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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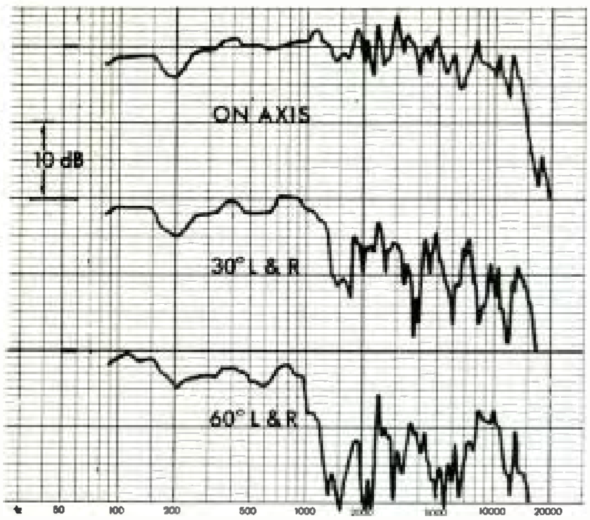


Figure 1. A selection of anechoic frequency response curves for a popular studio monitor speaker. Below about 200 Hz the output is dependent upon mounting and room acoustics.

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 mid-ranges. 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.

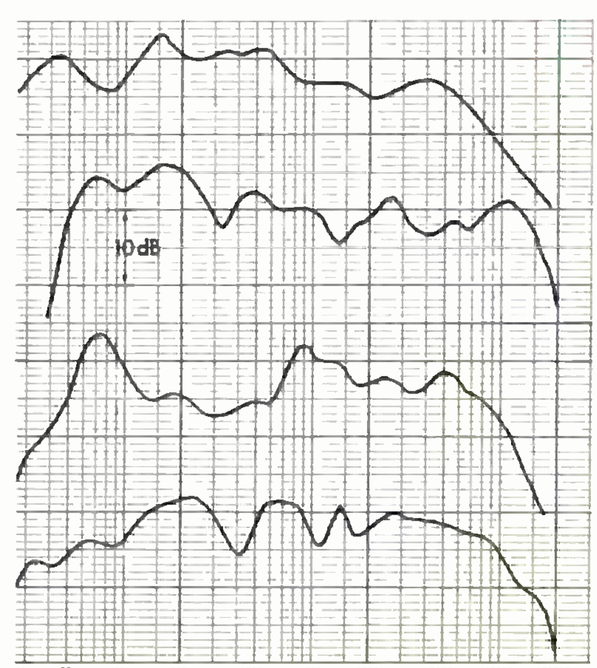
ANECHOIC TESTS

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.



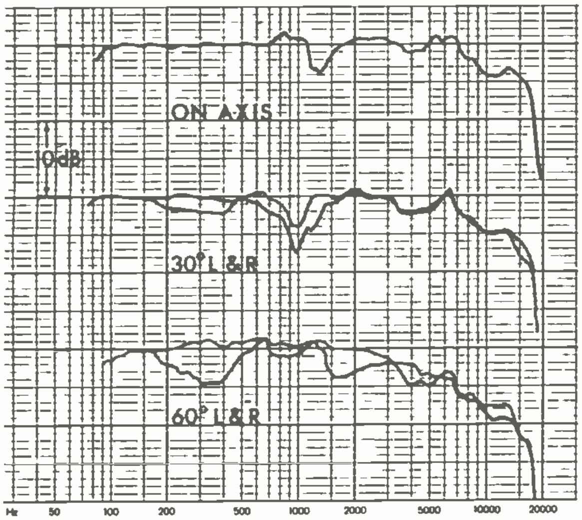


Figure 3. A selection of anechoic frequency response curves for a prototype studio monitor speaker.

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.

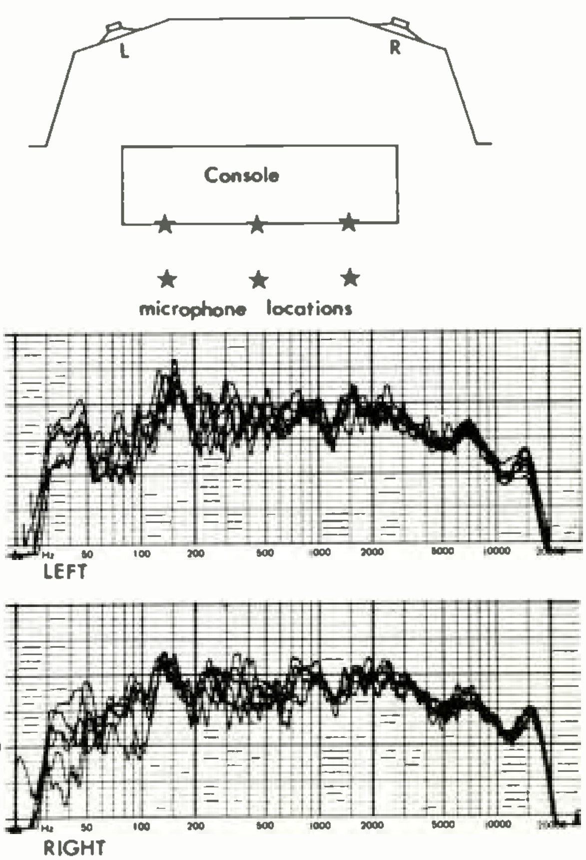
As part of a new recording studio project an opportunity arose to try out the high principles being expounded here. Using the same criteria that one would use in designing a hi-fi speaker (flat frequency responses, low distortion, wide dispersion, clean transient response) a number of speaker units were 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.



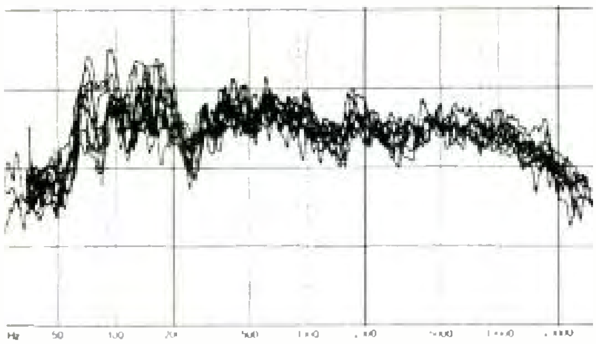


Figure 5. As Figure 4, but with a good domestic speaker substituted for the right monitor. Nine mic locations were used.

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-

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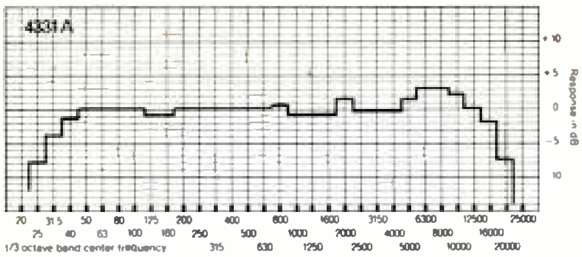

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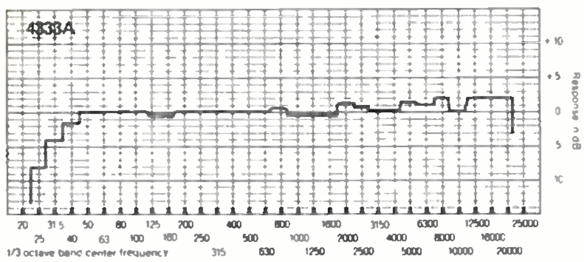

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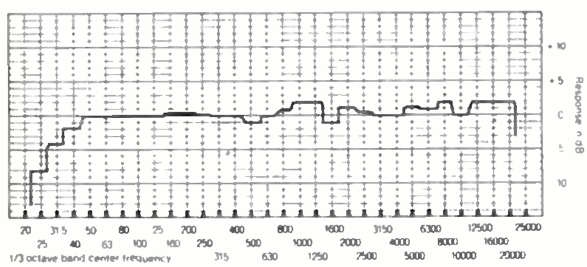


(C)

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)

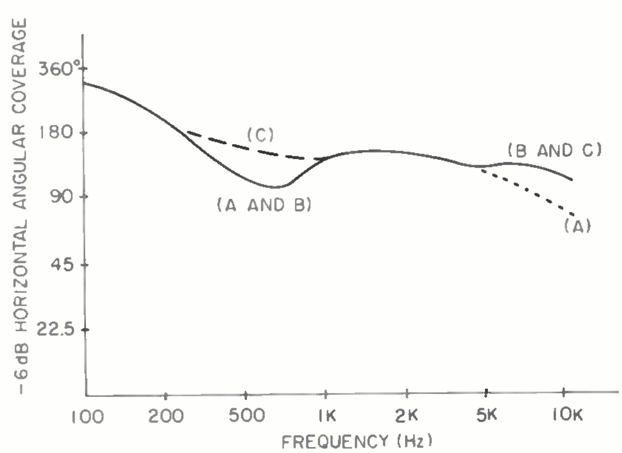


Figure 3. A three-way enclosure system. Note the UHF driver to the left of the regular high-frequency system. (JBL 4333)

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

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 frequency was then measured graphically. This slope ($d\varnothing/d\omega$) 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.

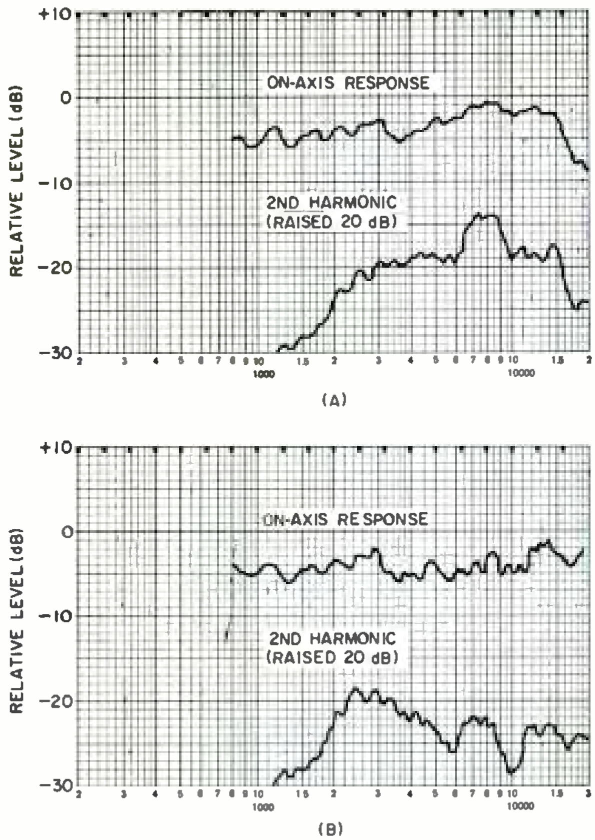


Figure 5. Harmonic distortion in two- and three-way systems.
(A) Two-way system (JBL 4331)
(B) Three-way system (JBL 4333)

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)

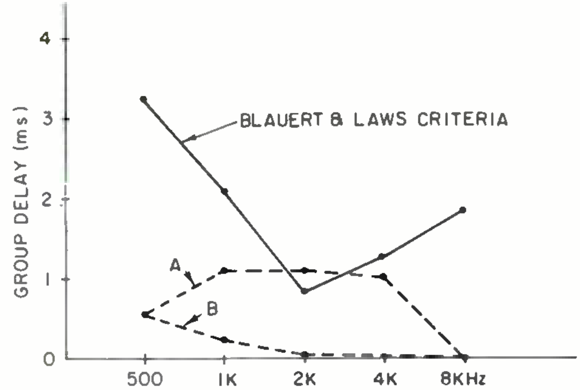


Figure 7. The Blauert and Laws criteria for non-audibility of delay effects.
(A) Time domain response for a professional three-way system (JBL 4333).
(B) Time domain response for a bookshelf three-way system (JBL 4313).

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.

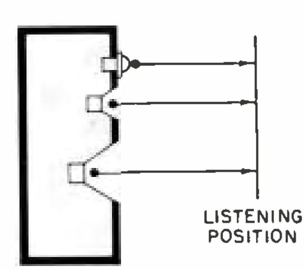
FIRST, a word about the subjective and objective aspects of loudspeaker systems. An artist spends most of his working time in the "subjective" domain. A scientist (or recording engineer) spends most of his time in the "objective" domain. Objective aspects are measurable, quantifiable and repeatable. 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.

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.



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

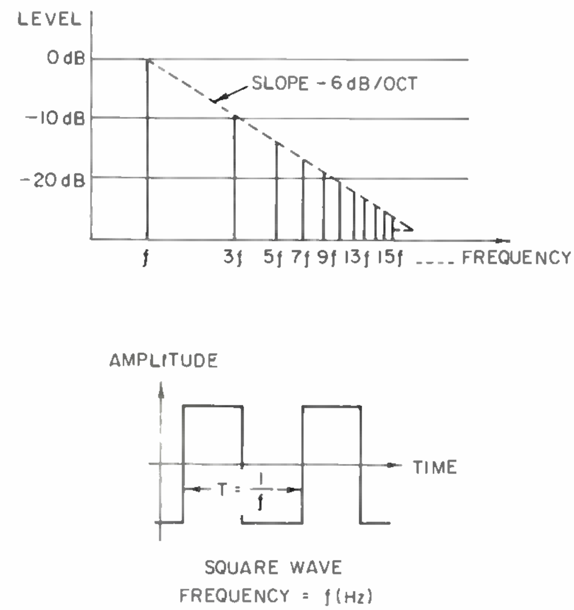


Figure 2. Spectrum of square wave.

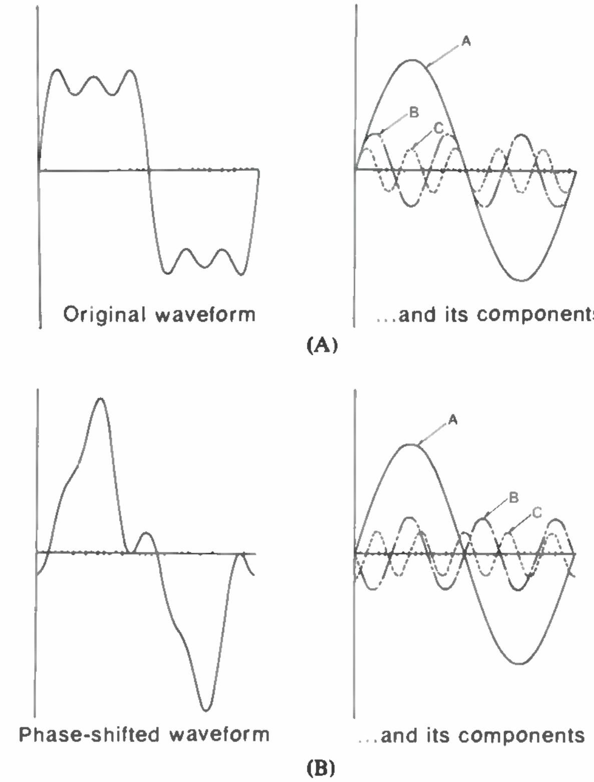


Figure 3. Effect of phase shift on waveform.
(A) Top diagram: Original waveform and its components.
(B) Bottom diagram: Phase-shifted waveform and its components.
A=Fundamental
B=Third Harmonic
C=Fifth Harmonic

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:

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

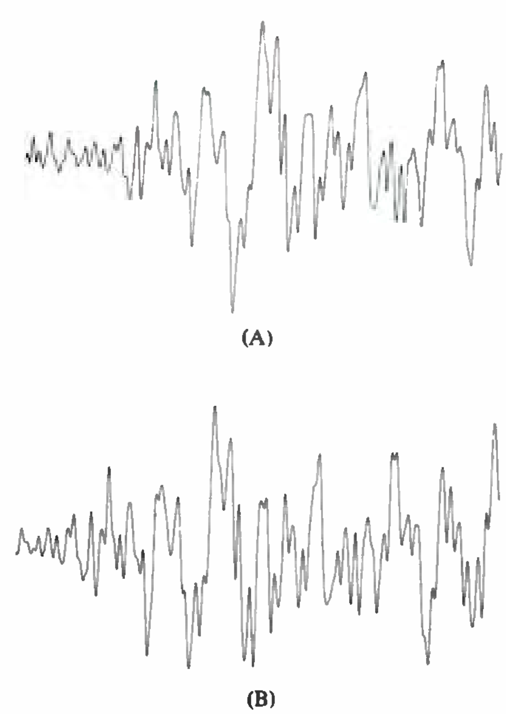
This means that a square wave can be generated by taking an infinite number of sine wave generators, with amplitudes and frequencies set exactly as the equation dictates. In other words, the amplitude of the third harmonic ($\sin 3\omega t$) is one-third the amplitude of the fundamental ($\sin \omega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



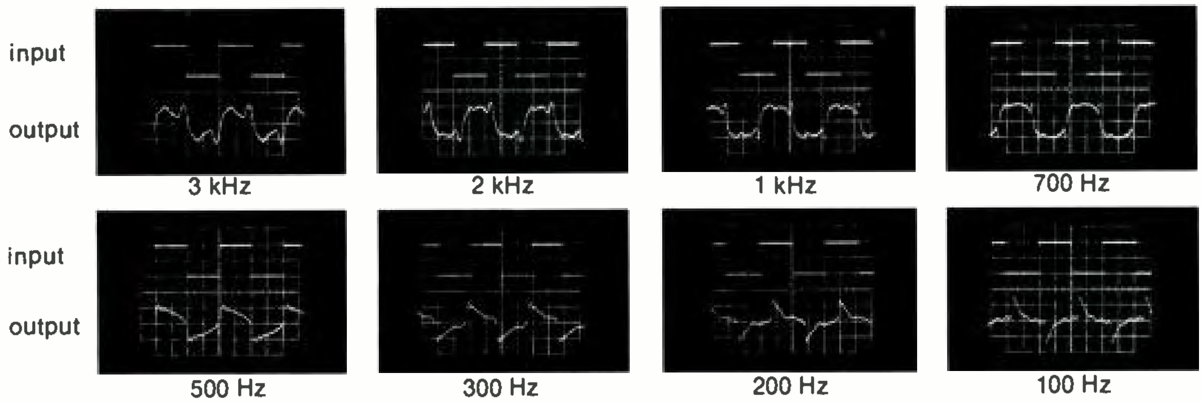


Figure 5. Square wave responses of a linear phase loudspeaker system.

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.

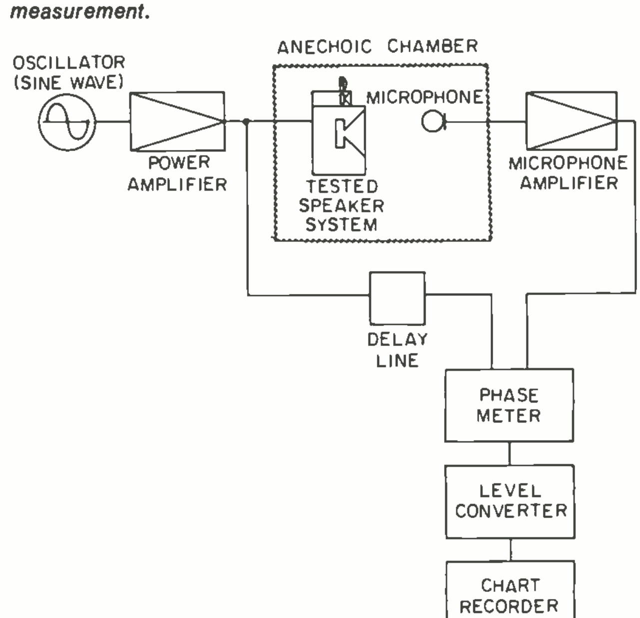
After careful attention is given by the designer to all of the above mentioned details, the final product can be measured. There are no universally accepted measurement standards for waveform fidelity, but there are three ways of showing it that will be discussed. Of course, an actual musical signal 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.



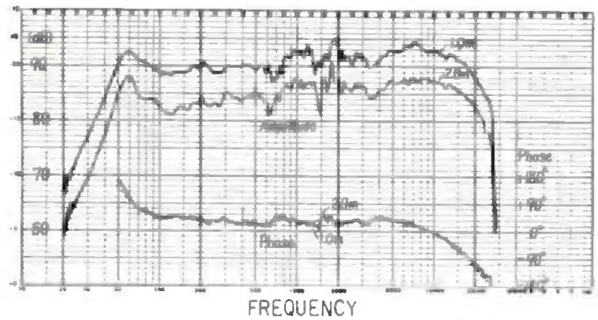


Figure 7. Amplitude and phase response for a linear phase speaker system (Technics SB7000A).

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.

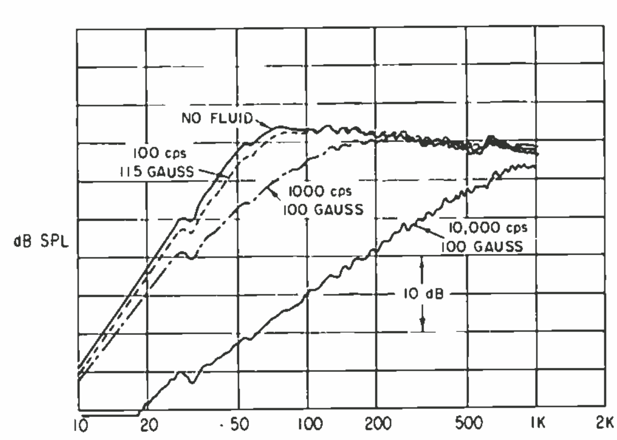
MAGNETITE PARTICLES

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 = 1×10^{-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.

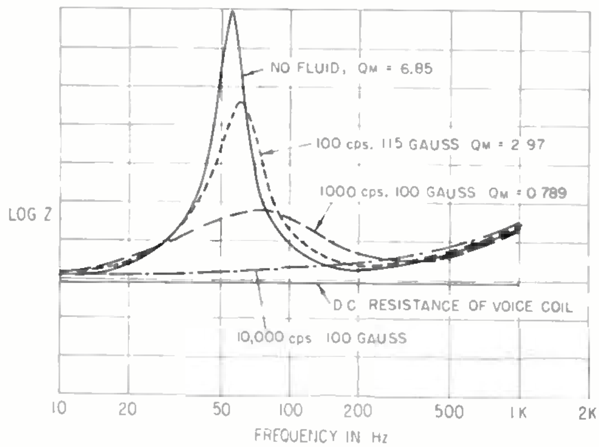


Figure 2. A Modulus of Impedance Curve (Log Z vs. Frequency) may also be used to determine the effect of ferrofluids on voice coil motion. Note the decrease in Q_M as the viscosity is increased.

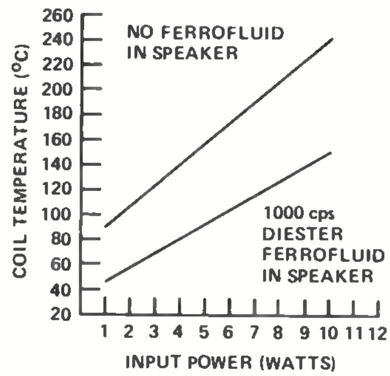


Figure 3. Voice Coil Temperature vs. Input Power.

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. ■

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,

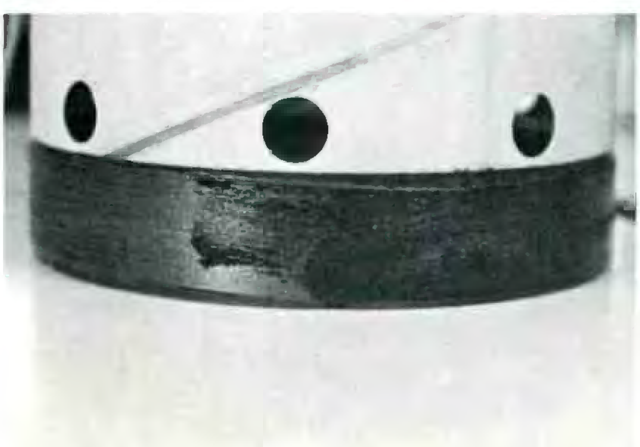
A Gauss speaker cutaway, showing the voice coil in place.

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.

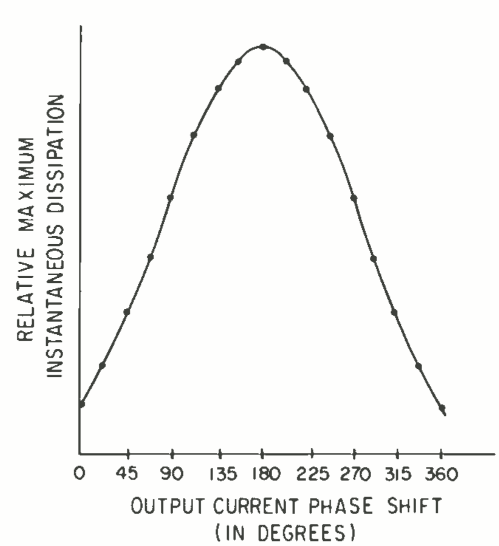
In connecting console components together, the easiest situation to handle is that in which the input of one component bridges the output of the other. Thus, a component should have a very low output impedance, which will provide a constant voltage to the load. The loading component should have virtually 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.

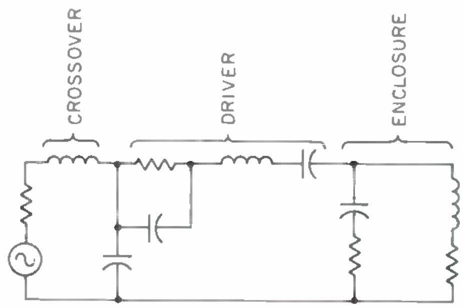


Figure 2. An equivalent circuit for a loudspeaker.

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.

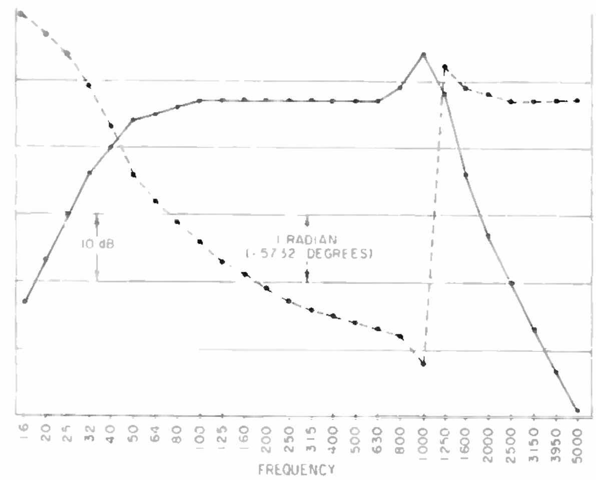
MAXIMUM POWER TRANSFER

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

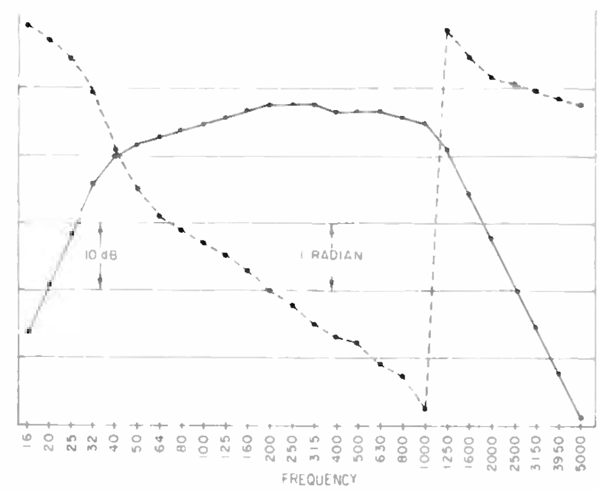
$$W_{MAX} = e_g^2/2(R_g + R_E)$$

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 $X_g = -X_L$.

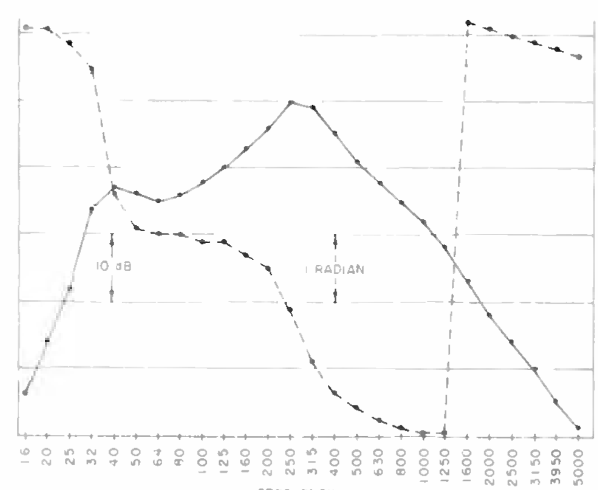
This expression shows that for maximum power transfer to the resistive portion of the load, the reactance of the amplifier must be the conjugate of the reactance of the load. In other words, an inductive speaker must be matched to a capacative amplifier, and *vice-versa*. Thus, the reactances must "cancel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



(A) 0.8 ohm source

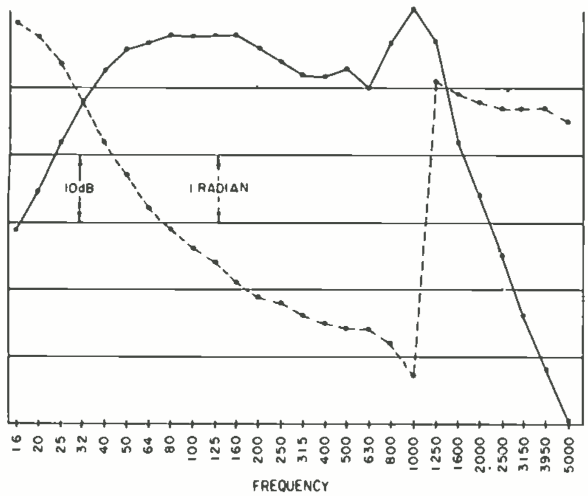


(B) 8 ohm source

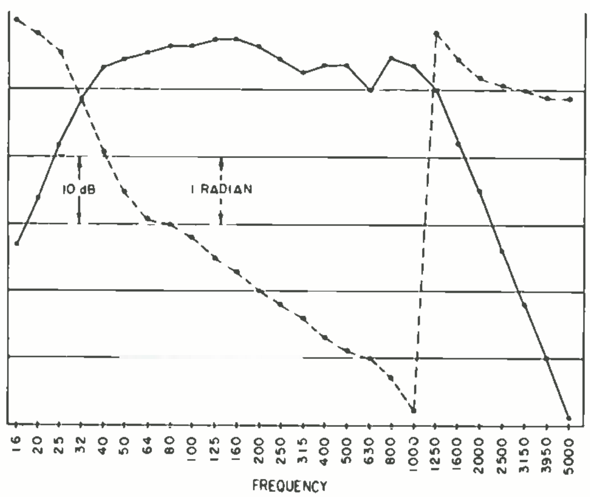


(C) 80 ohm source

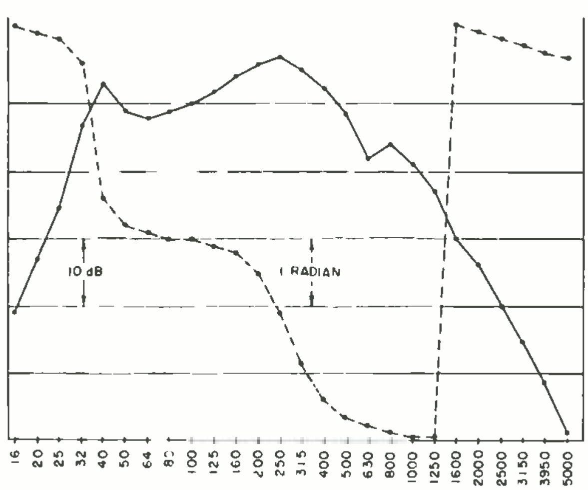
Figure 3. 10-inch woofer in vented box. Solid line represents amplitude response; Dashed line represents phase response.



(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

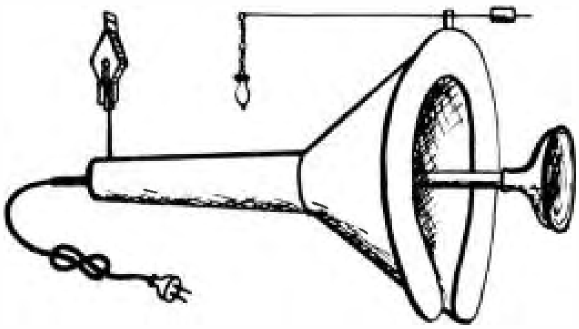


Fig. 13 — Model SP13.5TRBXWK

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.
"[illegible]!"

Fig. 5 — Actual Testimonials from Satisfactory Customers

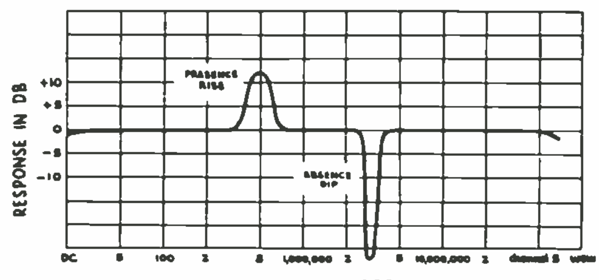


Fig. 3 — Frequency Response at 791-watt Level

*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!

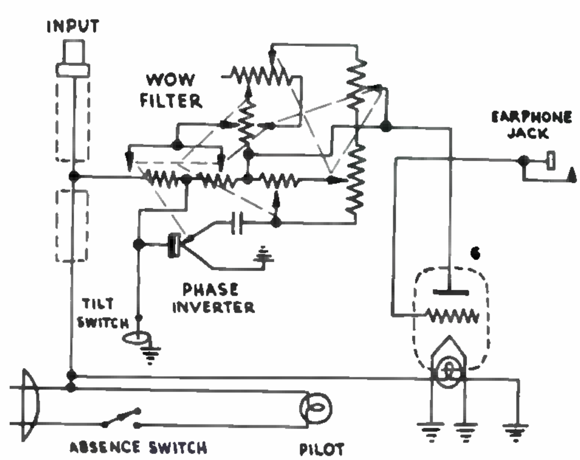
CAUTION NOTES

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

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