ON MEASUREING LOUDSPEAKERS

BACKGROUND

For the most part loudspeakers in the audio marketplace are only evaluated with subjective tests. Historically the dominance of these tests made sense because acoustic measurements are notoriously difficult to make and often show data that is far worse looking than what we think we hear. There has not been much confidence in these measurements so people relied on subjective assessments.

Measuring a loudspeaker in-situ (in a listening room), one gets a result that invariably looks far worse than what we sense subjectively. That is because of all the room reflections. The ear/brain system is sophisticated enough to hear through a great deal of this "hash", but alas it cannot hear through all of it. The problem is that it is impossible to sort out what is a problem from what is benign by looking at a normal listening room response. At low frequencies a steady state room response (one that contains all the reflections) makes sense, because that is in essence what we actually do hear. But above the modal region, this is not the case and if we want to assess the loudspeaker response and not that of the room, then we need to look at what is called its "free field" response, i.e. the response of the loudspeaker sans room reflections.

Years ago measuring the free field response required an anechoic chamber and lots of expensive equipment. Today one can make state-of-the-art measurements in a good sized living room with only a few dollars' worth of test equipment (I do this myself.) I say a "few" dollars because one can do this for less than say $US100, which, compared to the old days, **is** a <u>few</u> dollars.

The purpose of this paper is to define how one can setup to make these measurements and how to get the analysis done.

In this paper I will describe a proprietary piece of software that I have developed that does the data analysis. At this point in time this software is not available publicly, although it may be someday, I have not decided (I would need some more IP protections in place before this could be done.) I have a long standing offer to run this analysis for anyone as long as the resulting data goes into my database of various systems. This data is now about 30 systems and growing. I would like to see it as a repository of objective data so that the audio world can begin to move away from using the subjective as the standard of evaluation to an objective one. I believe that only in this way can the discipline of loudspeaker design move forward. As long as the standard for loudspeaker performance is subjective there will always be a serious limitation to progress in their design. Two people will seldom agree on the subjective aspects, but we should all be able to agree on objective ones.

Why not standardize on a commercially available package that everyone has access to? There are two reasons not to do this. The first is that we should not put some company in a preferable market position simply because we standardized on their software. I do not do commercial software and never will so (again, I did in the past), hence I have no commercial interest in the software being used, but others might. The second is that I wrote this software precisely because I could not find a commercial system that did a good enough job for me. I still don't see it in the marketplace, although it is getting better. The results that I get and show are far more detailed than what I see others showing using other software. This high level of detail is critical, especially when one is looking to compare two very good designs – *the devil is in the details*.

POLAR MAPS

For those unfamiliar with a polar map, you can see them at
http://gedlee.azurewebsites.net/Application%20Files/RunPolarMap.aspx

These plots contain an enormous amount of data. The "map" at the bottom shows SPL as a color gradient going from white (maximum SPL) to black (minimum SPL) through the colors yellow and blue. Red is used to indicate SPLs > 6 dB above the mean. This color choice shows uses the intensity of the color as directly related to the intensity of the sound at that point. The polar map shows frequency along the bottom and angle along the vertical. In the polar map there are 200 point frequency responses at 45 different off axis locations (the plots as they stand right now are symmetric about $\theta = 0$, but this could, and should evolve.) in the horizontal plane. Any location on this map can be observed in the other two plots with the drag of the mouse. Shown in the top graph is the frequency response at that angle in black, the power response in red (top plot) and the Directivity Index (DI) in White. The top plot changes to track the angle shown in the bottom map as does the side plot. The DI and frequency response depend on the angle, but the power does not.

The data in these plots is smoothed using an algorithm based on Brian Moore's concept of critical bands. The smoothing is greater at low frequencies, about 1/3 octave to fairly narrow at high frequencies, about 1/20 octave. At 1 kHz it is about 1/6 octave wide. The angular data is interpolated in a very unique way to yield an extremely high resolution display of the polar response data of a loudspeaker, which has about the same resolution in the angular axis as it does in the frequency axis. Normally, the frequency axis is highly resolved, but the angular one is poorly resolved and in some cases not even accurate.

It is well known that polar data cannot be interpolated between angles because nulls tend to move location in frequency as the angle is changed. Interpolating this effect simply smears it out of existence. The software that will be used here is free from this defect.

THE SOFTWARE

What is it that makes my software so unique? I will now give a brief overview of its features.

Some 12 years ago I became convinced that the polar response of a loudspeaker was an essential part of its design. I am firmly of this believe today and in fact believe that the polar response is the single most important aspect of any design. Remember that the classic "axial frequency response" is contained in a polar map, but a polar map contains so much more.

In a measurement system, doing angular data is the most difficult because this involves some form of rotation or a whole lot of microphones. There are automated systems for this, but that is not in the "low cost" option. Taking polar data is not difficult, but it can be very tedious, hence one wants to minimize the angular data points. There is always more than enough frequency data points, so this is never an issue.

These days an anechoic chamber is not required because we can use what is called "windowing". This means that we simply "zero out" the data once the reflections start to occur, which yields a very good resolution of the higher frequencies. Unfortunately, by doing this the lower frequencies get "spoiled' by the process. We will see how my new approach has a very elegant way around this problem.

What I came to realize was that for any loudspeaker, there is a spherical surface could surround the loudspeaker through which the sound would have to pass and that I could always represent the radiated far field response by some set of velocities on this fictional surface. A technique very similar to this was developed by Gabriel Weinriech at the University of Michigan way back in the 50's (and published in JASA), so it is not completely new, however I have modified it in many distinct ways.

It is know that the sound radiated from a sphere is composed of radiation modes. The two modes that everyone is familiar with are the *monopole* and the *dipole*, but these are simply the first two of the infinite number of modes that can exist to represent the complete sound radiation to ever higher frequencies. These modes, like all modes in acoustics, have a cut-off aspect to them. This means that if we are looking at a band limited signal, then one needs to only look at a finite number of the modes. This number is a variable in my system, but seldom requires more than 18-20. The bigger the speaker the more modes that are required. The lower the frequency the less modes that are required.

The first limitation in the software comes about at this point. I am doing this analysis only in two dimensions, not a full three (actually on a spherical surface, it is one dimension, lateral angle, instead of two angles, the lateral and vertical.) It is possible to expand the analysis to its full extent, but at this point I do not consider it necessary - desirable and maybe interesting, perhaps, but a luxury that I can easily live without. One can always look at the horizontal and vertical planes as two different measurements with the device turned 90°, but this is not exactly correct. To develop a full analysis instead of just one or two separate ones would add a couple orders of magnitude complexity to the system. To me the horizontal is by far the more important. I occasionally look at the vertical, but not that often as I don't find anything surprising or interesting. (Except in cases where I might suspect bad things to happen based on the design and then I usually find them - i.e. across the diagonal of a rectangular waveguide.)

I take data on a fairly sparse grid, but then by expanding each frequency in its modal representation, I can get a radiation model that has infinite resolution in the angular sense (up to some frequency, which can be found by inspection.) It can be shown that the highest sensitivity of a forward facing loudspeaker is directly near the central axis. Hence I take a fine grid at small angles and a coarse grid elsewhere. The angles that I use are 0, 5, 10, 15, 20, 30, 40, 50, 60, 80, 100, 120, 150, 180 – 14 angles in total. The angular data is then fit to the angular shape of each mode using a similar expansion to the Fourier series (in fact the expansion for a square piston in a baffle actually is the Fourier series, for a circular piston in a baffle it is a Bessel series, for the spherical case it is the Legendre series and the full analysis would use Spherical Harmonics – see Morse and Ingard as the only Acoustics source that I know of for these later functions.) For a Fourier series one would want equal angular spacing, but for the Legendre series one wants a finer spacing near zero.

I take data using HolmImpulse. I cannot recommend this software more highly because it is ideal for what I do. I take data to find the impulse response at each angle using a handmade turntable (more on that later). I use the Log sweep capability in Holm which creates a whole array of interesting data, like the impulse responses of each harmonic, etc. The measurements need to be "in-synch" time wise and this is done by running the impulse on the axial data and then locking the *time alignment* position for the rest of the data points. My software are fully complex number calculations, so phase becomes critical and only a locked time alignment will maintain the correct phase between the angular data sets.

Once I have taken all fourteen angular data sets, I export them using *File/Export all Measurements (Dialog)*. Set the "*from*" point to 100 or 200 and the "*to*" point to 4800 or more. You need to export at least 5000 data points. *Include Sample numbers* and *header information*. Use the ";" *separator* and the"dot" *Decimal separator*. Export the data to a .txt file. It must look like the sample posted on my web site.

Once this data file has been created it should be sent to me (for now) for analysis.

In my software, the data file is read, windowed, and each angle then has it frequency response calculated on the windowed data. The FFT points are first interpolated into a log frequency base of 300 points using a spline interpolation routine. There is no loss of resolution at this step. With an array of 300 log frequency points by 14 angle, I interpolate each frequency into its radiation modes. I end up with an array of 300 frequency points by m modal coefficients. Actually this matrix is only about half full since at LFs only the first few modes will contribute. The higher modes are not efficient enough to radiate. As the frequency goes up more and more modes need to be included.

It turns out that the power of the source is equal to the sum of the squares of the modal coefficients (see Morse, *Vibration and sound*), which is very convenient. With these coefficients I can now calculate the SPL response at any angle, so I do so at 45 positions from 0 to 90°. I also have the power response as well. Finally I can then calculate the DI at any angle as well. The DI is the power response normalized by the response along the particular axis. The higher the DI the more directional the source is. The true way to see Constant Directivity (CD) is that the DI curve is flat. If the DI is not flat along some axis then the system is not CD along that axis. An interesting and incredibly important aspect of the DI curve is that it is not correctable by any electronic means. It is a physical acoustic property of the system which cannot be corrected without changing the acoustical design. This is why I pay so much attention to the DI curve because if this is bad then it is time to start over, it cannot be corrected.

I strive for a high DI of 9 dB or more which is flat from several hundred Hz to about 10 kHz. Along the flat DI axis – this need not be the normal axis – the frequency response should also be near flat with perhaps a slight fall off towards 20 kHz. This means that the power response must also fall slightly towards 20 kHz.

The final advantage of my approach that I alluded to before has to do with the LF problem that occurs from windowing. When one windows the data it makes the LF response below some frequency invalid. This needs to be corrected if a good view of the frequency response in LFs is desired. The modal approach offers a unique opportunity to do this LF extrapolation. This is because at very LFs a piston in a sphere, which is the model that the modal approach uses, is dominated by only two terms, the monopole and the dipole. All higher order terms are insignificant because of “cut-off”. These two terms are well represented down to about 200 Hz or so in the data (typical for a small room) and do not need to be altered. By simply inputting a few parameters, such as *box volume*, woofer *piston radius*, *resonance frequency* and *Q*, the two lowest terms can be accurately determined from a simple LF spherical model. The modeled terms and then sliced/merged together (across about 20 data points) with the measured terms for a very accurate representation of the source down to well below what could normally be done in a small room. Details such as porting or not are ignored as these are simply secondary effects on the LF response. (This could easily be changed in the future should it turn out to be an issue. I only use closed boxes so it is not an issue for me.)

These calculations are all done in the software and what is produced is the simplified set of modal coefficients (about 6000) that are a very high resolution representation of a massive set (about 70,000) of mostly superfluous data. The calculations take only about 30 seconds on an Intel I-7 system.

The *Polarmap* program on my web site then reads this data from the database that I have assembled.

THE TESTING.

What one needs to do these tests are first a microphone (i.e. a Behringer ECM8000), a mic preamp with phantom power supply (i.e. Behringer MIC800). I find that any PC soundcard works just fine. You just need to make sure that there is not a feedback of the output signals into the input, such as “listen through”, because this will cause problems with a double impulse. The default condition on most sound cards usually does not do this, but sometimes it does.

You need an amplifier – not critical. Connect the PC sound card output to the amp, the amp to the speaker. Take the microphone into the preamp and its output into the sound card input. Run a test and you should see a nice clean impulse in Holm. You should hear the chirp from the speakers as well. It should not be too loud nor too soft.

The next step takes some fooling around initially but once you find the answer then it doesn’t change. You need to adjust the preamp gain so that the mic does not clip with the signal level from the amp (which is set in either Holm or with the amps gain control) but does cause some swing and the *signal* light to light. The signal in Holm (*Peak PCM* – bottom left) should be at about -3 dB on the conclusion of the test sweep. It can take some fidgeting of all these gains to get that just right. There is no unique set of gains here either so it is best to be on the slightly loud side to maximize the SNR in the test. But be careful not to overload the speaker either. HolmImpulse can show this by looking at the THD plot. It should not be excessive or there is a problem somewhere.

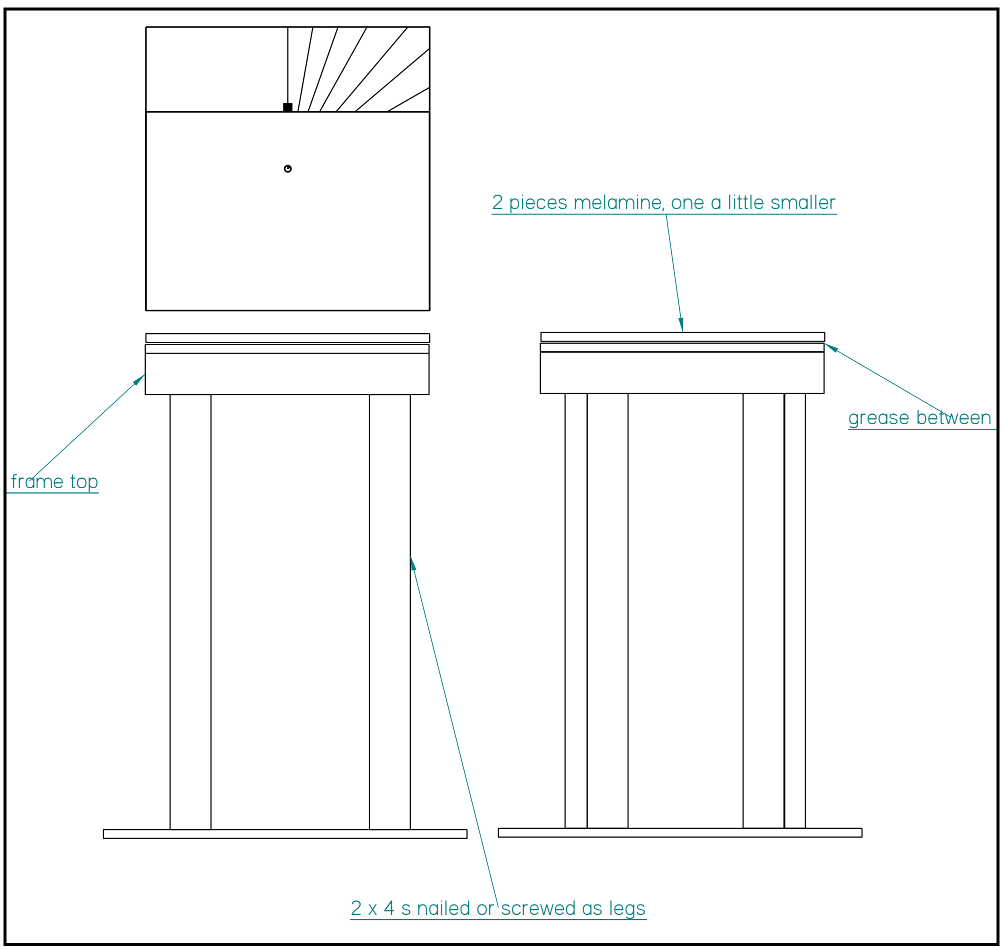
Once you have a nice clean axial impulse response (See example on my site), then you simply lock the time alignment (don’t forget to do this! Been there, done that!), rotate the speaker on the test stand to the next angular position and retest accumulating the datasets into Holms *Measurements* array. While not essential, the curves should be labeled as “00”, “05”, “10”, etc. I always use a mic distance of 1.5 meters, but any distance is allowed. It is a good practice to put this distance into the header which you will have to do after the file is exported in a Text processor like *Notepad*. Try not to go below 1 meter as this might be problematic.

THE TEST STAND

You must use a test stand, otherwise the data won't be any good for my software. The idea of the test stand is to place the speaker as far away from the room boundaries as possible. This is the free-field situation. It is possible to test in an infinite baffle and this is the classic method, but it is extremely cumbersome to do this in a small room. That and the results will be different and harder to analyze. At any rate this type of condition is not supported in the software at this time.

The test stands height should be designed such that it places the speakers center at the center of the room vertically. (Sources must be in enclosures as this is what the model assumes.) This will yield the greatest time before the first reflections from the floor and the ceiling – usually the soonest.

The test stand should be made sturdy with two tops pieces as shown below. I use ¾" melamine and place the two high gloss surfaces together. They are both attached to a common pin. I use a ½" piece of brass rod, but anything cylindrical will do here. A larger shaft will last longer and stay more secure. In my system the larger top plate is 24" x 24". The smaller is 22" x 18". These sizes are not critical – my speakers tend to be on the large size.

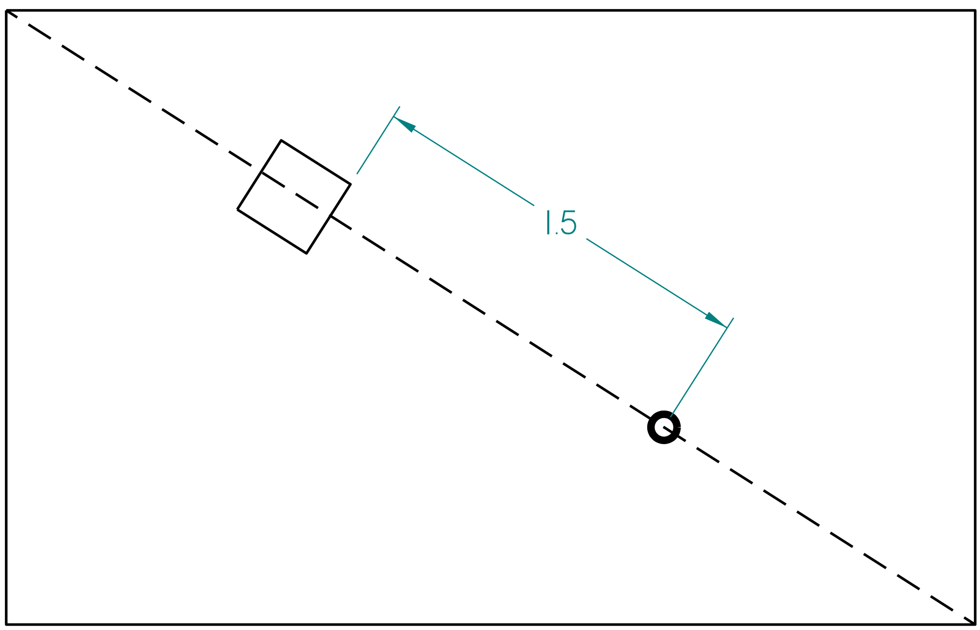


Draw lines from the rotation point out to the edge on ½ of the bottom plate (or you can do it all the way around.) It is best just to mark those angles that are actually used. I only mark a minimum of 10° because below this is hard to draw. Just measure between the 10° markings to get the 5° ones. A reference marker (I use a nail! Real high tech.) is placed in the center of the top plate such that when this plate is rotated it lines up with the lines drawn on the lower plate. With the two opposing melamine surfaces lightly greased with a heavy grease, the top plate will rotate very smoothly on the lower plate even with a 100 lb. speaker placed on it.

I attached two ½" plywood plates to two side surfaces of the stand to add rigidity, but my speakers can get very heavy. In most cases this will not be required. Some Liquid Nails for Subfloors during assembly will help to dampen the stand.

You now have your test stand.

Place the mic and test stand roughly as shown in the figure below. This placement will maximize the time of arrival of the first side wall reflections, which, in general, will not be the biggest problem. The floor and ceiling will.



CONCLUSION

I think that pretty much highlights all that is required. In the end it is really quite simple, except that the test stand can take up a lot of room to store if it is to stay in one piece.

If It appears that I have left anything out please notify me through my website.

I am hoping that we can begin to establish a database of very high resolution polar maps so that we can all talk from objective data instead of from weak unreliable subjective tests. As Floyd Tool often says; "There is more information on a tire about its performance than an audiophile gets about their speakers." This is oh, so true. Let's change that.

Is the polar map the end all of loudspeaker performance, no. But I believe, as others in this field do, that is a good 90-90% of the criteria important for defining a loudspeakers performance. Spending much time on the other things is not really warranted until we have a good established basis in Polar Maps.