Ground-Plane Acoustic Measurement of Loudspeaker Systems*

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Elaborate electronic techniques or specially constructed environments are usually required to measure only the direct field from a loudspeaker without contamination from reflected signals. By placing the measurement object on a flat, rigid, unobstructed ground surface, and placing the measurement microphone flush with the ground, an accurate simulation of the anechoic response of the object together with its image source may be obtained.

0 INTRODUCTION

When measuring the sound output of a loudspeaker, it is usually desirable to separate the direct sound field produced by the source from the reverberant field generated by the reflections in any particular environment. This has traditionally been accomplished by placing the loudspeaker source in an artificial free-field environment, an anechoic chamber, specially constructed for this purpose [1; 2, pp. 96–99]. This provides accurate measurement conditions for mid and high frequencies at small measurement distances. However, the finite size of the chamber necessarily limits low-frequency accuracy, and large measurement objects and long measurement distances also decrease the usefulness of such environments.

Another approach to simulating a free field is to suspend the source up in the air, far enough above the ground, to eliminate significant reflections [2, p. 102; 3]. This is awkward at best, particularly for large measurement objects and long measurement distances. Near-field measurements may be taken, but they are accurate only for the low frequencies [4].

Recently gating and other delay techniques have been developed to extract the direct field from the total output through an adjustable time window, but these require complex manipulation of elaborate electronics and still suffer from bandwidth restrictions at low frequencies [5, 6].

The previous approaches try to remove the effect of the environment entirely from the measured signal. Another approach is to stabilize the environment such that any effect introduced by it can be easily qualified. The most common example of this is the hemispherical free field, where the front surface of the loudspeaker is mounted flush with a large baffle surface, changing the effective radiation volume from a solid angle of 4π steradians to 2π steradians. This method has been adopted in order to better simulate the low-frequency load on a loudspeaker in typical use conditions [7, p. 460], but also allows somewhat less complexity in the measurement environment. The 2π baffle may be surrounded by the wedges of an anechoic chamber, or it may be generated by placing the loudspeaker in a pit flush with a hard ground surface, the unobstructed open air providing the hemispherical free field. In either case, the change in integrating area provides a different, but stable and known environment in which to assess loudspeaker performance.

In the same way that we can change the low-frequency loading and still extract the necessary information, we can allow a single reflection if its effect can be accurately predicted. In the presence of boundaries where drastic changes in acoustic impedance occur, acoustic image sources can be employed to model the resulting wavefronts. The method of images has been utilized extensively in underwater acoustics [8, pp. 427, 474] and in the field of aircraft noise [9, pp. 15–16]. Placing microphones at or near a rigid boundary, to make con-

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structive use of the acoustical image and eliminate destructive interference, has been used in both measurement and sound recording applications [10]. Malter was the first to make use of the ground reflection for loudspeaker measurements, utilizing the technique for low frequencies only [11, p. 614], and Kellogg described the ground-plane method as being theoretically valid [12, pp. 175–176].

If a loudspeaker is mounted resting on a smooth rigid ground surface and the microphone is placed flush with the ground, a mirror image can be thought of as being produced below the ground surface next to the real source. The presence of the image source effectively doubles the axial pressure, adding 6 dB to the sound pressure level. The measurement then approximates two identical loudspeakers next to each other in free space (Fig. 1).

1 THEORY

A spherical sound source radiating uniformly in all directions close to a rigidly reflecting boundary may be considered to be a pair of sources in a free field vibrating in phase and equal in strength, due to the presence of an image source beyond the boundary. This is shown in Fig. 2.

From Beranek [13, pp. 92–96] the equaton for the magnitude of the rms sound pressure $|p|$ is

$$|p| = \frac{2A}{r} \frac{\sin[(2\pi b/\lambda)\sin\theta]}{2\sin[(\pi b/\lambda)\sin\theta]}$$

where A is the magnitude of the rms sound pressure at unit distance from the center of each source, b is the distance separating the sources, r is the measurement of distance, and θ is the angle to the perpendicular bisecting the sources. At low frequencies b is very small compared to a wavelength and the two sources essentially coalesce. The pressure at any distance r and at any angle θ is then double that for one source acting alone.

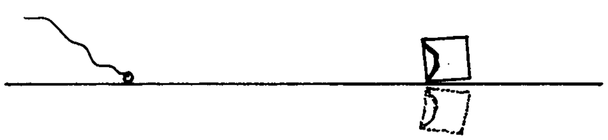

Fig. 1. Illustration of the ground-plane technique. Measurement microphone on axis of measurement object and image source created by ground plane.

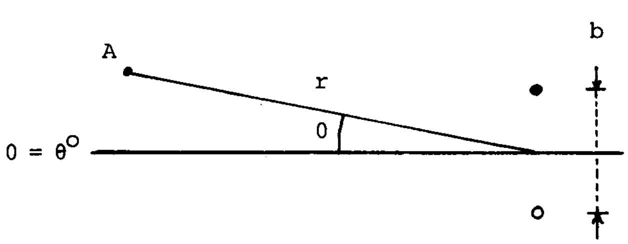

Fig. 2. Two equal-strength sound sources vibrating in phase located a distance b apart, at distance r and angle θ with respect to point of measurement A. The same conditions apply for a single source located ½b from a rigid boundary.

As the wavelength decreases with increasing frequency, the pressures arriving from the two sources will be different in phase at various angles, forming changing directivity patterns. The axis bisecting the sources, $\theta = 0°$, is the principal lobe of the directivity pattern, however, and the pressure on this axis is always double that without the boundary and image source. This doubling of the pressure will add 6 dB to the axial sound pressure level. It is important to remember that while we have doubled the pressure and generated four times the intensity, the power generated has only doubled. Olson's chart is useful in understanding these differences [14, p. 32; 15, p. 13].

2 PRACTICAL MEASUREMENT CONSIDERATIONS

2.1 Measurement Site

The most appealing aspect of the ground-plane measurement technique is that the only special environment necessary is a large, flat, smooth, and rigid (reflective) ground surface. Any asphalt or concrete parking lot, playground field, or roof, free from obstructions, can be utilized. The smoother and more reflective the surface and the more consistent the absorption coefficient, the higher in frequency the accuracy of the measurement can be extended (smooth painted concrete being preferable to rough porous asphalt). There should be no obstructions in any direction for a distance which will depend on the measurement distance and signal frequency. For guaranteed safety, the distance from the source to any obstacle large compared to a wavelength should be at least five times the measurement distance. This ensures that in the worst case a reflection will be more than 20 dB down, contributing less than 1 dB to the total pressure. Obstacles must also be greater than a wavelength away in distance to ensure consistent radiation loading.

2.2 Environmental Conditions

Since the area chosen for the ground plane will usually be outdoors, measurements will be subject to the prevailing atmospheric conditions. While it is obvious that inclement weather can prevent measurements due to simple practical considerations, it can also affect the measurement accuracy.

Wind turbulence can create spurious low-frequency signals. These may be sufficiently reduced by a suitable microphone windscreen. Temperature and wind gradients can cause refraction of the sound rays at the ground surface [16]. These effects are most pronounced at frequencies above 5 kHz at the small grazing angles normally associated with the ground-plane technique [9, pp. 15–16]. For this reason it is best to make ground-plane measurements during calm, overcast periods to avoid the possible effects of wind and heating from direct sunlight. Orienting the source location at a higher angle relative to the ground and microphone can reduce these effects.

2.3 Source and Microphone Positioning

The microphone must be placed flush with the ground and at a distance sufficient to be in the far field, usually greater than about three times the maximum extent of the source (the largest dimension) which here includes both the source and the image. The microphone must be of the pressure-responding type with an omnidirectional response characteristic [10, p. 2].

The loudspeaker should be placed on the ground and tilted such that the transducer axis is aimed directly at the measurement microphone. The microphone's position relative to the source should be exactly the same as for a free-field measurement were the ground plane not there. Only the relative position of the microphone and the measurement object is important, and the fact that the microphone is flush with the ground plane. As the directivity of the source changes with frequency, the microphone sees the same frequency response from the image source as it sees from the actual source. By the reciprocity theorem [8, pp. 324–325] only a microphone position which deviates from a flush position relative to the ground plane limits the high-frequency accuracy of the ground-plane technique, not the spacing of the source and its image.

For convenience it is often desirable to merely place the microphone on the ground rather than actually mount it flush with the surface. In this case, it is necessary to place the transducer, or the high-frequency radiating elements in multiway systems, as well as the microphone as close to the ground as possible to extend high-frequency accuracy. The derivation in the Appendix indicates to what extent this will limit the high-frequency accuracy of the method.

2.4 Baffle and Directivity Effects

The orientation of the loudspeaker source can be varied to investigate various boundary effects [17] and to determine how the baffle and box size affect the radiation pattern [18]. Care must be taken, however, in that the ground-plane technique simulates two sources positioned in mirror image along the measurement axis in free space. The baffle size is hence twice as large and the shape is different than that of a single system alone. If it is practical to raise the loudspeaker some angle above the ground or directly above the microphone position while maintaining the microphone flush with the ground plane, the increased-baffle-size effect can be reduced or eliminated along with mutual-coupling effects between the source and the image [19, pp. 11–14]. Normally, however, the physical advantages of simply placing the loudspeaker on the ground and tilting its axis toward the microphone will outweigh the slight measurement inaccuracy.

2.5 Polar and Distortion Measurements

While box effects must be studied carefully, other measurements are easily made on the ground plane. Since dispersion is only affected in the vertical direction, off-axis measurements can be taken by moving the microphone to various points along the ground plane. Turning the cabinet 90° will allow the other angles of dispersion to be measured along the ground plane. The exact degree of tilting of the source will have to be adjusted for each off-axis position to ensure the proper relative orientation of the microphone and source. Alternately, the source cannot be tilted, and polar curves or off-axis information can be taken at an angle slightly below the normal 0° plane bisecting the source by simply moving or rotating either the source or the measurement microphone. The placement of the actual source location as close as possible to the ground plane will minimize error. Distortion and other measurements may be taken just as in any other environment. Where moving a large system or obtaining a large enough measurement distance were once insurmountable problems, all that is required here is moving a few more cars off the parking lot.

2.6 Multiple and Array Sources

In sound-reinforcement work, where working with multiple systems in suspended clusters, the ground-plane measurement provides the unique advantage of being able to evaluate the performance of twice as many systems in free space by merely stacking the systems on the ground. Similarly, large line arrays, arc, plane and spherical segments may be synthesized and evaluated with only half as many components. When attempting to utilize the image to simulate a second actual source, the loudspeaker should normally not be tilted toward the microphone. The unit should be positioned such that it and its mirror image together are oriented as the total actual array would be. In this case vertical off-axis measurements can be made on the combined source by raising the microphone up off the ground plane, since positions above the ground will see the ground reflection as the off-axis angles from the image source combining with those from the real source. The path length difference will cause peaks and nulls in the response, which will be the actual combined response of the array at that point in space. This is the situation often encountered when attempting to measure a single loudspeaker resting on the ground, and it is seldom realized that the response abberrations can be so easily qualified as the effect of the image-source reflection.

2.7 Suggested Measurement Distance

When evaluating small-size systems or single transducers, current practice has standardized on a 1-m microphone distance. Since the ground-plane image adds 6 dB to the axial sound pressure level, and doubling the measurement distance in the far field decreases the sound pressure level by 6 dB, it is convenient to standardize on a nominal 2-m distance for ground-plane measurements. In this way, with the same power input, a ground-plane measurement at 2 m will have the same apparent mid- and high-frequency sensitivity as a half-space or whole-space measurement at 1 m. At low frequencies the output will approximate that in whole space, increased by any mutual coupling from the image. In between will be a region where the source directivity increases from omnidirectionality to half-space radiation, determined by the total effective baffle size of the source and its image [20].

3 EXPERIMENTAL MEASUREMENTS

3.1 Test System

A single 200-mm full-range loudspeaker in a 25-L stuffed sealed-box enclosure, 0.505 m high by 0.355 m wide by 0.215 m deep, was measured in various environments. The acoustical system parameters were $f_C = 60$ Hz, $Q_{TC} = 0.82$ [21]. The driver was mounted equidistant from the long sides and 0.2 m from one short side.

3.2 Half-Space

Fig. 3 shows the response of this system with the baffle mounted flush in the pit of a 2π gound platform, microphone 1 m on driver axis, 2.8-V input. The response is very smooth through the mid and low frequencies due to the true half-space loading. The response is 2 dB down at resonance, as predicted by the Q of the system.

3.3 Anechoic Chamber

Fig. 4 shows the response of the same system in an anechoic chamber designed for high-frequency work only, 2.8-V input, microphone 1 m on driver axis. At this measurement distance, the chamber provides an

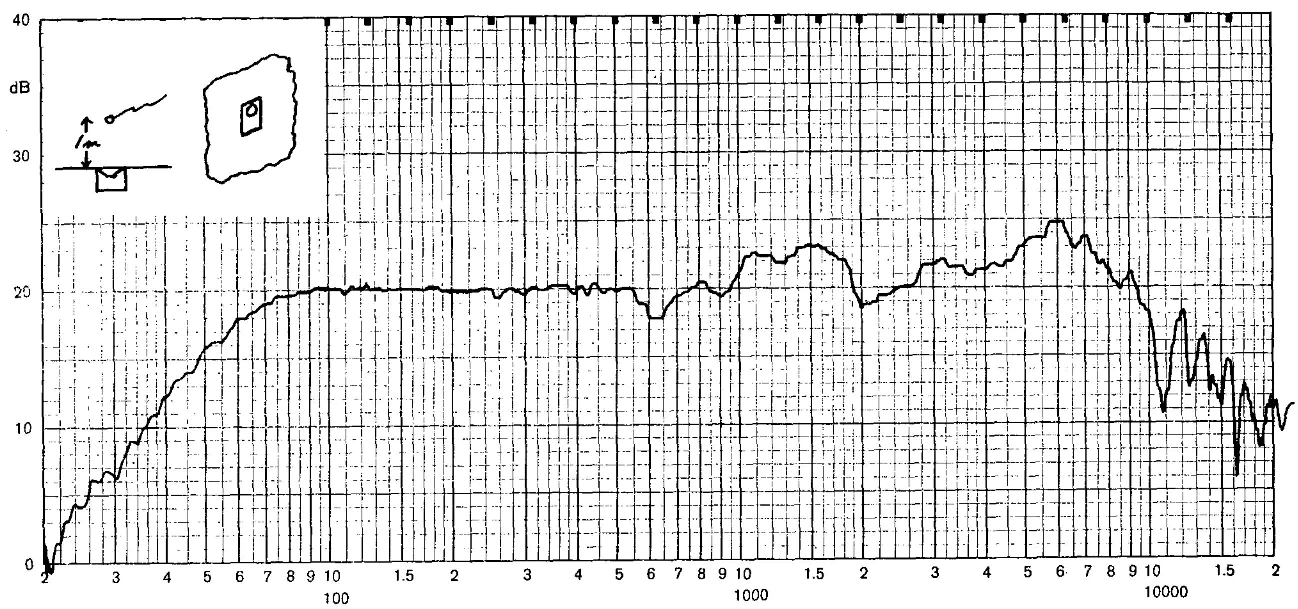

Fig. 3. Half-space response of a single 200-mm full-range loudspeaker in a 25-L stuffed sealed-box enclosure, 0.505 m high by 0.355 m wide by 0.215 m deep, $f_C = 60$ Hz, $Q_{TC} = 0.82$, driver mounted equidistant from the long sides and 0.2 m from one short side. Baffle mounted flush in the pit of a 2π ground platform, microphone 1 m on driver axis, 2.8-V input. Measurement range is 40 dB, 70 dB SPL bottom line, re 20 $\mu N/m^2$.

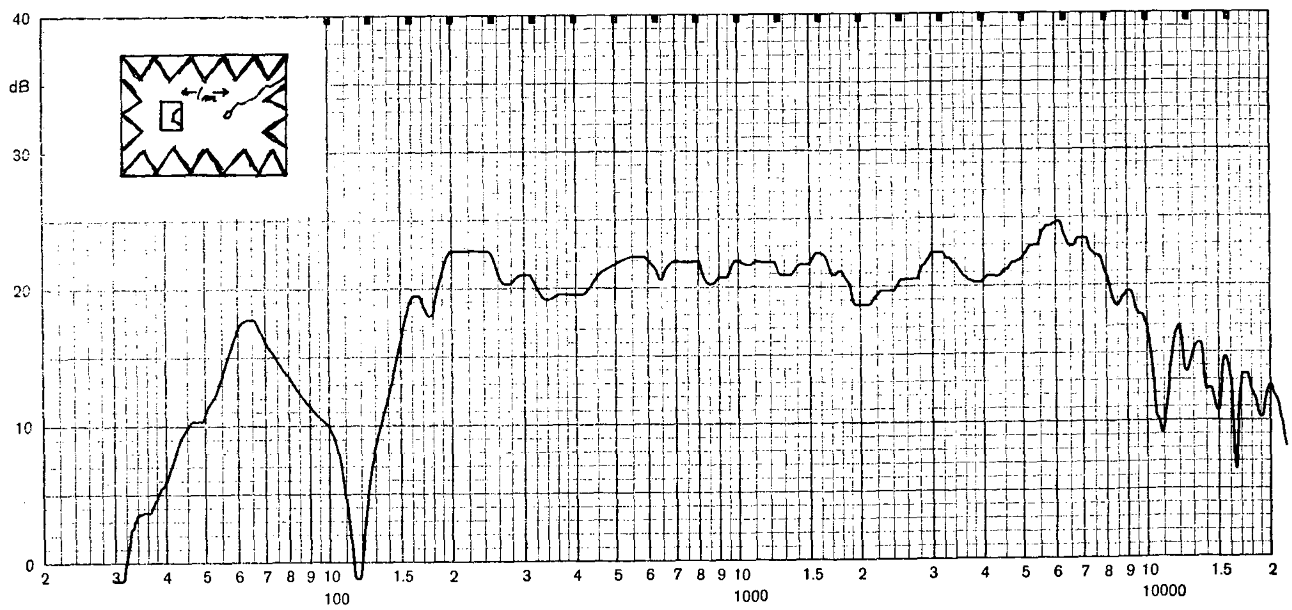

Fig. 4. Response of same system as in Fig. 3, in a 500-Hz anechoic chamber. Microphone 1 m on driver axis, 2.8-V input.

effective free field only above 500 Hz. While the low-frequency limitations of the chamber are quite evident, there is a characteristic rise in the midrange output below 1 kHz due to diffraction effects caused by the enclosure size and shape [18, p. 28]. The high-frequency response is substantially unaffected.

3.4 Suspended

Fig. 5 shows the response of the same system suspended outdoors 4.5 m above the ground surface, 2.8-V input, microphone 1 m on driver axis. The mid- and high-frequency response is exactly the same as that in the chamber, and here the total free-field response of the system is intact. The response at low frequencies has only been reduced by a maximum of about 4.5 dB from half-space loading rather than the theoretically expected 6 dB due to an imperfect free-field load for the very long wavelengths below 50 Hz. Had the system been suspended higher above the ground surface, true free-field loading could have been maintained to a lower frequency.

3.5 Ground Plane

Fig. 6 shows the system response resting flat on the ground plane with the driver nearest to the ground surface, microphone at 1-m distance resting on the ground surface, 2.8-V input. Since the 13-mm (½-in) diameter microphone capsule is centered 6.5 mm (¼ in) above the ground plane, and the loudspeaker source is 0.2 m above, by the derivation in the Appendix the measure-

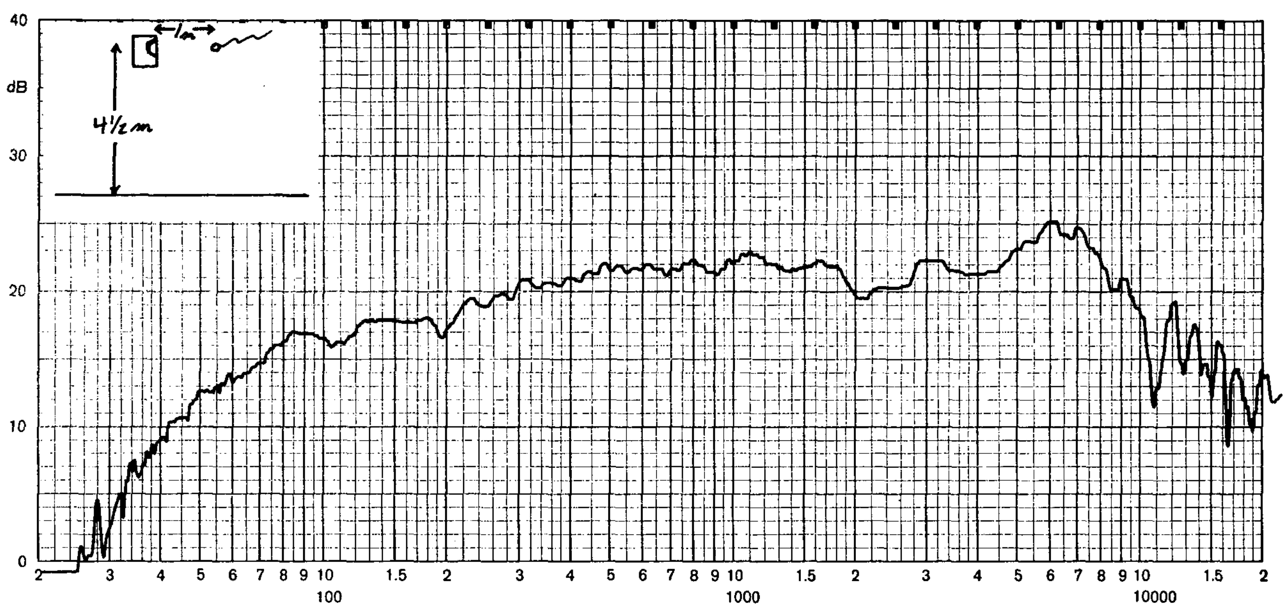

Fig. 5. Response of same system as in Fig. 3, suspended outdoors 4.5 m above ground surface. Microphone 1 m on driver axis, 2.8-V input.

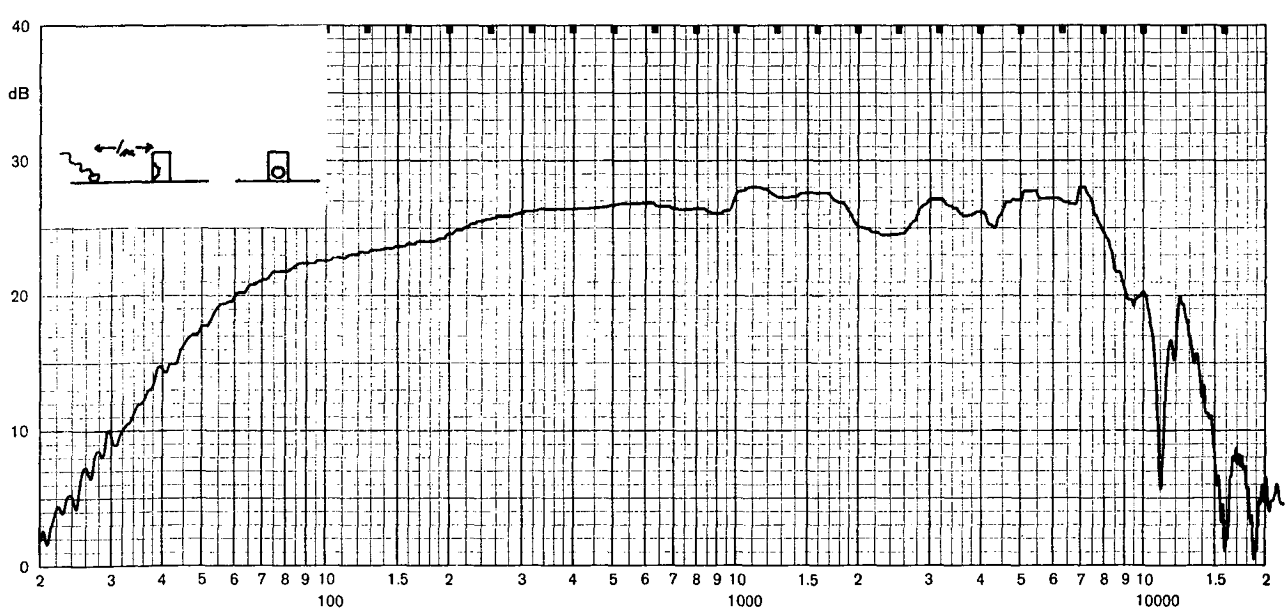

Fig. 6. Response of same system as in Fig. 3, resting flat on ground plane, with driver nearest the ground surface. Microphone resting on ground at 1-m distance, 2.8-V input.

ment will still be capable of accuracy to above 20 kHz. The low-frequency response closely duplicates the suspended free-field curve, but raised 6 dB due to the contribution of the image source.

Fig. 7 shows the system ground-plane response with the microphone moved to a 2-m distance, 2.8-V input. The 6-dB loss incurred in doubling the measurement distance has canceled the 6-dB increase due to the image source, and the low-frequency response now closely matches the suspended free-field curve. The mid- and high-frequency response is reduced, however, since the microphone is off axis from the source.

Fig. 8 shows the same 2.8-V 2-m ground-plane response of Fig. 7, but with the system slightly tilted such that the microphone is positioned on the driver axis, as in previous curves. Using this method, the agreement is within about 1 dB up to about 13 kHz, the variations above this point being due to skew in the exact relative positioning of source and microphone.

Figs. 9 and 10 show the two other possible orientations of object and image source, creating different baffle shapes and source placements, hence different diffraction effects in the 200-Hz to 2-kHz region.

4 CONCLUSION

The ground-plane measurement technique can be employed to simulate the free-field response of a loudspeaker source, together with its acoustic image. The simplicity of the required measurement environment, together with the relative ease of performing a variety of measurements, make the method particularly appeal-

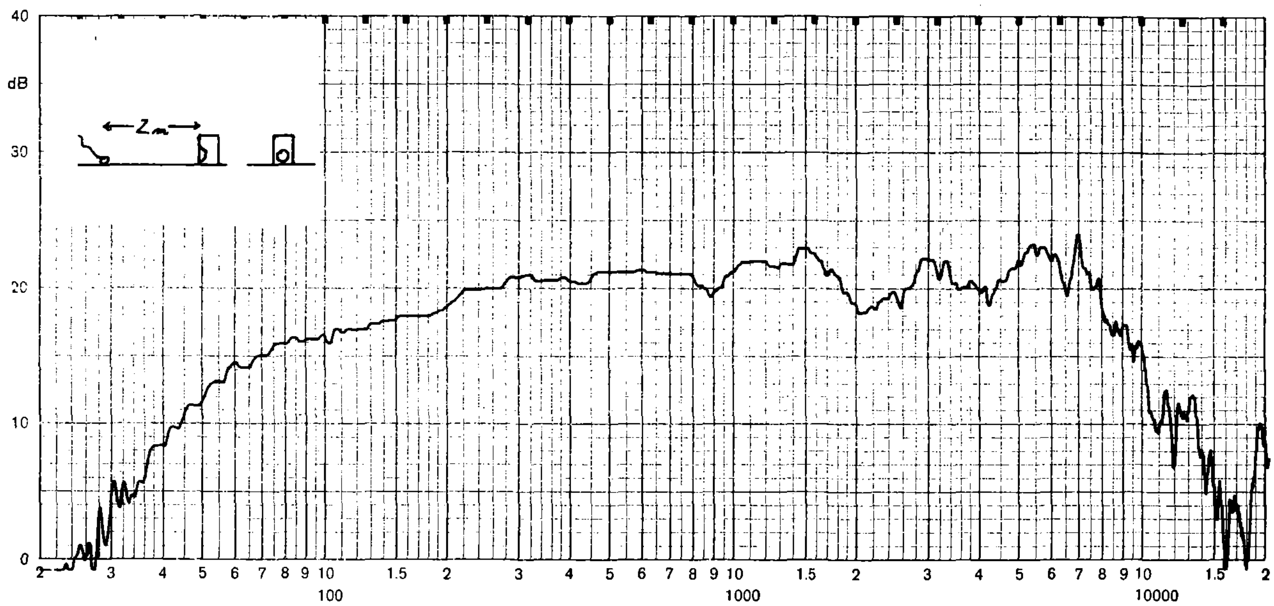

Fig. 7. Response of same system as in Fig. 3, on ground plane with microphone moved to 2-m distance, 2.8-V input.

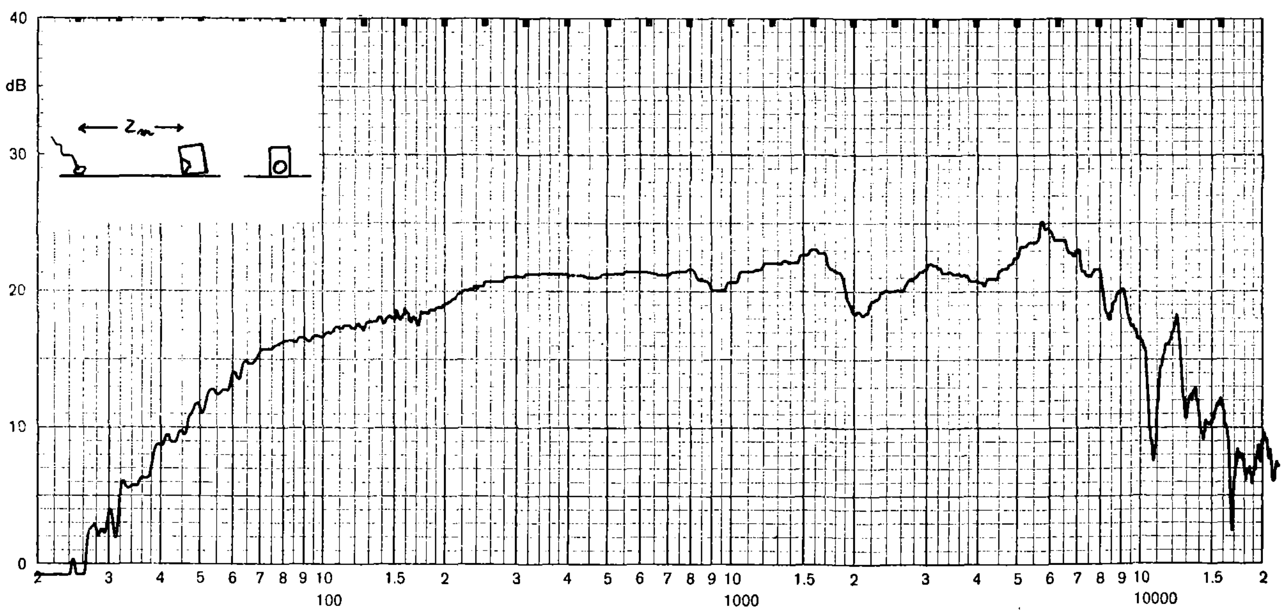

Fig. 8. Response of same system as in Fig. 3, on ground plane, but with system slightly tilted so that microphone is on driver axis at 2-m distance, 2.8-V input.

ing and useful for many general design and evaluation applications.

5 ACKNOWLEDGMENT

The derivation in the Appendix is the work of D. B. Keele, Jr. The author would like to thank him as well as Terry Sorensen and Greg Timbers, all of the JBL Transducer Engineering Department, for their aid in developing and refining the ground-plane technique.

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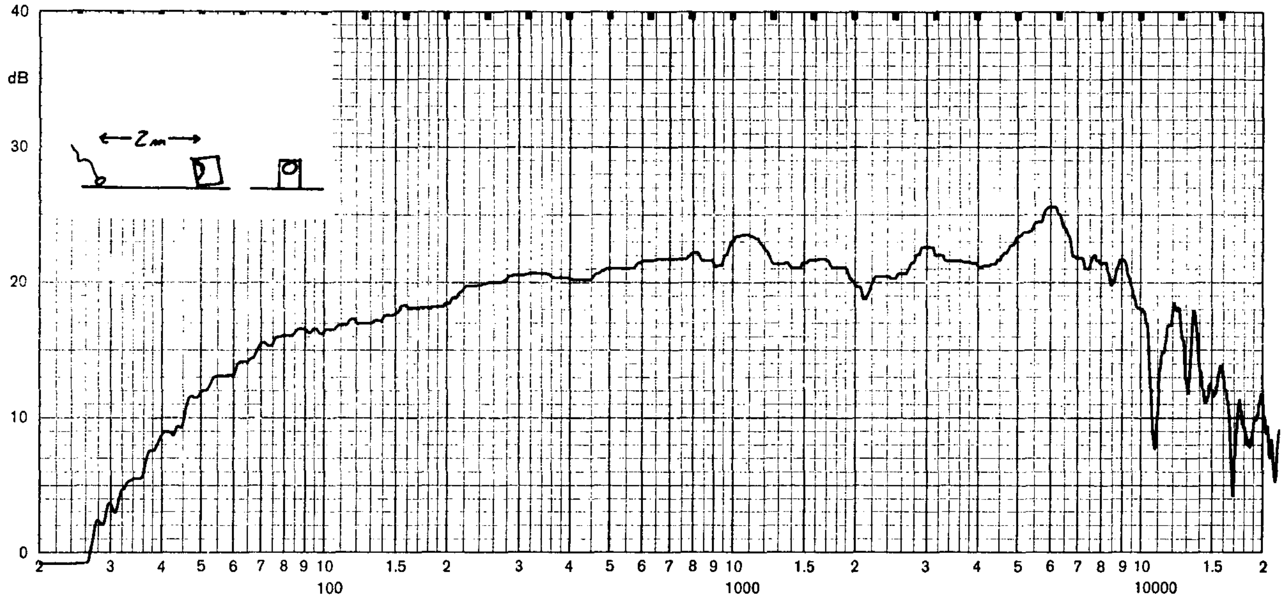

Fig. 9. Response of same system as in Fig. 3, on ground plane, but oriented so that driver is furthest away from ground surface. Tilted so that microphone is on driver axis at 2-m distance, 2.8-V input.

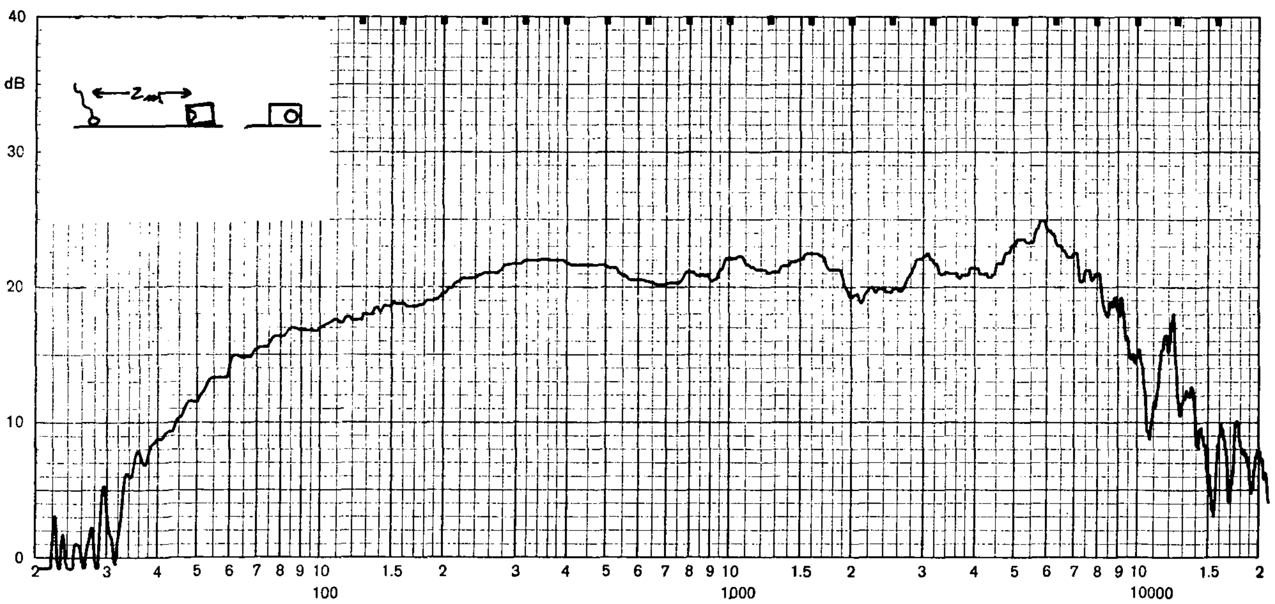

Fig. 10. Response of same system as in Fig. 3, on ground plane, but oriented so that driver is placed to one side of baffle. Tilted so that microphone is on driver axis at 2-m distance, 2.8-V input.

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APPENDIX

RELATIONSHIP BETWEEN UPPER FREQUENCY LIMIT AND DEVICE HEIGHTS

Consider Fig. 11, which shows the device under test (DUT) and the measurement microphone (M) along with their ground-plane reflections and distances.

The following brief derivation computes the difference ΔD between the direct D_d and the reflected D_r distances and relates it to the sound pressure phase and magnitude at the microphone position.

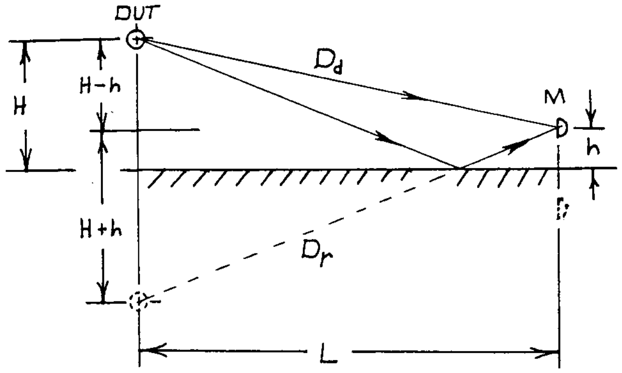

Fig. 11. Device under test (DUT) and measurement microphone (M) with ground-plane reflections and distances.

Using right-angle trigonometric relationships yields the following expressions for the direct and reflected distances:

$$D_d = \sqrt{L^2 + (H - h)^2} = \sqrt{L^2 + H^2 - 2Hh + h^2} \quad (1)$$

$$D_r = \sqrt{L^2 + (H + h)^2} = \sqrt{L^2 + H^2 + 2Hh + h^2} \quad (2)$$

Assuming that $h \ll L$ or $2Hh + h^2 \ll L + H^2$ (a reasonable assumption if the mic height is small with respect to the measurement distance) and the approximate relationship

$$\sqrt{1 + \Delta X} \approx 1 + \frac{\Delta X}{2}, \quad \text{if } \Delta X \ll 1 \quad (3)$$

yields

$$D_d \approx L^2 + H^2 + \frac{-2Hh + h^2}{2} \quad (4)$$

$$D_r \approx L^2 + H^2 + \frac{2Hh + h^2}{2}. \quad (5)$$

The path-length difference is then

$$\begin{aligned}
\Delta D &= D_r - D_d \\
&= L^2 + H^2 + \frac{2Hh + h^2}{2} \\
&\quad - (L^2 + H^2) - \frac{-2Hh + h^2}{2} \\
&= \frac{2Hh + h^2}{2} - \frac{-2Hh + h^2}{2} \\
&= Hh + \frac{h^2}{2} + Hh - \frac{h^2}{2} \\
&= 2Hh. \qquad (6)
\end{aligned}$$

This path-length difference yields a phase difference at the microphone of

$$\Delta\theta = k\Delta D \tag{7}$$

where

k = wave number ($= \omega/c = 2\pi f/c$)

c = speed of sound ($= 343$ m/s)

f = frequency, in Hz

Therefore using Eqs. (6) and (7) yields

$$\Delta\theta = k2Hh = \frac{2\omega Hh}{c} = \frac{4\pi}{c} f Hh. \tag{8}$$

Eq. (8) can in turn be solved for f, giving

$$f_{\max} = \frac{c}{4\pi} \cdot \frac{\Delta\theta}{Hh} . \tag{9}$$

For the magnitude of the sum of two equal-level sine waves of phase difference $\Delta\theta$ it can be shown that

$$\Delta\theta = 0.300\pi \text{rad} = 53.9°, \quad \text{for } -1\text{-dB loss} \tag{10}$$

$$\Delta\theta = 0.5\pi \text{ rad} = 90°, \quad \text{for } -3\text{-dB loss.} \tag{11}$$

Neglecting amplitude differences due to inverse-square-law variations and using relationships (9)–(11) yields

$$f_{\max} = \frac{25.73}{Hh}, \quad \text{for } -1\text{-dB loss} \tag{12}$$

$$f_{\max} = \frac{42.88}{Hh}, \quad \text{for } -3\text{-dB loss} \tag{13}$$

THE AUTHOR

Mark R. Gander was born in New Brunswick, New Jersey, in 1952. Prior to college he received an extensive musical education. He later earned a B.S. degree from Syracuse University in 1974, and worked as a sound engineer and audio systems designer in broadcasting, studio recording, and concert reinforcement.

In 1976 he received an M.S.E.E. from The Georgia Institute of Technology, specializing in audio electronics and instrumentation, including the multidisciplinary program in acoustical engineering. That same year he joined James B. Lansing Sound, Inc. as a transducer engineer where he had design responsibility for various consumer and professional loudspeaker products, including the Cabaret Series and E Series musical instrument loudspeakers. Since 1980 he has been an applications engineer for the JBL Professional Division, and currently has additional responsibilities as product manager for JBL Professional Products.

Mr. Gander's articles have been published in the *Journal of the Audio Engineering Society, db* magazine, *Sound Arts*, and *Modern Recording*. He holds an FCC First Class Radiotelephone Operator License, and is a member of the AES, The Acoustical Society of America, and the IEEE.

CORRECTION TO "GROUND-PLANE ACOUSTIC MEASUREMENT OF LOUDSPEAKER SYSTEMS"

An error has been pointed out in the Appendix to the above engineering report.[1] The derivation of Eqs. (4) and (5) is incorrect. This should have been obvious since the units are $(\text{length})^2$ instead of (length). Assuming that $h << L$ or $2Hh + h^2 << L^2 + H^2$, and using Eq. (3), the correct equations are

$$D_d \simeq \sqrt{L^2 + H^2} + \frac{-2Hh + h^2}{2\sqrt{L^2 + H^2}} \tag{4}$$

$$D_r \simeq \sqrt{L^2 + H^2} + \frac{2Hh + h^2}{2\sqrt{L^2 + H^2}} . \tag{5}$$

These mistakes propagated through to the conclusion. The equations should have read as follows;

$$\Delta D = D_r - D_d = \frac{2Hh}{\sqrt{L^2 + H^2}} \tag{6}$$

$$\Delta\theta = \frac{2\pi f}{c} \frac{2Hh}{\sqrt{L^2 + H^2}} \tag{8}$$

$$f_{max} = \frac{c}{4\pi} \frac{\Delta\theta\sqrt{L^2 + H^2}}{Hh} \tag{9}$$

$$f_{max} = \frac{25.73\sqrt{L^2 + H^2}}{Hh} \tag{12}$$

$$f_{max} = \frac{42.88\sqrt{L^2 + H^2}}{Hh} \tag{13}$$

The effect of these corrections is to change the upper frequency limitation for a desired accuracy, device height, and microphone height. For the usual condition of $L > 1$, the upper frequency is increased. The author is grateful to Mr. Dan Joffe for pointing out the error.

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[1] M. R. Gander, *J. Audio Eng. Soc. (Engineering Reports)*, vol. 30, pp. 723–731 (1982 Oct.).