# The Murphy Corner-Line-Array

## An Open Loudspeaker Design Project

by John L. Murphy
Physicist/Audio Engineer
TrueAudio.com

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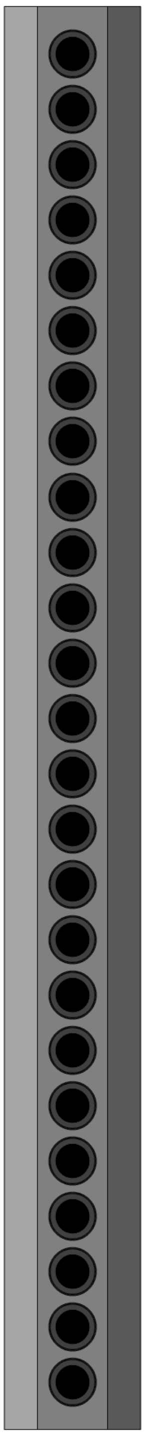
# 1. Project Overview



The MCLA

The Murphy Corner-Line-Array (MCLA) is a new type of line array loudspeaker system that I have designed to meet my own requirements for reference monitor loudspeakers with sufficient output to also serve as the sound reinforcement system for my band. I am pleased to make this novel loudspeaker design freely available as a DIY design for everyone who wishes to enjoy this new level of quality in sound reproduction. I hope to provide sufficient detail here for DIY speaker enthusiasts to reproduce the system for their own enjoyment and/or profit.

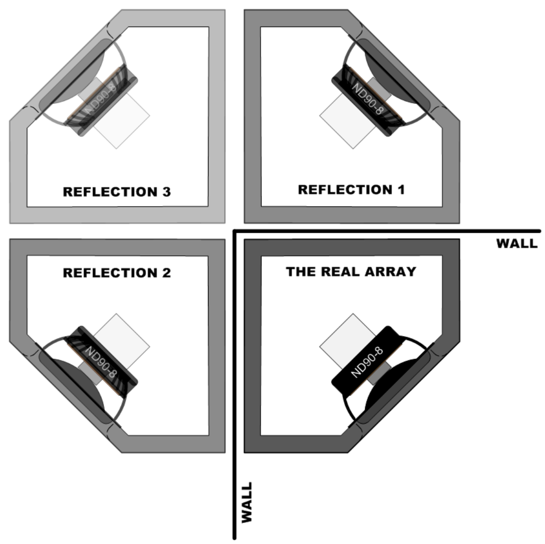



Figure 1.1: Top View of a Single Corner-Line-Array Along With Three Reflected Images

Having been fully disclosed in the public domain since 9July 2009, my "corner-line-array" invention cannot now be claimed or patented by any third party. As far as I am concerned, everyone is free to use it as they wish with no strings attached. I humbly request only that you openly acknowledge the origin and name of the design.

As an audio product designer for the past 30 years I am not easy to please when it comes to loudspeakers. I've heard them all and have definite opinions about what I want when it comes to sound reproduction. The role of these speakers will be to serve as reference monitors in both my home studio and my home theater. I recommend this system wherever a high performance sound playback system is required. My recommended applications include: two-channel home music playback systems, home theater (especially the front corners!) and small sound reinforcement systems for small rooms. **There is one basic requirement for the room:** The room must have about an 8 foot ceiling which is parallel to the floor with two corners available for placement of the line arrays. If your listening room conforms to this description then I expect the MCLA will perform in your room very much like it performs in my rooms. This should allow you to benefit from my testing and fine tuning as I voice the system.

Each array employs 25 identical full range drivers in a floor-to-ceiling enclosure specially designed to take full advantage of corner placement. After searching the available drivers I have selected the new Dayton Audio ND90-8. This is a 3.5" driver with solid aluminum cone and 4mm of linear excursion.

## A Little History

I first came to admire line array type loudspeakers around 1979 when I was still simulating enclosure responses on a hand-held calculator…point-by-point and plotting them by hand. Now, after having lived with a large (16 cu ft!) pair of line arrays over the last 30 years I have designed this new corner-line-array speaker system for use in both my home studio and my home theater. You are invited to review my design and my measured performance and perhaps you will decide to construct a pair for your own use.

My original line arrays (which I designed and built in 1980) were a two-way system employing eight 8" woofers and sixteen 1" soft dome tweeters. The crossover varied between active and passive types through the years but always hovered in the 1.5 kHz range. Butterworth, Linkwitz-Riley filter types…I've tried them all. Regardless of the crossover details I wanted to keep the crossover frequency as low as possible for optimum off-axis frequency response in the crossover range. By design, the incredibly high power handling capability of the tweeter array allows them to be safely crossed at a much lower frequency than if using only a single driver. Over the years these two line-array systems have performed marvelously and typically make a strong positive impression on all who have auditioned them. They have wonderfully low distortion even at the highest sound levels a body can tolerate. Many of the audio industry insiders who auditioned my line arrays back in the 1980's had never heard high performance line arrays before. Today I see concert line array loudspeakers offered by at least one company whose founders could have been seen leaving my place with a smile after hearing the MLA 8/16 system back in the 80's, long before concert line-arrays came into popular use.

From the start these two-way line arrays employed active (line level) equalization in various forms. Initially it took the form of a fixed equalizer which was designed based on in-room measurements of the actual system response. Eventually I settled on an off the shelf one-third-octave equalizer for flexibility in voicing the system's frequency response. The responses I keep handy in the EQ's memories include (echoic in-room) "flat", "X-curve" and "half X-curve". The flat response is reserved for very select recordings which have been recorded with flat microphones and mixed and mastered on flat audio monitoring systems. My typical setting is what I consider to be the norm for most recorded music and film audio: the X-curve. Now with my beloved "MLA 8/16" line-arrays approaching 30 years in age I have developed the MCLA full-range corner-line-array in a more compact enclosure to replace them as my master reference monitors.

## Design Overview

A: Line Array Source Geometry

> Based on 30 years of listening to my previous line array speakers the new system could only be another line array.

B: Crossover-Free Full-Range Operation:

> The array employs a high quality full-range transducer of small diameter but large excursion capability. The exact native frequency response of the driver is less important than the ability to be equalized to cover from 30 Hz to 20 kHz (-3 dB points). I first elected to use a 3 to 4 inch transducer based on my long term experience with a 4 inch full range speaker I enjoy in one room at my home. This speaker convinced me that there is no clearer midrange than that of a crossover-free full-range speaker. Splitting the frequency band into two (or more) parts and then recombining them via separate different sized speakers inevitably introduces audible coloration. The 2.5 inch piston of the full-range driver may have narrower high frequency dispersion than a typical 1 inch dome tweeter but the geometry of the corner-line-array actually prevents the listener from ever being more than 45 degrees off axis. Thus the requirement for wide dispersion is reduced compared to a speaker located in the room that might be heard at 90 or even 180 degrees off axis. The dispersion of the 2.5 inch piston is certainly wide enough considering that the MCLA listening angle is limited to 45 degrees. Ultimately, the very best crossover...is NO crossover. The power amplifier is connected directly to the full

range drivers with no passive components in-between.

C: Corner Loading:

By designing the system specifically for corner placement the side and front wall reflections are forced into close proximity to the real speaker. As expected from corner loading, the multiple images increase low frequency output capability because of the close proximity of the images to the real speaker. This standardized placement in the room eliminates the major source of variations in the frequency response as the listening position changes within the room. It also minimizes the difference from room to room. This standardized room placement actually gives everyone a good chance of achieving a system response very close to my own documented response by reproducing the line arrays exactly and then using my carefully constructed equalization settings.

D: Extended Bass to below 30 Hz (-3 dB)

The arrays are recommended as full range loudspeakers for both home audio and home theater applications. A subwoofer is not required. When used as a small room sound reinforcement system in live music applications the goal is to provide bass extension to 50 Hz. Restricting the bass extension for sound reinforcement applications will allow for an overall higher SPL level before any limit is reached. I deliberately elected a tradeoff of lower efficiency for a smaller sized enclosure that would provide bass response to around 100 Hz. This means that below 100 Hz the system has increased power requirements compared to a non-equalized system. Given the tremendous efficiency of the overall system however, my 100 Watt (at 8 Ohms) power amp seems adequate for my applications. So I am currently using a power amp that is rated far below the actual power handling of the system. (20 Watts/driver = 480 Watts per system). Even though they need EQ to get bass extension the arrays are still operating with a generous amount of headroom. This keeps distortion very low compared to other speakers which typically operate at much higher individual driver stress levels.

E: High SPL Capability for Sound Reinforcement Applications

I want the system to be capable of serving as the PA system for my band when we perform in my studio. For this application it is reasonable to limit the bass response to around 50 Hz (-3 dB) to reduce the excursion requirement and allow for the increased overall system SPL required for live music. With the ability to perform sound reinforcement for a live band, they are just loafing when playing recorded music at loud home playback levels. So far I have actually been using the 30 Hz mode without any signs of stress during our band rehearsals. I suspect that in a larger room the 50 Hz limit would be needed to avoid problems.

F: Active Line Level Equalization:

The system employs a line-level equalizer in order to achieve various target responses including (echoic in-room) “flat”, X-curve and Small Room X-curve. Equalization is currently achieved by an off the shelf 1/3 octave digital equalizer: the Behringer Ultra-Curve Pro DEQ2496. But I may also design a custom analog equalizer to replace the digital EQ and lower the total cost of the project. Stay tuned...

## Driver Selection

The performance of the full-range driver is critical for the project. The characteristics of the driver which are most important are upper frequency extension, distortion and excursion capability (both Xmax and maximum mechanical excursion). The driver has to be correctable to 20 kHz and must have enough linear volume displacement such that 24 drivers will provide adequate output to 30 Hz. Surprisingly, the F(s), Q(ts) and V(as) parameters, and even the box volume are not critical for this project as long as they are "reasonable" because the frequency response will be equalized to flat and the closed box design "transformed" to a new second order response with the desired Q and F3.

The driver I have selected for the project is Dayton Audio's new ND90-8. This is a high performance 3.5" driver with aluminum cone and 4mm of Xmax (peak). The driver is being custom manufactured to Dayton Audio's specifications by one of their OEM sources and this new design takes full advantage of Dayton's long and broad experience in driver design. Dayton currently has a large supply of ND90 drivers on hand and the supply should be ongoing as long as there is demand for the driver. I have followed the development of the driver and have been testing and auditioning samples beginning in March 2009.

As one of my listening tests I routinely used a single ND90 test box as the speaker for my electric guitar during personal practice sessions. Playing my Telecaster's single coil pickups through my Quad X guitar preamp followed by a power amp and ND90 at close range provides an intimate musical connection that is very enjoyable. My ear tells me that this is a very low distortion speaker when driven within its power handling range. Using just a single driver it is possible to explore how the driver sounds when it is overdriven without getting terribly loud. If I turn up my guitar (keeping it squeaky clean at the preamp) I can hear exactly how the ND90 sounds as it overdrives. I was pleased to hear that it distorts softly with no surprises and even though I definitely abused it I have never heard it bottom out on the bass as many drivers would do in this situation. This is likely due to the very generous maximum excursion capability of 10mm (peak). This allows a full 6mm of headroom beyond the 4mm (peak) linear excursion. It's nice to know the driver will overdrive gracefully when pushed to its limits even though the full array will never be pushed that hard in actual use. To drive the complete array into distortion would require a much larger power amp than I plan to use and would produce dangerously high sound levels. At the loudest sound levels I can tolerate the individual drivers will be loafing. That's the power advantage of an array.

The availability in recent years of small diameter drivers (3 or 4 inches) with low distortion and high excursion makes this kind of loudspeaker system feasible. While I will be standardizing the project on the ND90 note that other 3" to 3.5" drivers with adequate performance could be used to create a high performance corner-line-array. Upon surveying the small number of qualifying drivers I have elected to specify the Dayton ND90 for the project. Below are some photos of the driver:


Figure 1.2: The Dayton Audio ND90 Full Range Driver

## Variations on the Design

At this time I am documenting here one very specific implementation of the MCLA for the DIY loudspeaker community. But, in general, the corner-line-array design can be implemented with broad variations in components and scale. Certainly the array might be built with different full-range drivers or as a multi-way system with different types of transducers. The number of drivers used and the total length of the array could be varied to suit many different applications. Smaller clubs and theaters could certainly take advantage of corner-line-arrays.

Corner arrays for rooms with ceiling heights other than eight feet come to mind immediately. It would be simple to increase the number of drivers to span a ten foot height and then come up with a wiring arrangement to achieve a net impedance around 8 Ohms. The overall size of the enclosure could be increased or decreased to vary the native bass response or explore the performance of even smaller full-range drivers. While equalization can be used to achieve extended bass response from small drivers, I have made it a point to have a generous degree of headroom in the "25xND90" system detailed here but it is reasonable that others might explore implementations of arrays of smaller drivers of less output capability. Scaling the enclosure down would tighten up the spacing of the corner reflections further. Shorter arrays (perhaps 16 drivers) placed horizontally at the ceiling and floor could provide good center channel performance for home theaters. Enclosures could be designed for mid-wall placement to serve as home theater surround speakers. Thus it is possible to employ MCLAs for a complete home theater surround sound system.

While my prototype enclosures are implemented in wood the enclosure might also be made as a two piece aluminum or plastic extrusion. An extrusion would allow the enclosure to be shaped as a quarter-round with a flat area for mounting the driver. The effective shape of the enclosure plus its reflections could be made even closer to a perfect cylinder with edges more rounded than my "octagon".

Arrays of typical dynamic transducers, such as the ND90, will always require frequency response correction in order to neutralize the inherent 3 dB per octave slope (falling with increasing frequency) that is characteristic of long line arrays using many drivers. The equalizer used to perform the frequency response correction could be either digital or analog, and either custom or off-the-shelf.

# 2. Design Concepts

## Loudspeakers in Rooms

Loudspeakers always seem to be at odds with the room in which they must operate. In order to deal with this designers resort to all sorts of clever tricks. We go to great lengths to either test our speakers in a anechoic chamber or to take quasi-anechoic measurements with fancy test instruments all in an attempt to show what the speaker should sound like...if it weren't for that darn room. But like it or not, virtually all loudspeakers find themselves playing into a room and competing with their own reflections for the listeners attention. My new corner-line-array is an attempt to bring some peace in the war between loudspeakers and rooms. Instead of fighting the room or pretending it's not there, the corner-line-array design joins the loudspeaker with the room in such a way that the performance of the loudspeaker in the room is more predictable and repeatable than with previous point, line or planar type loudspeakers.

I am often asked "but what about comb filtering?" by those who know that combining multiple drivers can lead to frequency response irregularities. This is due to the fact that the sound is arriving at the listener by more than one path. To the extent that the path lengths are different the two sound sources will have different travel times and will arrive at the listener's ears at different times. When two sound sources of the same level but with different time delays are combined the effect is to create a comb filter response with deep notches closely spaced so as to resemble a comb. It seems reasonable to assume that this garish comb filter frequency response will spoil the sound of any loudspeaker. But in the real world, in real rooms even if we were to play just a single speaker (we never actually do this!) it would be heard by the listener by multiple paths, each path with a unique delay. This is why a typical in-room measured frequency response looks just as garish as the dreaded comb filter. A simple comb filter response might actually seem to be an improvement over the response of a typical listening room with its multiple pathways from speaker to listener. Comb filtering due to multiple signal paths is a given for in-room audio playback systems unless you listen in mono in an anechoic chamber. The MCLA design takes positive control of the room reflections in order to both standardize and optimize the complete loudspeaker-room system and thereby provide a more consistent frequency response than past approaches to in-room monitoring.

## Image Analysis

Image analysis is a powerful technique used in the field of acoustics to study reverberation and room reflections. In this analysis the walls of the room are considered to be mirrors and reflected images of sound sources can be readily located. Before proceeding let's review the image analysis method as it applies to loudspeakers in rooms.

The famous acoustician/physicist Carl F. Eyring commented on the image method in 1930 as follows:

"This necessary analysis is aided by the method of images. Just as a plane mirror produces an image of a source of light, so also will a reflecting wall with dimensions large as compared with the wave length of the sound wave produce the image of a source of sound. An image will be produced at each reflection. In a rectangular room, the source images will be discretely located through space. This infinity of image sources may replace the walls of the room, for they will produce an energy density at a point in the room just as if they were absent and the walls were present." [2-1]

Here Eyring is saying explicitly that the image sound sources can be substituted for the room and the resulting sound field will be the same. The assembly of reflected images is exactly equivalent to the effect of the room. A sound source in a room is exactly equivalent to that same sound source in free space along with an infinite array of sound sources representing the mirror image reflections of the single sound source in the room.

A rigorous mathematical foundation for the image method can be found in the 1978 paper by Allen and Berkley titled:

"Image method for efficiently simulating small-room acoustics" [2-2]

It is also worth mentioning that in 1957 the well known acoustician Richard V. Waterhouse also used the image method to analyze the effect of reflecting surfaces on sound sources. [2-3]

Floyd Toole in his recent (2008) book titled "Sound Reproduction, The Acoustics and Psychoacoustics of Loudspeakers in Rooms" [2-4] writes:

"Explanations of sound fields in concert halls begin with notions of ray (geometrical) acoustics, showing direct sound and discrete reflections from large surfaces. The rules are simple: the angle of incidence equals the angle of reflection." [2-4, p. 43]

The simple rule of equal angles of incidence and reflection is shared with the field of optics, a sister field of acoustics falling under the fundamental science we call physics. This shared rule is why acoustical reflections are located in exactly the same positions as the visual images of the sound source that we would see if the walls were mirrored.

In Figure 2-1 below we see an object reflected in a single mirror. If the object in front of the mirror were a sound source that sound source would have a reflected acoustic image at the same apparent location as the optical reflection. Placing the object against the mirror would result in the object and the image being back to back. The acoustic equivalent would be the difference between full-space and half-space loading (see Appendix 3, “Loudspeaker Spatial Loading” for more on acoustic loading). The acoustic image located close to the sound source as a result of a reflecting surface boosts the sound level by 6 dB exactly as if a second sound source were placed next to the original in free space. These are different but compatible views of the same observed phenomena. See Refs. [2-1], [2-2] and [2-3] below for a entry points into the scientific literature on the image method in acoustics.

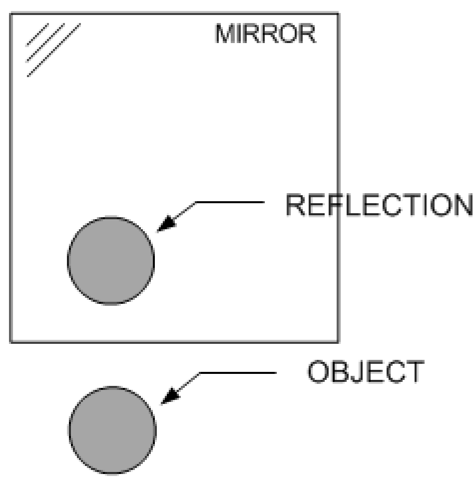



Figure 2-1: An Object Reflected in a Single Mirror

Now let’s consider what happens when viewing an object in two mirrors placed at 90 degrees to one another as in Figure 2-2 below.

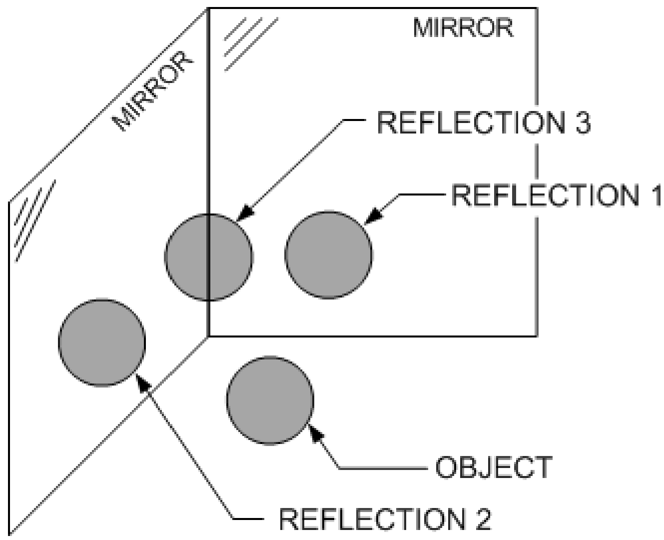


Figure 2-2: An Object Reflected in Two Mirrors

The object has picked up not just one, but two more reflections. Reflection 1 is the original reflection. Reflection 2 is a new reflection from the second mirror that corresponds to reflection 1 in the first mirror. Reflection 3 always falls in the corner and is the reflection of reflection 1 and reflection 2 as seen by the opposite mirror. This 3rd reflection is best understood by doing the experiment with two mirrors, and if you actually do the experiment you will see the original object along with 3 reflected images. As the object is moved nearer to the corner all 3 images move closer together to give the appearance of four tightly spaced objects. If the object is a sound source then it will also have three reflected images when placed close to an acoustically reflective surface (such as a wall of a room). This is what happens when a speaker is loaded by a quarter-space load as at the intersection of two walls of a room. Compared to the original speaker in free space the sound level is increased by 12 dB (6 dB for the first reflection and another 6 dB as the number of sound sources is doubled from 2 to 4). You can view this problem two different ways. First, we can view it as a speaker with different spatial loads. Alternately, and just as valid, we can view it as multiple speakers in free space. For the purposes of the present discussion I will continue to use the acoustic image model.

In Figure 2-3 below I show a simple example with two mirrors placed at right angles similar to two walls of a room. You can see the small hand mirror I placed at 90 degrees to the larger mirror. There is only one paper speaker mockup in the photo but you can clearly see all four speakers around the faces of the overall octagon formation. The acoustic images will appear in the same locations as seen here.

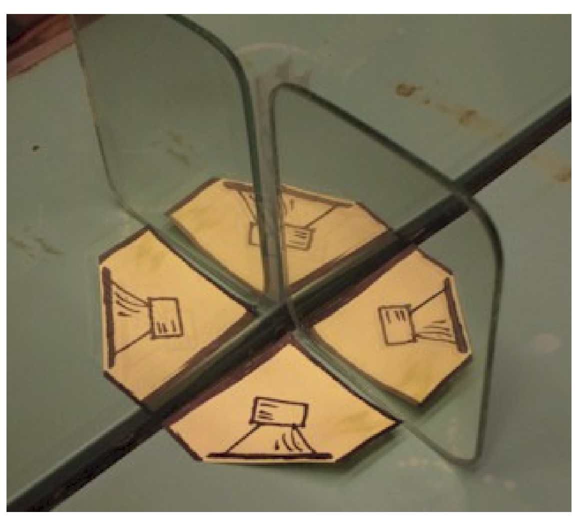


Figure 2-3: This Photograph Shows a Single Drawing of a Speaker Along With Three Reflections in Two Perpendicular Mirrors

## The View From The Top

Consider again the simple case of the object reflected in a single mirror. The view from above would be as seen in Figure 2-4 below.

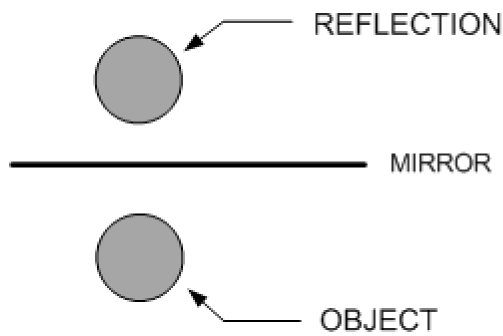


Figure 2-4: The Top View of an Object Reflected in a Single Mirror

Now, consider the top view for the two mirror case shown in Figure 2-5 below. In this figure I have shaded the reflections lighter to indicate that they are slightly weakened by the absorption of the walls...sort of like a dirty mirror. Because reflection 3 is a second order reflection (that is, a reflection of a reflection) it is shown even more dim than the first order reflections. It doesn't matter whether the reflected images are optical or acoustic as their positions are the same.

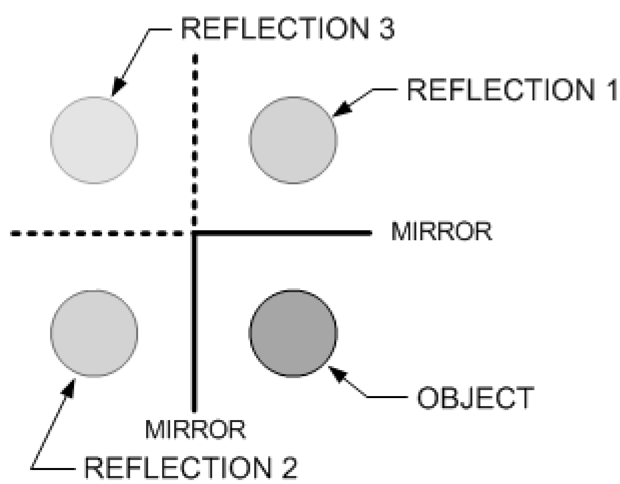


Figure 2-5: Top View for the Two Mirror Case

Notice that moving the object all the way into the corner results in the four images coming together in a tight pattern as seen in Figure 2-6 below.

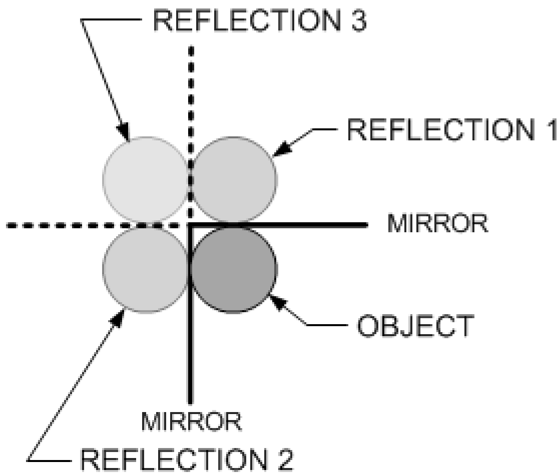


Figure 2-6: Top View with Object In the Corner

## Back to the Room

In a normal home listening room the sound reflected from the walls, floor and ceiling creates reflected sound images in the same way the optical images appear in the mirrors in the discussion above. "Ray Tracing" is a method that has long been used in the study of both optics and acoustics. Ray tracing allows us to precisely locate reflected images using a simple form of geometric analysis. Rays follow this rule for reflection: ***the angle of reflection is equal to the angle of incidence***.

Let's examine how a single reflected image is created in a listening room. In Figure 2-7 below we see "rays" of sound leaving a sound source and arriving at the listener by way of two paths. The ray A-B from sound source A to listener B follows the direct path. There is only one location on the side wall where a reflection occurs such that the incident and reflection angles are equal and that passes through the listening position B. That reflection path is through point C. The sound ray A-C propagates from source A to reflection point C and is then reflected from the wall as ray C-B from the reflection point to the listener. The presence of the wall creates a very real reflected image at D, just as if the wall were absent and a second sound source were added at that location. The ear is tricked into hearing the sound that arrives via ray path A-C-B just as if the path were straightened out and arrived via the phantom path D-B. Just as the eye sees the reflections in the mirrors in the previous discussion, so too does the ear hear sound appearing to arrive from the location of the reflected image behind the wall. From an energy point of view, we would say that the wall redirects sound energy that would have gone unheard toward the listener thereby creating a higher sound level at the listeners position than if the wall were absent.

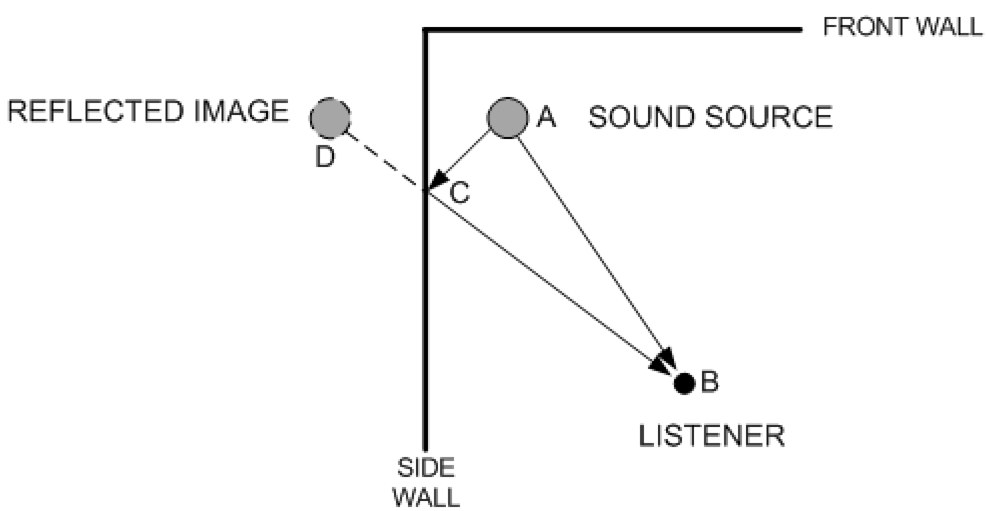


Figure 2-7: Sound Source with a Single Reflected Image

Once we see how a single reflected image is formed we might notice that in Figure 2-7 above there would be additional images from the front wall and corner reflections just as in the optical cases we saw in Figures 2-5 and 2-6. The optical and acoustical examples are wonderfully analogous cases where strict mathematical analogies can be applied. In Figure 2-8 we see the additional sound images E and F that in addition to image D would be created by the two walls shown. The added rays show the path from A to B via a precisely located reflection from the front wall to create image E. Rays to define the corner image are not shown but would go through the corner of the room. Once the room is enclosed with all six reflecting surfaces (four walls plus ceiling and floor) the number of reflections is increased to infinity giving rise to the sound of reverberation in the room. The finite absorption of the surfaces of the room attenuates the images as they repeat causing the reverberation to eventually decay to inaudibility. Once aware of the reflected images we can begin to take control of the "house of mirrors" game that inevitably ensues when we experiment with loudspeaker placement in rooms. My solution here is to merge the reflections with the original sound source by moving the sound source (line array) into the corner. Moving all the sound sources into a tight group increases the upper frequency limit for their coherent summation and keeps arrival times tightly clustered well enough for the multiple sources to fuse into a single perceived sound source. Moved out of the corner the array would create more widely separated images which would have a wider spread in arrival times and would be in danger of not fusing into a single perceived sound source.

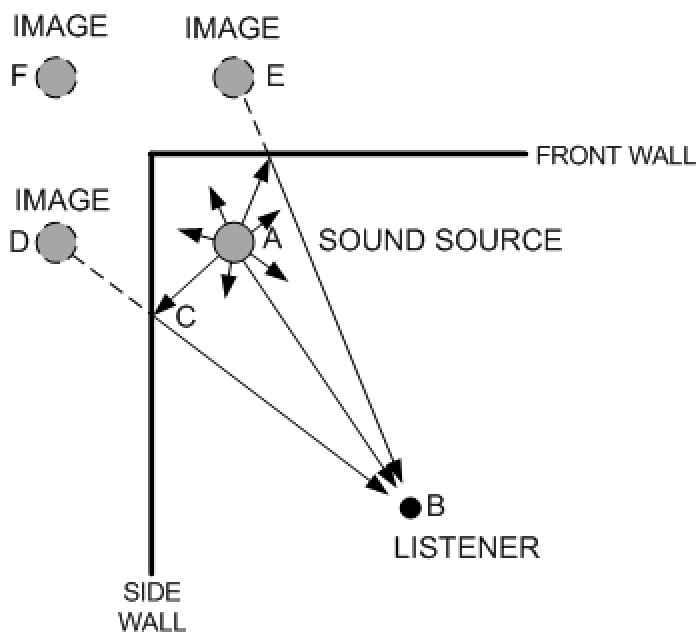



Figure 2-8: Sound Source with Multiple Reflected Images

Figure 2-9 shows a pair of point source radiators located against the front wall of the listening room along with their first few reflected acoustic images in the floor, ceiling and side walls. Only the first one or two reflections are shown but the array of images actually continues to infinity in all directions...just as regular optical mirrors would show if each wall were mirrored. Each successive reflection grows weaker due to the finite sound absorption of the walls. The listener hears the speakers directly along with all the reflected images. The delay of each image is determined by its path length to the listener. The shortest path to the listeners ears is the direct path from the speakers. Those reflections that follow within 20 -30 milliseconds of the direct sound more or less fuse together into a single perceived sound. Those reflections arriving after 30 milliseconds or so are heard as early reflections and reverberation. As you move away from the speakers and toward the rear of the room the direct sound from the speakers falls compared to the total combined energy of the reflections. Note that the SPL falloff rate for each image is the same as for the direct sound: 6 dB per doubling of distance.

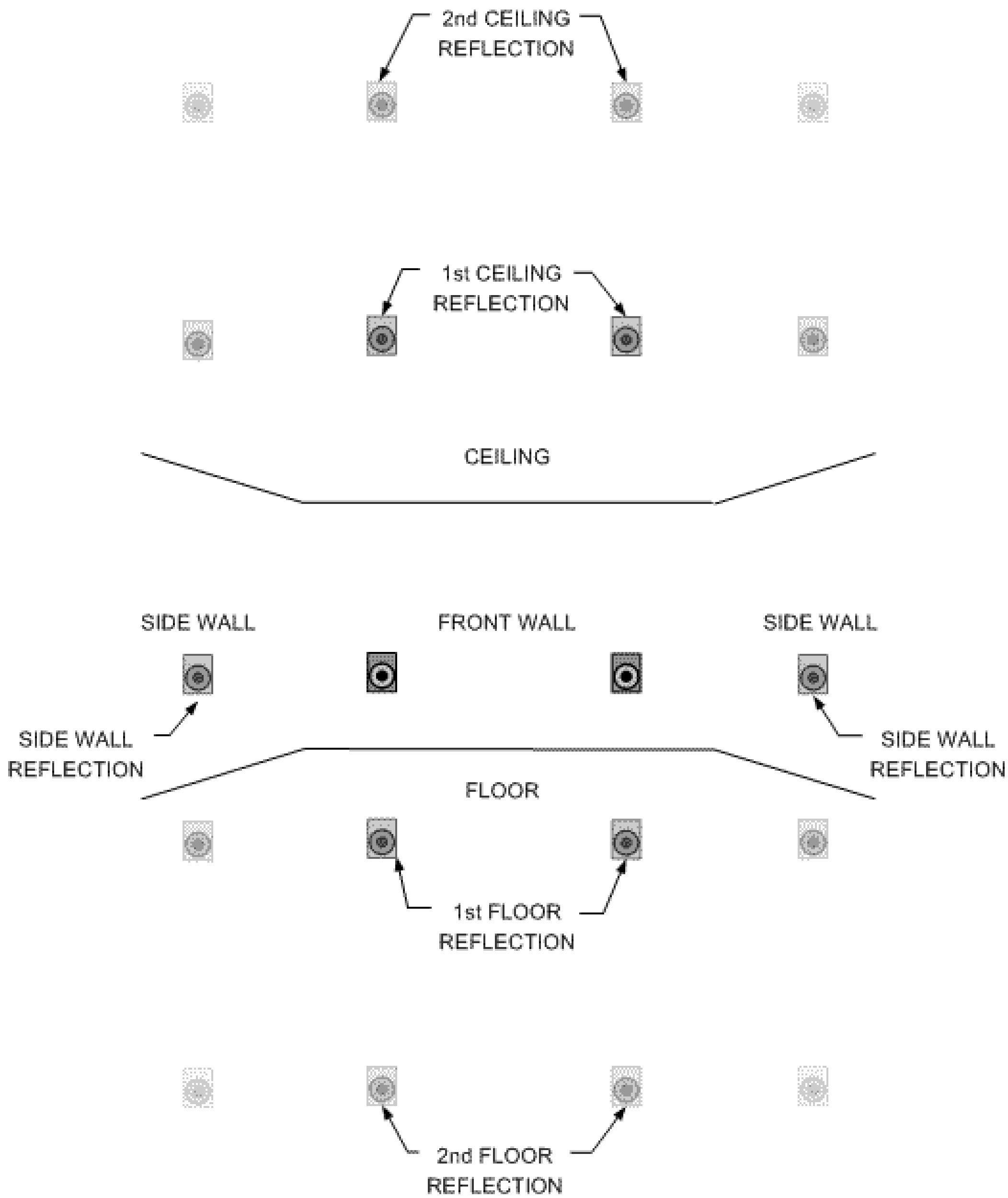


Figure 2-9: A Stereo Pair of Point Source Speakers and Their First Few mages

My response to questions about an array's "comb filtering" is to point out that our human hearing systems are not sensitive to this particular type of frequency coloration. Besides, point source loudspeakers suffer the same comb filtering effects when used in rooms and/or in multiples, that is, in normal use. So comb filtering results whenever any speaker is used in a normal residential environment, if is not something that happens only with arrays. My goal with the corner-line-array is to include the inevitable room reflections in the design from the start in order to achieve a frequency response that is more consistent throughout the room and from room to room. I would hope that if you reproduce the MCLA's in your room that you would achieve very nearly the same frequency response as I achieve in my own reference system. This is rarely the case with point source speakers for which there is no standardized room location.

## Lines in the Room

Now consider what happens when we place a line array loudspeaker in a room where the line array spans from floor to ceiling. Note the ceiling and floor reflections shown in Figure 2-10. The first order reflections TRIPLE the effective length of the array. Including the second order reflections we see the height of the array increased FIVE FOLD over the actual speaker. A seven foot long array is reflected into a 35 foot array by consideration of just the first two reflections. A seven foot array of 25 speakers is transformed into a 35 foot array with 125 sound sources based on two reflections. In reality the higher order reflections are significant and the array is effectively longer with even more sound sources. In a perfectly reflective room the lines would extend to infinity giving the room an infinite reverberation time. The finite reverberation time of real rooms indicates that the reflections actually fade to inaudibility as they extend toward infinity in all directions. The combination of direct and reflected sound sources actually forms a very long array with the output progressively more tapered toward each end.

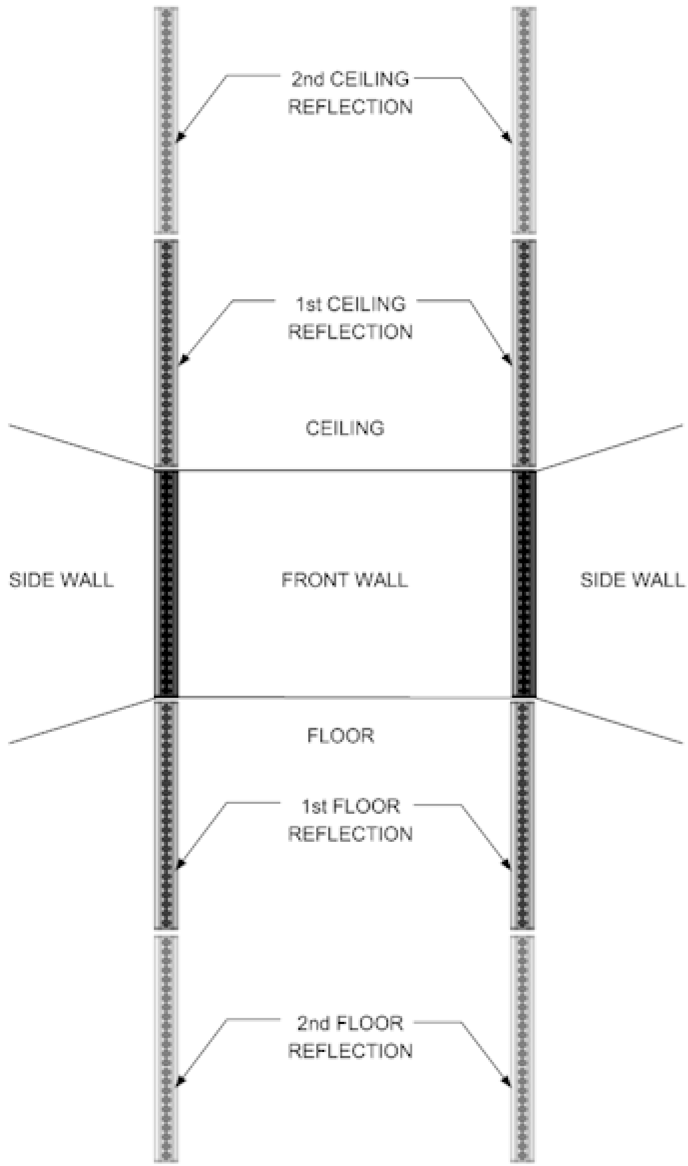



Figure 2-10: Line Arrays with Ceiling and Floor Reflections

## Fun With Mirrors

Now let's look at the view from above when a line array is placed in the corner of the room. Figure 2-11 shows the corner-line-array placed in the corner of a room. In light of the front and side wall reflections we now have four arrays tightly packed into the corner.

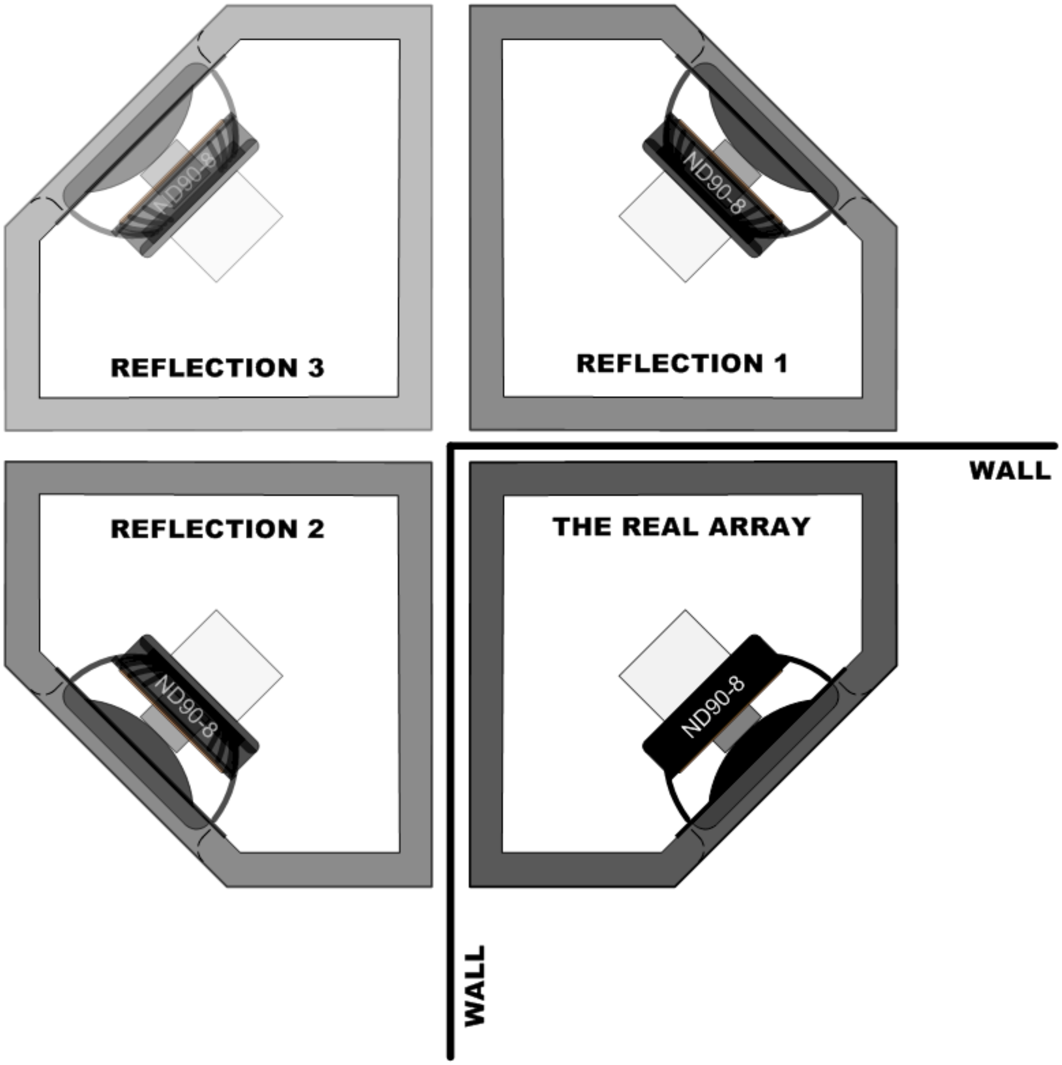


Figure 2-11: Top View of the Single Corner-Line-Array with Three Reflections

I deliberately shaped the enclosure to fit in the corner in such a way that it joins with its reflections to form a tight octagonal cluster of four line arrays. If you imagine moving the array out of the corner you would see all four arrays moving apart. Bringing the array into the corner causes the front and side wall images to merge with the real line to form a very real acoustic cluster of four arrays extending about 16 feet above and below the actual arrays. Instead of having one driver every 3.5 inches we actually have four drivers for every 3.5 inches of height. You could argue that from the listening area we now effectively have an average driver spacing of less than 1 inch. A single physical array near 8 feet in length with 24 drivers integrates with the room to form a new acoustic system of four clustered arrays extending another 16 feet beyond the ceiling and floor and tapering off toward infinity. Figure 2-12 shows a 3D view of one array with its corner reflections and repeating floor reflection.

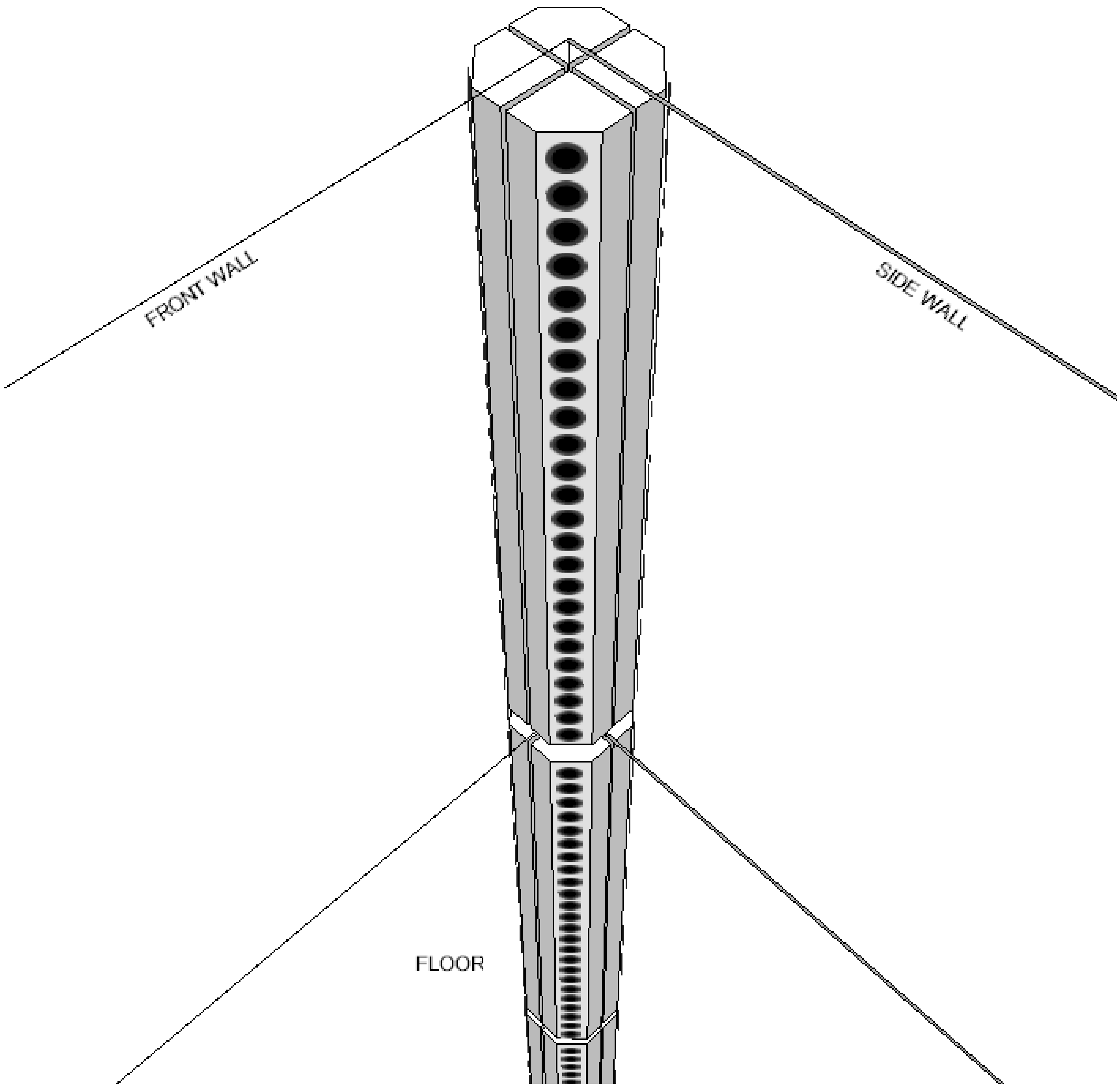



Figure 2-12: 3D View of the Corner-Line-Array with Corner and the First Two Floor Reflections

This net acoustic system now has the power of not just 24 three inch radiators but a total of N = 5 x (24 x 4) = 480 radiators. That's for just one line. With two MCLAs in the room you effectively have about 1000 sound sources configured as long octagonal tubes with speakers on 4 faces of the octagon. A 3D view of two arrays in a room along with corner reflections and two ceiling and floor reflections is shown in Figure 2-13. If you wanted to reproduce the MCLA system in free space outdoors that is exactly how you would construct it...using about 1000 drivers. But remember that the ceiling and floor reflections don't actually stop at just two reflections as shown but rather continue to infinity in each direction fading out as they go. A simple array of 24 drivers is reflected into a monstrous acoustic tower when carefully placed in the corner of a room. As if this is not enough to boggle the mind, each of these two "towers" has additional repeating reflections in the side walls and rear walls (not shown).

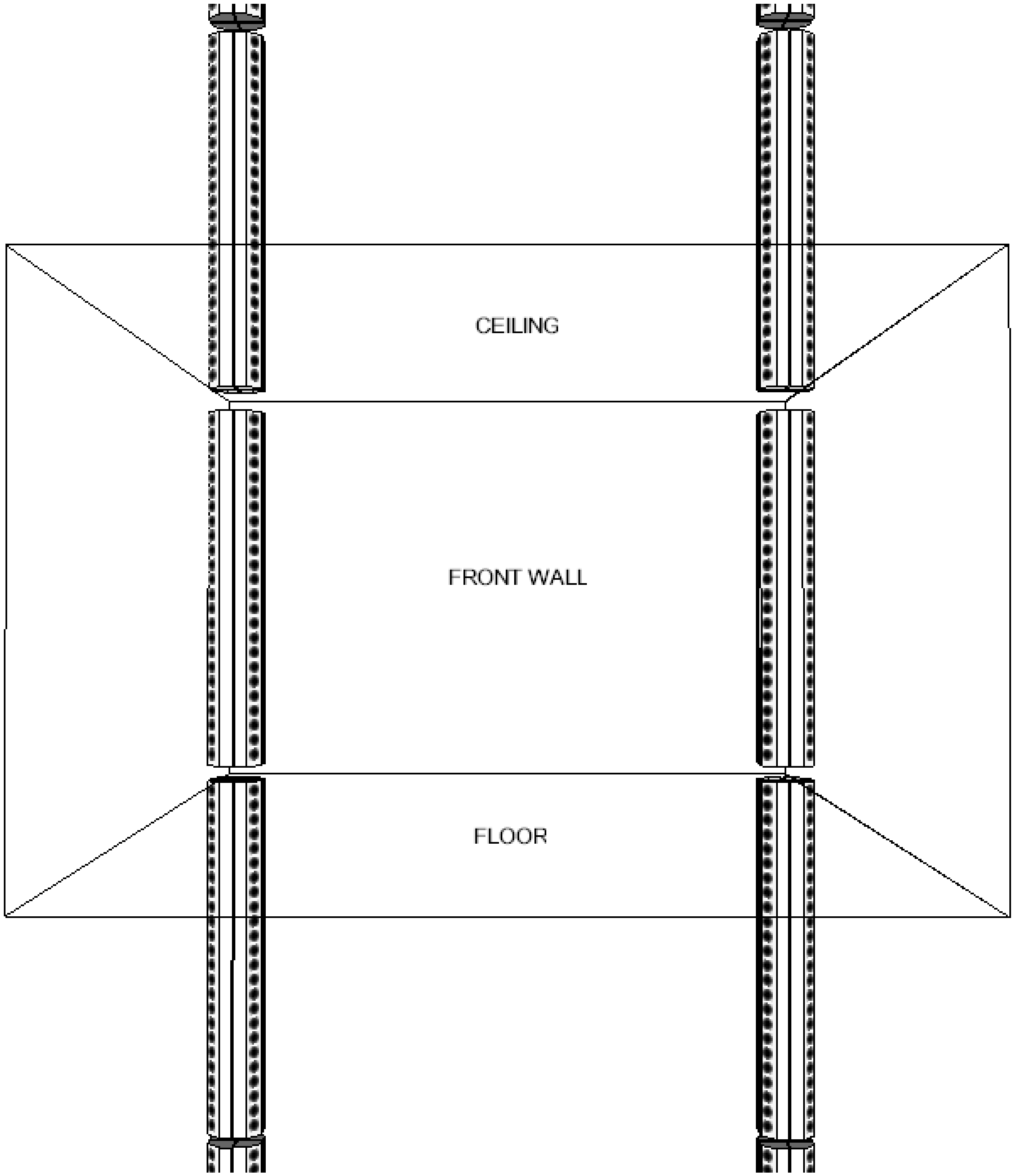



Figure 2-13: 3D View of Two Corner-Line-Array with Corner Reflections and the First Two Ceiling and Floor Reflections

Simulating arrays with up to 24 drivers reveals that the net frequency response consistently improves as the number of drivers is increased and their spacing reduced. The simulated response for 24 drivers averaged over the listening area is quite excellent so I have every reason to expect that a simulation of 480 drivers (one array with first reflections) would prove to be very much smoother throughout the listening area...just as I measure. The large number of sources and their tight spacing very effectively resolve many of the issues that are raised concerning line array loudspeakers. By "managing" the room reflections instead of ignoring them, the MCLA loudspeaker system can achieve a frequency response that has a very good chance of being consistent from room to room. I look forward to seeing reports and measurements from those of you who build the MCLA.

Here, just for fun, in Figure 2-14 is another 3D view from outside the front top-left corner of the room looking toward the front bottom-right corner. A more evolved design might move toward closing the gap by making the array length more closely match the exact floor to ceiling distance.

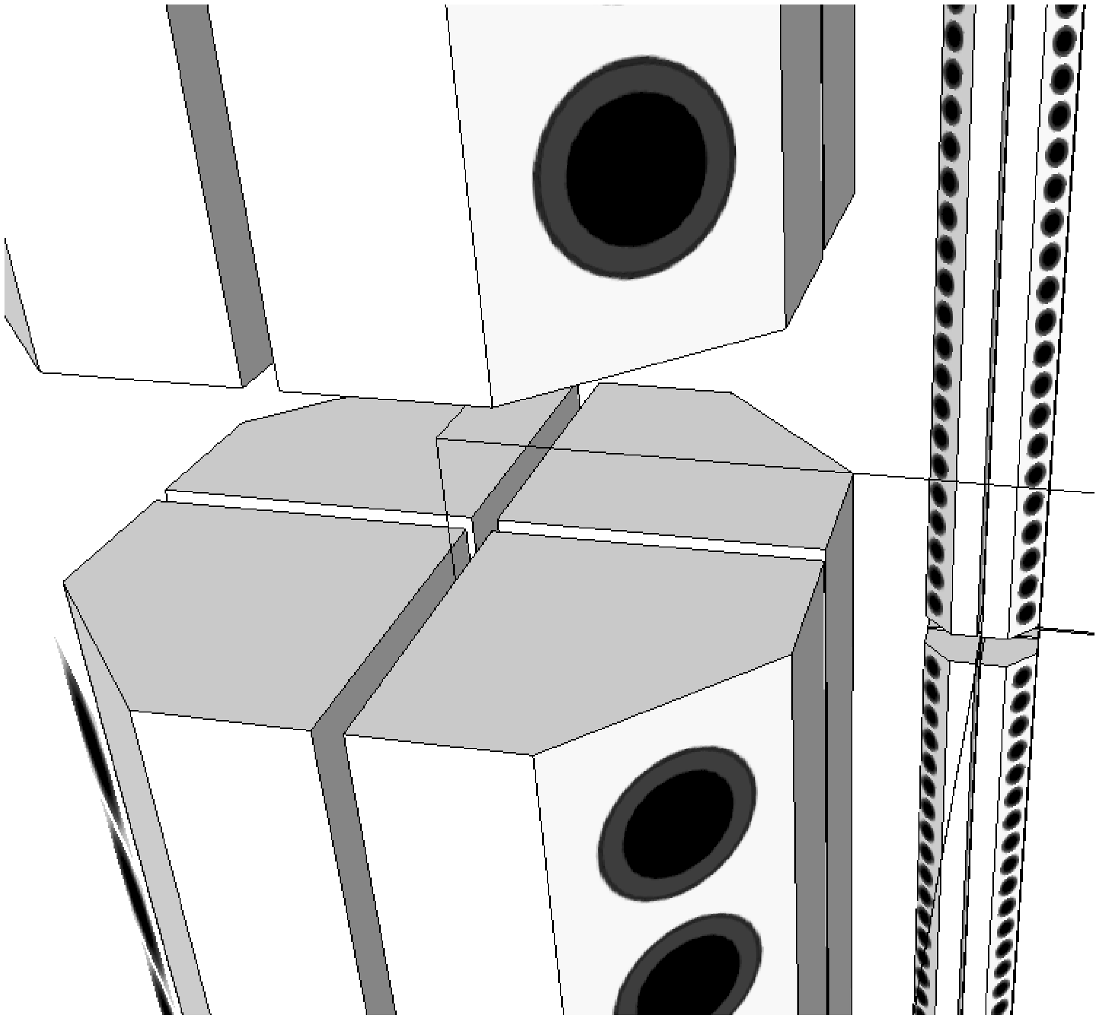


Figure 2-14: Alternate 3D View of a Pair of Line Arrays with Reflected Images

The corner-line-array when located in the room as prescribed provides an overall acoustic playback system that is generally superior to the typical point-source playback system (Figure 2-9 above). The widely spaced images of the point-source playback system produce a colored frequency response that is unique to each room and varies with speaker position in the room making consistent performance almost impossible. The corner-line-array with its standard speaker placement and well managed reflections appears, in my opinion, to be a superior solution to the overall application of loudspeaker playback systems in home environments.

Those who might still be concerned about "comb filtering" effects are encouraged to reconsider their concerns in light of the image analysis and measured data presented here for the corner line array. Comb filtering is inescapable when any speaker is placed in a typical residential listening room due to the multi-path reflections always present. The typical placement of point source, line or planar speakers in our listening rooms results in especially bad comb filtering because the reflected images are so distant from the sound source. Ultimately what really matters in a high performance audio monitoring system is the frequency response that the system achieves in-room at the listening area, not the anechoic response that WOULD occur IF the system were auditioned in an anechoic chamber...in mono.

As a summary of the image analysis of the MCLA loudspeaker-room system I present Figure 2-15 which depicts the effective acoustic system that results from the placement of just two MCLA enclosures in a listening room. By precisely combining the real enclosures with their reflected images each enclosure is effectively multiplied first into a column of four enclosures by virtue of the side wall reflections. Each quad enclosure is then multiplied again into tall towers of quad enclosures extending far above the ceiling of the room and below the floor of the room. This is by consideration of just the first two reflections. In fact the towers extend much further in each direction fading at each extreme. Also, not shown are the subsequent reflections from the side walls that result in further repeating images of the two towers in all directions. As astonishing as the system shown in Figure 2-15 may be, the acoustic measurements presented later will confirm that the in-room behavior of just two MCLA enclosures is acoustically equivalent to 40 enclosures in free space. While my claims for the MCLA may seem almost magical I assure you that this wonderful invention is simply the result of careful application of the image method to the design process.

Figure 2-15: The In-Room Acoustic Equivalent of Just Two Murphy Corner-Line-Array Enclosures

## References:


[2-1] C.F. Eyring, "Reverberation Time in 'Dead' Rooms," *J. Acoust. Soc. Am.*, (1930, Jan.).

[2-2] J. B. Allen and D. A. Berkley, "Image method for efficiently simulating small-room acoustics." *J. Acoust. Soc. Am.*, vol. 65, no. 4, pp. 943–950 (1979).

[2-3] Richard V. Waterhouse, "Output of a Sound Source in a Reverberation Chamber and Other Reflecting Environments." *J. Acoust. Soc. Am.*, vol. 30, no. 1 (1958).

[2-4] Floyd D. Toole, "Sound Reproduction, The Acoustics and Psychoacoustics of Loudspeakers and Rooms.", Focal Press, 2008

# 3. MCLA Project Details

## The Transducers

The Dayton Audio ND90 3.5 inch speaker is available from Parts Express.

Per the Parts Express web site pricing for the ND90 is as follows:

1-10 units: $19.70 each

10-49 units: $16.90 each

**50+ units: $12.90 each (this is nearly 35% off the single unit price)**

The MCLA system employs 24 speakers per corner line array enclosure. If you are considering building a pair of MCLAs I recommend buying at least 50 speakers to allow you the freedom to reject one or two for quality reasons and/or to have a replacement driver to set aside.

**Total Cost for 50 ND90 speakers: $645.00**

At $645 for the drivers alone this may not be a low cost speaker system, but when you consider the high level of performance that this system delivers, the value is quite large.

## The Enclosure

My initial prototype enclosures (Prototype #1) were built out of 1/2" (actually 15/32") plywood. I am a minimally skilled woodworker so my prototypes are a bit rough, but they are fully functional. I ended up covering the entire face of the line enclosures in black grill cloth. I have seen some of the fine looking enclosures that DIY speaker builders create so I am looking forward to seeing better implementations of the MCLA than my Prototype #1. Please send pictures of your finished systems and I will post them here.

Figure 3-1 shows an overview of the plans for Prototype #2. The detailed plans are available in .pdf form below.

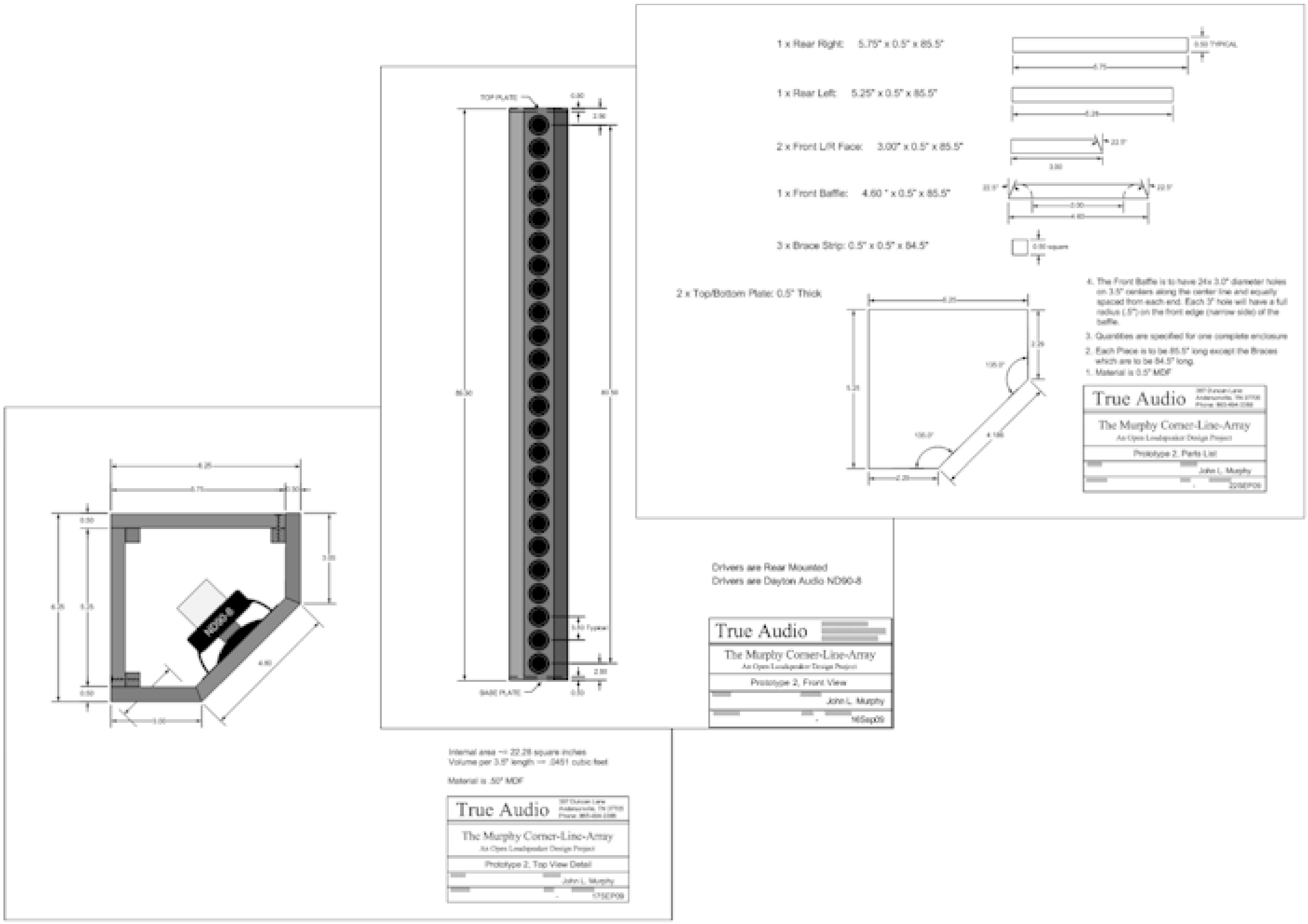



Figure 3-1: The Enclosure Drawings

The recommended enclosure at this time is Prototype #2. The plans are provided in a .pdf file for easy reference and easy printing.

| Enclosure Revision | Revision Date | Download |
|---|---|---|
| | | |
| Prototype #2 | 16Dec09 | MCLA_Proto_Enclosure_2.pdf |
| | | |

## Wiring Configuration

With 24 drivers of 8 Ohms each it is necessary to employ a series/parallel wiring arrangement in order to achieve a net load impedance near 8 Ohms. The simplest ways to wire the drivers are either six parallel groups of four in series or four parallel groups of six in series. I chose the first case in order to achieve an impedance that falls between 8 and 4 Ohms (5.33 Ohms). Figure 3-2 shows the wiring diagram. The other arrangement with four groups of six drivers in series would yield a net impedance of 12 Ohms which I decided was just too high.

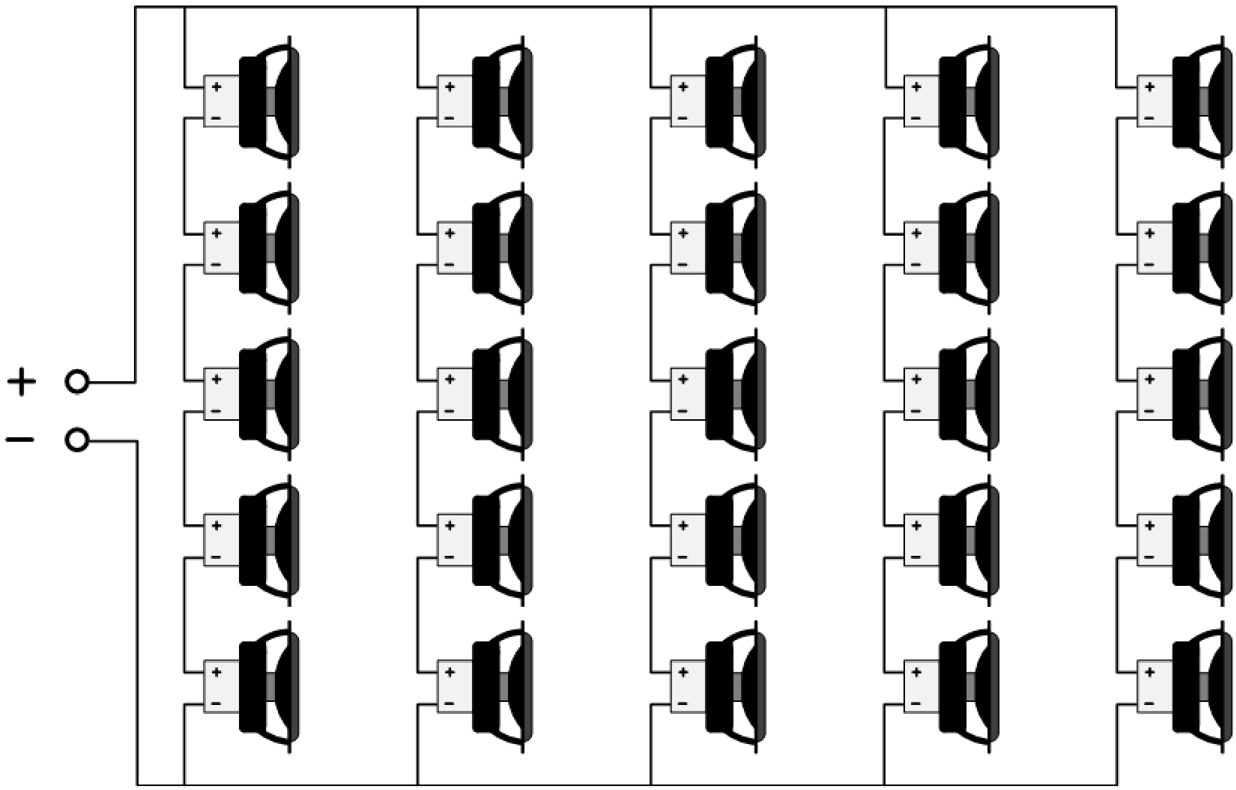


Figure 3-2: The MCLA Wiring Diagram for 25 Drivers

## The Equalizer

I am designing a custom analog EQ to voice the MCLA. For now however, I am using an off-the-shelf digital 1/3rd octave equalizer. I strongly recommend that you use this same EQ in order to directly implement my EQ settings for the MCLA. The equalizer I am using is the Behringer Ultra-Curve Pro DEQ2496 shown in Figure 3-3 below. This unit is available from Parts Express for $299.99. The thing I especially like about this unit is the accurate correspondence between the boost/cut settings and the amount of boost/cut actually achieved. There is minimal interaction between bands and being digitally implemented the EQ is perfectly repeatable. It would not be possible to convey my settings to other users and get precisely repeatable results with anything besides a digitally implemented EQ. If you want to precisely reproduce the MCLA voicing you will need to use this equalizer. I will tell you exactly how to set it so that should not be a concern.


Figure 3-3: The Behringer Ultra-Curve Pro DEQ2496 Programmable EQ

## The Equalizer Settings

Here are the equalizer settings I have arrived at for the MCLA systems in my music studio room. Shown in figure 3-4 below are the average settings for the two channels.

## MCLA Equalization

### Average EQ for Proto Unit #1 at Right and Unit #2 at Left

1-Oct-09 Note: values in **Bold** are calculated
3-Oct-09

| | GRAPHIC EQ SETTINGS | | | X-CURVE ADJUSTMENTS | |
|---|---|---|---|---|---|
| | FLAT | Small Room X-Curve | X CURVE | | |
| Frequency (Hz) | EQ SETTINGS (dB) | EQ SETTINGS (dB) | EQ SETTINGS (dB) | Small Room X-Curve (dB) | X-Curve (dB) |
| 20 | 4.0 | **4.0** | **4.0** | 0 | 0 |
| 25 | 7.8 | **7.8** | **7.8** | 0 | 0 |
| 31.5 | 3.0 | **3.0** | **3.0** | 0 | 0 |
| 40 | 5.8 | **5.8** | **5.8** | 0 | 0 |
| 50 | -1.3 | **-1.3** | **-1.3** | 0 | 0 |
| 63 | -6.5 | **-6.5** | **-6.5** | 0 | 0 |
| 80 | -8.3 | **-8.3** | **-8.3** | 0 | 0 |
| 100 | -14.8 | **-14.8** | **-14.8** | 0 | 0 |
| 125 | -10.0 | **-10.0** | **-10.0** | 0 | 0 |
| 160 | -14.5 | **-14.5** | **-14.5** | 0 | 0 |
| 200 | -15.0 | **-15.0** | **-15.0** | 0 | 0 |
| 250 | -10.3 | **-10.3** | **-10.3** | 0 | 0 |
| 315 | -9.8 | **-9.8** | **-9.8** | 0 | 0 |
| 400 | -5.0 | **-5.0** | **-5.0** | 0 | 0 |
| 500 | -7.0 | **-7.0** | **-7.0** | 0 | 0 |
| 630 | -5.5 | **-5.5** | **-5.5** | 0 | 0 |
| 800 | -7.3 | **-7.3** | **-7.3** | 0 | 0 |
| 1000 | -4.8 | **-4.8** | **-4.8** | 0 | 0 |
| 1250 | -4.3 | **-4.3** | **-4.3** | 0 | 0 |
| 1600 | -3.5 | **-3.5** | **-3.5** | 0 | 0 |
| 2000 | -3.0 | **-3.0** | **-3.0** | 0 | 0 |
| 2500 | 0.0 | **-0.5** | **-1.0** | -0.5 | -1 |
| 3150 | -0.5 | **-1.5** | **-2.5** | -1 | -2 |
| 4000 | 2.3 | **0.8** | **-0.8** | -1.5 | -3 |
| 5000 | 3.8 | **1.8** | **-0.3** | -2 | -4 |
| 6300 | 2.8 | **0.3** | **-2.3** | -2.5 | -5 |
| 8000 | 10.8 | **7.8** | **4.8** | -3 | -6 |
| 10000 | 14.8 | **11.3** | **7.8** | -3.5 | -7 |
| 12500 | 10.3 | **6.3** | **2.3** | -4 | -8 |
| 16000 | 15.0 | **10.5** | **6.0** | -4.5 | -9 |
| 20000 | 15.0 | **10.0** | **5.0** | -5 | -10 |

Note: the parametric EQ settings are common to all three graphic EQ setups.

| | PARAMETRIC EQ SETTINGS | | |
|---|---|---|---|
| Parametric Band | Freq (Hz) | BW (octaves) | Gain (dB) |
| 1 | 35.6 | 1/4. | 5 |
| 2 | 44.8 | 1/4. | -6.5 |
| 3 | 563 | 1/4. | 4 |
| 4 | 8933 | 1/4. | -3.5 |
| 5 | 11246 | 1/4. | 4.5 |
| 6 | | | |
| 7 | 15704 | 1/5. | 3 |
| 8 | 180 | 1/5. | 3.5 |
| 9 | 17825 | 1/5. | -2 |
| 10 | 20000 | 1/2. | 4 |

Figure 3-4: Equalizer settings for the MCLA, Page 3 Average Settings

I use separate EQ settings for my Left and Right channels. By swapping the two enclosures and measuring each way I determined that the two prototype enclosures are very consistent but that there is a small difference between the two positions. My corners are actually not ideal as one has windows at either side and the other has a window to one side and a door to the other. So I have prepared a PDF file containing one page of settings (each) for my current left and right positions along with an average of the two. In general, I would recommend using my average EQ unless your corner configuration closely matches one of mine. The EQ settings PDF file can be downloaded from the project site at www.trueaudio.com/array. Each page of the document has the three settings for the Graph Equalizer portion of the Behringer EQ. These settings correspond to:

A: Flat
B: Small-Room X-curve
C: X-curve

In addition to the Graphic EQ settings I have included settings for the Parametric EQ portion of the Behringer unit as well. These adjustments are just as important as the graphic EQ settings. I recommend you enter the data for the FLAT setting and then save this to three different memories. Then you just need to edit the second two memory locations to create the small-room X-curve and X-curve EQ settings. Note that the only modules of the EQ unit that I am using are the Graphic EQ and Parametric EQ. All other modules are bypassed at the unit's bypass pages.

# 4. Single Driver Test Results

## Parameter Measurement


Figure 4-1: The Dayton Audio ND90-8 Full Range Speaker

I measured the ND90 driver parameters using the Dayton Audio WT3 system. This allowed me to simulate the system's response using data from the actual drivers I was going to use. Figure 4-2 shows my measured impedance along with parameters for unit #1. Unit #1 was typical of several units I tested.

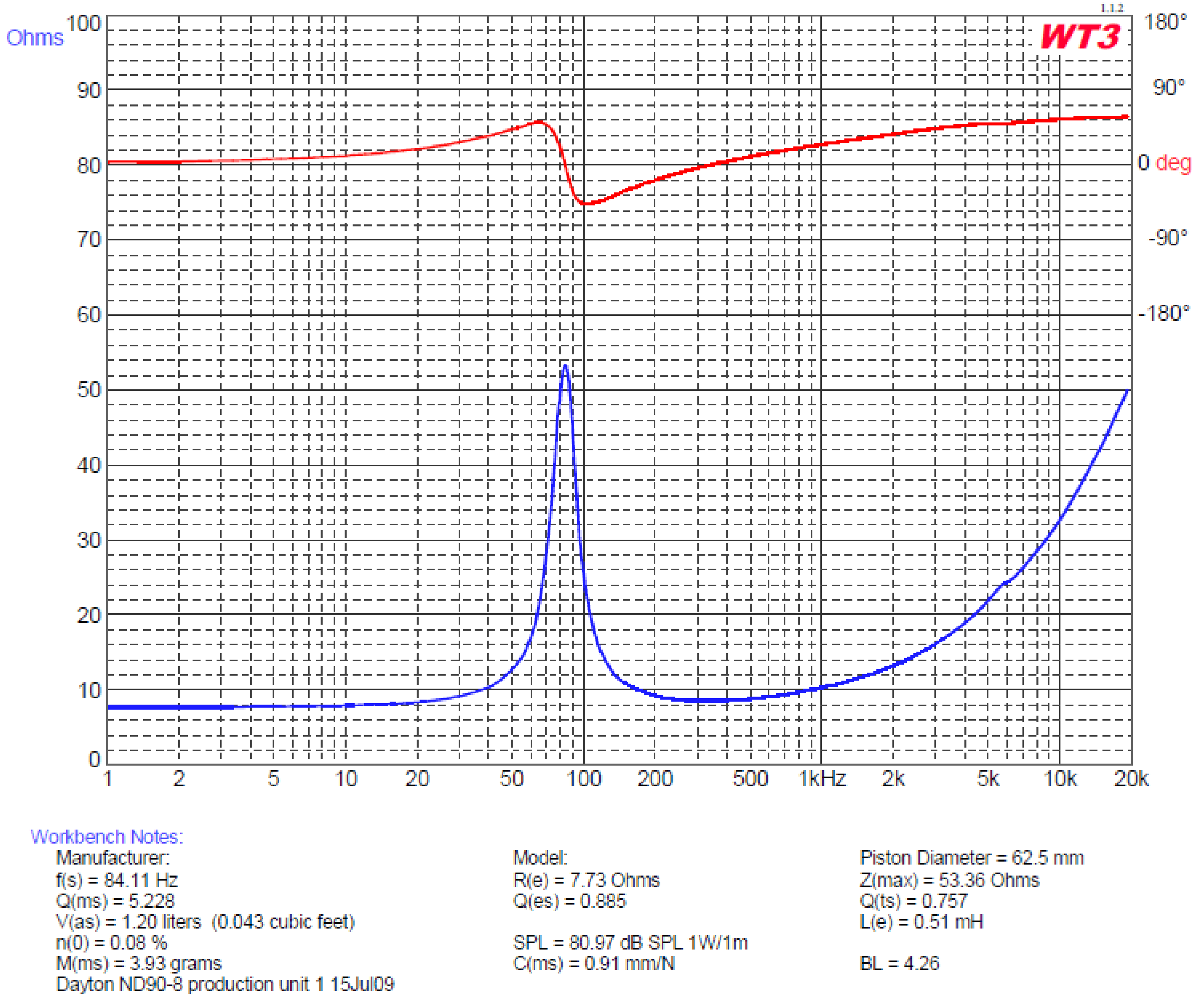



Figure 4-2: ND90 Impedance and Thiele/Small Parameters measured using the WT3 Woofer Tester

## Measuring Xmax

As part of the driver qualification testing I especially wanted to verify the driver's Xmax as this was a critical parameter for allowing the system to achieve adequate acoustic output in the lowest octaves. Because of the difficulty in measuring Xmax I decided to perform a direct observation of the excursion with the driver was operating at 50 Hz and distortion increasing to 10% 3rd harmonic (i.e. 3rd harmonic 20 dB below the fundamental). I set up my camera to catch the actual excursion as the system was operated at the excursion limit.

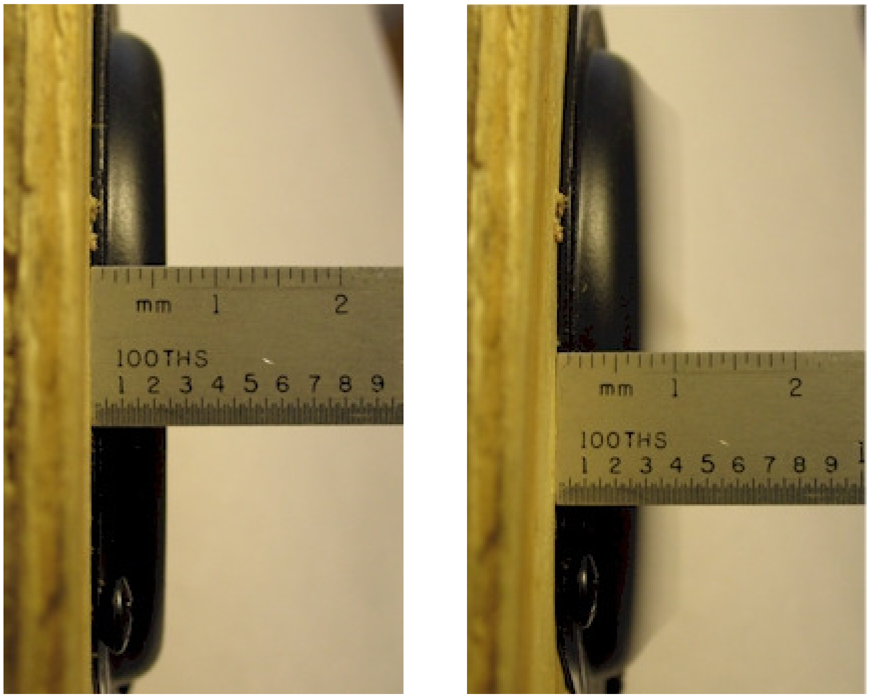



Figure 4-3: The ND90-8: excursion baseline (left) and 50 Hz at 10% 3rd harmonic distortion. (right)

The baseline displacement is 6mm as seen in the Figure 4-3 (above) at the left. Operating at 10% 3rd harmonic the peak excursion is seen to reach the 11mm mark in the right photo. That equates to a peak excursion of 5mm at 50 Hz as the 3rd harmonic reaches 10%. From this observation I'd say the driver has an effective Xmax of 5mm. At no point did the driver bottom out or otherwise appear to reach any hard mechanical limit in the course of my testing. Compared to most of the speakers I've evaluated over the years the ND90 performs exceptionally well up to and even beyond the limits of its excursion capability.

## Frequency Response

I measured the frequency response of an individual ND90-8 driver using nearfield and ground plane methods. The nearfield measurement method provides a very accurate response up to about 3kHz for this size driver. This upper limit on the nearfield data is a limitation of the nearfield measurement method itself as described by Don Keele. [Ref. 4-1] The nearfield measurement was very smooth and highly repeatable so that no averaging or smoothing was required. Figure 4-4 shows a single unsmoothed nearfield measured response. Note that the nearfield response is equivalent (in shape) to a farfield half-space response. That is, it exhibits no diffraction loss as would normally be seen in the farfield measurement with the enclosure in free space.

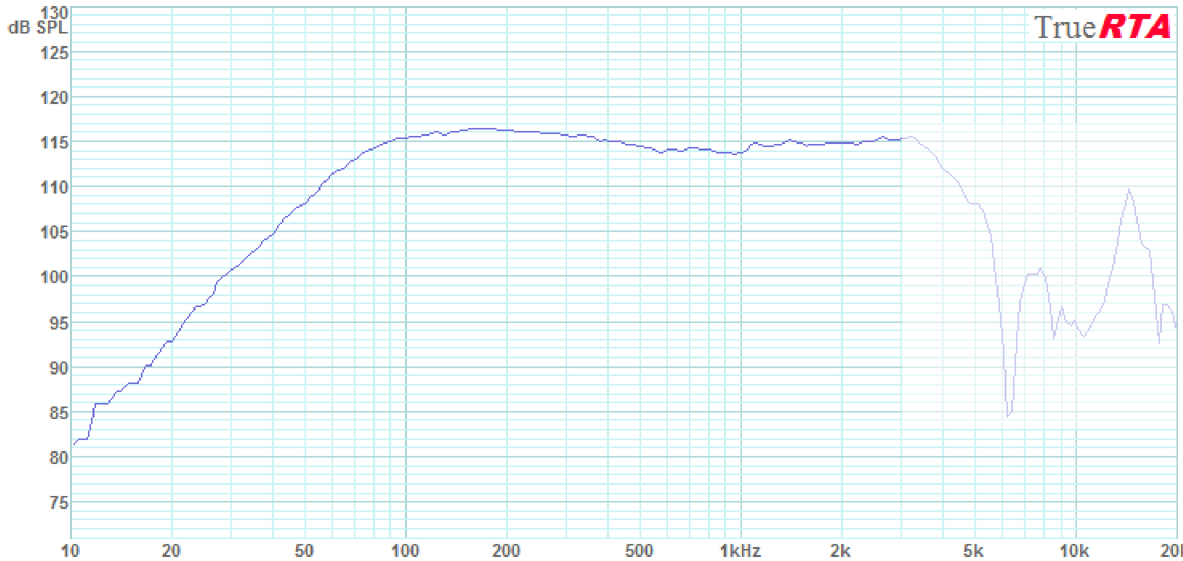


Figure 4-4: ND90-8 Nearfield Frequency Response in 0.1 cubic foot closed box (ignore above 3kHz)

The ground plane frequency response was measured at 1W (2.83 Vrms), 1 meter outdoors about 15 feet from my house. The ground plane measurement is equivalent to a farfield measurement with two identical speakers placed side-by-side (or one over the other). My ground plane measurement setup was not ideal and definitely included some local reflections. Still, it does give a good picture of the overall frequency response of an individual driver in a small enclosure in free space. The measurement seen below includes the effect of diffraction loss which appears as a 6 dB decrease in response below 2 kHz. For more on diffraction loss see my Tech Topic on the subject.

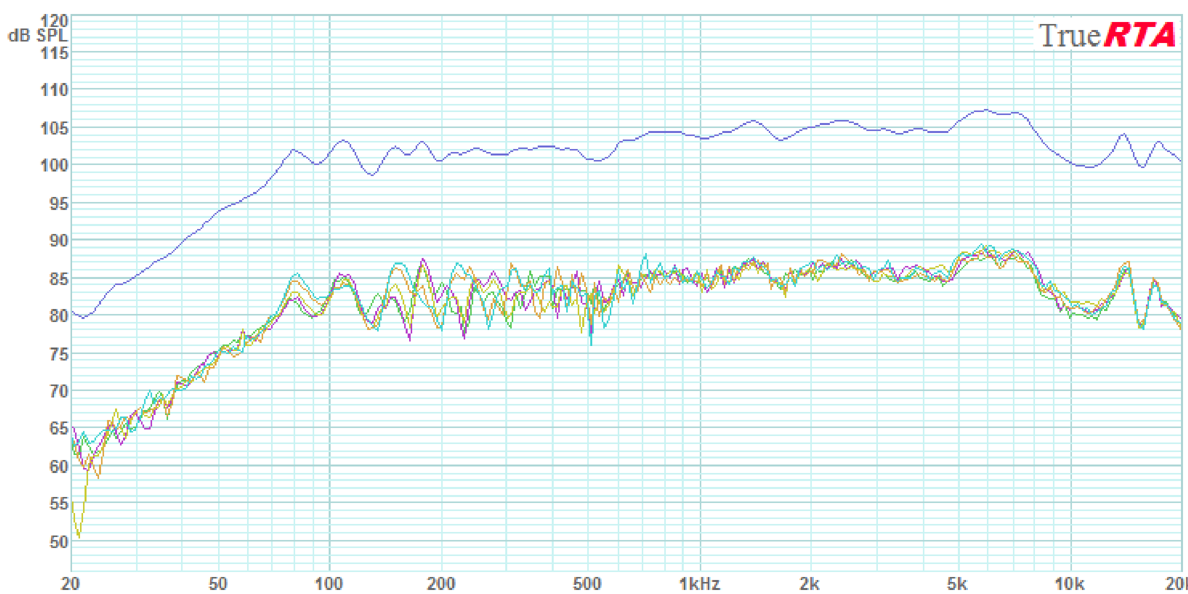


Figure 4-5: ND90 Ground Plane Frequency Response, Closed Box (0.1 cu ft), 1W/1m

The ground plane response shown at the top of Figure 4-5 is an average of the 5 responses below with (slight) 1/6th octave smoothing. The five responses were obtained by varying the outdoor measurement geometry slightly (rotation and/or translation) between measurements in an attempt to average out variations due to reflections from the side of the house.

In order to create a frequency response that best represents the response of the ND90 I combined the nearfield response below 2 kHz with the ground plane response above 2 kHz to get the hybrid response shown in Figure 4-6. This response is representative of the response of the driver in a small closed box with a half-space acoustic load.

Note that these frequency responses are displayed in relatively high resolution with minor divisions equal to just 1 dB.

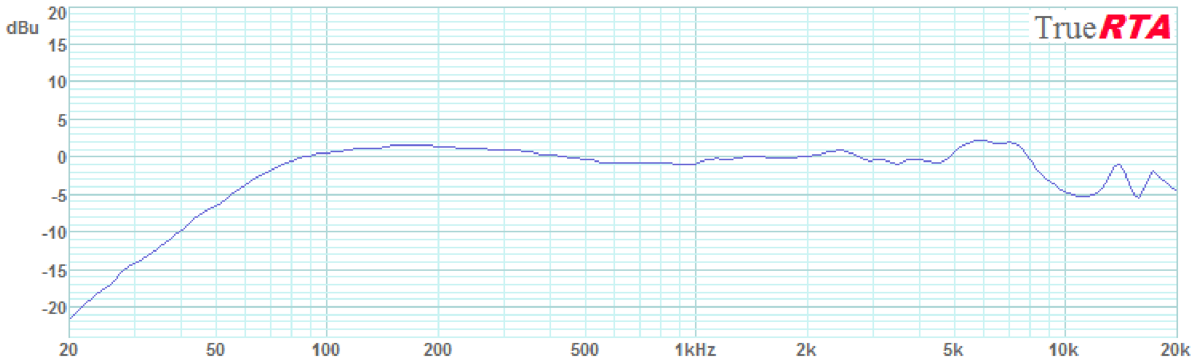


Figure 4-6: ND90 Nearfield/Ground Plane Hybrid Closed Box Response (0.1 cubic foot)

The raw response of the above closed box system is within +/- 2 dB from about 70 Hz to 8.5 kHz. The response is within +/- 4 dB from about 55 Hz to 20 kHz. It is the very wide frequency response range of the driver that makes this project possible.

## Distortion Performance

Figures 4-7 through 4-10 show the distortion performance of a single ND90 driver in the 0.1 cubic foot test enclosure. The distortion was measured at 1 Watt (2.83 Vrms), 1 meter.

We see the ND90 reproducing 100 Hz in Figure 4-7. The plot shows the 100Hz sine wave at a level of 80 dB SPL with higher harmonics (distortion components) all below 40 dB SPL. Note that the lines at 60, 120 and 680 Hz are components of ambient room noise...the notebook PC primarily. The 2nd harmonic at 200 Hz is at 38 dB SPL versus 80 dB SPL for the fundamental. This puts the 2nd harmonic 42 dB below the fundamental which equates to 0.8% second harmonic distortion. The 3rd harmonic at 300 Hz is down 44 dB from the fundamental for a 3rd harmonic distortion of 0.6%. The 5th harmonic is at -54 dB or 0.2%. The distortion components above the 5th harmonic vanish into the noise floor.

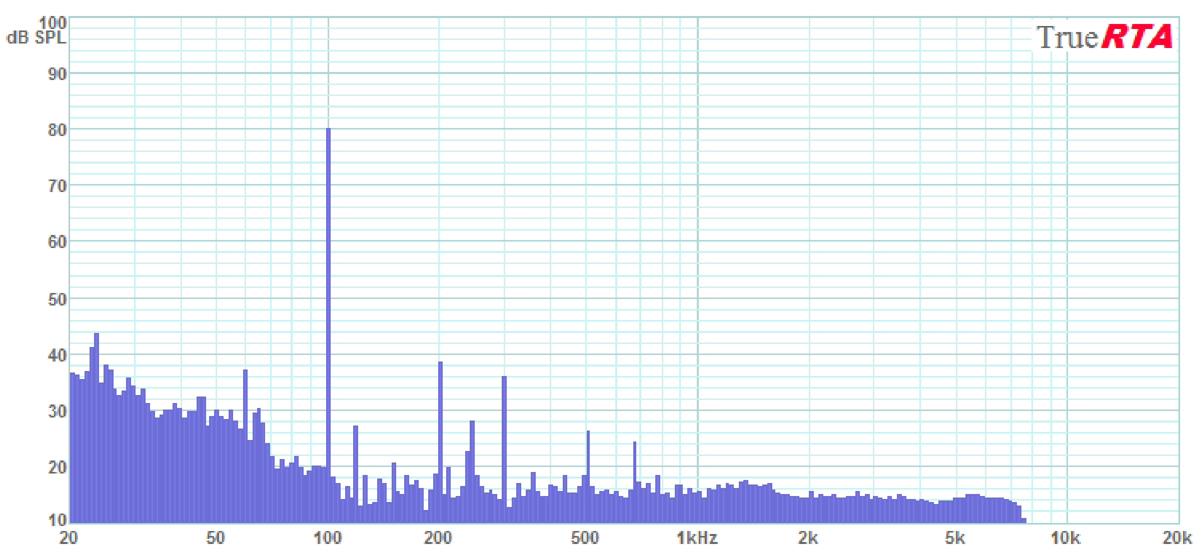


Figure 4-7: The ND90 at 100 Hz, 1Watt, 1 meter

Next in Figure 4-8 we see the driver at 200 Hz. The 2nd harmonic at 400 Hz is at -51 dB or 0.3%. The 3rd harmonic, 600 Hz is at -55 dB or 0.18%. The 5th harmonic is at -57 dB or 0.14%.

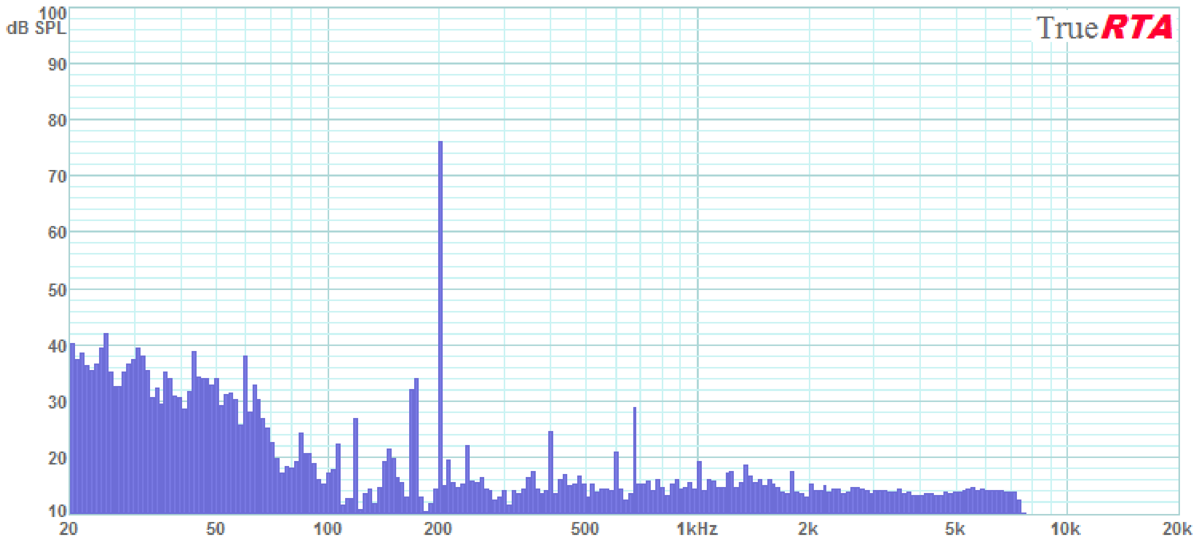


Figure 4-8: The ND90 at 200 Hz, 1Watt, 1 meter

The 500 Hz distortion performance is shown in Figure 4-9. The 2nd harmonic is at least 63 dB below the fundamental or .07% or lower. The 3rd harmonic at 1.5 kHz is at -55 dB or 0.18%. The 5th harmonic is at -59 dB or 0.11%.

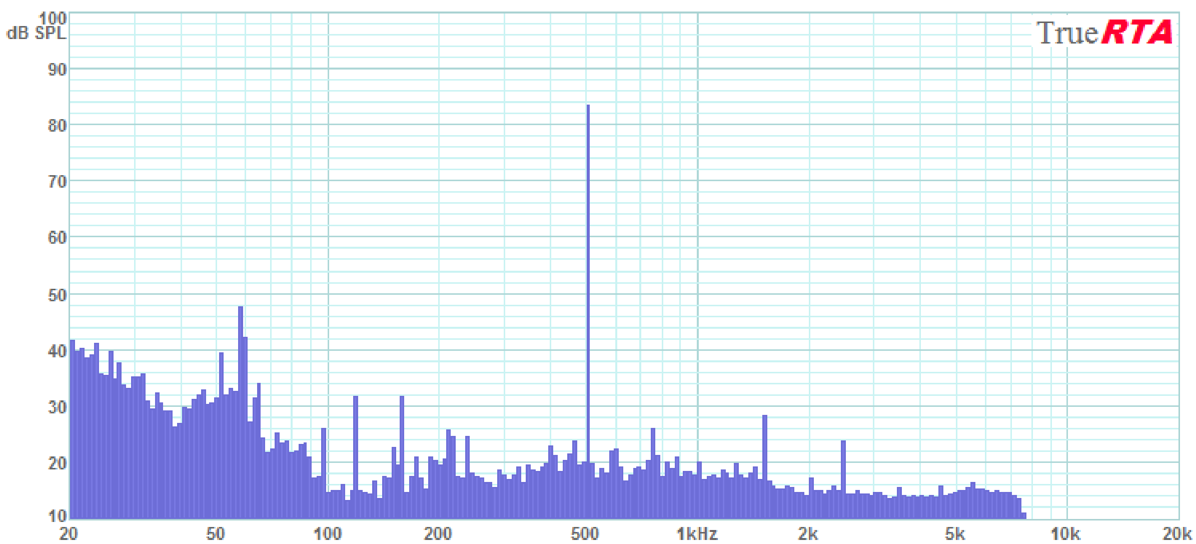


Figure 4-9: The ND90 at 500 Hz, 1Watt, 1 meter

The performance of the ND90 at 1 kHz is shown below in Figure 4-10 where we see that the dominant distortion component is the 3rd harmonic at -49 dB with respect to the fundamental or 0.35%. The 5th harmonic is at -55 dB or 0.18%.

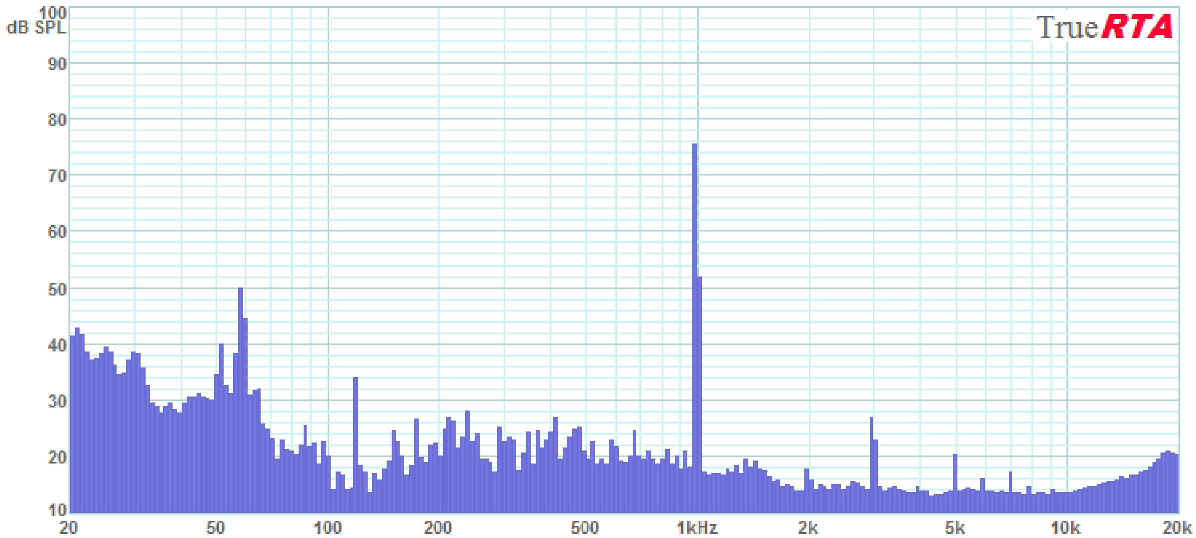


Figure 4-10: The ND90 at 1000 Hz, 1Watt, 1 meter

The distortion measurements for the ND90 show a driver with less than 1% harmonic distortion at the 1 Watt drive level. No other hidden distortions were evident in the course of my distortion testing. These numbers confirm what my ear had already told me: this is a nice clean little driver with no audible distortions of any type.

While healthy ND90s exhibit the low distortion shown above I did have one driver (out of 50) that "buzzed" and had to be replaced. For this reason when performing final assembly on the loudspeakers it is a good idea to quickly audition each driver with a low frequency tone (say 80 Hz) to identify any "buzzing" units before they are installed. In production environments a "rub and buzz" test is often performed on drivers as they are unpackaged and just before they are installed into a speaker system.

# 5. Array Test Results

## Equalized In-Room Frequency Response

Here are the current frequency response measurements as of 3Oct09. Each measurement below is an average of the Left and Right systems in-room measured response. Each of the Left and Right measured responses consists of an average of 16 unsmoothed responses measured in the listening area over a range from 1 to 3 meters from each array. The final average is smoothed just one time if smoothing is specified. The unsmoothed average is an average of completely unsmoothed data. These responses represent the array with my best equalization to date. Note that the responses are shown at a high resolution with 1 dB per division on the vertical scale.

Figure 5-1 shows the MCLA measured spatial average frequency response with 1/3rd octave smoothing. This is what would typically be given as the measured response if this were a commercial loudspeaker. The frequency response is seen to be within +/- 1 dB from 28 Hz to 20 kHz.

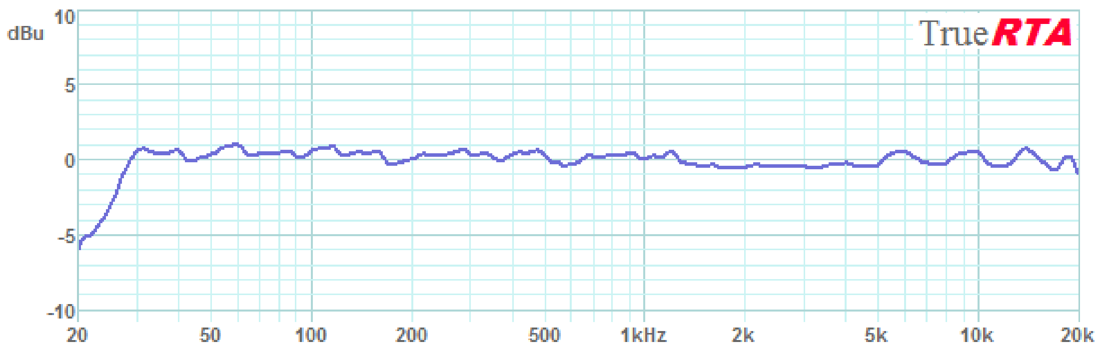


Figure 5-1: The MCLA In-Room Measured Response with 1/3rd Octave Smoothing

Figure 5-2 shows the same measured frequency response as above but this time with more conservative 1/6th octave smoothing. The frequency response is seen to be within +/- 1.5 dB from 28 Hz to 20 kHz. This 1/6th octave smoothed response probably corresponds most closely with what you would actually hear from the system as 1/6th octave corresponds to the "critical bandwidth" of human hearing in the range above 1 kHz. Below 1 kHz the critical bandwidth gets progressively wider as the ear becomes less discerning and greater smoothing could be used without misrepresenting what we would hear. So above 1 kHz we don't want to use any more than 1/6th octave smoothing. Below 1 kHz the 1/3rd octave smoothed response of Figure 5-1 above is more representative of our ear's response.

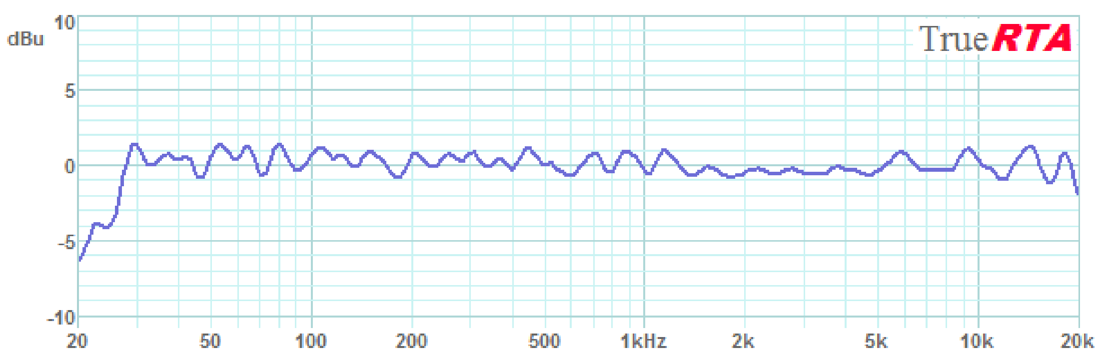


Figure 5-2: The MCLA In-Room Measured Response with 1/6th Octave Smoothing

Figure 5-3 shows the completely unsmoothed spatial average response of the MCLA system. The unsmoothed response is seen to be within about +/- 2.5 dB from 28 Hz to 20 kHz. Keep in mind that loudspeaker manufacturers rarely (if ever) show this high level of detailed data for their commercial loudspeaker systems.

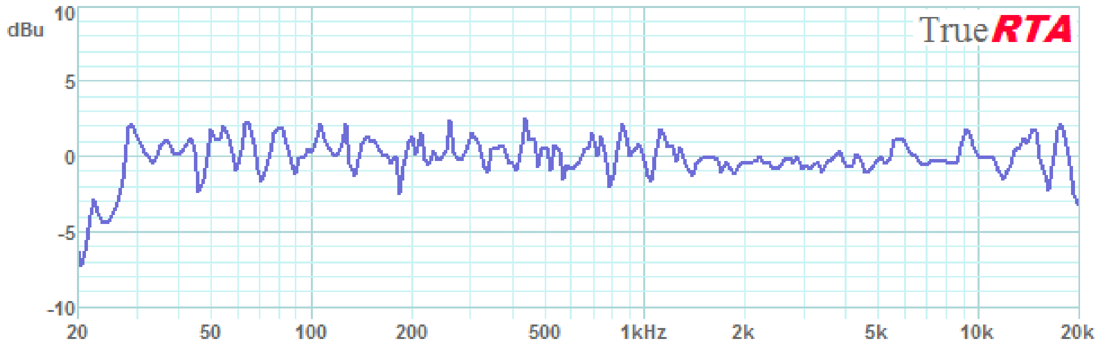


Figure 5-3: The MCLA In-Room Measured Response with No Smoothing

## Un-equalized In-Room Frequency Response

The equalized frequency responses shown in the previous section show the arrays with their equalization switched in as they are normally used. In this section I will show the un-equalized frequency responses in order to understand the required components of the corrective equalization.

It is reasonable to ask what frequency response we expect from the arrays. Based on my understanding of line arrays I expect to see the half space response of the driver modified by the -3 dB per octave slope resulting from the effect of array. There will also be some lumps and bumps resulting from the finite spacing of the drivers. The 3 dB per octave boost in the bass from the array effect may not continue all the way to 20 Hz due to the finite length of the array. Remember, the reflections taper off due to finite absorption of the room surfaces so the bass build is expected to be limited. In Figure 5-4 below we see the single driver's hybrid measured response along with a -3 dB per octave reference slope. If we tilt the single driver response the -3 dB per octave we expect the array to add then we should get a response similar to what we expect from the raw (un-equalized) array. Just for fun I'll add the two responses shown here to get some idea of a "predicted" array response.

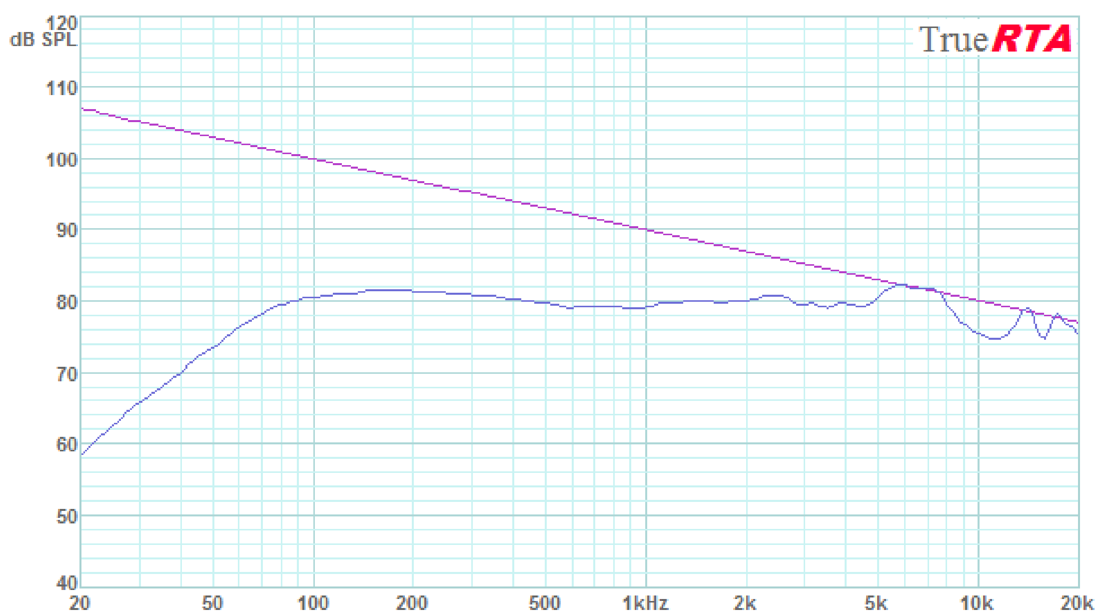


Figure 5-4: The Single Driver Half-Space Response along with a -3 dB/octave Reference

Summing the two responses shown in Figure 5-4 gives the "predicted" response for the array shown in Figure 5-5.

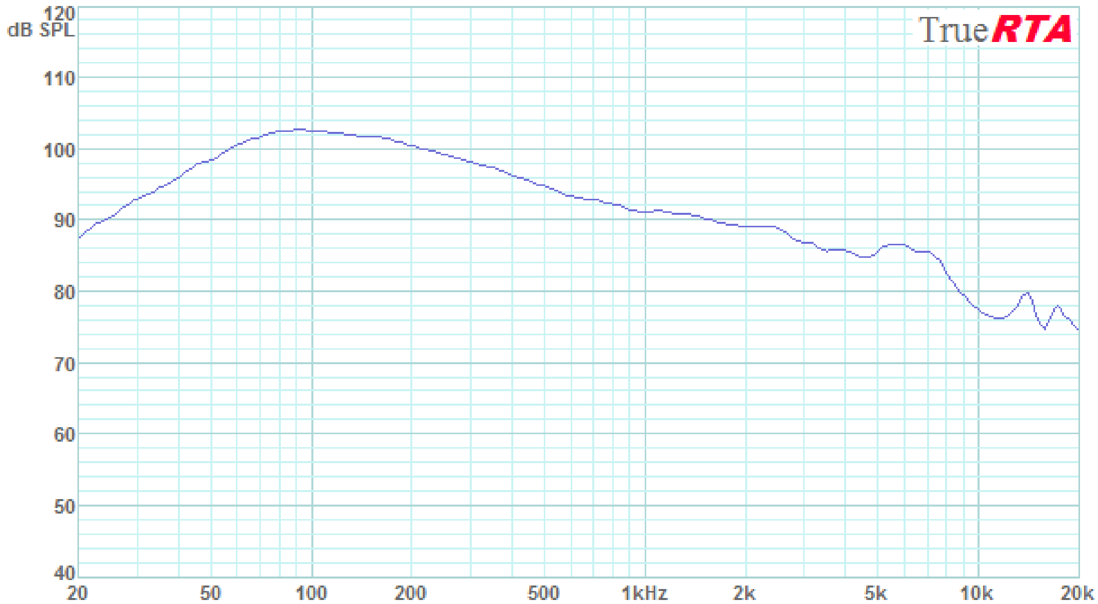


Figure 5-5: Predicted Response of the Array based on the Single Driver Response summed with the -3 dB/octave Reference

In order to restore the flat response of the single driver we would need to equalize the above response with a 3 dB per octave rise. Then, in order to extend the bass response to 30 Hz we would need bass equalization. The bass equalization might take the form of a "Linkwitz Transform" circuit to precisely create a new target response or could be achieved with a general purpose digital EQ such as I am doing.

Finally, here in Figure 5-6 is the un-equalized measured response of the two array prototypes. This response is an average of the two measured responses for each array in-room. Each individual array was measured at sixteen locations around the listening area at a distance from 1 to 3 meters. The individual array responses were smoothed at 1/6th octave before the two were averaged. The similarity to the predicted response is notable, especially regarding the overall -3 dB per octave slope of the array. The low frequency corner is expected to be slightly different than the single driver response because the single driver was measured in a 0.1 cubic foot closed box. The array has .045 cubic feet per driver so the array has a higher closed box Q(tc) than the single driver test system.

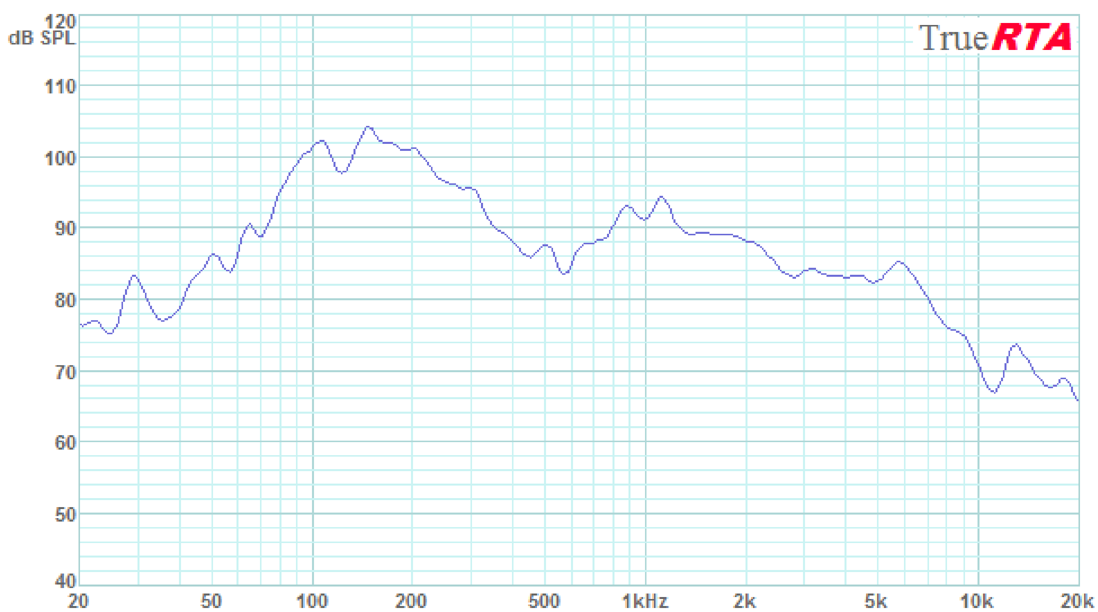


Figure 5-6: The Measured Response of the Un-Equalized Arrays Exhibiting the Expected -3 dB Slope

The frequency response in Figure 5-6 was measured with a drive voltage of 2.309 Vrms which corresponds to 1 Watt into the array's nominal impedance of 5.33 Ohms (proto 1 with 24 drivers). Note the very large build in efficiency in the 100 to 200 octave where the array has an efficiency exceeding 100 dB SPL for 1 Watt over the 1 to 3 meter range. With the array receiving an input of 1 Watt each driver was being driven at 1/24 Watt or .042 Watts. While producing over 100 dB SPL in the 100-200 Hz bass range the drivers are just being tickled a bit with 42 milliwatts of signal. Operating the array at 100 Watts (or just over 4 Watts per driver) would add 20 dB to the above levels. The usual 1W/1meter sensitivity specification of a typical speaker is not appropriate for an array. Instead we would have to indicate sensitivity at a specific frequency or preferably, just show sensitivity a graph as above. Figure 5-7 shows the measured response along with a -3 dB per octave slope for comparison.

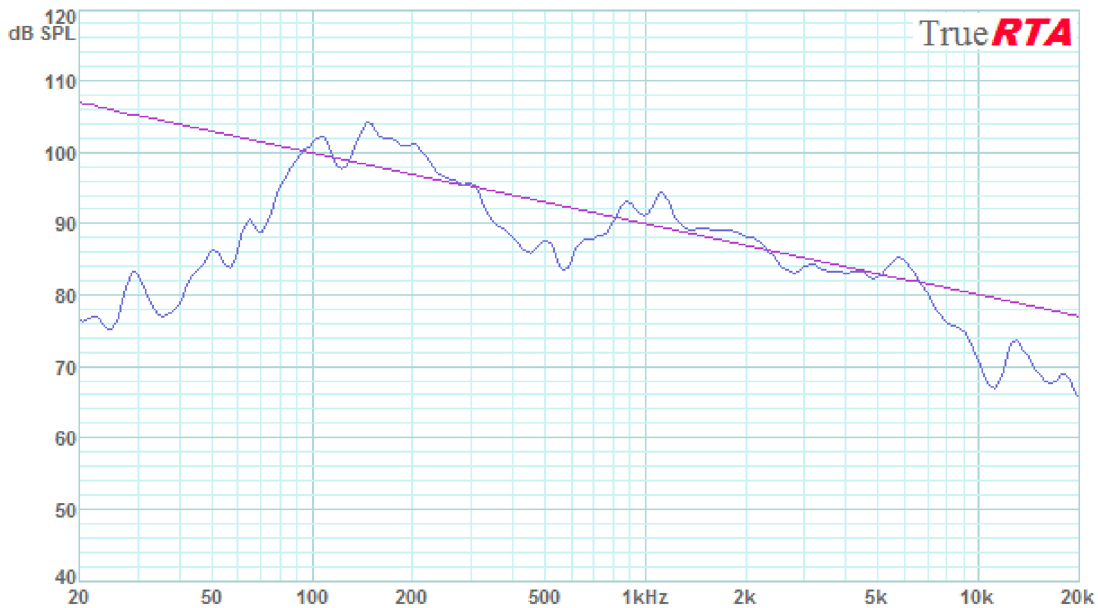


Figure 5-7: The Measured Response, Un-Equalized, along with a Reference -3 dB/octave Slope

All in all the arrays achieve a frequency response much like what is expected based on the single driver measurements. The essential components required of the corrective EQ are a 3 dB upward slope (a blue filter) and EQ to extend the bass response. In practice the final EQ is best achieved by a measuring the response and adjusting the EQ a few times in succession until a sufficiently flat response is obtained.

## Distortion Performance

The spectrum of the completed array was measured with 1 Watt at 1 meter at key frequencies in order to evaluate its distortion performance. The only significant distortion seen was at 2nd, 3rd and 5th harmonics. The relatively benign 2nd harmonic dominates the distortion makeup below 1 kHz. Harmonics above the 5th are very well behaved as they continue to descend in level.

Figure 5-8 shows the spectrum from one array driven at 2.31 Vrms (1 Watt into 5.33 Ohms) with a 100 Hz sine wave. The array creates a very efficient 96 dB SPL for 1 Watt/1meter. The 2nd harmonic, 200 Hz is measured at 41 dB below the 100 Hz fundamental for a distortion level of 0.9%. The 3rd harmonic is -51 dB with respect to the fundamental for a level of 0.3%. The 5th harmonic at 500 Hz is even lower at -70 dB or 0.03%. Higher harmonics continue to descend in level and vanish into the noise floor. The line at 60 Hz is just ambient noise.

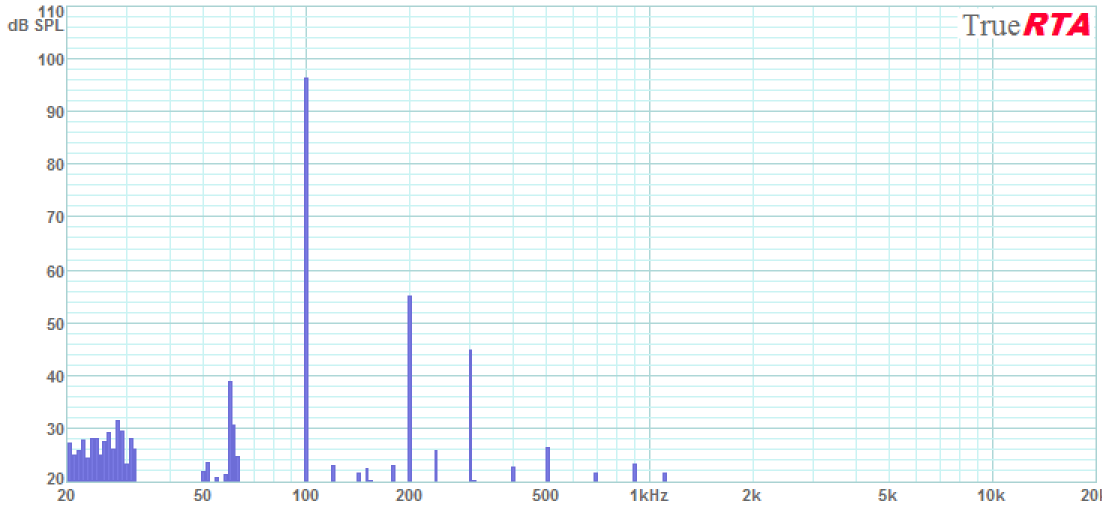



Figure 5-8: The Array's Sine Wave Spectrum at 100 Hz for 1 Watt at 1 meter

Next in Figure 5-9 we see the array at reproducing a 200 Hz sine wave at an incredible 98 dB SPL with only 1 Watt of input. The distortion is quite low compared to any loudspeaker I've measured. The 2nd harmonic at 400 Hz is down 52 dB or 0.25% distortion. The 3rd harmonic (600 Hz) is at -66 dB or 0.05% distortion. The higher harmonics continue to descend in level even further.

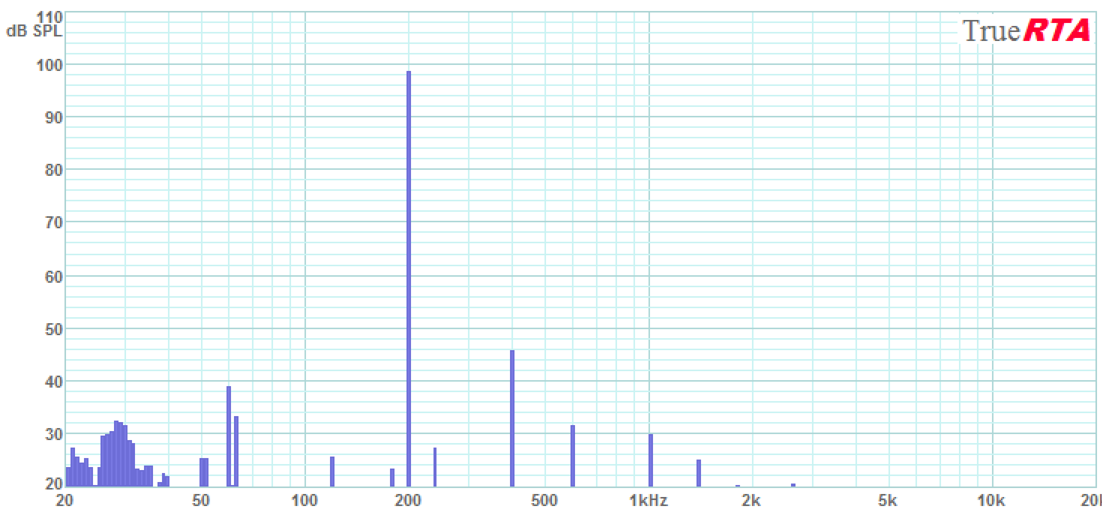



Figure 5-9: The Array's Sine Wave Spectrum at 200 Hz for 1 Watt at 1 meter

Note the results above are for just 1 Watt into the array or just 1/24th of a Watt per driver. If the array were driven at 24 Watts each driver would then get 1 Watt and the output of the array would increase by 14 dB. This means that at 200 Hz a single array would generate 98 + 14 = 112 dB SPL with each driver at 1 Watt (24 Watts total input). Yet the distortion would be similar to a single driver driven at 1 Watt, that is, very low. How many other speakers do you know that can generate 112 dB at 200 Hz with distortion on the order of 0.2%?

The array's performance at 500 Hz is shown in Figure 5-10. Output has fallen to 86 dB SPL for 1 Watt at 1 meter but the distortion continues to be quite low. The 2nd harmonic is at -47 dB or 0.4% distortion. The 3rd harmonic is at -53 dB or 0.2%. The 5th harmonic is at -58 dB or 0.13%. Higher harmonics descend further in level.

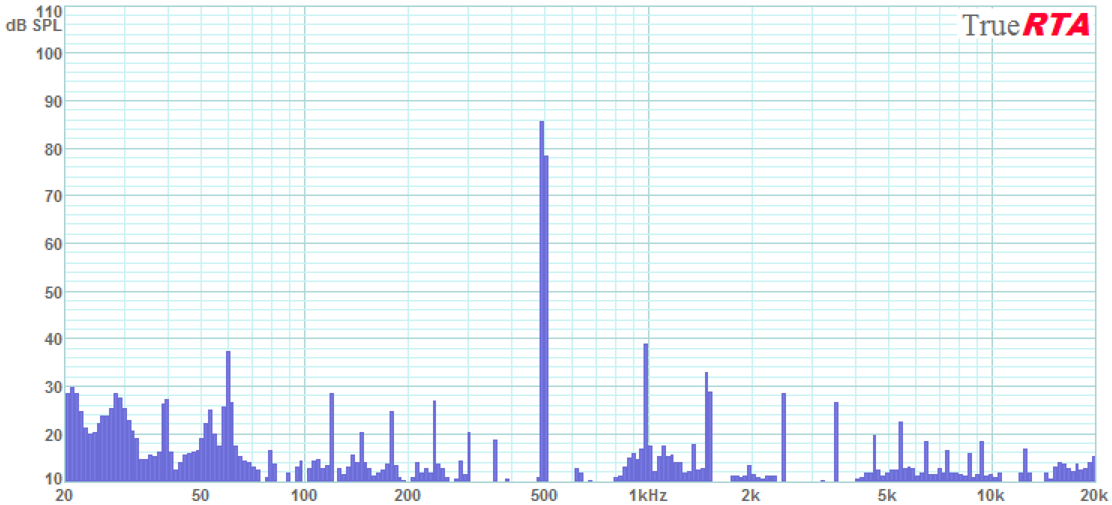


Figure 5-10: The Array's Sine Wave Spectrum at 500 Hz for 1 Watt  at 1 meter

Figure 5-11 shows the array at 1 kHz with 1 Watt of input creating 89 dB SPL. The 2nd harmonic is absent with the 3rd harmonic (3 kHz) at -57 dB (0.14% distortion) compared to the fundamental at 1 kHz. The 5th harmonic is at -67 dB or 0.04%. The 7th harmonic pops up to 0.1% but the 9th and higher continue the fall to lower levels.

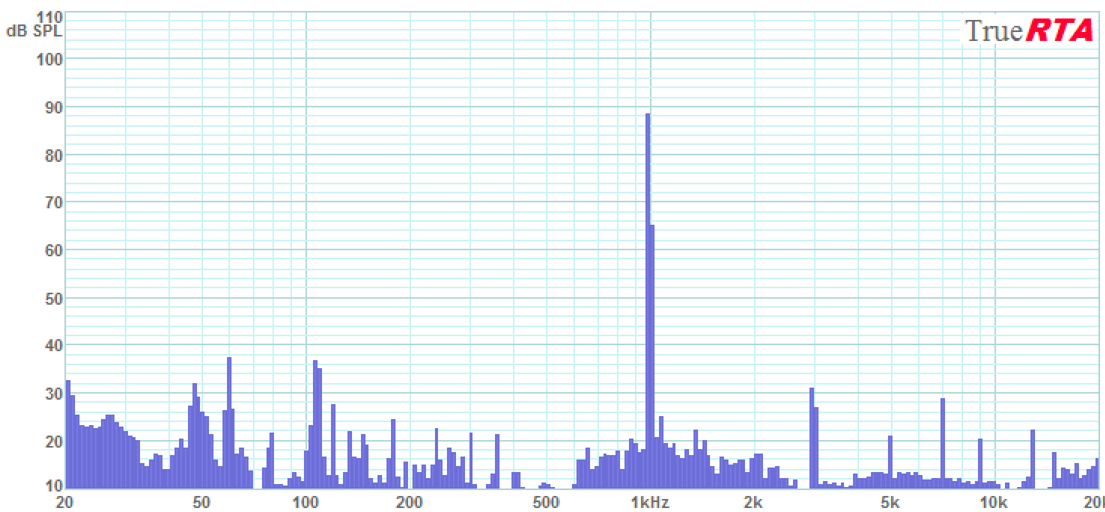


Figure 5-11: The Array's Sine Wave Spectrum at 1000 Hz for 1 Watt  at 1 meter

For a look at the low bass performance here in Figure 5-12 below is the performance at 50 Hz for 1 Watt at 1 meter. The output level is still quite high at 94 dB SPL indicating the strong low frequency boost from the combination of the 24 drivers (prototype 1) and the corner placement. Remember even though the array is putting out 94 dB SPL with 1 Watt it has a power handling capability of 480 Watts or 27 dB greater than the 94 dB SPL we see here. This suggests the array could generate 121 dB SPL at 50 Hz at its max rated power of 240 Watts. There is no shortage of low frequency output capability. At 50 Hz the 2nd harmonic is -41 dB down for 0.9% distortion. The 3rd harmonic increases to -37 dB or 1.4% distortion. Higher harmonics are much lower in level.

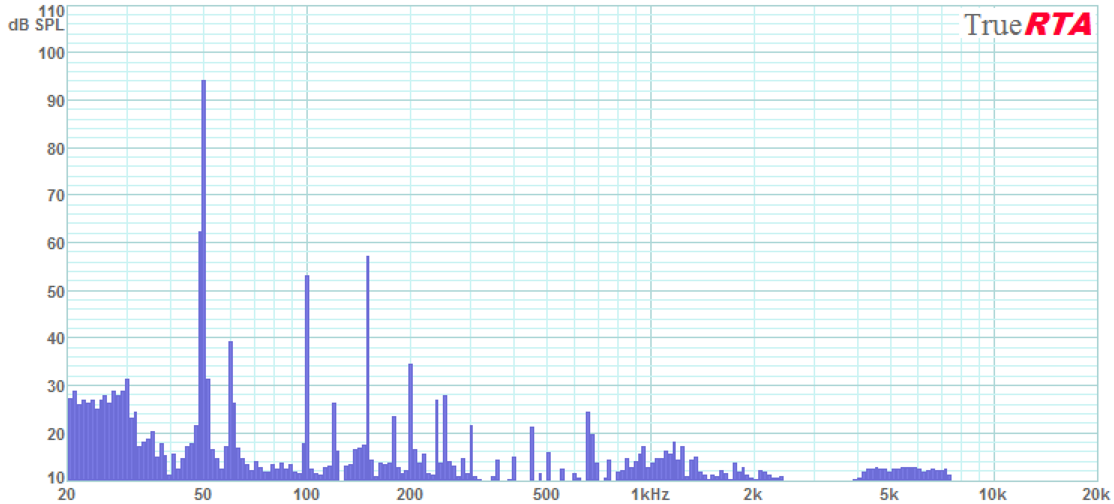


Figure 5-12: The Array's Sine Wave Spectrum at 50 Hz for 1 Watt at 1 meter

As a low frequency stress test I tried to drive the array to 10% distortion at 50 Hz. This is one measure of the maximum operating level of a speaker in the low bass range. Figure 5-13 shows the array operating at the highest level I could tolerate before everything in the room started vibrating loudly. The system reached 113 dB SPL at 50 Hz with 65 Watts of drive level (18.7 Vrms into 5.33 Ohms). The dominant distortion is 2nd harmonic at -26 dB or 5% distortion. The 3rd harmonic is at -29 dB or 3.5%. The 5th harmonic was only at -48 dB or 0.4%. This is excellent high SPL low bass performance! A single array can drive the ROOM to the limit of what it can take with just 65 Watts at 50 Hz. At the highest levels a person (or room) can stand (113 dB SPL in this case) the distortion does not exceed 5% at 50 Hz.

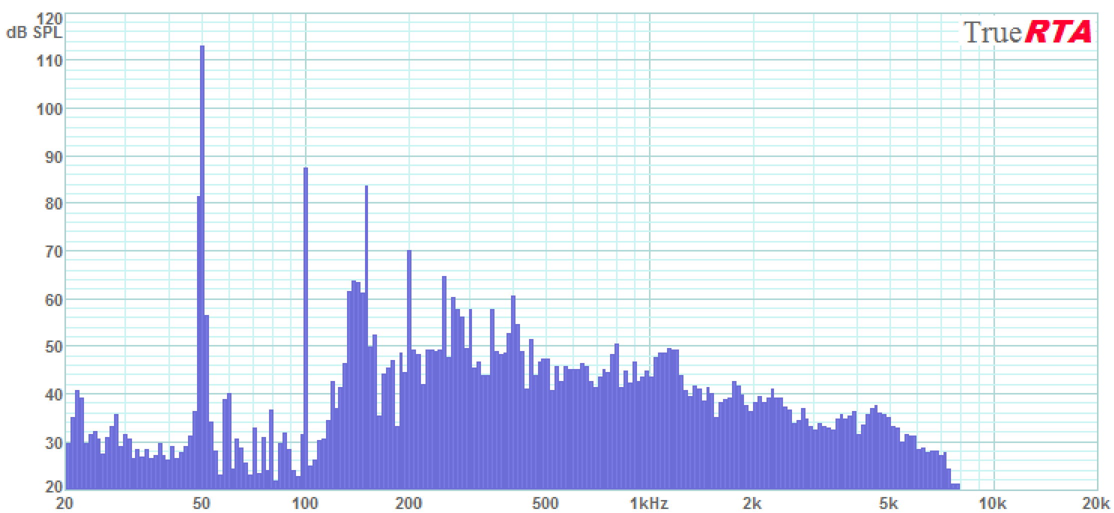


Figure 5-13: The Array's Sine Wave Spectrum at 50 Hz for 65 Watts Input at 1 meter

The distortion test show a speaker system capable of very high output levels with very low distortion. In all cases the distortion is dominated by the low order harmonics and is frequently dominated by the (relatively benign) 2nd harmonic. Distortion should not be a concern at even the highest sound levels...even for a single enclosure.

The distortion tests confirm what I heard while stress testing a single drive unit with my guitar. Playing my telecaster electric guitar through the corner line arrays provides a pristine clean sound in the room at any level I wish. If I crank it up they can really rock without any hint of stress. Palm muted guitar delivers an awesome thump. Bass guitar likewise sounds fantastic through the arrays. When rehearsing with my buddy I see the average SPL easily breaking 90-95 dB SPL with peaks up to 110 dB SPL. I hear no sound of stress from the system and my small power amp remains cool to the touch. Microphone feedback is less of a problem with the new speakers as we seem to be able to get more level from the vocals before feedback sets in.

# 6. Line Array References and Links

## Line Array Technical References:

Smith, David L. , "Discrete-Element Line Arrays - Their Modeling and Optimization", J. Audio Eng. Soc., vol.45, no.11, 1997 November

Stanley P. Lipshitz and John Vanderkooy, “The Acoustic Radiation of Line Sources of Finite Length”, AES Preprint No. 2417, presented at the 81st AES Convention, November 12-16, 1986

Taylor, Paul H., "The Line-Source Loudspeaker and Its Applications," British Kinematography (J. Brit. Kinematograph Soc.), vol. 44, pp. 64-83, 1964 March

L. L. Beranek, Acoustics (McGraw-Hill, New York, 1954)

Available online:

Jim Griffin's paper, "Design Guidelines for Practical Near Field Line Arrays", dated 2003
http://www.audioroundtable.com/misc/nflawp.pdf

Roger Russell's Sound Column History
http://www.roger-russell.com/columns/columns.htm

NOTE: The following paper discusses line arrays in the FAR FIELD as used for large scale concert sound systems. This analysis does not apply to the MCLA which is used in the near field but is included for those wishing a wider study of line array loudspeaker systems.

Mark S. Ureda, “Line Arrays: Theory and Applications”, presented at the 110th AES Convention, Amsterdam, May 12-15,2001.
http://www.jblpro.com/products/Tour/Vertec/support/pdf/AES%20May%2001%20Ureda%20Line%20Arrays.pdf

## Image Method References:

Available online:

C.F. Eyring, "Reverberation Time in 'Dead' Rooms," *J. Acoust. Soc. Am.*, (1930, Jan.).

J. B. Allen and D. A. Berkley, "Image method for efficiently simulating small-room acoustics." *J. Acoust. Soc. Am.*, vol. 65, no. 4, pp. 943–950 (1979).

Richard V. Waterhouse, "Output of a Sound Source in a Reverberation Chamber and Other Reflecting Environments." *J. Acoust. Soc. Am.*, vol. 30, no. 1 (1958).

Advances in Edge Diffraction Modeling for Virtual-Acoustic Simulations
(PhD Dissertation) by Paul Calamia
http://www.cs.princeton.edu/~pcalamia/thesis/calamia-phd.pdf
(see section 2.1.2 for discussion of the image method)

**Other Loudspeaker References:**

Floyd D. Toole, "Sound Reproduction, The Acoustics and Psychoacoustics of Loudspeakers and Rooms.", Focal Press, 2008

**Other DIY Line Array Projects for Residential Use:**

Darren Kuzma's Kuze 3201 line arrays using 32 Tang Band 2" drivers per side:
http://www.partsexpress.com/projectshowcase/Kuze3201/Kuze3201.html

Louis Coraggio's Thin Blue Line Array:
http://www.lonesaguaro.com/speakers/array/array.htm

Bill Fiztmaurice's TLAH 2-way line array
http://www.billfitzmaurice.com/TLAH.html

**Commercial Line Array Speakers for Residential Use:**

PipeDreams (2-way plus sub) line array systems: Suggested retail prices from $20,000 - $80,000
http://www.nearfieldacoustics.com

Roger Russell's IDS-25 full range line array: Priced at $18,900
http://www.ids25.com/

Rick Craig's Selah Audio: Line arrays from $7,990
http://www.selahaudio.com/id73.html

Dali Megaline: Priced at $40,000
http://www.dali.dk/display_content.php/INT/speakers.html/120/912

Monacor Lightning (kit):
http://www.lautsprechershop.de/hifi/index_en.htm?/hifi/lightning_en.htm

Scaena Iso-Linear Array:  A line array with price tag ranging from $44,000 to $75,000
http://www.scaena.com/iso-linear%20array.htm

ClaireAudient Line Source Array: A full range bi-pole line array of 3" drivers, $35,000 (16 driver version)
http://www.audience-av.com/loudspeakers/index.php

**FREE Array Simulation Software:**

Vertical Polar Response Line Array at the FRD Consortium:
http://www.pvconsultants.com/audio/radiation/vpr.htm

# Appendix 1. The MLA 8/16 Line Array (circa 1980)

Figure A1-1: The Original Murphy-Line-Array 8/16 designed in 1980 (MLA 8/16)

I designed the MLA 8/16 Line Array System around 1980 when I was working as the Chief Engineer for Ford Audio in Oklahoma City, OK. I was doing a lot of speaker design (using Don Keele's calculator method) and developed an interest in linear arrays of loudspeakers. The prototype seen here was built in the Ford Audio wood shop where a small number of similar systems (half this height and a more compact enclosure) were also built. Those systems were used in two long lines above the dance floor at a large new club where the company was installing the audio systems. The MLA 8/16 system uses eight 8" woofers and sixteen 1" dome tweeters made by Peerless.

It is fairly normal that anyone who auditions these speakers comes away positively impressed by the listening experience. They play at any sound level with absolutely no hint of stress.

The configuration has changed several times over the last 29 years. In the early years they were bi-amped with 3rd order Butterworth crossovers at 1.5 kHz. That eventually changed to 6th order Linkwitz-Riley crossovers. During one period they were used with passive crossovers, digital delay units and multiple stereo amplifiers for array focusing experiments.

These speakers were moved from Oklahoma City to San Diego, CA in 1981 and from San Diego to Knoxville, TN in 1998 where they remain in use today at my home on Norris Lake.

# Appendix 2. Public Disclosure

## PUBLIC DISCLOSURE OF INVENTION

Title of the Invention: **Murphy Corner-Line-Array Loudspeaker**


Inventor: **John L. Murphy**
**Physicist/Audio Engineer**
**Andersonville, TN USA**


Summary of the Invention:

The corner line array loudspeaker consists of a line array loudspeaker designed specifically to optimize the placement of the speaker's reflected images when placed in the corner of a room. More precisely, the enclosure is designed to fit in the corner in such a way as to smoothly merge the real enclosure with its reflections into the acoustic equivalent of an octagonal enclosure located in free space. See Figure 1.1 below for the details.

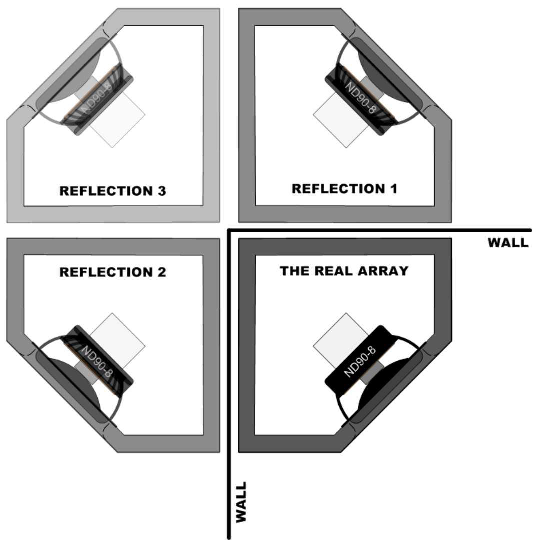



Figure A2-1: A Single Murphy Corner-Line-Array As Viewed From Above Along With Three Reflections

The "corner-line-array" loudspeaker constitutes an improvement over previous point, line and planar loudspeakers by combining the reflected acoustic images with the direct sound radiation in such a way as to effectively create an infinitely long line source radiator where the side wall and front wall reflections of the line are uniquely located in very close proximity to the real speaker system by virtue of the corner placement. When viewed from above along with its reflections, the unique corner-fitting shape is seen to join with its reflected images to form an overall octagonal shaped line source with loudspeakers on four sides.

As a direct result of controlling the precise reflection placement, the in-room frequency response that is achieved by the corner-line-array loudspeaker system is more consistent as the listener moves about the room than previous types of loudspeaker systems. It is also more consistent from room to room than other types of loudspeakers. When equalized to have a flat frequency response or other targeted response (such as the X-curve) the Murphy Corner-Line-Array provides a consistently accurate frequency response about much of the listening room and from room to room. Previous loudspeakers experienced highly variable frequency response because of the lack of standardized placement in the room and the resulting variable and widely spaced room reflections.

This invention is being disclosed publically in order to deliberately and immediately place the invention fully in the public domain for the benefit of all.

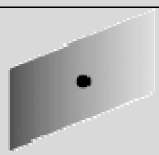
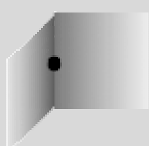
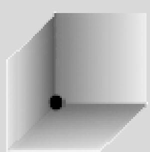
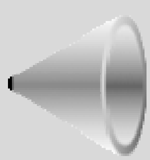
John L. Murphy
9July2009

(a signed and witnessed disclosure is on file at True Audio)

# Appendix 3. Loudspeaker Spatial Loading

The frequency response of a loudspeaker system depends on how the system is "loaded" in much the same way that the output from a power amp depends on the load impedance. The power amp drives an electrical load specified in Ohms. The speaker drives an acoustic load specified as a "solid angle" measured in steradians. The most frequently specified speaker load, and the one typically specified in the loudspeaker literature, is half space.

The following table presents a summary of fractional space acoustic loads:

| | |
|---|---|
|  | **Full Space = 4 pi steradians**<br>This represents radiation into free space, that is, in the open with no walls, floor or surfaces nearby. |
|  | **Half Space = 2 pi steradians**<br>(the most frequently specified speaker load)<br>If you imagine putting a speaker on an infinitely large baffle then the front of the speaker would be radiating into half space. The plane divides all of space into two halves. |
|  | **Quarter Space = pi steradians**<br>Imagine the speaker placed at the intersection of two infinitely large perpendicular planes. Approximated by the intersection of two walls. The two planes divide all of space into four quarters. |
|  | **Eighth Space = pi/2 steradians**<br>Now, imagine the speaker placed at the intersection of three walls, such as in the corner of a room and at the ceiling. The three planes divide all of space into eight parts. |
|  | **Smaller Acoustic Load Angles**<br>In horn type loudspeakers the acoustic radiation from the horn driver is funneled into an even smaller solid angle than "eighth space" and achieves increased power output as a result of the horn load. The smaller solid angles constitute a more "stiff" acoustic load and draw more power from the source analogous to the way that a lower impedance speaker draws more power from the amplifier driving it. Indeed, horns can be called "acoustic transformers" for the way that they can match the impedance of the driving source to the impedance of the load to effect maximum power transfer. |

The notion of the acoustic load that a speaker is driving is very important to understanding the reproduction of sound and especially important if you are trying to understand the net output of a speaker into a listening room. Speakers rarely see a constant acoustic load. It is almost always frequency dependent.

Here is a sketch of how it plays out in a typical listening room. Regardless of how it is placed in the room, a typical hi-fi speaker system sees a half space load at frequencies from the midrange up as a result of the drivers being placed on the baffle (front face) of the box. At high frequencies the drivers only radiate to the front of the box. Starting in the midrange (depending on the baffle size) the system shifts from radiation into half space to radiation into full space at lower frequencies. Another way to say this is that the bass radiates in all directions . . .even to the back. This transition from half space loading to full space loading results in what is commonly called the "6 dB baffle step", or "diffraction loss", and results in a 6 dB loss of bass with respect to the midrange output when a speaker enclosure is placed in a full space environment.

At even lower frequencies, say from 100 Hz down, the wavelength of the radiated sound is such that the walls and cavity (the listening room) begin to load the system in a way that results in a complex load that is less than half space and results in increased output from the system. This effect in the bass region is called variously "room gain", "boundary effects", "room resonance", or "cavity effect". In general, real speaker enclosures and listening rooms present a frequency dependent radiation impedance to the loudspeaker.

# Appendix 4. Point, Line and Plane Sound Sources

It is generally recognized in the audio community that, within their near-field, the SPL of line sources falls at a rate of 3 dB per doubling of listening distance versus point sources which fall off at a rate of 6 dB per doubling of distance. As a practical example consider the observed behavior of the noise pollution from a highway. Folks living near highways are the unfortunate victims of the highway noise only falling at only 3dB per doubling of distance from the highway. That's because the highway noise source precisely fits the model of an infinite line source. A pretty good approximation really! The engineers who battle noise know this behavior all too well. But let's look even further . . . at the behavior of a *plane* source. . .

**Point Sources**

The SPL from an ideal point source radiator falls at the rate of 6 dB per doubling of distance. The Intensity of sound from the point source falls off as the inverse square of the distance. This is known as the inverse square law. The energy radiated from the point source is evenly distributed over the surface of an expanding sphere. The surface area of the sphere is inversely proportional to the distance (radius of the sphere) squared.

**Infinite Line Sources**

The SPL from an infinitely long line source falls off at a rate of 3 dB per doubling of distance. This is because the energy distribution is now over the surface of a cylinder, rather than a sphere as in the case of the point source. Because the surface area of the expanding cylinder is inversely proportional to distance, NOT distance squared, it follows that the energy density falls simply with distance from the source, rather than distance squared.

**Infinite Plane Sources**

Imagine an infinitely large flat surface that radiates sound. The SPL from an infinitely large plane sound source is constant with distance from the source. The energy distribution from the source is now over the surface of another plane some distance from the source. As the wave propagates it does not expand but rather continues to pass through precisely the same area as the source itself. Therefore the energy density at any point in space is equal to the energy density at the source plane itself. The SPL is constant everywhere in the vicinity of a plane radiator!

**Some Comments on Finite Sources**

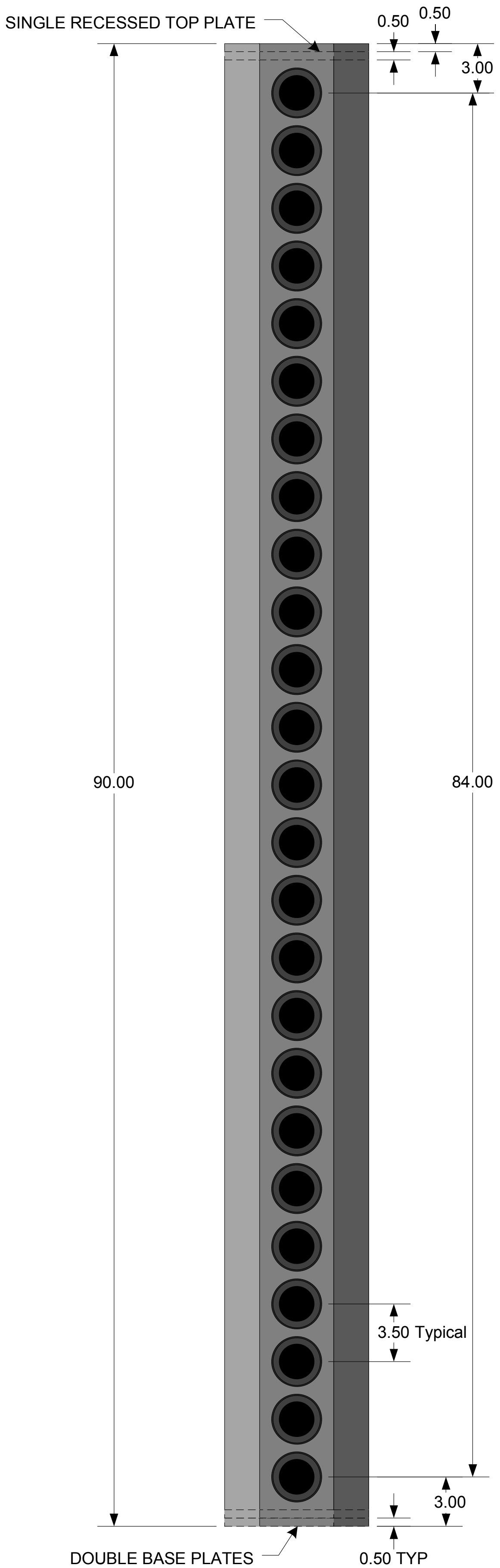
It is important to note that we have been discussing the behaviors of an *infinitesimal* point source and *infinite* line and plane sources. Real sound sources with finite geometries will exhibit different characteristics from these ideal extremes.

A **finite line source** will behave more as an infinite line when the observation point is close to the line compared to its length (near-field). At greater distances, that is in the far-field, the source looks more like a point radiator and the SPL will fall at 6 dB per doubling of distance.

A **finite plane source** likewise will exhibit constant SPL with distance only when the observation point is very close to the plane compared to the dimensions of the plane. At larger distances the finite plane source behaves more like a point radiator and the SPL will fall at 6 dB per doubling of distance.

# Appendix 5. Construction Documentation

Following are drawings detailing MCLA enclosure construction, wiring and essential EQ adjustments. The enclosure plans are for “Prototype 3” the latest incarnation of the system employing 25 drivers per enclosure. If the 90 inch Proto 3 is a too tall for your room then you might consider building the slightly shorter Proto 2 which uses 24 drivers. Plans are available at the project site at: http://www.trueaudio.com/array



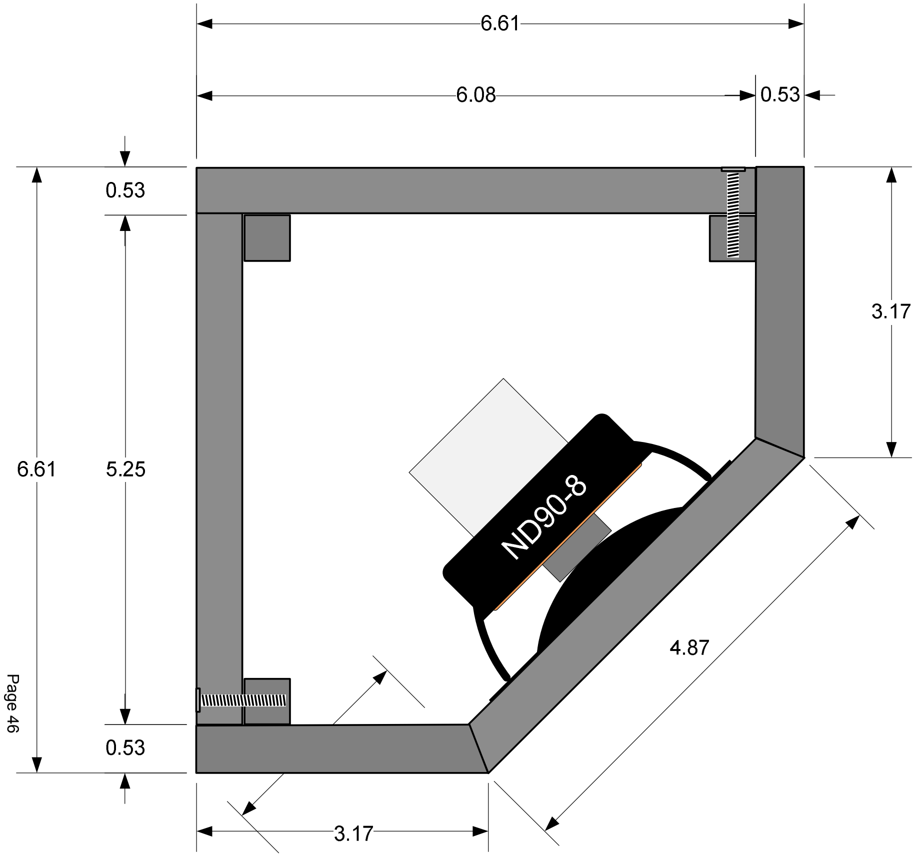

Drivers are Rear Mounted
Drivers are Dayton Audio ND90-8
25 Full Range Drivers are Employed

True Audio
387 Duncan Lane
Andersonville, TN 37705
www.trueaudio.com

The Murphy Corner-Line-Array
An Open Loudspeaker Design Project

Prototype 3, Front View

| | |
|---|---|
| SCALE: | REV BY: John L. Murphy |
| PART NO.: | REV: - REV DATE: 26Jun10 |



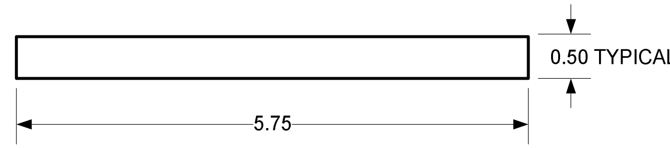


Internal area ~= 22.28 square inches
Volume per 3.5" length ~= .0451 cubic feet

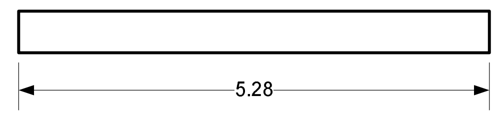
Material is .50" MDF

| | |
|---|---|
| True Audio | 387 Duncan Lane<br>Andersonville, TN 37705<br>www.trueaudio.com. |
| The Murphy Corner-Line-Array<br>An Open Loudspeaker Design Project | |
| Prototype 3, Top View Detail | |
| SCALE: | REV BY: John L. Murphy |
| PART NO: | REV: - REV DATE: 26Jun10 |

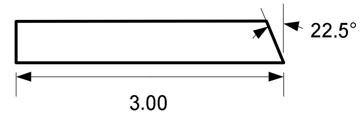
1 x Rear Right: 5.75" x 0.5" x 89.5"



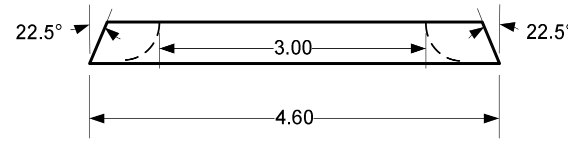

1 x Rear Left: 5.25" x 0.5" x 89.5"



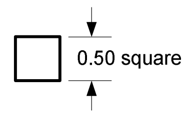

2 x Front L/R Face: 3.00" x 0.5" x 90"



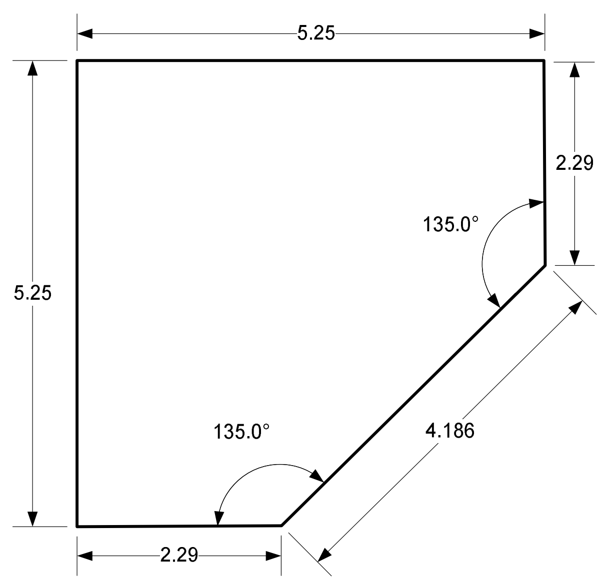

1 x Front Baffle: 4.60 " x 0.5" x 90"




3 x Brace Strip: 0.5" x 0.5" x 88"




2 x Top/Bottom Plate: 0.5" Thick




3. The Front Baffle is to have 25x 3.0" diameter holes on 3.5" centers along the center line and equally spaced from each end. Each 3" hole will have a radius (.25") on the front edge (wide side) of the baffle.

2. Quantities are specified for one complete enclosure

1. Material is 0.5" MDF

True Audio

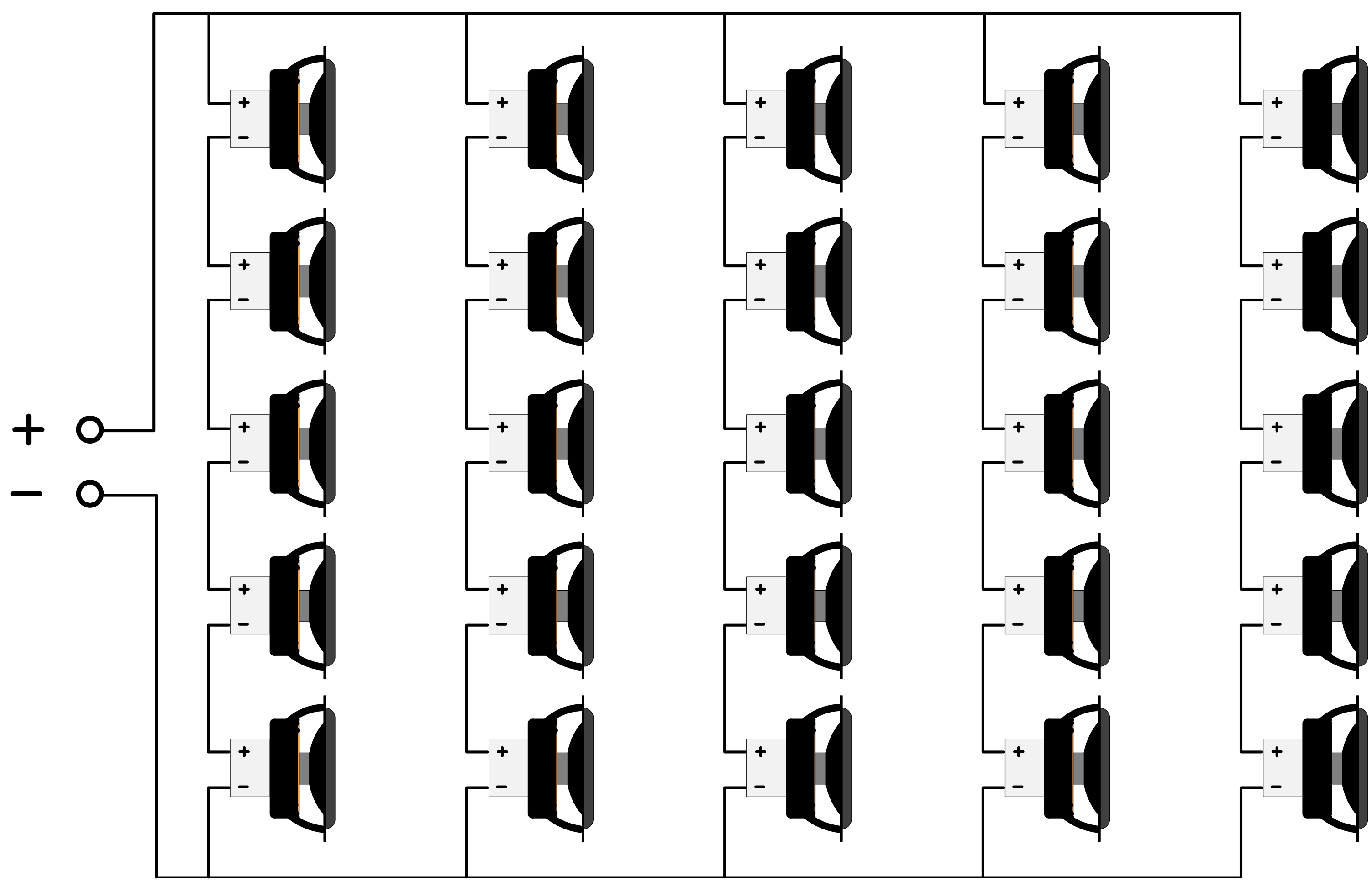
387 Duncan Lane
Andersonville, TN 37705
www.trueaudio.com

The Murphy Corner-Line-Array

An Open Loudspeaker Design Project

Prototype 3, Parts List

| | |
|---|---|
| SCALE: | REV BY: John L. Murphy |
| PART NO: | REV: - REV DATE: 26Jun10 |




1. 25x Dayton Audio ND90-8 Drivers, 8 Ohms each, 8 Ohms Net Load

| | |
|---|---|
| True Audio | 387 Duncan Lane<br>Andersonville, TN 37705<br>www.trueaudio.com |
| Murphy Corner-Line-Array | |
| Wiring Diagram for 25 Drivers | |
| SCALE: | REV BY: John L. Murphy |
| PART NO.: | REV: - REV DATE: 26Jun10 |

# MCLA Equalization

## Proto Unit #2 at Left Side (windows at each side)

1-Oct-09 1:21 PM　　Note: values in **Bold** are calculated

**GRAPHIC EQ SETTINGS**

| | FLAT | Small Room X-Curve | X CURVE |
|---|---|---|---|
| Frequency (Hz) | EQ SETTINGS (dB) | EQ SETTINGS (dB) | EQ SETTINGS (dB) |
| 20 | 4.0 | **4.0** | **4.0** |
| 25 | 7.0 | **7.0** | **7.0** |
| 31.5 | 3.0 | **3.0** | **3.0** |
| 40 | 5.0 | **5.0** | **5.0** |
| 50 | -0.5 | **-0.5** | **-0.5** |
| 63 | -6.5 | **-6.5** | **-6.5** |
| 80 | -8.5 | **-8.5** | **-8.5** |
| 100 | -15.0 | **-15.0** | **-15.0** |
| 125 | -10.5 | **-10.5** | **-10.5** |
| 160 | -14.5 | **-14.5** | **-14.5** |
| 200 | -15.0 | **-15.0** | **-15.0** |
| 250 | -11.0 | **-11.0** | **-11.0** |
| 315 | -10.0 | **-10.0** | **-10.0** |
| 400 | -4.0 | **-4.0** | **-4.0** |
| 500 | -9.0 | **-9.0** | **-9.0** |
| 630 | -8.0 | **-8.0** | **-8.0** |
| 800 | -9.0 | **-9.0** | **-9.0** |
| 1000 | -7.5 | **-7.5** | **-7.5** |
| 1250 | -3.5 | **-3.5** | **-3.5** |
| 1600 | -3.5 | **-3.5** | **-3.5** |
| 2000 | -3.0 | **-3.0** | **-3.0** |
| 2500 | -1.0 | **-1.5** | **-2.0** |
| 3150 | -1.0 | **-2.0** | **-3.0** |
| 4000 | 1.5 | **0.0** | **-1.5** |
| 5000 | 3.0 | **1.0** | **-1.0** |
| 6300 | 2.5 | **0.0** | **-2.5** |
| 8000 | 9.5 | **6.5** | **3.5** |
| 10000 | 14.5 | **11.0** | **7.5** |
| 12500 | 9.5 | **5.5** | **1.5** |
| 16000 | 15.0 | **10.5** | **6.0** |
| 20000 | 15.0 | **10.0** | **5.0** |

**X-CURVE ADJUSTMENTS**

| Small Room X-Curve (dB) | X-Curve (dB) |
|---|---|
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| -0.5 | -1 |
| -1 | -2 |
| -1.5 | -3 |
| -2 | -4 |
| -2.5 | -5 |
| -3 | -6 |
| -3.5 | -7 |
| -4 | -8 |
| -4.5 | -9 |
| -5 | -10 |

Note: the parametric EQ settings are common to all three graphic EQ setups.

| | PARAMETRIC EQ SETTINGS | | |
|---|---|---|---|
| Parametric Band | Freq (Hz) | BW (octaves) | Gain (dB) |
| 1 | 35.6 | 1/4. | 4.5 |
| 2 | 44.8 | 1/5. | -7.5 |
| 3 | 563 | 1/4. | 5 |
| 4 | 8933 | 1/4. | -3.5 |
| 5 | 11246 | 1/4. | 4 |
| 6 | 109 | 1/4. | -2.5 |
| 7 | 15704 | 1/4. | 1.5 |
| 8 | 180 | 1/4. | 2 |
| 9 | 17825 | 1/6. | -3 |
| 10 | 20000 | 1/2. | 3.5 |

# MCLA Equalization

## Proto Unit #1 at Right side (door at left, window at right)

3-Oct-09 1:30 PM Note: values in **Bold** are calculated

**GRAPHIC EQ SETTINGS**

| | FLAT | Small Room X-Curve | X CURVE |
|---|---|---|---|
| Frequency (Hz) | EQ SETTINGS (dB) | EQ SETTINGS (dB) | EQ SETTINGS (dB) |
| | | | |
| 20 | 4.0 | **4.0** | **4.0** |
| 25 | 8.5 | **8.5** | **8.5** |
| 31.5 | 3.0 | **3.0** | **3.0** |
| 40 | 6.5 | **6.5** | **6.5** |
| 50 | -2.0 | **-2.0** | **-2.0** |
| 63 | -6.5 | **-6.5** | **-6.5** |
| 80 | -8.0 | **-8.0** | **-8.0** |
| 100 | -14.5 | **-14.5** | **-14.5** |
| 125 | -9.5 | **-9.5** | **-9.5** |
| 160 | -14.5 | **-14.5** | **-14.5** |
| 200 | -15.0 | **-15.0** | **-15.0** |
| 250 | -9.5 | **-9.5** | **-9.5** |
| 315 | -9.5 | **-9.5** | **-9.5** |
| 400 | -6.0 | **-6.0** | **-6.0** |
| 500 | -5.0 | **-5.0** | **-5.0** |
| 630 | -3.0 | **-3.0** | **-3.0** |
| 800 | -5.5 | **-5.5** | **-5.5** |
| 1000 | -2.0 | **-2.0** | **-2.0** |
| 1250 | -5.0 | **-5.0** | **-5.0** |
| 1600 | -3.5 | **-3.5** | **-3.5** |
| 2000 | -3.0 | **-3.0** | **-3.0** |
| 2500 | 1.0 | **0.5** | **0.0** |
| 3150 | 0.0 | **-1.0** | **-2.0** |
| 4000 | 3.0 | **1.5** | **0.0** |
| 5000 | 4.5 | **2.5** | **0.5** |
| 6300 | 3.0 | **0.5** | **-2.0** |
| 8000 | 12.0 | **9.0** | **6.0** |
| 10000 | 15.0 | **11.5** | **8.0** |
| 12500 | 11.0 | **7.0** | **3.0** |
| 16000 | 15.0 | **10.5** | **6.0** |
| 20000 | 15.0 | **10.0** | **5.0** |

**X-CURVE ADJUSTMENTS**

| Small Room X-Curve (dB) | X-Curve (dB) |
|---|---|
| | |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| -0.5 | -1 |
| -1 | -2 |
| -1.5 | -3 |
| -2 | -4 |
| -2.5 | -5 |
| -3 | -6 |
| -3.5 | -7 |
| -4 | -8 |
| -4.5 | -9 |
| -5 | -10 |

Note: the parametric EQ settings are common to all three graphic EQ setups.

| | PARAMETRIC EQ SETTINGS | | |
|---|---|---|---|
| Parametric Band | Freq (Hz) | BW (octaves) | Gain (dB) |
| 1 | 35.6 | 1/4. | 6 |
| 2 | 44.8 | 1/4. | -5.5 |
| 3 | 563 | 1/4. | 3.5 |
| 4 | 8933 | 1/4. | -3.5 |
| 5 | 11246 | 1/4. | 5 |
| 6 | 276 | 1/4. | 1 |
| 7 | 15704 | 1/6. | 4.5 |
| 8 | 180 | 1/6. | 5.5 |
| 9 | 17825 | 1/4. | -1.5 |
| 10 | 20000 | 1/2. | 5 |

# MCLA Equalization

## Average EQ for Proto Unit #1 at Right and Unit #2 at Left

1-Oct-09 Note: values in **Bold** are calculated

3-Oct-09

**GRAPHIC EQ SETTINGS**

| | FLAT | Small Room X-Curve | X CURVE |
|---|---|---|---|
| Frequency (Hz) | EQ SETTINGS (dB) | EQ SETTINGS (dB) | EQ SETTINGS (dB) |
| 20 | 4.0 | **4.0** | **4.0** |
| 25 | 7.8 | **7.8** | **7.8** |
| 31.5 | 3.0 | **3.0** | **3.0** |
| 40 | 5.8 | **5.8** | **5.8** |
| 50 | -1.3 | **-1.3** | **-1.3** |
| 63 | -6.5 | **-6.5** | **-6.5** |
| 80 | -8.3 | **-8.3** | **-8.3** |
| 100 | -14.8 | **-14.8** | **-14.8** |
| 125 | -10.0 | **-10.0** | **-10.0** |
| 160 | -14.5 | **-14.5** | **-14.5** |
| 200 | -15.0 | **-15.0** | **-15.0** |
| 250 | -10.3 | **-10.3** | **-10.3** |
| 315 | -9.8 | **-9.8** | **-9.8** |
| 400 | -5.0 | **-5.0** | **-5.0** |
| 500 | -7.0 | **-7.0** | **-7.0** |
| 630 | -5.5 | **-5.5** | **-5.5** |
| 800 | -7.3 | **-7.3** | **-7.3** |
| 1000 | -4.8 | **-4.8** | **-4.8** |
| 1250 | -4.3 | **-4.3** | **-4.3** |
| 1600 | -3.5 | **-3.5** | **-3.5** |
| 2000 | -3.0 | **-3.0** | **-3.0** |
| 2500 | 0.0 | **-0.5** | **-1.0** |
| 3150 | -0.5 | **-1.5** | **-2.5** |
| 4000 | 2.3 | **0.8** | **-0.8** |
| 5000 | 3.8 | **1.8** | **-0.3** |
| 6300 | 2.8 | **0.3** | **-2.3** |
| 8000 | 10.8 | **7.8** | **4.8** |
| 10000 | 14.8 | **11.3** | **7.8** |
| 12500 | 10.3 | **6.3** | **2.3** |
| 16000 | 15.0 | **10.5** | **6.0** |
| 20000 | 15.0 | **10.0** | **5.0** |

**X-CURVE ADJUSTMENTS**

| Small Room X-Curve (dB) | X-Curve (dB) |
|---|---|
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| 0 | 0 |
| -0.5 | -1 |
| -1 | -2 |
| -1.5 | -3 |
| -2 | -4 |
| -2.5 | -5 |
| -3 | -6 |
| -3.5 | -7 |
| -4 | -8 |
| -4.5 | -9 |
| -5 | -10 |

Note: the parametric EQ settings are common to all three graphic EQ setups.

**PARAMETRIC EQ SETTINGS**

| Parametric Band | Freq (Hz) | BW (octaves) | Gain (dB) |
|---|---|---|---|
| 1 | 35.6 | 1/4. | 5 |
| 2 | 44.8 | 1/4. | -6.5 |
| 3 | 563 | 1/4. | 4 |
| 4 | 8933 | 1/4. | -3.5 |
| 5 | 11246 | 1/4. | 4.5 |
| 6 | | | |
| 7 | 15704 | 1/5. | 3 |
| 8 | 180 | 1/5. | 3.5 |
| 9 | 17825 | 1/5. | -2 |
| 10 | 20000 | 1/2. | 4 |