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THE SOUND ENGINEERING MAGAZINE

FEBRUARY 1979
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Coming Next Month

● Next month, international audio is our subject, as we visit recording studios in New York, Mexico and Japan. Then, it's off to Paris, with John Borwick as our guide through the incredible IRCAM (Institute for Research and Coordination of Acoustics and Music) at the *Centre Georges Pompidou.*

● We'll also have a brief report on the eleventh *Tonmeistertagung*, held late last year in Berlin, and a look at "an improved audio pipeline"—that is, National Public Radio's multi-channel satellite system.

All this, and a little bit more, in the March issue of **db**, The Sound Engineering Magazine.

About The Cover

● A view of Criteria Studios' Studio D, touted by the Miami-based operation as their "newest pride and joy." Equipment visible in the control room: an MCI automated 32-channel console and a custom-built tri-amped monitoring system, containing components from JBL and Cetec.

db

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db Letters

To The Editor:

Thanks for that segment of your December **db** article which dealt with microphone connectors. The habit of describing the "XLR" connector type with ITT Cannon's trade name tends to obscure the comparable products of other manufacturers, when it comes to purchasing time. This frustrates the other manufacturers and might cause the purchaser to spend more money, or get less quality, than would be the case if component comparability were more obvious.

The solution is to describe this connector design by its E.I.A. standard number—RS-297-A. In this context, "297-A" is both easy to use and precise.

Also, note that there are dozens of different military connector designs, so "Mil-Spec" is not a very useful descriptor. The design-standard number is the preferred description. For instance, Sescom uses Mil-C-5015 types ("Fifty-Fifteens") on its mike splitter and snake cables.

Again, thanks for the article.

John W. Schaefer
mpd Electronics Inc.
Long Island City, NY

WHY NOT A PEAK VU METER?

To The Editor:

The controversy of VU and PPM still goes on in the United States. While the standard VU meter is a good instrument for watching dissipated average power in loudspeakers, it is a known fact that VU meters cannot inform us about those occurring peak levels that are encountered in a running sound program, but only give us a rough estimation. Is the United States ready for PPM's? Well, this is still a matter of what we are used to.

I know that the PPM is a better instrument for broadcasting and recording use. It is just that we are so orientated to the VU meter that it seems difficult to conform. But if we had a PPM that employed the same

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letters (cont.)

type of scale as the standard VU meter, it may find its way around much faster. Let me clarify on this point. Most European and a few American meter manufacturers of PPM's have their PPM's with a black or charcoal background and use white markings for their scale calibrations.

We could call our PPM "VU_p Peak Meter" and use the same scale spacing as those that conform with the EBU standard for PPM's. The only difference is O $VU_p = 0.775$ volts rms and that we would employ the buff-colored VU meter background with black and red for scale calibrations.

This type of scale has been proven to be easier on eye fatigue. I, therefore, believe that if we employ a VU_p scale on our PPM's it would make for an easier transition to a PPM.

Our current standard on VU meters ANSI-C16.5 is undergoing revision, and this VU_p Peak Meter could be added to that standard as a different class of VU meter. I would greatly appreciate comments on this idea.

RON AJEMIAN
New York Telephone

TO THE EDITOR:

I've been reading with the greatest of interest the crabs, gripes, and "atta-boys" engineers have sent in to your letters column, and I can't pass the opportunity to have my say.

We at KRAV (Tulsa, Oklahoma) will match our air sound with any station anywhere. Since we're all cart, we do process very lightly in the recording process. This is done to overcome the problems which appear in some pressings, and to overcome the problems in the cart medium itself. The audio chain is unequalized, using only 3-5 dB maximum of limiting. Perhaps, with another format, we might find that a different processing set-up would give us a better sound.

The answer to good sound, though, is so simple it's corny. Our entire operation is committed to working together to achieve this. The owner, manager, program director, music director, news director, announcers, newspeople, and (hopefully) engineer are ordinary people, who talk to one another rather than about one another.

True, each of us has his or her field of specialization, but we try to know enough about the other guy's job to be able to appreciate his needs and problems. If we don't know, we ask, and then we listen to the answer. Just keep in mind the old adage "there are no

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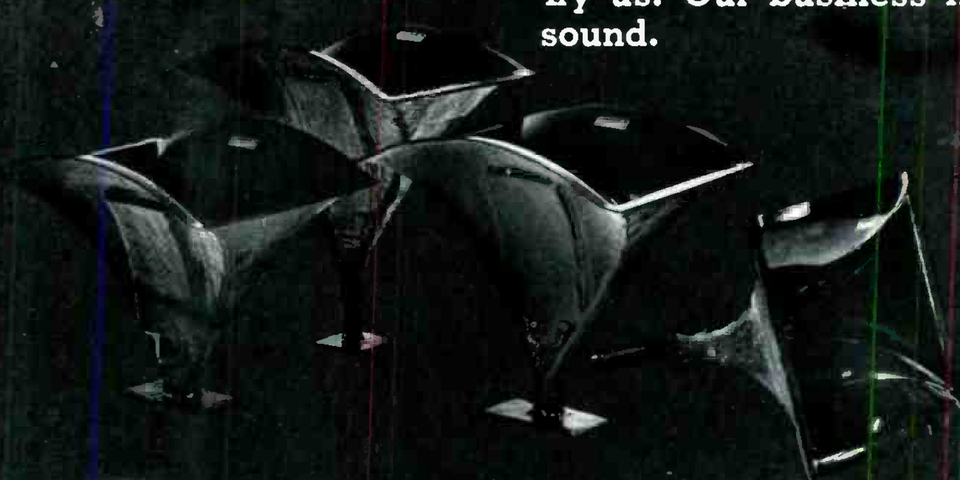

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stupid questions, only stupid answers." After a short time, you'll find that the questions become more and more intelligent.

The undertone of discontent in some of the letters leads me to believe that the writers are unhappy in their present positions. Having found myself in the same position recently, let me share with you the answer to the problem. If you're a reasonably competent engineer, you're in a seller's market. When it becomes obvious that you can't stay in the situation you're in, do what the jocks do: start sending resumes everywhere you can think of. Then, pick the offered situation which suits you best. You'll wind up with as good a job as mine, and it's excellent.

You're invited to drop in when you're in Tulsa, and see how we do it —we have no secrets. And, when any of us are in your town, we'll drop by and ask to see your station. We've gotten some of our best ideas this way, and you may too.

JOHNNY BRIDGES
Chief Engineer
KFMJ, Inc./KRAV Radio
Tulsa, Oklahoma

TO THE EDITOR:

I love to read your "Letters"; they give me a lot of pleasure. And the letters about audio quality remind me how right I was, some years ago, when I saw the future and didn't like what I saw: automatic master control switching; a good many more automatic devices for producing utmost coverage of broadcast area, regardless of quality of sound and all the rest of the things that Paul Dunn is griping about.

But not one of us knows very much about the crucial piece of equipment —human hearing. How, then, can we decide what is the right kind of sound to feed it? We simply do not know whether the ear is a straight line device or not. Some people, who *should* know, say that the ear is immensely more sensitive to sounds between 3,000 to 5,000 cps (or Hertz to you kids!). But no one knows how these sounds are processed in the brain, or whether the brain can routinely shut out what it does not want to hear. I incline to the latter idea. Let's not fight, fellows, about things we know nothing about.

About programming—I had a dream once, just after I had listened, for the first time, to my first crystal set in 1916. I thought I was listening to the music of the spheres when, I think, it was just KDKA playing an old phonograph record. I dreamt that radio was the perfect medium, for what was Samuel Johnson's ideal. The Great Cham said that writing should be both instructive and amusing. Radio, I dreamt, would be the great means of universal education the world was ready for. My dream, as you know, never came true. But that does not mean it can never happen. I tried again in 1952 or so, when Masaru Ibuka (retired chairman of Sony) came to visit me in New York to talk about tape recording. There was, as yet, no Sony Corporation. I asked him if he could design a tape playback that would be completely self-contained, and did not require any external power or batteries. He said he could, magnetically. If he could do it then, how much easier could it be accomplished now? I proposed to the United Nations that this could educate anyone in the world; they turned it down. So much for dreams.

Maybe, in another thousand years, we'll know enough about the brain and human hearing to know what to program and how to deliver the programs, both to amuse AND INSTRUCT, so that the general level of listeners will no longer be 11-year olds (as someone, maybe Bill Paley, once claimed), but persons educated to appreciate the music they hear and not the distortion accompanying it. There is enough for everyone to do, without quarreling about who is right and who is wrong; when none of us really knows much about what is really important.

JOEL TALL
Washington, D.C.

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db Calendar

FEBRUARY

27-3/1 **Eynergetic Audio Concepts Sound Engineering Seminar;** Quality Inn, Anaheim, Ca. For registration forms or information, contact: SYN-AUD-CON, P.O. Box 1134, Tustin, CA 92680. (714) 838-2288.

MARCH

13-16 **62nd AES Convention (Europe),** Brussels/Sheraton Hotel. Belgium. Herman A. O. Wilms, Zevenburderslaan 142/9, B-1190 Vorst, Brussels. Belgium.

Synergetic Audio Concepts Sound Engineering Seminars:

13-15 The San Francisco Airport Hilton, San Francisco Airport. CA.

20-22 Hilton Inn. Salt Lake City. Utah. For registration forms or information on either seminar. contact: SYN-AUD-CON. P.O. Box 1134. Tustin. CA 92680. (714) 838-2288.

25-28 **National Association of Broadcasters (NAB) Convention.** Dallas Convention Center. Dallas. Texas. For more information contact: Dallas Convention & Visitor's Bureau. Dallas Chamber of Commerce. 1507 Pacific Avenue. Dallas. Tex. 75021 (214) 651-1020.

APRIL

2-5 **First Annual Architectural Acoustics Exposition and Seminar.** Hyatt Regency O'Hara. Chicago, Ill. Contact: Wayne V. Montone. Executive Director. 464 Armour Circle, N.E.. Atlanta. Georgia 30324.

10-12 **Synergetic Audio Concepts Sound Engineering Seminar:** Sheraton Harbor Island. San Diego, CA. For registration forms or information. contact: SYN-AUD-CON. P.O. Box 1134. Tustin. CA 92680. (714) 838-2288.

23-26 **Audio-Visual '79.** Wembley Conference Centre. London. Contact: British Information Services, 845 Third Avenue. New York. NY 10022. (212) 752-8400.

MAY

12 **1979 Midwest Acoustics Conference. Topic:** Digital Technology: Impact on Recorded Sound. Norris Center. Northwestern University. Contact: William R. Bevan, Shure Bros.. Inc., 222 Hartrey Ave., Evanston, Illinois 60204. (312) 866-2364.

15-18 **63rd AES Convention (Los Angeles),** Los Angeles Hilton, California; Chairman will be Martin Polon, Director, Audio Visual, U.C.L.A., C.A.S.O., Rice Hall 130, 405 Hilgard, Los Angeles, Calif. 90024, (213) 825-8981.

22-24 **Synergetic Audio Concepts Sound Engineering Seminar:** Sheraton-Universal Hotel, No. Hollywood. CA. SYN-AUD-CON. P.O. Box 1134, Tustin. CA 92680. (714) 838-2288.

JUNE

3-6 **13th Annual Summer Consumer Electronics Show.** McCormick Place. McCormick Inn. and the Pick Congress Hotel. Chicago. Illinois.

20-22 **APRS '79—Annual Exh. of Professional Recording Equipment.** Connaught Rooms. London.

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PATRICK S. FINNEGAN

db Broadcast Sound

Telco Line Problems

• Transmission of program audio over local Telco lines can meet with problems that reduce quality of transmission, or in some cases the program is lost altogether. Line problems are not within the jurisdiction of station personnel to correct, nor always within their competence either. Telco personnel are generally very cooperative in correcting line problems which are brought to their attention. An important fact to remember is that Telco does not monitor program transmissions on local loops. It is essential, therefore, when problems do develop, that station personnel recognize that problems exist, and alert Telco immediately so that corrective measures can be taken.

INITIAL SET-UP

When a local line has been ordered for a remote broadcast, the installation will be made some time in advance of program time—often days ahead. Once the installer has completed his work at the remote site, he should check through to the studio and with station personnel at the studio end. This check through is important since the circuit will route through the local Telco testboard and must be cross-connected there. While the installer is still at the remote site, it is a good practice to make some measurements on the circuit yourself. These are essentially resistance measurements and perhaps a listening test. Save this data for future reference should troubles develop on that circuit.

The first measurement to make is the resistance of the circuit itself. The installer should have placed a 10k resistor across the terminals, so in essence you are measuring the total series resistance of the two wires and that resistor. Next, measure from each side of the line to ground. Since this is a balanced circuit, the resistance should be infinity. When first placing the test leads on the circuit and ground, there will be a quick kick of the ohmmeter, this is normal. It is the battery charging up the capacity to ground of the circuit. The reading should only be momentary and settle back to infinity.

Patch the line into the console to make a relative listening test for noise on the circuit. Set the console fader to the usual position for local remotes and observe the VU meter, and listen to the monitor. With no movement of the meter, the noise is at least 20 dB below normal program. Then open the fader wide open. You can estimate any noise that results by the fader steps which are usually about 2 dB per step.

The installer is really not needed to help you make these tests, but if any serious problems show up you can report them immediately. He knows who to contact at the testboard, and they may even find another pair to use. But if another pair is going to be used, then make the same measurements on that pair.

NO CIRCUIT

A relatively common problem is that there is *no circuit* at program time. Many things can happen to the circuit once it has been installed and checked out. An open circuit prevents the announcer from calling in over the line and the console over-ride. He must then spend time finding a telephone, calling Telco and getting someone out to correct the problem. However, enough time can be lost that part or, worse yet, all of the program may be lost. The control room operator, of course, can detect such problems by checking out the circuit some time in advance of the program air time—well enough in advance to get the problem corrected before air time. This is where those initial measurements come in handy. If the measurements are still the same, then he can reasonably expect the line to be operating properly.

The most common cause of an open circuit occurs at the Telco testboard itself! Someone may misread the line orders, and thinking the circuit is no longer active—pull it down. (The author had this happen one time right in the middle of a 3 hour remote broadcast!) Yet another cause of open circuits is damage to cables by digging equipment or cranes on a construction project. This doesn't happen too often, fortunately, but it does happen and can disrupt telephone service over a large area—including broadcast circuits. Should the station's News Department have information that such an event has occurred in your town, check out all your permanent circuits, as well as those temporary circuits not yet used. Since

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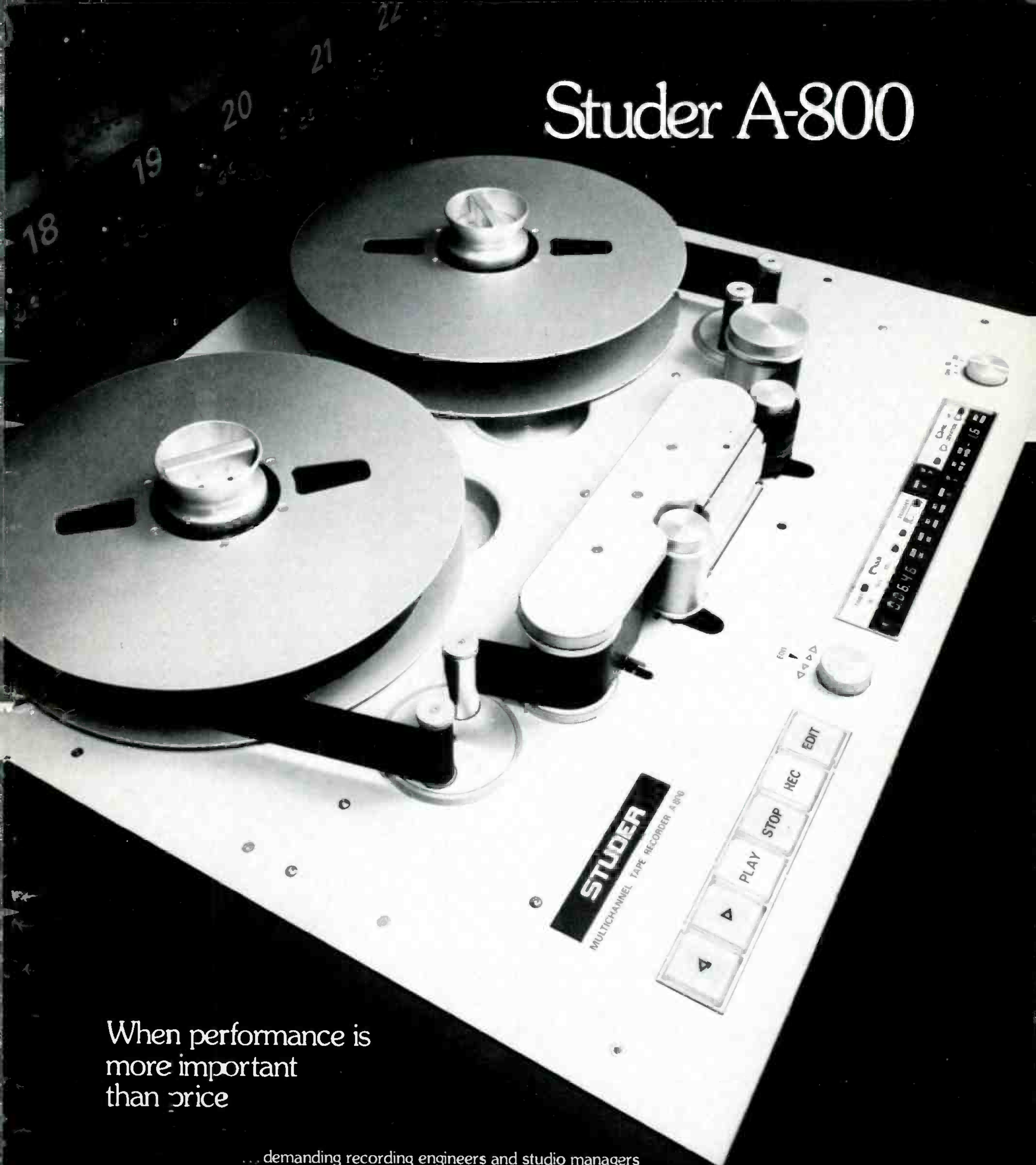

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there is no way of knowing the actual routing of any of your circuits. it is best to check them all out. Again, using that original set of measurements, taken on each circuit, it is easy to determine whether the circuit in question was in that damaged cable. If so, report these immediately to Telco, and give them the day and time of its next use. This gives them a time frame in which to work at the correction. But don't leave it at that. Shortly before the air time for the circuit, make the measurements again to be sure it has been properly restored.

POOR CIRCUIT

The most common situation is a *poor* circuit rather than an open circuit. Unless station personnel are alert and get the problems corrected early, very poor quality program audio can result. (Here we are not referring to the normal poor bandpass of a "good" line, but things which happen to make a "good" line go bad.) This is yet another reason why regular checks should be made on all permanent circuits on a regular basis, or at least some time well ahead of the use of a circuit.

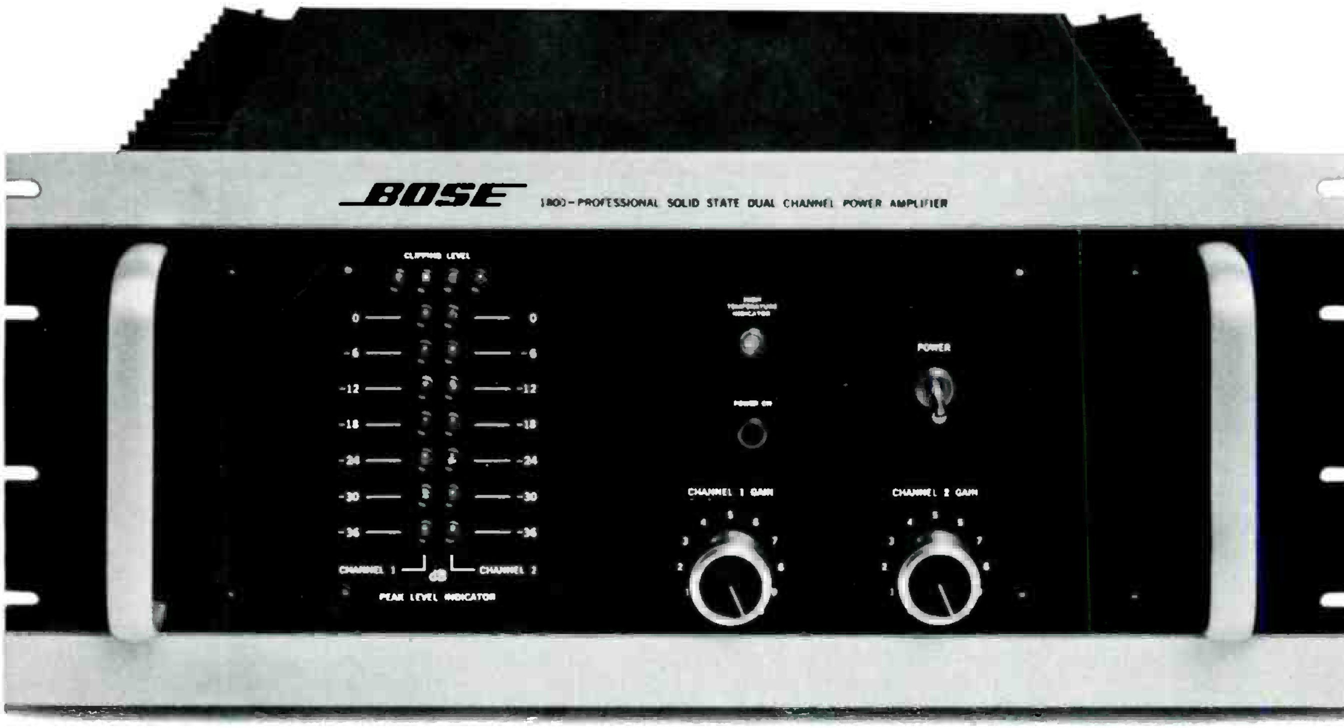
Assume you are checking your permanent circuits and measure one where the resistance across the circuit is very high (not infinity). This can be caused by a series loading coil (if still in the circuit) that is faulty, a poor jack, or connection in the testboard or anywhere along the circuit. Signal loss will be high and the audio quality may be very poor on the circuit under those conditions. Report such a condition immediately.

Instead of a high resistance across the circuit, the ohmmeter may indicate a *low* resistance of a couple hundred ohms or less. This can be a *normal* situation, or a shorted circuit. It is normal if there is an amplifier connected across the remote end of the line, or if the console is connected to the line at the studio. The transformer in those units will have a low value d.c. resistance and the line will appear shorted—but it is not. To get a true reading on the line itself, the amplifier must be removed. And always unpatch the console before attempting to measure the circuit. It may not be practical to go out and remove the amplifier in many cases—especially in various churches with a permanent amplifier arrangement—however the measurements can still be of use if measurements were carefully made, at the time the equipment was first installed. A shorted cable or moisture will still effect these indications to alert the observant operator that there is a problem. Without equipment on the line, a low reading indicated on the Ohmmeter means the line is shorted. Report this immediately to Telco for correction. The line will be unusable for broadcast.

STILL MORE

A broadcast loop is a balanced circuit, which means both sides of the circuit are above ground. It is important to measure *each side* of the circuit to ground. In making this measurement, let's assume you measure a *very high* resistance to ground (not infinity), on one or both sides of the circuit. The resistance may also be varying or intermittent. Most likely, the cause is a defective *heat coil.* These coils are lightning (and power line) protective devices to shunt transient spikes or large voltages to ground, and are placed at various intervals along the circuit. Report such a resistance reading to Telco immediately for correction. The circuit is usuable to some degree, but look for an increase in noise and crosstalk.

A high resistance reading to ground may not be resistance at all, it may be voltage. When a high resistance is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indicated, switch the meter over to measure a.c. or d.c. voltages. Long cable runs often pick up induced a.c. voltage from the power lines on the same poles. This will measure approximately 1½ to 3 volts. There isn't much Telco can do about this problem without major construction work, but that a.c. voltage can create hum in the program audio. The hum can often be reduced or eliminated by the use of a repeat coil at the studio end.

On the other hand, you may measure a *dead short* from one side of the line to ground. A typical cause here is the small carbon blocks used as lightning protection devices. A strong surge may have burned through one of the blocks and it now has a direct connection to ground. If these are mounted at the Telco terminal box in the studio, pull out the carbon block. If it is shorted, the short will disappear from the ohmmeter. But if this does not correct the problem, the short is elsewhere along the route. Report the problem immediately as the circuit is now definitely unbalanced. Hum and crosstalk may become so high as to make the line unusable. As an emergency measure, try running the circuit through a repeat coil for isolation. On some broadcasts such as a basketball game, the crowd noise is very high and this may override the line noises enough to at least save the broadcast—if not the quality.

NETWORKS

While the previous discussions have centered on local Telco loops for remote broadcasts, stations have a variety of Telco circuits in use. One such circuit in many stations is the connection to a National Radio Network. Besides the local loop to the Toll Test section of the Telephone Company, there will be a cross-connection from there to the long lines division of A.T.&T. The long lines section of the Network can be either wire circuits or multiplexed microwave carriers. There is some degree of monitoring along the Network which can catch problems quickly, but there are other problems that can develop unnoticed.

Most of these Networks send along response and other test signals at various times and on a scheduled basis. The local station should make use of the tests to keep a check on the quality of the circuit. It is not necessary to set up test equipment to take the response runs on a regular basis. Simply switch up the Net on the Console and use the VU meter. Set the fader to a convenient place around Zero VU on the set level tone, and then note and write down the difference in signal level of each of the tones sent in the test. These should all be within 2 or 3 db from the reference. Watch for roll-offs, sharp peaks or notches in the response curve. These are normally equalized circuits, but one may have been accidentally misadjusted somewhere, or one inserted that doesn't belong in the lineup. Report such deviations from normal to the Toll Test section of Telco. And if the fault isn't corrected over a practical period of time, you may have to complain further. The real problem may be that the fault is only in your "leg" of the Network. If, after several complaints, it doesn't get corrected, go to the Network itself. They can often get results the local station can't.

RECAP

Local loops can develop problems. To maintain consistant line quality as well as save programs from being lost, check the lines on a regular basis. Compare new test results with the original resistance measurements which were made at the time of installation. If you have a national Network, make use of the test signals to keep tabs on line quality. ■

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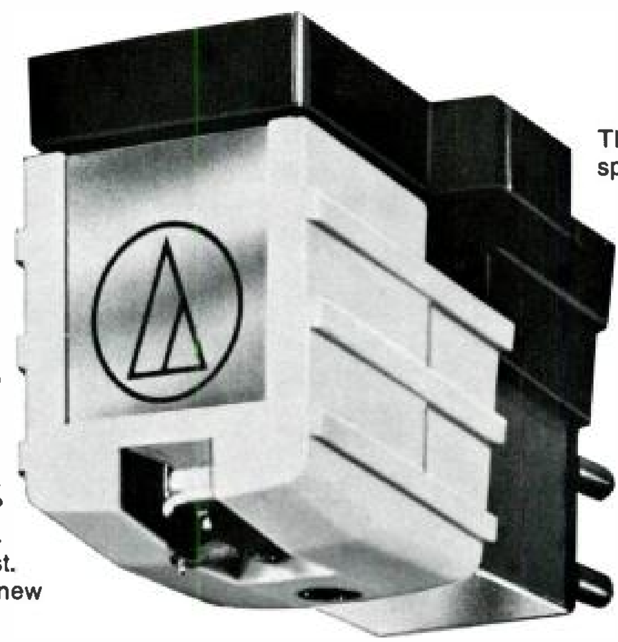

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NORMAN S. CROWHURST

db Theory & Practice

The Role of Semantics

• As my wife and I are of English origin, we are well aware of differences between our former and present usage of that language: English and American. A friend once said, "The English and the American are two peoples separated by a common language." Perhaps the best illustration of this aspect occurs in the usage of the word "homely."

In England, to call a woman "homely" is a pretty high compliment. Perhaps the best American equivalent to the English usage would be "affable." It means easy to like, get along with. It describes someone who quickly makes you feel at home, comfortable, which is surely a desirable trait. Over there, it has nothing whatever to do with looks.

Getting nearer to our kind of business, there are spanners and wrenches, boot and trunk, bonnet and hood, valves and tubes, and a whole lot more translations where the English and the American use different words for the same object.

But that is not what this column will be about. However, it may relate to it in a way: people sometimes read what I write and assume that, because I am (or was) English, I express myself differently, and therefore I do not mean what I appear to say, although actually I am using English in the American way. So in that sense, the above obstacle may remain, to an extent.

In most instances, exactly the same misunderstanding could occur on either side of the Atlantic. It is a basic misunderstanding, not a difference in

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meaning. Let me illustrate with a few examples, to make my point.

RESISTANCE OR IMPEDANCE?

We have frequently gone over the use of technical terms like input impedance and output impedance. For many perhaps, the term impedance is difficult to distinguish from resistance, since both terms determine a relationship between voltage and current. But at the input or output of an amplifier, for example, what is it that determines this relationship?

Put simply, it is the relationship between voltage and current. You can measure voltage between the terminals, and you can measure current between one side and the other. Now, what determines this relationship? At the input, it is the impedance inside the amplifier. At the output, it is the load impedance to which the amplifier is connected.

If you disconnect, in either case, the voltage will still be there, possibly increased by removal of the load, on the input side of the connection. But it will have vanished from the other side, because you have disconnected the two: now no current flows.

However, as generally used in specifications, "output impedance" signifies the impedance to which the amplifier *should be connected,* rather than any impedance inside the amplifier itself.

The complication in understanding comes because, whenever you make such a connection, you are connecting two impedances together. At the input, you connect the impedance of a microphone, for example, to the internal input impedance of the amplifier. At the output, you connect the output of the amplifier to the impedance of the loudspeaker—or whatever.

Maximum power transfer theory says that these two impedances, in each case, should match, or be the same. We have discussed in other columns why that is not always the best arrangement. In fact, the microphone's internal impedance is usually somewhat lower than the amplifier's input impedance. But if it is specified as a "50-ohm mike," that usually means it should be connected to an amplifier whose input impedance is rated as 50 ohms. The mike's own actual impedance may be, say 25 ohms.

At the output end, the difference is usually even more marked. If the impedance is given as 8 ohms, that should be the nominal impedance of the loudspeaker. It is also the designation used for the amplifier output, for that reason. But the amplifier's own internal output impedance will usually be only a fraction of an ohm.

A few paragraphs back, I started with the words, "Put simply." Modern educators would write that, "Put simplistically." Why all those extra syllables? That is another instance of special usage, or in this case, a "sophisticated" development of words. Either way, it implies there may be more to it than first meets the eye.

These complexities are understood, to those who understand the technology involved. In our case, this means they have studied, or are educated in audio technology. A person who can explain all this to you, will sometimes be called "erudite," especially if the person making that description happens to be an "educator."

Erudition, the noun form of erudite, means "book learning." Someone who is well-read is properly termed erudite. Unfortunately, the word often implies more than that: from the way it is said, you would think that the person so described knows a lot. Perhaps in fact, he knows very little, but can

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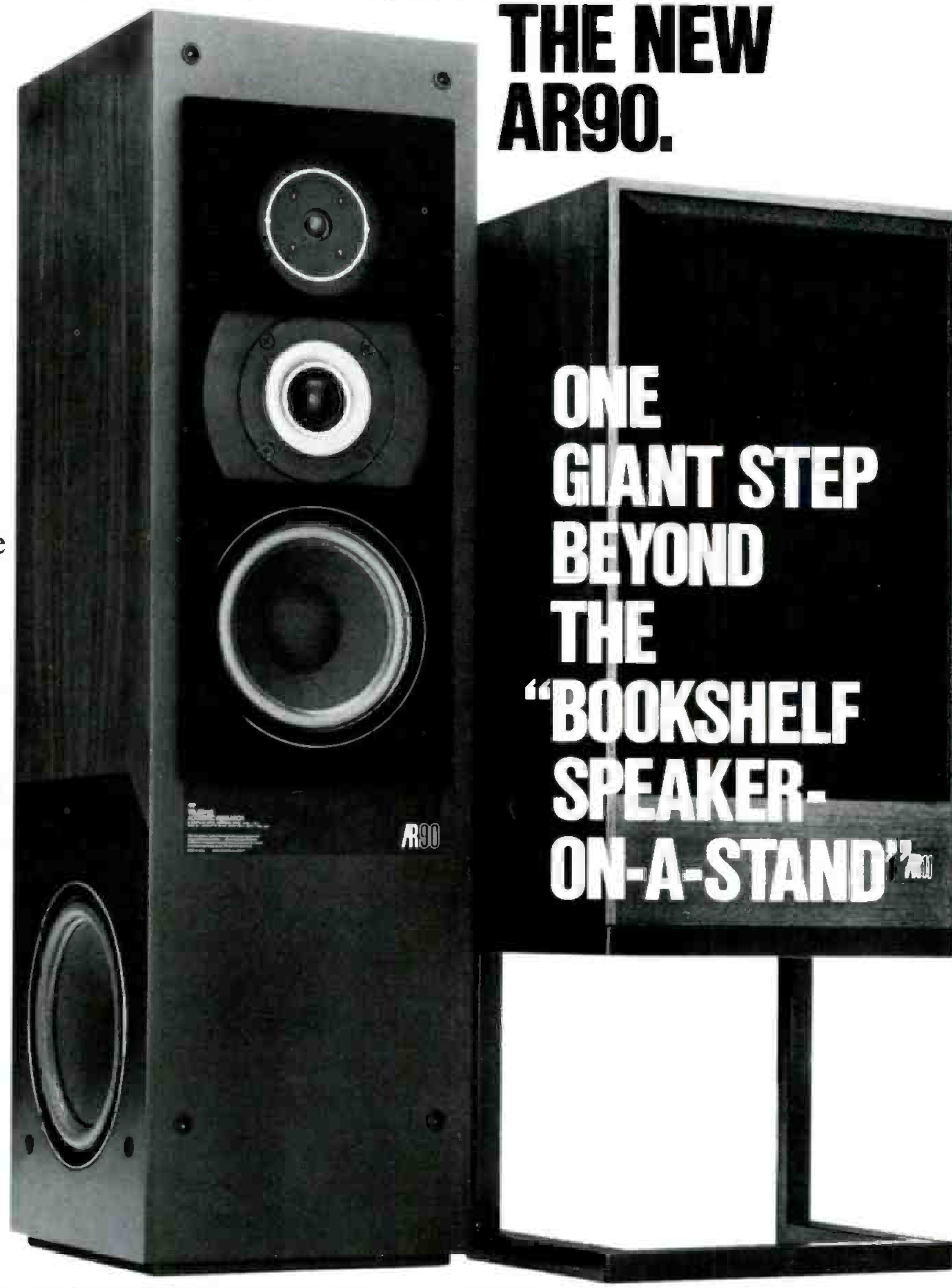

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theory and practice (cont.)

quote a lot of what he has previously read.

This further relates to the distinction between education and instruction. Today's "educators," which unfortunately has come to mean those employed in the business of schooling. commonly misnamed education, equate the two terms: to them, instruction *is* education. The concept seems to be, "I have told you; therefore, if you remember what I told you, you know."

But what if I tell you black is white? Do you now know that black is white? You may very well remember that I said black is white, because you probably thought, "That guy must be nuts! Anybody knows black is not white." If you thought my statement was instruction, your reaction was that it cannot be true. But it got you thinking, didn't it? Which means it was a step in your education.

In this instance, I just said something obviously wrong, to make a point. But in more constructive circumstances, it might only appear to be wrong, until your thinking about it brought the understanding, based on all the pertinent facts, that it is true.

To summarize the distinction between instruction and education: instruction is accepting, and repeating where appropriate, what the teacher or the textbook says, as true; education, on the other hand, is applying yourself to understand what it is all about, so you know, and can reconstruct or apply what you know.

NO. 2 PENCIL STRIKES AGAIN

This brings me to the way my column in the November issue was edited. It failed to say what I wrote it to say, because the editor, innocently enough I am sure, failed to understand my intent.

Early in that column, my manuscript referred to some documentation I put together while at Tannoy, on what later became known as systems design methods. It was used, I went on to describe, to enable newly engaged engineers to apply the method to any new system they might be assigned to design.

The editor apparently thought this documentation was a "manual," so substituted that word. A *manual* is a handbook that tells you what to do, in one, two, three fashion. *Documentation* is a setting of information on paper, or putting it on record. The information can be anything that thus needs recording, and it could well be in the form of a handbook, or manual.

But in this case it was not. The editor could scarcely be expected to know the nature of that documentation, because he had not seen it, and I did not describe it in any detail. It was hardly appropriate to try to do that in the limited space of a monthly column.

Later in the column, I referred to my now-out-of-print book, "Taking the Mysticism from Mathematics." That is not a manual, or handbook. Nor is it a textbook. But having concluded my "documentation" was a "manual," my reference to this book seemed irrelevant, so he blue-penciled it out.

The very point of that column was that a caller, asking to buy half a dozen copies if I had them, drew attention to something I myself had not realized: that the same method that proved successful in systems design, contained in the documentation I prepared at Tannoy, I applied in the book to education in general, and to mathematics in particular.

He would not have known this, had not the documentation referred to, which I had completely forgotten by that time, been preserved in secret government files to which he and the other people who wanted copies of the

theory and practice (cont.)

book, had access. My purpose in last November's column was to draw attention to the importance and universality of this concept of systems design.

As the documentation was there recorded, it could not be put into a manual. Last November's column, as published, did call attention to some manual-type things that do not work (at least as well), which in those days (back in the '30s) we described as "the brute force method." Nowadays that would be described as "doing it by the book"—or the manual.

By editing out the reference to the book that applied the same basic method in a completely different context, November's column omitted the positive side of what I intended it to say. Perhaps the same distinction can be shown by reference to two more words, one of which appeared in that column as printed: application and utilization.

Application means taking something and applying it, in perhaps one, two, three fashion, as you might from a manual. Utilization means putting something to use. Utilization would more appropriately fit with the form of documentation I was talking about. But the word the column printed was application.

The book, "Taking the Mysticism from Mathematics" showed quite clearly how hang-ups in math occur—and by application, that could include other subjects—and from that, how to remove your own hang-ups and become proficient at learning that subject. I have a file full of testimony that it works.

PROBLEM-SOLVING

Fairly obviously, such a book is not a manual: you could call it a problem-solving procedure. First you find out why you have a hang-up. But then you have not necessarily solved that hang-up, unless you also remove the cause. And part of the cause is often just the thing we have been discussing here: that you want simple, a-b-c-type answers, without bothering to "put your brain in gear."

The same comment goes for true engineering, in any context. It is from such constructive efforts, based on true ingenuity—which can never be put into a manual—that progress results. As an earlier column of mine once said, Edison did not invent the light bulb by going down to the corner newsstand and buying a paperback on "How to invent the light bulb." ■

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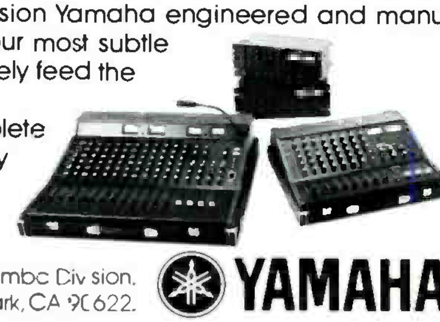

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YAMAHA

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Editorial

NEXT TIME some well-intentioned friend tells you all about his favorite speaker—you know, the one that tells him what's *really* on the tape—you might want to hand him this issue of **db**. Better yet, slap him across the face with it. For such people need to be awakened quickly, and maybe we can help, one way or another.

Most of us know that the loudspeaker is merely a small part of the audio reproduction system. Certainly, its a very important part (just try listening without one sometime), but there are an incredible number of complex variables in between speaker and listener, and each of these plays an important role in creating that "true" sound your friend thinks is coming out of the speaker itself.

Prove it by moving the speaker into a corner (or out of one). Turn the volume up (or down). Move the speaker into another room. If you've got the patience for all this, and need convincing yourself. you'll find that the same speaker has different "truths," depending upon its environment, and yours. Therefore, which one is the *real* truth? We'll leave that one unanswered here, for truth is in the ear of the listener, and not necessarily in editorials. Or at least not in this one.

On the off-chance that you are currently in the presence of a "true" loudspeaker, how will you recognize it for what it is? And, what will it sound like in someone else's control room? Will it still be true? As you can sec, we're quickly getting irto trouble here, but there's no need to worry too much: the ultimate loudspeaker is not apt to be built by the time we go to press. In fact, it's unlikely that the "experts" will even get around to agreeing on what goes into the perfect system.

For example, take the matter of waveform fidelity. You demand it of your amplifier, but what about the square wave response (i.e., waveform fidelity) of your favorite monitor system? One designer states ". . . the human ear is unable to distinguish time domain errors of the kind introduced hy loudspeakers during music reproduction." But others have measured the amount of time delay that *is* perceptible to the listener, and present figures that are greater than the time delay found in many professional monitor systems.

And then, there's the matter of absolute phase, or should we say, polarity? Everyone knows that the two speakers in a stereo monitor system should be wired "in phase" (with each other, that is). Assuming they are, would it matter if we reversed the polarity of *both* speaker cables?

Before leaping to conclusions, think about some percussive sound that reaches your eardrum, unassisted by any electronics. An instantaneous increase in air pressure is followed by a decrease, and you hear sound. But what if a polarity reversal inverted that sequence? On paper, this would be an unnatural phenomenon, but over a control room monitor (or at home), could you tell the difference? Some say yes, others, no.

Even those who already realize that loudspeakers are no more truth tellers than microphones are, might like to know a bit more about what makes them tick (or woof, squawk, tweet, howl, etc.). And so, we start off with the basics—that is, the **Anatomy of a Loudspeaker.** Author Steve Boak guides us through that collection of baskets, gaskets, spiders, coils and magnets that somehow or other go together to produce the ultimate sound.

With the consumer audio industry supplying evermore spectacular speakers for your hi-fi set at home. Floyd E. Toole raises an interesting thought. That is: **High Fidelity in the Control Room—Why Not?** Well, why not? Is there a law against it? You might think so, for there seems to be a line drawn between studio monitors and "hi-fi" speakers. Yet, Mr. Toole feels there are really no insurmountable technical problems to moving hi-fi into the control room. But, will engineers accept such a thing? And, what about you?

Next, John Eargle takes a look at **Requirements for Studio Monitoring,** giving close attention to the differences between two-, three- and four-way designs. And, as if confirming that there is indeed a difference between "hi-fi" and "pro" sound, we discover at least one parameter where the hi-fi speaker may measure a bit better than its professional counterpart.

We'll say no more about that particular measurement here, but we will ask—is time delay distortion really significant in loudspeakers? As we suggested earlier, the answer is a definite yes—or a definite no—depending on what you read. Almon Clegg steers a reasonable course down the middle of the controversy by reminding us that waveform fidelity in loudspeakers is certainly not harmful to audio. His article, **It's About Time,** is the first in what will probably be a series on the subject, as we hear from various authors who support, or reject, the concept.

When was the last time you blew out a loudspeaker? Or heard a strange noise as the voice coil rubbed up against the magnet? Maybe its time to investigate ferrofluids. Dana Hathaway has been using them for several years, and he offers us some insight into the theory behind **The Use of Ferrofluid in Moving-Coil Loudspeakers.**

Next an **Application Note on Speaker Protection,** in which Dave Rosen offers a few practical points on applying ferrofluids to your own voice coils.

And then, Dan Queen's feature on **Impedance Matching** guides us through the simple act of connecting a speaker to an amplifier. Of course, there's nothing to it. Or is there? After reading the article. you should know a bit more about the relationships that should—and fortunately, usually do—exist between amplifier and speaker.

If you've absorbed all of the above, you may finally be prepared to tackle the complexities of **The Electro-Voice Rearaxial Softspeaker.** Unfortunately, we don't seem to have the author's name in our files, so we cannot give him the recognition he so richly deserves. As a matter of fact, we seem to recall that this offering came to us in a plain brown envelope. addressed to "occupant." If you have trouble interpreting the specs, we suggest you read them upside down, or perhaps backwards, or maybe through a mirror.

Once you understand how the softspeaker works, it's time to conclude, with Michael P. Rogalski's explanation of **Murphy's Theorem of Systematic Catastrophe Applied to P.A. Work.** Once again, we are indebted to Murphy for reminding us that, if anything can go wrong, it will. ■

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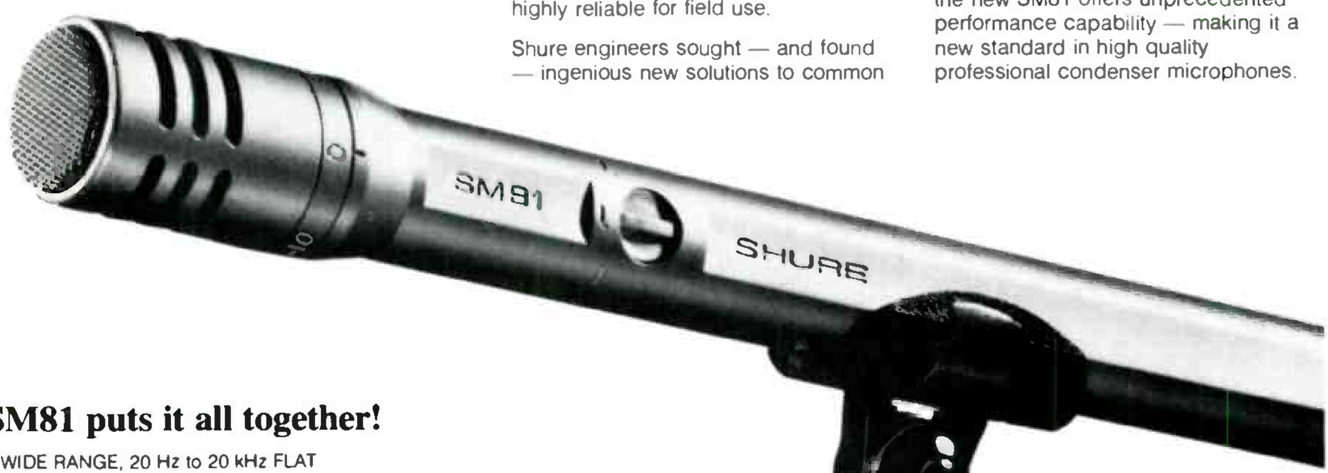

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STEPHEN BOAK

Anatomy of a Loudspeaker

A basic look at the complex design of loudspeakers.

A LOUDSPEAKER, like a microphone or a phonograph cartridge, is a transducer, or converter. It is a device that converts electrical energy into mechanical, and then acoustical, energy. Though there are many types, any speaker serves but one purpose; connecting the electrical amplifier circuits and the listener's ears.

THE ROLE OF THE LOUDSPEAKER IN AN AUDIO SYSTEM

The loudspeaker is among the simplest of audio components; yet, its design can be enormously complex. It is simple because it has relatively few parts (as compared to a tape recorder, for example) and even fewer moving parts. With very few exceptions, it is virtually impossible to alter or change one part of a loudspeaker without noticeably affecting its performance. Also, the speaker system is the only audio component whose performance is radically influenced by its environment, which is usually beyond the control of the speaker designer. Further, it is the only component in direct contact with the ear, which is far from consistent. To make matters more complicated, there are no accepted specifications which will describe the way a speaker should perform. Here, we shall try to analyze some of the variables that affect loudspeakers and loudspeaker system performance.

Before we plunge into the simple theory behind speaker operation, it would be helpful to have a small section devoted to the parts that make up a speaker. These parts are usually called "piece parts." In the average speaker, there are some twelve to fifteen piece parts, four or five of which move.

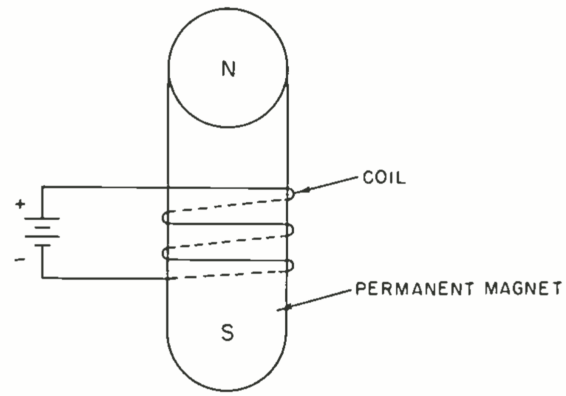
The moving parts are the surround, cone (diaphragm), dust cap, bobbin, voice coil, and spider. The cone and surround come joined together, as do the bobbin and voice coil. The stationary parts are the gasket, basket, front and back plates, magnet, and pole piece.

Stephen Boak is product specialist at Panasonic Company.

A description of some of the key pieces and their functions follows:

1. Basket—Sometimes called a frame or cone housing. The basket is the foundation for the speaker and holds all the parts. It may be made from stamped steel or cast aluminum, finished by static painting or plating.
2. Cone—Sometimes called the diaphragm or piston. The cone moves the air in front and behind it to make sound. It is usually made from a compound of paper, cloth and carbon fibre. Material for dome diaphragms range from Nylon 66 to Boron.
3. Surround—Also called the edge or anulus—part of the suspension. The surround attaches the cone to the basket. What the surround of a speaker is made of depends on the type. Woofers usually have a foam or butyl rubber edge. Musical instrument and full range types usually have a paper edge, which is just an extension of the cone. The kinds used for midranges and tweeters are too numerous to catalog here.
4. Dust Cap—Also called the center dome. Keeps foreign particles out of the air gap.
5. Spider—May be called a damper or back spider. A flexible ridged cloth disc attached to the cone and basket to keep the voice coil centered around the pole piece.

Figure 1. Current flow through the coil induces a magnetic field around the coil.



Introducing the system that can tape itself. Tune itself. All by itself. For up to a week.

Up to now using a tape deck to record Tchaikovsky's 4th at 4 PM on one station and Beethoven's 5th at 5 PM on another was easy. As long as you were home. Now Technics makes it just as easy when you're not, with the ST-9038 quartz synthesizer FM stereo tuner and its matching SH-9038 microprocessor.

When used with the ST-9038, the SH-9038 microprocessor can be programmed to tune eight FM stations in any order, at any time, on any day for one week. In fact, the SH-9038 can be programmed to remember 32 individual steps. Starting with the day of the week, the time, the FM station and AC line on/off.

All you do is simply select the "write" mode and the SH-9038's computerized memory does the rest. Then select the "read" mode for a readout of the programs you have selected. You can also override your preselected program by switching to the manual mode. What's more, the SH-9038 can be programmed to turn on or off three other components in addition to the ST-9038 tuner.

That's what the SH-9038 microprocessor can do. What the ST-9038 quartz synthesizer tuner can do is just as impressive. Unlike conventional tuners which use a series of variable capacitors to tune in FM frequencies, the ST-9038 uses the quartz synthesizer tuning system. With this system the quartz crystal, one of the world's most accurate reference devices, becomes the reference for the local oscillator frequency and the broadcast frequency. The results: Only the frequencies on which a broadcast signal might exist can be received. At precisely spaced 200 kHz steps. And that means you don't have to worry about drift or misalignment due to temperature, time or mistuning.

The SH-9038 microprocessor and the ST-9038 tuner. Because you can't beat the memory of a computer or the accuracy of quartz.

SENSITIVITY: 1.2 µV (75Ω). 50dB QUIETING SENSITIVITY (New IHF): Mono 18.1 dBf. Stereo 38.1 dBf. THD (100% modulation): Mono 0.1%. Stereo 0.15%. FREQ. RESP.: 20Hz to 18 kHz +0.1dB −0.5dB. SELECTIVITY: 75dB. STEREO SEPARATION: 45dB (1 kHz), 35dB (10 kHz). IMAGE REJECTION AT 98 MHz: 105dB.

Technics SH-9038 and ST-9038. A rare combination of audio technology. A new standard of audio excellence.

Technics
Professional Series

Circle [illegible]6 on Reader Service Card

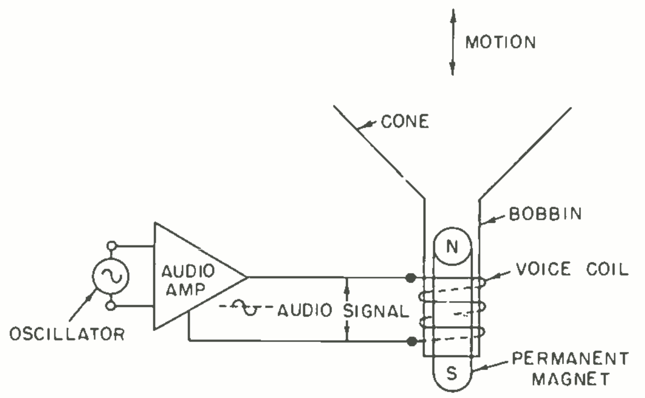


Figure 2. Basic speaker design.

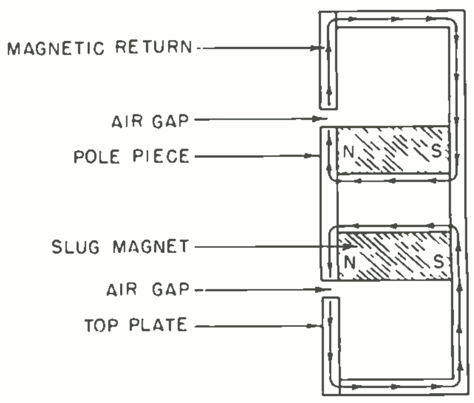


Figure 3. "Alnico" magnetic structure.

6. Voice Coil Assembly—Consists of a thin paper. cloth or aluminum tube (the bobbin) with a copper or aluminum wire coil around it.
7. Motor Structure— Front and Back Plates
   —Pole Piece
   —Magnet—popular types are Ceramic and Alnico. Ceramic magnets are a compound of clay and sintered iron. "Alnico" stands for *al*uminum-*ni*ckel-*co*balt. Other types include ferrite barium, cerium cobalt, and "Ticonal" (*ti*n-*co*balt-*n*ickel-*al*uminum).

The motor structure may also be called a "magnet pot." The motor consists of the magnet, and the magnetic return path to allow the diaphragm to move when voltage is applied to the voice coil.

TYPES OF LOUDSPEAKERS

Among loudspeakers available today, there are three basic categories (when classified by operating principle): 1. electrodynamic, 2. electrostatic, 3. piezoelectric. These classifications refer to the principle by which electrical energy is transformed into mechanical energy.

ELECTRODYNAMIC LOUDSPEAKER—BASIC PRINCIPLES

Electrodynamic loudspeakers are, by far, the most common. If it has a magnet, it's electrodynamic. When electrical current flows through wire, a magnetic field is induced around that wire. The polarity of the input voltage determines the polarity of the magnetic field around the wire. The polarity of the input signal is expressed either as a positive or a negative voltage, while the polarity of the magnetic field is expressed as north or south. Remembering high school science class, "Like poles repel, but unlike poles attract." These principles of polar attraction and the principle of induced magnetism are those upon which all electromagnetic loudspeaker designs are based.

Keeping in mind the two principles, polar attraction and induced magnetism, let's look at FIGURE 1.

The voltage from the battery will cause a current to flow through the coil, inducing a magnetic field around the coil. Because the polarity of the battery, with respect to the magnet, is positive, the polarity of the induced magnetic field will be north, causing the coil to move away from the magnet's north pole, towards its south pole.

Conversely, if we reverse the battery connections, the direction of current flow will reverse, as will the polarity of the induced magnetic field, and the coil will move in the opposite direction. If we alternate the battery connections or pass a varying signal in the coil, the direction of current flow will reverse or change with each alternation (alternating current), and the polarity of the induced magnetic field will be alternating, in turn. When this happens, the coil will move back and forth along the magnet as the direction of current flow alternates.

To convert this mechanical motion to acoustical energy, we must attach a cone or diaphragm to the coil and replace the battery with an amplifier driven by an audio signal, as in FIGURE 2. The amplifier drives the coil (now called a voice coil) with a signal of alternating potential. causing an induced magnetic field around the coil and a back-and-forth movement of the coil along the magnet. The coil is attached to the diaphragm and causes air to be moved back and forth as the coil moves back and forth. This, oversimplified, is how an electrodynamic loudspeaker produces sound.

Magnetism, like electricity, is always trying to neutralize itself; in this case, by trying to reduce the energy potential between one pole and the other. It is this principle that permits the electrodynamic loudspeaker to operate. FIGURE 3 shows a close-up view of an Alnico-type magnetic structure. The magnetic energy flows through the magnet pot, top plate, and pole piece to get the opposite pole of the magnet. To accomplish this, it must "jump" between the pole piece and top plate. This area is called the air gap. Just how much energy (called "flux density") is allowed to cross the gap, is a function of the size and type of magnet as well as the types and masses of materials used in the magnetic return circuit. The amount of magnetic energy, or flux, that crosses the gap is one of the major determing factors in the sensitivity of the loudspeaker to the input signal.

Types of magnets. As mentioned earlier, there are two popular types of magnets used in electrodynamic loudspeaker designs: Alinco (slug type) and ceramic (ring type). Magnet manufacturing methods dictate the shapes into which the magnets can be formed. The difference between the two is that the magnet slug is the pole piece in the Alnico design. In the ceramic design, the magnet surrounds the voice coil and the pole piece. Each structure has its own advantages and disadvantages.

Although both magnets are relatively efficient, the Alnico design has a better concentration of flux, thereby permitting relatively lighter structures when compared to ceramic designs. The basic drawback to the Alnico design, is that the magnet is inside the voice coil. If the coil is a high-power type, sustained amounts of high input power can actually result in a partial demagnetization of the magnet, which will alter the loudspeaker's performance. In the ceramic design, this is avoided. The ceramic design, however, has its own disadvantages. Because the

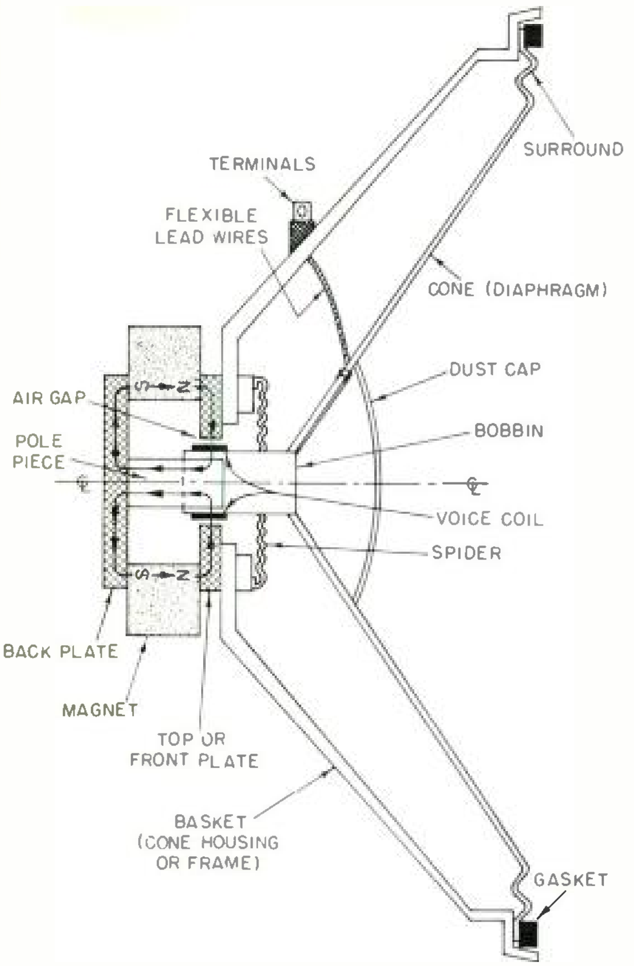


Figure 4. Parts diagram for a ceramic magnet-electrodynamic loudspeaker.

ceramic design requires that the magnet edges be exposed, a greater degree of fringing (lack of flux concentration) occurs, as compared to the Alnico design. This results in proportionately less flux being transferred to the air gap, requiring proportionately larger (than Alnico) magnets to perform a given job. This excess weight increases the chance of the magnet structure shifting, if the speaker is subjected to a sharp jolt. The ceramic magnet is also more susceptible than Alnico structures to being fractured, due to its material composition. FIGURE 4 shows the parts and assembly of a typical ceramic magnet electrodynamic loudspeaker.

ELECTROSTATIC OPERATION

Electrostatic designs, although not as common as electrodynamic designs, do occupy a very important position in the world of high fidelity. Although these designs tend to have excellent frequency response and transient response abilities, they do have limitations which are more severe than electrodynamic designs. In the areas of total acoustic output and power handling capacity they are defficient; they also lack good low frequency response and user convenience (for example, they require an extra power supply, which must be located near the speaker).

The electrostatic speaker operates in a similar fashion to an electrodynamic unit; that is, it derives its motion from the attraction of unlike charges. At the risk of being redundant, however, the charges are electrostatic, not magnetic.

FIGURE 5 indicates the basic parts of the electrostatic driver. A fixed high voltage is applied to a fixed plate.

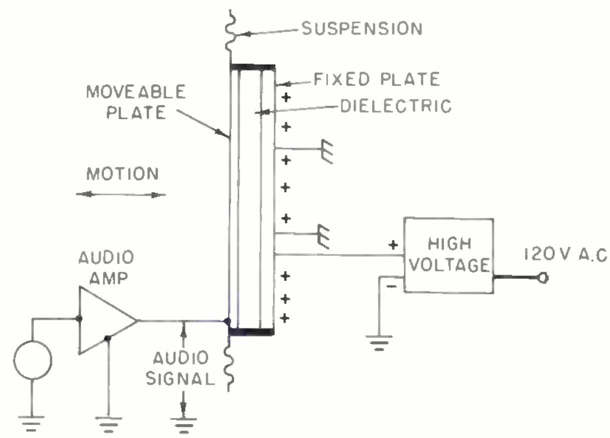


Figure 5. Electrostatic driver circuit.

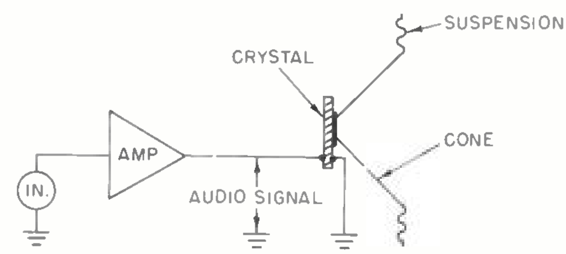


Figure 6. Piezoelectric driver circuit.

An amplified audio signal is applied to a movable plate. The two plates are separated by a dielectric (a special type of insulator). As the audio signal varies, the charge to the movable plate—which has the same polarity as the charge on the fixed plate—causes the movable plate to move away from the fixed plate. (Remember, like charges repel). Conversely, as the audio signal becomes more negative or less positive, the direction of motion is reversed because the movable plate is attracted to the charge on the fixed plate (unlike charges attract). This mechanical motion moves the air mass around it and produces sound.

PIEZOELECTRIC OPERATION

The third basic operating principle is the piezoelectric type. Loudspeakers of this design are sometimes referred to as solid state drivers.

As the name implies, this type of driver operates on the principle of piezoelectricity. It is also called a solid state driver because its operation is based on crystal, which is the basis for today's transistorized circuits.

The principle of piezoelectricity states that a crystal, when squeezed by a mechanical force, will emit a proportional electrical voltage. The earliest microphones worked this way. A diaphragm was attached to a crystal. The diaphragm would receive sound waves and push against the crystal. The crystal would emit a proportional electrical current which could then be amplified. The piezoelectric loudspeaker works in just the opposite way of the crystal microphone. It is shown in FIGURE 6.

An amplified audio signal is applied to the crystal. This causes a squeezing motion by the crystal. A diaphragm, connected to the crystal, moves with it, causing the air mass to move at the same rate, producing sound.

Because of the limited size of the crystals available, current piezoelectric designs have been restricted to high frequency reproducers only. These units are sometimes referred to as piezoelectric or solid state tweeters. ■

FLOYD E. TOOLE

High Fidelity in the Control Room—Why Not?

With "Rolls-Royce" loudspeakers in most living rooms, why do engineers continue to mix music on "Mack Trucks?"

THE CONTROL ROOM of a recording studio is the technical heart of the audio industry. Here, important decisions are made and an artistic idea becomes an acoustical reality. For the most part, nowadays, the original 'live' performance takes place in the control room during the final mix. The audience, of course, is small; an engineer, a producer and perhaps a handful of observers. It would be almost criminal if it all ended there—yet, all too often it does.

For a variety of reasons it may turn out that nobody else will ever hear the performance as these few people did. This is a pity, since presumably the musicians and the studio staff care about what was created. Fortunately, in many cases, music prevails, hits are made, and pleasure given. Nevertheless the fact remains that, by and large, the audience hears but a variation—if not a distorted caricature—of the original performance.

Prime factors in this dilemma are the control room monitor speakers. It is presumed that the sounds heard in the control room relate somehow to repoductions in the real world. Considering the colossal variety of home radios, tape recorders, car radios and 'hi-fi' sets that range from junk to jewels, it is clearly impossible to cater to all conditions. The sounds are simply too diverse and the differences too great for a single "compromise" to work.

CONSUMER "HI-FI"

In recent years, the consumer audio industry has grown enormously. Once a pastime for an enthusiastic minority, "hi-fi" is now an almost universal adjunct to domestic life. Once components varied considerably in performance and only the knowledgeable (and well heeled) had quality sound. Now genuine high fidelity sound is available to the masses. Only in speakers do large performance differences exist and rapidly even they are conforming to the same high standard of quality. Audiophiles have become a significant portion of the population and their buying power supports whole stores in larger communities. Even car stereo has moved into the hi-fi domain.

Studios have changed too. There are bigger and more sophisticated boards, tape recorders and signal processing devices. Computers are making inroads as studio equipment and digital recording makes its first appearances. It makes the mind swim, at least until somebody rolls the tape and, there it is!—the sound of the last decade.

With due respect to those studios that are making a conscientious effort, it does seem to be a fact that a great many are working with speakers that no self-respecting audiophile would give house room to. In terms of sound quantity they do fine, but sound quality . . .?

At this point there are introduced all manner of rationalizations: "it works great for AM mixes" (who should know that business better than the stations themselves—let them EQ, compress and limit, they do it anyway); "Fred Zinq had his last two hits mixed on them." (maybe Fred is just a great musician); Studios X, Y and Z use them and they are successful (thanks probably to good music and clever engineers—perhaps they all have good cafeterias); and so on. The justifications range from absurd to obvious.

Putting it plainly, I think the recording industry needs to get its act together. Nowadays I think it is professionally "unsound" to strive for anything less than high fidelity in the control room. Studio owners are in a real quandry. As it is, they must try to cater to the preferences of their customers, and they frequently fail. It can be an expensive and frustrating affair.

We demand technical excellence and conformity in all components up to the speaker, so why stop there? In consumer audio the better speakers are measuring and sounding more and more alike, and more like the real thing. It *can* be done. The only thing peculiar to professional speakers is that they must be utterly reliable, almost indestructable, and capable of undistorted sound at very high levels. Traditionally this has meant dipping into the sound reinforcement catalogs to find speaker components.

Designing high-fidelity speakers is a specialized business. To be up with the front runners one must have

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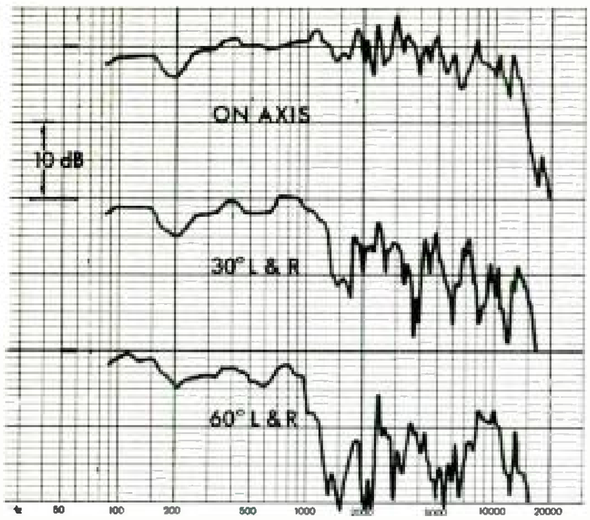


Figure 1. A selection of anechoic frequency response curves for a popular studio monitor speaker. Below about 200 Hz the output is dependent upon mounting and room acoustics.

knowledge, measurement facilities and the experience to know when and how to compromise. In spite of this, countless do-it-yourselfers put together systems in their basements or garages. Most of them are mediocre, but that doesn't matter. It matters more when studios get into the do-it-yourself speaker business, as many have out of sheer frustration with the tired old war horses. Woofers get put into inappropriate enclosures. Wrong crossover frequencies are used. Tweeters with 'flashlight beam' directivity are mated with medium dispersion midranges. Speakers are operated into frequency ranges where cone breakup abounds—and perhaps a bad choice has been made to begin with. And so it goes, through the catalog of errors made in the absence of useful technical information.

In the end, it is assumed, all will be made right with a graphic equalizer. Speakers will become silky smooth and room resonances will vanish. Nonsense!

PUT UP OR SHUT UP

It is one thing to complain about a state of affairs; it is another to do something constructive. Being both an audiophile and an acoustical consultant to professional audio I feel particularly strongly about a situation that seems almost to be self-perpetuating in a detrimental direction.

One of the myths that is slowly being hammered out of audiophiles is that the opinion of the individual is all that matters in the choice of a loudspeaker. It is not that personal opinions *don't* matter; quite the contrary, it is just that, by and large, they are unreliable. Human judgements are notoriously susceptible to bias by non-acoustical factors, such as brand names, size, price and 'reputation'.

For over a decade I have been responsible for a program of speaker evaluation wherein both measurements and listening evaluations are conducted. The former consist of more-than-usually comprehensive anechoic and real-room measurements. The latter are 'blind' tests in which great care is taken to ensure equal listening levels from speakers being compared. Perhaps even more importantly, much time is devoted to randomizing the position of the speakers and the listeners in the room during the test, which involves listening to a wide variety of music. In this manner, we deal statistically with acoustical aberrations of room and source material that can influence a judgement. The result is that most people, most of the time, agree on the relative merits of a group of speakers. Moreover, the results are in basic agreement with the measurements. That this should be such an uncommon occurrence reflects mainly on how unusual it is for good measurements *and* adequately controlled listening tests to be conducted.

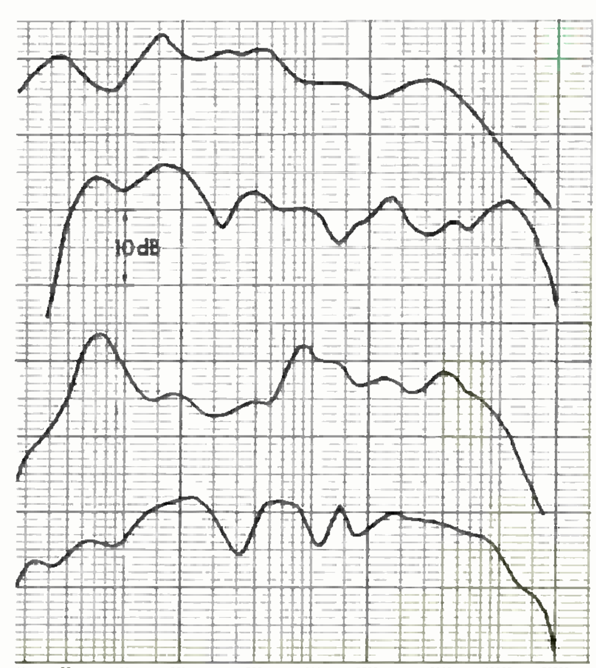
ANECHOIC TESTS

Anechoic measurements are the most revealing of speaker performance although, in the final listening situation, room measurements are useful in examining low-frequency problems due to room dimensions or speaker and listener placement.

Let us take an example. FIGURE 1 shows anechoic frequency responses measured at a distance of 6 feet from a very well known and widely used studio monitor speaker system. The on-axis response is the most flattering but, truthfully, not indicative of how it sounds. Off-axis radiation creates the reverberant sound field and accounts for the bulk of the acoustic output, therefore the 30°, 60° and other off-axis measurements must be regarded with more than passing interest. In the curves one sees a lot of roughness due simply to interference caused by multiple sources, diffraction, reflections, etc. These irregularities are more offensive to the eye than to the ear, and they are different for different mic positions. Some peaks, though, are visible in all the curves; for example, look at 1800 Hz, 2500 Hz, 3000 Hz and 5000 Hz. These and several other persistent features are indicative of resonances that are sources of sound coloration.

Above 1 kHz, the sound output becomes very directional, limiting the listening area over which the direct sound has a balanced frequency response, and contributing to an imbalance in the acoustical power output. One can anticipate a somewhat bass-heavy sound and prominent midrange coloration due to the directional discontinuity near 1 kHz.

Figure 2. Room curves measured at the engineer's head location in four different studio control rooms.



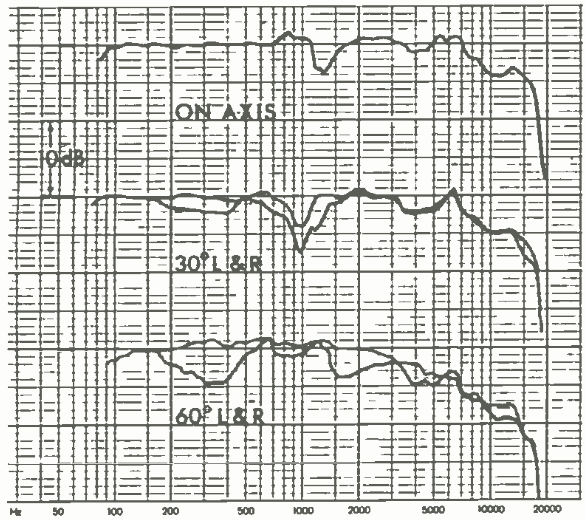


Figure 3. A selection of anechoic frequency response curves for a prototype studio monitor speaker.

Without regard for other aspects of performance: distortion, phase linearity, etc., we can predict that, in terms of the accuracy of sound reproduction, this speaker has some problems. Compared with better quality consumer speakers, including some made by the same company, it rates poorly.

EQUALIZATION?

Advocates of equalization should aproach this one with caution. Since room measurements are basically indicative of total radiated energy, they may show the high frequencies to be attenuated. Using an equalizer to flatten the curve will simply boost the on-axis response, which is already reasonably flat. The room curve may look better, but the ears will complain of strident highs. Hence there have developed a number of empirical 'ideal' room curves that, in fact depend very much on the intrinsic performance of the speaker being equalized. This, though, is not information that manufacturers commonly dispense (sometimes for obvious reasons). As a result, equalizations are usually done either to produce room curves that conform, in blind faith, to someone's 'ideal' contour or, simply adjusted by measurement and trial and error until the sound is 'right.' Neither of these can be regarded as much of an industry standard.

Room measurements and equalizers seem to be most useful at frequencies below about 500 Hz, where the audible defects of an unfortunate room shape and/or speaker placement can usually be somewhat alleviated.

Sadly, there are still those misguided individuals who believe that a control room ought to have a 'sound' of its own, in which case EQ may be used to produce a quality of sound that may exist *nowhere* else in the world. This is an ego trip the industry can ill afford. Distinctive sounds should exist on the records and be thus conveyed, through the medium of hi-fi, to the consumers home or car. Just imagine the different and idiosyncratic EQ's and mixes that have been produced in the four control rooms with the performances depicted in Figure 2.

THE RIGHT DIRECTION

For some time now the technology and products have existed to do much better than this. That there has not been widespread acceptance of *truly* accurate sound reproduction in the control room is a reflection of the most serious problem of all—human judgement. Money is poured into all manner of exotic electronics while the sound quality is too often dictated by speakers of demonstrably questionable merit.

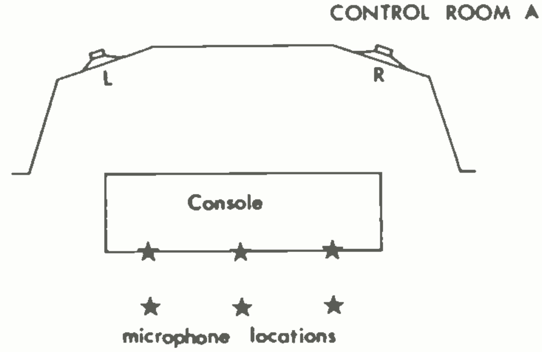
As part of a new recording studio project an opportunity arose to try out the high principles being expounded here. Using the same criteria that one would use in designing a hi-fi speaker (flat frequency responses, low distortion, wide dispersion, clean transient response) a number of speaker units were selected and assembled into systems. It was an interesting exercise. Along the way, some widely-used components were found to have serious technical and audible flaws, while other widely-ignored components turned out to be eminently useable. And, without the encumbrance of having to use components by a particular manufacturer, many options opened up.

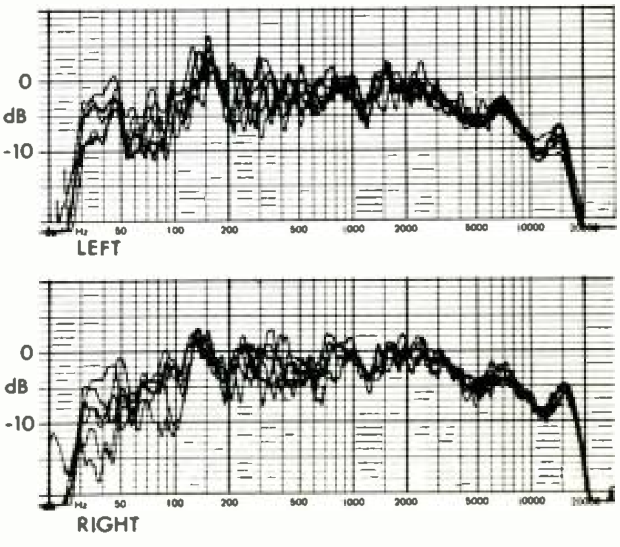
The result of this pilot project was that two prototype systems were built: both 4-way tri-amplified systems, combining direct radiators and horn drivers. Not only did they sound remarkably similar but the competition, in terms of listener preference, came not from the old standard monitor speakers but from the ranks of better quality hi-fi speakers.

That much had, in fact, been predicted from the performance measurements. Figure 3 shows anechoic measurements on the prototype speaker system that was eventually used in the new studio. It is evident that, not only are the frequency responses rather smooth, but there is little change up to very large angles off-axis. Incidentally, the high-frequency roll-off was designed-in, as a concession to reality: at present, even very good hi-fi speakers exhibit an energy deficiency at the upper frequency extreme.

Distortion was very low, and maximum sound output

Figure 4. Swept-tone frequency responses measured at six locations behind the console in a studio control room.





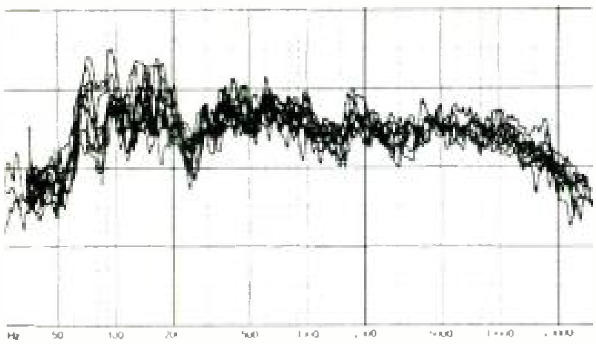


Figure 5. As Figure 4, but with a good domestic speaker substituted for the right monitor. Nine mic locations were used.

Figures 3, 4 and 5 are shown with the permission of Marc Productions Ltd., 1163 Parisien Street, Ottawa, Ontario K1B 4W4.

very high. This was a system that could make ears hurt, and yet it exhibited the essential qualities of a high-accuracy speaker. It wasn't perfect by any means. Any transition from direct radiator to horn driver is a difficult one, and the ears find it hard to ignore the crossover points. Besides that, high efficiency and horn speakers tend intrinsically to exhibit more colorations than the highly damped inefficient direct radiators that are used in accurate consumer speakers. Still, it was an impressive sound.

The next step was the risky one. Could this performance be maintained in the special environment of a control room?

To avoid acoustical interference effects due to the adjacent room boundaries, the speakers were built into the wall and adjacent walls were angled away from the face of the system. The control room was large, unsymmetrical and irregularly shaped, with a highly absorptive surface behind the mixing console. In short, attention was given to maintaining good acoustical practice in the room itself.

When the speaker systems and the rooms were assembled, the results were as shown in FIGURE 4. Here are superimposed six separate frequency-response measurements made at three locations spaced along a 12-foot console and at three locations four feet behind the console, all at the height of a seated listener's head.

The resolution of these measurements was improved by the use of swept pure tones, rather than the usual 1/3-octave bands of noise. The equipment used was all by Bruel and Kjaer. (A small amount of smoothing was introduced by limiting the pen writing speed of the model 2205 level recorder to 80 mm/sec at a paper speed of 10 mm/sec).

The results are unusually good by most standards. The shape of the curves at frequencies above about 300 Hz is about what would be expected from an interpretation of the anechoic measurements in FIGURE 3. The good behaviour at low frequencies is an indication that the room appears not to have any insurmountably bad characteristics. Perfection has been, as ever, elusive but for a first attempt it is not bad.

Unusual in a control room is the fact that an almost uniformly high quality of sound is audible over a large area. Stereo imaging suffers, of course, but otherwise critical listening is now possible for several people at once.

Educated eyes will have spotted some aberrations that can stand improvement. There is a shallow depression between 60 and 90 Hz, and others centered on 5 kHz and 10 kHz. A slight elevation around 150 Hz also needs attention. A simple 4-filter parametric equalizer is all that is needed to turn this into a superb performance. Furthermore, because the corrections needed are not due to any gross maladies of the speakers themselves, the benefits of the equalization will be apparent to all listeners in the room.

A final confirmation that the exercise was not an accidental success comes from FIGURE 5. Here a small domestic speaker (a tested and proven favorite of critical audiophiles), was substituted for the monster monitor and the same swept-tone measurements were made. So consistent were the curves that three more were run at mic positions 7 feet behind the console and further to the sides. The small woofer cannot compete at the lowest frequencies but there's little to criticize at middle and high frequencies.

Clearly, there are no insurmountable technical problems to moving hi-fi into the control room. The acceptance of it by engineers is quite a different thing.

The present example is but a forerunner of things to come—if there is a demand for it by the industry. Ironically, it still seems to be a fact that a lot of successful and influential engineers feel the necessity to promote the use of speakers that are not just different from one another, but that are apparently deliberately less than the state-of-the-art.

Consumer audio is getting its act together amazingly well, in spite of poor odds. It is time all professionals took a long look at what is happening out in the world of hi-fi —it may be a surprise.

Change of any kind is difficult, but it must come if we find we are adhering to an arbitrary or an obsolete standard. That, I think, is the kind of detachment that identifies a true professional. ■

JOHN EARGLE

Requirements for Studio Monitoring

The monitor system and its control room environment remain an ongoing challenge to both studio designer and component manufacturer.

FOR MOST OF ITS HALF CENTURY, electrical recording has made do with inferior monitoring speakers and conditions. The early requirements were fairly simple; monitors were used to check signal continuity and detect possible interference levels from hum and other sources. Esthetic judgments were rarely made over these early systems.

The advent of tape recording in the post-war years brought greater artistic freedom, in terms of increased bandwidth and dynamic range, and the role of the monitor speaker changed dramatically. The technology which had been developed for motion picture sound provided the basis for monitor systems over which esthetic judgments could be made. A handful of manufacturers dominated the field; in the United States, the Altec 604 coaxial loudspeaker became the reference standard, while the Tannoy 15-inch dual-concentric loudspeaker played a similar role in Europe.

In the early sixties, the monitor designs of James B. Lansing Sound, Inc. began drawing attention, primarily through joint efforts with a major record company and its affiliates around the world. The company's technical traditions were firmly rooted in those of Western Electric as well as the design philosophies which originated on the west coast during the early years of sound motion pictures. This technology stressed efficiency and ruggedness as well as the use of compression drivers and their associated horns and acoustic lenses for high-frequency applications.

The most recent epoch in monitor system design dates from the early seventies. Professional design consultants are responsible for many studios today, and they have integrated their own monitor designs, constructed from standard componentry, into control room environments which stress uniform acoustical absorption and diffusion across the audio range.

MONITOR SYSTEM REQUIREMENTS

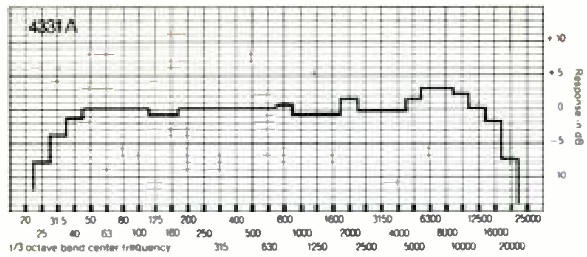
In general, we can outline present day-requirements for the professional monitor system and its environment as follows:

1. Ruggedness. Monitor systems must be able to withstand considerable electrical abuse, unintentional or otherwise.
2. High output capability with low distortion. Monitor systems must be able to reproduce cleanly the sound pressure levels in the control room typical of pop-rock performances. The ready availability of high amplifier power has allowed a beneficial trade-off between system sensitivity and low-frequency bandwidth extension.
3. Accurate time domain response. No firm criteria exist for this yet, but it is surprising how accurate in this regard many present monitor designs are.
4. Reasonably flat energy response across the audio band. Whether wide or narrow, the horizontal dispersion angle should be maintained as evenly as possible.
5. Lateral symmetry in the control room, along with smooth boundary conditions and smooth absorption characteristics across the audio range.

ANALYSIS OF TYPICAL SYSTEMS

It is curious that the high fidelity industry realized the advantages of three-way designs long before the designers of monitor systems did. Up to the early seventies, most monitors were two-way systems. In fact, for certain "close-

John Eargle is Vice President, Product Development at James B. Lansing Sound, Incorporated.



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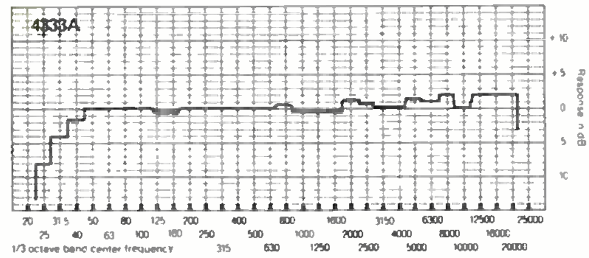

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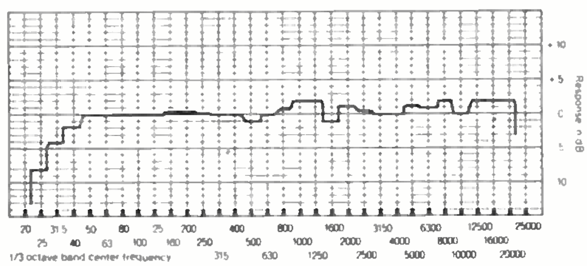


(C)

Figure 1. Frequency response for three speaker systems.
(A) Two-way system (JBL 4331)
(B) Three-way system (JBL 4333)
(C) Four-Way system (JBL 4343)

in" monitoring conditions, a two-way system may still be preferable to three- or four-way designs, because of the spatial integrity of high frequencies emanating from a single source.

The chief drawbacks of two-way systems have to do with uneven energy response and a tendency for high-frequency distortion at high levels. A typical two-way system may have a 15-inch LF unit crossing over to a horn-loaded HF assembly in the region of 1 kHz to 1.5 kHz. In terms of energy response, the dispersion of the 15-inch LF unit narrows considerably as it approaches the 1 kHz range crossover point. The transition to the HF assembly once again broadens the dispersion angle, but beyond 10 kHz the response is apt to narrow again unless the design is an exemplary one.

In FIGURES 1 and 2, the frequency response and angular coverage of representative two-, three-, and four-way systems are compared. The frequency response plots were made using ⅓-octave pink noise signals, averaged over a 60 degree horizontal arc and a 30 degree vertical arc.

The JBL model 4331 is a typical two-way design. This system is an updated version of the model 4320, introduced in the early sixties. In the early seventies, the model 4333 added a UHF driver to the two-speaker array of the 4331.

From these figures, it will be readily seen that the additional UHF driver permits an extended high-frequency response, as well as an improvement in effective angular coverage. The same enclosure and baffle configuration is used for both the 4331 and 4333, and is shown in FIGURE 3.

FIGURE 4 shows details of the four-way model 4343, introduced in the mid 1970's. This system added a 10-inch lower mid-range cone element to the three-way configuration. As seen in FIGURE 2, the effect of the lower mid-range driver on angular coverage is apparent: it effectively broadens the system's coverage in the 500-1000 Hz octave.

While the 4331 is inherently symmetrical, the 4333 and 4343 provide for mirror imaging of all components through alternate component mounting as well as (in the case of the 4343) baffle rotation.

The effect of a separate UHF element in an array serves two purposes; dispersion at high frequencies is ensured (as is evident from the dispersion curves), and second harmonic distortion is reduced. FIGURE 5 shows the advantage of a three-way system over a two-way system as regards second harmonic distortion. In FIGURE 5(A) we see the on-axis high-frequency response of a two-way system with a nominal input level of one watt. The second harmonic distortion is shown raised in level by 20 dB for ease of comparison. Note that the level of the second harmonic component tends to rise with frequency and remain at a level about 35-40 dB below the fundamental. At FIGURE 5(B), we see the response of a three-way system under the same conditions. Here, the second harmonic distortion decreases as the UHF element comes into the picture. The same mechanism which causes harmonic distortion will of course cause intermodulation distortion well within the audio band on complex signals. The three-way system will therefore be less prone to IM effects than the two-way system.

SPECIAL PURPOSE SYSTEMS

The three systems we have just discussed represent elaborations on the basic two-way theme, and should satisfy most normal monitor requirements. However, a "no holds barred" approach is sometimes required, in order to meet the demands of high-level rock monitoring. The JBL 4350 is a representative four-way design, making use of two LF drivers, and it is designed to be bi-amplified. Nominal specifications are:

	LF Section	HF Section
Sensitivity	93.5 dB/watt/metre	93.5 dB/watt/metre
Power Handling	200 watts	100 watts

Figure 2. Angular coverage for three speaker systems.
(A) Two-way system (JBL 4331)
(B) Three-way system (JBL 4333)
(C) Four-way system (JBL 4343)

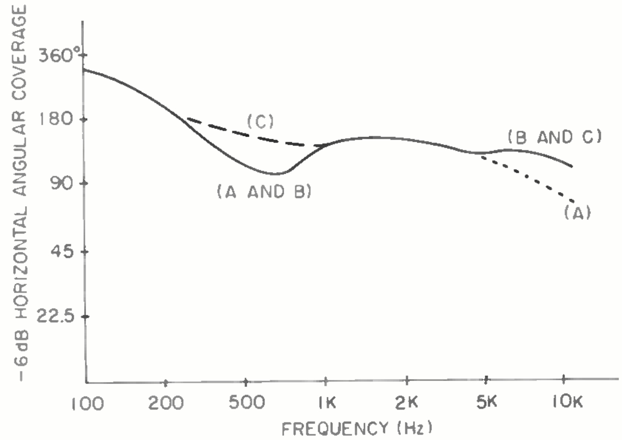


Figure 3. A three-way enclosure system. Note the UHF driver to the left of the regular high-frequency system. (JBL 4333)

With these characteristics, the 4350 can easily produce levels in a normal environment of 110 dB at distances of 10 feet. The system is shown in Figure 6.

For many broadcast and semi-pro recording applications, fairly straight-forward two- and three-way direct radiator systems are more than adequate as monitor speakers. These are generally bookshelf systems, and as such are limited in power handling capability when compared with their big brothers in the compression driver class. Typical sensitivity and power ratings for such systems are listed below.

Number of Elements	Sensitivity	Power Rating (steady State)	JBL Model
2	88 dB/watt/metre	15 watts	4301
3	91 dB/watt/metre	40 watts	4311
3	89 dB/watt/metre	40 watts	4313
4	89 dB/watt/metre	60 watts	4315

TIME DOMAIN ACCURACY

We have heard much in the last two years of the importance of time and phase accuracy in high fidelity speaker designs. These concerns, if they are important at all. should have relevance in the monitor area as well. Writing in the *Journal of the Acoustical Society of America,* Blauert and Laws established criteria for non-audibility of delay effects, in the paper, "Group Delay Distortion in Electroacoustical Systems," vol. 63, no. 5, May, 1978.

While it is true that a number of consumer high-fidelity systems exceed the Blauert and Laws criteria, it may be argued that this level of performance is really not necessary.

It is surprising how well behaved the modest three-way monitor systems are in their time domain response; they are better in this regard than the larger designs with compression drivers. This may be seen in FIGURE 7, where the time domain response of the 4313 is compared with its big brother—the 4333. The displacement due to the midrange horn structures account for these differences, as opposed to a typical three-way direct radiator system with the acoustic centers of its elements located on the plane.

In computing the group delay characteristics of the models 4313 and 4333 shown in FIGURE 7, the phase response was first measured using a time delay adjusted to the acoustic path length between the system and the microphone. The slope of the phase response with respect to

Figure 4. A four-way system. In the photo, the UHF driver is to the right of the high-frequency system.

frequency was then measured graphically. This slope ($d\O/d\omega$) represents the group delay characteristic of the system.

THE MONITORING ENVIRONMENT

The professional studio designers we referred to earlier have not only designed their own monitor systems but have established criteria for studio and control room acoustics as well. A handful of these design consultants have been very successful and have established impressive "track records," designing rooms in which absorption is evenly distributed and further, is uniformly calculated as a function of frequency.

Often, the monitor enclosures are flush-mounted into the environment; this ensures that uneven response from diffraction effects due to sharp boundary discontinuities will be minimized.

Another characteristic of a well-designed control room is the avoidance of uneven bass response through the use of selective absorption. Such "bass traps" effectively damp out low-frequency resonances due to the normal mode or eigentone structure characteristic of the room.

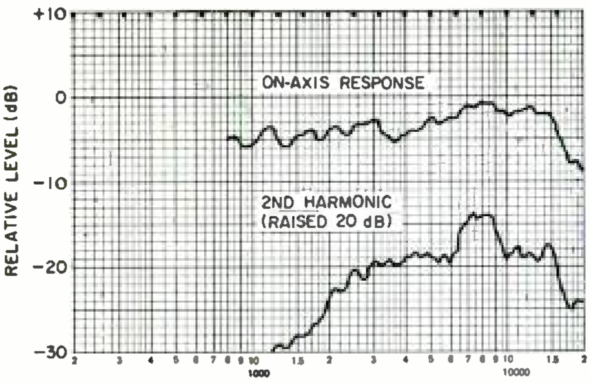
Finally, a canting inward of the monitors, along with the use of wide-dispersion HF devices, will ensure that smooth response will be maintained over a relatively large space, enabling both engineer and producer to hear equally well.

MONITOR EQUALIZATION

Monitor system equalization has become an accepted practice in professional control room design. If the monitor componentry has been properly specified at the outset, and if the acoustical design is proper, then the amount of equalization required for smoothly-tailored response at the operator's position may be quite small.

Typically, one-third-octave, minimum-phase, band-rejection equalizer designs are used, and these are now available from many manufacturers. After some years of field experience in monitor equalization, most practitioners of the art are pretty much in agreement on the following:

1. The last equalization is the best. This rule is almost self-fulfilling if due attention has been paid to monitor "hardware and horse power" as well as acoustical matters.



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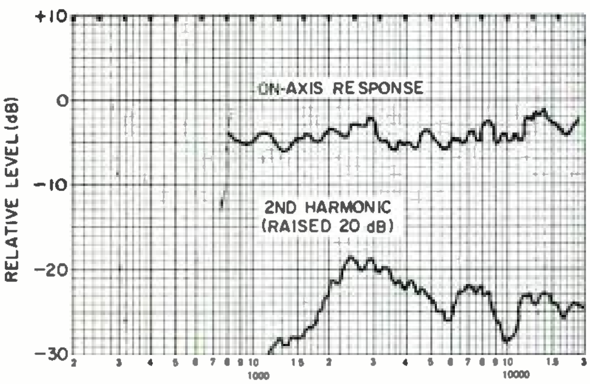


(B)

Figure 5. Harmonic distortion in two- and three-way systems.
(A) Two-way system (JBL 4331)
(B) Three-way system (JBL 4333)

2. Where the room design is laterally symmetrical, it is apparent that the same equalization curves should apply to both left and right monitor channels. This is highly desirable, as it guarantees that stereophonic imaging—a function of the first arrival sound at the listener—will be precise and unambiguous.

Preferred equalization contours will vary according to tastes and traditions. In general, an adequate monitor in a properly designed control room can be equalized for flat response in the prime listening area out to 15 kHz. More usually, the response is held flat out to about 7 or 8 kHz and allowed to roll-off 3 dB/octave above that point.

Figure 6. The "no-holds barred" approach. A four-way system with two low-frequency drivers. (JBL 4350)

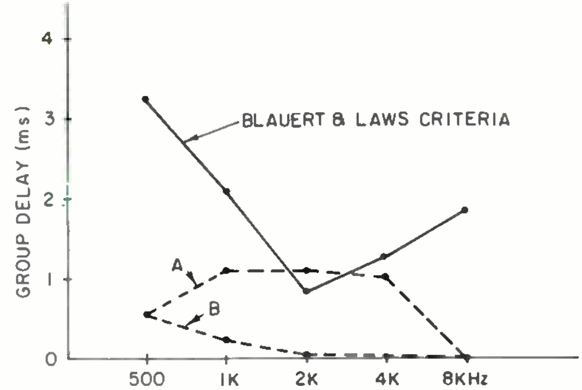


Figure 7. The Blauert and Laws criteria for non-audibility of delay effects.
(A) Time domain response for a professional three-way system (JBL 4333).
(B) Time domain response for a bookshelf three-way system (JBL 4313).

BI-AMPLIFICATION

The chief benefit of bi-amplification is the reduction in intermodulation distortion which it affords. Low-frequency power demands (and they are invariably greater than the high-frequency demands) may drive even a large amplifier into clipping, and the products of the clipping will show up as distortion through the HF portion of the system. With bi-amplification, both LF and HF portions of the system have their respective amplifiers, with the frequency-dividing action taking place before their inputs. Therefore, there is no possibility of intermodulation taking place between LF and HF parts of the monitor system.

An additional, but more subtle, advantage of bi-amping results from the elimination of lossy inductances in the LF portion of a conventional dividing network, and the result may be a significantly better amplifier damping factor, as seen by the LF transducer.

One should *never* skimp on power allotments in a bi-amped system. Even though it can easily be shown that bi-amping can provide a two-to-one power advantage over a standard system on certain kinds of program material, this will not be true in the general case. In any event, amplifier power is cheap these days, and there is absolutely no reason in a well-engineered system not to use rated power—with an additional 6 dB of head room for good measure. Many bi-amplified systems are equalized as well, and this is only one more reason to power the system adequately.

Bi-amping is sometimes hard to implement, and the user is often left to his own devices. It should not be undertaken without first asking the manufacturer's advice. Larger monitor systems should provide for proper component access through external switching and additional terminals. Many manufacturers, including JBL, also provide electronic dividing networks for use in bi-amping.

CONCLUSIONS

The monitor system and its environmental requirements remain an ongoing challenge to both studio designer and component manufacturer. Responsiveness to the needs of all segments of professional audio is an obligation of any company wishing to stay in the forefront of the industry. Progress over the last eight years has been rapid, and we can look forward to significant developments as we move into the decade of the eighties. ■

ALMON H. CLEGG

It's About Time

Waveform fidelity throughout the signal path, let's not exclude the loudspeaker.

FIRST, a word about the subjective and objective aspects of loudspeaker systems. An artist spends most of his working time in the "subjective" domain. A scientist (or recording engineer) spends most of his time in the "objective" domain. Objective aspects are measurable, quantifiable and repeatable. Temperature, voltage. frequency, impedance—these characteristics are readily measured, using the appropriate instrument, and then objective data of one kind or another can be recorded. So long as consistent measuring procedures and calibrated instruments are used, two different observers can obtain similar results. Take, for instance, the impedance value of a loudspeaker at 400 Hz. The value is easily measured and recorded, with little dispute about it among different engineers. The only argument may be over the precision of the measurement. Is it *really* 7.985 ohms or just 7.984 ohms? On the other hand, subjective aspects, are unique to the observer, or "subject." Consider a fine oil painting: You think of it as a beautiful masterpiece, yet another person tells you it's repulsive. Consider again the loudspeaker system which is to one observer "boomy," to another "strident" and to yet another "hollow." Each may be an "expert in the field," willing to defend his evaluation with his reputation, if not with his life.

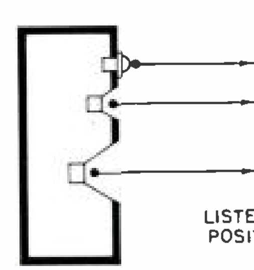
Thus, the difference between objective and subjective. Now, in loudspeaker analysis, evaluation is usually a combination of objective measurements, such as frequency response under certain conditions, polar or dispersion characteristics, etc., and of listening tests, which are subjective. A great historical difficulty with all audio devices—particularly loudspeakers—has been in correlating the objective measurements with the subjective tests. A great deal of devotion has been committed to this subject by audio engineers throughout the world. New measurement concepts, such as TIM and slew rate in amplifiers, have caused a great furor. In recent decades, one of the most significant design directions in loudspeakers has been the search for waveform fidelity, which would add a much greater degree of objectivity to analytical techniques and at the same time remove some subjective guess work. Hopefully, waveform fidelity may make the system sound better, while it measures better.

Almon H. Clegg is the manager of the Audio Engineering Department Product-Engineering Division at Technics.

THE WAVEFORM FIDELITY CONCEPT

The concept of waveform fidelity is certainly not new to the measurement of amplifiers and preamplifiers. Briefly stated, the technique is to present some complex waveform at the input terminals and compare the output with the input. Using a dual-trace scope (or more sophisticated means. such as comparator devices), it is easy to detect the resulting waveform distortion caused by the device under test.

Figure 1. A conventional loudspeaker system. Note the different path lengths from each speaker within the system.



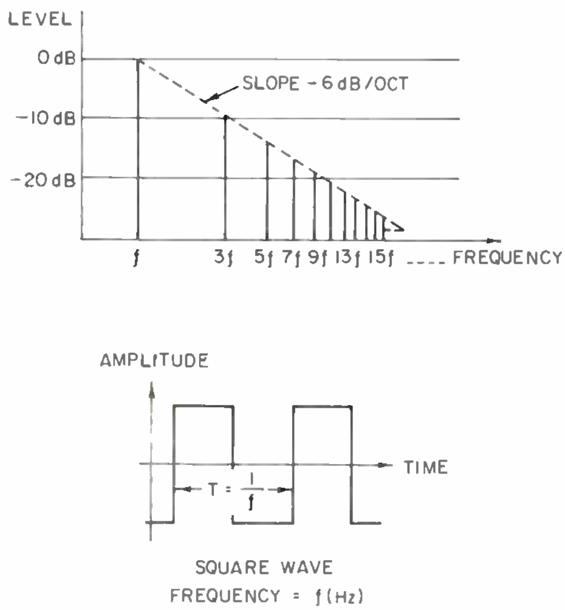


Figure 2. Spectrum of square wave.

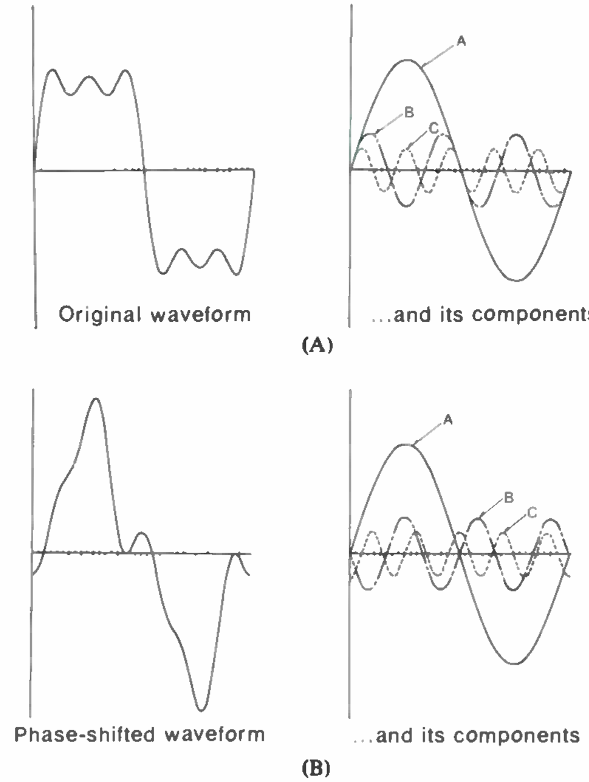


Figure 3. Effect of phase shift on waveform.
(A) Top diagram: Original waveform and its components.
(B) Bottom diagram: Phase-shifted waveform and its components.
A=Fundamental
B=Third Harmonic
C=Fifth Harmonic

Complex waveforms are interesting because they contain at least several frequencies, with a certain phase relationship between them that should remain fixed. Using mathematical theorems, it can be shown (by Fourier transforms) that any non-sinusoidal, continuous, repetitive wave shape can be considered to be a number of pure sine waves, with certain amplitude and phase relationships. Thus, when a device passes such a signal, it must not alter the amplitude or phase of *any* frequency throughout the spectrum. or it will cause waveform distortion.

For years, the square wave has been used in testing amplifiers. Let's take a look at one, in order to see why it is such a severe test, and why conventional loudspeakers do so poorly at passing it. Using Fourier analysis, it has been shown that the equation for a square wave is:

$$e = \frac{4}{\pi}(\sin \omega t + 1/3 \sin 3\omega t + 1/3 \sin 3\omega t + \ldots\ldots\ldots)$$

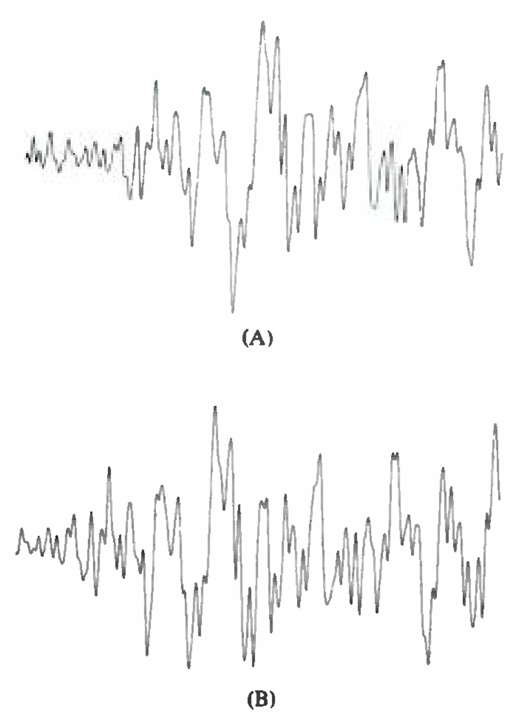
This means that a square wave can be generated by taking an infinite number of sine wave generators, with amplitudes and frequencies set exactly as the equation dictates. In other words, the amplitude of the third harmonic ($\sin 3\omega t$) is one-third the amplitude of the fundamental ($\sin \omega t$). The fifth harmonic is one-fifth the amplitude of the fundamental, and so on.

But now, let's see what happens when this square wave is reproduced by a conventional loudspeaker system, such as that shown in FIGURE 1. The sound pressure wave arrives at the listener's ears from the tweeter before it does from the woofer, because the tweeter is physically closer. Consider what effect this has on the waveform. First, let's look at the frequency spectrum of the square wave, in FIGURE 2. This shows the relative amplitude of each frequency, or harmonic, but it does not show the phase.

For simplicity, let's take just the fundamental and the first two odd harmonics, and consider that the amplitude is not altered, i.e., the speaker has absolutely flat response. The equation for our truncated square wave is:

$$e = \frac{4}{\pi}[\sin(\omega t + \theta_1) + 1/3 \sin(3\omega t + \theta_2) + 1/5 \sin(5\omega t + \theta_2)]$$

Figure 4. Evaluating waveform fidelity by comparing input and output waveforms. (A) represents input waveform. (B) represents output waveform. Speaker used in test—Technics SB—7070.



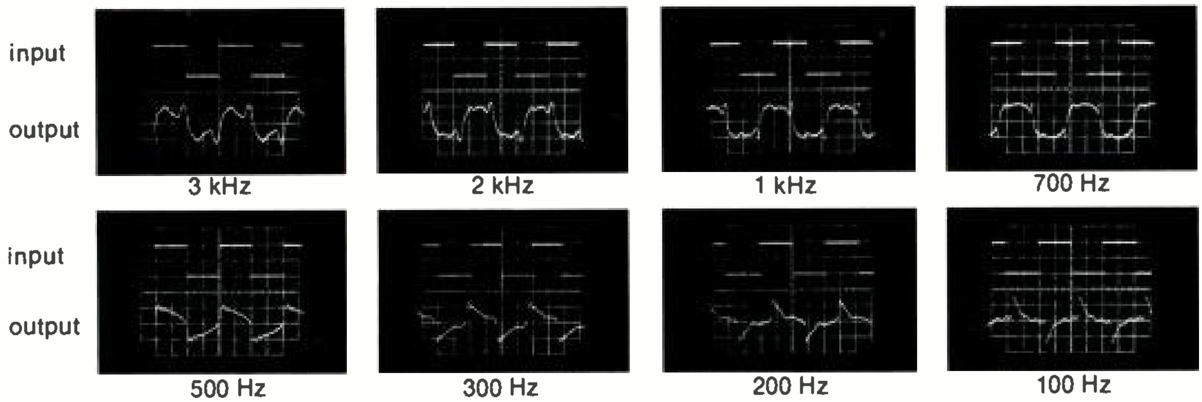


Figure 5. Square wave responses of a linear phase loudspeaker system.

The new terms, θ_1, θ_2, θ_3, represent phase shifts caused by the loudspeaker system. It is a simple matter to write a program for a pocket programmable calculator to compute the values of the final waveform. The result with zero phase shift ($\theta = 0$) is shown in FIGURE 3(A). Notice the waveform approximates that of a square wave. (Adding still more odd harmonics would give us a closer approximation.) But now, let the phase of each frequency be shifted slightly and see what happens. The result is shown in FIGURE 3(B), where the various values of θ are no longer equal to zero. This is a very typical waveform, as seen from conventional loudspeaker systems.

An interesting observation about the distorted waveform is that *only* phase has been changed. The amplitude and the frequency of each harmonic were not changed. If they had been, even greater distortion in the waveform would have appeared.

Thus, as we can see from the foregoing discussion, a loudspeaker system must have flat frequency response throughout the range of harmonics and flat phase response in order to insure waveform fidelity. Many designers have merely stopped here, aligned the driver's voice coils in a vertical plane and assumed the job is completed. However, many more problems must be dealt with because of the inherent phase shift within the individual drivers and the crossover network.

Conventional dynamic drivers have phase shift because of acoustical and mechanical resonances. These phase shifts must be compensated for by adding some RLC components between the amplifier and the terminals of the driver. Also, the very nature of crossover networks causes phase shift throughout the crossover region, and specially designed networks must be created to allow acceptable overall system performance.

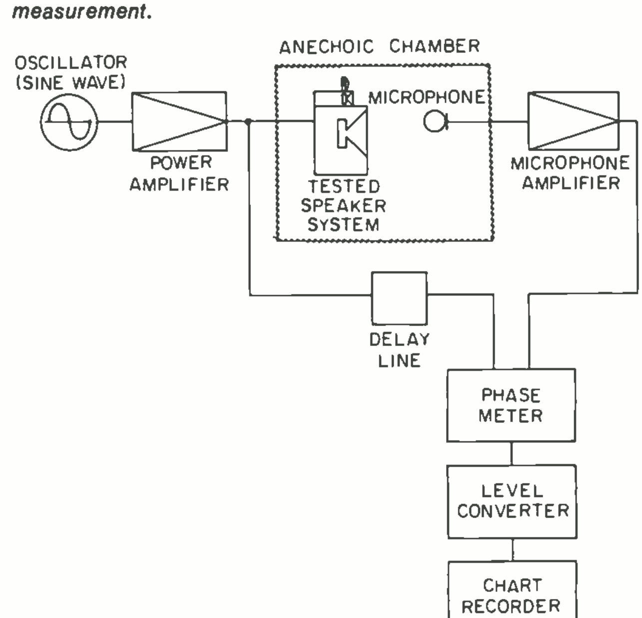
After careful attention is given by the designer to all of the above mentioned details, the final product can be measured. There are no universally accepted measurement standards for waveform fidelity, but there are three ways of showing it that will be discussed. Of course, an actual musical signal is of interest since this is what it is all about. Using two calibrated microphones (one placed in front of a musical instrument and the other in front of the loudspeaker under test) and a storage oscilloscope, it is possible to compare the output to the input. This procedure is quite difficult because of synchronizing problems with the instrumentation and of course, deciding which portion of the musical waveform to show is judgemental. Nonetheless, it is a worthwhile test and FIGURE 4 shows the result of such a test, using a piano as a sound source. The speaker under test is a Technics SB-7070.

LINEAR PHASE LOUDSPEAKER SYSTEM

Prior to the development of linear phase loudspeaker systems (also called time aligned, time coherent, phased array, etc., by various manufacturers) it was unheard of to expect worthwhile results. We are now brave enough to apply the square wave to the terminals of a speaker system and thrust a pickup microphone in front of it and see the results. FIGURE 5 shows the results of several different frequencies as reproduced thru a linear phase loudspeaker system. While these resulting waveforms are certainly not as good as those of an amplifier, they do represent a great step forward in state-of-the-art design of loudspeaker systems. As the years roll on, it is certain that many significant improvements will be forthcoming.

The two previously mentioned test methods are in the time domain. It is possible to characterize the linearity of the amplitude and phase *vs.* frequency by direct measurement. The system is complicated somewhat by virtue of the time taken for the sound wave to travel from the loud-

Figure 6. Block diagram of test set-up for phase measurement.



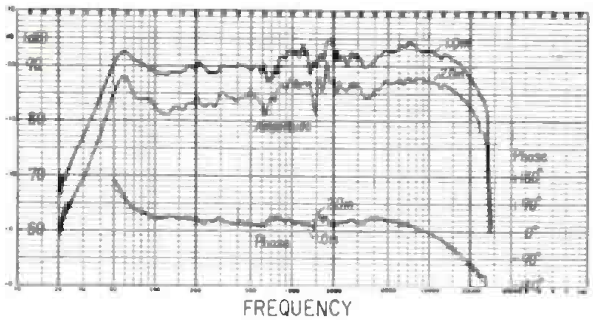


Figure 7. Amplitude and phase response for a linear phase speaker system (Technics SB7000A).

speaker under test to the pick-up microphone. This is solved by utilizing an adjustable delay system which can be set to equal the transit time of the sound wave. By placing the delay in the signal generation path, it is possible to compare the phase of the input and the phase of the loudspeaker output by feeding the two signals directly into a phase meter. A test set-up is shown in FIGURE 6. By adding chart recording equipment, it is possible to plot phase *vs.* frequency in the same fashion that frequency is plotted. In FIGURE 7 both amplitude and phase have been recorded on the same chart paper. You will note that phase is within ± 90° from about 70Hz to 15 kHz.

What does all this mean? To the designer it adds one more "objective" characteristic and removes a little bit of "witchcraft." To the advertising copy writer it gives a new dimension to write about. To the corporate lawyer, it gives an ulcer, worrying whether ad claims can stand FTC scrutiny. To the sales manager it means a whole new concept to sell: product differentiation (at least until all manufacturers have it!). But what about the professional user? Does it give him any benefits? Do linear phase loudspeakers reproduce sound more realistically?

Skeptics point out that no scientific tests have yet proven phase shift to be audible—at least not significantly so. Thus, there are those who challenge the need for linear phase (particularly those who don't have it). To the question of whether or not it can do harm, there seems to be no disagreement. All things being equal, time alignment can't possibly do any harm. So, what are the advantages? Listed below are three points worth considering:

1) The great attention to design and performance details required to achieve waveform fidelity result in a better product.

2) Waveform fidelity insures that frequency response and phase/linearity cannot be too far off regardless of what one may think of its subjective sound quality.

3 Stereo and multi-channel playback is spatially more accurate when phase shifts are linear and uniform.

As with any product, the final test is; how well does it do the job, compared to the alternatives? And where does that lead? Back to the listening room so that you can decide for yourself. To this listener, and to many friends and associates who have listened and carefully compared the time-corrected design philosophy to the conventional designs, there seems to be a general agreement: linear phase systems do sound better.

Of course, still other listeners will continue to disagree. But will anyone argue that waveform fidelity throughout the signal path is an invalid concept? If not, then why not strive for waveform fidelity at the loudspeaker, as well as in the amplifier? ■

D. B. HATHAWAY

The Use of Ferrofluid in Moving-Coil Loudspeakers

Ferromagnetic fluids applied to moving-coil loudspeakers provide efficient heat sinking capabilities and viscous damping of the voice coil's motion.

FERROFLUID TECHNOLOGY, developed in conjunction with the NASA space program, has been commercially available from Ferrofluidics Corporation of Burlington for over ten years. However, it was not until 1974 that ferrofluids were used in loudspeakers. At that time, Epicure Products of Newburyport, Massachusetts incorporated the fluid in the design of a new tweeter. Since then, the use and application of ferrofluid in loudspeakers has expanded tremendously, with research continuing to add to our knowledge of its behavior in loudspeakers.

The use of a ferromagnetic fluid in the air gap of a moving-coil loudspeaker not only provides the designer with a method to provide voice coil heat sinking, but also the capability of adjusting the damping of voice coil motion, without affecting speaker efficiency.

Before discussing the use of ferromagnetic fluid in moving-coil loudspeakers, it will be helpful to describe exactly what a ferromagnetic fluid is, and what some of its physical properties are. This will provide an immediate idea of the uses and limitations of a ferromagnetic fluid.

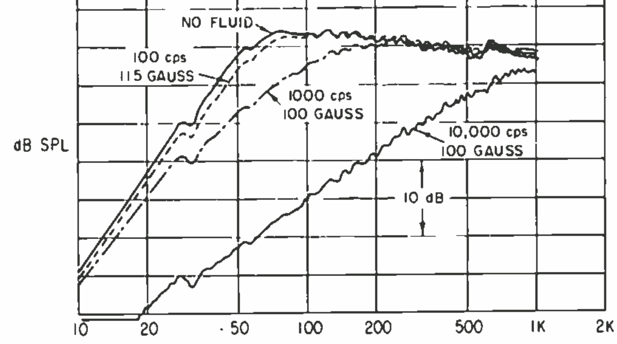
MAGNETITE PARTICLES

Ferromagnetic fluid is a suspension of magnetite (the stable, inert magnetic oxide of iron) particles in a liquid carrier. The particles range in size from 90 to 100 angstrom units (1 angstrom unit; abbr. $Å = 1 \times 10^{-10}$ meter, or approximately 100 hydrogen atomic diameters). The particles are so small that the thermal Brownian motion of the liquid carrier molecules keeps them permanently suspended. (The irregular movement of small particles—known as Brownian motion—is attributed to the bombardment of the particles by the molecules of the medium—Ed.)

The particles are coated with a stabilizer (known as a surfactant) which keeps the particles from clumping together and ensures a homogenous colloidal suspension (that is, an even dispersion) under the influence of strong magnetic fields. The magnetite particles provide the fluid with its magnetic properties. Higher concentrations of particles increase the saturation magnetization of the fluid, thus allowing strong forces to hold the fluid in place. This magnetization may be measured in Gauss.

Presently, ferromagnetic fluid is commonly available in 100 to 200 Gauss concentrations. This corresponds to fluid which contains by volume, 1.8 to 3.6 per cent magnetite particles. However, even the highest conductivity fluids, which contain the most magnetite particles, do not contain enough magnetic material to substantially influence the operation of the speaker magnet circuit. The viscosity of the fluid is essentially controlled by the viscosity of the liquid carrier, because the concentration of particles is so low. This is convenient because a wide range of viscosities is necessary to suit different types of speakers and design goals.

Figure 1. The effect of various viscosity fluids on the damping of a mid-range speaker.
cps = Centipoise; a CGS (centimeter-gram-second) measurement of viscosity.



D. B. Hathaway is Director of Research & Development at Epicure Products Incorporated.

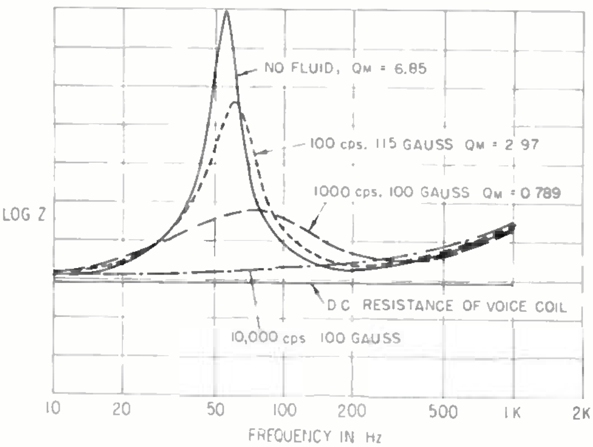


Figure 2. A Modulus of Impedance Curve (Log Z vs. Frequency) may also be used to determine the effect of ferrofluids on voice coil motion. Note the decrease in Q_M as the viscosity is increased.

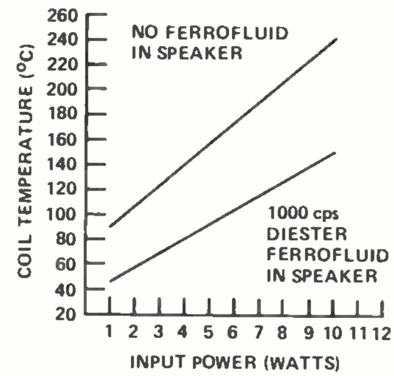


Figure 3. Voice Coil Temperature vs. Input Power.

DIESTER LIQUID

An oil-like synthetic liquid called diester is the carrier most commonly used for ferromagnetic fluids in loudspeakers. Diester liquid is a high-temperature, oxidation-resistant lubricant with very low vapor pressure between —10°C and 100°C. Diester liquid will evaporate and eventually boil as its temperature exceeds 200°C. Like many oil-like substances, its viscosity is exponentially temperature-dependent. The diester liquid also conducts heat approximately five times better than air.

To summarize, the magnetization (expressed in Gauss) which is responsible for the strength with which the ferrofluid holds itself in a magnetic field is determined by the concentration of ferromagnetic particles in a fluid.

VISCOUS DAMPING

The liquid carrier determines the viscosity as well as the evaporation rate and boiling point of the fluid. The lubrication qualities of a fluid's liquid carrier provide viscous damping of a voice coil's motion. Viscous damping is a constant and linear resistance to motion, which is proportional to the velocity of the voice coil. Viscous damping is unlike many previous forms of damping used in speakers, in that there is no tendency towards energy storage or sticking of voice coil motion, such as can be encountered when grease or other sticky material is used for damping.

Because viscous damping is proportional to velocity, the damping applied is frequency selective; that is, damping is applied in the frequency range where the voice coil has its highest velocity.

A dynamic speaker's coil and diaphram velocity increases at 6 dB/octave below motional resonance and decreases at 6 dB/octave above motional resonance. This means that damping is applied to the moving coil in the area of its natural resonance frequency. (It has also been demonstrated that the use of ferrofluid may damp high-order parasitic resonances in speakers, which are fed back to the voice coil bobbin.)

The amount of damping depends on the viscosity of the fluid used and the geometry of speaker gap and voice coil. The amount of damping provided by various viscosity fluids in the gap of a mid-range speaker is reflected in the SPL curves shown in Figure 1.

SMALL-THEILE PARAMETERS

One way to quantitatively measure the amount of damping provided by various fluids, is to investigate the loudspeaker's Small-Theile parameters. These parameters describe the motion of the speaker's voice-coil moving system. Measurement of the Modulus of Impedance curve of a speaker describes (among other things) how much the motion of a voice coil is damped electrically by the magnetic field force between the voice coil and permanent magnet, and how much it is damped mechanically by the resistance to motion of the mechanical suspension and air load on the surfaces of the diaphram. In most dynamic speakers, fluid-viscous damping will predominate over the other forms of mechanical damping. A measure of the mechanical damping, expressed as Q_M, then provides a measure of the effect of the fluid, as seen in Figure 2.

Viscosity of the fluid is also temperature dependent. This means that the damping and Q_M is also temperature dependent. Therefore, in order to arrive at the viscosity of fluid necessary to achieve the desired amount of damping, all testing must be done when the fluid has reached its actual operating temperature. This can be accomplished by testing the entire speaker structure after it has risen to the fluid operating temperature. Selection of a fluid evaluated cold may result in insufficient damping at operating temperature.

If a speaker operates at a low natural resonance frequency, large voice coil excursions will occur. The resulting motion of the dust cap or dome causes air pressure on the ferrofluid which forms an air seal in the speaker gap. Venting (which may have other advantages as well) through the dust cap or center pole is then necessary. If very high excursions are present (such as might occur in high power low frequency woofers) a fluid with higher magnetic saturation may be necessary.

Because of the convergence of magnetic lines toward the center pole of a magnetic circuit, the fluid obtains a greater magnetic force near the center pole face than at the outer gap face. This difference in force at the inner and outer pole faces will symmetrically force a voice coil towards the outer pole face, thus providing a force which tends to center the voice coil in its gap.

HEAT SINKING

Ferrofluid also has the important benefit of lowering voice coil operating temperature. Since the fluid can conduct heat many times better than air, the metal and magnet structure becomes a more effective heat sink for the voice coil. This has two benefits. First, there is a direct lowering of voice coil temperature as the temperature of the voice coil seeks equilibrium with the temperature of the much larger thermal mass in the magnet and metal work. Second, the time it takes the voice coil to transfer heat to the speaker structure is greatly reduced. The time required for thermal equilibrium between voice coil and speaker structure is the inverse of the thermal conduction between the coil and speaker structure. This means that—depending on the amount of thermal conduction that the fluid provides in a loudspeaker gap—the heat generated in the voice coil by an electrical power transient may be sunk into the speaker structure and not fully felt by the voice coil itself. The exact thermal conduction between a voice coil and structure is determined by the amount and position of the fluid, and the size and geometry of the air gap and voice coil. If the temperature of the voice coil (which can be found by monitoring its rise in resistance) is plotted as a function of input power at a constant frequency, a relationship such as that in FIGURE 3 is found.

The inverse slope of the curve in FIGURE 3 is related to the thermal conductivity or the amount of heat that has moved from the voice coil to the speaker structure. The slope of the curve is related to the thermal time constant or the time it takes for heat to flow from the voice coil to the speaker structure.

Plots of coil temperature, made with and without fluid, illustrate the cooling effect the fluid has in a particular speaker.

CONCLUSION

The introduction of a ferromagnetic fluid surrounding the voice coil will lower the coil temperature by sinking its heat to the surrounding speaker structure. This means that for a given coil temperature, more electrical power may be dissipated with fluid than with air as the transfer medium between coil and structure. The voice coil should not be constantly operated above 100°C for prolonged periods of time in order to protect the fluid carrier from evaporation. However, the power requirements made by music are transitory, and steady-state power is very rare and usually accidental.

The best use of the fluid's cooling ability then, is to average out the thermal peaks or transients by allowing the entire speaker structure to dissipate heat, rather than the voice coil alone. This, of course, is best demonstrated when low-duty-cycle pulses of power are used to test a loudspeaker for maximum power handling.

If a voice coil could handle higher temperatures without the use of fluid, then its power handling would be increased by not using the fluid. But very few coils have this capability—which can easily be determined by measuring the maximum input power possible, with and without the fluid. ■

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db Application Notes

Speaker Protection

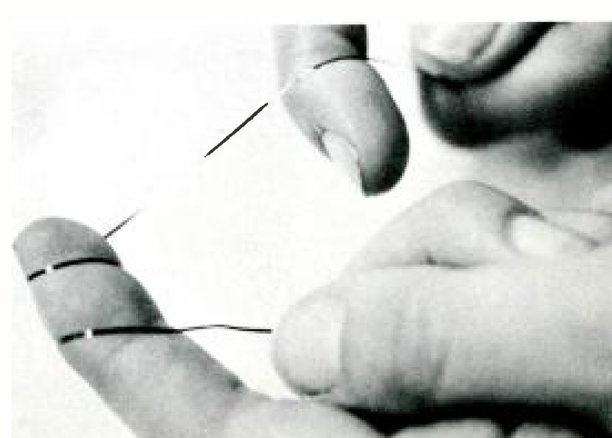
DAVE ROSEN at Audio by Zimet in Manhasset, New York, sells and installs a lot of high-powered audio systems for disco and sound reinforcement applications. Therefore, he's seen more than his share of demolished loudspeakers. Although many such installations are quite extensive, speaker failure is still not an unknown phenomenon. After all, as the night wears on, the disco gets more and more crowded, the volume controls go higher and higher, until suddenly—the sound of silence! More often than not, the voice coil has gone into some sort of self-destruct mode.

The first thing that comes to mind is a failure due to over-heating, but Rosen decided to do a little in-shop investigation anyway, in an effort to reduce his return rate. Using a calibrated strobe light and a signal generator, he noted that in many cases, there was considerable voice coil shift within the gap at high input levels.

A coating of hard epoxy over the voice coil gave better performance at high power, but didn't stop the overheating and voice coil burn-outs.

Here, ferrofluids came to the rescue. During re-coning jobs, Rosen "paints" the voice coil, using a cotton swab. The coil is then re-inserted into the gap. When working on an intact speaker, the dust cover is removed and—using a hyperdermic needle—the ferrofluid is forced between the coil and the air gap. (See Figure 4 of "Anatomy of a Loudspeaker" in this issue for an idea of what's involved. Note that the hyperdermic needle must first penetrate the diaphragm and spider—and *be careful*!—Ed.)

A sample of a voice coil ribbon, prior to winding. Photo courtesy of JBL.

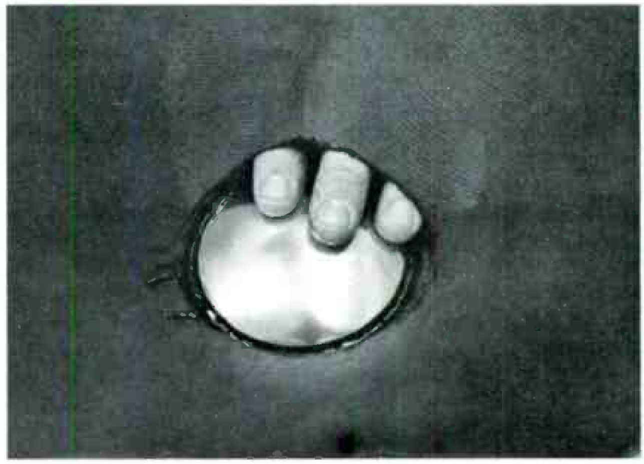


Alternatively, extend the cone forward, and inject the fluid through the lower spider and onto the coil. Although Rosen prefers the cotton swab approach (whenever possible), he feels that coil motion will eventually smooth out any uneven fluid coating caused by the hyperdermic technique.

After treating some 75 coils with ferrofluid, four were returned—one because the spider blew apart, another due to an open coil winding (later traced to a manufacturing defect). One of the other failures was caused when a flexible lead wire snapped, due to excessive movement. As for the fourth, it looked as though it had been plugged into an AC power line, which shows that even ferrofluids do not make speakers idiot-proof.

Of the 95 percent that are still out there blasting away. Rosen even claims to note a slight improvement in overall performance. This is confirmed by an Application Note from the Ferrofluidics Corporation which reports that,

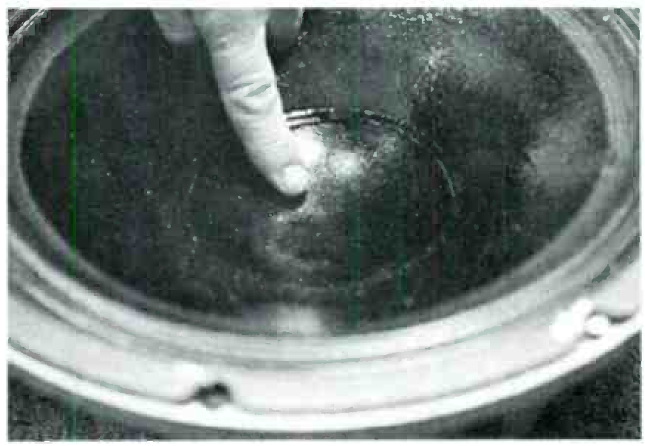
A Gauss speaker cutaway, showing the voice coil in place.



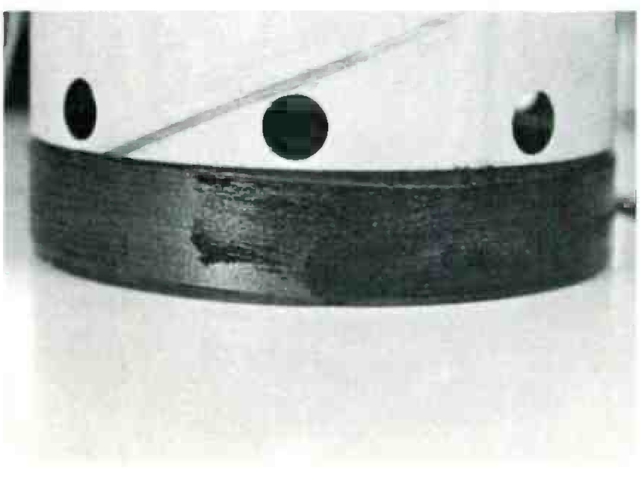
This one definitely needs a re-coning job. Just another example of what can happen when the SPL goes up.

Where's the Rub? Note the extreme wear patterns at the top of the coil.

Note that the new dust cover has been inverted from its usual orientation. Speaker manufacturers may shudder, but Rosen reports better performance.



A new voice coil on its bobbin, partially coated with ferrofluid.



"Some customers report to us an improvement in output efficiency (by as much as 1 to 3 dB) which is contrary to damping principles. The ferrofluid can contribute to the magnetic circuit by reducing reluctance in the gap approximately 1 percent. Normally, that is not significant enough to noticeably improve efficiency. The improvement is attributed to thermal effects. Ferrofluid heat convection controls voice coil temperatures and therefore sustains efficiency during performances at elevated power levels. Testing for this improvement can only be realized when comparing the speakers with and without ferrofluids under actual power-handling conditions, just as would be heard by the music listener."

In any case, keep in mind that although the application of a ferrofluid to your monitor speakers may indeed improve performance and lower your failure rate, the treatment requires great care. Remember, a mis-directed hyperdermic needle can do more harm than good. As we said before, be careful! ■

DANIEL QUEEN

Impedance Matching

Impedance matching loudspeakers and amplifiers, there's no neat rule of thumb.

AT ONE TIME, when power amplifiers and loudspeakers were usually sold in the same box, their relationship was well-known to engineers, who would design amplifiers and speakers to compliment one another. Even in the field of sound reinforcement, where the speaker and amplifier were not in the same box, the loudspeaker designer could be reasonably confident of the nature of the amplifier for which he was designing. However, with the coming of component high fidelity, both loudspeaker and amplifier designers retired to their own worlds—each to assume minimum interaction with the other.

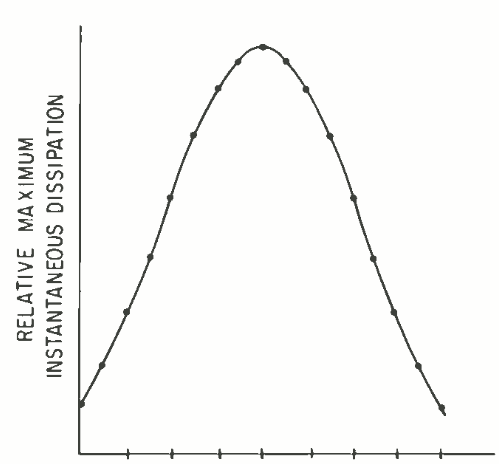
In connecting console components together, the easiest situation to handle is that in which the input of one component bridges the output of the other. Thus, a component should have a very low output impedance, which will provide a constant voltage to the load. The loading component should have virtually infinite input impedance, thus drawing no current from the source.

Lately, the trend in power amplifier design has been in the same direction. Thus, we see damping factors of over 200—that is, the amplifier output impedance is 1/200th that of the loudspeaker load at its nominal impedance. (For more on damping factors, see Theory & Practice, Feb. 1978 and The Sync Track, July 1978—Ed.)

Nevertheless, the opposite choice would have had substantially the same effect if history and coil winding techniques had not brought about the 8 ohm *de facto* standard. Perhaps, had the transformer been eliminated from its critical position in the power amplifier earlier, then we might have amplifiers with 200 ohm output impedances driving loudspeakers with 1 ohm impedances—the amplifier thus providing constant current drive to the source, much in the manner in which a record amplifier drives a tape recorder head.

The fact that high damping factors are not necessarily the optimum way to drive loudspeakers had always been recognized in commercial products. The juke boxes of the 1930s utilized 15-inch loudspeakers driven by relatively high impedance push-pull triodes (and even tetrodes) to obtain a ringing bass while sharply cutting off the response at a frequency high enough to prevent too much transmission of vibration into the record playing chamber—thus reducing tracking problems for the 700 gram cartridges.

Figure 1. As the phase of the current shifts through 360 degrees, the maximum instantaneous dissipation occurs at 180 degrees.



Daniel Queen is head of Daniel Queen Associates, consultants, of Chicago, Illinois.

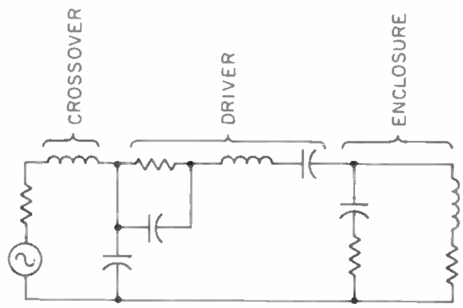


Figure 2. An equivalent circuit for a loudspeaker.

In the late 50's several manufacturers of high fidelity component amplifiers, notably Fisher, provided controls which, by changing feedback, would vary the output impedance thus providing a variable damping factor. In the late 60's, Shure Bros. introduced its Vocalmaster System with an amplifier designed to provide very low damping factor—in fact, current drive—to its loudspeakers, a practice which had actually been used for some years by manufacturers of electronic musical instruments. And now, with the advent of bi-amplification, some loudspeaker manufacturers are offering sub-woofers with built-in amplifiers, providing impedance complements to the impedance of the loudspeaker.

OPTIMUM APPROACH

In his February, 1978, column, Norman Crowhurst ably points out that the operation of an amplifier for maximum power transfer can result in high dissipation in that amplifier. However, a "zero" impedance amplifier is, in effect, operating for maximum *voltage* transfer. It requires a high supply voltage with the result that power is still dissipated in the amplifier. Furthermore, if the phase of the voltage and the current in the load differ, then the amplifier can begin dissipating instantaneously much higher powers than would have been indicated by its operation into a resistive load, as shown in FIGURE 1. Anticipating this, amplifier designers build load-dependent protection circuitry into their amplifiers, so that when the load calls for a voltage and current relationship which would cause high instantaneous dissipation, the amplifier goes into a form of clipping, giving rise to the recently coined term, "chirping."

However, were the amplifier designed to provide current drive, the output devices would have to handle very high current. Again, instantaneous power dissipation could increase to the point where protection is required. Thus, the goal of making the amplifier and speaker independent is not really achievable.

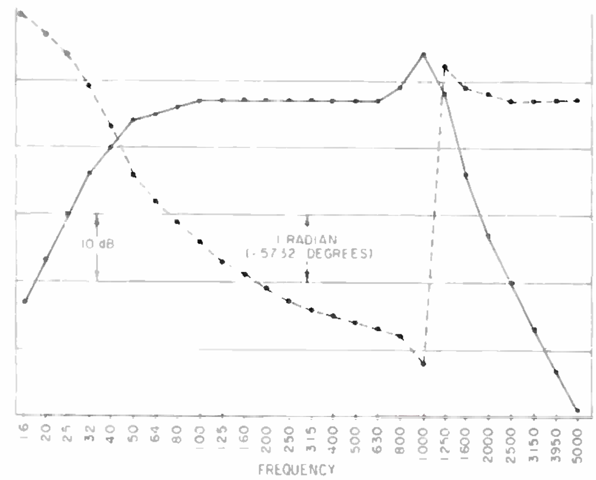
MAXIMUM POWER TRANSFER

This brings up the key problem in the power transfer discussion. In the classic expression, the maximum power available occurs according to the equation:

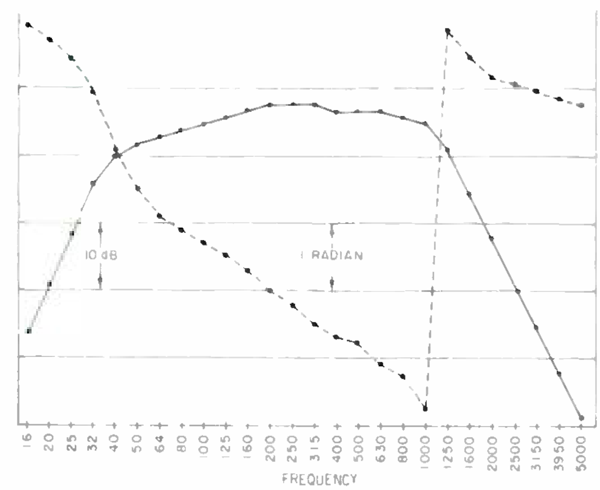
$$W_{MAX} = e_g^2/2(R_g + R_E)$$

where e_g is the source voltage, R_g is the source resistance, and R_E is the load resistance. But note that this equation calls for an equality of the impedances. Actually, maximum power transfer can occur only when $X_g = -X_L$.

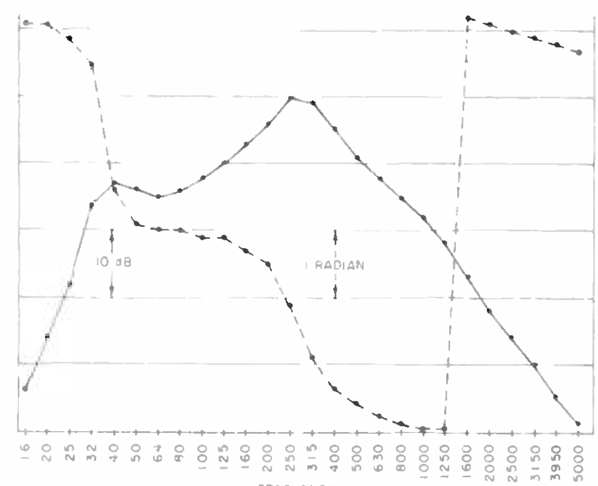
This expression shows that for maximum power transfer to the resistive portion of the load, the reactance of the amplifier must be the conjugate of the reactance of the load. In other words, an inductive speaker must be matched to a capacative amplifier, and *vice-versa*. Thus, the reactances must "cancel out" in order for maximum power transfer to occur.



(A) 0.8 ohm source

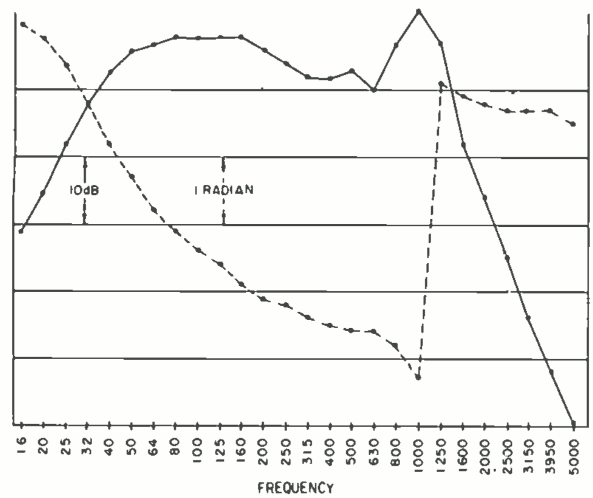


(B) 8 ohm source

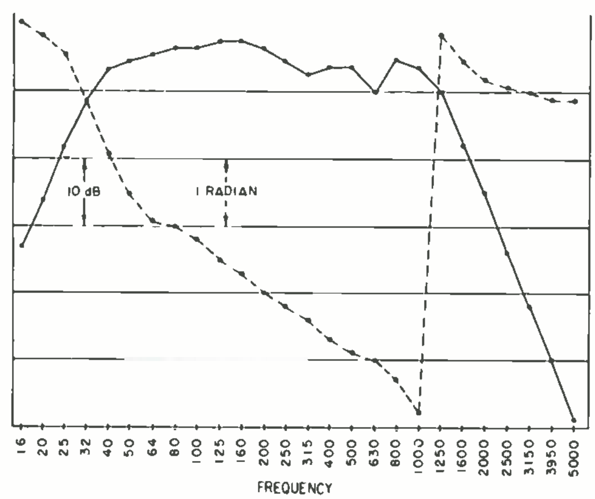


(C) 80 ohm source

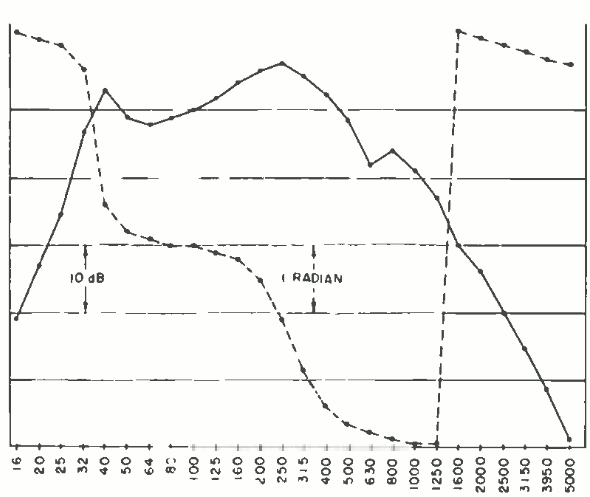
Figure 3. 10-inch woofer in vented box. Solid line represents amplitude response; Dashed line represents phase response.



(A) 0.8 ohm source



(B) 8 ohm source



(C) 80 ohm source

Figure 4. Same as Figure 3, but with a change in crossover frequency. Solid line represents amplitude response; Dashed line represents phase response.

In a long transmission line such as in a telephone network or power cable—where maximum power transfer is a necessity in order to reduce standing wave losses—loading coils are used to achieve and maintain this relationship. However, in the lines much shorter than the wavelengths encountered in the connection of loudspeakers, such techniques are seldom possible (except perhaps in very long line distributed systems).

A look at a simplified equivalent circuit, in FIGURE 2, for a loudspeaker shows how difficult it would be to make the amplifier look like a loudspeaker.

SOURCE IMPEDANCE

The effects of the output impedance of the amplifier on the loudspeaker are evident in the equation given earlier, which shows the loudspeaker efficiency. The amplifier resistance obviously affects the Q of the circuits. In FIGURE 3 we see computer-plotted responses of this network for three source impedances; (A) near zero ohms, (B) equivalent to the d.c. resistance, and (C) ten times the d.c. resistance. In terms of the average deviation from flatness, the 8 ohm source impedance case [FIGURE 3(B)] might be thought to be more desirable, although its effects on transient response would have to be evaluated, particularly where the sharp slopes are encountered. Nevertheless, the resonant peaks in the response shown are really not serious enough to predict any substantial transient problems. One could say that this loudspeaker was designed for current drive rather than voltage drive.

In FIGURE 4, a change has been made in the crossover frequency. Now, the difference between low impedance drive and high impedance drive creates a problem in the crossovers. Since protection circuits tend to raise the output impedance of the amplifier when they operate, this is a possible source of the "chirp" often experienced.

In a direct radiator designed entirely for low frequencies, the equivalent circuit of FIGURE 2 can be greatly simplified to show essentially only the moving mass and suspension compliance of the loudspeaker as the principal reactances. One can then design the amplifier to provide the conjugate output impedances—in this manner essentially canceling out the resonance frequency of the loudspeaker and allowing improved low frequency response. Today—using active filter concepts in the amplifier feedback network—this is achieved with relative ease, and is embodied in some recent sub-woofer designs. The approach is not new; it was put forward more than ten years ago by Keith Johnson at an Audio Engineering Society meeting.

A POOR MATCH

A direct radiator loudspeaker is a low efficiency device because it is poorly matched to its load (the air). To afford maximum power transfer from the loudspeaker diaphragm to its load, horns are used. However, once the loudspeaker-to-air transfer is maximized, the effect of amplifier-to-loudspeaker transfer becomes greater. In *Acoustics*, Beranek showed that ten-fold improvements in horn loudspeaker efficiency can be achieved by matching the amplifier to the horn driver.

In this case, the problem of loudspeaker reactance is also minimal, since the masses and compliances of the loudspeaker are largely swamped by the throat impedance of the horn; so that a well designed horn provides an essentially resistive load to the amplifier—making more practical the goal of maximum power transfer.

Thus, impedance matching, which is a virtual necessity to achieve power flow through transmission lines, can be a versatile design tool in achieving the flow of power from an amplifier to a loudspeaker, but cannot be approached with the rule book in hand. ■

db Speaker Spotlight

In a field, such as the audio industry, one which is constantly changing—we at **db** endeavor to keep our readers abreast of the latest developments and current practices.

Since this issue is centered squarely on the topic of loudspeakers (design, response characteristics, use of ferrofluids, etc.), it would certainly be negligent and perhaps down-right foolish on our part, if we didn't take this opportunity to unveil a dramatic break-through in loudspeaker design and performance.

Sure there are those who'll complain that for every loudspeaker currently on the market, there's a new and different design. But we feel confident that, upon careful examination of the Engineering Data for the new "Rearaxial Softspeaker" by Electro-Voice, you'll quickly come to understand the *significance* and possible *impact* such a device could have, in turning the whole audio industry on its "?".

S.Z.

FEATURES

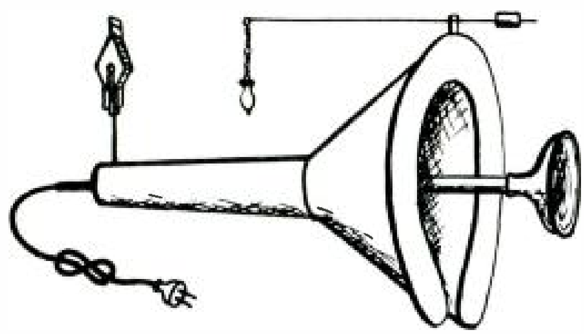


Fig. 13 — Model SP13.5TRBXWK

IT'S HEAR!

NEW! THE SPEAKER THEY SAID "COULDN'T BE MADE"

You won't believe this! WOW!

Top Extra Highest Fidelity!

Unprecedented! Superb! Unexcelled! Amazing!

Unnerving! Undirectional!

Unnatural! Customized! WOW!

Unequivocally, undeniably, doubtlessly, without question and really this is the most amazing, remarkable, revolutionary, sensational new development ever revealed, ever, honest! When you see this new terrific unit you wont't be able to contain yourself! WOW! Silky highs-woolen lows! Complete with new vital "presence" and "absence" controls! Your friends will scream with envy when they hear this new speaker! Amazing new exclusive 3-way, 12-D, binaural relief port eliminates spurious propagation of intermediate, and undesirable biped, tertiary grid-leaks! No messy floors to mop! WOW! This new reproducer incorporates a new, ridiculously simple principle which our engineers can't explain yet! 13 extra octaves of added bass when the unit is coupled to a bowl of oatmeal. Think of that-just think of that! WOW! Complete, self - contained including new "wow" filter; nothing else to buy! (See page 2 for accessories.) Comes with new 5" cable and new genuine combination "Good-luck" charm and stylus pressure gauge!

SPECIFICATIONS

Frequency Response: DC to middle of Channel 5 ± 3 inches (See Fig. 39.)
NRA Sensitivity Rating: 98.6°
Free-Space Cone Resonance: Huh?
Power Handling Capacity: 110-220V 25 cycle AC-DC 3 phase
Critical Damping Factor:
In an infinite Baffle: .001
In recommended orange crate: WOW!
Distortion: Don't mention that word
Magnet Weight: 3 tons
Size: 5'4" wide x 17' narrow x 13 1/16" high x $7 short
Mounting: 13 miscellaneous size holes randomly spaced at uneven intervals in a haphazard way
Net Weight: 73 tons
Shipping Weight: 68 tons
Price: $7,907* Audiopill Net, F.O.B. your nearest tar pit. Comes completely unassembled in three box cars

Mr. Fafnir N., Horse Cave, Ky.
"Send me another—my canaries love it—it's for the birds!"

Miss Brunhilde S., Horse Cave, Ky.
"This thing scares me!"

Mr. Clyde T., Horse Cave, Ky.
"I'd send it back, but the postmaster won't touch it."

Mr. Wong Wong Ago, Horse Cave, Ky.

Fig. 5 — Actual Testimonials from Satisfactory Customers

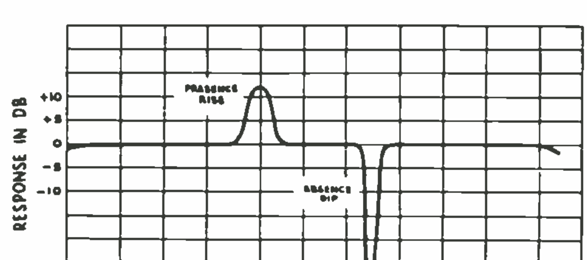


Fig. 3 — Frequency Response at 791-watt Level

*Zone 2 includes: West Texas

ENGINEERING DATA

SP13.5TRXWK Rearaxial Softspeaker

INSTALLATION

If you are left handed, start with step "f" and reverse procedure.

a. Connect the 300-ohm twin-lead from the UHF Antenna to the terminals marked VHF ANT on the rear of the chassis.

b. Connect a short length of 300-ohm twin-lead between the terminals marked UHF REC on the rear of the front chassis and the UHF antenna terminals marked AFC on the front of the rear chassis.

c. Connect the 300-ohm lead from the UHF antenna to the terminals marked WOW ANT on the side of the front chassis.

d. Connect a short length of 300-ohm twin-lead between the terminals marked VHF REC on the front of the side chassis.

e. Connect a short length of 300-ohm twin-lead between the terminals marked TV REC on the side of the rear chassis and the UHF antenna terminals marked REC ANT on the receiver chassis.

f. Connect the VHF antenna to the terminals marked FM ANT on the rear of the rear chassis.

NOTE: If a short length of 300-ohm twin-lead is not available, use two short lengths of 150-ohm twin-lead, or cut a long length of 600-ohm twin-lead in half. If more than one UHF station is to be received, the automatic on-off switch may still be used as outlined above under Automatic "Operation." If this feature is not desired the front panel on-off switch may be used and the AC receptacle on the rear chassis disregarded; in this case, disconnect a short length of 300-ohm twin-lead.

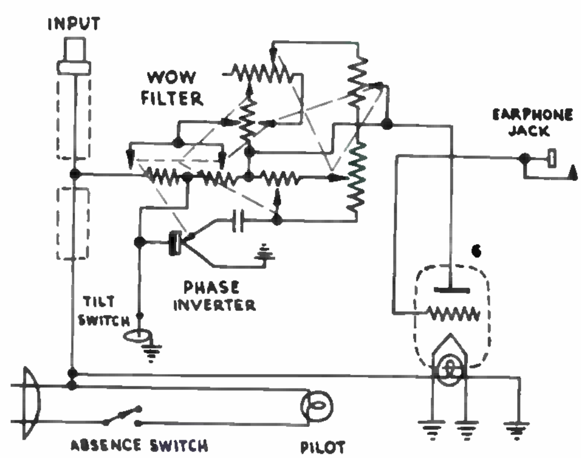
When connecting interconnected connectors, take care to connect the connected connecting connectors to the unconnected connections, interconnecting the disconnected connectors with a short length of 300-ohm twin-lead. Otherwise constrampulation of the transmognifactor will cause interpolation of the controcacoustic control. WOW!

CAUTION NOTES

1. Do not operate this speaker within 200 feet of Aardvarks and 3-toed Sloths as this thing will cause untold agony and immediate disintegration to the aforementioned above.
2. It is imperative that all inputs and outputs are grounded so that output is reduced to the doorstep of pain when used indoors.
3. Close cover before striking.
N.B. Unless instructions are carefully followed with utmost caution and circumspection, shock waves and turbulant buffeting may be experienced and the darn thing won't work.

ACCESSORIES

Model BRAK-1. 75-ton derrick. Dainty and inconspicuous for quick, easy mobility when moving unit for dusting, relocating, mobile operation. Audiopill Net (with steam engine).......$7,907
Model BRAAK-1. Furnishing kit. For interior of packing crate. Use this handy little accessory to convert the SP13.5TRBXWK packing crate to an additional room for your home. Comes in Battered Blonde, Weather-beaten Mahogany, and Box Car Red. Audiopill Net.................. $790.70
Model BRAAAK-1. Birdcall adapter. Special sonic generator either attracts or repels birds of all types. Complete with 1,239 position bird selector switch and 7x50 coated binoculars for watching birds come and go. Please specify birds desired. Audiopill Net................................ $79.07



Schematic Diaphram

ELECTRIC-VOICE, LTD. / Largest makers of ear trumpets and stuff in the country.
Electro-Voice, Ltd., 7 North Southwestern Ave., East Westerly, South Dakota
Plants located in Horse Cave, Ky., and Kinnikinnick, Connecticut
Shopping for Real Estate in Horse Cave, Calif.
Cable Address: Madglockenschpielermitgongennoisenlab

Part No. A7907
Stock No. 7907
Form No. HB-6
Bulletin No. 21
Run No. 12218L
Phone No. 1000

Printed in Lhasa, Tibet

MICHAEL P. ROGALSKI

Murphy's Theorem Applied to P.A. Work

A neurotic sound fantasy/experience.

EARLIER, you were very secure. Naturally—you had a nice safe job as a bench technician. But now, you are a P.A. sound man for the first time. Why? Simple; there weren't enough field service people in your company to handle the entire National Convention being held in a Tijuana border town. So here you are, poised at the controls, awaiting the first words of the first speaker, reflecting upon the circumstances and nightmares which brought you to this half-way mark. Somehow, you've managed to get all the equipment set up, and now, for the production part—the actual event. As you reflect upon what it took to get you this far, the bad memories come back to you again—how you quoted "An extension cord! An extension cord! My Kingdom for an extension cord!" when you found that your power line was only four feet away from the outlet before you ran out of the means to reach it. And when someone did contrive an extension cord for you, it was the three prong variety. You didn't have a three prong adaptor. And then, being that it was in a border town, you didn't have a step-down transformer for the P.A. equipment you brought. Of course that was minor. The troubles really began when the people running the conference told you where they wanted you to put your equipment. Then, the hotel's "sound man" (he has a degree in food management) insisted that the podium must be immediately under the ceiling speakers because "it looks nice."

Fortunately, you brought your own equipment with you, including power amps and speakers. You did so because your boss insisted it wasn't necessary, since there was an in-house sound system. [You had already memorized Murphy's Laws on In-House Sound Systems.] What nobody noticed was that the local help utilized the ceiling speakers for a rock concert during off-hours and pieces of cardboard fall through the grillwork whenever the system is in operation.

Of course, you made sure to set up two hours before the program. Of course, the program director arrived ten minutes before the program, to let you know that you were going to have to record it. He brought along a tape recorder, but no cables or mikes, and you have to wire into the chassis of the machine (manufactured in Albania) because the input jacks are a rejected version of a serbo-Croatian standard d.c. power plug.

There was no dress rehearsal for this program. All you can decipher from the program sheet, is that the topic is Yoga, and there are four speakers. You assume that each speaker will need a lavalier mike, because of apparent mobility. Naturally, you are right. Naturally, each speaker leaves his lav. mike on the podium, and then moves back three feet, to speak in a spiritual-sounding manner that can best be described as Inverse Fletcher-Munson, divided by the negative square root of infinite articulation loss. People place their ears against the wall, in search of a better sound medium, while gazing at dummy you. And since they said the sound man must not be too obvious, you cannot even see the speaker, to make use of your newly found lip-reading skills.

The next man to speak stands at the podium, picks up the only condenser lav. mic, bangs a nubby forefinger on the diaphragm to confirm that it is working, and then wonders why his voice sounds fuzzy. His speaking is punctuated in all the proper places by microphone failure.

Dust, and more falling cardboard, concludes his speech as the resident dishwasher turns on the A.M. rock he favors, and generously distributes it to all of the rooms in the meeting hall as well as to the speaker over his dishwashing machine. Just as you reach the cleverly concealed house P.A. system closet and turn off the power to the whole system, a blaaaah sound emanates through the meeting room. When you recover, you realize it is a klaxon horn for shift change, and not the dying embers of your power amps. You are of course, unshaken, for this is typical of what can go on at such affairs. The minister gives the invocation for the dinner banquet, beating everyone else to the meal by prematurely devouring the microphone and spitting it out on alternate syllables. Your job is done. And if you've done it well, no one will never know you were there. Except you, your psychiatrist, and three gastrointestinal specialists. ■

Michael P. Rogalski is audio engineer for Philadelphia Community College and does free-lance design.

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People/Places/Happenings

• Joining **Koss Corporation** as research engineer, **Dr. T. Don Mathis** will be involved in acoustic engineering research for the firm's stereophone product line. Dr. Mathis was formerly a member of the technical staff of **Bell Labortories** in Atlanta, Georgia.

• In a major realignment within the sales organization of **Switchcraft, Inc.,** sales of products to original equipment manufacturers have been grouped into three businesses, with **Kenneth Kline** appointed marketing manager, Plug and Jack Products; **Ronald Larson** marketing manager, Microphone and Audio Connectors and Assemblies; and **Ken Yerama** named marketing manager, Switch Products. Others named in key marketing appointments were: **Ronald Pitchford,** distributor sales manager; **Fred Fitzpatrick,** telecommunications marketing manager; **George Zib,** advertising and sales promotion manager; and **Norman Luksik** appointed administrative coordinator for the marketing department.

• Joining **Magnetic Controls Company,** Minneapolis, Minn., as vice president, **John S. Donnelly** will oversee the sales, marketing and engineering functions of the company. Mr. Donnelly previously was senior vice president of operations at **Graco Inc.**

• **Abe Voron** has been named permanent convention agenda chairman for the **National Radio Broadcasters Association** (NRBA) in addition to his present duties as NRBA's executive vice president/government affairs. **Tim Ives,** president of **Bloomington Broadcasting** (Bloomington, Il) and an NRBA regional director, will head the newly formed exhibitor committee.

• **James P. Broderick** has been appointed audio product manager for the audio-video systems division of **Ampex Corporation.** Mr. Broderick returns to Ampex after six years at **Scully Audio,** where he held the position of international sales manager.

• An established 16 track studio the past five years, **Recording Associates** of Portland, Oregon has increased its track capability to 24, in addition to enlarging and re-designing the control room of Studio A.

• In his new capacity as president of **TEAC Corporation of America** (TCA) **Norio Tamura** will direct operations of TCA and provide liaison with TEAC Japan. Mr. Tamura, previously director of the international department of TEAC Japan, replaced **Masaji Takahashi** in the top post. A nine year veteran of TCA, **Tay Hotta** has been named assistant to Mr. Tamura.

• **Ewald J. Consen** has been appointed vice president, marketing of **United Recording Electronics Industries (UREI)** in Sun Valley, California. Mr. Consen was formerly with **JBL** as their national field sales manager, professional products division.

• **Signet** has named **John D. McGurk** assistant sales manager. Prior to joining Signet, Mr. McGurk was national sales manager for **Adcom,** New Brunswick, NJ.

• **Philip R. North** has been named chairman of the board and president of **Tandy Corporation,** parent company of the **Radio Shack** store chain. Mr. North, director of the Tandy Corporation since 1966, succeeds **Charles D. Tandy,** founder of the company, who died November 4, 1978. **John V. Roach,** named executive vice president of Radio Shack, will be responsible for overseeing and coordinating all phases of the company's operations worldwide.

• Based in the Philadelphia area, **Paul W. Spillane** has been appointed to the position of eastern area manager for the OEM field sales organization of **Amperex Electronic Corporation.** Mr. Spillane will be responsible for the sale of Amperex and Ferroxcube products, including semiconductors, electron tubes, hybrid integrated circuits, electro-optical devices, ferrites, piezoelectric ceramics, ceramic permanent magnets and loudspeakers.

• Leaving **Bolt Beranek and Newman, Inc., Dr. J. Jacek Figwer** has established the consulting firm of **Figwer Associates, Inc.** The firm, based in Concord, Massachusetts, provides consulting services in architectural acoustics, the design of professional sound systems, noise and vibration control and audio-visual systems.

• **Sony Corporation of America** has established a new digital audio products division, naming **Roger Pryor** general manager and **Louis Nanassy** manager of engineering. In addition, **Jason Farrow** has joined the Sony Corporation as advertising manager for high fidelity and digital audio products.

• In a change at the top management level at **Soundcraftsmen, Inc.,** the entire sales division responsibilities and general management will be assumed by **Ralph Yeomans,** president and chairman of Soundcraftsmen, Inc., and by **Paul Rolfes,** vice president; as **Charles "Chuck" Gassett,** general manager of the domestic and international sales division leaves the company to pursue other business endeavors. Sales offices will be consolidated with present corporate headquarters and manufacturing facilities at 2200 South Richey, Santa Ana, CA 92705. (714) 540-4961.

• Resulting from an increased need for product training on both the representative and dealer sales levels, **Bernard J. Gaffney** was appointed to the newly created position of national sales representative/audio visual products group for **Telex Communications, Inc.**

• Appointed sales representative for **RCA Broadcast Systems, Donald R. Musson,** based in Wellesley, Massachusetts, will be responsible for the sale of radio and television studio and transmitting equipment in the New England States.

• Coming from **Infinity Systems, Inc., Denis A. Wratten** has been appointed to the position of executive vice president of **KLH Research and Development Corporation.** Developing long-range programs for new products at KLH will be the responsibility of **Frank Jones,** named vice persident of product development.

• Named market development manager by **3M** Company's Magnetic Audio/Video Products division, **Dennis A. Farmer** will head up the marketing and promotion of audio and video recording tapes in the professional recording and broadcast fields. Mr. Farmer was formerly sales manager for the 3M division in Chicago.

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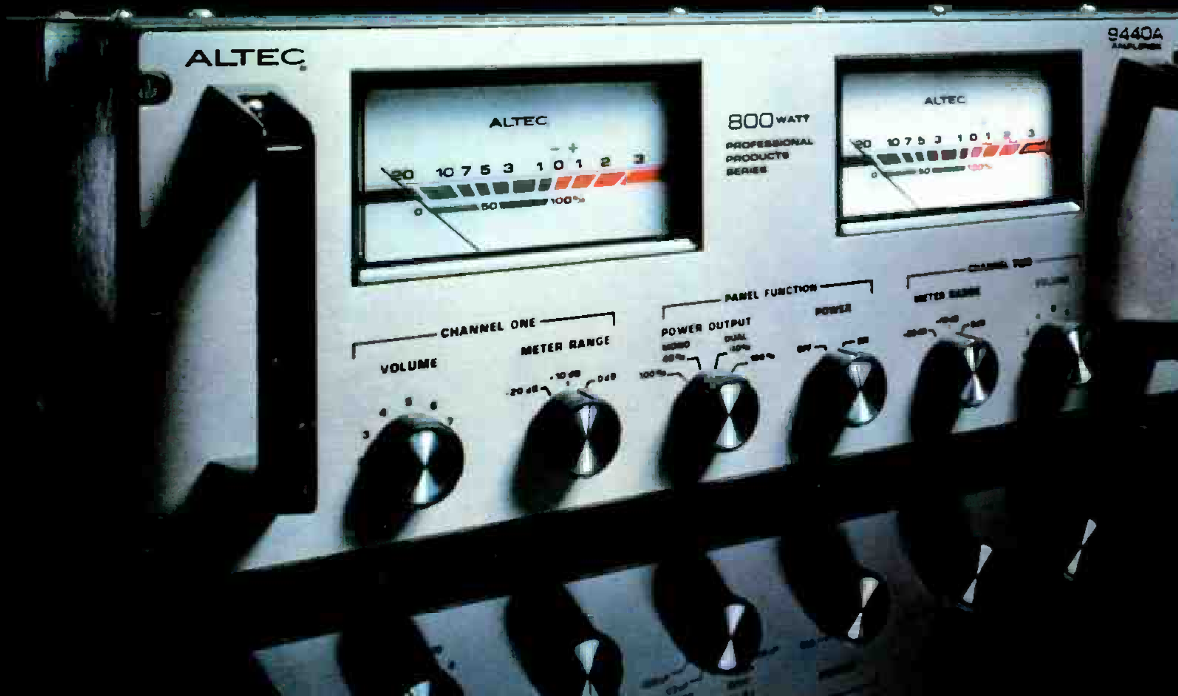

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