db

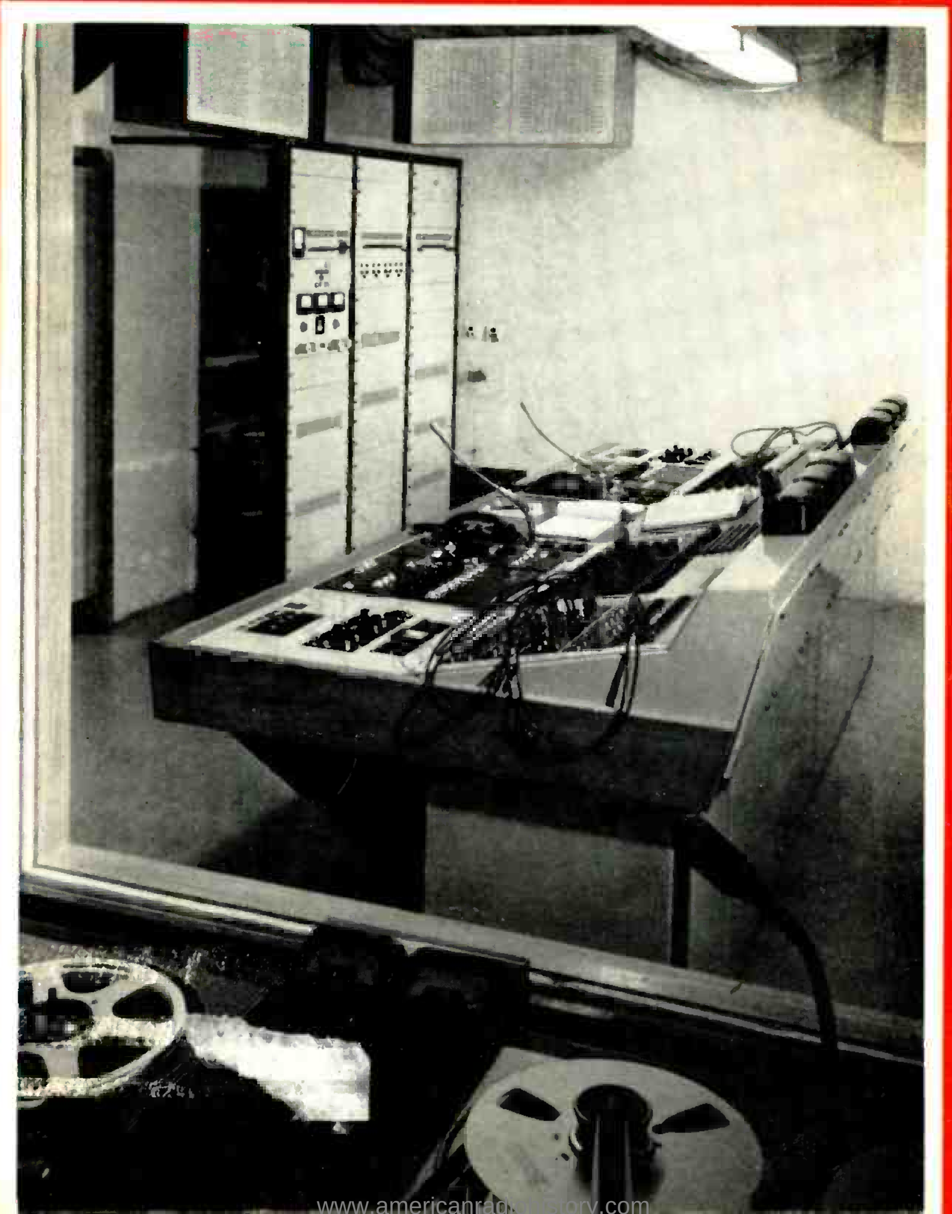
THE SOUND ENGINEERING MAGAZINE

January 1970 75c

Decca's Vienna Venue

A Primer on Noise Measurement, Part 1

A Different View of Speaker Coverage



Just listen to the money-making jingle of multi-sound recordings. That's where the money is.

The chart is now overflowing with new multi-channel recording creativity from studios with Ampex MM-1000's.

This unique multi-channel recorder/reproducer has given the audio engineer a front seat in the talent team. He shapes new sounds, part by part . . . Controls each take and balance, track by track . . . Works with artists to master money-makers in totally new sound dimensions.

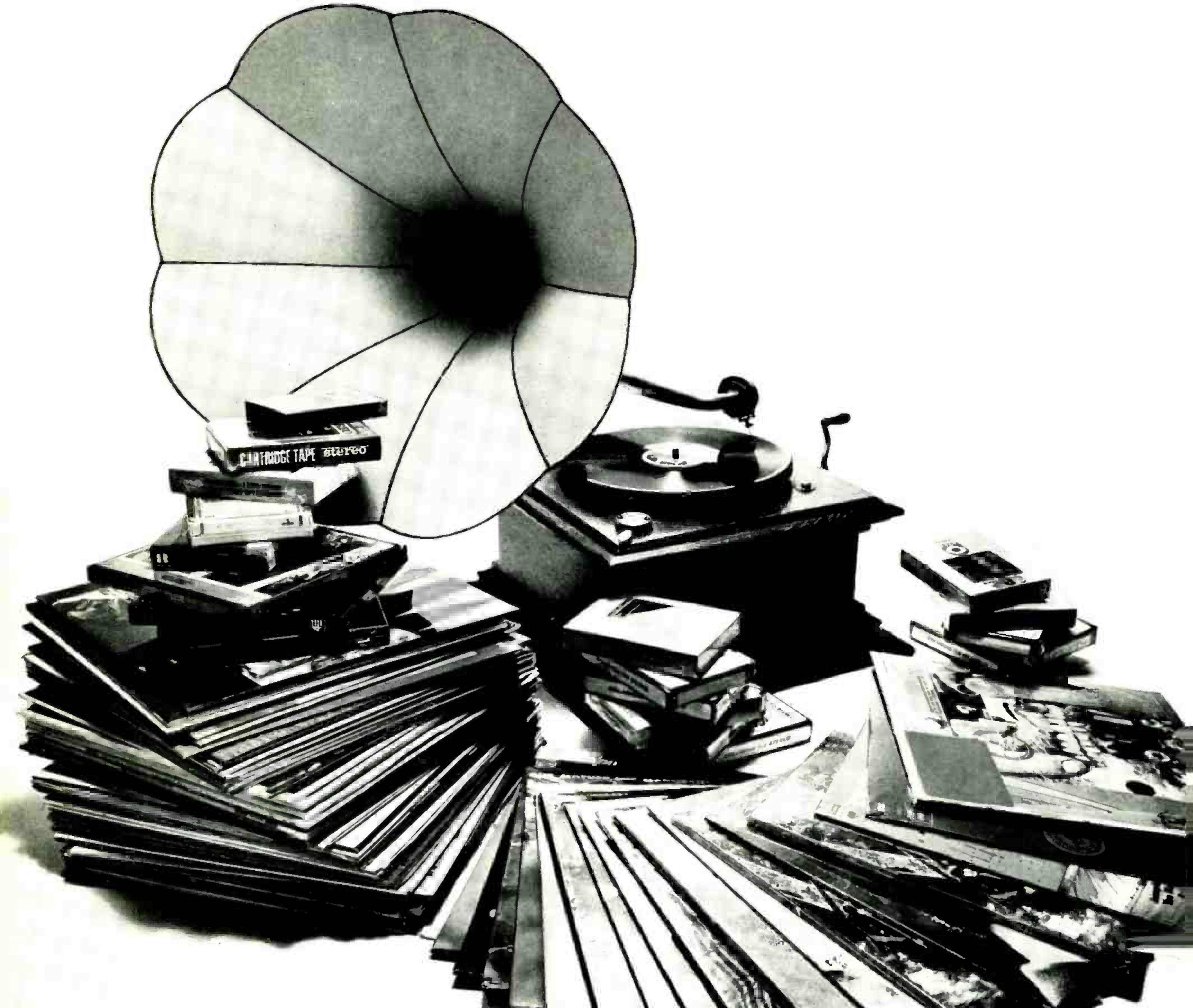
The MM-1000 is the only field-proven recorder that lets you start at 8-track one-inch tape and expand on up to a fantastic 24-track on two-inch tape.

Of multi-track money-makers throughout the world, virtually all studios have chosen Ampex MM-1000. Are you listening?

Call Ampex collect (415) 367-4400. Or write: Ampex, M.S. 7-13, 401 Broadway, Redwood City, Calif. 94063.

AMPEX MM-1000
RECORDER/REPRODUCER

AMPEX



The sound of money as recorded by the Ampex MM-1000

Coming Next Month

AUTOMATED RADIO BROADCASTING is the title of a survey article by Gene Hostetter. In it, he strips away the mystique that seems to surround automated broadcast operations. The fundamental basics are covered in a way that will provide the semi-technically equipped broadcaster as well as the non-broadcaster with a clearer picture of what transpires.

Robert Orban has submitted a paper that describes the development and operation of a Stereo Synthesizer and a Stereo Synthesizer Matrix. These products serve to provide synthetic stereo from mono sources and, in the case of the matrix system, to give a phase-controlled stereo (for stereo/mono compatible discs) from multi-channel sources.

Part 2 of Wayne Rudmose's PRIMER ON METHODS AND SCALES OF NOISE MEASUREMENT will discuss frequencies and the relation to sound measures and go on to the equipment of sound measurement.

And there will be our regular columnists, George Alexandrovich, Norman H. Crowhurst, Martin Dickstein, Arthur Schwartz, and John Woram. Coming in **db**, The Sound Engineering Magazine.

About the Cover

● A view from the tape room into the control room where many of London Records' most impressive productions originate. See John Borwick's description of English Decca's Vienna facilities beginning on page 26.

← Ampex Circle 10

db

THE SOUND ENGINEERING MAGAZINE

JANUARY, 1970 • Volume 4, Number 1

Table of Contents

FEATURE ARTICLES

MONTHLY DEPARTMENTS

db, the Sound Engineering Magazine is published monthly by Sagamore Publishing Company, Inc. Entire contents copyright ©1970 by Sagamore Publishing Co., Inc., 980 Old Country Road, Plainview, L.I., N.Y. 11803. Telephone (516) 433-6530. **db** is published for those individuals and firms in professional audio—recording, broadcast, audio-visual, sound reinforcement, consultants, video recording, film sound, etc. Application should be made on the subscription form in the rear of each issue. Subscriptions are $6.00 per year ($7.00 per year outside U. S. Possessions, Canada, and Mexico) in U. S. funds. Single copies are 75c each Controlled Circulation postage paid at Easton, Pa. 18042. Editorial, Publishing, and Sales Offices: 980 Old Country Road, Plainview, New York 11803. Postmaster: Form 3579 should be sent to above address.

Letters

Robert Bach
PUBLISHER

Larry Zide
EDITOR

Bob Laurie
ART DIRECTOR

Marilyn Gold
COPY EDITOR

Richard L. Lerner
ASSISTANT EDITOR

A. F. Gordon
CIRCULATION MANAGER

Eloise Beach
ASST. CIRCULATION MGR.

SALES OFFICES

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

Denver
Roy McDonald Associates, Inc.
846 Lincoln Street
Denver, Colorado 80203
303-825-3325

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

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

San Francisco
Roy McDonald Associates, Inc.
625 Market Street
San Francisco, California 94105
415-397-5377

Los Angeles
Roy McDonald Associates, Inc.
1313 West 8th Street
Los Angeles, California 90018
213-483-1304

Portland
Roy McDonald Associates, Inc.
2305 S. W. 58th Avenue
Portland, Oregon 97221
503-292-8521

Japan
TECHNOSERVICES, LTD.
31-16 Akazutsumi 1-Chome
Setagaya-Ku
Tokyo 156, Japan

The Editor:
I have just read the article JUST STEP UO TO THE MIC by Robert Hawkins in the November issue. The author is in error in attributing the cardioid microphone to Western Electric.

I presented the first paper on the unidirectional or cardioid microphone at the Cleveland Meeting of the Acoustical Society of America, December 1. 1931. The first commercialization of the cardioid microphone was the RCA77-A introduced in 1935.

The Western Electric cardioid microphone is described in a paper published in the Journal of the Acoustical Society of America in April 1941. In this paper they give us credit for the first cardioid microphone.

Harry F. Olson
RCA Laboratories
David Sarnoff Research Center
Princeton, N.J.

The author responds:

Dr. Olson's letter included documentation so there is no doubt of his correctness.

I want to clear up the identification of the microphones pictured on the cover and again within the text of my article.

The microphones on the cover from left to right are:

1. Western Electric double-button carbon mic, model #600-A. The carbon element was given me by KNBR and a pencilled date indicates its last use in August, 1931.

2. A mystery mic with no identifying marks. It obviously is a velocity mic and appears to be twenty or more years old. Strangely enough, in a fan letter just received, another collector says he has the identical model, but he too can find no markings to indicate who manufactured it. Perhaps I will find the time to take it apart and look for information inside.

3. Carbon microphone made by the American Microphone Company, Los Angeles. Circa, 1925.

4. Electro-Voice velocity microphone model V-3 made by that company in South Bend, Indiana.

5. An old carbon mic, by the Universal Microphone Company, Inglewood, California.

6. A crystal press-to-talk model T-3 by the Astatic Corporation of Conneaut, Ohio.

The line-up inside the article and shown on this page is:

1. American
2. Electro-Voice
3. Universal
4. Astatic
5. Western Electric
6. Unknown

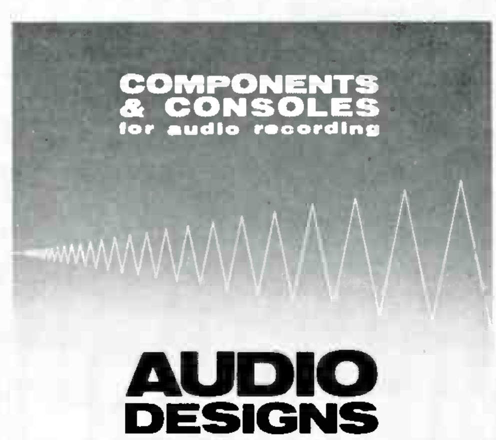
Robert Hawkins
KVI
Seattle, Washington

CUSTOM 16 CHANNEL CONSOLE

by Audio Designs

Fine Recording, New York*, feels they're onto something good in consoles. When they needed a 4 channel console in January, 1968, they ordered it made by Audio Designs & Manufacturing, whose reputation for design and quality was already well established. In the summer of 1968, Fine Recording called again on Audio Designs to construct an 8 channel console. Audio Designs' consoles apparently work well — Fine Recording Studios are now using this remarkable 16 channel console made by Audio Designs, including their unique Audex Switchers.

If you want to get onto something good in consoles . . . call Audio Designs. You can get a quality console designed, engineered, manufactured and delivered at a price competitive with most "stock" consoles. You'll be getting a console that does "exactly" what you want it to do . . . fits "exactly" where you want it to fit. For consultation and estimates on a console to meet your exact audio recording requirements, write or call . . .



*Fine Recording, Inc., Subsidiary of Vidcom Electronics, Inc., New York

AUDIO DESIGNS / 15645 Sturgeon / Roseville (Detroit), Michigan 48066 / (313) 778-8400

Circle 13 on Reader Service Card

The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

EQUALIZATION IN SOUND REPRODUCTION AND REINFORCEMENT SYSTEMS.

• Until recently it was up to an acoustical engineer to apply acoustical treatment to a studio, theater, or auditorium and to select microphones and speakers which would acoustically match the response of the environment. Lately the acoustical deficiencies of many rooms are being compensated by electronically equalizing the frequency response of the amplifying system. Equalization accomplishes almost everything that acoustical treatment does except decreasing reverberation of the room.

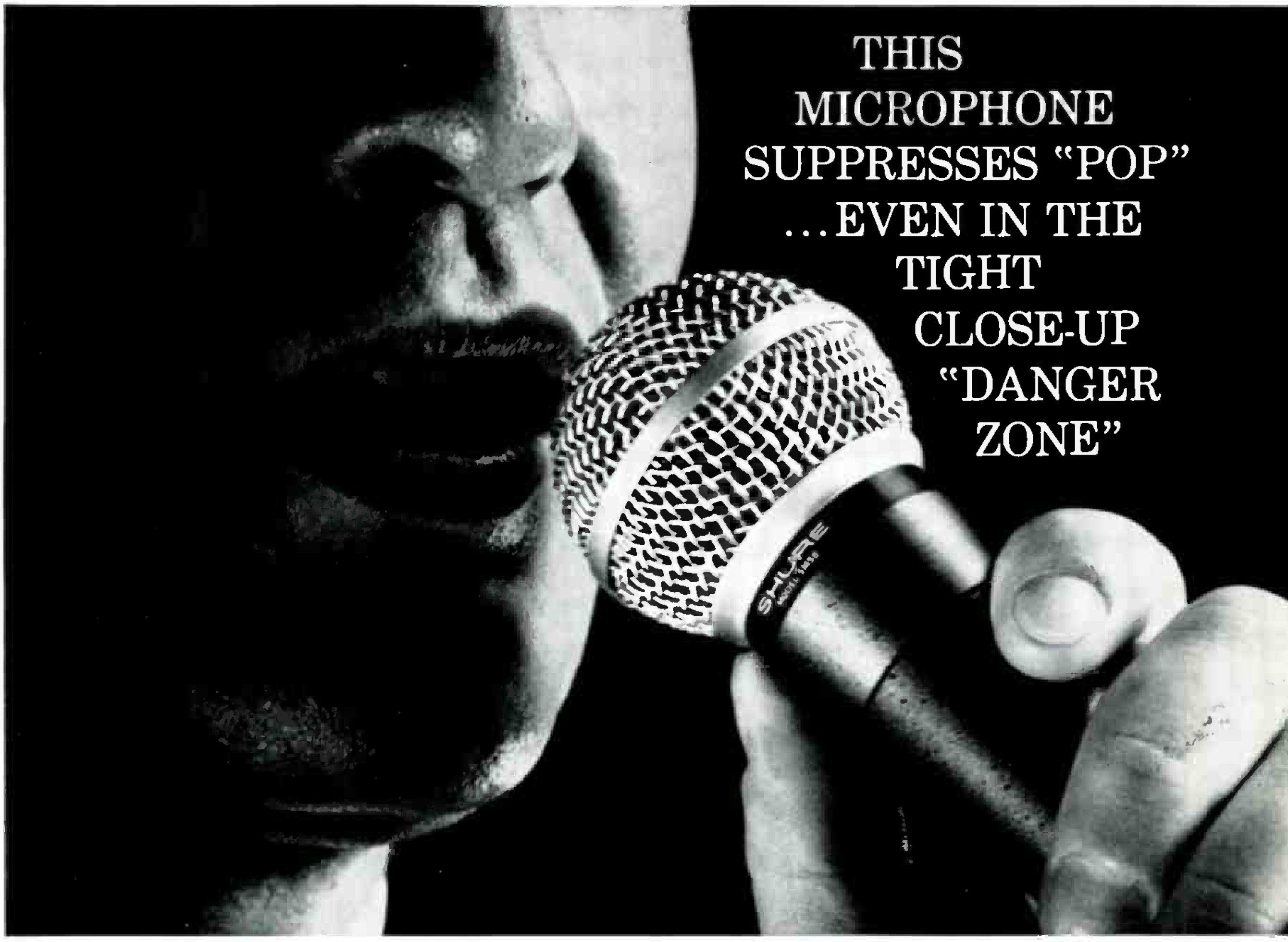
The idea of equalization is not new. In the professional sound-recording field, equalizers have been used for years to compensate for the varying characteristics of microphones, room (studio) acoustics, and also to reduce the inter-mic crosstalk by limiting the bandwidth of the individual mic input circuit. In home-entertainment equipment the loudness contour and tone controls help accomplish this. One of the main reasons sound-reinforcement equalization thasn't been extensively practiced before was a lack of the right equipment. Several manufacturers finally jumped on the bandwagon and a variety of variable environmental equalizers with high Q and narrow bandwidth have become available.

Because of the relative precision and knowledge of what has to be done and how equalizers have to be set, manufacturers prefer to distribute the equipment through their dealers who are trained in survey installation and adjustment of the equalizers. They are equipped with automatic plotting devices in order to speed up and simplify their job.

Equalization of a system consists of acoustically obtaining the response of the system and then using a full complement of filters for equalizing until response is reasonably flat. At least one manufacturer, after this is done, then sells the customer only the filters he needs to accomplish this. Other manufacturers sell a package consisting of which cover the entire equalizers range. Even if you need just one filter you have to buy them all. On the other hand if the acoustics of the auditorium or room change you are equipped to cope with them. Another system offers filters which can be tuned over a limited range of frequencies. This method offers higher precision in tuning out individual sharp peaks but accomplishment requires the services of a better equipped and more experienced specialist.

All of these methods are more than satisfactory and all of them, executed properly, can enhance the performance of a system twofold at least. Aside from the fact that a correctly equalized monitor system would reproduce sounds more faithfully, sound reinforcement systems, can be set for much higher reproduce levels without feedback. However there are scores of systems and installations with much less critical applications and they don't justify the high cost of professional equalization. The purpose of this month's column is to work out simple methods of charting your own frequency-response characteristics of the installation and determine what has to be done to make the system sound and perform better. In future columns the design of simple filters and equalization techniques will be described.

As you know, most speakers and microphones are designed and tested in anechoic chambers. They are designed to work well within an environ-



The Shure SM58 *self-windscreened unidirectional* microphone is ideal for broadcast uses such as remote news, sports, interview and vocal recordings because it eliminates or minimizes the irritating "pop" caused by explosive breath sounds. With the SM58 you will have the peace-of-mind assurance that you're delivering the quality audio that goes with pop-free pickup. It's great for studio announcing, too—or wherever the announcer or vocalist has the audio-degrading habit of "mouthing" the microphone. Of course, the same filters that eliminate pop also do away with the necessity for an add-on windscreen in outdoor uses.

On the other hand, the unusually effective unidirectional cardioid pickup pattern (uniform at *all* frequencies, in *all* planes) means that it is a real problem-solver where background noise is high or where the microphone must be operated at some distance from the performer. Incidentally, but very important, the SM58 tends to control the low frequency "boominess" that is usually accented by close-up microphones.

All in all, close up or at a distance, the Shure SM58 solves the kind of ever-present perplexing problems the audio engineer may have felt were necessary evils. The SM58 might well be the finest all-purpose hand-held microphone in manufacture today. And, all things considered, it is moderate in cost.

Other features: the complete pop-proof filter assembly is instantly replaceable in the field, without tools. Filters can be easily cleaned, too. Stand or hand operation. Detachable cable. Rubber-mounted cartridge minimizes handling noise. Special TV-tested non-glare finish.

For additional information, write directly to Shure Brothers Inc., 222 Hartrey Ave., Evanston, Illinois 60204.

SHURE SM58

SELF-WINDSCREENED UNIDIRECTIONAL DYNAMIC MICROPHONE

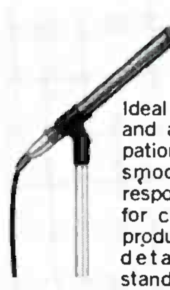
SHURE PROFESSIONAL MICROPHONES . . . FOR BETTER AUDIO

MODEL SM56
CARDIOID
DYNAMIC

Extremely versatile in studio, control room, and remote use. Also widely acclaimed for rhythm recording. Bright, clean sound. Exceptionally uniform cardioid pattern gives optimum control of environment.

MODEL SM33
UNIDIRECTIONAL
RIBBON

Warm, smooth sound for studio, control room, and scoring stage. Super-cardioid directional pattern. Compact, yet rugged.



MODEL SM76
¾" OMNIDIRECTIONAL
DYNAMIC

Ideal for interviews and audience participation, yet unusually smooth wide range response (40-20 KC) for critical music reproduction. Instantly detachable from stand. Steel case with Cannon connector.

MODEL SM50
OMNIDIRECTIONAL
DYNAMIC

Self-windscreened and pop-free for news, sports, remotes, and interviews. Also ideal for many studio and control room applications. Comfortably balanced for hand or stand use. Natural response.

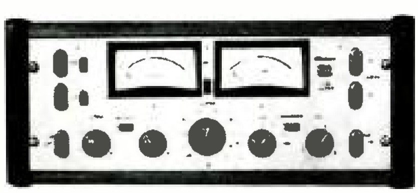
Circle 15 on Reader Service Card

NEW IM ANALYZER

ADVANCES STATE OF THE ART

Extended measuring range and high-speed readings are the outstanding features of a unique new Intermodulation Distortion Analyzer introduced by Crown International recently. The American firm is known for its line of Crown precision professional tape recorders.

This analyzer was developed to meet the production line requirements of the Crown DC300 lab standard amplifier. The first need was for accurate measuring capability through 0.01%. This analyzer guarantees a residual IM level of less than 0.005%, with seven full-scale ranges from 100% to 0.1%.



The second requirement was for an instrument simple enough to be operated by production-line personnel and rapid enough to make sequential readings across the entire power band. The Crown analyzer meets the challenge by reducing measuring time from minutes per reading to just seconds. This is accomplished by a "tracking" function, using two meters and a ganged input/output gain control. The input level is set using the calibrate meter, and distortion is immediately read on the percentage distortion meter. Successive readings at 5db increments take under five seconds each. The entire operation is completed in less than one minute.

Solid state construction, utilizing FETs, makes the Crown analyzer highly stable and uniquely compact, measuring 7x19x7 inches. Rack mount list price is $595. Write for spec sheet to CROWN, Dept. DB-1, Box 1000, Elkhart, Indiana, 46514.

Circle 14 on Reader Service Card

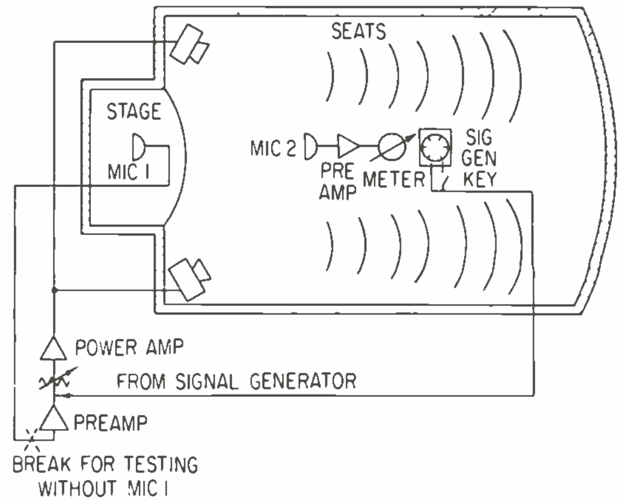


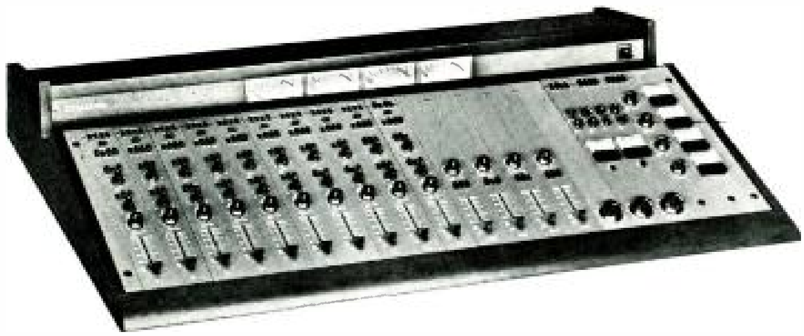
Figure 1. A typical test setup for measuring frequency response of a theater system.

ment which is passive and doesn't influence the performance of the transducers (speakers and microphones). Heavily-damped speakers are less influenced by the acoustical qualities of the room while high-efficiency types are more affected. But with today's extremely low output impedance power amplifiers, electrical damping of the speaker makes it less susceptible to external influences. Mechanical coupling of the driving coil and the speaker cone still remains the determining factor. Some reflected frequencies from the walls of the room return to the speaker and beat with vibrations of the speaker cone.

Those are the factors that make speakers sound differently when they are set in a room and listened to. In a simple and sound reproducing system for pre-recorded information, equalization serves the purpose of helping to reproduce the sound more faithfully to the original and serves no other purpose. In sound-reinforcement systems, aside from equalizing for better sound, we are seeking higher acoustic operating levels. If the system reproduces a certain frequency more strongly than others this frequency will be picked up by the mic sooner and stronger and will therefore be amplified more, sometimes enough to start acoustical feedback. If we remove all those troublesome peaks we can increase the maximum operating level of the system (sometimes) as much as 10 dB. Considering that 6 dB of voltage gain is twice the power, this gain in loudness can be substantial.

How do we measure the response of the system and draw a usable curve? First of all we need test instruments. If sound-level meter is available it will replace the otherwise needed calibrated microphone, preamplifier and VTVM. We also need a variable frequency signal generator and a push-button momentary switch or key. This is needed to send the generator signals for short periods since long-duration signals at high levels are not only objectionable to the person who is measuring the response, but can also damage the speakers. Most speakers are rated for music power but not for r.m.s. tones. FIGURE 1 shows the suggested location of test gear for a typical theater-type job. It is important that the response of the mic used for measurements if a sound-level meter is not at hand) be known as accurately as possible. Even if the mic is not flat a derived curve can be corrected for the mic response deficiency. Place the mic where most critical listening will be done. In a control room place the microphone where the mixing engineer sits. Just as in your recreation room you would like the sound to be best in your favorite arm chair, so in the theater you aim for the best sound in the favorite middle front of the orchestra.) Make sure that stage mics and speakers are installed permanently and that they will not be disturbed after the tests, since a change in location of the speakers and mics will change the response of the system. This is not so much because of a change of the basic response of the speakers dut due to the ratio of the direct sound we hear to the reflected waves which bounce off the walls and come back.

Once the setup is completed (without sending any signals through the system) measure the ambient noise of the room or hall. Reproduced sounds from the speakers should register on the meter at least several dB higher so that the obtained results will be as accurate as possible a (because high ambient noise may mask sharp dips in response). In a sound-reinforcement system take two readings—one with the microphone open and one with it closed. The difference in results will serve as an indication of the effect mic-to-speaker coupling has on the response of the system. The more pronounced the effect, the higher the coupling and the lower the reproduced levels can be expected from the setup. If a sound meter is used, set it to the flat position. A VTVM has



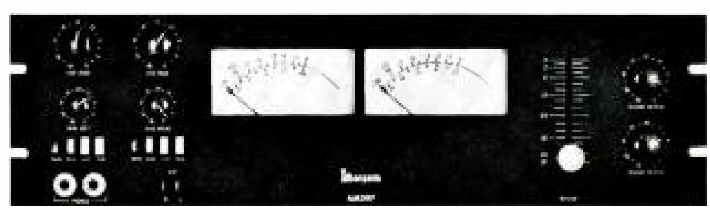
AM4A MIXER SYSTEM CONSOLE $9,200

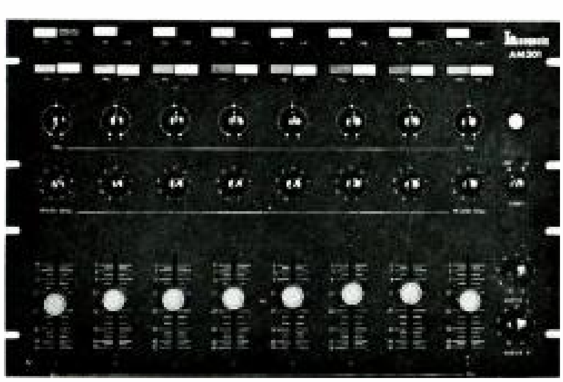
Monaural and Stereo Consoles with 13 modules, factory wired with one year warranty . . . from $3,000.

A distinctive New Trio that never misses a note from Langevin

AM-307A OUTPUT SYSTEM $475.

Designed for general system application, as an accessory to AM-301A for Radio Master Control or Recording Mastering Systems.





AM-301A MIXER SYSTEM $1,450.

A complete two channel mixer system for either Studio or remote use. Radio station monaural or stereo model available soon.

• SEND FOR OUR NEW 1970 ATTENUATOR AND EQUALIZER CATALOG •

Langevin's Engineering Department can help you plan **your** new Systems. Please let us know your requirements.
LANGEVIN 1801 East Carnegie Avenue, Santa Ana, California 92705. Telephone (714) 546-8830.

Circle 16 on Reader Service Card

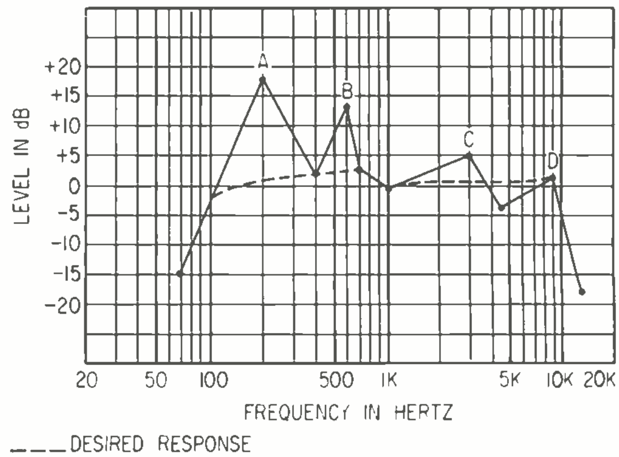


Figure 2. An example of a frequency response plot of a system, showing high and low points.

to be least sensitive enough to measure levels as low as −60dB otherwise an additional preamplifier has to be used after the mic. Sometimes 1:10 step up transformer will do the job if the system lacks 20 dB or so in sensitivity.

Adjust the signal generator to 1000 Hz. Adjust the output of the generator until the meter registers an increase in level over the ambient noise of at least 10 dB. Observe a 'scope for a clean sine wave that is clearly distinguishable from the noise signals. If the wave form is not clear, raise the level further until it is fairly clear. From this point on log all meter readings. Consider the level shown on the meter from 1000 Hz to be the reference level and mark it 0 dB. All other readings above 1000 Hz will carry a positive sign and ones below will be negative. Depress the key at the signal generator and start changing frequency upwards. While you change the frequency observe the meter. All fluctuation of less than 2 dB can be disregarded. Keep increasing the frequency until the meter registers a well-defined peak or dip (several dB) in the frequency response. As the meter goes through the maximum deviation, note the frequency and magnitude. Log the result. Continue changing the frequency until you reach a second peak or a dip—and so on. Don't be surprised if you find peaks 10-15 dB above the average level. After reaching the upper limit of the audio spectrum return the signal generator back to 1 kHz and continue changing the frequency downward. Keep signal going through only long enough to achieve the reading. With a little practice a few seconds should be plenty of time to get a good reading. When all frequencies are covered and all readings are marked down on paper it is time to take either the frequency-response paper (linear Y or vertical graphing, periodic log for X or horizontal divisions)—or prepare your own grid paper. It is really important to have a visual presentation of the system's response, so that you know whether a particular point on the graph is a dip or peak and which peak is more important to equalize out than the other. Figure 2 shows such a typical graph. Peak A is the highest and should be removed first, peak B is almost as high as peak A and can cause as much problems. Peak C although it measures 5 dB above the 1 kHz level may not be as objectionable because if all frequencies below peak C are equalized to the level of 2-3 dB above the 0 level then the deviation of peak C is only 3 dB or so. Sharp rolloff in the sample graph of the high frequencies can probably be explained by poor speaker construction or absorption characteristics of the room. Also, directionality of high-frequency waves should be taken into account. Speakers may not be pointed directly at the test mic and you may not be measuring much of direct sound from the speaker but may hear considerable amounts of reflections. Although this month we only covered the method of plotting freqiuency response, it would be advisable to be prepared to perform some sort of equalization once you are set up for testing.

Quite a few systems can be improved considerably by just removing a single peak, others may need a simple band pass or band stop filters to improve performance. Whatever we do to a system we must remember that even the most thorough efforts may not bring the completely desired degree of improvement because the system performs differently with an audience. Also if you listen from a different place in the house than the one where response measurements were taken, results may be different. There are other factors creeping in we shouldn't forget about such as phase cancellation and additional effects.

Without the trying to discourage the reader, I am stating the fact that the technique of system equalization is a big step in the right direction which eventually will be amplified by other improvements in technology and equipment.

Walter Carlos, creator of "Switched-On Bach" and "The Well-Tempered Synthesizer," uses the Dolby System.

Mr. Carlos says, "The raw materials of electronic music — the outputs of my Synthesizer, for example — are sounds which can be varied from striking purity to extreme complexity. After a desired sound is created, often with considerable effort, it must be preserved with care, to be combined later with others in a meticulous layer by layer process. The noises of magnetic recording are significant hazards in this regard, since they are particularly noticeable in electronic music. However, my experience confirms that the Dolby System effectively attenuates the noise build-up in electronic music synthesis. My studio at TEMPI is equipped with ten Dolby units, which I consider to be indespensible in my work."

Whatever your recording activities, the dependable Dolby System can help you make good recordings even better. Now in use in over 100 companies in 21 countries around the world.

DOLBY LABORATORIES INC
333 Avenue of the Americas New York N Y 10014
telephone (212) 243-2525 cables Dolbylabs New York
for international inquiries contact UK address:
346 Clapham Road London S W 9 England
telephone 01-720 1111 cables Dolbylabs London

S. Calif. **Audio Industries Corp.**
1419 N LaBrea Ave., Hollywood, Calif. 90028
Tel. 213-HO 5-4111

N. Calif. **Audio-Video Systems Engineering**
1525 Tennessee Street, San Francisco, Calif. 94107
Tel 415-647-2420

Midwest **Expert Electronics, Inc.**
7201 S Western Avenue, Chicago, Illinois 60636
Tel. 312-HE 6-2700

Canada **J-Mar Electronics, Ltd.**
6 Banigan Drive, Toronto 17, Ontario, Canada
Tel 416-421-9080

Circle 21 on Reader Service Card

The Sync Track

JOHN WORAM

Herewith, we begin a new monthly column to be devoted to the special provinces of the recording industry. Its author is no stranger to this field. He is a recording engineer based with RCA Victor in New York and, as such, has put in more than his share of long days and nights at the console.

Mr. Woram promises us that he will be responsive to reader inquiry and request on any subject pertinent to the field. That which he does not himself know the answer for, will be properly researched to give you the information you need. So look at this new effort in the same light as our other columnists, all of whom are ready to feed back inputs you may place. Ed.

THE CASE FOR HUMAN ENGINEERING

● There is apt to be some amount of mental pressure involved in sitting behind a mixing console during an important live session. On a modern console, there may be as many as 600 variables (mixers, echo selectors, equalization, *et al*) any of which may need adjustment during a take. As the number of controls increase, so does the probability that you will sooner or later adjust the wrong one, and maybe blow a take. Very embarrassing. The musicians make funny remarks, and the producer makes remarks that are not so funny.

But what if a little mistake meant disaster rather than embarrassment? Suppose you were at a missile site and you pushed the rocket-launch button because you mistook it for a cigarette lighter, or, on the way into lunar orbit you fired a left retro when you really meant the right one? Obviously, little goofs like these must never occur. Once you've blown up Washington or crashed into the dark side, you can't apologize and ask for a re-take. There's got to be some way of stopping mistakes before they happen, and of course, there is. It involves, among other things, intensive training, a good night's sleep, and a confusion-free set of instruments in front of you. All of these little amenities have been built in to such critical programs as missiles and space probes.

Meanwhile, back at the recording studio, the training may not have been so intensive, and you probably haven't had a good night's sleep all week. And to top things off, there may be much confusion built into your console.

There may not be much that can be done about the training, or getting to bed on time, but there certainly is something to be done about the console. It's called *human engineering*. It involves not so much the electrical, as the *physical layout* of a piece of equipment. It means designing equipment without built in confusion. For space science, it helps keep astronauts alive. For recording science, it helps make life just a little easier.

For example, most recording consoles are of modular design. With minor variations, a module will contain: preamp pad, fader, preview, equalization, muting, and so on. Controls are usually all placed above the slide fader, and are designed to look practically identical. More often than not, a row of four or more round black knobs are found on each module. And it looks very impressive—until you get 24 modules in a row. It then looks very confusing.

For instance, in the middle of a take, the producer has a vision—"more lows on the bones", he says. The trombones' microphone was plugged into position 13, so you quickly count over to 13 and then up to knob 3, which is the low-end equalizer. If you haven't arrived too late, you twist the knob, and *voila*!—nothing happens. Why should anything happen, the eq. switch is in the eq. out position? As soon as you catch this, you quickly punch it in, and the lows come riding in on a sonic tidal wave. Surprise, surprise! It seems the low end was already up a few notches from flat before you cranked it up. But you didn't notice it because the knob is round and black, and the marker is on the upper part, well hidden from your eye by the knob's bulk.

Preposterous? What do you think? After a few hours staring into the lap of a console, the knobs have a way of staring back at you, and being where they shouldn't be when you want them.

This is where *human engineering* comes to the rescue. To over-simplify, human engineering says that all knobs don't have to be round, black and shiny. The military spends a fortune designing equipment that is, in addition to being electronically A-O.K., as human proof as is possible. One prominent military contractor has spent years analysing the behavior of equipment operators. The idea is, that by studying the relationship between man and machine, the machine can eventually be refined into a device that is almost impossible to mis-operate.

Much of this human engineering technology is available to the recording industry practically for the asking. A lot of it is, to be sure, irrelevant or impractical. But there is a nucleus of information that would be worth investigating by anyone planning a new construction project. In any man/machine situation, there must be a two-way communication system. The operator communicates with the machine—in our case, a console—by manipulating the various controls. The console communicates via its visual read-outs, which may be colored lights, switch positions, patch cords, etc. The more obscure this feed back is, the more you will be apt to make errors.

One obvious example of human engineering is the *tactile shape* control knob. That is, a knob with physical characteristics that indicate its function. Thus, at the missile site, the *self destruct* button is square, and the *light dimmer* is oval. Now, if you want to raise the lights a bit and you can feel corners, you know you've got your hand on a no-no. Reducing this to our needs, a square *echo-send* knob will not be confused for an oval *equalization knob*.

To design the ultimate console in terms of human engineering is no slight task. The designer must be able to separate the essential reliability requirements from the variations suggested by the personal tastes of various operators. Think about the little built-in stumbling blocks you may encounter in your work, whether it's behind a mixing console, or at a lacquer cutting lathe, or, wherever. Jot down your thoughts and send them to this column, along with any proposals you may have for improvements. We'll try to process the information received, and with a little help from our friends in the human-engineering game, perhaps we can come up with some ideas for a more confusion-free way of doing things.

The Feedback Loop

ARNOLD SCHWARTZ

● The presence of two major all-news radio stations in the New York City metropolitan area attests to the popularity of this format. Back in August we took a look at one of these stations, WINS. This month we will talk about the other all-news station, WCBS, located on the Avenue of Americas and 52nd street in New York City. While WINS' studios were designed to be an all-news operation, the present WCBS installation was originally intended for a less specialized format. Nevertheless, the original installation, completed in 1965, was designed with sufficient flexibility to allow the changeover to be made with relatively few changes and additions.

The previous WCBS format was one where much of the programming (talk shows, panel shows and music) originated in the studios. In the present format, news and other information is constantly flowing from outside sources into the studio complex. Typically, incoming news is recorded in production studios, and the edited program is then packaged on a cartridge tape to be utilized in the on-air studio. When the change came it became necessary to have a large number of incoming lines available. Accordingly, lines for the incoming feeds were installed, and fourteen-position selector switches were located at control consoles and tape recorders to enable selection of the feeds at these locations.

Now, virtually all broadcasting originates in control room B (FIGURE 1) which has a ten-channel McCurdy console. To accommodate the frequent live newscasts, involving two or more reporters, there are four microphone inputs from studio B. Studio B is shown in FIGURE 2, with control room B seen through the window. The largest of the three cabinets on the desk (directly in front of the announcer, Jim Harper) houses a selector switch which makes incoming lines available for monitor purposes. As can be seen from the floor plan, studio B is large and easily accommodates four people. The three remaining control rooms are used as production areas. In addition there are two edit rooms. Control room A, adjacent to B, (see FIGURE 1), is also the central control where jack fields, transmission equipment, switchers, and f.m. automation are located. FIGURE 3 is a photograph of control room A showing the logging recorders (left), and the f.m. automation equipment.

Originally all four control rooms alternated as on-air studios, and a relatively sophisticated master switcher was installed to transfer the outgoing program line from one studio to another on cue. The engineer in the upcoming control room punches a preselect button mounted on his console, and the outgoing program line will then be transferred to that console when the engineer in the on-air control room actuates the release button at the end of his program. The master switcher has eight inputs, and four outputs; *a.m.*, *f.m.*, *network*, and *spare*. An overide in the central control room gives instant access to the four outgoing lines. In the present format the master switcher is not as active as originally planned, but it does provide a very convenient method of switching studios.

The CBS Netalert system is a vital part of the WCBS operation. Netalert is a two-tone signalling system which is sent over program transmission lines simultaneously with the program signal itself. The transmitted code actuates a receiving device that monitors the network program line, but its presence on the line does not disturb the listener. Netalert signals originate at the CBS network facilities in New York, Washington D.C., and KNX in Los Angeles. The code is initiated by dialing a number from one to nine which then sends out one of the nine available messages. Each of the CBS affiliates is supplied with a receiver that decodes the tones. Remote indicators operated by the receiver unit are installed at control consoles and other strategic locations. The receiver output may be used to automatically initiate a sequence, or used as a cue signal to the engineer.

Here are some examples of how the Netalert system is used. Code number *one* signals a cutaway cue for local commercials during the hourly network

The Professionals.

Bill Bell is known as "The Ear." He's the owner of Bell Sound Studios, Hollywood. Bill does commercials, some of the best. You've heard a lot of them. He orchestrates each one, every element of sound from the soft spoken solo voice of Marvin Miller to the high dB blare of acid rock.

Bill's fussy about sound, and so are his engineers. So are the advertising agency production men, the creative people and the account executives. If you're going to take three or more hours to get the right sound in sixty seconds of commercial, you want to make sure the sound is the best possible.

So, as a starter, Bill uses Altec "The Voice of the Theatre"® speaker systems, sixteen of them. And in his custom consoles, Bill Bell also uses Altec audio controls. Again, because he thinks they're the best. After over thirty years of developing sound systems for the broadcast and motion picture industries, that's a nice reputation for Altec to have.

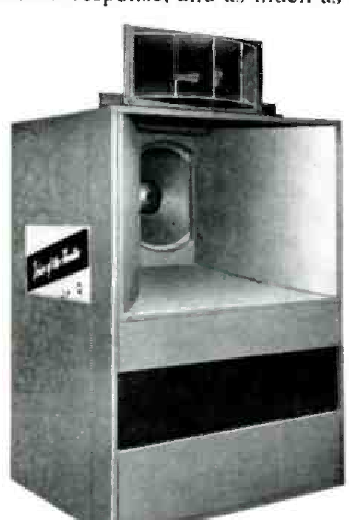
Specifically, each new Third Generation Voice of the Theatre speaker system features a 15″ low frequency speaker that's the finest made. It has a 10½ pound magnet structure, a cast aluminum frame and a 3″ edge-wound voice coil of copper ribbon. All this—plus the efficient new Symbiotik™ diaphragm—provides outstanding bass and transient response, and as much as two times more power handling capability than previous designs. The 18″ massive cast-aluminum sectoral horn has a very wide sound dispersion angle at all frequencies.

The driver. It works from 500 to 22,000 Hz in the A-7-500-8. (The A7 crosses at 500 Hz.) It's so efficient, there's need for only one crossover in the system which eliminates those high frequency peaks and dips. All this means the crispest, cleanest, most undistorted sound you can get from low end through the high.

For complete specs on our sound equipment, just write to Altec Lansing, 1515 So. Manchester Ave., Anaheim, California 92803.

ALTEC LANSING®

A QUALITY COMPANY OF LTV LING ALTEC, INC.



Circle 23 on Reader Service Card

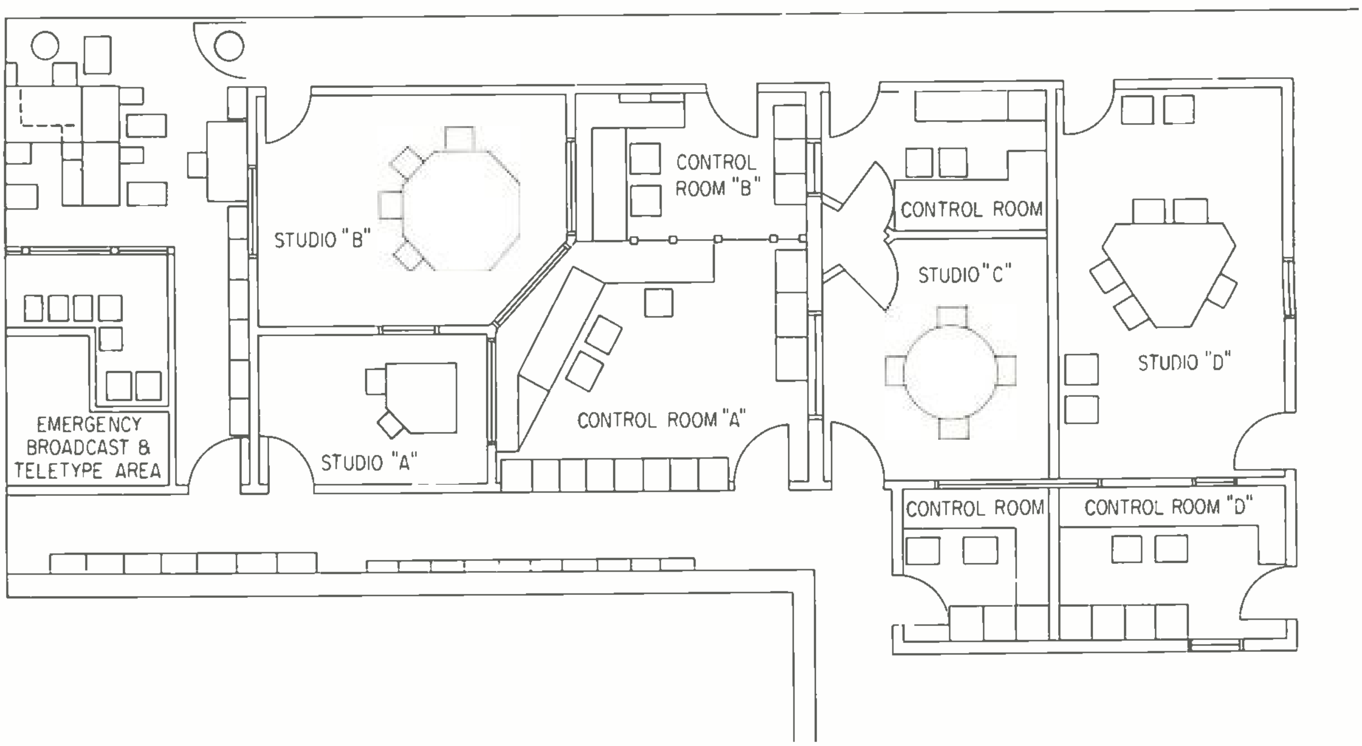


Figure 1. The floor plan layout of WCBS radio showing the studios and control rooms.

newscast. Code *two* is used similarly when the Washington news bureau is feeding the network line. If an exclusive news bulletin comes up, code *four* is sent and the program is fed following a ten-second delay to allow for breakaway. Code *nine* is used in the event of a national emergency, and in this case special alarms are actuated in the homes and offices of key station personnel. According to Mort Goldberg, director of technical operations of WCBS Radio, the Netalert system has proven to be virtually foolproof against accidental triggering in its more than ten years of operation.

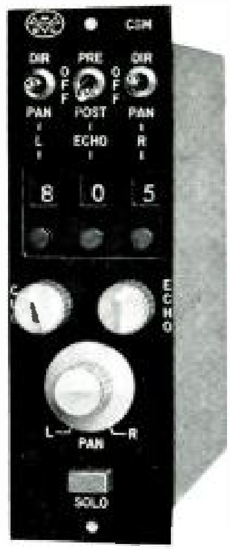
To expand their coverage, in an area with a bewildering complex of news stories and events, WCBS operates three mobile units. These miniature studios on wheels are equipped with two-way radio, and mobile telephone communication. The base station of the two-way radio is on top of one of the world's tallest structures, the Chrysler Building. This is an important feature for reliable coverage because of the numerous tall buildings and structures in New York. Each mobile unit (there are two Ford Econoline Vans, and a Jeep Wagoneer) has a four-channel console and tape recorders with editing facilities. Normally a newsman and a technician man these vehicles. Reports can be broadcast live directly from the mobile unit, or a news spot can be completely prepared and the tape played back on cue.

Figure 2. Studio B at WCBS. The control room is behind announcer Jim Harper.

Figure 3. The racks in control room A. Metrotech logging recorders are on the left, f.m. automation Scully recorders are in the center, and a cartridge Carousel for the automatic insert of commercials and station i.d. are on the right.

The twice-daily rush-hour traffic syndrome is covered by a WCBS helicopter. In discussing the helicopter coverage with Mort Goldberg I mentioned the noise problem encountered by the police helicopter which reports traffic for the city-owned station. Here the cockpit noise tends to mask the pilot's reporting. I asked Mort how the WCBS pilot's voice was kept so far out of the noise. He explained that a special noise-cancelling microphone is used and that it is mounted virtually on the lips of the speaker. The pilot has a split headset, with one earphone monitoring the FAA channel for flight instructions, and one monitoring the WCBS two-way radio communication system. When the time approaches for the pilot's report, he switches over from the two-way radio to the output of an off-air receiver to get his cue.

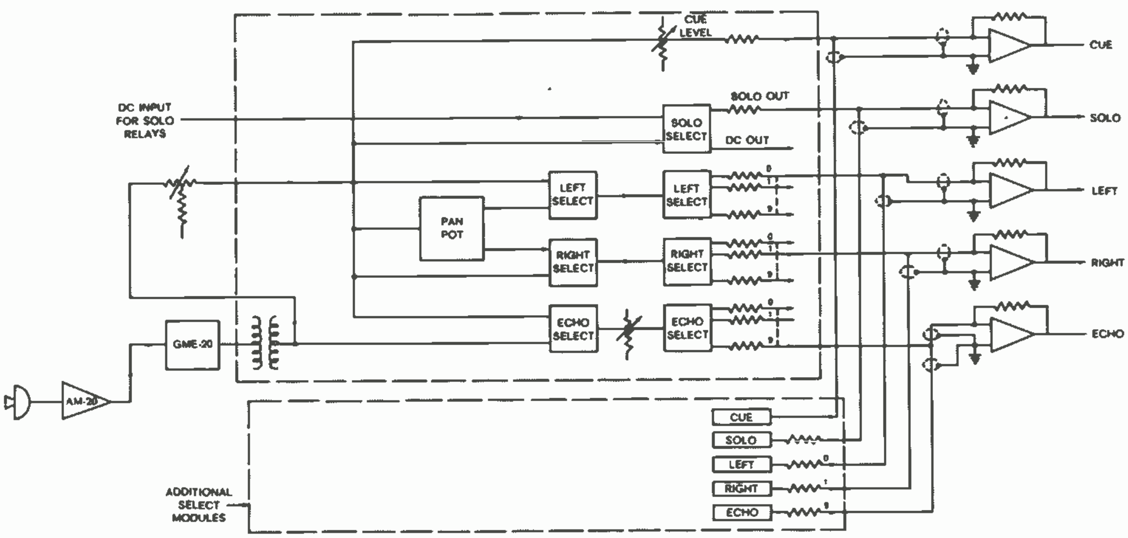
The use of the helicopter need not be confined to reporting traffic conditions. On the recent parade of the Apollo 11 astronauts, the WCBS helicopter participated in a three-way coverage of the event. A mobile unit was dispatched to be in the parade, and was in communication over the two-way radio. Simultaneously the helicopter, with a two-man crew, hovered over the area and also was in communication over the two-way radio. A third form of coverage was provided by stationary positions along the line of the parade which were connected to the studios by telephone lines. The only thing missing from this array was a satellite in orbit—and we may have that before long.



MELCOR CHANNEL SELECT MODULE

The Melcor Channel Select Module is designed to process inputs such as microphone channels in recording consoles, or tape channels in rerecording consoles, and distribute them to selected mixing busses. Three illuminated, ten-position, rotary, push-button activated switches are provided to route input signal to three independent output lines. A pan pot is included, which may be used between two of the three output lines. The third output is available for Echo Send and has its own level control. A Cue output with level control is available as well as push-button activated Solo output. Also included, is an isolation transformer for breaking possible ground loops.

reduces console wiring



CSM Functional Diagram

Module provides capability of switching an audio input signal to 3 sets (left, right, echo) of 10 channel outputs for a total of 30 outputs ■ Internal pan control can continuously vary the signal level from left to right output ■ Echo and cue levels can be set independently and have separate outputs ■ Echo output can be switched for pre or post fader output ■ A momentary action push-button solo output switch provides for monitoring an individual module input ■ Input impedance: input isolation transformer provides an input impedance of 600Ω or 150Ω depending on primary connections ■ Insertion Loss: 1db MAX @ 1KHz when outputs are summed at an operational amplifier with a 6.2 K feedback resistor ■ Maximum input level: +20dbm ■ Cross Channel isolation is 80 db Min. ■ All level controls are audio taper ■ Pan control is 3 db down on both channels at center position ■ Power for lighting — 70MA at 24 to 30 volts D.C.; can also be ±15 V.D.C.

For more detailed information write or call

MELCOR ELECTRONICS CORPORATION

A subsidiary of Newton Electronic Systems, Inc.

1750 NEW HIGHWAY, FARMINGDALE, NEW YORK 11735 ■ 516 / 694-5570 ■ TWX 510-224-6429

Circle 20 on Reader Service Card

Theory and Practice

NORMAN H. CROWHURST

● We must all live busy lives. Sometimes months go by without my receiving any technical queries of the type suited for discussion in this column, and I begin to wonder whether anybody has any queries and so fall back on bits and pieces I know haven't been covered lately.

Than last August's column took a different turn and I suppose several readers were interested to learn that I also write books. But a mere half dozen managed to find time to write and enquire further. Maybe more thought about it but then got busy. . . .

Next, in October's issue, the editor published a letter from a reader, asking how my book "Taking the Mysticism from Mathematics" could be obtained, to which the editor replied that I would autograph a copy for any reader who sent me $4.50 with his name and address.

Well, that brought a response, not only asking for that book, but whether I had any other books published, and also several technical queries that will be grist for this column. So rather than continuing where I left off last month, I'll take up some of these, because these are "live ones".

And in response to those who ask whether I have books on this or that, I've had a list of books still in print made up, which I'll be pleased to send free to anyone who sends me his name and address. Mine is P.O. Box 651, Gold Beach, Oregon 97444. The list gives a brief outline of each book, enough to show whether or not it includes anything of use to you.

Now here's a question that came with one of the book orders: Do I like emitter followers any better than I did the infamous cathode follower?

Well, that's a way of putting it! Many readers must have joined the scene long since I last joined issue about proper and improper use of cathode followers. And when I say joined issue, that brings up memories: a few people really did join issue with me, over a matter that is really a theory-and-practice question.

To answer that reader's question briefly, I'd say that, while there is a similarity between the emitter follower and the cathode follower, there are a couple of significant differences, that make it virtually a different animal. But get matters straight, let me briefly recap the cathode-follower situation.

When a triode tube, or some other tube strapped as a triode, is operated with the plate load in the cathode circuit and the plate connected directly to B+ (FIGURE 1), we have the "infamous cathode-follower" circuit.

This circuit has the property that, regardless of the d.c. plate resistor used, provided it achieves the operating mutual conductance of the tube, the source resistance, to an a.-c. signal, as seen at the cathode, is very approximately the transconductance of the tube.

Thus, if a tube with transconductance of 2,000 micromhos (2 milliamps per volt, to those who know that term better), is connected this way, the a.-c. resistance seen at the cathode is 500 ohms, which is the resistance equivalent of 2,000 micromhos.

Added to this, the textbooks pointed out that the voltage feedback in a cathode-follower stage is very nearly 100 per cent, so distortion is greatly reduced, as compared with the same tube operating with a plate-connected load of the same value. What the textbooks often omitted to point out is that this part of the theory assumed a normal plate load for that kind of tube is used, whether in cathode or plate.

My beef—which led to this notion that I "didn't like" cathode followers—was that you can't have your cake and eat it too. A suitable plate load for a tube with mutual conductance of 2,000 micromhos might be, say 20,000 ohms. This could give a voltage gain, with plate-connected load, approaching 40—say 35.

If, and only if, this is the a.-c. resistance connected in the cathode circuit, then a feedback factor of, say 36, is effective in reducing distortion. If the tube normally gives 5 per cent distortion at a certain level, working it as a cathode follower, with 20,000-ohm load, would reduce that to $5/36 = 0.14$ per cent distortion, at the same output level.

But—and here comes the beef—many people would use such a cathode follower to feed a 500-ohm load, because it provides a source impedance of 500 ohms.

While this is matching, of a sort, the low distortion often claimed doesn't follow through, as anyone who tried it found out. To start with, loading the same tube with a 500-ohm load probably raised the distortion to about 10%, without the feedback. And how much feedback is there?

With the figures used above, the plate a.-c. resistance of the tube would be about 2,850 ohms. Working this tube into an a.c. load of 500 ohms, makes the gain fractional, and the feedback factor about 1.17. So the 10 per cent is reduced only by this factor, to about 8.5 per cent which isn't so awfully good!

While I pointed out this fact, I also pointed out that the cathode follower had some very definite uses. If it was used in the output circuit of a preamp, connected to a high-impedance input, over whatever length line—but not long enough to cause mis-match problems—it could save hum problems just as well as any other form of 500-ohm line. But it must be terminated at the other end with a relatively high impedance (FIGURE 2).

Observing that little rule, the cathode follower could serve in audio circuits very well, and did. Just don't try and use it interchangeably with transformers!

But now, as I say, the emitter follower is something quite different again. The tube had an inherent value of mutual conductance (sometimes called transconductance nowadays), upon which its source impedance is based. Provided the tube was operated in conducting mode (not cut off or saturated) the performance would not deviate far from this.

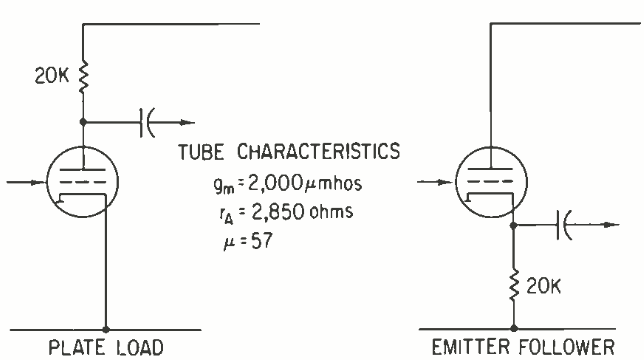


Figure 1. How the parameters of a tube produce the properties of a cathode follower (see the text).

KTIS AM/FM STEREO, MINNEAPOLIS

Every Hour of Every Day

Magnecord tape recorders run hour after hour, every day, under the toughest broadcast conditions. The die-cast mainplate assures permanent mechanical alignment. Timing accuracy is constant with the hysteresis synchronous capstan tape drive. Payout and take up reels have their own heavy duty split-capacitor motors. In short, solid state Magnecords are built to take it, day after day. You can depend on it.

The quality of a Magnecord does not vary from model to model. Features do:

MODEL 1021. Monaural broadcast unit. Inputs: Mixing bridge and choice of Lo-Z microphone or balanced bridge or unbalanced bridge. Cue Speaker. Monitor Amplifier. Two speeds. Balanced 150/600 ohm output.

MODEL 1022. Studio or broadcast stereo unit. Inputs Per Channel: Auxiliary bridge and choice of Lo-Z microphone or balanced bridge or unbalanced bridge. Separate Playback and Record/Gain Controls for each channel. Master Playback and Record/Gain Controls. Balanced 150/600 ohm output. Choice of speeds and head configurations. Full remote option.

MODEL 1024. Commercial or personal stereo unit. Inputs Per Channel: Hi-Z microphone, mixing bridge, auxiliary bridge. Full mixing facilities. 1K ohm emitter follower output. Choice of speeds and head configurations.

Write for the full story on a Magnecord tape recorder. You can depend on a prompt answer.

PRODUCTS OF SOUND RESEARCH
TELEX®
COMMUNICATIONS DIVISION
9600 Aldrich Avenue South
Minneapolis, Minnesota 55420

Circle 26 on Reader Service Card

The emitter follower is much more flexible and has somewhat different properties. The important parameter about the transistors it uses is current gain, commonly known as beta. This means that collector current is approximately beta times base current, or conversely that base current is collector current divided by beta.

It also means (FIGURE 3) that the a.-c. at the base is approximately the a.c. delivered by the emitter, divided by beta, and that the a.c. delivered by the emitter is beta times the a.c. fed to the base.

As voltage transfer through an emitter follower is virtually zero, these changes in a.c. occur without changing the associated a.c. voltages.

Thus the emitter follower provides two-way impedance transformation. Suppose beta is 100, and that the a.-c. emitter load is 500-ohms: the a.-c. impedance reflected at the base will be 500 times 100, or 50K. And if the base circuit has an a.-c. source impedance of 50K, this will look like 500 ohms at the emitter.

This transformation is effective over a fairly wide range of values, and can be used in far more ways than was possible with the cathode follower. And feedback is effective, whatever values are used, calculated by methods we have outlined before, and probably will again.

So, to answer that question in a nutshell, I'd say that the emitter follower definitely does not suffer from the limitations to which I objected with the cathode follower. While I don't suppose I'll be designing any more circuits to use cathode followers, I have certainly used enough of them in the past. But nothing to be compared with the uses

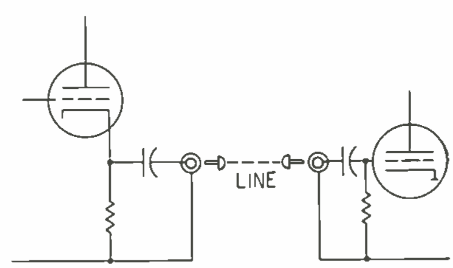


Figure 2. A valid way to use the old cathode follower.

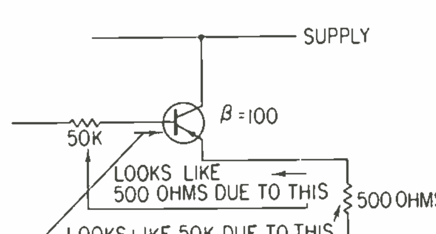


Figure 3. The basic properties of an emitter follower, which are far more versatile than the cathode follower ever was.

to which I have already put the emitter follower, with variations of which the cathode follower was not capable.

Incidentally, the responses to which I referred at the beginning of this column have led me to realize that the publishing scene today has a weakness: with all the junk mail coming out, promoting my books, and those of other authors who have something to say, buried in the pile somewhere, many readers cannot, or do not, find out about books they would really like to know about.

Also, authors have virtually no contact, or means of contact, with their readers. Many a reader would like to contact a book's author and ask a relevant question, but he doesn't know how. What I am now doing, in making a book-list available, not only enables readers to find out what I've written: they can write to me, too, if they have a question I might be able to answer for them.

And I might sell a few more books that way: I guess I deserve to. (*I think so too. Ed*)

The Sound Of Koss Surpasses Every Traditional Stereo Listening System!

MODEL ESP-9
ELECTROSTATIC STEREOPHONES
GIVE YOU ALL 10 OCTAVES

Rear view of the E-9 Energizer.

Functional Block Schematic showing switching functions and hook-up of the ESP-9.

NEW!

SUPER, WIDE-RANGE RESPONSE for critical, controlled monitoring of finest recording sources. Delivers all 10 audible octaves, 15-15,000 Hz $\pm$ 2 db, 4 octaves beyond ordinary headphones.

VIRTUALLY DISTORTION-FREE PERFORMANCE through precision electrical balancing of push-pull acoustical circuitry to give fatigue-free listening through long, intense recording sessions. Elements cancel all 2nd harmonic distortion, unlike conventional units.

LIGHTWEIGHT—HUMAN ENGINEERED FOR COMFORT— Uses fluid-filled cushions for distributed gentle pressure with good seal; coupling transformers and circuitry located in external housing; extendable stainless steel headband with wide cushion for perfect fit and restful listening.

CALIBRATED, PRECISELY CONTROLLED OUTPUT—IDEAL FOR AUDIOMETRIC USES— Switch on front panel of energizer selects ac operation for precision measurements of output; in self-energized switch position no connection to ac lines is required; this gives maximum convenience.

HIGH-POWER CAPABILITY IN VERY LOW BASS RANGE—Large, oversize coupling transformers mounted in E-9 energizer unit give good wave form at 30 Hz with up to 10 volts input.

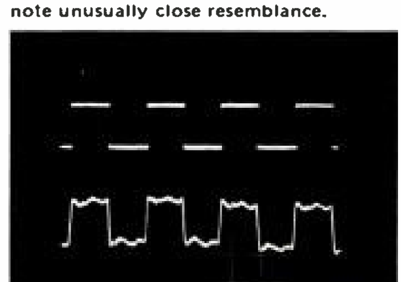
NO SPECIAL AMPLIFIERS REQUIRED—CONNECTS TO LOW-IMPEDANCE SPEAKER TERMINALS—Easy, quick hook-up to any good amplifier delivers performance to specification.

The ESP-9 is a refinement of the famous ESP 6 Electrostatic Stereophones. The most important new feature is a response range of 10 octaves, **the widest ever attained in a headset.** A new cup design promotes virtually linear response to below 20 Hz.

The ESP-9 has a signal handling capacity of 10 volts at 30 Hz with good wave form versus 6 volts for the ESP-6. This is made possible by increasing the size of the coupling transformers by a factor of 4, and mounting them externally to the cup in the E-9 Energizer.

The E-9 Energizer offers the option of self-energizing for the bias supply, or energizing through the ac line; choice is made with a selector switch on the front panel. When energized through the ac line, very precise level measurements can be made. Thus the unit is ideal for audiometry, and for evaluating the spectral character of very low level noise in equipment like tape mastering machines and recording consoles. In contrast to the ESP-6 and ESP-7, both cups are independently energized; a left cup signal is not required to supply bias to the right cup.

TYPICAL SQUARE WAVE RESPONSE AT 400 Hz.
Trace at top is input, lower trace is ESP-9; note unusually close resemblance.



Circle 24 on Reader Service Card

ELECTRICAL SPECIFICATIONS

Frequency Response Range, Typical: 15-15,000 Hz ± 2 db (10 octaves) 10-19,000 Hz ± 5 db. An individual, machine-run calibration curve accompanies each headset. This curve uses standard 3-1/2 log-cycle chart paper, and reads from 20 to 20,000 Hz only.

Sensitivity: 90 db SPL at 1kHz ± 1 db referred to 0.0002 dynes/cm^2 with 1 volt at the input. Variations from calibration furnished are less than 1/2 db at 25°C.

Total Harmonic Distortion: Less than 1/5 of 1% at 110 db SPL.

Isolation From External Noise: 40 db average through fluid-filled cushions provided as an integral part of the headset.

Power Handling Capability: Maximum continuous program material should not exceed 10 volts (12 watts) as read by an ac VTVM (Ballantine meter 310B or equal) with average indicating circuitry and rms calibrated scale; provides for transient peaks 14 db beyond the continuous level of 10 volts.

Source Impedance: Designed to work from 4-16 ohm amplifier outputs. At higher impedances response at the extremes of the frequency range will progressively reduce; e.g., 50 ohms causes a loss of 5 db at 30 and 10,000 Hz.

External Power Requirements: None, except when used for precise low level signal measurement, when external ac line can be selected by a front panel switch on the E-9 Energizer (1/16 amp, 117 VAC, 50-60 Hz normally; 234 VAC with internal strap for foreign use).

PHYSICAL SPECIFICATIONS

Size of Cup: 4-1/4" h x 3-3/4" w x 1-1/4" d.

Cushions: Fluid filled for high ambient noise isolation.

Headband: Extendable, stainless steel bands with self-adjusting pivoting yokes; conforms to any head size.

Headband Cover: Formed of wide, soft molded-rubber with 1/2" polyethylene sponge cushion on underside.

Boom Mount for Microphone: Knurled, anodized, aluminum knob on left cup with threaded shaft and 2 compressible rubber washers; accepts all standard booms.

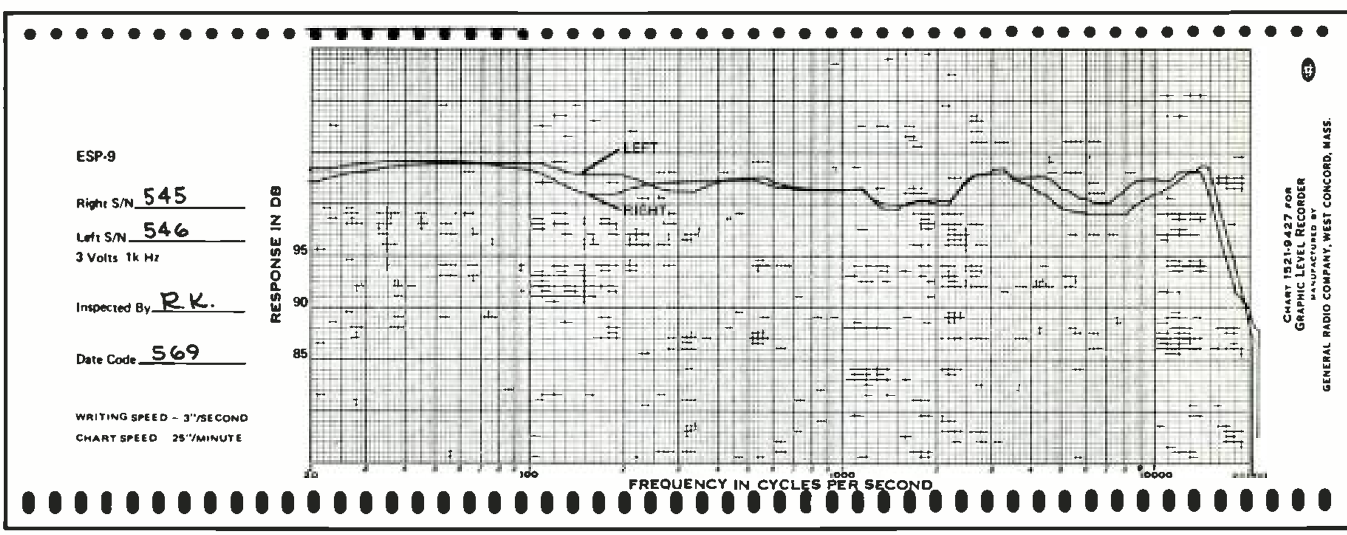
Headset Cable: Flexible, polyvinyl, 5 conductor, shielded, 6' long, black, with 5 prong plug keyed to E-9 Energizer receptacle.

Weight of Headset Only: 19 ounces

E-9 Energizer: Contains 2 coupling transformers, self-energizing circuitry, speaker/headphone transfer key-switch and ac pilot light on black anodized front panel. Also contains ac power transformer, ac on-off switch, ac line fuse, and speaker terminals. Size is 4-1/2" h x 3-3/4" w x 6-1/4" d; weight 3 pounds. Has 6' 4 conductor input cable terminated with 4 spade lugs to connect to amplifier output terminals.

Accessory Provided: 6' ac line-cord P/N 41-0235 for optional use, with plug on one end and plug-receptacle on the other.

Model ESP-9 Studio Monitor: Electrostatic Stereophones, complete with E-9 Energizer, ac line-cord, machine-run calibrated response curve and instructions; Shipping weight 6 pounds; Price **$150.00**



MACHINE RUN RESPONSE CURVE OF THE MODEL ESP-9

KOSS

KOSS ELECTRONICS INC / MILWAUKEE, WISCONSIN
2227 NORTH THIRTY FIRST STREET / ZIP 53208

Primer on Methods and Scales of Noise Measurement

WAYNE RUDMOSE

Noise is with us all the time. We have discussed noise in the past and will continue to do so on the future. The understanding of basics to be gained in this article should be of considerable assistance to the reader.

Noise—in the sense of unwanted sound—has been a problem since Eve first poked Adam in the remaining ribs and told him to stop snoring. As time has gone on, and the population has increased, we have become even noisier—until today it seems that man-made sound of one sort or another bombards our ears almost continuously. The problem of noise has become so great that it is regarded as a special type of air pollution and, as such, is judged by many as a problem which might well lie within Federal jurisdiction.

In June 1968 a conference on noise as a public health hazard was held in Washington, D.C. It attempted to define the extent to which noise has become a public health hazard. Speakers from this country and Europe (where the problems are more extreme due to high population densities and where noise control is accepted as a matter of course) were called on for papers relevent to this problem. Many of these papers are of a highly technical character and not pertinent to readers of db. However, we feel that the paper that follows, by Wayne Rudmose, is outstanding as a basic summary of the methods used to measure the physical and psychological attributes of sound. It is, in fact, now being used at several universities to provide a more than basic level of understanding of sound measurements.

We are therefore pleased to offer this Primer—to appear in several installments. We wish to thank Mr. Rudmose and the American Speech and Hearing Association for their permission to reprint this material which originally appeared in the ASHA Reports 4 (February 1969). Ed.

It is appropriate to begin by commenting briefly on the nature of sound waves since many of the measurements under consideration are keyed to the fundamental properties of sound and because the reasons and necessities for using special procedures are intimately involved with the properties of the sound wave.

Sound, as we know it, is propagated in the form of a compressional wave. The fact that it is a compressional wave demands that the wave must move in a physical medium of some sort which can be compressed and rarefied. Sound waves are almost always referred to as compressional rather than rarefactional waves. We tend to overlook that, for airborne sounds, symmetry of waveforms ceases to exist when the amplitude of rarefaction reaches a peak value of one atmosphere, because there is nothing left to "rarefy." In a world where energy is sometimes measured in megatons of explosives, we must constantly remind ourselves that even for extremely loud sounds the total amount of energy in a sound wave is usually quite small. It follows then that the forces which sound waves can exert should be categorized as moderate to small. Only when one talks of extremely loud sounds will this classification be subject to modification.

Most of the sounds to be discussed here will be those propagated through our atmosphere. As the atmosphere is compressed and rarefied by these sound waves, a very small fluctuation in the barometric pressure, attributable to the passage of the sound wave, occurs. Although there are other measurable attributes of the sound wave, it is almost a certainty that, at this Conference, all of the data reported relating to the amplitude of the sound wave will have been measured by pressure-operated microphones sensitive to fluctuations in barometric pressure.

Two major classifications of sound wave types are used in acoustics: plane waves and spherical waves. These names are derived from the geometric description of the wave front, the area occupied by the forward part of the pressure disturbance. A plane wave results, in general, if the source is large, if the measuring point is at a long distance from the source, and if there are no reflecting surfaces of any kind between the source and the measuring point. Under these conditions the forward edge of the wave will lie on the surface of a flat sheet or plane which is perpendicular to the direction of propagation of the wave. Measurement of a plane wave's pressure is simplified because the microphone location is essentially unimportant as long as it remains in the plane or one parallel to it.

The spherical wave is classically described as the type of radiation coming from a small pulsating sphere where the point of measurement is reasonably close to the sphere and there are no reflecting surfaces in the vicinity of the source or the point of measurement. If one were to obtain an instantaneous snapshot of the pressure distribution of the forward part of such a wave, it would be evident that the maximum pressure, for example, would lie on the surface of a sphere with the center of the sphere located at the pulsating source. It should be pointed out that as one gets farther and farther from a spherical source, a reasonably small section of the forward wave front would occupy an area approaching a flat plane.

One important feature of a plane wave is that the pressure amplitude does not change with distance unless there is some form of absorption or dissipation in the medium as the wave travels through the medium. In the case of the spherical wave, the pressure amplitude decreases as the wave travels farther from its source, and this decrease is commonly referred to as the "inverse square law" change in amplitude. Since any real medium departs from an "ideal" medium, there is always some amount of dissipation of a sound wave as it advances through a medium. This dissipation represents a loss of energy, and the amplitude of the sound wave is affected by this loss of energy. Thus the amplitudes of plane waves do decrease with increasing distance and spherical waves lose energy at a greater rate than the inverse square law (6dB/double distance) would predict.

FIELDS OF MEASUREMENT

It is important in measuring sound waves to recognize the various characteristics of plane and spherical waves. One must also recognize the interpretation of data depends not only upon the measurement of a sound wave's pressure amplitude but also upon the location of the measuring microphone. Points of measurement are commonly referred to as "far field" and "near field." These words are so descriptive that they need little interpretation except to explain one fundamental question: When is the measuring point in the far field and when is it in the near field? For simplicity let us assume that measurements are being made in an environment which is far removed from any reflecting surfaces. Such environments can be realized in a laboratory by providing highly efficient absorbing materials for wall surfaces, as in an anechoic chamber. In such an environment, a measuring point is considered to be in the far field if, by increasing somewhat the distance of the measuring point from a small source, the pressure measurement change follows the inverse square law, i.e., as the distance from the source is doubled the measured sound pressure level decreases by 6 dB. An engineering rule of thumb commonly used to specify a measuring position as being in the far field is that the measuring point should be at least several wavelengths, or several times the circumference of the source, whichever is larger, from the source.

The near field of a source is defined as that region where changes in pressure amplitudes do not obey the inverse square law. Normally such positions are within a few wavelengths of the source. A transitional region clearly exists between the near field and the far field as one increases the distance from the source to the measuring point. Unfortunately, the mathematical description of the wave in this transitional region is far from simple. In practice one tries to avoid making measurements in the transitional region, but in real life it is sometimes impossible to avoid making measurements within this ill-defined space.

RELATIONSHIP OF MEDIA TO SOUND PROPAGATION

At this point I wish to consider how the media affect the movement, or propagation, of sound waves, and to point out certain features of the waves that are altered when sound travels from one medium to another. Since a sound wave represents a pressure disturbance in a medium, and since this pressure disturbance moves through the medium, there must clearly be a certain speed of propagation of this pressure disturbance. The speed of sound within any medium depends upon the compressibility of the medium and upon the density of the medium. If the medium is solid, such as a metal bar, and the sound wave is being transmitted along the bar in the direction of its length, then the compressibility is defined by a term called "Young's modulus." If the medium is a fluid or a gas, the compressibility is given in terms of the bulk modulus of the fluid or gas. For most of the sounds to be discussed, the elasticity, or bulk modulus, of the medium is solely a property of the medium and does not depend upon the rate of compression or upon the amplitude of compression of the medium. Thus the speed of sound does not depend upon the frequency or amplitude of the sound wave but only upon the medium. The uninitiated cannot appreciate the simplification which results as a consequence of this statement, but if this were not true, then sounds would change in quality as a function of distance from the source, and waveform or shape would change as the wave moves etc.

As a sound wave travels from one medium to another, certain changes in the wave do, however, take place. It is important to understand these changes and to recognize how the characteristics of the media affect the changes. Since the frequency of a sound wave depends upon the source which created it; since the speed of propagation of a sound wave depends only upon the medium; and since the wavelength of a sound wave equals the speed of propagation divided by the frequency (a relationship which I just draw from thin air for lack of time), it follows that when the sound moves from one medium to another there is a change in wavelength. To illustrate the point, consider a sound wave traveling from air into water. Since the speed of sound is approximately four times as great in water as in air, the wavelength of a given frequency of sound in water will be four times as great as the wavelength of the same frequency in air.

Another important change that takes place as sound travels from one medium to another is a change in the pressure amplitude of the sound wave. As a sound wave strikes the boundary between two media, a reflected wave comes back with respect to the original wave and travels in the same medium as the original wave. In general, there will also be a wave transmitted into the second medium, and the pressure amplitude of this wave must obviously be less than the pressure amplitude of the original wave. Similarly, the pressure amplitude of the reflected wave must also be less than the pressure amplitude of the original wave, otherwise the law of conservation of energy is violated. The question of whether the pressure amplitude of the reflected wave is greater or less than the pressure amplitude of the transmitted wave depends in a somewhat complicated way upon the speeds of propagation of the waves and the densities of the two media. Crudely speaking, the more nearly the second medium resembles the first medium, the greater will be the pressure amplitude of the transmitted wave and the smaller will be the amplitude of the reflected wave; however, as the second medium becomes distinctly different from the first medium in the product of its density and the speed of sound, the reflected pressure amplitude will become greater than the transmitted pressure amplitude. This knowledge is important when one wishes either to absorb sound energy or to reflect sound energy ,and as subsequent speakers describe their subjects this fact will most likely become quite evident.

METHODS OF DEFINING A NOISE SOURCE

Turning now to the problem of how one defines a noise source, there are two commonly used methods. The first method describes the free field pressure radiation measured as a function of the angle of radiation from the source. Measurements using this method are normally made in the far field and represent the sound pressure radiated in various directions from the source under the condition that there are no reflecting surfaces in the vicinity of the source. Such measurements are commonly made in an anechoic chamber.

The second method leads to data which define the power radiated by the source. Measurements under this method are normally conducted either in a reverberation chamber or in a semireverberant room. Here the directional characteristics of the source are ignored, and the measurements yield only the total energy radiated. If one knows the free field radiation characteristics, the power of the source can be calculated; however, if only the power of the source is known, one cannot calculate the free field radiation characteristics.

The choice of method is made on the basis of the intended application. If one is interested in calculating the noise a machine will generate if it is placed in a room of known acoustical characteristics, then he should use the power definition. However, if one is calculating the effect of a machine as it radiates its noise toward a neighborhood, the free field pressure definition is generally most applicable.

In either case the measurements, whether they are sound pressure level or power level, are normally specified as a function of frequency. The most common way of specifying the frequency function is to supply data indicating sound pressure level in octave bands (or one-third octave bands) or sound power in octave bands (or one-third octave bands). It is conventional to use the decibel as the measurement of either sound pressure level or power level, but one must be careful to recognize that the reference levels for these two units are different. These reference levels will be discussed next month.

THE SOUND MEASUREMENT SCALE

Let us turn now to the physical measurement of sounds. Sound pressure is the attribute which relates to the amplitude of the sound, and frequency is the attribute which relates to the pitch of the sound. The range of sound pressures of interest to us is represented on the low end by the threshold of hearing of normal young people and on the upper end by the noise of small arms measured in the near field. Stated in physical terms, this sound pressure range is approximately from 0.0001 to 100,000 dynes per square centimeter (or microbars). (Atmospheric pressure represents 1,000,000 dynes per square centimeter.) It is clear that we are dealing with a tremendous range of sound pressures. Because in acoustics we are just as interested in observing the effects of small changes near the threshold of hearing as we are in observing the effects of small changes near the upper end of the scale, it would be impossible to construct a linear scale which would be applicable to our problem. An analogous problem which might be more meaningful would be one of measuring lengths where we are interested in having a scale ranging from one inch to 16,000 miles, and we need a ruler to measure changes of a few inches or changes of a few miles with the same ruler. The simplest mathematical scale available for either purpose is the logarithmic or decibel scale. One characteristic of the decibel scale is that it is possible to show, on an ordinary sheet of graph paper, a large range of sound pressures in such a manner that the small variations are as accurately portrayed as are the large variations. In my opinion this is the principal reason that the decibel scale is so useful in the field of acoustics.

Even though the decibel scale is a useful scale, it still creates many problems for the beginner and the uninitiated. I do not feel justified at this point in entering into mathematical definitions and calculations, but I shall try to give you some feeling for the units which you will hear and see in most, if not all, of the subsequent presentations. In our field of acoustics whenever the sound pressure level is designated in decibels, the reference level will be 0.0002 dyne per square centimeter (or microbar). On this scale zero dB sound pressure level corresponds to a pressure of 0.0002 microbar. A sound pressure level of 60 dB corresponds to a pressure not 60 times the reference pressure but 1000 times the reference pressure or 0.2 microbar. A sound pressure level of 100 dB corresponds to a pressure of 20 microbars, and finally a sound pressure level of 160 dB corresponds to a sound pressure of 20,000 microbars.

Sound-power level is also expressed in decibels, but here the reference level is 10^{-12} watt.[1] This means that a power level of zero dB represents a power of 10^{-12} watt. A power level of 60 dB represents a sound power of 10^{-6} watt; a power level of 100 dB corresponds to a power of 10^{-2} watt; and a sound-power level of 120 dB corresponds to a power of only 1 watt. These examples illustrate that for most of the sounds encountered in our daily lives the actual amount of power involved is less than 1 watt, which seems truly insignificant in comparison with our normal sources of power.

continued next month

A Different View of Speaker Coverage

ELLIOTT FULL

The choice of equipment to go into a public-address system should be made, at least in part, on the purpose for the system, rather than just on the basis for coverage. The author describes several examples.

The questions of psycho-acoustic considerations of a public-address system are brought up in this article. This is an area that has been explored extensively in print so far as we know. Accordingly, we would be most interested in hearing from other operators as to their experiences and practices to make a p.-a system do the really best it can

"Keep the level low and use a lot of speakers"
"Put 'em in a circle and locate them high over the center of the crowd"
"Phase 'em and aim them at the crowd"

There are almost as many speaker placement theories as there are practitioners of the art. How a large crowd can be covered depends on many factors: What power will be needed to cover the background noise? What fidelity needed, will music be transmitted? Are there going to be reflections that will adversely effect understandability? Amplifier power, speaker efficiency and crowd density have to be calculated, sometimes empirically, and related to cover a crowd which itself may vary in size. How much money is available?

The above are some of the usual considerations that are, or should be, the concern of the p.-a. designer. Other factors are in the habit of cropping up. Each installation has its own problems.

Indoor p.-a. has many of the same problems, but the designer must be aware of the room or auditorium's time constant, so that the reverberation won't make the program material difficult or impossible to understand. Cardioid mikes and linear equipment will often help all p. a. systems, particularly the indoor variety.

There is one more problem that few, if any, designers account for. *What is the interrelationship between the audience and the man at the microphone*: is he just entertaining them, giving them their money's worth or is he going to demand something of them?

The premise of this article, admittedly only partially tested is that the audio for these two situations should be basically different. If the audio is to be used to warn of a hazard, such as the propellers at an air show, swimming pool precautions, race track dangers, unruly crowd warnings, etc., a different brand of audio should be used! In such cases we feel that the ordinary p.-a. system with its several medium-or low quality widely-separated speakers, restricted-range microphone and amplifier, will not elicit the audience response needed or desired. A commanding voice coming from one location, as with ordinary person-to-person communication, becomes more demanding, more real and less mechanical than other types of p.-a. coverage. This amplified voice becomes the "superman" that will take care of the problem situation. Nietze, the philosopher, outlined this concept in his "Man and Superman". If possible, this high quality, clear voice should be above or removed from the crowd. It should be used as little as possible.

Where unruly outdoor crowds are becoming more frequent, how would this theory be applied? First, a loud commanding voice cannot come from a squad car's fender mounted reflex trumpet. The *commanding voice* should be just as spoken by the man, but with the level raised about 30 dB or more.

Continuing with the police example, what hardware might be used? (1) A high-quality, medium-sized speaker such as a Klipsch La Scala mounted on a van, protected, water proofed and rotatable, (2) A good-quality solid-state 25-watt amplifier, (3) a broadcast-type cardioid microphone such as an EV RE15. The man at the microphone has to have a good voice and be trained and capable of using the tool to it's greatest advantage.

In a large auditorium with a high quality, low-level, multi-speaker installation, it might be well to shift to a central dominant speaker with amplifier gains and tone controls moved to marked locations in case of panic or fire. The announcer would then shift to an assured, controlled lower tone of voice, then point out the exits and minimize the concerns of the listeners. We have used this theory at several athletic contests. We bypassed the existing poorly-maintained multiple-speaker installation and installed an identical amplifier, well maintained, and a single speaker with a 26-inch mouth and four drivers. Among other requests made of the audience, we told them not to litter the grounds. After the all day events involving about a thousand people, the grounds were still clean when the spectators departed.

Elliott Full is vice-president of radio station KXIC-AM & FM in Iowa City, Iowa.

Decca's Vienna Venue

JOHN BORWICK

One of the worlds great serious music halls is also a recording studio (venue). The author guides us through this unique installation located in the city of classical music.

Surely the world's most famous recording venue is the Sophiensaal in Vienna. I have just been on a visit to Vienna and spent a fascinating time in the Sophiensaal at the invitation of Gordon Parry, Decca's senior recording engineer.

The Sophiensaal — there are three splendid halls in fact, superbly architectured *en suite* and decorated in Vienna's richest style — has hit the audio headlines in various ways. First, we know it is the place where Decca (London) recorded their monumentous complete cycle of Wagner's *Ring* — and numerous other recordings with the Vienna Philharmonic Orchestra, Willi Boskovsky, etc. Then a wider public saw Decca at work on recording *Gotterdammerung* in the historic film *The Golden Ring* made by BBC Television and since shown all over the world. And John Culshaw, formerly classical record director of Decca, has now set down in print the whole story of these Sophiensaal adventures in his book *Ring Resounding*.

Perhaps the most unique feature of this special "remote studio" is the thoroughness with which John Culshaw and Gordon Parry (as long ago as 1956, set about locating a suitable hall in Vienna and, having been delighted with the Sophiensaal acoustics) dug themselves in. They did this both technically and domestically, wiring the whole place for sound, designing and building a huge custom control console and even setting up house in an apartment of rooms upstairs. (This flat is the scene of many between-recordings parties and discussions and while lunching there I was very conscious of the great names who had preceded me, Flagstad, Solti, Nilsson, Karajan, Fischer-Dieskau, and so on. There were many momentoes too, like the actual steerhorns used in *Gotterdammerung*).

THE CONTROL CONSOLE

A general view of the massive control desk is shown in Figure 1. The now-familiar concept of slide fader modular strips each containing all the facilities for that channel was adopted. There are basically 20 Channels, in the unusual configuration of 10 to each operator position with the producer's chair (originally for John Culshaw) in the center.

This mode of working is a Decca speciality and it is common for Gordon Parry and his colleague Jimmy Lock to work in tandem, mixing the orchestra on one panel and voices on the other. Sometimes, as in the orchestral interludes in the recently completed *Der Rosenkavalier* recording, the whole desk will be used for the orchestra alone, even though the 8 switchable echo-return faders can be used as extra microphone channels. So a modification is planned, using space-saving integrated circuits, to increase the number of channels.

Four echo sources are on tap, two echo chambers and two stereo EMT echo plates, with remote-control reverberation period. The send/return circuitry is novel in that the return can be controlled either separately or ganged in with the channel fader. For truer orchestral reverberation (hardly ever needed) or special effects, the subsidiary Blauersaal can be co-opted as giant echo-room. It is effectively soundproofed from the main Grosse Saal and has an 8 seconds reverberation time. It is heard to impressive effect in the *Ring* recordings when Fafner's voice was bounced in there from 12 different loudspeakers! More subtly, I was given a foretaste of its use in the still-to-be-issued *Der Rosenkavalier* master tape where a beautiful airy perspective was achieved by placing the stage orchestra of some 40 players in the Blauersaal (with a separate conductor) and mixing this with the main performers in the Grosse Saal.

The distribution networks are unusually versatile. Outputs from individual channels, grouped in any required manner, can be sent to a variety of listening points. (It is important that the cue signal sent to off-stage performers does not contain their own sound.) Tape replay too is tricky, and can be stereo or mono and either selected or reduced from four-track as required. The Dolby noise reduction system is always used and this was certainly the first location in the world to install 8 Dolby units to give 4-track stretch and de-stretch (for monitoring) operation.

OTHER TECHNICAL FEATURES

Closed-circuit television is widely used. The control room has no window into the main hall and so the console faces two large t.v. monitors flanked by Tannoy loudspeakers (behind gauze curtains). British Quad 50E power amplifiers are used for line and speaker distribution etc., but the channel limiters and equalizers are all of Decca design.

With so many expensive and busy artists to record, great care has been taken to double up on equipment. The rack-

---

John Borwick is technical editor of the British publication The Gramophone and is currently secretary of the Association of Professional Recording Studios with close on 150 member studios in Britain. He has contributed before on the professional scene in Europe.

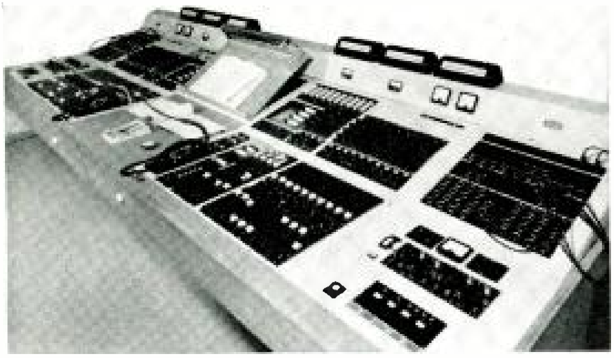


Figure 1. The console installed by Decca (London in the U.S.) at Vienna's Sophiensaal. Note the use of strip-type peak-reading meters.

Figure 2. A rear view of the console shown in Figure 1.

mounted amplifiers, for instance (see FIGURE 2) have instant change-over switches and pull out for physical replacement. This is true of the power supply banks too, which identify faulty units by warning lights and can be replaced in a few seconds. Taping too is normally double-banked, using the EMT/Studer C37 and J37 studio machines so widely employed in European studios. Final editing of the tapes is carried out on the spot, so that it is the final *masters* which are sent back to Decca headquarters in London. They are Dolby-ized, of course, and so are compatible with the rest of Decca's classical record program.

Neumann electrostatic microphones have been first choice for a number of years but a revolutionary new microphone technique is presently being tried. This includes a trial of new AKG dynamic microphones and has been prompted by the need to look ahead to combined audio/visual presentations. As cameras and microphones will increasingly need to work together, re-thinking of the microphone's job is vital. Film and television needs have now been joined by EVR (electronic video recording) and Decca wants to be ready for any move towards the audio visual gramophone record of the future. During my visit, indeed, they were completing the sessions on a t.v. film soundtrack of *Cosi fan tutte*. The Decca engineers and the singers reluctantly agreed to voice dubbing for the arias but they felt that miming the recitatives was too prone to timing errors and so these were filmed and recorded simultaneously.

Nobody can measure the total effect on gramophone records (and equipment) sales brought about by the public impact of a major recording like the Decca *Ring* cycle. No doubt it is considerable, and clearly we can expect further treats and maybe surprises from Decca and from the Sophiensaal.

A Preamp/Balanced Modulator for Audio Experimentation

ROBERT C. EHLE

The author describes an audio frequency, balanced, amplitude modulator useful as a volume expander as well as an envelope shaper, gate, or bell generator and amplitude modulator for experimental audio and electronic music projects. A low-level phono preamp is included.

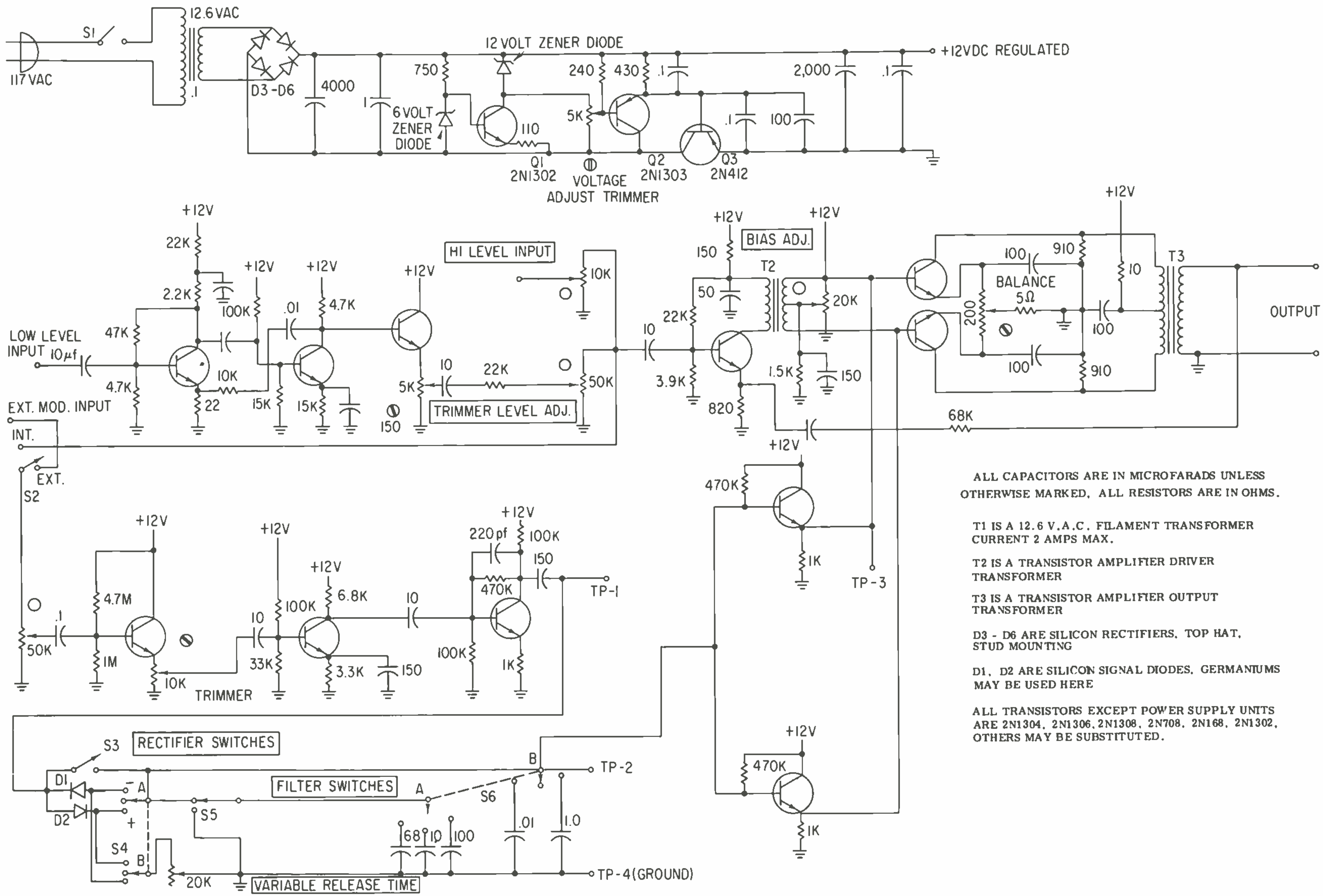
MANY PEOPLE are not merely content to turn their sound systems on and enjoy the music but must be constantly experimenting with various techniques in order to provide new musical experiences through the medium of high fidelity. To such people, high fidelity means more than faithful reproduction of a live performance and often becomes an attempt to apply electronics to a particular approach or interpretation of music. Involving themselves with synthetic stereo, enhanced reverberation, volume expansion, etc., they turn high fidelity into a creative medium and, if they continue far enough may find themselves actually composing original music for electronic instruments. This *electronic music* as it is called is a potential haven for all those musically inclined engineers and technically capable composers.

The balanced, amplitude modulator is a device which is capable of performing many special and interesting modifications to music and other sound material. It does this by a control signal automatically adjusting the amplitude of the program material to its own instantaneous changes in amplitude. If the control signal is a slowly changing, d.c. voltage the result is a slow change in amplitude of the controlled program. This type of operation is used in both the volume expander and the gate or envelope control mode of operation. On the other hand, if the control signal is an audio frequency range signal, the program material will be changed in amplitude at an audio frequency rate where each *loud* peak will correspond to the maximum amplitude of the waveform doing the controlling while the *soft* trough will correspond to the moment of minimum amplitude in the control signal waveform. This non-linear mixing or amplitude modulation results in the generation of a series of sum and difference frequencies from the interaction of the program and the control signals.

For many operations where amplitude modulation is to be used, a balanced modulator is desired so that the control signal does not appear in the output. With such an arrangement, the control signal may be constant but it will not be heard in the output while the sum and difference products will appear only when a program signal is fed into the main input.

The specific circuit designed by the author is shown in

Robert C. Ehle is a consultant and teacher of electronic music. He holds a master of music degree in composition and has taken advanced courses in mathematics, electronics, and computer-systems technology.



ALL CAPACITORS ARE IN MICROFARADS UNLESS OTHERWISE MARKED, ALL RESISTORS ARE IN OHMS.

T1 IS A 12.6 V.A.C. FILAMENT TRANSFORMER CURRENT 2 AMPS MAX.

T2 IS A TRANSISTOR AMPLIFIER DRIVER TRANSFORMER

T3 IS A TRANSISTOR AMPLIFIER OUTPUT TRANSFORMER

D3 - D6 ARE SILICON RECTIFIERS, TOP HAT, STUD MOUNTING

D1, D2 ARE SILICON SIGNAL DIODES, GERMANIUMS MAY BE USED HERE

ALL TRANSISTORS EXCEPT POWER SUPPLY UNITS ARE 2N1304, 2N1306, 2N1308, 2N708, 2N168, 2N1302, OTHERS MAY BE SUBSTITUTED.

Figure 1. The solid-state balanced modulator circuit described in the text.

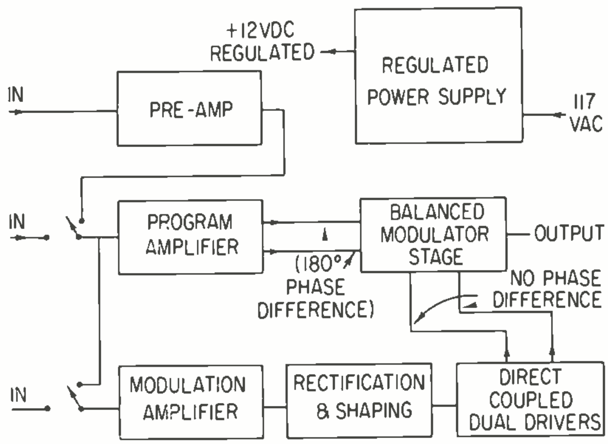


Figure 2. A block diagram of the complete balanced modulator.

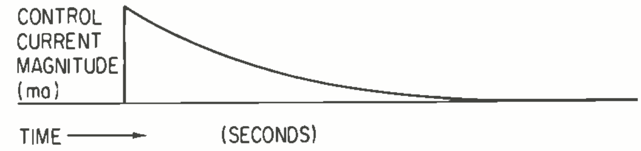


Figure 3. A graph of the current envelope used to generate synthetic bell-type tones.

Figure 4. The wave-form of pulse bursts, useful for testing audio equipment and other experimental applications including electronic music.

Figure 1 and 2, and is a solid state, amplitude modulator containing the following sub-sections:

1. A pre-amp for low-level signals, (this is a simple design; a more elaborate circuit with variable equalization might be substituted).

2. A modulation signal amplifier.

3. An envelope follower shaper with positive and negative rectification and six time-constant capacitors.

4. A balanced, amplitude modulator with one stage of program amplification and direct coupled, low impedance modulation drivers.

5. A regulated power supply.

Articles have been published previously concerning the design and applications of vacuum-tube designs for balanced, amplitude modulators.[1] In the design of a transistorized version, several new situations must be considered which require different approaches than were used with vacuum tubes. Perhaps the most important of these is the fact that since transistors are current operated rather than voltage operated as are vacuum tubes, we must provide a control current to the modulation stage instead of a control voltage which worked in the tube versions. In order to provide the necessary current, a low-impedance output driver stage must be used to drive the modulator stage. Since both halves of the modulator must be driven in parallel, yet a low impedance path may not exist between the two halves because they are 180 degrees out of phase with each other in the program material input, it is necessary to employ a pair of driver stages to drive the modulator.

In the volume-expansion mode, it is necessary to provide a relatively great direct current by means of a rectification and filtering of the audio input. In order to obtain a wide enough range in dynamic control, three stages of audio amplification have been used before the envelope shaping circuits. With this plan it is possible to drive the modulator hard enough that only the loudest passages are heard if such is desired. Such effects, while ludicrous in conventional music, may be useful to the electronic music composer who is working with unconventional material from the outset. The reader is referred elsewhere for information on electronic music equipment and circuits which may be employed with the modulator.[2]

On the schematic diagram of Figure 1, front-panel controls are marked with a circle while those best suited to be screwdriver adjusted are marked with that symbol — a circle cut by a diagonal line. The test points have been mounted on the panel and are used for metering points and, alternatively, as auxiliary inputs for special applications.

The most important and critical adjustment of the instrument is the balance control. This control is set by driving the external modulation input with an audio signal and adjusting the balance control for minimum output. The operation of the balanced circuitry can be improved if matched components, particularly transistors, are employed in the push-pull stage. Even greater improvement could be obtained by purchasing higher quality, balanced transformers. Cancellation actually takes place in the output transformer so this is a critical part of the balanced circuit.

After setting the balance control, the bias and the function switches are set to suit the various desired functions. For volume expansion, it was discovered that the saturation end of the output transistor curve was more linear and gradual than the cut-off portion of the curve which tended to be quite abrupt. The procedure for setting the unit for volume expansion is this:

Turn the bias control to a point where the output transistors are beginning to saturate. This point can be metered by a voltmeter at TP-3 and will be approximately 1.4 volts to ground; also the volume of a signal passing through the unit will be less than at some lower voltage measured at TP-3. Now set the switch, S-2, to internal and increase the modulation volume control to maximum. Open S-3 and turn S-4 to negative. Select a suitable time constant with S-5 and S-6 (fast for jazz and piano music, slower for orchestral music), increasing the shunt capacitance increases the reaction time of the circuit, hence, a longer time constant.

In the operation of the circuit in the expansion mode the amplified, rectified, and filtered signal is used to oppose the bias current of the output transistors, thereby bringing them out of saturation and into a range where their amplification is greater. Although a transistor is not particularly linear near saturation, this causes very little distortion if the signal amplitude at the input is kept small. Since negative control current causes an increase in volume by the mechanism described above, the diode switch, S-4, is turned to negative for volume expansion. Various amounts of volume expansion may be selected by adjusting the modulation volume control and the bias adjustment. The user should experiment with various settings of these and all the controls, particularly if he is interested in some of the unusual effects possible with this circuit.

Finally a discussion of some of the more unusual applications of the balanced modulator is in order. Most significant of these are the applications in electronic music and sound synthesis. One basic use of the balanced modulator is in the generation of complex tones. To do this, the external modulation input is driven by a set of continuously running oscillators. These signals will be balanced out in the modulator and will not be heard in the output as such: however, when a signal enters the signal input of the modulator, the output consists of the input plus sets of sum and difference frequencies made up of the input frequency and each of the modulating frequencies.

The unit may be used as a gate or envelope control device in many ways. If the signal which is to provide the gate is an audio signal, it can be driven into the external modulation input and rectified to produce the necessary varying direct current for increasing and decreasing the volume of the main signal channel. In this mode of operation, the set of the controls duplicates that for volume expansion with the exception of the modulation input switch set for external modulation. A more direct method of providing envelope control is to generate a varying direct current by some means such as a battery in series with a potentiometer (or a cadmium-sulfide cell) which is controlled by the operator, or by a mechanical device (such as a punched paper tape through which light passes to fall on the CdS light-sensitive cell). This direct current can be applied between test points *two* and *four* and the amplitude of the signal channel of the modulator will follow the variations in current.

Generation of synthesized bell tones (see FIGURE 3) is a simple matter with the device mentioned above. All that is required is a control current with a rapid rise and a gradual decay, obtainable with a push button in series with the battery and a potentiometer for decay control. Anyone interested in sound-effect generation or sound synthesis will find many more applications for the unit, for example, with a switch in series with a battery a tone-burst generator (FIGURE 4) can be devised. A tone-burst generator has many applications including the testing of the transient response of audio equipment. Driving test points *two* and *four* with a pulse generator gives a high-speed, electronic switch for audio signals. This arrangement is useful for testing transient response, recovery, and for synthesizing many special effects.

SUMMARY

This article has described a practical circuit for a transistorized, balanced modulator and, in particular, has stressed the adaptability and the versatility of the circuit. The unit described is capable of performing many functions including some remaining for the builder to discover.

Described is an audio frequency, balanced, amplitude modulator which is useful as a volume expander as well as an envelope shaper, gate or bell generator and amplitude modulator for various experimental audio techniques including electronic music procedures. A pre-amp is included for phono and other low-level signals.

REFERENCES

1. R. C. Ehle, A Complex Tone Generator for Electronic Music' **Audio Magazine,** October 1966
2. R. C. Ehle, Design of an Electronic Music Synthesizer, **db, The Sound Engineering Magazine,** November 1968

Sound with Images

MARTIN DICKSTEIN

● Just as in the audio field, where it is necessary to take advantage of (or circumvent) certain characteristics of the ear, it is also necessary in the visual field to know some of the operational shortcomings of the eye and then develop equipment and devices which perform in a manner which will provide pleasing sensations to the viewers.

The eye is complex enough to occupy a complete discussion by itself, but one characteristic which plays a most important part in the operation of various pieces of equipment is the eye's relatively slow reaction to light stimulation. Human vision can not distinguish two separate light impulses which have reached the eye in less than about 1/10th of a second. The principles of film projection and television are based on the fact that any succession of impulses reaching the eye in less than this time period result in the viewer seeing what seems to be a continuous and unbroken scene.

The simple stroboscope also operates on this principle. If a fluorescent light, pulsing at 120 impulses per second, were to shine on a rotating disc which had on it 2 pie-wedges painted on in a dark color for easy visibility, then it would be possible to tell how fast the disc was turning by the relative apparent rotation of the wedges. If the dark areas turned 180 degrees during a dark moment of the lamp, it would seem as though the disc were not turning at all. This would mean, then, that the disc was actually turning in sync with the pulsing lamp. If the pie-shapes seemed to be turning slowly in a direction opposite to the motion of the disc, the disc was turning slower than 60 revolutions per second. If the wedges were rotating forward, the disc's speed was faster than that of the lamp. The speed of the rotation of the wedges also indicates how far off the speed of the disc is relative to that of the lamp pulses. This results in automobile and stagecoach wheels on t.v. or film seem as if they are turning backwards while the car is moving forward. Of course, this principle is applied to testing the speed of turntables, tape recorder drive shafts, projector motors, many shop and factory rotary devices and is also used to observe fast repetitive motions at a slowed-down pace. Even the kids benefit from this effect when they look at the little books with animated figures on successive pages which, when flipped rapidly, seem to be showing a moving picture.

In the projection of motion pictures, a similar trick is played on the eye to provide what appears to be smooth movement in the images. However, the operation of the projector itself does not keep the film moving continuously past the light source port. The movement of the film is actually frame-by-frame past the light port with each frame stopping completely while it is being projected; the construction of the projector provides this motion.

A constant-speed rotary sprocket wheel pulls the film from the feed-reel in a steady motion. However, when the film gets to the film gate, it is intermittently jerked ahead by a claw which fits into the sprocket holes, pulls the film ahead one frame and then recedes, rises, projects, and pulls the next frame down, etc. It is, therefore, imperative that a loop be made in the film between the feed sprocket wheel and the gate to take up and release the quickly changing slack resulting from the two different types of motion and speeds feeding the film through the projector. Some projectors may have the claw mounted on a cam wheel imparting an eccentric drive to the claw shaft which is also pivoted at its center, while other projectors may have a Maltese-cross shaped device rotated eccentrically on a constantly rotating wheel—but the film motion has to be the same past the gate, frame-by-frame with a stop at each frame. To eliminate the flutter effect which would be visible if the film were to move past the gate with no precautions, a rotating shutter is inserted to cover the gate opening during the movement of film in the gate. A three-blade shutter is used most frequently, especially on the smaller film sizes, and the frame being projected is also obscured a small portion of the time it is in the gate to further prevent flutter in the image. Another loop is then formed by the film between the gate and the take-up sprocket wheel. The motion is, therefore, again smoothed out before the film passes over the sound drum and between the exciter lamp and the light-sensing unit. This provides low flutter sound from the optical track and also protects the film from being pulled too sharply by the take-up reel.

In the days of the silent movies, the film frames were close together and the spacing between them was very small. The aspect ratio of width-to-height was maintained and the frame spacing was kept to a minimum. When sound was introduced to the film, the width was decreased to make room for the track. This meant that the height had to be decreased in proportion to maintain the proper ratio.

It was discovered, however, that the aspect ratio considered proper did not really correspond with the true field of vision of the eye. Along with this, came the desire to improve the sound distribution. A new process was developed and called *CinemaScope*. When the film used for this process (35mm) was provided with the usual optical track, the picture width was optically squeezed on the film by a special lens but the height used all the available space to a very narrow dividing line between frames. The aspect ratio was, therefore, not the one that had always been used. It was filmed with an anamorphic lens which distorted the picture to a narrower width than usual. However, when the film was projected, the special lens on the projector compensated for this difference and the resulting picture was made wide-screen, more in keeping with the proper horizontal field of vision of the eye. To accomplish this expansion in the horizontal, the method used could be a lens system with the proper width-expansion factor built in, a mirror system which allowed the picture to expand by the normal reflective process, or with prisms which would spread the image depending on the angular relationship between them. To provide the viewer with distortionless images no matter where in the theater he sat, the screen is usually curved and this creates the illustion of having the viewer in the action. To further this illusion, the sound track can consist of four magnetic stripes on the film with three of them used for dialogue and music and the fourth (narrower than the others) for sound effects. The separate tracks can then be distributed to separate speakers located to enhance the involvement of the viewer. When four tracks are used, the film has two tracks between the image and the sprocket holes (one each side of the image) and two tracks on the outsides of the sprocket holes. In this case, the sprocket holes are made narrower to permit the image width to remain the same.

Another innovation in the film industry was the development of a process using 35mm film but covering a field of vision of approximately the same width as the normal human eye, about 146 degrees. This process, Cinerama, makes use of three projectors locked together in sync and shooting simultaneously onto a curved screen to provide a wide image. The two side projectors fire across the line of throw of the center projector to eliminate image distortion. In this process, the sound is recorded on six tracks and reproduced from screen and surround speakers in the theater. However, the sound tracks are on a separate blind film and played from a fourth projector which does not throw any visual image. Five of the tracks are voice and music with the last for effects.

A different method of producing wide-screen images makes use of the 70mm film, twice as wide as the other processes. Todd-AO uses one camera and one projector with a width of coverage of about 128 degrees. One advantage to using larger film is that there is less magnification required in projection with the result that the image is sharper. Sound is recorded on six magnetic tracks (one on each side of the image inside the sprocket holes, and two on each side of the film outside the sprocket holes). One interesting feature of this wide-screen projection is that the image can lose sharpness due to reflected light coming from the curvature of the screen itself. To prevent this, the screen is made with vertical ridges that eliminate spurious reflections and to keep the screen's reflected light directed to the audience.

As long as we are discussing various processes used to provide visual effects that are pleasant and exciting to the viewer, we should like to take this opportunity to add some previously omitted information. The omission was by no means deliberate. In our discussion of the color camera used on the recent Apollo flights to the moon, we mentioned that the color wheel utilized was an adaptation of the original color t.v. process developed by CBS. Actually, the process is still being used and further developed at CBS. The camera owes a good deal of its excellent functional capabilities to the latest developments of this process and the associated equipment and modifications made especially for the Apollo missions. We thank CBS for keeping us informed and up-to-date so that we may provide our readers with the latest and the best.

New Products

MODULAR AMPLIFIER

● The model Z 30 is a complete power amplifier employing 9 silicon epitaxial planar transistors. Total harmonic distortion is only 0.02 per cent at full output. Power output is 15 watts continuous into 8 ohms using a 35-volt supply. Input sensitivity is 250 mV into 100k ohms and output impedance is 3-15 ohms. The amplifier may be powered from any supply delivering d.c. over an 8 to 35 volt range. A 35-volt supply model PZ6 is available.

Mfr.: Sinclair (England), distributed by Audionics, Inc.
Price: $17.95 (amplifier); $23.95 (power supply).
Circle 51 on Reader Service Card.

BACKGROUND MUSIC

● Expanded programming and a new track-selection feature have been announced in this product, the Cantata Mark II, for the leased music field. A choice of on-location music systems is now offered to fit the budget and environmental desires of every type of prospect. The unit is offered for leasing with a basic package of several 1200-selection music libraries. The system also has a full line of features and accessories to permit operation with public address and payment of music royalties by the manufacturer. Operation is completely automatic, operation is designed for 24-hour use, automatic fading between cycles, and preset, non-tamperable controls.

Mfr.: 3M Company
Circle 54 on Reader Service Card.

DIP FILTER

● A line of inexpensive dip filters is now offered with dip attenuation variable in 2 dB steps with a maximum of 10 dB attenuation. Dip frequencies offered are 3-5-7 kHz. A 14 kHz sync filter model will attenuate the 14 kHz sync signal greater than 45 dB. All filters are 600-ohm and have a 1.5 dB insertion loss. Tolerance is 1 dB non-accumulative. A plotted curve, knob, and dial plate is provided.

Mfr.: United Research Laboratory Corp.
Price: $69.95 (14 kHz filter is $39.95)
Circle 55 on Reader Service Card.

'SCOPE CAMERA

● There is no need to remove film and install focus plates when focusing this new camera. An external indicator eliminates the need for through-the-lens sighting. In addition the camera is adaptable to a variety of oscilloscope bezels so a single camera can easily accommodate several different applications. The new camera, model 198A uses a Polaroid back and standard flat-pack Polaroid films. Shutter and lens controls are readily accessible on the outside of the housing. Shutter speeds range from 1/60 second to 1 second. Shutter operation closes a contact so that the camera can trigger the 'scope or other equipment the instant the shutter opens. The shutter also has B and time exposure settings. The lens is a 75 mm 3.5 lens. A full 8 x 10 cm 'scope trace can be photographed.

Mfr.: Hewlett-Packard
Price: $350.00
Circle 56 on Reader Service Card.

The db Bookcase

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

Use the coupon at the bottom of the page or give the complete title, author and coupon number. Be sure to indicate quantity on the special instructions line if more than one copy of a title is wanted. Full payment must accompany your order. We cannot ship c.o.d. Checks or money orders should be made payable to Sagamore Publishing Company, Inc. Because of the time required to process orders, allow several weeks for the receipt of books.

Broadcasting

RADIO BROADCASTING

edited by Robert L. Hilliard. Each of the five chapters has been written by a prominent educator with an extensive background of practical experience in commercial and educational broadcasting The areas covered include: management and programming, operating and studio facilities, producing and directing, writing, performing. For those of you who want to, or must, operate on both sides of the control room, this is virtually required reading. 190 pages; 6¼ x 9½; indexed; clothbound.

$6.95 ($8.35 in Canada)
Circle 2 on Coupon Below

Reference

New 17th Edition of the famous RADIO HANDBOOK

Tells how to design, build and operate the latest types of amateur transmitters, receivers, transceivers, amplifiers and test equipment. Provides extensive, simplified theory on practically every phase of radio. Broad coverage; all original data, up to date, complete. 847 pages.

$12.95 ($15.50 in Canada)
Circle 10 on Coupon Below

How To

PRACTICAL PA GUIDEBOOK: HOW TO INSTALL, OPERATE AND SERVICE PUBLIC ADDRESS SYSTEMS

by *Norman H. Crowhurst. 1967.* This book gives all the basics needed to become a successful PA operator, in any situation where the reinforcement, relay, or distribution of sound can provide a service. It shows how to properly install, operate and service public address systems. All aspects of the subject, from survey to the selection of appropriate equipment, to installation, to routine operation and the maintenance of a finished system, are covered. Attention is given to solving problems encountered in providing successful service. The book's systematic and practical approach makes it highly useful to radio-TV servicemen, hobbyists, and PA equipment manufacturers. 136 pages; 6 x 9; illus; softbound.

$3.95 ($4.60 in Canada)
Circle 15 on Coupon Below

CLOSED-CIRCUIT TELEVISION HANDBOOK

by **Leon Wortman.** Gives comprehensive detailed information about the field in an easy-to-understand presentation. It's particularly suited to those who plan to use, install, and service cctv. Covers the subject from the simple single-camera system to the most exotic systems. 288 pages; 5½ x 8½; clothbound.

$7.95 ($9.50 in Canada)
Circle 18 on Coupon Below

Electronic Music

ELECTRONIC MUSICAL INSTRUMENTS

by Richard H. Dorf. Now in its third edition and sixth printing since its first appearance in 1954, this is considered the authority on electronic organs. This edition is completely rewritten to explain everything technical about today's organs. The book is of special value to organ designers and service technicians as well as electronics-minded hobbyists and prospective organ purchasers. Of special value are the author's many practical comments and expressions of opinion based on his years of musical, engineering, and management experience with electronic musical instruments. 393 pages; 239 diagrams and photographs.

$10.00 ($11.95 in Canada)
Circle 19 on Coupon Below

General Audio

THE TECHNIQUE OF THE SOUND STUDIO

by *Alec Nisbett.* This is a handbook on radio and recording techniques, but the principles described are equally applicable to film and television sound It describes how the highest standards may be achieved not only in the elaborately equipped studio but also with simple equipment out on location. 264 pages; 60 diagrams; glossary; indexed; 5½ x 8½; clothbound.

$10.50 ($11.95 in Canada)
Circle 1 on Coupon Below

DESIGN OF LOW-NOISE TRANSISTOR INPUT CIRCUITS

by *William A. Rheinfelder.* 1964. Written for students as well as circuit design engineers interested in low-noise circuit design. Throughout, the book gives a multitude of time-saving graphs and design curves for the practical circuit designer. Simple derivations of all important formulas are also presented to help the reader obtain a deeper insight into the fundamentals of practical low-noise design. 128 pages; 6 x 9; illus., clothbound.

$5.50 ($6.50 in Canada)
Circle 14 on Coupon Below

NEW!

THE AUDIO CYCLOPEDIA (2nd EDITION)

by Dr. Howard M. Tremaine. New and updated, here is the complete audio reference library in a single volume. It provides the most comprehensive information on every aspect of the audio art. This new edition includes the latest audio developments including the most recent solid-state systems and integrated circuits. It covers all subjects in the fields of acoustics, recording, and reproduction with more than 3400 related topics. Each topic can be instantly located by a unique index and reference system. More than 1600 illustrations and schematics help make complicated topics masterpieces of clarity. 1760 pages; 6½ x 9-3/8; hardbound.

$29.95 ($35.95 in Canada)
Circle 20 on Coupon Below

ACOUSTICS—ROOM DESIGN AND NOISE CONTROL

by Michael Rettinger. 1968. The enormous problems and hazards presented by noise are dealt within an orderly and practical manner. With many charts, graphs, and practical examples, the text covers the physics of sound, room acoustics, and design, noise and noise reduction. The author's many years of experience in the field and extensive knowledge (set down in two earlier books on acoustics) make this work a boon to architects, builders, designers, planners, and engineers. 392 pages; hardbound.

$17.50 ($20.90 in Canada)
Circle 21 on Coupon Below

Sagamore Publishing Company, Inc.
980 Old Country Road, Plainview, N.Y. 11803
Please send me the books I have circled below. My full remittance in the amount of $......... is enclosed. N.Y. State residents add 5% sales tax.

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

Name..

Street Address..

City........................State..........................Zip.........

Special Instructions..

HANDBOOK OF ELECTRONIC TABLES & FORMULAS, (3rd Edition)
A one-stop source for all charts, tables, formulas, laws, symbols, and standards used in electronics. Devotes complete sections to items of interest to service technicians and to circuit design data. Includes a 8-page, full-color fold-out chart showing latest FCC allocations for the entire frequency spectrum. 232 pages; 5½ x 8½; hardbound.
$5.50 ($6.60 in Canada)
Circle 8 on Coupon Below

Test and Maintenance

INTERMODULATION AND HARMONIC DISTORTION HANDBOOK
by Howard M. Tremaine. A complete reference guidebook on audio signal intermodulation and harmonic distortion. 172 pages; 5½ x 8½; softbound.
$4.25 ($5.25 in Canada)
Circle 9 on Coupon Below

Circuitry and Components

PASSIVE AUDIO NETWORK DESIGN
by Howard M. Tremaine. A complete and comprehensive guide to the design, construction, and testing of all types of attenuators, equalizers, and wave filters for the practicing audio technician or engineer. This authoritative text is one of the few written on the subject, and requires only nominal mathematical background. Written in easy-to-understand language, the content presents the basic design, construction, and testing considerations without the confusion often associated with passive networks. 288 pages; 5½ x 8½; softbound.
$5.50 ($6.45 in Canada)
Circle 5 on Coupon Below

TRANSISTORS FOR AUDIO FREQUENCY (AUDIO-FREQUENCY AMPLIFICATION)
by Guy Fontaine. **1967.** This systematic and detailed treatment of the application of transistors in audio-frequency amplifiers shows how the published transistor characteristics are related to the principles of design. To assure clarity, the figures are rendered in several colors and placed opposite the related text. Simple equations reinforce the lucid approach. An ideal textbook or reference on the subject for engineers and advanced technicians. 384 pages; 5½ x 8; illus.; clothbound.
$7.95 ($9.55 in Canada)
Circle 12 on Coupon Below

ACOUSTICAL TESTS AND MEASUREMENTS
by Don Davis. Provides a solid understanding of the entire subject of acoustical measurements; based on actual field test work, using commercial equipment. Contains practical, time-saving solutions to actual problems encountered in the field; minimum math is required for understanding. The author is an expert in this field, and an authority on auditorium acoustics. An invaluable book for phone company special service engineers, plant maintenance engineers, communications engineers, noise control specialists, architectural engineers, broadcast engineers and technicians, hi-fi fans and students. 192 pages; 5½ x 8½; hardbound.
$6.95 ($8.35 in Canada)
Circle 7 on Coupon Below

Classified

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

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

FOR SALE

SCULLY TAPE RECORDERS — one to twenty-four track and model 270 auto players, many models in stock for immediate delivery. SCULLY LATHES — Previously owned and rebuilt. Variable or automatic pitch. Complete cutting systems with Westrex heads. MIXING CONSOLES — Custom designed using Wiegand Audio Lab modules. From $7,000.00. **Wiegand Audio Laboratories, 3402 Windsor Road, Wall, New Jersey 07719. Phone: 201 681-6443.**

SOLID-STATE AUDIO PLUG-IN OCTAL (1" Dia x 2" H) modules. Mic preamps, disc & tape preamp-equalizers, tape bias osc. & record ampl., power amps & power supplies. Send for free catalog and audio applications. **Opamp Labs., 172 So. Alta Vista Blvd., Los Angeles, California 90036.**

AMPEX AG-350-2 in MR-70 console, stereo half-track plus quarter-track play, 7½-15—$1850. Crown CX-744 4-track with sync panels—$1600. Ampex F-4450 (2)—$200 each. Set of 12 speed-change capstan sleeves for F-4450—$50. Pultec EQP program equalizer—$300. Fisher Series 80-C preamp/mixers (4)—$35 each. **R. S. Field, 833 Las Lomas, Pacific Palisades, California. (213)454-1050.**

12 POSITION AUDIO DESIGNS CONSOLE, with producer area. $10,800. **Sound 80, Inc. 4444 West 76th Street, Minneapolis, Minnesota (612)920-7781**

AMPEX 601-2 PORTABLE broadcast recorder, 200-ohm mike inputs, Samsonite luggage case, supply flutter filter, heads recently re-lapped for first time. With service manual and almost-complete spare-parts kit. Slightly marred case but in excellent operating condition: $350 plus shipping. **J. G. Holt, RD2, Box 314, Glen Mills, Pa. 19342.**

MICROPHONES — dynamic, condenser, ribbon. Precision-built in Germany. Top value, performance and appearance. Write for your catalog, distributor cost sheet, and free booklet "How to Choose and Use Microphones". **Stanford International, San Carlos, California 94070.**

AMPEX 300-350 half-inch 4-track; 602-2 7½ and 15—$450. 350-2 playback electronics—$100. 300 mono electronics with meter panel—$75. 350 type sel-sync—$35. Sigma/350 4-mic mixer—$75. **Guy Heitmann, 12 Lake Road, Peekskill, N. Y. 10566.**

ELECTRONIC FOR AMPEX 350 WITH POWER SUPPLIES FOR 1,2,3, or 4 track. **Box 11-B, db Magazine, 980 Old Country Road, Plainview, N. Y. 11803.**

EMPLOYMENT

PROFESSIONAL RECORDING PERSONNEL SPECIALISTS. A selective service for employers and job seekers: engineers, tape editors, production and studio mgrs, traffic assts, etc. Call us today! **Smith's Personnel Service, 1457 Broadway, N.Y.C. 10036. Alayne Spertell 212 WI 7-3806.**

HELP WANTED—SOUND SYSTEM ENGINEERS. Experience in audio/visual and CCTV helpful. East side-midtown Manhattan. Immediate openings. All information held in strict confidence. **Call (212)679-0400.**

DISTRICT SALES MANAGER for leading commercial sound line. To qualify you must have a minimum of 5 years' experience in commercial sound sales (industrial school, etc.) plus solid technical understanding of the field. Applicants must be capable of building sales volume through effective selection, training, and supervision of commercial sound distribution in multi-state territories. Send full resume and references to **Rauland-Borg Corporation, 3535 W. Addison St., Chicago, Ill. 60618, attn: C. Dorwaldt.**

YOUNG SALES ENGINEER needed by reputable manufacturing firm of professional audio equipment. Some knowledge of application engineering. Opportunity for advancement. Little travel, offices in New York City. Write **Box 1A, db Magazine, 980 Old Country Road, Plainview, N. Y. 11803.**

EDUCATION

FCC LICENSE TRAINING by correspondence. G.I. Bill approved. Free brochure. Write: **Dept. E-4, Pathfinder School of Electronics, 1505 N. Western Avenue, Hollywood, California 90027.**

SERVICES

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

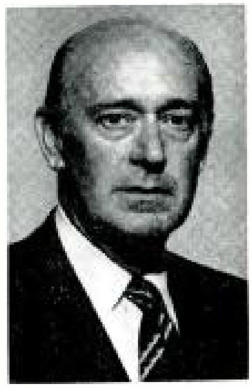
CUSTOM STYLUS — cartridge re-tipping, re-building, replacements. **International Audio Stylus Corp., 111-D Lake Ave., Tuckahoe, New York, 10707 (Telephone: (914) SP9-1297.**

People, Places, Happenings

●The **Audio Engineering Society** has made its call for papers for the 38th convention to be held at the Los Angeles Hilton the 4th through the 7th of May. If you wish to present a paper at this convention check with the AES at its New York headquarters at 60 East 42nd St., New York, N. Y. 10017. Address titles and abstracts directly to the California chairman, **Hugh S. Allen, Jr., Gotham Audio Corp.,** 1710 N. La Brea Ave., Hollywood, Calif. 90046.

●**Bolt Beranek and Newman** are sponsoring a symposium on organ and church acoustics to be held on Tuesday, February 17th at the main Sanctuary of North Shore Congregation Israel at 1185 Sheridan Road in Glencoe, Illinois. The seminar will place particular emphasis on achieving fine acoustical environments for worship music while assuring high speech intelligibility. Organ design, room-acoustic design, sound systems, and noise control will be discussed. Among the speakers will be **Nils Schweizer** and **Harold Spitznagel,** architects; **Lawrence I. Phelps** and **Walter Holtkamp, Jr.,** organ builders; and **Robert B. Newman** and other members of the BBN staffs. There will also be a recital by **Margaret McElwain Kempber,** Dean of the North Shore Chapter of the American. Guild of Organists, on the temple's Casavant organ. Particulars may be had from BBN's Downers Grove, Ill. office at 1740 Ogden Avenue. The telephone is (312) 969-6150.

●**Advent Corporation** of Cambridge, Mass., has announced that it has reached an agreement in principle with **Dolby Laboratories** of London, England for the manufacture and marketing of a consumer version of the Dolby audio noise reduction system. The Advent unit will be a single-band device designed specifically to reduce background hiss in home-type tape recording. It uses the new B-Parameter circuitry. Advent expects to market its version in April of this year at a retail price of under $300. According to the announcement by **Stanley Pressman** Advent's vice-president for marketing, the product will make dramatic improvements in the noise level of and over-all dynamic range capability of any good home-type tape recorder.



●**C/M Laboratories,** manufacturers of audio amplification equipment has a new team of operating officers and directors, after an agreement was signed with an investment group to provide capital for the expansion and development of the company into a major factor in the professional quality field. New president is **Maury Jungman,** formerly president of **Belmont Electric Co., Inc.,** a division of **Arc Industries.**

Wayne Chou, the founder with **Nick Morris** of the company and recent president will remain as vice-president of engineering. Mr. Morris remains as vice-president production.

G. T. Thalberg has been appointed vice-president of sales and marketing. He was formerly a co-founder of **Benjamin Electronic Sound Corp.** He has also held executive positions with other high-fidelity manufacturers.

Among the new board of directors are such well-known figures in electronics as **Ira Kamen, A. B. "Burt" Covey,** and **John R. Poppele.** Mr. Jungman and Mr. Chou also serve on the board along with members of the investment group (not named in the release).

●**Hammond Hunt** is the new general manager of **Jensen Manufacturing Division of The Muter Company.** According to the announcement by **Herbert J. Rowe,** president of Muter, Mr. Hunt will be responsible for the total operations of the division. Prior to this appointment, he held several executive positions with Jensen.

●The **NAB** continues to grow. New membership figures for the new year include two more a.m. stations to a total of 2214, 30 new f.m. stations to a total of 1209 and 6 new t.v. stations to a total of 544.

●**Trevor Kendall** has been named president of the **Island Magnetic Electronics Corp.** The company is currently producing a line of production high-speed duplicating and peripheral equipment for the magnetic tape cartridge and cassette industry. A high-speed common-capstan tape duplicator is now being scheduled for early delivery along with supporting production tailoring equipment.

●A new company, **Satellite Film Service** has been formed to handle film strips, film editing, music scoring, video tape recording, tape duplication, and sync resolving. The owners are two brothers, **Warren and Fred Berney.** Fred has had extensive recording and film experience most recently with Academy-McLarty Productions of Buffalo, N. Y. Warren was formerly with the Army Missile Command dealing with the procurement of missiles for the Redstone Arsenal. He is a graduate of Temple University with a BS in Business Administration. The new company is located in San Jose, California and intends to service the San Francisco Bay area.

●**Michael Thaler** is the new sales manager for **Dubbings Electronics, Inc. Paul C. Smith,** president of Dubbings said that Mr. Thaler had been selected because of "his impressive background in the production and marketing of cassettes." Before joining Dubbings, he was with **Plastic-Ware,** a Bronx. N. Y. manufacturer of cassettes and parts. He was with this company for nine years. After that, and just before joining Dubbings he was vice-president of sales for **Allison Radio,** of Hauppauge, a manufacturer of stereo tape cassettes and cartridges.

REVOX GUARANTEES THESE 4 PARTS FOR ONE YEAR.

THE REMAINING 842 PARTS ARE GUARANTEED FOR LIFE.

Until now, equipment guarantees were problematical. Some companies guaranteed their products for 90 days, some for a year or two. And one rather exceptional company went so far as to offer a five year guarantee on its speakers.

Now, the Revox Corporation becomes the first to offer a lifetime guarantee, on what is regarded by many as the most complex link in the high fidelity chain, the tape recorder.

There are 846 basic parts, exclusive of wiring and connectors in the Revox A77 tape recorder and every one of them, with the exception of the four pictured above is guaranteed for life.

This unprecedented offer becomes effective immediately and has been made retroactive to include the very first model A77 distributed by the Revox Corporation in the U.S.A.

Wouldn't it be nice if everyone could make the same offer?

Revox Corporation guarantees to the original purchaser of a Revox A77 tape recorder purchased from it in the U.S.A., except as to fuses and bulbs· 1) to replace without charge any part failing within twelve months after purchase; and 2) to provide a free replacement in exchange for any part thereafter failing except the record and playback heads, capstan and pressure roller. This guarantee shall be void if the purchase has not been registered with the Revox Corporation within the time specified in the card supplied the purchaser with the recorder, or if the recorder has been modified or altered by anyone other than the Revox Corporation or its authorized representatives, or if the recorder has been damaged by misuse or accident. Transportation charges are not included in this guarantee. There are no warranties or guarantees except those expressed herein.

REVOX delivers what all the rest only promise.

Revox Corporation, 212 Mineola Avenue, New York, N.Y. 11577 In Canada: Tri-Tel Associates, Ltd., Toronto, Canada

Circle 11 on Reader Service Card

Can a tough, little $49.20 microphone make the big time?

(A success story.)

A good little microphone, the E-V 635A. But just how good? After all, it was intended to replace the "workhorse" Model 635... a dynamic microphone that had earned its title under fire in studios and on remotes all around the world.

So when we introduced the 635A we put it to a critical test. A major recording studio was loaned a dozen 635A's and asked to test them. The engineers weren't told the price, but they got the idea that it was somewhere near $300.00.

They were so delighted with the sound that they cut several big band recordings with nothing but 635A's. "Best $300.00 microphone we've got." Then we told them the price. They were shocked. $49.20? They couldn't believe their ears.

Meanwhile, 635A's were beginning to appear in force on music and variety shows on every TV network. Mostly hand held. Something to do with ruggedness and good balance... but mostly because of the sound. Especially during ultra-close miking.

The rest is history. Radio and TV newsmen quickly adopted the 635A as their new "workhorse". After all, news only happens once, and the 635A was their best insurance against bad sound.

To most professional sound engineers, the E-V 635A is already an old friend, although it's only been around since 1965.

At the price, they can afford to use it almost everywhere. And they do. (We told you it was a success story.)

ELECTRO-VOICE, INC., Dept. 101BD
686 Cecil Street, Buchanan,
Michigan 49107

MODEL 635A Omnidirectional dynamic. Response 80—13,000 Hz. Output —55db. Balanced low impedance. Includes Model 310A stand clamp and lavalier neck cord. Fawn beige Micomatte finish.

• high fidelity speakers and systems • tuners, amplifiers, receivers • public address loudspeakers • microphones • phonograph needles and cartridges • space and defense electronics

Circle 12 on Reader Service Card

Electro-Voice®

A SUBSIDIARY OF GULTON INDUSTRIES, INC.