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Measurements for Loudspeaker Modeling Files

Loudspeaker modeling files are most often used with acoustical modeling programs such as EASE, Focus, CATT-Acoustic, and others to help determine how well the loudspeakers cover the audience areas and how much "spill" there is on the room's surfaces. Some of these programs also allow for the investigation of additional metrics like reverberation time, clarity, speech transmission index (STI), and more. For many of these items it is important to have accurate data about the characteristics of the loudspeakers that are used in the models. Errors in the data used for these modeling programs will often lead to errors in the results the programs generate. In this article we will details some of the requirements to help assure good measurement data for use with loudspeaker modeling.

It is also possible to use some of these loudspeaker modeling files to help optimize the design of crossover filters for the loudspeaker system. Since these modeling files have both on-axis and off-axis data the effects of the crossover and equalization filters used for each pass band can be seen in the overall directivity response of the loudspeaker system. We will also show an example of this in a follow-up article.

While it is sometimes possible to measure a multi-transducer loudspeaker system as a single radiating, full-range device, these measurements may be limited in their use. We certainly cannot use such measurements to help design the crossovers for the loudspeaker system. It is preferred that each separate pass band of a loudspeaker system is measured individually, when possible. These measurements are best performed with no filters in place. This allows the natural response of the drivers mounted in their enclosure to be captured.

There are some loudspeaker systems that do not allow for these separate pass band measurements. This can be a result of their design or operation. Each type of loudspeaker system must be evaluated as to how it functions acoustically. It is only possible to measure each pass band separately if each pass band actually radiates acoustical energy separately from the other pass bands in the loudspeaker.

Measurement Distance

The first thing we must do is to make sure the measurements are made in the far-field of the device we are measuring. Too often I hear people indicating they perform measurements at 1 m. After all, that's the standard distance, right? WRONG! For all but very small loudspeakers a distance of 1 m is way too close to the loudspeaker. At that distance we are most likely still in the near-field of the loudspeaker. In the near-field of a device the frequency response can change as a function of distance from the device. In the far-field the frequency response of the device itself will not change. Although air absorption can occur at higher frequencies when the

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measurement distance is relatively large, this is typically not a concern for the measurements we will perform.

The far-field of an acoustical radiator can be a function of both its size and the frequency (wavelength) which it radiates. In the low frequency region the distance at which the transition from near-field to far-field occurs is dependent only on the wavelength radiated. It is given by Equation 1. From this we can see that our measurement distance needs to be at least one wavelength away from the device we are measuring.

$$Distance \geq \lambda \qquad \text{or} \qquad Distance \geq {}^{f}/_{c}$$

Equation 1: Far-field distance based on wavelength (frequency)

Where λ is the wavelength, f is the frequency, and c is the speed of sound.

In the high frequency region the distance at which the transition from near-field to far-field occurs is dependent on both the size of the source and the wavelength radiated. For a line source, or a source that is like a line, the transition distance is given by Equation 2.

$$Distance \geq h^2 * {}^{f}/_{2c}$$

Equation 2: Far-field distance based on the height of a line source

Here h denotes the height of line-like radiator.

For non-line source devices, like pistons, cone drivers, and horns, the transition distance is given by Equation 3. These devices have appreciable surface area and/or a relatively low aspect ratio (height/width).

$$Distance \geq S * {}^{f}/_{c}$$

Equation 3: Far-field distance based on a the radiation surface area of a non-line source

Here S denotes the surface area of the radiator.

When evaluating S for a loudspeaker we should often include the entire front surface of the loudspeaker, not just the surface area of drivers themselves. This is necessary to account for the diffraction that occurs at the edges of the loudspeaker. These, too, are sources of radiation that are generally excited by woofers, midrange drivers, and dome tweeters.

Let's look at an example loudspeaker and see how far away we need to be for far-field measurements. We will use the loudspeaker shown in Figure 1. This loudspeaker has a height of 0.7 m (27.5 in.) and a width of 0.4 m (15.8 in.) This loudspeaker uses a horn loaded HF pass band and a direct radiator LF pass band. A rough rule of thumb often used for the measurement distance is 3x the baffle diagonal. This would be about 2.4 m (7.9 ft.) and is shown by the purple line (Figure 2).

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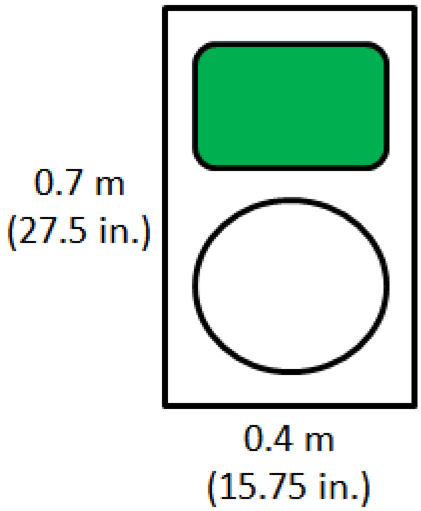

Figure 1: Example loudspeaker

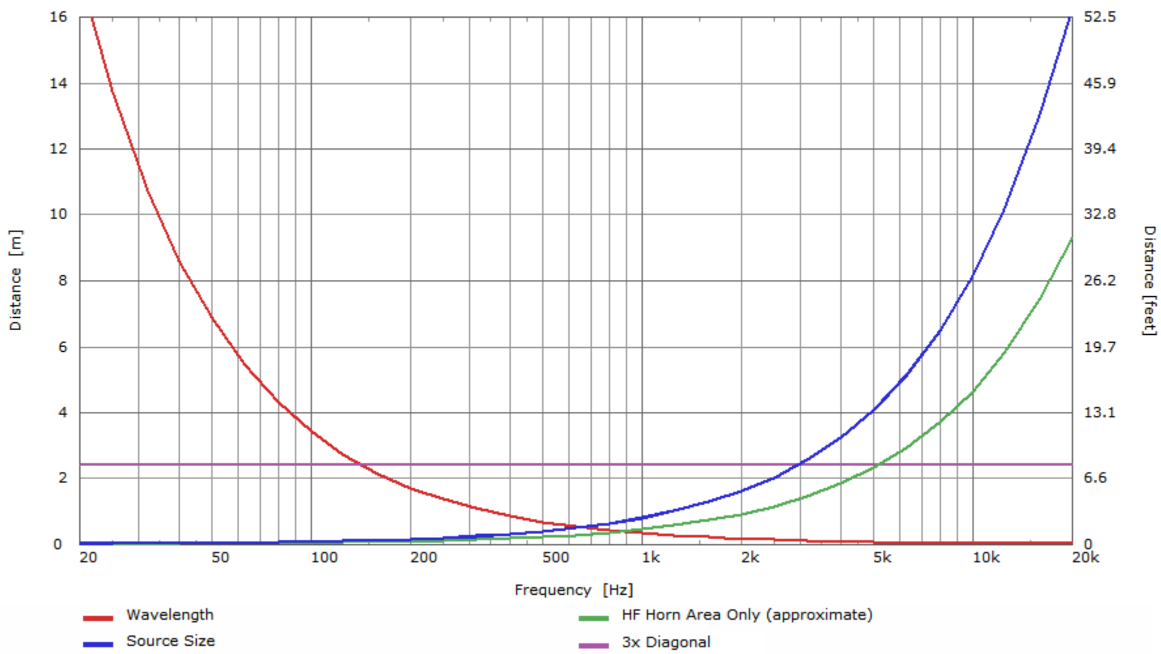

Figure 2: Graph of far-field distance requirements for the example loudspeaker shown in Figure 1

The red curve is the minimum distance as defined by wavelength only. The blue curve is the minimum distance as defined by the source size (the entire baffle area). The green curve is the minimum distance required when we consider only the area of the HF horn mouth. Since the horn has good directivity control at high frequencies its radiation is not "illuminating" the baffle edges. We can typically use this distance in such cases.

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We can see from the curves in Figure 2 that the 3x diagonal rule of thumb ensures we are in the far-field from only about 150 Hz to just above 5 kHz for this loudspeaker. If we need accurate data up to at least 10 kHz we need to perform our measurements at least 4.7 m away from this loudspeaker.

Directivity & Angular Resolution [1]

When creating loudspeaker modeling files we need to know how the loudspeaker system radiates sound into 3D space. We must measure the off-axis radiation in order to characterize the loudspeaker's directivity response. To do this accurately our angular resolution must be finer than the lobes and nulls that the loudspeaker radiates. In other words, our spatial sampling interval must be finer than the spatial attributes of the directivity. This can be likened to the sampling frequency for an audio signal. If there are frequencies higher than one-half the sampling frequency in the signal there will be aliasing in the result. Similarly, if our spatial (angular) sampling does not have at least two measurement points for a lobe or a null our measurement data will suffer from spatial aliasing.

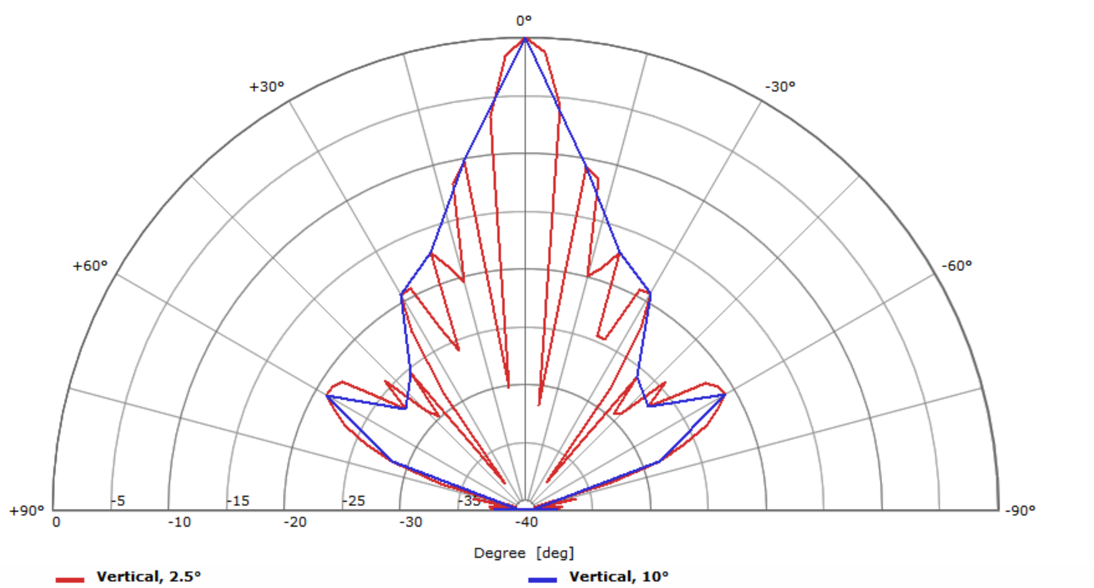

Figure 3: Polar response plots for a line source measured at 10° (blue) and 2.5° angular (red) resolution

In Figure 3 we can see polar response plots for a line source device measured at 10° resolution and at 2.5° resolution. It is clear from looking at this graph that a spatial sampling of 10° is not sufficient to accurately characterize the directivity of this device. At some of measurement points for the 10° resolution data the SPL is shown to be higher than it actually is. At other points it is shown to be lower than it actually is.

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So how can we know what angular resolution is required before we make our measurements? The off-axis angles at which nulls occur are related to the wavelength (frequency) radiated and the dimension of the source in the plane of interest. Mathematically, this relationship is shown in Equation 4.

$$\theta_i = \sin^{-1}\left(\pm i\,\frac{c}{fl}\right)$$

Equation 4: Off-axis angles at which nulls occur

Where f is the frequency, l is the length of the source in the plane of interest, i is a non-zero integer, and c is the speed of sound.

We can use this relationship to calculate the angular difference between two adjacent nulls. To assure we do not have undersampling errors (aliasing) we must have an angular resolution for our measurements of at least half the angle between these adjacent nulls (Equation 5).

$$\Delta\theta = \frac{1}{2}\left[\sin^{-1}\left(2\,\frac{c}{fl}\right) - \sin^{-1}\left(\frac{c}{fl}\right)\right]$$

Equation 5: Angular resolution requirement for directivity measurements

As an example, let's say we want to accurately measure the vertical directivity up to at least 10 kHz for a ribbon driver that is 30 cm (11.8 in.) tall. At 10 kHz the first and second vertical nulls occur at approximately 6.6° and 13.3°, respectively. One-half the difference between these angles is about 3.4°. The angular resolution of our measurements must be at least this. If we were to measure at an angular resolution of 2.5° our data would be accurate up to about 13 kHz.

Directivity & Point of Rotation [2]

When performing off-axis measurements of a loudspeaker system a point of rotation (POR) must be selected. This is the point about which the loudspeaker is rotated so that the measurement microphone is at the desired off-axis angle. If we are only interested in the magnitude of the off-axis frequency response the location for the POR is not too important, within reason. However, if we will be using the measurement data for simulations in order to calculate the combined response of multiple devices the location of the POR can be quite critical.

The reason for the importance of the POR has to do with the acoustic center (the point from which the radiation appears to originate) of the device under test (DUT). Ideally we would like for the POR to be coincident with the acoustic center of the DUT. This is not always possible. A primary reason for this is that the acoustic center of an individual driver, much less a complete loudspeaker system, is rarely confined to a single location for all the frequencies radiated by the transducer or loudspeaker system. That is to say, the acoustic center can change location as a function of frequency.

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If our measurements contain complex data (e.g. magnitude and phase) it is possible to deal with the POR not being coincident with the acoustic center, up to a certain point. The measured phase data can act as an error correction mechanism for the offset distance between the POR and the acoustic center of the DUT. Note that this data must be the actual measured phase response of the DUT, not the phase response derived from the Hilbert transform of the magnitude-only data. There is a limit to the offset distance that the phase data can effectively handle. This can generally be stated as a maximum of one-quarter wavelength at the highest frequency of interest.

This limit for the offset distance, which has been termed the critical distance, manifests itself in the form of two different criteria. One is based on the measurement distance (the distance between the POR and the measurement microphone) and the other is based on the angular resolution used for the off-axis measurements. Each of these criteria are shown in Equation 6 & Equation 7, respectively. The lesser of these two values for the critical distance will be the governing value for the critical distance.

$$x_{crit} \approx \sqrt{{}^{cd}/_{4f}}$$

Equation 6: Critical distance based on measurement distance

$$x_{crit} \approx \frac{c}{4f\,\sin(\Delta\theta)}$$

Equation 7: Critical distance based on angular resolution

Where c is the speed of sound, d is the distance from the POR to the measurement microphone, f is the highest frequency of interest, and $\Delta\theta$ is the angular resolution for the measurements.

Let's look at an example selection of the POR for a set of directivity measurements. We'll use the same loudspeaker shown in Figure 1. This is a two-way loudspeaker system with a 12 dB/octave passive crossover at a frequency of about 1.2 kHz. We want to be able to accurately model this loudspeaker in clusters/arrays up to about 10 kHz. We will be measuring at a distance of 4 m using an angular resolution of 5°. From this information we can calculate the critical distances from the woofer's acoustic center and the horn's acoustic center. The radii for these distances will define spheres, shown here as circles, around the acoustical center of each radiator. The area where the circles (spheres) intersect is where the POR must be located for our measurements (Figure 4).

The highest frequency of interest for the woofer is about 4.8 kHz. This is 2 octaves above the crossover frequency. If the crossover used higher order filters (steeper slopes) the woofer would not be radiating to as high of a frequency. Using the other values stated above gives us a critical distance (x_{crit}) of 0.268 m (10.6 in.) based on the measurement distance of 4 m and a critical distance of 0.206 m (8.11 in.) based on the angular resolution of 5°. Since the smaller

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value governs, the critical distance for the woofer is 0.206 m. The purple dashed circle in Figure 4 denotes this. It has a radius of 0.206 m, centered at the acoustic center of the woofer.

The highest frequency of interest for the HF horn is 10 kHz, as this is the upper limit frequency for our modeling. Using the other values stated above gives us a critical distance (x_{crit}) of 0.186 m (7.32 in.) based on the measurement distance of 4 m and a critical distance of 0.098 m (3.68 in.) based on the angular resolution of 5°. Since the smaller value governs, the critical distance for the HF horn is 0.098 m. The orange dashed circle in Figure 4 shows this. It has a radius of 0.098 m, centered at the acoustic center at the mouth of the HF horn. Note in the side view of Figure 4 that the acoustic center can move from the mouth of the horn towards the throat of the horn at higher frequencies. The critical distance around this acoustic center closer to the throat of the horn is shown by the blue dashed circle. The entire range of locations for the HF horn's acoustic center must be considered.

The point of rotation for the off-axis measurements must be located where all of these circles intersect. Keep in mind that the POR and acoustic centers are actually defined in 3D space so the circles represent spheres of the same radius as the circles shown.

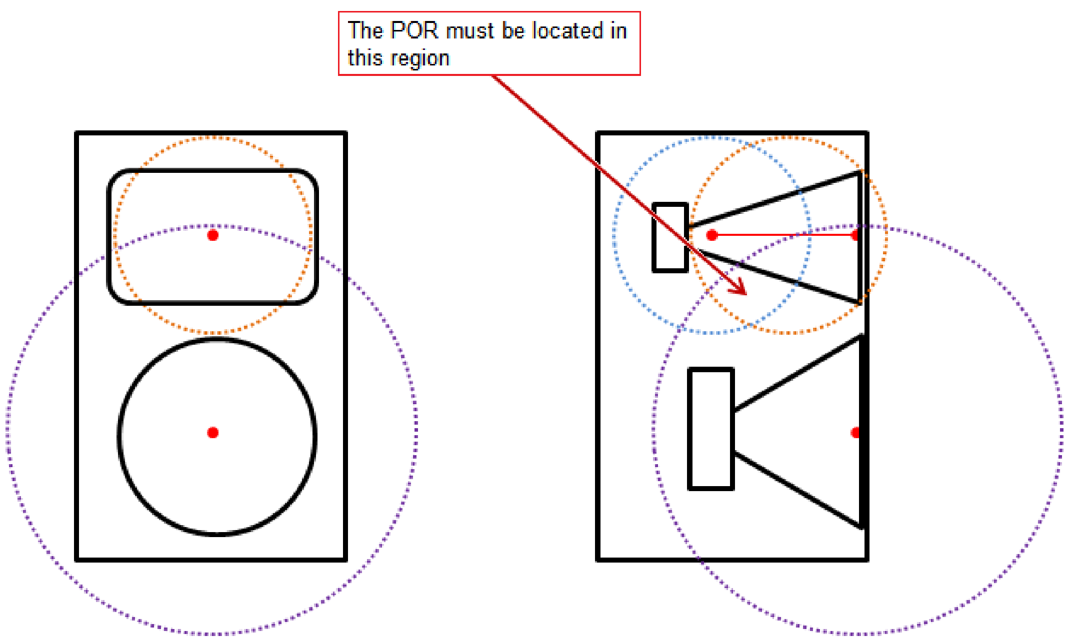

Figure 4: Front and side view of a loudspeaker with the acoustic centers & the circles defining the critical distance for each source of radiation

Crossover Measurements

When the individual pass bands of a loudspeaker system are measured separately, with no filters in place, we also need to measure the crossover filters that will be used with the

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loudspeaker system so they, too, can be included in the loudspeaker modeling file. It is a relatively straight forward process to measure the crossover outputs but there are some things that must be considered.

These measurements must contain complex data, both magnitude and phase. If they are magnitude-only the resulting summation of the individual pass bands will not be correct. So a transfer function measurement must be performed. Here again it is important that the measured phase response is used and not the Hilbert transform of the magnitude-only data. If an active or DSP-based crossover uses delay for some pass bands this delay must be included in the phase response. Only the measured phase response will contain this. The Hilbert transform will just have the minimum phase component and not the complete phase response.

For measuring the transfer function of a passive crossover it is best to measure at the input terminals of the driver(s) for each pass band. This gives an accurate representation of the signal present at the input of each pass band. Since a power amplifier is typically required to drive the input to the passive crossover care should be taken when setting the output voltage from the amplifier. An rms voltage of 1 V is probably a good starting point. The inputs of most audio analyzers should be able to accommodate this voltage without any problems.

The polarity of the measurement data must be maintained compared to the polarity of the signal at the input to the loudspeaker drivers. That is to say, the positive lead of the measurement system should be connected to the positive input terminal of the driver and the negative lead of the measurement system should be connected to the negative input terminal of the driver. If the measurement system leads are not connected in this manner the polarity of the transfer function measurement will be reversed. This can cause errors in the simulations using this data.

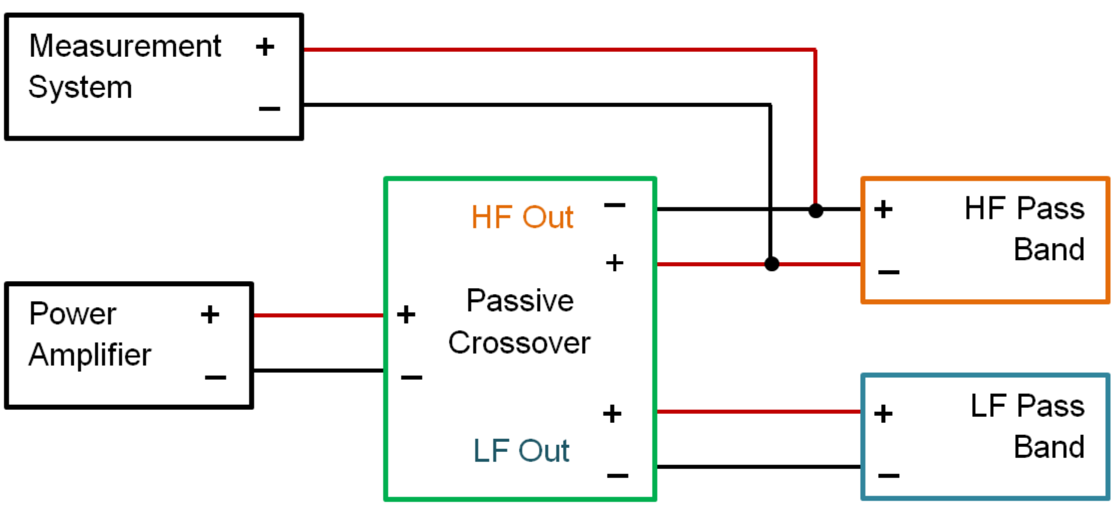

Figure 5: Recommend connection of measurement system with balanced inputs (ground-isolated) for measuring crossovers

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It is strongly recommended that only ground-isolated, balanced inputs be used for measuring crossover outputs, particularly passive crossovers. It is not uncommon for passive crossovers to have an intentional polarity reversal for one or more of the outputs as part of the intended design. This is shown in Figure 5 where the HF output of the crossover is reversed polarity. If a measurement system with a grounded, unbalanced input is used to measure a crossover output having reversed polarity, the positive output of the amplifier driving the crossover will be grounded. This will result in the output of the amplifier being shorted to ground. Not good!

Hopefully this has given you some ideas about how to make good measurements for use in loudspeaker modeling data files. In the next article we will take a look at using loudspeaker modeling files to help design a crossover for optimum directivity response through the crossover region. Stay tuned.

References

Some of the material presented in the referenced sections was originally published in the presentations at AES Conventions and the Journal of the Audio Engineering Society. For more details on these topics please see the papers listed below.

[1] S. Feistel and W. Ahnert. “Methods & Limitations of Line Source Simulation”, *J. Audio Eng. Soc.*, vol. 57, No. 6, p. 379 (2009 June)

[2] S. Feistel and W. Ahnert. “Modeling of Loudspeaker Systems Using High-Resolution Data”, *J. Audio Eng. Soc.*, vol. 55, No. 7/8, p. 571 (2007 July/August)

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