# db

THE SOUND ENGINEERING MAGAZINE

SEPTEMBER 1973 $1.00

DAVID R BLOCK M3
APT 303
499 S SNELLING
ST PAUL MN 55116

Photographed at Record Plant, Sausilito, Calif.

# The Model 2824...one of a series of great consoles.

In aiming for creative expression, producers and musicians make great demands on a console and its operator. The key to a successful session is the ability to satisfy these demands quickly and easily.

A *great* console...the MODEL 2824 for example ...has the built-in flexibility and versatility to be responsive to the wide range of dynamic requirements encountered in studio applications. The MODEL 2824 provides up to 28 inputs, each with the Model 550 three knob equalizer, four echo sends, three position high-pass filter, channel phase reverse, switchable mike-pad and gain control, panner, two independent cue feeds, a Program direct output feed, and input solo. There are sixteen metered mixdown busses, with optional 24 track metering and monitoring, simultaneous quad and stereo facilities, overdub facilities, four separate headset or cue busses with master level controls, and a complete patch bay with jacks for over 350 separate circuits.

The exceptional quality and reliability of its components means a console that consistently delivers fine performance under the most demanding conditions.

For a look at the MODEL 2824 and a complete description of its features, contact us today.

Meanwhile, start thinking of the great things you'll be able to do with it.

**AUTOMATED PROCESSES, INC.**
**80 MARCUS DRIVE, MELVILLE, NEW YORK 11746 · 516-694-9212**

West of Rockies: WESTLAKE AUDIO
Los Angeles, Calif.

Europe: 3M FRANCE
Paris, France

*Circle 10 on Reader Service Card*

# COMING NEXT MONTH

• The European concept of training recording engineers is radically different from the haphazard way in which one gets to sit at a console in the United States and elsewhere. Noted English writer John Borwick has submitted a paper WHAT IS A TONMEISTER? that appeared originally in a British publication. You will find it illuminating in many ways.

db VISITS—CARIBOU RANCH STUDIOS. Not every successful studio has to be in the middle of a city—or even close by. John Woram visited Caribou recently (it's in Colorado) and has this story to tell. Caribou must be doing something right—they've already recorded an album for Blood, Sweat, and Tears that is out on Columbia Records.

The audio professional has little choice when it comes to saving money if he wants quality components. We've built two new kits made by Gately Electronics that seem to satisfy this impossible dream and will report on their construction and the quality of the materials that go into them.

And there will be our usual contributors: Norman H. Crowhurst, Martin Dickstein, and John Woram. Coming in **db, The Sound Engineering Magazine.**

# ABOUT THE COVER

• Every now and then we ask our art director Bob Laurie to come up with something interesting in response to a theme we give him. We told him that this issue was going to be distributed at the AES Convention celebrating their 25th year. This art dates from the 19th century and was originally used (probably) to decorate a music score for publication. We commend everyone to visit the Convention at the Waldorf-Astoria in New York, information on which is to be found on page 26.


# db

THE SOUND ENGINEERING MAGAZINE

SEPTEMBER 1973 VOLUME 7, NUMBER 9





**db** is listed in **Current Contents: Engineering and Technology,**





db, the Sound Engineering Magazine is published monthly by Sagamore Publishing Company, Inc. Entire contents copyright © 1973 by Sagamore Publishing Co., Inc., 980 Old Country Road, Plainview, L.I., N.Y. 11803. Telephone (516) 433 6530. db is published for those individuals and firms in professional audio-recording, broadcast, audio-visual, sound reinforcement, consultants, video recording, film sound, etc. Application should be made on the subscription form in the rear of each issue. Subscriptions are $6.00 per year ($12.00 per year outside U. S. Possessions, Canada, and Mexico) in U. S. funds. Single copies are $1.00 each. Controlled Circulation postage paid at Harrisburg, Pa. 17105. Editorial, Publishing, and Sales Offices: 980 Old Country Road, Plainview, New York 11803. Postmaster: Form 3579 should be sent to above address.

# advertisers index




db

THE SOUND ENGINEERING MAGAZINE

SALES OFFICES

**New York**
980 Old Country Road
Plainview, N.Y. 11803
516-433-6530

**Dallas**
Roy McDonald Associates, Inc.
Stemmons Tower West
Suite 714
Dallas, Texas 75207
214-637-2444

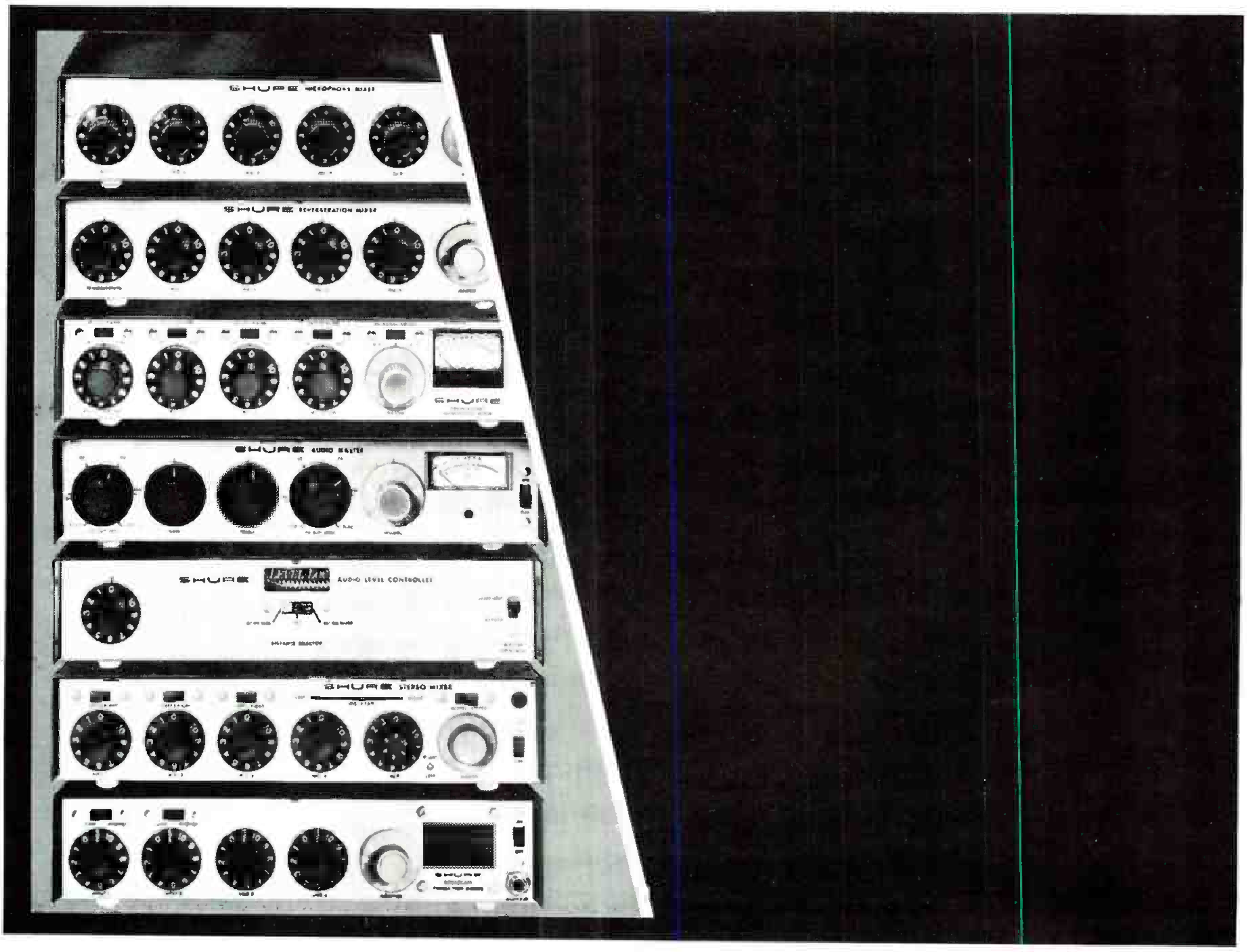
**Denver**
Roy McDonald Associates, Inc.
3540 South Poplar Street
Denver, Colorado 80237
303-758-3325

**Houston**
Roy McDonald Associates, Inc.
3130 Southwest Freeway
Houston, Texas 77006
713-529-6711

**Los Angeles**
Roy McDonald Associates, Inc.
500 S. Virgil
Suite 360
Los Angeles, California 90020
213-381-6106

**Portland**
Roy McDonald Associates, Inc.
2035 S. W. 58th Avenue
Portland, Oregon 97221
503-292-8521

**San Francisco**
Roy McDonald Associates, Inc.
Baybridge Office Plaza, Suite 265
5801 Christie Avenue
Emeryville, California 94608
415-653-2122



# The Magnificent Seven

We've been hearing unsolicited rave reviews from soundmen across the country concerning our seven ingeniously versatile problem-solving audio control components (1) *M68 Microphone Mixer,* vanguard of the low-cost, high-performance portable mixers; (2) *M68-RM Mixer,* with built-in reverb for vocalists and special effects; (3) *M67 Mixer,* the trail-blazing low-cost professional mixer; (4) *M63 Audio Control Center,* that gives you variable response shaping; (5) *M62V Level-Loc,* the audio level controller that automatically limits output level; (6) *M688 Stereo Mixer,* made to order for stereo recording and audio-visual work; and finally, (7) *M675 Broadcast Production Master,* that teams up with our M67 to give a complete broadcast production console (with cuing) for about $325. Write for the new Shure Circuitry catalog that shows them all:

**Shure Brothers Inc.**
**222 Hartrey Ave., Evanston, Ill. 60204**

SHURE

*Circle 12 on Reader Service Card*

Norman H. Crowhurst

# THEORY AND PRACTICE

● Back in August, 1969 when I first started digressing from those aspects of theory and practice that related strictly to the technicalities of audio, I commented on the compartmented thinking that seems to have overtaken all of us, largely as an outgrowth of the way educational curricula is compartmented during our schooling. Since then, more and more, I am impressed with a sense of what today's younger generation calls "getting it together."

This basic question has actually been on my mind since the day I took my final exam that graduated me from engineering school. The head of the department asked me to come back the next year to teach, but I had already accepted a job as a chief engineer. Subsequently, however, I was able to combine practical engineering with teaching, holding two lectureships at London's engineering colleges. I found out how well these jobs can complement each other. In those days, most electronic engineering was very sloppy; few written records were kept and how new products were put together was recorded only in the heads of the people who had worked on them. And on the educational side, there was nothing in electronics planned systematically; in fact, the electrical courses available would have been suitable for helping Noah install an electrical system in the ark!

Applying teaching disciplines to engineering resulted in my finding that writing things down did far more than merely providing a record: often it led to discoveries we would otherwise have missed. In course of time—not very long, as I look back—we shortened up our methods of attacking research and development problems so that we could get results (always at a premium in private enterprise) much more rapidly.

On the educational side, my work in engineering, and my contacts with other companies in the field benefited my work at the colleges. Beside enabling me to update curriculum content and to devise new approaches more relevant to the requirements of a rapidly changing world, close working contact with students and a knowledge of the work environments of various companies enabled me to help the students to find suitable jobs when they graduated.

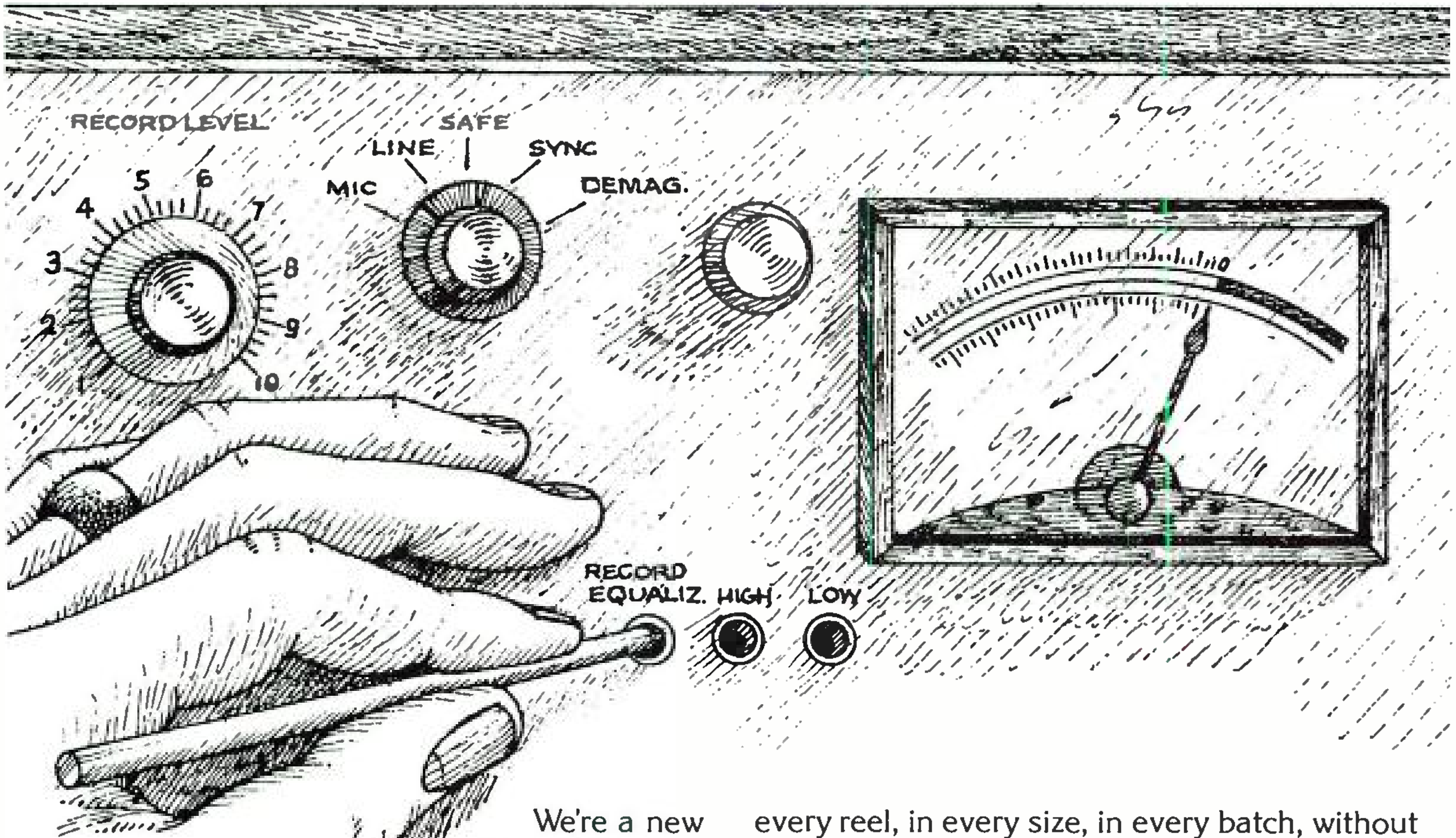
On that issue, it still appalls me to observe the activity of placement agencies and personnel departments, who measure suitability of candidates for job openings by the content of pieces of paper that have no relevance to whether the person will fit well into the environment, humanly speaking.

But relative to what I started to say, this was really the beginning of my own "getting it together" program. Since then I have had many successes in this area—articles, books, and whole files full of letters from grateful readers. So when I started branching out in this column, discussing educational matters as well as technical and engineering aspects, this was really continuing something that had its beginnings much earlier.

About a year ago, I was working on a very promising math program funded by the government under contract to a regional educational laboratory. But bureaucrats and politicians being what they are, the fact that it turned kids on, enabling them to learn much faster and more effectively, in no way influenced their decision about continuing to fund the work until the job was finished—unless it was, as John Woram recently suggested, a negative influence, based on a desire to promote incompetence and penalize competence! In any event, the project was discontinued, much to the disgust of everyone associated with it.

A little before that happened, I had learned about a relatively new organization in education, called Educational Research Associates, based in Portland, Oregon (97205). Curiously, I learned about it, not because of their educational philosophy, but because the president read a letter of mine published in the paper, suggesting that schools could be made more economical, and give more education, for less money.

# We offer you better quality masters, but you'll have to work for them.




We're a new mastering tape formulation, so the limits of older tapes don't apply to us. Initially, our extra quality needs some getting used to. To upgrade your original recordings, you'll have to adjust your equalization and bias controls to our specifications. That's not much work, but it's necessary. And it's worth it.

When you play back your first reel of Maxell mastering tape you'll notice how clean it works. Our tape doesn't use a carbon backing so there's no messy carbon shedding. Sheddings cause noise, thumping and interference. On the reel, carbon can also transfer to the recording surface and cause dropouts.

Because of the care we take in the manufacturing process, you won't find any pinholes in our oxide coating. No pinholes and no shedding mean Maxell mastering tape is virtually 100% free from dropouts.

And you get Maxell's low noise level, wide range and freedom from distortion with every reel, in every size, in every batch, without fail. That's a statement about quality performance you won't find other manufacturers making anymore. We're eager to draw your attention to the edge-to-edge, end-to-end quality you'll find in Maxell mastering tape.

Send in the coupon for documented evidence about Maxell tape performance. Then think about the repeat sessions Maxell quality will eliminate. That's a lot of work you won't have to do.

Send to: Maxell Corporation of America
130 W. Commercial Avenue **Dept. 9DM**
Moonachie, N.J. 07074

Gentlemen:
Send me technical information and tell me how I can get Maxell mastering tape.

name/title ____________________

company ____________________

address ____________________

city ____________ state ____________ zip ______

**maxell.**
The answer to all your tape needs.

*Circle 31 on Reader Service Card*




Educational Research Associates turned out to be an organization that operates entirely without government funding. This, in itself, as I have found since then, produces a negative reaction with most teachers and administrators, their view being that if it is not paid for with public funds, it cannot be any good!

Those who are bold enough to try the products anyway are soon "sold." Instead of applying the methods insisted upon by institutions that accept government funding, which almost guarantees failure as far as useful education is concerned, ERA applies the economic principles that any good business would use. First they must produce the best products they know how, as economically as possible, and then improve them based on experience gained in the process. Pretty obvious to you and me, but quite foreign to most educators, I am afraid.

They specialize in "individualized instruction." In fact, I believe they started it. I first heard about it while I was with Teaching Research, working for government money, and it seemed by the reports I received then, that individualized instruction was developed with government funds. But the facts are that ERA was developing materials several years before any government sponsored "work" started copying the idea.

Carl Salser, executive director of ERA and I found we have a lot in common. It was only a matter of finding how we could best work together. Right now I have almost finished the first part of a course on basic electronics, which applies the principles I have been learning over all these years. The credit for much that I am able to put into this course belongs to many of you readers: to the questions you ask, to the discussions we have had at various times.

And once again, putting it all together serves its purpose. For example, an early outcome of various efforts to analyze circuit behavior resulted in my developing a simpler alternative to Fourier analysis and other analytical problems, about which I then wrote an article, "The Finite Approach to the Infinite."

Since then, other applications for the same kind of approach led me to extend it, so that now, in this new course, I can make the whole thing much easier for the student—and much more meaningful. In one lesson I make a comparison, from which I will borrow an illustration (FIGURE 1). This compares the first few terms of the Fourier series with summation to the same number of

# "Meet our new Recorder/Reproducer. You'll call it fantastic. We call it the 280-B"

What's so great about it? First off, it's really up to the state-of-the-art. In electronics, control logic and architecture. Everything's changed but the transport. (Why mess with that? It's field proven.)

Here's what you get:

S/N ratios at a consistent 72 dB at mastering speeds on full track, .25 inch tapes. Improved headroom and increased record level for maximum signal utilizing high output tapes.

Band widths are just about flat at ±2dB, 30 Hz to 18 KHz. Tape handling is incredible. Our new motion sensing system (OPTAC™) gets rid of deck top sensor mechanisms. Internal logic lets you select a new mode and activate it without having to hit the STOP button. And enter and leave RECORD while the transport's in PLAY.

Maintenance is a snap, too. We got rid of all the spaghetti and replaced it with plug in IC's. And TTL logic for high reliability and low power consumption. Test and adjustments are made without bending all over the place. Individual channel modules come out easily. And all electronics are in slide out drawers.

Oh yes. Selectable synchronization with extended response. Standard on 2 and 4 channel machines.

*You can get all the specs (or a machine) by writing or phoning (415) 968-8389 TLX 345524 475 Ellis Street, Mountain View, California 94040*

**Scully/Metrotech**

**Recording Divisions of Dictaphone**

Scully and OPTAC are trade marks of Dictaphone Corporation. Rye, New York.

*Circle 43 on Reader Service Card*

terms of what I call the successive finite series, which can be a real problem-solving tool.

The big difference is that the Fourier series "never gets there." However many terms you add, the resultant curve still swings back and forth across the ultimate shape being approached. True the deviations get smaller, but since their frequency goes up, they also get sharper. However many terms you take, cutting them off will always leave something that looks anything but straight-line in its shaping.

Long before I developed a simple mathematical way to handle it, I realized that what really happens—the practical side again—is that the terms drop off gradually—a slow high end roll off. But there was no way of showing that with Fourier. If the right kind of roll off is used, the shape can come almost as close as you want it to, to the ultimate shape designated—square, sawtooth, triangular, etc.

This was another instance of putting theory and practice together. An infinite series, postulated by the Fourier approach, is not practical, because no electronic system can produce, or reproduce, frequencies out to infinity: there is always a top limit. We know too, that this top limit is not sudden. But from there on, in my head, the thing got a bit vague. I wanted answers to questions like "How sharp can I make the actual shape, by putting the roll-off where?"

For some time I labored and kept coming up with n-order equations that were not easy to solve. Finally I came in "the back way," as one so often does in mathematics (usually with help from someone who already went that path), and found that the easy way is to work from known solutions, to which all you have to do is to substitute in to get the coefficients.

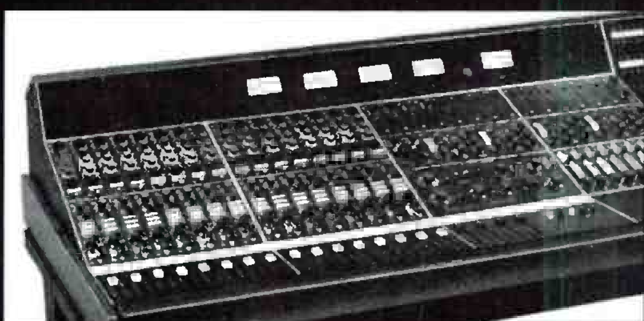
For the square wave, you use a first derivative of the series, and find a set of roots, all of which concentrate at the 90° point for zero slope, and thus get a maximal flatness of top and bottom. Substitution gives the coefficients of the finite series to replace the Fourier infinite set.

For the sawtooth, second derivative is used. One slope reversal goes at the zero point, and all the rest go at the 180° point, for maximal flatness of the downward slope. For the triangular waveform there are several approaches. The double pulse, shown here, is similar to the single pulse we discussed in an earlier issue, but symmetrical.

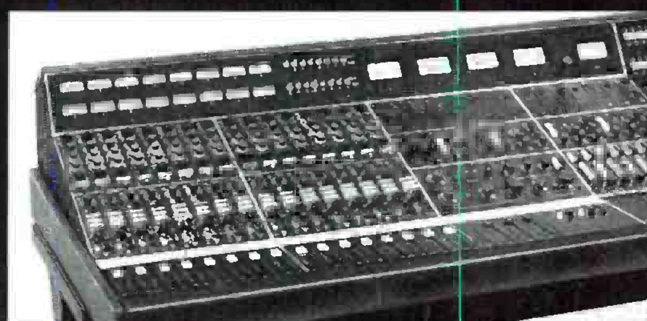
The further the series goes, the

# Neve presents the 3-in-1 console.

An unbeatable 8-track console.



A deluxe quadraphonic mixdown console.



A great 16-track console for small studio budgets.

It's the Neve 8014. Take your choice of three different meter configurations (shown above). Whether you need a 4-track or 8-track or economy 16-track console, the Neve 8014 will meet your needs to perfection. If you wish, start small and expand later. This versatile console can "grow" as your studio grows.

Fully fitted, the 8014 features 16 input channels; 4 Pgm, 2 Rev and 2 Cue mixing buses; 8-track monitoring and metering; 4 limiter/compressors; talkback and slating; test oscillator; and solo function. Delivery is almost immediate. As in all Neve Consoles, total harmonic distortion is guaranteed to be less than 0.075%.

The sound of Neve is worldwide. Our audio control consoles are used for music recording, film production and broadcasting in studios in 40 countries. Call or write us for more facts. Chances are, we can help you record some new highs in profits.

**Rupert Neve Incorporated,** Berkshire Industrial Park, Bethel, Conn. 06801. (203) 744-6230
**Hollywood:** Suite 616, 1800 N. Highland Ave., Hollywood, Ca. 90028. (213) 465-4822
**Nashville:** Telephone (615) 255-4315
**Rupert Neve of Canada, Ltd.,** 7528 Bath Rd., Malton, Toronto, Ontario, Canada. (416) 677-6611
**Rupert Neve & Co., Ltd.,** Cambridge House, Melbourn, Royston, Herts, England.

**Neve**

*Circle 16 on Reader Service Card*

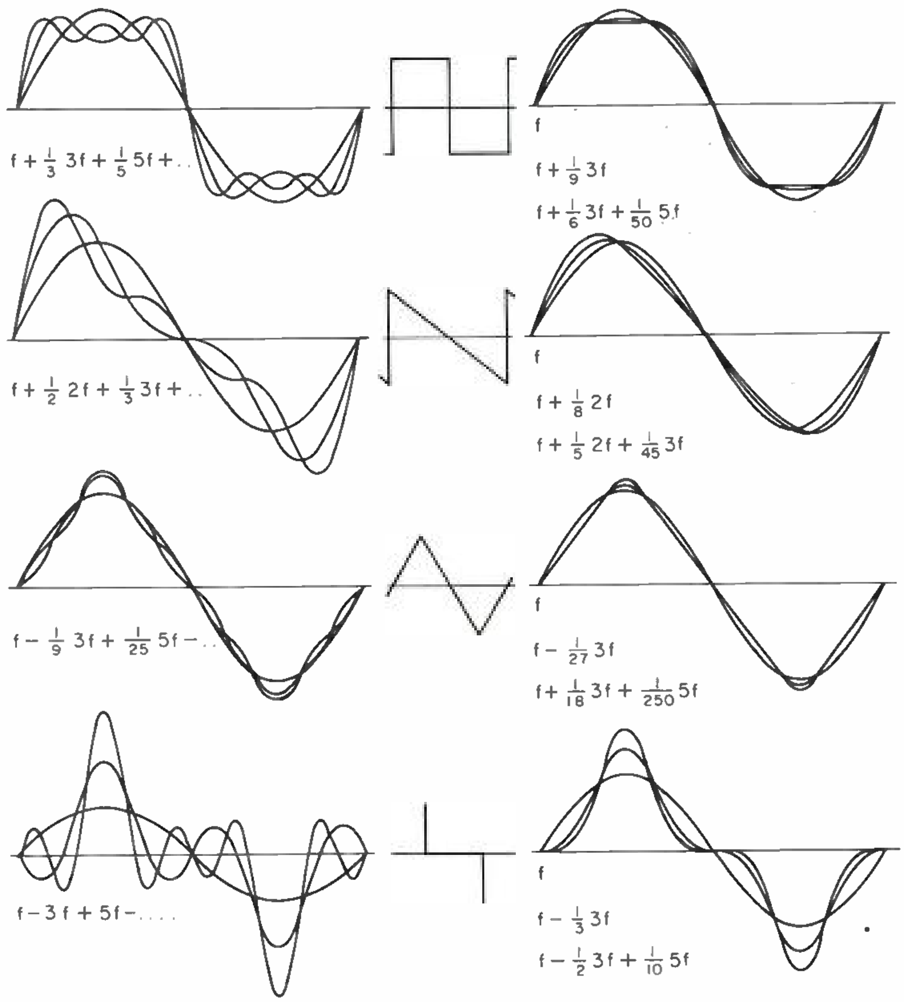


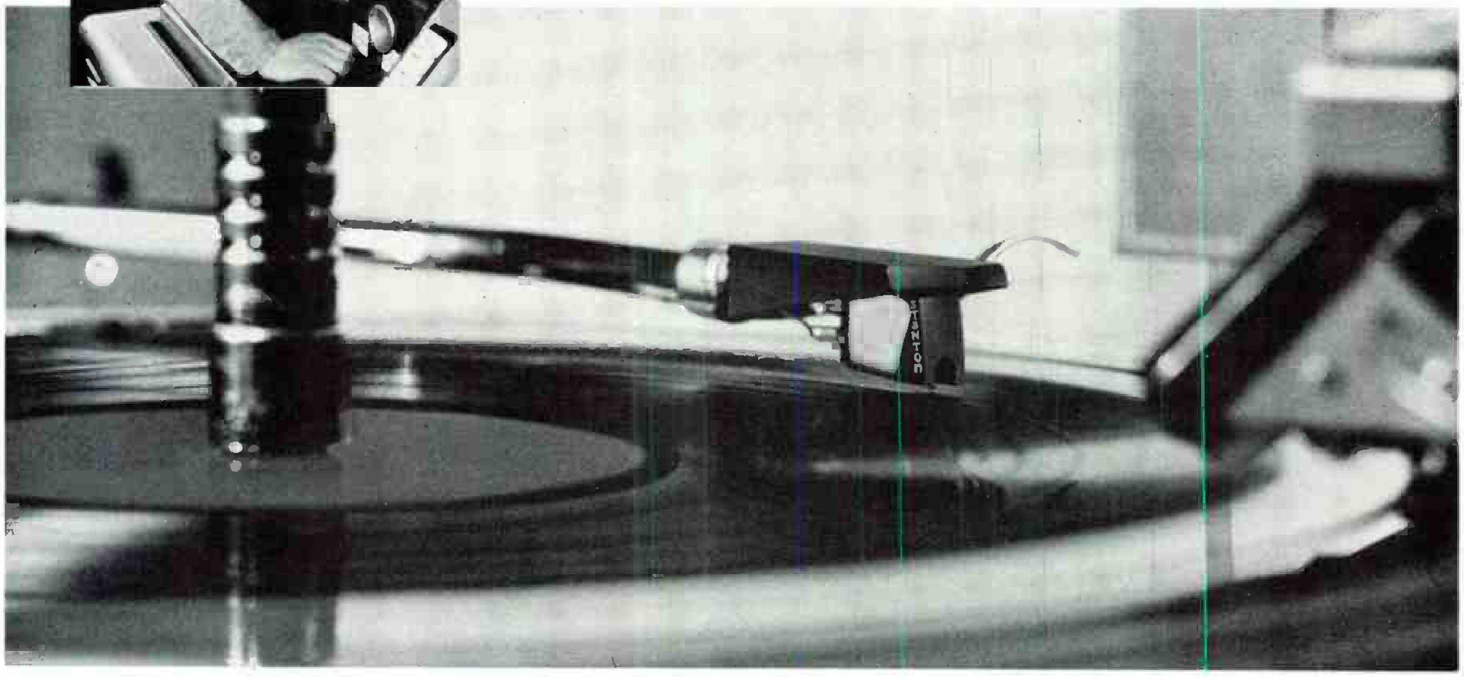

*Figure 1. Part of a page from the new course discussed in the text.*

more closely the early coefficients match the Fourier set. But they start falling off long before you reach the last term of whatever number of terms you choose to use.

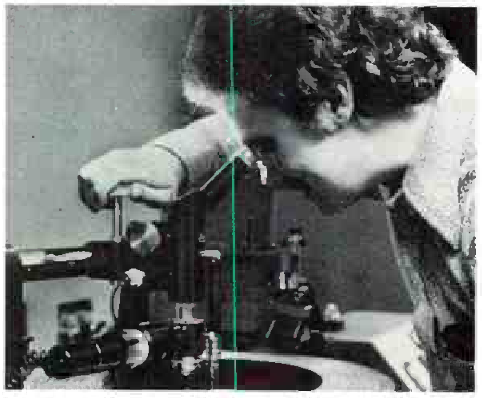
Although this is an easier approach to use than Fourier, running any considerable number of terms through it can get quite protracted. This, coupled with other, somewhat similar, activities, have led me to start work on a whole new approach to learning what has always been called mathematics. Because of what that word has come to mean to most people, I will call it "problem solving arts."

In a three-year course on problem solving arts, of which a student may take as much or little as he or she chooses, he can start as a dropout

# Stanton.
## Benchmark for an Industry.



## Stanton's 681 Series is the Calibration Standard to recording engineers such as Robert Ludwig.



Whatever the requirements for recording and playback, Stanton's Series 681 cartridges are the Calibration Standard. And there is a 681 model engineered specifically for each of these critical applications. That's why Stanton is truly the Benchmark for the industry.

**The Stanton 681A – For Cutting Head Calibration.** With Stanton's Model 681A, cutting heads can be accurately calibrated with the cartridge, for it has been primarily designed as a calibration standard in recording system checkouts. Frequency response is factory calibrated to the most rigid tolerances and the flattest possible response is assured for precise alignment of recording channels.

**The Stanton 681EE–for Critical Listening.** Stanton's Model 681EE is designed for low-distortion tracking with minimum stylus force, regardless of the recorded velocity or the distance of the groove from the disc center. High compliance, low mass and low pressure assure perfect safety even on irreplaceable records.

All Stanton Calibration Standard cartridges are guaranteed to meet the specifications with exacting limits. Their warranty comes packed with each unit – the calibration test results for that individual cartridge.

*For complete information and specifications write Stanton Magnetics, Inc., Terminal Drive, Plainview, L.I., New York.*

STANTON

*All* Stanton cartridges are designed for use with *all* two- and four-channel matrix derived compatible systems.

*Circle 22 on Reader Service Card*

from existing math courses with problems in adding, subtracting and multiplying or dividing. Instead of using those artificial approaches, he will start by looking at problems people have to solve, using progressively easier ways to solve harder problems.

The first year will pursue this through the kinds of problems for which operations that mathematics knows as counting, adding, etc., up to measurements with dimensions, areas, volumes, and graphing, but all related to problems that need to be solved, developing necessary abstractions from necessary uses. Important disciplines often not emphasized in conventional math courses, like being sure you always get a correct answer, finding your mistakes, and so forth, are stressed. Also, how to use what you know how to do, to find new ways of doing something else.

The second year chunks into the body of math, as now taught, but much more directly, still using the method of finding easier ways to tackle more difficult problems. For example, the nomenclature of algebra is introduced only after the methods are already being used to short-cut solutions of more difficult problems, so it makes sense, which it didn't to most who learned algebra the way it is still taught, even in "modern math" courses.

The third year, which gets really interesting, contains some math which may have been taught at various college levels, but in a totally new organization, which makes it really easy to digest. It starts by tracing man's development of calculators or computers from things like the abacus or multiplication table, to modern analog and digital types. It concentrates on how people, who are able to think and reason, utilize such machines, by suitably programming them to do what they want. Of course, it leaves computer programming to courses that specialize in that. What this shows is that everything so far has followed the same pattern of development, and how to make that whole pattern serve a problem-solving use.

A man who heads a head-start program in these parts was so impressed by the outline, that he believes some of his pre-school tots could master the course—all 3 years of it, by the time they are ready for first grade! Quite a thought, I'll admit, but I'm wondering what would be the effect of an onslaught of first graders demanding to be taught advanced calculus! Maybe that is what the system needs, to wake it up. ■


**Due to conditions beyond our control you'll have to pay a little more to get the best.**

We don't like to think about the decreasing value of the dollar.

Or about the fact that, as a result, it now takes more dollars to buy Neumann microphones—and virtually every other product produced overseas.

And we know you don't like to think about it, either.

But we're sure that you wouldn't want us to maintain price simply by reducing quality to match the devalued dollar.

People don't buy Neumann to save money.

They buy Neumann microphones for their 45 years of unquestioned dependability. For their unqualified technical superiority. And for their distinctive sound. That's the way it's always been. And we think that's the way you want it to continue.

So, count on us to keep Neumann's price tag as low as we possibly can during the currency crisis. But never so low that we have to devalue Neumann quality.

GOTHAM
AUDIO CORPORATION

*Headquarters*: 741 Washington Street,
New York, NY 10014 (212) 741-7411
*West Coast Sales Office*: 1710 N. LaBrea Ave.,
Hollywood, CA 90046 (213) 874-4444

John M. Woram

# THE SYNC TRACK

● As usual, I am behind schedule in answering letters sent in by readers. And as usual, I have all sorts of elaborate excuses. I thought about writing a colum of reasons for not answering letters immediately, but the editor didn't seem to feel it would be worth printing. He's probably right. [Editors are *always* right. Ed.]

However, some letters are all but unanswerable—even by writers who are not lazy. For example, ". . . if I study such and such, what sort of job can I expect to get?" Studying such and such will not guarantee anyone anything. If you're looking for guarantees, try the army. If you can make it through basic training, you'll have a career guaranteed for as long as you like. Outside of that, there are few guarantees in life, and certainly none in this business. Success comes from some indefinable mixture of education, personality, and good luck. You need all three, and the absence of the latter two cannot be compensated for by enrolling in some school, unless perhaps it's run by Dale Carnegie.

I hope the regular readers of this column (if any) can endure the constant repetition of this theme, but the letters on the subject are endless, and there is just no way to give each writer the immediate answers he demands. Enough said?

Other readers have a positive talent for writing thought-provoking letters that seem harmless enough until you try to answer them. I recently received a four-pager from a fellow in Birmingham, Michigan who is particularly interested in recording solo piano. He writes, "Solo piano may be the most difficult instrument with which to achieve the sound I want—which simply stated is the best. To me, the Philips sound is the best available; Columbia and RCA Red Seal are tied for second place. I don't know the relationship between the sterile studio approach and the indefinable pleasant sound that Columbia and others achieve by recording in one of the churches in lower Manhattan."

Well, for one thing, the churches have a lot to do with it. Or any of the other great acoustic environments that exist here and there. Most of them have one thing in common —they were not designed as recording studios. Economics and the mass market being a consideration, the record company that can design and construct a studio suitable for solo piano recordings is rare indeed.

RCA's Studio A in New York City is a notable exception. But a studio of this size (60′ x 100′ x 30′) is really more than one can expect to find on every street corner. Most studios must be designed on a less heroic scale, and of course the natural acoustics suffer. But, the bulk of today's recordings do not demand superlative acoustics in the classical tradition. In fact, it would be difficult to impossible to produce the kind of record that becomes a million seller in a hall with great acoustics.

And so, the solo piano recordist must often seek out a room that was never intended for recording. That is, if he wants that illusive "concert hall" ambience. Before the construction of its Studio A, RCA did much of its solo piano work at Webster Hall, an ancient ballroom in lower Manhattan that happened to have a good "room sound." It also had a post office with noisy mail trucks next door, no air conditioning, and a perverse heating system that would inevitably rattle its pipes at all the wrong times.

The engineer may be caught between a room with good acoustics and terrible working conditions on one hand, and better conditions but poor acoustics on the other. Depending on the final choice, the recorded sound will vary. And, depending on the taste of the listener, one or another of these variations will be most acceptable. Some will prefer the character of a fine hall—others the drier clarity of a modern studio.

Solo piano—or solo anything for that matter—is of course difficult to record well. The instrument is completely exposed, and any weakness in technique (engineering *or* artistic) will be quite obvious. And, *differences* in technique will likewise be more apparent.

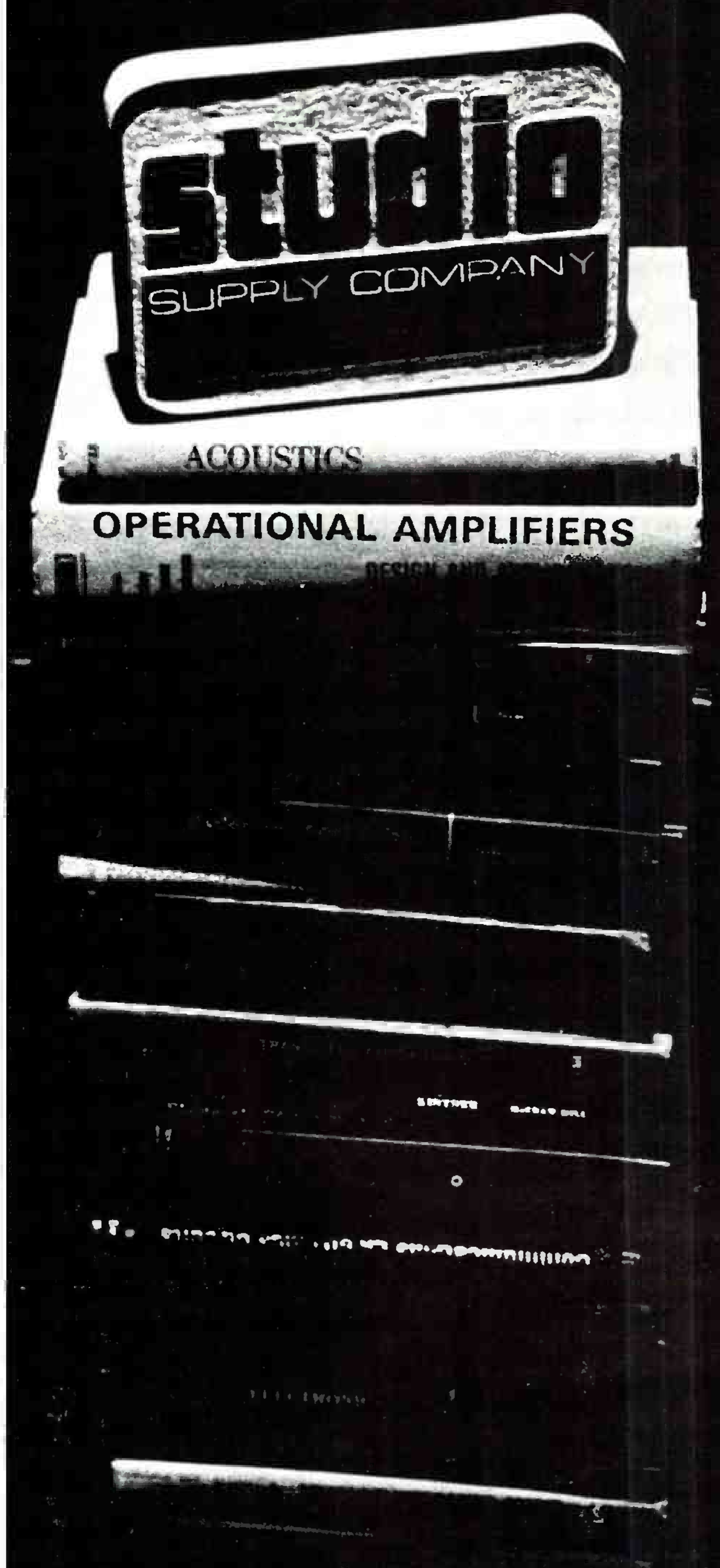
Summing up, technical and acoustic differences exist from one record to another. As usual, there are no "absolutes." Each recording will offer a different "sound" and the listener will make his own decision as to which is best for him.

For his own work, the letter writer notes that after many comparisons, he gets the best results with a pair of Beyer M101N microphones (dynamic, omnidirectional). That's inter-



*Circle 38 on Reader Service Card*

# WHERE THE BOOK ENDS STUDIO SUPPLY EXPERIENCE BEGINS




EXPERIENCE — wrought from extensive years of hard work by many people participating in all phases of recording. People now brought together to serve you.

KNOWLEDGE — gained from long successful involvement in mixing, studio management, maintenance, equipment design, accoustical consultations, and turn-key studio construction that uniquely qualifies us to mold an unlimited selection of the world's finest recording gear into a facility that fits you.

WISDOM — dictating that our designs and specifications are generated after careful consultations as to your style of recording, personal room and equipment preferences, and your fiscal requirements.

STUDIO SUPPLY designs and equips rooms that feel good, sound good, and work good. After all, isn't that what a good studio is all about?

STUDIO SUPPLY HAS ACCESS TO MANY LEASING PLANS AND OTHER SOURCES . . .

CALL DAVE, STEVE, EMIL OR CLAUDE TODAY . . .

(615) 327-3075

**studio**
SUPPLY COMPANY

P.O. BOX 280
NASHVILLE, TENN. 37202

*Circle 44 on Reader Service Card*

esting to me. I've never seen these mics used on solo piano, although many studios are now using one M101N on piano during pop rhythm section sessions.

He also inquires " . . . about the variations that occur in making masters from tapes, and in the record pressing process, and the quality of material used in the record."

Try to answer *that* in less than five thousand words!

Many engineers feel that once a master tape has been made, the session is over, and it's time to go on to something else.

Wrong! The tape-to-disc transfer session is just as demanding as the mixdown session. The program is to be transferred from one type of storage medium to another. And, consider some of the differences between tape and disc.

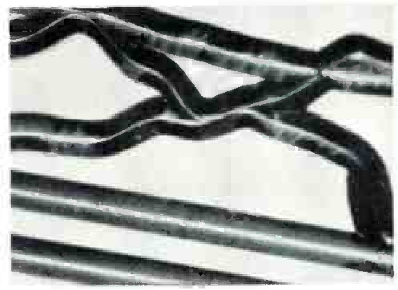
A tape program is stored as a magnetic field which the playback head converts into electrical energy. The disc stores the program in the form of modulated groove walls which later supply mechanical energy to the playback stylus.

The length of a tape program is in no way influenced by the maximum program level. On a disc, every loud section eats up a little more space on the record surface. Once you reach the label area, the record is over, even if the tape still has another minute or so to go. The name of the game is to run out of tape *before* the cutting stylus automatically goes into lead-out.

If you don't make it, the disc must be re-cut, probably at a lower level. Or, some of the low end may have to be attenuated, since lower frequencies take up more disc surface area.

Percussion instruments have been know to drive disc cutters into early retirement. Those marvelous tambourine sounds on your favorite tape may be more than the cutting head can take. Or, a particularly loud instrument on only one side may cause vertical overcut problems.

In short, if the final product is to be a disc, the limitations of this medium must be considered in the preparation of the intermediate medium —the master tape. But more often than not, the unique characeristics of the tape itself demand full time attention during mixdown, and so a certain amount of "reprocessing" is inevitably required at the disc cutting session.



(A)

(B)

*Figure 1.*
*Examples of poor cutting. At (A) grooves meet and overlap while at (B) a groove discontinuity exists.*

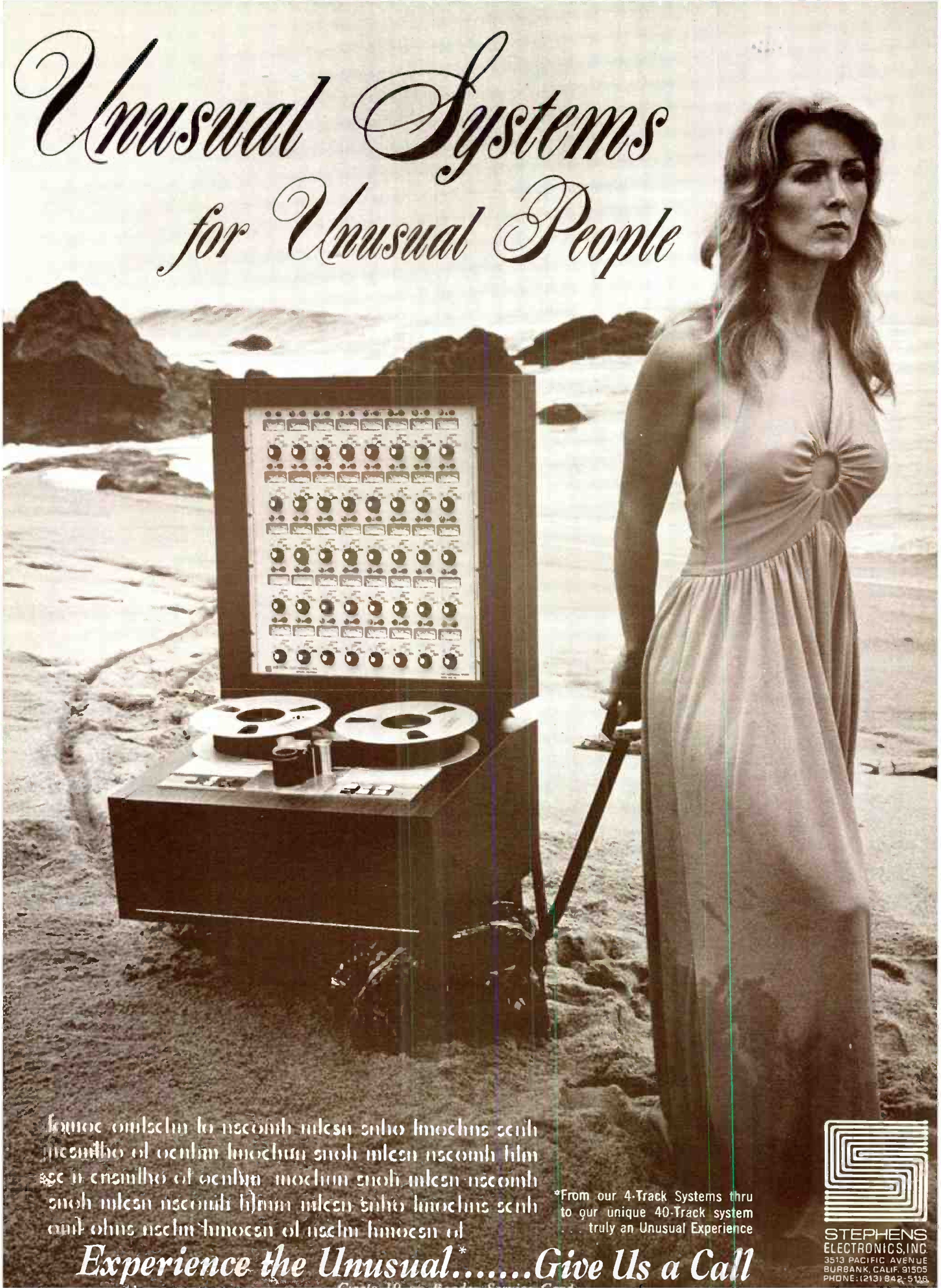
The quality of the final disc master depends on the expertise of both the recording engineer and the tape-to-disc engineer. If the former does not understand the disc cutting process, the latter will have a harder time making the transfer. Hence, the variations that sometimes show up in a tape/disc A-B comparison.

Even the most routine tape-to-disc transfer demands careful attention. Disc cutting must never be considered in the same league as tape copying. A well-equipped tape-to-disc transfer system will be packed full of elaborate equipment, some of which is unique to this operation. And even the more familiar items—limiters, for example—may have to be doubled up for use in both the preview and the transfer circuits.

The "variations" that may occur during the tape-to-disc transfer are many and varied. Some are to compensate for an ill-prepared tape; others, to make a good tape into the best disc master possible.

As program-storage media, tape and disc are not the same. The recordist seeking the best recording possible will do well to acquaint himself with tape-to-disc fundamentals, as well as learning how to operate a tape recorder.

As for the pressing process, and the quality of material used in the record, that's a book in itself. I'll have to hold off comment until a later issue—which will give me more time to figure out what it's all about. ■


Unusual Systems
for Unusual People
*From our 4-Track Systems thru
to our unique 40-Track system
. . . truly an Unusual Experience
Experience the Unusual*......Give Us a Call
STEPHENS
ELECTRONICS, INC.
3513 PACIFIC AVENUE
BURBANK, CALIF. 91505
PHONE: (213) 842-5116

Martin Dickstein

# SOUND WITH IMAGES

## Convention Time

• We are in the midst of conventions again, and the question comes up as to which to attend. Attitudes run from "See one, see all," or "They're different; choose carefully," to "There's something for everyone at every one; attend all." For some, it's something to put on an expense account, so might as well go, or, if the convention is held in an attractive city, it's an opportunity to squeeze in an extra vacation with a good conscience. Whatever your outlook on conventions, here is a preliminary rundown on those conventions you may have missed and mention of several you may want to attend.

Just over the horizon, as you receive this issue, is the Audio Engineering Society powwow in New York City, *Video Expo IV*, to be held at the Hotel Commodore on September 18, 19, and 20. At this conference, International Industrial Television Association and media and methods workshops are planned to bring together the most knowledgeable people in the fields of industrial, educational, and medical communications, pooling their viewpoints on problems in video management and production. For example, under the heading of *Connecting Up the Organization*, such subjects as "Management and the Tube," "Managing Private Video," and "How to Succeed in Video" will be discussed.

On the second day, the total subject, *Production Problems in Multi-Camera Systems* will be talked about and discussed. The session on the last day is entitled "Production Problems in Single-Camera Systems" and will cover portable vtr system operation, basic half-inch electronic editing, incorporating other media into videotapes, and selecting and trouble-shooting in a single camera vtr system.

Some of the scheduled exhibitors of hardware to be displayed in conjunction with the sessions include Audio Magnetics, EVR Systems, Goldmark Communications, Harvey Radio, Image Magnification, IVC, Magnavox Consumer Electronics, Modern Talking Picture Service, Panasonic, Sony, Telemation, Teletronics, and many more. A total of 10,000 square feet of space has been alloted to the display of new and projected products. For those of you who don't get to this one, we hope to have a complete report on what's new as soon after the convention as possible.

Cannes, France will be the site of *Vidca '73*, running from September 28 to October 3. More formally, this convention is named the Third International Market for Videocassettes and Video Disc Programs and Equipment. Experts from all over the video world will get together to discuss current developments in the cassette/disc fields. The main objectives of the convention are to present complete and unbiased information on hardware, as well as software, developments and the latest applications for these and to provide workshops dealing with current information regarding international distribution of video material, as well as applications and problems in various industries.

The total operations workshop program will be broken down into five parts: *Companies Workshop*, which will fully discuss the use of video by Coca-Cola, Ford, and Japan Air Lines; *Particular Market Workshop*, during which the case histories of Japan Video Association and Primary Medical will be discussed; *Education Workshop*, for the study of various school systems in the United States, Japan, the Netherlands, and Jamaica and their use of video; *Public or Private Locations Workshop*, in which several industrial applications will be discussed; and the *General Public Workshop*, in which the case of Cartrivision will be discussed. In addition, there will be a workshop on legal matters, designated *Contractual Obligations Between Creative Contributors and Producers*. This will take up legal aspects of video production and distribution, including international considerations.

Among the exhibitors will be video disc proponents, including Teldec, makers of special-film video systems like the EVR, and manufacturers of video tape systems, such as Philips, with its VCR system, Grundig with compatible-to-Philips VCR equipment, Shibaden with the CVR (Cartridge Video Recorder) system, IVC with



*Circle 21 on Reader Service Card*

# At last, a studio mastering tape that's better than the one everybody's been using.

A while ago, someone came along with a new tape that, admittedly, was a better mousetrap. But it was not the ultimate mousetrap.

There were problems with that tape, good as it was. Problems which we at Audio Devices have been able to solve.

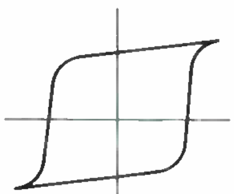
**First problem:**

How do you get even more energy out of each particle?

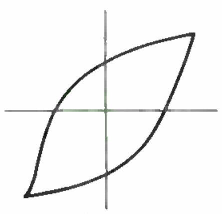
**Solution:**

By improving the dispersion.

Poorly dispersed particles clump together causing magnetic losses due to interaction and energy cancellations.



Hysteresis loop of Audiotape HOLN.



Hysteresis loop of poorly dispersed tape.

The new Audiotape HOLN has higher output and lower noise than the tape you switched to years ago. The new Audio tape iron-oxide particles deliver more energy.

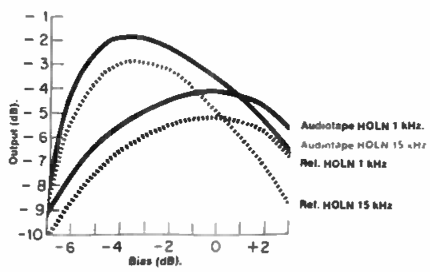
**Second problem:**

How do you reduce print-through?

**Solution:**

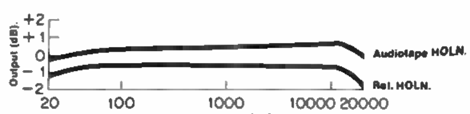
Uniform particle size and dispersion are part of the solution.


Output vs. bias.




Scully 1 recorder at 15 ips. Input level 0 dB.

Frequency response.




Scully 1 recorder at 15 ips. Bias 0 dB. Input level 0 dB.

Audio's secret processes are the other part. The results aren't secret, though: Audiotape HOLN has reduced print through by at least 2 dB, and typically 3 dB over the tape you switched to a few years ago.

**Third problem:**

How do you reduce headwear?

**Solution:**

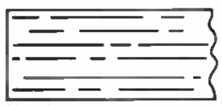
Use a smooth coating surface with a built-in permanent lubricant. That'll reduce your headwear.

**Fourth problem:**

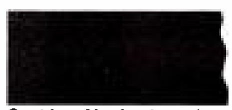
How do you improve handling and storage reliability?

**Solution:**

Use a backcoating that's super-durable. Cushion-Aire™ backcoating, with its controlled surface texture, improves high-speed handling and winding, and eliminates the causes of cinching, spoking, and other pack deformities.



Non-backcoated polyester after 1000 passes.



Cushion-Aire backcoating after 1000 passes.

Another problem is maintaining consistency of product. We've solved that too. By having a superior degree of quality control.

For a complete discussion of the improvements Audiotape features over the leading backcoated tape, send for our detailed literature.

**audiotape**™

Audio Devices, Inc., A Capitol Industries Company, 100 Research Drive, Glenbrook, Conn. 06906

*Circle 26 on Reader Service Card*

the one inch VCR cartridge system, Sony with the U-Matic cassette equipment, and Nivico with the compatible-to-Sony JVC Type-U system, as well as others in the forefront of new developments. The question of international compatibility, still very much in the air, will no doubt be an issue at this conference, if not as part of the program, certainly discussed by many of the attendees.

If your expense account won't carry you as far as Cannes, while we can't deliver a Riviera suntan, we'll have a complete report of news coming from this conference available for you right here.

Turning from things to come to retrospect, here are gleanings from some recent conferences you might have missed or wish to review. The first of the summer-fall cycle took place early in August at the Americana Hotel in New York City, listed as the *Ninth Education and Training Equipment Exposition*, run by the American Management Association. Although the tenor of the convention was that of business and education, it contained many aspects of interest to technical people. Exhibitors presented both hardware and software which also find application in many fields other than those being discussed. Trends and new concepts in education and training, of course, are of vital interest to many involved in audio/visual/video activity.

Just in case you are not familiar with the AMA, it celebrates its fiftieth anniversary this year. Through the years, educators and business executives have solved management problems with AMA seminars, courses, software material, and conferences. This year's session featured an education conference and a training briefing. The conference on education provided a forum for experts to discuss professional management, management theory, and management problems at every educational level. The briefing focused on developments in the utilization of human resources and the latest techniques aimed at humanizing industry. Exhibitors included the big names in the electronic and photographic field. We'll have a more complete report on this convention in our next issue.

*VidExpo '73*, sponsored by Billboard Publications, the third international video marketing conference and exhibition, was held at the Hotel Plaza in New York City on September 4-6. One of the unique features of this shindig, which draws as many as 1,500 people, was the VidShow. There were two of these three hour telecasts, which brought program material and news of commercial and in-house software, via closed circuit t.v. done by Trans-World Communications' *TeleVention*, right into the rooms of registrants, thereby enabling one to attend the convention without taking off his shoes.

The program session focused on *Management Communications, Professional Training* and *Specialized Applications*. More specifically, the scheduled topics were "How Management is Using Video Networks to Improve Communications," "How Business is Using Video to Improve Employee Performance," and "Specialty Applications in Advertising, Medicine, Hotel Pay/Free t.v., and Retailing." Speakers included representatives from a major advertising agency, a frontranking bank, the American Academy of Family Physicians, city government, and one of the huge international industrial corporations.

Another session dealing with government agencies, education, and entertainment, discussed the use by government agencies of video, the actual effectiveness of the "entertainment concept" in getting across messages in the public interest, and a consumers' eye view of home entertainment. Speakers were recruited from educational t.v., the International Tape Association, the Veterans' Administration, a large university, and several organizations associated with video communications.

One of the interesting exhibits of new products was G.E.'s demonstration of its video projector, showing the same video material set to be fed into the registrant's rooms. A partial list of other exhibitors of major hardware, software, duplicating, and distribution systems included Akai, Ampex, G.E., Image Communications, ITA, JVC, Martin Audio, Panasonic, Philips/Norelco, and RCA. We'll be having a more complete report on this convention in our next issue.

Whether, as we said, you charge conventions off to expenses or enjoying a well-deserved (naturally) vacation, or to an urge to become better informed, you are bound to come away from them with fresh perspective and an improved understanding of video and a/v situations, problems, and applications that will make you better equipped to satisfy your clients. And in case you just can't get away to attend in person, we'll do our best in future more specific discussions of this spate of conventions to deliver the information you want right here. ■

# $1890

## You pay for what you need

If you're seriously into music or sound reinforcement you want more than hi-fi products can give you. But full professional studio gear costs an arm and a leg, and you pay for a lot of things you may not really need.

That's why there's a TASCAM Model 10. It's an 8-in, 4-out mixing console, and it's just $1890.

With the Model 10 you get what you have to have. Without sacrificing a single necessary function.

Each input module gives you mic and line attenuation, three bands of peak and dip equalization (two with frequency selection), pre- and post-echo send and receive circuitry, pan function, and a unique straight-line fader.

Each of the four submasters has a meter control switch (line/echo), independent monitor level control, echo receive level control, and a straight-line fader. You also get a master gain module and 4" VU meters with LED peak indicators. Plus pre-wired facilities for up to four additional input modules and other optional accessories including talkback, remote transport control, quad panner, and headphone monitor.

That's what you need and that's what you pay for. Some things, however, you may or may not need, and we leave that choice up to you. For instance, the basic Model 10 is high impedance in and out, but studio line impedances are available optionally. You'll probably want low impedance mic inputs, but you may not need all low impedance line inputs. So we don't make you pay for them. You can order any combination of high and low input/output impedances according to your application.

Details and specs on the Model 10 are available for the asking. At the same time we'll tell you about our new Series 70 Recorder/reproducers.

We've got what you need.

**TASCAM CORPORATION**
5440 McConnell Avenue
Los Angeles, California 90066

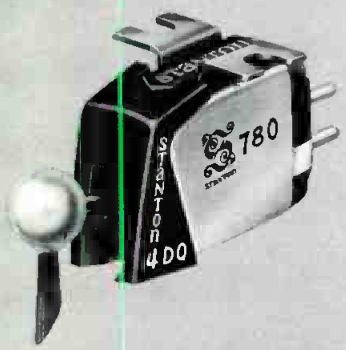
*Circle 45 on Reader Service Card*

# NEW PRODUCTS AND SERVICES

### HAND-HELD TAPE TENSION GAGE

● A virtually frictionless mechanism, according to the manufacturer, provides excellent static and dynamic response with a high degree of accuracy on tape widths ranging from ⅛ inch to two inches. Designated model T-2, the "Tentelometer" is factory calibrated to within ± five per cent and measures tape tension to control: CCTV picture distortion; head wear; clutch wear; tape wear; "servo wander," motor surges; head cleanliness; belt slippage; other operating parameters. Easily inserted onto stationary or moving tape, the gage has a single expanded scale for accurate low tension readability and a balanced mechanism to provide accurate readings with gage in any position. There are no wires, batteries, or levers.
*Mfr: Tentel*
*Price: $198.00*
*Circle 53 on Reader Service Card.*


### DISCRETE 4-CHANNEL STYLUS




● For users of discrete four-channel records, cartridge 780/4DQ, featuring a quadrahedral stylus expressly designed for this type of record, is being offered. According to the manufacturer, the new stylus has functioned well under testing conditions and is now ready for the consumer.
*Mfr: Stanton Magnetics, Inc.*
*Price: $125.00*
*Circle 54 on Reader Service Card.*



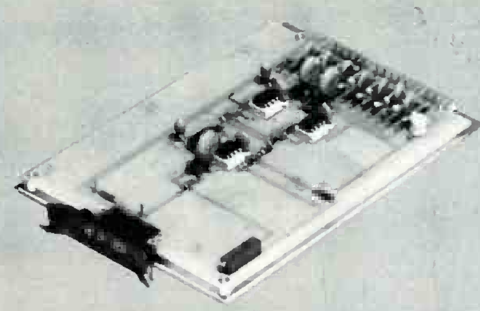
*Circle 39 on Reader Service Card*

## CLOSED-CIRCUIT INTERCOMMUNICATION SYSTEM


• Designed specifically for the high noise level encountered in the entertainment business, System CCX100 is designed to provide clear headphone closed-circuit communication for production crews working in theaters and studios. The system consists of two basic units . . . the main station, which houses the power unit, overall system controls, and station controls for the main station operator; and remote stations — lightweight belt-packs with headphones. Up to thirty remote stations can be looped into the system. Shielded two-conductor microphone cables connect stations to the main station or to other remote stations. The system also includes call lights which back up the audio system to cue operators who have removed headphones. An auxiliary input and gain allows the operator at the main station to input additional audio signals to all stations.

*Mfr: Clear-Com (Lumiere Productions, Inc.)*
*Price: CS-100 Main Unit . . . $275. (With relay $290.)*
*Other components at comparable prices.*
*Circle 50 on Reader Service Card.*

## ACTIVE COMBINING NETWORK



• Differential inputs for use with 600 ohm balanced circuitry are featured in the model 213 ACN 6X1 active combining network module, designed for use as a mixer for up to six audio sources. The unit produces no signal loss while maintaining source isolation. Specifications include: 10 K ohm balanced bridging inputs; 60 dB isolation between inputs; 50 dB minimum CMR for each input; unity to 20 dB voltage gain provided by potentiometer adjustment.

*Mfr: Roh Corporation*
*Price: $75.00*
*Circle 52 on Reader Service Card.*

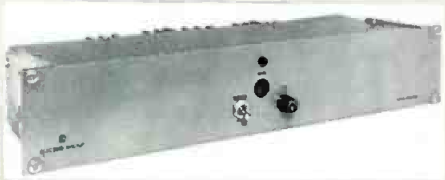


## "HANDS-FREE" GAIN RIDING COMPRESSOR

• Mixing and automatic gain riding are delivered in a portable package by SE30 gated compressor/mixer, designed for broadcasting and sound recording. The component combines a three input mixer with 600-ohm line output and a compressor adjustable to input requirements within a 40 dB range. Once the compressor is set, it rides gain automatically; a gated memory circuit holds compression level when input signals stop or drop below a certain threshold. When desired program material returns, the hold is released. This application can be useful in situations where crowd noises are to be eliminated. This compression is accomplished before the signal enters telephone lines or voice couplers. SE30 can be powered by either a.c., d.c. or its own self-contained batteries. Should the a.c. power fail, the unit automatically switches to d.c. (battery). Epoxy-glass p.c. boards. Input and output options available.

*Mfr: Shure Brothers*
*Price: $310.20*
*Circle 51 on Reader Service Card.*

## CONTINUOUS DUTY POWER AMPLIFIER



• Model 950 is especially designed for those situations where the instrument is to be left unattended for long periods. Long term reliability is provided by an internal current sensing circuit which protects the unit from overloads and holds its output within safe operating ranges even if an accidental short should occur. There are also oversized heat sinks for the output transistors. The compact, 3½ inches wide, unit provides a full 50 watt rms output with less than 0.5 per cent total harmonic distortion. Standard output impedances of 4, 8, and 16 ohms are provided, plus 100 ohms (70 vol line.) Will operate at ambient temperatures up to 55 degrees centigrade.

*Mfr: Electro Sound*
*Circle 55 on Reader Service Card.*

---

## PHONE BUTLER

• An inexpensive telephone answering device, suitable for both home and business, Phone Butler incorporates both a professionally pre-recorded answering message and provision for personal message dictated into a condensor microphone. In addition to answering the telephone, the device can be used to record memos. Operated with simple pushbuttons, the Phone Butler can record up to 30 messages. It can be set to answer after three rings or five rings. The a.c. operated unit meets all applicable telephone company regulations and may be used with or without a telephone company coupler. UL approved. Computer-type logic circuitry. Automatic level control maintains proper voice volume. Dimensions are 9 x 10 x 2¾".

*Mfr: BSR/Metrotec Electronics, Inc.*
*Price: $99.95*
*Circle 56 on Reader Service Card.*

# 46th AES Convention and Exhibition

On these pages we present the essential program and a map of the Audio Engineering Society's 46th Convention and Exhibition to be held in New York City at the Waldorf-Astoria Hotel. The dates are September 10 through September 13, 1973.

## Schedule of Events

Exhibits and Technical Sessions—Ballroom (Third) Floor
Demonstrations—Fifth Floor

### Registration

Monday, September 10—8:00 a.m. to 10:00 p.m.
Tuesday, September 11—8:30 a.m.—9:00 p.m.
Wednesday, September 12—9:00 a.m. to 5:00 p.m.
Thursday, September 13—9:00 a.m. to 5:00 p.m.

### Exhibit Hours

Monday, September 10—1:00 p.m. to 10:00 p.m.
Tuesday, September 11—11:00 a.m. to 10:00 p.m.
Wednesday & Thursday, September 12 & 13—11:00 a.m. to 5:00 p.m.

### Technical Sessions

Jade Room: Sessions A, B, D, E, G, I, J, K, M
Astor Gallery: Sessions C, F, H, L

Monday, September 10

9:00 a.m.—Annual Business Meeting
9:30 a.m.—Broadcast Audio
2:00 p.m.—Transducers

2:00 p.m.—Audio in Medicine
7:30 p.m.—Home Music Production

Tuesday, September 11

9:30 a.m.—Electronics and Automation
9:30 a.m.—Noise Control
2:00 p.m.—Architectural Acoustics and Sound Reinforcement
2:00 p.m.—Digital Techniques
7:30 p.m.—"What did they do to my song, Ma?" (Open AES New York Section Meeting—no registration fee)

Wednesday, September 12

9:30 a.m.—Magnetic Recording
2:00 p.m.—Tape Duplication

Thursday, September 13

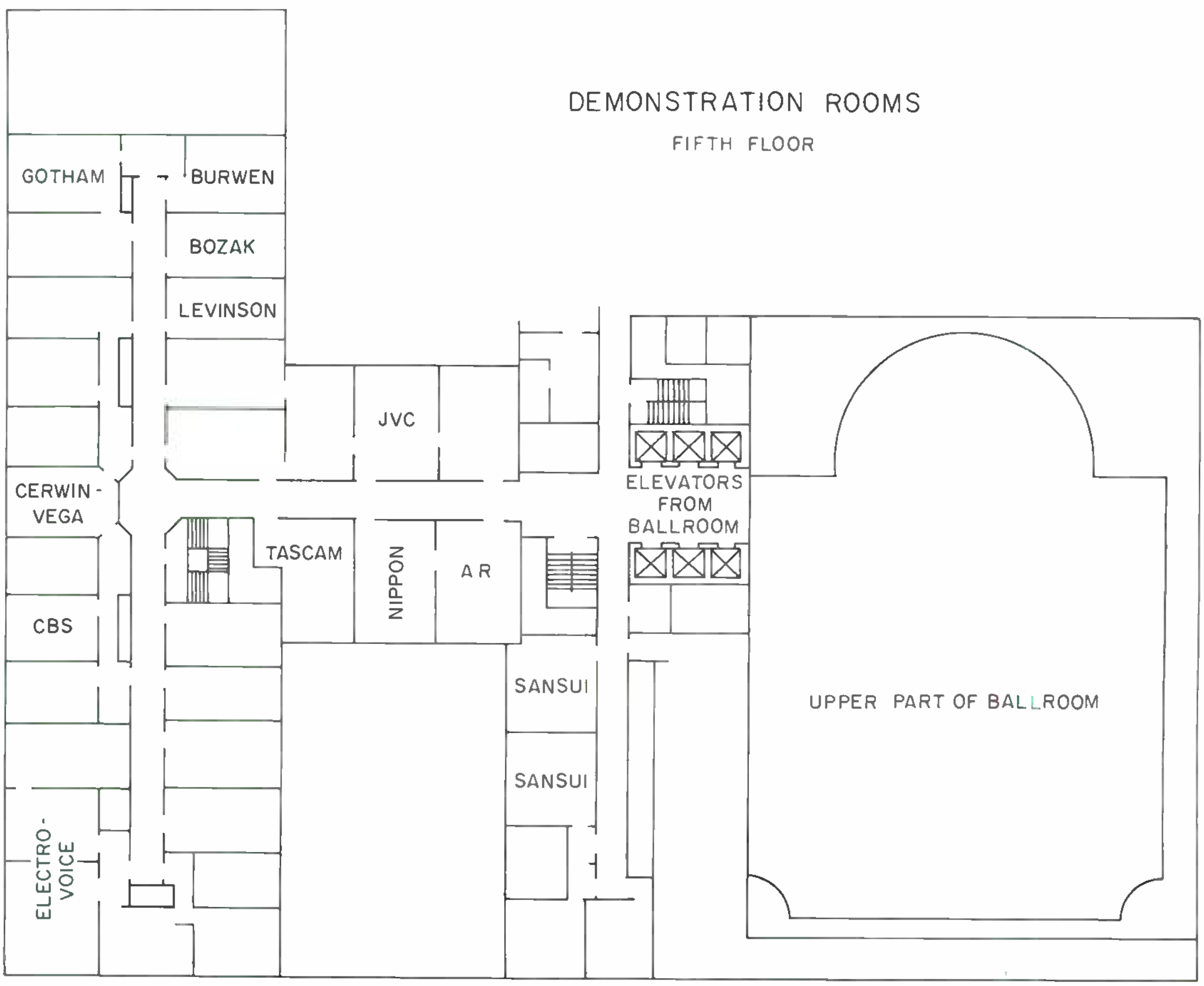
9:30 a.m.—Disc Recording and Reproduction
2:00 p.m.—Electronic Music/Electronics in Music Education

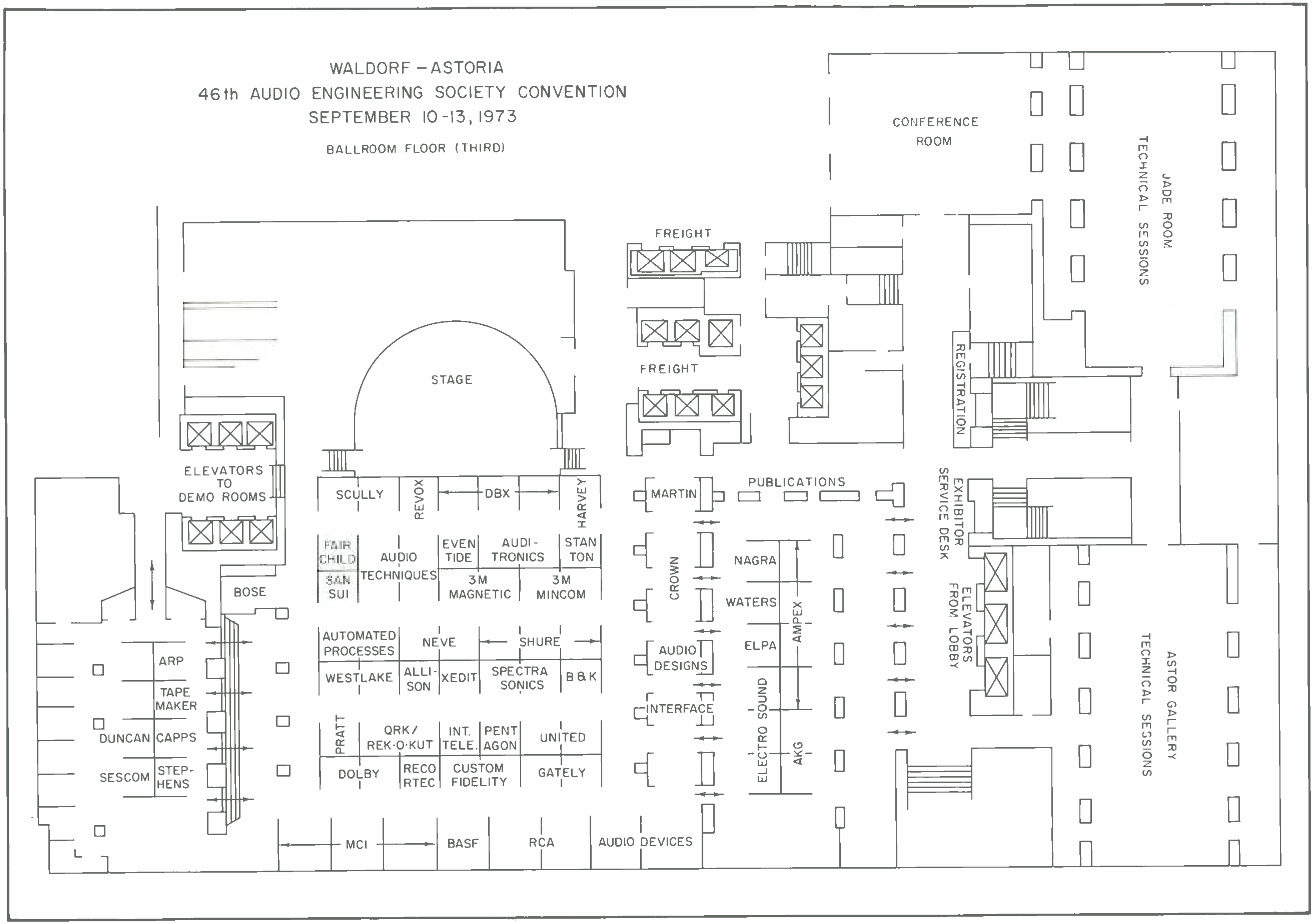
**Social and Cultural Activities**

Wednesday, September 12

7:00 p.m.—Social Hour—Starlight Roof
8:00 p.m.—25th Anniversary Awards Banquet

A program of activities for those not totally involved in the Convention is planned by Marjorie & Donald W. Powers. Continental breakfast will be served at 9:00 a.m. each day before commencing the day's activities—suite number will be posted.



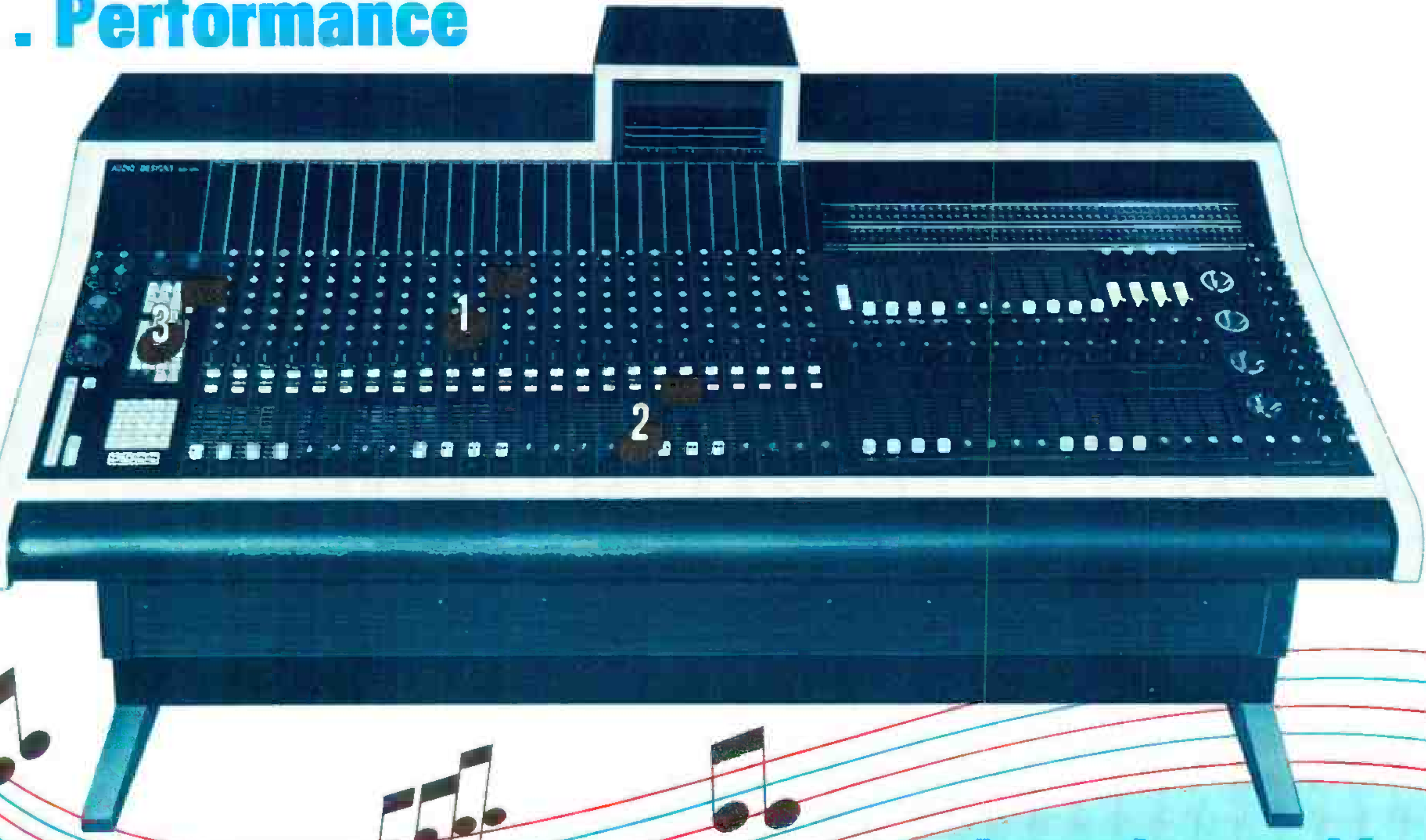


WALDORF – ASTORIA
46th AUDIO ENGINEERING SOCIETY CONVENTION
SEPTEMBER 10-13, 1973
BALLROOM FLOOR (THIRD)
CONFERENCE ROOM
JADE ROOM
TECHNICAL SESSIONS
ASTOR GALLERY
TECHNICAL SESSIONS
REGISTRATION
EXHIBITOR SERVICE DESK
ELEVATORS FROM LOBBY
FREIGHT
FREIGHT
STAGE
ELEVATORS TO DEMO ROOMS
BOSE
SCULLY
REVOX
DBX
HARVEY
FAIR CHILD
SAN SUI
AUDIO TECHNIQUES
EVEN TIDE
AUDI-TRONICS
STAN TON
3M MAGNETIC
3M MINCOM
AUTOMATED PROCESSES
NEVE
SHURE
WESTLAKE
ALLI-SON
XEDIT
SPECTRA SONICS
B&K
PRATT
QRK/REK-O-KUT
INT. TELE.
PENT AGON
UNITED
DOLBY
RECO RTEC
CUSTOM FIDELITY
GATELY
MCI
BASF
RCA
AUDIO DEVICES
MARTIN
PUBLICATIONS
CROWN
AUDIO DESIGNS
INTERFACE
NAGRA
WATERS
ELPA
AMPEX
ELECTRO SOUND
AKG
ARP
TAPE MAKER
DUNCAN
CAPPS
SESCOM
STEP-HENS

You never
lose a
hemidemisemiquaver
with Audio Designs
Consoles and Components
COMPONENTS
& CONSOLES
for audio control
AUDIO
DESIGNS

# ADM NRC SERIES CONSOLES

## hit a high note in Quality . . . Versatility . . . Performance



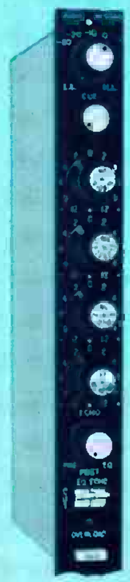
## Full-featured 8-16-24 Channel models at amazingly low prices

No one gives you more for your money than ADM. All ADM NRC Series Consoles are professionally engineered with such precise controls that you never miss or mess up a note. Yet we have held prices down through use of trouble-free "Mother Board" construction that eliminates literally miles of costly wiring . . . without sacrifice of quality, performance or "state-of-the-art" specifications.

You get high reliability features that ADM engineers into its custom consoles, including Slidex, Audex, four Cue Busses, two Solo Systems, complete Sync System, complete Tip, Ring and Sleeve Patching, Full Quadraphonic Capabilities including four Joy Sticks. And you get the flexibility and convenience of interchangeable plug-in modules, which make it easy to expand your console for future needs.

Chances are that one of our NRC models will meet your needs nicely. If not, we will be glad to work with you to integrate our components into your existing system, or in designing a custom console to your exact requirements.

## Interchangeable Plug-in Modules for complete flexibility



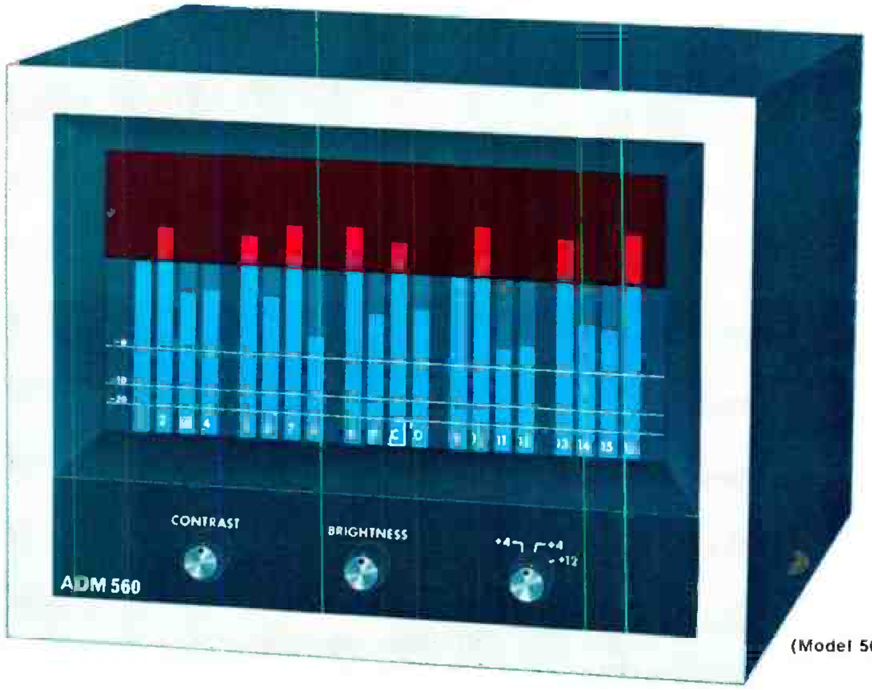
**2 SLIDEX**
Unique, noise-free attenuator uses sealed elements. No contacts to clean. Even spilled coffee won't cause noise problems. Guaranteed free from noise or other defects for five years.

**1 INPUT MODULES**
Versatile input strip features op-amp circuitry and four-section equalization. Input overload light and Solo button are also included. Everything required at an input is in one convenient plug-in module.

**3 AUDEX**
Ultra-reliable switching system uses magnetic reeds governed by this compact control panel. Switch modules, mounted above inputs, include illuminated readouts for instant reference.

# ADM VUE-SCAN

## Sweet music for frustrated "head swivelers"




(Model 560)

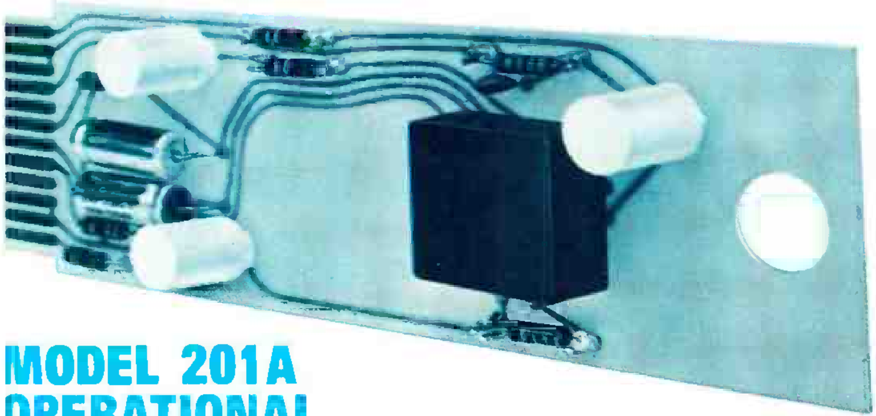
## Replaces up to 28 VU meters with an easy-to-read bar graph monitor

ADM's new Vue-Scan solves one of the most troublesome and annoying problems of multi-channel recording . . . visual monitoring of the recording channels. Vue-Scan takes up to 28 channels of audio information, and displays each channel as an easy-to-read bar graph on a TV monitor screen.

Each illuminated bar represents one analog channel. The vertical bars are always present as a background hue—the lower two-thirds in blue and the upper one-third in red. As the level of a channel increases, the bars representing that channel increase in height and intensity—making it almost impossible to overlook an overloaded channel.

Vue-Scan is standard on all NRC Series Consoles, replacing VU meters, and is also available as an option on Custom Consoles. In addition, it can be used as a self-contained accessory with any ADM or competitive console.

# More sound ideas from ADM



## MODEL 201A OPERATIONAL AMPLIFIER CARD

An audio operational amplifier designed on a 2.5" x 7.5" plug-in card, which can serve as a pre-amp, line amp or booster amp. Feedback stabilization is provided by "state-of-the-art" feedforward technique, providing the fastest possible response, and extending the frequency at which maximum feedback is available. Specially designed for low noise, and includes short circuit protection in the output stage.

## MODEL 1500/1501 EQUALIZER

A unity gain equalizer, designed to provide EQ anywhere in an audio system. Features ADM's exclusive four band equalizer, providing reciprocal EQ functions in low frequency (LF), low mid frequency (LMF), high mid frequency (HMF), and high frequency (HF) bands. Four bands and reciprocal functions provide absolute control over the entire frequency spectrum. Model 1500 is designed for vertical mounting in a console; the 1501 mounts horizontally in the CH15 Frame.

## MODEL 301 NOISE SUPPRESSOR

A new, fast gain expander that puts silence where previously there was noise. It will turn on from dead silence with a cymbal crash and not miss a single cycle of sound. Because it is a true gain expander, with a continuously varying gain, there are no threshold clicks, nor is the ear aware of the threshold. Extremely fast attack time, and smooth operational characteristics. Packaged on a 4" x 7½" card.


16005 Sturgeon
Roseville (Detroit), Michigan 48066
Phone: (313) 778-8400

SIDNEY L. SILVER

# The Psychoacoustic Aspects of Sound

*We take certain hearing effects for granted, but an understanding of what they are and how they work will be of benefit to the audio engineer.*

THE PERCEPTION OF SOUND by humans is a complex process involving the correlation of various psychological and physiological functions of the hearing mechanism. Although the properties of sound in terms of the subjective responses have been extensively investigated by many scientists, e.g., Helmholtz, Bekesy, Fletcher, etc., a full understanding of the hearing process has not yet been achieved. On the one hand, we must take into account the succession of events starting with the entry of sound waves into the ear, leading to the patterns of activity set off in the auditory regions of the brain. On the other hand, we must also be concerned with the problem of evaluating our reaction to sound as expressed by the sensational and emotional response in hearing. These aspects of our hearing experience are embraced by the term *psychoacoustics* which attempts to establish a relationship between the mechanism of hearing and our subjective response to sound stimuli.

In dealing with these phenomena, certain questions arise: Why, for example, do loud sounds tend to mask less intense sounds so that the weaker sounds are inaudible? And why do low-frequency sounds tend to obscure high-frequency sounds more readily than vice versa? Also, how can we explain our ability to locate a sound as coming from a definite source in space? A useful approach in seeking answers to these psychoacoustic questions would be to examine the structure of the auditory system, observe how the system functions, and determine the relationship between the input and output of the system.

The human ear is a highly sophisticated transducer system which establishes the final criteria for the judgment of sound quality. Although it is not a perfect transducer, it is a remarkably well-engineered device. Included in the hearing mechanism is a multichannel transmission system that can handle the conversion of mechanical energy to electrical energy, a system for maintaining a delicate hydraulic balance, an amplifying unit with provision for impedance matching, a time-frequency analyzer, and an information-processing system—and all of these are contained in a miniaturized package. A simplified representation of the human ear is shown in FIGURE 1. Paradoxically, the auditory functions of the entire hearing mechanism transcend the apparent limitations of the individual parts.

## ACOUSTIC RESONANCE

The hearing process begins in the external portion of the ear, or *auricle,* which collects sound pressure waves and channels them through the auditory canal. In humans, the acoustic effect of the auricle is almost negligible, largely because it is too small to deflect sounds into the ear efficiently. In order to modify a sound path, the surface would have to be large enough with respect to the wavelengths of interest. this condition is hardly satisfied, even for the highest audible frequencies. Actually, the side of the head forms an even larger baffle in a similar plane.

Further into the ear is the *auditory canal,* which functions in a free field as an acoustic resonator, loaded down by the pressure-sensitive eardrum. The resonating chamber reaches quarter-wave resonance at about 3000 Hz and increases the ear's sensitivity by about 10 dB at that frequency. In some respects, this combination of an open end and a shaped end enclosing a resonating column of air can be likened to an organ pipe.

## MECHANICAL LINKAGE

Beyond the eardrum is a chain of three articulated bones, collectively known as the *ossicles,* which act as a matching transformer for the efficient transfer of sound energy through the middle ear. Referring to FIGURE 1, acoustic pressure on the drum sets up vibratory motion through the ossicles which, in turn, is transmitted to the oval window membrane forming the entrance to the liquid-filled inner ear. Due to an apparent lever action, the ossicles create a mechanical advantage or force-multiplication of three. Since the area of the ossicle contacting the oval window is much smaller than that loading the drum (about 1/20th), the vibrations are transformed into a greater force with less excursion, thereby effectively matching the low impedance of sound waves in air to the high impedance of sound in fluid. As shown in FIGURE 2, this yields an additional multiplication factor of twenty, so that the overall vibrations are magnified about sixty times. This is equivalent to a total gain of about 36 dB at the natural resonant frequency of the system. Essentially, this transmission path acts as a broadly tuned bandpass filter whose response includes a peak, fortunately within the range of speech frequencies.

---

*Sidney L. Silver has done extensive research on the effects of noise pollution on the human ear and audiometric measurements. He comes to this work as an engineering graduate who worked on the development of the microgroove record under Dr. Peter Goldmark, and is presently on the supervisory staff of the Telecommunications Section of the United Nations where he is in charge of sound and recording.*

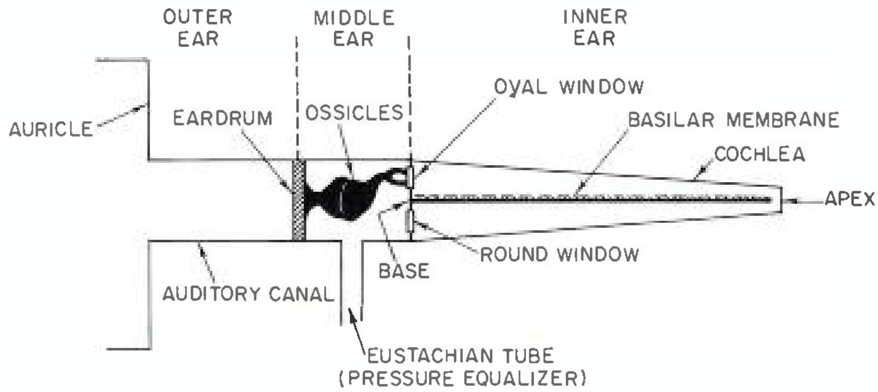



*Figure 1. A simplified schematic of the human hearing mechanism. For the sake of clarity, the cochlea is shown straight; it is in fact a coiled spiral form much as a snail shell.*

Another important function of the middle ear is that of a safety device to help protect the hearing mechanism from excessively loud sounds. This is accomplished by a muscular reflex action which modifies the mechanical impedance of the ossicles functioning as an amplifying unit, somewhat like the operation of an automatic volume control. Small muscles stiffen the drum and simultaneously pull the output ossicle away from the oval window, resulting in a transmission loss of the order of 10 dB. The reaction time is about 10 msec, so that the reflex action is most effective when the ear is stimulated by sound waves that build up slowly enough to permit the muscles to be actuated. A steep, high-intensity wavefront, for example, may force the drum and ossicles into a destructive motion before the AVC action comes into play, and cause irreparable damage to the delicate structure of the inner ear.

## HYDRAULIC BALANCE

In the next stage of sound transmission, the mechanical vibrations impinging on the outer window are transformed into hydraulic pressure waves. This conversion takes place in a spiral-shaped tubular structure called the *cochlea,* which is completely filled with a viscous fluid. In FIGURE 1, the cochlea is shown in an unwound position for the sake of simplicity. Because the liquid is virtually incompressible, a pressure relief point is provided by the flexible round window, and the resulting reciprocal motion between the oval and round windows makes possible the almost instantaneous propagation of sound waves in fluid.

During acoustical stimulation, traveling waves are set up in the cochlear fluid, causing a ripple-like motion of the basilar membrae within the cochlea. Owing to the elastic properties of the cochlea, different sound frequencies tend to produce maximum displacement along different portions of the membrane. The position of highest crest is important because it determines which sensory receptors will be used to process the sound signals. High-frequency sounds produce maximum amplitudes toward the base of the membrane, while low-frequencies cause the membrane to ripple throughout its whole length, but with greatest displacement at the tip. This transformation of sound frequencies into regions of maximum stimulation is the basis of the ear's ability to perceive pitch and to resolve complex sounds into their various components.

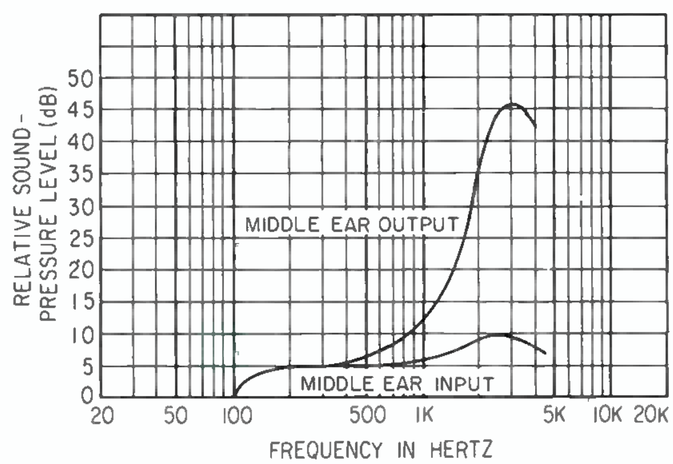
## PULSE TRANSMISSION

Acoustic motion of the cochlea stimulates a complex of sensory cells scattered along the basilar membrane, and this produces a mechanical bending, or shearing effect, against the surrounding area. The resultant force generates an electrical trigger potential which, in turn, initiates neural impulses in the auditory nerve. These impulses are presumably analyzed by the brain so that we can reach some conclusion about the sounds we hear.

The *auditory nerve* can be regarded as a multichannel cable composed of thousands of nerve fibers, each fiber constructed very much like an insulated wire. But the nerve impulses which give rise to the sensation of hearing are in no way comparable to an electric current flowing through a conductor. These impulses are characterized by an all-or-none action; that is, they are of normal strength or zero strength regardless of how they are initiated. Accordingly, an increase in the intensity of the sound stimulus excites more sensory cells along the basilar membrane and hence, neural impulses are discharged in more of the nerve fibers of the auditory nerve. This produces shifts in location of the cochlear response which the ear associates with changes in pitch. Conversely, variations in frequency cause changes in loudness since the shift in location of maximum-response areas along the basilar membrane involve different intensity responses of adjacent nerve fibers. Clearly, the sensation of loudness and the perception of pitch are interdependent factors of our hearing experience.

Sound energy, in effect, is translated by the nervous system into a succession of discrete pulses wherein recognition depends upon the particular pattern of the coded signal. It is the familiarity of these patterns that provides a basis for the brain's interpretation of a specific group of impulses representing certain sounds. Depending on an individual's previous conditioning, various sounds can evoke a wide spectrum of emotions and reactions. This process is somewhat analogous to the manner in which a data processor can run through a great number of punch cards and select those groups that have common characteristics.

*Figure 2. Frequency response curves indicate sound pressure amplification in the middle ear.*



# What would you do if you saw an orchestra drowning?

**We didn't just watch**
Seven years ago, musicians all over the world were drowning in tape noise. We jumped in and began saving everyone we could, with our Dolby professional noise reduction system.

**We made the best even better**
At first, not many people were sure we could do it without damage to the music, but we showed them, one by one, that if the recordings were made and played with our system, the tape noise was greatly reduced with no change in the sound. Now, recording companies all over the world use the Dolby system to make quieter master tapes. With more than 12,000 tracks Dolby-equipped in studios around the world, Dolby noise reduction has become the standard professional method of making original and duplicate masters for disc cutting, duplicating, and international mastering. Use of the system is rapidly growing in film and TV production as well.

That takes care of the professionals.

**Meanwhile, at home . . .**
Then we made a simpler Dolby system to save orchestras from drowning at home, especially on cassettes. Now, more than forty companies are making recorders and adapters with the Dolby B-type circuit. With one of these, home listeners can make Dolbyized cassette recordings that sound every bit as good as discs – and stay that way longer.

**Dolby noise reduction helps everyone**
Many pre-recorded cassettes are already made in this way. In fact, more than half the pre-recorded cassettes sold in America, England and Japan are Dolbyized.

On an ordinary cassette player these cassettes sound ordinary; but played with the Dolby circuit they are unlike any cassettes ever heard before. The same system is already being used in FM broadcasting, too, bringing Dolby-equipped listeners reception improvement which would otherwise require a tenfold increase in transmitter power.

Write or call for more information.

**Dolby®**

**Dolby Laboratories Inc**
1133 Avenue of the Americas
New York NY 10036
Telephone (212) 489-6652

346 Clapham Road, London SW9
Telephone 01-720 1111



*Circle 23 on Reader Service Card*

## PSYCHOACOUSTIC PHENOMENA

There are a number of pecularities in our hearing experience which alter, to some extent, the quality of sound impressed upon our ears. Consequently, under certain conditions, the sound we perceive does not accurately represent the variations of air pressure in the auditory canal. Vibratory waves processed by the cochlea will thus contain frequencies other than those making up the original sound. At one time, this nonlinear relationship was attributed to asymmetrical action of the eardrum, but subsequent tests have proven the drum to be virtually free of amplitude distortion—except when displaced by unusually intense sound levels. For moderate intensities, the movement of the ossicles is apparently a linear function of the input pressure waveform. It is generally accepted that the primary source of nonlinear distortion is the cochlea, but whether it is generated in the cochlear fluid or derived from the sensory cells is not known.

The failure of the hearing mechanism to respond linearly to sound pressure waveforms represents a limitation of the ear's powers of frequency analysis. Thus, when a pure tone of suitable intensity is impressed upon the ear, a series of harmonics of the original frequency is perceived. These harmonics, referred to as aural, or subjective, harmonics, have no external or physical existence but originate entirely within the ear itself. FIGURE 3 shows a set of curves indicating the sensation levels at which aural harmonics are barely detectable when the ear is stimulated by pure sinusoidal waveforms. Clearly, for tones above 2000 Hz, no subjective harmonics appear until the sensation level is about 50 dB, whereupon the second harmonic is just barely perceptible. In the low-frequency range, harmonic distortion appears even at comparatively low levels of the stimulating tone.

It is also interesting to note the effects of introducing two fairly intense pure tones, say $f_1$ and $f_2$, into one ear simultaneously. Here the perceived sound consists not only of the fundamentals and harmonics, but also other tones in the composite sound. These so-called "combination tones" may comprise the difference tone, $f_1 - f_2$; the summation tone $f_1 + f_2$; and possibly other tones formed by the combination of fundamentals and harmonics, e.g., $2f_1 - f_2$, $2f_1 - 2f_2$, etc. By far, the most conspicuous of these subjective tones are the difference frequencies. Generally, these are less easily perceived when the frequencies lie between the stimulating tones; thus tones of 800 Hz and 1800 Hz will give a difference frequency of 1000 Hz which careful perception will reveal. On the other hand, 1000 Hz and 1500 Hz will yield a difference tone of 500 Hz which can readily be observed. If the two tones are close in frequency, they may alternately reinforce or cancel each other resulting in a beating, or throbbing, effect.

When the intensity of one of the tones exposed to the ear is considerably greater than the other, the louder sound interferes with, or reduces the ability to sense the other sound. This phenomenon, known as *auditory masking*, is expressed as a shift in the threshold of audibility caused by the louder sound, and depends upon the frequency difference between the two sounds. Tests have been performed to discover the nature of masking; one of these is shown in FIGURE 4. Here a masking tone is kept at a constant level, while various tones are gradually increased in level until they are just barely perceptible in the presence of the masking tone. It can be seen that the higher the level of the masking tone, the greater the masking effect. There is also a greater degree of masking at frequencies higher than the masking tone than below it. Note that when the masking tone is near the frequency of the masked tone, the presence of the masking tone is perceived by the beats it produces. The beats, which account for the dips in the

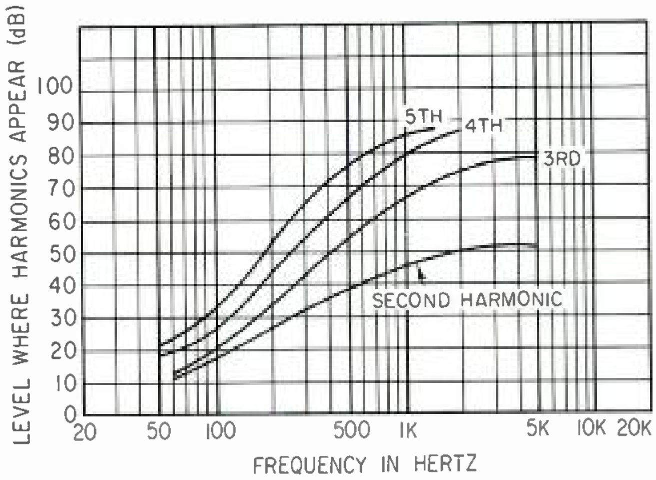



*Figure 3. The sensation levels at which aural harmonics are barely perceptible.*

curves at these frequencies, also may be observed at the harmonics of the masking tone.

In terms of motion of the basilar membrane, masking can be explained by the fact that low frequencies tend to cause undulations throughout the entire length of the membrane, so that there is considerable overlapping, or spillover, of the response areas. As a result, low-frequency movement at the base conceals the smaller high-frequency motion of the membrane. Thus, low frequencies tend to mask high frequencies more easily than the reverse situation.

Now let us consider the case where two tones are heard binaurally; that is, one tone in each ear respectively. Here the fundamentals and harmonics heard exclusively by one ear do not appreciably add, subtract, heterodyne, or interfere in any other way with the tones confined entirely to the other ear. Since these sounds are perceived individually, they do not produce beat notes with each other, even if the difference in the frequencies is very small. These binaural phenomena derive from the fact that the two ears have separate and practically independent effects on the auditory regions of the brain. For very loud sounds, however, intermodulation effects may be observed when a small amount of sound energy is cross-coupled between each ear. This is due to bone conduction in the skull, where solid-borne vibrations set up the same patterns of excitation of the basilar membrane as do air-borne sounds.

Another aspect of our hearing experience is the phenomenon of binaural localization, which relates to our ability to locate the direction of a sound source. From the psychoacoustic standpoint, this is an important consideration in the understanding of stereophonic reproduction, and its rational extension, four-channel sound. If we assume that the ear (in conjunction with the brain) is an information-processing system, then we can proceed to explain how the auditory system is able to tell direction, or more specifically, how it is capable of responding to slight temporal differences in sound stimuli presented to the two ears.

Complex sounds that reach our ears from a definite source in space have differences in arrival time and intensity, both of which are important for binaural localization. Let us consider, as an example, how timing information is maintained by each ear. At low frequencies (below 400 Hz), the nerve impulses are synchronous, or phase-locked, with the frequency of the sound stimulus. Thus, for each cycle per second of a sinusoidal input wave, there is a burst of nerve impulses at a time corresponding to a particular phase of the input. The brain uses this information for pitch discrimination. For the frequency range from 400 Hz to 5000 Hz, synchronism is still maintained in the audi-

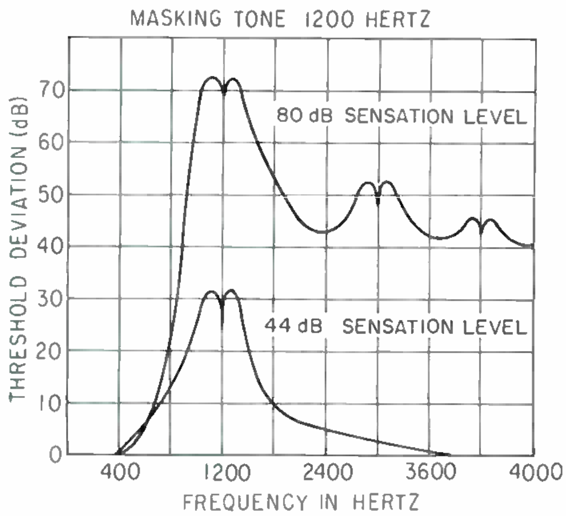


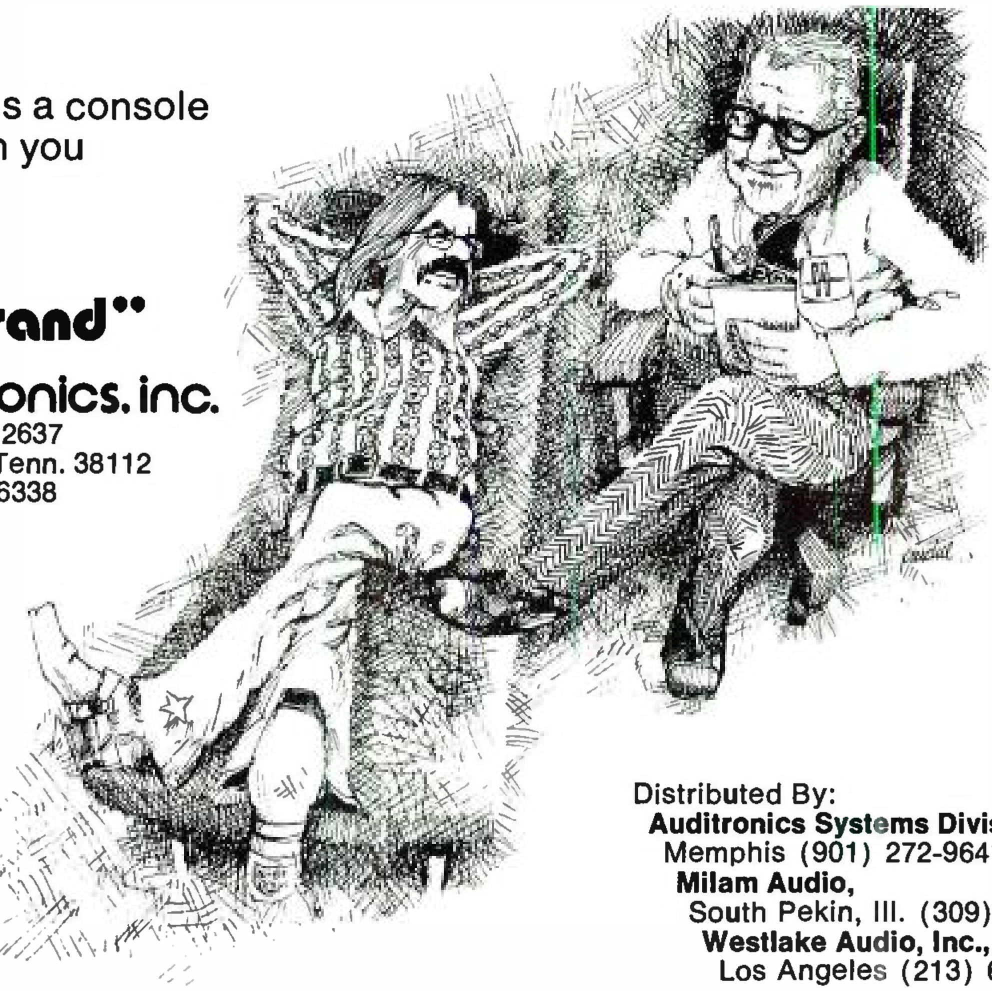

*Figure 4. The rise in threshold level for various frequencies during continuous stimulation by a masking tone. This is after H. Fletcher "Speech and Hearing in Communication" published by D. Van Nostrand and Co.*

tory nerve system, even though any particular nerve fiber is limited in response to 400 impulses per second. According to the "volley principle," the individual nerve fibers stagger their response, or alternately fire in groups, so that there is synchronization in the medium frequency range. The timing data contained in the nerve discharges are combined in the brain to produce a binaurally localized sound image. For the audible range above 5000 Hz, the rhythm of volleying is lost and synchronization can no longer be established. In this case, the random nerve impulses cannot follow the high frequency sounds, but they can follow the displacement envelope, or place of maximum excitation, along the basilar membrane.

Curiously, despite the fact that each ear has a resolution of about 2.5 msec, the two ears in concert are much more sensitive to temporal patterns. The relative time of impulse volleys between both ears can be small enough so that interaural time differences of the order of 10 usec are detectable. This remarkable faculty is attributed to the fact that the layout of the auditory nervous system is such that nerve impulses travel by several "relay stages" on their way to the brain. Some nerve fibers, for example, terminate in junctions which introduce time delays. Others cross over between the left and right auditory channels, while still others proceed directly to the brain. Consequently, the brain can perform a cross-correlation of nerve impulses, enabling the auditory nervous system to shape and define the binaural sensation.

It should be pointed out that the phenomenon of hearing is sufficiently complex so that no single theory thus far advanced by auditory researchers can fully account for every aspect of the hearing mechanism. At the present time, the most widely accepted theory involves a compromise between the space-pattern and time-pattern concepts in the transformation of sound energy to neural impulses—the main difference being the manner in which frequency analysis, and hence, pitch perception is handled by the inner ear. Each of these principles has helped to organize some of the psychoacoustic data on hearing, but until a truly unified theory is conceived, many questions have yet to be answered. ■



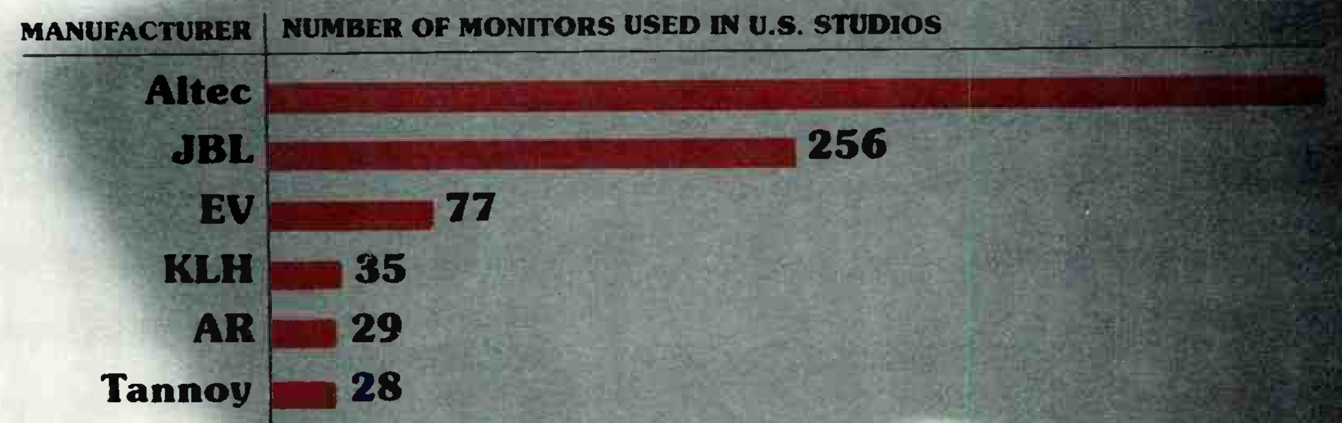


# The hungry leader.

At Altec, we're not taking our leadership position for granted. We're always trying harder—challenging ourselves to develop studio monitor speakers that stay a step ahead of constant improvements in the contemporary recording process.

Throughout the United States, recording industry professionals—engineers, producers, A&R men, musicians—have responded to our continuing ability to produce the finest monitor loudspeakers by installing more Altec systems in studios than any other brand.

And we can prove it. Here's the latest data on monitors installed in U.S. studios, as published in Billboard's 1973 International Directory of Recording Studios.



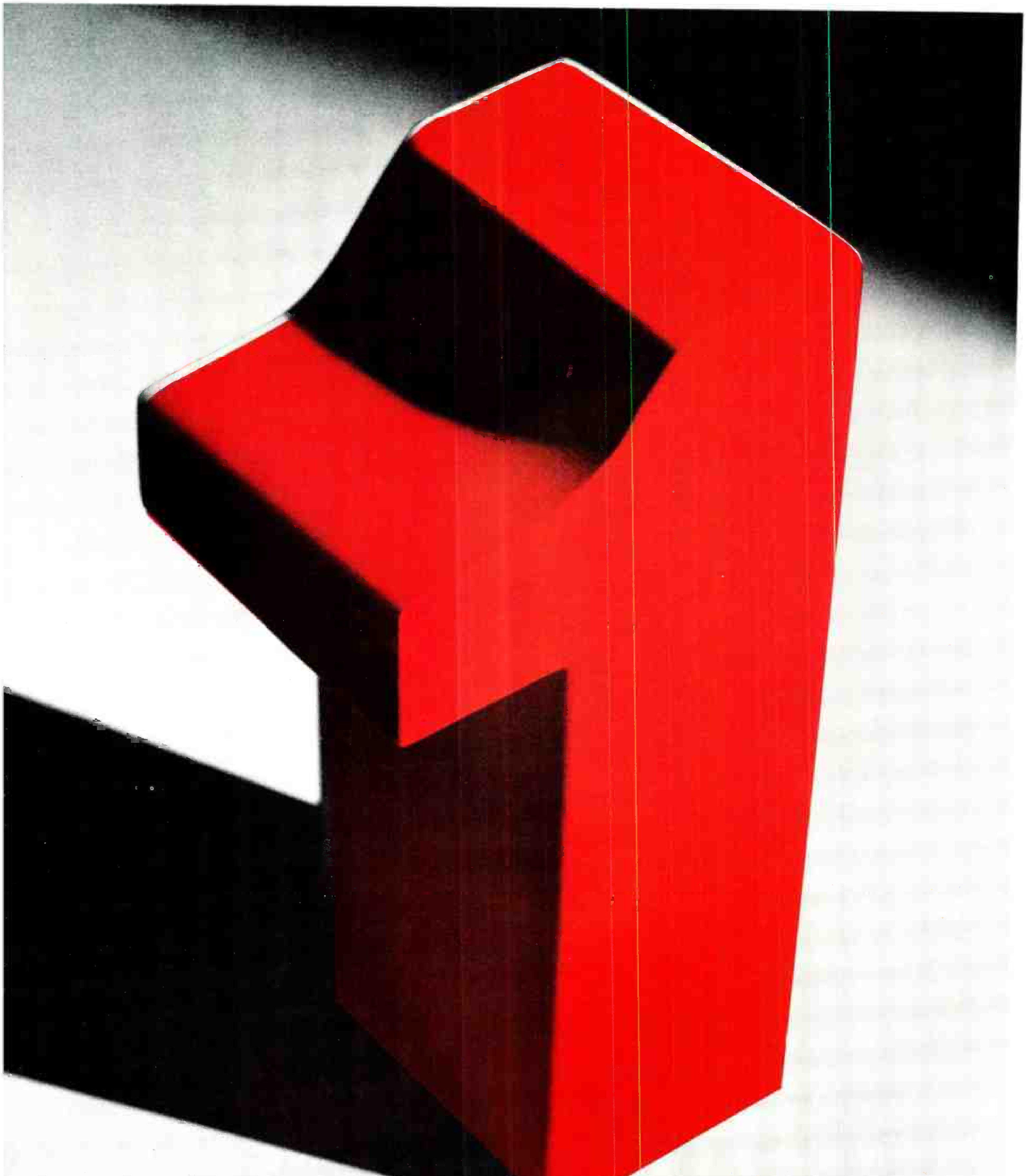

But we're not really satisfied—even with this impressive track record. We're still trying to better ourselves. In fact, Altec has three all-new studio monitors available right now. They're a whole new generation of speakers designed to meet the whole new range of tomorrow's dynamic recording techniques. Your studio may need them. Why not call your local Altec representative to find out?
Or write us for full details.

Altec gives you the best of both worlds: proven leadership, plus an unrelenting commitment to doing a better job. That's because we've really grown to enjoy being #1 in studio monitor sales during the past three decades. And we intend to stay right there for at least the next three decades by always being our own biggest competitor—in research, in quality, in service and in satisfying the demanding needs of an ever-evolving industry.

Altec. We're the hungry leader.



*Circle 24 on Reader Service Card*



W. A. RHEINFELDER

# New Approaches to Improved Tonal Reproduction, Part 2

*In this concluding section the author explores the problems with tonal balance systems and shows how they can be made satisfactory from a musician's viewpoint.*

NORMAL METHODS of tone control prove generally unsatisfactory from a musician's viewpoint. Determination of the exact reason for this behavior was the objective of this investigation. As part of this work, some specialized test equipment was constructed.[1] It is most interesting to examine the rather unexpected results obtained with these equalizers in critical listening tests with different types of music and program material. The investigation also relates to the removal of noise and extraneous frequencies as well as improvement of tonal quality, overall balance and sound enhancement.

## SELECTIVITY AND USEFUL Q

The Q of an equalizing or tuned circuit is defined by the center frequency divided by the 3 dB bandwidth, and is a measure for the sharpness of the frequency response. After some initial rather unsuccessful tests with an excessive Q in the vicinity of 20, the tunable filter described in Part I was set to a Q of 2. With the circuit set for boosting, it was quite remarkable how much ringing a low Q of 2 caused in the reproduction of music. This effect is noticeable at boosts above 5 dB. A Q of 1 proved too low to give a sufficient effect, and eventually a $Q = \sqrt{2} = 1.41$ was chosen and it appears to be the right

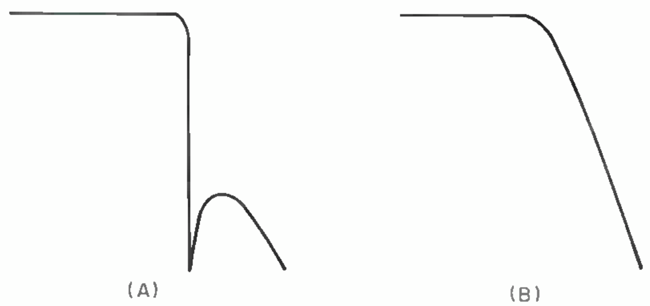



*Figure 1. Low pass filter types with (A) a null in response and (B) a normal rolloff.*

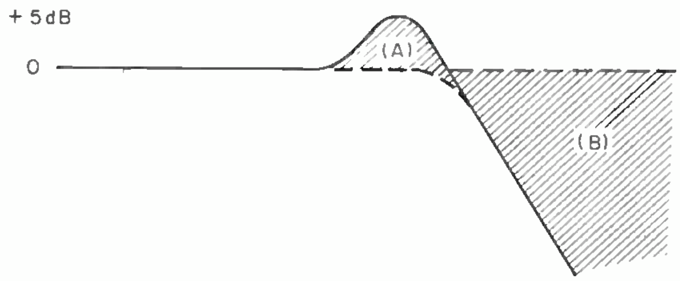



*Figure 2. An uncompensated filter with boost before rolloff. The musical energy in areas (A) and (B) should be about equal.*

value. At the moment there is no explanation why this value appears best; maybe it is somehow related to the physiology of hearing. With a Q of 1.41, frequency spacing on an octave basis (frequency ratio 2 to 1) in a comb filter leads to overlapping ranges. For convenience in equalization research, such a multi-frequency equalizer was constructed, based on octave bands with an individual Q of 1.41.

The individual tone filter was also used to eliminate interference tones. Contrary to the boost position, it was found that a higher Q was desirable for best results. For equalization purposes, if a band of frequencies was to be attenuated, a Q of 2.8 was found best, or twice the Q used for boosting. For single tone elimination, with little or no flutter present, a Q of 30 was best; this was the maximum available in the test equipment. A tape containing both hum and a high frequency beat was used for this test. It took several passes to clean the tape up, one at 60 Hz, one at 120 Hz, and one at approximately 5 kHz. Also it was easily verified that when flutter was high, a lower Q had to be used to reduce the resulting frequency modulated signal. In fact there is a direct relationship between the amount of flutter and the filter Q giving the greatest reduction of a recorded signal, and flutter could be measured with this method.

For analyzing individual overtones, a high Q on the order of 30 was desirable, particularly when analyzing a closed loop recording. Of course there are many other techniques available today based on the use of rather costly instruments such as spectrum analyzers. Much of this work pertaining to musical instrument tones and their harmonics has been done already and is available in the literature (Bell Laboratories, Western Electric reports). We can now examine what was learned in using a multifrequency equalized with different musical selections.

## LISTENING RESULTS WITH MULTI-FREQUENCY EQUALIZER

The first part of this investigation centered on better bass reproduction, bass in the musical sense, to include the bass, drums, bass clarinet, and not including the organ for the moment. In trying all kinds of program sources the best effect for bass boosting was found by using both the 80 and 160 Hz circuits, even though the fundamental tone of the musical instrument was lower. It was startling that the 40 Hz circuit did not improve the bass sound. On examining the harmonic outputs of all bass instruments in the literature, it was found that in most cases more than 90 per cent of the sound output was above 80 Hz, even if a lower note was played. Actually this stands to reason, because only physically large instruments are reasonably efficient transducers for coupling to the air as a transmission medium. Only the bass itself (and the organ of course) extend to lower frequencies. However, even the bass does become quite inefficient below 80 Hz. Consequently in boosting an instrument, the frequency range containing the significant sound output (90 per cent of sound energy) must be emphasized. The listening tests confirmed this theorem very well. Further listening tests showed that the musical bass is centered around 130 Hz. It was also found that a single filter with a Q of 1.41 centered at 130 Hz had insufficient bandwidth. Since a lower Q filter cannot be used, two stagger-tuned circuits are needed to achieve the desired response. Ultimately, best results were obtained with two circuits of Q = 1.41 tuned to 80 and 160 Hz, and boosted equally. The amount of boost ranged from 4 to 10 dB for really outstanding effect. There were minute advantages by boosting the two frequencies to amounts differing up to 3 dB depending on program material. From a practical viewpoint this was unimportant. A far more serious shortcoming was the lack of boosting simultaneously some significant high frequency component contained in the bass instrument. After many listening tests, it was found that the high frequency component is centered between 5 and 5.5 kHz and wants to be boosted by a single circuit with a Q of 1.41. The correct amount of boost was found to nearly equal the low frequency boost. The ideal bass control should therefore have the characteristics of Table I.

Considerable A-B listening confirmed that on average material 7 dB boost led to very nice results. It is not imperative that this control be continuously variable, particularly if it is combined with a standard bass control. All these findings were verified with different speaker systems extending to very low frequencies. We are excluding the organ here because it is a strange instrument with significant fundamental output down to 16 Hz. Also it does not have a high frequency component associated with bass tones other than extraneous noise. With the organ, bass boosting must be defined differently and a standard bass control is workable for the organ. Therefore, both types of bass control are really needed for a universal control. An example of how this can be achieved economically is given in the musician's control below.

In addition to better bass sound, experimentation was carried out to improve the middle register, to emphasize soloists or increase the brilliance of a piano. Best frequency for solo instruments was 1000 to 1500 Hz with boost up to 8 dB and a Q of 1.41. Added brilliance was achieved in some cases by moderate boost at frequencies

**Table I**
**Effective Bass Boost Frequencies**

| Frequency | Q | Amount of Boost |
|---|---|---|
| 80 Hz | 1.41 | 4 to 10 dB, 7 dB average |
| 160 | 1.41 | |
| 5200 | 1.41 | |

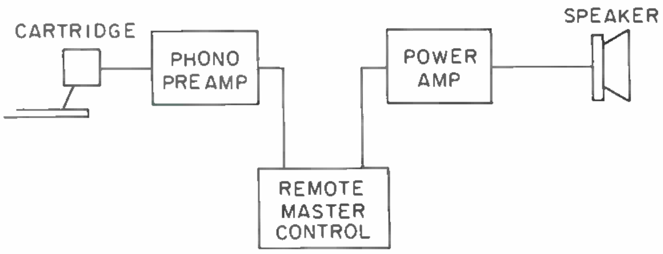



*Figure 3. A complete listening system. The remote master control contains: loudness, volume, bass, treble, musical bass, presence, five position equalizer, and filter.*

up to 2500 Hz. The effect of such a control can be astounding; such a control was at one time called a presence control.

For tonal balance tests, several hundred records, tapes and other music sources were individually adjusted for best tonal balance and the equalization noted. Only a handful of sound sources were balanced well to start with. The various equalization settings were repeated under different conditions and without referral to earlier tests. In analyzing the different equalizations some definite conclusions can be drawn.

The effect of the various frequency bands is given in Table II. All these tests indicate that a better center reference would be 600 Hz, instead of the nice decimal number 1000 Hz. 600 Hz lies closer to the musical energy center, and bass and treble adjustments made with reference to 600 Hz appear to give better tonal balance (400 Hz for male speech). It is interesting to note that in Europe 800 Hz is used for reference.

**Table II**
**Relative Effect of Frequency Bands**
(important bands marked by *)

| Frequency Band Center | Effect |
|---|---|
| 40 Hz and below | Little use except for organ |
| 80 Hz* | Significant bass tones |
| 160 Hz* | Significant bass tones |
| 320 Hz | Can corect weak middle register, but excess responsible for "muddy" bass |
| 640 Hz | Changes volume mostly, suppression of this band is much more pleasing than boosting |
| 1200 Hz* | Increases "presence" of soloists and brilliance of instruments |
| 2500 Hz | Some effect on brilliance of piano |
| 5000 Hz* | Important "noise" component of bass instruments; little effect on musical tones |
| 10 kHz and up | Only slight effect on musical sound |

In summary, it can be said that the four frequency bands marked appear to be the only ones of major interest for tone control, and a more elaborate equalizer is rarely required. It is now necessary to look briefly at the elimination of noise.

## EXPERIMENTS WITH NOISE LIMITING FILTERS

Since noise power is proportional to bandwidth, reducing the bandwidth is a most effective way of reducing noise. In order to preserve as much of the music as possible, a relatively sharp cut-off filter is desirable. For the first tests the extremely sharp filters described earlier were used. Reducing the bandwidth from 20 kHz to 10 kHz should reduce the noise by half, with little effect on the music, because there is no significant musical content in this frequency range. Going from 10 kHz to 5 kHz again cuts noise in half, etc. The effect on the music becomes more pronounced even with a sharp cut-off filter as the cut-off frequency falls into the area of significant tones.

Tests were made with the filter described earlier having cut-off frequencies of 4, 5.5, 7.5, 10, 13.5 and 18 kHz. It was immediately apparent that the cut-off frequencies were spaced too closely to be really effective. Octave spacing is much better; also the frequency range was too high. Subsequently, a filter was built with cut-offs at 3, 6, 12 kHz and flat. The action on noise was good, but the first two bands had a pronounced effect on the musical sound. The filter was flat and had the general shape of Figure (A). Next, the slope was varied and it was found that a slope of 18 dB/octave was nearly as good as far as noise is concerned; also, only the top 20 dB were significant. The effect on the music was pronounced at lower cutoff frequencies as before. A filter as shown in Figure 1(B) behaved similar to 1(A).

Purely by accident, it was discovered that an uncompensated filter with some overshoot, as in Figure 2, had a much more pleasing effect with music. This was further investigated and the optimum Q of the boost portion before the roll-off was 1.41. With the proper boost of about 5 dB, the noise was gone and the music did not appear to have lost substantially. In fact, all the effort spent to compensate the filter and make its response flat before the roll-off was a waste, since for best sound the filter should have a peak before roll-off. It is stated here without proof, that apparently the gain x bandwidth product lost by the filter should be equal to the gain x bandwidth product gained by the peak. Or stated differently, the energy lost is replaced by an extra energy peak preceding the roll-off. Under this condition in listening

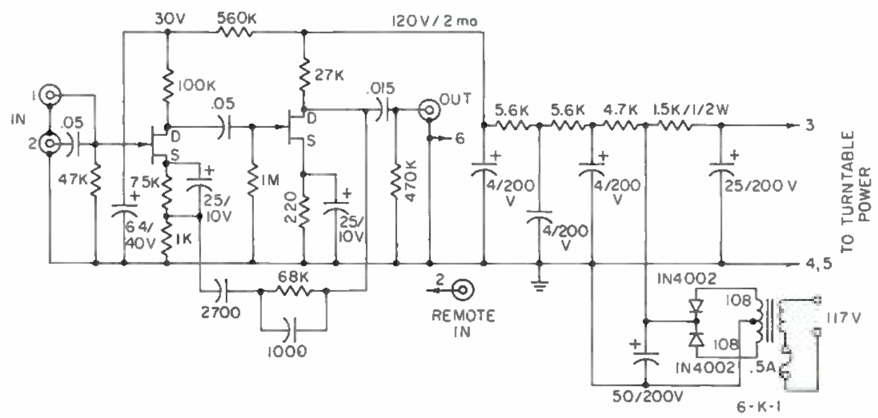



*Figure 4. A phono equalizer using fet's and having RIAA equalization.*

# Stellavox.
# Your best investment in professional sound recording equipment.

All Stellavox sound systems are designed and manufactured in Switzerland (land of engineering perfection) and incorporate the latest, most advanced "state-of-the-art" features* which provide *the* highest quality sound reproduction *every time* (even under the most demanding on-location recording conditions).

- ***Plug-In Head Modules** (Only Stellavox recorders feature the capability of being able to accept all the mono and stereo track configurations in modules that are each individually tailored for ultimate performance on a specific recorder)
- ***Increased Reel Capacity** (With the Stellavox ABR extension accessory, reel capacity is increased up to 10½" for longer recording time)
- ***Battery Operation** (In addition to AC operation, all Stellavox recorders operate on batteries thus permitting the utmost in portability and flexibility)
- ***Professional Accessory Line** (A full selection of professional accessories are available for each Stellavox recorder)

**Quadrasonic SQ-7**
The only professional 4-channel battery operated unit of its kind available today. Wow and flutter (DIN weighted): ≦ ± 0,06%. Overall frequency response: 25 to 25 kHz ± 2 dB. Signal to noise ratio (rel. 800 nW/m): ≦ 66 dB ASA A.

**Stellavox SP-7** The ruggedly built SP-7 is so compactly designed that you'll be amazed at how little it weighs (only 8 lbs. with tape and batteries). Wow and flutter: < ± 0,07%. Overall frequency response: 30 to 16,000 Hz ± 2 dB. Signal to noise ratio, weighted (ASA A filter) > 65 dB stereo and > 70 dB full track (at 50 mMx/mm).

**Stellamaster SM-7**
A totally unique, one-of-a-kind system, the Stellamaster two-channel recorder provides performance abilities unequalled in either studio or portable recorders. Wow and flutter (weighted): ± 0,05% (0,04% typical). Overall frequency range: 20 Hz to 28 kHz ± 2 dB. Signal to noise ratio: 74 dB (Maxell UD 50)

Each Stellavox recorder is backed by warranty service which is available through our service center in Los Angeles. Factory trained technicians, utilizing a full stock of parts and the latest in laboratory measuring devices, will handle all aspects of quality control, maintenance and repair of Stellavox products.

If you would like to receive full technical information on the Stellavox line of professional sound recorders and mixers, please write to:

**Hervic** ELECTRONICS, INC.
1508 Cotner Avenue
Los Angeles, California 90025
(213) 478-5086

*Circle 27 on Reader Service Card*

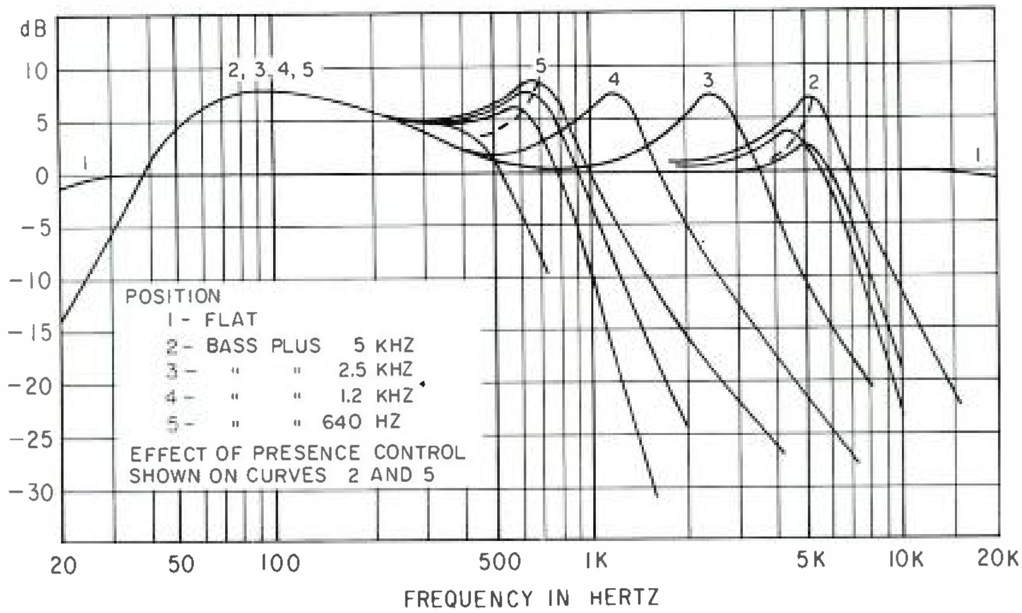


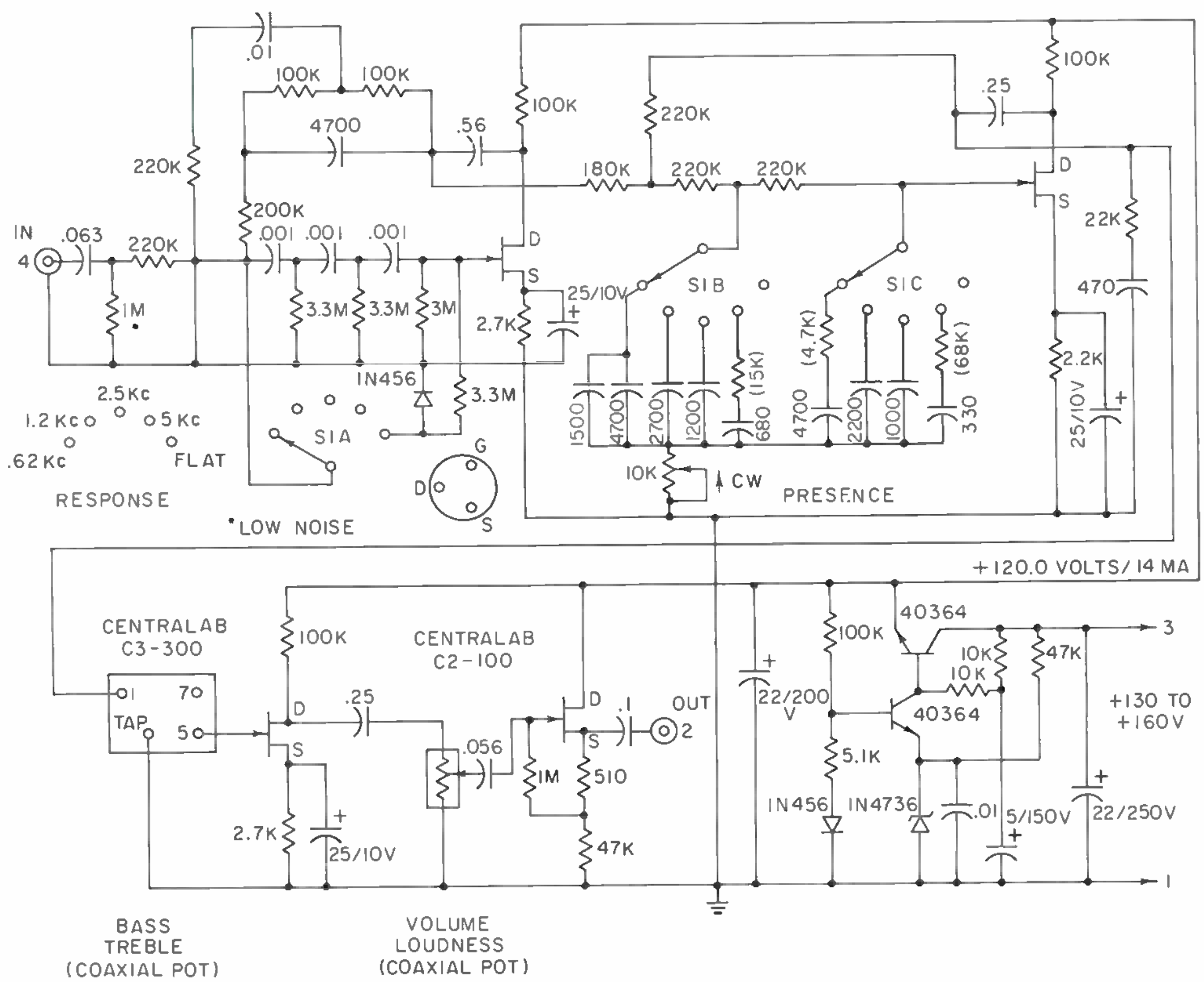

*Figure 5. The various responses of the musician's tone control. Other responses of bass, treble, volume, and loudness are not shown.*

tests, the improvement in musical quality was unbelievable. Further tests with older noisy recordings indicated an optimum cut-off frequency all the way down to 640 Hz! To get any semblance of listenable music at this low cut-off frequency is totally impossible without the peak preceding the roll-off. This is true even for cut-offs around 3 kHz. The peaks want to be about 5 dB, except at 5 kHz, where a smaller peak was desirable with noisy records. As we know, the sound energy lost above 5 kHz is relatively small, and apparently less of a peak is sufficient to compensate for it. For better clarity, we describe briefly the complete system.

## THE OVERALL LISTENING SYSTEM

The complete system is shown in FIGURE 3. In the author's opinion, the tone control circuitry must be adjustable from the listening position. It is impossible to adjust the sound while standing at a cabinet and then walk over to an easy chair in the listening position and expect the adjustment to be still correct. The critical listener

*Figure 6. The schematic of the musician's tone control.*



must be able to adjust the sound while he is sitting in his listening chair. Consequently, the tone control must be small and unobtrusive, connected preferably with a single cable into the system. Another change from the usual is the use of a separate preamplifier. This is almost a necessity, since most cartridges, even with only three feet of connecting cable, show a completely erratic response at the high end. This is easily verified with a test record, or by making a so-called "injection" test, where the audio generator is connected through a simple network to the bottom end of the cartridge. Since the cable often cannot be made shorter and a cable of much lower capacitance is not available, this means that the preamplifier must be built into the turntable. Listening tests showed a remarkable improvement in smoothness of sound. The author's preamplifier is based on high voltage fets and the schematic is shown in FIGURE 4. Only RIAA equalization is provided and no controls are needed, so the unit is hidden easily within the turntable enclosure and wired across the turntable motor. The only precaution is that the power transformer must be located and rotated in a way as not to introduce hum into the magnetic cartridge. This preamplifier also furnishes power to the remote tone control unit described below.

### THE MUSICIAN'S SOUND CONTROL

This control was to incorporate normal bass and treble controls, musical bass, noise filters and different tone equalizations, as well as volume and loudness, not a small task! The normal tone controls of the Baxandall type and the volume/loudness presented no problem. Standard Centralab coaxial controls were used, together with their associated ceramic circuits (C-1-100 and C-1-300). The musical bass, filters, and different equalizations posed a more serious problem. However, in examining the end results, certain redundancies can be found. For example, excellent bass sound was achieved with a boost of about 5 dB at 5 kHz (in addition to low frequency boost of course). If this peak also used for the cut-off filter at the proper frequency, a great simplification might be achieved. Also if this peak were controllable by a separate control in amplitude, a certain amount of presence control would be possible. For the low bass frequencies, the same response was always desirable. Going from a flat response, the bass would be boosted and the very low frequencies attenuated. This bass would be held the same in other switching positions, which changed the bandwidth and high frequency cutoff. The responses of the final control are shown in FIGURE 5 and the schematic of the complete circuit in FIGURE 6. Bass boosting is achieved by dual twin-T circuits set for stagger tuning, since a single tuned circuit of any Q is unsatisfactory. This circuit is followed by a filter circuit based on a phaseshift network with degeneration. The amplitude at the peak is controllable by the 10 kilohm potentiometer in the second stage. Making the responses switchable is much preferred over a continuously variable control. The only parameters lost in this design are the 1 kHz presence control for the 2.5, 5 kHz or flat position. Also lost was the flat position with the addition of the bass boost. This makes a difference above 7 kHz, which we did not feel was important enough to get into a vastly more complex circuit.

As said before, the tonal range and effect on musical balance of this unorthodox but well founded circuit, is rather incredible and must be heard to be believed. ■

#### References

1. Part I of this paper.
2. Audio-Baton by Blonder-Tongue Laboratories.

MICHAEL RETTINGER

# Studio Rumbles

*Here's what happens and what to do when your studio is not as bass quiet as you think it ought to be.*

WHEN A HIGH-QUALITY stiffly supported microphone is placed in an apparently well insulated studio and one listens in the monitoring room to the greatly amplified noise signal, he may hear a pronounced low-frequency disturbance which is not directly audible when he is stationed in the studio itself. The rumble often lies in the range from 30 to 40 hertz, and tends to disappear when the transducer is supported in a highly resilient mounting applied to the floor-placed microphone stand or overhead-supported hanger. It appears, then, that the floor or ceiling is subject to a mechanical vibration which exists at many points of the floor or ceiling, because relocation of the microphone support (with its stiffly mounted microphone) tends to have but a slight effect on the reproduced rumble. In other words, since the floor does not move as a piston, the disturbance is rarely audible in the studio, but can be transmitted into the microphone support.

An examination of the vibration spectrum of such a floor has in two instances confirmed the existence of strong low-frequency components within the concrete slab of ceiling and floor of the studio. It is the purpose of this report to account for the existence of such rumbles even when apparently extensive vibration isolation measures had been taken in the design and construction of the enclosure.

*Michael Rettinger is a consultant on acoustics operating out of Encino, California.*

A floating concrete floor may be likened to a mass supported on a spring—a mechanical system with lumped constants and only one natural frequency.[1] The term "lumped" derives from the fact that the inertia characteristic of the system is concentrated in one part—the mass—and the elastic characteristic is concentrated in another part—the spring. In such a system, the kinetic energy is associated with the moving mass and the potential energy is associated with the deformed spring.

The structural slab under the floating concrete floor is a mechanical system with distributed elements; every part of the slab has both inertia and elastic characteristics and there may be numerous possible independent displacements. Such a mechanical system has many natural frequencies. There are several ways by which the energy in the two forms (kinetic and potential) can be exchanged. At the perimeter of the structural slab, for instance, there must be a node, or place of no displacement amplitude, because the slab is tied into the walls and the steel columns of the building and cannot move there (unlike the overhead isolated concrete floor). Between the perimeters, the slab may move in various ways, exhibiting nodes and anti-nodes dependent on the excitation frequency. When one or more of the natural frequencies of the slab correspond with one or more of the frequencies of the forcing vibration, a condition of resonance exists.

1. The natural frequency of a spring-supported mass, or any other mechanical system having matter, stiffness, and resistance, exists when the displaced mass is allowed to oscillate freely under its own elastic and inertial forces.

# WE JUST MADE BROADCAST MIXING A LITTLE EASIER. AND A LOT LESS EXPENSIVE.

Meet our 30,000-series off-the-shelf broadcast audio consoles. Available in versions from a 5-channel mono mixer to a full 8-channel stereo unit, its compact, 25½ x 12½ x 22" package features big-board styling and comfort on the outside.

And state-of-the-art circuitry inside. At an almost-unbelievable price.

Starting at $750 for a 5-channel mono model, all 30,000-series consoles offer a maximum of performance and reliability at absolute minimum cost.

From the moment you sit at the controls, you see. Maximum operator visibility. Functional control groupings. Even the padded armrest* you'd expect only on a custom console.

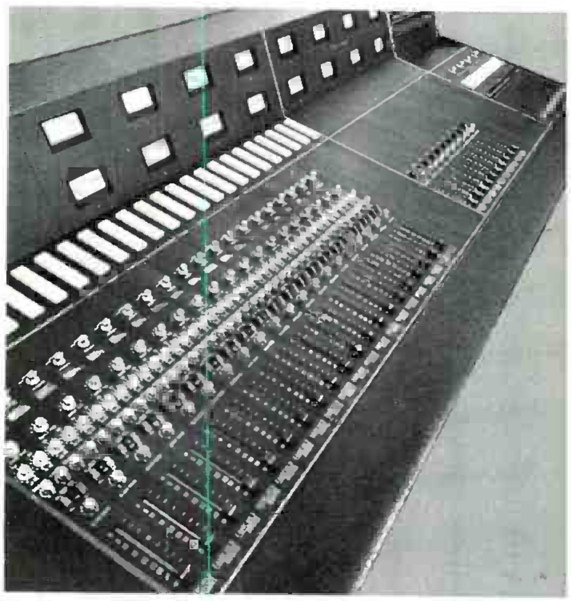
But outside's only half the story. Swing up the easy-access hinged control panel, and there are more surprises. IC op-amp circuitry. Stepless, linear-motion faders with cue switches, in all mixing channels. And modular plug-in PC boards, to let you interchange various input, output, monitor and cue amplifiers. Modifying or expanding the console with your needs.

(If you're familiar with our larger consoles, you'll note we use the same modular boards.)

But without a schematic, you still won't see all the built-in flexibility. Dual independent outputs, for instance, permitting "2-in-1" console operation. While programming "on the air," for instance, you can use the audition channel for pre-listening or recording. Our remote line channels work two-way, as well: for mixi or for cue/talkback. Switched with our monitor inputs are contacts for use with warning lights—or remote control of tape machines and turntables, if you wish

The list of features is considerably longer. But specs and a discussion will be far more convincing. Send for our data sheet, or better yet, call Sales Manager Rick Belmont. You'll see why small news can be big news when it comes from Robins/Fairchild.

*Not available on all models



**We build our small consoles the way we build our big ones.**

ROBINS FAIRCHILD

**The first name-and the last word-in broadcast consoles and equipment.**

A Robins Industries Corporation

75 Austin Boulevard, Commack, N.Y. 11725 (516) 543-5200

*Circle 40 on Reader Service Card*

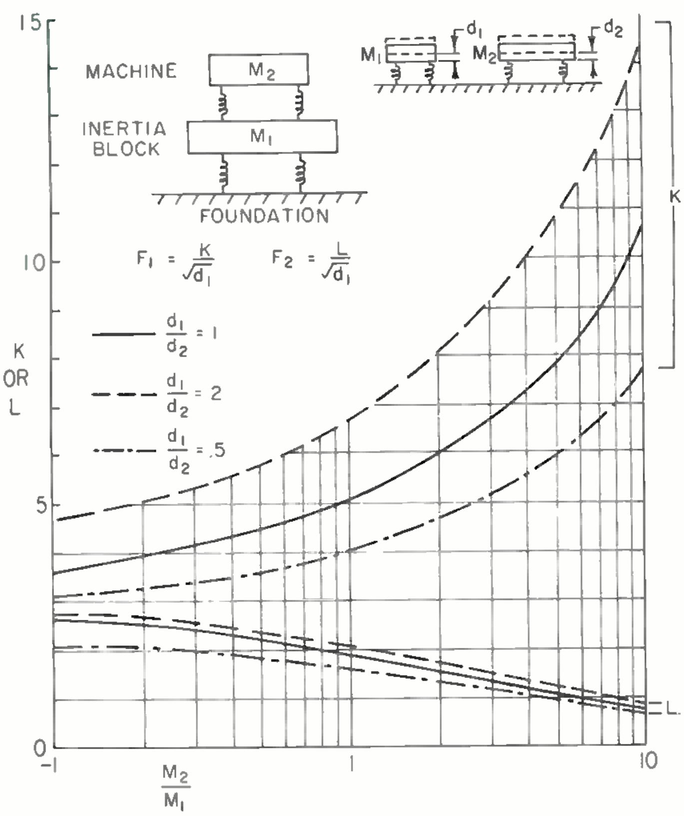



*Figure 1. A floating floor on an elastic body.*

It may at this point be desirable to discuss vibration isolation in a general way. When a mass rests on a spring, which rests on a solid body, the system acts as a low-pass filter when a vibratory force is applied to the mass. The natural frequency of the mechanical arrangement is the cut-off frequency of the filter. Components with a higher frequency are not passed through the spring into the solid but are retained by the mass and cause it to vibrate against its own inertial forces. Exciting forces of lower frequency are, however, transmitted through the spring into the solid. The ratio of force transmitted into the solid body to the force applied to the mass is spoken of as the force transmissibility of the system.

When the solid body itself moves, the system is still a low-pass filter, setting the mass into vibration at frequencies below the natural frequency of the system or the cut-off frequency of the filter. In this instance, one speaks of the displacement transmissibiltiy of the system, which is the ratio of the displacement of the mass to the displacement of the solid body.

The natural frequency of the two systems is the same, namely, $f_o = 3.13/(d)^{1/2}$, where d is the static deflection of the spring in inches, that is, the difference in spring height before and after the mass is applied to the spring.

In the first instance, one seeks to prevent the transmission of exciting forces from the vibrating mass into its foundation, as in the case of a piece of machinery or a man walking on a floor. In the second case one wishes to reduce the transmission of motion of a foundation to a noise-sensitive device resiliently mounted on it.

For the sake of analytic simplicity a floating floor on an elastic body is illustrated on FIGURE 1 as two isolated masses, $M_2$ (the floating concrete floor) and $M_1$ (the structural slab). $M_2$ is coupled elastically to $M_1$ by means of a spring (the elastic pads under the floating concrete floor) and $M_1$ is coupled to the rigid building frame by another spring (its own elasticity). When either mass is placed on its supporting spring (with this spring resting on firm ground), the spring will experience a static deflection, which is given by the difference in the height of the spring unsupported and when the mass is on it.

Such a mechanical system of two degrees of freedom has two natural frequencies which are not the same as when each mass is resting on its spring only, without the other coupled system. Of the two coupled natural frequencies, one is higher than either one of the two uncoupled natural frequencies, and the other is lower. It is generally the higher of these two uncoupled frequencies which, when coinciding with one of the frequencies of the exciting vibration, causes large displacement amplitudes in the lower mass, the structural slab, particularly when little damping exists in the slab. This is generally true for concrete, but not necessarily for macadam.

The vibrations with coupled frequencies $F_1$ and $F_2$ do not exist exclusively in either one of the two moving masses, but alternate between the two. Thus, while for a short time the floating floor may vibrate at $F_1$ and the foundation at $F_2$, a little later the floor will vibrate at $F_2$ and the slab at $F_1$, if the complete exciting force is continuously active.

$$F = \frac{2.21}{\sqrt{d_1}} \left\{ \left[ 1 + \frac{d_1}{d_2} \left( \frac{M_2}{M_1} + 1 \right) \right] \pm \sqrt{ \left[ 1 + \frac{d_1}{d_2} \left( \frac{M_2}{M_1} + 1 \right) \right]^2 - 4 \frac{d_1}{d_2} } \right\}^{1/2}$$

The higher of the frequencies is obtained by using the plus sign in the above equation, and the lower by taking the minus sign.

As an example, consider the case where the floating concrete floor weighs as much as the structural slab, so that $M_2/M_1 = 1$. Let $d_1 = 1/8$ inch and $d_2 = 1/16$ inch, so that $d_1/d_2 = 2$. For this case, the K on the dashed curve of FIGURE 1 is 6.7 and the and the corresponding L is 2, so that the two coupled natural frequencies come to 8 and 27 hertz. The two uncoupled frequencies were $f_1 = 3.13/(.125)^{1/2} = 8.85$ hertz and $f_2 = 3.13/(.0625)^{1/2} = 12.42$ hertz.

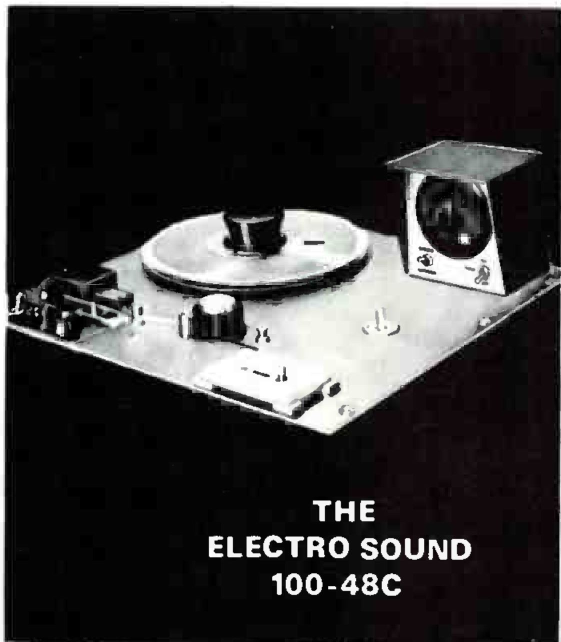
We see, therefore, that although the natural frequencies of the uncoupled systems, that is, the cut-off frequencies of each of the low-pass filters represented by them, were relatively low and outside the audible frequency range, the higher of the two coupled natural frequencies is well within the audible range.

The curves of FIGURE 1 show that the high coupled frequency of the foundation can be avoided by making the ratio of $M_2/M_1$ small, that is, by making the slab heavier than the floating concrete floor, and by making $d_2$ large, that is, by providing relatively soft pads for the floating floor.

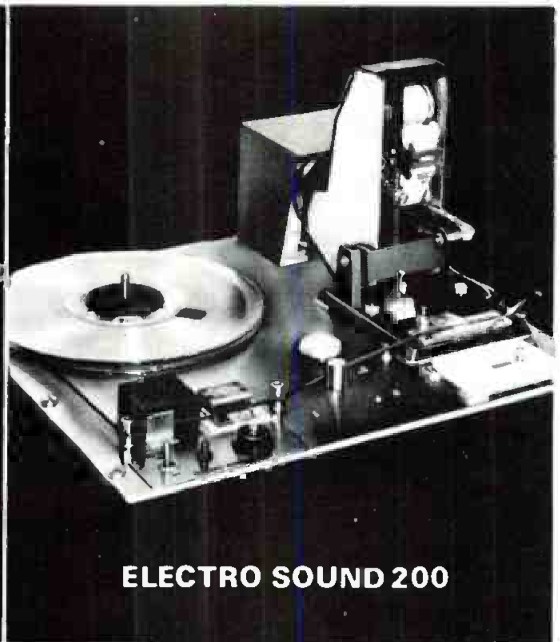
While equal benefits can be achieved by reducing the amplitude of the exciting disturbance, as by providing additional vibration isolation on vibratory equipment within the building, little can often be done when the noise is caused by ground vehicular traffic from a nearby street. In such a case, a highly resilient microphone mounting is the cheapest solution to the studio rumble problem.

RCA, CBS, EMI,
JAPAN, U.S.S.R., NEW ZEALAND, AUSTRALIA,
AND SOUTH AMERICA...

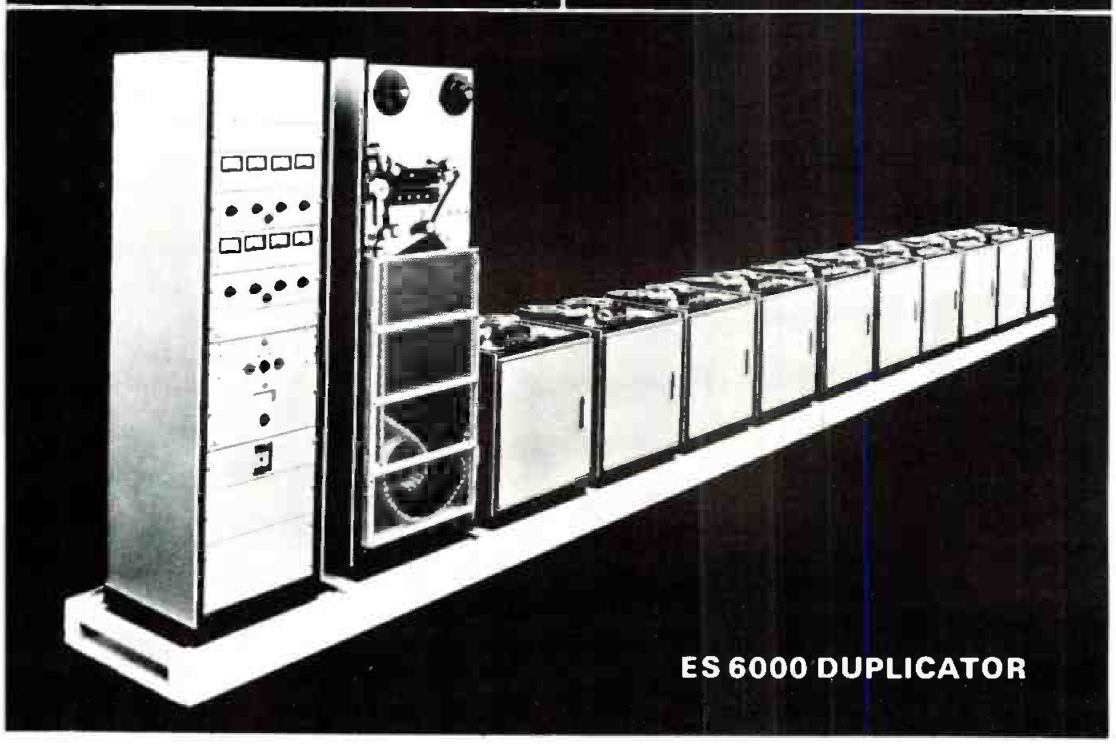
# Audio sells Electro Sound tape duplicating equipment to everybody!



THE ELECTRO SOUND 100-48C



ELECTRO SOUND 200



ES 6000 DUPLICATOR

Tape industry leaders on six continents are duplicating, winding and splicing on Electro Sound machines supplied by Audio.

Here's why:

**THE ES 4000 AND THE NEWER ES 6000**
at 32:1 are the world's most versatile tape duplicating systems. We have all the equipment including all the accessories necessary for a fully functioning duplicating plant.

**THE ES 100-48C**
is the most popular tape winding unit in the world for both cartridges and cassettes.

**THE MODEL 200**
is the only moderate priced automatic tape splicer on the market. It increases production and makes uniform perfect splices every time. (Shown here is the cassette splicer. A cartridge splicer is also available.)

**AND OUR BRAND NEW ES 100-2C**
winder which has two cassette take-up positions plus an automatic splicer is the most reasonably priced and troublefree high speed automatic cassette winder/splicer available.

Write or call Milton B. Gelfand for Electro Sound, the finest in tape duplicating equipment.

**audio**
AUDIOMATIC CORPORATION

1290 AVENUE OF THE AMERICAS
NEW YORK, NEW YORK 10019
TELEPHONE (212) 582-4870
CABLE: AUDIOMATIC/TELEX: 12-6419

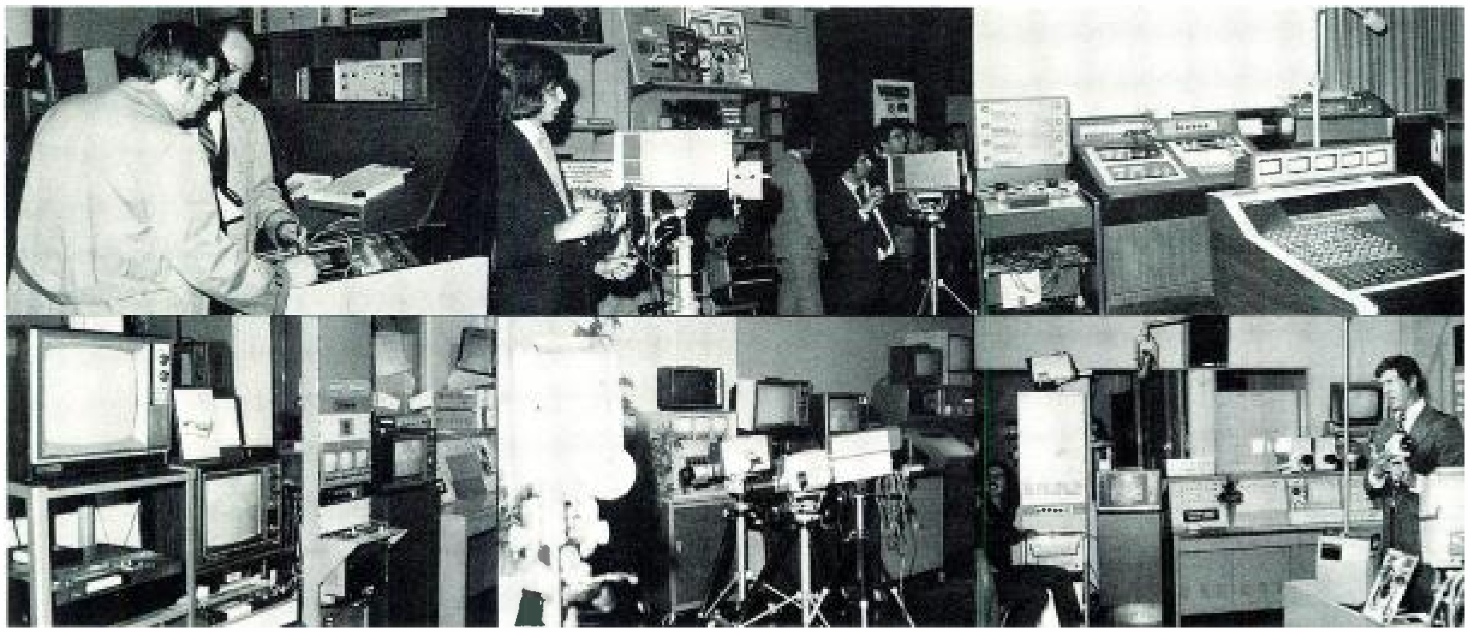
*Circle 25 on Reader Service Card*

# db Visits—Harvey Radio

The professional audio video division of Harvey Radio Co. held an open house at their New York showroom/office complex on May 24th and 25th to give a demonstration of the video equipment presently available in the marketplace. The Harvey offices are located on the second floor of 444 Madison Ave. right in the middle of "Advertising Alley." The suite consists of several working offices plus two operating showrooms where potential buyers may see operating demonstrations of various units. Highlights of the two day show were: a display of the JVC cassette v.t.r., the Panasonic Omnivision ½-inch v.t.r. and the Panasonic cassette v.t.r. and the Sony cassette v.t.r. One of the show's highlights was the Panasonic low cost color camera and special effects generator units which toal under $2,000.

Don Plunkett, the genial vice president of Harvey radio, indicated that it would take about $25,000 worth of equipment to get started in a color v.t.r. studio. Also on demonstration was the Philips ½-inch v.t.r. and the IVC 1-inch unit. And accessory items were in great abundance—the Telestrator was a rather expensive "toy" that everyone just had to try. Plunkett also pointed out the new Delta Time Base Corrector which makes possible the use of ½-inch portable video equipment in news recording and may soon replace 16-mm film which up till now has been the standard for on-the-spot news coverage. In a side alcove, the Harvey staff demonstrated the $3,500 Sony Projection System. This is the first color projection system we have seen.

Audio products are also on display in the showroom but were somewhat in the background during this video oriented show. We noted a wide variety of microphones, consoles, monitor speakers, etc.: which indicate only a small part of the total equipment offered by Harvey Radio. Harvey's warehouse, located in Long Island City—across the East River, stocks a wide selection of pro audio equipment since they are major suppliers to many of the east coast's broadcast and recording studios. ■



*The illustrations, photographed at Harvey Radio's open house show some of the equipment and excitement that were there. The showroom is always there, but the pitchman, Bruce Morrow (New York radio's Cousin Brucie), is not; operating equipment and personalized attention, however, is.*

# StopClock

## A New, Accurate, Digital Timing Instrument for Recording Studios, Broadcasters, and Other Demanding Applications

Timekeeper is proud to introduce a new inexpensive *Electronic* StopClock—a compact instrument featuring an easily-read visual display.

Only 5⅛″ wide x 3¼″ high x 5½″ deep, it uses modern digital circuitry to provide accuracy of a very high order with exceptional long term stability. The large ⅜″ high, seven-segment numerals can easily be read from a distance of more than 15 feet. Maximum count is one hour (59:59:9).

Three remote-mounted push buttons are used for manual control: START, STOP, and RESET. These buttons may be placed in a console, operating desk or any convenient location. The clock may be remotely located in the equipment, or on a desk or table. (A mounting flange is provided.)

Operation is similar to any conventional stopwatch. The green button starts the clock; the red button stops it. The black button resets it to zero. These may be depressed in any order, or all at once, without damage to the clock. If it is desired to start the clock by releasing a button rather than depressing one, the green and black button are pressed simultaneously. When the black button is released, the clock will start automatically.

The accessory plug on the rear panel may be used for all remote operations. All BCD information is available at this plug for accessory units such as digital printers, slave units, etc.

The Model T-1 is supplied for 120 volt 60 Hz operation, in an attractive simulated walnut grained enclosure with a red lucite face. However, 120-volt 50 Hz operation is available at no extra charge if so specified at the time you place your order.

The Model T-1 is priced at $185.00. As with all Timekeeper products, it is fully guaranteed to meet with your complete satisfaction, or your money will be promptly refunded. It is guaranteed for one year against any defects in manufacturing.

The Timekeeper Electronic StopClock is a must. More than a high quality timer—it provides the added convenience of full visual display, high accuracy and stability plus operational flexibility. Order one soon. You will be delighted with it.

**TIMEKEEPER**

**P.O. Box 835, Great Neck, N.Y. 11021**

**Please send me ____ Model T-1 Timekeeper Electronic StopClock(s) at $185.00. N.Y.S. residents add 7% sales tax. Add $1.00 shipping per order. I enclose $________**

**Name**____________________

**Address**____________________

**City**____________________

**State**__________ **Zip**__________

DOUGLAS R. KAYE

# Better Film Sound Using Lavaliers

*The correct use of lavaliers for low-budget film use can materially improve sound quality without adding strain to costs.*

IN THE DAYS when ninety percent of all motion pictures were filmed within the confines of Hollywood's sound stages, there was not a great deal of controversy over the selection or placement of microphones. The cardioid or multi-patterned microphone, placed at the end of a studio boom, was almost universally accepted. But modern technological advances and economic pressures have moved film production out of the sound stages. A considerable majority of feature film shooting is now done on location and, inevitably, in less than ideal acoustical environments.

The industry-wide acceptance of location shooting, and the availability of portable sound and camera equipment have helped to create an altogether new film format: the low-budget feature. These are full length feature films produced for less than $100,000 and sometimes for as little as $25,000. Such productions have little contingency for experimentation in techniques or for duplication of equipment. Although every department is forced into a makeshift operation at one time or another, it seems that the recording of sound is always one of the last considerations. In fact, it is usually the poor quality of the sound track that is the giveaway to the "low budget look."

There have emerged two distinct schools of low budget sound techniques. Perhaps the more common is the "shotgun-microphone-for-everything" school of recording. The other is "lavaliers-or-wireless-microphones-on-everybody." Although the proponents of each technique insist that their way is the best, each will admit that their system is far from perfect and has definite disadvantages. It comes down to a "lesser of two evils" approach.

## ADVANTAGES AND DISADVANTAGES

The shotgun, or interference type, of microphone (the Electro-Voice 642 or Sennheiser 815) is a highly directional microphone which has become increasingly popular in television and film production. Although these microphones can bring truly incredible results under the proper conditions, it is primarily an ignorance of simple acoustics that is at fault in their misuse. The producer of a typical low budget film can usually afford to pay only one man in the sound department—the recordist. This means that if a boom-mounted microphone is used, either the recordist has to hold the boom himself, which keeps him from utilizing a good microphone position and keeps his eyes away from his meter, or somebody's brother-in-law is appointed at the last minute as the boom man. If this instant boom man knows nothing at all we are better off; at least he will avoid the mistakes made by someone with a little knowledge. For, if he "knows" (1) that a shotgun picks up sound from as much as one hundred feet away,

(2) that it is as directionally selective as a laser beam, and (3) that all he has to do is to aim it between the actor's lips and he gets "Hollywood" sound, he's going to get some strange results.

Most shotguns have a reasonably good angle of acceptance of thirty to forty-five degrees. Beyond this, the angle response falls off sharply.

But the polar response is measured in an anechoic chamber and the only place one can practically take advantage of such response is out of doors. Even if you are in the proximity of buildings or paved streets, the acoustic environment is far from anechoic, and our "club-shaped" polar response becomes lost in a jumble of reflections and standing waves. In fact, outdoors, where many recordists favor the use of lavaliers, is where shotgun microphones become the most useful. But in low budget work it is interiors, not exteriors, that pose the greatest challenge.

The typical interior set is a New York apartment living room: maybe fifteen by fifteen by eight feet high with a hardwood floor, brick walls and a plaster ceiling—far from anechoic. In a room this size, the actors are always within a few feet of a magnificently reflective wall. Our novice boom operator yells over, "Don't worry, I know just where to put it."

Worry. The microphone becomes attached to a "fish-pole" (certainly one of the worst gadgets ever invented) and is swung out towards the actor. After changing positions four or five times to avoid the light (those damned eight-foot ceilings!) the microphone is in textbook position: perhaps two feet out and a foot above the actor's head. When the set quiets down and the director calls for rehearsal, the recordist discovers that this plush New York apartment now sounds like Grand Central Bathroom.

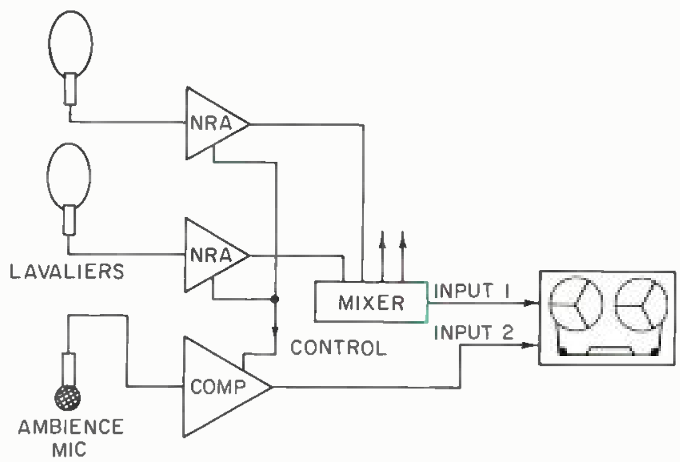



*Figure 1. This is the approach that seemed to solve everything.*

The "laser beam" microphone is picking up not only the actor's voice but also every bit of sound being reflected from the wall behind him.

**REMEDIES**

There are certainly many things that can be done to remedy this situation. First, the microphone can be directed towards the actor's chest rather than towards his mouth. His clothing should help some, but certainly not enough. If time (or money) permits, acoustically absorbent material can be strung up all over the set to reduce reflections from the walls, ceilings and floors, but this has to be re-rigged every time the camera position is changed. Let's be realistic; our low budget feature has neither the time nor the money for such acrobatics so it is probably useless even to begin this procedure. If we start this in the beginning of the scene, and by three o'clock in the afternoon (when we've only completed two shots) the producer orders, "No more sound blankets!," the track from the first two shots is rendered useless because it will not match the sound we get from now on. (I hate being the one to tell a producer that eight hours of effort are down the drain.)

As if bathroom acoustics were not our only problem, we also have to deal with less-than-professional actors who have little or no vocal projection, and with noisy cameras which always seem to be right next to the microphone since that is the only space on the set where there isn't a light stand in position. And there are still the endless, polite explanations to the inexperienced boom man who insists on panning the microphone about like a tennis racket, with matching audible results. Even assuming all of these difficulties, the recordist's headaches are not yet over, for all of these problems are variables and change drastically from take to take, making it nearly impossible to record sound that will match from one shot to another. (Of course the recordist can always mumble, "You can fix it in the mix." But that isn't very nice, is it?)

## AN ALTERNATIVE?

Now I'll get to those of you who have had your hands up for the last ten minutes. You are going to ask about lavaliers: those lovely little miniature microphones that can so easily be hidden in the actors' clothing (when they are wearing any), that eliminate the athletic boom man, and totally isolate us from the acoustical environment. Well, lavaliers have their own set of problems. We've either got to deal with a little wire running down every actor's pants leg into a jumbled mess of cables on the floor and a birdsnest of patches behind the mixer ("No full length shots, please.") or we have to deal with an equal number of transmitters and receivers scattered about the set, giving the impression that we're shooting an episode of *Mission Impossible.*

Contrary to manufacturers' claims, lavaliers are very sensitive to the movement of clothing. A sexy silk blouse sounds like a forest fire in the headphones. This can be minimized by using microphones which incorporate an internal suspension system to isolate the diaphragm from the outside shell, or by taping the layers of clothing together to restrict the movement of the fabric; but neither trick will completely solve the problem.

Professional quality lavaliers have a slightly peaked high frequency response to eliminate the effects of chest cavity resonance and the rather indirect microphone position. However, equalization will still be required at the time of the mix. This really wouldn't be so bad if there were only one actor wearing one lavalier, but each principal actor will have a different lavalier, different clothing and a different anatomy which means that each actor's dialogue will have to be equalized separately. To the editor, this means that the track has to be split up and a separate track made for each actor, and that each word has to be placed on the track assigned to the actor who said it. Aside from creating a pain in the splicer, the results are rarely acceptable, for as the original single sound track is now run through the separate dubbers, a major change in equalization occurs every time another person on the screen begins to speak. Although the e.q. tends to naturalize the voices, the background noise gives the impression that someone is opening and closing a window every time he speaks.

A final but basic and inherent problem with lavaliers arises from their placement so close to the source of sound. Although they are omni-directional in response, they are placed so much closer to the actor than to other sounds such as footsteps, doors, props, and so on, that these other sounds will not be recorded in the proper perspective. Effects and room sounds that are recorded while no one is speaking can be replaced by conventional dubbing, but effects which occur during dialogue cannot be dubbed without also replacing the overlapping dialogue as well in costly and time-consuming looping sessions.

## THE CHOICE

Thus we are left with a choice between shotgun microphones which require patience, understanding and experienced help, and lavaliers which create post-production havoc. All of this makes that expensive shotgun and those lovely lavaliers sound pretty useless, doesn't it? Those of you who are considering taking on the responsibility of low budget film sound recording can just ask someone who has already tried it and you'll find that these conditions are far from rare.

Now to the dilemma, which microphones should the low budget recordist use? With the exception of extremely high noise situations, lavaliers used in the traditional manner are probably inferior to single boom systems when considering the overall balance of time, economy, convenience, post-production and quality of results. However, by combining lavalier and boom techniques, a different philosophy may arise with methods and results far superior to those associated with either traditional approach. The technique is not new; it has been used for decades in recording classical music. However, it is rarely employed in the recording of sound for low budget feature films.

The theory behind this technique was in my mind from

the beginning, but the practical solution came about only after a series of trial-and-error experiments. The discussion that follows includes my experimental failures so that you will appreciate the simplicity of the final solution and understand the theory behind its development.

From this point on, we shall assume that the use of lavaliers or radio microphones is economically and physically practical. (If it isn't, you don't have many alternatives, do you?) Assuming the actors are all wired up and ready to go, let's begin to solve the lavalier's problems. To compensate for the poor proportion between dialogue and sound effects when using lavaliers we shall hang an "ambience" microphone over the set which, if centrally located, will record all sounds in their "realistic" perspective. One should, however, not confuse realistic perspective with the desired perspective. We must increase the dynamic separation between dialogue and noise beyond the realistic range in order to isolate the dialogue and render it understandable. (Films do not yet have the advantage of binaural sound.)

One might think that the use of of an omni-directional or cardioid microphone placed to respond to the entire room would negate the primary advantages of lavaliers: *i.e.*, isolation from unwanted noise and acoustical extremes. But what about those desirable "noises," such as footsteps and props?

If the ambience microphone and all lavaliers are fed through a mixer, the recordist may create many subtle variations in the recording. The level of ambience relative to the level of the dialogue should be judged by many factors. Adding ambience will restore the proper perspective to all sounds in the area (including those sounds made off camera). However, the apparent noise and reverberation will be increased as well. But if the recordist is nimble, and has adequate rehearsal, he can vary the relative mix of these two inputs. When the actor is silent, his lavalier may be greatly attenuated. Besides allowing the ambience microphone to record accurately the effects on the set, this technique can also eliminate unwanted clothing noise while the actor is not speaking. Most actors tend to move less while they are speaking and more when they are silent, thus creating clothing noise primarily during a time when we don't need to keep their microphones open. If there were no ambience microphone, the lavalier would have to be left at full level, for it would be the only input for natural room tone and effects. As the actor begins to speak, his lavalier can be restored to full level, and, if necessary, the ambience level can be lowered at the same tim to reduce the effects of apparent reverberation. This attenuation of ambience level will be unnoticeable, due to the natural masking of this sound by the much louder dialogue input.

## WAS THIS THE SOLUTION?

My initial reaction to this overly-complicated procedure was that it negated the advantages of using a modern recorder with a sixty dB dynamic range. One reason for the existence of such equipment (far beyond the capabilities of release print sound tracks) is that it allows the recordist to set his levels by the maximum level during a scene and leave the controls untouched during the scene, thus avoiding the distracting changes in background noise level that result from any adjustment to the dialogue level. To avoid this seat-of-the-pants mixing (inevitably the recordist is late in opening an actor's microphone channel) my first approach was to automate the mixing, using a system of noise reduction and compression equipment as shown in the accompanying diagram, of FIGURE 1. When an actor speaks, the noise reduction amplifier (NRA) goes to zero attenuation. The presence of this signal also triggers the compressor and attenuates the ambience channel. Any actor's voice will compress the ambience, but it will affect only his own NRA. When no one is speaking, all dialogue NRA's are in attenuation and the ambience is recorded without compression. This really is a handy system and works quite well. I was tempted to stop right there, thinking I had discovered "the" answer to location sound recording except for three problems: (1) not only the actors' voices, but any sound present at any lavalier, including that clothing noise we've been trying to eliminate, will trigger this whole thing, (2) it takes a good deal of time to set this stuff up and adjust each NRA for each actor, and perhaps the most relevant, (3) by the time all this equipment is purchased or rented, I have spent so much money that I could have rented a stage from Universal. But it was a nice idea.

What I needed next was an approach that solved everything. No matter how low a production budget may be, there is little choice in the matter of the final mix. Both the $5,000,000 and the $50,000 one must go to a professional studio for the mix. True, there is a great difference in studios, but any that survive must have a minimum collection of gadgets at their fingertips. Since most studios already have compressors and noise reduction equipment we can take advantage of this equipment at that later stage in production and avoid bringing a mass of studio equipment onto the set. This is perhaps one of the few instances where it is best practice to "fix it in the mix."

## THE BETTER WAY

An approach far superior to using either automatic or manual compression and noise reduction during shooting is to record the ambience on a separate synchronous recorder or on one of the recently introduced stereo sync machines. One advantage of this method is that the ambience or reverberation information remains isolated from the dialogue track and the level of apparent reverberation can be accurately controlled during the mix. As any mixer will tell you, reverberation can always be added but never removed. Ambience recorded on location will also be far superior to the reverb obtained with electrical or mechanical devices. If the ambience microphone is discreetly located on the set, it may be left in one location during the entire filming of the particular scene, and if the ambience is recorded on a separate track from the dialogue, the recording level may be left untouched throughout every take and camera set-up.

## ADVANTAGES

The advantages of such a constant ambience track are enormous. During the mix such a track will completely mask any splices in the dialogue track. (For best results splice the ambience track one frame after the dialogue track throughout the picture to avoid "pops" from simultaneous splices.) True, since we are using lavaliers, the dialogue track will still have to be split into separate tracks for each actor, but the editor need no longer be concerned with trying to find a fill track: a separate fill track is already completed and in sync! The equalization shifts caused by track-splitting will also be far less noticeable, due to masking by the constantly-equalized ambience track.

In essence, the technique described is a combination of the techniques used for recording with a single boom and those for recording with multiple lavaliers. Experience has shown that this combination employs the advantages of both systems and does well to minimize the disadvantages of each. By assigning a specific lavalier to each actor for the entire production period, the quality of each actor's voice can be kept constant throughout the film. This of course could not be done if microphone placement were constantly being changed, as when using a shotgun microphone. If, however, the mixer wishes to alter the sound of one actor's voice or create some unnatural effect, he can work separately with each actor's track or alter only the ambience track. Some very dramatic changes in the acoustics of the set can be made during post-production without destroying the intelligibility of the dialogue.

One final suggestion: when you begin the mix, play around with the ambience track dubber. Advancing or retarding the sync of this track from one-quarter frame to two or three frames can also drastically alter the apparent acoustics of the scene. Some truly remarkable effects can be created. One surprising result comes from advancing the ambience one-quarter frame in relation to the dialogue track. This has the similar (though not identical) effect as having the ambience microphone ten feet closer to the actor in the original recording. If the appropriate degree of advance can be made on the ambience track, the apparent reverberation time can be significantly reduced. (Just be sure that all microphones are in phase during recording.)

Although this technique will never replace a well-handled boom, this two-track technique has the following advantages: No boom operator is required. There will be no interference with the lighting or the camera department. Reverberation in interiors will be significantly reduced. There is no need for acoustical treatment (*i.e.*, sound blankets) on the set. There will be better continuity of sound from take to take and from scene to scene.

As compared to using lavaliers alone, this technique has the following advantages: (1) There is less difficulty with editing out clothing noises. (2) Track splitting is greatly simplified and becomes merely a mechanical operation without the need for hunting for fill material. (3) A room-tone track is recorded (in sync!) without the need to hold up production for this purpose. (4) A smoother final track will result since equalization shifts are made only on the dialogue, not the ambience. (5) The perspective of sounds will be more realistic, yet fuly controllable. (6) Fewer sound effects will require dubbing. (7) Effects such as doors, etc., are recorded in sync instead of wildly and without the need to spend separate production time for such recordings.

## CONCLUSIONS

Over-all, this approach should create a minimum increase in cost. The extra track on a sync recorder and the additional ambience microphone are probably less expensive than the cost of a qualified boom operator. However, the greatest savings will be realized in set-up and rehearsal time. Give each actor a lavalier and leave it alone! Hang up the ambience microphone and leave it there all day! Set a peak ambience level and don't touch it during the entire scene!

The advantages of lavaliers in noisy or extremely reverberant locations are generally accepted. However, the post-production problems created by the traditional method of lavalier use are rarely considered by location recordists. The simple addition of an independent, but synchronous ambience track can, as shown, prove invaluable in post production. This does not mean to imply, please understand, that this method is superior to those obtainable with a boom microphone under proper conditions. We're talking here about low-budget situations where ingenuity takes the place of cash.

Next time you're involved in a low-budget production, try to convince the producer to use this technique. He may think you're a bit strange and that you're only trying to complicate his life, but when he sits down at the final mix, he'll be glad he let you do it this way. ■

# CLASSIFIED

Closing date is the fifteenth of the second month preceding the date of issue.
Send copy to: Classified Ad Dept.
db THE SOUND ENGINEERING MAGAZINE
980 Old Country Road, Plainview, New York 11803

Rates are 50¢ a word for commercial advertisements. Non-commercial and employment offered or wanted placements are accepted at 25¢ per word.

## FOR SALE

BROADCAST AND RECORDING EQUIPMENT: Scully; Metrotech; Langevine; Electrodyne: Q.R.K.; Micro-Trak; M.R.L.; Nortronics; McMartin; U.R.E.I.; used Neumann; E.V.; A.K.G.; Sennheiser; Atlas; Ferrograph; HAECO; Stevenson; Gately; D.B.X.; Advent; Altec; Fairchild; Audio Designs; 3 M; Magnacord; Telex; Inovonics; disc recording systems; package deals; installations; service. **Wiegand Audio, Middleburg, Pennsylvania 17842. (717) 837-1444.**

SPLICE FASTER, BETTER BY SHEARING . . . replaces razor; specify .250-inch or cassette: $16.95. With attached splicing tape mechanism, $24.95. Details- **NRP Box 289, McLean, Va. 22101.**

RECORDING CONSOLE FOR SALE due to expansion of facilities. Custom built; 16-in/8-out; sold new for $40,000, asking $9,000. For more information call **Mark XVI Recording Studios. (901) 526-2194.**

NEUMANN DISC MASTERING SYSTEM; AM 131 lathe; HAECO automation system with variable pitch; Grampion system with table; excellent condition. $4,700 FOB Omaha. **Call Grant (402) 553-1164.**

CEE MODEL DC1542MDA mono cassette duplicator is jealous of my new stereo model and needs a new home. $1,500 will do it. **138 Farber Lane, Williamsville, N.Y., 14221. (716) 632-1975.**

10.5" RECORDER SPECIALISTS. Wanted: used metal 10.5" reels and flanges for ¼" tape in good condition; $.75 each with box; subtract $.15 for no box; occasionally buy salvage reels and tape; write giving condition. **P.O. Box 338, Dunwoody, Georgia 30338.**

FOR SALE: AUTOMATED PROCESSES model 2061 console, 2 years old; 16-in/8-out; $18,500. For detailed information, call **(901) 388-1032.**

PRE-EQUALIZED J.B.L. & ALTEC MONITORS; D.B.X. Companders; Little Dipper hum/buzz notch filters; Cooper Time Cube echo delay; B.B.C. & I.M.F. reference monitors; Lamb "B" Dolby processors; Ortofon and B & O ultra-track cartridges; Schoeps & A.K.G. condensers; Beyer ribbons; U.R.E.I. comp/limiters; Gately pro-kits; Infinity electrostatics; Crown amplifiers/recorders; Tascam, Community Light & Sound fiberglass horns; Q.R.K.; dozens more. Shipped prepaid/insured from **Music & Sound, Ltd., 11-1/2 Old York Rd., Willow Grove, Pa. 19090. (215) 659-9251.**

TASCAM MIXING CONSOLES—$1,890. Tascam ½-inch recorders — $1,770. Shipped prepaid/insured including free alignment and calibration. **Music & Sound, Ltd., 11½ Old York Rd., Willow Grove, Pa. 19090. (215) 659-9251.**

AMPEX, SCULLY, TASCAM; all major professional audio lines. Service, experience, integrity. 15 minutes George Washington Bridge. **Professional Audio Video Corporation, 342 Main St., Paterson, N.J. 07505. (201) 523-3333.**

CROWN TAPE RECORDERS, amps; IMF monitor speakers; AKG professional microphones; UREI limiting, Little Dipper; Fairchild; Crown warranty service station, more. **Barclay, 503 Haverford Avenue, Narberth, Pa. 19072. (215) 667-3048. or 120 Villa Crest Dr., State College, Pa. 16801. (814) 238-1598.**

HAECO announces complete repair service and overhaul for all Westrex cutterheads. Conversions of 3D-II and older models to higher performance standards and reliability. Helium cooling systems and hi-temp coils can protect your investment. Repair insurance program available. Rapid service. Lower cost. **HAECO, 14110 Aetna, Van Nuys, California 91401.**

**MULTI-TRACK**
**8 and 16**
**TRACK RECORDING CONSOLES**
**THE SOUNDEST DOLLAR SPENT IN PRO AUDIO TODAY**
1965 CHEREMOYA AVE.,
HOLLYWOOD, CALIF. 90028
P.O. Box 3187, Hollywood, CA. 90028
(213) 467-7890

LOWEST PRICES for stereo masters expertly cut on a Scully lathe with the Westrex 3D stereo cutting system. Stereo: 12-inch—$33 per side; 7-inch—$14 per side. Mono: 12-inch—$22 per side; 7-inch—$9 per side. We also cut demos—prices even lower. **Trutone Records, 6411 Bergenwood Ave., North Bergen, N.J. (201) 868-9332.**

ONE STOP FOR ALL your professional audio requirements. Bottom line oriented. **F.T.C. Brewer Company, P.O. Box 8057, Pensacola, Florida 32505.**

CROWN CX 822, used. **Barclay (dealer) 503 Haverford Ave., Narberth, Pa. 19072. (215) 667-3048.**

PRE-EQUALIZED J.B.L. MONITORS; Little Dipper hum/buzz notch filters; Cooper Time Cube echo-send-delay; B.B.C. reference monitors; Lamb "B" Dolby processors; Ortofon ultra-track cartridges; Schöeps/A.K.G. condensers; Buyer ribbons; U.R.E.I. comp/limiters; Gately pro-kits; Infinity electrostatics; Crown amplifiers/recorders; dozens more, plus class (A) warranty service station. Shipped prepaid/insured from **Music & Sound Ltd., 11½ Old York Rd., Willow Grove, Pa. 19090. (215) 659-9251.**

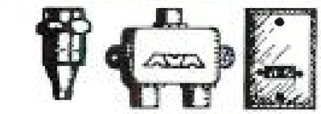
TWO-CHANNEL MONITOR EQUALIZERS for your Altec's and J.B.L.'s are a steal at $150. **Music & Sound, Ltd., 11½ Old York Rd., Willow Grove, Pa. 19090. (215) 659-9251.**

SOLID-STATE AUDIO MODULES. Console kits, power amplifier kits, power supplies. Octal plug-ins—mic, eq, line, disc, tape play, tape record, amplifiers. Audio and tape bias oscillators. Over 50 audio products, send for free catalog and applications. **Opamp Labs. Inc., 172 So. Alta Vista Blvd., Los Angeles, Ca. 90036. (213) 934-3566.**

NEW YORK'S LEADING supplier of professional audio/video equipment and hi-fi stereo components. All major brands in stock. Call for quote—sales—service—leasing—trade-ins. **Martin Audio, 320 West 46th Street, New York, N.Y. 10036. Telephone: (212) 265-6470.**

AMERICA'S LARGEST SELECTION of new and used broadcast and recording equipment! Latest bulletins available. **The Maze Corporation, P.O. Box 6636, Birmingham, Ala. 35210.**

WHATEVER YOUR EQUIPMENT NEEDS —new or used—check us first. Trade your used equipment for new. Write for our complete listings. **Broadcast Equipment & Supply Co., Box 3141, Bristol, Tenn. 37620.**



**CATV—MATV PRODUCTS**

- CONNECTORS
- SPLITTERS
- WALLPLATES
- AMPLIFIERS

**AVA ELECTRONICS CORP.**
416 LONG LANE, UPPER DARBY, PA. 19082
215—MA 6-9100

AUDIO EQUIPMENT, new and used; custom consoles built to your specifications using the components of your choice. Whether you're building a new studio or remodeling your present one, check us first for a package price. **Amboy Audio Associates, 236 Walnut St., South Amboy, N.J. 08879. (201) 721-5121.**

## EMPLOYMENT

MEDIASOUND has an opening for an experienced disc-mastering engineer. Call or write **Mediasound, 311 W. 57th St., New York, N.Y. 10019. (212) 765-4700.**

**WANTED: RECORDING ENGINEER**
**$12,000 - $18,000/Yr. Negotiable**

*Do you have a total knowledge of all aspects of audio recordings?
*Can you appreciate all forms of rock and soul and get along with all types of personalities?
*Can you take raw musical talent and convert it into a sellable product on tape?
*Do you know the sound of a hit? Do you want to cut hits? Do you want success badly enough to eat every top selling single and LP you're not on?
*In short, are you a bom winner?

If you can honestly answer "yes" to all the above, we want you to join us and we'll pay whatever's fair. Track Recorders has had eight national chart records in the last year. Washington, D.C. is the last major music frontier and we're the leaders. Our studio has all the standard quality equipment — 3M 16-track, 25-in/16-out custom console, EMT reverb, JBL 4320 monitors, Dolby, Kepex, varispeed, grand piano, Hammond B3 organ, amps, drums, excellent test gear and maintenance. Your weekends will generally be free. The Washington area offers great entertainment plus Blue Ridge Mountains, Shenandoah Valley, Chesapeake Bay, Atlantic Ocean.

Call or write to: **TRACK RECORDS, INC., 8226 Georgia Ave. #11-2, Silver Spring, Md. 20910. (301) 589-4349.**

# MODERN SOUND REPRODUCTION

**By Harry F. Olson**

**Retired Staff Vice President**
**Acoustical and Electro-mechanical Research Laboratory**
**RCA Laboratories**
**Princeton, N. J.**

Dr. Olson has written a basic text that presents a detailed technical exposition on the significant and essential elements and systems of modern sound reproduction for a wide range of readers, including scientists, engineers, technicians and audio enthusiasts.

The major portion of the book employs simple physical explanations and descriptions which can be read and understood without any special training in engineering, physics, or mathematics. However, for the trained engineer and scientist, the book is backed up by technical descriptions of the action, performance, and characteristics of sound-reproducing systems, including the use of dynamical analogies.

Detailed explanations cover the elements employed in modern sound-reproduction, microphones, amplifiers, loudspeakers, earphones, magnetic tape systems, radio, phonographs, film sound, television and sound reinforcement. Included are monaural, binaural, stereo and four-channel systems. The subject of room acoustics as applied to studios, theaters and auditoriums is also covered.

This is a highly recommended text and will serve as a useful tool and reference for all those working in the audio field. 328 pages (Plus index), $17.50

**SAGAMORE PUBLISHING CO., INC.**
**980 Old Country Road**
**Plainview, N. Y. 11803**

**Please send me ____ copies of MODERN SOUND REPRODUCTION by Dr. Harry F. Olson. My full remittance in the amount of $........ is enclosed. (N. Y. State residents add 7% sales tax)**

Name: ____________________

Address: ____________________

City: ____________________

State: ____________ Zip Code: ____________

**As a service to our readers we are pleased to offer books from prominent technical publishers. All prices listed are the publishers' net. Shipping charges are included.**

**To order use the coupon at the bottom of the page. Indicate quantity on the special instructions line**

**If more than one copy of a title is wanted. Full payment must accompany your order. We cannot ship c.o.d. Checks or money orders should be made payable to Sagamore Publishing Company, Inc. Because of the time required to process orders, allow several weeks for the receipt of books.**

# BOOKCASE

**20. The Audio Cyclopedia (2nd ed.).** *Dr. Howard M. Tremaine.* New and updated, here is the complete audio reference library in a single volume. It provides the most comprehensive information on every aspect of the audio art. This new edition includes the latest audio developments including the most recent solid-state systems and integrated circuits. It covers all subjects in the fields of acoustics, recording, and reproduction with more than 3400 related topics. Each topic can be instantly located by a unique index and reference system. More than 1600 illustrations and schematics help make complicated topics masterpieces of clarity. 1760 pages; 6½ x 9⅜ hardbound. **$29.95**

**1. The Technique of the Sound Studio.** *Alec Nisbett.* This is a handbook on radio and recording techniques, but the principles described are equally applicable to film and television sound. 264 pages; 60 diagrams; glossary; indexed; 5½ x 8½; clothbound. **$14.50**

**7. Acoustical Tests and Measurements.** *Don Davis.* Provides solid understanding of the entire subject of acoustical measurements; based on actual field test work, using commercial equipment. 192 pages; 5½ x 8½; hardbound. **$6.95**

**8. Handbook of Electronic Tables & Formulas, (3rd edition).** A one-stop source for all charts, tables, formulas, laws, symbols, and standards used in electronics. Includes an 8-page, full-color fold-out chart showing latest FCC allocations for the entire frequency spectrum. 232 pages; 5½ x 8½; hardbound. **$5.50**

**Sagamore Publishing Company, Inc.**
**980 Old Country Road,**
**Plainview, N.Y. 11803**

Please send me the books I have circled below. My full remittance in the amount of $________ is enclosed. N.Y. State residents add 7% sales tax.

**1 2 3 4 5 6 7 8 9 10 11**
**12 13 14 15 16 17 18 19 20**
**21 22 23 24 25 26 27 28 29**
**30 31 32 33 34 35 36 37 38**
**39 40**

Name ____________________

Address ____________________

City ____________________

State ____________ Zip ________

Special Instructions ____________________

Canada and foreign: Add $1.00 per book

**24. Basic Electronic Instrument Handbook.** *Edited by Clyde F. Coombs, Jr. Hewlett-Packard Co.* A basic reference background for all instruments. Offers saving in time and effort by having complete information in one volume on how to get the most benefit from available devices, how to buy the best instrument for specific needs. Reduces chances of costly errors. Ideal reference book, it is an excellent source for the beginner, technician, the non-electrical engineering man, or general non-engineering scientific and technical personnel. 800 pages. Hardbound. **$28.50**

**25. Operational Amplifiers-Design and Applications.** *Burr-Brown Research Corp.* A comprehensive new work devoted entirely to every aspect of selection, use, and design of op amps—from basic theory to specific applications. Circuit design techniques including i.c. op amps. Applications cover linear and non-linear circuits, A/D conversion techniques, active filters, signal generation, modulation and demodulation. Complete test circuits and methods. 474 pages. **$15.00**

**26. The Design of Digital Systems.** *John B. Peatman.* Textbook for students desiring to develop a creative approach design capability through digital systems approach. Answers these question: Under what circumstances it is desirable to implement a system digitally? What are some of the components available for implementing the system? How do we go about designing it? 448 pages. **$15.50**

**31. Solid-State Electronics.** *Hibberd.* A Basic Course for Engineers and Technicians. An extremely practical reference book for anyone who wants to acquire a good but general understanding of semiconductor principles. Features questions and answers, problems to solve. 1968. 169 pp. **$9.95**

**32. Circuit Design for Audio, AM/FM, and TV.** *Texas Instruments.* Texas Instruments Electronics Series. Discusses the latest advances in design and application which represent the results of several years research and development by TI communications applications engineers. Emphasizes time- and cost-saving procedures. 1967. 352 pp. **$14.50**

**35. An Alphabetical Guide to Motion Picture, Television, and Videotape Productions.** *Levitan.* This all-inclusive, authoritative, and profusely illustrated encyclopedia is a practical source of information about techniques of all kinds used for making and processing film and TV presentations. Gives full technical information on materials and equipment, processes and techniques, lighting, color balance, special effects, animation procedures, lenses and filters, high-speed photography, etc. 1970. 480 pp. **$24.50**

**40. Radio Transmitters.** *Gray and Graham.* Provides, in a logical, easy-to-understand manner, a working knowledge of radio transmitters for quick solution of problems in operation and maintenance. 1961. 462 pp. **$16.00**

**23. Wide Screen Cinema & Stereophonic Sound.** *M.Z. Wystozky.* First published in USSR in 1965 this excellent English translation covers wide gauge films, panoramic films, circular panoramic cinematography; technical fundamentals of stero sound recording for film, as well as details of the Soviet systems now in use. 284 pages. **$15.00**

**33. Noise Reduction.** *Beranek.* Designed for the engineer with no special training in acoustics, this practical text on noise control treats the nature of sound and its measurement, fundamentals of noise control, criteria, and case histories. Covers advanced topics in the field. 1960. 752 pp. **$19.50**

**16. Magnetic Recording.** *Charles E. Lowman.* Reference guide to the technology of magnetic recorders used in audio recording, broadcast and closed-circuit TV, instrument recording, and computer data systems. Includes latest information on cassette and cartridge recorders; TV recorders; direct and FM signal electronics from low to wideband; servo-control and signal record/playback circuitry; capstan, reel, and head-drum servos for longitudinal, rotary, helical-scan, and disc recorders. Glossary, index, bibliographical information. 274. pp. **$14.50**

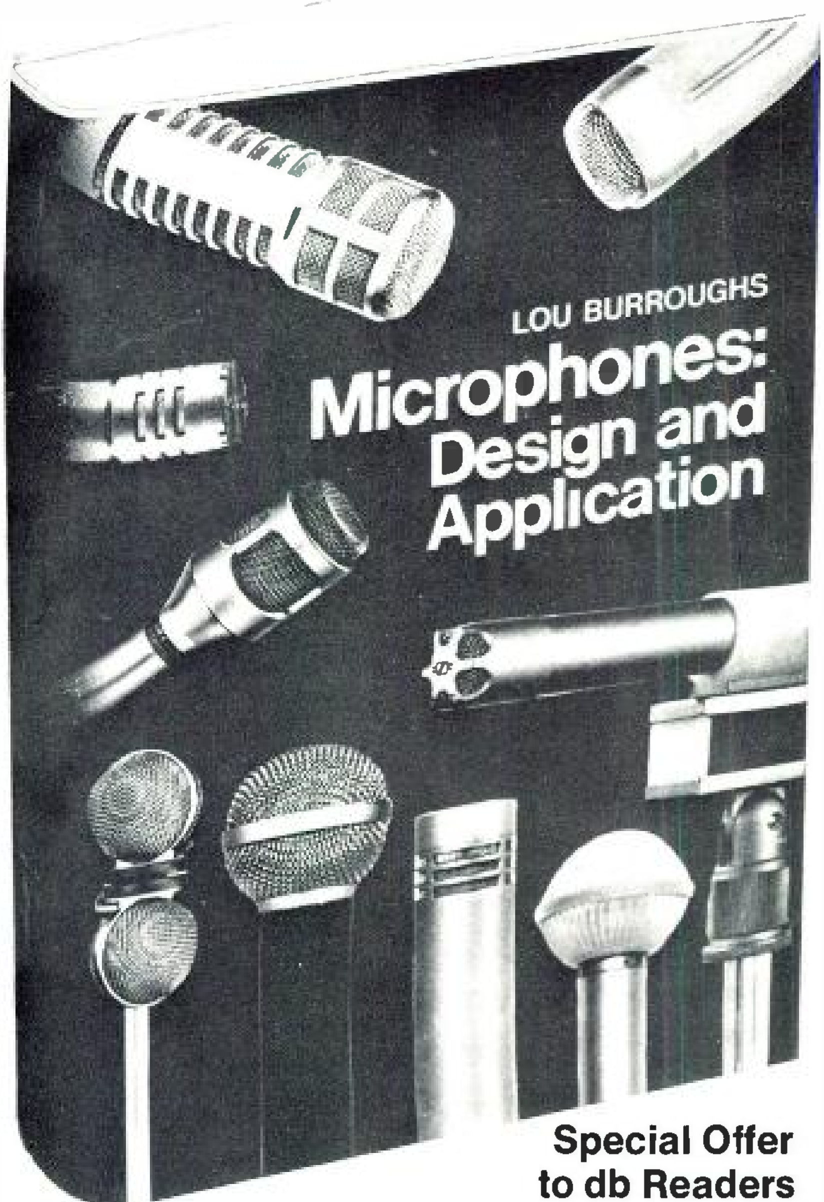
**28. Environmental Acoustics.** *Leslie L. Doelle.* Applied acoustics for those in environmental noise control who lack specialized acoustical training. Basic information in comprehensible and practical form for solving straightforward problems. Explains fundamental concepts; pure theory minimized. Practical applications stressed, acoustical properties of materials and construction listed, actual installations with photos and drawings. Appendixes illustrate details of 53 wall types and 32 floor plans and other useful data. 246 pgs. **$18.50**

**13. Acoustic Design & Noise Control.** *Michael Rettinger. 1973.* NEW, revised and enlarged edition covers physics of sound, room acoustics and design, noise and noise reduction, plus noise and its problems. Many charts and graphs. A practical and useful book. 562 pages. **$22.50**

**22. Acoustics of Studios and Auditoria.** *V.S. Mankovsky.* Basic theory plus a mass of design data covers the field with special reference to studios and places of public performance. For acoustical designers and specialists in sound transmission in cinema and broadcasting. Features exhaustive treatment of studio acoustics by the statistical, geometric and wave methods in parallel. 416 pgs. **$15.00**

# The most important microphone book ever published.

## Covers every significant aspect of theory and use from A to Z!




**Special Offer to db Readers Save $6.50!**

At last, the whole field of microphone design and application has been prepared and explained in one concise, fact-filled volume by one of audio's outstanding experts. This soon-to-be published book is complete, up-to-the-minute and so full of useful information, we think you'll use it every time you face a new or unusual microphone problem.

Sagamore Publishing Co. (publishers of **db The Sound Engineering Magazine**) is so convinced that this authoritative manual will meet the needs of our industry that we are offering it exclusively to our readers in a special, money-saving offer with a money-back guarantee.

### Perfect for Reference or Trouble-Shooting

The twenty-six fact-packed chapters in this indispensable volume cover the whole field of microphones from theory, physical limitations, electro-acoustic limitations, maintenance and evaluation to applications, accessories and associated equipment. Each section is crammed with experience-tested, detailed information. And everything is arranged for easy reference because this is one handbook you'll turn to again and again. Whatever your audio specialty—you need this book!

Along with down-to-earth advice on trouble-free microphone applications, author Lou Burroughs passes on dozens of invaluable secrets learned through his many years of experience. He solves the practical problems you meet in day-to-day situations. For example:

- **How does dirt in the microphone rob you of response?**
- **Which mic would you pick for a large auditorium?**
- **How are omni-directional mics used for orchestral pickup?**
- **When would you choose a cardioid, omni-directional or bi-directional mic?**
- **How do you space your microphones to bring out the best in each performer?**

### Author

Lou Burroughs is widely known for his pioneering work with Electro-Voice and is one of the universally recognized experts in the field. He helped design and develop many of the microphones which made modern broadcasting possible. In fact, he holds 23 patents on electro-acoustical products! Lou Burroughs knows microphones inside out. This book is based on his many years of research, field studies and lectures given throughout the world.

### Exclusive Pre-Publication Offer

**Microphones: Design and Application** will be offered to the general public at $20.00. But for a limited time, **db** readers can buy this same book at the special introductory price of $13.50 (including shipping and handling)—a saving of $6.50.

To take advantage of this reduced rate, your order must be received before October 30, 1973. Just send your check with the coupon below and we will reserve your copy at the special pre-publication price.

## PRE-PUBLICATION RESERVATION

**Sagamore Publishing Co. Inc. 980 Old Country Rd. Plainview, N.Y. 11803**

Please reserve ( ) copies of MICROPHONES: DESIGN AND APPLICATION at $13.50 each postpaid. Rush shipment upon publication.

Name ______________________ **Total amount $** ________

Address ______________________ **New York State Residents add 7% tax $** ________

City ________ State ________ Zip ________ **Enclosed is check for $** ________

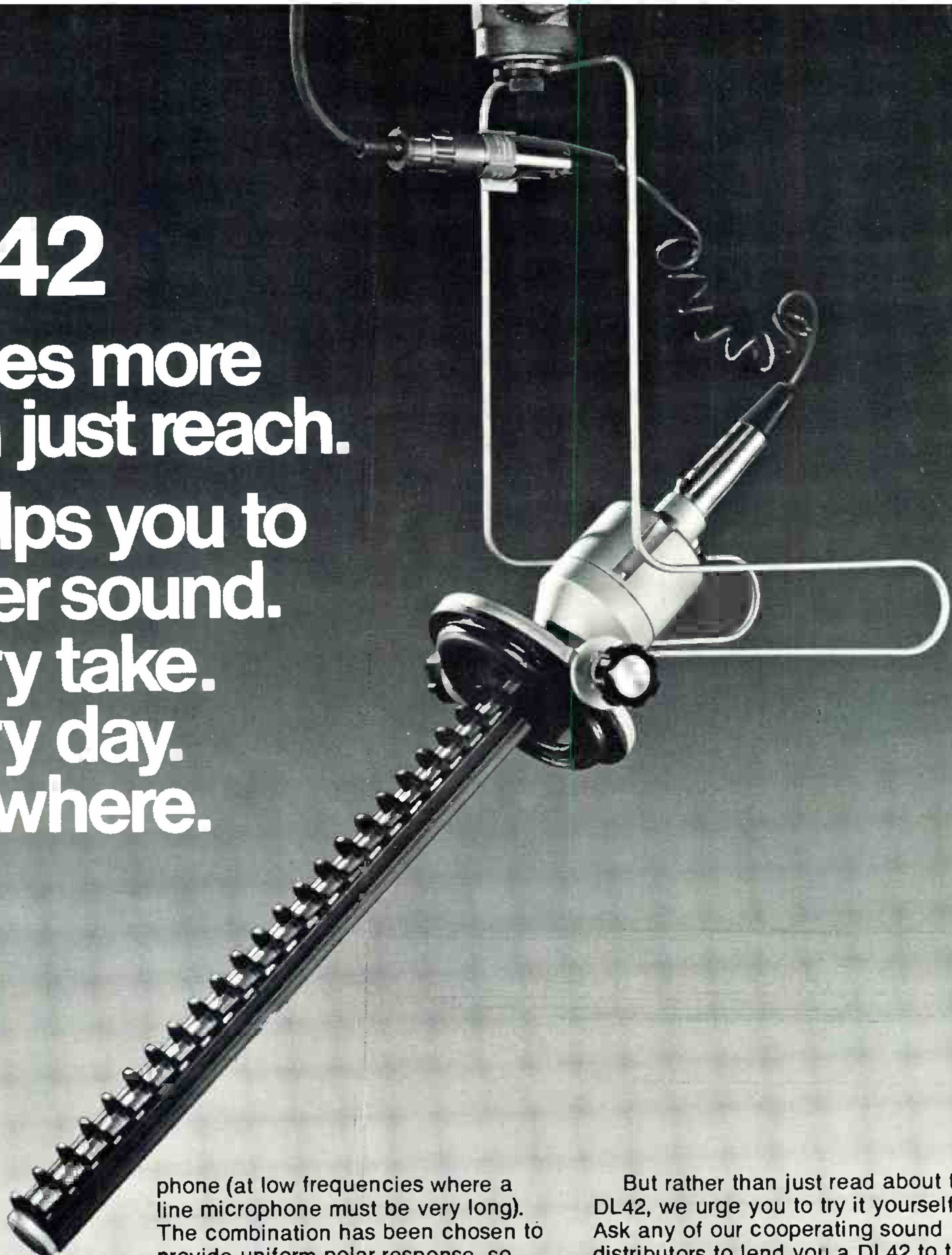
**Foreign orders add $1.00 postage & handling.**

**Please Note: THIS PRE-PUBLICATION ORDER FORM IS VALID TO OCTOBER 30, 1973, AFTER WHICH THE PRICE WILL BE $20.00.**

# DL42

## It does more than just reach.

## It helps you to better sound. Every take. Every day. Anywhere.



Since 1962 when the E-V Model 642 earned an Academy Award certificate for contributions to motion picture sound, there have been a lot of attempts to better this pioneering design.

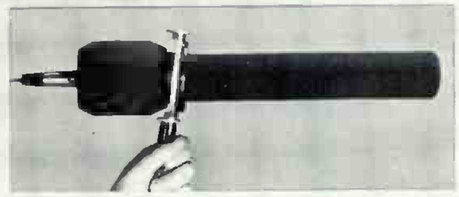
Admittedly, some similar-appearing microphones were lighter and a bit smaller. But often it was at the expense of uniform polar response and wide range. Now there's a long reach microphone that's both smaller and lighter, yet maintains wide range response even off mike.

It's the E-V DL42.

Weight has been reduced to 1/4 of the 642 . . . even less than most highly-directional condenser microphones. And size has been shaved wherever possible. The DL42 is a unique combination of line microphone (at the high frequencies) and cardioid microphone (at low frequencies where a line microphone must be very long). The combination has been chosen to provide uniform polar response, so important to consistent sound quality. Off-mike pickups even sound good (although lower in level), a particular advantage to documentary units and free-wheeling shows where the unexpected is always happening.

The good pickup quality off axis has another practical benefit in the studio. Because maximum rejection is at the sides (where most of the noise comes from) you can work at a remarkable distance when necessary. So when a long shot is called for, the DL42 can be moved upward and cover the entire area with good quality and level. In fact the DL42 is more like a *super*-super-cardioid. And it covers with less racking and panning of the microphone than you ever needed before.

But rather than just read about the DL42, we urge you to try it yourself. Ask any of our cooperating sound distributors to lend you a DL42 to try on your next production. No cost or obligation to serious professional users. Write today for a DL42 technical data sheet. Good reach with good sound. We think you'll like what you hear.



Electro-Voice®

a Gulton COMPANY

Model DL42 Cardiline® Dynamic unidirectional microphone including shock-mounted bail, noise-isolating coil cord, screw-on handle for hand-held use, Acoustifoam™ 2-piece windscreen, and carrying case. $300 suggested professional net.

ELECTRO-VOICE, INC., Dept. 931BD, 686 Cecil Street, Buchanan, Michigan 49107

*Circle 11 on Reader Service Card*