Measuring Loudspeaker Low-Frequency Response
by
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Introduction

With the availability of low-cost PC-based acoustic data acquisition systems experienced loudspeaker designers can make accurate loudspeaker frequency response measurements without an anechoic chamber. Popular systems include CLIO, MLSSA, Praxis, and Sound Easy to name a few. The power of these systems comes at a price. These systems window the time domain measurement to eliminate room reflections. This in turn limits low-frequency response. Typically, response below 200 to 300Hz is not possible in rooms of reasonable size.

The near-field technique proposed by D.B. Keele circa 1973 [1, 2] is the commonly accepted way to get low-frequency data without an anechoic chamber. However, there is another technique proposed by R.H. Small in an AES paper circa 1971[4]. His technique is simple. It does not require phase information and avoids some of the complexity of the Keele procedure especially when there are multiple radiating surfaces. In this article I review the Keele procedure and then present Small's approach. Examples of both are given and discussed. The emphasis in this article is on the practical application of the Keele and Small techniques and some of the problems that come up in their application. The theory of each technique is well covered in the sited references.

The Problem

Our goal is to measure the on-axis far-field response of a loudspeaker free of any room effects. Reflected energy from the nearby floor, walls and ceiling will arrive at the test microphone later than the direct wave. Depending on the path length difference and therefore the phase difference between the two arrivals, the reflected waves may add to or subtract from the direct wave.

Let's look at the effect of a single reflecting surface. Typically, if the speaker under test is placed on the middle of the floor, far from other reflecting surfaces, the first reflection will come from the floor. This condition is illustrated in Fig. 1. In this figure the driver to be tested and the test microphone are both at a height of one meter. The direct distance, d_1, from driver to microphone is also one meter. The wave reflected by the floor travels the longer path, $2d_2$.

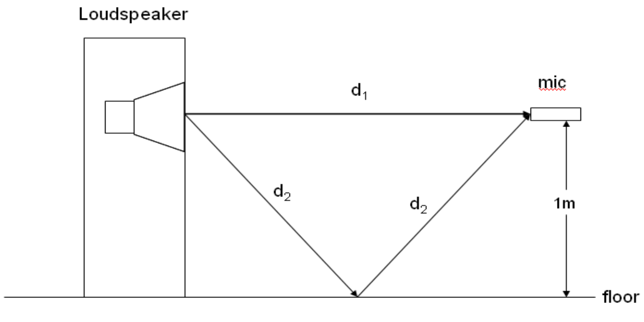

Fig 1 Floor reflection geometry

Whenever the distance $2d_2$-d_1 is equal to an even multiple of a wavelength the direct and reflected waves will add directly. Whenever this distance is an odd number of half wavelengths, the reflected wave will be 180° out of phase and subtract from the direct wave. At intermediate distance-to-wavelength ratios, there will be partial addition or subtraction. The reflected wave will be somewhat weaker than the direct wave since it travels a longer distance so complete cancellation of the direct wave will not occur.

Fig 2 is a plot of the measured impulse response of a small two-way monitor loudspeaker. The CLIO electrical and acoustical measurement system was used for this and all subsequent measurements. Additional details on the measurement equipment are outlined at the end of this article. The measurement geometry is similar to that of Fig 1. Remember the impulse response is the time-domain equivalent of frequency response. The two are related by the Fourier Transform.

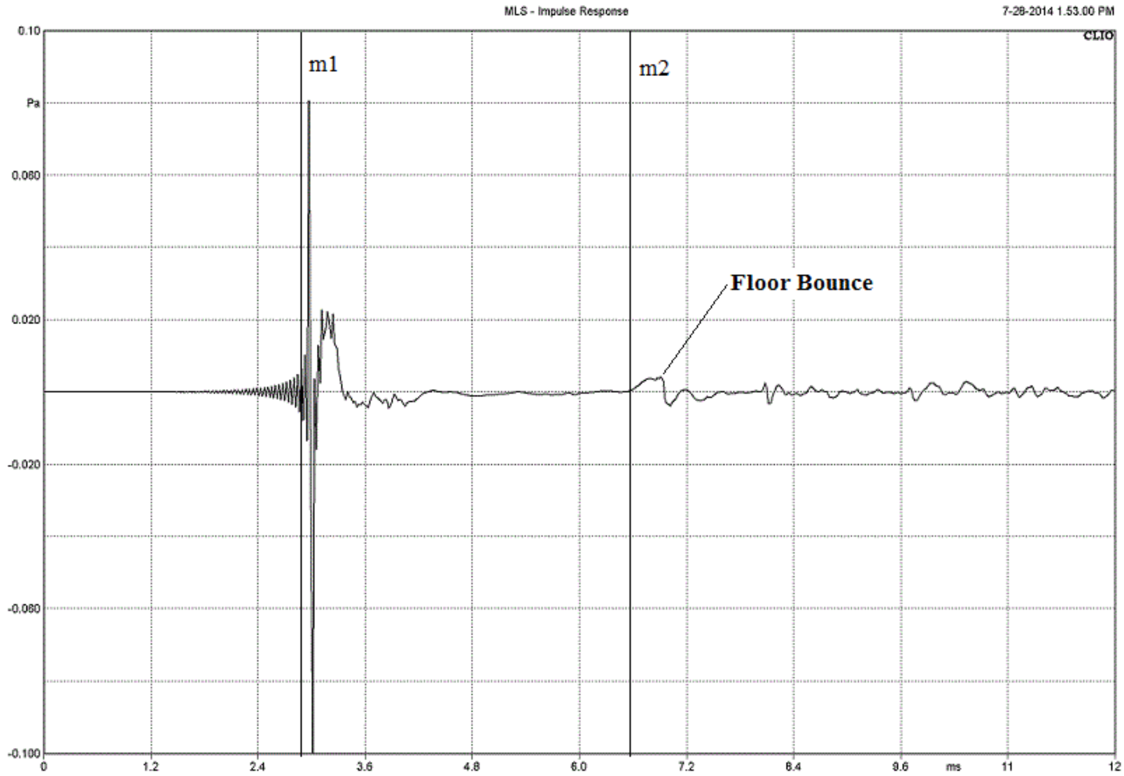

Fig 2 Measured impulse response of small two-way loudspeaker

Referring to Fig 2, the direct on-axis arrival of the impulse response at the microphone is somewhat obscured by the transient build-up of the finite-impulse-response (FIR) anti-aliasing filter used in CLIO. Computation of the excess group delay, a post-processing option in CLIO, indicates the arrival at 2.99msec after the test signal is applied to the loudspeaker. The floor reflection arrives at 6.53msec. It follows that we have 3.54msec of reflection free data. If we analyze only the data between markers m1 and m2 we get the relatively smooth frequency response shown by the green curve of Fig. 3

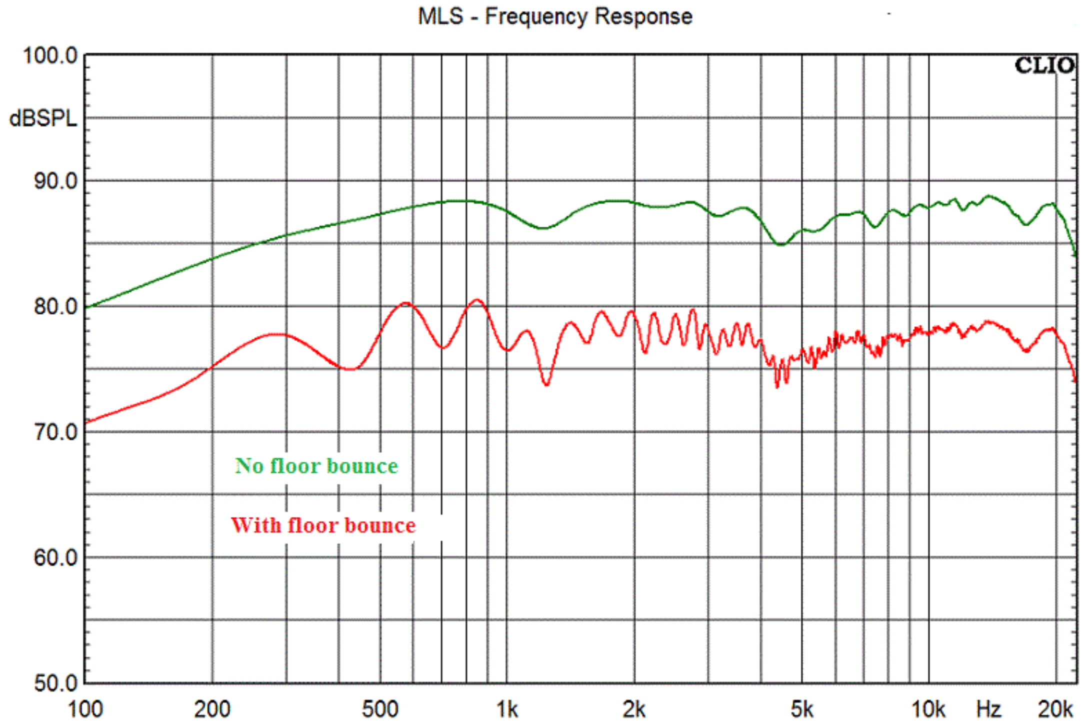

Fig 3 Two-way frequency response with and without floor bounce

Extending our analysis out to 12msec we now include the floor reflection and get the red response curve in Fig. 3. (The curves are offset by 10dB for clarity.) Now you see the effect of the alternating addition and subtraction on the direct response caused by the reflected wave.

Because we do not get the full anechoic response with the windowing process this response is often termed “quasi-anechoic”. And this is the catch! By using only 3.54msec of data in this example the lowest frequency we can resolve is:

$$f_{\text{min}} = \frac{1}{0.00354} = 282\text{Hz}$$

Any part of the curve plotted below that frequency is simply an artifact of the Fourier Transform and does not represent valid data. Fortunately there are ways to get the low-frequency data which can then be spliced to the high-frequency data to get the full range response. I’ll talk about them next.

One Solution: The Near-field Approach

In this technique, the microphone is placed very close to the driver diaphragm to swamp out baffle and room effects. At low frequencies where the driver diaphragm behaves like a rigid piston, the measured near-field response is directly proportional to the far-field response and independent of the environment into which the driver radiates. D.B. Keele describes this technique in his excellent paper [1]. I will summarize the approach and its limitations here.

For the near-field technique to work properly the microphone should be placed as near to the center of the diaphragm as possible. Keele shows that a microphone distance less than 0.11 times the diaphragm effective radius results in measurement errors of less than 1 dB. As an example, a 7" driver will typically have an effective cone diameter of 5" or an effective radius of 2.5". For this driver, the microphone should be placed within 0.275" of the driver dust cap.

At higher frequencies where cone break up begins, pressure waves from various areas of the diaphragm they may arrive at the microphone out of phase causing near-field response cancellations not observed at normal listening distances. For this reason there is a practical upper limit to the near-field technique given in terms of driver diaphragm diameter. For a driver mounted in an infinite baffle the limit is

$$f_{MAX} = \frac{4311}{D}$$

Here f_{MAX} is in Hertz and the driver diameter, D, is in inches. For closed-box or ported systems with finite baffles this limit may be slightly lower. For our 7" driver example we have

$$f_{MAX} = \frac{4,311}{5} = 862\text{Hz}$$

What happens when we have multiple radiating surfaces as in a vented loudspeaker? Keele has the answer for that too. He shows that the individual near-field responses may be added with proper weighting to get the total near-field response. If all of the radiating surfaces are circular the addition looks like this:

$$p_{tot} = D_1 p_1 + D_2 p_2 + D_3 p_3 + ... + D_n p_n \qquad (1)$$

Where:

p_{tot} = the total near-field pressure

p_n = near-field pressure of the n^th^ circular radiating surface

and: D_n = the corresponding diameter of the n^th^ radiating surface.

For example, if you are testing a vented loudspeaker with two woofers and two port tubes you would take a total of four near-field measurements and add them together after multiplying each one buy its respective diameter.

If some of the radiating surfaces are rectangular you can use the diameter of a circle with the same area. Alternatively, you can weight all measured near-field pressures by the square root of the area each respective radiating surface before adding them together. In his paper Keele goes on to show how the near-field response can be extended to the far-field. However, in this paper we are only interested in the low-frequency response shape which will be merged with a measured far-field response.

The Keele approach seems pretty straightforward, but there are some things to be careful of. First off, Keele assumes all radiating surfaces are mounted on an infinite baffle. Under this condition the radiation is into a "half-space" or a solid angle of 2π. However, most loudspeakers have relatively narrow baffles so that they become omni-directional at low frequencies causing the on-axis response to fall by as much as 6dB in this frequency range. Loudspeaker designers generally try for flat on-axis response by tailoring the crossover to compensate for this drop in frequency response. If a speaker designed this way is now tested in a half-space it will show a rise in low-frequency response. For this reason the Keele approach may over estimate the low-frequency sound pressure level.

Second, the near-field pressures are complex quantities. That is they have both magnitude and phase. If more than one radiating surface is involved in the testing, a simple pressure magnitude measurement is not enough; you need a system that measures both magnitude and phase to sum the responses correctly. If radiating surfaces are close together your measurements may be contaminated by crosstalk. Also, without direct acoustical measurement, the effective areas of each radiating surface are only approximated.
Finally, the upper frequency limit on port pressure measurements tends to be much lower than that for a diaphragm of the same diameter. Even with these caveats the technique is useful when no anechoic chamber is available.

It's time for an example. We have the far-field on-axis response of the two-way monitor first examined in Fig 2. Let's look at the low-frequency response using the Keele approach. This speaker is vented so we will have to measure both the woofer and port near-field responses. The results are plotted in Fig. 4A. For the port measurement the microphone was placed in the plane of the port exit.

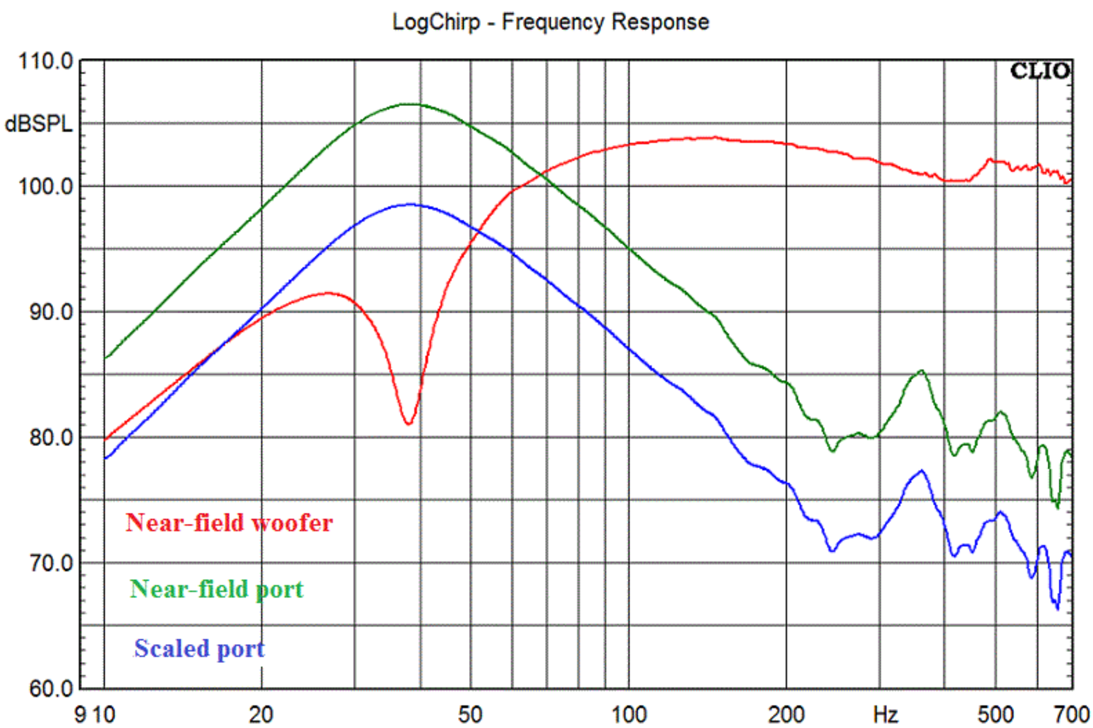

Fig 4A Near-field woofer, port and scaled port responses

At first glance the port output seems to be about 8dB higher than the woofer output; a counter intuitive result. This is due to the diameter difference between the woofer and port. We can scale the port output to the correct relative level by writing the summing equation in a different form, namely:

$$p_{\text{tot}} = p_{\text{woofer}} + \frac{D_{\text{port}}}{D_{\text{woofer}}} p_{\text{port}} \qquad (2)$$

For this example the woofer effective diameter is 13.8 cm and the port diameter is 5.5cm, so we have:

$$\frac{D_{\text{port}}}{D_{\text{woofer}}} = \frac{5.5}{13.8} = 0.398 = -8.0\text{dB}$$

The scaled version of the port response is also plotted in Fig. 4A. Now you can see that it is more in line with the woofer level.

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Before leaving Fig. 4A there are two interesting points not directly related to near-field testing. First, the sharp dip in woofer response at 36.8Hz indicates the tuning frequency of the vented enclosure. In general, this value is more accurate than one obtained from the impedance curve since it is not corrupted by voice coil inductance. Second, the up tick in port response around 400Hz is possibly caused by woofer back wave leaking out through the port. When added to the front firing woofer response it produces a small dip in total near-field response at the same frequency. However, this is not heard in practice because it is almost 15dB lower than the woofer output and because the port exhausts to the rear of our two-way monitor example.

Adding the woofer and scaled port near-field responses we get the complete low-frequency response. This result is plotted in Fig. 4B. At this point we have the *shape* of the loudspeaker's low-end response. Now we have to splice it to the measured far-field on-axis response of our two-way monitor example to get the full range response. We can use CLIO's post-processing "merge" function to do this.

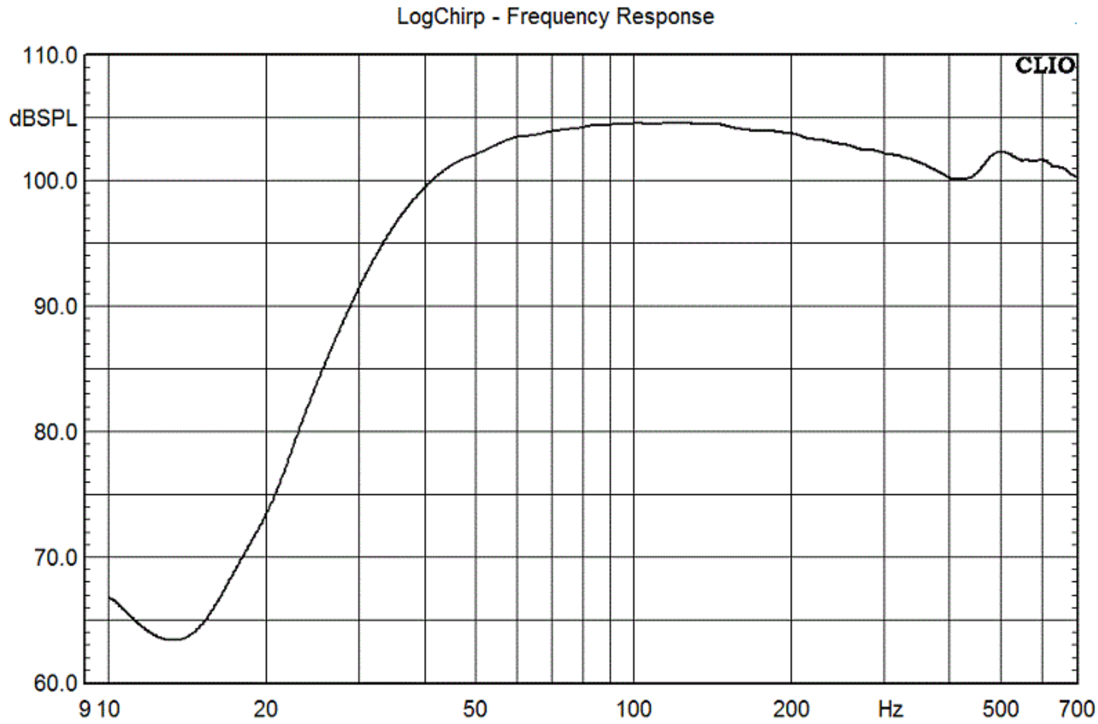

Fig 4B Weighted sum of near-field woofer and port responses

The near-field data are valid below 862Hz, but we have that small dip at 400Hz. The far-field data are valid above 282Hz. Clearly, the two graphs should be merged somewhere is the 282 to 400Hz range. There is no good theory for picking the merge point. This is where the designers experience comes in to play. Regardless of the point chosen, however, the near-field response should always be brought into coincidence with the far-field response since the latter represents the measured loudspeaker sensitivity.

In Fig. 5A I have used CLIO's level shifting post-processing option to align the near-field response level to meet the far-field curve at 300Hz using. This point selection is somewhat arbitrary, but the result shown in Fig. 5B looks reasonable. Absent an anechoic chamber, the near-field approach gives us a good estimate of the low-frequency extension of our two-way monitor.

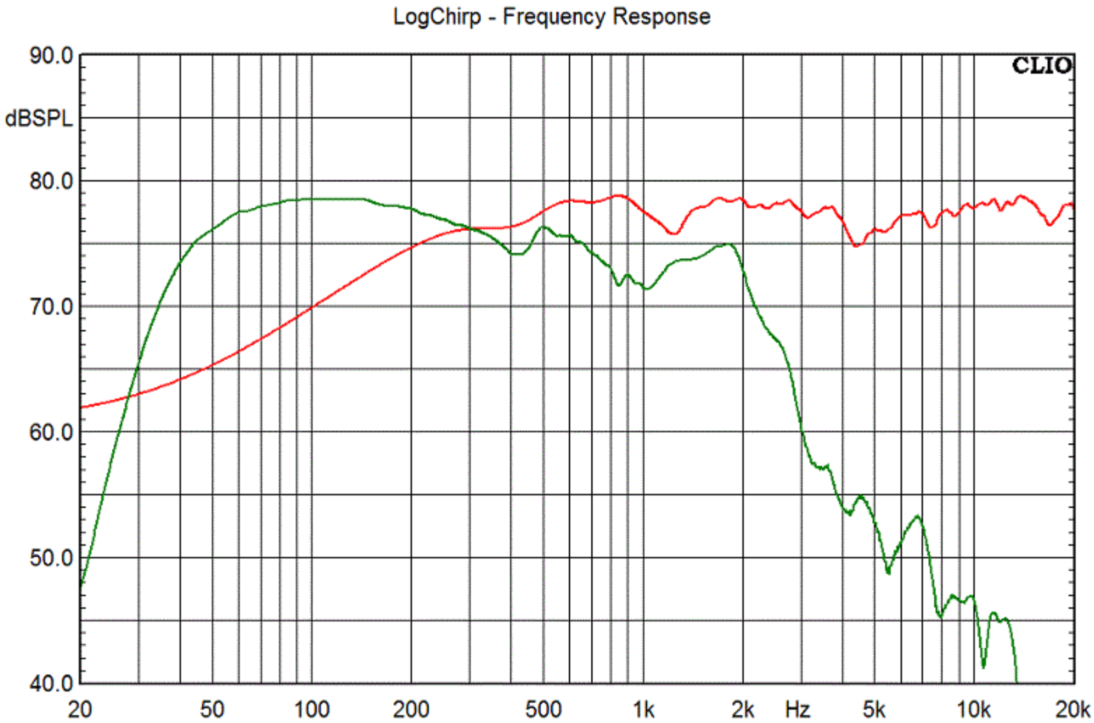

Fig 5A Two-way fa-field and near-field responses

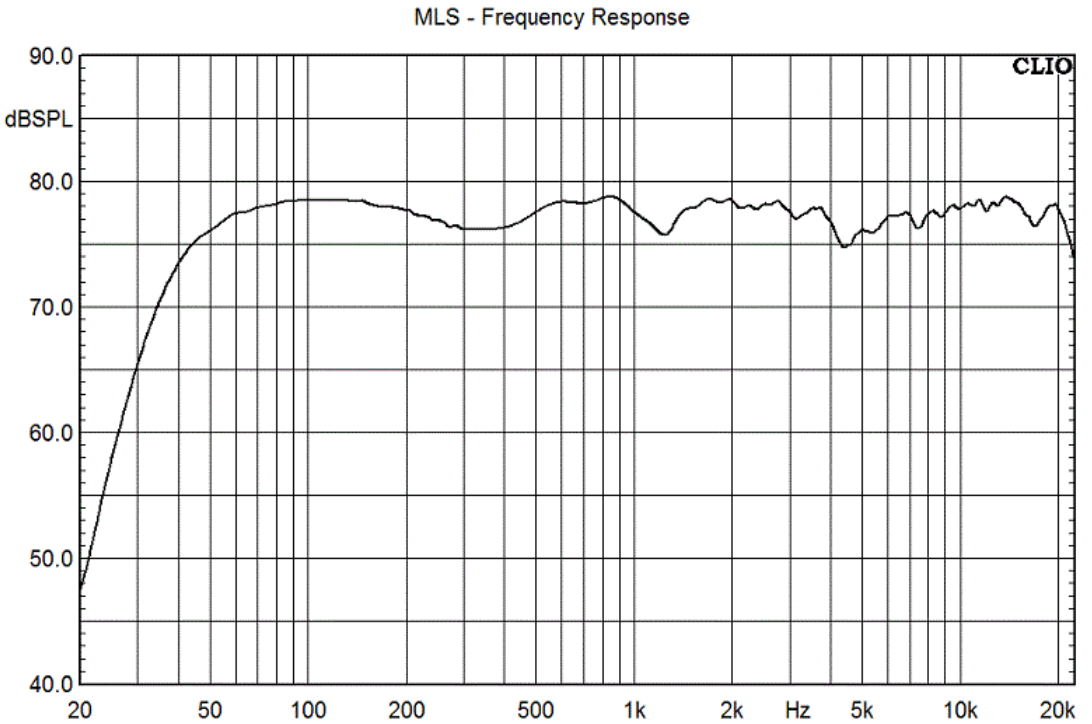

Fig 5B Two-way full range response

The Microphone-in-Box Technique

Using the work of Benson [3], Small [4] showed that at low frequencies there is a simple relationship between the sound pressure level at a distance from an enclosure and the internal pressure within the enclosure regardless of the number of radiating surfaces. This is an amazing result! To determine the low frequency response of our two-way example we need only one measurement of pressure inside the enclosure! Not only that, but because there is only one measurement, phase information is not required. The governing equation can be written as follows:

$$p_{\mathrm{r}} = kf^2 p_{\mathrm{B}} \tag{3}$$

where:

p_{r} = pressure at a distance outside the enclosure

p_{B} = pressure inside the enclosure

k = a constant

and: f = frequency in Hertz

Equation (3) is not too useful in its present form. The squaring operation will lead to rather large numbers. We can avoid this by normalizing the equation to a reference frequency, f_0. Then the equation for p_{r} becomes:

$$p_{\mathrm{r}} = k\left(\frac{f}{f_0}\right)^2 p_{\mathrm{B}} \tag{4}$$

Where k has a new value. I have dubbed this process the “Microphone-in-Box” technique or just MIB. p_{r} is the pressure response shape. We still have to scale it to the measured far-field response.

As with the near-field approach, there are some caveats to consider when using the MIB technique. First off we have the same half-space assumption as in the near-field approach. Secondly, it is assumed that the pressure, p_{B} within the enclosure is uniform. Once standing waves build up the equation breaks down. Small thought that the data would be good up to a frequency where the largest dimension of the enclosure equals 1/8th of a wavelength. In practice it is fairly obvious from the data where the technique breaks down. There are also some effects at higher frequencies due to enclosure losses that I will not discuss here (see Ref. 4).

Let’s take a second look at our two-way monitor using Small's MIB technique. I passed a microphone through the port tube and placed it close to the geometric center of the enclosure. The red curve in Fig. 7 shows the in-box pressure measurement taken at that location. The in-box pressure response peaks close to the box tuning frequency of 36.8Hz. Above and below that frequency in-box pressure response falls off by 12dB/octave.

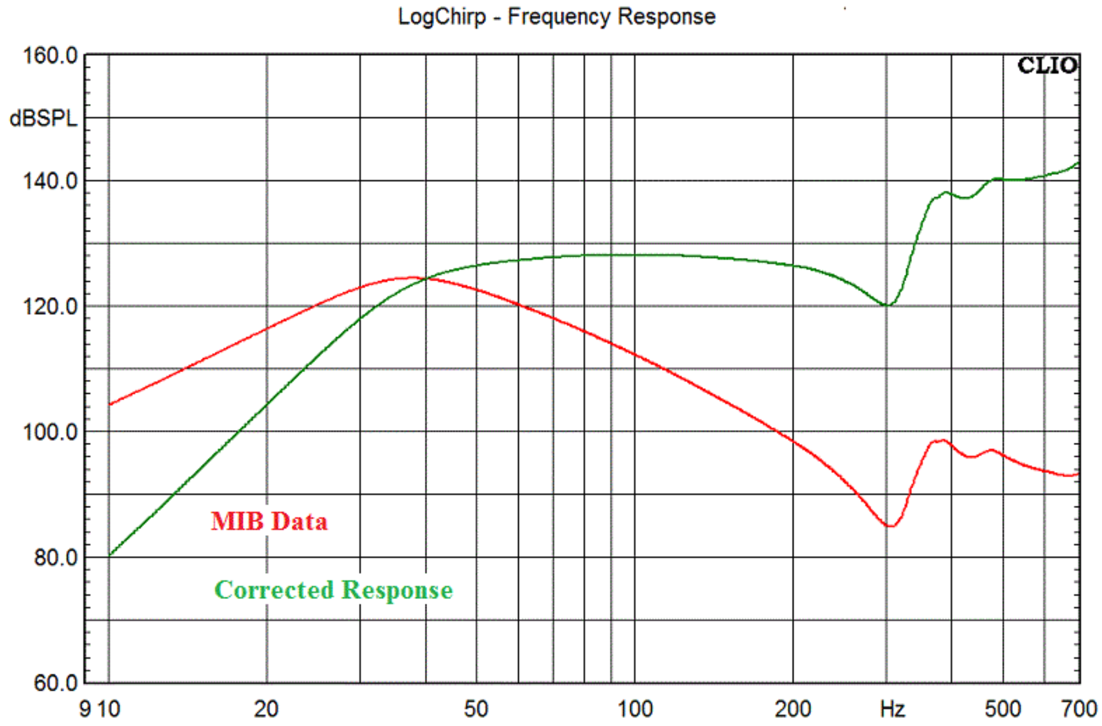

Fig 7 Two-way MIB measurements

Another of CLIO's post-processing functions multiplies a measured response by frequency. Selecting the reference frequency f_B of 36.8Hz and applying this post-processing function to the MIB measurement twice yields the low-frequency response shown by the green curve on Fig. 7. Looking at Fig. 7 it is clear that the MIB technique breaks down somewhere above 300Hz in this example.

It is interesting to compare the two low-frequency response methods. This comparison is plotted in Fig. 8. In the plot the MIB response has been level shifted to meet the near-field result at 80Hz. Both responses agree within 1dB between 40Hz and 200Hz. However, below 40Hz the near-field result rolls off more quickly than the MIB result. In the octave between 10 and 20Hz the MIB response falls by 24.1dB which matches the accepted fourth-order hi-pass model for a vented loudspeaker to well within experimental accuracy.

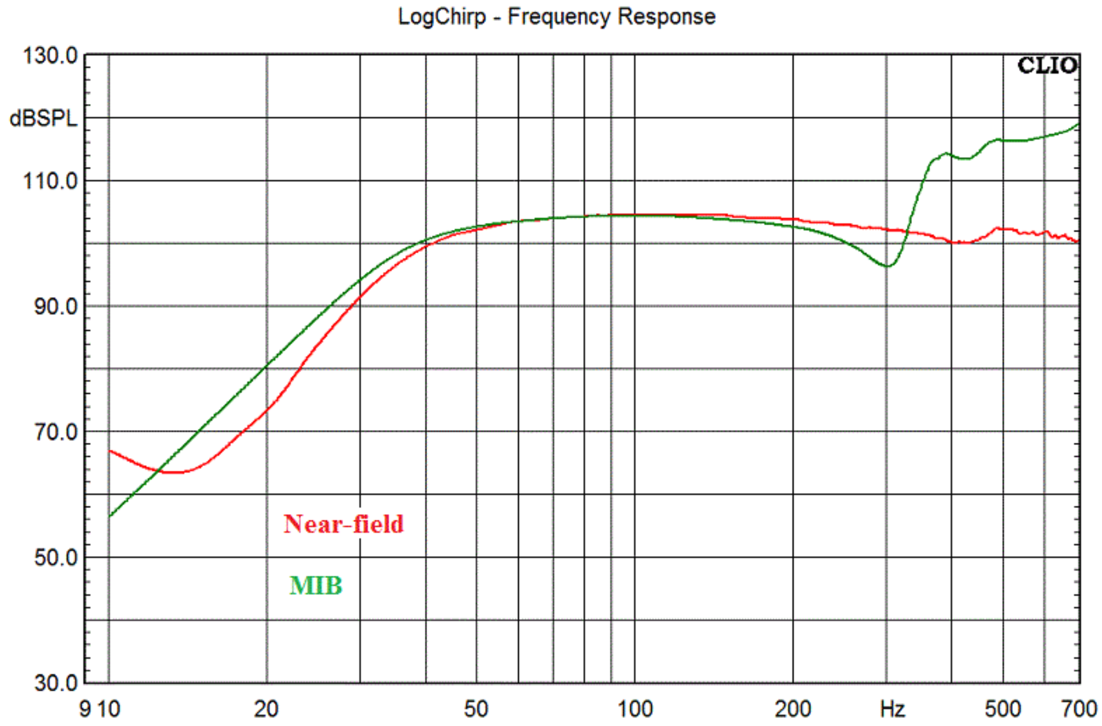

Fig 8 Comparing near-field and MIB responses

By contrast, in the Keele approach response falls at a rate of 28db/oct in the 20-30Hz range. Below 20Hz the slope declines, reaches zero at 15Hz and then turns upwards. This is an unexpected result. Referring to Fig 4A we see that this occurs because port near-field response has fallen below the woofer response at roughly 18Hz. The woofer and port responses are out of phase below f_B so the response in that range results from a *subtraction* of two relatively large quantities to get a very small difference. Clearly a 1dB error in either near-field measurement and/or an error in the effective diameters of either the port or woofer or both could lead to response error seen here.

Based on this discussion I tend to believe the MIB result below 40Hz and the near-field result above 200Hz for this example. This suggests that the best estimate of the low-frequency response for our two-example would be a combination of the two results. Fig 9 shows the result of merging the MIB response below 180Hz with the near-field response above that frequency again using CLIO's "merge" post-processing function.

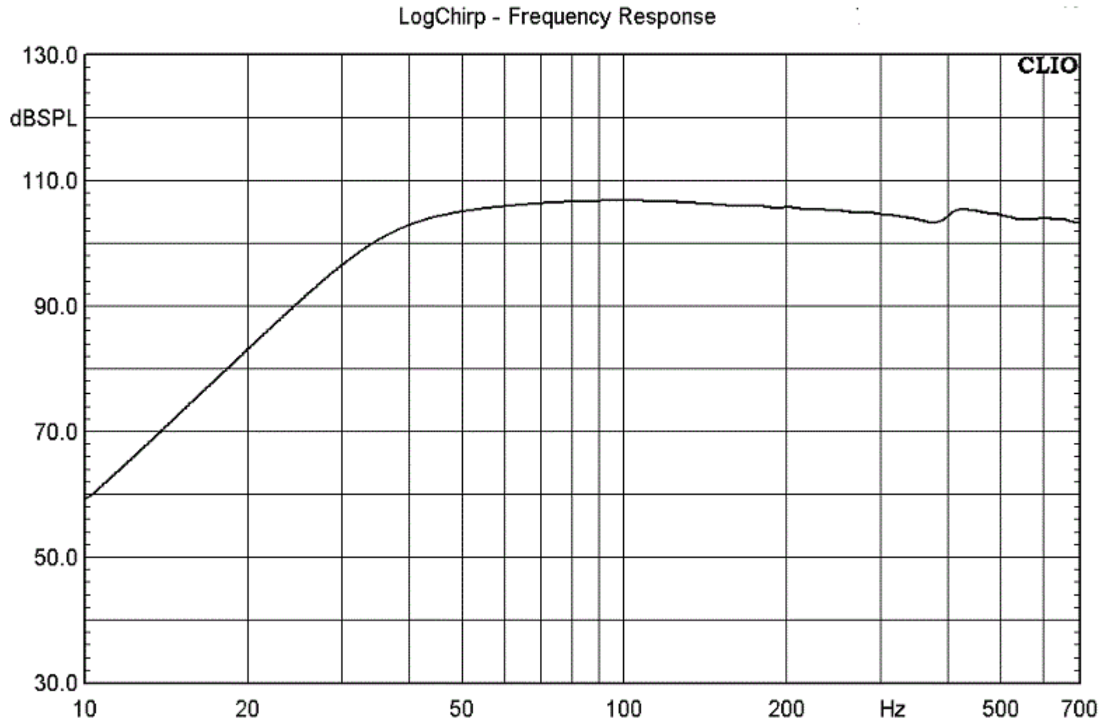

Fig 9 Two-way MIB and near-field responses merged at 180Hz

A Seal-Box Subwoofer

Let's look at results from the testing of a sealed-box subwoofer with two 12" drivers. These tests were run to get the unequalzed subwoofer response. The data was then used to design of an electronic equalizer for the subwoofer to extend low-end response down to 20Hz.

Near-field and MIB data are plotted in Fig. 10. To aid comparison, the MIB response was level shifted to meet the near-field response at 45Hz. Near-field measurements of both woofers were taken and added together. The woofers are mounted on opposite sides of the enclosure so there was little chance of cross-contamination. The near-field magnitude responses of the two woofers were identical and could be added together without regard to phase. Fortunately, in this example there was no port response to consider. A 3/4" hole had to be drilled in the test box to insert the microphone for the MIB test. Blue Tac sealed the space around the microphone cable.

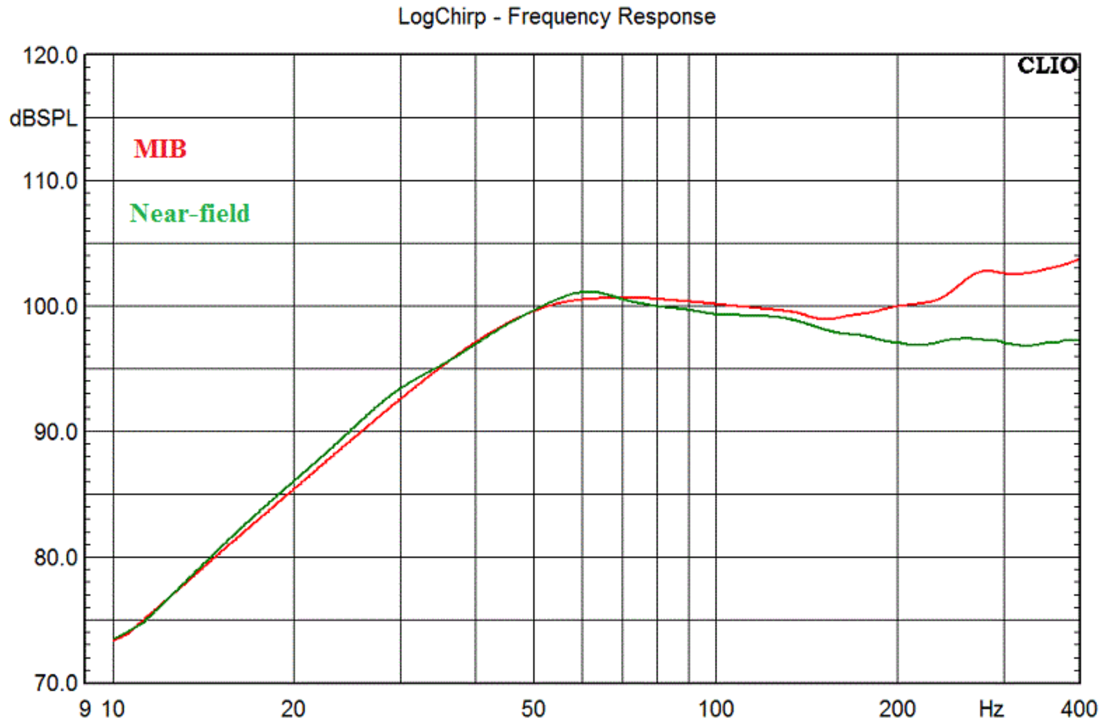

Fig 10 Comparing MIB and near-field subwoofer responses

Looking first at the near-field result, there is some ripple in the response, but it is relatively smooth from 10Hz all the way out to 400Hz. Response peaks at 60Hz falling gently by about 3dB to 200Hz and then flat beyond that point. Below 40Hz the rising slope varies from 11.4dB/oct to 13.3dB/oct.

The MIB response is very smooth out to about 150Hz, but clearly breaks down above that frequency due to the effects of a standing-wave centered on 280Hz. Below 40Hz MIB shows the expected 12dB/oct rise exactly.

Both responses agree within 1dB out to about 150Hz. Both show a gentle roll off above 60Hz. The differences are simply due to the different theories underlying each approach and errors associated with realizing them.

Response data out to 200Hz should be sufficient for designing electronic compensation for a sealed-box home theater subwoofer. The most critical frequency range is below 80Hz where substantial boost is required to extend response down to 20Hz. To facilitate the compensator design a composite response curve was used where the near-field response above 150Hz was merged with the MIB response below 150Hz again using CLIO's merge post-processing. The result is shown in Fig 11.

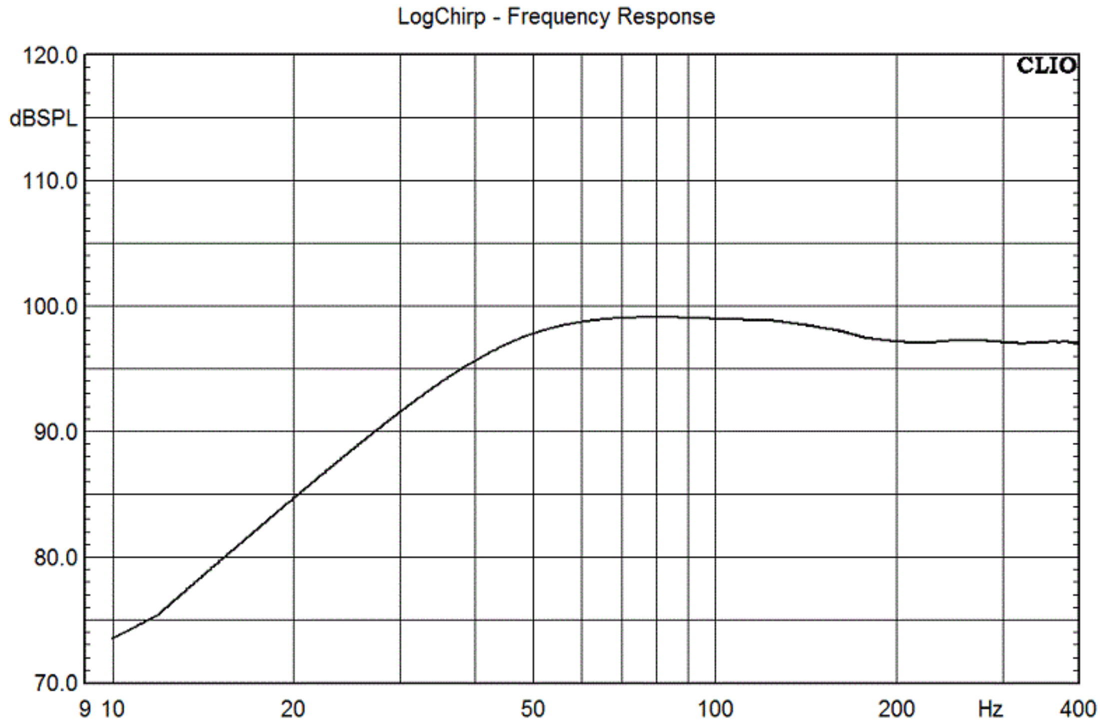

Fig 11 Subwoofer near-field and MIB responses merged at 150Hz

Crosstalk in the Near-Field Approach

Earlier we noted that if radiating surfaces are close together your measurements may be contaminated by crosstalk. In a typical Keele procedure the port output is maximum at the woofer dip. Because of its much higher level, even a small amount of crosstalk in the measurements from the port will contaminate the woofer data at and near this frequency. Considering the earlier two-way example and referring to Fig 4A, the raw port output (green curve) is about 25db above the woofer output at f_B. Assume the port output is added to the woofer near-field measurement at a level 20dB down. The resulting near-field woofer measurement is shown by the green curve in Fig. 12 along with the original uncontaminated measurement (red).

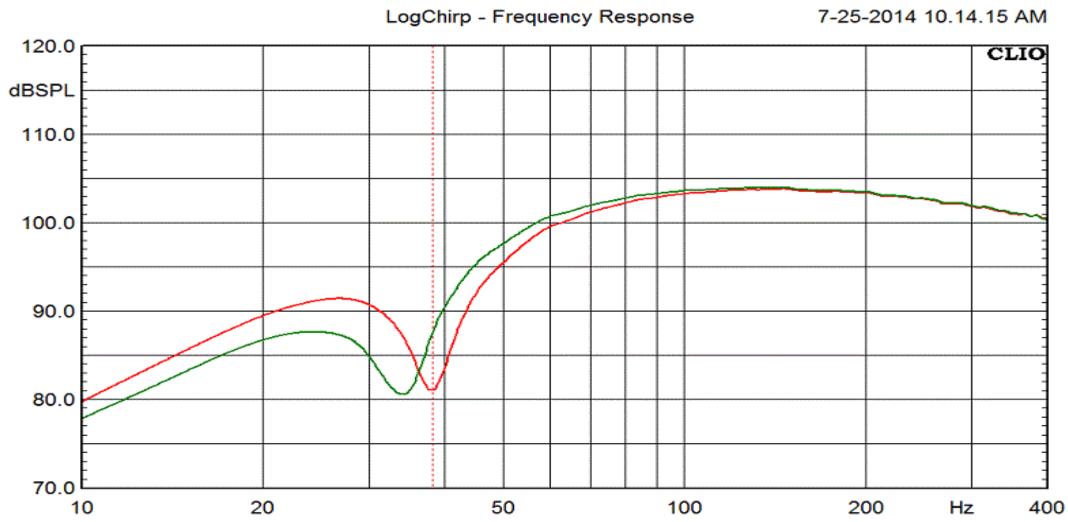

Fig 12 Showing the Effect of Port Contamination on Near-Field Woofer Data

In The woofer minimum has been shifted down by 3.8Hz. Notice that the woofer level is increased above f_B and decreased below f_B by the port contamination.

That this happens in practice is shown by the following example. Consider another small two-way ported speaker with a 110mm woofer and a 50mm port mounted on the front baffle. The woofer-port center-to-center spacing is only 10cm. Measured near-field woofer and port responses along with their combined response are plotted in Fig 13.

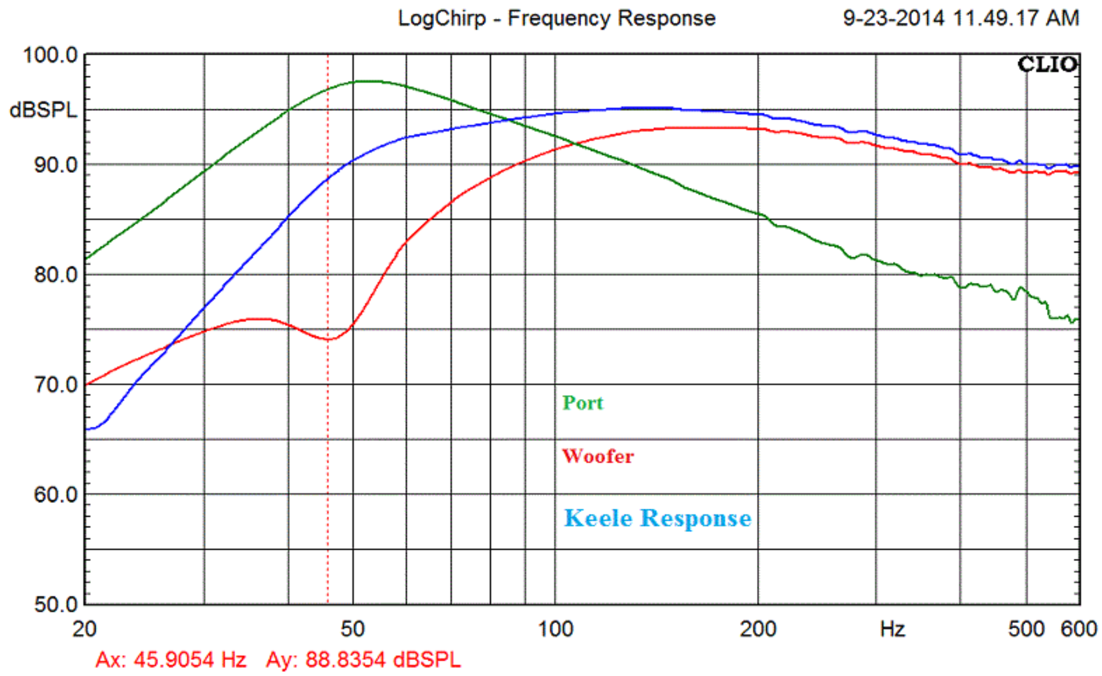

Fig 13 Small Two-Way Near-field Responses

The results look reasonable and in particular, the woofer response indicates an f_B of 45.9Hz. However, if we look at the impedance data for this speaker as shown in Fig 14, the minimum impedance at the saddle point indicates an f_B of 54.8Hz.

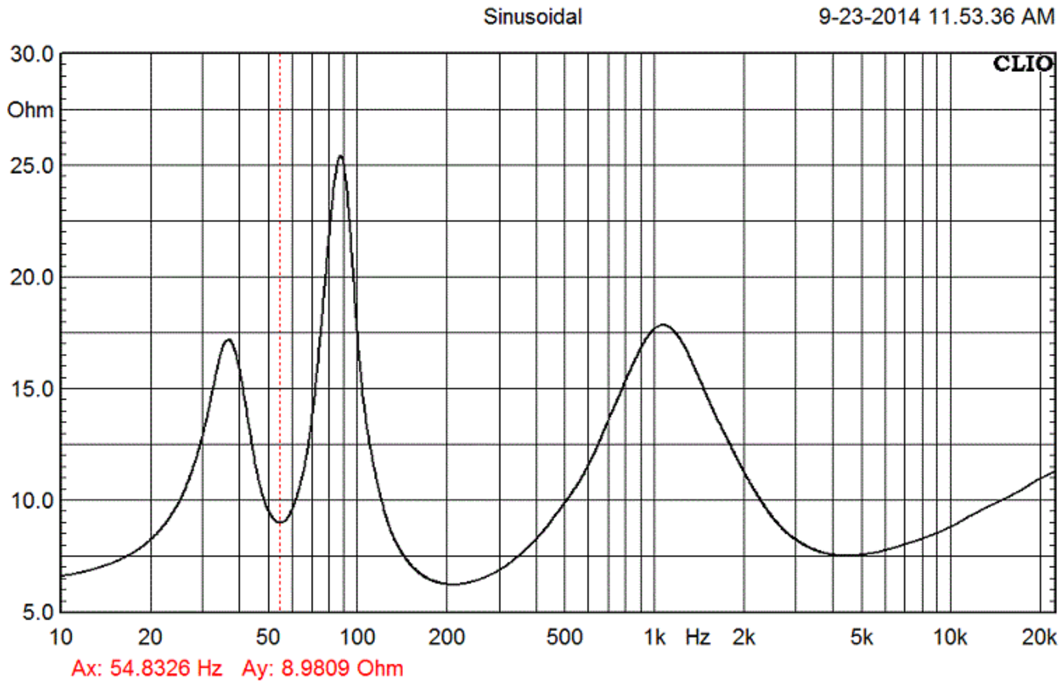

Fig 14 Small Two-Way Impedance Plot

In this example the presence woofer-port crosstalk leads to an incorrect prediction of f_B which is 8.9Hz below the actual f_B. Generally f_B predicted by the impedance plot and the woofer near-field response agree quite well. However, a large difference between the two is a good indication of crosstalk in the near-field data. In this case the impedance minimum is a better estimate of F_B.

Perhaps surprising even with crosstalk in the data, the computed low-frequency response is not badly distorted. Fig 15 compares the Keele response with the MIB response for this example.

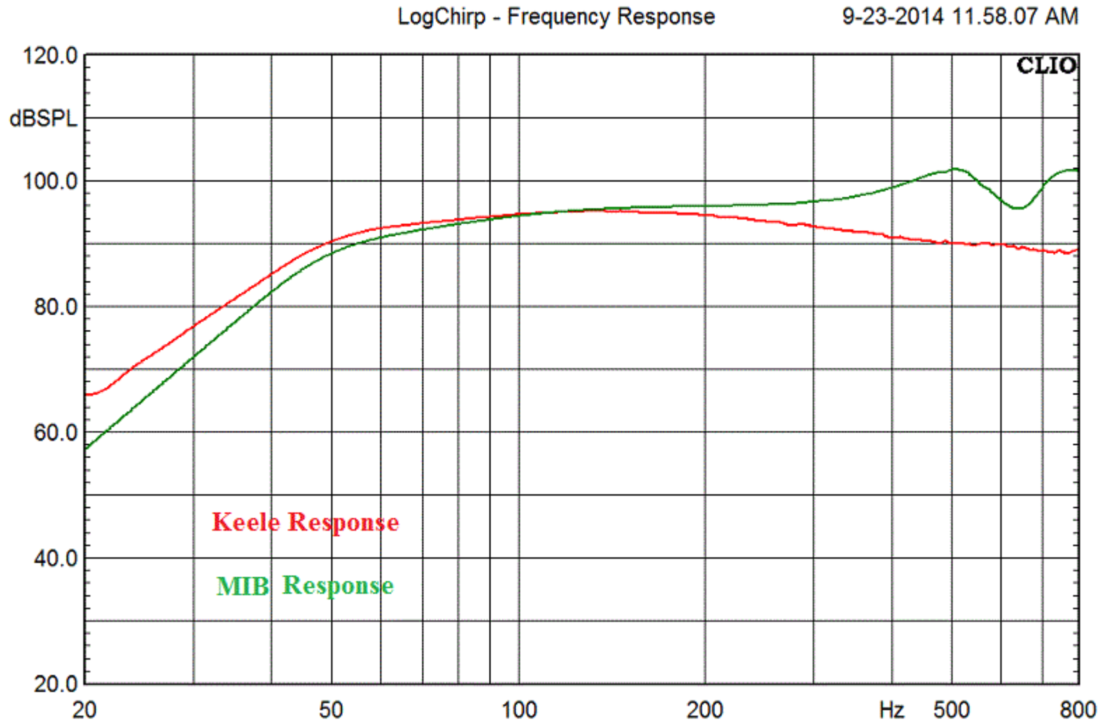

Fig 15 Comparing Keele and MIB woofer responses

The plots have been level shifted to meet at 150Hz. Below 40Hz the MIB response falls at exactly 24dB/oct below 40Hz while the Keele response falls at 22dB/oct. The MIB response appears suspect beyond 200Hz and clearly breaks up at 500Hz. This is typical of what we have seen before. The MIB response appears more accurate below f_B while the Keele response is to be preferred above f_B.

Multiple Radiating Surfaces

There is strong evidence based on extensive listening tests that listeners most prefer loudspeakers with flat on-axis frequency response for stereo and home theater use. (See Ref. [5]). However, in the home listening environment most loudspeakers become omni directional at low frequencies. That is their radiation pattern transitions from half-space to full-space as the baffle width becomes small relative to the radiated frequency. This causes the on-axis SPL may fall by as much as 6dB. This process is sometimes referred to as "spreading loss". The transition typically begins at a frequency corresponding to a baffle width of one-half wave length.

Our final example involves a small tower speaker with a single 25mm tweeter and two 170mm woofers placed vertically on the front baffle. A 75mm port faces to the rear. The tweeter is stacked above the two woofers in a vertical line. The upper woofer operates full range up to the crossover frequency of 2.2kHz. Via it's crossover network, the lower woofer frequency response is shaped to compliment the spreading loss to produce flat on-axis response. For this example the spreading loss begins slowly at about 700Hz.

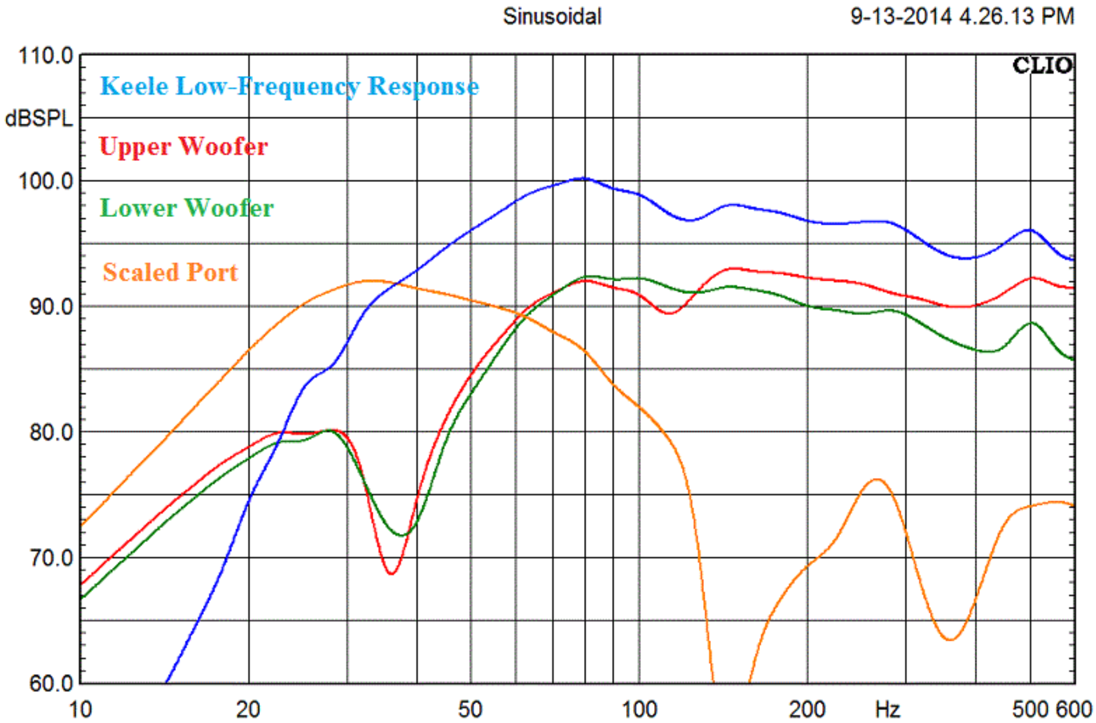

Fig 16 Tower Speaker Low-Frequency Measurements and Keele Response

Fig 16 shows the frequency responses of the scaled port (orange), upper and lower woofers (red and green) and the summed low-frequency response (blue). The effective diameter of the woofers is 130mm requiring a level shifting factor of

$$20\log(75/130) = -4.78dB$$

Notice that the measurements in this example were made using CLIO's Sinusoidal Analysis mode. The low-frequency data was next merged with the far-field measurement at 500Hz to obtain the full range response shown in Fig 17.

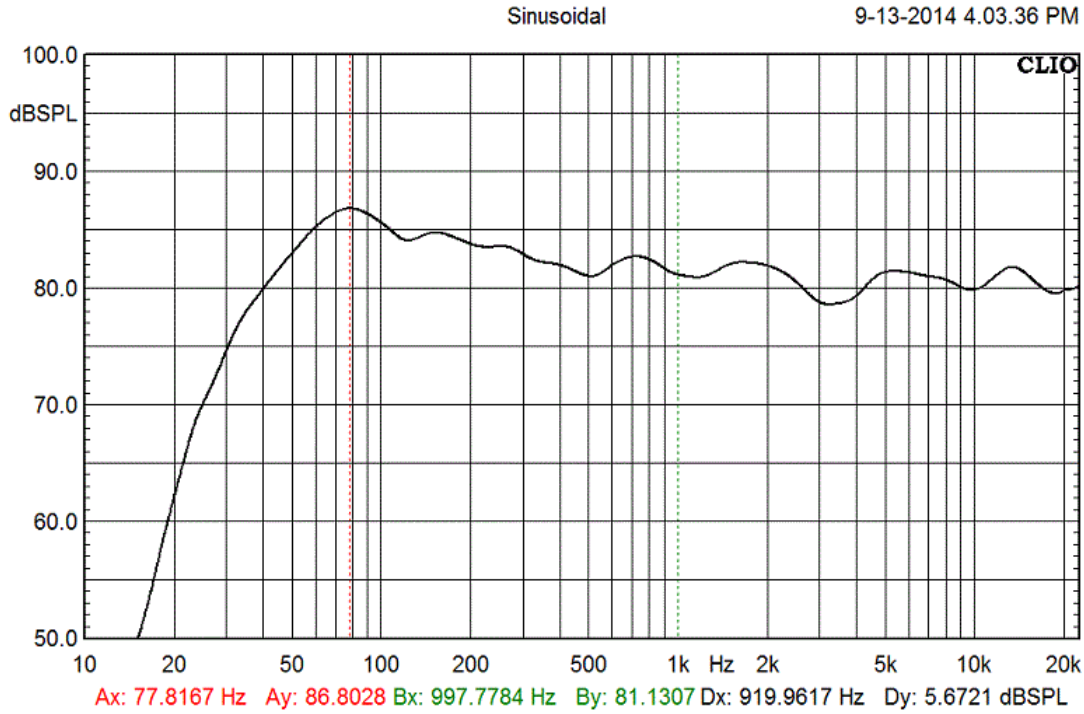

Fig 17 Tower Speaker Full range response via Keele Approach

Relative to 1kHz the response rises gently, to a maximum of 5.67dB at 77.8Hz. Remembering that the Keele approach gives us the half-space response it appears that the small tower is properly tuned to give flat on-axis response in a typical home environment.

Unfortunately, the MIB response is of no use in this example as the internal height of the tower has a corresponding one-eighth wavelength frequency of 44Hz. The MIB response (not shown) is clearly unreliable above this frequency.

Summary

In both the of the two-way examples and the subwoofer example we see that the near-field technique shows more variability then the MIB approach for frequencies below f_B. In addition to the reasons already given signal level may be another factor. For a fixed drive level, the acoustic pressure inside an enclosure is substantially higher than the pressure at any radiating surface. Differences between in-box pressure and pressures at the radiating surfaces were typically on the order of 20dB. Both MIB and near-field examples were obtained at the same drive level of 2.83 volts. This was done to eliminate any nonlinear effects that could corrupt the results. This is particularly true for the ported example where simple ports can be exhibit nonlinearity at relatively low drive levels. Our final example shows the impact enclosure dimensions have limiting the frequency range of the MIB technique.

Conclusion

This article presents the Keele and Small procedures for measuring loudspeaker low-frequency response and discusses some practical issues that arise with their use. Within the limits of their applicability, both techniques produce reasonable *estimates* of loudspeaker low-frequency response in the absence of an anechoic chamber. Both techniques agree reasonably well given their different approaches to the problem.

On the plus side, the near-field approach tends to be valid over a wider frequency range. Except for the single woofer case or the sealed-box subwoofer example given above, phase information is required to properly add the individual radiating surface responses. The possibility of error increases with multiple radiating surfaces and measurement crosstalk can be a problem. Also, response error appears to increase below f_B whether sealed or vented. Increasing the drive level for better SNR can help in this case, but be careful to avoid port nonlinearity.

The big advantage of the MIB technique is that it requires only a single measurement of in-box pressure to get speaker response regardless of the number of radiating surfaces. As a result phase data is not needed. Generally, the valid frequency range for the MIB measurement is smaller then that of the near-field approach. Based on the examples given above, MIB accuracy appears to be better than near-field below f_B. The microphone should be placed near the geometric center of the enclosure away from walls and interior baffles. With sealed-box systems getting the microphone into the enclosure may present a problem.

Measurement Equipment

The following test equipment was used to develop the data presented in this article:

CLIO electrical and acoustical measurement system using version 10.6 software
B&K 4191 ½″ laboratory grade condenser microphone
B&K Type 2669 microphone preamp
Listen Sound Connect microphone power supply and amplifier
B & K Microphone calibrator
CLIO QC Box Model 4 power amp for speaker testing

References

[1] D. B. Keele, "Low-Frequency Loudspeaker Assessment by Near-Field Sound Pressure Measurement", J. Audio Engineering Society, Vol. 22, pp. 154-162 (April 1974)

[2] J.A. D'Appolito, *Testing Loudspeakers,* (Chapt. 4), Audio Amateur Press, 1998

[3] J.E. Benson, "Theory and Design of Loudspeaker Enclosures", Proc. IREE (Australia), vol. 30, p. 261 (Sept 1969)

[4] R.H. Small, "Simplified Loudspeaker Measurements at low Frequencies", J. Audio Engineering Society, Vol. 20, pp. 28-33, (Jan/Feb 1972)

[5] Toole, Floyd E., "Audio-Science in the Service of Art", available at http://www.harman.com/about_harman/technology_leadership.aspx

Bio

Dr. D'Appolito has been an independent consultant in audio and acoustics for 22 years. He is a long time contributor to AudioXpress and its predecessor, Speaker Builder. He heads his own firm, D'Appolito Laboratories, Ltd., specializing in the design, test and evaluation of loudspeaker systems for two-channel and home theater applications. He also served as Chief Engineer for Snell Acoustics from 2003 to 2010. In that position he designed or led the design of some 80 loudspeaker systems for retail and custom home installation. He is the author of *Testing Loudspeakers,* the acknowledged bible on the subject, which has been translated into four languages, including Chinese. Prior to his work in audio he led a group developing advanced non-linear signal processing algorithms for passive sonar under government contract.