thinking sound

Frequency Response Measurements and the Meyer Sound HD-1 High Definition Audio Monitor

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Meyer Sound

I. Introduction

The Meyer Sound HD-1 High Definition Audio Monitor is a precision nearfield loudspeaker for music recording and mastering, audio post production, microphone evaluation, and psychoacoustics research.

To satisfy the rigorous requirements of this broad range of applications, the HD-1 is designed to exhibit ±1.5 dB amplitude response tolerance from 40 Hz to 20 kHz, and ±20° phase response tolerance from 200 Hz to 17 kHz, when measured at 1/100th octave frequency resolution with loading by a single boundary at a distance of 8 feet. Adjustable electronic circuitry for amplitude and phase correction is integrated into the system design, and each HD-1 is individually calibrated in an anechoic testing environment.

The criteria established for the HD-1 challenge the limits of contemporary acoustical measurement technique. While it is common to see loudspeaker frequency response plots that appear to resolve amplitude variations of a decibel or less, the resolution of most acoustical measurements is far more gross than such representations would indicate. In practice, the compounded effects of acoustical factors, instrumentation and measurement methodology routinely introduce amplitude errors that can exceed 3 dB and vary in magnitude with frequency.

In developing the HD-1, Meyer Sound embarked upon extensive research to formulate methods for obtaining acoustical measurements that are accurate to within 0.25 dB of free field. This paper describes the predominant factors that must be considered in any attempt to reproduce those measurements, and presents comprehensive HD-1 polar performance data in the form of frequency response plots.

It is our hope that this discussion will promote increased understanding of the variables that affect acoustical measurements, and perhaps serve as a starting point from which nearfield amplitude response measurement standards may be derived. *

DATA PRESENTATION & RESOLUTION

The frequency response data presented in this paper are taken from multiple FFT (Fast Fourier Transform) measurements made with 1/100th octave or greater frequency resolution.

Data displayed at this resolution are frequently subject to misinterpretation. As this paper will show, many high-Q features in such plots can be, in fact, measurement artifacts rather than actual facets of the test object's response. Correct interpretation of these artifacts depends upon a full knowledge of the test apparatus and conditions, coupled with an understanding of the effects of the many factors that enter into acoustic measurement.

Partially for this reason, it is often useful to present frequency response data at lower frequency resolution, as illustrated in Figure 1. Here, the top curve is a high-resolution plot of the response of an HD-1 measured at 1/2 meter on axis of the tweeter, with a single boundary surface 8 feet away from the cabinet.

The middle curve was generated from the same data by computationally simulating a rectangular 1/3rd octave filter swept over several thousand points. This presentation clarifies the contour of the response while retaining suitable detail, yet eliminates very fine amplitude ripple artifacts. It is for this reason that we have chosen this format for the polar response plots presented at the end of this paper.

Being derived from the original high-resolution data using a large number of points, the middle curve is quite accurate as long as acoustical errors are minimized in the original measurement. A swept-filter apparatus with random noise excitation would require a sweep time of many hours to achieve equivalent accuracy.

For comparison, the bottom curve illustrates the results obtained by computationally processing the original high-resolution data using parallel, fixed (RTA-style) 1/3rd octave filters at ISO standard center frequencies. The deficiencies of this presentation, when compared with the first two, are apparent.

* While our discussion here is centered on frequency response, a true reference monitor must possess more than flat amplitude response. Other key design criteria include linear phase response, minimal phase intercept distortion, good linearity (low harmonic and intermodulation distortion), consistent amplitude and phase response over a large listening window, sufficient dynamic range, high RMS and burst SPL, self protection, graceful overload characteristics, good power bandwidth and low-frequency accuracy. All of these performance criteria are optimized in the HD-1.

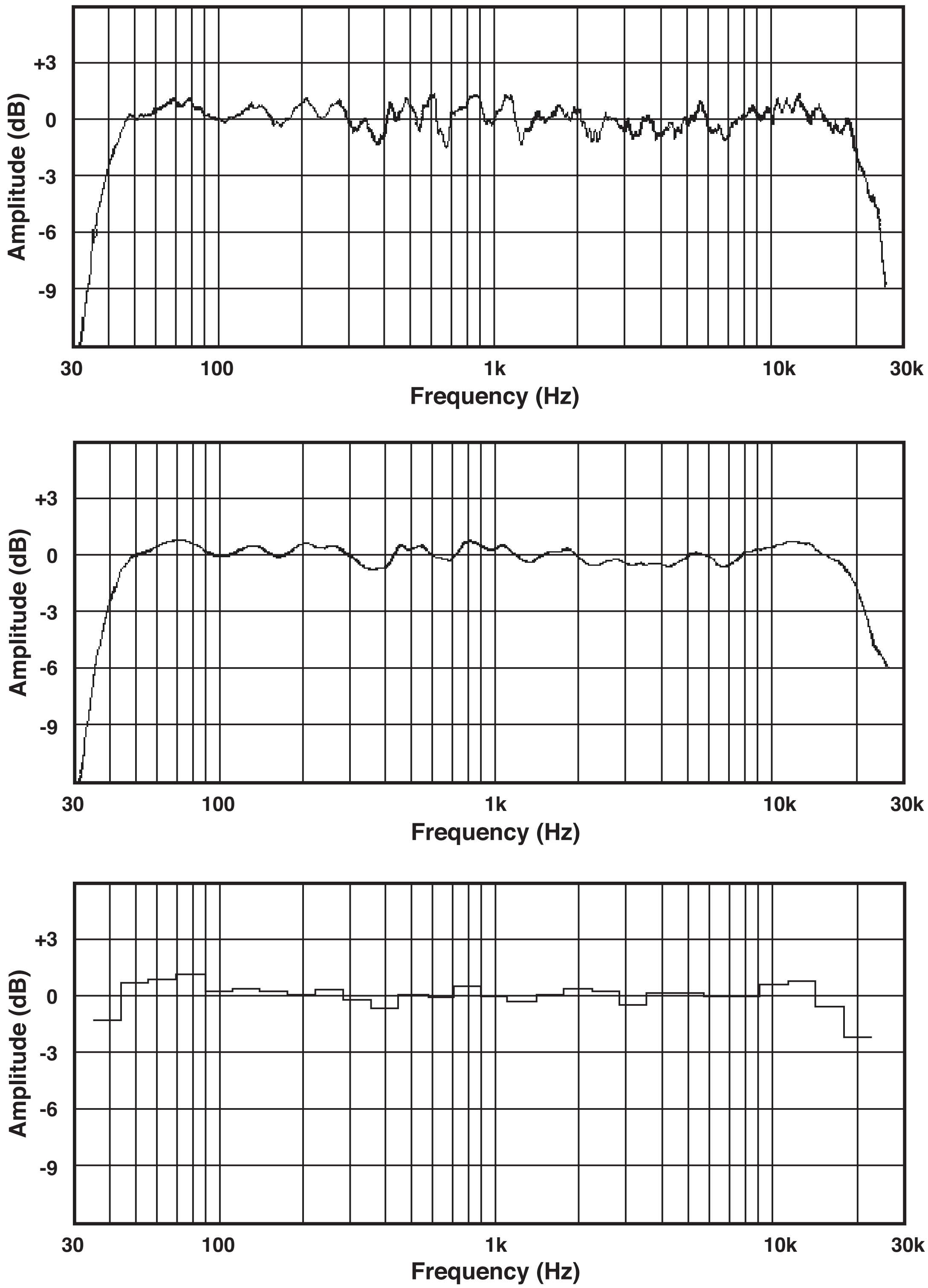


Figure 1 – HD-1 frequency response at 0.5 meter on tweeter axis with single boundary at 8 feet. 1/100th octave (top); continuous 1/3rd octave (middle); fixed 1/3rd octave (bottom).

II. Factors Affecting Accuracy of Acoustic Measurements

Loudspeaker measurements are complex, and are subject to many variables that can substantially affect accuracy. Among these are the acoustical characteristics of the measurement environment, the test method, and the instrumentation employed.

INSTRUMENTATION

Meyer Sound customarily employs FFT measurement, often coupled with customized instrumentation. The advantages of the FFT technique include good statistical noise removal, rapid update, and high resolution with the ability to evaluate some of the predominant acoustical error sources that can contaminate loudspeaker measurements. (The data presented in this paper were collected using an FFT analyzer and compiled in a computer.)

It would be a mistake to consider any single measurement method preeminent. Nonetheless, when performing any measurement, the limitations of the instrumentation must be recognized and evaluated.

For example, a test instrument may have both an absolute accuracy specification and a separate amplitude flatness specification. Most analyzers have an A/D front end to digitize the signal in order to process it, and this A/D converter will exhibit such limitations as finite precision and resolution, 12/16/24-bit internal processing, a specific analog noise floor, and limited linearity. These factors may combine to yield an absolute accuracy of 1% and amplitude resolution of 0.1% full scale.

If a precise 1.0 vpk 1 kHz sine wave were applied to it, such an analyzer would display the signal amplitude as anywhere from 1.01 vpk to 0.99 vpk, with harmonics approximately 60dB below that. With swept sine or noise source excitation, if the amplitude flatness specification were 0.25 dB, a perfectly flat input spectrum might be displayed with variations of 0.25 dB in addition to an offset attributable to absolute accuracy. Furthermore, such aspects of the analysis method as windowing or truncation can also have unique effects on the spectrum (if that is what is being measured).

In measurements of the HD-1, where ± 1 dB variations are being studied, the characteristics of the microphone become a factor as well. Meyer Sound uses the Brüel & Kjaer model 4133 microphone capsule because it has good overall characteristics for practical measurements. To verify its performance in manufacture, this microphone is measured with an electrostatic actuator, but its actual acoustic response varies from the actuator response — the most dominant variation being a high frequency rise of approximately 8 dB starting at 3 kHz (for the acoustic response compared to the actuator response).

Since such microphones cannot easily be measured acoustically, the actuator measurements employ a “designed-in correction” to yield flat acoustic response for 0 degree incidence; this correction classifies the 4133 as a “free-field”mic. Microphones without this correction, such as the Brüel & Kjaer model 4134, are classified as “pressure” microphones. The two types are designed to be used in different ways, and measurements performed without knowledge of these classifications can yield grossly misleading results.

Another, less dominant effect is a fine (±0.5 dB) amplitude ripple in the response of the model 4133, which occurs predominantly in the high frequencies and is a function of the microphone’s finite geometry. This effect is very difficult to measure and study, but by comparing the 4133 to a Brüel & Kjaer 4138 eighth-inch capsule, the error may be demonstrated. (The eighth-inch capsule makes a good reference because its geometry is small, and any errors that appear are effectively outside the audio spectrum.)

While it is difficult to identify all the mechanisms that cause these variations, they nonetheless appear in high resolution loudspeaker measurements, including the data presented at some points in this paper. These high Q ripples must not be interpreted as variations in the speaker: most of them are caused by noise, microphone variations, and other instrumentation errors.

ACOUSTICAL ERROR SOURCES

The three most important error sources in loudspeaker measurement are noise, reflections (including loading) and reflected acoustical impedance.

These error sources affect all loudspeaker measurements, regardless of the environment — be it a concert hall, an anechoic chamber, or an outdoor space. The degree of the error will vary in each case, and it is necessary to identify these errors to determine the validity of the data.

Some instruments claim the ability to remove these errors from any measurement. Such methods are based on many theoretical assumptions, and should be used with caution. It is generally better to minimize errors before taking any measurement, rather than attempt to remove them afterward through data processing techniques. For this reason, our reference measurements are performed in essentially free field conditions (usually suspended at least 25 feet in the air), significantly reducing reflection and loading error. A brief description of these three sources of error that occur in all acoustic measurements follows.

Noise

Noise is any extraneous, undesired sound that enters into a measurement. It might be random (like wind) or periodic (like that arising from motors or other cyclic mechanisms). Most measurement methods have little or no capability to remove noise — and of those that can, some remove only random noise.

Averaging is one of the most common methods for removing noise. There are many different ways to average a signal, including RMS (PWR), Peak, Linear, and Log Mag techniques. All of these leave some error term in the result, however, and this error must be evaluated carefully, since it may be significant when the noise is large compared to the source signal. The only mathematically correct way to statistically remove both periodic and random noise without new errors is Vector averaging.

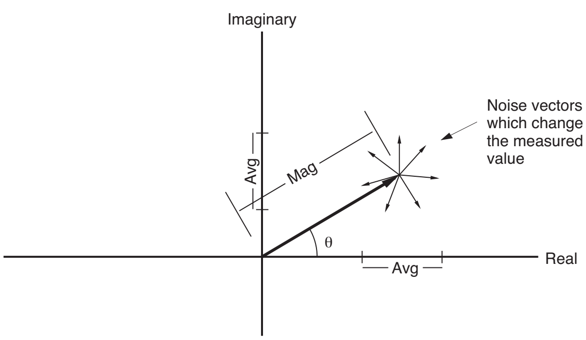


Figure 2 – Vector representation of test signal with noise

We often view audio signals as magnitude and phase, but they may also be converted to a vector containing real and imaginary components for each frequency. A noise-contaminated test signal thus may be visualized as the actual vector with the noise vectors surrounding it as shown in Figure 2.

In Vector averaging, the real and imaginary parts are linearly averaged simultaneously to yield the correct magnitude and phase for a given frequency. Since the noise vectors are uncorrelated to the source stimulus, over many averages they will statistically reach a value of zero, yielding the actual magnitude and phase value for that frequency. This is true for both random and periodic noise, since they are both uncorrelated to the source (if the noise is large with respect to the signal, many averages will be required but, theoretically, the correct value can be reached).

Vector averaging requires either a dual-channel instrument or a synchronized source stimulus so that complex data can be gathered multiple times in the frequency domain. It also demands stationary phase, which may be difficult to achieve at high frequencies if positional stability of the test object and/or microphone is poor. Except for some newer FFT instruments, most analyzers do not offer Vector averaging. *

Ultimately, it is preferable that noise be minimized and measured to quantify worst-case contamination. The use of an anechoic room with good transmission loss is the best way to eliminate errors produced by noise.

Reflections

Reflections may be caused by any object or boundary surface within the measurement site.

Reflections always occur later than the direct signal, so it might appear that they can be removed by gating or truncation. Unfortunately, this works only at high frequencies, because it takes a minimum of one period to properly characterize a linear system for a given frequency. In order to characterize a speaker at 100 Hz, then, the measurement must be sustained for at least 10 ms (1/100 Hz) — assuming that the system under test settles within the first period.

* Regardless of the averaging technique, the exact algorithm utilized by the analyzer is important, because it can produce its own errors. For example, the use of an accumulator, versus a true “n” size buffer, introduces errors that increase with greater numbers of averages. Unfortunately, information provided with the instrumentation about methods and algorithms is often insufficient.

Further low-frequency errors may be introduced by what is often referred to as "loading" (the increase of power at low frequencies by boundaries). Loading is also a reflection but, because of the size of the wavelengths involved, its effect is perceived as a phase-shifted signal combined with the direct signal rather than as an echo.

The difficulty in removing reflections post-measurement makes minimizing them before testing imperative. While it might be assumed that the use of an anechoic room accomplishes this, that is often not the case.

In most instances, such rooms are not "anechoic" (<10% pressure reflection) below 150 Hz. Pressure reflections at these low frequencies can be as great as 100%, significantly contaminating that part of the spectrum. Additionally, within the "anechoic" region, reflections can still reach 10% from a single surface, introducing about a ±1 dB error. Multiple surfaces and diffraction (scattering) compound the errors, which may reach several dB.

Anechoic rooms normally are qualified after being built, using an inverse distance law transversing method that will indicate the distance sensitivity. These data must be reviewed before undertaking a measurement, in order to evaluate any possible error that might skew the results.

Figure 3 illustrates the error (in dB) introduced by reflections in a typical large anechoic room across the audio spectrum. Similarly, Figure 4 presents deviations from inverse-distance law at low frequencies for various measurement distances. From these examples, we can see that the error from reflections in anechoic rooms can be significant, particularly at greater distances, at low frequencies, and when working close to the treated surfaces.

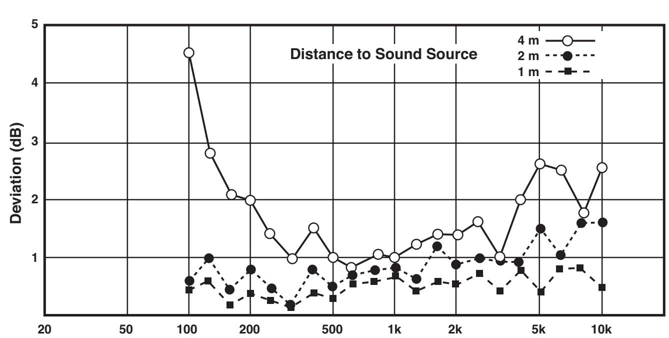


Figure 3 – Broadband deviation from inverse-distance law vs frequency for a typical large anechoic chamber

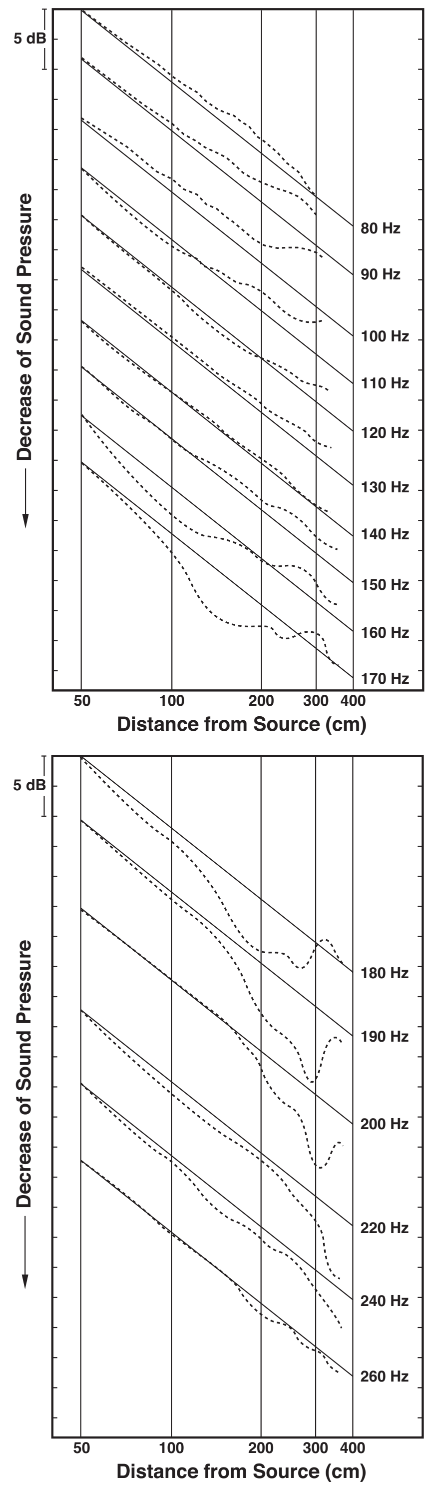


Figure 4 – Actual sound pressure decrease vs measurement distance (dotted lines) compared with inverse distance law (solid lines), 80 ~ 260 Hz

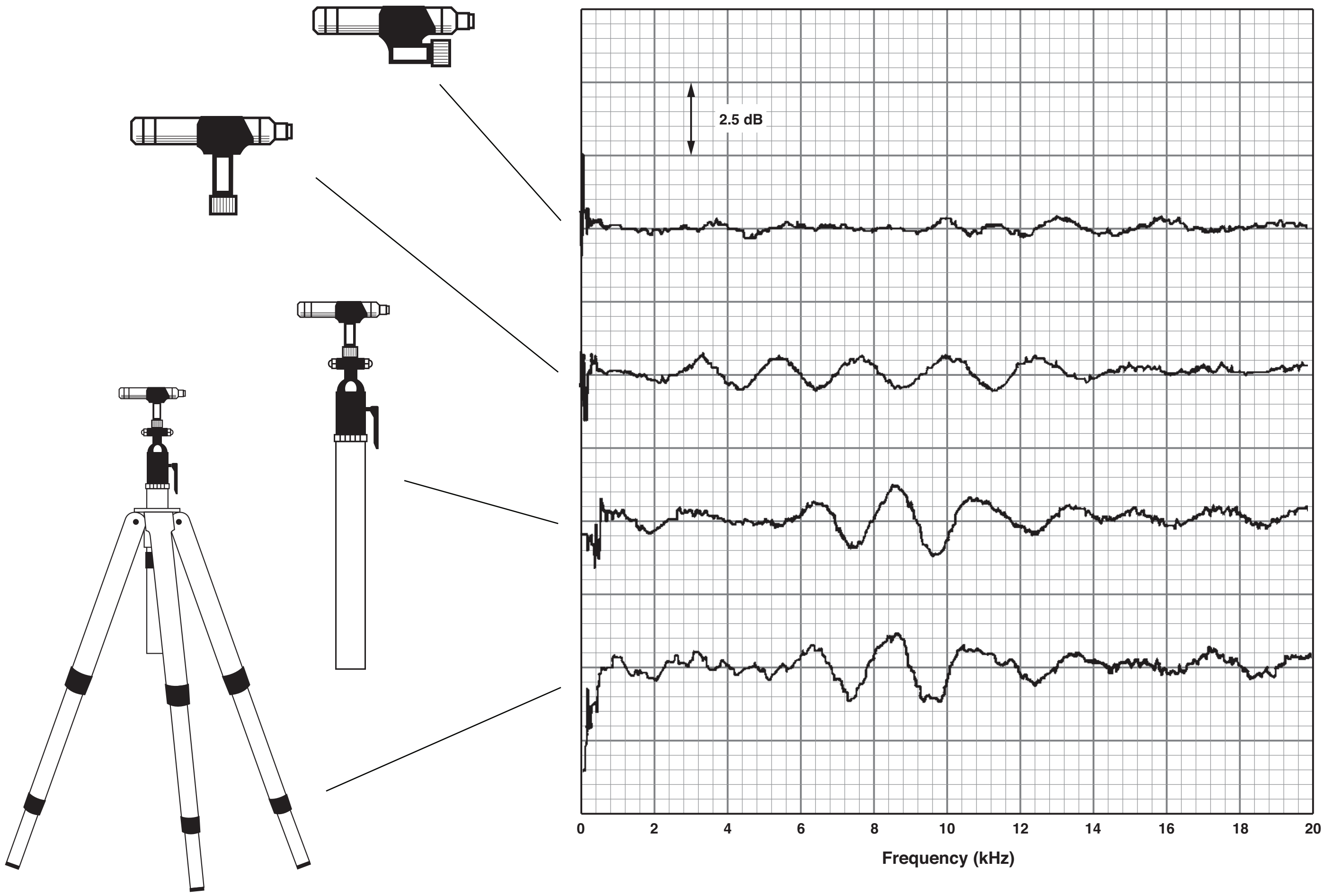


Figure 5 – Amplitude errors introduced by various test microphone mounts

Such errors could easily affect measurements of an HD-1, where 1 dB variations are critical.

Microphone mounts can also cause significant errors at high frequencies since they are, of necessity, quite close to the object of measurement. Examples of such errors are illustrated in Figure 5.

Note that even a simple mic clip introduces approximately 1 dB of error above 2 kHz when extended perpendicular to the mic housing. The situation worsens when a ball mount is attached. The very common practice of mounting a test microphone directly to a ball joint atop a tripod stand exacerbates the effects of the ball mount, and causes additional ripple (bottom curve).

These errors are extremely difficult to identify due to their early arrival time. They are best minimized by moving the test setup sufficiently far away from all objects that the reflections are attenuated significantly by distance alone.

Suspending the microphone by its cable, with a 25-foot distance from all reflective objects, results in a short frequency-domain ripple with propagation loss, reducing the errors from reflections to approximately 1/4 dB.

Impedance of the Environment

This error source is rarely discussed, but has significant consequences in loudspeaker measurements. Within several wavelengths' distance from the loudspeaker under test, the environment appears as an acoustic load to the acoustical source impedance of the speaker.

This acoustic load alters the frequency response of the speaker – just as, when a horn is coupled to a transducer, its impedance affects the transducer's response. In loudspeaker measurements, reflected impedance errors usually occur in the lower frequencies.

It is important to note that errors from boundary reflections and environmental impedance occur concurrently. The only practical way to avoid these errors is to move the loudspeaker far away from all physical boundaries.

MEASUREMENT METHODS

In addition to acoustical contaminations, measurement accuracy also is affected by all of the components in the instrumentation chain, such as microphones (including proper use of pressure and free-field types), repeatability of physical setups and, perhaps most importantly, the specific measurement technique employed.

The choice of the measurement method determines the sensitivity to contamination. Many different methods can yield the same results — if properly implemented. Unfortunately, proper implementation remains a substantial problem with the majority of commercially-available acoustic test tools.

Measurement systems such as TEF and MLSSA allow the user enormous freedom to control variables that can drastically affect the quality of the result (arbitrary or custom windows, variable tracking-filter width, user-selectable start/stop impulse truncation points, and other data manipulation options). As more variables are introduced into a measurement, the requirement for user sophistication rises, and it becomes increasingly difficult to verify that a valid result has been obtained.

One popular method to "remove reflections" from a non-anechoic or quasi-anechoic measurement is truncation of the impulse response (this technique is implemented in MLSSA through direct editing of the measured time response, and indirectly in TEF). But any integral transform (Fourier, Inverse Fourier, Hilbert, or other) that converts data from the time domain to the frequency domain must be continuous — or assumed continuous — and contain both real and imaginary parts.

To illustrate this point, let's examine the example of a Fourier transform of a low-pass filter, as depicted in Figure 6. (In this figure, and those that follow it, the script F refers to a Fourier transform, while F^{-1} designates an inverse Fourier transform.)

In the case of actual audio signals, the negative frequencies shown in the diagram are usually not displayed, but are assumed conjugate of the real part in order to yield a zero value imaginary part in the time domain.

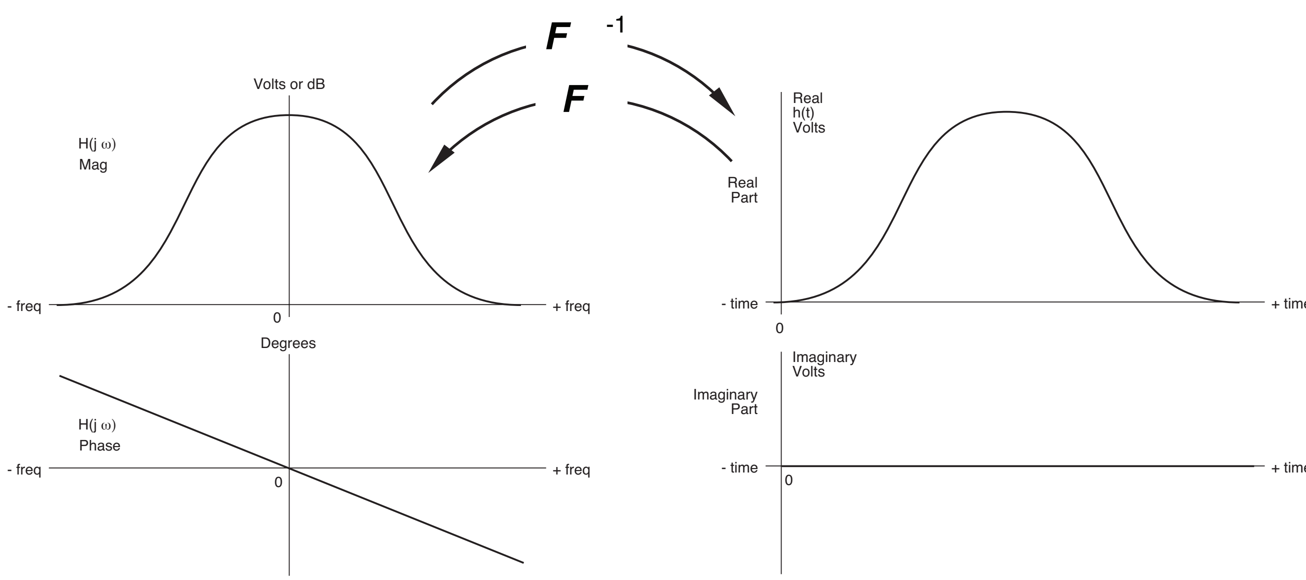


Figure 6 – Ideal linear phase low-pass filter

A discontinuity in either the time or frequency domain will produce errors that appear as ripples in the other domain. For example, the truncated frequency response depicted on the left in Figure 7, when transformed to the time domain, causes ripple in the time function as shown on the right side of the figure. The important observation here is that the discontinuities at the edges of the frequency response produce ripple throughout the time domain. The reverse is also true, as depicted in Figure 8.

In theory, the frequency response from DC (0 Hz) to infinity must be measured if the "impulse" (time) response of a speaker is to be derived. Otherwise, if we stop somewhere and arbitrarily set the remaining frequencies' magnitude to zero, the resulting discontinuity will produce errors throughout the time response.

Similarly, valid data at low frequencies are often difficult to obtain, so the data cutoff might occur, for example, at

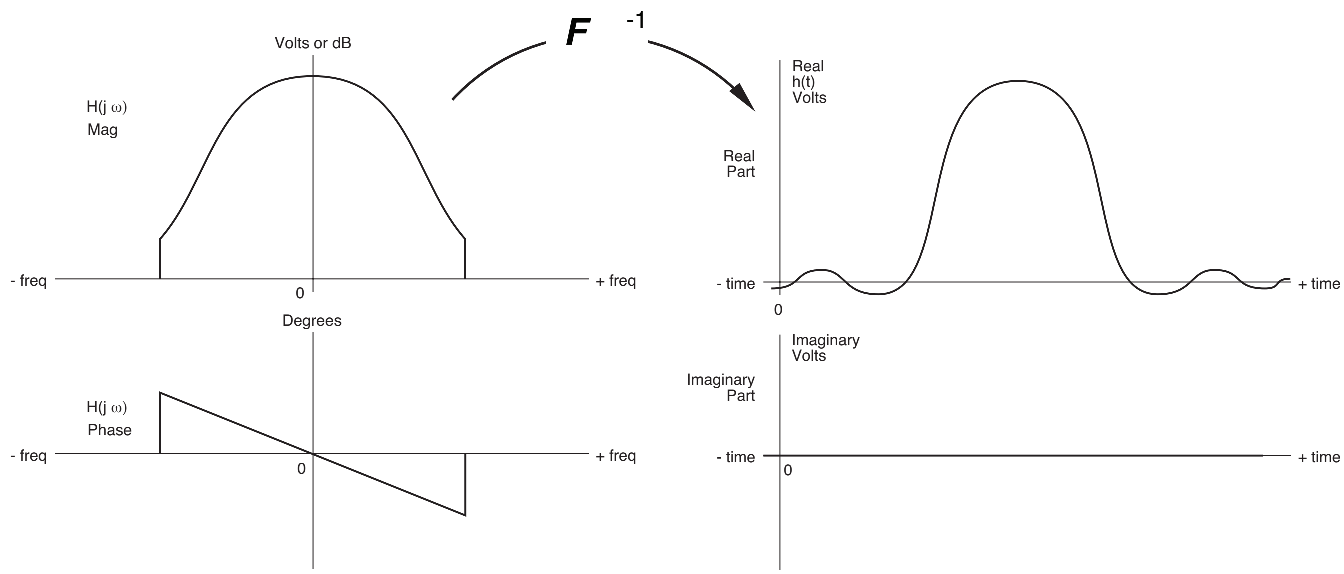


Figure 7 – Frequency-domain truncations cause ripple in the time domain

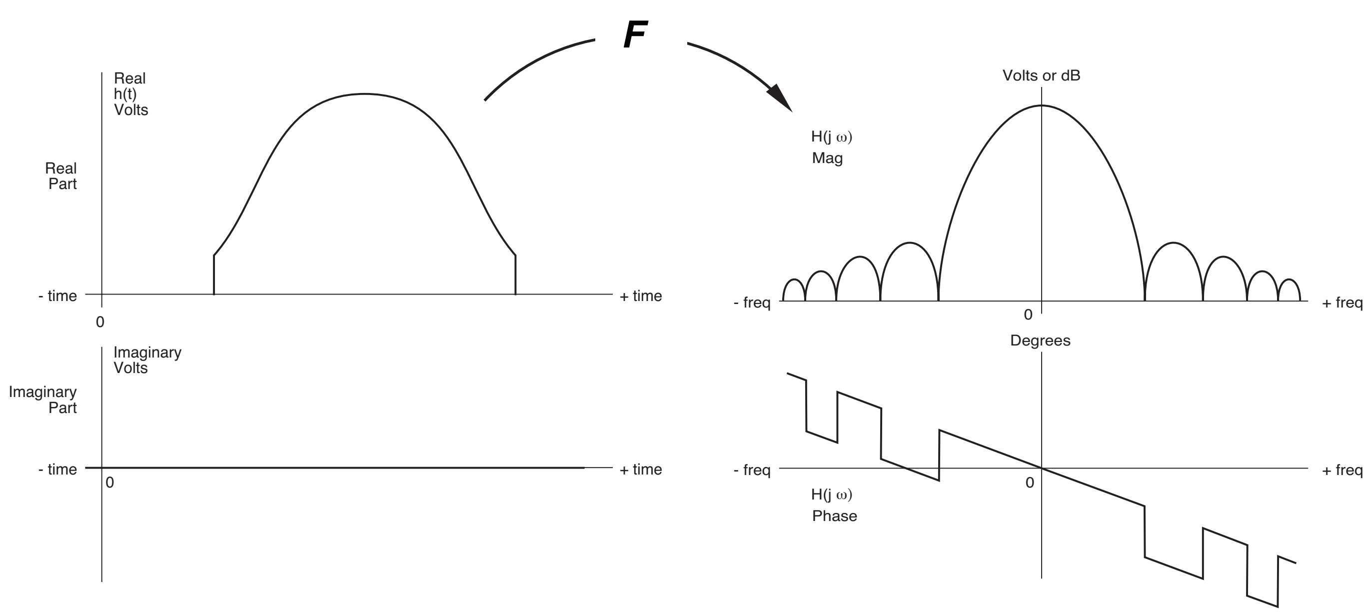


Figure 8 – Time-domain truncations cause disturbances in the frequency domain

30 Hz. If we merely set the lower frequencies to zero, as shown in Figure 9, the results would be virtually meaningless.

How, then, can we tell what part of the impulse response is the speaker and what part is the ripple from "truncation" (discontinuity)? This is a problem for all analyzers that perform integral transforms. *

It has become very popular to attempt to obtain the free-field response of a speaker in an ordinary reflective room by truncating anything that appears in the time domain to be an echo. This practice carries the potential for substantial error.

As an example, if reflective objects are placed within 5 feet of the speaker being measured, the first reflections will occur approximately 9 ms later. Were the measurement collected in the time domain (or transformed to the time domain), it might appear as in Figure 10a.

If the signal is truncated just before 9 ms, problems occur:

1) The sharp discontinuity will cause amplitude ripples in the frequency response. Even if a zero-value point on the time record is chosen, or a window is applied, the frequency and phase response will still be affected.

2) The frequency response shown below 111 Hz (1/9ms) is invalid (see Figure 8).

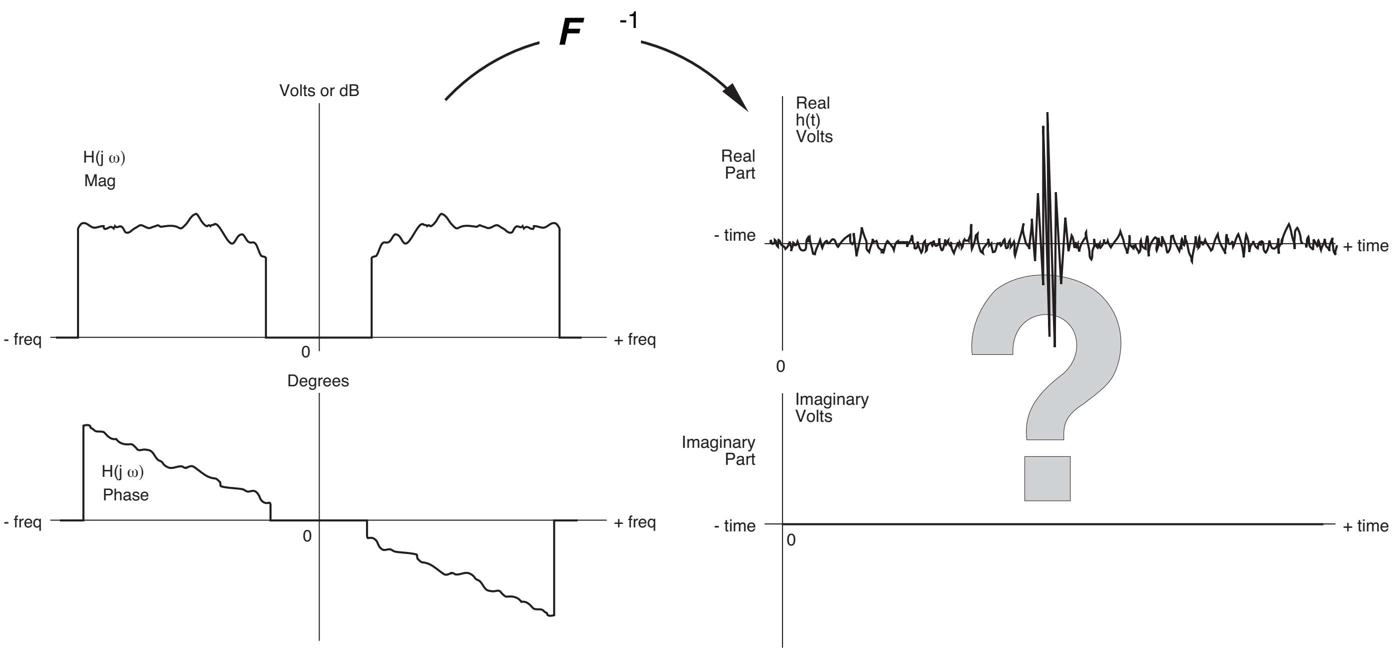


Figure 9 – Arbitrarily setting frequencies below 30 Hz to zero amplitude in frequency domain produces meaningless time-domain transform

* One mathematical solution to this dilemma is the use of windows: rather than truncating the response abruptly, the edges are gradually attenuated to zero in an attempt to minimize undesired effects. But this creates additional characteristics in the other domain. It is beyond the scope of this report to discuss all the effects of windowing, but it should be mentioned that the use and implications of windowing are often misunderstood. Some of the older FFT analyzers offer a limited number of windows, the effects of which are well understood (Rect, Hanning, Hamming). Unfortunately, many of the newer analyzers allow arbitrary, custom windows, further complicating evaluation of measurement accuracy.

Moreover, what appear to be echoes in the time domain may actually be part of the natural response of the speaker. Removing that portion of the speaker's impulse response can alter the computed frequency response to make the loudspeaker appear to be "better" than it really is, as depicted in Figure 11.

While such a result may soothe the loudspeaker designer's ego, and probably would enhance the impressiveness of a published data sheet or technical paper, from an engineering standpoint its validity is nil.

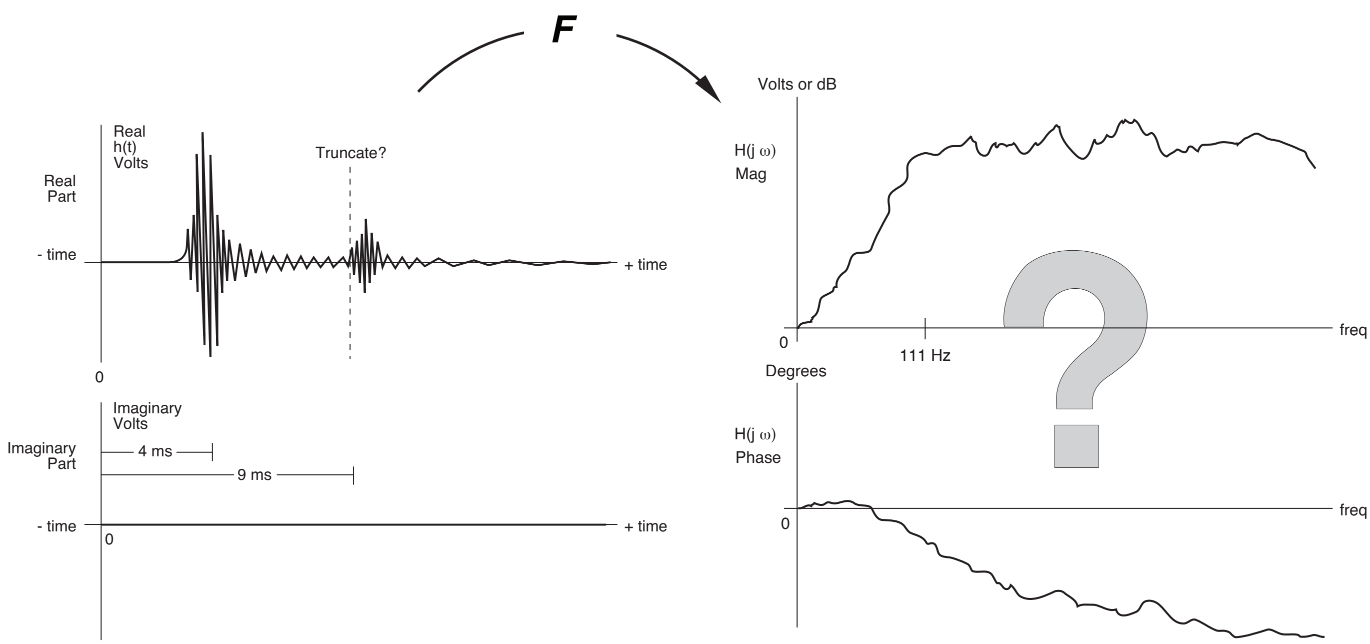


Figure 10a – Apparent echo in time domain

Figure 10b – Truncation affects frequency and phase response

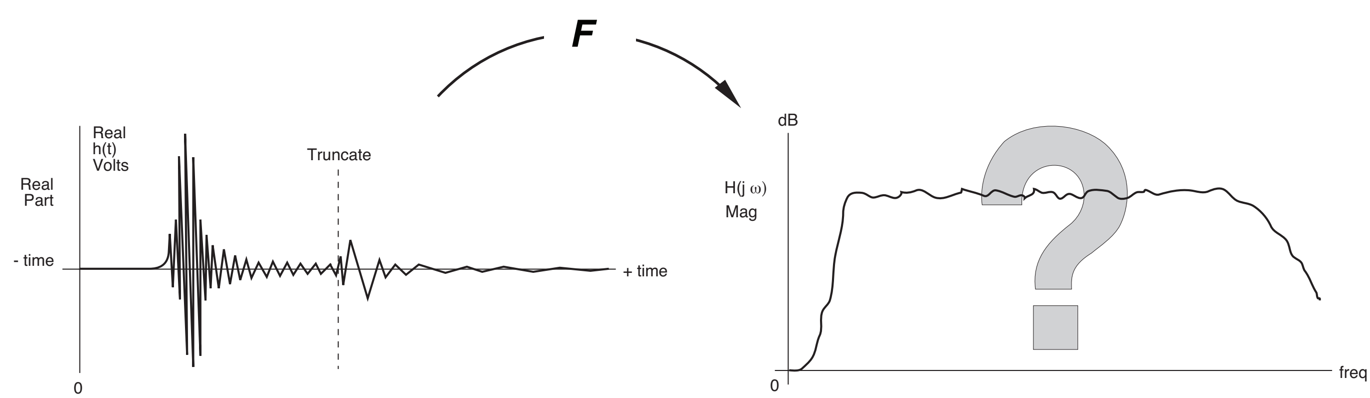


Figure 11 – Impulse response truncation can make a speaker appear "better" than it is

III. HD-1 Polar Response Measurement Data

PROPAGATION CHARACTERISTICS OF THE HD-1

In the development of the HD-1, a proprietary alignment scheme was devised to preserve its magnitude and phase response over a large vertical and horizontal listening window. This scheme is electronic and transducer-dependent, and is patented by Meyer Sound.

A unique property of this proprietary alignment is that the HD-1 listening window is justified upward. This is quite suitable for a reference monitor, because both full-bandwidth (on axis) measurements and recording studio monitoring often occur in the upper portion of the vertical polar pattern. In recording applications, an additional immediate benefit is that mid-band reflections from the console surface are minimized.

Within the listening window shown in Figure 12, the HD-1's amplitude and phase response is extremely uniform, and the unit acts as a virtual point source — that is, it appears as far-field (measured SPL output drops 6 dB per doubling of distance) throughout its bandwidth.

This can be observed by comparing the frequency response at 1/2 meter to that at 1 meter, as shown in Figure 13. The response at 1 meter drops 6 dB below that at 1/2 meter throughout the unit's bandwidth, following an ideal 1/R loss as would a spherical wave front. This effect does not occur with conventional loudspeaker designs, in which the low-frequency response in the near-field may only drop 3 dB or less with doubled distance.

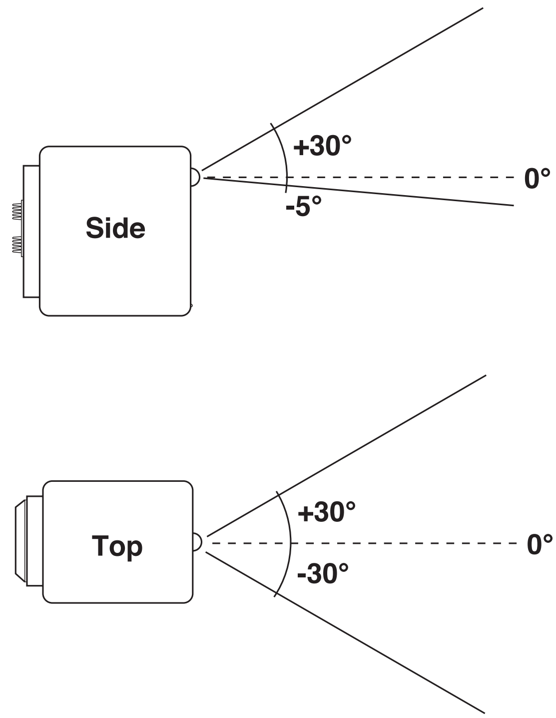


Figure 12 – HD-1 optimum listening window

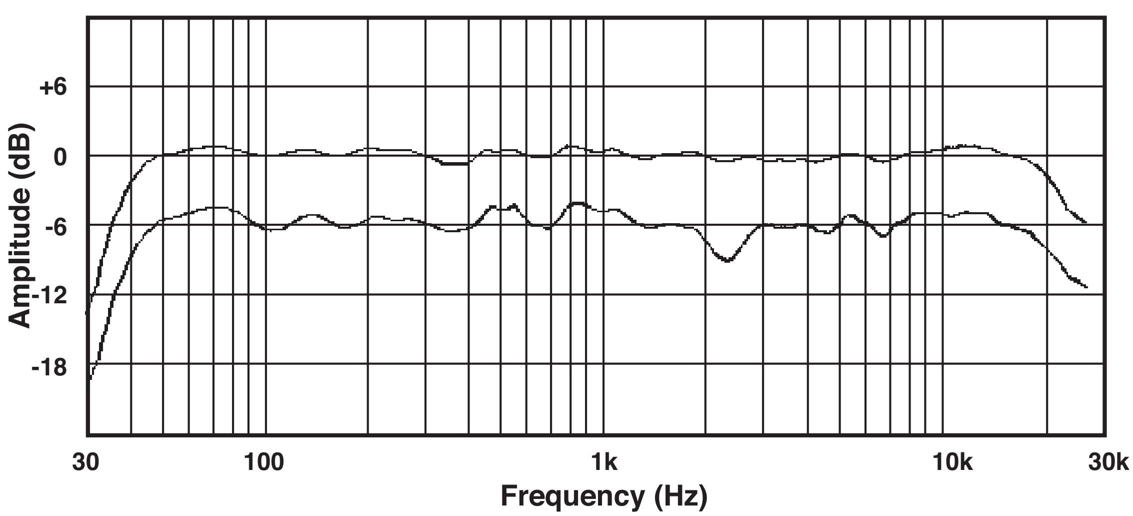


Figure 13 – HD-1 measured at 0.5 meter (upper curve) and 1 meter (lower curve) on tweeter axis, both with loading by a single boundary at 8 feet from cabinet, continuous 1/3rd octave frequency resolution

ALIGNMENT REFERENCE DISTANCE

The HD-1's frequency response is specified and aligned at 1/2 meter on tweeter axis. The reason behind this 1/2 meter measurement reference point is that one of the largest sources of error in a real measurement is acoustic reflections.

The magnitude of the error is inversely proportional to the ratio between the direct and reflected energy. The farther the test microphone is moved from the source, the larger the error. In fact, the error increases exponentially with distance, so when the measurement point moves from 1/2 meter to 1 meter, the error more than doubles. With reflections being a concern in all environments, including anechoic chambers, 1 meter measurements are far more susceptible to environment-related error than those made at 1/2 meter.

The 1/2 meter measurement reference works only because the HD-1 is a good point-source radiator: from other loudspeakers, it could yield meaningless results. It is understood that practical near-field listening is generally done at distances greater than 1/2 meter, and that is why 1 meter response plots have been included in this document. But, as we have shown, the HD-1's performance tracks closely from 1/2 to 1 meter.

POLAR RESPONSE MEASUREMENTS

In contrast to conventional polar coordinate plots, we have chosen to present full bandwidth frequency response data at representative measurement positions. This method reveals much more information about the HD-1's true polar performance.

Data Acquisition and Presentation

The polar measurements presented here were taken utilizing FFT instrumentation. All data taken are stored on disk, and subsequently downloaded to a computer. Various math programs operating on the stored data assist in analysis and identification of error sources.

From the resulting data, continuous 1/3rd octave plots have been made on a log frequency axis with an approximate constant percentage (constant Q) frequency resolution and 6 dB/division amplitude resolution. These are presented in a single group at the end of this paper, and will be referred to in the discussion that follows.

Data Analysis

A feature that may be observed in all of the response plots is a degree of amplitude "ripple" caused by reflections from surfaces surrounding the apparatus.

The amplitude plot of Figure 14 shows a 1/100th octave measurement made at 25 feet above the ground outdoors (virtual free-field, the nearest object being the ground 25 feet below the speaker). This plot exhibits a small distinctive amplitude ripple between 20 ~ 400 Hz, which we have made more visible by scaling the vertical resolution to 3 dB/division. The residual error is ≈ ±0.25 dB, which is both negligible and definable. Its period is ≈ 20 Hz (the amplitude peaks occur at 20 Hz intervals), which equates to a reflection path 50 feet in length, or double the distance to the ground.

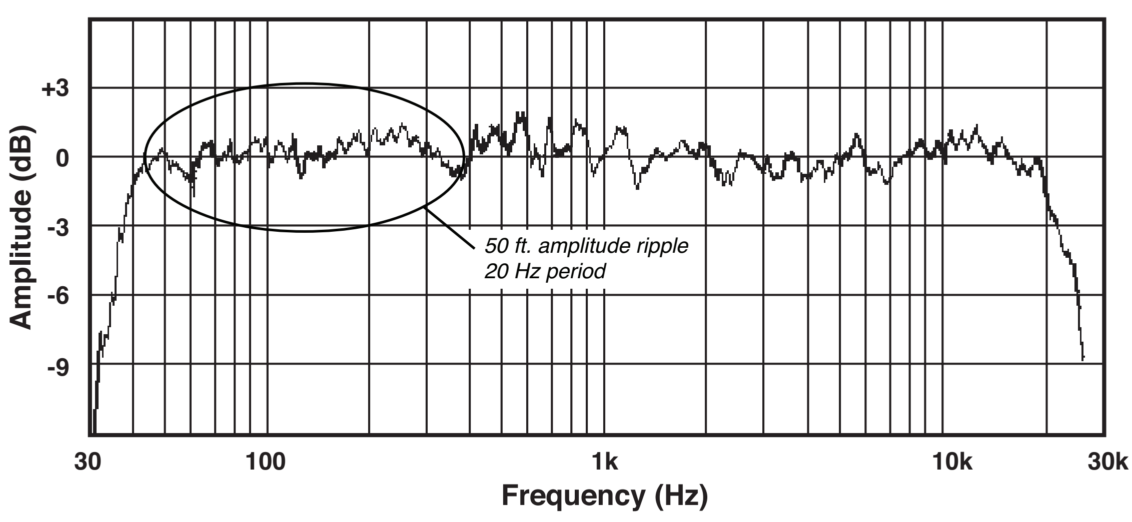


Figure 14 – Magnitude response of HD-1 measured 25 feet above ground (virtual free space), 1/2 meter on tweeter axis, 1/100th octave frequency resolution

Since the data are plotted on a log frequency axis with constant Q (%BW) resolution, the 20 Hz ripple period appears narrower as frequency increases, and eventually is so closely spaced that it is indistinguishable at 600 Hz and above. (The reflection also is attenuated at mid and high frequencies because the reflective object, the ground, is behind the speaker.)

Because the logistics of performing measurements at 25 feet are restrictive, the succeeding polar measurements were taken at 8 feet above ground, with the microphone suspended by its cable, on axis of the HD-1 tweeter.

At 8 feet there is a greater effect from ground reflection, amounting to almost ±1 dB amplitude ripple between 200 ~ 400 Hz. The effect is shown in the upper plot of Figure 15, which is plotted at 1/100th octave resolution. This ripple is also detectable in a continuous 1/3rd octave 6 dB/division plot at the bottom of Figure 15.

The ripple has a period of about 70 Hz. With the 25-foot reference measurement available, it is much easier to visually normalize the amplitude ripple from the 8-foot plots.

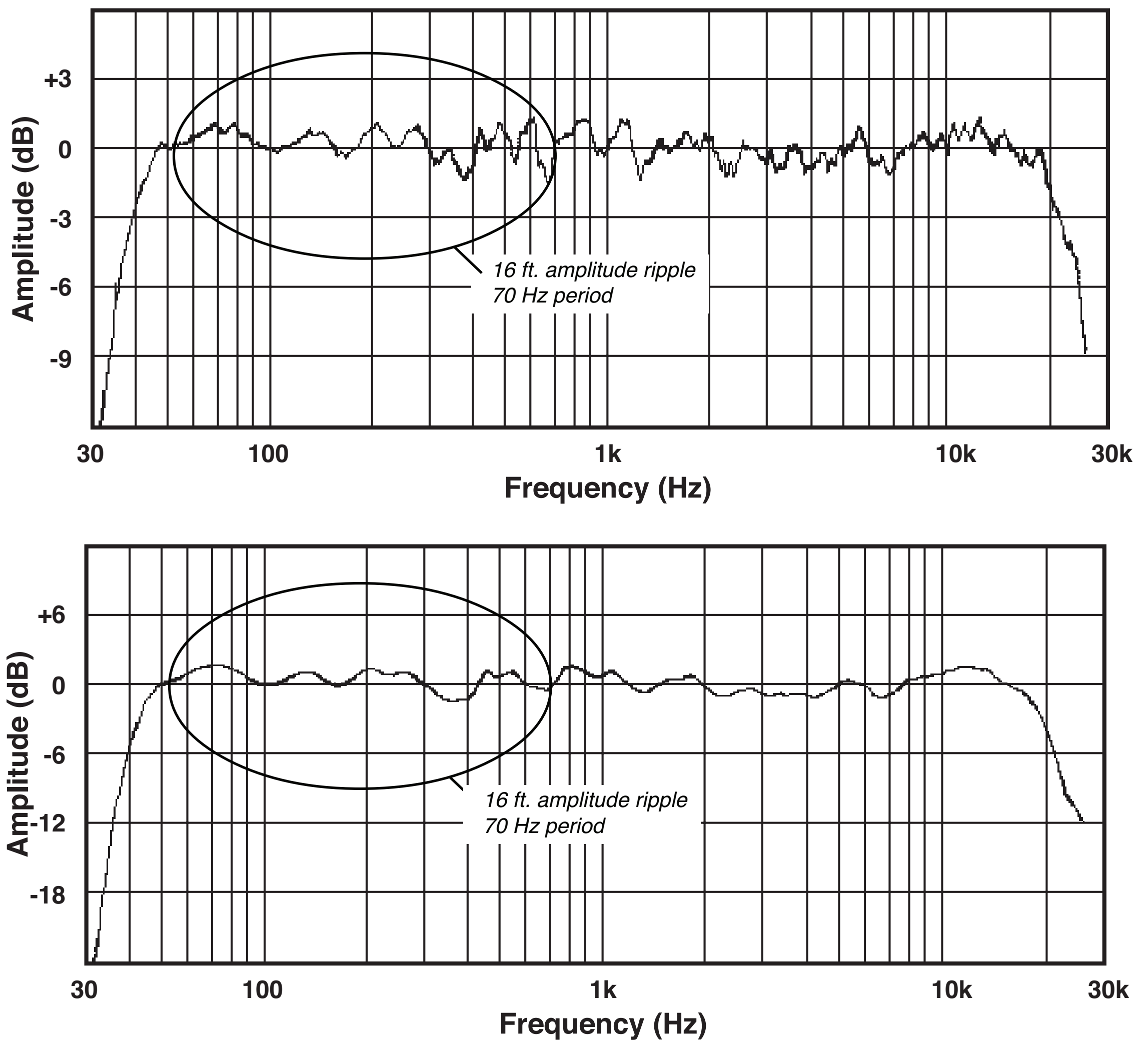


Figure 15 – Magnitude response of HD-1 at 8 feet above ground, 1/2 meter on tweeter axis, at 1/100th octave frequency resolution, 3 dB/div (top); and continuous 1/3rd octave frequency resolution, 6 dB/div (bottom)

When the microphone is moved to 1 meter on axis, an attenuation tendency appears in the 1 ~ 3 kHz region (see the upper curve in Figure 16). This anomaly is caused by both the complex diffraction effect from the transducers to all four edges of the baffle, and the relative phase of the two transducers as the mic is moved. If the mic is moved even farther than 1 meter, the response takes on a new variation in the 1 ~ 3 kHz region — again, because of diffraction.

It is important to note that this effect changes with different microphone positions (see lower curve in Figure 16).

If the initial tendency at 1 meter were to be corrected with peaking equalization, it would cause large response peaks at other positions in the polar response. Certainly, in any real listening situation, the engineer or listeners (and there may be several) will need to move about, so overall uniformity is critical.

Many more observations can be made from the polar response plots that follow. Taken overall, these plots indicate what we believe to be reference quality performance. The reader should be cautious in interpretation, and consider what has been described in this paper.

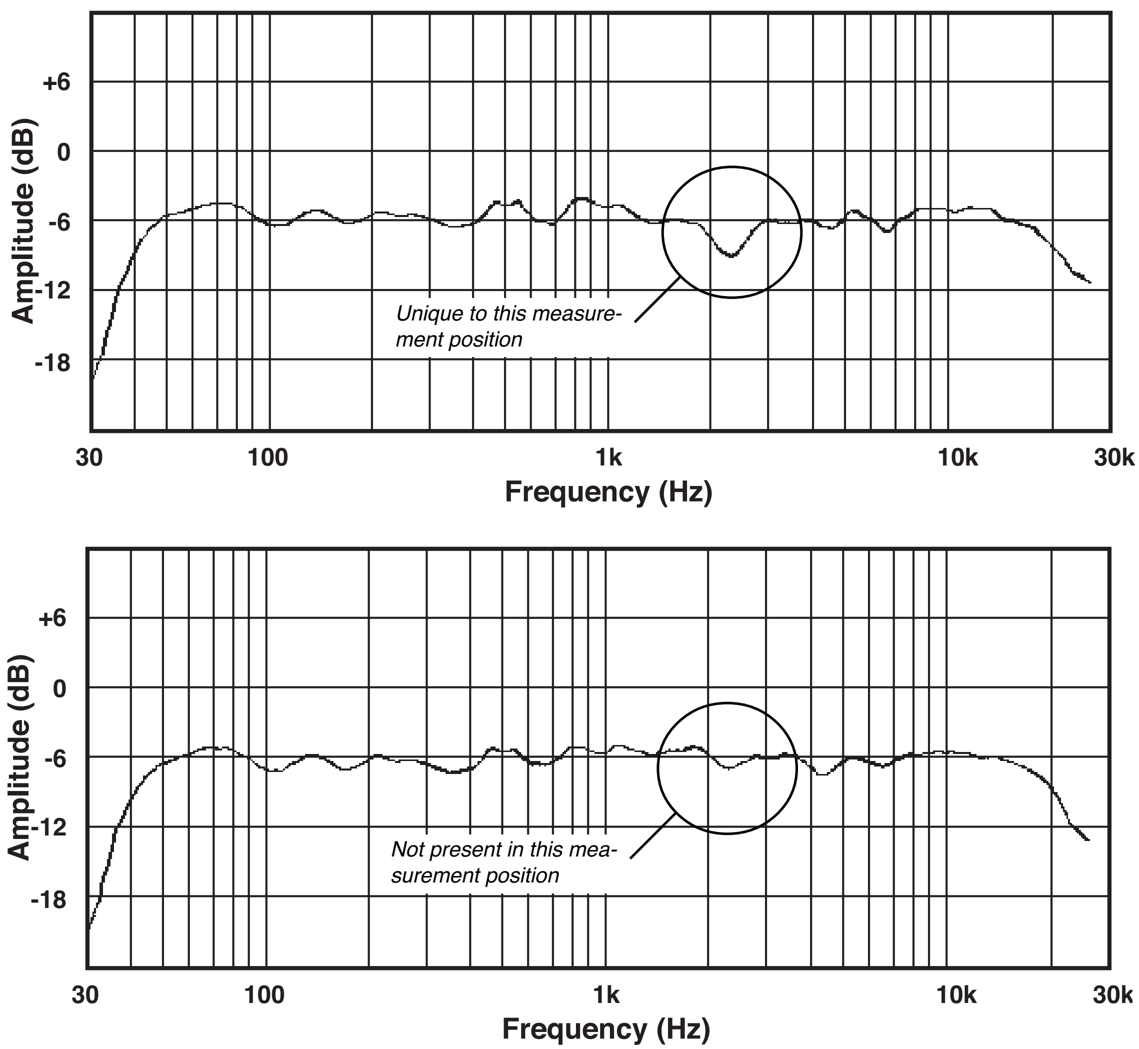
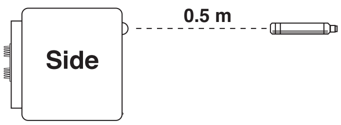
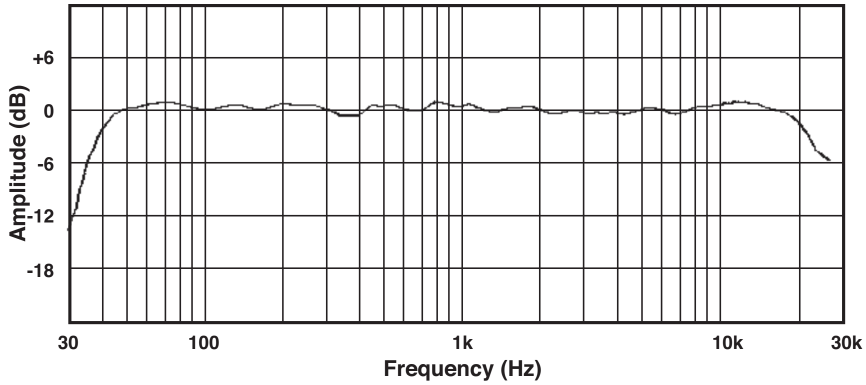
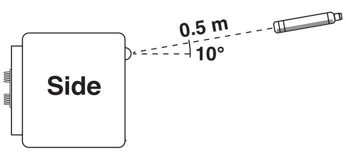


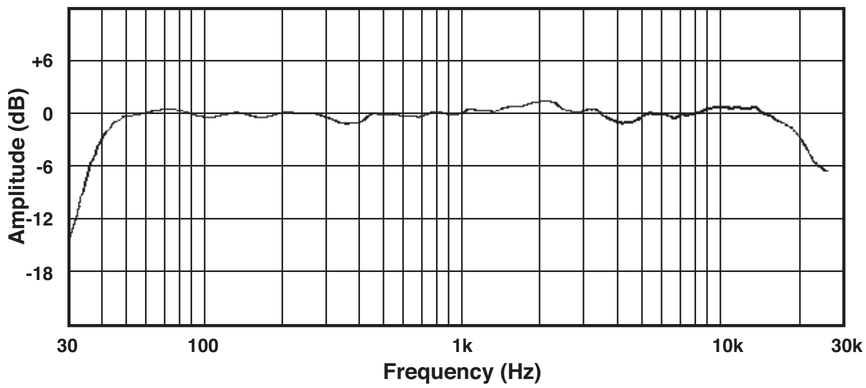
Figure 16 – Measurements of the HD-1 at 1 meter; on tweeter axis (top), and 10° off axis vertically (bottom).

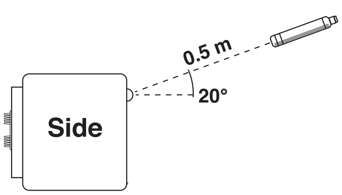
Meyer Sound HD-1
0.5 Meter Polar Response

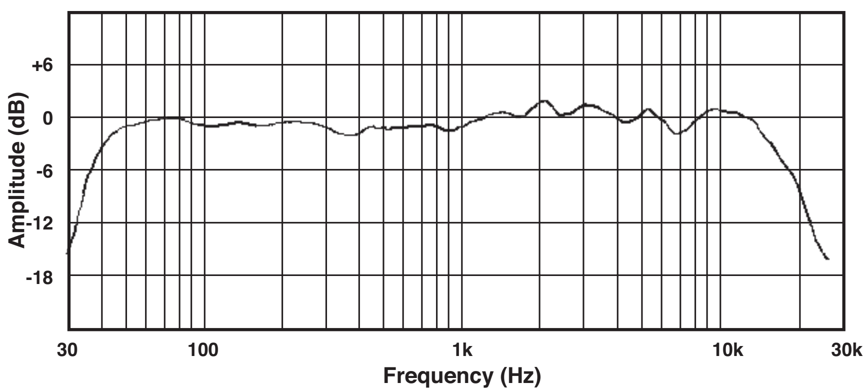


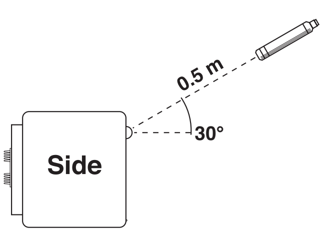


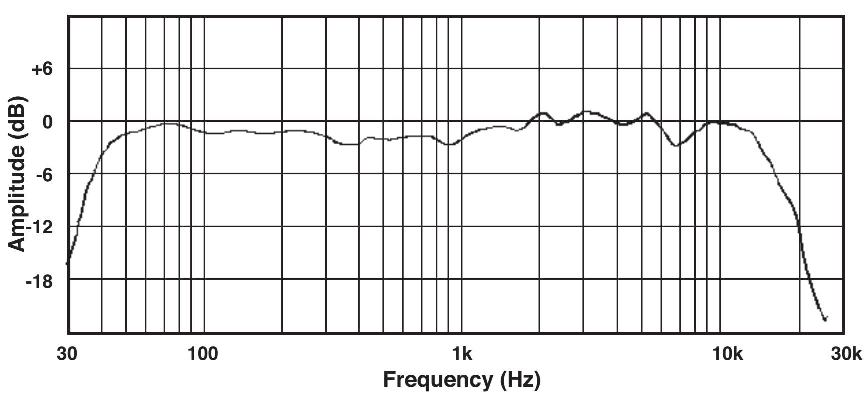


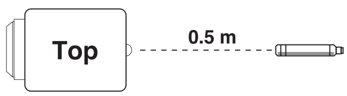


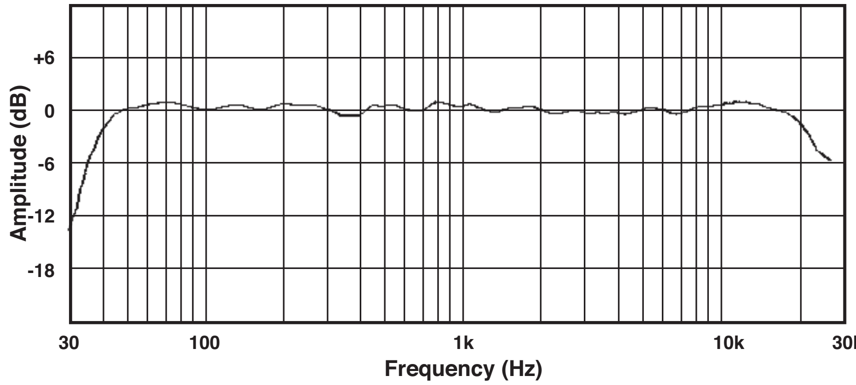


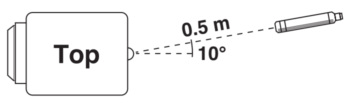


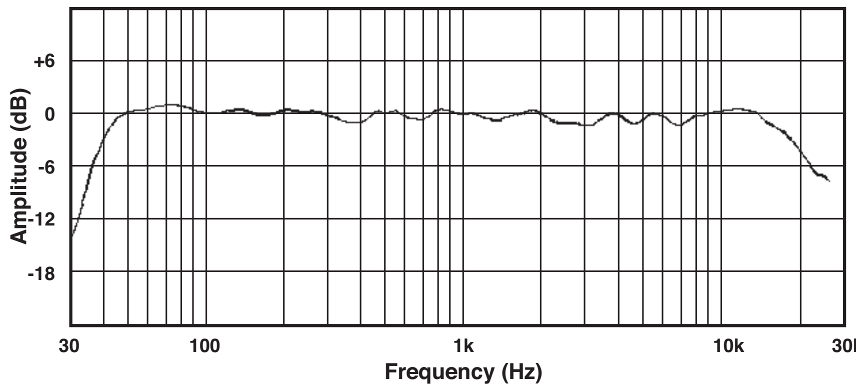


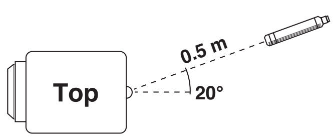


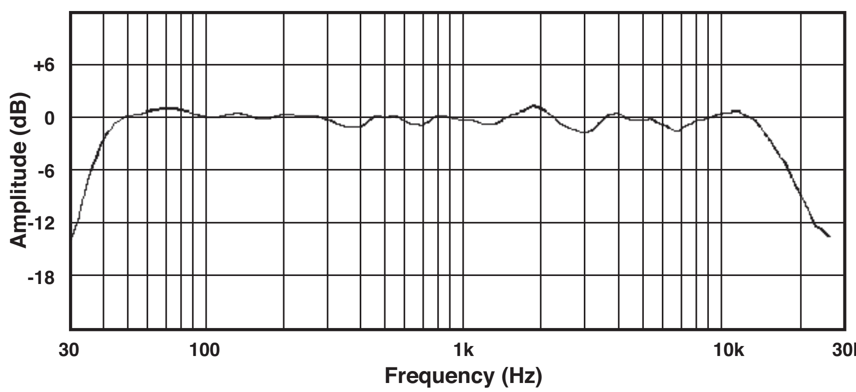

Top
0.5 m

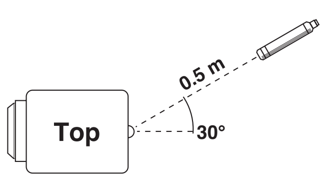

+6
0
-6
-12
-18
Amplitude (dB)
30
100
1k
10k
30k
Frequency (Hz)

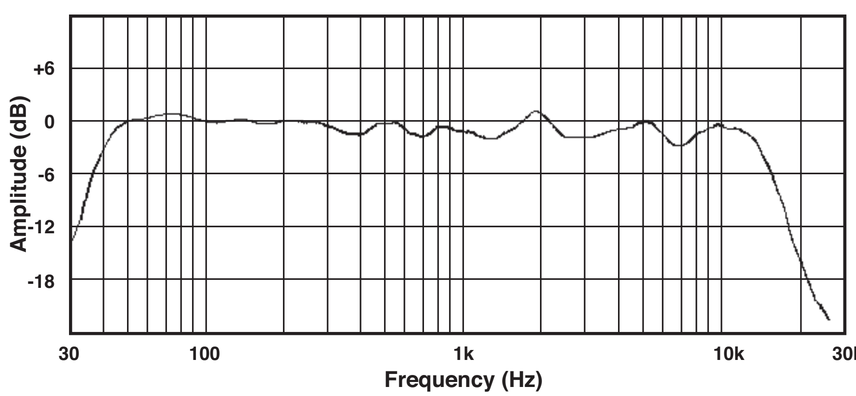

Top
0.5 m
10°


+6
0
-6
-12
-18
Amplitude (dB)
30
100
1k
10k
30k
Frequency (Hz)


Top
0.5 m
20°


+6
0
-6
-12
-18
Amplitude (dB)
30
100
1k
10k
30k
Frequency (Hz)


Top
0.5 m
30°


+6
0
-6
-12
-18
Amplitude (dB)
30
100
1k
10k
30k
Frequency (Hz)

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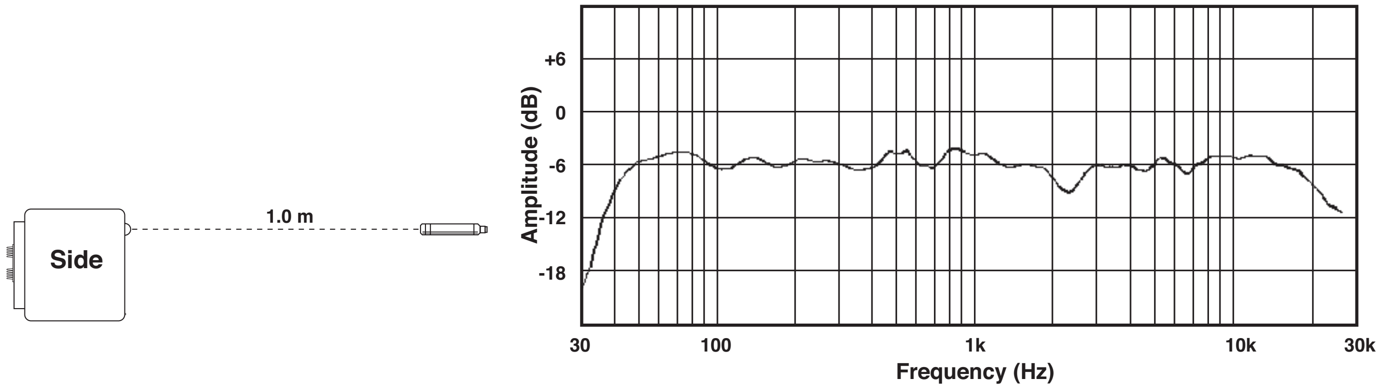
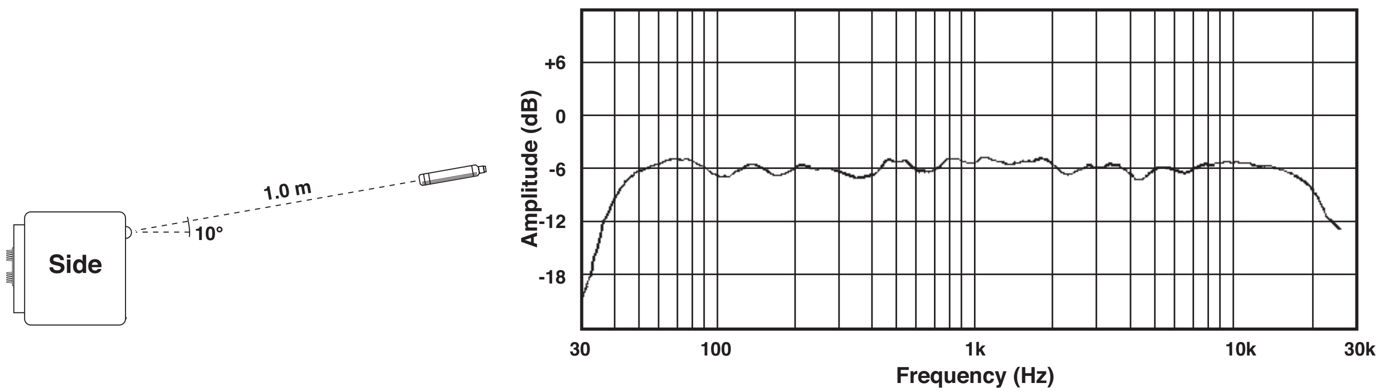
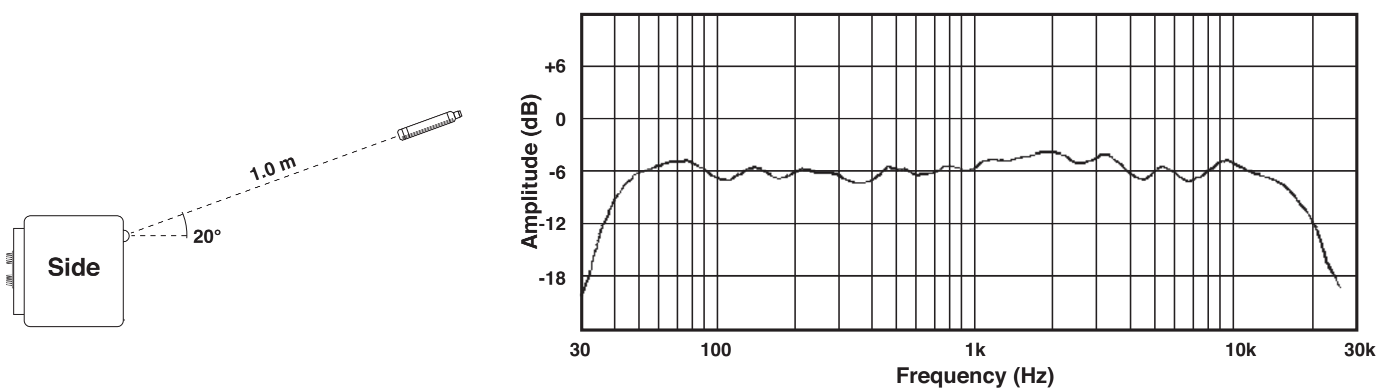
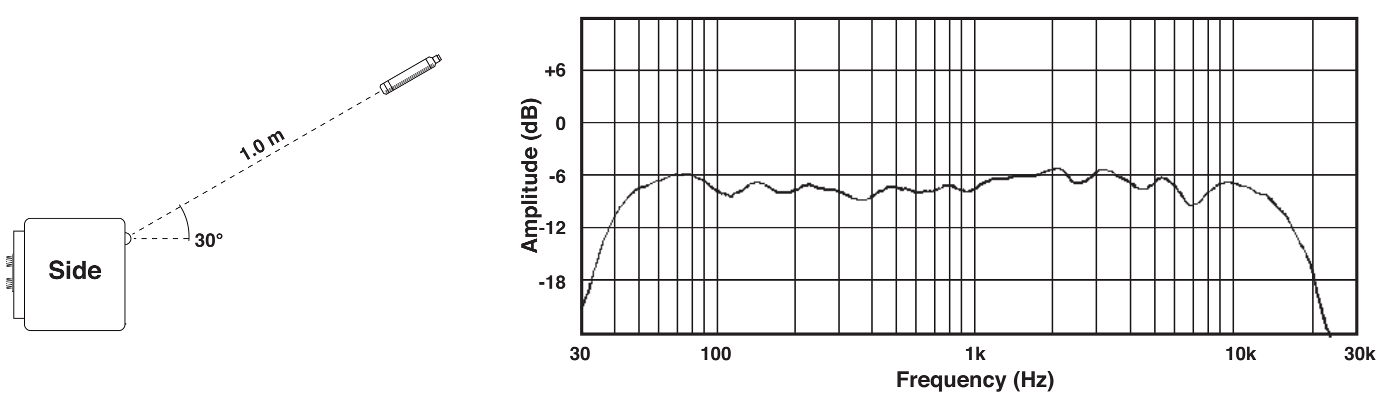
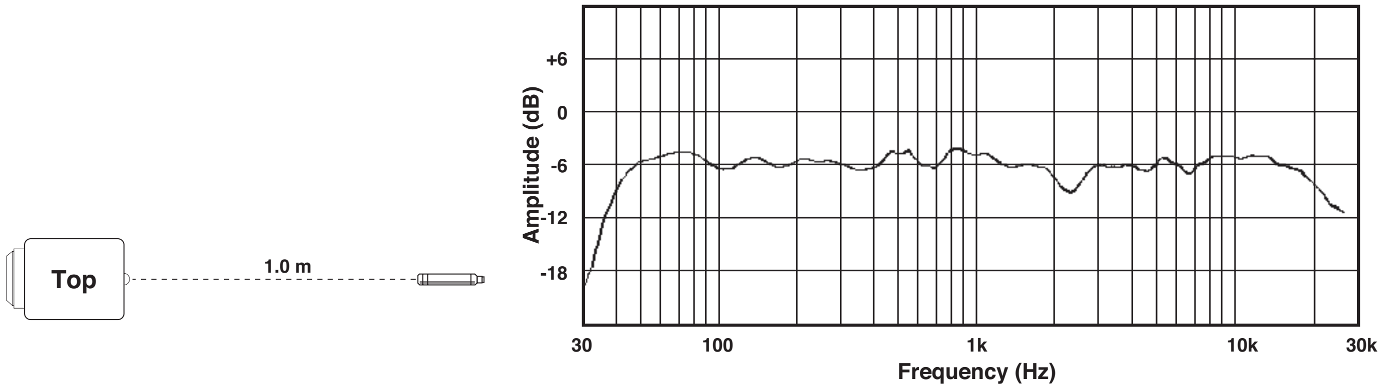
Meyer Sound HD-1
1 Meter Polar Response

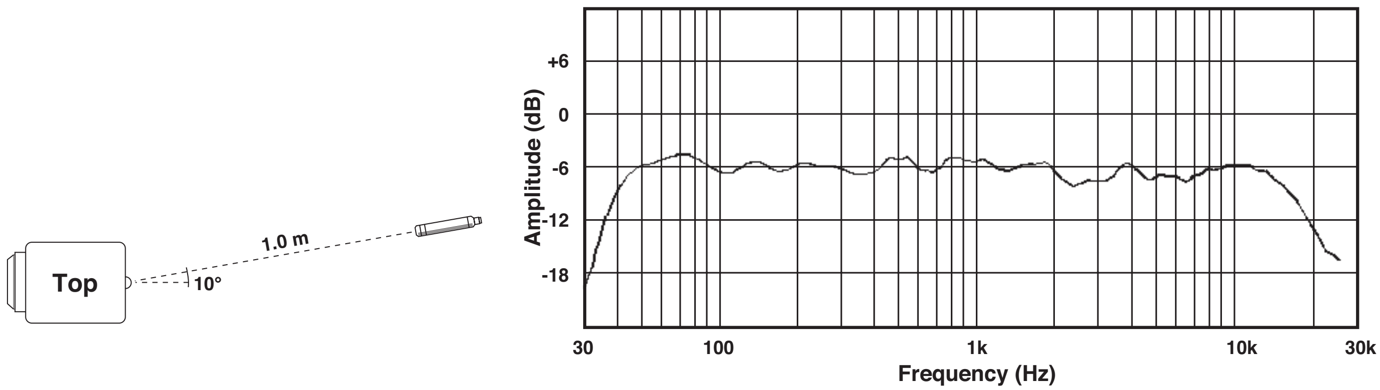


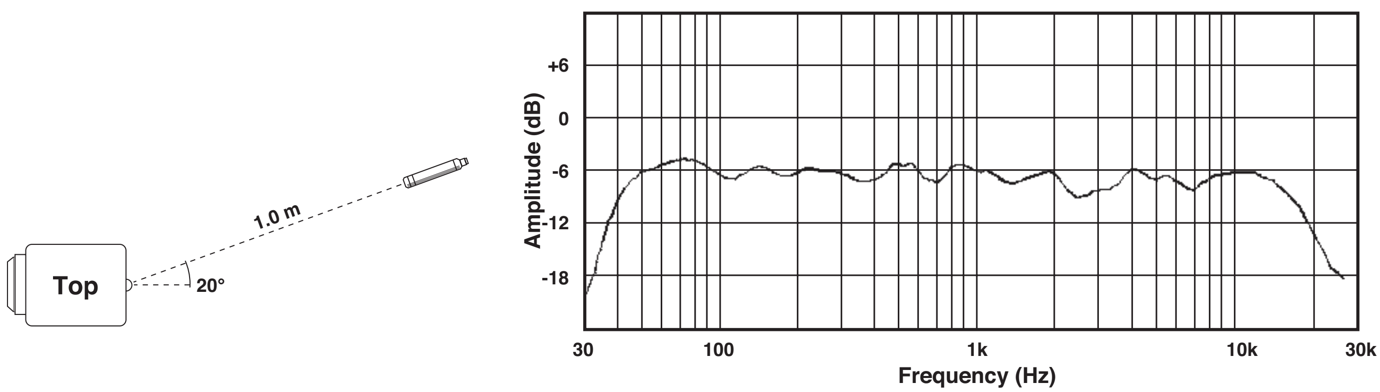


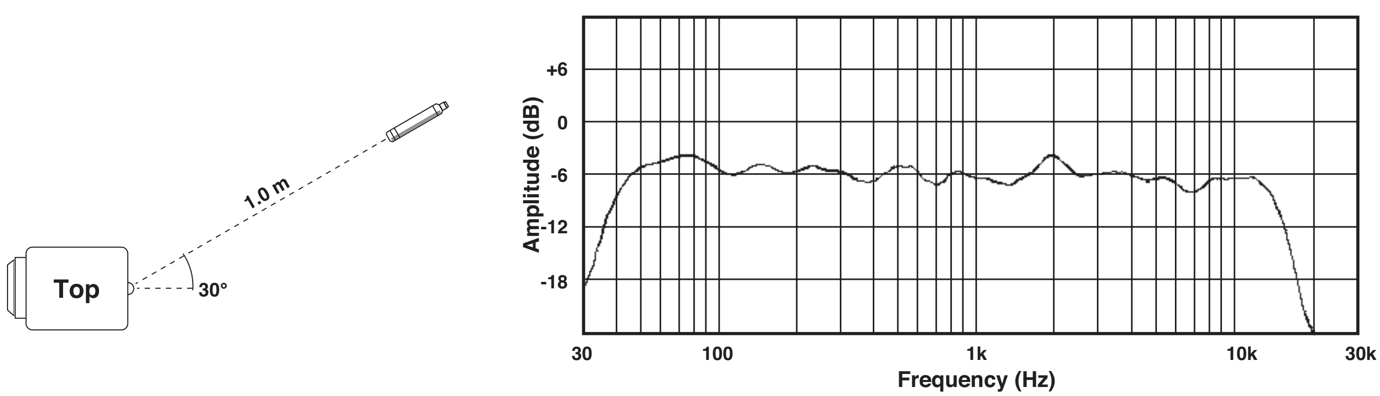





Top
1.0 m
+6
0
-6
-12
-18
Amplitude (dB)
30
100
1k
10k
30k
Frequency (Hz)


Top
1.0 m
10°
+6
0
-6
-12
-18
Amplitude (dB)
30
100
1k
10k
30k
Frequency (Hz)


Top
1.0 m
20°
+6
0
-6
-12
-18
Amplitude (dB)
30
100
1k
10k
30k
Frequency (Hz)


Top
1.0 m
30°
+6
0
-6
-12
-18
Amplitude (dB)
30
100
1k
10k
30k
Frequency (Hz)

Acknowledgements

Figure 3 is reprinted from *The Anechoic Chambers at the Technical University of Denmark*, Brüel & Kjaer Technical Review no. 2 (1968), with the permission of Brüel & Kjaer.

Figure 4 is reprinted from “Acoustic Properties of Anechoic Chamber” by N. Olson, J. Acous. Soc. Am. vol. 33 no. 6 (June 1961), with the permission of the Acoustical Society of America.

Figure 5 is reprinted from *Validity of Intensity Measurements: Influence of Tripods and Mic Clips*, Brüel & Kjaer Technical Review no. 4 (1985), with the permission of Brüel & Kjaer.

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