

SOUND TECHNIQUE

For Professional Recording • Broadcasting and Sound Reinforcement

Volume 1 • Number 2

September 1969



BILL RAVENTOS

Finally, another edition of **Sound Technique** is ready to be sent out! We're sorry it's been so long, but I know you'll find the information in this issue informative and interesting!

Over the past few months, activity has been increasing in the labs here at Electro-Voice. New products that will influence your professional lives are being given final tests in preparation for a busy season. A great deal of research is being done in several areas pertinent to recording and sound reinforcement as well as broadcasting. We are currently using several auditoriums to conduct tests that we hope will further substantiate some of our own theories. We are sure that our research will open up new areas in which research is needed. It is exciting to experiment in the field of audio, where much of what is now being done has never been done before!

Lou Burroughs and his professional microphone group are introducing several new products, and tests are being conducted to demonstrate their versatility and worth to the professional user. In addition, Lou has prepared some very special demonstrations and lectures involving placement of Line Radiator speakers, acoustic phase cancellation/coloration, and on-stage monitoring for performers. Many of the accepted practices for microphone placement in recording and sound reinforcement situations are being evaluated, with some shocking results. Lou will be writing articles in **Sound Technique** to share findings with you.

Another area in which we are doing a great deal of work has to do with speaker response in rooms. Much has been written about the acoustical problems of "typical" auditoriums, and the affect of poor room acoustics on sound systems. Our studies are showing us that the use of truly flat transducers can achieve very desirable house curves.

Unfortunately, many highly regarded sound reinforcement transducers are far from flat, and may themselves introduce

serious flaws in system response. (Flaws which are sometimes attributed to the room.) Faulty placement of speakers can also create response problems, and hinder good coverage. The addition of equalization into such systems may achieve the desired final results, but at greatly increased cost compared to the result achievable by proper selection and placement of flat, unfiltered, peak-free components. We mean in no way to intimate that proper component selection and placement is the total answer, or can be a solution to every condition, but it's certainly a good way to start a system design.

Of course there is no "end of the road" in research and investigation. Conclusions, drawn or implied, may offer practical solutions to problems, but their greatest benefit lies in the opening of new avenues of investigation. We hope that, as we progress and learn more about these fascinating areas, we can share our findings with you. Any suggestions you may have regarding areas of interest, or perhaps work that you have done in this field, will be received gratefully.

William A. Raventos
E-V Assistant Marketing Manager
Commercial Products

P.S.

At the spring, 1969 Los Angeles Convention of the Audio Engineering Society, Electro-Voice presented a paper at the Sound Reinforcement Session. The subject had to do with the correlation of anechoic speaker response (the type you see published in specification sheets for speakers, etc.) with the response of the same loudspeaker(s), in a room with given characteristics, etc. Although research in this area is by no means complete, the paper does offer some interesting thoughts that we feel would be of benefit to the sound installer, architect, or operating engineer.

Look for the paper to be printed in the Journal of the A.E.S.



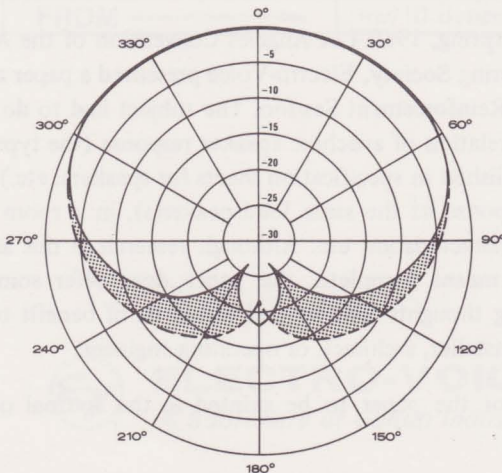
LOU BURROUGHS

As I travel across the country and talk with professionals at motion picture, television, and radio stations, recording studios and many other places, I am asked many questions regarding microphones. In each field there may be specific questions that are related to that particular field. But I find

that there are a few questions common to almost all fields of audio. One of those questions has to do with the advantages of the super-cardioid polar pattern as opposed to the true or pure cardioid polar pattern. Before giving the RE15, RE16, RE10, and RE11 their now-famed super-cardioid pattern, our Electro-Voice professional microphone engineers put a great deal of time and effort into the selection of the polar pattern.

THE BENEFITS OF A SUPER-CARDIOID POLAR PATTERN

In an effort to reduce the level of off-mic pickup and thus increase working distance, many variations in the shape of the polar curve were investigated. Beginning with pure cardioid, the pattern was varied until we arrived at the response shown below. The dotted line indicates a pure cardioid pattern, the solid line indicates the super-cardioid polar pattern of the RE15. The shaded area shows the amount of improvement (reduction of pickup at the rear region) gained by tailoring the polar pattern to be slightly bi-directional. By allowing a loss of 2 db at 180 degrees off-axis, much has been gained in the rest of the rear region. This has been accomplished with only a very slight narrowing of the angle of acceptance at the front. You will see that both patterns maintain essentially the same character-



istics to approximately 65 degrees off axis, or a 130 degree arc.

This super-cardioid polar response, in conjunction with smooth, peak-free axial response, has greatly increased the working distance available to the microphone. When comparing it with a typical well designed normal cardioid unit, you will find that a properly designed super-cardioid microphone can be used at from 30% to 50% greater working distance.

You will note from the polar graph that the greatest rejection of unwanted noise with the super-cardioid pattern is offered at approximately 135 to 150 degrees off axis. This is an advantage when you consider the angle at which a microphone is most often mounted on floor stands (tilting slightly upwards) or on a boom (tilting downwards).

Under conditions where high gain-before-feedback is desired, consider the relationship of the microphone to the location of the speakers. Usually the speakers are located above the microphone (as in a proscenium arch) or to the sides of the microphone (as at the sides of the stage). Most of the sound returning from the speakers to the microphone is returned from above the microphone or from the sides. Comparatively little of the sound returning is at exactly 180 degrees off axis. As a result, the super-cardioid pattern offers much greater rejection to general off-axis sound than does the pure cardioid. The same has been proven true in recording studios and broadcast stations with regard to ambient noise in the studio.

Of course, the final test to you, the user, should be a practical one; a comparison test. Here are two or three tests that you can easily perform in your own auditorium or studio:

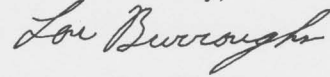
Under Sound Reinforcement Conditions: Set up a cardioid microphone alongside a super-cardioid RE15. Speak in a clear level voice equidistant from the microphones. Begin backing away, still speaking. Adjust for maximum gain-before-feedback. You will notice that the super-cardioid microphone will allow you to work at the same or greater distance than the cardioid microphone, while maintaining the same presence.

In Studio and Auditorium Conditions: Make the same set up - a cardioid microphone along side an RE15 super-cardioid. Set up a recorder to make comparison recordings. It's best if you can make a stereo recording with the cardioid mic on one channel and the super-cardioid mic on the other channel. This facilitates much easier A-B switching. Begin speaking into the microphones about one foot equidistant from them. Now gradually move back, still

speaking into the microphones, and have someone adjust the gain on the recording to keep the level approximately the same. After a few feet you will begin to notice that the room noise is coming up much faster on the cardioid unit than on the super-cardioid RE15. 30% greater working distance can be expected with the RE15.

Though there may be discussion as to which pattern is best under all conditions, we at Electro-Voice have found that the professional users' preference as well as the academic choice indicates the super-cardioid performs better than the "pure" cardioid. Try it yourself and see!

Sincerely,



Lou Burroughs
Vice President

Broadcast and Recording Equipment

"FOLDBACK" SYSTEM

Many interesting and unusual ideas come to us here at E-V from our friends in the field. Truly, where is there a better place to originate solutions to the industry's problems than in the field? From time to time we'll share with you some practical ideas that have been contributed by the "pros in the know" who are kind enough to share them with us.

In school assemblies, public speeches, community and civic events, etc., the situation is often encountered where the talker stands behind a podium and delivers a message to the crowd of people in front. (The standard set-up for one talker distributing information to a group.) Often the soundman has little or no trouble with his set-up in distributing good sound to almost all in the audience.

But what about the "V.I.P.'s" who are sitting on the stage, band-stand, or platform just behind the person speaking? Usually these important people are in the worst position for hearing. With little or no direct radiation from the talker who's back is to them, they depend on the sound system for hearing. The sound operator is often blamed for inadequate sound to the whole house simply because the man in charge of the program (who is sitting on the speaker's platform) can't hear.

If the sound operator turns up the gain to the house, he often gets feedback long before the people on the platform are satisfied. One method to combat the problem is to use directional horns of some sort, and focus them on the platform. This may or may not bring up the gain before feedback on the platform, but the *quality* of the sound will be less than that of the full-range system feeding the house.

In this issue, Edward Jones, Supervisor of Audio Operations at Brigham Young University, gives us a tip on how to solve an all-too-common problem in sound reinforcement. With

Ed's solution, the "V.I.P.'s" on the platform can have more than adequate level to hear comfortably without affecting the gain-before-feedback or the house level. Keep in mind, too, that the "foldback" sound is high quality.

We'll let Ed's own letter explain the system. The concert hall he describes is modern, has average-to-good acoustics, and seats approximately 1200.

BRIGHAM YOUNG UNIVERSITY
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ERNEST L. WILKINSON, PRESIDENT

ELECTRONIC MEDIA-AUDIO OPERATIONS
184 HMCB

June 27, 1969

William Raventos
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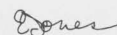
Dear Bill:

We have just found a solution to a long-standing problem which you may wish to discuss in "Sound Technique". During assemblies in our Concert Hall, we have had the problem of the officials sitting behind the podium not being able to hear. We have tried a number of methods of providing foldback (on stage monitoring), but we could not achieve the required volume without feedback.

Lou Bourroughs' lecture on out-of-phase transducers and Mel Draper's paper at this year's Aes convention inspired us to try this approach: we place two speakers (two hi-fi bookshelf type) back to back about five feet behind the podium. The speakers are wired out of phase which causes a dead spot about eighteen inches wide perpendicular to the speaker axis. The podium with two microphones (EV RE 16's) is then positioned in this dead area. We only use one of the microphones; the second microphone is for standby only. This foldback system provides 15 db more sound on the stage, to the delight of those sitting behind the podium.

We have also increased the gain before feedback in the entire house by using transducers with smoother response curves. We are using two Electro-Voice Sentry speakers in the house producing about 6 db more gain before feedback over our old system which used two well-known speakers with notable peaks.

Yours truly,



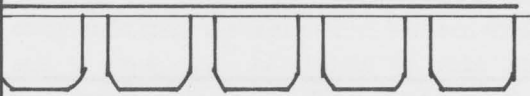
Edward S. Jones, Supervisor
Audio Operations Section

ESJ/md

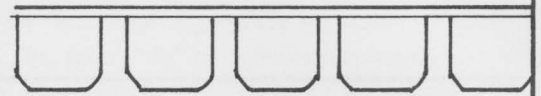
2 foldback speakers out of phase



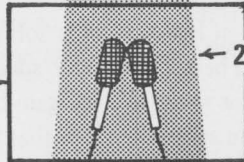
Chairs



Chairs



Podium



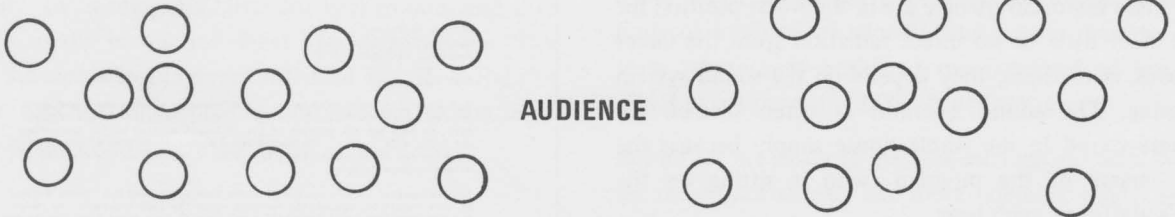
← 2 RE16's (in phase and close together)

CANCELLATION AREA

Overhead Sentry II
feeding house from
proscenium arch

Overhead Sentry II
feeding house from
proscenium arch

AUDIENCE





THE "E-V RULE"

Here at Electro-Voice, we are constantly receiving letters and telephone calls asking about the methods used for measuring microphone output. There are several common methods, and each manufacturer has his own "pet". While the methods of measuring microphone output may be useful in their own realm, it is certain that each of the methods is related to the others. While one manufacturer may rate his microphone output one way, others do it differently. The microphone engineering department here at Electro-Voice hopes that with the handy "E-V Rule," operating engineers in the industry will be able to more easily understand the relationships between these standards, and in some cases be enabled to make and correlate measurements themselves. By becoming familiar with the background material in this letter, and then following the included instructions, the "E-V Rule" should prove to be a valuable tool to both the interested novice and the vitally involved professional.

Much like the slide rule, the "E-V Rule" can be used for simple problems, or, with background and practice, it can be used to solve more complex and involved problems having to do with microphone level and the various systems of rating the same. Bob Herrold, the Electro-Voice Microphone Engineer who originated and designed the "E-V Rule," wrote me the following letter giving some fundamental background having to do with sound pressure level and how microphone output is measured. He wrote the letter with the goal of establishing certain ideas basic to the understanding of microphone level measurement, and then went on to basically describe the method we use at Electro-Voice for measuring microphone output level. Hopefully, by reading his letter for background, and then going on to the instruction sheet provided with the "E-V Rule," the operating engineer will find the "E-V Rule" a valuable tool for understanding and relating to the terminology of the industry—*W.A.R.*



BOB HERROLD

more easily understood and useful to the operating engineer.

Dear Bill:

Following are some thoughts that might add some light and background to the instruction sheet provided with the "E-V Rule." I have tried to keep this letter basic so that, after reading it and the instruction sheet the "E-V Rule" itself may be

What is Microphone Level?

We may say that level is generally expressed in "db" or decibels. The term "db" is meaningless unless we can compare it to something with which we are familiar. When we compare the term db, or decibel, to the loudness or volume of sound, an obvious point at which to start would be the faintest sound discernable to the human ear — commonly called the "threshold of hearing." Through a series of scientific experiments, the threshold of hearing was found to be $.0002$ dynes per cm^2 . That is a measurement of force per unit area, or pressure applied to a given area (your eardrum, a microphone diaphragm, or whatever). Generally, measurements made with this threshold of hearing as a reference are called sound pressure level (SPL) measurements.

Now that we have established a reference point (the threshold of hearing), we need to relate our term of decibels so we may work with the terminology of the industry. When we say that normal conversation is around 74 db in sound pressure level, we mean that normal conversation is 74 decibels (units of sound pressure level) *above* $.0002$ dynes per cm^2 . In other words, decibels are units of loudness or sound pressure level that appear on a scale whose zero begins at the threshold of hearing, or $.0002$ dynes per cm^2 ($.0002$ dynes/ cm^2).

Before we are ready to measure microphone level, however, we must have at least one more reference point. Since the most common method of measuring microphone level takes power output of the microphone into consideration, we need a power output reference point. The point has been chosen as 1 milliwatt (milliwatt equals 1/1000 of a watt).

Finally, an orientation having to do with sound pressure level reference points may be helpful. Our 74 db SPL (the conversation level previously mentioned) just happens to be equal to 1 dyne per cm^2 , or 1 microbar (1 microbar = 1 dyne/per cm^2). Since normal conversation is actually rather quiet, when measuring microphones we use a sound pressure level of 10 dynes/per cm^2 , which is 94 db of sound

pressure level. This additional level allows us to make accurate measurement of the microphone without interference from the usual low levels of ambient noise found in measuring chambers.

The actual measurement of the microphone output is then made with the microphone in a position on axis to a sound source at a distance of, for example, five feet. (Most directional or cardioid type microphones have an associated effect called "proximity effect". This is true in particular of "single-D" type dynamic or condenser microphones. The proximity effect, in reality, is a "bumping up" of the microphone response in the low frequencies when the sound source is very close to the microphone. Therefore, it is good practice to measure the level of these microphones at a minimum of two feet from the sound source). No matter what distance the microphone is located from the sound source, the sound pressure level, at the diaphragm of the microphone, would be 10 dynes/per cm² (94 db) in a free field, or non-echoing environment.

Under these conditions, the open circuit output voltage of the microphone is noted. A decade resistance is then placed across the microphone output and adjusted so the output voltage is halved, or decreased by a value of 6 db. The value of the microphone Z then must be equal to the decade resistance and is noted (see Figure 1).

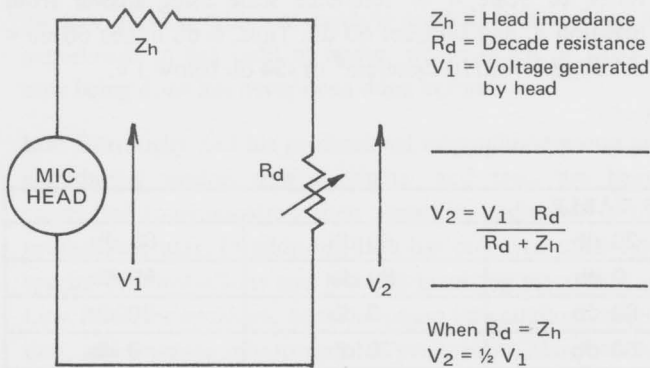


FIGURE 1

We now know the microphone Z and the open circuit voltage outputs. Only the level remains to be found. Now it is possible for us to use the "E-V Rule."

Sincerely,
Robert F. Herrold III
 Robert F. Herrold III
 Microphone Engineer

Instructions

The circular nomograph was created for those individuals who desire to know the relationship between various rated microphone impedances and/or systems of level determination. Obtain two pieces of rigid cardboard, such as 1/32" chipboard (frequently used on the back of note paper). Cut around the two nomograph discs, leaving about 1/4 inch of excess paper, and cement them to their respective backings. Trim excess. Punch holes where indicated by X and fasten together, preferably using an eyelet. A nail or thumb tack will also work.

The nomograph was set up so that scales 1 and 2 are always used in conjunction with scales 3 and 4. The pointer determines an answer on scale 5 or 6, e.g. rotate to known mic level on scale 6. Find known impedance on scale 2, and read open circuit output voltage opposite known impedance on scale 3.

Scales 3 and 4 are tied directly into the (voltage - db) reference scale in the following manner: The db portion of the reference scale is an expansion of the 0 to 20 db portion of scale 4, whereas the voltage portion is an expansion of the 1 to 10 mv range of scale 3. If, for example, one wanted to find the db value on scale 4 corresponding to two mv on scale 3, he could read 6 db at a glance, by referring to the (voltage - db) scale. This permits comparison in db between two different voltages.

Useful Information

.0002 bar = threshold of hearing = 0db sound pressure level (SPL).

SPL = sound pressure level.

1 μ bar = 1 dyne/cm² = 74db SPL.

2 μ bar = 74db + 6db = 80db SPL.

10 μ bars = 74db + 20db = 94db SPL.

If a mic is repositioned at twice the distance from a source of constant loudness, the level will drop 6db, and the output voltage will be halved. The reverse situation will obviously increase the level by 6db, and double the output voltage.

If a microphone has an open circuit voltage of 1mv in a free field of 10 dynes/cm², it is said to have an output in db of 0db, referenced to 1 mv/10 dynes/cm². If, under the same conditions, the mic had an output of 2mv, it would have an output in db of 6db above 1mv, etc. If a microphone has a power output of 1mw in a 10 dyne/cm² field its power output in db would be 0dbm, the "m" denoting a reference to 1mw. Therefore, a mic rated at -55db re 1 mw/10dynes/cm² (or -55dbm) would mean that the output was 55db below 1mw.

Referring to the conversion table: *e.g.* when converting from mv/10 dynes/cm², to v/dyne/cm² add (-80db). In other words, a mic whose output is 5db above 1mv re 1 mv/10 dynes/cm² would have an output of -75db below 1v re 1v/dyne/cm².

Example

A microphone spec sheet gives the mic impedance as 250 ohms and the level as -54 db re 1 mw/10 dynes/cm².

Question

Answer

1. Open circuit output in mv?

Rotate pointer to -54 on scale 6. Find 250 ohms on scale 2. Read answer opposite 250 ohms on scale 3. (Approximately 2 mv).

2. Open circuit EIA (RETMA) output in db?

EIA (RETMA) nominal impedance rating for 250 ohm actual is 150 ohms. Rotate 2 millivolts (answer to 1) on scale 3 to 150 ohms on scale 1. Read answer opposite pointer on scale 5 (-146.5 db EIA).

3. Open circuit output in db re 1 millivolt?

Refer to reference (voltage – db) scale. Two mv (answer is to question 1) is 6 db above one mv or two times the voltage, for comparison purposes. Scale 4 determines the range above/below 1 mv, *e.g.*, if voltage output was .2mv, the answer would be 6 db minus 20 db = -14 db below 1 mv.

4. Open circuit output in db re 1 volt/dyne/cm²?

Refer to Scale 4 or reference scale using answer from question 3, and subtract 80 db. Thus, 6 db minus 80 db = -74 db re 1 volt/dyne/cm², or -74 db below 1 volt.

5. Open circuit output in db re 1 volt/10 dyne/cm²?

Refer to Scale 4 or reference scale using answer from question 3, and subtract 60 db. Thus, 6 db minus 60 db = -54 db re 1 volt/10 dyne/cm² or -54 db below 1 v.

| CONVERSION TABLE | | | | | |
|------------------|-----------------------------|-----------------------------|-------------------------|---------------------------|------------------------|
| T O | mv/10 dynes/cm ² | 0 db | -20 db | 60 db | 80 db |
| | mv/dyne/cm ² | 20 db | 0 db | 80 db | 60 db |
| | v/10 dyne/cm ² | 60 db | -80 db | 0 db | -20 db |
| | v/dyne/cm ² | -80 db | -60 db | 20 db | 0 db |
| | FROM → | mv/10 dynes/cm ² | mv/dyne/cm ² | v/10 dyne/cm ² | v/dyne/cm ² |



ELECTRO-VOICE, INC. / Buchanan, Michigan

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