Simulated Free Field Measurements

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ABSTRACT

The development of time selective techniques has enabled measurements of the Free Field response of a loudspeaker to be performed without the need for an anechoic chamber. The low frequency resolution of both time selective techniques and anechoic measurements is, however, limited by the size of the room. A technique is presented enabling measurements of the complex Free Field response of a loudspeaker to be performed, without an anechoic room, throughout the entire audio frequency range. It is shown that this technique can also be used for measurements of harmonic distortion.

GLOSSARY OF IMPORTANT SYMBOLS

a	radius of driver (or equivalent rigid, flat, circular piston)
b	length of major semi-axis of ellipsoid
c	speed of sound (344 m/s)
d	distance from source to measurement microphone (direct sound path)
d_0	reference distance (e.g. 1 metre)
d_R	distance from source to first reflecting surface to microphone (path of first reflection)
F	frequency range
f	frequency
f_S	transition frequency – Near Field / Far Field measurement
Δf	frequency resolution (lower limiting frequency)
$H_{NF}(f)$	Near Field response of loudspeaker
$H_{FF}(f)$	Far Field response of loudspeaker
h	length of minor semi-axis of ellipsoid
k	wave number $(2\pi / \lambda)$
λ	wavelength of sound
M	largest linear dimension of source
N	harmonic order
P_0	reference electrical power level (e.g. 1 Watt)
p_0	reference sound pressure level (e.g. 20 µPa)
S_D	effective surface area of driver
S_P	effective surface area of port
T	time range
Δt	time resolution
τ	time delay
τ_0	delay in sound arrival corresponding to reference distance
x_0	dB reference
Z_0	nominal impedance of loudspeaker

1 INTRODUCTION

The purpose of this paper is to present a convenient method for obtaining the complete Free Field response of a loudspeaker in an ordinary room, making as few assumptions about the device under test as possible. Before attempting to obtain the Free Field response of a loudspeaker, it is useful to know what is meant by "Free Field". A Free Field describes idealized sound propagation with no reflections, i.e. only the direct sound from a source can be observed. This occurs naturally in open outdoor spaces, high enough above the reflective surface of the earth. It can also be created artificially by constructing an anechoic room. In this case, a large room is built using highly absorptive material on the walls, floor, and ceiling. The lowest usable frequency is determined by the depth of the absorptive material and smallest dimension of open space available, usually floor to ceiling height. Normally the volume of the device under test must be less than approximately 1% of the volume of the open space available in the anechoic room. The depth of the absorptive material is approximately equal to 1/4 λ of the lowest frequency that can be effectively absorbed. In practical terms, this translates to a minimum room dimension of

$$h = 1.5\,\lambda \tag{1}$$

(see Fig. 1). Using the definition of the speed of sound

$$c = f\,\lambda \tag{2}$$

this becomes

$$h = \frac{1.5\,c}{f} \tag{3}$$

For measurements down to $f = 20$ Hz, $h = 25.8$ meters (without absorptive material).

The opposite case, a Diffuse Field, is created when the sound pressure arrives from all directions with equal magnitude and probability. A Diffuse Field can be approximated with a reverberation chamber, which is constructed with many irregular, highly reflective surfaces.

A sound source can be considered to be a point if the wavelength of sound radiated is large compared to the size of the source. This occurs in practice when the distance from the observer to the source is large compared to size of the source. Such a point source will radiate spherically (4π sr) into a Free Field in all directions, if there are no reflections (see Fig. 2). Note that under these conditions, the sound field is very well behaved and the sound pressure level changes –6 dB for every doubling of distance from the source, as shown in Fig. 3. At this point, one is said to be in the Far Field of the source. For long, narrow line sources, the sound radiation will be cylindrical (see Fig. 4). In this case, the sound pressure level will change by –3 dB for every doubling of distance from the source. In practice, these conditions only exist over a limited frequency and distance range for a finite size source and finite size room. When the distance to the source is small compared to the wavelength of sound, one is said to be in the Near Field of the source. This typically occurs at low frequencies, where the wavelength (and radius) of a spherical wave is so large that the wave is essentially a plane. Theoretically, there is no change in sound pressure level with distance for plane wave radiation.

Therefore, it can be said that the Far Field and the Near Field are determined by the *relative size of the source* compared to the frequency of sound radiated. The Free Field and Diffuse Field are functions of *the room or environment.*

2 TIME SELECTIVE TECHNIQUES

All time selective techniques rely upon the constant speed of sound in order to separate the desired direct sound component from reflections, which arrive at the measurement microphone delayed due to a longer travelled path [1]. Therefore, differences between the various techniques are found only in terms of signal to noise, measurement time, and frequency selectivity [2]. In practice, the measurement time range, T, is limited in order to exclude unwanted reflections and noise. This can be done as part of the measurement or in post-processing. The Uncertainty Principle, in turn, determines the frequency resolution, Δf, of the measurement. Disregarding the actual shape of the time window for the moment, this is

$$\Delta f = \frac{1}{T} \tag{4}$$

The relation holds for <u>any</u> analysis or technique.

Using time selective techniques, it is possible to obtain the Free Field response of a loudspeaker by measuring in an ordinary room. This is done by restricting the time range to prevent reflections from entering the measurement. Fortunately, no special treatment of the walls, floor, or ceiling is necessary. The absorption influences the reverberation time, however, and consequently the total measurement time for impulse and gating techniques [2].

Fig 5 shows a loudspeaker and a measurement microphone situated in an ordinary room. The distance travelled by the direct sound is d. From the loudspeaker to the nearest reflecting surface to the microphone, the sound travels a total distance of

$$d_R = d_{R1} + d_{R2} \tag{5}$$

The difference in path length between the direct sound and the first reflection determines the time available for a Free Field measurement

$$T = \frac{d_R - d}{c} \tag{6}$$

where c is the speed of sound. This, in turn, determines the lower frequency limit for the measurement. From Eq. (4)

$$\Delta f = \frac{c}{d_R - d} \tag{7}$$

It is easily seen that the loudspeaker, measurement microphone, and nearest reflecting surface define an ellipsoid, with the loudspeaker and microphone at the focal points. The ellipsoid is a solid body of rotation, containing no reflective surfaces, about the major semi-axis which contains the microphone and loudspeaker focal points. Any surface touching the outer shell of the ellipsoid will cause a reflection to appear at the microphone that has travelled a distance d_R.

In general, the distance travelled by the first reflection is equal to the length of the ellipsoid along the major semi-axis

$$\begin{aligned} d_R &= b \\ &= \sqrt{h^2 - d^2} \end{aligned} \tag{8}$$

For the typical case where

$$d_R = 2d \tag{9}$$

the height of the ellipsoid along the minor semi-axis, h, is found to be

$$\begin{aligned} h &= \sqrt{d_R^2 - d^2} \\ &= d\sqrt{3} \end{aligned} \tag{10}$$

usually the floor to ceiling height. The eccentricity of the ellipse, ε, is defined as the ratio of the distance between the focal points and the length of the ellipse along the major semi-axis

$$\begin{aligned} \varepsilon &= \frac{d}{b} \\ &= 0.5 \end{aligned} \tag{11}$$

and the ratio of the length of the major semi-axis to the length of the minor semi-axis is

$$\frac{b}{h} \approx 1.15 \qquad (12)$$

The problem of optimization then becomes that of placing the largest possible ellipsoid with eccentricity $\varepsilon = 0.5$ within a given room. In an ordinary rectangular room, this is usually done by centering the major semi-axis of the ellipsoid, b, along the longest horizontal room dimension, L. If $L > 1.15\, h$, from Eq. (10) we obtain

$$\begin{aligned} d &= \frac{h}{\sqrt{3}} \\ &\approx 0.58\, h \end{aligned} \qquad (13)$$

For $L < 1.15\, h$

$$d = \frac{L}{2} \qquad (14)$$

In general, it is not worthwhile to carry the optimization too far, as will be seen in Section 3. It is useful, however, to keep a mental picture of the ellipsoid in mind when arranging the test set-up in the room, to avoid the influence of disturbing objects that may cause reflections [2]. For this application, the magnitude of the Time Response is useful for viewing the arrival of the direct sound component, accurately fixing the measuring microphone distance, and determining the delay before the arrival of the first reflection.

For the response to be measured in the Far Field of the source, d should be at least 3 times the largest linear dimension of the source, M

$$d > 3M \qquad (15)$$

Depending upon the size of the source, this may be some distance other than 1 metre (or other reference distance). Because simulated Free Field conditions have been obtained (i.e. the sound field is well known) the measured response can easily be normalized to any desired reference (see Appendix A). For measurements in the Far Field of the source, the distance, d, is then determined by the size of the device under test. In general, assuming Eq. (15) is fulfilled and substituting Eq. (8) into Eq. (7) we obtain

$$\Delta f = \frac{c}{\sqrt{h^2 - d^2} - d} \qquad (16)$$

This expression is deliberately left unsimplified (using the assumptions of Eqs. (9), (10), (12), and (14)) in order to show the dependence of Δf on both room size and test object size if Eq. (15) is also fulfilled. Therefore, the lowest measurable frequency using any time selective technique is a function of the room size, where h is the smallest room dimension (usually floor to ceiling height) and test object size, which determines the distance to the measurement microphone, d, according to Eq. (15).

Whenever attempting to mentally visualize this problem, it is useful to think of a "small box (the loudspeaker) inside a large box (the room)". When the ratio between the room size and the loudspeaker size is large, it will be relatively easy to perform Free Field measurements over a wide frequency range, F. As the ratio decreases (i.e. the room size is decreased or the loudspeaker size is increased), it will become increasingly difficult to perform Free Field measurements.

In contrast to the Time Delay Spectrometry technique, where the time window is determined by the use of a tracking filter [3], it is advantageous to apply the time window as a post-processing operation [4]. This is done in order to have a more well defined time window and to obtain the optimum frequency resolution. It also allows the entire measurement, including reflections, to be viewed in the time domain before applying the window. This implies an inherently linear, constant bandwidth data format for the Far Field measurement for use of the forward and inverse Fourier Transforms. Reflections are removed from the Far Field measurement by multiplying the measured response (in the time domain) by a window function. Consequently, the frequency response is convoluted with the Fourier transform of this window. The best time selectivity and minimum resulting frequency domain spreading are obtained by the use of a variable width rectangular time window with user definable leading and trailing Hanning ($\cos ine^2$) tapers (see Fig. 6). With respect to "worst case" frequency resolution, the effective time range, T, is determined by the width of the rectangular portion of the time window. It will be seen in Section 4 that the actual shape of the time window becomes less critical if the low frequency response of the system can be obtained by another method.

As mentioned previously, the choice of test method should be based on speed, its effect on the test object, selectivity in both time and frequency, and the maximum available signal to noise ratio. In addition, it should be possible to accurately specify the input power to the device under test, unless the system is known to be perfectly linear. Transducers, and loudspeakers in particular, do not normally fall into the category of perfectly linear systems, particularly not over a wide dynamic range. Using sinusoidal excitation, the time required for the Far Field measurement is determined entirely by background noise. In an ordinary room, this should be no more than about 1 second [5]. An improvement of 3 dB in S/N is gained for every successive doubling of measurement time [3], [6]. The sine wave has a low crest factor and its power can be determined easily and precisely. Sinusoidal excitation enables ideal frequency selectivity, offering the possibility for harmonic measurements.

3 NEAR FIELD MEASUREMENTS

The low frequency response of a loudspeaker can be obtained using the Near Field technique. This is done by simply moving the microphone very close to the low frequency driver(s) in the loudspeaker system (see Fig. 7). This of course, assumes that there is little or no passive radiation from the walls of the enclosure, i.e. the structure is essentially rigid. To avoid measurement errors, the measurement microphone should be as close as possible to the driver under test.

A microphone distance of

$$d < 0.11a \tag{17}$$

results in measurement errors of less than 1 dB. The microphone should be placed as near to the centre of the diaphragm as possible [7]. This technique physically eliminates reflections and noise. Because there is no time delay, the measurements can be carried out entirely in the frequency domain. The theoretical upper frequency limit for performing Near Field measurements is given by

$$ka = 1 \tag{18}$$

where k is the wave number. Substituting Eq. (2) into Eq. (18), this becomes

$$f_{ka=1} = \frac{c}{2\pi a} \tag{19}$$

or

$$f_{ka=1} = \frac{58.749}{a} \tag{20}$$

for driver radius, a, in metres. For convenience, this can be rewritten as

$$f_{ka=1} \leq \frac{10949.86}{2a} \tag{21}$$

in terms of the driver diameter, $2a$, in centimetres. Fig. 8 shows the upper limiting frequency for Near Field measurements versus driver diameter in centimetres. For driver diameters specified in inches, this becomes

$$f_{ka=1} \leq \frac{4310.97}{2a} \tag{22}$$

For ported loudspeakers or enclosures containing multiple drivers, additional Near Field measurements of the individual sources are required. The complete Near Field response is found by first scaling the measurement of the port(s), and then taking the complex sum of all measurements [7], (see Appendix B).

Because the microphone is so close to the driver (or port), the measured sensitivity appears considerably higher than an equivalent Far Field measurement. The Far Field sensitivity can be calculated from

$$H_{FF}(f) = \frac{H_{NF}(f)\, a}{2\sqrt{2}\, d} \tag{24}$$

or

$$H_{FF}(f) = H_{NF}(f) - 20\log_{10}\frac{2\sqrt{2}\, d}{a} \quad \text{[dB]} \tag{25}$$

where $H_{NF}(f)$ is the measured Near Field response and $H_{FF}(f)$ is the Far Field response at a microphone distance, d, for a driver of radius a. For ported loudspeakers, the radius, a, is taken as the radius of the driver. This means that at low frequencies, where the loudspeaker behaves like a rigid piston, the measured Near Field response is directly proportional to Far Field response and is independent of the environment into which the loudspeaker radiates [7], [8]. Alternatively, the level offset can be determined directly by comparison to a Far Field measurement in the "overlap" frequency range

$$\Delta f \leq f \leq f_{ka=1} \tag{26}$$

where both the Near Field and Far Field measurements should be valid (see Fig. 9). The actual upper frequency limit for the "overlap" range will be somewhat lower, as will be explained in Section 5. As it is usually not possible to reduce the size of the loudspeaker once it is created, it is necessary to perform the Far Field measurements in a room of sufficient size so that an "overlap" frequency range exists. Combining the two methods of determining the sensitivity offset and solving Eq. (24) for the radius, a, yields

$$a = \frac{2\sqrt{2}\, d\, H_{FF}(f)}{H_{NF}(f)} \tag{27}$$

for a frequency, f, in the "overlap" range, when $H_{NF}(f)$ and $H_{FF}(f)$ are measured independently. For individual drivers, this interesting result provides an empirical method for determining the *effective* radius, a, and subsequently the effective radiating surface area of the driver, S_D ($S_D = \pi a^2$). This parameter is critical in the determination of the Theile-Small parameters.

For frequencies above $f_{ka=1}$, most drivers no longer behave like a rigid piston (i.e. modal behaviour or "cone break-up") and the relationship between Near Field and Far Field response becomes complex. The frequency range for the Near Field measurement should therefore be chosen to an upper frequency of no greater than $f_{ka=1}$. The lower frequency should be less than or equal to 20 Hz, (lower for large diameter or extended range woofers) in order to include the low frequency roll-off of the system. This is important for examining the time domain behaviour of the system.

Using stepped sine excitation, Near Field measurements can be performed at discrete frequencies in a logarithmic (ISO preferred) frequency format, typically 1/12 octave (ISO R40) or 1/24 octave (ISO R80). This optimizes low frequency resolution and enables measurements of harmonic distortion to be performed [6].

4 FULL RANGE RESPONSE MEASUREMENTS

If a full audio range response (typically 20 Hz – 20 kHz) is desired, the measurement can be performed in two passes. The first measurement is performed in the Far Field using a calibrated microphone. The microphone can then be repositioned into the Near Field to measure the low frequency response. Assuming an "overlap" frequency range exists, a transition frequency, f_S, within this range, can be selected at which to join these measurements. With the exception of the size of the room, no other assumptions are made. Post-processing is then performed to account for the time delay to the Far Field microphone, for the application of a time window to the Far Field measurement to isolate the direct sound component, and to account for the sensitivity offset between the Far Field and Near Field measurements. All block arithmetic operations must be complex so that other functions such as phase, group delay, and the time response are available for the final, full frequency range response data.

To test the application of this method, measurements using a Brüel & Kjær Type 2012 Audio Analyzer and a calibrated Brüel & Kjær Type 4133 Free Field microphone were performed on a small (18 x 11 x 11 cm) two-way closed box loudspeaker, with a 8.5 cm diameter woofer. The output of the measurement system was calibrated to account for the gain and frequency response of the power amplifier used to drive the loudspeaker under test. This enabled direct measurement of the system response at the input and output terminals of the device under test. All post-processing was carried out directly in the analyzer. All measurement results can be exported as ASCII data for further external post-processing (e.g. polar response, directivity, statistics, etc.).

The first measurement was of the loudspeaker response in the Far Field. Fig. 10 shows the result of this measurement, including the effect of room reflections. The Time Response Magnitude clearly shows the arrival of the direct sound as well as the delay before the arrival of reflections. The delay before the arrival of reflections determines the time range,

T, available for a time selective measurement. If the time range is insufficient, the microphone/loudspeaker setup must be repositioned or the measurements may need to be performed in a larger room. Reflections are convoluted with the response of the loudspeaker in the frequency domain, resulting in a "ragged" frequency response.

The peak in the Time Response indicates the delay, τ, until the arrival of the direct sound at the microphone, and consequently the exact distance, d, to the microphone position. A time shift can be applied to the Far Field measurement to compensate for this delay, which is not an actual part of the loudspeaker response (see Fig. 11). This allows the final phase response to be displayed without "wrapping". This is followed by the application of the time window to the Far Field measurement (see Fig 12). The resulting frequency response appears "smoothed", due to the removal of time domain reflections which cause frequency domain ripple.

The microphone is then repositioned in the Near Field of the woofer and the low frequency response is measured. This appears with a higher sensitivity than the Far Field Measurement (see Fig 13). The cursor can be used to read the offset between the measurements at f_S. Note that this is also done for the phase response. Alternatively, the Far Field measurement can be divided by the Near Field measurement to obtain the offset at f_S. The low frequency measurement is then multiplied by this complex offset constant (magnitude and phase) to match the sensitivity of the high frequency measurement at f_S. The magnitude of the offset can also be calculated according to Eq. 25. The scaled Near Field measurement is shown along with the Far Field measurement in Fig. 14, after the application of Eq. 25. Note the "overlap" range. The transition frequency, f_s, is then selected within this region.

In order to preserve low frequency resolution and to avoid a discontinuity in frequency resolution between the upper and lower frequency ranges at f_S, the Far Field data is converted to a logarithmic (constant percent bandwidth) format equivalent to the ISO R80 format of the Near Field measurement. This is a straight line interpolation algorithm on a dB vs. logarithmic frequency axis. This has the added advantage of showing the response in the frequency domain in an optimum format, as it is universally accepted as standard industry practice to show the frequency response in dB, on a logarithmic frequency scale, preferably with the IEC 87263 aspect ratio. A linear data set in the frequency domain tends to concentrate resolution at the higher frequencies on a logarithmic frequency axis. An additional benefit is that the final response contains the ISO preferred frequencies, so it is a simple matter to obtain the necessary values for the calculation of sensitivity, directivity, etc. Due to the logarithmic format, the data for both measurements now exists only in the frequency domain.

Afterwards, rectangular frequency windows are applied to both measurements. The Near Field, low frequency measurement is windowed to eliminate data at frequencies above f_S. The Far Field measurement is windowed to exclude data at frequencies below f_S. Assuming $f_S > 1/T$, this also eliminates invalid data remaining after the application of the time window. Now the two responses no longer overlap. The final process is to add the two responses together to obtain a single, continuous complex data set. The magnitude of the frequency response for the complete system, from 20 Hz to 20 kHz, appears in Fig 15. The phase and group delay are shown in Fig. 16.

Because the data is complex, the final response can be reconverted to a linear format (without the effect of reflections) before applying an Inverse Fourier Transform to calculate the time response. In this way, the measured data is displayed optimally in both the frequency and time domains. Extending the measurement frequency range, F, to 40 kHz normally eliminates the need for any windowing of the frequency response prior to calculation of the time response. Most loudspeaker systems designed for the normal audio range will be sufficiently rolled off, or "self-windowed" at both the upper and lower ends of a 1 Hz to 40 kHz frequency spectrum. This subsequently eliminates artifacts caused by such windowing. The resulting time resolution, Δt, given by the uncertainty principle is

$$\Delta t \approx \frac{1}{F} \tag{28}$$

For an "extended range" loudspeaker, with a frequency response that has not rolled off sufficiently at 40 kHz, a half-Hanning frequency window from 20 kHz – 40 kHz can be applied before calculation of the time response, as recommended in reference [9]. This has the advantage of not affecting the magnitude of the frequency response in the audio bandwidth and especially not affecting the response at low frequencies. The magnitude of the time response is seen in Fig. 17. The real part of the time response (traditional Impulse Response) can also be displayed (see Fig. 18).

The measurements and post-processing were performed in less than one minute, including the time to manually move the microphone for the Near Field measurement. The entire process can be automated using the Autosequence facility of the analyzer. If desired, text can be displayed on the screen to prompt the user for movement of the microphone, choice of the transition frequency, etc. or to document measured results.

Fig. 19 shows a comparison of the response of this loudspeaker measured in an ordinary room using this technique to the response of the same loudspeaker measured using traditional techniques (i.e. not time selective) in an anechoic chamber. Differences at low frequencies are due to errors introduced by the size of the anechoic chamber (7.7 x 6.5 x 6.6 m), according to Eq. 3.

To further test the technique, a ported loudspeaker (39 x 23 x 22 cm) was also measured. Fig. 20 shows the individual measurements of the driver and of the port. The measurement of the port is first scaled by the ratio of its surface area to the effective radiating surface area of the driver (see Appendix B). This scaled response is then summed (complex summation) with the Near Field response of the driver. The complete Near Field response of the ported loudspeaker is shown in Fig. 21. The resulting simulated Free Field response of the ported loudspeaker system (20 Hz – 20 kHz) appears in Fig. 22. It should be noted that the majority of the measurements were carried out in one of the author's apartments, in a room 4.5 x 2.7 x 4.5 m, with the furniture in place, and that the initial tests of the method were done in an even smaller office cubicle !

Fig. 23 shows a comparison of the response of this loudspeaker measured using the simulated Free Field technique to the response of the same loudspeaker measured in an anechoic chamber. Even greater differences can be observed at low frequencies. This is due to the use of a greater microphone distance for the Far Field measurement, necessary because of a larger source size. Fig. 24 clearly reveals the effect of the anechoic chamber on the measured response with increasing microphone distance, particularly at low frequencies, even for an optimization of source/microphone positioning within the room. This is caused by the limited size of the anechoic chamber and insufficient absorption of low frequency reflections. Increasing the microphone distance correspondingly lowers the path length difference between the direct and reflected sound, and consequently increases the lower frequency limit according to Eq. 6. Note that the level of the measured response decreases by 6 dB for each doubling of microphone distance, empirically establishing that the microphone is in the Far Field of the source.

5 CHOICE OF TRANSITION FREQUENCY

Several possibilities exist for choosing the exact transition frequency f_S. In general, f_S should be chosen as high as possible, in order to preserve low frequency resolution, but above $1/T$. In any case, it should be the Near Field response which is manipulated relative to the Far Field response. The level of the Far Field response should not altered, assuming it was calibrated. The first possibility would be to simply apply Eq. (25) and to choose $f_S = 1/T$. Choosing this transition frequency reduces the apparent resolution around f_S due to the "smoothing" effect of the time window on the measured Far Field response at its lower frequencies, i.e. in this frequency range, there is more detail in the Near Field measurement. Another method is to see if, after using Eq. (25), the two measurements intersect at some frequency. This frequency can then be used as f_S if it conforms to the aforementioned restrictions. Alternatively, the magnitude of the Near Field response could be adjusted for a visual "best fit".

Because the drivers are generally mounted in an enclosure somewhat larger than the driver itself, at high frequencies (smaller wavelengths), the low frequency driver(s) will no longer radiate spherically in the Near Field, due to the "baffle" effect of the enclosure [4], [7]. This causes a transition from 4π (spherical) to 2π (hemispherical) radiation to occur over several octaves as the wavelength approaches the size of the source for a Near Field measurement. Therefore, the useful upper frequency limit for Near Field measurements is always lower than $f_{ka=1}$. This limit seems to be governed by the overall size of the test object, rather than by the driver radius. The Near Field measurement will progressively under-estimate the true Free Field response at higher frequencies. In practice, we have seen that the Near Field response can be used with errors of less than 1 dB (compared to anechoic measurements) at frequencies where the wavelength is greater than approximately 3 times the major dimension of the source, M. An attempt to explain this behaviour can be found by substituting M for $2a$ in Eq. (18). In terms of an upper limiting frequency, this is

$$f_S \leq \frac{c}{\pi M} \tag{29}$$

This is therefore the most critical factor governing the selection of f_S.

For multi-way systems with a crossover frequency, f_X, less than f_S, a lower transition frequency, f_S, should be chosen such that

$$f_S < f_X \tag{30}$$

to avoid roll-off in the Near Field response due to the crossover filter.

6 HARMONIC DISTORTION

Using sinusoidal excitation also enables time selective and Near Field measurements of harmonic distortion to be performed [5],[6]. While the Brüel & Kjær Type 2012 allows any selected harmonic(s) up to the 20th to be measured, practical considerations for full audio bandwidth tests (such as the range of human hearing, the frequency range of the measurement microphone, and the low energy level of upper harmonics) typically limit full audio bandwidth measurements (20 Hz – 20 kHz) to the 2nd and 3rd harmonics. More useful results for nonlinear measurements can be obtained using two tone techniques, but here the one-to-one relationship between time and frequency is lost and we are limited to measurements in an anechoic room or Near Field measurements [10].

The measurement of a harmonic response results in a frequency multiplication of the desired response by the harmonic order. Consequently, the upper limiting frequency for Near Field measurements is correspondingly reduced to the frequency where

$$ka = \frac{1}{N} \tag{31}$$

or

$$f_{ka=1} = \frac{c}{2\pi \, a \, N} \tag{32}$$

where N is the harmonic order. Fortunately, the lower limiting frequency for Far Field time selective measurements is correspondingly lowered as well

$$\Delta f = \frac{1}{NT} \tag{33}$$

Fig. 25 shows the separation of individual harmonic components in time and frequency for a linearly swept sinusoidal excitation. Each Far Field harmonic must therefore be measured with an individual sweep, in order to isolate the desired component from reflections and other harmonics. The Near Field harmonics can be tested at discrete frequencies in a single pass. The transition frequency for each harmonic, $f_S(N)$, should also be a function of the harmonic order

$$f_S(N) = \frac{f_s(1)}{N} \tag{34}$$

where $f_S(1)$ is the transition frequency selected for the fundamental. The measurement algorithms optimize accuracy and measurement time, reject uncorrelated background noise and suppress adjacent harmonics [5], [6].

Harmonic distortion measurements were performed on the closed box loudspeaker using this technique (see Fig. 26). Note that these are true harmonic responses (not THD plus noise). Total Harmonic Distortion is shown in Fig. 27. Good correlation was found with equivalent measurements performed in an anechoic chamber, especially at higher frequencies (see Fig. 28).

7 CONCLUSION

A technique has been presented enabling measurements of the complex Free Field response of a loudspeaker to be performed, without an anechoic room, throughout the entire audio frequency range. One measurement is performed using a time selective technique in the Far Field of the source. The low frequency response is then obtained using the Near Field technique. Assuming an "overlap" frequency range exists, a transition frequency can be selected at which to join these measurements. The sine-based test methods employed for this technique optimize speed, selectivity in both time and frequency, and maximize the available signal to noise ratio. In addition, sinusoidal excitation provides a low crest factor signal and enables accurate specification of the input power to the device under test. This technique also presents the resulting data optimally in both the frequency and time domains. These results are available in any desired coordinates and can be exported for further processing. Furthermore, it has been demonstrated that, using a sine-based analysis, this technique can be extended to include measurements of harmonic distortion. The only limits for this technique are, in fact, imposed by the size of the room used for performing the tests. These limits are, however, much less critical than with either purely time selective techniques alone or with traditional measurements in an anechoic chamber. The effects of the anechoic chamber on the measured response at low frequencies, caused by its limited size and insufficient absorption of low frequency reflections, are eliminated. A comparison to traditional anechoic measurements revealed the magnitude of these chamber induced errors. This technique, therefore offers a convenience hitherto unavailable to the majority of loudspeaker designers.

8 APPENDIX A

dB REFERENCE

The magnitude of the frequency response of a loudspeaker is usually plotted in decibels versus frequency. In order to be able to correctly interpret this curve, it is necessary to know the decibel reference. Recall that decibels express a logarithmic power ratio

$$dB = 10\log_{10}\frac{P}{P_0} \tag{35}$$

where P_0 is the reference power.

Normally, power is not measured directly. Therefore, this expression can be rewritten as

$$dB = 20\log_{10}\frac{x}{x_0} \tag{36}$$

where x_0 is the corresponding signal reference. This is derived from the fact that the ratio of the square of two signals is equal to the ratio of their powers. The reference for a transfer function has dimensions and can be found by dividing the reference for the output by the reference for the input. For a loudspeaker, x_0 is in Pa/V. Therefore, if the output is desired in dB SPL, the output reference is 20 μPa.

Typically, the reference for the input is given as a power (such as 1 Watt). The signal applied, however, is in Volts, so the power will depend upon the impedance of the loudspeaker. In general, the voltage is found that will dissipate the reference power across a resistance equal to the loudspeaker's specified nominal impedance

$$P_0 = \frac{V^2}{Z_0} \quad [W] \tag{37}$$

so

$$V = \sqrt{P_0 Z_0} \quad [V] \tag{38}$$

where V is the corresponding input reference voltage. This refers the measured response to the reference power, regardless of the actual level of the excitation signal. This of course assumes that the device under test behaves linearly (i.e. no power compression).

Because the output terminals of a loudspeaker are not intrinsically obvious, it is also necessary to specify a reference distance. Typically, this is 1 metre. For Free Field measurements of a point source (i.e. spherical sound radiation), the sound pressure level is inversely proportional to distance (-6 dB change in output level for every doubling of distance). This is easily verified empirically. As explained earlier, it may be necessary to perform measurements at some distance other than the reference distance, in order to be in the Far Field of the source. Fortunately, this can easily be incorporated into the dB reference using either distance or time delay, assuming a constant speed of sound. The complete reference then becomes

$$x_0 = \frac{p_0}{\sqrt{P_0 Z_0}} \cdot \frac{d_0}{d} \quad [\mathrm{Pa/V}] \tag{39}$$

Typically, p_0 is 20 µPa, P_0 is 1 Watt, and d_0 is 1 metre; e.g. for an 8Ω loudspeaker measured at 1 metre (Z_0 = 8Ω and d = 1m), x_0 = 7.07µ Pa/V.

For a constant speed of sound, c, time delay can be equated to distance as

$$\tau = \frac{d}{c} \quad [\mathrm{s}] \tag{40}$$

In terms of time delay, the dB reference can then be written as

$$x_0 = \frac{p_0}{\sqrt{P_0 Z_0}} \cdot \frac{\tau_0}{\tau} \quad [\mathrm{Pa/V}] \tag{41}$$

For c = 344 m/s and d_0 = 1 metre, τ_0 = 2.91 ms. Normally, the time delay can be obtained directly from the magnitude of the Time Response by locating the largest peak, corresponding to the arrival of the direct sound.

For line sources, such as tall, narrow ribbon loudspeakers and tweeter line arrays, the sound pressure level changes by -3 dB for every doubling of distance of the measurement microphone, under Free Field conditions (recall Fig. 4). Applying this relationship, the dB reference becomes

$$x_0 = \frac{p_0}{\sqrt{P_0 Z_0}} \cdot \sqrt{\frac{d_0}{d}} \quad [\mathrm{Pa/V}] \tag{42}$$

or, in terms of time delay

$$x_0 = \frac{p_0}{\sqrt{P_0 Z_0}} \cdot \sqrt{\frac{\tau_0}{\tau}} \quad [\mathrm{Pa/V}] \tag{43}$$

This is also easily verified empirically. Of course, if d is increased sufficiently, even a line source will appear to behave like a point source. Provided the measurements are calibrated, the loudspeaker sensitivity, and Directivity can easily be calculated from these dB values.

9 APPENDIX B

Near Field Measurements of Ported Enclosures and Multiple Driver Sources

The low frequency response of ported loudspeakers and loudspeakers containing multiple drivers can also be measured using the Near Field technique described by Keele [7]. The complex response of each source (driver or port) should be measured individually. These complex responses can then be summed. The overall Near Field response is given by

$$H_{NF}(f) = H_D(f) + \frac{S_P}{S_D} H_P(f) \tag{44}$$

where $H_D(f)$ is the Near Field measurement of the driver, S_P is the total radiating surface area of the port(s), S_D is the total radiating surface area of the driver(s), and $H_P(f)$ is the Near Field measurement of the port. Passive radiator systems can also be measured with this method. In this case, the passive elements are simply treated as ports.

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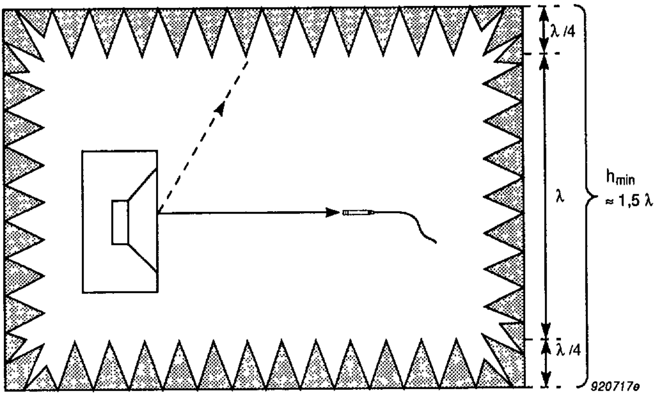

Fig. 1
The lower limiting frequency for anechoic measurements is determined by the depth of the absorptive material and the size of the room.

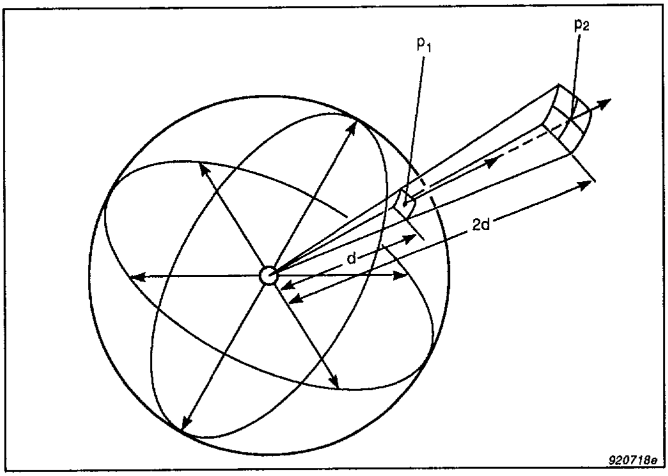

Fig. 2
Point source radiating into a Free Field. The sound pressure level decreases by 6 dB for every doubling of distance from the source.

Fig. 3
Definition of sound fields for a point source. The Near and Far Fields are functions of the source. The Free and Diffuse fields are functions of the environment.

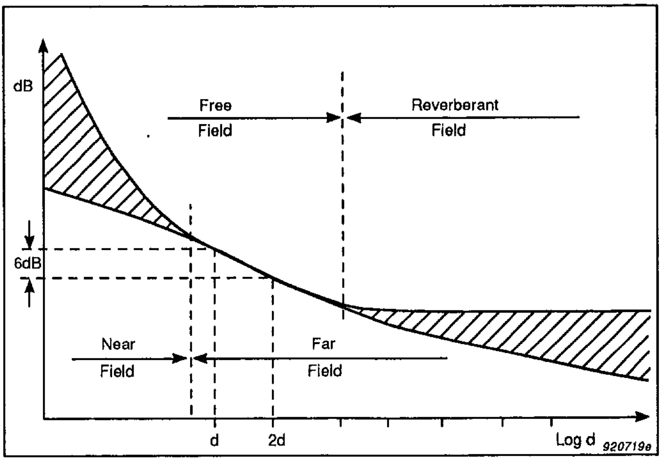

Fig. 4
Line source radiating into a Free Field.The sound pressure level decreases by 3 dB for every doubling of distance from the source.

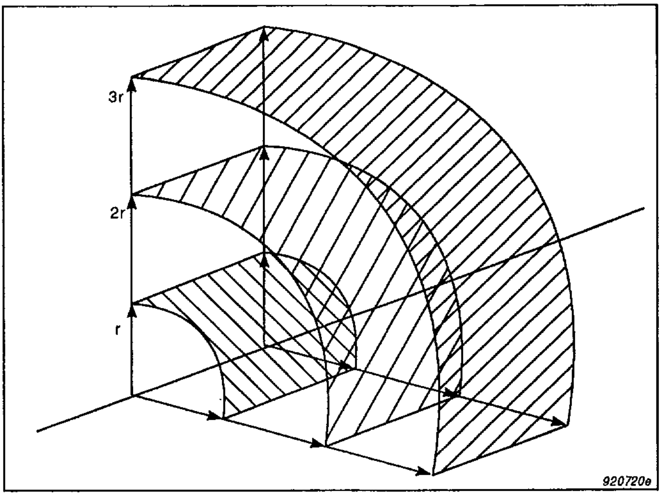

Fig. 5
The loudspeaker, measurement microphone and the nearest reflecting surface define an ellipsoid within which Free Field measurements can be performed using time selective techniques.

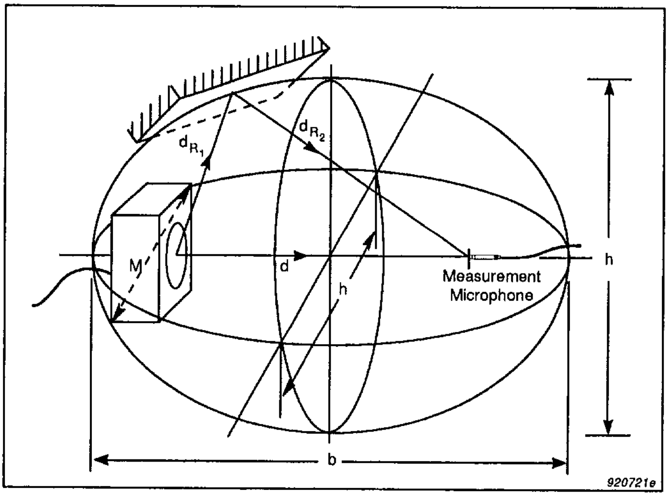

Fig. 6
The effect of reflections can be removed from the Far Field measurement by multiplying the measured time response by a Time window. This window is Hanning tapered to reduce frequency domain spreading.

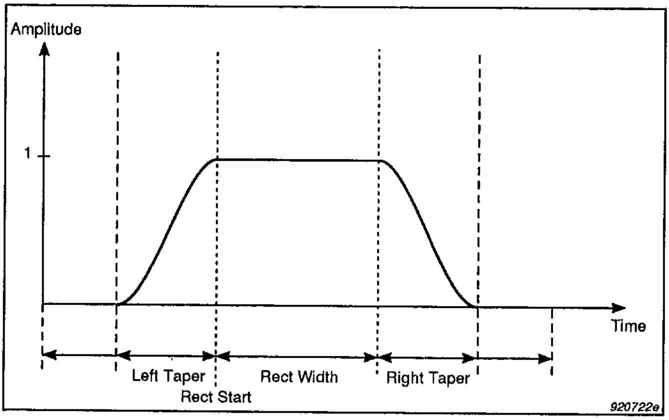

Fig. 7
Low frequency measurements can be performed by placing the microphone in the Near Field of the loudspeaker. The microphone should be placed as near as possible to the centre of the diaphragm.

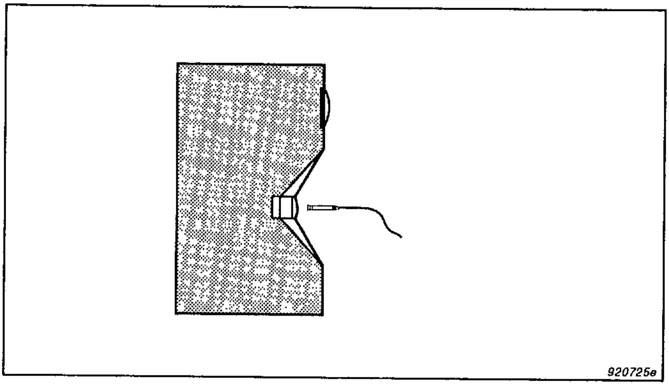

Fig. 8
The highest frequency at which Near Field measurements can be performed is determined by the size of the driver. Upper frequency limits are shown vs. driver diameter (in cm).

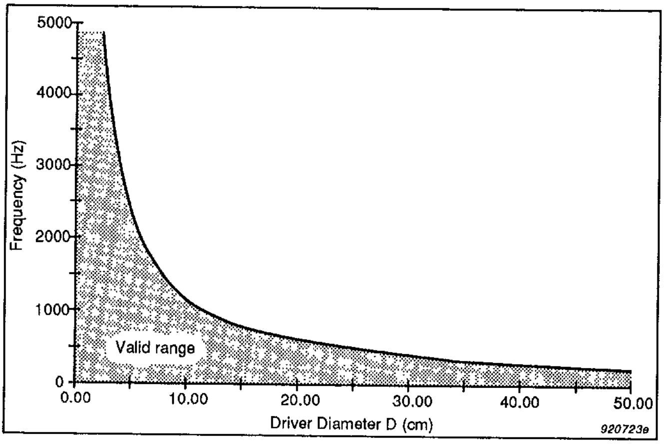

Fig. 9
For a sufficiently large room, a frequency range exists where both the Near Field and Far Field measurements are valid.

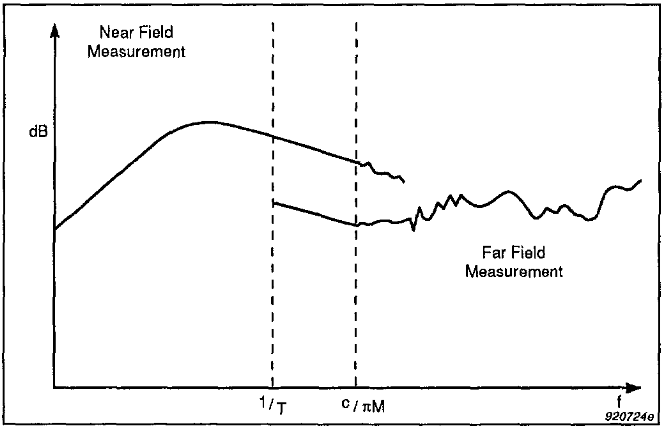

Fig. 10
The magnitude of the Time Response (lower graph) shows the arrival of the direct sound at the measurement microphone followed by the arrival of reflections. The effect of the reflections appears as "ripple" in the Frequency Response (upper graph).

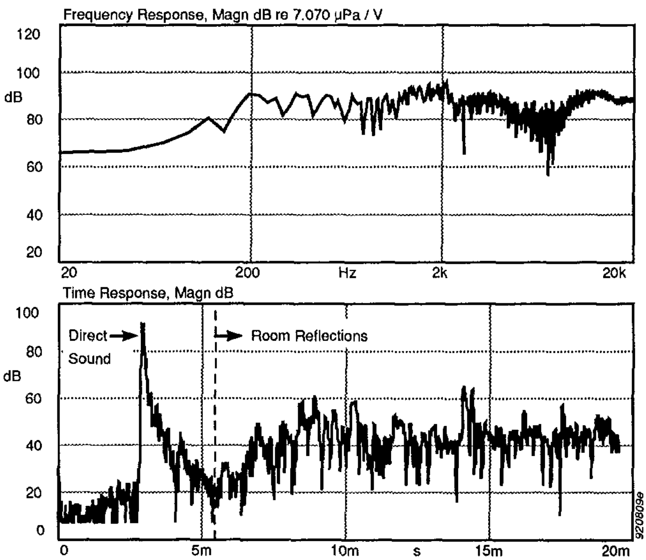

Fig. 11
A time shift is applied to the measured response to account for the delay to the measurement microphone. This is followed by the application of a time window to isolate the direct sound component. The inner lines indicate the rectangular portion of the window. The outer lines indicate the leading and trailing Hanning tapers.

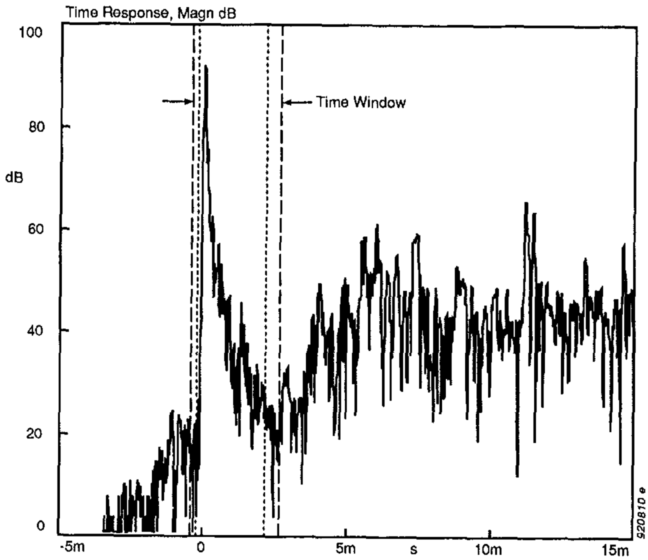

Fig. 12
The Time Response after the application of the Time window (lower graph). The resulting Frequency Response is shown in the upper graph.

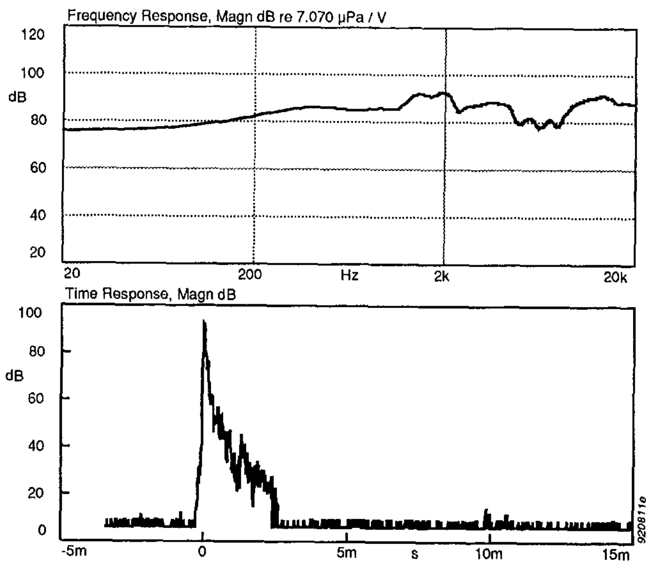

Fig. 13
The low frequency Near Field measurement (upper curve) shows a higher apparent sensitivity, compared to the Far Field measurement, due to the decreased distance to the microphone.

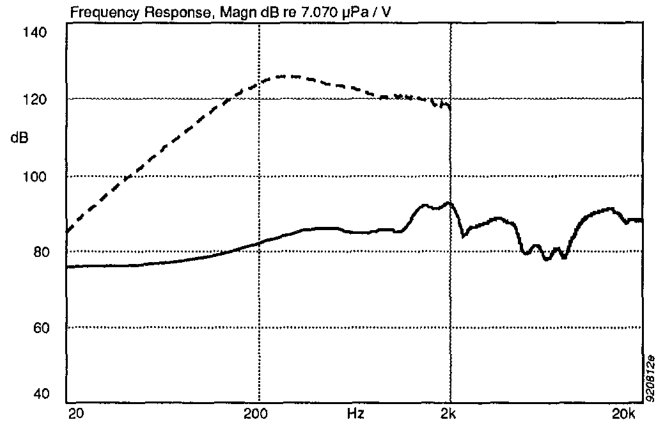

Fig. 14
The Near Field response after the application of the level offset. Note the "overlap" range where both measurements yield the same result. The cursor indicates the selected transition frequency, f_S.

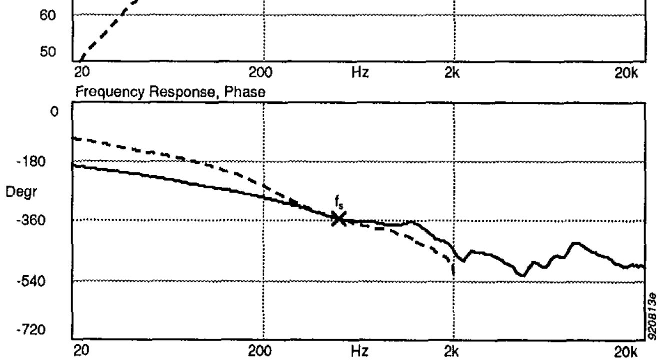

Fig. 15
Resulting on-axis Frequency response, 20 Hz – 20 kHz, ISO R80 format (1/24 octaves), ref. 20μPa / 1W at 1m, grille off, f_s = 545 Hz (David Visonik 5001, 2-way, 4 Ω, closed box loudspeaker).

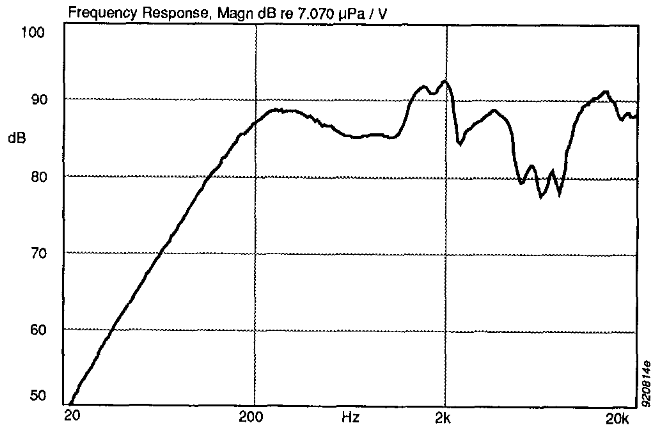

Fig. 16
Phase (upper graph) and Group Delay (lower graph) for the same 2-way closed box loudspeaker.

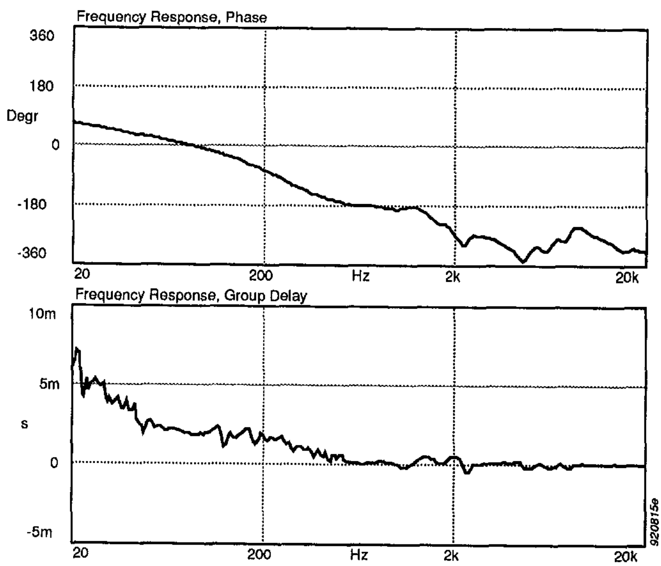

Fig. 17
The Time Response (Magnitude) of the system can be calculated from the full range Frequency response by converting the Free Field data to a linear format (half-Hanning window, 20 kHz – 40 kHz, applied to Frequency Response).

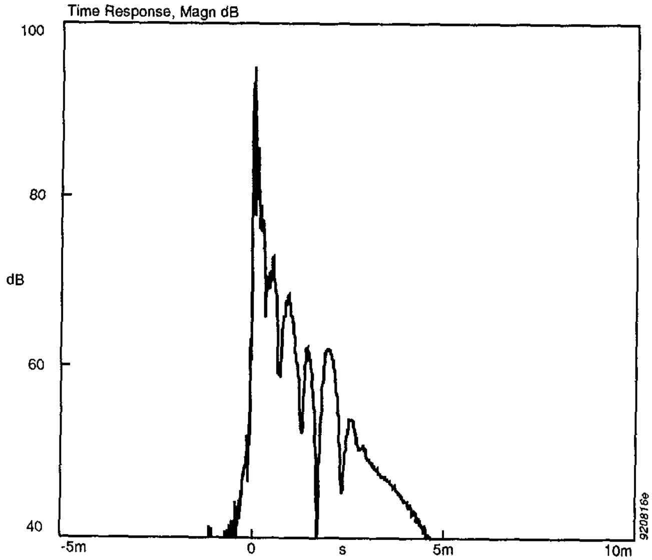

Fig. 18
The Real part of the Time Response (Impulse Response).

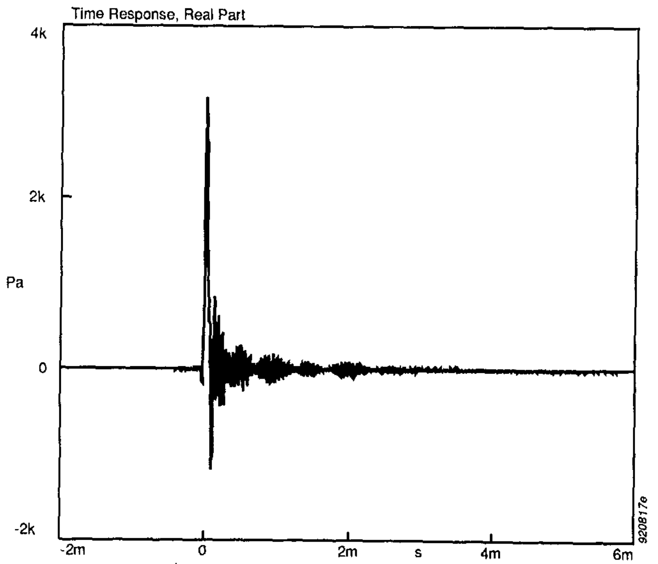

Fig. 19 a
The simulated Free Field technique shows good correlation with measurements performed in an anechoic chamber at high frequencies and a significant improvement at low frequencies.

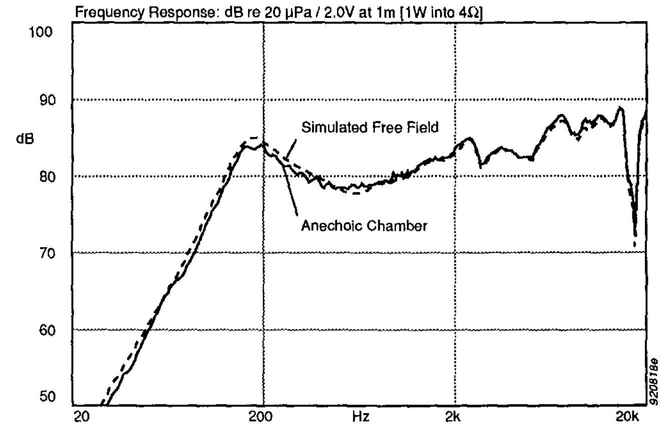

Fig. 19 b
Phase response. Note that the resulting data set is complex.

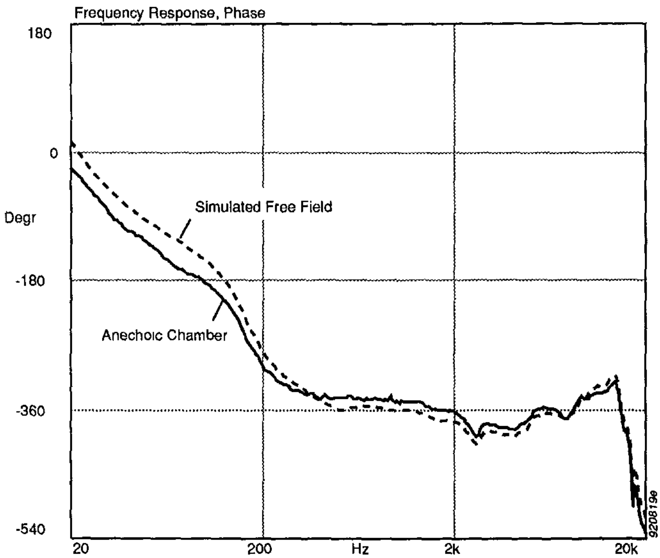

Fig. 20
The low frequency response of the ported loudspeaker system is found by performing individual Near Field measurements of the driver(s) and of the port(s). The response of the port is scaled relative to the surface area of the driver. These complex responses are then summed.

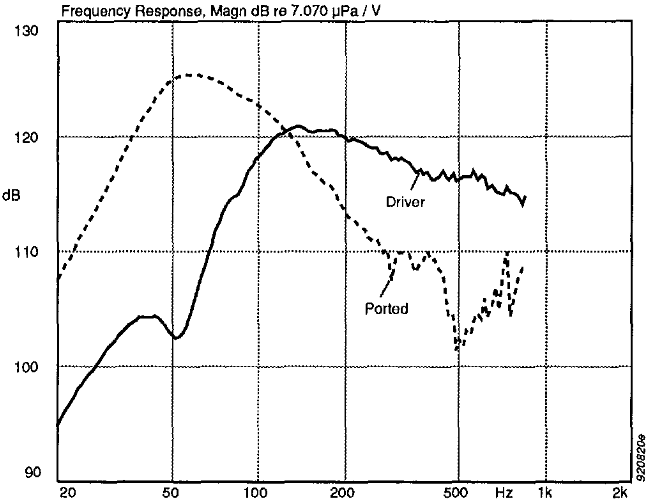

Fig. 21
Resulting Near Field response of ported system (S_D = 127.68 cm^2, S_p = 21.32 cm^2).

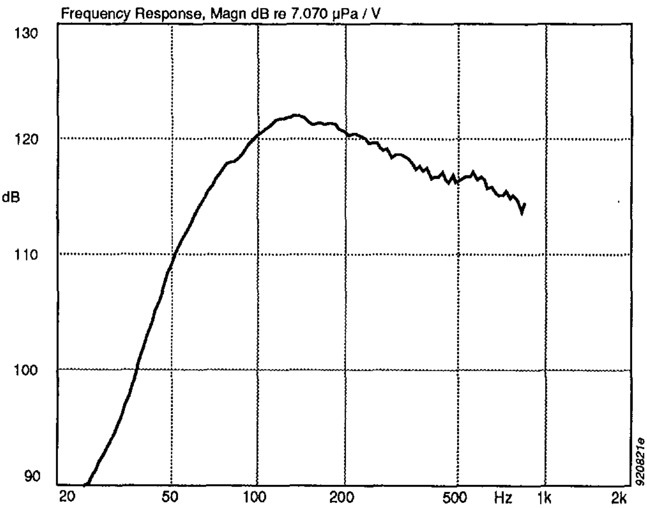

Fig. 22
Complete simulated Free Field Frequency Response of 2-way, 8 Ω, ported loudspeaker, ref. 20 μPa / 1W at 1m, measured on-axis, grille off, f_S = 224 Hz (JBL 4406 Studio Monitor).

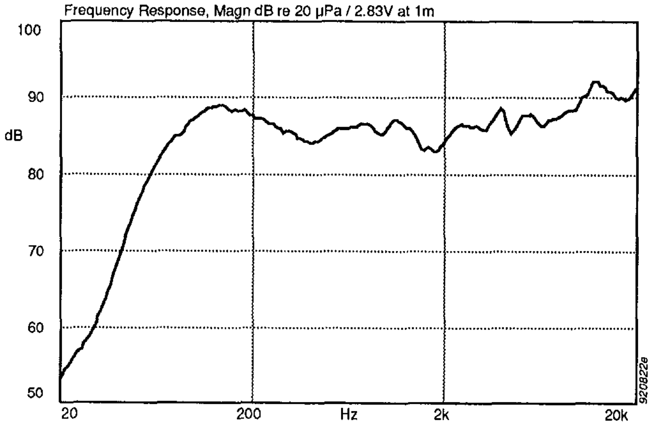

Fig. 23
Frequency Response of the ported loudspeaker measured using the simulated Free Field technique and measured in an anechoic chamber.

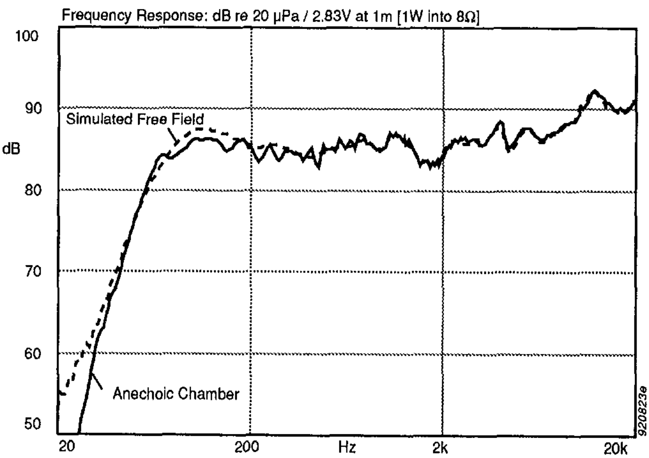

Fig. 24
Frequency Response of a loudspeaker measured in an anechoic chamber for various microphone distances. Ripple in the response is due to insufficient absorption of reflections at low frequencies.

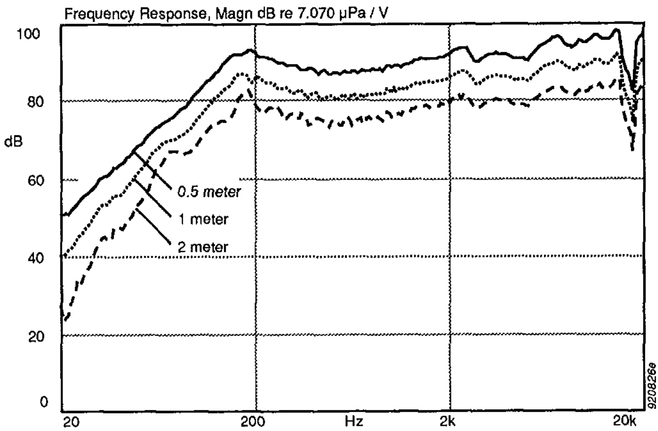

Fig. 25
The use of a linearly swept sinusoidal excitation signal enables time selective measurements of individual harmonic components to be performed. The diagram shows the separation of the 2nd harmonic in time and frequency .

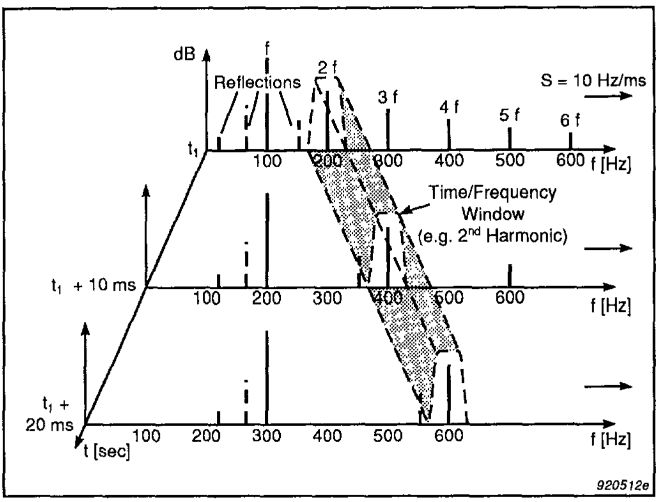

Fig. 26
Fundamental, 2nd, and 3rd harmonic responses of closed box loudspeaker for a 1W input, measured at 1m, using the simulated Free Field technique.

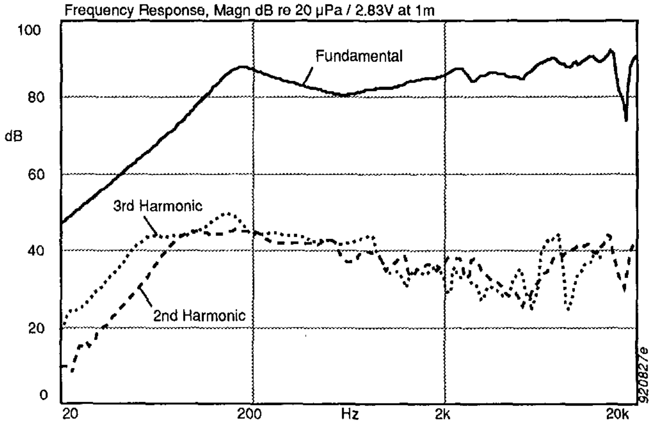

Fig. 27
Resulting Total Harmonic Distortion in dB re. Fundamental and in %.

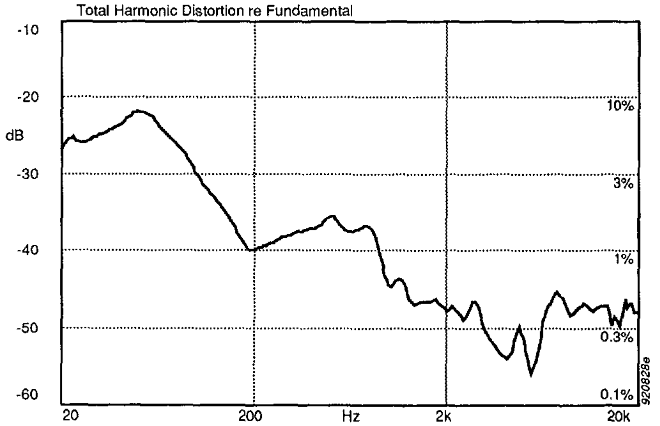

Fig. 28
Comparison between harmonic measurements performed using the simulated Free Field technique and measurements performed in an anechoic chamber (2nd & 3rd Harmonics only).