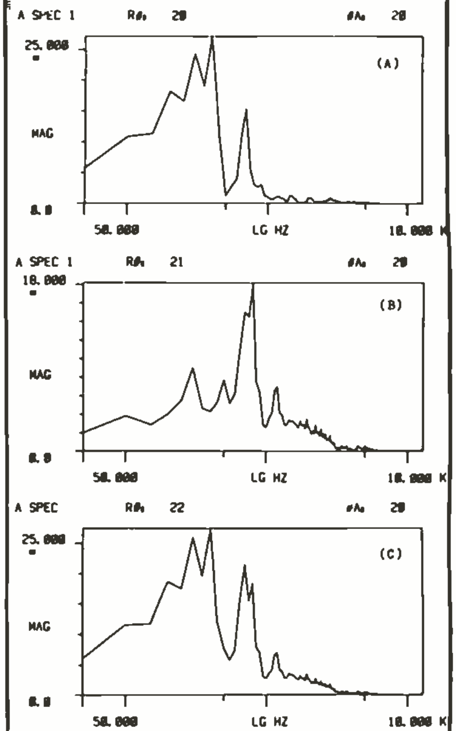

Figure 14. The Baritone Saxophone G2 (98 Hz): The averaged spectrum density analysis of the signal received from: A) the side microphone, B) the front microphone (both as in Figure 4), C) the spectrums of both microphones summed arithmetically.

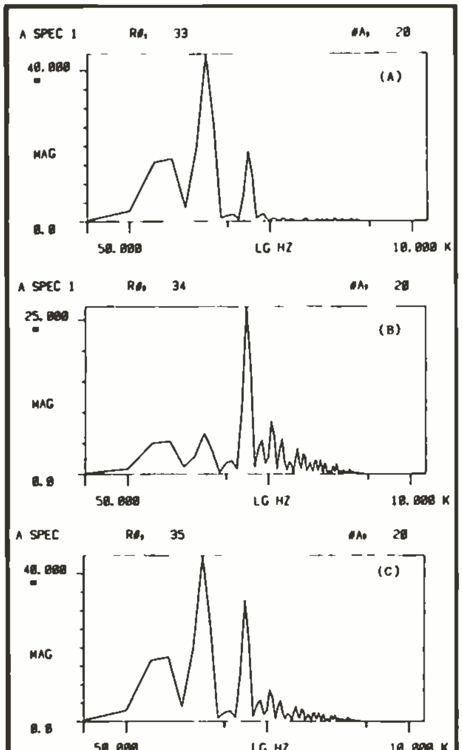

Figure 15: The Baritone Saxophone F3 (175 Hz): The averaged spectrum density analysis of the signal received from: A) the side microphone, B) the front microphone (both as in Figure 4), C) the spectrums of both microphones summed arithmetically.

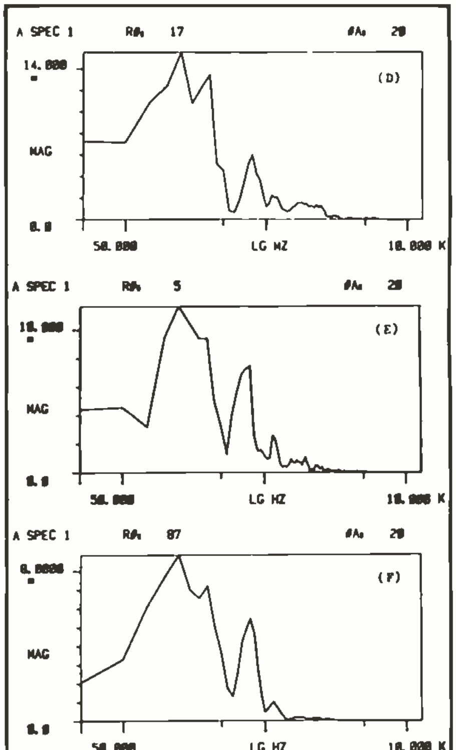

Figure 16. The Baritone Saxophone Eb2 (77 Hz): The averaged spectrum density analysis of the signal received from: D) the two[2] microphones seen in Figure 4 mixed electrically, E) the distant microphone in a concert hall, F) the distant microphone in a reverberant room.

Listening Tests

Listening tests support this statement. The participants in the tests included professional musicians and music students. A tape was prepared providing instant comparisons between the sounds obtained from the close and distant microphones with their levels adjusted for equal loudness. Participants consistently chose the mixed microphones over the single microphone sounds to correspond to the timbre quality of the instrument in the concert hall, and most preferred the timbre of the mixed sounds. Heard separately neither of the closely placed microphones reproduced a sound satisfactory to the listeners.

Mixed close microphone signals maintain the desirable sound qualities of close microphone techniques while introducing some of the frequency domains inherent in the distant mike technique, without allowing the time domain processing of room reflections and reverberation. Phase interference from reflected sound is largely reduced. The engineer has a more effective control of the natural spectral balance in the mixing process with ingredients that are pure products of the instrument. Close multi-miking eliminates the need for excessive electronic equalization. Microphone choice, microphone placement, and close multi-miking should be the primary means of spectrum equalization, leaving electrical equalization for spectrum correction in the mix-down.

During additional listening tests artificial stereophonic reverberation was added to the individual and mixed microphone signals. The listeners preferred the tonal quality of reverberation from the mixed microphone signals. They described this sound as "rounder" and "fuller." A single microphone feeding an artificial reverberation system did not have the timbral quality or the quantity of information that was received and processed by a room in a live concert. A mixed multi-microphone signal provides an almost complete and balanced spectrum of the instrument output and contained a good proportion of low frequency energy that was characteristic of live situations.

It should be understood that this system of miking is not meant to supersede or replace other microphone techniques. The close multi-microphone technique is just another method, but gives very good results compared to single close miking in a damped environment.

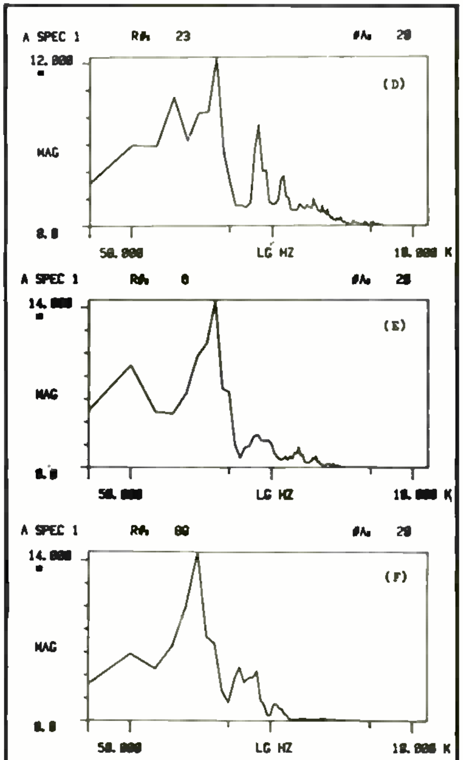

Figure 17. The Baritone Saxophone G2 (98 Hz): The averaged spectrum density analysis of the signal received from: D) the two[2] microphones seen in Figure 4 mixed electrically, E) the distant microphone in a concert hall, F) the distant microphone in a reverberant room.

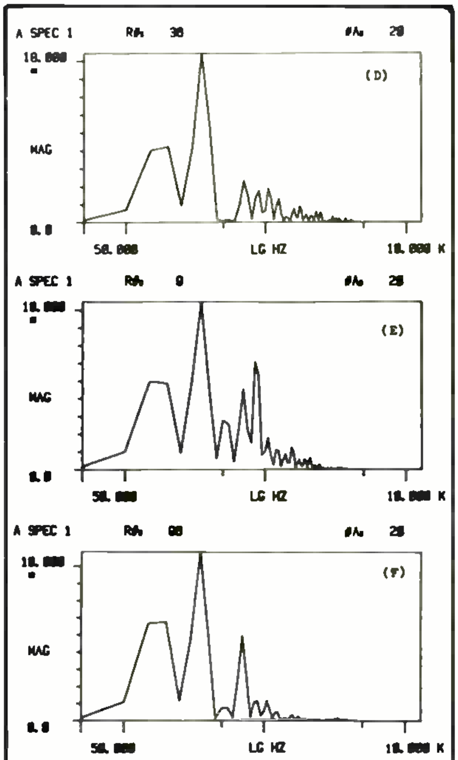

Figure 18. The Baritone Saxophone F3 (175 Hz): The averaged spectrum density analysis of the signal received from: D) the two[3] microphones seen in Figure 4 mixed electrically, E) the distant microphone in a concert hall, F) the distant microphone in a reverberant room.

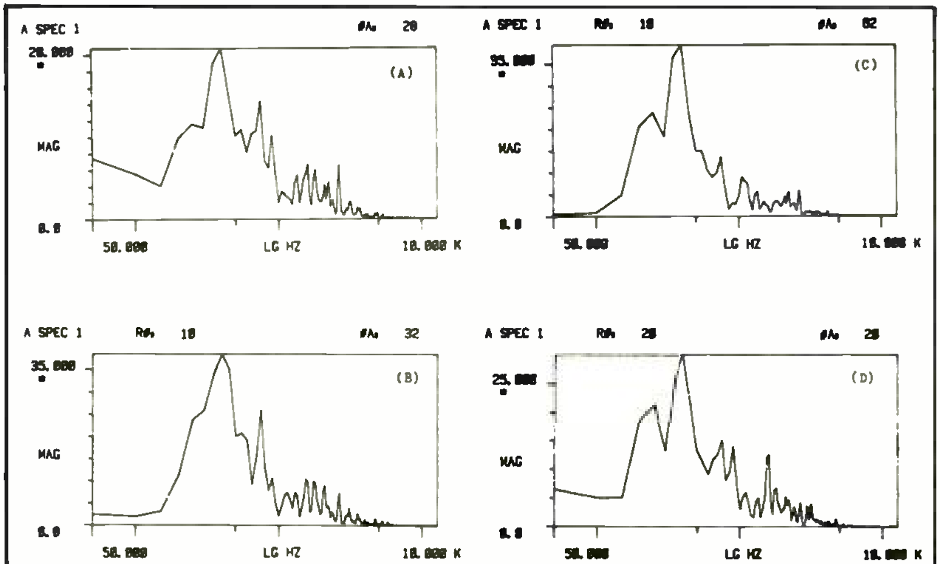

Figure 19. The Viola Scale: The long-time averaged spectral density analysis of the signal received from: A) The distant microphone in a reverberant room, B) the distant microphone in a concert hall, C) the two microphones from Figure 6 mixed electrically, D) the distant microphone in a concert hall with a reflective baffle behind the musician.

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. . . an answer to the drummers' need for more cue volume —

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by Ethan Winer

There are few hard, fast rules in this world of music and art, but one thing is always for sure — the drummer wants it louder in the phones. The singers could be wracked with pain, tears in their eyes and still the drummer will want more volume. You could swear they're all deaf. (H-m-mmm? . . .) So here, to the rescue, we have a booster amplifier packing — hold on to your hats — a whopping 1.25 watts into a pair of 8 ohm stereo headphones. Coupled with a relatively efficient pair of cans, this combination could be lethal. But seriously, the idea is to reduce the overall cue feed level to something more comfortable and then give the booster to the percussionist. (Here you go, Jack, blow your brains out.) A volume control has been provided allowing the drummer to be in complete command, though obviously anyone can use it. But it has been designed for a drummer's use since it includes an electronic metronome which allows the drummer or other principal timekeeper to hear a steady beat along with the music. If you want everybody to hear the metronome in their phones, its' output is available for feeding into the control room.

Since the electronic "click" sound is

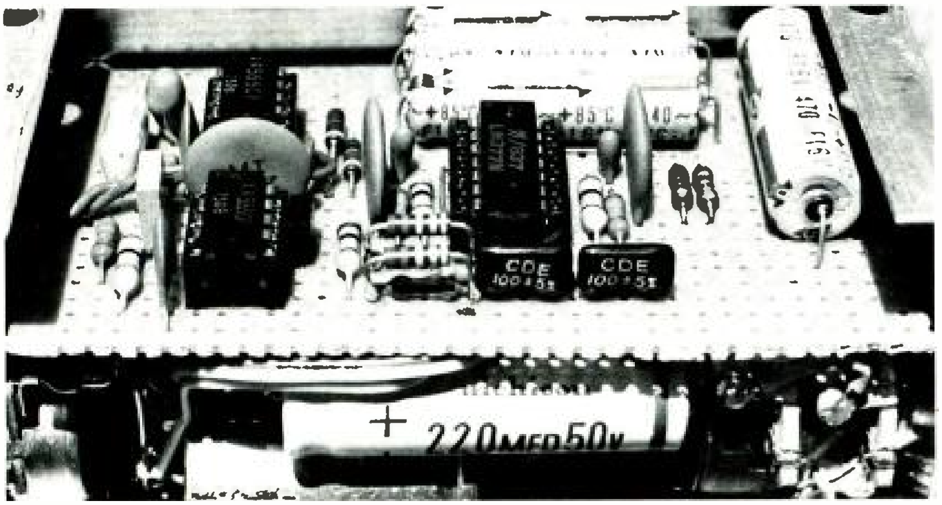
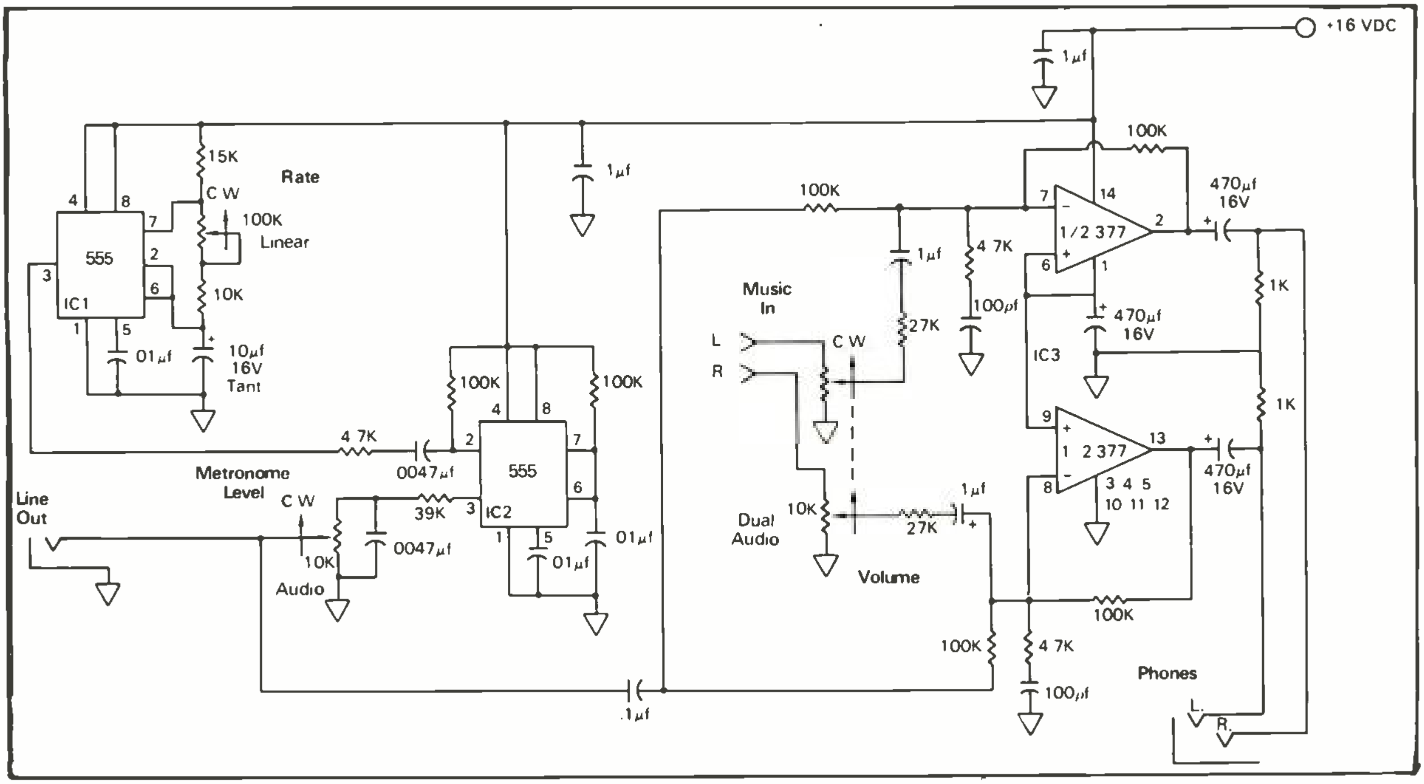


Photo by Pete Hodgson

Parts List

- 2 - 555 timer ICs
- 1 - 377 audio power amp IC
- 1 - bridge rectifier, 50 volt/1 amp
- 2 - 1K ¼ watt 5% resistors
- 3 - 4.7K ¼ watt 5% resistors
- 1 - 10K ¼ watt 5% resistor
- 1 - 15K ¼ watt 5% resistor
- 2 - 27K ¼ watt 5% resistors
- 1 - 39K ¼ watt 5% resistor
- 6 - 100K ¼ watt 5% resistors
- 1 - 10K potentiometer, audio taper
- 1 - dual 10K pot, audio taper or dual 100K pot linear taper (see text)
- 1 - 100K pot, linear taper
- 2 - 100 pf 10% disc capacitors
- 2 - .0047 mf 10% disc capacitors
- 3 - .01 mf 10% disc capacitors
- 2 - .1 mf 10% disc capacitors
- 2 - 1 mf/16 V electrolytic capacitors
- 1 - 10 mf/16 V tantalum capacitor
- 3 - 470 mf/16 V electrolytic capacitors
- 1 - 1,000 mf/25 V electrolytic capacitor
- 1 - on/off switch
- 1 - ⅛ amp slo-blo fuse in holder
- 1 - line cord and grommet
- 1 - power transformer, 12.6 V/400 mA
- suitable enclosure, hardware, input and output connectors.

Note: A drilled and plated PC board for the metronome/amplifier is available for $9.50 post paid from The Recording Center, Inc., 25 Van Zant, East Norwalk, CT 06855.



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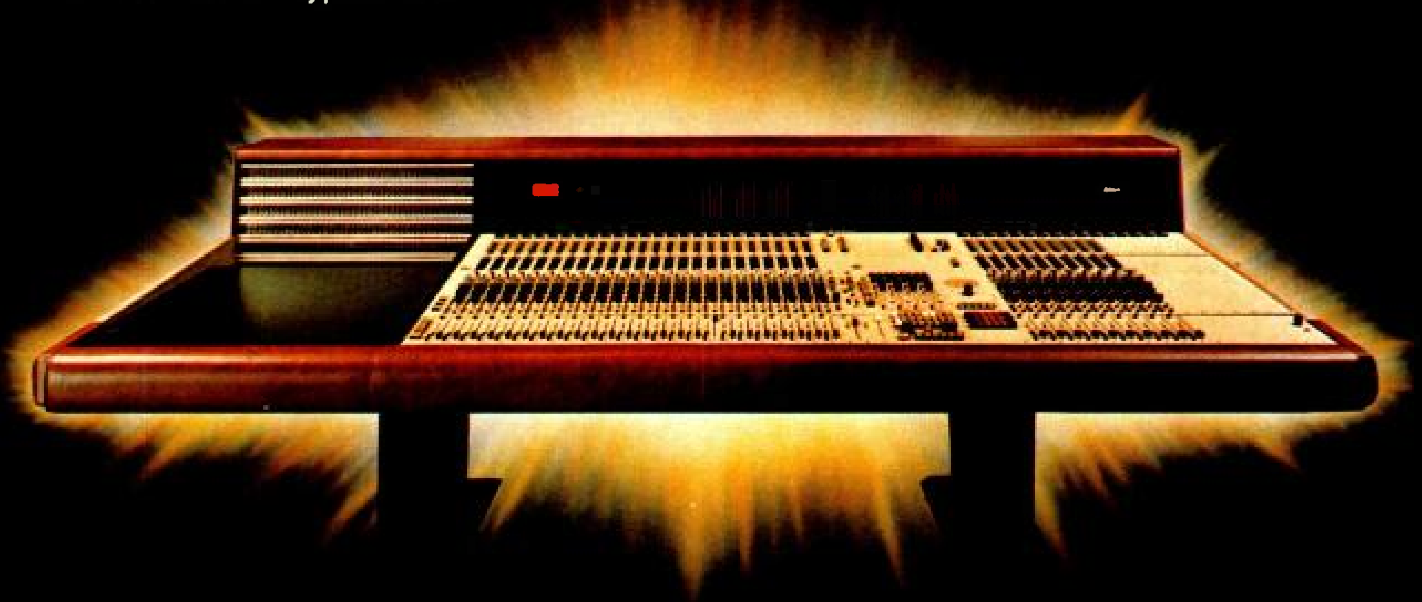

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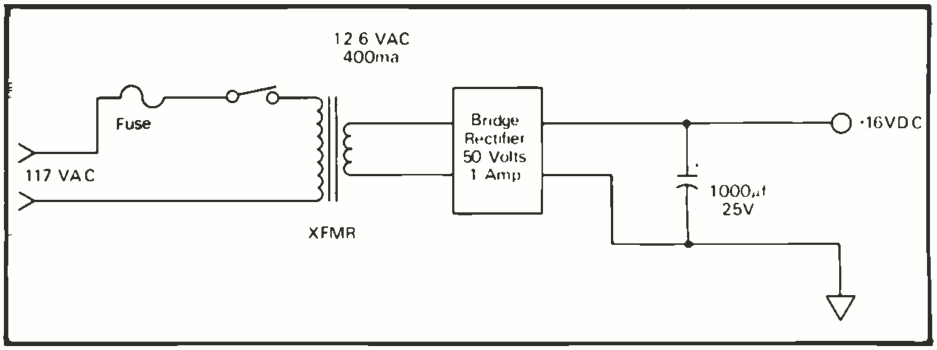
relatively neutral and resonance-free, an equalizer may be used to create a tone quality that will help to stand out in the mix. A parametric EQ would be the most useful — set for a narrow peak and give it a healthy boost. In the midrange, this will bring out a wood block or claves type sound. Lower down, near 80 or 100 Hz, a boost (use less here) will sound like a kick drum. Cut out some midrange and it will sound even better. Play it through a good speaker and then mike that — you'll think it's the real thing. But there are even more uses for a metronome in the control room. For one, you can keep your 30-second spots to 30 seconds. Or, you can lay a steady beat on the master tape to hold things together if the drums won't be added till later. In fact, a metronome can be so handy in the studio that many readers may want to build only that part. No problem. Eliminating the power amplifier will allow battery operation which further simplifies construction. Simply hook up a 9-volt battery to the timer ICs and away you go. No other changes are required, though the output level will drop by 6 dB or so.

Most mechanical metronomes have a range of from 40 to 208 beats per minute. The electronic version presented here has been designed to slightly exceed this range to take component tolerances into account.



Final calibration amounts to little more than counting beats and timing with a stopwatch. Marks may be made on the panel indicating the tempo or you could go all out and have your unit professionally painted and screened.

As you might imagine, operation isn't very complicated. In addition to the tempo adjustment, there are two volume controls — one for the music and one for the metronome. The user can set any balance desired without disturbing the main cue feed. Incidentally, the music is maintained in full stereo, while the metronome is fed equally to the two channels.

Construction

Little difficulty should be encountered, even for novice builders, as the circuit is relatively simple. Also, a printed circuit board is available from the author for a nominal cost, though some readers may prefer to use perf board. Either method should work fine, but another decision will need to be made regarding input and output connectors. The prototype was built into a fairly small aluminum box with a ¼" stereo phone plug sticking out the side allowing the whole thing to plug directly into the cue jack on the wall. This may not be practical if a larger enclosure is used and you may be better off with a two or three foot wire coming out the side for the input. This also allows you to use any connector type you want, or even to change later on.

About The Circuit

For the metronome, two 555 timer ICs are used — one to set the tempo and the other to generate the click sound. The 555 was chosen for several reasons. The accuracy of the timing is independent of the power supply voltage so the markings, once established, will not change. This becomes especially important when powering the device from batteries since the timing must not vary as the batteries age. Variations in temperature won't have much affect either on this little guy. The 555 is very common, inexpensive, and available nearly anywhere electronic parts are sold. The 377 power amp IC is also pretty common and has similar tolerance to variations in operating conditions. But to stay with the timing circuit for a moment, we should quickly go over some of the other components.

No matter how stable the IC may be, unless the remaining parts are of a similar high quality, full advantage may not be had. The 10 mf capacitor should be tantalum if possible and the 10K and 15K resistors should be carbon film (or better, metal film if you can find them). A wirewound potentiometer is also preferable, but this may be even harder to track down. If you can't locate these special parts, or if you're too lazy to even try, don't let that stop you from building this. In fact, I didn't bother with the wirewound pot either and mine works fine. Still, there is some satisfaction to be had from knowing it's the best that it can be. No other parts are critical at all, though don't buy the cheapest stuff you can find either.

Regarding hard to find parts, one real

winner is the dual taper potentiometer. One good place to order this and other hard to find parts is Mouser Electronics, 11511 Woodside Avenue, Lakeside, California 92040. While even they may not have in stock dual audios, you can get a dual linear in a higher value, which in many cases can approximate the audio curve. Incidentally, Mouser is the only source that I know of for audio inductors and radio coils. (Stay tuned for an article on a barely legal AM transmitter you can use to broadcast your mixes to a car radio.) But back to tapers, for those of you who don't understand the different kinds, a brief explanation is in order.

A linear tapered pot is used to "divide" a voltage or an audio signal linearly. That is, when the control is turned half way up, the output voltage is half that of the input. A quarter way up gives a quarter of the signal out. While this makes sense for many applications, it is less useful as a volume control since the ear doesn't perceive loudness in a linear fashion. With an audio tapered pot (sometimes called logarithmic), when the control is set half way up, only one *tenth* of the total output has been reached, allowing another 20 dB of range till full open. (Remember, increasing by 20 dB corresponds to multiplying the voltage by ten.) Many guitar amps are intentionally made with a linear pot for a volume control to give the impression of a large power reserve. (Gee, it's really loud and it's only on two and a half!) Unfortunately, most of the useful range has been crammed into the first quarter of a turn, making adjustment more difficult than it needs to be.

In the case of our metronome/booster, a dual 100K linear pot may be used instead of the dual 10K audio pot specified and the music will be about 12 dB down from maximum with the control set in the middle. While this may be less than optimum, it does help spread out the range some.

Performance

For the amplifier, you can expect the worst case THD to be less than .1% when using 8 ohm phones, and even less with higher impedance models. Frequency response is equally impressive, being within 1 dB from 20 Hz to well beyond 100 kHz. Gain of the audio program will be just over 10 dB with the volume control all the way up. If you need even more, you can reduce the 27K resistors to 10K which will allow up to 20 dB of increase. Regarding the metronome, there isn't much to spec except possibly for stability of the tempo. The biggest factor here will be variations in temperature, since all electronic components are affected this way. In fact, this is the main reason film resistors and tantalum caps were mentioned as being preferable. The 555 IC varies less than .01% for each degree (F) of change in the ambient room temperature.

Photo by Sundance Light

by Patrick Maloney

The fact is, musicians are becoming more demanding in their requirements for good monitors and monitoring systems are growing in complexity and sophistication to satisfy this demand. The trend is to more and more independent mixes on stage and finer control of these mixes. Of course, there will always be shows that are best controlled by a simple monitor send pot on the house console—thank God. But many systems which are now appearing can be much more complicated and difficult to operate than the house console. It is no longer a place to train future house mixers nor is it a position for the faint hearted. It is a highly specialized sound system in itself — complete with its own unique set of problems and solutions whose ultimate purpose is to provide an environment on stage that totally supports the artist acoustically, musically and emotionally.

The most complex system I've encountered to date is owned by "The Captain and Tennille." Designed and operated by Rodney Pearson, the system utilizes 15 separate output channels, 15 amplifiers, 18 speaker cabinets and even includes an intercom for onstage communication between the monitor mix operator and the musicians. The system was recently set up in The Captain and Tennille's Rumbo Recorders facility in Canoga Park while the band was in rehearsal for an upcoming Las Vegas appearance at the MGM Grand Hotel.

In discussing the system with Rodney, I quickly learned that Daryl Dragon (The Captain), and Toni Tennille's dedication to high quality sound is evident not just in their monitor system, but also extends to their concert touring systems, their first class recording studio and their Las Vegas showroom engagements. The key word that kept coming up in our discussion was "consistency" and is one of the most important reasons why a system of this size and complexity ultimately works.

Rodney Pearson is certainly no stranger to the sound reinforcement business. He started his career with BBC, in London, and eventually came to the States where he went to work for Stanal Sound as a house mixer on tours for Liza Minnelli, The 5th Dimension, Paul Simon, Mac Davis, Dolly Parton and numerous others. He then travelled with Mac Davis as an independent mixer and eventually started working with The Captain and Tennille when they went to England to do a television special a few years ago. It was at this point that he started working with them on their monitor system.

Prior to Rod's arrival, Daryl had his own onstage monitor system which was based on a Yamaha PM-1000 console (16 in by 4 out) that he was using for a keyboard mixer. He fed a few other instruments through it as well and routed them through all four outputs to separate monitor speakers. Basically he wanted to mix his own monitors and was quite capable of the task — stemming no doubt from the necessity of having to do so at the *Smokehouse*, in Encino, California, where he and Toni first started out. In those days there were no roadies or monitor mixers.

Two outputs from his PM-1000 also fed the house system in a novel manner: the keyboards that Daryl played with his left hand were mixed together on one buss and the right hand keyboards went down the other line, thus giving the house mixer some control over the sound.

Basically, the present system all grew up around Daryl's monitors and then expanded to include the whole band. For several reasons the decision was made to custom build their own system from scratch instead of renting a system only when they needed it. From his experience while working for Stanal Sound and with other sound rental companies, Rodney didn't feel he'd have complete control of a rented monitor system. Based on a situation he had with Mac Davis where in one ten-day stretch he had used ten different sound systems from five different companies. He observed that the house generally wasn't too much trouble but the monitor mixer had all kinds of problems since there was no repeatability.

The Captain and Tennille had shows coming up in places of varying sizes and acoustics (i.e.: State Fairs, arenas, Las Vegas-type showrooms, theaters-in-the-round and smaller theaters). They felt they would have gotten different gear for each venue depending on the specific monitor equipment available. Obviously, this would not have given them what they were really looking for which was consistency in the monitors. The decision was made to develop their own onstage set-up while continuing to rent the main house system since they knew that the house sound would change depending on the venue and that was part of the concept. For instance they had a segment of a tour where they did theaters-in-the-round and had Randy Weitzel from Clair Brothers mix the house sound with a Yamaha PM-1000-32 that Clair supplied. The Captain and Tennille supplied virtually everything else, including McCune

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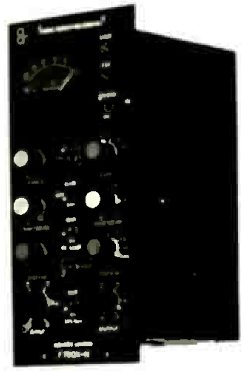

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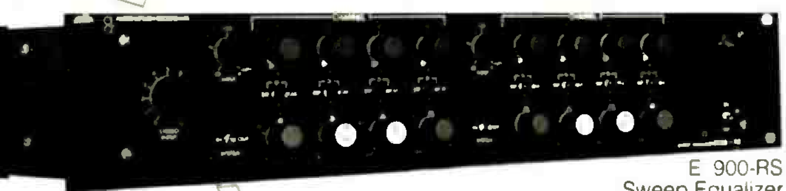

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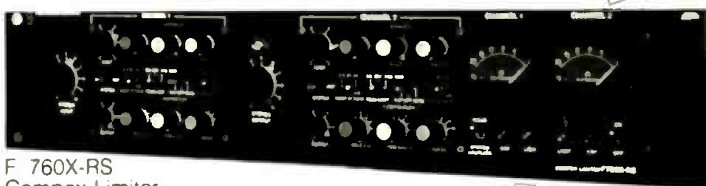

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Sound Service's SM-4B hanging speaker system, specifically developed for theater-in-the-round shows. In Las Vegas, they augmented the house sound with speakers at stage level to give more of a feeling of sound coming from the performers rather than from above the stage where the house speakers are usually located. Clair Brothers S-4 system was used for State Fairs and large venues but throughout the tour they maintained a consistency of sound on stage for the performers by using their own monitoring system at every show. Furthermore, Daryl's concern that the sound in the house be consistent with their recorded sound was so strong that he has his studio recording engineer, Roger Young, operate the controls of the house console.

Before that tour could start, however, the group needed to assemble a system that would be flexible enough to adjust to these different acoustical environments and roadworthy enough to withstand the many thousands of miles the tour would take them.

The Speakers

Rodney's intention was to design a workable system for Daryl and then adapt it to work for everyone else. He started out by doing some research using a 24-track tape that they had recorded live at the Greek Theater in Los Angeles. They played various instrument tracks through different monitors and checked to see how well the speakers reproduced these individual instruments.

Daryl was especially concerned with getting a good kick drum sound and the final tests were to determine which woofer could best reproduce it. They used a two-way monitor and kept changing the woofer until they were satisfied. By this time they had already decided on the high frequency

element they wanted to go with — the Emilar EH-800 coupled to an Emilar EA-175 16-ohm driver. The woofer continued to be in question, but a decision was made soon after they started adding other instruments on top of the kick drum sound. Adding the conga really started showing the difference between speakers and led them to choose a Yamaha woofer. A cabinet was then designed to house the woofer and horn. They had also picked up some monitors made by TASCO which were used with success at Harrah's Club in Lake Tahoe. It used the same Emilar driver but with a different horn — the Emilar EH-500 — and a JBL 2220 woofer which was subsequently replaced with the Yamaha (Figure 1).

There are certain things that are unique to this band and one of them is that the basis of their monitors is a good tight kick drum to keep everyone in time. The system needed to deliver a fair amount of kick without sounding boomy and out of control. This being achieved the rest of the system was designed around it.

Some of the speakers are bi-amped while others have passive crossovers but the basic speaker elements are the same. Rod first used a Crown VFX-2 variable crossover to find the optimum turnover point. He then orderd a White type 4016 - 800 Hz crossover with an 18 dB per octave rolloff. The passive crossovers used in a few of the monitors are made by Histrionics and work well.

The speaker system was designed to sound good to start with and does not require a lot of equalization. According to Rodney, "we didn't take a system and say 'we can *make* it sound good.' It had to sound good to start with. When EQ is found to be necessary it is done with a combination of ear and real time analysis using the

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Inovonics Model 500 analyzer which I'm very happy with." Orban parametrics (model 622-B) are available for use on each channel but are used only when necessary. Generally, the units are in the circuit in the bypass mode and Rod hasn't noticed any coloration or other problems by leaving them in. On occasion he will not connect through the unit at all — for instance, in the case of the bass player whose signal is not put through his own monitor and there is little danger of feedback. In a situation like the conga player's, however, where his signal is fed back to him through a close by monitor, then the parametric is used to cut down on feedback. The units aren't used much for EQ purposes as this can be handled adequately from the EQ section in the board.

However, Rod and company don't believe in a lot of equalization. They chose their microphones as carefully as they chose their speakers — with an ear to faithfully reproducing the sound of each instrument as it occurs at the source. In a system as complex as this one is, the fewer paths that a signal takes and the fewer modifications that are made to it, the better it's ultimately going to sound.

The Main Mixing Console

The heart and soul of the system is a muchly modified Yamaha PM-1000 32-input console. John Windt, who worked very closely with Yamaha in modifying this console, took out four of the input modules and replaced them with four more output modules. He then replaced the four output assign buttons on each input strip with eight rotary send pots. This gave Rod 28 inputs and 8 totally separate outputs, plus the two echo sends. It was necessary to re-wire the mother board to accept these last changes but it was done in such a way that it is still possible to put back the original 4 input strips if that should ever be desired. No re-wiring would be necessary. According to Rod, this was the second console of its type customized in this manner by Mr. Windt and is currently available as a standard modification through Windt Audio in Los Angeles.

The eight rotary output pots are all color-coded in order to simplify the operation of the console as well as to help speed up the installation of the entire system.

The 4 x 4 output matrix familiar to all Yamaha PM-1000 users has been expanded to 8 x 4 to accommodate the additional 4 output busses but Rod is not using it as part of his monitor system. However, it is occasionally used to make reference tapes during the course of the show. The eight matrix masters were converted into 8 auxiliary inputs and the "direct/PB" switch at the top of each output module was changed into a headphone cue switch to provide monitoring of each output buss individually.

The cue system of the console has been further modified by Rod for a louder output and cleaner signal by removing some terminating resistors in the cue circuitry and by bypassing the headphone amp. Other modifications Rod has made to the console include making the echo sends appear on the cue buss; re-wiring them to be post fader so they could be used as monitor busses; and installing kill switches on each input channel. He doesn't change levels a lot during the show but instead chooses to use the kill switches to bring signals in and out as needed. This helps keep the monitors clean sounding. Toni, for instance, uses only one of her three vocal mikes at a time and keeping the other two off prevents unwanted noise from cluttering up the monitors.

It was thought at the time that 28 inputs and 8 outputs would be sufficient — famous last words, etc. It's never been a real problem but there have been a couple of times when they could have used more inputs depending on the requirements of the lead-on act. However, it is usually more than adequate, and according to Rod, the board's flexibility and size came in handy when they had Gene Cotton opening for them. "He had a guitar-based band and we were able to use seven inputs for his group which we didn't need for our part of the show. So we could give him the exact mix he wanted and then re-patch the mikes to different input modules for our part of the show. So everybody had exactly what they wanted." By making the system bigger — and in some people'a opinion more complicated — they've actually made it easier. To be sure, the system is involved but it has been designed by Rod and Daryl along simple workable principles and is set up in a logical straightforward manner. Complex, yes; complicated, no.

Fifteen Independent Mixes

The system is now set up to handle 15 separate mixes — 10 being controlled from the main console and the remaining 5 through Daryl's PM-1000-16. Referring to Figures 2 and 3 should clear up any confusion resulting from my forthcoming description of the routing of these 15 mixes.

The way it breaks down is somethng like this: When Toni sits at her piano she has two monitors connected in parallel which are fed off one buss of the main console. The three background singers have two speakers which are fed off another channel. The horn players have the same setup. The drummer gets one mix from Rod which contains his kick drum, bass, vocals and piano. He also gets another mix directly from Daryl's keyboards which goes to a separate monitor behind him. The bass player gets an independent mix from Rod, as does the percussionist. The second keyboard player has a small Bi-Amp model 8802 mixer which he uses as a keyboard mixer. The 8802 has instrument level inputs as well as balanced mike level inputs on each channel — a feature which eliminates the need for direct boxes for each keyboard. This mixer in turn feeds the main monitor mixer which routes the signal back to the keyboard player as well as throughout the rest of the band. The bass guitar is taken direct into the monitors and PA although the bass player does use an amp on stage for his main monitor. However, none of the electric keyboards have amps. They are all taken direct or submixed, sent to the monitor mixer and then fed back out to the various monitor speakers at the appropriate levels. The idea is to cut down on a lot of unnecessary and to some extent uncontrollable sound on stage. They have been using SESCOM passive direct boxes (model SM-1A) to interface the instruments with the various consoles, as well as an 8 input box custom made by Stanal Sound for Daryl.

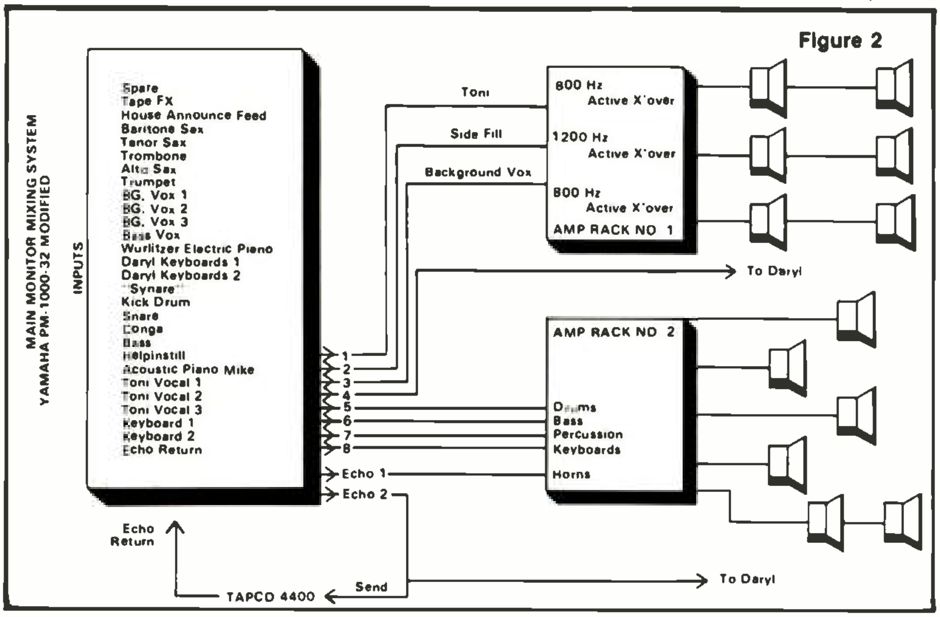


Figure 2

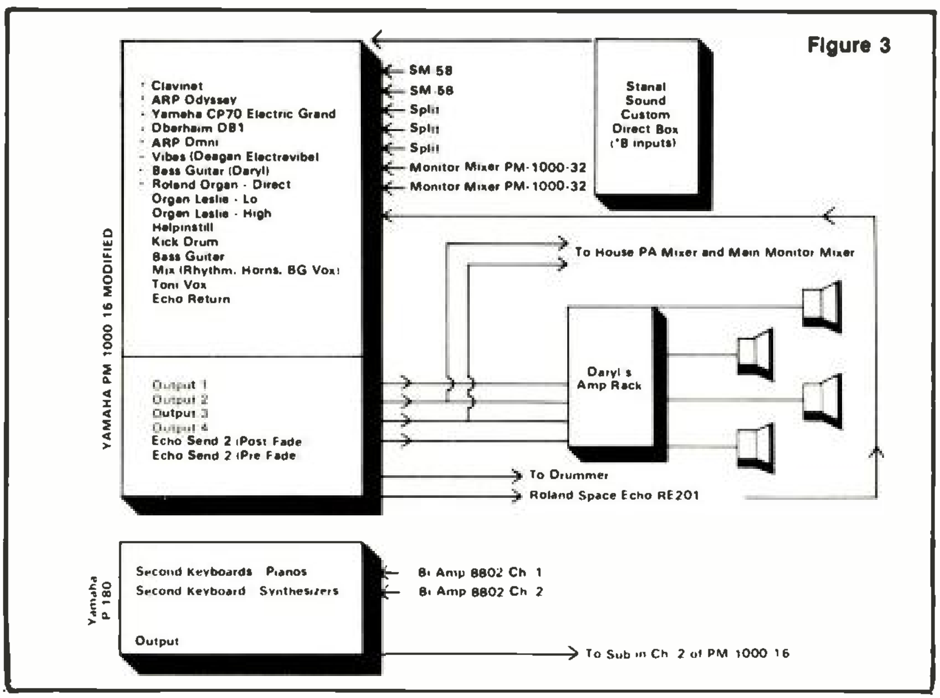


Figure 3

Sidefill speakers are on yet another buss (lost count yet?) and are basically designed by Stanal Sound to combine two Altec 816s into one cabinet. The ports were moved from the bottom of the cabinet to the sides so that it became more of a squat, square shape instead of a tall narrow one. They were further modified to incorporate a 90° JBL 2350 horn, an Altec 288-16G driver and two of their beloved Yamaha woofers. When it was first assembled in this fashion it was flat from 125 Hz to 6 kHz with no EQ. These side fills cover the front part of the stage and provide an even monitoring field for Toni when she's not sitting at the pianos. When Toni is playing the Wurlitzer electric piano she hears herself through a combination of side fills and one of Daryl's monitors.

Daryl's System

Daryl's four monitor speakers are fed directly from the four output channels of his own Yamaha PM-1000-16. Two of these outputs are also used to feed both the house system and the main monitor console in the novel method described earlier.

Daryl uses four discrete monitors instead of simply combining all the signals and sending them to one speaker for several reasons, one of which is greater clarity. For example, Toni's vocal doesn't have to fight with the bass guitar for the woofer's attention. Localization of sound with respect to feedback is another consideration — the idea being that it's easier to isolate which microphones are feeding back if the signals come from different speaker positions. Also Daryl has better control — he can, for instance, bring up just the vocal master during a particular song in order to more easily listen to the harmonies if he so desires.

Microphones

Toni uses a total of three vocal mikes, all Beyer M500 ribbons. One is located at the Yamaha grand piano, one at the Wurlitzer and the third one is used as a hand held mike down front. Beyer M-69 dynamics are used for the background vocal mikes and Rod has found them to give a combination of smooth sound, good low end and significant

freedom from feedback. The manner in which the drums are miked differs slightly depending on the venue. On large concert dates all the drums are individually miked although only the kick, snare and high hat ever go through the monitors (Figure 5).

The toms have the bottom heads removed and are individually miked for the PA with Sennheiser 421s mounted on clips underneath the toms. The mike heads are placed inside the shell just far enough to provide some isolation from the other toms and cymbals. Beyer M201s are used both on the percussion and on the hi-hat while a Shure SM-57 picks up the snare. Overhead drum mikes are AKG 452s. The two congas are also miked with a 452 in the middle above the heads. Horns are all miked with Shure SM-58s.

A good deal of attention was placed on the kick drum miking since it plays such a large part in this particular system. It's miked with a Sennheiser 441 which is fed directly into a UREI LA-4 limiter. One

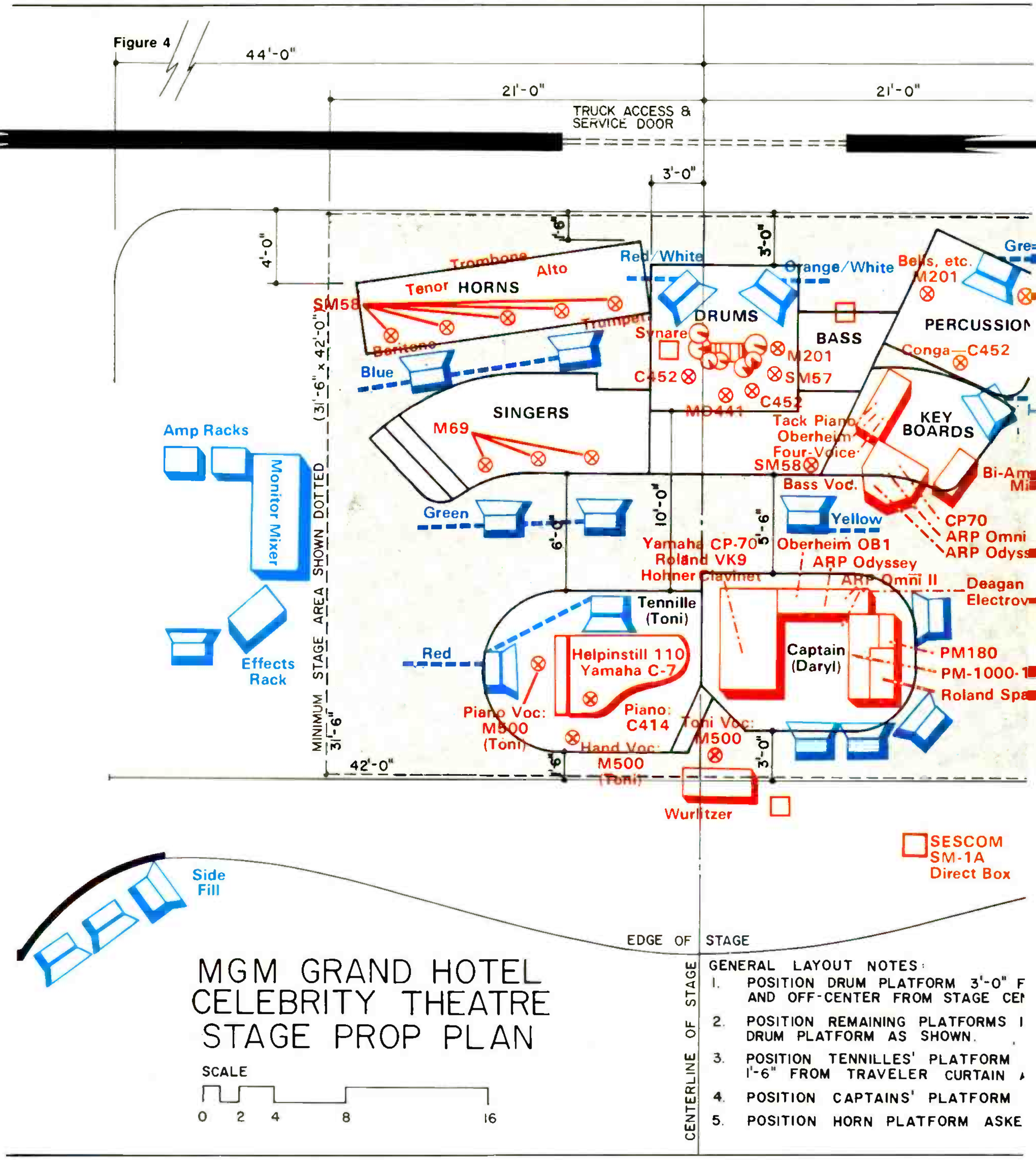
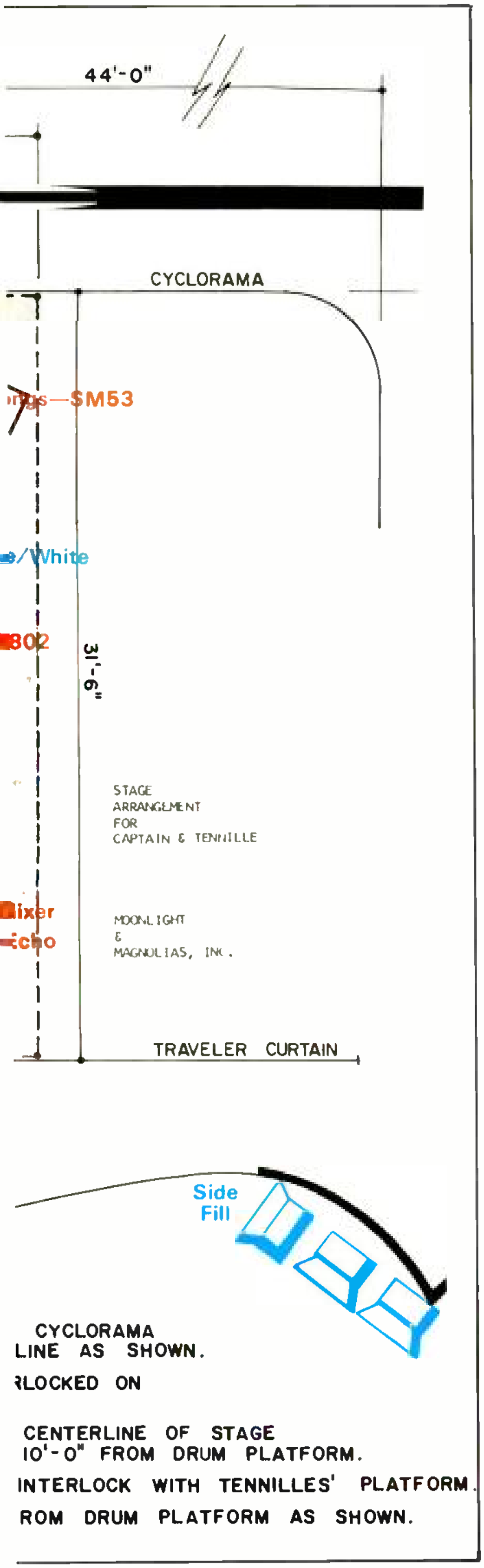


Figure 4

advantage of this unit is that it has a high gain position which boosts the output of the 441 enough to drive the circuitry without just limiting a lot of noise. Of course, having a pretty hot signal from the kick drum helps. I would hesitate to try this with a soft singing vocalist. The LA-4 tightens up the kick drum and saves a lot of headroom in the amps.



The limiter is usually set on a 12:1 ratio so there is actually some compression going on. It basically is set up to prevent the woofers in the monitors from kissing the grill cloth. This way they can get a little more level without the associated amplifier problems and blown woofers. The outer head of the kick is removed and the 441 is placed as close to the remaining head as possible. It's positioned off to the side away from the beater area.

There are no other outboard devices used in the main monitoring system other than a TAPCO 4400 reverb unit for Toni which is used to create a little fuller sound onstage. A Master-Room reverb model MR III is used for the house PA and has proven to be perfectly suited for the job, says Rodney.

The piano uses a Helpinstill 110 pickup which is basically used just for the monitors. The isolation and freedom from feedback it provides is a welcome aid to the monitor mixer. An AKG 414 mike set for a unidirectional pattern over the second sound hole is the main source of piano sound for the house PA, although it can be mixed into the monitors if so desired. This also provides a backup for the piano in case either the pickup or microphone malfunction.

The Yamaha piano always travels with the group and is another example of their dedication to consistency. They also carry many spare instruments as well as a spare monitor mixer for Daryl. Figure 4 shows the entire microphone instrument set-up, and speaker layout.

Redundancies and Backups

This recurring aspect of The Captain and Tennille's monitor system impressed me a great deal. Spares, backups and instant alternative course of action are all part of the philosophy of a good monitor system. Pulling out the soldering iron on stage in the middle of a set is considered bad form. It upstages the musicians and you can never find an empty AC socket anyway.

Daryl recently started playing bass guitar on a few tunes which necessitated the additon of another mixer since he had long since run out of available inputs. So a Yamaha PM-180 was pressed into service. Daryl uses two Leslie organ speakers located offstage which are miked top and bottom with Shure SM-58s and these two signals are then fed to his mixer where they are combined and sent to the main monitor mixer. Two Leslies are not needed from a power standpoint since they are re-amplified anyway; but the interaction of the various speakers all rotating at slightly different speeds provide a fuller, richer sound. Also, again we have the backup factor. If one Leslie fails, the other one will still supply the necessary effect.

The microphone cables are made by

Neumann and are all of various colors to match the color coding on the console output buses. If you have ten or so cables bundled together and there is a problem in the red channel, it helps to be able to immediately pick out that color cable from the rest and trace it.

All monitors are marked so that they always go to the same location on stage and are always connected to the same amplifier with the same cable. This simplifies trouble shooting considerably by preventing a particular problem from turning up at a different location every time the stage is set up, due to the same piece of faulty equipment being in a different place each time.

Another built-in safeguard of Rod's console setup is the fact that he sets all input faders at the same position — around 8 on the linear scale. This is the position Yamaha recommends for optimum signal-to-noise. The various signals are actually mixed via the 8 individual rotary pots at the top of each input. Therefore, if someone comes and plays with the console there is less likelihood that they'll seriously foul anything up — the feeling being that the 8 small knobs at the top of the strip are not as accessible and fun to play with as the faders are. If these faders are touched Rod will notice it immediatley, reset them back to 8 and then check the rest of the board for any other changes.

All speaker cables are color-coded with strips of colored tape which are then sealed with clear heat shrink to keep them from unravelling or getting dirty. Connections to the speaker cabinets are made by three-wire Hubbell Twist-Lok connectors which provide a very positive, polarity correct connection — no phone plugs here, folks. Stagehands can't confuse them with AC plugs, mike cables or guitar cords, as they can with other connectors, and they are about as foolproof as the man who wires them. Available from most electrical supply houses, they come in almost limitless sizes and styles. The one used by The Captain and Tennille is especially handy in that it is a right angle connector, thus allowing the cable to lie alongside the speaker cabinet instead of sticking straight out of it. This makes for a neater appearance and fewer tripping accidents. Recommended. Color coding even extends to the banana plug connectors on the back of the amplifiers. These amps are mounted in self-contained shippable racks which have a simple yet extremely useful feature: small music stand lamps mounted in the back. No more flashlights in the teeth for Rodney.

Housed in the amp racks are Yamaha 2200s (200 watts per channel into 8 ohms) and Yamaha 2100s (95 watts per channel into 8 ohms). One of the nice things about using Yamahas for monitor amps is the fact that they have meters on them which help keep track of the signal flow through the system. With 10 outputs from the main board alone, it helps to have a visual indication that signals are getting to all the amplifiers, especially since it is possible to only listen to one of the outputs at any one time. At Rod's position there are two power amp racks and one effects rack which houses two Pioneer CTF919 cassette decks, the Clearcom CS200K base station, the TAPCO 4400 reverb, a spare power amp, and a Crown D-60 which is used to power the headphones at the console (Figure 6).

Daryl has his own rack which houses two Yamaha P2200s, two P2100s and a UREI 1176 limiter which can be connected to any of his outputs. The rack also contains a spare Yamaha amp.

Many of the major components of The Captain and Tennille's monitor system are manufactured by Yamaha — a factor Rodney feels plays a large part in the dependability and consistently high quality performance of the system as a whole. The excellent backup, cooperation, and continuous support they have received from Yamaha has made Rod a firm believer in the company and their products.

Figure 5

Snake System

In the past, Rod had too many bad experiences using other people's multi-cables and going through house patching systems, so now the group owns all its own snakes and never do a show without them. The snakes were built by Stanal Sound and are designed around a heavy duty gold plated pin conector manufactured by AMP. A similar system was pioneered by McCune Sound a few years earlier using a slightly larger AMP connector. At about that time several other professional companies — among them, the Record Plant, Wally Heider Recording, Filmways Audio and A-1 Audio — followed McCune Sound's lead and built their snake systems using the same connector and pin configuration. Stanal has constructed adapters to properly interface with this earlier system. So, today, remote recordings at live concerts that involve any of these companies are a lot easier to do since the snake systems are all the same. This is a good example of competitors cooperating with each other for the sake of compatibility and ease of doing a show. This is especially helpful in the situation where more than one sound company's equipment is being used at the same time — a situation The Captain and Tennille are often involved in. Their particular snake system makes the hookup in theaters that have a sound system installed by Stanal a lot quicker since all Rod has to do is connect to the existing cables.

The Captain and Tennille use a 27-pair snake and a 15-pair. The 27-pair system incorporates a simple 1 in by 2 out splitter box which feeds signals to both the house and monitor mixers while the 15-pair goes directly out to the house console. The splitter is not active nor are there any transformers in it. Each input is simply hard wired to the two separate outputs although there are ground lift switches on each line that goes to Rodney. He reports there have not been any problems with hums or buzzes in the system as careful attention is paid to the AC grounds and the interconnected consoles are always Yamaha's. In addition, the connections and cable routing from microphone or direct box to the two consoles are always the same — a factor which decreases the hum and buzz potential considerably. They do have some SESCOM transformer-type splitter boxes if they ever do get into a problem. A few of the instruments are actually split three ways.

— *continued on page 112*

what have YOU done for your console lately?

Oh sure, you occasionally clean the panels and polish up the meters. And... if you're real considerate, you might even disassemble the modules and remove the cigarette ashes and remains of the last producer's shrimp salad.

But a pretty suit of clothes does not impress all clients. In fact, it's what they hear that brings them back. And that's our point!

***A clean console is important, but not so important as* CLEAN SOUND.**

The VCA employed relates directly to clean sound. Chances are you're still using a VCA that defeats the richness, clarity and definition you strive to achieve. That's why we make a transparent VCA for YOUR console.

We offer three complete VCA packages to clean up your sound...

The EGC-2500 which plugs directly into the MCI 500 series consoles without any modification.
($50.00 in quantities of 30-99)

The EGC-202 which is a physical replacement for the dbx 202 series VCA, only requiring a few resistor changes.
($29.00 in quantities of 30-99)

The EGC-205M which is a physical and electrical replacement for the Allison VCA-5M. That's right, we even improved our own VCA.
($34.00 in quantities of 30-99)

***You provide the* GLEEM....**

***We'll provide the* CLEAN**

EGC VCA's. From Allison Research.

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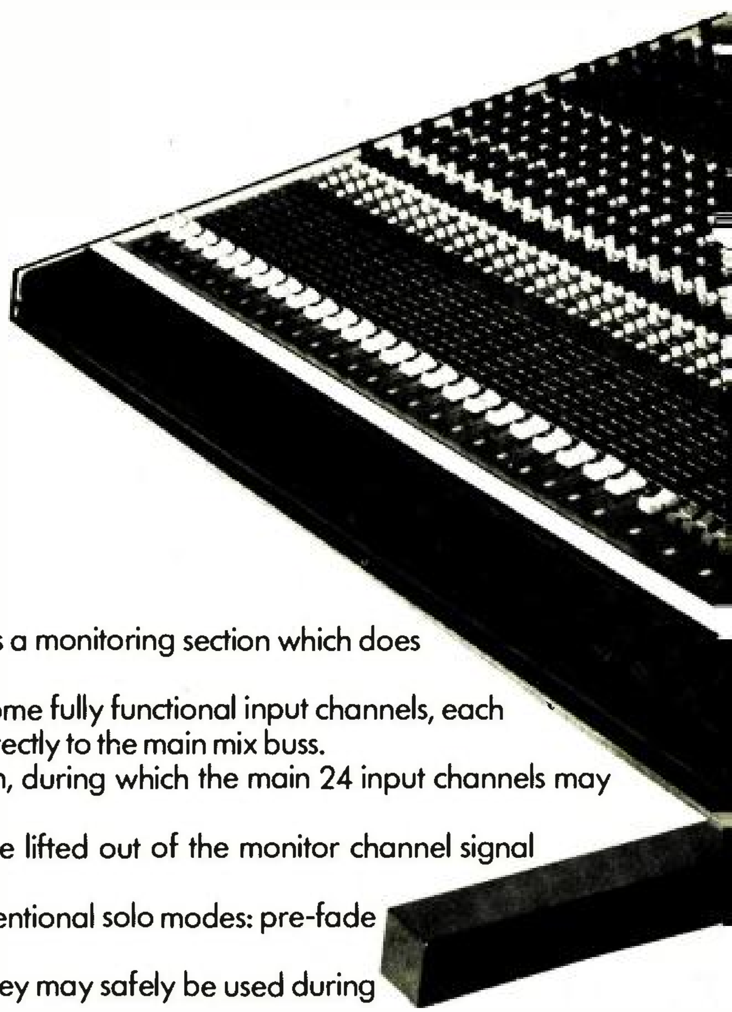


Revolutionary, in that, unlike any other split console, it has a monitoring section which does not become redundant during mixdown.

Instead, with simple switching, the monitor channels become fully functional input channels, each with EQ, aux, pan, fader, solo and mute, and are assigned directly to the main mix buss.

Therefore, a 24/16 console has 40 channels in mixdown, during which the main 24 input channels may still be assigned to the 16 group faders as sub groups.

Furthermore, the auxiliary sends (cues) and pan may be lifted out of the monitor channel signal path and inserted into the subgroup signal path.

Revolutionary, in that the console provides all three conventional solo modes: pre-fade (mono), post-fade (stereo) and solo in-place.

The first two modes do not disturb any signal paths, so they may safely be used during recording or mixdown.

The in-place mode mutes all channels not soloed, except monitor channels being used as effects returns or input channels in "safe" mode.

Other sophisticated features include: two programmable mute busses; six auxiliary sends, two of which may be assigned to follow the pan pot; a proprietary transformerless mic pre-amplifier; 41-position detented potentiometers, which are so precise that volume tracking between two similar controls will be typically within 1dB, and frequency tracking within 2 semitones.

Console equalisation is particularly versatile. Input channel equalisers have 4 variable-frequency bands, and a separate variable high-pass filter, while the monitor channel equalisers have 3 bands; the mid band with variable frequency.

All sections of the console electronics have been carefully designed to minimise phase deviation through the signal path, so that, typically, channel to track phase error is within 20° at 20kHz.

Conventional VU meters with peak level LED indicators are standard (as illustrated), but Soundcraft bargraph displays are available as an option.

Series 1624 is available in two mainframe sizes–24/16 (which with an optional 8 channel module provides 24 track monitoring) and 16/16, either of which can be supplied part filled.

SOUNDCRAFT ELECTRONICS LIMITED

Soundcraft Electronics Ltd 5–8 Great Sutton St London EC1V 0BX England Telephone 01-251 3631 Telex 21198
Soundcraft Inc PO Box 2023 Kalamazoo Michigan 49003 USA Tel (616) 382 6300 Telex 22-4408 Soundcraft KMZ

continued from page 108 —

The kick drum for instance goes to Daryl's monitor, to the main monitor console, and out to the house as well.

On Stage Intercom

After a while, most monitor mixers get quite adept at sign language and secret signals. A tug on the right earlobe could be a pre-determined signal to turn the monitor up while a left earlobe tug means to turn it down. You hardly ever see left tugs, however. Covering the ear with the hand is a sure sign of low monitor level while an index finger in the ear could either mean just the opposite or the same thing. I love to watch the movements a band goes through for the first couple of tunes. Well there is an easier way; unless, of course, you really enjoy having internationally known stars scream at you in front of 10,000 people. The Captain and Tennille solve the problem with a two-channel Clearom intercom with the base station located at the monitor console. One channel is for communication with the house mixer and the other channel allows Rod to communicate with various musicians onstage. The drummer has a station, the second keyboard player has one and Daryl has one. The other people generally have good eye contact with Rod and communication with them is not a problem. Daryl's system is quite interesting — it uses a Clearcom King Biscuit speaker box coupled to an AKG model D58-E close talking microphone. There is a switch on the mike but it is generally live all the time since it is of a noise-cancelling design and Rod doesn't hear too much sound through it unless Daryl gets right on top of it. Clearcom telephone handsets are now used at the other stations instead of the standard headsets because they don't require two hands to put on and they automatically shut off when the push-to-talk switch is released. If the drummer had a regular headset and forgot to turn the mike off, the system would be rendered useless due to the constant barrage of drum sound in all the headsets.

Daryl doesn't have to pick anything up — all he has to do is go over to his mike and speak into it. Rodney can either see him do this and either picks up his handset or listens to Daryl on a regular headset if he happens to be wearing it at the time. Sometimes Rod uses a Shure SM12 combination headset/microphone which has the advantage of leaving one ear free at all times. Rod has installed a remote signal light on top of his console which is wired up to the Clearcom and alleviates the necessity of having to look over at the master station in the effects rack to see if someone needs his attention. A red and green light indicate either channel one or channel two. The belt pack out at the house console has also been modified by wiring in a Mallory Sonalert which puts out a high pitched beep to attract the mixer's attention. There is a switch to defeat it if necessary. The drummer's belt pak is fitted with a clip to attach it to the high hat microphone stand and thus keep it within easy reach.

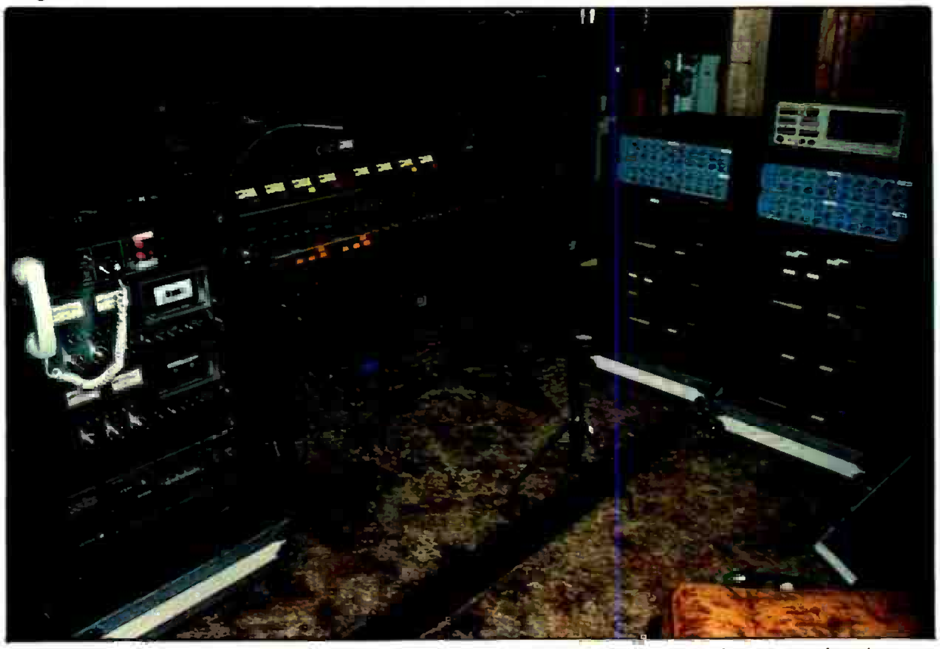
A further aid to onstage communication is the talkback microphone which is a standard feature on the Yamaha PM-1000. The signal from this mike can be routed to any of the outputs including the cue output. This enables Rodney to speak to any individual through the separate monitor busses without disturbing the rest of the musicians during a sound check. Of course, it also allows him to talk to anyone without a Clearcom during the show if necessary.

Mixing Approach

Rodney Pearson's approach to mixing monitors can best be described as a passive one — most of the action taking place during set-up and sound checks.

There are exceptions, of course, but generally very little mixing is done during the course of the show. One thing Rod had seen to be a problem back when he was mixing the house system for other groups was the prevailing attitude whereby someone new was put on the monitors and then eventually moved up to the house mix position — providing he lasted that long. Rod felt that these aspiring house mixers "seemed to

Figure 6



think that they were going to mix the monitors and play around with the levels a lot. I have found from my experience that the less you do the better it turns out to be." If a situation does come up where someone wants something changed, Rod generally tries to find some other way of handling the problem before changing the mix. The difficulty may have been due to someone not using a mike properly during a particular song and sometimes just a word concerning this will take care of the situation. It's necessary for performers to have a reference. If you take away their frame of reference by turning monitors up and down it becomes more difficult for them to determine if they are loud enough or not. So The Captain and Tennille try to maintain an accurate frame of reference at all times. Consistency in monitors is very important, especially on a string of one nighters where *something* has to stay the same show after show. It's difficult enough for the house mixer to have to adjust to all the various changing conditions. And, since the sound of the monitor system, especially one of this size, does get out into the house any change in the monitor mix can be heard to some extent. This can throw off the house sound mixer when he's least expecting it. He ends up reacting to and compensating for whatever the monitor mixer is doing. In small halls, Rod will sometimes back off on the overall monitor level to prevent interferring with the house sound. Everyone in the group is very understanding of this — it is discussed at the sound check and adjustments are made at this time.

With most groups, just turning down the monitors in an attempt to lower the overall stage level doesn't always work due to the live guitar amps on stage which the artist has total control over. And telling all the musicians to turn down isn't always successful since they all have their own concept of what "down" means. When the balance between instruments changes the mixer finds he has to readjust all his levels. And then, of course, the musicians re-balance depending on what they then hear through their monitors. Ad infinitum.

With The Captain and Tennille's system,

— *continued overleaf*

for additional information circle no. 55

however, where just about everything goes direct and is then fed back to the musician through his own monitor, the situation is much more controllable. The overall level can be lowered while still maintaining the relative balance between instruments. Seasoned monitor mixers will immediately recognize the benefits of such a system. Imagine giving everyone in the group just the amount of lead guitar they want instead of fighting with and compensating for the killer-watt output of the guitarist's amp on stage. House mixers no longer have the problem of a loud localized sound source destroying the mix for the first twenty rows. The hard part, of course, is getting the guitarist to trade in his pre-CBS Twin Reverb for a direct box. And rightly so — *that* sound is extremely important to the guitarist and is quite difficult to electronically recreate to his satisfaction. Most amp heads made nowadays however, have extra gain stages that approximate the sound and sustain of an overdriven amp and speaker quite well. They also tend to have direct outputs designed to feed console inputs directly at line level. Hopefully, more sound men will explore these possibilities — I think the benefits are well worth the effort.

One of the biggest advantages of this technique is immediately apparent in theaters-in-the-round that use rotating circular stages in the middle of the audience. A stack of amps onstage becomes a focused sound barage that sweeps across the audience and totally destroys the mix that the audience hears. The house mixer finds himself choosing between mixing for the people that face the amps or for those who are momentarily on the back side of them — a decision made extremely difficult by a constantly rotating stage. A much more even mix results when the instruments are taken direct and all amplified sounds eminate from uniformly hung speakers above the stage.

As complex as the system is, it had to be designed to go together quickly and flawlessly if it was to work at its full potential. Much thought was put into the packaging and hookup procedures and the result is a remarkably quick installation. Since the stage setup is the same for each show, it's possible to bunch a lot of the cable together that always go to the same location on stage. This is another advantage to custom designing a system — if you were renting a different system for each show all these individual cable lengths would have to be figured out each time.

In this case, howeve., all eight speaker lines that go to the back of the stage are laid out and connected the first time and are then tie-wrapped together into one neat bundle. This makes for a more efficient and accurate installation and as a result sound checks are quite quick and fairly predictable — generally lasting only two or three numbers. In one instance they went from the Tennessee State Fair, which was an outdoor show with a loud PA to an indoor arena with typical arena acoustics. The monitor system was set up and when the group arrived for the sound check they were happy right away. No adjustments had to be made at all, not even to any individual mixes. Rod says the system is so predictable and consistant that, given a time problem, it is possible to set up and do a show without a sound check. He's confident that the system will be pretty accurate to start with, and only minor tweeking during the first song will be necessary.

Visually Attractive

The speakers are designed to look like part of the set instead of being dirty black boxes coverd with shipping labels. The cabinets are painted white with dark grey foam grill cloths and are transported in individual shipping cases. There are no handles or recessed latches visible. They blend in perfectly with the black and white set which comes complete with a portable white floor and white grand piano. You'll also notice from the performance photos that there are no monitors down front to block sight lines. This is an extremely important consideration in places like Las Vegas where people sitting up next to the stage pay $30 or so for the privilege of being close to a real, live performer and not a loud wooden box, however attractive. In The Captain and Tennille's, case monitoring up front is handled by the side fill speakers.

Cable grouping also contributes to the clean line.

This concept of setup and appearance was developed for the Vegas-style showroom and is carried over on tour whether the venues be theaters-in-the-round or state fairs. It takes a little more time and attention originally but the hours saved setting up on the road make it all worthwhile.

The system has been operating for over 18 months without any major hitches. The only real problem resulted from the time Daryl attempted to recreate the sound of an earthquake with his synthesizers. Needless to say, a few woofers decided they'd rather quit than fight, so the UREI 1176 limiter was brought in to help keep the woofer cones within their specified excursion range.

Rumbo Recorders

As I mentioned earlier, this entire system was set up for rehearsal sessions inside the comfortably spacious studio at Daryl and Toni's new Rumbo Recorders facilities. The room was so large and well-designed that there was actually enough room left in the studio for over one hundred packing and shipping cases with lots of room to walk around. A large cleverly designed artificial skylight overhead helps contribute to the bright open effect of the room.

Located about twenty-five minutes from downtown Los Angeles at 20215 Saticoy, in Canoga Park, the studio is yet another example of The Captain and Tennille's non-compromising dedication to quality. Daryl and Toni wish to stress that the facility was not built for their exclusive personal use but was designed as a top-of-the-line, cream-of-the-crop, pick-of-the-pack, state-of-the-art studio available to anyone who picks up the phone and dials (213) 998-5398.

Built by Rudi Breuer, the overall design was a result of the cooperative input of all involved: Daryl, Toni, Rodney, engineer Roger Young, and of course, Rudi himself. "This was one of the more-or-less ideal jobs I've ever done," Rudi explained. "It was determined that the only way to go was first class. It may prove to be more expensive to build that way but the end result is worth it."

Of the many unique features of this facility, the first to catch my eye was the sunken drum booth. As can be seen from the photo on the front of this issue, the booth is totally glassed in and the drums are all sitting under the level of the main studio floor where extensive sub-floor trapping prevents the drums from leaking into the main room. (Sketches of this design were shown in the April 1979 *R-e/p*, Riordan, J., *The Rudi Breuer Approach*, page 45, Volume 10, Number 2.) The slanting glass walls are all non-parallel and close enough together to prevent any smearing slap-back or echo. The space actually sounds a lot bigger than it is and reports are that it gives a nice open live sound to the drums.

This was the first of several such sunken drum booths Rudi is building. Besides the drum pit there are three separate isolation rooms adjoining the main studio and are all quite visible from the large control room. All the rooms are floating on separate concrete slabs of differing thicknesses to cut down on sound transmission and sympathetic vibrations between areas.

The studio acoustics have been made variable to adjust to a client's preferences. Pivoting wall panels are hard on one side and absorbent on the other. There is no permanent flat ceiling surface — instead ceiling traps of various densities and construction are individually covered with material to match the decor and are hung in such a way that they can be moved around to different positions or removed altogether to provide a totally variable acoustic environment.

They have just taken delivery of a new 52 input x 48 out Neve Model 8088 console fitted with NECAM faders. This is the largest complete Neve recording console in the United States at the moment and is specifically designed for dual 24-track recording. A separate monitor section was specified with the board, a system they feel is quicker to operate than the in-line system. From their extensive experience with monitor systems in general, they know that if things don't happen quickly, even in a "relaxed" studio situation, they often don't happen at all. In addition, again based on their road experience and in keeping with the current trend toward separate headphone mixes in the studio, Rumbo will be offering nine separate cue mix channels, all musician controllable from the studio.

The two 24-track recorders are Studer A800s and the control room monitors are UREI 813 Time Aligned™ units. The studio will, of course, offer all the standard goody boxes.

Oh yes, one more thing. The hot tub is 8 feet in diameter — at last there will be room for the roadies! □□□

"TURN-ONS" for AUDIO ENGINEERS

by **BEN W. HARRIS**, *Chief Engineer*
Ground Star Laboratories
Division of Ronnie Milsap Enterprises, Nashville

Through the years, I have had to produce numerous circuits to accomplish various tasks and functions. Many times those circuits evolved around audio switching and indication. The following article describes some of those circuits and their uses. I do not mean to imply that these designs are totally mine, having had input from many knowledgeable people.

The first circuit described will be the building block for later circuits covered in this article. Figure 1 consists of one SPST switch, one ½-watt 4.7 kilohm resistor, and one 2N3904 transistor. The load can be any DC voltage device, including relays and lamps. For discussion purposes we will assume the load to be a 24 volt relay, which is close to the maximum voltage this circuit can handle. First, the relay is connected to the +24 V supply, and its common or negative pole to the collector of the 2N3904 transistor. With the switch in the "open" position, the relay is not activated for current cannot flow from collector to emitter to "ground" because the transistor's base is open. When the switch is connected to the +6 VDC, a bias voltage is applied to the base of the 2N3904, forcing the collector to emitter to conduct, activating the relay. The reason for such a circuit is to eliminate the need to switch directly heavy voltages such as the 24 volt supply, thus reducing "pops" and "clicks" in the audio chain. The diode across the relay further reduces transient spikes.

In some very sensitive circuits, "pops" and "clicks" may still be present. If this is the case, the circuit of Figure 2 will still further reduce this problem. The two circuits of Figure 1 and Figure 2 are practically the same, except that the SPST switch is now shunting or opening the +6 VDC to "ground" through the 4.7 kilohm series resistor. Also, there are two 1N4001 diodes in series with the B+ bias supply. If this circuit is to be used with the relay or load, normally closed, or push-push switch. All other parameters are the same as Figure 1.

Figure 3 is a much more complicated switching scheme. This time I have included a CMOS CD4069 as part of our circuit. The CMOS, as shown, provides a latching operation, while using a momentary SPST normally open switch to control the flip-flop. Each time the switch is momentarily engaged, the CMOS CD4069 alternately provides a B+ bias to the base of the 2N3904 transistor. The B+ in this case would be +6 VDC. However, up to 14 VDC may be used. The alternate action works extremely well and can be used in countless variations.

Figure 4 depicts an actual circuit you Ampex ATR-100 users will appreciate. Many consoles now come standard with remote tape transport controls, complete with 24 volt bulbs. Now, as you may already know, the ATR's remote indication was designed to power the LEDs in the ATR's remote transport control unit, and are not capable of supplying neither the current nor the voltage that the 24 V lamps require. The remote switching works fine with the ATR-100's transport functions alone, therefore, an interface to a +24 V lamp supply for function indication must be designed.

The logic of the ATR is a bit unusual and requires an interverting type circuit. Each indicator send is at a +5 VDC until that function is selected, where it then drops to zero volts or "goes low." There is also a constant 5 VDC supply designed to be the common power rail for the LEDs of the factory remote, as well as supplying logic B+ for the transport functions. The play, stop, fast forward, rewind, record and edit may be remoted with visual indication as follows:

The constant +5 VDC is connected to R1 which, in turn, is connected to the collector of Q1 and the base of Q2. One of the transport function indicator sends is then connected to R2, which is in series with the base of Q1. When the transport function is idle, Q1 remains biased by the +5 VDC from the indicator function. This, in turn, causes the collector to emitter of Q1 to conduct, shunting the constant +5 VDC to "ground," starving the base of transistor Q2 of bias, and forbidding the collector and emiter of Q2 to conduct, now allowing the 24 V lamp to be "grounded." When the transport function is selected, the +5 VDC at R2 goes "low" or to zero volts, relaxing the collector to emitter of Q2 due to insufficient bias at its base. This allows the B+ at R1 to bias the base of Q2, forcing the collector and emitter to conduct, turning "on" the 24 volt indicator lamp for as long as that function is in use. There has to be one such circuit for each function that you wish to have indication on. Therefore, to provide indication for all six transport functions, you must have six such circuits. One other requirement is that 24 VDC "ground" and Ampex ATR-100 "ground" must be common, which is the normal in most studio applications. This inverting type circuit may be used with other recorders with similar transport logic.

In all of the preceding circuits, I have used no more than +6 VDC to bias the transistors, however, up to +24 VDC may be used, except in the case of the CMOS CD4069 which can only handle about +14 VDC. The series resistors may have to be changed in value with other bias voltages, depending upon the circuit application.

I would like to thank the following people who provided much of the information contained in this article: Paul Buff, Allison Research; Gary Carrelli, Valley Audio; and George Cumbee, Audio Creations.

Happy "turn ons!" □□□

FIGURE 1

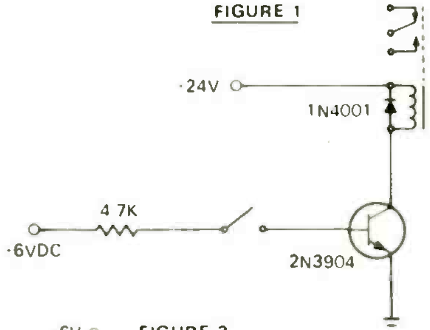


FIGURE 2

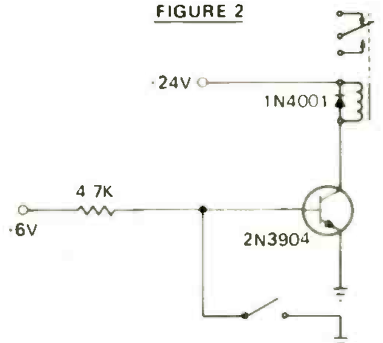


FIGURE 3

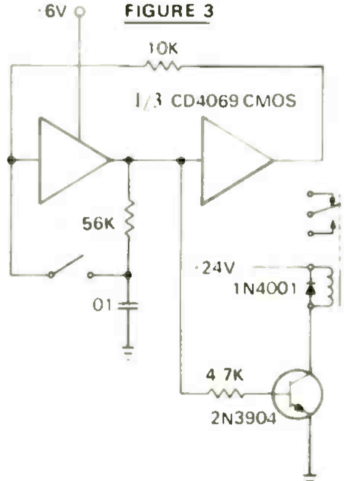
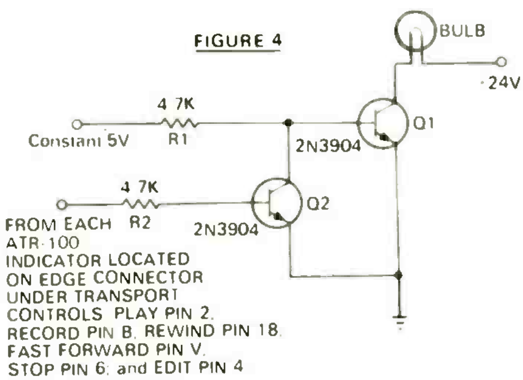


FIGURE 4



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for additional information circle no. 58

Generating the Low Frequency Audio Energy for Apocalypse Now

By John Meyer and Terry Tomaselli

Photo by Robert Peterson, courtesy of American Zoetrope & Dolby Laboratories, Inc.

American Zoetrope, the producers of the film *Apocalypse Now*, wanted to extend the bass response of existing playback systems in the 70 mm theaters that they were going to show their film in this fall. They felt if they could hear and feel the power produced by the low frequencies in their soundtrack, especially in the battle scenes the audience would experience a greater realism.

The existing "standard" theater playback system produced by Altec was designed many years ago before low frequencies were considered desirable or even possible to reproduce; hence, they produced very little energy below 100 Hz. Tom Scott, the theater sound consultant from Zoetrope, was looking for sub-woofer systems with a range of 30 to 100 Hz. We invited him to hear our ACD/John Meyer Studio Reference Monitor with the accessory sub-woofer which extends the power bandwidth down to 25 Hz. Passively crossed over at 100 Hz and driven by the ACD's 125 watt low frequency bi-amplifier, it produced 120 dB at 50 Hz at 1 meter. We had designed signal processing electronics to flatten the response and almost eliminate the effects of energy storage, which is responsible for the "juke box" syndrome or boominess associated with many sub-woofer designs. Ours is a seventh order reflex design with an 18-inch driver and measures 48" x 24" x 24" and weighs 110 lbs.

Mating The New To Old

Tom liked the sound of the system and wanted to hear them at the North Point Theater in San Francisco where American Zoetrope does most of its testing and development. Briefly, the North Point has a 70 mm format with six discrete channels for playback. It has five Altec A-4 speaker systems behind the screen and several small surround speakers placed on both sides of the theater. The Altec A-4s consist of two 515-B Altec 15-inch drivers in a horn loaded enclosure and a multi-cellular horn driven by two 288 drivers. This system is passively crossed over. Each system is driven by an Electro Sound amplifier rated at 125 watts. The surround system consists of Altec 409 8-inch coaxial speakers. Channels 1 through 5 are used for the front speakers and the 6th channel feeds the surrounds. Channels 2 and 4 were used by American Zoetrope for their low frequency effects. The entire system utilizes Dolby noise reduction.

At this early stage, none of us knew how much low frequency level was going to be necessary; very few people have had much experience with low bass.

We placed two of our sub-woofers behind the screen under the scaffolding that holds the Altec system. We interrupted the speaker wires that fed channels 2 and 4 of the Altec system just before the 500 Hz crossover, so that our crossover and speakers were passively fed from the house amplifier while the information above 100 Hz was passed onto the Altec crossover. This procedure is similar to the way we drive the sub-woofers for the ACD/John Meyer system. In the projection booth we ran sine wave sweeps from 30 to 125 Hz into the amplifiers and obtained almost nothing from our sub-woofers. When we looked at the output of the 125 watt rated amplifiers, we found that they produced less than 25 watts at 30 Hz. When we tried to drive the amps harder, they broke into oscillation. The amplifiers hadn't been designed to reproduce low frequencies because it was part of a theater system where no bass was expected.

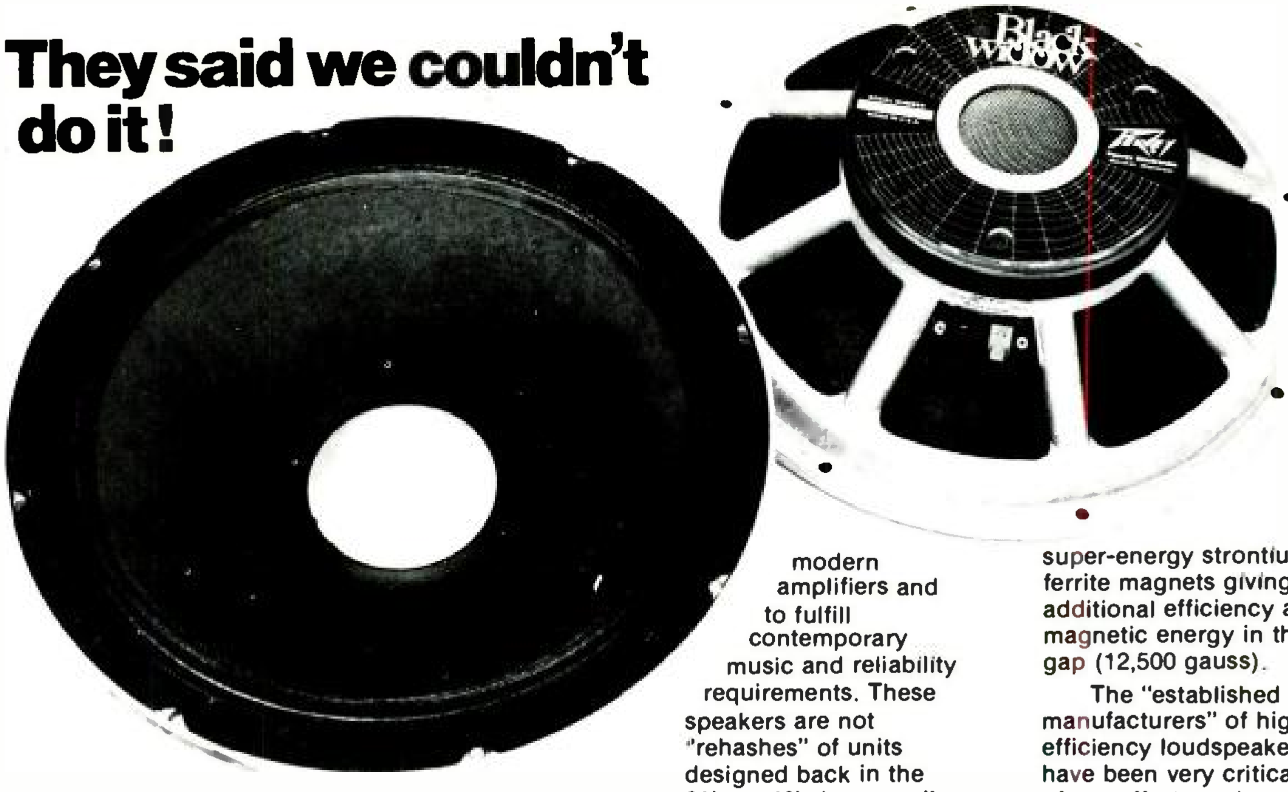
It was obvious that we needed an external power amplifier. This meant that we had to design active control electronics so that we could take a line level feed from the Dolby noise reduction unit and drive our 125 watt stereo amplifier to get 120 dB. Once these were designed, we brought the electronics and the amplifier behind the screen and fed tone into the system and obtained the expected 120 dB at 1 meter. We were now ready to run the sound effect which was a recording of a Howitzer on a 35 mm magnetic film loop played back at half speed. We were still trying to determine if 125 watts would be sufficient power to create the desired low frequency effect. If so we could replace the amplifiers in channels 2 and 4 and drive our sub-woofers with a passive crossover. Tom Scott started the film loop of the Howitzer and we took sound level readings in the center section of the theater.

We were using a Bruel & Kjaer 2209 Precision Sound Level Meter to obtain our readings. We were getting readings of a little over 90 dB SPL measured with an integration time constant of 500 milliseconds (slow). The peak reading was in the high 90s. This was right before the 125 watt amplifier commenced clipping. Due to the high peak-to-average power factor of an explosion, we discovered we were not generating enough bass energy without clipping the amplifier. It was obvious we needed more bass energy to

Meyer Sound Laboratories, Inc., (MSLI) is a new company located in San Leandro, California, building sound equipment for the professional market. John Meyer, the president and founder of MSLI, has worked over ten years in professional sound reinforcement and recording studios. He is the creator of the JM-3 sound system being used currently for Beatlemania and other popular tours. He worked in 1974 for the Institute for Advanced Musical Studies in Switzerland during which time he developed a studio monitor system (the ACD/John Meyer system). John has been technical director for Crystal Clear Records, in San Francisco, and has been a consultant to JVC Cutting Center, in Los Angeles.

Terry Tomaselli, the general manager of Meyer Sound Laboratories, has worked for numerous recording studios as a sound engineer, has toured with several concert groups and has been involved in the development of high-powered lasers.

They said we couldn't do it!



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

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

We did!

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

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

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

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

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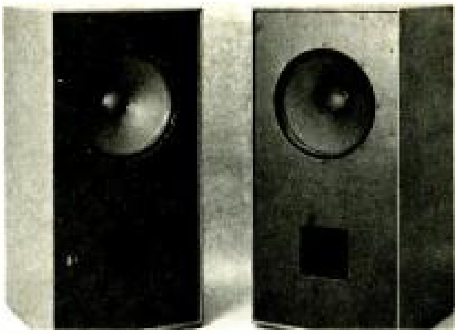

J.M. SUBWOOFERS

be effective. Also, the North Point (which seats 1,000) is a very dry sounding theater which added to the problem of effectiveness. The floors are concrete so you do not feel the sound up through the floor.

We brought in a 250 watt stereo amp, placed the speakers together against the back wall to help them couple better, ran our tones and obtained readings of about 125 dB at 50 Hz at 1 meter. It appeared that we had sufficient power, so Tom set up some pink noise tests using 35 mm magnetic film run through the house systems. A real time analyser showed that the response was down almost 6 dB at 30 Hz when the signal was run through the house electronics. Tom called Dolby labs to have them check their system. Apparently, Dolby, like almost everyone else, was led to believe that the projectors and sync motors would produce audible rumble, that film mixers never put any low bass on tracks and that the speakers wouldn't reproduce it if it were there anyway.

We demonstrated to Dolby that we could accurately reproduce and track bass and that this was not a sense-surround type rumble device that simply injected low frequency monotone noise at predetermined intervals with a toggle switch type on/off response curve. Dolby labs, not wanting to be a limiting factor in the playback chain, took about a week and redesigned their system so that it was flat down to 30 Hz.

The entire system was working well and the Howitzer firing on the film loop had a sharp crack, a low frequency blast, and finally a rumble at the end. Tom Scott brought in Walter Murch, the sound designer and mixer for *Apocalypse Now* to hear the system. The sub-woofers were A-B'ed in and out from the middle of the theater by means of a remote control cable which enabled Tom to turn on or off any channel and control overall system level. Walter decided that there was not enough power to give him the intensity he considered essential.

At this stage in the project, American Zoetrope didn't want to bring in additional speakers and amplifiers because the cost would have to be born by the theaters and/or film distributor. We told Tom that we could probably redesign our system to obtain more power.

Essentially we needed a more powerful amplifier. After testing several, we found the Crest 3501 which produced 800 watts continuous into an 8 ohm load in bridging configuration. The 18-inch drivers were modified to handle more power. At these power levels we decided that we needed to protect the system in some way without the necessity for an operator to monitor the system. The speakers couldn't be fused because they were located in hard to get to areas behind the screen.

We designed an RMS limiter circuit to put into our control electronics and a low pass filter to filter out the high frequencies. It allowed undistorted 1,500 watt peaks to pass through the amplifier unaltered if they are of short duration. When the continuous power exceeds 325 watts per speaker, 650 watts for the two speakers on the one amplifier, the limiter will hold the power to this maximum level. This method of limiting is very soft and unnoticeable. It keeps the voice coil from overheating to the point of failure.

130 dB At 50 Hz

In this configuration, we were able to obtain 130 dB at 50 Hz at 1 meter in front of the speakers. At these levels it was very difficult to talk as our voice boxes were being modulated. The type of feeling produced by low frequencies at this level is very different, more a feeling of pressure than anything else. The Howitzer test tape was again run for Walter, who insisted that an earlier system he had heard was louder. We couldn't believe that anything else anywhere near our size and cost could be louder so we brought the other system in to evaluate it. It produced 10 dB less level than ours on our sound level meter, but subjectively sounded loud. We checked this system with tone bursts, slowly varying the frequency from 30 Hz to 100 Hz and noticed there was very little subjective change in the sound regardless of where the fundamental frequency of the tone was set. We also noticed that the system exhibited an incredible amount of hangover — in other words, was a very boomy box. As the North Point was very dry, this ringing and long decay time gave the system subjectively enormous power. Our system measured 10 dB more, but didn't sound "loud." We had purposely gone to a great deal of design effort to eliminate "boominess" as unusable in studios because it tended to obscure information we wanted recording engineers to be able to hear clearly in our monitor system. Our system was very powerful, sounded tight and tracked bass accurately, but Walter was right — subjectively, it didn't sound "loud" in a large dry place like the North Point Theater.

After talking to Walter and other mixers,

Imagination fuses two powerful illusions.

Experimentation leads to the discovery of new realities.

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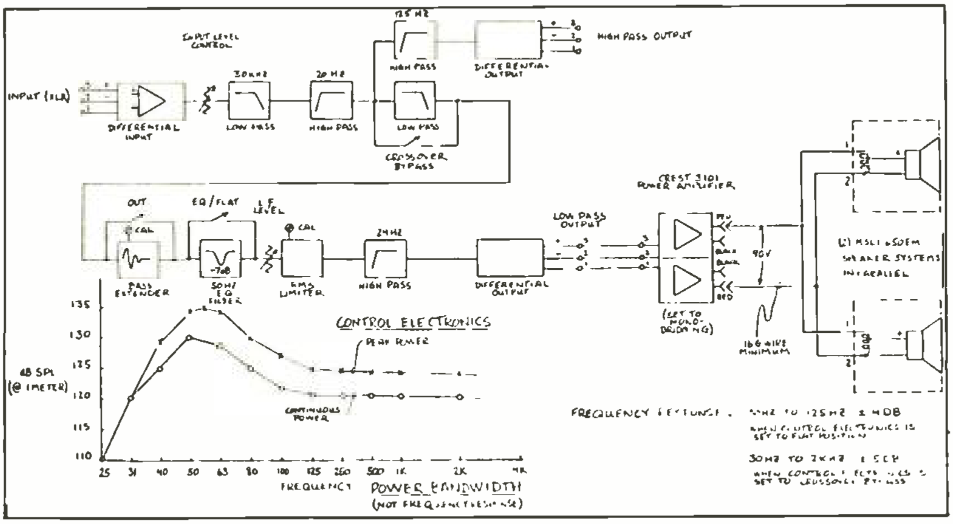

we decided to design a user variable "bass extender" circuit which would allow the duration of the signal fed into it to be stretched to an almost continous ring, or anywhere in between from an ultra-tight monitor-accurate snap to a super-boom box, depending upon preference or environment.

We went back to our laboratory and designed an electronic circuit which would simulate this boomy property and included it in our control electronics. This circuit could be adjusted by a small trim pot underneath the front panel from a very short ring to a long one or could be switched out altogether by a toggle switch. We also included a peaking filter set to 50 Hz which could also be switched in and out. The ringing circuit only controls the phase of the signal, therefore has no effect on continuous tones. It will only alter an impulse by causing it to ring. When the ringing circuit, called the bass extender, is set to maximum (fully clockwise on the trim control) there will be a certain amplitude loss due to the energy storage nature of the circuit. The peaking filter is included to restore the amplitude of the stretched impulse. Also included in the control electronics is the active crossover so that the high frequencies can be sent to the house system and the low frequencies are sent to the sub-woofer system.

Sounding Loud

When we brought our new control electronics back to the theater, and set the bass extender to ¾ of the maximum ring, everyone liked the sound and subjectively it was very loud except at the back of the theater.

We now have learned, through experience, just how much power and duration of signal was necessary in order for the listener to hear and feel low bass. It is necessary to have enormous power available to be effective. Everyone concurred that another two speakers and amplifier were necessary. After installing a total of four speakers and two amplifiers plus control electronics in the projection booth we were able to obtain 100 dB SPL at the back of the theater with the Howitzer track. Walter Murch was very pleased and decided to have a sub-woofer system installed at the American Zoetrope studio so that he could mix the sound and hear the bass reproduced accurately.

The North Point was showing (in the afternoon) a science fiction film *Alien* which has a giant space ship landing scene. With our system running on the effects channel the sound of the ship landing was incredible. With accurate tracking of amplitude, frequency and extended duration, the realism factor was greatly increased. The ship's approach was perceived as higher frequencies at low level when at a distance. As it came nearer the amplitude and lower frequencies increased to thunderous, long continual pounding as it actually landed.

By actually testing our system during the running of the film, we became aware of the extensive frequency and dynamic range possible with 70 mm magnetic film. It became necessary to further protect the speakers from excessive voltage peaks by placing an audioformer in each speaker which allows approximately 75% of the peak voltage to pass to the voice coil, limiting the peaks that each speaker would see to under 500 watts, while not effecting continuous power at all.

The variable bass extender has proved to be very useful for different installations. For instance, the Cinerama Dome, in Hollywood, is very reverberant and the system worked best with the bass extender circuit switched off. While the North Point, in San Francisco, being dry, needs more bass extender, other theaters have proved to be somewhere in between.

FM Productions, Bill Graham's sound company, is handling installations and provides leasing arrangements for the sub-woofer system. Stephen Neal, from FM, used 16 theater systems stacked up for a Grateful Dead concert in a 7,500 seat hall to shake the concrete walls. It may be that the bass will become a new dimension for more than just theaters. □□□

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dBs can be Hazardous to your Health

by Martin Polon

Many people are involved in the development and advancement of audio as an art form for our times. Audio professionals enjoy their creative work and enjoy the music that is being enhanced. State-of-the-art electronics have brought audio systems to the limits of measurable distortion. Digital electronics are gradually expanding the flexibilities of the 3 Cs of audio. These three Cs: create audio, control audio, and communicate audio, are at the center of the professional audio experience.

One element is common to the creation, controlling, and communicating of audio. Whether sound is being recorded, transferred, re-recorded, mixed down, duplicated, blended, reinforced or broadcast, it has to be monitored. The process of monitoring becomes the electro-acoustic connection between the electronic element of audio and the creative interaction of man.

The human exposure to the monitored audio becomes more than a link in the creative chain of the three Cs when that exposure leads to whole body damage. The human body can be compared to a programmable calculator with continuous memory that stores everything entered into it. The body, when exposed to excessive energy, assumes these characteristics by permanent unalterable tissue change. Exposure to energy in any form can lead to mutagenic or carcinogenic damage. The only variables are the duration of exposure and the intensity; how long and how much. In this way, the body is an unforgiving storage medium, compiling exposure.

Energy of any kind has the property of causing body damage. A gas stove can burn your hand. The sun will cause severe sunburn if exposure is not limited, or even lead to skin cancer for susceptible individuals. Nuclear energy has a long list of mutagenic or carcinogenic consequences of exposure. All three energy forms share the characteristics of toxicity. But, the three energy forms have varying levels of visibility. No one will put their hand into a gas flame because the consequences are so obvious. Less obvious is exposure to the sun, and worse still the sun is quite attractive. Nuclear energy is not attractive at all, but it is totally invisible.

Audio is an invisible source of energy. The use of high level audio is attractive, both to the audio professional and to the skilled musician. The invisibility and attractiveness of electro-acoustic energy is a paradox because at the high levels considered desirable for listening and monitoring, the physiological tools used for evaluating quality of performance are severly impaired.

The lexicon of sound induced damage to the body includes such terms as noise, pollution, invisibility, exposure, duration, intensity, and susceptibility. The study of the effect of negative factors on mass populations is called Epidemiology. The word is derived from the study of epidemics, and describes the impact of a disease or pollutant upon susceptible groups of people. Epidemiologically speaking, high level sound is noise polluting the environment. It is an invisible source of energy, which damages the body based on how much exposure there is (time × level). The affect on any given individual is difficult to predict precisely, since each human being has a different genetic signature based on the DNA (deoxyribonucleic acid). What is known is the scale of damage that can be predicted with exposure to sound pressure levels in excess of 90 decibels.

Damage to the body from energy occurs over a long time span. High energy levels in audio monitoring and reproduction are a fairly recent phenomena, and some of the predictable sequences of damage, especially in the non-hearing areas, are occuring on a continuing basis. Hearing damage, compared to whole body damage, begins rapidly and deterioration can be accelerated in a shorter time frame. The "inflation" of audio levels has reached a point where it is

Martin Polon is director of Audio-Visual Services at UCLA, in Westwood, California, where he has been since 1961. He is a graduate of UCLA and is currently working on a doctorate in Educational Technology. Mr. Polon is also a noise consultant for the aerospace industry and related technologies. He is session chairman at the November New York AES Convention for the topic on Enrivronmental Audio/Acoustical and Medical Impact on Man. *His field of research covers the epidemiological effects of sound on the human body.*

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Martin Polon:

dBs can be Hazardous to your Health

not uncommon to find concert environments where backstage monitoring is done with six theatre-type speaker systems and the amplifier drive is 2,000 watts. Sound levels in excess of 110 dB (A) will be found during the entire performance. Similarly, a studio mixdown session might take as long as 24 hours, during which time the monitoring of as many as 32 tracks will take place at levels ranging from 100 to 120 dB (A). Thousands of watts of amplification are found in most studios, and the trend is towards more power in most applications.

The point is that most audio professionals and enthusiasts involved with high level sound are currently experiencing greater levels of exposure than their predecessors did in years past. Just as an inflated economy leads to eventual acceptance of higher prices by the population; the use of high sound levels constitutes an "audio inflation" that is eventually accepted as the norm by people working in the audio profession.

Acceptance of audio inflation is like a double edged sword. Extreme caution has to be exhibited in its use, or somebody is going to get hurt. On one side there is the acceptance of higher amplifier power and loudspeaker power handling capacity as tools for achieving extraordinary transient response. On the other side, we have an acquiescence because the hearing mechanism may be damaged without the condition being recognized.

EXPOSURE STANDARDS TO HIGH SOUND LEVELS

COMPARISON OF EXPOSURE TIMES PERMITTED BY THE BRITISH OCCUPATIONAL HYGIENE SOCIETY (BOHS), AND THE INTERNATIONAL ORGANIZATION FOR FOR STANDARDIZATION (ISO)

BOHS Permitted Duration, Hours per Day	*Sound Level SPL, dB (A)*	*ISO Permitted Duration, Per Week*
12	88	
8	90	40 hours
7	90.5	35
6	91	30
5	92	25
4	93	20
3	94	15
2	96	10
1	99	5
0.5	102	150 minutes
0.25	105	75
0.125	108	40
0.0625	111	20
0.03125	114	10
0.015625	117	
0.0078125	120	
0.00390625	123	
0.001953125	126	
0.0009765625	129	

PERMISSABLE EXPOSURE UNDER THE OCCUPATIONAL SAFETY & HEALTH ACT

Sound Level dB (A)	*Permissable Daily Exposure, Hours*
90	8
92	6
95	4
97	3
100	2
102	1½
105	1
110	½
115	¼ or less

One way to state the potential for hearing and whole body damage from high level sound is to view various standards used for occupational exposure to noise. In all cases, these standards are designed to prevent damage by restricting exposure; regardless of content. At sound levels higher than 110 dB SPL (A), the damage caused by high level sound can be calculated with a stop watch. American standards were established by the Department of Labor, and the input of the U.S. Public Health Service, and the Environmental Protection Agency. The current Occupational Safety and Health Act standards allow a maximum of 15 minutes exposure, per day, to a level of 115 dB. Far less lenient are the standards imposed in the United Kingdom by Her Majesty's Factory Inspectorate based on "The Code of Practice for reducing the Exposure of employed Persons to Noise."

These standards in the U.K. are also endorsed by the British Occupational Hygiene Society. The standard calls for a maximum exposure of 112 seconds, per day, to a sound level of 114 dB (A). Even more restrictive, are the figures used by the International Organization for Standardization and recommended to the entire world. The ISO document #R 1999-1975, calls for a weekly totalized exposure of 10 minutes to a sound pressure level of 114 dB (A).

That these standards do not agree, with the American standards being the least restrictive, illustrates the dichotomy of risk assessment in high SPL exposures. The U.S. standards were based on the

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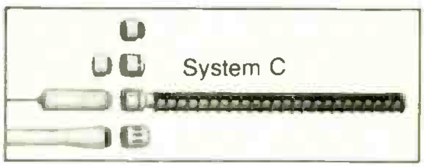
When boom mounted, the CS15P has better gain-before-feedback and a better signal-to-noise ratio than most shotguns. It's phantom powered and it's rugged.

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

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

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



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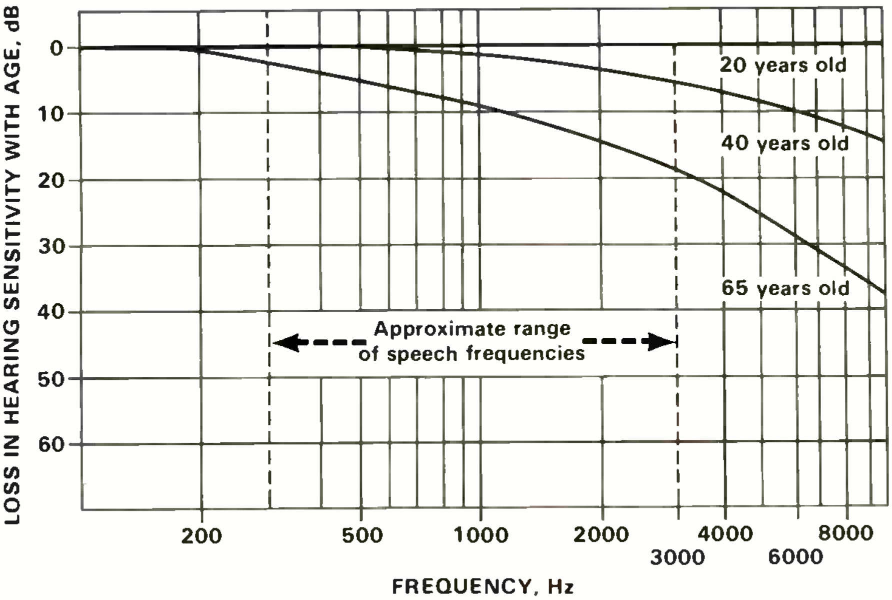

Figure 1. Audiograms showing the typical reduction of hearing sensitivity with increasing age. The approximate range of speech frequencies is shown.

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assumption that 90 dB was the maximum allowable level for 8 hours of exposure. The British and International standards also calculate base exposure with an 8 hour/90 dB (A) basis. But, the method of calculating the time - intensity trade-off for exposure above 90 dB is the point of departure between the systems used on the two continents. The question of intermittency was used as the justification for the so-called 5 dB rule in the U.S. That is, each increase of 5 dB above 90 dB, would be matched by a corresponding decrease in time by one half. Thusly, the 95 dB exposure time would be only 4 hours. The hypothesis was that intermittent sound of low or moderate levels does not cause the same kinds of temporary damage to the hearing mechanism as more persistent sources of sound. But, the supposition is based on low level, evenly spaced sources of energy. At high levels, such as 100 dB or greater, and without regard to content or spacing, damage is far more severe than this criteria allows for. The European approach is to use a 3 dB increase to halve exposure time. This is becoming favored in this country as well. Thusly, at 99 dB, exposure would be reduced to a time frame of 1 hour per day, five days per week.

What is significant to the audio professional, is that the criteria for risk assessment from sound exposure do not agree, but that the most lax interpretation allows for only 15 minutes exposure per day at a level of 115 dB SPL (A). *The continued exposure to higher levels and/or at longer durations can cause damage to the entire body.* What is most disturbing clinically is that the damage to the hearing mechanism constantly increases over a long time frame, reducing the awareness of any change in level, while the rest of the body is being impacted. Damage attenuates hearing perception. (Figure 1)

The human hearing mechanism has a built-in loss factor to start with, that is a result of aging. This condition is known as Presbycusis. Although it is as inevitable as the other consequences of growing old, it is not of itself a condition which will necessarily disenfranchise the working audio professional. The deteriorative affects are predictable and are confined primarily to high frequncy information. The condition is not totally understood medically, but the celular degeneration brought by advancing age seems to affect the hair cells of the inner ear. The cells are not destroyed, but are deteriorated, reducing function. The basilar membrane, also a part of the inner ear hearing mechanism, stiffens with age limiting response. Blood supply reduction to the inner ear is involved with the process as is the general brain cell deterioration common to aging. A normal hearing curve for a twenty year old male will be relatively flat, while the same individual will have a 15 dB rolloff at 40 years of age, and a 30 dB rolloff at age 65 (rolloff at 8 kHz). This function of aging varies from individual to individual and becomes serious when other pathological impact is present. The presence of high level sound can and does become additive to the degeneration of presbycusis.

The mechanism of hearing has three divisions in the human ear. (Figure 2) The outer and middle ear serve to gather and condition sound for reception and perception in the inner ear. The mechanism of the inner ear consists of a complex system of fluid mechanics, frequency selective reception, and transmission paths to the brain. Sound stimuli reach the inner ear after being funneled through the external ear canal, via the tympanic membrane or eardrum of the external or outer ear, to the three vibration transfering bones of the air filled middle ear (malleus, incus, stapes). (Figure 3)

The damage that high level sound induces in the human hearing mechanism takes place primarily in the inner ear. The inner ear consists of three sections known as the vestibule, the semi-circular canals, and the cochlea. It is the cochlea which houses the mechanism to transfer the mechanical impulses of sound into neural impulses for transmission to the brain. The inner ear is filed with lymphatic fluids, which serve as transfer mediums for various ear nutrients and waste products. The mechanical vestibule and canals are filled with perilymph fluid, while the cochlea operates with endolymph fluid. The cochlea is essentially a triple canal coiled up spirally around a bony axis. At the base of the cochlea, a progressively widening membrane, known as the basilar membrane,

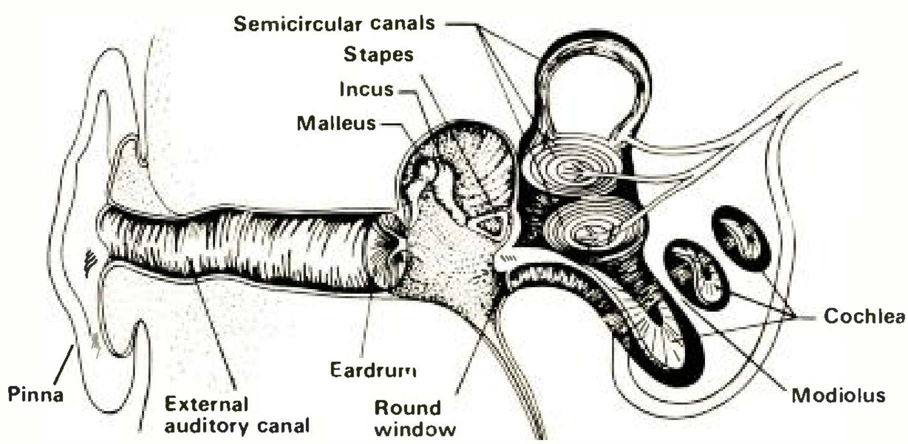

Figure 2. Drawing of human ear showing three subdivisions in cross section.


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Glossary:

Audiological Checkup — A diagnostic experience, using an audiometer to measure the sensitivity of hearing. A record is produced, usually a graph, showing as a function of frequency, the amount measured in decibels of deviation from the standard hearing thresholds.

A-Weighted Sound Level — A measure of sound pressure in which that pressure corresponds to the levels of the various frequency bands; the bands having been weighted to conform approximately with the frequency senstivity of the human hearing apparatus. The A-weighting conforms most closely to the susceptibilities of the human hearing system to energy damage. Defined as a standard internationally in IEC 179 (1973), and in the U.S. as ANSI S1.4 - 1971 (same standards defined B, C and D weightings).

Cerebellum — The structure of the brain responsible for regulation and coordination of voluntary complex muscle movement.

Criterion — The factor which is used to judge noise or sound. As an example, criteria for an acceptable sound would be that it causes no annoyance, does not impact or modify any patterns of behavior, and damages nothing and no one.

dB (A) — The unit of A-weighted sound level. (Also dB (B), (C) and (D).)

Deafness — Total degradation of the hearing function.

Decibel — A unit used to connote relative difference in power, usually between acoustic or electric signals, equal to ten times the common logarithm of the ratio of the two levels. In defining sound pressure levels, it is ten times the logarithm (base ten) of the ratio of the root-mean-square sound pressure squared, and a reference pressure squared.

Diastolic — Relating to the normally rhythmically occurring relaxation and dilation of the heart cavities during which time the cavities are filled with blood.

Frequency — The time ratio of repetition of a sound pressure. This characteristic determines whether human perceptions identify sounds as high or low in pitch.

Hypothalamus — That part of the brain that lies beneath the thalamus, functioning to regulate the body temperature, certain metabolic processes, and other autonomic activities.

Limitations — Legal, and extra-legal prohibitions on the quantity of sound. Amounts to be limited can be identified in terms of sound level, sound duration, exposure, or any other appropriate expression of acoustic output.

Lymphatic Fluids — Clear, transparent, watery fluids that contain white blood cells and red blood cells. The fluids travel through the lymph system to return to the blood stream and act as an agent for the removal of bacteria and certain proteins from tissue, and as an agent of exchange for various other substances.

Medulla — Nervous tissue at the bottom of the brain responsible for circulation, respiration, and certain other body functions.

Motility — Movement, or having the power to move, possibly spontaneously depending on stimulus.

Thalamus — A large mass of grey matter that relays sensory stimulus to the cerebral cortex, and acts with integrative and non-specific functions.

Threshold of Perception — The intensity below which a stimulus, mental or physical, cannot be preceived and will not produce any response or recognition.

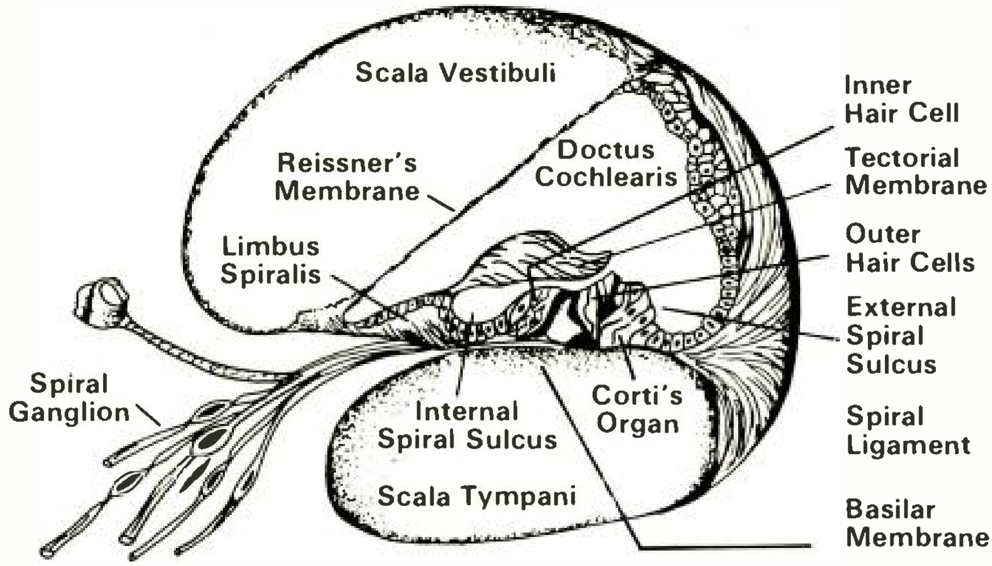

Figure 3. Enlarged cross-section drawing of the cochlea.

supports the hair cells. These hair cells, comprising four rows (one inner row, three outer rows) on the membrane, number about 30,000.

The hair cells are the sensory receptors for hearing which, with supporting cells constitute the Organ of Corti, the human auditory sense organ. The hair cells are innervated by nerve fibers which have their cell bodies grouped to form the spiral ganglion. Axions from this ganglion collect at the base of the cochlea, and pass out to become the auditory branch of the 8th cranial nerve. Information transmitted upon this neural path goes to the medulla in the brain, and then on to the higher nerve centers of the thalamus and cerebellum.

The sensory function of hearing seems to be related to the transfer of acoustical energy via fluid motion in the inner ear, which bends the hair cells triggering electrochemical impulses, which are transmitted as neural energy to the brain. (Figure 4) Discrimination of different sounds results from the analysis in the cochlea, which is mechanical in nature, and in the brain, which utilizes the neural impulses. The language of hearing and of hearing damage uses terms such as "can," "seems," and "will." It is a given that precise examination of human hearing and hearing damage can take place only as an autopsy exercise. While this has contributed a remarkable body of information, especially in terms of damage assessment, the exact mechanisms in a living person remain subject to hypothesis. There has been considerable work done with laboratory animals such as rats, gerbils, and human ear analogs such as chinchillas and monkeys so that the body of medical information is definitive; but open to interpretation in a living creature.

In damaging the hearing mechanism, high energy sound is a function of the "constant energy" principle. The presence of a measured amount of "A" weighted sound energy will cause a corresponding amount of damage to the hearing mechanism. A trading relationship exists between the exposure time and the sound level in dB (A), the product of the two being a measurement of the total acoustical energy received at the

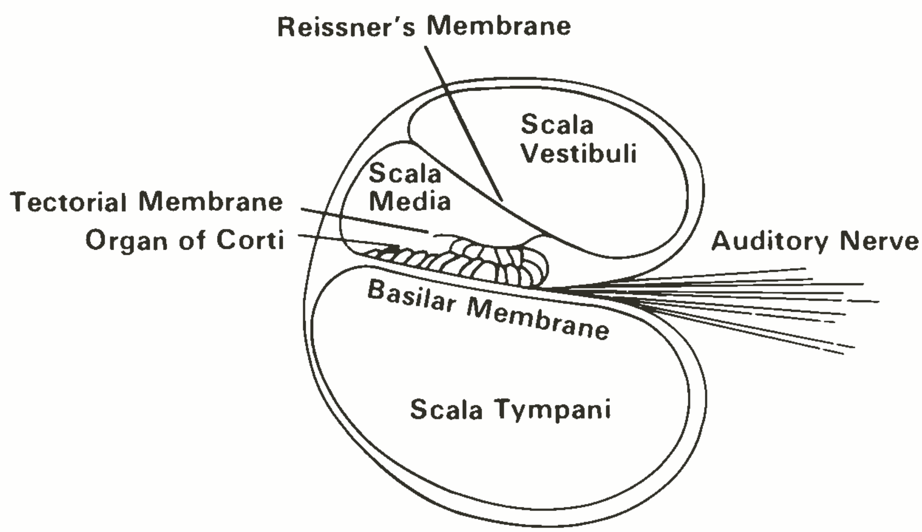

Figure 4. Section through one turn of the cochlea showing its three sections. The basilar membrane, tectorial membrane, and hair cells are not shown in detail.


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inner ear. There is some proportional relationship to frequency in the acoustical energy received, although not necessarily on a linear basis.

Physical damage to the receptors in the inner ear has been identified as having a greater high frequency component. Histological examination (microscopic tissue and cellular analysis) has identified those regions of the basilar membrane having greatest cellular injury. These areas are usually in the base area of the membrane where high frequency discrimination takes place. There are several factors that operate in the concentration of damage at the high frequency sensory cells:

1. The regions of the membrane receiving high frequency information consign higher amplitudes and more sharply defined patterns of response to incoming stimulation.

2. Energy protective reflex actions of the middle ear provide less attenuation at high frequencies.

3. Resonant frequency of the external ear canal operates to acoustically amplify high frequencies relative to low frequency information.

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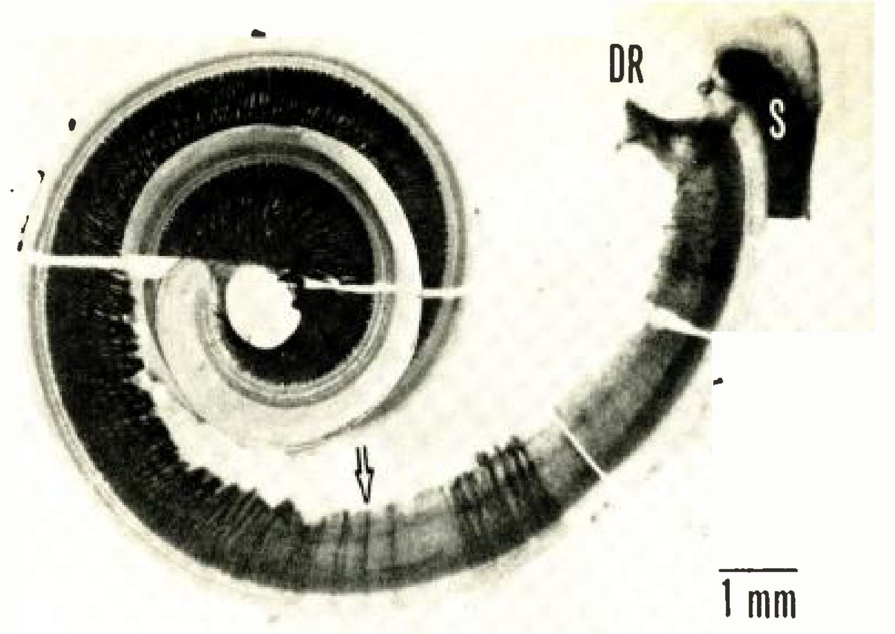

FIG. 5. Membranous spiral and osseous lamina dissected from the left cochlea of RS, age 25, a hunter, showing extensive loss of organ of Corti and myelinated nerve fibers from the first quadrant. Note the surviving nerve fibers at and below the arrow and even near the cecum vestibulare, wherever small patches of supporting cells and hair cells persist. DR, ductus reuniens; S, stria vascularis. (Reprinted with permission from *The Annals of Otology, Rhinology and Laryngology*, 83:294–303, 1974.)

Medically defined, hearing damage from high level sound exposure assumes the following syndrome:

1. High intensity sound will result in a detachment of a portion of the Organ of Corti (hair cells and supporting cells) from the basilar membrane. This appears to be a mechanical consequence of stress developing in the organ during high intensity sound exposure. This is in contrast to the other patterns of cellular degeneration which are not solely mechanical.

2. Damage to the hair cells varies in function. The outer row of cells seems the most vulnerable to noise exposure, while the inner cells degenerate from secondary biochemical degradation.

3. The permeability of the membrane is changed during episodes of high level sound exposure. This seems to accelerate the severe damage to a number of cells during the exposure.

4. When a particularly severe episode of high intensity sound exposure causes a number of sensory cells to degenerate, simultaneously, small holes will exist for a given amount of time, allowing the leakage of fluids.

5. The presence of unwanted potassium ions or endolymph fluids can cause secondary damage in the Organ of Corti. Uninjured cells and nerve fibers will undergo an isosmotic (osmosis) swelling that will eventually rupture many of them. This is thought to be the phenomena that is responsible for the degeneration of many supporting cells and nerve fibers after sound exposure. Figure 5 illustrates cochlea damage.

The scenario just given is the medical explanation of degeneration that occurs when the human ear is exposed to high intensity sound. Each and every exposure produces the affects described. The only variables are the susceptibility to damage that varies from individual to individual, and the duration of time between exposure. Time between exposures becomes important since it is this unexposed time that allows the ear to regenerate and recover sensitivity.

The mechanism by which temporary hearing damage becomes permanent hearing damage is known as TTS (Temporary Threshold Shift), and PTS (Permanent Threshold Shift). A warning of the damage to the hair cells of the hearing mechanism is the on-set of a condition known as "tinnitus." This is a ringing, buzzing or whistling in the ears that is a precursor of inner ear damage. Unfortunately, the use of sound levels above 100 dB (A) can usually mask this reaction. The previously described "equal energy" principle comes into effect as exposure takes place, and the ears accumulate exposure as damage. The terms threshold shift refer to the fact that attenuation of response in the ear is maximum at a frequency of one-half octave above that involved in the exposure. If an exposure takes place, the net result is a notching of perceptive sensitivity in the 1,000 to 6,000 Hertz range.

The goal for the audio professional is avoidance of this notching condition, the threshold shift, and PTS. There is no way to know for a given individual what combination of intensity and duration is going to induce damage, but it is medically advisable to separate exposures. The ideal situation would be to have no exposure over 80 dB (A) for a period of 7 to 14 days after a significant high level episode. It is possible to recover in 16 to 24 hours, if an exposure is not "pathologically fatiguing." That presumes regularized exposure patterns to moderate sound levels in the 85 to 95 dB range. In between the two axis, there is a compromise which would grant some immunity from severity in permanent damage. If there was a staggering of work assignments with levels in excess of 100 dB, the intervening time would allow for recovery.

For the person who uses audio regularly, in excess of 100 or 110 dB, the long term consequences could be viewed similarly to the affect of smoking a pack of cigarettes a day over twenty or thirty years. Mutagenic change will occur in the hearing mechanism. This change can be further complicated by the use of ototraumatic drugs; i.e. those agents used for medical or recreational purposes that will accelerate inner ear damage. Aminoglycoside antibiotics, certain diuretics and analgesics (pain killers), and some stimulants and intoxicants (drugs, marijuana and alcohol) ingested in varying ways fall into this category.

It is apparent that the practicing audio professional or musician who three or four times a week undergoes sustained (2 hours plus) exposure to sound pressure levels of 110 to 120 dB (A) or better will gradually reduce sensitivity by the process of converting TTS to Permanent Threshold Shift. There are two significant results of this syndrome. Firstly, the person involved is going to need more energy to reach the same sense of perceptive accomplishment held prior to the attenuation of the PTS. This slowly increasing need for higher level of input matches the deterioration of sensitivity from hearing damage. The higher the new level, the greater the new level of damage. Secondly, the lack of sensitivity will mask the continuing exposure of the rest of the body to high level sound.

The affect of sound on the body does not stop with the gradual degeneration of the inner ear hearing mechanism. Whole body damage is a phenomena in which the activation of basic defense mechanisms by high level sound involves many different body organs and interferes with certain necessary bodily functions. The perceptions of high level sound trigger primordial reactions that evolved to protect man from dangers of the environment. Attack by predatory enemies was presaged by certain sounds that would give warning. The body responds today as it did thousands of years ago. A noise outside at three o'clock in the morning will activate modern man the same way a saber-toothed cat alarmed the cave man.

The defensive response to perceived

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danger is just that. The body cannot detect the nature of the danger; just the projected perceptions. High level sound triggers regularly body responses that were intended to function infrequently.. The perception of danger from high level sound causes the operation of a complex chain from the inner ear to the brain, where the hypothalamus triggers hormonal outputs, stimulates the pituitary gland and involves the endocrine system. The voluntary nervous system, the autonomic nervous system and the central nervous system are involved to prepare for danger. Muscles are tensed by the main motor nerves. Heart activity, rate of breathing, blood pressure, and other functions are altered to prepare for fight or flight. This series of reactions in man complicate the health pattern by adding stress systemically. Stressing the system to prepare for self-protection is the body's response to high level sound. The consequences are not positive, especially viewed over a long time frame and/or for susceptible individuals.

The most dramatic involvement in terms of health, of high level sound can be seen in interference with the cardiovascular system. In a high level sound episode, the body goes into an Orienting Response, and then assumes a sustained reaction known as the Defense Response. The blood vessels undergo a narrowing of caliber, known as vaso-constriction. This changing in size of the blood vessels occurs at the onset of the exposure episode, and does not disappear in some individuals until the sound ceases or even continues after extinction of the sound source. Other cardiovascular changes include the blood pressure rate (especially diastolic), heart rate, cardiac output, and pulse volume. All of these heart functions are interrelated, and may create further affects. For example, while high level sound does not seriously modify the long term blood pressure rate for a normal population group, patients already suffering from hypertension will probably undergo a moderate further increase in blood pressure. Another area of concern is the interference of certain hormones released during a high level sound episode. These hormones actually cause an increased blood level of serum chloresterol, which the body produces normally. The deposition of placque, which is involved in arterial blockage, is also enhanced during the sound episodes.

There is an interesting contradiction in the vasoconstrictive response to high energy sound. The ear's blood supply is cut down along with all of the other areas supplied by the peripheral blood supply. The ear, however, is undergoing the extreme trauma previously described. The sensory cells need an increased blood flow to remove the metabolic products and to provide ionic exchange. The lack of blood from cardiovascular response to high level sound further accelerates the degenerative process going on in the inner ear. One might say the hearing mechanism "gets it from both sides" when this condition known as cochlear anemia or ischemia occurs.

The continuing exposure to high energy sound creates a stress reaction in the body that significantly involves the gastro-intestinal system. Certain stomach functions are disrupted by abnormal contractions of the abdominal area, and increased infusion of hydrochloric acid causing dyspepsia. Recurring activation of this syndrome will lead to peptic ulceration in susceptible individuals. The intestinal system has increased motility with waves of contractions which coupled with the highly acidic condition of the stomach produce diarrhea and/or frequent loose motions. This syndrome in the gastro-intestinal system, when combined with a diet of junk food, and the on-going stress of demanding creative activity, could explain "Sound-man's Stomach."

Less easily explained, but far more fascinating is the interaction of the creative environment with intestinal motility. Research has established that those individuals who can control the high level sound source, but continue their exposure to it suffer far greater motility problems than those who have no control over the exposure. Even more interesting is the disco related problem with unwanted release of intestinal contents during episodes of high level sound on the dance floor. This reaction to sound has been parenthetically named the "Disco Dump" although it is by no means confined to discos. The presence of stimulus that confuses the brain, along with the reaction to high level sound triggers a simultaneous release of control by the sphincter muscle while the intestines are in extreme motility. Fortunately for the disco business, this phenomena appears to be restricted to susceptible individuals, although susceptibility is enhanced by the presence of alcohol and other ingested stimulants.

The hormonal outpouring during high level sound exposure episodes has a definable consequence of negatives. The hypothalamus in the brain signals the release of a number of hormones throughout the body. (Figure 6)

1. The pituitary is stimulated directly to

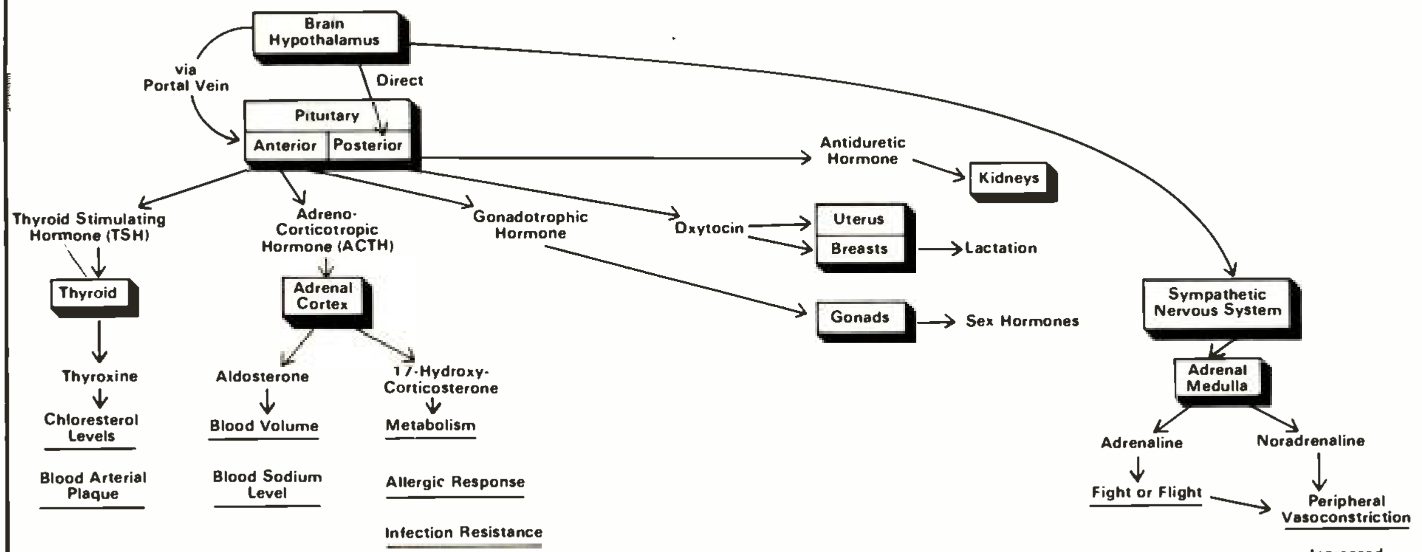


Figure 6. The hormonal chain reaction throughout the body from the brain hypothalamus due to high sound levels.

the posterior lobe by the hypothalamus; the anterior lobe is stimulated by the portal vein from the hypothalamus.

2. The posterior lobe of the pituitary gland (neurohypophysis), releases oxytocin which affects the uterus and lactation in the breast of females, and vasopressin which acts as an antidiuretic to the kidneys (possibly upsetting the body's fluid and sodium balance).

3. The anterior lobe of the pituitary (adenohypophysis) releases three substances of interest:

A. Thyroid stimulating hormone (TSH) stimulates the thyroid which in turn produces thyroxine, which is responsible for blood levels of serum cholesterol, and the deposition of atheromatous plaques.

B. Adrenocorticotropic hormone (ACTH) activates the adrenal cortex, which releases aldosterone and 17-hydroxycorticosterone. These corticosteroids tend to remain active for a long time frame with high level sound exposure. Blood levels of sodium, blood volume, kidney effectiveness, metabolic regulation of carbohydrates and proteins and fats, body inflamation, and body allergic response are all modified by these steroid hormones. Resistance to infection may also be impaired with continued long term high intensity sound exposure.

C. Gonadotrophic hormone is released which may decrease potency in the male by gonadial interference.

4. The hypothalamus activates the sympathetic nervous system, which in turn involves the adrenal medulla which mediates the release of adrenaline and noradrenaline to the blood. These hormones trigger the fight-flight response affecting heart rate, vasoconstriction, etc.

High level sound episodes can also affect the ability to concentrate on performing a task. The phenomena is known as the distraction effect, and is "invisible" as are most of the affects of high level sound exposure. The presence of sound levels as low as 80 dB (A) can interfere with the achievement of cognitive tasks such as those involving memory. Complex tasks, especially new ones that have to be learned in the presence of high level sound (90 dB+), can be seriously interfered with. This interference is especially significant where multiple components or high information loads are involved. The capacity to communicate can be interfered with beyond the inability to detect speech as the brain's ability to process speech information is degraded as well.

Lastly, the presence of continued exposure to high level sound can trigger psychopathological impacts on individuals who have otherwise been primed by internal or external factors. These impacts can range from depressions noted among females during the menstrual period to actual presence in the brain of chemicals normally found in schizophrenia and psychosis. There are a number of other interesting reactions to the presence of high level sound which involve the brain, including interference with vision.

The evidence of interference, degradation and/or mutagenic impact of high level sound on the human body is overwhelming. The facts point to the dangers of continued exposure of the body to acoustical energy greater than 90 dB (A). The current argument circulating among noise damage specialists is where between 70 and 90 dB (A), the real threshold of damage to the body is located for 90% of the population.

This is not to say that music in a working and/or recreational environment does not have value medically. The therapeutic value of sound constituted as music is as well documented as the extent of sound induced damage. The keystone to understanding the paradox of high level sound is that it ceases to be music when the brain cannot discriminate it as music. Certainly at levels in excess of 110 dB (A), if the brain cannot process or detect the articulation of speech, the real cognizance of music has to be diminished.

It is fortunate perhaps that the laws of logarithmic progression require such quantum doubling of power that the much dreaded figures above 160 dB will never be reached with audio systems. This is the hypothetical point where human life is imperiled by audiogenic seizure producing respiratory failure. It is sobering to note two facts, however. Firstly, laboratory animals die from audiogenic seizure at levels far lower than 160 dB. While it has never been documented in man, the other similarities found in the effect of sound on man and animals are defined. This syndrome of painful, sudden death remains an unknown factor in the equation. Secondly, the current measurements made of bands performing in the United States have seen levels in the high 130s and low 140s, "A" weighted. It would seem time to put an end to the inflation of levels in audio and stabilize the technology of monitoring, reproducing, and reinforcing sound. (Figure 7)

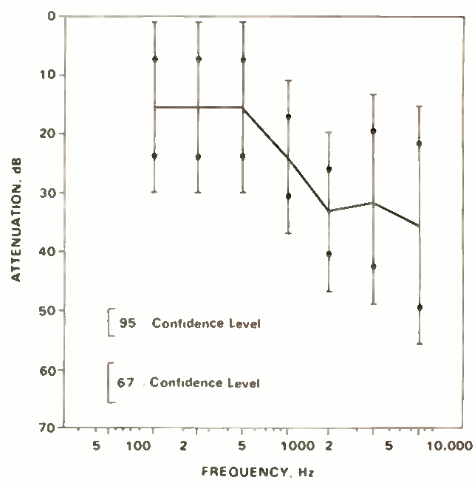


FIGURE 7: Mean attenuation characteristics of an earplug plotted with one and two standard deviations.

There are several techniques that could be used profitably without a noticeable deterioration of sound effectiveness in various applications.

1. Traveling concert systems should be

designed in modules that can be easily split apart and identified for various sizes of halls and auditoriums. Most concert sound systems are designed for large arenas and pavillions. Often, during a tour middle sized halls and small auditoriums are also booked. The standard system is often used for all three kinds of engagements. A well-designed system with thousands of watts is not dangerous where there is a large absorption and a large volume of air to be moved. It is when the huge systems are placed into a small hall with highly reflective surfaces that real difficulty is engendered.

2. Performers should not feel the need to keep up with the "Joneses." Just because "Eddie Machete and The Switchblades" use 20,000 watts of power to the audience or 5,000 watts for stage monitoring is no reason for a group with a well-designed reinforcement system that has had good reviews and which the band enjoys, to replace it.

3. Each band should consider the services of a consultant in acoustical damage, or at least have the sound mixer for the unit well versed in damage versus protection. For instance, a complete audiological checkup would give a specialist the tools to alter both the monitoring setup and the information necessary to equip band members with hearing protection. It might be that some band members could benefit from certain types of hearing protectors while others would use different types of protectors. By tailoring the protection to the band, its music, and its instruments, the protection package could be designed to have the least impact on the creative aspects of the music.

4. The purchase and use of a sound level meter is recommended to all involved in audio: professionals, semi-professionals and talented enthusiasts. The meter could be a professional meter capable of calibration and of meeting legal standards, but it could just as well be an inexpensive unit which may be 1 or 2 dB off, but which would still provide a relative indication. Knowing the level being encountered is the first step in dealing with high level sound.

5. Investigate studio monitoring acoustics to determine if levels in the 120 to 130 dB range are being used to bulldoze improperly designed studio acoustics, especially in old studios or studios where no acoustic consultant was employed.

6. If customers consistently demand high monitoring levels, studios could consider a "hot" room with isolation to keep the high levels from the working staff.

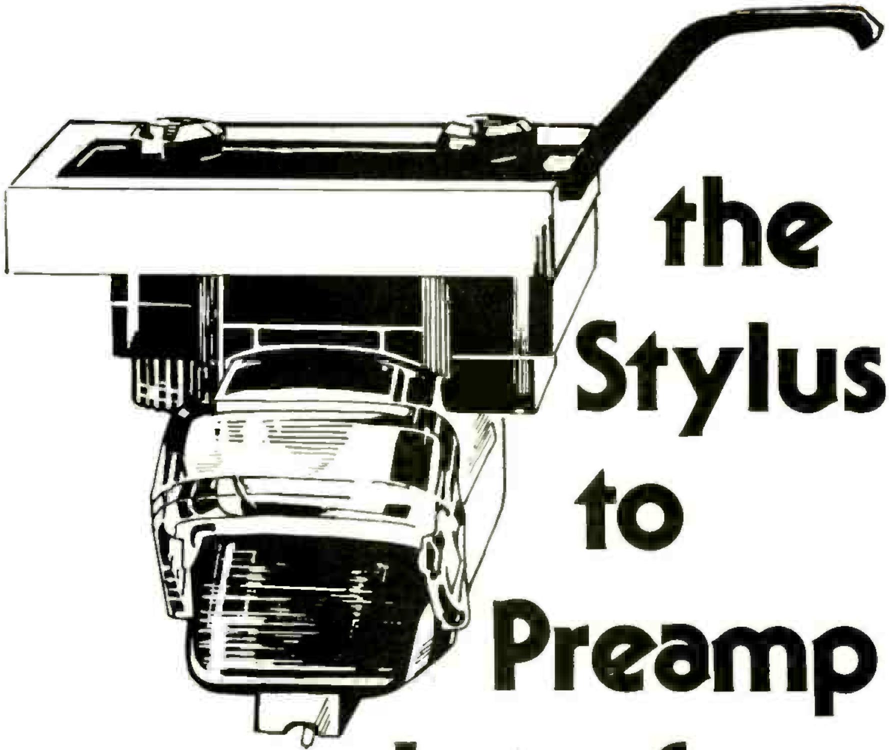
7. Each audio professional should consider the acquisition of some kind of hearing protectors to be used when creative interference is not a problem, or hopefully when perception might be enhanced by a little less level. Certainly 30 or 40 dB of attenuation is not going to hamper anyone at a level of 140 dB.

8. Sound system designers can consider the use of headroom limitations that would bring in distortion for disco or reinforcement systems at a given sound level to protect the audience and performers/users. A peak limiter could be used in the same way as an alternative.

9. Careful use of headphones by all who have need of them. The headphone's direct coupling to the head can create high levels with a minimum of input power.

10. Equalization can be used creatively to reduce monitoring levels. Correct voicing of a room or measured adjustment of room equalization might correct a problem being dealt with at the power amplifier's gain controls. Insufficient bass can be corrected in other ways besides moving up the overall system gain.

The above suggestions are not of real interference to the creative aspects of audio; creating, controlling, and communicating. The other side of the coin is that the future holds no other alternatives. If the industry does not recognize the health hazards, the regulatory implications of well used Federal and State laws will soon reach the audio profession. If self-regulation, or more properly moderation can become the norm, then the heavy hand of regulatory enforcement might be avoided. The industry would benefit. □□□



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. . . the weakest link in the audio reproduction chain

by William Isenberg

In every record project there is a crucial moment when everyone concerned gathers round to find out what has happened to the sound when the final mix was transferred to disk. No matter what was done at the mastering facility, inevitably a reference cut or test pressing must be compared to the master tape. Ideally this means the playing of that disk in the same room used for mixdown, so as to be absolutely sure of the sound.

Such comparisons are truly unfair, but there is no alternative. Even under ideal conditions, a record cannot sound as good as the original tape since no analog system can improve fidelity or reduce distortion. All we can hope for is the least possible number of devices in the signal path. If a certain piece of equipment is not being used, it should be bypassed with patchcords, rather than running program through it.

When it comes to that all-important playback, it is equally important that top quality gear be used. This means a high quality belt or direct driven turntable properly fitted with a grade "A" tonearm and the best cartridge available. The playback equipment must be above suspicion, or there is little point in listening. If any changes

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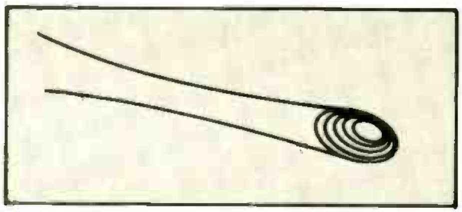

Figure 1: **Flux Loop for Phono Cartridges**

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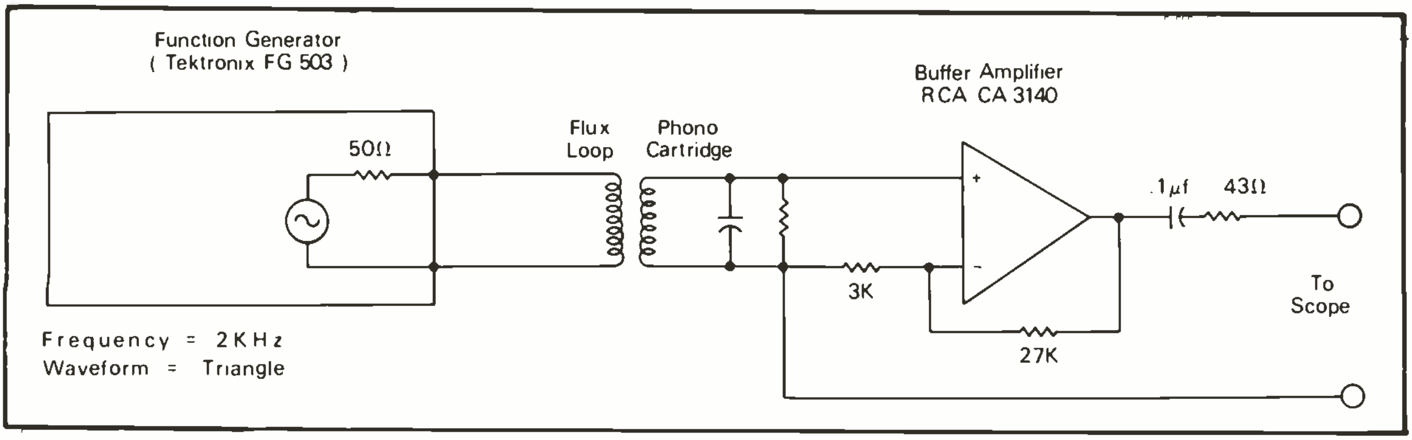

Figure 2: **Phono Cartridge Flux Loop Test Setup**

are made, particularly to cartridge or electronics, it would be a very good idea for all concerned to listen first to a representative sample of familiar discs, including some direct-to-disk material. Since we are talking about a professional situation, one might think that professional equipment would be the best choice to insure reliable service. However, this may not be the case.

The prime users of professional turntables are radio stations, syndication services and discos. In these applications rugged, reliable construction and ease of maintenance come first, but a recording studio user will also be very picky about the sound quality as well. This can be a problem because it is quite possible that the slip-cue felt-covered turntable which works so well over at the local radio station has so much rumble that it becomes unusable in a recording studio. The electronics can also be a problem, because the type made for radio usage often has to perform in a strong radio field. Our friends in radio know how hard it is to eliminate RF pickup. Thus we find that a preamp designed for radio probably has lots of RF chokes and bypass capacitors to kill RFI. Sad to say, these things tend to kill the high frequency response as well. Again, the fact that we are picky about sound quality is causing us to wonder about alternatives.

As far as sound quality is concerned, we can probably get what we want by using hi-fi gear. This will result in good sound, but not for long, as most hi-fi gear isn't rugged enough for studio use and has many features that are great for the hi-fi market but too complex for professional application.

Certain hi-fi products will be okay if not abused. The trade refers to this type as "esoteric." In this category, extra gadgets and features are left out and the prime thrust is better sound. Sad to say, there are many opinions of what good sound is in the hi-fi business. The only way to find out is borrow some promising units and take them to the studio and listen carefully.

If test equipment is available the units should be checked out *before* listening. It is possible to waste a good deal of time trying to figure out what sounds good or bad since the objective measurement figures may run contrary to the subjective evaluation. However, some folks insist on listening first because "if it sounds bad you'll hear it." If there weren't so many factors involved I could agree, but experience has convinced me that complex problems are solved more easily if divided into bite-sized chunks.

A Playback Cart

Since a disk playback is not done every day in most studios, the most practical package is probably a roll-around cart. If the cart can be locked up when not in use it will be more likely to have a cartridge with a good stylus when needed.

Once the gross mechanics have been dealt with, what about those all-important details that affect sound quality? The two most critical areas are the cartridge chosen and the preamp. In addition, we face a serious challenge in proper matching of these components.

Until recently proper termination of phono cartridges has been one of the more neglected areas in disk reproduction. A flat response is difficult to maintain because the mechanical and electrical systems tend to resonate at different frequencies in the upper end of the audio spectrum. For many years cartridge loading was done on a compromise basis which included a standard load resistance of 47K and the assumption that anywhere from 200 to 500 picofarads of capacitive loading would be presented to the cartridge. Typical cartridge inductance evolved to be anywhere from 500 to 900 millihenrys. A glance at a reactance chart shows resonance to fall in the octave from 10 kHz to 20 kHz. This electrical resonance is somewhat damped by the 47K load resistance, but the overall result is not outstanding as far as transient performance is concerned.

Having done some work with microphone transformers I became very aware that termination makes vast differences in the sound quality of microphone preamps. Square wave testing is a powerful tool for evaluating the transient performance of transformers which are step-up in the case of mike preamps. The usual ratio is 1 to 8 or so with the primary source impedance usually 150 ohms which the transformer reflects into the amplifier as 10K ohms. This is done to obtain the best noise performance using a typical bi-polar transistor amplifier. Unless the secondary of the transformer is properly terminated, the capacity present in circuit will resonate with the inductance of the secondary winding and cause a peak in the response at some very high frequency. On the surface of it there would be no harm in this, but contrary to some opinions, many musical instruments have substantial transients and overtones well above 20 kHz. When bells and triangles are struck a whole spectrum of ultrasonic information appears to excite any spurious resonances in a mike transformer. This is plainly audible as a shattering sound on impact instead of the distinctive click heard in person. Broadband sources such as ride cymbals generally sound harsh and nasty through a preamp with transient distortion.

Measurement Procedure

There is no substantive difference between the secondary of a microphone transformer and a phono cartridge. It is not hard to measure the response of a mike preamp as long as the generator has the proper source impedance, but how to go about measuring a phono cartridge? Vibrate the stylus? With what, and with what kind of precision? Even if it were possible, the stylus mechanical resonance would influence the results. In order to treat the electrical and mechanical areas separately there must be an electrical method of exciting the cartridge without the stylus being used.

There seemed to be no answer until Lyman Miller and I were part of a group listening to Peter Butt describe the calibration of magnetic tape duplicators

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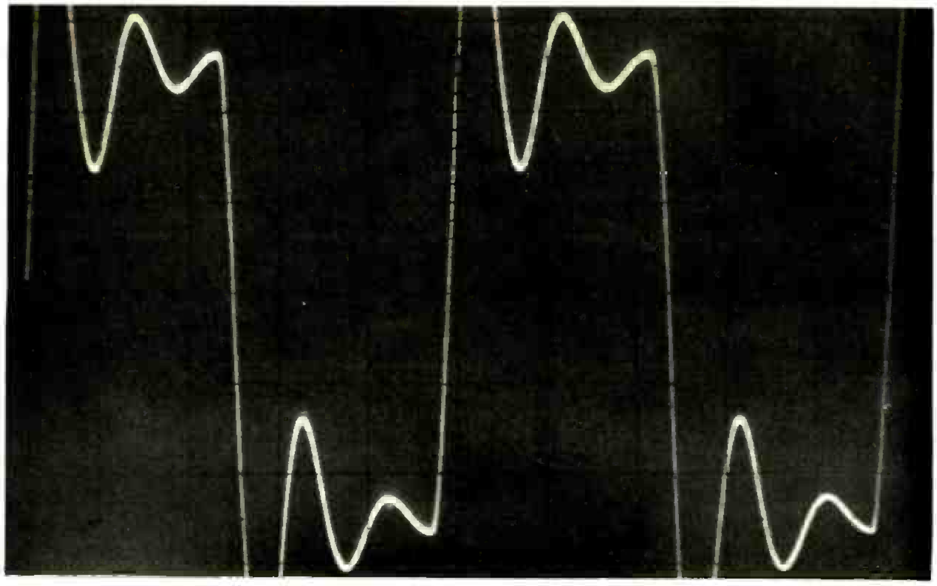

Figure 3: F = 2kHz, 100 μsec/div.
Capacitive Load Only, Overshoot 42%.

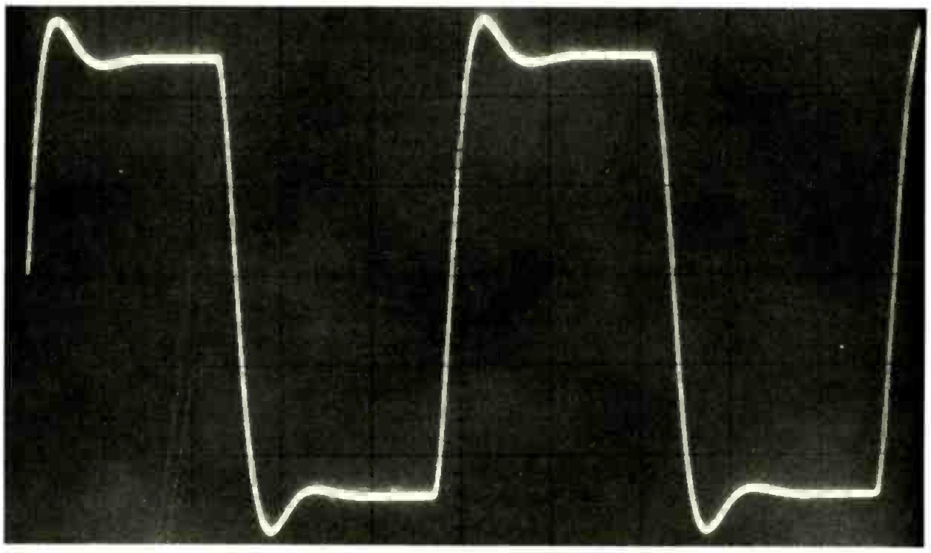

Figure 4: **Resistive and Capacitive Loading.**
Risetime 26μsec, Overshoot 9%.

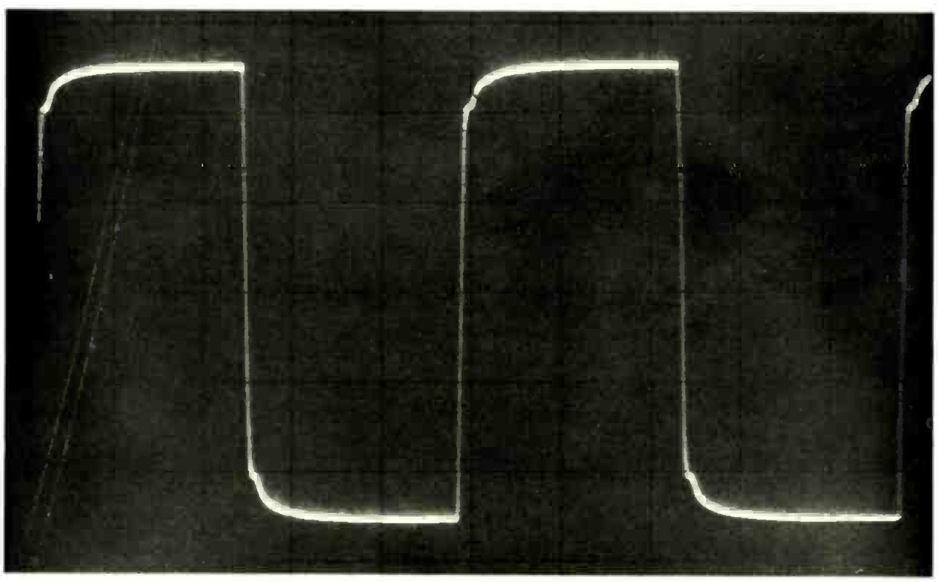

Figure 5: **F = 2 kHz, 100 μsec/div.**
No Load, Rise Time 8 μsec.

using a *flux loop*. This gadget makes it possible to check performance of a reproduce channel without worrying about tape-to-head interface problems such as poor tape wrap or azimuth. Eureka! All we had to do was remove the stylus assembly from a phono cartridge and insert a ten-turn flux loop (Figure 1). Lyman pursued the concept and read a paper on flux loop calibration of phonograph reproducing systems at the AES convention in Los Angeles in May, 1976. My own effort along these lines did not progress as rapidly, but the insight obtained proved to be quite an eye-opener.

The biggest single problem with the phono cartridge is that it has to be installed in a tone arm away from the preamp. Tone arm lead capacity is difficult to reduce below 100 pf. To simulate this a capacitor was connected in parallel with the cartridge while using a flux loop to excite the coils (diagram in Figure 2). This caused substantial overshoot and ringing; the square wave was badly distorted, as shown in Figure 3. By connecting a 47K resistor in parallel, the damping of the square wave got much better, but the rise time became unacceptable (Figure 4). Now what? By transformer standards this performance was *terrible*. Was this the only way to listen to records? Not exactly. Moving coil cartridges which operate at a lower impedance and thus are relatively immune to capacitive loading are available but they have problems of their own, such as non-replaceable styli, high cost, and more hum and noise. What really intrigued me was that with both resistive and capacitive loads removed from the cartridge driven by the flux loop, square wave performance improved incredibly (Figure 5). If the cartridge could somehow be operated with *no loading whatever*, a dramatic sonic improvement could be expected. The only thing to do was either put the entire phono preamp in the tonearm headshell, or use some kind of buffer to lower the impedance such that the cartridge would not be loaded by tonearm wiring. A method was devised using two transistors to buffer the cartridge as required. With electrical problems bypassed, it became possible to investigate the mechanical situation. Various cartridges were installed into the buffered headshell and the DIN 45541 test record (sold by Gotham Audio) was played. Incredible! Some cartridges performed very well, but others had very pronounced peaks in the high frequency end. One highly rated cartridge had a stylus assembly which resonated so strongly that it reached a 10 dB peak at 22 kHz! Flat as a pancake all the way from 10 Hz (tonearm resonance) until 10 kHz and then a curve that shot up like a rocket from Vandenburg! No wonder the spec sheets recommended a 500 pf load. Needless to say, my personal system did not use this cartridge.

New Cartridge Designs

Fortunately, new cartridges appear with regularity and the CD-4 quadraphonic system has served as a stimulus to improve the status quo. The latest cartridges available have lower inductance which reduces electrical problems and better designs for the stylus/cantilever assembly which addresses mechanical resonance difficulties. These benefits are complemented by a growing consumer awareness of the situation and the appearance of preamps with adjustable cartridge loading. Some preamps have switches on the front panel to make it easy.

Once the cartridge output is presented to the preamp, a very important process begins. The amplifying devices used and the connection of circuits in conjunction with them can affect listening greatly. It is very important that no spurious responses be generated by the preamp. The two types likely to cause the most trouble are harmonic and intermodulation distortion.

By far the most noticeable is third harmonic. Third harmonics do not normally occur in music (other than fuzz-tone guitars and synthesizers) and as a result just a little bit is audible. If a preamp has a lot of third harmonic distortion, voices will buzz, brass will sound bigger than life, and everything will tend to smear in a concave manner. In fact, this sonic effect is often called "transistor smear" or "transistor sound." It is not musical and also causes listening fatigue.

Second harmonic distortion can also cause trouble, but in a different way. Since music naturally contains second harmonics (octaves), it would seem that some of this distortion would be a good thing. Having too much of a good thing can be a problem, however, when it slowly dawns on you that vocals sound *very* closed miked and intimate, woodwinds swell and bloat, brass sounds muffled and strings get sleek and fat. The perspective becomes convex as things seem to bulge towards you. (Fat) Vacuum tube equipment is prone to this type of distortion. Some people cherish old (or new) tube equipment because of the less annoying characteristic sound and relative lack of fatigue.

Intermodulation distortion is difficult to measure using the SMPTE method because it always turns out to be very low. A more stringent test has been devised which uses two tones much higher in frequency. This process goes by the general name of twin-tone and involves two tones separated by 1 kHz. If the amplifier were perfectly linear, no 1 kHz beat note would be produced. Real amplifiers aren't so good. The sonic result is muddiness and loss of articulation.

Phono Pre-Amp Circuitry

At this point it might be a good thing to consider the amplifier internal circuitry, sometimes called topology. Remember that an amplifier does not amplify anything. What is called an amplifier and looks simple is in fact a complex servo system which takes power obtained from the power supply and uses that power to create a replica of the incoming signals.

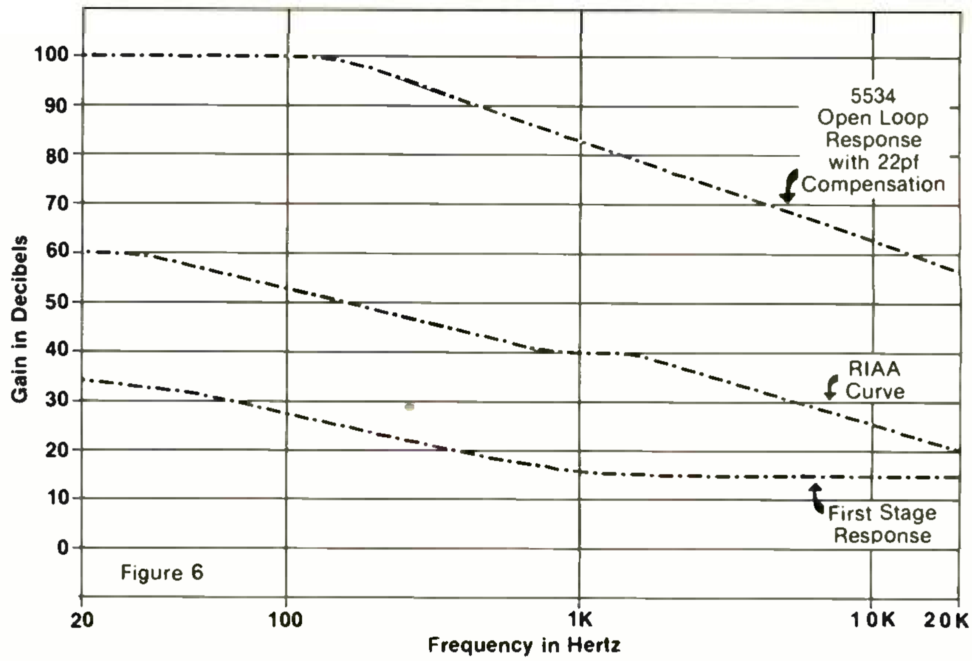

Figure 6

Most amplifiers used for audio signal processing are known as operational amplifiers or "op-amps." Originally developed for analog computers, this flexible device has found application in many audio amplifiers and signal processors. When monolithic op-amps were first designed, performance was limited. The first integrated circuit op-amp to be widely accepted is the Fairchild µA709 which came out in 1965. Although many other devices have appeared since and come with as many as four units per package, they all have certain things in common as far as the audio designer is concerned.

- Low cost.
- Ease of application (fewer parts)
- Small size
- Low power consumption

So much for the good side. But you don't get something for nothing. Here are some of their deficiencies:

- Distortion rarely specified
- Noise can be a problem
- Large product variation
- Occasional stability problems
- Inconsistant sonic characteristics.

This puts the designer in a tough spot. To keep costs down and make a salable product, it is necessary to use ICs, but a truly high performance design requires a discrete transistor approach. The most desirable advantage of discrete amplifiers is the total control which is available to the designer. This doesn't come cheap, however. It takes about 8 transistors to make a good discrete op-amp and lots of other parts as well.

RIAA

Just as important is the overall circuit configuration used to implement the RIAA playback equilization. Almost all phono preamps on the market use a feedback circuit which attempts to accomplish the entire task with a single network/amplifier. This may be good enough for a compact stereo or a small receiver, but component quality stereo and professional usage requires less compromise.

The RIAA curve requires a 40 dB change over the frequency range of 20 to 20 kHz. Since a 40 dB change represents a voltage ratio of 100 to 1, the network used imposes severe demands on the amplifier used. This is shown graphically in Figure 6, where a standard RIAA curve (40 dB @ 1 kHz) is the middle trace. The open loop (no feedback) response of a signetics NE5534 op-amp with 22 pf compensation is shown at the top for comparison. Note that there is 43 dB of loop gain (difference between open loop and closed loop) at 1 kHz which degrades to 37 dB at higher frequencies. Feedback theory indicates that error caused by having only 40 dB of loop gain will be approximately -84 dB or about .006%. This is not as good as it is possible to do with the 5534 or a good discrete amplifier. If the loop gain is 60 dB instead of 40 dB, theoretical error becomes -126 dB or .00005%. This is much lower than the -70/80 dB signal-to-noise ratio obtained by most good phono preamps. It is obviously preferable to have error (distortion) be 40 dB below the noise level than approximately equal to it.

Given the fact that this amplifier chip has only 83 dB of gain at 1 kHz, it appears that the maximum closed loop gain shouldn't be

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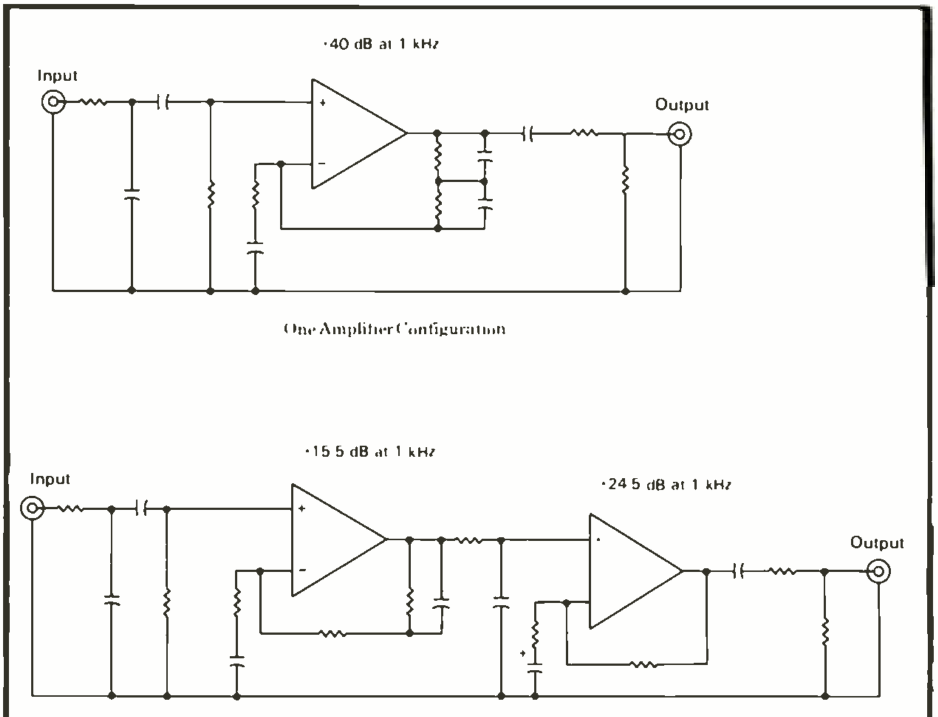

One Amplifier Configuration

Two Amplifier Configuration

Figure 7: Comparison of Amplifier Configurations (Note that overall gain is the same for both.)

any greater than 23 dB @ 1 kHz in order to maintain 60 dB of loop gain. Since 40 dB overall is required, the best way to get it is to cascade two stages. The lower curve on the graph, Figure 6, shows the 50 - 500 Hz portion of the RIAA curve implemented with 67.5 dB of loop gain at 1 kHz, 66 dB at 20 Hz, degrading to 42 dB at 20 kHz. This configuration has 15.5 dB of gain at 1 kHz, requiring an additional 24.50 dB to reach a total of 40 dB. A second amplifier with no EQ network is quite acceptable as there will be 58.5 dB of loop gain at 1 kHz and 32.5 dB @ 20 kHz. In order to accomplish the HF equalization (-3 dB @ 2,122 Hz) a passive RC network is used between the amplifiers. This network is independent of the 50 - 500 network used in the first stage and because it is passive there is no departure from the ideal 20 dB/decade (≈6 dB/octave) slope as unity gain is approached. The output of the second amplifier is available to drive cables without changing equalization due to capacitive loading. Schematic diagrams of the one- and two-amplifier configurations are shown in Figure 7.

To evaluate sonic differences the author has built many "little grey box" preamps in the past five years (Figure 8). The first few used IC op-amps of various types, and, when ICs available at the time proved to be less than perfect, a discrete op-amp was designed using 8 transistors. This made the phono preamp a bulky box containing 32 transistors and many other parts (Figure 9), however the effort was justified as sound quality developed an "effortless" character which I have not heard from any IC op-amp. As better sounding IC's became available, the IC version was re-designed to take advantage of the newer devices (Figure 10). One advantage is that newer ICs can be plugged into old sockets, which produces *amazing* changes in sound quality. All of these grey boxes use the two stage configuration described earlier.

Calibration

To get the best performance, the turntable-tonearm-cartridge-preamp system should be calibrated. This means that all members of the group play in tune, so the sound doesn't suffer.

Calibration should not prove to be a serious problem if it is undertaken carefully. *Warning: Not all test records are created equal. Some are more equal than others!* (My personal favorite is DIN standard 45541 available through Gotham Audio). With careful adjustment of cartridge termination and perhaps the tone controls (if any) it should be possible to get response within ±0.5 dB of flat at least to 15 kHz. Do not use a VU meter on the console or a tape machine to measure frequency response. Some of them are unacceptable above 10 kHz. If you have a Hewlett-Packard or Sound Technology distortion analyzer the meter section is ideal. You will probably notice that the meter responds to rumble and record warp well enough to make measurements difficult. The preamp rumble filter may help as you adjust the top end termination. At the

FIGURE 8: **GREY BOX PREAMPS**

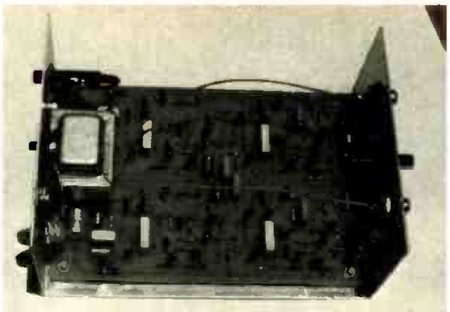

FIGURE 9: **DISCRETE PREAMP INTERIOR**

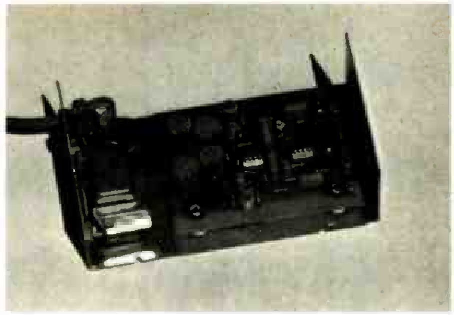

FIGURE 10: **I.C. PREAMP INTERIOR**

bottom end it is likely there will be a slight rise in response caused by tonearm/stylus resonance. Very few tonearms have any damping mechanism to adjust, but some cartridges have an integral brush which is an effective substitute. Now that frequency response is as flat as it will get, don't forget to set the two channels equal in level. This should be done while playing the mono section of the test record.

If an oscilloscope is available you are ready to undertake an important step toward better reproduction. Compare the two channels for phase difference as you would to check azimuth on a two-track tape machine. Adjust the phase by loosening the cartridge screws slightly and swiveling the cartridge as viewed from the top. Then tighten the screws again. This assumes that overhang, tracking force and all the routine things have been done as well.

If you are really brave and have a distortion analyzer you can adjust anti-skating to optimum instead of taking their word for it. Find a tone on the record long enough to permit the analyzer to lock on. Don't expect any marvels of low distortion, either. Anything under 3 percent is an incredible miracle. Before the tone quits, adjust anti-skating and stylus pressure for the least odious result.

Now you have a calibrated disk playback system. It may not be quite as good as the stuff at your friendly disk mastering facility, but at least when you play a reference on your monitor system, you'll be giving it a fair chance.

Hears to fidelity. □□□

References:

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new product review

. . . a new approach to audio data transmission: a digital snake —

THE JHD MAINLINE

by Peter Butt

The Mainline Line Return is a new product recently introduced by JHD Audio, of Costa Mesa, California. In simplest terms, the device could be called a "digital snake" inasmuch as it permits the transmission of eight channels of 22 kHz bandpass signals over a single pair of wires for a distance of 25 to 600 feet (7.6 m to 183 m).

The Mainline system consists of a transmission unit, (a transmission cable,) and a receiving unit. The system is supplied in three versions for balanced low-impedance microphone applications, high-impedance instrument pick-up applications, and a high-impedance, low gain model for line level transmission. The receiving terminal is the same basic unit for the three different models. The transmitting modules are adapted for each input signal requirement. The transmission cable itself can be any length of shielded pair microphone cable with the familiar three-pin XLR-type connectors. The only really stringent requirement of the transmission cable is that it have low leakage and well-soldered connections so that non-linearities are minimized.

Earlier models of the Mainline system have been supplanted with an up-graded version that includes a line compensation network at the transmitting module. Adjustment of the compensation controls permits the optimization of transmission and isolation characteristics for the specific length and type of interconnecting cable used. The recommended method of achieving this optimization is to simply feed program material through one of the system channels and to listen for crosstalk on the next lower-number channel while the compensation network is adjusted for maximum attenuation. All tests described were preceded with the optimization procedure.

— continued overleaf

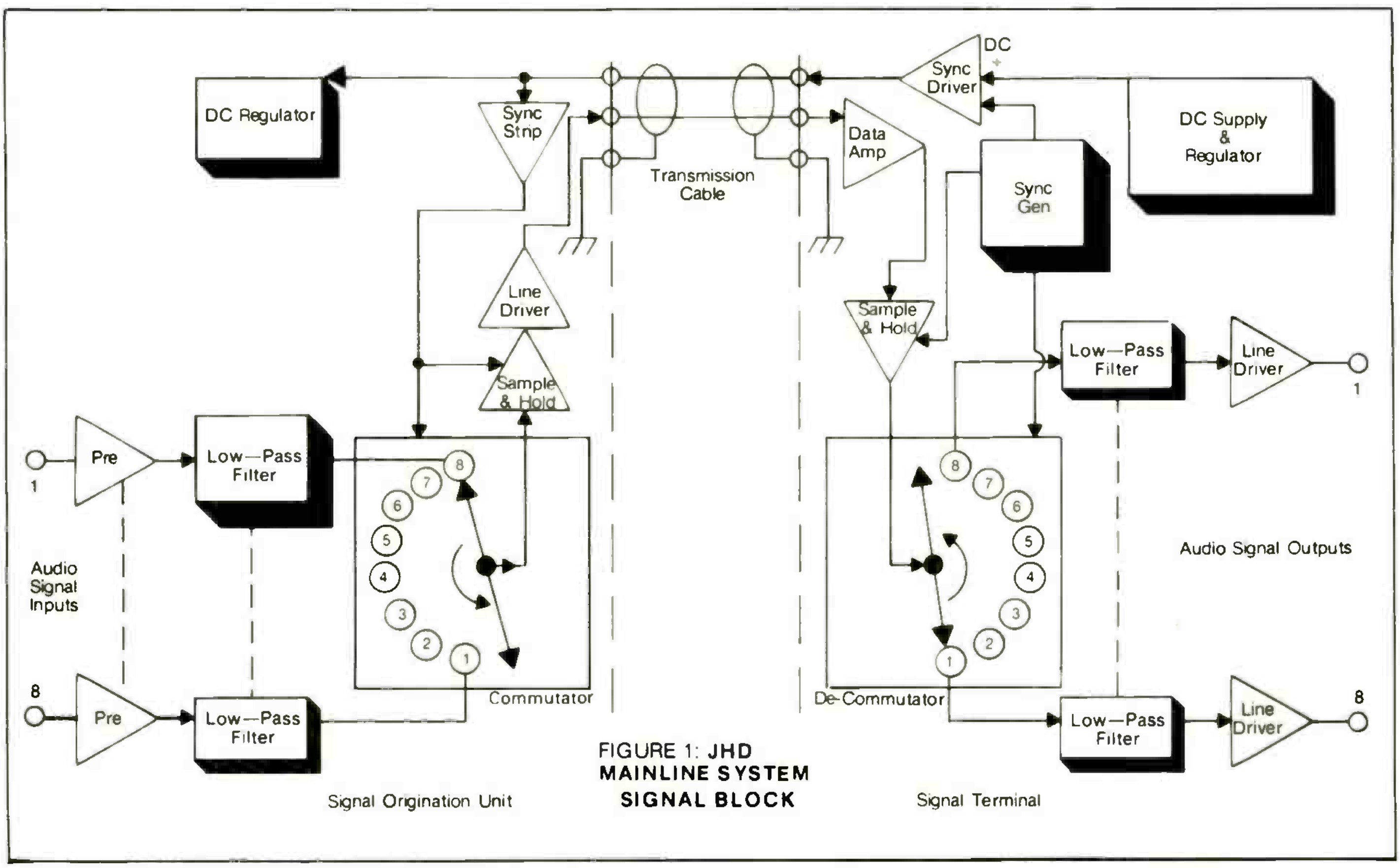

FIGURE 1: **JHD MAINLINE SYSTEM SIGNAL BLOCK**

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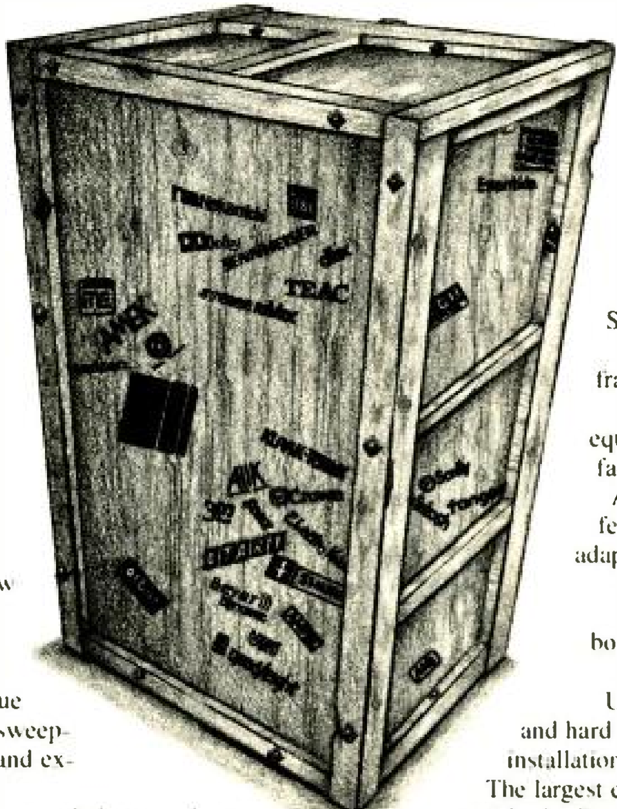
We offer a wide range of multitrack consoles and machines. Among the best and most cost effective is the new Amek M2000A, an automated 36/24 multitrack recording console which offers 10 VCA subgroups, 4 band true parametric equalization and sweeping filters on each channel, and extremely versatile routing.

With the MTR-90, Otari has made its entry into the world of no-compromise recording. A symmetrical tape path and a pinchroller-free direct drive capstan assure optimum head-to-tape contact, with minimal tape wear. The MTR-90's advanced electronics, with active bias circuitry, offers sonic performance second to none in the 2" multitrack format.



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The Concept

The Mainline concept is a rather clever one in which digital data acquisition technology is applied to a common problem encountered in stage performance sound reinforcement: the transmission of a multi-channel signal package from one point to a distant one.

The historical solution to this problem has been to simply run as many pairs of shielded wire between the two or more points as may be needed. That is simple enough in concept but suffers a bit in actual application. The Mainline system approaches the problem from a different direction. The eight audio channels to be transmitted are each fed into the transmitting terminal box where each is independently amplified and low-pass filtered. Then each of the eight signals is sampled in sequence by a solid state commutating switching device that feeds samples of the input signals to the transmission circuitry.

The data package is then sent down the interconnecting cable to the receiving unit which also provides power to both units of the system. Here the data sample stream is picked off of the cable and converted to analog form at a rate and sequence corresponding to the conversion at the transmitting end. These reconstructed slices of analog signals are then passed to a decommutator similar to the commutator at the transmitting end. Commutation and decommutation occur synchronously, permitting separation of the eight signals at the receiving end. Each decommutated signal stream is then low-pass filtered, smoothing the spaces between the decommutated signal segments. The filtered channel signals are then fed to the outside world by eight buffering amplifiers.

The block diagram of Figure 1 shows the general progress of events within the Mainline signal system. The rotary switch devices are, of course, not anything like that in reality. The commutation concept seems easier to portray as pictured rather than by the strict specifics of what is really going on.

All well and good, one may say. The real question is: "How well does all of this exotica work as compared to the wire and cable approach?" Given the following data, one may well conclude that it is not much worse and probably a good bit better than the traditional method.

Table 1 summarizes the technical specifications of

TABLE 1
SUMMARY OF MAINLINE SPECIFICATIONS

	HI-Z SPECIFICATIONS	LO-Z SPECIFICATIONS		LINE RETURN SPECIFICATIONS	
PARAMETER	PUBLISHED	PUBLISHED	MEASURED	PUBLISHED	MEASURED
MAXIMUM OUTPUT LEVEL	·10 dBv	·10 dBv	·12 2 ·13 5 2 kilohms	·10 dBv	·12 2 ·12 8 2 kilohms
SYSTEM GAIN	·13 dB	·20 dB	19 20 5 dB 1 kHz	·3 dB	·2 5 ·3 3 dB 1 kHz
FREQUENCY RESPONSE 20 Hz - 20 kHz	·0 2 dB	·0 2 dB	·0 1 7 dB	·0 2 dB	·0 1 8 dB
INPUT IMPEDANCE	220 kilohm	150 ohms nominal 10 K actual	1 kilohm 1 kHz	220 kilohm	greater than 160 K
OUTPUT IMPEDANCE	500 ohms nominal 10 ohms actual	500 ohms nominal 10 ohms actual	no measurement	500 ohms nominal 10 ohms actual	no measurement
THD · HUM · NOISE	12% maximum 06% typical	12% maximum 06% typical	065% maximum 1 kHz	12% maximum 06% typical	11% maximum 1 kHz
CROSSTALK WORST CASE	60 dB	60 dB	44 dB @ 20 Hz	60 dB	42 dB @ 20 Hz
CROSSTALK TYPICAL	75 dB	75 dB	54 5 avg @ 1 kHz	75 dB	58 5 dB @ 1 kHz
NOISE · HUM	103 dB	92 dB	93 dB re clip 30 kHz bandwidth	102 dB	104 dB re clip 30 kHz bandwidth

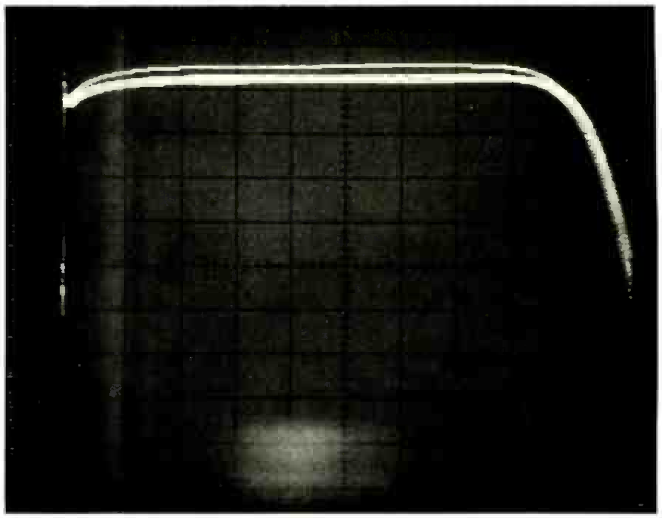


FIGURE 2: ***Mainline return frequency response, 8-channels overlaid. Top graticule line is 0 dBv reference, 2 dB/div. vertical, log frequency horizontal, 20 Hz to 43 kHz.***

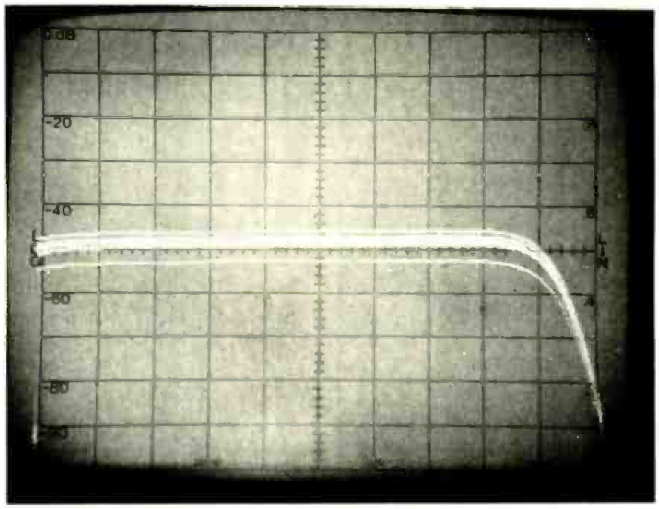


FIGURE 3: ***Mainline Lo-Z microphone input system frequency response traces, 8-channels overlaid. Top graticule line is +10 dBv, other scale factors as in Figure 2.***

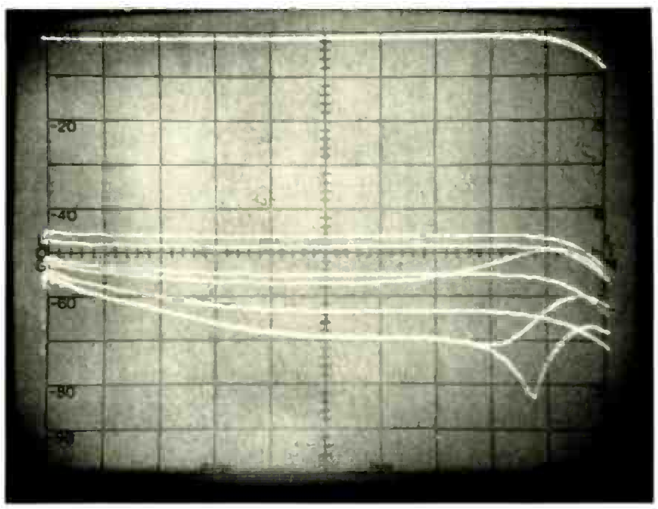
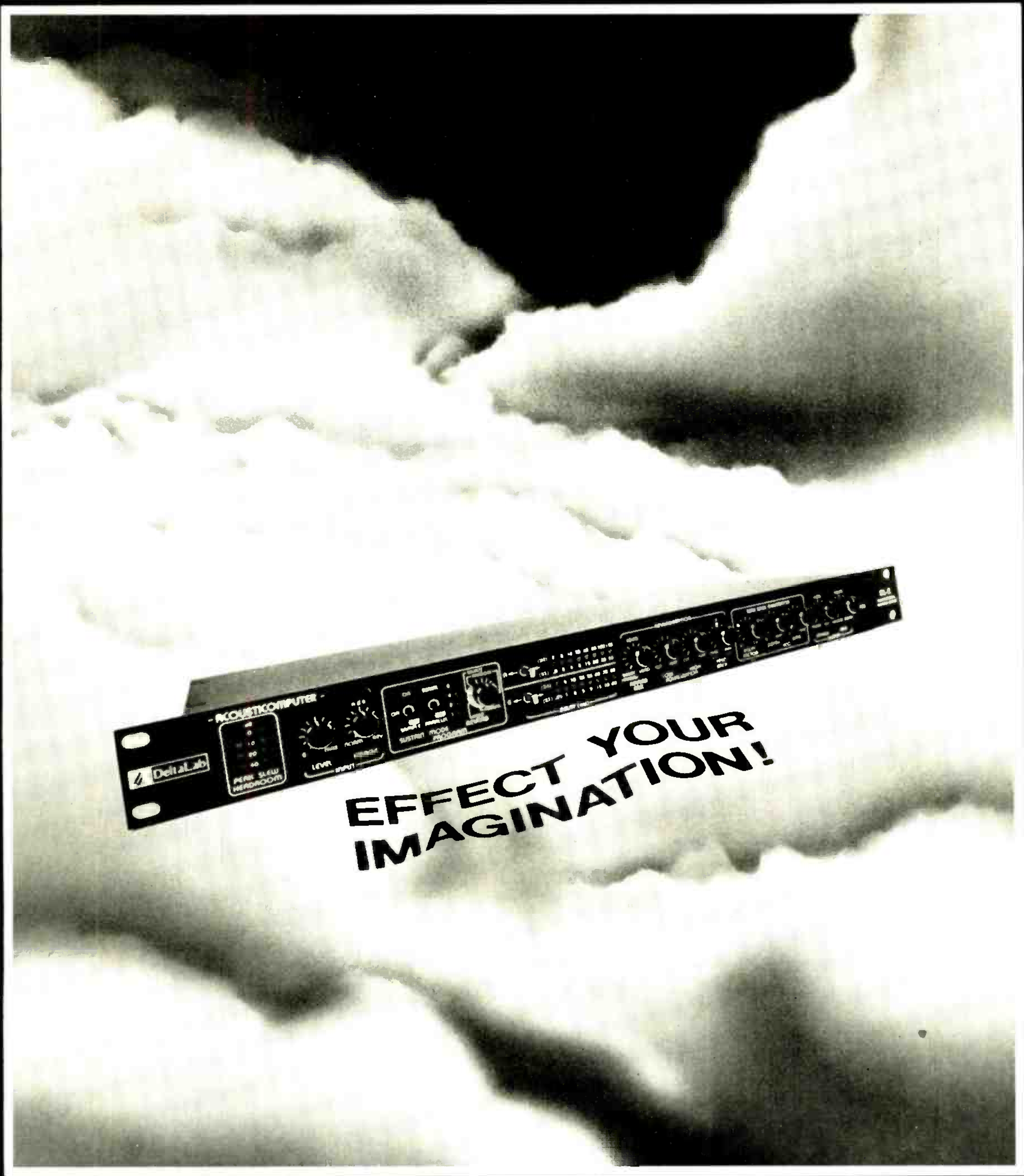


FIGURE 4: ***Crosstalk attenuation of Lo-Z transmission unit, channel 1 driven. Top graticule ref is 0 dBv/div. vertical, log f horizontal.***



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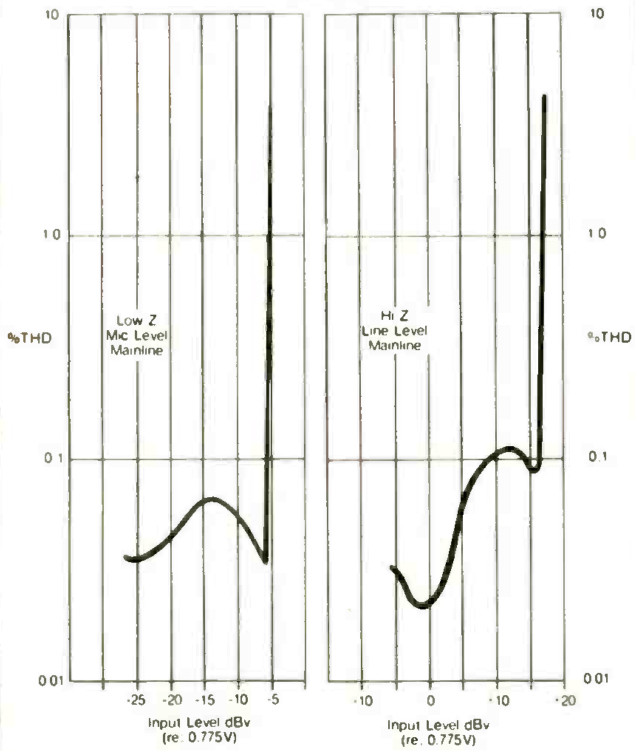
the three types of device. Only the balanced Lo-Z and Line Return units were tested as the only difference between the Line Return and the Hi-Z unit is the system gain specification. The only observed performance parameter that differed from the published specifications were the crosstalk data.

Figure 2 shows the frequency response of the Line Return unit. The response curves for the balanced Lo-Z microphone unit are shown in Figure 3. The traces of all eight channels are shown overlaid on a logarithmic frequency scale extending from 20 Hz to 43 kHz and having a 2 dB/div. vertical scale. Figure 4 shows the crosstalk traces of the Lo-Z unit. Again, a log frequency sweep is shown, however, with a 10 dB/div. vertical scale factor. The top trace is the output of the driven signal channel while the lower traces are the crosstalk levels of the other seven channels. The data for the Line Return system was not very much different.

Distortion (THD) versus input level for a 1 kHz signal is shown plotted in Figure 5. The onset of clipping is fairly abrupt and is shown about 3 dB into overload in Figure 6. Again, the data is similar for both units.

Squarewave response at 1 kHz is shown in Figure 7. Close examination of the photo will reveal a slight delay between the upper, input, trace and lower output trace. System propagation time, through a 100 foot (30.5 m) cable was measured at 16.462 microseconds at half pulse height. Squarewave rise time was 15.102 microseconds, fall time was 15.960 microseconds, with a 4% overshoot. The preservation of the squarewave shape and lack of discontinuities on the slopes indicate that the phase

FIGURE 5
INPUT LEVEL vs OUTPUT SIGNAL THD, %



response, and therefore group delay, are quite acceptable for most audio applications. A comparison of the Mainline system against a 100 foot run of common microphone cable was not

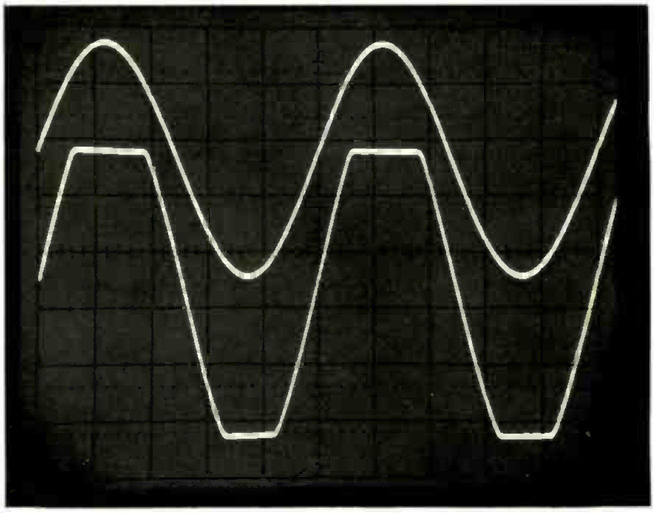


Figure 6: ***Clipping characteristic at 2 dB over threshold. 1 kHz signal, 2V/div. vertical sensitivity.***

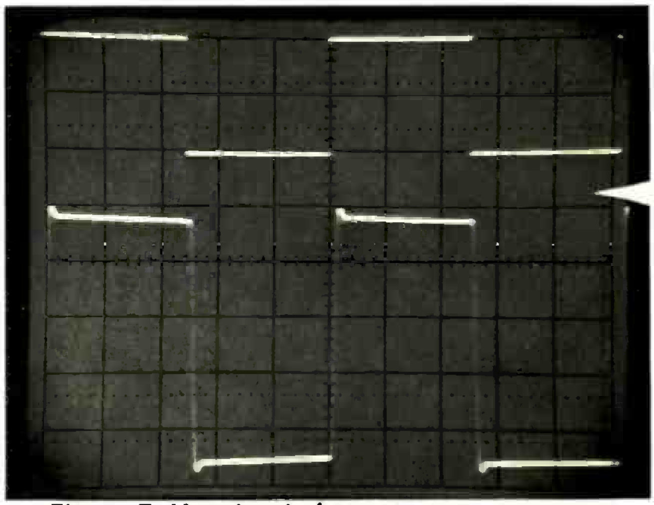


Figure 7: ***Unsaturated square-wave response. 2 V/div. vertical, .2 msec/div. horizontal.***

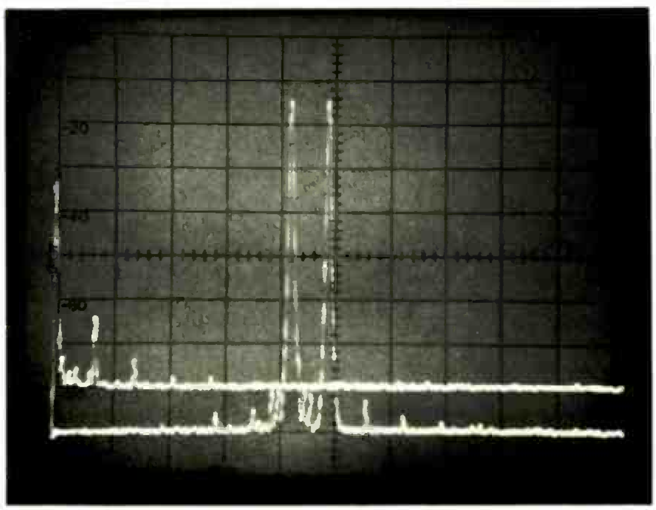


Figure 8: ***Twin-tone IM spectra. +20 dBv ref. level, .5 kHz/div. horizontal, 10 dB/div. vertical. Center frequency 2.5 kHz top trace, 10 kHz bottom trace.***

attempted. It is likely that the squarewave would not fare quite so well on its own.

Twin-tone intermodulation distortion was determined using a 9.4 and a 10.0 kHz pair of signals in 1:1 ratio. Figure 8 shows the resultant output with a 0.5 kHz/div. sweep and a 10 dB/div. vertical scale. The lower trace is centered at 10.0 kHz while the upper one is centered at 2.5 kHz. A 100 Hz resolution bandwidth was used for both traces. The upper trace was displaced upward 10 dB for clarity, and shows second order IM products while the lower trace shows third order IM. The top graticule line is +20 dBv. IM distortion measures 0.01% second and 0.05% third order for a signal input approximately 2 dB below clipping. This should be a fair approximation of a worst-case situation.

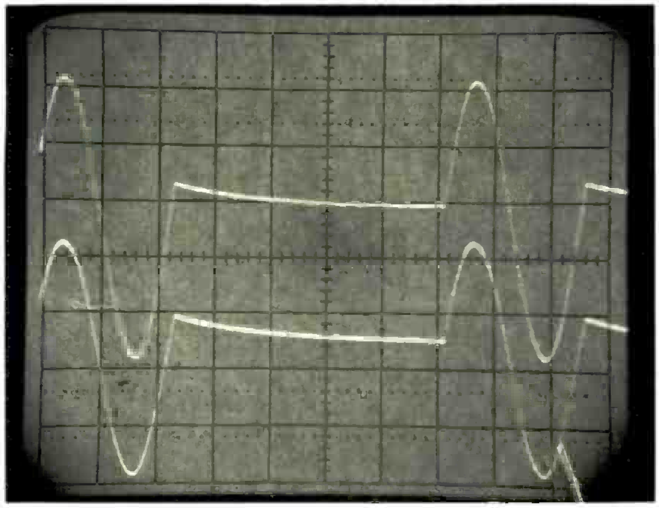


Figure 9: ***20 Hz tone burst response.***

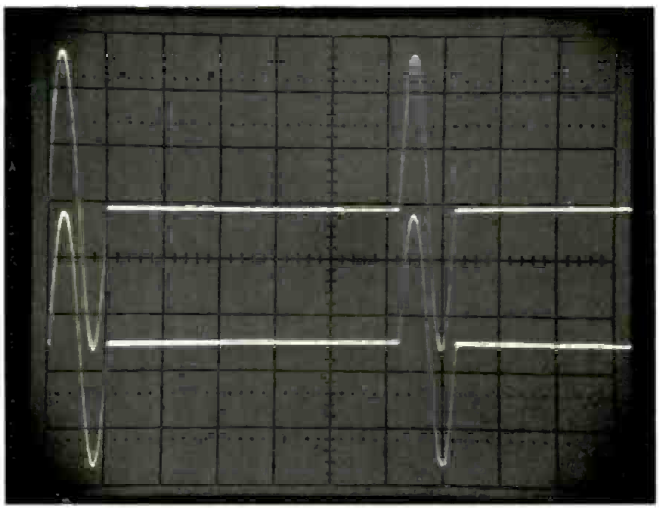


Figure 10: ***1 kHz tone burst response.***

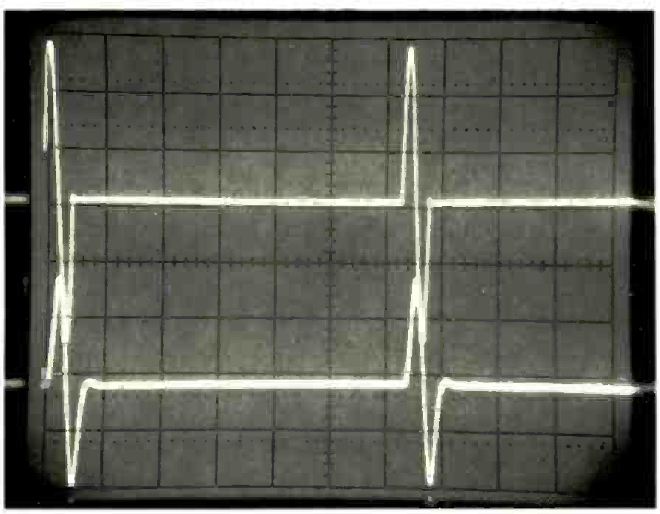


Figure 11: ***20 kHz tone burst response.***

Figures 9, 10 and 11 show the system response to single-cycle tone bursts at frequencies of 20 Hz, 1 kHz and 20 kHz, respectively. Vertical sensitivity is 2 V/div. in each case. Response of the Lo-Z microphone unit is shown, although the Line Return system behaved similarly.

The polarity response of all three systems is positive output for a positive input, case taken as common. The microphone Lo-Z device responds positively to a positive-going transition at pin 3 of the input XL connector, pin 2 common, pin 1 shield. There are no blocking capacitors in the microphone input circuit, so phantom powered microphones will have to be used with an appropriate powering device.

In taking the crosstalk attenuation measurements, it was found that ground loop coupling between generator and detector degrade performance noticeably. System grounding is definitely a consideration in application of the device as output ports are AC-coupled unbalanced. Only the Lo-Z microphone transmitting module has balanced inputs.

The most obvious application for the Mainline signal transmission systems would seem to be the live performance situation. Multiple instrument pick-up and microphone feeds to a central mixing console would seem to be greatly simplified. The quality of the signal as received at the terminal point would seem to incur much less degradation than for the case of multiple shielded pairs.

The use of the Line Return system to feed actively-crossed power amplifier systems would permit isolation of the crossovers at the sound console and power amplifiers at the speaker array location, thus eliminating line losses. For the cases where all signals within an eight-channel group are related as part of a common program system, the crosstalk figures observed should not impair separation of signals in the final mix to any noticeable extent. Multi-track tape machines show similar or worse separations between adjacent tracks, and consoles often are not very much better. The cost of a single Mainline system varies between $450 and $550. This is fairly comparable to the cost of an eight-channel snake of about 200 ft. or more. The microphone Lo-Z unit shows a high noise figure that will probably preclude its use in classical music or other relatively low level pick-up applications. The majority of studio and live situations will probably find use of the device practical. □□□

New Products

AMPEX UNVEILS ATR-124 24-TRACK RECORDER

Called the analog recorder of the '80s, the Ampex ATR-124 24-track audio recorder will make its debut at the Audio Engineering Society convention, November 2 - 5 in New York City. Demonstrations will be held during the show.

"The ATR-102 and ATR-104 are the best analog mastering-mixdown machines on the market today," said Lee Cochran, general manager, audio products group, of the Ampex audio-video systems divisio.

"We have built on those ATR advances and on technology completely new to the recording industry to produce a multitrack recorder that will satisfy the most demanding requirements of the audio professional for years to come," he added. "It is truly the system and format for the 1980s."

The ATR Series of multitrack recorders, also available in a 16-track configuration, offer exclusive features that enhance the creative capabilitites of the recording professional.

The ATR multitrack series features the smooth tape handling characteristics developed by Ampex for computer tape transports and first offered to the audio industry in the ATR-100 recorder/reproducer.

The record employs a closed loop DC servoed transport that maintains constant tape tension at each reel in all operating modes without pinch rollers.

Both reel motors and the single drive capstan are servo controlled. Unlike ordinary records that use pinch rollers to pull the tape and reels to take up the slack, the capstan-controlled reels on the ATR control the motion of the tape at all times.

It senses the motion of the capstan, the direction it is moving, and then automatically adjusts the tension accordingly.

Flux gate™ record heads that combine the recording and sync playback windings on one head are utilized in the recorder, giving the user Sel Sync™ response that approximates normal reproduce response. The system also offers a unique transformerless I/O capability that eliminates annoying distortions while offering excellent frequency response.

A variable speed shuttle control lets the engineer control the forward and reverse motion of the tape by merely sliding his finger along the switch. Shuttle speeds can be regulated from slow to 300 ips.

The recorder has a 16-inch reel capability, making it ideal for double system recording in a quad videotape recorder environment.

Membrane switches are utilized on the setup panel for greater reliability. Setup is further enhanced by the recorder's I/O bus capability, which provides evaluation of each channel without continual moving of the I/O cables.

The system features programmable monitoring with memory and a battery-powered backup memory that retains setup instructions in the event of a power failure.

Dual microprocessor controls are utilized in all ATR multitrack recorders for greater reliability and less down time. The system also features record mode diagnostics that alert the engineer through flashing VU meter lights if there is a record malfunction.

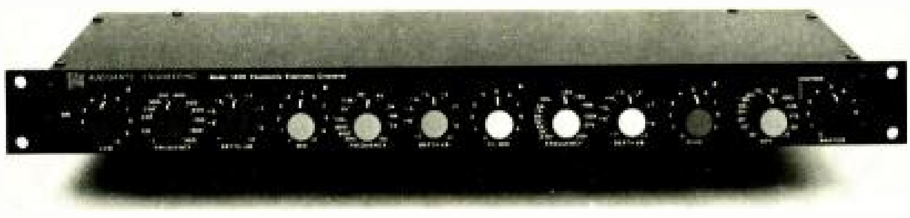
Other standard features of the recorder include Pick Up Recording Capability (PURC), which permits the editing or dubbing of new material without creating errors at either end of the new insert.

The ATR multitrack series also provides four assignable record, playback and Sel Sync equalizers per channel.

Another convenience feature stops the transport or slows the reels at either end-of-tape or near-end-of-tape.

Single point search-to-cue with tape looping is also standard. With the tape looping feature, the recorder will automatically go into rewind when it reaches a preset stop point, return to the start point, and then go back into play continuously.

The recorder's vary-speed control will display the tape speed as a percentage of the



nominal tape speed and in $\frac{1}{4}$-tones, and the phase-loc capstan provides precise control of the rotation of the capstan to insure absolutely accurate tape speed.

The ATR also provides the noise reduction interface needed to use many of the professional noise reduction systems in use today.

Dynamic braking in all modes means fewer mechanical problems and breakdowns. A universal power supply is offered in all versions of the ATR, along with NAB or CCIR equalization standards. The recorders are also designed for top and front end service accessibility for easy maintenance.

Several option features are also available on the recorders.

A new multipoint search-to-cue, designed to replace the standard single point search-to-cue, provides a capacity of 99 memories. A complete remote control panel that is identical to all the functions on the main panel is also available in the same panel.

Auxiliary output monitoring amplifiers give the engineer the ability to have separate output for sync playback. A two-to-one conversion kit for one-inch, eight-track heads is also available.

AMPEX CORPORATION
401 BROADWAY
REDWOOD CITY, CA 94063
(415) 367-4151

for additional information circle no. 1

AUDIOARTS 4-WAY PARAMETRIC CROSSOVER

The Audioarts Engineering Model 1400 is a monophonic four-way parametric electronic crossover intended for use with high power four-way and three-way speaker systems. It is equipped with front panel crossover frequency controls, enabling it to be easily matched to virtually any combination of drivers. Furthermore, crossover depth controls (-7 to +1 dB) are also provided to compensate for speaker frequency abnormalities in the crossover region. The Model 1400 also has four front panel level controls, four phase reversal switches, a system overload indicator and a master level control. A variable high pass filter is incorporated.

Specifications: Frequency response 20 Hz to 100 kHz ($\pm\frac{1}{2}$ dB); THD .004%; Dynamic range 110 dB; Slope 12 dB/octave; Maximum input +26 dB; Maximum output +22 dB into 600 ohms. The Model 1400 mounts in one standard rack space ($1\frac{3}{4}$" high).

AUDIOARTS ENGINEERING
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(203) 393-0887

for additional information circle no. 2

STUDER REVOX ANNOUNCES NEW MULTI-SPEED CONFIGURATIONS IN B77 OPEN-REEL TAPE RECORDER

The B77 open-reel tape recorder is now available in four different speed configuration: 15/16 and $1\frac{7}{8}$, or $1\frac{7}{8}$ and $3\frac{3}{4}$, or $3\frac{3}{4}$ and $7\frac{1}{2}$, or $7\frac{1}{2}$ and 15 ips. These speed options make the B77 ideal for logging purposes, extended recording and playback, background music systems or professional, studio or portable use.

The B77 is available in both half-track and

quarter-track formats, and may be ordered with an A/V head option which permits recording sync pulses between stereo tracks, for slide-show presentations.

A versatile input-switching system allows selection or low- or high-impedance microphones, preamplified "Aux" or "Radio" sources.

The Revox B77 operates in either stereo or mono modes, and may be configured to record monaurally from either or both inputs, onto one or two tracks. In the mono mode, any combination of input sources (microphone, auxiliary, radio) may be mixed together. This feature facilitates "voice-overs" and other techniques important to audio-visual and broadcast production.

The B77 stereo tape recorder is also equipped for the sound-on-sound transfer of a previously-recorded track. Sound-on-sound is accomplished by a simple switch

for additional information circle no. 4

SYNCON

Logic and Music in Harmony

It is a fact that many medium priced consoles use ungraded VCAs and ICs resulting in signal degradation and unpredictable performance. Syncon uses top quality discrete circuitry on interchangeable cards which allow not only instant replacement but future upgrading.

Sophisticated PCB design has virtually eliminated hardwiring making Syncon not only cost effective but incredibly reliable and serviceable, an important factor for studios without resident 'boffins.'

Add to this a superb status, routing and grouping system enabling 28 tracks or effects to be mixed through 14 stereo subgroups and you have a very logical alternative to the headaches of cut price automation.

SYNCON FEATURES

- 28 Input output capacity.
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- 6 Auxiliaries.
- 2 Stereo and quad echo.
- Master and channel status switching.
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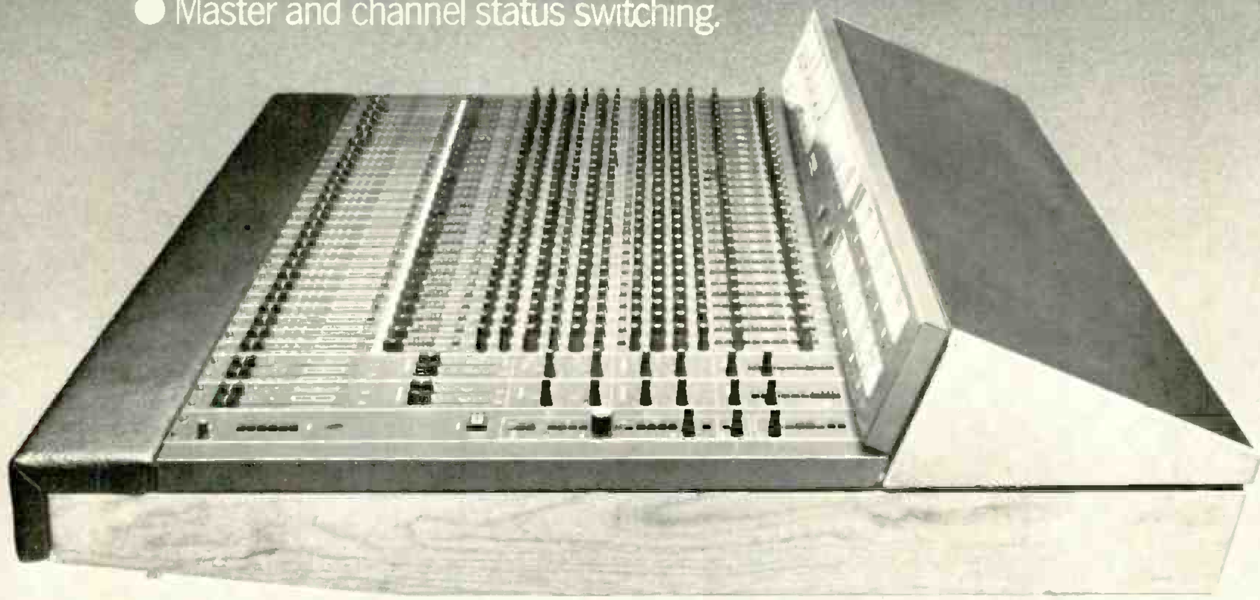
- Parametric eq.
- 3 Module inserts.
- Producer's desk and patchbay.
- Price range $20,000-$30,000.



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selection. Simultaneous with the transfer, a microphone or line-level source may be combined and balanced with the track being transferred. The ease with which "bouncing tracks" is accomplished emphasizes the B77's practicality for even the most complex small-production requirements.

Front panel ¼" stereo phone jacks are provided for two headphone pairs, driven by built-in amplifiers. A most-flexible monitor switching arrangement selects either input or tape source, with normal or reverse-stereo monitoring, as well as left-only or right-only applied to both outputs. When auditioning in mono, another monitor position allows stereo/mono compatibility checks, without affecting the recording.

A dual-concentric front-panel volume control adjusts headphone levels without affecting the recorder's main output level. Separate screwdriver adjustments are used to calibrate the B77's output level, over a 26 dB range. Illuminated, ASA-standard VU meters, combined with LED peak indicators, provide comprehensive monitoring of signal levels.

A three-motor drive system, logic-controlled and electronically regulated, assures stable and precise tape motion, at both 3¾ ips and 7½ ips. Wow and flutter are less than 0.1% at 3¾ ips and less than 0.08% at 7½ ips.

Other performance features at 7½ ips are: frequency response from 30 - 20 kHz, +2, -3 dB (50 - 15 kHz, ±1.5 dB); distortion of less than 0.6% at 0 VU; signal-to-noise ratio of better than -67 dB (half-track).

Revox B77 has a built-in tape cutter for editing, and accepts either cine or NAB-hub reels.

The extensive complement of functions and superior performance features make

the B77 stereo tape recorder well-suited to all professional production requirements, as well as those of the critical audiophile. Suggested retail price for 3¾ and 7½ ips is $1,499.00.

STUDER REVOX AMERICA, INC.
1819 BROADWAY
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for additional information circle no. 89

PORTASTUDIO FROM TEAC

TEAC's new lightweight M-144 Portastudio — a complete portable studio that combines a four-in, two-out mixer with a multi-track cassette recorder and weighs less than 20 pounds.

"The Portastudio really is a musical instrument," according to Bill Mohrhoff, national sales manager for TEAC/Tascam Series, under whose banner the M-144 will be marketed in the United States.

Mohrhoff pointed to the Portastudio's unique features — lightweight, compact, extremely mobile. "You can pack it under your arm, plug it in anywhere — and record merely by plugging in a microphone and a headphone. The Portastudio is not forcing the musician to pay for specifications and other features he doesn't want or need. But it does fulfill the needs of the songwriter/composer."

"What the Portastudio is not," he emphasized, "is an audio/high fidelity product. The Portastudio has a new head configuration that is not compatible with other audio products on the market. It will not replace other audio products in the

for additional information circle no. 90

"GIVASHIP"

(giv'ə•ship) *vt.*, to present for the gratification or acceptance of others: to transfer to another in exchange: *n.* Having the state, condition, or quality of giving of one's concern, knowledge, or interest, in behalf of others, as in *Friendship*, or *Stewardship*. The art or skill of, as in *penmanship, marksmanship*.

That's why we do everything in our power to supply what you need (and want) in professional audio products and service. Our staff members were chosen because "they care", and because they're good at what they do... two of the same qualities we look for when we select manufacturers of product lines. Before we make it available to you, all equipment and instrumentation is first tested and approved by our testing division. Our research, quality control, hard work and, most of all, our level of caring, have earned for us a large number of the most consistently satisfied customers in the recording service business.

Sure, we could give you a bunch of "hype" and make a lot of impossible promises. But remember what happened the last time someone did that? When time came to back up their claim, you got stuck holding a bag of hot air... or worse.

The difference is people. The difference between equipment working or not, or the service coming through on time or not, is the people behind it. Customers and clients know it, manufacturers and reps know it, and I think you know it. But if you haven't seen it in action for a while, call us. Tell us about that studio you're ready to build, or that new piece of equipment you've been debating about buying.

We've served and created, designed and assisted all over America. Our prices and fees include free access to our imagination... make use of it.

When it seems like nobody does,
We Givaship!

VALLEY AUDIO

DIVISION OF TODRANK TECHNICAL SYSTEMS, INC.

P.O. Box 40743 • 2821 Erica Place • Nashville, Tennessee 37204 • 615-383-4732

MASTERS OF AURAL GRATIFICATION

for additional information circle no. 91

Tascam or TEAC multi-track lines. It is a creative tool or musical instrument, a recording system capable of making practice tapes with amazing ease at a very low cost."

Mohrhoff said the Portastudio carries a suggested retail of $1,100 and will be available nationally in October.

Up to 10 instruments, or vocals, can be recorded using TEAC simul-sync "ping-pong" recording with only one-time dubbing for each instrument. Mohrhoff said the Portastudio records only two tracks in sync at one time, but will play back all four tracks simultaneously.

The mixer has four line or mike inputs, a pan pot, individual bass and treble controls on each track, easy track-to-track dubbing without cabling, mix down from 4 to 2 channels for dubbing to an external recorder or audio system, tape cue monitoring, stereo auxiliary return input for external echo unit hook-up, and four large VU meters. Mixer specifications include 68 dB (weighted) signal-to-noise ratio, 20 to 20,000 Hz frequency range, -60 dB mike input (nominal level unbalanced for 200 ohms or more), -10 dB line input (nominal level unbalanced).

The Portastudio's cassette section has a two-motor, soft-touch logic control transport (including an FG-servo DC motor for the capstan), pitch control for precise tuning or special effects, full-time Dolby noise reduction, and a faster-than-normal cassette tape speed of 3¾ ips for wider dynamic range. The cassette has less than 0.04% wow and flutter, 20 to 18,000 Hz frequency response, 63 dB signal-to-noise ratio (with Dolby), higher than 50 dB (at 1 kHz) crosstalk track-to-track, and pitch control of ±15%.

The M-144 measures 18-1/8 W x 4-5/8 H x 14-9/16 D (in inches).

TEAC CORPORATION OF AMERICA
7733 TELEGRAPH ROAD
MONTEBELLO, CA 90640
(213) 726-0303

for additional information circle no. 93

CYBERSONICS DM 2002 DISCMASTER

Cybersonics announces the release and delivery of the DM 2002 Discmaster. The DM 2002 is a revolutionary concept in Discmastering lathes. One of the most interesting features of this lathe is its compact size.

Tom Lippel, president of the company, located in North Hollywood, says its unique concept is one of the reasons it was over five years in development. This product was totally designed from scratch, with all the features, functions and quality of any product available today. "Don't let the size fool you."

The Electro Mechanical systems employed are the ultimate in simplicity and accuracy from its quartz lock servo drive turntable to its patented cutter head suspension system, totally unique and unlike anything in use today.

The design criteria of compactness was intentional with the thought in mind of expanding discmastering to unusual locations such as the recording control room environment, mobile applications, studios with limited space, and the possibility of high production, multiple unit installation. Or, just being able to put two lathes in a normal mastering room where one older unit stood before.

One of the outstanding features is its integral computer. The pitch control function of the computer is most interesting. All control signals are taken from the

for additional information circle no. 92

Jimmy Messina
OASIS
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Engineered by Don Murray
ON COLUMBIA RECORDS
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33 West Haley St. Santa Barbara CA 93101
(805) 963 4425
MCI
4007 N.E. 6th AVE. FT. LAUDERDALE, FLA. 33334
for additional information circle no.

standard preview head on the tape playback system to allow circuits to anticipate what is going to happen to the groove when it is cut. The preview signal is also stored for one revolution in the computer, where a comparison is made between what the previous groove looked like and what the upcoming groove will look like. This information is converted into pitch and depth control signals.

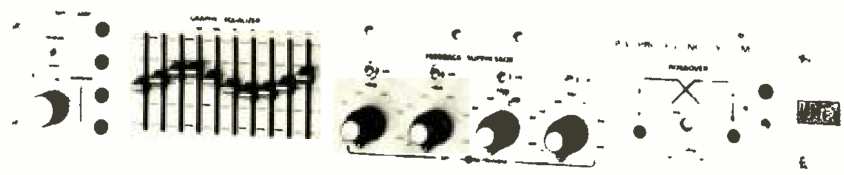
The computer receives new information constantly and is capable of up-dating pitch and depth information from two to eighteen times per turntable revolution. A programmable delay time in the computer's logic compensates for preview head-to-playback head distances, disc speed, and tape speed, thereby eliminating the need for complicated tape paths on the master tape machine.

All control signals are formed into eight-bit words, in order to facilitate automated disc recording, in conjunction with presently available computer mixdown system, or with future microprocessor-based system. This feature also makes it possible to link several lathes together electronically, for high production or direct-to-disc recording.

The vacuum disc hold-down system is incorporated into the turntable drive motor shaft, thereby eliminating the need for external drive motor shaft, thereby eliminating the need for external hoses and manual disc diameter selection. The diameter selection is automatically achieved.

The inspection microscope is on a leadscrew that is directly driven by a servo system for easy front panel control of movement across the disc.

Front panel controls are kept to the essentials for day to day operation. On a



sub-panel located directly under the front panel are specialized control functions, such as stylus head, adjust, band time, deepening control, etc.

Also located on the front panel is a digital electronic ruler which reads out cutting diameter to within .01 inches, digital stylus heater current indicator and digital strobe indicating speed of turntable.

Analog meters are used for LPI and depth of cut.

All lathe functions are controlled by push buttons on the front panel with LED indication of current operating status. These are the only mechanical switches in the entire machine. All functions related to cutter head position are accomplished through the use of linear position decoding in conjunction with solid state switching devices incorporated within the lathe computer.

Cybersonics has not only eliminated most mechanical switches but has also eliminated any belts, pulleys, springs, cables and hydraulic dampers, all of which are prone to periodic failure or change and adjustment.

Total direct drive and digital electronic simplicity insure reliability and ease of operation, allowing the creative process of the mastering engineer to take precedence in the final act of transfering of tape to disk.

CYBERSONICS
11128 WEDDINGTON
NO. HOLLYWOOD, CA 91601
(213) 766-7104

for additional information circle no. 95

PA PROCESSING SYSTEM FOUR EQUALIZATION TOOLS IN ONE

The UREI Model 567 PA Processing System offers compact, economical signal processing for small auditorium and church public address and sound reinforcement installations where full 1/3 octave equalization and/or elaborate tuning is too costly.

The 567 includes: an input amplifier/gain control, level monitor, pink noise source; for setup; a 10 band Graphic Equalizer; a four frequency Feedback Suppressor; and a two-way Electronic Crossover with continuously adjustable crossover frequency point. These are contained in a single rack mounted unit with a regulated power supply.

The 567 operates at high or low impedances, and at any nominal level from -20 dB. Rear panel patch points allow for other signal processing equipment (i.e.

for additional information circle no. 96

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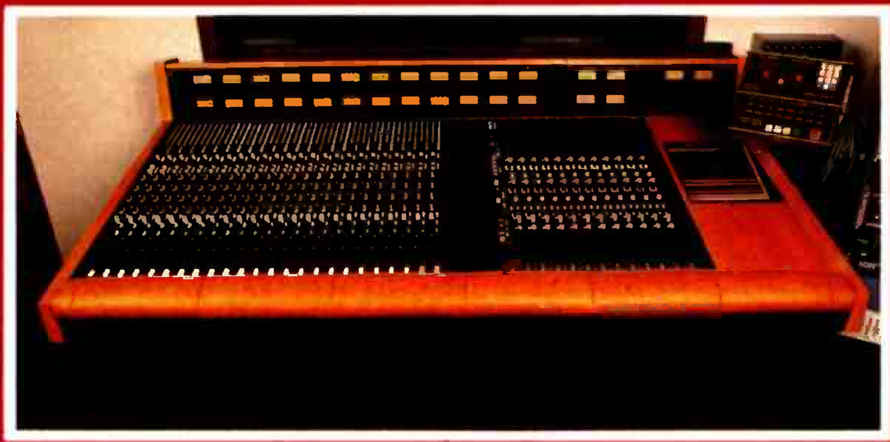


The Logical Choice

For Further Information Contact Loft Modular Devices
91 Elm Street, Manchester, CT 06040 (203) 646-7806

for additional information circle no. 97

Limiter or Leveling amplifier) between the Feedback Suppressor and the Electronic Crossover sections.

UNITED RECORDING ELECTRONICS INDUSTRIES (UREI)
8460 SAN FERNANDO ROAD
SUN VALLEY, CA 91352

for additional information circle no. 98

BGW INTRODUCES THEIR NEW AMP — THE 50A

The 50A features teflon interconnecting wires, totally modular construction, an all-steel chassis and cover, metal-cased output transistors, and large aluminum heat-sink extrusions.

At 8 ohms over a power band from 20 Hz to 20 kHz, the BGW 50A delivers 25 watts per channel with a maximum THD of no more than 0.02% at any power level from 250 milliwatts to 25 watts, and 0.03% THD from any power level from 250 milliwatts to 50 watts monaural.

The manufacturer emphasizes the following features: less than 0.01% intermodulation distortion from 250 milliwatts to rated power; a small signal frequency response of +0, 3 dB, 1 Hz to 100 kHz, +0, 0.25 dB, 20 Hz to 20 kHz; hum and noise level (unweighted, 20 Hz to 20 kHz) better than 102 dB below 25 watts; and 0.7 volt input sensitivity for maximum power output and an input impedance of 15K ohms.

Further, the damping factor is greater than 400 to 1 at 8 ohms and output impedance designed for any load impedance equal to or greater than 4 ohms.

The 50A is 1-3/4" x 19" standard rack front panel by 11-1/2" deep, and weighes 15 lbs.

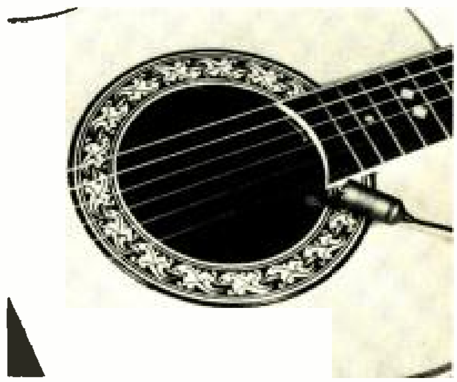
BGW SYSTEMS
13130 SOUTH YUKON AVENUE
HAWTHORNE, CA 90250

for additional information circle no. 100

MINIATURE INSTRUMENT PICKUP ANNOUNCED BY SHURE

Specifically designed to be mounted on acoustic stringed instruments and other acoustic musical instruments, the SM17 is a very small, lightweight unit. The SM17 comes with two mounting options: an expansion mount for string hole mounting on violins, violas, and cellos, and a clip that fits on the sound hole of acoustic guitars, and edges of other instruments, such as the bell of a saxophone.

By enabling the performer to mount the microphone right on the instrument, the SM17 allows high quality acoustic pickup with excellent isolation from other instruments and freedom from feedback. The SM17 also features an omnidirectional pickup pattern and a frequency response well suited for instrument use.



The SM17 is supplied wired for low-impedance microphone inputs. The low-impedance, balanced line operation is desirable where long cable lengths are required or conditions where severe hum pickup may exist.

The unit has a rugged aluminum case construction with a smooth exterior and a recessed grille screen to minimize accidental rubbing noise.

Other supplied accessories include an attached 3 m (10 ft.) cable, additional bushings for the expansion mount, and two cable clips.

User net price for the Shure Model SM17 Microphone is $76.80.

SHURE BROTHERS INC.
222 HARTREY AVENUE
EVANSTON, ILL 60204

for additional information circle no. 101

please mention . . . YOU SAW IT IN R-E/P

HOW TO CHOOSE FROM SOME OF THE CHOICEST MICROPHONES WE'VE EVER MADE.

ECM-33F ECM-150 ECM-260F ECM-990F ECM-56F ECM-23F

Among recording professionals, Sony is widely recognized as an expert on microphones. That's because we're continually applying new technology to deliver better sound.

Our latest innovation is the exclusive Back Electret condenser microphone capsule, which delivers response truer than ever thought possible.

You can get this capsule in a variety of Sony mikes. And that's a bit of a problem: it's hard to know which mike is appropriate for your recording needs.

Therefore, let us clear up any confusion:

MICROPHONES THAT ARE AT HOME IN YOUR HOME STUDIO.

If you're involved in the music business and have a home studio, you need a microphone as professional as the rest of your equipment.

For all-purpose recording, we recommend the Sony ECM-56F. It's a uni-directional Back Electret condenser mike with excellent transient response, good for close miking of both instruments and voices.

For recording instruments only, the uni-directional Back Electret condenser ECM-33F is ideal. It provides flat frequency response over the entire range, and picks up amplified and non-amplified instruments equally well.

Both of the above plug into mixers for multi-channel recording.

LOCATION MIKES, FOR STUDIO SOUND WITHOUT THE STUDIO.

But suppose you want to record on location. At a rock concert, say, or a performance of your church choir or glee club. Sony has mikes that, combined with your tape recorder, practically make up a portable studio.

Take the ECM-990F, an especially versatile and lightweight stereo Back Electret condenser mike. You can vary its directional quality to adapt for everything from solo voice to small groups to full orchestra.

Or choose an ECM-23F. It runs more than 6,500 hours on a single AA battery, and it's uni-directional. Use a pair when you want to create a stereo effect. The ECM-23F also incorporates Sony Back Electret technology.

RECORD FOR RECREATION AND STILL RECREATE NATURAL SOUND.

Maybe you just need a mike to use at home, to record family sing-alongs. Or someone's performance on guitar or piano, for your own enjoyment.

You can still get a Sony Back Electret mike at a very affordable price. It's the ECM-260F, which plugs into a tape recorder and makes whatever you record—instrumentals, singing or speech—sound true to life.

For greatest versatility, use our ECM-150 omni-directional condenser mike. It's Sony's tiniest mike, smaller than a dime in circumference, and you can clip it to the fingerboard of a guitar or use it as a lapel or tie tack mike. (Incidentally, it's great for business conferences or any occasion when you want the mike to be inconspicuous.)

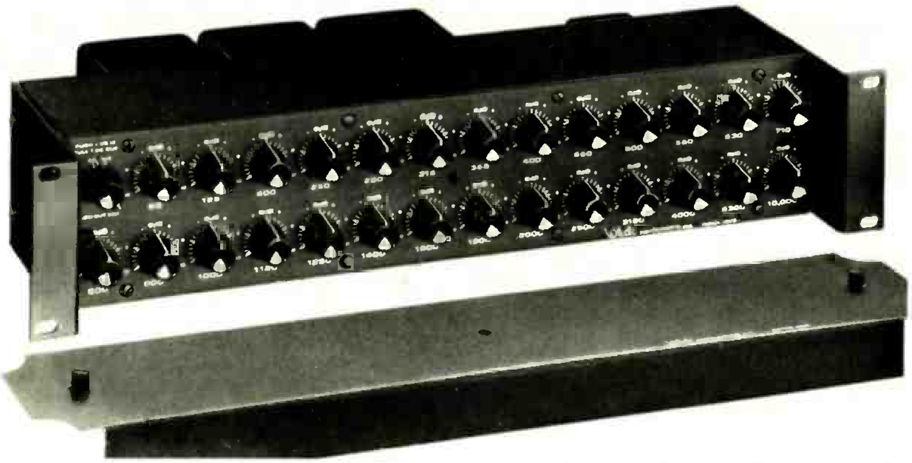
Whatever you need to record, and wherever you need to record it, there's a choice Sony mike to do the job.

And now that you know which mikes to choose, all you need to do is see your Sony dealer.

SONY

We've never put our name on anything that wasn't the best.

for additional information circle no. 102



WHITE INSTRUMENTS ANNOUNCES TWO NEW ONE-SIXTH OCTAVE ACTIVE EQUALIZERS

The Model 4240 is a cost effective one-sixth octave hybrid which concentrates its double resolution in the speech intelligibility region between 200 Hz and 2 kHz. End shaping is accomplished with broader filters. The intended application of the Model 4240 is the equalization of sound systems employing speech as the primary program material.

The Model 4310 is similar to White Instrument's first one-sixth octave equalizer, the Model 4301, but the one-sixth octave resolution has been moved up in frequency to accommodate the modes found in larger rooms. It features 29 one-sixth octave bands from 180 Hz through 4,500 Hz. End shaping is accomplished with 11 one-third octave filters from 31.5 Hz through 160 Hz and 5,000 Hz through 8,000 Hz plus a single octave wide filter centered at 12.5 kHz.

To support these new equalizers, White Instruments reports that they are in full production of their new System 200 Signal Analyzer which features interchangeable filter sets, total software dependent microprocessor control, RT-60, 8 non-volatile memories, dual mode display, and a number of software dependent features and functions.

WHITE INSTRUMENTS INC.
P. O. BOX 698
AUSTIN, TX 78767
(512) 892-0752

for additional information circle no. 103

NEW LOWER MID-RANGE REPRODUCER FROM EASTERN ACOUSTIC WORKS

Designated the MR-109, the new lower mid-range unit is designed to meet the demanding requirements of high level sound reinforcement. The MR-109 is said to excel in both power handling and distortion as compared to other compression driver/horn combinations. This new level of performance results in exceptional clarity and definition in the principal music band of 200 to 2,000 Hz. In addition, high frequency distortion is reduced by allowing a higher crossover frequency to the high frequency drivers.

The MR-109 is a straight exponential horn with a 200 Hz cutoff frequency. The horn flare is constructed of hand-laminated fiberglass with high density sheets inserted to damp resonances to well below cut-off. Additional damping material is used inside the enclosure to reduce resonances, even at 130 dB SPL in the throat region.

The drive unit incorporates a large Alcomax magnet thermally bonded to a heatsink to improve heat dissipation, minimize stray magnetic fields and maximize the efficiency of return magnetic fields. This magnetic structure produces 12,000 Gauss on the 75 mm copper edgewound ribbon voice coil.

The horn box is constructed of cross grain laminated Baltic Birch, and comes standard with metal corners, skids, handles and a protective lid that clamps on with recessed spring loaded twist catches. An internal passive 200 Hz crossover is provided to protect the system from inadvertant low frequency signals. Input connections are male and female 15 amp twist-lock connectors, dual 14" phone jacks and dual banana plugs.

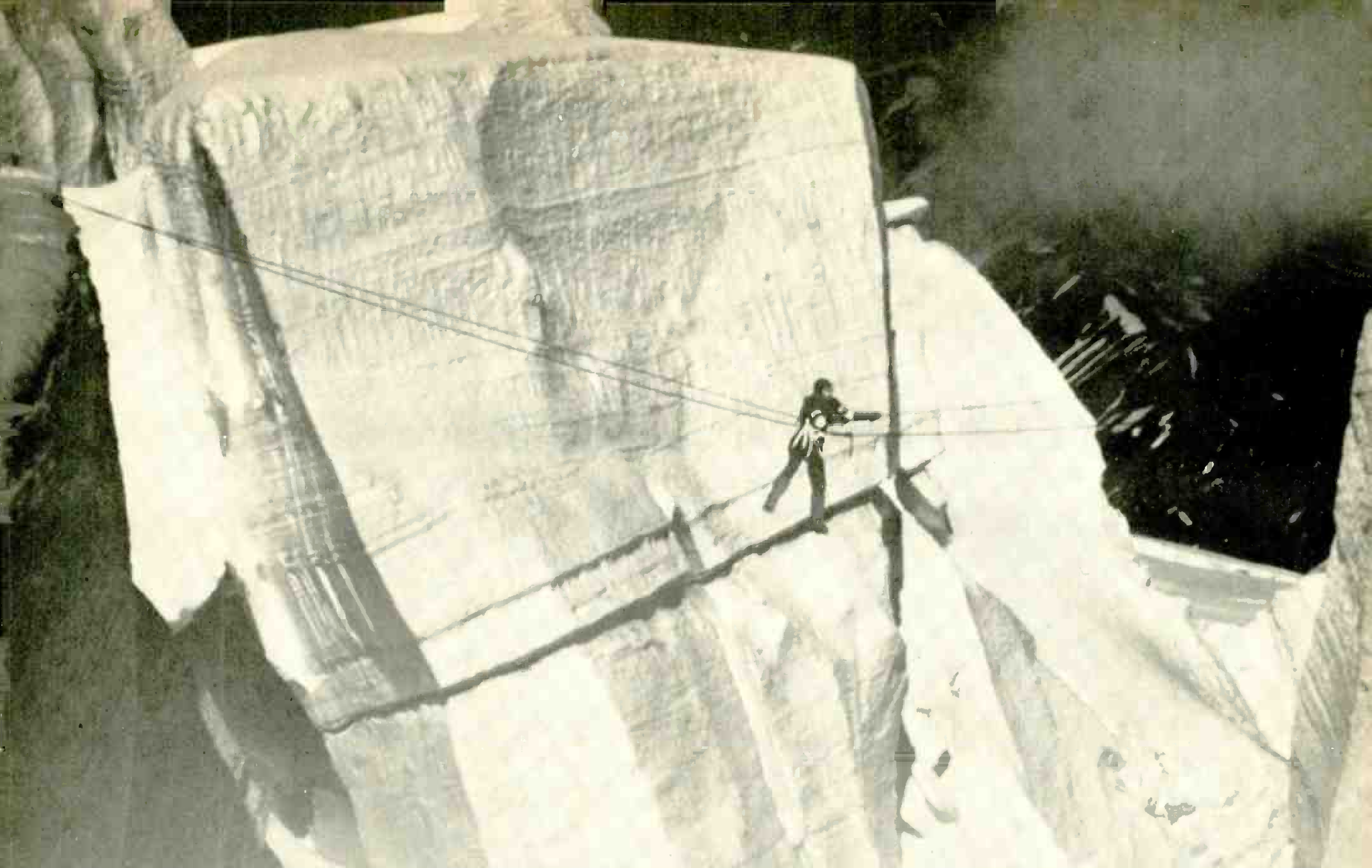
The drivers are built to withstand continuous use at high power inputs without overheating. When operated within the units specifications, field failures are virtually unknown, making is unnecessary to haul spares around on location.

EASTERN ACOUSTIC WORKS
59 FOUNTAIN STREET
FRAMINGHAM, MA 01701
(617) 620-1478

for additional information circle no. 104

ORANGE COUNTY ELECTRONICS INTRODUCES NEW DEQ FULL PARAMETRIC EQUALIZER MODULE

The new DEQ module is a four-band parametric with center frequencies variable from 20 Hz to 20 kHz in overlapping five octave (32:1) ranges. Each section tunes over an 80 dB control range (60 dB cut and 20 dB boost). Bandwidth is variable from .15



Professionals depend on their equipment.

Like their BGW amplifiers. Why is it so many have come to rely on BGW? Why in less than ten years have BGW amps become the number one choice among audio pros worldwide?

Because their legendary performance refuses to fail even under the most severe conditions you can throw at them. Rugged, awesome power that's been tamed by continuous common-sense engineering. That's why there are more BGW amps in discos than any other kind, and why there are so many in recording studios and on concert stages. BGW has earned a reputation for building superbly engineered products ... massive heat sinks, large safe operating area, redundant output stages, welded steel modular construction are all synonymous with a BGW product.

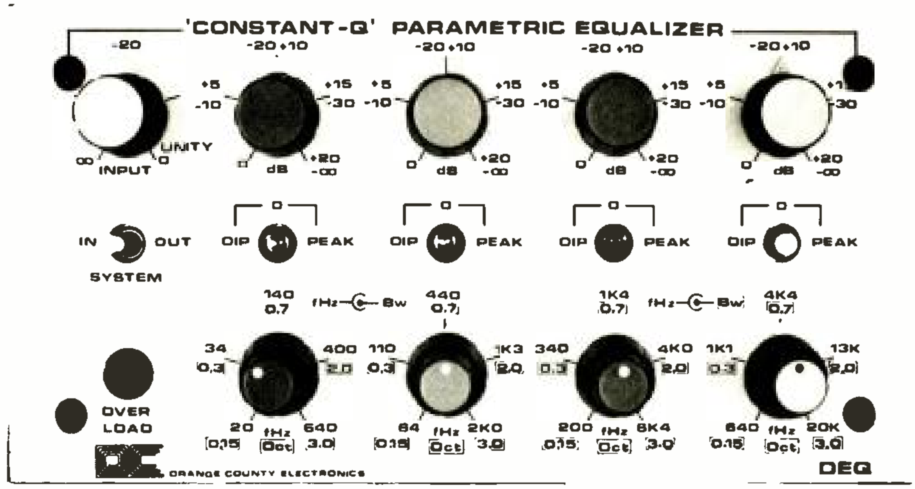
We are now proud to introduce a new cost-effective 175 watt per channel power amplifier ... the Model 600. It's a quality basic power amp, built around our super reliable 750 B/C output modules. It's in a big 8¾" high rack-mount package so it runs cool and costs substantially less than a 750C. It's a quality BGW amp and the answer to the professional who wants BGW on a budget.

Check out the new 600 at your dealer. He'll show you an amp that lives up to your expectations with performance you can compare to anyone ... and reliability that compares to no one.

BGW SYSTEMS **Depend On Us.**

BGW Systems Inc., 13130 S. Yukon Ave., Hawthorne, CA 90250 (213) 973-8090 In Canada: Omnimedia Corp., 9653 Cote de Liesse, Dorval, Quebec H9P 1A3

for additional information circle no. 105



- 3 octaves (Q = 10 - 0.33) as well.

A unique feature of this new parametric equalizer module is that all controls are non-interacting. This means that when the bandwidth is varied, for example, there is no change in level.

This module also offers extremely low noise operations. Signal-to-noise is 110 dB with all sections in 20 dB boost. Distortion is 0.05% THD @ 18 dBm output. Standard balanced or unbalanced operation is available. An overload indicator warns of excessive levels in any stage of the module. Output capability is +30 dB (10K load); +24 dB (600 ohm load).

The DEQ is designed to be used as a stereo parametric equalizer in the standard Orange County FR-1 rack frame or as a mono unit in the FR-2 Desk Housing. It can also be utilized as part of the Orange County VS-1 Stressor, thereby offering more control in a signal-processing system than is available elsewhere in the market, according to the manufacturer.

**PARASOUND INC.
680 BEACH STREET, SUITE 414
SAN FRANCISCO, CA 94109
(415) 673-4544**

for additional information circle no. 108

SUB-SONIC PROCESSOR SAVES DISCO AND MONITOR SPEAKERS

UREI's new Model 501 two channel Sub-Sonic Processor attentuates frequencies below the audible range that may cause excessive speaker cone excursions in high level sound programs.

Sub-sonic sounds from turntable rumble, warped records, acoustic coupling of turntables to speakers and wind blast in microphones are removed saving amplifier power and costly speaker repairs.

The Model 501 has a two-position response switch: In the "Flat" position, response is -3 dB at 30 Hz and down more than 50 dB at 5 Hz, with a rolloff of 18 dB/octave. The "Boost" position actually enhances the LF response by adding a 5 dB peak at 40 Hz. The signal is down 3 dB at 27 Hz and more than 40 dB at 5 Hz also at 18 dB/octave. The boost compensates for the drop in low frequency response of many speaker systems.

The Model 501 comes with its own power supply. It can be connected to any source up to +30 VDC . . . current required is less than 20 mA. Uses ¼" phone plugs In/Out.

**UNITED RECORDING ELECTRONIC INDUSTRIES (UREI)
8460 SAN FERNANDO ROAD
SUN VALLEY, CA 91352**

for additional information circle no. 109

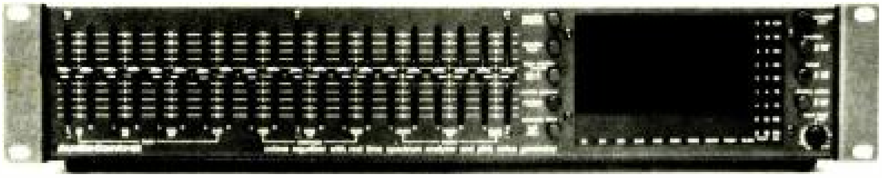
AUDIO CONTROL INTRODUCES GRAIPHIC EQUALIZER/REAL TIME SPECTRUM ANALYZER

The Audio Control C-101 is said to be the world's first graphic equalizer with a real time spectrum analyzer built in.

"For the first time, the listener can actually see music broken down into ten separate frequency bands and watch the effect of equalizer controls," states Greg Mackie, Audio Control president.

The Audio Control Spectrum Analyzer Equalizer features a 92 LED display divided into ten frequency bands with nine vertical level LED's and two green center level reference LED's activated in the sound pressure level mode.

Four push buttons control speed of display change, meter range, choice of



sound pressure level or real time analyzer and on/off. In addition, a center detent input level control is built in for calibrating input sensitivity.

To aid in using the Spectrum Analyzer Equalizer for room acoustic analysis, tape bias and speaker adjustment, a high quality pink noise generator is built into the unit.

The equalizer portion of the Audio Control Spectrum Analyzer features stereo paired center detent slide pots in ten octave bands from 32 to 15.5 kHz. In addition an 18 dB per octave Tchebychev subsonic filter is included. Finally, the Equalizer includes a phase correlation rumble reducing circuit for reduction of rumble, phase and intermodulation distortion below 200 Hz. It effectively mono's bass below this frequency. Since master tapes are often mixed or mastered with mono bass at these frequencies, and since in the very low bass source directionality is undiscernable, no reduction in program material quality results. Frequency response of the unit is rated at 3 Hz to 100 kHz with THD at less than 0.025%. EQ range is ±15 dB.

AUDIO CONTROL CORP.
6520 - 212th S.W.
LYNWOOD, WA 98036

for additional information circle no. 111

OMNITEC™ PREMIUM OMEGA-SHAPED MOVING MAGNET CARTRIDGES INTRODUCED BY AUDIO-TECHNIC

Innovative doughnut-shaped coil dual moving magnet cartridge design is featured in the new Omnitec™ line of phono cartridges from Audio-Technica.

Each of four cartridges in this line has two omega-shaped coils, whose laminated cores also serve as pole pieces for the moving magnets. Technical benefits of the single-piece design are low inductance and greater efficiency, according to Audio-Technica. Audible benefits include cleaner reproduction of transients and high-level, high-frequency signals.

Omnitec cartridges use rigid, lightweight square shanked beryllium cantilever, and nude-mounted 0.2 x 0.7 mil elliptical tip diamond stylus. Stylus assembly is secured to the cartridge body with a set screw to eliminate unwanted resonances that can occur with conventional stylus assemblies.

Like other Audio-Technica phono cartridges, the Omnitec models feature two separate magnetic systems for maximum stereo separation.

The top-of-the-line AT-25 cartridge, with integrated headshell and built-in device to adjust overhang for adjusting stylus overhang to optimum tonearm specifications, is nationally advertised at $275.

AT-24, the same cartridge minus the headshell and overhang adjustment, has a suggested price of $250.

A more economical dual magnet cartridge is the AT-23, which is nationally advertised at $225 with integral headshell and built-in overhang adjustment. For $200, audiophiles can choose the AT-22 phono cartridge, identical to the AT-23, minus the integrated headshell and distance gauge.

AUDIO TECHNICA
33 SHIAWASSEE AVENUE
FAIRLAWN, OHIO 44313
(216) 836-0246

for additional information circle no. 112

ALTAIR INTRODUCES NEW FUSE CLIP

Altair Corporation recently introduced the FC-1 Fuse Clip. The FC-1 solves the problem of not having the proper replacement fuse handy when the fuse on an amplifier or other electronics blows. The fuse clip has four clips mounted on a 1⅛" x 1⅝" phenolic block and holds two ¼" x 1¼"

spare fuses. Double-sided foam adhesive on the back of the FC-1 allows it to be conveniently attached to the amplifier at a location close to the fuse holder.

ALTAIR CORPORATION
202 WEST BENNETT STREET
SALINE, MI 48176
(313) 429-5454

for additional information circle no. 113

VOCODER PLUS INTRODUCED BY ROLAND

The Roland Vocoder Plus is an instrument that combines vocoder circuitry with two other tone-generating sections (String, Human Voice) to achieve a dramatic and usable effect.

Each of the three sections may be independently assigned to cover the whole keyboard, or either the upper or lower half. In addition, each half ot the keyboard feeds into its own output so that the Vocoder Plus can be run in stereo.

The String section produces orchestral string sounds that are warm and natural feeling. The tone and attack time of this section can be controlled independently,

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WHITE INSTRUMENTS

VISION-SOUND PROFESSIONAL AUDIO • 110 GRAND AVENUE, ENGLEWOOD, N.J. 07631 • (201) 871-471-4101
toll-free info line:
(800) 526-4710

R-e p 177

for additional information circle no. 115

the release time is shared with the Human Voice section.

The Human Voice section is said to produce an incredibly lifelike chorus of human voices. The upper half of the keyboard contains one female and one male chorus, the lower half contains two male choruses.

The Vocoder section processes the spoken or sung human voice, and uses this information (or program) to modify another musical signal (known as the carrier). The Vocoder section uses the Human Voice section as its carrier, but will also process an external signal if desired.

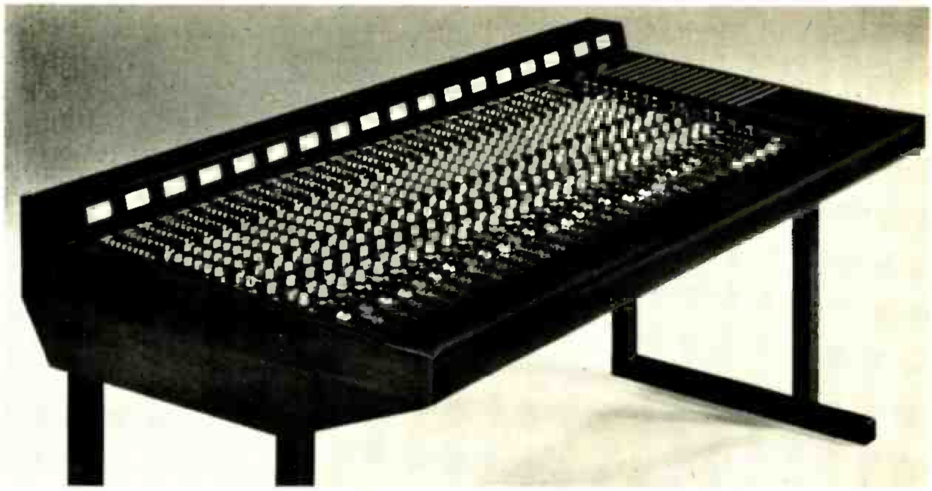
In a live performance, the Vocoder Plus can be used to strengthen a band's vocal capabilities by literally adding a chorus of voices singing the same part. The String and Human Voice sections give additional enhancement.

The Vocoder Plus contains a Balance control between all sections as well as vibrato controls that allow selection of rate, depth and delayed vibrato. The Microphone Input will accept either phone plug or XLR connector.

The list price is $2,695.

ROLAND CORP. U.S.
2401 SAYBROOK AVENUE
LOS ANGELES, CA 90040
(213) 685-5141

for additional information circle no. 116



NEW FROM SPECK: MODEL 800-D MIXING CONSOLE

The Speck 800-D is a 28-input, 16/28 output studio mixing console. The console is totally modular with 28 input modules, a master module, and a complete communications module housed in a sturdy mainframe that contains 16 large illuminated VU meters.

Each input has 8 panable assigns, 3 band parametric equalizers, 3 sends, pan, stereo solo, a long throw slide fader, and most important: a second line input with an independent slide fader, a 2 band equalizer and pan.

Since the Speck 800-D has two discrete line input circuits for each input module and 28 assignable direct outputs in addition to the 8 submasters, the 800-D is well suited for 16-, 24-, or 32-track studio operations. The stereo program buss is independent of the multitrack assign section which allows the console to feed a full compliment of ½-track, ¼-track, and cassette recorders simultaneously during mixdown.

A 384 point patch bay is standard on all "D" models, and is wired to accept two 16-track, 24-track, or 32-track tape recorders. All connections for tape recorders, power amps, chambers and outboard equipment

for additional information circle no. 117

are made via eight high density multi-pin connectors at the rear of the console.

Price of the Model SP-800-D-28 is listed at $23,900.00.

SPECK ELECTRONICS
7400 GREENBUSH AVENUE
NORTH HOLLYWOOD, CA 91605
(213) 764-1200

for additional information circle no. 118

NADY WIRELESS AUDIO TRANSMISSION SYSTEMS

Nady Systems, Inc., (formerly Nasty Cordless) will be showing their two lines of professional wireless audio transmission systems. They have both a tunable system as well as a fixed frequency approach.

Nady Systems claim the tunable unit is interference free since it operates on blank spots on the 88 - 108 MHz bandwidth of radio frequencies. Its normal range is 250 feet.

Three different tunable systems are available. The Nasty Cordless "Black Transmitter" was designed for the professional on a budget. It is used in conjunction with the users' own receiver, hence the quietness of operation is determined by the inherent signal-to-noise of the receiver that is used. The "Blue Transmitter" requires a "Pro 400" receiver. This system gives a frequency response of ±3 dB from 20 to 20 kHz with less than 1% distortion and a better than 99 dB signal-to-noise ratio.

For the most exacting tour requirements Nady offers the "Pro 500" receiver which when used with a "Blue Transmitter" and the "Pro 400" receiver will guarantee no "null spot" or "drop out" of the received signal. (An area where the radio signal is momentarily not received properly.) The Pro 500 combiner circuitry continuously monitors the received RF signal strength of both receivers and instantaneously selects the stronger of the two.

The fixed frequency lines offered by Nady include the "VHF 600" and the "VHF 700 True Diversity." Designed for applications that don't require tunable frequencies this line offers a 102 dB signal-to-noise ratio, 20 to 20 kHz frequency response, and over 100 dB image rejection. This last feature gives complete channel isolation without phasing or squelching which commonly plagues many other VHF systems.

Nasty Cordless products are currently being used by many top acts, including Rod Stewart, The Rolling Stones, Bruce Springsteen, and Alice Cooper.

NADY SYSTEMS
1145 - 65TH STREET
OAKLAND, CA 94608
(415) 652-2411

for additional information circle no. 119

for additional information circle no. 120

JBL ANNOUNCES THE CABARET SERIES

James B. Lansing Sound, Inc., announces the Cabaret Series, developed for club sound reinforcement applications. Each unit is designed without compromise for a specific application. The present line consists of the 4602 Stage Monitor, the 4622 Lead Instrument System and the 4680 Line Array.

Cabaret Series products are completely self-contained, rugged enclosures which provide flush-mounted professional road handles and offer ease of set-up and tear-down. Loaded with JBL K Series musical instrument loudspeakers, they are built to provide greatest accuracy, efficiency and power handling capability.

The 4602 Stage Monitor offers smooth, wide frequency response and controlled directiviity for stage and general purpose sound reinforcement applications. A wedge-shaped enclosure allows it to be placed in either an upright position, or at 30- or 60-degree angles to the floor. The enclosure houses the K120 12-inch loudspeaker, 2402 high frequency ring radiator, and a specially designed crossover network: a highly accurate and efficient system, capable of handling up to 100 watts continuous sine wave power.

The 4622 Lead Instrument system features two K120 12-inch loudspeakers

mounted in an enclosure engineered specifically for the distortion-free great power handling capability and clarity characteristic of K Series products. Suitable for lead guitar and keyboards, the 4622 can accommodate up to 200 watts continuous sine wave power.

The 4680 Line Array features four K110 10-inch loudspeakers and a 2902 High Frequency Power Pack for clean, crisp sound reproduction. A JBL Professional Series 4682 Line Array housed in a special Cabaret Series enclosure, the 4680 is ideal for any application demanding uncompromising sound quality over a wide spectrum of power levels and frequency ranges.

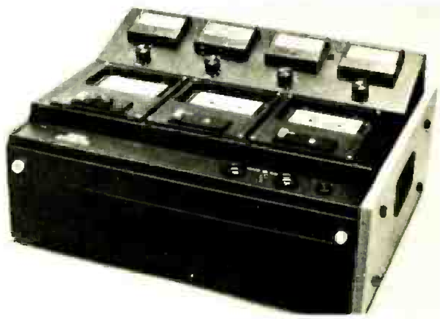
Specially designed corners on the 4622 and 4680 provide additional protection and also permit stacking of the systems in a secure vertical array.

JAMES B. LANSING SOUND, INC.
8500 BALBOA BOULEVARD
NORTHRIDGE, CA 91329
(213) 893-8411

for additional information circle no. 121

NEW PROFESSIONAL-QUALITY CASSETTE MASTER TWO SLAVE IN CASSETTE DUPLICATOR

Otari has announced the introduction of the DP-4050-C2, a new compact version of its widely-used professional-quality DP-4050 in-cassette duplicator system. Featuring a cassette master and two slaves, it has the capability of adding up to nine additional slaves in groups of three, for a total of eleven slaves all driven from one master.



Although new in configuration, the DP-4050-C2 retains all of the proven features of the more expensive DP-4050-OCF and

for additional information circle no. 122

long-life ferrite heads, flip-down panel for easy access to VU meters, and duplication of all four tracks simultaneously in one pass. From two to eleven C-30 cassettes can be produced in less than two minutes, depending on the number of slaves.

Servo-controlled modular transports are an important benefit of the DP-4050-C2. Each is an individual unit, such that failure of one unit does not disable the entire machine. The master transport features automatic rewind and stop. In addition, it has a ± 4% speed control to compensate for non-optimum cassette masters.

The add-on slave unit is called the DP-4050-Z3 and contains three cassette slave decks. It is easily attached to the master by means of a multi-pin connector cable.

Suggested professional user price of the DP-4050-C2 with cassette master and two slaves is $2,950. Price of the DP-4050-Z3 three slave add-on unit is $2,750.

OTARI CORPORATION
1559 INDUSTRIAL ROAD
SAN CARLOS, CA 94970
(415) 592-8311

for additional information circle no. 123

ALTAIR CT-3 MIKE CABLE TESTER

The CT-3 is said to be an entirely new concept in microphone cable-testers. It is six inches long, 3/4 inch in diameter, is made from extremely rugged epoxy fiberglass and weighs only four ounces. It easily fits into a shirt pocket (and even has a pocket clip). The CT-3 has an XLR-type connector in

each end and is switched on by pushing in on the cable's female conenctor. There are no buttons to push, so both hands are free to wiggle the cable and connectors to check for loose, intermittent connections. All three of the cable's conductors are tested simultaneously and continuously. A failure in any one of them will cause one of the two light emitting diodes to go out. One stays on to let you know the cable is being tested. The CT-3 checks for all the common wiring faults: shorts, open circuits and cross wiring (including reverse phase). With the CT-3½ Remote Testing Accessory, which is provided with the CT-3, a mike cable can be tested without bringing its ends together. This means a snake cable can be tested after a sound system has been set up, without moving the cable. Permanent studio wiring can likewise be tested. This method will detect many faults (such as shorts to ground, open ground, and reversed phase) that would be overlooked by a simple mike check. The CT-3 is powered by a readily available mercury battery which will give many thousands of tests.

ALTAIR CORPORATION
202 WEST BENNETT STREET
SALINE, MI 48176
(313) 429-5454

for additional information circle no. 124

MEYER SOUND LABS ANNOUNCES ACD/JOHN MEYER STUDIO-STANDARD REFERENCE MONITOR

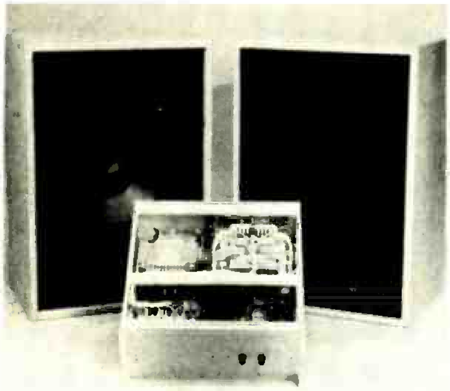
Designed to provide the analogue and digital recording engineer with a high output monitor system of unparalleled accuracy and transparency, this system is said to be an acoustic testing device, a listening tool with no perceptible tonal coloration or noise.

The Swiss constructed ACD system consists of two biamplifiers and electronic crossovers in a single chassis and two speaker cabinets with horns and 12-inch

for additional information circle no. 125

cone drivers.

The crossover and each channel of the power amplifier are electronically phase and amplitude synchronized to each of the elements of the speaker system. This provides the smooth linear phase response necessary for true stereo imaging and tonal balance.



The system features signal-to-noise ratio that exceeds that of digital tape recorders, 110 dB. No perceptual harmonic or intermodulation distortion. Accurate response ±3 dB from 27 Hz to 18 kHz at every frequency, not just 1/3 octave intervals, which hide many irregularities. Tested by sine wave generator swept through entire range, outdoors, pointed up, into B&K mike. Extremely fast rise and settling time eliminates transient smearing, 500 ns. High power output maintained continuously, no thermal losses, 115 dB. Smooth frequency response on and off axis. RMS power limiting allows undistorted peaks in excess of 125 dB.

Limiting occurs only when average power exceeds continuous rated output. This protects speakers without fuses or breakers and instantly resets when overload ceases.

Every system exhibits almost identical response due to absolute factory commitment of quality control and system-to-system consistency. Individual testing and calibration assures that each system is within specs.

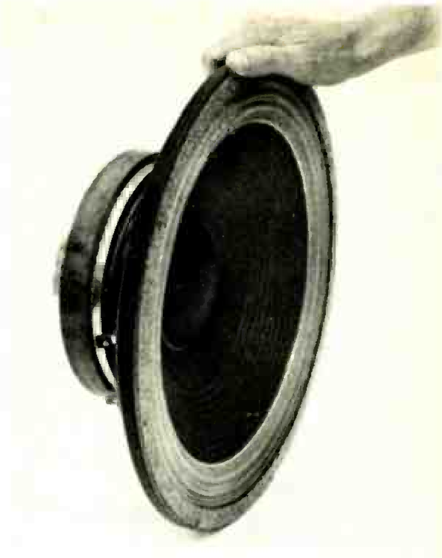
Options include the 200 X Subwoofers which move the power bandwidth of the ACD system one full octave lower. It produces 4 acoustic watts, 114 dB at 30 Hz and 120 dB at 50 Hz continuously, with higher peaks. It is an eight order system design.

MEYER SOUND LABORATORIES
2194 EDISON AVENUE
SAN LEANDRO, CA 94577
(415) 569-2866

for additional information circle no. 126

ESS/RCF INTRODUCES ITALIAN WOOFER

The ESS/RCF L15P/06C is a high powered bass driver conservatively rated to



handle 300 watts of continuous program material with a sensitivity rating of 97.5 dB sound pressure at a distance of 1 meter with 1 watt input. This 15-inch woofer can be crossed over at anywhere from 100 Hz to 1,500 Hz and its treated corrugated front surround insures linearity even at high outputs.

Konrad Kratz, ESS/RCF operations manager for the pro products division in the U.S. and Canada, points out that "it's the precision construction of the L15P/06C that gives it such outstanding capability. The

for additional information circle no. 127

— continued from page 15 . . .

four-inch copper wire voice coil is edge wound in *two* layers on the aluminum form, one layer wrapped around the outside, the other on the inside. This type of winding serves to prevent flexing and separation under heavy loads."

Kraft further explains "the coil assembly coupled with the long-throw linear excursion of the cone assembly allows the L15P/06C to deliver high powered low frequency response with detailed clarity." The cone itself is fabricated from heavy duty corrugated fiber to minimize cone breakup, and is supported by a cast aluminum frame for structural rigidity.

Most of the 23.3 lb. (10.6 kg) total weight of the driver is centered in the 20.5 lb. (9.3 kg) magnet assembly which is rear-vented to dissipate heat.

The L15/06C is suitable for disco systems, musical instruments and studio or home system monitors. Included in the sale of each woofer is an "Enclosure Size Chart" with precise measurements for enclosure and port size to achieve several desired frequency responses.

ESS, INC.
9613 OATES DRIVE
SACRAMENTO, CA 95827
(916) 362-4102

for additional information circle no. 128

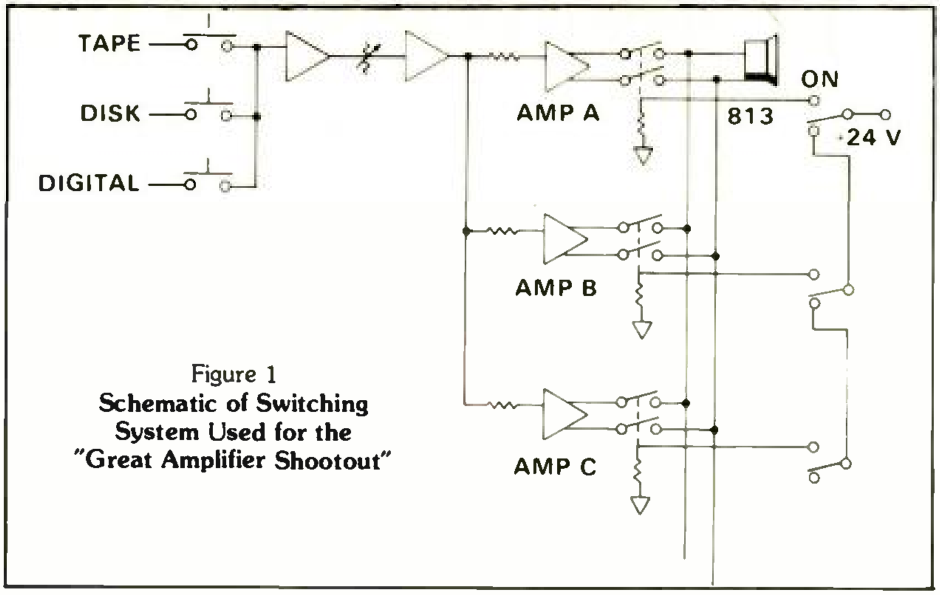


Figure 1
Schematic of Switching System Used for the "Great Amplifier Shootout"

the ideal power amplifier for use with their system, UREI sponsored what is thought to be the most impartial and accurate listening shootout, conducted in an actual control-room environment, comparing ten of the most prominent power amplifiers currently available in the marketplace. Each of the ten selected for the comparison had proven capable of providing the maximum SPL's typically demanded by studio clients, without observable clipping on the monitor oscilloscope. In alphabetical order they are:

1 - AB Systems 410A
2 - BGW 750B
3 - Cerwin Vega A600
4 - Crest P3501
5 - Crown PSA-2
6 - H H S500D
7 - MacIntosh 2300 (4 ohm output tap)
8 - Studer A68
9 - Unisync 350
10 - Yamaha P-2200

— *continued overleaf*

for additional information circle no. 129

— continued . . .

THE GREAT AMPLIFIER SHOOTOUT

Several hundred invitations were sent to, in the words of UREI, "an impressive list of the *golden-eared* people in the recording industry." The event took place on August 29, in the control-room of URC'c Western-United Studio 'A'. A buffet lunch was served.

The Impartial System

The competing amplifiers were assembled into a system with 10 switches so that any one of the 10 stereo amps could be switched into the circuit at the listeners direction. *(Figure 1)* Care was taken so that the level match between amplifiers was .25 dB and the frequency response of all amplifiers was within 1 dB from 20 Hz to 20 kHz. The third-octave pink-noise response at the listening position was flat, plus or minus 2 dB from 50 Hz to 10 kHz. A variety of music from different sources could be selected. These included tape, disk and digital multi-track. The controls for switching were located at the console mixing position.

Each participant was asked to mark the supplied ballot with the three best amplifiers, as well as the three worst from among the ten. Many taking part could discern no difference among the amps, most having taken considerable time A-B'ing one against the other at the console.

"From the ballots that were marked," according to UREI, "we had something of a shock when we tabulated the results and found that each amplifier had almost as many votes for it as against it. This seems to suggest that the personal tastes of human beings tend to average themselves out if the number is large enough. There are, however, too many complex variables at work here to go much farther than this statement. A safe conclusion might be, that for the majority of listeners, any one of the ten amplifiers would be a good match for the 813 loudspeaker system."

BEATRICE FOODS ANNOUNCES LETTER OF INTENT TO SELL JBL TO DR. SIDNEY HARMAN AND GROUP OF ASSOCIATES

Beatrice Foods Company has announced it has signed a letter of intent to sell James B. Lansing Sound, Inc., and several overseas distribution units of its stereo components company, Harman International Industries, Inc., to Dr. Sidney Harman and a group of associates. Dr. Harman is the founder of Harman International, which was purchased by Beatrice in August 1977.

The transaction, the terms of which were not announced, is subject to the approval of Beatrice's board of directors. The anticipated closing date is early next year. JBL produces loudspeakers for the home and sound systems for recording studios and other professional applications. The distribution units to be sold include operations in England, France, Germany, Belgium, Australia, and Japan.

Beatrice previously had announced its intention to sell the Harman Kardon unit of Harman International to Shin Shirasuna Electric Corporation. Beatrice retains ownership of the Harman Automative Mirror Division, as well as the Tannoy Group, a U.K.-based sound distribution company, and Ortofon Manufacturing, a Danish-based producer of cartridges and tonearms.

HARRISON TO INTRODUCE TOTALLY NEW CONSOLES

David Harrison, president of Harrison Systems, Inc., has announced that the company will begin delivery of a long-rumored, totally new, music recording console during the first quarter of 1980.

Harrison said, "Although sales of the eighteen-month-old, top-of-the-line 32C Series consoles have never been better, there has been an intensive engineering program at Harrison to develop advanced new concepts, both operationally and technically, for the eighties. A major development in this program is the concept of 'Distributed Control Intelligence' ("DCI").

for additional information circle no. 132

This involves placing an individual microcomputer in each input module of the console and in most of the other major modules."

Harrison Systems demonstrated "DCI" for the first time in its new, modular Post Production Series console at the May 1979 convention of the Audio Engineering Society in Los Angeles. The console displayed was configured as a four-operator, forty-eight-input, film post production re-recording console, containing seventy-seven individual microcomputers.

Subsequent to its public introduction, the console was delivered to Walt Disney Productions in Burbank. Late in June it was installed in their newly refurbished re-recording theater, where production is currently in progress on *The Black Hole*, Disney's latest motion picture undertaking.

According to Harrison, "The 'DCI' concept of placing an intelligent software-controlled microcomputer in each module offers the end-user many advantages over the older 'hardware logic-controlled' analog consoles. Included in these advantages are tremendously expanded automation opportunities, improved ergonomics (human engineering), better reliability, easier maintenance, and many side benefits such as noiseless switching on most functions."

Harrison continued, "One of the most signficiant advantages of 'DCI' is that the 'personality' or operational characteristics of the console are under the control of software (computer code) rather than hardware. This software control allows the console to be modified for unique applications by simple programming rather than laborious, often-irreversible hardware modifications. Additionally, many new and unique features can be incorporated with little additional expense when using the 'DCI' concept."

Harrison tells us that among the new features to be included in the new music recording console are automated panning, echo sends, group assignment, echo return, and automated insertion of patch points, filter, and equalization.

Harrison Systems plans to show the input module from the 'DCI' music console and a completely operational 'DCI' post production console at the fall convention of the AES to be held the weekend of November 2, at the Waldorf-Astoria in New York.

Although pricing has not been announced, Harrison has taken production reservations from several major clients, including Streeterville Studios in Chicago.

DIGITAL EDITING DEMONSTRATED AT 3M AUDIO SYMPOSIUM

Some two dozen recording studio executives recently witnessed the first demonstration of 3M's electronic digital editing system during a three-day audio symposium presented in St. Paul, Minnesota, at 3M headquarters.

The demonstration consisted of an assembly edit performed by what was described as a "simplified" editor. According to Bob Youngquist, research manager for 3M's Mincom Division, a deluxe automated system prototype with video display is still being refined and evaluated.

— *continued overleaf*

The edit was performed via a compact 20x10x10-inch console wired to two 32-track digital pre-mix recorders. A sequence of command buttons was punched to select a pair of edit points, preview them separately, precisely redefine these locations in terms of aural aesthetics, repeatedly audition the proposed assembly edit, then accomplish the noise-free edit by microprocessor-controlled copying onto the new master tape.

Plans calls for this editing system or a variation of it to be sold to studios following studio evaluations this fall. Pricing and delivery information will follow this phase.

On the subject of recording equipment, Bob Brown, marketing director for the Mincom Division, indicated the company plans to deliver six systems by the end of the year and produce at least several per month next year. Noting industry enthusiasm for the sound recreated by digital recording, he also predicted that between 30 and 40 per cent of the studios will be "into digital" in the next five years.

Primary host of the symposium, the company's Magnetic Audio/Video Products Division, provided attendees with an overview of its broad range of Scotch brand tapes and an update on the latest technical and product developments. This included a look at Scotch 265 digital audio mastering tape, developed for 3M's digital recorders, and Scotch Metafine audio cassettes, the first fine metal particle cassettes commercially introduced for a new generation of consumer recorders.

Following the presentations, the studio executives toured the St. Paul research facilities of the Magnetic Audio/Video Products Division and one of its tape manufacturing plants. They then convened at a 3M conference center to participate in an open exchange of information on a variety of business subjects.

NEW HOLLYWOOD BOWL SYSTEM FEATURES QUAD/EIGHT CONSOLE

Quad/Eight Electronics, the North Hollywood, California, based console manufacturer, delivered a large sound reinforcement/recording console to the Hollywood Bowl in May. The console features up to 160 microphone inputs, six electronic sub-masters, full VCA inputs on a basic Ventura Model 40 input by 40 output configuration.

In June, the Bowl's first major summer show was *The Playboy Jazz Festival* where the console found its first service. The Quad/Eight console works into a JBL speaker system. Local reviews in the *Los Angeles Times* have given raves to the greatly improved sound quality at California's most famous outdoor amphitheater.

OTARI MOVES TO NEW FACILITY

Effective September 24th, the new address for Otari Corporation is 1559 Industrial Road, San Carlos, California 94070. The new telephone number is (415) 592-8311.

The new facility contains 50% more office space and 100% more lab space. The new telephone number allows Otari to utilize a new, modern, expanded telephone system. What it all means is increased capability to serve you, according to Otari officials.

Otari wishes to extend a "thank you" for making this expansion necessary, and a wish that "we may all continue to grow and prosper."

AMERICAN AUDIO FIRM TRAINS SOVIET TECHNICIANS IN ADVANCE OF 1980 OLYMPIC GAMES

Mr. Max W. Scholfield, president of Crown International, Inc., has announced that nine Soviet technicians have successfully completed a special audio service and repair program provided by this American manufacturer of audio electronics.

Audio power amplifiers manufactured in Elkhart, Indiana, by Crown International will be used in the Moscow World Trade Center. The Trade Center, equipped with over 50 Crown amplifiers capable of delivering over 10,000 watts of audio power, is expected to play a vital role in the 1980 Summer Olympics.

Detailed classroom instruction and actual test-bench experience were provided on the Crown D-60, D-150A, DC-300A, and M-600 power amplifiers. Lecture content included design theory, trouble shooting techniques and field serivce procedures.

SID ZIMET JOINS AUDIOTECHNIQUES

Sid Zimet, founder of Audio by Zimet, Roslyn, New York, and co-founder of Sound Workshop, Hauppauge, New York, has joined the New York offices of Audiotechniques, Inc., a professional audio sales and engineering firm headquartered in Stamford, Connecticut.

"A more qualified man would be hard to find," says Hamilton Brosious, the company's president. "With this move, we have acquired not only one of the industry's top technicians, but also a real recording engineering pro."

Zimet's first assignment, according to Brosious, will be to expand the services of the firm's Audiotechniques Rentals division at 1619 Broadway, New York City, in the heart of the recording studio area. Rentals offers a unique service to both large and small studios and to musicians by providing, within the confines of Metropolitan New York, virtually any missing component in the time it takes a cab or messenger to reach its destination. Overnight air freight delivers needed equipment to neighboring states and as far West as California.

In his first moves toward expansion, Zimet has put into operation laboratory facilities for testing rental equipment berfore delivery and upon return; and an increased inventory covering a full range of components, including new equipment and the latest models of existing stock from 24-track tape recorders to microphones.

Sid Zimet draws on more than thirty years experience in professional audio. His career is, in effect, the story of the industry, and spans from his start as instructor of basic electronics in the U.S. Air Force to his present-day knowledge of the most sophisticated equipment in the field.

SONY UHF WIRELESS MIKE MAKE U.S. TELEVISION DEBUT

Sony's WRT-57 UHF wireless microphone recently made its U.S. television debut on the Jerry Lewis Muscular Dystrophy Telethon broadcast live nationwide from the Sahara Hotel, Las Vegas. The microphone was used by telethon host Lewis during the 21-hour-long show.

— continued on page 190 . . .

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Sony Wireless Microphones

. . . continued from page 187 —

"This marks a significant breakthrough in TV broadcasting," says Nick Morris, general manager of Sony's Professional Audio Division, which markets professional audio products. "The WRT-57 mike, about 50 per cent smaller than conventional hand-held wireless microphones, is the smallest hand-held mike in the industry. The electret condenser mike weighs 190 g (7 oz.), is 17 cm (6.8 inches) long and requires only 16 cm (6.4 inch) antenna wire.

"Until the telethon," Morris points out, "the mike had only been used in Las Vegas shows."

The Sony mike system, consisting of mike, tuners, antennas and diversity unit, was selected for use on the telethon by sound engineer Pete SanFillipo, of Western Media Entertainment.

According to SanFillipo, "The mike, with uni-directional characteristics, rejects noise from the sides and fights feedback for exceptional sound quality. In addition, the mike has less external noise interference than low band systems because it uses the UHF band (947 - 952 MHz — 15 separate channels are available)."

SanFillipo feels the mike "looks great" on TV. "Because of its smaller size and shorter antenna wire, the mike looks neater than conventional wireless mikes, and is easier for performers to use. The system is also very easy to set up."

"The WRT-57 mike," points out Morris, "features miniature helical resonator filters, plus hybrid ICs developed by Sony. With a frequency response of 50 - 15,000 Hz, the mike has a wide dynamic range that provides greater linearity than other wireless systems because the system uses no limiting or other noise reduction systems to accomplish this wide dynamic range."

Morris adds that since the UHF band requires only a 16 cm antenna wire, greater signal emission efficiency is attained. "And because transmission power for the WRT-57 is decreased, the unit has long battery life."

— Book Review —

HOW TO BUILD A SMALL BUDGET RECORDING STUDIO FROM SCRATCH . . .with 12 tested designs.
by F. Alton Everest

This unique volume contains all the background data needed to design, construct, and operate a budget recording studio . . . plus 12 plans for acoustical studios that can produce audio, radio, audiovisual, film, and television program material. The 12 studios featured include a budget audiovisual recording studio, a studio built in a residence, a small studio for instruction and campus radio, a small ad agency studio for audiovisuals and radio jingles, a multi-track studio built in a two-car garage, a radio program production studio, studios for commercial radio stations, a single control room for two studios, a television mini-studio, a television and multi-track studio, a film review theatre, and multiple studios. Among the items fully discussed in the designs are floor plans, control rooms, walls and ceiling treatment, reverberation time, air conditioning, observation windows, room proportions, acoustical treatment, noise considerations, etc.

The emphasis is on budget studios and the efficient production of material on a day-to-day, routine basis. The author describes all the proceedures, the equipment needed, and the likely costs.

F. Alton Everest is an acoustics consultant in Whittier, California, and is a member of several audio engineering societies. He has been involved in TV broadcasting since 1936, and has written several audio and broadcasting books including, The Handbook Of Multi-Track Recording.

How To Build a Samll Budget Recording Studio From Scratch . . . with 12 tested designs
by F. Alton Everest

Tab Books	326 pages
Soft Cover	$8.95

HOW TO BUILD A SMALL BUDGET RECORDING STUDIO FROM SCRATCH is available from R-e/p Books. Send check or money order for $8.95 (postage paid) to: R-e/p Books, P.O. Box 2449, Hollywood, CA 90028.

VISION-SOUND PROFESSIONAL AUDIO DEBUT

Audio consultant Michael Salafia has announced the formation of Vision-Sound Professional Audio, Inc., a new audio consulting firm and dealership. Vision-Sound will specialize in equipping recording

DOCTOR ROCK **by Petach & Leiendecker**

"I'M NOT SURE, BUT I THINK THE SPEAKERS ARE OUT OF PHASE."

For the last four years you have heard about the innovative design and technical superiority of Harrison consoles. In no place has the Harrison "no compromise" philosophy been more evident than in the Harrison 32C series recording console. This philosophy pays off for you in the knowledge and confidence that your Harrison 32C series console is the most reliable console built anywhere in the world. What does this innovative design, technical superiority and reliability mean to you

success

The Harrison 32C series console has become the industry standard with installations in every major recording center in the world. True, a Harrison console initially costs more than most other consoles, but in the long run this investment in quality pays off in less down time, increased operating efficiency, and a superior recorded product.
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success

Success is full-time utilization of your recording facility, and that translates into profit for you. Harrison consoles are used in major recording studios around the world. Many of these studios have purchased a second, and even a third Harrison console. Harrison 32 series consoles have been instrumental in the production of Grammy Award winning recordings for "Best Engineered Recording (non-classical)" for the last two years in a row, and you know what that means

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for additional information circle no. 137

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studio facilities and in providing sophisticated home recording systems for music industry pros.

Salafia, designer of elaborate audio systems for such music business personalities as Charles Koppleman and guitarist John McLaughlin, reports that Vision-Sound will represent a full range of state-of-the-art equipment, including Harrison consoles, Studer recorders, Calrec Sound Field Microphones, and a full complement of peripheral signal processing hardware. Vision-Sound is the exclusive East Coast representative for Neotek Recording Consoles, and the firm's new headquarters are being acoustically treated to provide clients with the opportunity to audit equipment functions in a studio environment.

Salafia, who previously served for two years as a sales representative for Audiotechniques, has established Vision-Sound to provide the recording industry with a "fresh approach" to the problems inherent in selecting appropriate audio systems.

In announcing the formation of Vision-Sound, Michael Salafia remarked, "We are experienced enough to be on top of new audio developments, and free enough to keep pace with them. The recording studio industry is experiencing a period of dramatic growth, and it is critical that studio owner/builders have as wide a range of quality options available to them as possible."

Vision-Sound Professional Audio, Inc., is located at 110 Grand Avenue, Englewood Cliffs, New Jersey. The phone number is (201) 871-4101.

ORBAN IMPROVES MODEL 111B DUAL SPRING REVERB

Orban Associates, Inc., has announced a significant product improvement to their Model 111B Dual Spring Reverb. This compact, professional-quality reverb is now being delivered with six springs per channel instead of four.

Based on component advances, this improvement provides lower flutter, higher echo density, and a smoother, more natural sound according to designer Bob Orban. It is anticipated that this sonic improvement will further increase the 111B's acceptance among professional users as the reverb of choice.

All of the standard Orban signal processing is retained, including the exclusive "floating threshold" limiter to minimize "spring twang" and a versatile equalizer with quasi-parametric midrange and shelving bass sections. This processing ideally complements the new six-spring arrays.

Orban is providing this product improvement at no increase in cost.

for additional information circle no. 138

WHY AUTO-SET

Auto-Set is part of a revolution that's happening right now. A micro-electronics revolution. Some folks call these times the "micro-computer age" because this decade has brought us revolutionary advances in digital control and processing technologies. These developments confront all of us with the question of how to best use our new-found computer power. Computer scientists call our relationship with computers the "man-machine interface," and have given us some terms to talk about.

For example, some computer systems are called "transparent" because the user is not usually aware of the presence of an information processing device . . . a it's a "black box." Systems of this type try to appear unobtrusive . . . they don't have display screens, or data cartridges, or keyboards. They are usually hidden somewhere under the "skin" of the host equipment, and typically are available to perform *only* those functions the designer has built-in, with no variations.

Other computers are called "interactive." Here, the user is more aware of the processing device because he can communicate with it. Systems of this type are intended to be more helpful to the user, and generally meet this objective. But, there are a few interactive systems around whose designer was probably more familiar with computers than with the user and his application. The user is sometimes required to function more like a computer programmer than a recording engineer.

Auto-Set is a computer system whose interactive capabilities were developed to help you create the best sounding products possible. You don't have to *know* much about computers to use Auto-Set...it tells you how. And you don't have to *do* much to use Auto-Set because its programs are menu-based . . . its memory helps your memory.

Auto-Set's menus and control commands are designed to promote a "learn-by-using" environment. You don't have to talk "computerese," just key-in simple command codes. For example, the letter "H" (the command code for HELP) displays a complete command reference menu . . . it's like having a built-in teacher.

Auto-Set is effective. Its applications design is geared to the real-world situations found in the modern recording studio. Its applications feature data manipulation functions and mix situation logic to introduce a new dimension of *creative capability* to the recording studio . . . it's a new creative tool.

Auto-Set is flexible. It's two computers in one, with different ways to store automation data. In real-time, like a motion-picture, on a spare channel of your multi-track audio tape; or as pre-set mix "snapshots" stored on special certified data cartridges which plug into the front of the unit.

Auto-Set is convenient. Whatever's happening with Auto-Set is in one place, right before your eyes, not spread out across all the I/O modules. And, if you want to make Auto-Set operate transparently, like some other automation programmers, a single keystroke is all that's required. There are no demands from Auto-Set . . . just the capabilty to handle your most complicated mix situations. Auto-Set . . . with single-point interaction, or fully transparent . . . it's the automation programmer that gives you the options.

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fact: you can choose your microphone to enhance your productions.

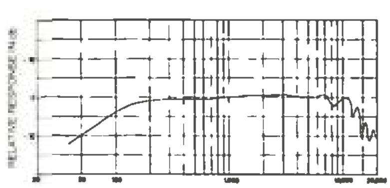
Shure makes microphones for every imaginable use.
Like musical instruments, each different type of Shure microphone has a distinctive "sound," or physical characteristic that optimizes it for particular applications, voices, or effects.
Take, for example, the Shure SM58 and SM59 microphones:

SM59

Mellow, smooth, silent...

The SM59 is a relatively new, dynamic cardioid microphone. Yet it is already widely accepted as a standard for distinguished studio productions. In fact, you'll often see it on TV . . . especially on musical shows where perfection of sound quality is a major consideration. This revolutionary cardioid microphone has an exceptionally flat frequency response and neutral sound that reproduces exactly what it hears. It's designed to give good bass response when miking at a distance. Remarkably rugged — it's built to shrug off rough handling. And, it is superb in rejecting mechanical stand noise such as floor and desk vibrations because of a unique, patented built-in shock mount. It also features a special hum-bucking coil for superior noise reduction!

Some like it essentially flat...

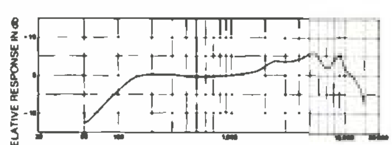


SM58

Crisp, bright "abuse proof"

Probably the most widely used on-stage, hand-held cardioid dynamic microphone. The SM58 dynamic microphone is preferred for its punch in live vocal applications . . . especially where close-up miking is important. It is THE world-standard professional stage microphone with the distinctive Shure upper mid-range presence peak for an intelligible, lively sound. World-renowned for its ability to withstand the kind of abuse that would destroy many other microphones. Designed to minimize the boominess you'd expect from close miking. Rugged, efficient spherical windscreen eliminates pops. Lightweight (15 ounces!) hand-sized. The first choice among rock, pop, R & B, country, gospel, and jazz vocalists.

...some like a "presence" peak.



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