Reprinted from RECORDING engineer/producer, Vol. 7, No. 5 (October 1976)

The JBL Way
INSIDE THE STUDIO MONITOR

by
GARRY MARGOLIS

Of all the machinery found in today's recording studios, the most mysterious device is the studio monitor. All artistic judgements about recordings are made subjectively through monitor speakers. Therefore, it is important that monitors inject as little personality of their own as possible. Accuracy of reproduction is paramount.

Unfortunately, more witchcraft has been promulgated about loudspeakers than any other recording tool, although microphones run a close second. Even though many people are aware of objective standards for electronic and magnetic products and are equipped to measure such parameters in their own environments, audio transducers remain a black art, evaluated subjectively and used without understanding.

There are, in fact, objective criteria for studio monitors. There are no magic boxes. This article will discuss some of the factors entering into the design and use of a studio monitor.

MONITOR DESIGN CRITERIA

When a studio monitor is designed, several parameters must be considered, all of which interact and affect the final system. Required bandwidth must be balanced against required efficiency, distortion, size and cost. Basic physical properties of generating and propagating sound in air limit the design. One cannot get maximum possible bandwidth and efficiency in a minimum size box at a minimum price. Tradeoffs are inevitable.

It is presently impossible to make a single piston which radiates all audible frequencies equally efficiently with equal dispersion and still maintains usable sound pressure level (SPL) and minimum distortion. Current technology therefore requires the use of two or more transducers for the system, and the audio spectrum must be divided between them. In most designs suitable for studio applications, the transducer which handles the lowest part of the audio spectrum is the least efficient. It must work harder than a high frequency unit to produce equivalent SPL. Thus, the studio system design effort usually begins with the woofer.

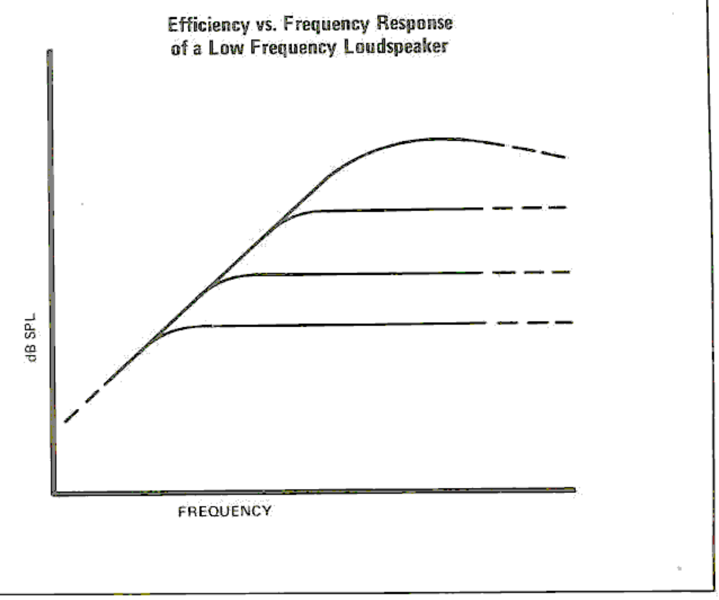
Figure 1 shows the options available in selecting the bandwidth and efficiency of a woofer. When extended low frequency response is chosen, efficiency must be sacrificed, given the same size piston. The woofer in a studio monitor is not required to have extremely high efficiency. Since the distance between the loudspeaker and the listener is small, it is possible to develop reasonable levels without inordinate amounts of power.

There is, however, a point of diminishing returns. The fundamental frequency of the lowest note on the bass guitar is 41.2 Hz, and 32 Hz is the lowest "C" on the piano. Choosing a woofer which provides flat response below 30 Hz needlessly sacrifices effieiency, since almost no musical sound exists below that frequency. Even flat response to 30 Hz may sacrifice too much maximum loudness for some applications. In such cases, two woofers may be needed to retain this bandwidth with high SPL, as in the JBL 4350 Studio Monitor.

If flat response to 30 Hz is required, but high SPL is not necessary, the woofer piston size may be reduced along with enclosure size. The relative low frequency response of the 12" woofer used in the JBL 4315 Studio Monitor, for example,

> *The Author*
> GARRY MARGOLIS is an applications engineer for JBL, responsible for professional product information. His background includes mixing and production for records, film, broadcast and live theatrical sound. He has led university-level courses in audio theory, music recording and motion picture re-recording. He is a member of AES, SMPTE and Local 47 of the American Federation of Musicians.

Figure 1:



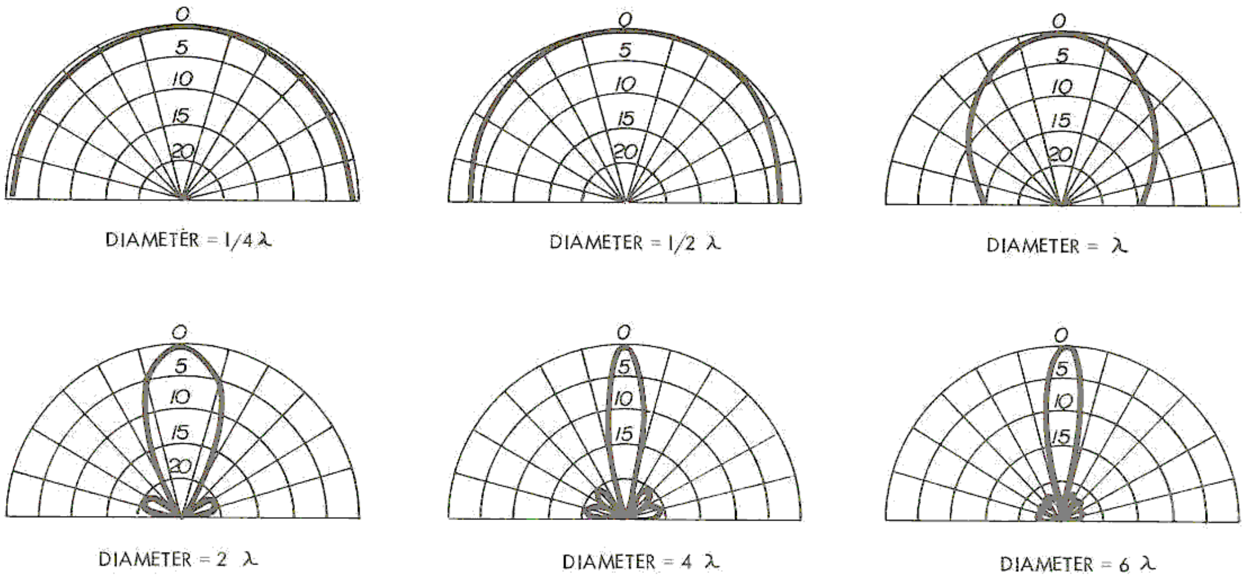


Figure 2: **Dispersion of a Single Direct Radiating Piston With Respect to the Wavelength of the Radiated Sound**

is the same as the low-end response from the 15" woofer used in all of the current JBL full-size monitors. The efficiency of the 12" speaker is lower and the maximum output level is consequently lower, but the bandwidth is the same.

A moderate-efficiency 15" woofer suitable for studio monitoring will have smooth on-axis response up to approximately 1 kHz. A transducer must then be selected to take over the response above that point. This transducer will probably be more efficient, so it will require less power to match the loudness of the woofer. Less power handling capacity, therefore, is needed. Of course, smoothest possible response, maximum bandwidth and lowest distortion are sought.

However, finding a single transducer to operate from approximately 1 kHz up to 20 kHz with reasonable characteristics is also nearly impossible. If smooth, flat response to 20 kHz is not required, a two-way system may be adequate, but flat response to 20 kHz almost always requires a third transducer.

One additional factor enters into our decisions. The dispersion of a direct-radiating piston is a function of both the wavelength generated and the size of the piston. If the wavelength is smaller than twice the diameter of the piston, dispersion will narrow and side lobes will develop, as shown in Figure 2. Also, if two pistons are mounted next to each other and are radiating the same signals, such as in a number of double-woofer monitors available from several manufacturers, dispersion in the plane corresponding to the long dimension of the array will narrow and lobe at a much lower frequency than in the opposite plane. To avoid these problems as well as improving intermodulation characteristics and phase response, another transducer can be added to cover the region between 300 and 1000 Hz.

We now have a number of options. We can have a two-way system which does not cover the full audible range. We can have a three-way system which does cover this range, or we can have a four-way system which covers the full range with greater accuracy than the three-way system.

Now we come to the most difficult part of the design. The acoustical outputs of the transducers must be combined in such a way that the transitions between them are as smooth as possible. Unfortunately, loudspeakers are not perfect devices, and they are not exactly linear in their responses. For example, loudspeaker impedance varies non-linearly with frequency, which makes crossover network design quite a challenge. Also, an otherwise usable transducer may have sloping response over a part of its desired operating range, which may be compensated in the network design.

Further, the physical spacing of the transducers on the enclosure baffle panel will affect their interactive responses in the transition regions. If a compression driver mounted on a horn or horn/lens combination is used, the length of the horn will also affect response through the region of transition. The horizontal displacement between the voice coils of the cone device and the compression driver should be even multiples of one-half the wavelength of the transition frequency to maintain proper acoustical phasing.

In order to obtain imperceptible transitions, JBL engineers design a crossover network which will work with ideal transducers. Connecting this network to the proposed system, the acoustical response is measured and the network is modified until the smoothest possible results with the actual transducers used in the system are obtained. This design method automatically takes into account the characteristics of the individual transducers and their interaction in that particular configuration. In the JBL 4343 (and the earlier 4341), as an example, a classical second-order Butterworth design (12 dB/octave, 3 dB down at the transition frequency) resulted in a 3 dB bump in acoustical response in the lowest transition area due to mutual coupling between the woofer and midrange cones. Modification of the network, so that the electrical response was down 6 dB at crossover, eliminated this acoustical anomaly. This same special crossover characteristic is required when the 4343 is bi-amplified.

It often happens that the proposed group of components and enclosure design need modification for smoothest possible response, at which point the engineers go back to the drawing boards and try again. Several systems may be designed in this manner until acceptable results are obtained.

If no transducer is available with characteristics suitable for the required application, a totally new transducer will be designed and built specifically for this purpose. The midrange drivers in all three of JBL's four-way monitors are examples of this. The midrange unit in the 4350 also happens to be a good compact reinforcement woofer for some applications, so it is available separately under the model number 2202A. The midrange drivers in the 4315 and 4343, however, are not suited for any other application.

After the designers are satisfied with the objective and subjective performance of the system, given the limitations on size and cost initially set, the JBL "Golden Ear" panel auditions the proposed system against existing units — both previous JBL designs and competitive systems. The members of this panel are drawn from various departments at JBL and have widely varying musical tastes. Many have extensive experience in the recording industry. Outside producers and mixers may also be invited to comment. If this panel is not satisfied with the proposed system, it goes back for further refinement. Only if the consensus recommends acceptance is the system released for production.

It can be seen that the development of a top-quality studio monitor is a difficult, exacting task. For this reason, JBL generally does not recommend construction of custom monitors unless the builder has considerable experience in designing systems which have the required reproduction accuracy and he has access to highly sophisticated acoustic laboratory equipment to evaluate the work in progress.

But what happens to all of this careful design when the finished system is installed in a real room? This aspect of monitor application is where most problems are found.

MONITOR/CONTROL ROOM INTERFACE

JBL publishes detailed specifications on all of its studio monitors, including frequency response and distortion figures. These specifications are measured in acoustically neutral "hemispherical free-field" conditions. Since JBL monitors are used in a large variety of rooms with many different acoustical characteristics, measuring in one size and type of room would yield results which would not be applicable for most other rooms.

As shown in Figure 3, the system is mounted in a 30' x 40' (10 x 12 m) test platform on the roof at JBL. The baffle is flush with the platform and the system radiates upward. A microphone is suspended directly above the system at a distance of from 6 to 30 feet (2 to 10 m) and measurements are taken for the curves and numbers published in JBL literature. It is reasonable to conclude that any significant deviations from these results when measurements are made with the system installed in a control room may be traced to room acoustics and the method of mounting the system in that room.

A number of consultants and studio designers have built different control room configurations which attempt to solve the myriad acoustical problems confronting the mixer and producer. JBL does not endorse any specific room design concept, since many have a number of advantages and none has clear superiority. Some of the characteristics of the interface between the room and the loudspeaker system, however, should be examined.

If the reverberation characteristics of the control room are not uniform in frequency response and/or there are acoustical standing waves present, the response of any system in that room will be affected. Monitor equalization is of limited value in smoothing out rough response. Third-octave equalization in the direct field of a loudspeaker system can be very difficult to accomplish successfully, since a microphone cannot discriminate between the direct and reverberant fields in a room in the same manner that the ear does. This can result in differences between perceived and measured response if the reverberation characteristics of the room are not uniform. It takes an extremely skilled person who has sufficient experience to make final equalization adjustments by ear in order to minimize this factor. Also, if more than approximately 3 dB of equalization is used, some subtle, yet difficult to describe effects can be audible, which may be related to phase shift and ringing. Finally, no amount of electrical equalization can eliminate standing wave effects, since these vary according to the precise position of the listener in the room.

If the woofer is at the intersection of two room boundaries — at the corner intersection between two walls, at the junction of the floor and wall, or at the wall/ceiling joint — low frequency radiation will be restricted to less than a hemisphere. Consequently, bass response will be accentuated. Still greater accentuation will be obtained with the woofer at the intersection of three room boundaries (two walls and the floor or ceiling). This effect may be diminished with acoustical treatment, but "bass trapping" requires a great deal of space to accomplish well.

Mounting a monitor away from the room walls will result in rough and diminished low frequency response, since

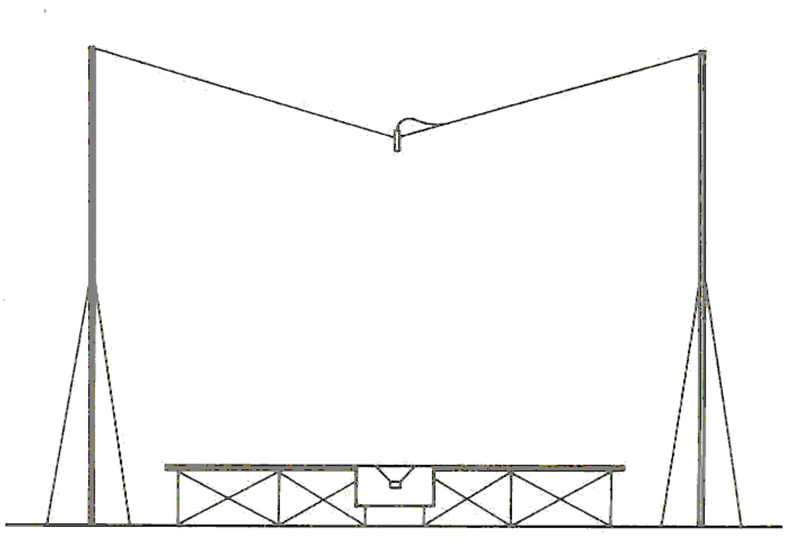
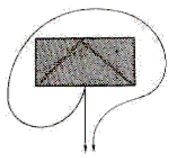


Figure 3: **System Testing in Hemispherical Free-Field Conditions**

low frequencies will have a double path — one directly from the woofer and another around the enclosure — as shown in Figure 4. The larger the enclosure, the lower the frequency at which this effect occurs. If the room configuration permits, flush mounting the monitors so that the baffles are even with the surrounding wall surface will yield the most uniform response.

Figure 4.



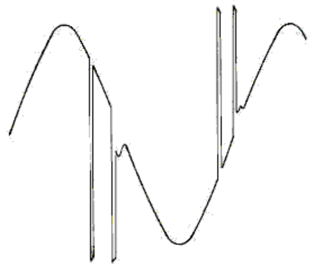
Low Frequency Radiation Around an Enclosure

If the monitor is installed too close to the listening position, the wavefronts from the individual transducers will not have a chance to combine properly, and the individual transducers will be audible as separate units. The minimum working distance from the JBL 4311 or 4315 is about 3 to 4 feet (1 m); single-woofer full-size JBL monitors should be at least 6 feet (2 m) from the listener; and the 4350 should be at least 8 feet (2.5 m) away.

Another problem which occurs is insufficient level in the room. As mentioned earlier, there are tradeoffs between bandwidth and efficiency (given equal size) as well as size and efficiency (given equal bandwidth). In order to achieve wide bandwidth, efficiency must be sacrificed. Many older monitors are quite efficient and do not need high amplifier power levels to achieve high SPL's. However, these older monitors do not have the extended bandwidth of the new JBL systems. If a current JBL monitor is installed in place of an older system, the original amplifier may not be adequate to the task and may be driven into clipping.

We now discover a totally new problem. Many popular amplifiers will work quite well until they reach the clipping point, after which their output protection circuits will "chatter" or produce large high frequency spikes. A typical chatter waveform taken from a popular studio amplifier is shown in Figure 5. Such spikes can destroy a high frequency driver. Cracking sounds on sharp low frequency impulses are an audible symptom of amplifier chatter. This phenomenon is much more likely to occur with reactive loads (such as those provided by large voice coils and massive magnetic assemblies) than with resistive loads, so it may not show up on a test bench. If chattering is a problem, increasing the available amplifier power or changing to an amplifier which does not exhibit chatter are recommended.

Figure 5.



TYPICAL AMPLIFIER CHATTER WAVEFORM CAUSED BY REACTIVE LOADS

Even if an amplifier does not chatter, it will produce square waves when it clips, which, by definition, are collections of odd-order harmonics. These harmonics have far more high frequency energy than normal program material and can burn out high frequency drivers.

Although most amplifiers used in studios are free from this problem, assuming that they are properly installed, there are some units, particularly inexpensive or "home-brew" designs, which may become unstable with reactive loading. The resulting high frequency oscillations can destroy high frequency transducer voice coils. Although studio monitors generally do not use electrostatic transducers, which obviously have large capacitive reactance, one hidden source of capacitance is the wire connecting the amplifier and the loudspeaker.

Assuming that a quality amplifier is used and that clipping is avoided, it is important to choose a monitor based on the expected maximum SPL to be achieved in the room. As mentioned before, one cannot get as much level out of a small system as from a large system, assuming equivalent bandwidth. For example, measured at 8 feet (2.4 m) in free-field conditions at rated power, a single JBL 4311 or 4315 will produce 99.5 dB SPL, while each of the current single-woofer full-size monitors will produce 104.5 dB and the 4350 will produce 111 dB SPL! Attempting to squeeze more level out of a monitor than it is capable of delivering will invariably result in component failure.

Unfortunately, it is extremely difficult for the user to make valid evaluations of new monitors, for several reasons. Most people, for example, will listen to tapes which they have previously mixed on other systems. The hidden problem with this approach is that if the original monitor was not smooth and flat in response, the recording will probably have an overall equalization which is the inverse of the original monitor's response. Further, room acoustics and system placement can markedly affect the sound of a monitor, as discussed earlier.

One evaluation method is to listen for detail. An accurate monitor, for example, will tend to reproduce acoustic guitar chords so that each individual string will be clearly audible. Subsidiary voices in large masses of sound should be well defined. High frequency response should be smooth across included angles of 60° horizontal and 30° vertical, so that the acoustical balance does not change with the location of the listener.

Stereo localization is another simple factor to evaluate. A monaural signal panned to the center should be sharply defined, without smearing or broadening. Incidentally, this last test is recommended for final critical balancing of monitors, since a 1-dB imbalance between the drivers of two systems may be heard by a discerning listener — if the acoustics are good.

There are many "magical" and "revolutionary" new speakers appearing in the marketplace every month. In order to achieve some of the claims made for these systems, however, the laws of physics would have to be repealed. As yet, JBL has not learned how to violate these laws. Current JBL monitors are as accurate as our present understanding of physics allows, given restraints on size and cost. JBL is constantly engaged in research into the mechanics of sound reproduction, and expects that in a few years, studio monitors will be available which will be as far ahead of current units as these present ones are superior to the previous generation of loudspeaker systems.