The Sound Engineering Magazine

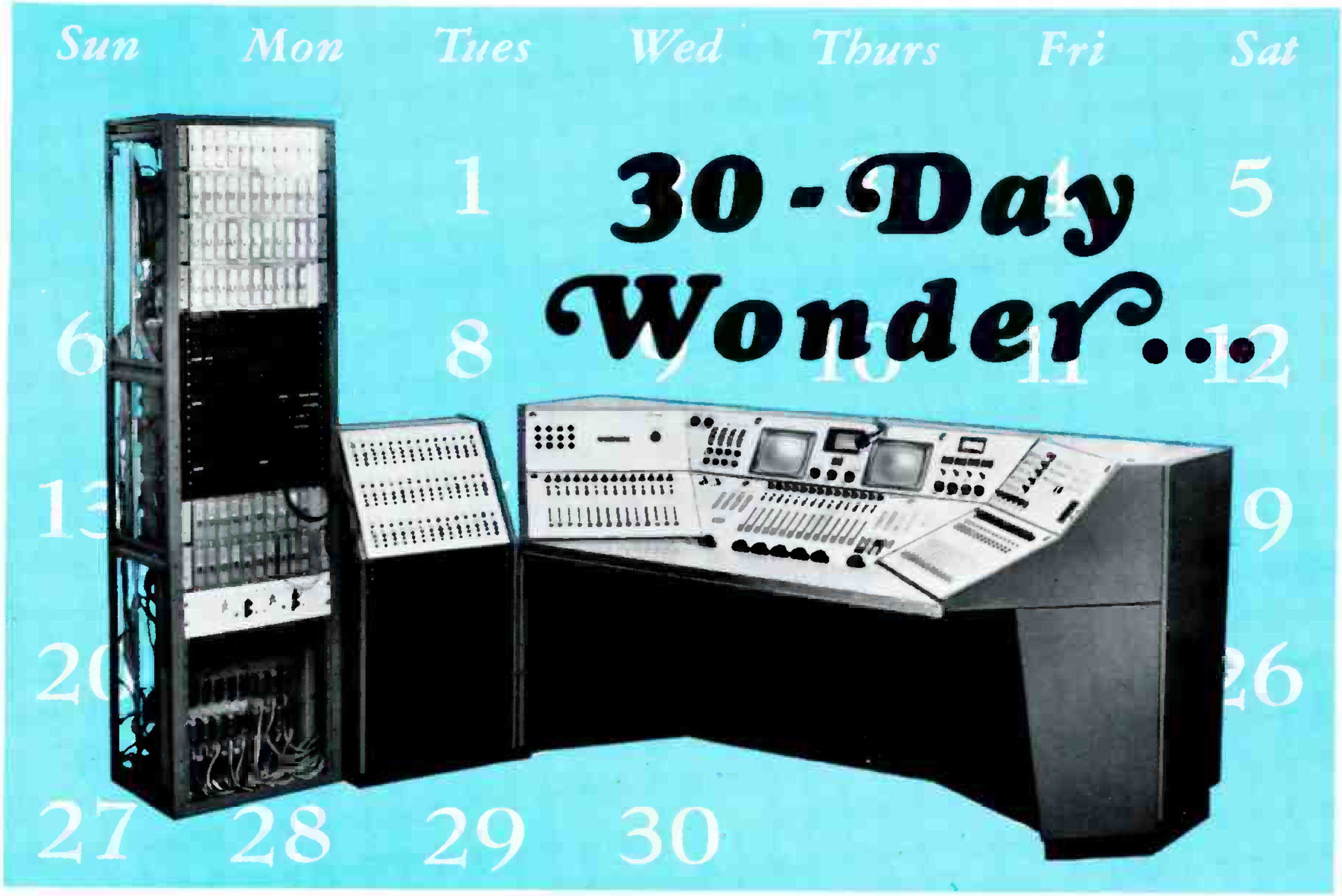
# db

DECEMBER, 1967 60¢

# Studio Operations

## Consoles, Monitor Speakers, Mixing, Compression

PLUS
a picture story on
the AES Exhibition




# New Fairchild Integra II Remote Control Custom Audio Console!

A major TV network selected FAIRCHILD RECORDING EQUIPMENT CORPORATION, from among several of the largest broadcast equipment manufacturers in this country, to design and construct a 42-input audio mixer console. Not only did FAIRCHILD deliver a remote control mixing console in substantially less than the required time, but the network's audio engineers were so deeply impressed with the INTEGRA II console's performance and compactness that additional consoles were ordered ... the next INTEGRA II console was constructed and delivered in thirty days. These consoles were installed and in operation within a matter of days after delivery.

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THE SOUND ENGINEERING MAGAZINE

December 1967 Vol. 1, No. 2


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# Coming Next Month

January's **db** will be devoted to the sound reinforcement field.

Martin Dickstein has prepared an analysis of the cream of the sound systems that went into Expo 67. Four expositions are examined for their audio-visual approach and the techniques used.

Ampex' recent entry into the manufacture of sound-reinforcement equipment has prompted an article from them on their building-block concept of design.

Robert Kerr of DuPont writes about his experiences with theater sound system installations and operation.

A new monthly column appropriately makes its bow in January. Written by Philip C. Erhorn the subject matter will be—Sound Reinforcement.

Plus: the next installment of George Alexandrovich's Handbook making its debut in this issue. And The Feedback Loop, New Products and Services, People, Places, Happenings, and others. Next month in **db The Sound Engineering Magazine.**

# About the Cover

A close look at a modern studio console can provide an almost surrealistic view. The subject at hand is an Electrodyne console photographed at the 1967 AES Exhibition. A full picture story on the exhibition and a more normal view of the cover subject begins on page 25.

# Letters

The Editor:
Congratulations upon your premiere issue of **db, The Sound Engineering Magazine.**

I am certain that the excellent, most informative articles will be appreciated by everyone in the business and the magazine may well become the standard of the industry.

Thank you for my opportunity to subscribe.

*Lawrence L. McQuown*
*Century Custom Recording Service*
*West Palm Beach, Florida*

The Editor:
Congratulations and best wishes on the new publication! I strongly agree with your editorial position that there is a need for a strong, practically oriented publication aimed at the professional audio practitioner.

As someone working in the audio "hinterlands," that is at a great distance from active AES sections, I have long felt the need for a publication that can offer some solutions to the day-to-day problems I encounter. A quick review of your Editorial Board and proposed articles convinces me that **db** has a great potential. I will anxiously await your next issue.

*Dick Bailey*
*Century Custom Recording Service*
*Columbia, Missouri*

*We appreciate these two letters and will certainly try to live up to the expectations stated.*

The Editor:
Thank you for your premiere issue. I have returned my card asking that I receive the publication on a regular basis. I've read and re-read your fine publication today, and find it most interesting and informative.

As the manager of this station I am concerned with our general sound as well as many other sound applications. For instance, our remote broadcast sound, both feed and live at location—response of our FM mobile news units—and the system we are currently developing to do away with loop-fed remotes and instead use an RPB system for both news and live music remote coverage. I am also interested in developing a versatile, high-quality wireless mike system at a reasonable cost. One that will work with our FM mobile units acting as a relay, but with the capability of remotely keying the mobile relay.

I'm interested in better audience participation sound via telephone live on the air, and/or taped from telephone for news playback. I always want a better, more versatile sound in my studios and visitors' areas, as well as facilities for feeding a quality sound over the 'phone to "audition" commercials and programs to prospective clients.

I'm interested in special recording studio effects such as filter mikes, reverberation, echo, compression, and repeating sounds, in electronically generated sounds, and many other areas.

Yet with all these needs, I have little or no engineering knowledge. So I hope that your magazine will develop a special segment with folks like me in mind. Folks who operate a small station with a small staff who must on occasion make minor sound equipment repairs, change systems, upgrade as funds allow, and specify or approve every item of equipment used.

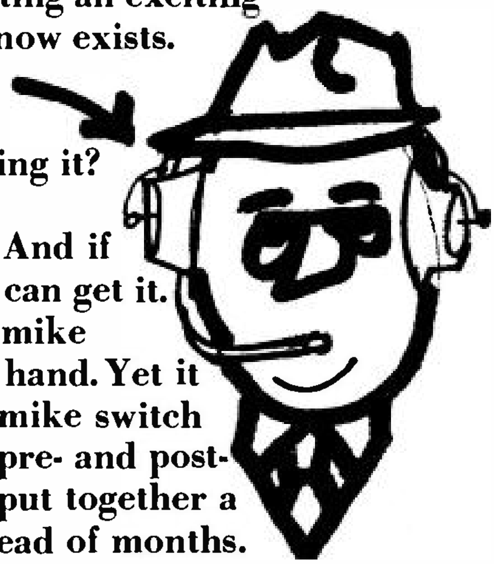
P.S. How about a complete issue on the regular production studio of commercial radio stations which usually doubles as an emergency studio. Then another issue on RPB, remote pickup broadcasting, covering two-way gear and how to acquire a better grade of sound with it—including live sound on location, mobile units, etc. Surely there are many stations that will gladly share with the industry their ideas now in use.

*Robert E. Pickett Jr.*
*Station Manager*
*WEEW*
*Washington, North Carolina*

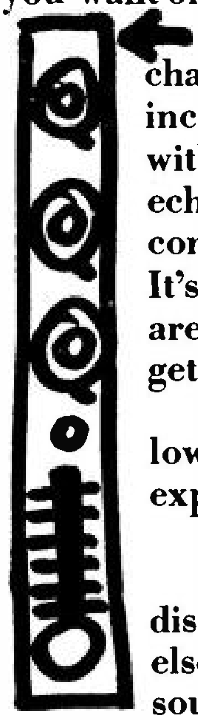
*Let this letter serve as a challenge to potential writers in stations around the country who have wrestled with and perhaps solved some of Mr. Pickett's very real problems. We want such meaningful articles and will publish them. Actually,* THE FEEDBACK LOOP *column starting in this issue is aimed in this direction.* LOOP *editor John A. McCulloch has items of this kind in preparation now. Use his column as a forum for exchange of ideas and information.—Ed.*

# Harvey's is selling stuff you may not know exists.

There's an information gap in the broadcast and recording fields today. Sometimes we find ourselves distributing an exciting new product that many professionals don't even know exists.



For example, did you know that there's a new boom headset with built-in microphone that's so light you can actually forget you're wearing it? It can pick up two different signals at once. It's interchangeable with any standard boom headset. And if you want one now, Harvey's is the only place you can get it.



Also, there's now a complete console mike channel so small, you can hold it in your hand. Yet it includes a fader, program equalizer, line-mike switch with input pad, reverb-send channel and pre- and post-echo switch. With a few of them you can put together a complete console in a couple of days instead of months. It's revolutionary—yet, just about the only people who know of it are Harvey customers. Because just about the only place you can get it is Harvey's.

There's much more that's new. We have several new low-cost microphones that sound just as good as the most expensive mikes of a few years ago.

And so on. And so on.

Harvey's is in the habit of finding new equipment and distributing it before anyone else does. Often before anyone else knows about it, in fact. That's why almost every major sound studio and radio station already deals with Harvey's.

Help us close the information gap in the broadcast and recording fields. Call or write Harvey's regularly.

Open an account, if you want, and we'll start sending you our newsletter.

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# The Feedback Loop

—JOHN A. McCULLOCH

*This is the first installment of what will be a regular monthly feature of* **db** Magazine. *The author brings to his task the qualifications of ten years in professional audio and a youthful, inquisitive outlook at the job at hand. He has worked as a recording engineer with Yankee Recordings in Boston, been a broadcast engineer/announcer with WRLB-FM in Long Branch, New Jersey, and done a stint with SHAPE headquarters in Paris, France where he was an audio specialist doing motion-picture sound recording. More recently he was at ABC Network headquarters in Hollywood as an audio maintenance engineer. For almost a year now he has been with Electro-Voice, Inc. in Buchanan, Michigan where he is in the professional microphone lab, as well as giving lectures, demonstrations, and field engineering.*

● When I was asked to write *The Feedback Loop*, I was also given the opportunity to make this column more than just a straight question-and-answer page. Our chosen field of audio engineering is a fast moving and very demanding one, calling upon each engineer to exercise his many talents in maintaining and advancing the state of the art. Thus to write a routine Q & A or Service Tips type of column would not provide you with other information which could be of great assistance to you, your co-workers, and—most important—your finished product: audio.

With Ye Olde Editor's permission, I would like to see this column, in addition to a place wherein may be found the answer to your technical problems, a place in which the operators, technicians, and manufacturers may exchange information.

To have this become a reality, you will (in the greatest part) have to supply the material and the ideas of *The Feedback Loop*. More concretely, your present problems, your unusual solutions to past problems, and other bits of operating and servicing information, that might just make the life of a fellow **db**'er easier, will form the basis of this column. (Perhaps the more complex and pertinent bits and solutions might induce Y.O.E. to a full coverage of that subject.) *(I induce easily. Ed.)*

Also, I would like to include, as this started out to be a Service Tips column, circuits and addenda that have been picked up from time to time, which would otherwise not be sufficient in length for a full article, but would be of use to many. Finally, with the hopeful cooperation of the manufacturers and those of you who are maintenance engineers, I would like to include small equipment modifications, which will help to improve the operation, performance, and/or reliability of your gear.

The accomplishment of these ideas that can set *The Feedback Loop* in operation in its most effective manner requires that names will be named, products specified, and particular models under discussion fully identified. This will be done. But in no manner is this procedure meant to carry an endorsement—favorable or otherwise—of a particular product, but merely to be completely professional and call the shots as they come.

In all equipment discussions I shall endeavor to have the manufacturer's data on hand. In cases of operational problems or other areas, I will, as far as is possible, give additional references for your use and possible aquisition for your reference library.

In the hope of accomplishing this I have cleared a file cabinet for the manufacturers' data. In addition to material already on hand, this will be filled out by each manufacturer wishing to participate in *The Feedback Loop*—by mailing a complete set of data (specifications, circuit diagrams, single lines, etc.) on each model in his present manufacture, and by continually updating his file as production changes are made and new models introduced. (New Products listings are covered by announcements in other sections of **db**.)

This then is the operating circuitry of *The Feedback Loop*. But, the operational stimulus must come from you. If you like the ideas set forth, and can suggest other areas of service or application, let me know. If you do not, let me know as well. If possible suggest changes which will put the column back into flat response. In either case this column is for you. And in the most part, by you. Information, problems, and comment must come from you—all of you—to make this column an effective tool in your work.

Because of the mechanics of the printing business, I am writing this column some months before it will be in your hands in this second issue of **db**. Therefore, some time may elapse between the time I receive information from you and may pass it along, answer your problems in print, or print addenda and modifications of equipment.

Therefore, in the answering of your problems and to assist you in the fastest manner possible, I shall answer all *problem* letters, and make acknowledgement of all letters containing information, tips, etc., within a month of their receipt, whether or not they are used in *The Feedback Loop*. However, to facilitate this policy, all such letters should be accompanied by a stamped, self-addressed envelope, otherwise I shall assume that an immediate reply is *not* requested.

Let's have an input signal from you.

## Next Month

A simple circuit to adapt any d.c. relay (single coil) to perform a latching type of operation with a single, momentary-contact, push button. Included will be facilities for remote operation, and/or remote release only.

There will be a discussion of output levels of dynamic microphones, and the possibility of overloading the input amplifier. Suggestions for its reduction or elimination will be offered.

# Letters, continued

The Editor:
Congratulations on *db*! I could hardly put the first issue down without reading it in its entirety. For some time I have enjoyed reading the AES journals and Langevin engineering newsletters. I find this new publication fresh and with exciting possibilities, especially in reference to establishing a dialogue.

The publication would be well worth the modest $5.00 per year you ask of non-qualified readers.

*J. O. Terry*
*Chief Engineer*
*Radio and Television Commission*
*of the Southern Baptist Convention*
*Fort Worth, Texas*

*Thank you for your kind sentiments, Mr. Terry. We've set high standards for* db *—and will make every effort to live up to them. If we should miss on occasion, we hope readers will be quick to let us know.*

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# A Handbook for the Audio Engineer

—GEORGE ALEXANDROVICH

*As the title indicates, this series will range wide in its search for meaningful material related to sound recording and processing. It will concern itself exclusively with the practical approaches to problems without going into any dry, theoretical approaches to the solutions of the multitude of tasks faced in the studio. This treatment will ensure that all sound engineers, with or without extensive theoretical trainings, can fully understand and benefit from it.*

*The author is well equipped to pursue this course. He is vice-president of engineering at Fairchild Recording Equipment Corporation. One of his many responsibilities is to be certain that the end user of the complex equipment of that company fully understands the operation, both in terms of versatility and limitations, of the equipment at hand.*

● The most important place for the sound recording engineer is the control room. So it is logical to begin our discussions with and evaluation of the control equipment and some basic performance tests.

## System Evaluation

Let us assume that you have just joined a recording company and have been assigned as a recording engineer to a particular studio. There you will spend long hours in front of the recording console with its multitude of switches, faders, knobs, buttons, lights, and meters. In the end, the success of your employment will depend on how well you will adapt to this system and how well you will be able to control it. It may well be that the system is well maintained, but it is up to you to determine the system's optimum operating points.

Each system has its limitations and each can be operated in a variety of ways. At the very beginning you should become familiar with the block diagram, the so-called *single line* drawing. This diagram shows all the possible paths a signal can be routed through the system, beginning at the microphone, through to the tape recorder and monitor speakers. Such a diagram is shown in fig. 1.

A properly laid out block diagram should indicate the level that exists at any point along the signal path. These levels should indicate the optimum condition of the mixer. From this block diagram you will learn the flexibility of the system, the equipment used, patching, monitoring, the switching facility, and other details that there are to know.

We judge the *quality* of sound by listening to the monitor speakers in the control room and studio. If we are to evaluate the system on the basis of listening tests it is obvious that we must be certain that our monitor system is sufficiently up to par so that we can rely on it.

Begin by listening to the speakers without turning on the console and without any signal into the monitor amplifiers. In a quiet control room concentrated listening to the normal hiss of the monitor amplifiers should convince us of the absence of hum, crackling noises, or other interference in the monitoring circuits. This will prevent the placing of blame on the console or other equipment for troubles that may well be in the monitor lines.

Once you are convinced of a sufficiently low monitor noise level, feed, by direct connection to the monitor circuits, a signal generator or program known to be clean. Now listen to the speakers at a fairly loud level. Listen for possible cabinet vibrations or room resonances. These are sometimes mistaken for distortion or clipping within the system.

Now reconnect the monitors to the main console and turn the whole system on. Turn down all faders. When you are listening to the console's output stage you will normally hear a slight increase in the white noise or hiss. With all input faders down and all switches off bring the master gain and sub-master gain controls of a particular channel up. You should, of course, hear an increase in noise. This is to be expected since each amplifier contributes to the over-all noise level. The noise to be expected is hiss; any

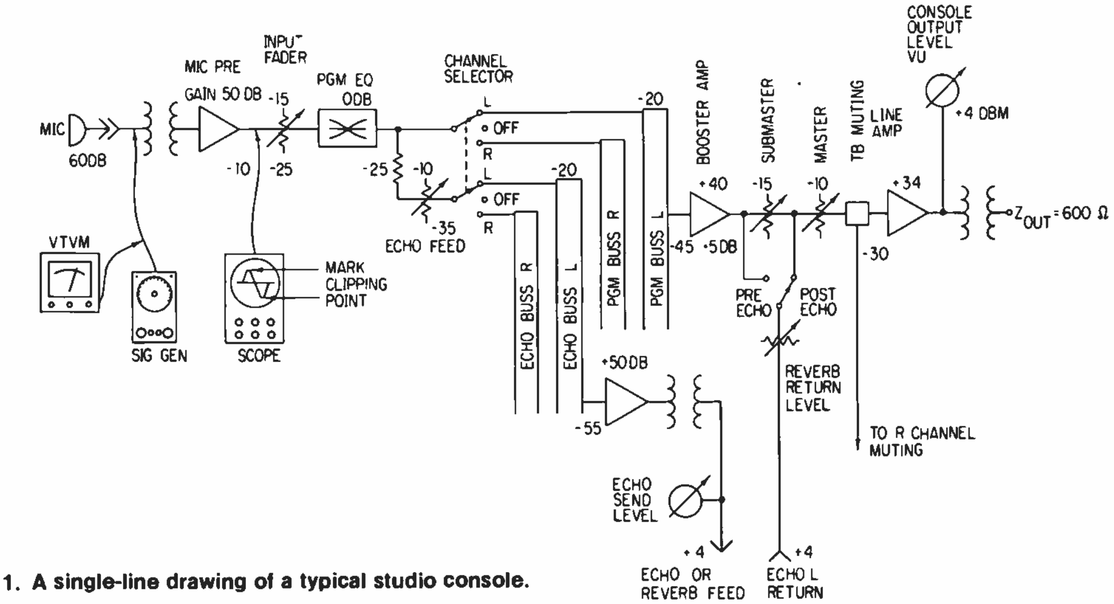


**Fig. 1. A single-line drawing of a typical studio console.**

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(D) **NEW Variable Regulated High Voltage Power Supply** Features new 6 v. and 12 v. AC filament voltages; also furnishes B+ from 0 to 400 volts DC, bias from 0 to −100 volts DC; separate panel meters monitor B+ output voltage & current; voltmeter switched to read C-volts; output terminals isolated for safety; high voltage and bias may be switched "off" with filaments still "on" for maximum testing efficiency and safety. **Kit IP-17,** no money dn., $6 mo. **$59.95; Wired IPW-17,** no money dn., $9 mo. **$89.95**

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crackling hash or clicks indicate trouble in the console—something to be taken care of without delay. This could later mean the difference between a successful or bad session.

Next, bring up your input faders. Don't connect the studio mikes. Continue listening to the ambient or inherent noise of the system. Set all faders so that there is approximately 10-15 dB of reserve gain for each. This sets up the system for its approximately optimum operating point — while allowing flexibility for gain riding during the session. Noise should still be low (hiss or white noise only).

Now connect the studio mikes and listen to the studio noise. With the controls set for normal operating condition, empty studio noise should be higher than the ambient noise of the system. Thus for all practical purposes, the noise level of the mixing console will be adequately low for good recordings. This is because studio noise is masking console noise preventing it from being heard.

But don't let this be the sole criterion for measuring the console noise. Not all studios are always quiet, and not all microphones are alike. What is being suggested here are the minimum practical requirements, certainly not the maximum theoretical limits. It is the duty of the maintenance man to see that the system is operating as close as possible to the theoretical optimum; it is the duty of the sound engineer to see that the system is performing within acceptable limits for good sound recording.

## The Input Stage

One of the vulnerable parts of the recording console is the input stage. All the other stages have a controlled feed; it is the input stage that is receiving a signal dependent on the type of microphone, its closeness to the instrument, or sound source. A 'scope and signal generator enable you to determine exactly how good the input stages are and how much level they can accept without clipping.

Connect the 'scope across the output of the mike preamplifier and the signal generator to the input (instead of a mike). It may be necessary to make an additional loss pad between the generator and the input to the console in order to obtain a simulation of mike levels.

Keep increasing the input signal until the sine-wave display on the 'scope just starts to show distortion or clipping. At this point adjust the 'scope so that the waveform occupies about three-fourths of the screen. Mark the peaks of the waveform with a grease pencil. From now on, don't touch the 'scope controls but substitute the microphone for the generator and listen to a hard-to-record instrument such as piano, drums, cymbals, or trumpet.

Observe the 'scope. If peaks of the incoming signal reach the markings on the 'scope face your input will overload at a regular session. You have left no safety margin. What's to be done? Pad the mike down if its output is too high. (It is known, of course, that different mikes produce different outputs with the same sound pressure levels. Condensers are notorious for high output levels while ribbon-types generate very small signals. It is a general practice to pad down a condenser mike with as much as a 20-30 dB pad.

(The European approach to this problem is to feed the outputs of the microphones directly into the faders and thus mix fairly low levels. This eliminates input preamplifiers but puts stringent quality requirements for extremely low noise and low contact resistance on the faders.)

Low-level mikes require a reverse treatment. Find more gain by using a step-up transformer or more gain from the amplifier.

In order to determine the proper signal strength at the output of the mike preamplifier repeat the earlier test using the signal generator and 'scope. But now reduce the signal generator output about 10 dB (measure with an a.c. meter, vtvm or equivalent) and again mark the screen. If program peaks are adjusted to reach the lower markings you will have ample headroom for overload, thus assuring clean performance of the input stage.

If instead of a vtvm, a volume indicator or VU meter is used be sure that the meter peaks do not exceed the zero point of the meter scale because the ballistics of the meter are such that it won't register sharp peaks (which can be as high as 10-14 dB). An average mike preamplifier is capable of outputs from 14-24 dBm above zero dBm so that 10 dB peaks could cause overload of the amplifier if the average level exceeds 4 dBm. One way to adjust the amplifier for proper output is to adjust its gain until specific mikes under certain conditions produce the most desirable output levels.

But every session requires different setups using different mikes. It is more practical to adjust all the microphones so they produce just about the same outputs. Condenser mikes are normally padded down, ribbon mikes are fitted with step up transformers while dynamic mikes are used the way they come, thus representing a group of average output level transducers.

What has been said here about the VU meter readings and peak information goes as well for the output stages of the console and for the meters measuring the signals leaving the console on the way to the tape machines.

## Pinning Meters

It is appropriate to here talk about the widespread practice of meter pinning during recording. There is no bigger fallacy, yet it happens every day. Not only the console is operated closer to the overload point but both recording engineer and producer loose their true reference indication. For what is gained in noise because of stronger signal is lost due to increased harmonic distortion, IM distortion, crosstalk, and tape print-through. The VU meter is in the system to indicate optimum operating level and is to be used as a reference for mixing. It seems that there is more respect today for a system which has its meter pointers more like fish hooks than precision level indicators.

This practice of pinning the meter is so much more dangerous with transistorized equipment. Even if the console output stages do not overload there is always a danger that tape electronics will. Transistors have a very definite overload point beyond which distortion skyrockets. If this point is approached clipping of the sound is inevitable. Relying on the compression properties of the tape when overdriven is foolish and dangerous because not only does distortion increase tremendously but raw tape has been made better and better so that today few tape machines may exhibit amplifier overload points before complete tape saturation.

## Set Up

Let us now review console controls and how to set them up correctly. A properly designed console should normally have up to 40 dB of additional gain in addition to the gain needed for normal operation. This extra gain has to be distributed between the gain controls evenly. Each fader in the line must be set so that at least 10-15 dB of extra gain are available for compensation of low-level signals. If all

the extra gain is shifted to one fader such as the mike fader, then the submaster fader has to be set to the other extreme. This condition will first send excessively high signals to the booster amplifier driving it into the higher distortion region, thus taking all flexibility out of the mike fader setting. And it will cause lower-level signals at the mixing network lowering the s/n ratio. The master fader or board gain control is usually given less extra gain than the other faders. Its general use is for trimming up the over-all level at the output and for over-all signal fade-out at the end of the take.

## Special Effect Equipment

Aside from faders and amplifiers, consoles usually contain a fair amount and variety of support or special-effect equipment. This may consist of compressors, equalizers, echo and reverberation devices, special switching circuits, and many other devices.

Although every console is different the same general rules apply for operating all of them. Stay safely below the overload point at any stage of the console. Don't allow the level to drop dangerously low exposing the noise of the amplifiers and support equipment. Watch out for those circuits where equalizers are used. Without applying boost the signal passing through the equalizer may be clean as a bell. But boost some frequency several dB and you may find signal peaks at that frequency clipping.

During a live recording the use of the compressors is generally limited only to the overload protection function. The reason for this is that:

1. the operating mode of the compressor if set improperly may ruin original tape or good take.
2. full dynamics of the original recording once lost can not be restored.
3. Compressors do not protect the input stages of the console. These are most vulnerable to overload.

If compressors are desirable as overload protectors then it must be emphasized that their thresholds should be set so that triggering of the compressor starts not lower than at normal operating levels of the stage preceding the compressor. Excessive compression ratios are also dangerous since they change dynamics too much, altering sound.

Compressors are altogether more desirable than limiters because they don't impose a harsh ceiling on peak information making their action less obvious to the ear. Amounts of compression in excess of 10 dB for live recording are generally not desirable; compression ratios ought not to exceed 4/1 (for 4 dB of input level change, a 1 dB change in output level). Compression ratios of 2/1 mean that by increasing input level by 20 dB, output level will increase only by 10 dB, thereby producing compression of 10 dB or 10 dB of level reduction.

Compression ratios for the limiters may range from 10/1 to 20/1. In some special instances limiters may be used for live recordings—in instances where it is hard to control or predict sudden program level changes and occasional peaks. In this case when the signal which is being recorded reaches limiter threshold, reduction of over-all gain starts taking place. Reduction of gain is proportional to the level of the peaks which exceed the limiters threshold. (More detailed treatment of limiter and compressor operation will be taken up in a future special discussion about them.)

## Reverb

Almost all consoles today are fitted with reverberation or echo busses. However very few of the consoles have the facility to measure the proper level to the reverberation device or echo chamber. The importance of correct level at this point is generally neglected causing improper utilization of the device or echo room, with resulting increased noise or distortion as well as a different character to the return signal. Overload of echo chambers and artificial reverberation devices is different in nature from amplifier overload. The simple expedient of a meter across the echo-send amplifier that is feeding the device usually eliminates unwanted hiss, rumble, and undesirable overload phenomena of the device.

A meter at this point serves as the feed level indicator and not a mixing-level indicator. For any levels of echo return, the echo monitor meter should be indicating the optimum signal supply level. Mixing echo signal with dry signal is judged by ear and can not be very well referenced to the meter reading. For some types of music with sustained tones, the amount of reverberation can be higher while for narration, signal should remain with a lesser amount of echo added. Basically, it is up to the producer's ear to determine how much reverberation to add. The engineer's job is to make sure that the reverberation device is functioning properly, producing the best possible results.

At this point it is appropriate to outline the basic difference between *reverberation* and *echo*. Reverberation of the signal exists in every accoustically live room or auditorium; in fact, in any enclosed area where sound waves can rebound from reflective surfaces and continue bouncing until the signal fades away. If reflections of the signal are numerous and frequent so that the over-all effect is one continuous fading sound, it is called *reverberation*.

The larger the hall the further sound reflections have to travel to bounce. When time between reflections becomes longer than 1/5 second the human ear is capable of distinctly disassociating each reflection. This phenomenon is called *echo*.

Echo is produced artificially for special effects by using a tape recorder. Sound is recorded on tape and played back with delay equal to the time that it takes for the tape to move from record head to the playback head. As the signal is played back it is again (or part of it) fed back into the record section along with fresh information creating an effect similar to the one obtained in a large hall or cave.

Although echo chambers use the word *echo*, very few of them can even approach the sound of real echo. All of them produce reverberation. The only difference between them is in the signal decay time and the frequency spectrum they can handle. Decay time is characterized by the total possible numbe of reflections of the sound before decaying beyond audibility. Properly designed and built reverberation chambers are an excellent addition to the recording studio. However, there are several devices on the market today which produce artificial reverberation without resorting to the need of the accoustical properties of special rooms.

These devices work on the principle of sending sound accoustically through a metal plate or spring. Plates or springs excited by the signal keep vibrating for a certain period of time before coming to rest. These vibrations are sensed by the special transducers and those signals are amplified. By careful design and execution results obtained by these artificial devices allow the creation of effects similar in quality and sound to the live reverberation chambers. Use of these devices today is widespread and gaining in popularity over expensive, large accoustical rooms and chambers.

# Editorial

THE 33RD CONVENTION of the Audio Engineering Society has come and gone. It was, in many ways, an exciting four days. Each year has seen a significant growth in the number of exhibitors. And the quantity and diversification of the papers has grown right along.

Technical sessions were divided into eleven categories: Audio Applications; Transducers; Speech and Music; Amplifiers; Control Systems; Sound Reinforcement; Broadcasting; Standards; Disc Recording and Reproduction; Tape Recording and Reproduction; and Film Recording and Reproduction. Each of these categories offered six to nine papers. Except for the one on Standards. A total of ten papers were given on this subject.

It is gratifying to see an organization of the stature of the AES exploring the need for standards and standardization. We cannot agree more strongly with the need to further pursue this area. The evidence of industry growth at this year's convention can not be ignored. Growth infers an infusion of new blood into the industry. If this growth is to be utilized to the maximum for everyone's benefit, it can only be done if a thorough system of standards are established for all facets of sound re-creation.

There is enough confusion now in the minds of those responsible for the purchase of audio equipment. New companies are only likely to add to the chaos unless they adhere to a standard of input and output that is fully compatible with other existing equipment. And certainly they can not do this unless standards exist on which everyone agrees.

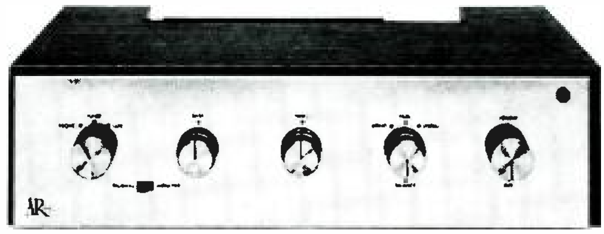
The AES has taken the correct steps in this direction. Organizations such as the SMPTE, IEEE, and the Acoustical Society are also moving toward more meaningful standards. The point is coming when it will be necessary for these groups to pool their thoughts so that each division of the audio field ends up with complementary rather than conflicting standards.

The pages of **db** can serve to cross individual society lines. As time goes on we will publish statements, comments, and papers that tend to forward the cause of over-all industry standards.

We invite your participation.

—L.Z.

# AR Inc. amplifier



Shown with optional oiled walnut wood cover.

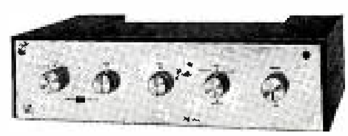
## Acoustic Research announces its first electronic product, *the AR amplifier,* an integrated stereo preamplifier/control and power amplifier, all silicon solid-state.

**1. PERFORMANCE** — The state of the art of electronic design has reached the point where it is possible to manufacture a nearly perfect amplifier. There are a few such now available. We believe the AR amplifier belongs in this select group.

**2. PRICE** — $225, in black finished aluminum case. Oiled walnut wood cover is $15 extra and optional. The AR amplifier costs considerably less than the few amplifiers capable of similar performance. However, it should be judged by professional standards and on an absolute basis without consideration of price.

**3. POWER OUTPUT*** — Enough to drive with optimum results any high fidelity loudspeaker designed for use in the home.

**4. GUARANTEE** — Establishes a new standard for reliability and durability. The product guarantee for the AR amplifier is unmatched in the industry by any other electronic component regardless of price.



The AR amplifier is sold under a two year guarantee that includes all parts, labor, reimbursement of freight charges to and from the factory or nearest service station. Packaging is also free if necessary.

Literature on other AR products — loudspeakers and turntables — will be sent on request.

*Power output, each channel, with both channels driven: 60 watts RMS, 4 ohms; 50 watts RMS, 8 ohms; 30 watts RMS, 16 ohms.

Distortion at any power output level up to and including full rated power; IM (60 & 7,000 Hz, 4:1), less than 0.25%; harmonic distortion, less than 0.5% from 20 Hz to 20 kHz. Distortion figures include phono preamplifier stages.

Frequency response: ±1db, 20 Hz to 20 kHz at indicated flat tone control settings, at full power or below.

Switched input circuits: magnetic phono; tuner; tape playback.

Outputs: Tape record; 4, 8 and 16-ohm speakers.

Damping factor: 8 to 20 for 4-ohm speakers; 16 to 40 for 8-ohm speakers; 32 to 80 for 16-ohm speakers. Lower figures apply at 20 Hz; higher figures apply from 75 Hz to 20 kHz. Measurements taken with AGC-3 speaker fuses in circuit.

ACOUSTIC RESEARCH, INC., 24 Thorndike Street, Cambridge, Massachusetts 02141

*Circle 18 on Reader Service Card*

# Modern Mixing Techniques Boost Efficiency

—Richard Vorisek*

*The modern recording studio control room is filled with sophisticated equipment and equally sophisticated engineers. They are an answer to the demands made by customers with special problems. One such group are those that make the sound tracks of TV commercial films. Here is one major studio's method of rising to the increasing challenges.*

CUE sheets in order? . . . Check; Footage counter reset? . . . Check; Projector ready? . . . Check; Sound tracks ready? . . . Check. Okay, now check dialogue equalization and present opening music . . . Everything ready . . . *stand by, this is a take!* . . . Kill the studio lights . . . Push the *start* button . . .

Okay, the picture is on the screen . . . 11 . . . 10. Slate it . . . Take # 1 . . . 7 . . . 6 . . . 5 . . . 4. Sync beep. Opening music begins . . . okay . . . watch it now. Dialog begins at 15 feet . . . be ready to duck the music . . . not too soon, easy does it. Here it is . . . balance seems fine . . . watch that brass passage . . . it's crowding the dialog. Dialog out at 25 feet. Goose the music . . . you've got ten feet. Sound effects come in soon . . . 30 feet . . . not too loud but be sure it's there. Okay. Sound effects # 2 comes in now . . . hit it, it's in the clear. Fade quickly, the dialog is in again at 35 feet . . . OOPS . . . that fade was a little late. Next cue 60 feet. The jingle comes in preset on Music "B" as soon as the dialog is over . . . 58 . . . 59 . . . 60. Now! Good level for the jingle and vocal . . . nice crisp sound . . . that high-frequency boost helps. Be ready to duck music at 80 feet for announcer's copy on voice track "B" as soon as the vocal is over . . . 78 . . . 79 . . . 80. Now! Balance seems fine . . . keep the music down. Don't let it build too soon. Be ready to change equalization on voice track "B" for retake of announcer tag at 29 feet . . . 90 . . . 91 . . . 92. Now! Okay . . . announcer out at 98 feet . . . music up for a button to 101 feet. Fade out.

Hit the *stop* button . . . studio lights on. Order the tracks and picture to be rewound and reset on their start marks . . . *relax*.

Not too bad except for that late fade at 35 feet. Let's see what the producer's comments are:

He feels the fade at 35 feet was a little late . . . (agreed). Also thinks sound effects a little too loud at 30 feet. (We discuss it and decide to lower it a bit to be on the safe side.) He was hopeful for a better match of announcer's voice quality from original recording to the retake at 92 feet. (I'd like that too.) Why didn't he at least record the retake in the same studio as the original?... I'll try cutting one more step out of the bottom . Anything else? . . . no . . . he likes the rest of the take. Okay . . . let's make another one.

*Stand by* . . . push Start . . . take # 2. Take # 3. Take # 4. Take # 5. . . finally got it . . . he likes take # 5. *There should be an easier way to do this.*

If all of this is familiar, you are probably one of a

**vice president, film, Reeves Sound Studios* relatively small group of people known as "sound mixers" who are engaged in the process of creating the final sound track for a TV commercial, TV show or motion picture.

If it is not familiar, I hope the preceding peek into a mixer's mind during a recording session will set the stage for an explanation of this process and a description of a new technique for recording the final sound track with greater speed and accuracy.

Basically, a "mix" involves combining two or more recordings to produce a new recording in which the original recordings are balanced with respect to each other and with appropriate regard for the visual image.

To illustrate a mix in its simplest form, let's use the example of a TV commercial whose audio consists of only two elements: Announcer and Music. Let us also assume that both tracks are of good quality and of constant volume. The mixer's job in this case is merely to adjust the music level below the announcer in a pleasing and tasteful balance and record this result on an equal or better quality synchronous magnetic recorder to produce a master. This master will be used to make the necessary photographic negative sound recordings or sprocketed magnetic recordings for ultimate release with the visual image.

At the other end of the scale, the mix for a feature motion picture is the most complex. In this case it is not unusual for the mixer to be working with as many as twelve tracks at a time. These sound tracks are a mixture of location and studio recordings, music, and original sound effects mixed with stock effects. Here the mixer's job is not only to balance the volume relationships of the tracks with each other, but also to relate them to the picture in the correct perspective necessary to create a sense of realism. Another problem is to equalize the tonal qualities of the tracks so that the audience is not aware that the film they are viewing is made from many individual shots over a period of possibly a month or more, all in a variety of acoustical conditions.

The traditional method of mixing requires the mixer (or mixers, if the number of tracks is more than one man can handle) to run the picture and sound tracks over and over, from the beginning of the reel to the end, the necessary number of times needed to familiarize himself with all the various adjustments required. This segment of the film is usually somewhere between ten and eleven minutes long.

As the rehearsals progress, the mixer will note on his cue sheets (with reference to elapsed footage computed from the start mark) any information that will be helpful to him in creating a smooth continuous sound track when he finally attempts to make the master recording.

The number of rehearsals and takes needed to complete a reel will of course depend on the skill, experience, dexterity, and speed of judgment of the mixer at work, the skillfulness of the editorial preparation, and the sophistication of the equipment at his disposal. Although it is possible to edit together a number of master takes, salvage the best sections of each, and thus achieve a higher degree of perfection, this is not general practice as it is somewhat time-consuming, thus increasing the cost.

An ideal mixing technique would certainly include the following:

Permit the mixer to rehearse any difficult section of a reel as many times as necessary without running the complete reel each time or rethreading the mix elements.

Permit simple editing together of the best parts of a number of takes.

Permit precise, quick, "pre-mixing" of difficult mix elements without loss of quality or time.

Permit instantaneous master playback synchronous with the picture.

Permit a simultaneous "minus dialogue" international track for foreign release.

At Reeves Sound Studios, we have installed new equipment and adopted as our normal procedure, a new mixing technique which gives us a near revolutionary opportunity to produce final sound tracks of the highest professional quality. This system could well be the greatest advance in motion picture re-recording since the introduction of magnetic recording, and easily meets all the preceding requirements of an ideal mixing technique.

The equipment consists of 35mm and 16mm projectors, 35mm and 16mm magnetic audio film playback machines, footage counters, and a special 3-track 35mm magnetic recorder for each studio. These are controlled by a distributor system capable of running all the machines interlocked in both forward and reverse directions at the push of a *forward, stop,* or *reverse* button located on the mixer's console. These controls are duplicated in the projection booths and the film playback machine room, but are normally operated by the mixer.

The special 3-track 35mm magnetic recorder has unique switching circuits which allow the erase and record bias to be switched on or off without creating a pop or other disturbance in the recording. This switching is precisely timed and permits any part of a Take # 1 recording to be replaced by a Take # 2 recording (of the same sound) with no noticeable change or drop-out as long as the volume and quality are matched.

The record switching circuit for each of the three tracks is activated by the mixer by pressing one of four buttons on his console. He has an individual *record* button for each of the recorder's tracks plus one more to activate all three at the same time.

Directly beneath each of these buttons is a matching *record off* button which disables the erase and record bias circuits. By matching "in" and "out," he can revise his master recording at any point without remixing or making a physical edit.

Since there are three recorder tracks available, we normally assign a specific category of sound to each track. Dialog and narration to Track # 1. Music to Track # 2, and Sound Effects to Track # 3. Combining the recorder's three playback head outputs results in the "mixed track." This normal assignment is not an absolute rule and each mixer is free to assign his recorder inputs to provide the most expeditious way of accomplishing his mix. Regardless of how he elects to patch his inputs, normally no two tracks have identical information recorded on them. Thus he can revise or remix any of the tracks without destroying those that do not need further work. If he wishes, he can use one of the recorder's tracks at a time and relieve himself of the need to struggle with all the tracks simultaneously. In effect, he can pre-mix directly to his master, subject to instant revision if necessary.

Six more buttons on the console, located in logical relationship with the *record on* and *record off* buttons, permit the mixer to switch his monitor speaker and volume indicator inputs as follows:

*Direct* feeds the output of the mixing console.

*1-2-3* feeds the combined output from the recorder's three playback heads.

*1* feeds the individual output of recorder playback head number one.

*2* feeds the individual output of recorder playback head number two.

*3* feeds the individual output of recorder playback head number three.

The sixth button is used to mute the monitor speaker and avoid hearing the audio during reverse, or when the tracks are slowing down to a stop or coming up to speed.

The recorder input and output volume controls are carefully adjusted so that regardless of the sensitivity of the master magnetic film oxide the monitor and volume level indicator remains constant in any of the switch positions.

When a take is to be made the *recorder on* switches are activated and the monitor selector is switched to the *1-2-3* position. The mixer is then listening to the combined mixed track he is recording.

If at any time he is dissatisfied with any part of the mix he is making, he can push the reverse button and prepare to correct his error. Upon pushing the reverse control, all the machines in use, including the footage counter, will come to a slow interlocked stop and the erase and bias circuits will automatically switch off. When all the equipment has come to a halt, the machines will automatically start again and run in reverse until the mixer has pushed either the *stop* or *forward* buttons. (If he does not push either *stop* or *forward*, the machines will run all the way back to the head of the reel and stop themselves at the start mark.) If he pushes *forward* at any time while the equipment is running in reverse, all machines will come to a stop and then go forward. If he elects to revise his mix, he merely has to match the previously recorded levels on the original tracks involved, and activate the *record on* circuits in the proper recorder tracks to be revised before he has again reached the footage of the error.

Since he is listening to the master playback immediately after he records it, any mismatch will be apparent the instant this new recording replaces the old one. If he has not reset his mixer volume controls accurately enough and does have a mismatch, he can easily correct this. He can either back up to the last place his cue sheet notes indicate a precise volume setting which he can duplicate, or by switching his monitor from any of the playback positions to *direct*, he can aurally compare the two and adjust the mixer volume controls for an absolute match.

In this brief attempt to describe the mechanics of a mix it was assumed that the original tracks or mix elements were of good quality and had standard frequency characteristics, good signal-to-noise ratio, etc.; were skillfully edited and prepared; and were accompanied by accurate, informative and readable cue sheets when presented to the mixer.

Unfortunately, more often than not, today the mixer is working with sub-standard original tracks, poorly prepared, from which he is expected to create a commercial-quality final sound track. The flexibility of this new mixing technique is immediately apparent under these conditions and the fortunate mixer who is using this system quickly finds his own best way to overcome his mixing problems.

I am confident that anyone using this technique will join me in saying, *"This truly is an ideal way to mix."*

# Automatic Audio Level Control

## A Review of the Current State of the Art

—Emil L. Torick*

*The modern recording studio has greater-than-ever need of automatic audio level controls. Product advancement has been extremely rapid. Here is the current state of the art*

THE automatic control of audio levels in recording, broadcasting, and sound reinforcement systems is a highly useful and accepted technique today. It is also a rather complex subject, due largely to the broad variation in operational requirements and the many brands of equipment offered to satisfy these needs. A fully comprehensive study of all the techniques commonly employed is beyond the scope of this article. However, it is hoped that a general presentation of the current state of the art will prove useful for many audio engineers.

In general, automatic audio control devices can be classified in either of two categories:

1. Those which prevent instantaneous overloading in recording or overmodulation in broadcasting and,
2. Those which are used to equalize variations in average level or reduce excessive dynamic range.

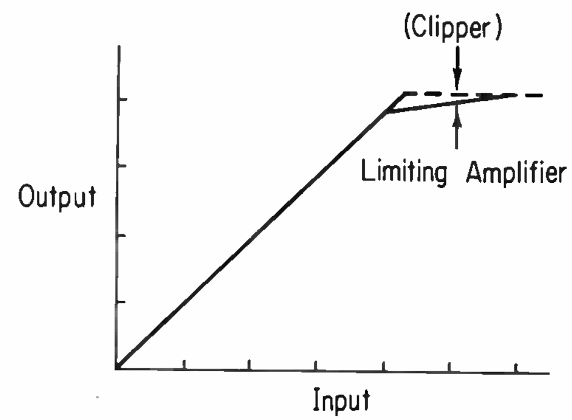



Fig. 1. The characteristics of a limiting amplifier.

*CBS Laboratories, Stamford, Conn.

### Limiting Amplifier Characteristics

The first category includes such devices as peak controllers, modulation limiters, limiting amplifiers and clippers. A bit of confusion exists regarding these terms. The IRE Dictionary defines a *limiter* as "a transducer whose output is constant for all inputs above a critical value" (52 IRE 17.S1). In current audio engineering terminology this definition describes a clipper. Thus it is more appropriate to use one of the first three above terms when referring to an amplifier whose gain is automatically adjusted as a function of program waveform amplitudes.

Figure 1 shows the familiar normalized transfer function for peak amplitude controllers. At low signal levels there is linear amplification. Above the so-called "verge of limiting", further increases in input level cause gain reduction, and the maximum output level remains nearly constant. In typical devices the slope of the limiting curve is 0.1—i.e. a 10 dB increased input will produce a 1 dB increase in output. For comparison, the response of a clipper is also shown, in dashed line. As per the definition, the output remains exactly constant above the critical value.

It should also be pointed out that the limiting amplifier characteristic of fig. 1 refers only to the steady-state sine-wave response. When complex program signals are fed to such a device, its action will be dictated by the time constants associated with the gain controlling function. At times there appears to be a competition for the title of "World's Fastest Limiting Amplifier." (In some respects this is similar to the horsepower race among automobile manufacturers.) Although it appears desirable for a limiting amplifier to react quickly enough to prevent any overshoot of the program signal, it must also be recognized that every time gain reduction occurs, there follows a less than optimum condition while gain recovers to its normal value. In general, attack time (the time required for gain reduction to within 1/e or 63% of steady-state value) is constrained by circuit limitations and recovery time (gain increase) by the lowest program frequency of interest. Attack times as fast as a hundred nanoseconds or less could be achieved if desirable, but recovery must be long enough to prevent the harmonic distortion which would be caused by gain changes during each single cycle of low frequency signals. This latter factor limits the theoretical recovery time to not less than approximately 100 milliseconds. Why then do we read in manufacturers' specifications of recovery times in the order of one-half to several seconds? The answer lies in the designers' attempt to minimize the audible effects caused by rapid and repetitious gain changes. The problem has been further complicated in the past by the use of a wide control range to accommodate the now obsolete practice of using modulation limiters to also perform an automatic gain riding function.

In order to reduce the loss of average program power associated with extremely fast attack times, a technique of combining peak clipping with medium-fast control action is currently used. The clipping limits the absolute output amplitude, while the gain control function prevents excessive clipping and resultant intermodulation distortion. The method is particularly popular with broadcasters because it permits higher average signal levels within modulation limits.

## Pre-Emphasis

A special requirement for peak amplitude control exists in systems where the maximum permissible signal level is not constant with respect to normal audio frequency. This condition commonly exists when high frequencies are boosted by pre-emphasis, as in tape, disc and optical recording, and in FM and TV broadcasting. Conventional automatic control before pre-emphasis will not prevent overloading of high frequencies. The same type of control after pre-emphasis will solve the overmodulation problem but at the potential cost of sudden annoying losses of level and other audible distortions. The optimum solution to this problem is to control the low and high frequencies separately. Two methods are available for high frequency control—clipping or automatically variable dynamic equalization. The latter method is preferred and most commonly used. It avoids the problem of distortion in clippers by automatically varying the upper-frequency response as a function of program spectral distribution.

## Automatic Level Control Characteristics

The second major category of automatic audio control devices includes compressors, and automatic level controls or automatic gain control (AGC)† amplifiers. In contrast with limiting amplifiers which are normally operated in the middle of their control range so that gain may be increased or decreased, as required. Furthermore, the output is based on average rather than peak considerations. Although generally successful in maintaining constant average levels (as read on a VU meter) the automatic level controls do not equalize true relative loudness levels. Only one such loudness-controlled device is available at the present time.

Figure 2 shows typical steady-state transfer functions for automatic level controls and compressors. The AGC and level control characteristic is similar to that of the limiting amplifier except for an increased control range. A compressor, on the other hand, is characterized by a slope which is less than 1 but not as severe as that of a level control. Compressors are generally used in recording and film studios because their control extends over a significant portion of the total program amplitude range. Their chief handicap in other applications is the non-constant output.

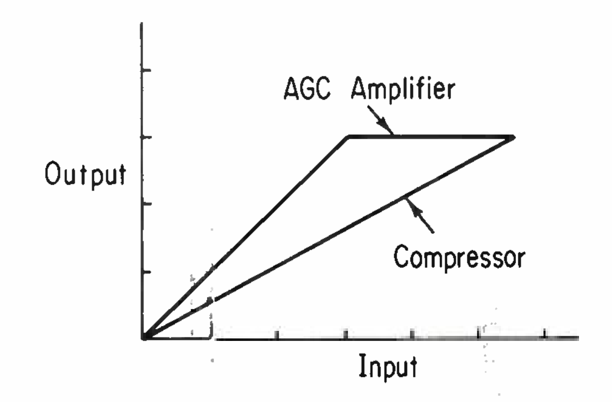



**Fig. 2. An automatic gain control characteristic.**

*†AGC should not be confused with AVC (automatic volume control), which is a technique for maintaining carrier levels in a receiver.*

AGC amplifiers, the most numerous and varied of all the automatic control devices, find application in broadcasting, public address, sound reinforcement and paging systems, recording, and radio and telephone communication systems. The basic circuit for nearly all such devices is shown in fig. 3. Signal is passed through an amplifier which has a variable gain stage. A portion of the output signal is rectified and fed back as control bias to this variable gain stage. Since the circuit arrangement constitutes a negative feedback loop, it tends to be self-regulating. A high level audio signal produces enough control bias to lower the gain to the point where the output level no longer produces excessive control bias. When low-level signals are encountered, the reverse action takes place and the gain is increased. In this way the output tends to be maintained at a constant level.

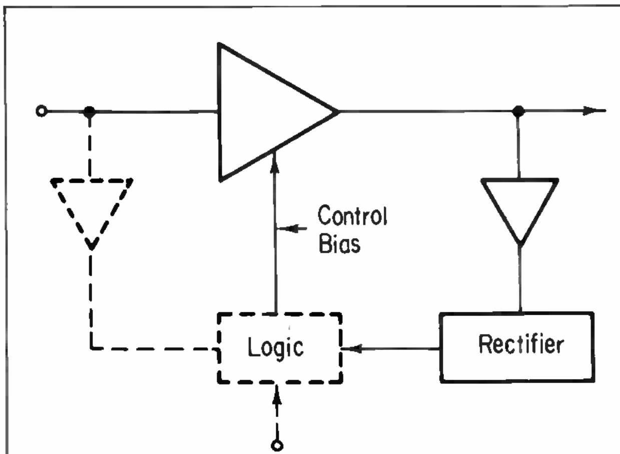



**Fig. 3. The block diagram of a basic AGC circuit.**

A number of variations on the circuit of fig. 3 have been employed by various manufacturers. For example, the choice of the variable gain element often receives a great deal of attention. The most common choices in semiconductor devices today are the variolosser—a biased diode shunt circuit—and the light dependent resistor (LDR)—a module containing a miniature lamp and a photocell. The AGC amplifier designer's choice of the variable gain element is based on circuit considerations and should be of little concern to the ultimate user. More important is the manner in which automatic gain adjustments are made. A garden hose transmission line controlled by a pair of pinch rollers might serve almost as well as the electronic methods if appropriate logic circuitry were employed in the control loop.

AGC amplifiers differ significantly from limiting amplifiers in the choice of time constants. Typical AGC attack times range between 10 and 20 milliseconds. This is appropriate for producing a control signal which is proportional to the average amplitude of the program signal. On this specification most manufacturers are in agreement. On the other hand, in the methods of recovery, we find the major differences among available units. If the basic AGC amplifier is operated normally in the center of its control

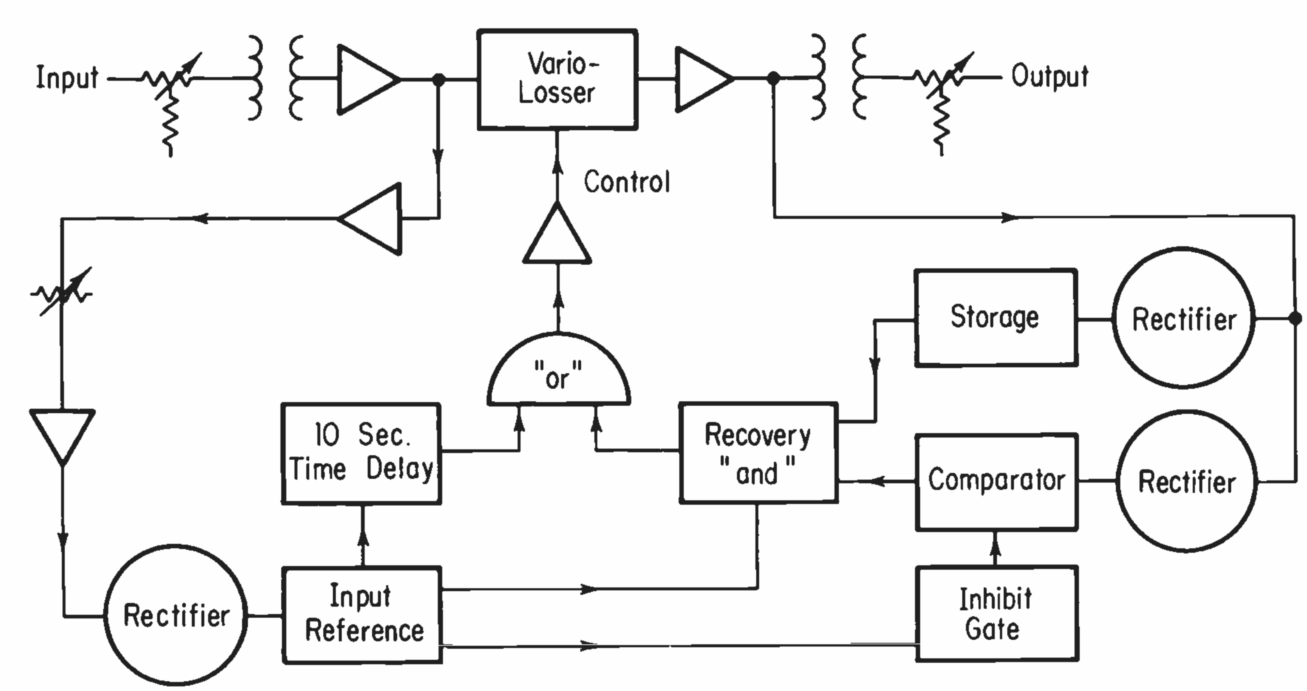



**Fig. 4. An automatic level control block diagram.**

region, a certain amount of gain reduction usually exists. When program ceases, no additional control bias is produced and the amplifier recovers to a condition of maximum gain. If the recovery time is short, the audible "pumping" distortion is produced and background noises may become annoyingly high. If the recovery time is increased, the amplifier will not react to correct low program level with adequate speed. One solution to this problem is to incorporate an additional circuit, operating off the input stage, to sense the absence of program. During such pauses, several manufacturers elect to reduce gain to a predetermined level in the manner of an expander or squelch. In another approach, which has proved to be the most popular and effective to date, gain is held stable at the level which existed just prior to the pause. If the pause lasts longer than 10 seconds, gain is slowly

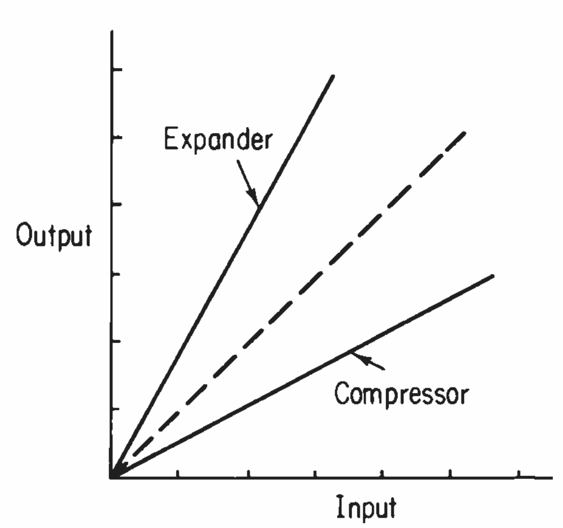



**Fig. 5. The characteristics of a compressor/expander system.**

returned to the center of the control range for standby operation. Figure 4 indicates the degree of circuit sophistication employed by this latter device.

## Compression-Expansion

An interesting variation on the use of compressors is currently receiving attention by a number of recording companies. The method involves the use of compensatory compression and expansion to achieve a reduction of recorded signal-to-noise ratio. It is in a sense analogous to the use of pre-emphasis in recording or FM transmission and de-emphasis in playback or reception. However, since the expandor device required for playback is as complex as its companion compressor, the technique has primarily been used only for making master tape recordings. The response of the compressor-expander system is shown in fig. 5. The compressor slope is less than 1, as was illustrated before, and the expander slope is greater than 1. When the two curves are combined, the original linear function results. Signal-to-noise improvement is achieved by operating the system as shown in fig. 6.

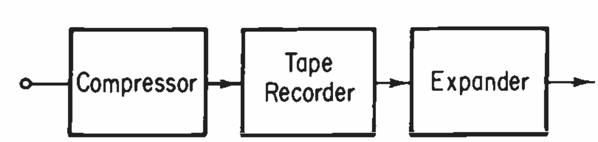



**Fig. 6. Tape compressor/expander systems are relatively straightforward.**

The compressed signal is recorded on the tape which contributes additional noise. During playback, the apparent noise is minimized by the expander action which reduces gain in the presence of low-level signals. In one current application of this principal, the Dolby system, the audio spectrum is divided into several regions, each with its own pair of control elements.

# Recording Consoles

—Philip C. Erhorn*

*The author presents his personal thoughts, born of long experience in console design, on the present status and future directions-of-design for this vital product.*

TODAY, those studios offering 8-channel recording facilities are booked solid, and if there is to be a new standard for large studio consoles, 8-track will be it. Indeed, one or two pioneering studios have been recording with 8-track machines for the past several years!

Just to compound (and confound) the thinking, a few studios are currently considering both 12- and 16-channel consoles, with up to thirty or more microphone input channels. At least two manufacturers are in a position to supply the necessary tape machines.

The voices of some wags have been heard to say, "Why bother with a mixing console? Just tie the mike lines directly to the tape recorder, and that's it!" While this approach is certainly feasible, all the goodies offered by a large mixing console, such as convenient monitoring, echo mixing, equalizers, compressors, simultaneous shrink-downs, etc. would be missing. The client might eventually get a finished tape to his satisfaction, but he might have to go to another studio (even in another city) to do it, so that the time and trouble spent in delivering the tape would be completely out of line.

Signal-to-noise ratio for these many-tracked machines, while still a major consideration, may not actually loom in importance as it would were they to be used for serious music recordings. These machines are destined for use in studios catering to rock 'n roll and other strenuous popular music recording, where tape noise is so far below average recording levels as to be insignificant. The dynamic range for much of this type material is very limited, and compression will further limit it.

## Compression

While it is my strong opinion that the shelving type of compression curve produced by a limiter, has no place between the console and the tape recorder, certainly compression is a great help to the mixing engineer. He is the one who has to cope with the unusually high studio levels produced by guitar combos and their associated hi-hum, hi-output electronic devices. At least two companies are making miniature LDR compressors well-suited for in-line panel mounting in the console. As many as four channels may be compressed simultaneously with the same time constant, and with only one compression device. This feature alleviates the possible degradation of the stereo effect when each channel is compressed to a different degree (depending upon level differences) by individual compressors.

Don't misunderstand me. I am well aware that "stereo" is incidental in multi-track recording. However, in the more elaborate consoles, simultaneous 2-track and mono feeds, made up from the main channels, provide tapes which the client may take with him at the end of a session. It is in these supplementary tape feed lines that the small compressor can serve most effectively. Of course, one or two single-channel compressors may readily be patched into the vocal mike channels for a worthwhile result. Some studios have considered the possibility of an in-line compressor in every input channel. Other than the set-up

*Pres. Philip C. Erhorn & Assoc.
Stony Brook, L.I. N.Y.*

time saved, this would seem to be expensive as well as redundant. Several single and perhaps multi-channel LDR compressors, placed conveniently, but terminating in jacks, will certainly provide an inexpensive yet very effective facility, with patching as the only penalty.

## Patching

Now the word patching brings up another vein of thought. Very few small consoles contain a patching facility, and the large ones are designed so that pushbuttons and lever keys relegate the jackfield to testing and emergency use. Harking back to some pioneering custom designs of more than 10 years ago, the in-line arrangement of all control components associated with each input channel, came into being as a piece of human engineering with the console operator in mind.

Today, several oldtimers, and some new manufacturers are offering complete, in-line input and output modules, containing a mixing pot, an equalizer, echo mixing, and several variations of channel, echo, and cue switching. A console may be constructed from these modules with a minimum of mechanical effort, and certainly a reduction in wiring complexity. Some lend themselves to the optional inclusion of a jackfield, and others do not. It may be argued that considering the apparent reliability of solid state electronics, (and who uses tubes today?) that the real need for emergency crosspatching is eliminated. This philosophy is further strengthened by the fact that switches are used with great versatility in modern consoles, eliminating the need for patching in normal operation.

## Console Kits?

Most recently, at least two manufacturers have placed on the market do-it-yourself console packages, utilizing combinations of their modular control strips to produce rather complex facilities with virtually all mechanical work done, particularly the fabrication of a suitable shell. Such an approach is naturally a great help, since the mechanical construction of anything but the most simple console can be very tedious. Eye appeal is excellent, and layout from an operating viewpoint can be functional as well as flexible.

I would like to dwell for a few moments upon one facet of this approach to completed consoles. The idea that future expansion is easy, is to me erroneous. The practical complications of adding to mixing networks (with the possible exception of active combining networks which include an operational amplifier); the physical business of adding much future wiring, including additional power supply wiring; the complexity of buss-switching wiring; and the age-old problem of proper grounding and possible ground loops, are of no small importance to the wire-it-yourself studio, even when suggestions and precautions accompany the package. And don't overlook the fact that adequate monitor switching and the mass of wiring associated with such a complex, is not for the uninitiated!

Returning to the subject of large custom recording consoles, several thoughts come to mind for consideration by those studios who have sufficient business to afford them. The only valid reason for multi-track tape feeds is the ability this conveys in matters of re-mixing for balance and effect; the tricks of adjusting echo, equalization, compression, and even the effectiveness of a board fade. It is vitally necessary for the successful studio to offer their clients all of these abilities on an almost instantaneous basis.

The ideal large console (even a conventional 4-channel system) must offer numerous mike and hi-level input/channels with the usual echo and equalization controls. In addition to four, eight, or more output channels, there must be 2-track and mono supplementary tape feeds made up in any combination of the main channels. Monitoring of any of these outputs must be available on a pushbutton switching complex, including the ability to flip from console to tape playback instantly. Echo return, including tape delay if desired, must be flexible. The client should be able to leave for the day with a 2-track or mono tape containing material with typical echo, equalization, and board fade effects already added. While the original recording may have been mixed in multi-track fashion, he probably has spent the day listening to a stereo or mono monitor mixdown.

After auditioning his tape at his office, the client may return to the studio another day for a remix session, without the burden of expensive talent. The basic material is on the studio's multi-track original tapes, and if the console is versatile, these originals will be relatively "dry", that is, they will be minus echo and board-fade effects. Equalization will be present, as virtually no client or studio is satisfied with the flat sound heard by the studio mikes. Now the business of adding just the right amount of echo, changing the balance, equalization, etc. may be done with the same console, or if the studio is fortunate in owning one, with a re-mixing console, which does not tie up the big studio. As was discussed earlier, compression should also be available in the over-all console flexibility, as well as cue circuits to talk to the conductor or vocalist while recording, and to feed cue for overdubbing or tracking. In a console of this general complexity, a large jackfield seems to be inevitable, as we are otherwise fighting something that has been with us for about 40 years!

But I think that the real need today is for a console considerably less complex than these big ones. There is a definite need for two- and four-channel consoles with small- to medium-sized input, and accompanying facilities. Because price is always a deterrent, the new in-line modules make the choice of a do-it-yourself approach quite interesting.

But for many, the purchase of a completed package is more desirable from a practical standpoint. And this market potential will go largely untapped until manufacturers manage to reduce their own costs on electronic components, particularly amplifiers, equalizers, and, of course, that important mixing pot. Then they will be able to pass along their savings, so that the smaller studios may purchase an adequate and up-to-date console forgoing the makeshift. This same console will also afford excellent remixing facilities, and the studio will be able to expand business based upon their new ability to compete more closely with the large studios.

The most important item any studio can offer is service to their clients. Those studios doing the biggest volume of business not only offer excellent technical facilities, but also quick and efficient service. That old cliche time-is-money applies to both client and studio. I personally like to think that money spent on modern, versatile technical facilities, will come back many times over, particularly when accompanied by a sensible attitude toward clients. And I perhaps selfishly believe that a fine mixing console is the single, most important piece of equipment in the studio.

# High-Quality Monitor Loud-speakers

—Harry F. Olson*

*The author, an outstanding authority, discusses the prime performance criteria that must be understood if the proper selection of a quality monitor speaker system is to be made.*

WIDE frequency range, low distortion, uniform directivity and relatively high sound-power output loudspeakers are required for monitoring in radio and television broadcasting, phonograph and sound motion-picture recording, and high-quality sound systems. The performance criteria of the sound reproducing systems for these applications should satisfy the conditions required to provide high-fidelity sound reproduction. The purpose of this article is to define high-fidelity sound reproduction, describe the loudspeaker characteristics involved in achieving high-fidelity sound reproduction, give specifications of the loudspeaker performance characteristics and the means for providing the required performance characteristics in a loudspeaker.

*RCA Laboratories
Princeton, New Jersey

## Performance Requirements For High-Fidelity Sound Reproduction

In the field of sound reproduction the term *high fidelity* is used to designate a quality of sound reproduction which provides a high order of realism.

To achieve this realism, that is high-fidelity sound reproduction, three fundamental conditions must be satisfied as follows:

1. *The frequency range must be such as to include without frequency discrimination all the audible components of the various sounds to be reproduced.*
2. *The volume range must be such as to permit noiseless and distortionless reproduction of the entire range of intensity associated with the sounds.*
3. *The subjective aspects involving psycho-acoustic effects must be used in an appropriate manner in order to obtain a close artistic resemblance of the original rendition; these include loudness and dynamics, noise, auditory perspective and reverberation.*

The elements of a sound reproducing system must exhibit such an order of excellence of performance so that the above conditions are satisfied if high-fidelity sound reproduction standards are to be met. The loudspeaker is one of the important elements of the sound reproducing chain. Its characteristics involved in achieving superior performance are: *frequency response, directivity, non-linear distortion, transient response, impedance, efficiency,* and *maximum sound-power output.*

## Frequency Response

*A loudspeaker's frequency response is the sound pressure on the axis as a function of the frequency.*

The loudspeaker should provide uniform response over the frequency range from 30 to 15,000 Hz. For most high-quality applications the response should be contained within the limits depicted in fig. 1. In general, there is no problem in achieving this performance today in a high-quality loudspeaker. The procedure is to divide the frequency range into two or more parts and employ separate mechanisms for each range. This expedient makes it possible to achieve uniform response in both direct radiator and horn loudspeakers. However, some care must be exercised in the geometric configuration of the mechanisms to avoid deleterious phase effects in overlap frequency regions due to a large physical separation compared to the wavelength between the mechanisms.

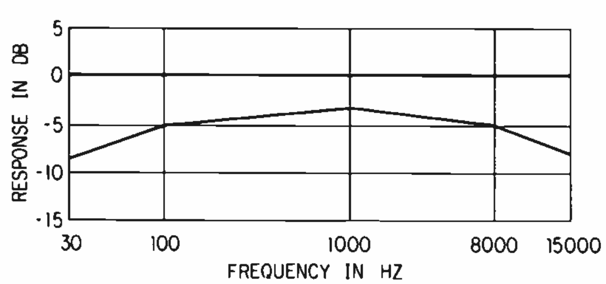



**Fig. 1. A high-quality loudspeaker system should fall between these upper and lower limits of frequency response.**

## Directivity

*The directional characteristic of a loudspeaker is its response as a function of the angle with respect to some reference axis of the system.*

The directional patterns are usually depicted in polar coordinates. If the directivity pattern varies with frequency, frequency discrimination will occur for observation and listening points removed from the axis. Uniform directivity is particularly important in stereophonic sound reproduction in order to provide realistic auditory perspective.

Limits on the directivity pattern of a loudspeaker over the response range of 30 to 15,000 Hz are depicted in fig. 2. The polar curves should fall within the amplitude limits shown for an angle of at least 45°. For really high quality, the polar curves should fall within the amplitude limits shown for an angle of 90°.

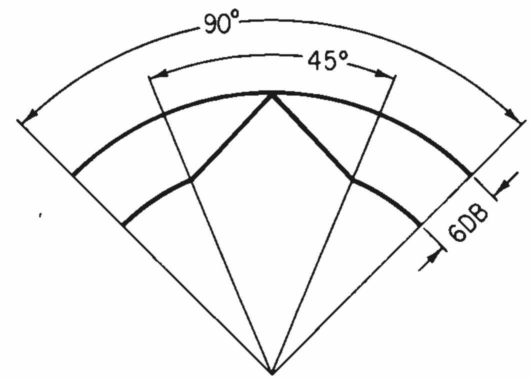



**Fig. 2. The directivity pattern of a good speaker system should fall between these extremes over the 30 to 15,000 Hz range.**

The directivity pattern of a direct radiator loudspeaker can be controlled by the shape of the cone. The directivity pattern of horn loudspeakers can be controlled by the shape and design of the horn. An additional powerful means which can be employed to control the directivity pattern over the wide frequency range is the idea of dividing the frequency range into two or more sections. Additional expedients which may be employed to control the directivity are the use of diffractors, reflectors and acoustic lens.

## Non-Linear Distortion

*The non-linear distortion frequency characteristic of a loudspeaker is the total non-linear distortion as a function of the frequency.*

The major result of non-linearity in the elements of the vibrating system of a direct radiator loudspeaker is the production of harmonics and subharmonics. Two major contributors to non-linear distortion in dynamic loudspeaker mechanisms are the driving and suspension elements. These elements are constant for small and moderate amplitudes but depart from constancy for large excursions of the cone or diaphragm. In the low-frequency range the amplitude of the cone in the direct-radiator loudspeaker must be inversely proportional to the square of the frequency and the amplitude of the diaphragm in the horn loudspeaker must be inversely proportional to the frequency in order to maintain constant output with respect to frequency. Thus it will be seen that amplitude increases rapidly with a decrease of the frequency. For these large amplitudes there is considerable departure from linearity in the operation of the driving and suspension systems.

A typical non-linear-distortion frequency characteristic of a direct-radiator loudspeaker with non-linear driving and suspension systems for full output is depicted by curve **A** of fig. 3. The distortion decreases as the power output decreases.

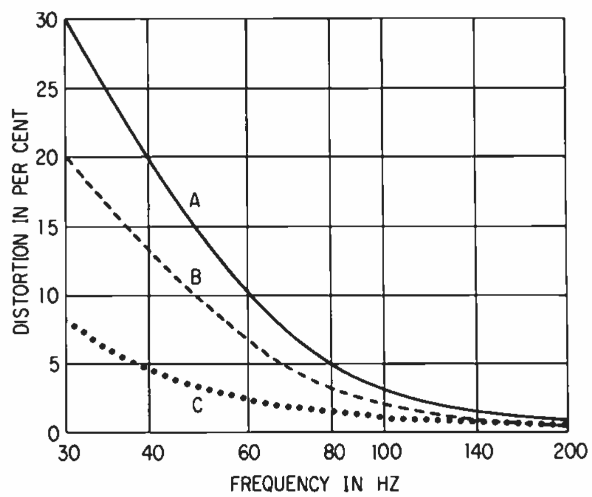



**Fig. 3. The non-linear distortion characteristics of a loudspeaker. (A) Typical of a non-linear driving and suspension system, (B) shows a non-linear suspension system only, and (C) is characteristic of a low-resonant mechanism mounted in a completely enclosed cabinet.**

Distortion due to the non-linear driving system can be eliminated by employing a voice-coil design as depicted in fig. 4. In this configuration the flux-turns product will be a constant if the amplitude of excursion is always less than *x* of fig. 4. The reduction in distortion by employing the expedient of fig. 4 is shown by the curve **B** of fig. 3.

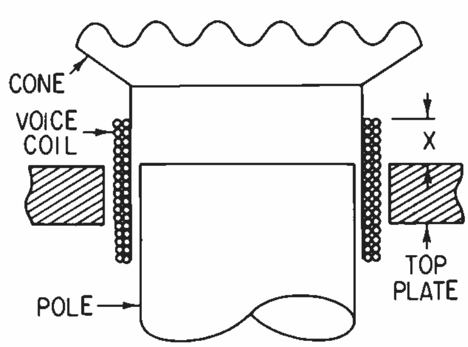



**Fig. 4. A voice-coil design for eliminating non-linear effects in the driving system.**

Distortion due to the non-linear suspension system can be reduced by the use of a loudspeaker mechanism with a low resonant frequency and a completely enclosed cabinet in which the linear compliance of the air in the cabinet is the controlling compliance as shown in fig. 5. The mechanical circuit of fig. 5 depicts the method for reducing the deleterious effects of a non-linear suspension. The non-linear distortion is reduced to curve **C** by the use of the low-frequency-resonant loudspeaker mechanisms housed in a completely enclosed cabinet of fig. 5 and the voice-coil-design of fig. 4.

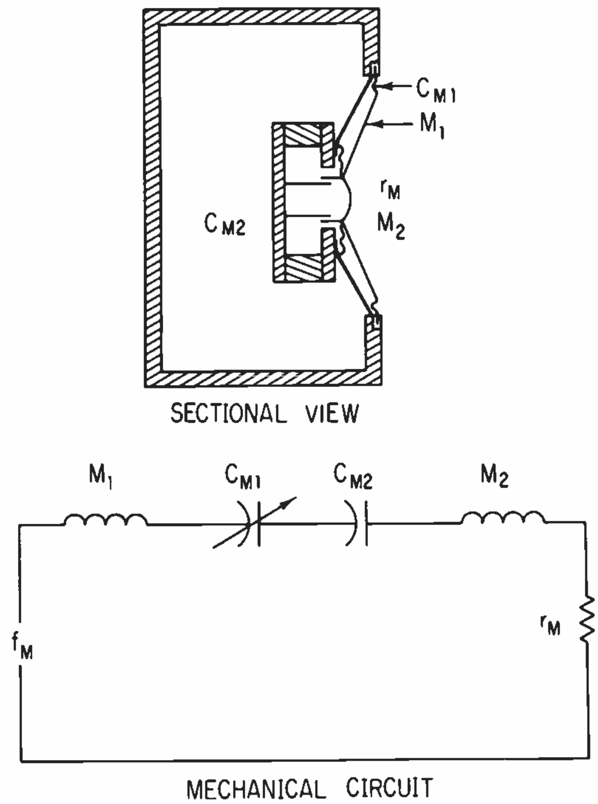



**Fig. 5. A sectional view and the equivalent mechanical direct-radiator loudspeaker mechanism mounted in a completely enclosed cabinet. In the mechanical circuit: $f_M$ driving force; $C_{M1}$ non-linear compliance of the compliance of the suspension system; $C_{M2}$ compliance of the air in the cabinet; $m_2$ mass of the air load; $r_M$ mechanical resistance due to the radiation of sound.**

The cones and diaphragms of loudspeakers should be designed so that the operation falls within the limits of Hooke's law for the material and construction. This leads to a relatively heavy cone or diaphragm. This poses no problem except for a reduction in sensitivity. The subject of sensitivity will be discussed in a later section. Employing a heavy cone or diaphragm, the distortion in the frequency range above 200 Hz will be exceedingly low.

## Transient Response

*Transient response in a loudspeaker refers to the faithfulness of response of the loudspeaker to a sudden change in the electrical input.*

For the most part the sounds of speech and music are of a transient rather than a steady-state character. Therefore, practically all sounds which are reproduced in a sound-reproducing system are of a transient nature. As a consequence, the transient response of a loudspeaker must be considered an important factor in the performance of the loudspeaker.

There are two general tests employed to depict the transient response of a loudspeaker: the *square wave* and *tone burst*.

Figure 6 shows the amplitude response of a loudspeaker and tone-burst response. The tone-burst response is quite faithful in the smooth part of the amplitude range. However, there is considerable deviation of the tone-burst response at the peak and the dip. In general, in a properly designed loudspeaker, if the response of the loudspeaker is smooth the transient response will be good.

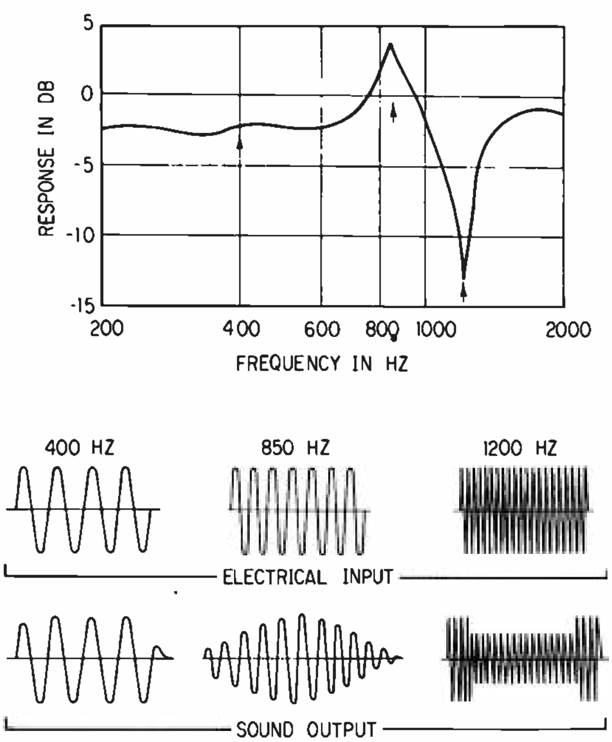



**Fig. 6. The relationship of the frequency-response characteristic and the tone-burst response of the same loudspeaker is shown at 400, 850, and 1200 Hz.**

## Impedance

*The frequency impedance characteristic is the electrical impedance of the loudspeaker as a function of frequency.*

In the dynamic loudspeaker the variation with frequency is relatively small and there is normally no problem on the transfer of power from the amplifier to the loudspeaker.

## Efficiency

*The efficiency of a loudspeaker is the ratio of the sound-power output to electrical-power input.*

For monitoring applications the low efficiency exhibited by most monitoring loudspeakers is of little concern because the maximum sound levels needed can be obtained with relatively modest amplifiers. However, for large-scale sound reproduction, as for example, sound reinforcing systems in large auditoriums, efficiency is a consideration if the amplifier requirements are not to become ridiculously large.

## Maximum Sound Power Output

*The maximum sound power output of a loudspeaker determines the level which the loudspeaker can deliver under a particular condition.*

For monitoring purposes the loudspeaker should be capable of delivering a level of at least 100 dB. The sound-power output required as a function of the room volume is shown in fig. 7. An alternate specification is 1 acoustic watt per 1000 square feet of area to be covered.

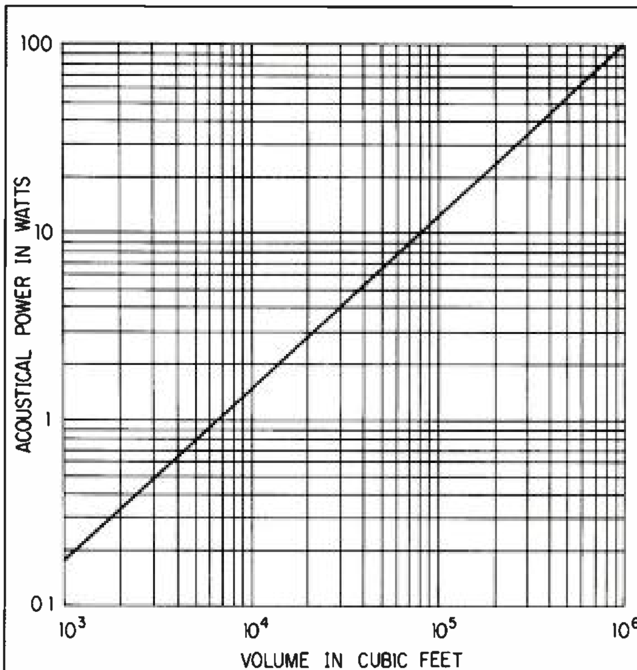



**Fig. 7. The acoustical power required to produce a 100 dB level in monitoring room, studio, or auditorium, is shown as a function of the room's cubic volume.**

These, in sum, are the performance characteristics used to determine high-quality loudspeakers for use in professional monitoring and sound systems. In general, this data represents the state of the loudspeaker art at this time.

## Additional Comments

The loudspeakers used for professional sound applications are practically all of the dynamic type mechanism operating either as a direct radiator or horn driver. The main reasons for this state of affairs are as follows:

1. *The mechanisms are relatively simple.*
2. *The mechanisms are comparatively easy to build.*
3. *The mechanisms are rugged and outlast some of the other electronic components.*
4. *The mechanisms are relatively reliable.*
5. *Very good performance characteristics can be obtained if there are no unreasonable limits on cost.*

Loudspeakers are often accused of being the weakest link in the sound reproducing chain. The question arises, namely, if the performance of a loudspeaker is so terrible, why is it possible by means of subjective tests employing a loudspeaker to demonstrate a minor improvement in one of the other elements of the sound-reproducing system? As a specific example, why is it possible by means of listening tests to detect a difference in the performance of two power amplifiers of different designs, both flat to within a decibel from 20 to 20,000 Hz, both with distortion of a small fraction of a per cent up to 20 watts, and both with excellent transient response?

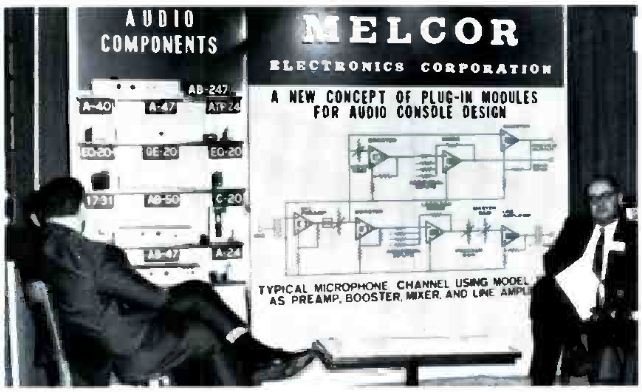
The answer is that the performance of a high-quality loudspeaker is pretty good for the various characteristics in the amplitude and frequency ranges that count.

# The New York AES Convention

☐ New York's Barbizon-Plaza Hotel bustled with activity this past October. For four days, some thirty-nine exhibitors displayed their wares. Our inquisitive camera managed to poke its lens into most of the booths . . .


Looking up over a **Crown** tape recorder that operates with automated fool-proof controls we see a smiling Clyde Moore telling all.


At **Ampex**, S. Bryan Hickox shows a prospect one of their eight-track machines.




At **Melcor**, one late evening, conversation went on at one end while rep Gil Miller sat at the other end.

One of the prettier sights around the exhibition was Gladys Hopkowitz holding forth at the **Sound Technique, Inc.** booth as she demonstrated one of their systems.

**Dolby Labs** had a hotel room set up to offer demonstrations of their noise suppressor. Here inventor Ray Dolby gestures in emphasis of a point made to Albert Grundy.

It's one of many busy moments at **Nortronics** as Thor Johnson (left center) makes one point while Mervin Kronfeld (Nortronics' board chairman) at right makes another.

Loren Ryder leans over to see into his display case while discussing a point about the **Nagra** recorder system.


At **B & K,** Bernie Katz (standing) discusses sophisticated measurement techniques.


Gene Rosen and Art Gruber (left) at the **Langevin** booth.

Who told the joke that broke up **Harvey Radio**'s Don Plunkett and Ed Lewin and the **Naval Training Device Center**'s Allen P. Smith (left to right)?

Tom Adams and Elliott Porte talk while a blond head listens at their **Audio Instrument** display.

**Scully's** H. H. Brosious (left center), **Bozak's** Paul Stone, and **3M's** Scotty Lyall (second from right—facing right) were wrapped around one corner of a room.

Ray McKinnon is all smiles as he demonstrates at the **Universal Audio** display.


Bill Watkins of **Altec-Lansing** plays the new 9200 series console. Of course, that's an Altec monitor speaker in the background.


**Hewlett Packard**'s handsome display. The flashgun washed out a 'scope display above that is being duplicated on the tracer below it.


Tom Dempsey of **Computron** tells Eli Passen of **Harvey Radio** something about **BASF** tape.

**Gotham Audio** rented a large room upstairs in the hotel to display its wares. Here Gotham's Hugh Allen (right) points out a **Neumann** console feature to **Capitol Records**' Irv Joel.


It was always busy at the **Electro-Voice** display. Directly behind E-V was **Haeco.**

George Alexandrovich (center, facing right) discusses **Fairchild** engineering with an interested listener.

Albert V. Siniscal listens at the **Spectra-Sonics** booth while a visitor discusses the virtues of a circuit board.

Our camera caught a studious moment at the **Bozak** booth. Eli Passen (left) looks over the shoulder of Rudy Bozak as he makes a point to Daniel Flickenger.

Black-and-white photography does not do justice to the multi-color effects used on modern studio consoles such as this one at **Electrodyne**'s display. This same console is the subject of our cover photograph.

# New Products and Services

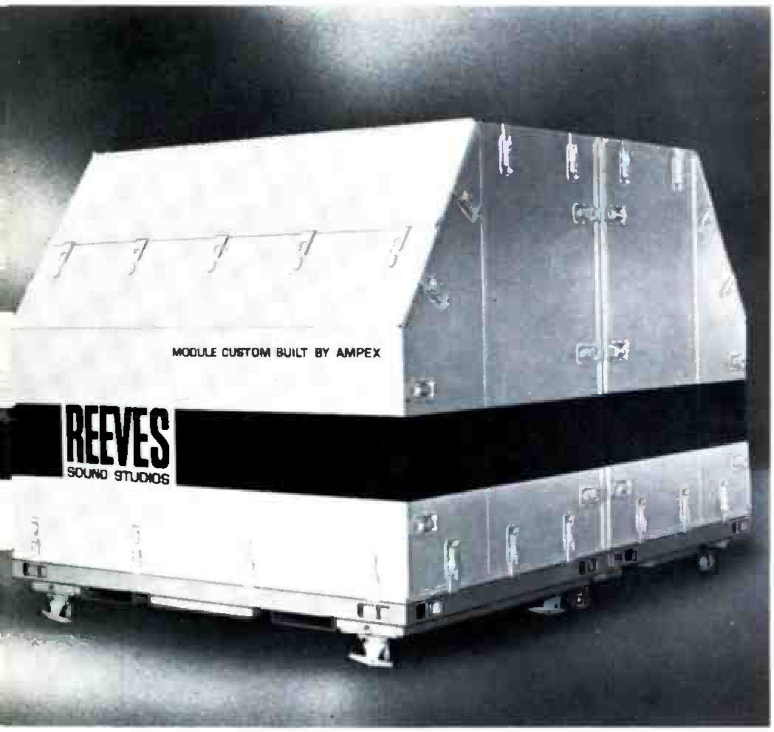



Fig. 1. The basic module designed to fit a standard jet cargo hold. It is pre-mounted on shipping skids.

Fig. 2. The innards of the module fit nicely into the body of a standard rental truck. Note that it still rides on its skids. Note also the camera bodies stored below the desk of the console.

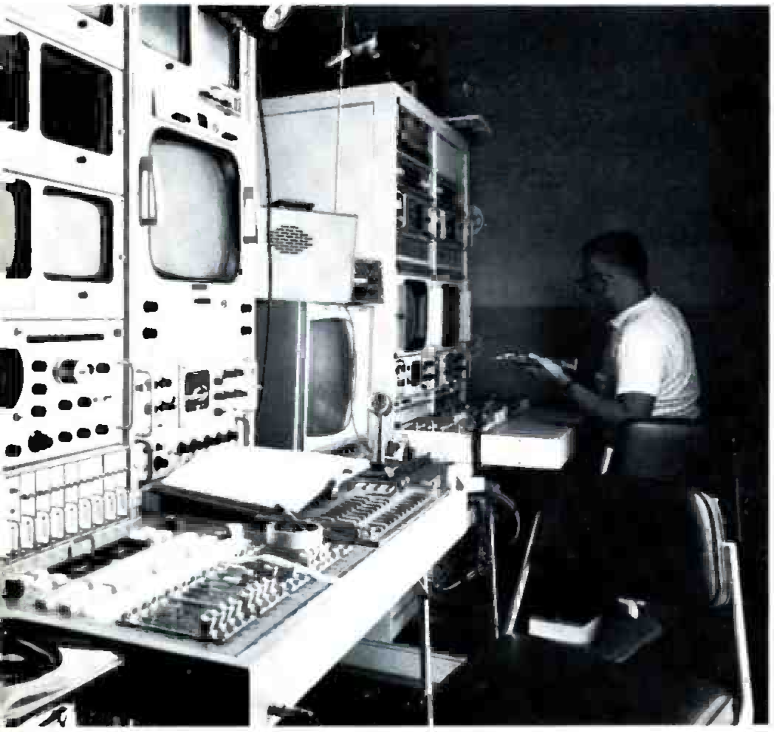


Fig. 3. With the cameras and accessory items removed from below the desk, a roomy control room is established inside a truck body.

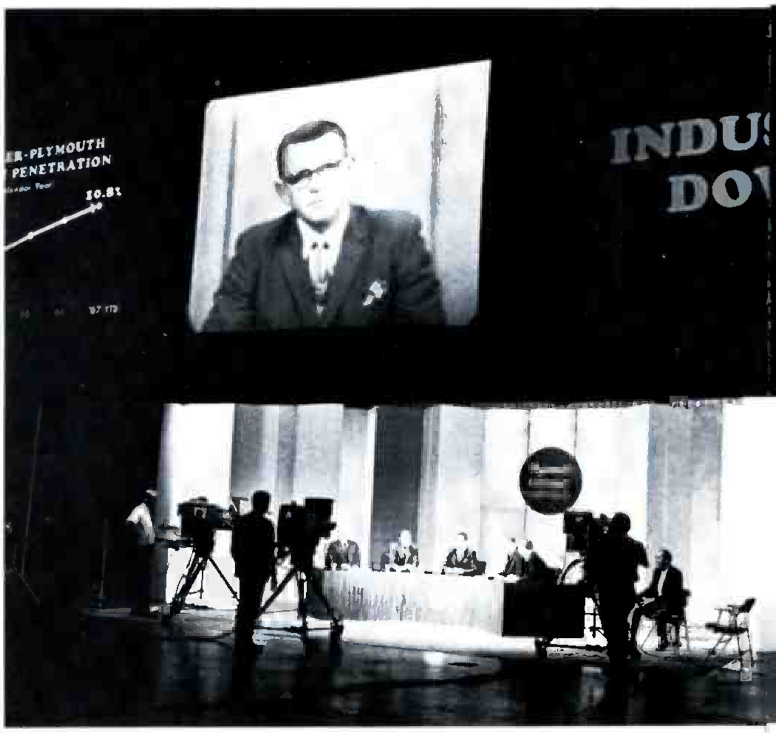


Fig. 4. In an incredibly short time, the system can be set up and rolling. It can feed a system, record on the spot, or go into live broadcast lines.

At the beginning of October, Reeves Sound Studios of New York unveiled, in a working demonstration, their Airmobile-Video* system. In a series of sessions, the system was displayed and operated for major tv producers, film production houses, network executives, and ad agencies.

The Airmobile-Video system is a "packaged" modular system of color television cameras, video tape recorders, and control equipment specifically designed for and installed in standard jet air-freight shipping containers. These self-contained packages can reach any jet port within a 24-hour period. On arrival, the package splits in half to fit a standard van-type rental truck. This system is available only on a rental or lease basis from Reeves.

Because of the modular design audio, switching, and monitoring equipment can be placed in one truck with video control operations in another. Actually, a two-camera, one-video-recorder production job will fit into one truck as will a four-camera, no-recorder job. Of course, the modules are easily installed indoors in hotel rooms and convention halls.

The cameras are Norelco Plumicon color cameras with Taylor-Hobson 21-210mm zoom lenses, Mitchell-Vinten camera heads, and contour enhancement and red masking in each supplied camera chain. The recorders are Ampex VR2000 units with Editec electronic editing, an auto velocity compensator, drop-out compensators, and color monitors on each unit supplied.

Switching is done on custom Riker switchers with inputs for up to eight cameras. There is chroma-key, a Riker special-effects generator, a color-bar generator, and eight Conrac preview monitors in each console. A Conrac color program monitor is supplied.

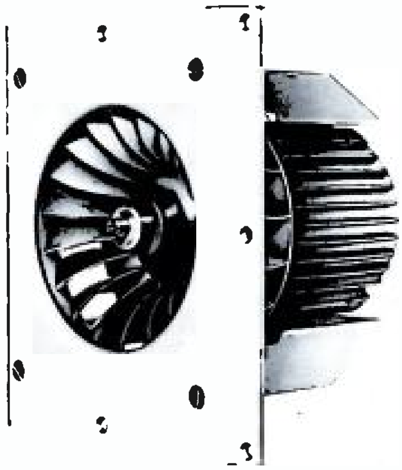
The audio console has sixteen fader positions with a custom Ward submaster and over-all master control pots. Two Nagra 1/4-in. recorders and up to fourteen mikes are included.

Onan 15-kw power generators and combination 3-ton air conditioning and heating units are also available.

This equipment can be made up in a wide variety of combinations. Any combination of 1-6 cameras and 0-2 video recorders can be ordered. A special package of one camera and one recorder is available for commercial production.

**T. M. Reeves Sound Studios*
*Mfgr: Reeves Sound Studios*
*Price: on request*
*Circle 50 on Reader Service Card*

## Centraxial Blower



Uniform air distribution problems for densly packed printed card arrays are claimed as solved by this new line of blowers. The model G Centraxial makes use of the characteristics of mixed-flow impellers which provide the high pressure normally available only from conventional squirrel-cage designs. This design claims that a uniform air discharge is superior to squirrel cages. Designed with ball-bearing induction motors, low-noise Centraxial G blowers distribute air evenly with adequate pressure for many rows of cards, without overcooling some and leaving others in stagnant air.

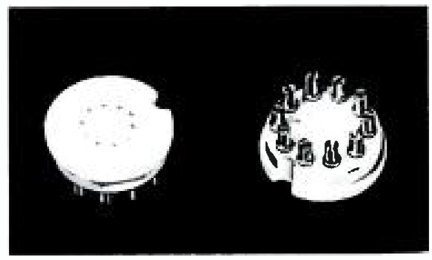
*Mfgr: Rotron Manufacturing Co.*
*Price: on request*
*Circle 51 on Reader Service Card*

## Button Power Rectifiers

Three series of flexible, space-saving, fast-recovery, silicon rectifiers in flat-button form—the TR, RU, and RD—with ratings of 20, 40, and 200 amps at 100°C case temperature. The RU series handles 40 amps and the RD units—200 amps. Both the TR and RU series have 500 nanosecond recovery times. The RD units offer a 700 nanosecond recovery time when measured from 1 amp forward current to 30 volts blocking. All rectifiers can be supplied singly or in stack combinations. PIV ratings are available from 50 to 1000 volts.

*Mfgr: Electronic Devices, Inc.*
*Price: on request*
*Circle 52 on Reader Service Card*

## IC Socket



This new integrated-circuit socket is engineered to accept a conventional 10-pin TO-5 package. Designated Press-Fit part number RTC-1010SL-C1, the new socket features ten 0.040-in. diameter holes in a 0.230-in. pitch circle. On the underside are to be found ten gold flash over silver-plated brass slotted solder lugs on a 0.40-in. pitch circle. Leads from the IC device are passed through the ten inner holes in the Teflon body, and soldered into the slots in the outer lugs. The unit can be supplied in any one of ten standard colors.

*Mfgr: Sealectro Corporation*
*Price: on request*
*Circle 53 on Reader Service Card*

## Molded Deck Switch

A new molded deck switch is now available for applications requiring protection against environmental hazards such as corrosion, humidity, sand and dust, fungus, careless handling, and explosion. The totally-enclosed switch features molded-in contact terminals that prevent open connections or intermittent continuity. One-piece contact terminals are of solid silver alloy. Outside terminals are gold-plated for ease of soldering. Other current-carrying parts are gold-plated beryllium copper. Diameter is two inches, 1 pole per deck with 2-24 positions per deck with ad-

justable stops, either shorting or non-shorting, and/or 24 position continuous rotation with 15° spacing. Up to twelve decks may be ganged on a single assembly. Load-carrying capacity is 10 amperes at 24 volts d.c.
*Mfgr: Aerovox Corporation*
*Price: on request*
*Circle 54 on Reader Service Card*

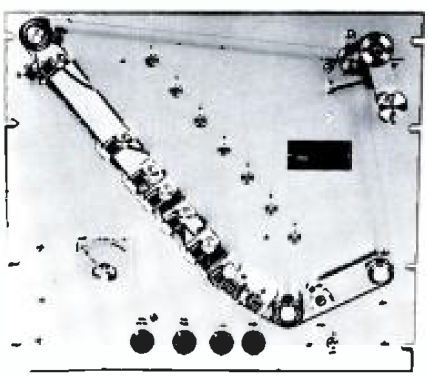
## Eight-Channel Recorder

This is actually an eight-track version of the established AG-440 series line. In order to accommodate the additional tracks the AG-440-8 uses 1-inch wide tape. (Standard AG-440's use 1/4- or 1/2-inch tape.) The additional width permits this unit to accommodate eight tracks with the high-frequency response and crosstalk rejection essential to master recording. The AG-440-8 includes the same plug-in solid-state modular electronics, plug-in head assemblies, tape lifters, adjustable azimuth for individual head stacks, three-way editing capability, and scrape-flutter idlers as the rest of the series. Electronic circuitry is designed to plug into the front of the chassis. The transport plate is precision mill die-cast to prevent flexing under demanding studio operations.
*Mfgr: Ampex Corporation*
*Price: $12,500*
*Circle 55 on Reader Service Card*

## PC Board Bias Oscillator


The T70-T2 bias oscillator features convenient plug-in dip-solder terminals and has the same efficient ferrite E-core construction and electrical characteristics as the popular Nortronics T60-T2 transformer. This unit will furnish up to 110 kHz of bias and erase power to full-track heads or to both channels of a stereo recorder.
*Mfgr: Nortronics Company, Inc.*
*Price: $4.50*
*Circle 56 on Reader Service Card*

## Delay Recorder

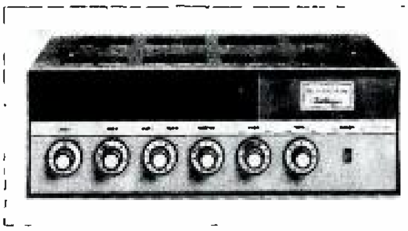


Recordists of pop music in particular will welcome the continuously variable delay range of 25 to 375 milliseconds at a tape speed of 30 inches-per-second. The model 303 Delay Recorder can accommodate as many as fifteen heads thus allowing the creation of highly specialized effects. Although the standard unit's speed is fixed at 30 inches-per-second, other speeds are available at slightly higher cost. Technical specifications include: input and output impedances of 600 Ω, balanced or unbalanced; maximum input and output levels: 0 VU; frequency range: 50-15,000 Hz, ±2 dB; s/n: 50 dB or better. The panel fits a standard rack and occupies 17 1/2 inches.
*Mfgr: Audio Instrument Company*
*Price: on request*
*Circle 57 on Reader Service Card*

## Cassette Duplicator

High-speed production of up to ten cassettes at a time is offered in this new duplicator. Cassettes may be loaded with predetermined tape lengths saving labor time, loading time, and the need for meticulous handling of previously recorded tape for subsequent installation in cassettes. Heads are optically aligned—no adjustment is necessary; duplicating speeds of 15 or 30 inches-per-second are available; all electronics are included and are solid-state; maintenance is made easy by rear-door access. Versions of the unit are available for master cassettes or reel-to-reel masters.
*Mfgr: Lang Electronics, Inc.*
*Price: on request*
*Circle 58 on Reader Service Card*

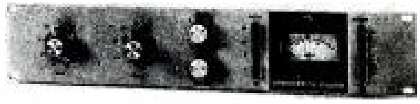
## Economy P.A. Line



New developments in solid-state engineering are claimed to provide advanced features in this new line. Designated the Challenger CHS series, availability is in four models ranging in power output from 20 watts to 100 watts. Designed for versatility and heavy use, the line combines the economy of the Bogen Challenger lines with the reliability of solid-state design. The amplifiers will operate continuously at full output from -20°C to +65°C. All major components are mounted on printed-circuit boards. The two higher-powered models, the CHS100 (pictured) and the CHS50 use all-silicon semiconductors. Each of these units has two mike inputs. The 35-watt model also features two mike inputs and may be operated from a 12-volt d.c. supply as well as conventional a.c. The CHS20 20 watt unit has one mike input. All models have two auxiliary inputs on a fader control. All models will accept a panel-mount preamp accessory that adds two additional mikes. Operating ease is served by the inclusion of high- or low-impedance mike inputs on all units. Mike precedence circuits that may be remotely operated are also included. Conveniences include memory markers to aid in returning controls to previously determined levels. All Challenger controls are of the single-function type to better avoid accidental switching or volume-level errors under the stresses of operation. Accessories include phono locking covers, rack-panel mounting kits, remote controllers, telephone line input or output plug-in transformers, and carrying cases.

*Mfgr: Bogen Communications Division/ Lear Siegler, Inc.*
*Prices: CHS100—$157.15*
*CHS50 —$127.45*
*CHS35 —$104.95*
*CHS20 —$ 92.25*
*Circle 59 on Reader Service Card*

## Limiting Amplifier



The application of a field-effect transistor to an all-transistor limiting amplifier, the model 1176 is claimed as a major breakthrough in limiter design. The FET is used as a voltage-variable resistor to achieve automatic gain reduction ahead of the first stage of amplification. This permits severe limiting without added distortion. Other special features include a fast attack time of less than 20 seconds, switch-selective compression ratios, remote metering, compact size (3 1/2-in rack space), and excellent stability. Input and output impedance is 600 Ω floating; external connection is via a Jones barrier terminal at the rear; gain is 50 dB; distortion is less than 0.5 per cent t.h.d. from 50 to 15 kHz with limiting, at 1.1 seconds release setting; s/n is greater than 70 dB referenced to a +10 dBm output level.

*Mfgr: Universal Audio*
*Price: $489.00*
*Circle 60 on Reader Service Card*



Circle 19 on Reader Service Card

# People, Places, Happenings

## People

The promotion of Howard Durbin to vice president of marketing was recently announced by Albert Kahn, president of Electro-Voice, Inc. Mr. Durbin has been E-V general product manager since May, 1967, and prior to serving in this position, he was sales manager of the company's OEM division. According to Mr. Kahn this appointment has been made as a part of the company's continuing effort to strengthen their marketing organization and to increase the responsibilities of competent management personnel.

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Cousins

A number of moves at Allied Radio's Stores Division have placed Robert Cousins as the new merchandising manager, Milton J. Blumberg as branch stores manager, and Paul J. Gilbertson as operating manager. The last two are newly created posts. All three men report to Shelby F. Young, Allied v-p and manager of the stores division. According to Mr. Young, "The assignment of important responsibilities to these experienced executives strengthens our organization, and will permit us to provide even better services and merchandise selection for our customers." Among the three appointees, there is a total of twenty-eight years of service to Allied Radio.

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James A. Thomas has been named manager, field engineering, and Richard L. Roule has been appointed manager, video products, according to an announcement by Alec Autote, marketing manager, CBS Laboratories professional products. Mr. Thomas had been the vice president and director of engineering of Broadcast Enterprises, Inc., of Melbourne, Florida. Mr. Roule has been associated with major developments in the area of miniature television cameras and image transmission systems, while a field-service engineer with the R & D departments of CBS Laboratories.

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Toshio Miyamoto, who has been general manager of the international division of Sony Corp. in Tokyo, has been elected a vice-president of the company's subsidiary, Sony Corporation of America. The announcement was made by Ernest B. Schwarzenbach, president of the U.S. based firm. Mr. Miyamoto will be an assistant to Mr. Schwarzenbach. Shigeru Inagaki will continue as executive vice-president and general sales manager for all consumer products.

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Kronfeld/Yngve

The board of directors of the Nortronics Company, Inc. has announced the election of John A. Yngve as president. Leonard Kronfeld, who founded the company in 1956 becomes chairman of the board of directors and chief executive officer. Mr. Kronfeld will have primary responsibility for engineering and technical aspects of the business. Mr. Yngve has been with Nortronics for over ten years. During that time he has been chief legal counsel, corporate secretary, member of the board of directors, and consultant on corporate development. His new responsibilities will be in the areas of general administration, marketing, finance, and other non-engineering business. Mr. Ygnve has also found time to serve three terms in the Minnesota State Legislature, and has practiced law for more than 16 years as a partner in the firm of Ygnve and Ygnve. According to Mr. Kronfeld, his election was made necessary by a "decade of extremely rapid growth from a small company with only a few employees to the point where it is now the largest manufacturer of magnetic tape heads in the world."

Blackburn/Gov. Volpe/Fried/Schneider

The "Faith in Massachusetts" award was presented by Governor John A. Volpe to executives of Computron Inc. in recognition of the new $8.5 million plant in Bedford, Mass. The participants were: R. Donald Blackburn, manager of community relations; Gov. Volpe; Albert A. Fried, chairman of the board; and Claus Schneider, president.

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## Happenings

Telex has announced the consolidation of U.S. representatives for the Telex Communications Group. This group includes Telex Acoustics, Magnecord, and Viking – all based in Minneapolis. According to director of marketing James S. Arrington, the consolidation is merely one more step of a master plan to merge all sales and marketing. The three divisions already operate under a common management.

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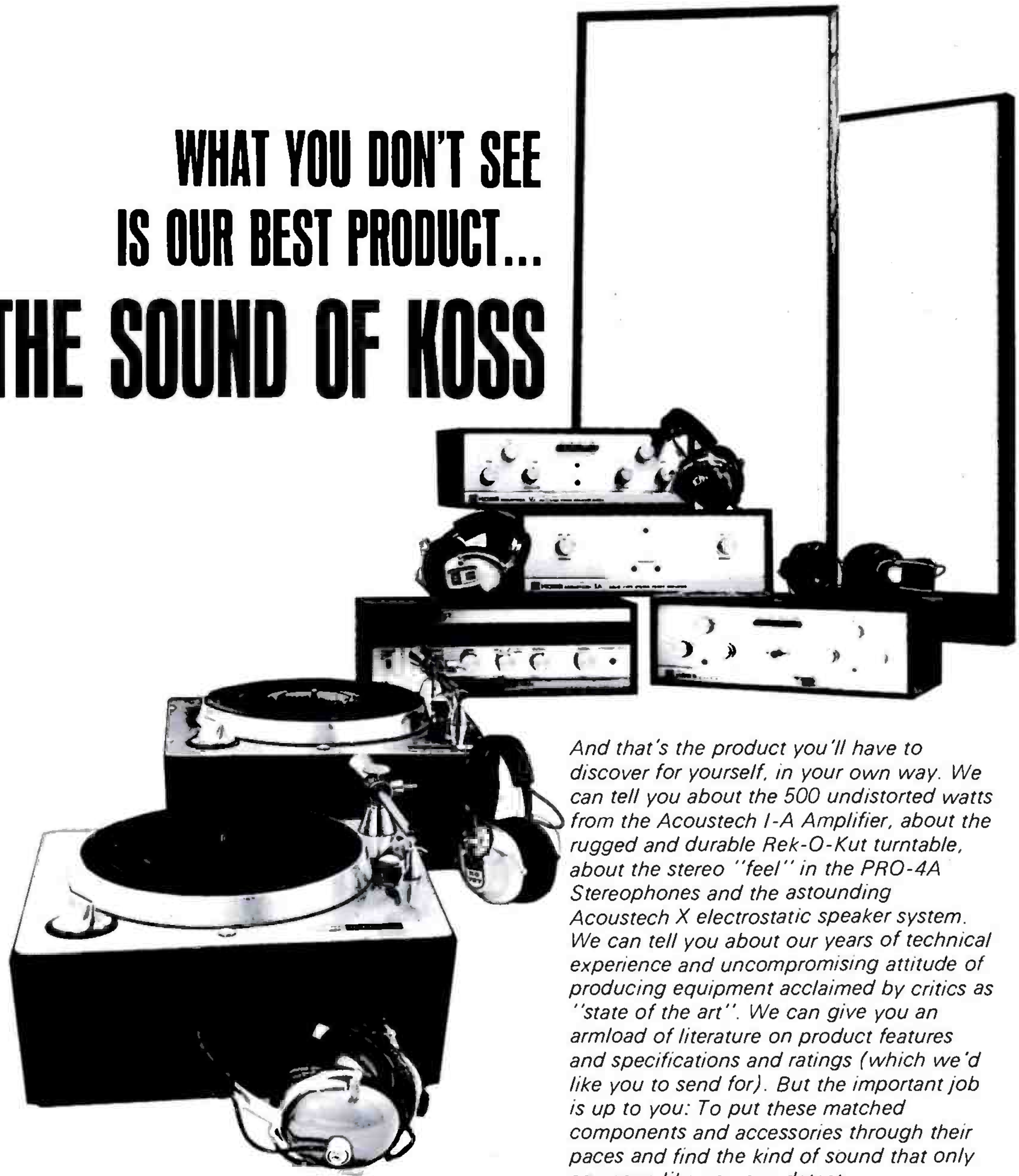
United Recording Corporation of Hollywood, California has recently announced the acquisition of Waveforms, Inc. Under the merger, URC has acquired the Waveforms' name and all assets including product line, designs, and engineering. URC is the largest independent custom recording complex with subsidiaries and affiliated operations that include Western Recorders, Inc. of Hollywood; Coast Recorders, Inc. of San Francisco; affiliated United Recording Corp. of Las Vegas, Nevada; and its wholly owned manufacturing division Studio Electronics located in North Hollywood, producers of Universal Audio and Teletronic products. Key personnel of the New York based Waveforms will be moved to the Los Angeles area in the near future. Mr. A. E. Byers, former Waveforms' president, has been appointed to the newly created post of product manager. Hugh McDonald continues in the new title of manufacturing engineer.

# WHAT YOU DON'T SEE IS OUR BEST PRODUCT...

# THE SOUND OF KOSS



*And that's the product you'll have to discover for yourself, in your own way. We can tell you about the 500 undistorted watts from the Acoustech I-A Amplifier, about the rugged and durable Rek-O-Kut turntable, about the stereo "feel" in the PRO-4A Stereophones and the astounding Acoustech X electrostatic speaker system. We can tell you about our years of technical experience and uncompromising attitude of producing equipment acclaimed by critics as "state of the art". We can give you an armload of literature on product features and specifications and ratings (which we'd like you to send for). But the important job is up to you: To put these matched components and accessories through their paces and find the kind of sound that only someone like you can detect.*

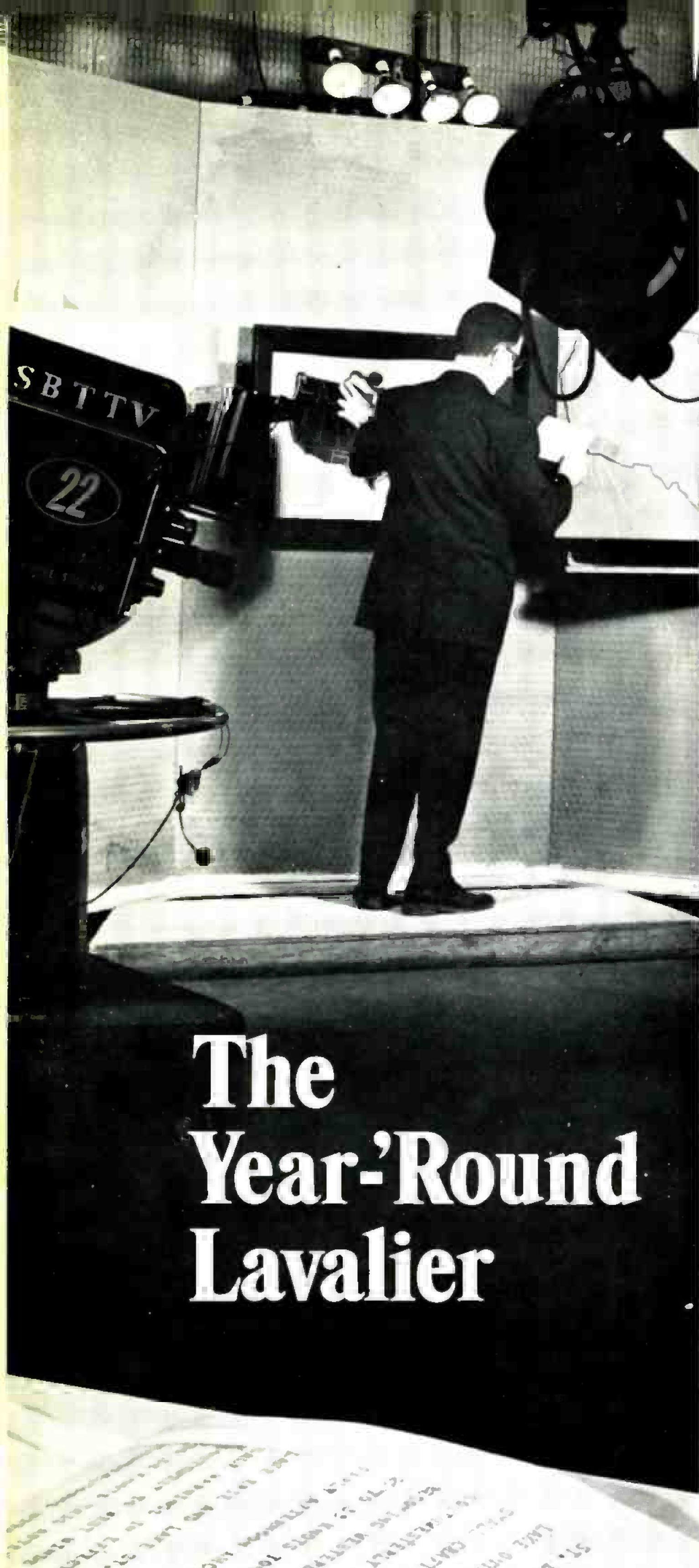
*Happy Hunting.*

*Equipment shown (clockwise from lower left): Koss Pro-4A Stereophones, $50.00; Rek-O-Kut B12-H turntable with S-320 tonearm, $199.95; Koss KO-727 Stereophones, $34.95; Rek-O-Kut B-12GH with S-320 tonearm, $144.90; Acoustech VIII tuner, $349.00; Koss K-6 Stereophones, $26.50; Acoustech I-A power amplifier, $395.00; Acoustech V-A integrated control amplifier, $399.00; Koss PRO-600A (600 ohm) Stereophones, $55.00; Acoustech X electrostatic speaker system, $1690.00; Koss SP-3XC Stereophones, $24.95; Acoustech VI pre-amp control center, $249.00.*

**KOSS** KOSS ELECTRONICS INC.

2227 N. 31st Street Milwaukee, Wisconsin 53208

Export Cable: STEREOFONE/Koss-Impetus
2 Via Berna/Lugano, Switzerland




# The Year-'Round Lavalier

Around most TV stations, E-V lavaliers are taken pretty much for granted. Just hang one around your neck, or clip it onto lapel or pocket—and start talking.

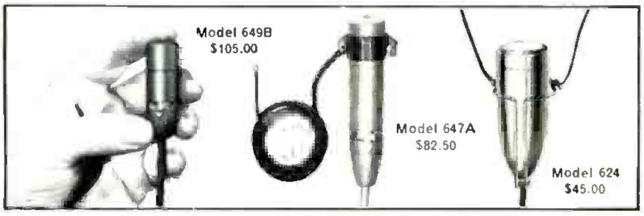
Nothing could make us happier. Because we take great pains to insure the absolute reliability of these tiny microphones. And frankly, no other type of microphone poses a bigger design problem. The lavalier gets dropped, stepped on, swung by its cord, smashed and banged—not once, but often during its life. Most of the abuse is accidental—but inevitable.

So we developed a "nesting" principle of construction that is based on tolerances so tight that the internal element acts as a solid mass, reducing damage due to shock. And we use nothing but Acoustalloy® diaphragms . . . almost indestructible despite heat, humidity, dirt, or high intensity noise or shock.

We've also spent years developing cable specifications—and methods for attaching it. We've taken into account all the tugs and twists that are the fate of any lavalier cable. That's why our strain relief is so effective. And knowing that no cable can last forever, we've made replacement easy and fast.

Of course reliability by itself is not enough. So our field testing of E-V lavaliers is also devoted to sound quality. We must satisfy major network and independent stations on every score. As a result, E-V lavaliers can be mixed in the same program with stand microphones with no change in voice quality.

In the process of developing the lavalier, we've also made it smaller. Our original model was 7" long and 1" in diameter. Today's Model 649B is just 2-1/4" long, 3/4" in diameter, and weighs a mere 31 grams!




Normal trade discounts apply to list prices shown.

Of course TV studios aren't the only places you'll find E-V lavaliers. They're used in classrooms, lecture halls, conferences, stages and business meetings. And they offer the same year-round reliability with no compromise of sound quality.

Every E-V professional lavalier is protected by our unique 2-year unconditional warranty against failure of any kind, plus the lifetime guarantee of workmanship and materials that is an integral part of every E-V microphone. Full details are waiting at your nearby Electro-Voice microphone headquarters. Or write us about your special needs. We're ready to solve the toughest sound problems—off the shelf—all year 'round!

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