The Question of Line Arrays

- or- Why Danley does not build line Arrays
-
By Douglas Jones, Danley Sound Labs

Intro.

Many of the leading loudspeaker manufactures are currently building line array systems. This paper will examine the underlying theory and assumptions behind line arrays and will present Danley's response to this trend.

What Do We Want Speakers to Do?

Well the most obvious answer is that they should turn electrical energy into acoustic energy, not heat! It should be *efficient*. Secondly, it needs to be *accurate*. That is, it needs to respond as accurately as possible to the electrical signal driving it. This can be measured and displayed as frequency response, or phase response or even time response. If we know what is going on to a loudspeaker, we can measure what comes out and compare it to the signal going in. Sometimes we can tolerate speakers which are not accurate. We tolerate them in our telephones. We tolerate them in our television sets, in our portable radios and in our guitar amps! We can tolerate them in emergency paging systems as long as we can understand the message. But we do not have to, nor should we tolerate bad loudspeakers in our sound systems! Furthermore, as manufacturers of professional loudspeaker systems, it should be our goal to build speakers that deliver the message as accurately as possible. In addition, this accuracy should be apparent over the entire coverage area of the speaker, not just at some "sweet" spot. We cannot claim accuracy if it is only demonstrable in the lab at some specific position.

Thirdly, we want some way to control the directional characteristics of the source. That is, we want to be able to choose where the sound should go and have it go there and no where else. The ideal speaker should be *directional*. There are a number of reasons why we might want to control where the sound goes. Indoors, we want to send sound to ears, not to walls, and ceilings where it does no good. Not only does it not do any good, adding to the reflected sound can be a bad thing. If intelligibility or the ability to understand the spoken word is a goal, one of the easiest ways of accomplishing this is to improve the ratio of direct to reflected sound. If the listener receives more direct than reflected sound, chances are good that intelligibility will be high. Loudspeaker systems which do a poor job of controlling the directionality of the sound and bounce as much sound off the ceiling and walls as is going to the audience will have a hard time delivering intelligible sound. Directional control is also important outdoors. Sports arenas often require that the sound form the PA be attenuated on the playing field. Municipalities often have noise laws which stipulate that sound may not bleed from a performance space into residential areas.

If a loudspeaker system is to truly control directionality, it must do it at all frequencies equally. It does no good if the high frequency sound is well controlled but the bass remains omnidirectional. Speaker systems which exhibit this sort of pattern where the high frequency is fairly well controlled but the low end isn't, wont satisfy either condition. The frequency response will vary over the coverage and the directivity will not be constant.

The Point Source

Sometimes we need to take complex problems and represent them with over-simplified idealized solutions in order to begin to make sense of what we observe. The behavior of sound sources is one topic where we frequently use idealized constructs in order to help us understand how real devices actually work.

The idea of a point source is one of those idealized constructs There is really no such thing as a point source. It is a mathematical fabrication which is extremely useful for helping us understand how sound waves propagate. If we could build an acoustic point source, it would be infinitesimally small, actually having no dimension at all, and would create sound waves that would travel in all directions equally well, and do so at all frequencies. It is important to note that a point source by definition would be *small* relative to the waves it produces. Of course, there are some considerable difficulties when it comes to building a point source. First, in the real world everything we build has some sort of mass and dimension to it. We simply cannot build a point. Secondly, we have to concern ourselves with wavelengths that range from over forty feet to under an inch. This is what makes audio and acoustics such a challenging field. In acoustics the way that sound waves actually interact with the real world is dependent on the relationship of the wavelength of the sound wave to the size of the elements in the universe in which the wave is propagating.   Another way of saying this is that in acoustics, nothing is large, nothing is small, it is all relative to the wavelength of the propagating wave.

Real world loudspeakers are not true points, however, when transducers produce wavelengths that are much larger than themselves they do in fact behave very much like point sources. This is due to the nature of how moving elements behave in

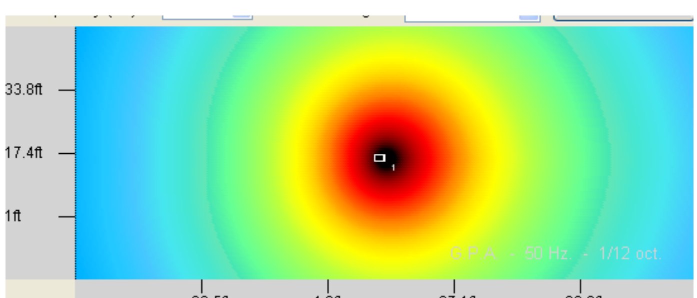

Figure 1, 50 Hz from a 15 inch woofer

the soup of air in which we exist. So although a 15 inch woofer is a pretty large device, If you send a very low frequency signal to a 15 inch woofer, it will behave pretty much like an ideal point source. A 50 Hertz wave is over 22.5 feet long. 22.5 feet is much larger that the 15 inch device that produced it. (note: This does not mean that the woofer is creating something that is 22 feet long. It means that the pressure wave will travel 22 feet away from the source in the completion of one cycle)

At 50 Hz, the 15 inch woofer in the box exhibits all the characteristics of a true point source. The sound seems to emanate from some point inside the box, and it radiates equally in all directions, that is to say it radiates a wave that, if we could see it, would look like a sphere with the speaker at the center as shown in figure 1. So if we knew how much energy went into the speaker, and we knew the efficiency of the speaker, that is to say how much of the energy that went into it was turned in to sound energy, we could use the point source model to predict the energy at different distances from the speaker. It would follow the so called “inverse square law”. As you doubled the distance from the source, the sound pressure would drop 6 db, and this would be observed at any position relative to the speaker. The speaker would not have to “point” in any direction. Indeed the notion of “pointing” a point source is kind of pointless! Things get interesting however when we choose other frequencies. What will happen when we double the frequency, cutting the wavelength in half? Now the 15 inch speaker is creating a wave which will travel 11 feet or so in the completion of one cycle. That is still large relative to the 15 inch speaker. It will still be a pretty good point source, but the apparent source of the sound will have changed slightly. It will appear to originate from a slightly different place inside the speaker box than the 50 Hz wave did. If you keep increasing the frequency, at some point you will notice that things begin to look very differently. At 500 Hz, this 15 inch speaker is not yet larger then the wave it is producing, but the dimensions are becoming similar. It is no longer a point source and cannot be modeled as one. It will be very directional, exhibiting what is called beaming. The apparent source of the sound will have moved, and the efficiency will be much less then it was at 50 Hz. See figure 2.

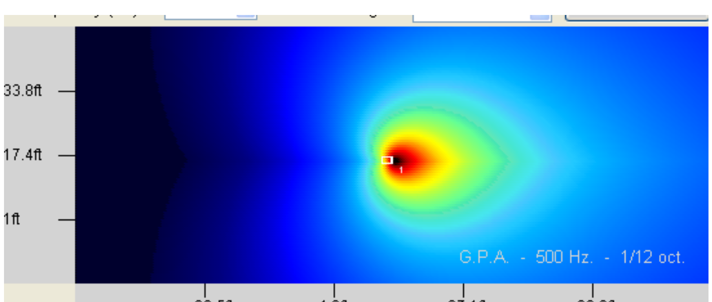

Figure 2, 500 Hz from a single 15inch woofer

It is one thing to imagine and to describe what is going on in a loudspeaker at discrete frequencies. Now try to imagine what is happening when music is sent into the loudspeaker. Everything changes depending on the frequency content of the music! Virtually everything about the speaker is dynamic, that is to say changing. The apparent source of the music is changing with frequency. The directional pattern is changing with frequency. The efficiency is changing with frequency. The load that the speaker presents to the power amplifier changes with frequency. Most loudspeaker systems are not single drivers in a box. Every time you add another crossover and driver the system gets more complex, and moves farther away from behaving anything like a point source. And to make matters worse, as you change your observation point, the speaker will appear to change. If you get very close to a loudspeaker, in what is known as the pressure zone, there will be no falling off of pressure as you move away form the speaker. So if you were to make a measurement at say 2mm from the cone, then you move out to 4 mm, you will not lose 6 db. You wont lose anything! It will be acting line a plane source (more on those later). Then for a short distance the pressure would drop off at 3 db for a doubling of distance, mimicking line source behavior. Then, in the far field, the device actually looks like a point source and the pressure drops off at 6 db for a doubling of distance. *None of this is the behavior of a true point source.* Remember, a point source by definition would always be smaller than the wavelength it is producing! So even though the level would drop off rather quickly, loosing 6 db for every doubling of distance, it would be very well behaved in the sense that it would have perfect frequency response and would be the same no matter where it was measured or observed.

So, *if* we could build a point source, or a device which truly behaved like one, would it be an ideal speaker? Would it be efficient, accurate and directional? Well, efficiency would be a function of the motor design and not necessarily a function of the "point-ness" of the source. A point source would inherently be accurate but of course a point source is omnidirectional by definition, so it is not directional. An ideal point source therefore would meet two of our three criterion.

The Line Source

Just like the point source, the line source is a mathematical fabrication of something we cant actually build. To be a true line source, the line has to be infinitely narrow, or skinny, and infinitely long that is to say it has no dimension in one axis and goes to infinity in the other. The perfect or ideal line source would produce a wave front that instead of looking like a sphere, would look like a cylinder. Like the ideal point source, the ideal line source would behave exactly the same way at all frequencies . However, unlike the ideal point source, the ***ideal*** line source would not behave the same at all observation points! More on this later. The advantage to such a device would be that instead of the pressure dropping off at a rate to 6 dB every time the distance from the source is doubled, it would drop off at only 3 dB!   Remember that the definition of a line source, is that the source is *large* in one dimension compared to the wave that it is producing. It follows then that if a system is to emulate a line source, it must be large relative to the wavelengths it is producing. According to one manufacturer of "line arrays", "it is theoretically possible to construct an audio line array that follows the theory at low frequencies. However the array requires more then 1000 fifteen-inch drivers spaced twenty inches center to center to do it!”[1]

So what happens if you try to create a line source out of discrete elements? This is what would happen if you took 16 omnidirectional boxes and stacked them up. This is the response at 100 Hz. It is clearly not omnidirectional. There is energy front and back, and lobes out the top and bottom.

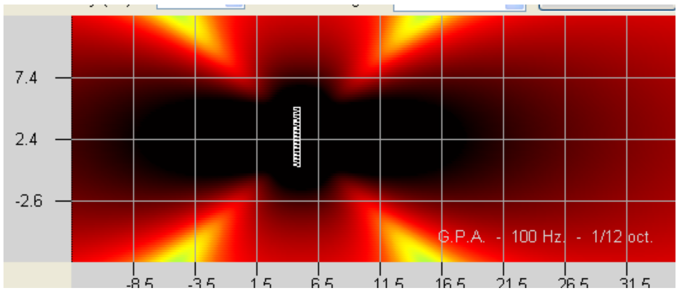

Figure 3; 16 omni sources, producing 100 Hz

At 500 Hz, the pattern would look like this.

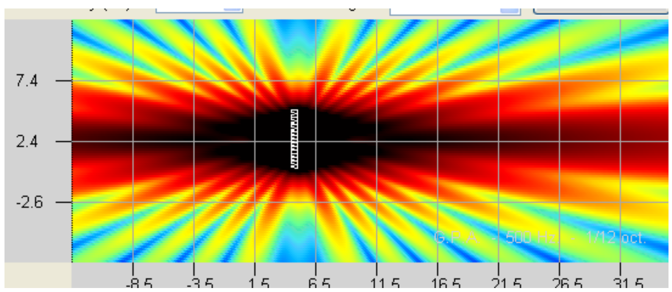

Figure 4; 16 omni sources producing 500 Hz

At 1000 Hz the lobing becomes even denser

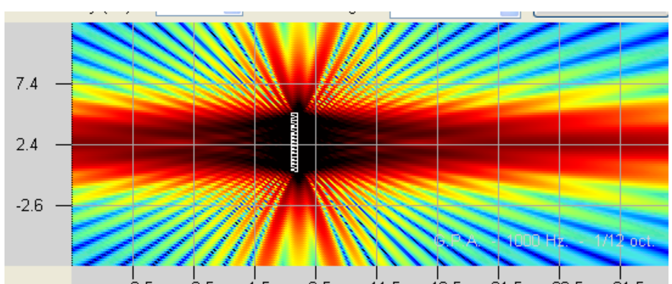

Figure 5; 16 omni sources producing 1000 Hz.

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[1] Meyer Sound Technical Report, Line Arrays, Theory fact and Myth page 3

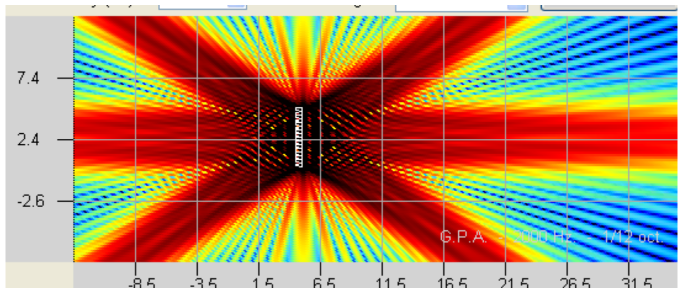

Figure 6; 16 omni sources producing 2000 Hz

At 2000 Hz, shown above, the patterns are even more interesting, and at 4000, the interference is extremely pronounced.

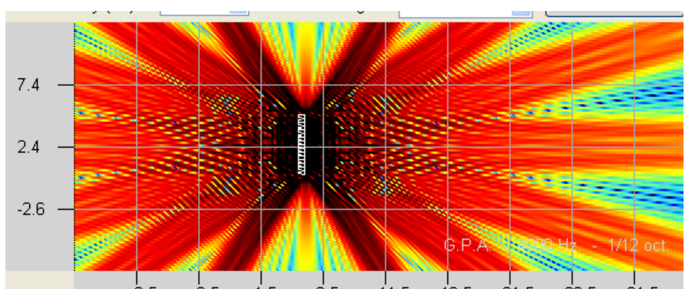

Figure 7; 16 omni sources producing 4000 Hz.

Clearly this does not look like our idea of an ideal speaker! It does get better if one uses directional devices instead of the theoretical omnidirectional devices shown in the graphs above. (see figure 8) It is possible to build a sort of line source using ribbons, maybe even easier than building a point source. Of course there is the small problem of needing to be infinitely long, but of course you can always shrink the

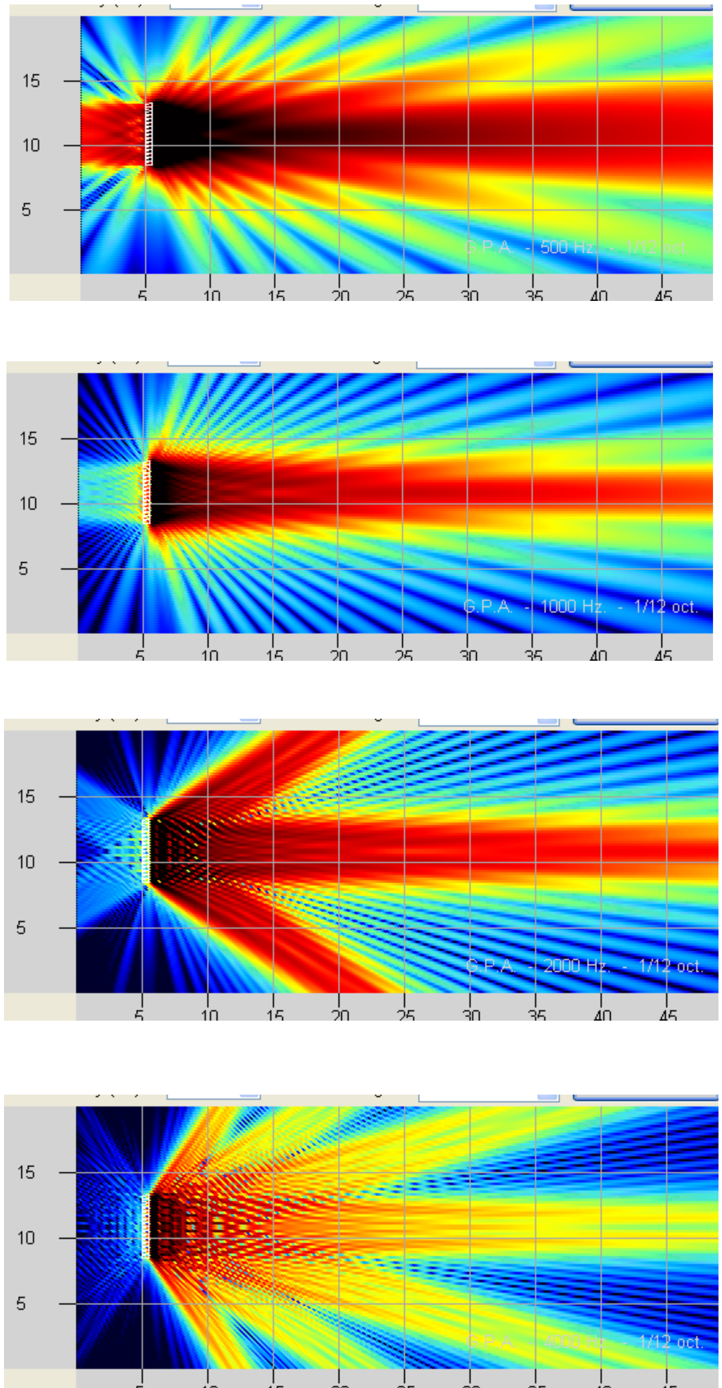

Figure 8, a commercially available 16 box line array, top to bottom, 500 Hz., 1KHz., 2KHz., and 4 KHz.

universe! This is not so far fetched as it sounds… line sources which extend from floor to ceiling for example may perform close to the ideal. In fact you can actually cheat a bit by placing the line source on the floor and effectively double its length.

The real problem is that even if one could build a line source that behaved like the mathematical construct it would not be an ideal loudspeaker. A point source looks like a point source no matter where you observe it from. Up close it behaves like a point, and far away it still looks like a point. With a line source, even an ideal one, up close in the near field, it acts first like a plane then like a line, but as you get further away from it where the difference in the distance from any point on the line to the observer is insignificant relative to the distance to the device, you discover that it begins to look like a point source. Of course if we are talking about the theoretical line source which is infinitely long, the "line" behavior would extend to infinity. In reality the length of the line will be finite, and at some distance from the source it will no longer behave as a line.

But what is the response in the region where it is acting like a true line? Mark Ureda wrote a fascinating paper on the behavior of line arrays. He uses a theoretical 4 meter long line source to examine the behavior of a line source. According to Ureda (Analysis of Loudspeaker Line Arrays JAES VOl52. No 5 May 2004) the pressure response….. exhibits undulations in this region, the magnitudes of which increase as the the distance approaches 100 meters. Beyond 100m the pressure amplitude no longer undulates and decreases monotonically at -6db per doubling of distance".[2] What Ureda is saying is that in the region where this array is behaving as a line source, the frequency response "undulates" (isn't flat) until the point where it no longer behaves as a line. At this point, Ureda calls it the transition point, the response smooths out and the line source actually behaves like a point source, exhibiting the flat, consistent response of a point source, with the level dropping off at 6 dB per doubling of distance. See figure 9.

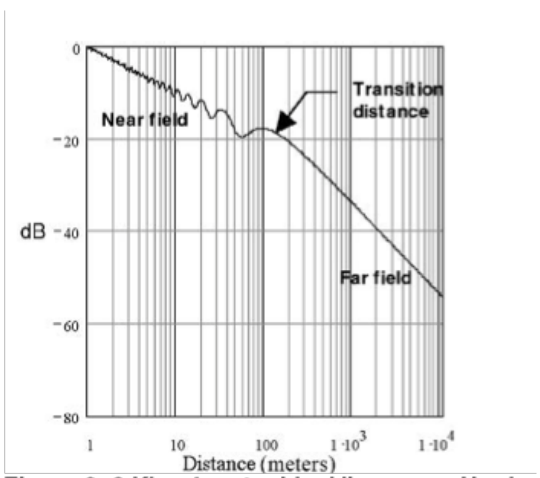

Figure 9; 8 Khz, 4 meter Ideal line array. Ureda P 472

Ureda notes that this phenomenon is frequency dependent, that is, the transition distance varies with frequency. See figure 10.

The problem with line sources, is that *ideal* line sources are self interfering devices. When you try to cover a large area, or move around in the coverage area, the observer will get sound from different parts of the array and the frequency response will not, indeed *cannot be consistent!* According to a major manufacture of line arrays, " While combing has traditionally been considered undesirable, line arrays use combing to work: without combing there would be no directivity."[3] Since when has "combing" become a desirable trait in a loudspeaker? Some have tried to suggest that the combing is not audible, or a

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[2] Ureda Analysis of Loudspeaker Line Arrays JAES VOl52. No 5 May 2004 p 472

[3] Meyer Sound Technical Report, Line Arrays, Theory fact and Myth page 2

reasonable trade-off for the “benefits” of the line source. But if you are trying to meet a performance specification, how can the combing or in Ureda’s words “undulations” not matter?

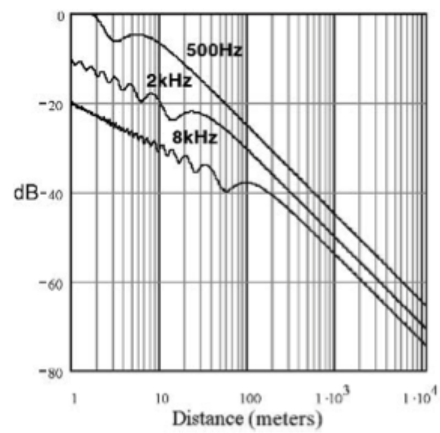


Figure 10, 500Hz, 2Khz, 8khz, ofset for clarity, Ureda p 472

Various manufacturers of line arrays have developed a variety of techniques to minimize the frequency response problems, with some limited success. It is interesting to note that all the attempts at fixing this coverage/frequency response problem actually are attempts at making the line array behave *less* like line sources and more like point sources! The popular J shape or the “arc source” are examples of this”fix”. This begs the question why are we going to such great lengths and expense to build something that *theoretically* cant work, much less practically? Figure 11 shows a 16 box commercially available array with no “J”.

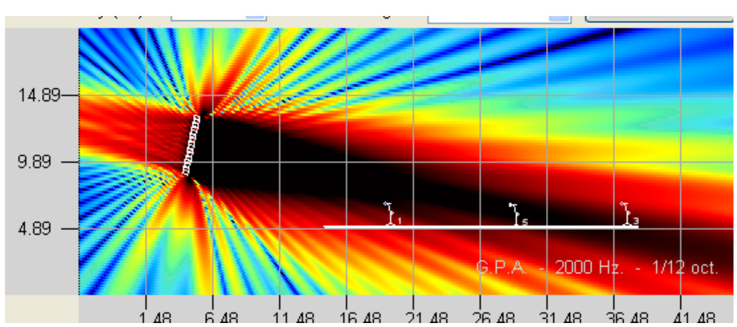


Figure 11. 16 box array, no “J”

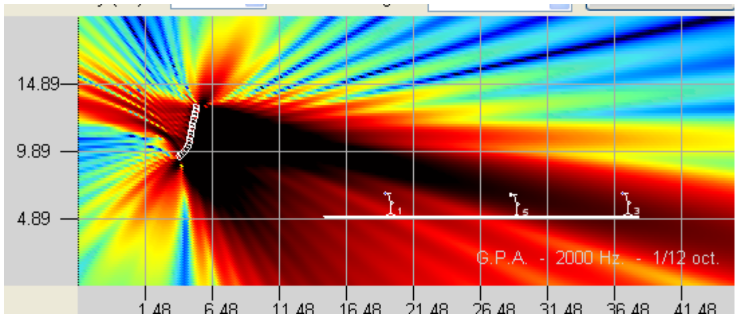


Figure 12. Same array as figure 11, with "J" correction

Notice in figure 12 the "J" correction actually increases the vertical coverage. but compare the frequency response. The response in this model is captured at the three microphone positions shown in figures 11 and 12. Figure 13a is the response before the "J" correction, Figure 13b is with the "J" correction.

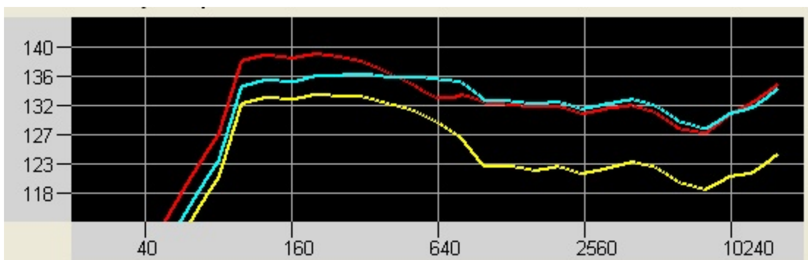


Figure 13a

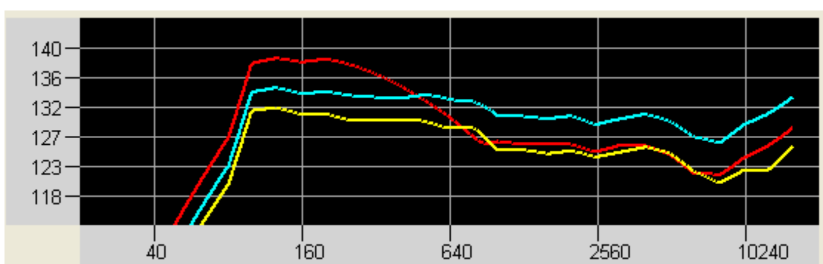


Figure 13b Red is closest mic, blue is middle mic, yellow is furthest mic.

To summarize then, the line source may be efficient, depending on the motor design, but it will not inherently be accurate, and it will not have any significant directivity.

The Plane Source

The third type of theoretical sound source is the plane source. The ideal point source has no dimension, the ideal line source is a line infinitely long and a plane source is infinitely long and wide. If we could build such a thing, it would produce a wave that did not loose any energy at all as it propagated! The amazing thing is that this is the easiest one of the three to build! Of course the source cannot be infinite. If you cant build the infinite, shrink the universe! If you take a tube of a given dimension and create a sound source that occupies the entire area of one end of the tube, you will have created a plane wave tube. The wave will propagate through the tube with virtually no loss. Plane wave tubes are useful in the laboratory for measuring the absorption characteristics of various materials. However, outside of the lab, it is hard to build a source that is very large relative to the lowest frequencies needed, and it is even more difficult to shrink the universe!

The Danley Solution

Lets review how we would like the ideal loudspeaker to behave. We said earlier that for a PA speaker, we want efficiency, accuracy, and directivity. An efficient loudspeaker will require less power resulting in smaller amplifiers. This is generally a good thing. An accurate speaker will arguably sound better and require less corrective signal processing (eq) to get it to meet a performance spec. Directivity is desirable indoors for a number of reasons, but primarily directivity will improve the ratio of direct to reflected energy at any given location resulting in improved intelligibility.

At Danley we add two criteria to the 3 mentioned above. We believe that the ideal speaker should be able to be mounted in clusters or arrays and exhibit little or no interference at the boundaries. We could call this “arrayability”. Finally we at Danley believe that a good loudspeaker should be easy to use and not require external processing to overcome issues in the speaker cabinet.

So, in our quest for the the ideal loudspeaker, which of the 3 mathematical models should we attempt to realize? In our opinion the point source is the best candidate, because of the three, it exhibits the most desirable traits. So, leaving the challenge of building a point source aside for a moment, if we are going to be successful we have to provide the point source with some mechanism to control the directivity. Fortunately the ancients have discovered the solution for us! The horn is actually a very good candidate for controlling the directivity of a point source. If you could couple a broadband point source to a properly designed horn, you would have close to an ideal situation; a phase coherent broadband source with constant directivity. It would have the same frequency response over its coverage area.

This may not be so intuitive as “we all know” that horns exhibit harmonic distortion and the classic horn “honk” along with all sorts of subjective evils. As it turns out, most if not all of these behaviors are understood and can be engineered out of a horn system. Take harmonic distortion for example. The harmonic distortion in typical horns comes from the very high pressures in the throat of the horn. In very high pressure situations, the air itself is not linear. You can keep compressing air, but you cant go beyond a vacuum. The best way to avoid this nonlinearity is to allow the horn to flair quickly. Conical horns with rapid flare rates have very low harmonic distortion. The other big complaint with horns is the “honky” sound which we have all heard emanating from cheap horns. This is the result of limited bandwidth. A horn operating over a wide bandwidth will not “honk”. So back to the problem of building a point source. A number of years ago, Tom Danley found that a horn could actually be used to create a system which behaves remarkably like an ideal broadband point source with directional control. By using an ingenious placement of drivers, the Synergy Horn™ shown below is able to

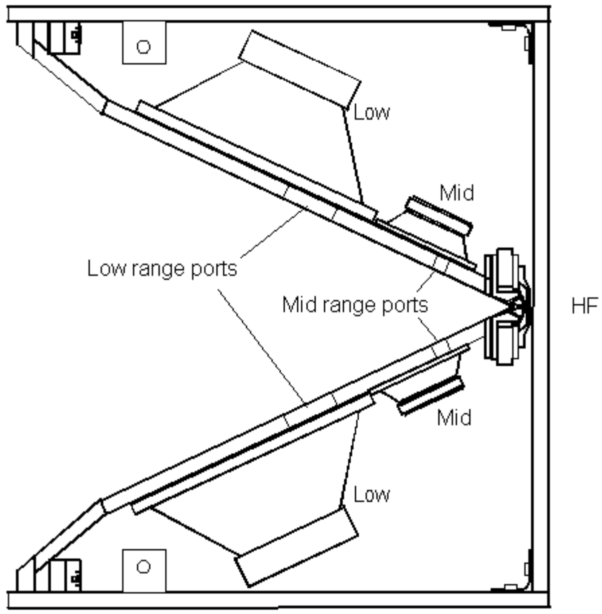


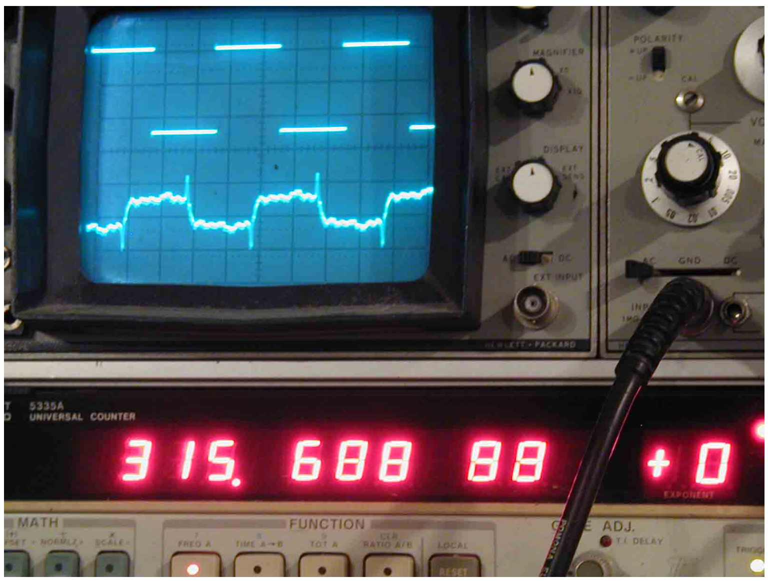
Figure 14. SH 50 cutaway

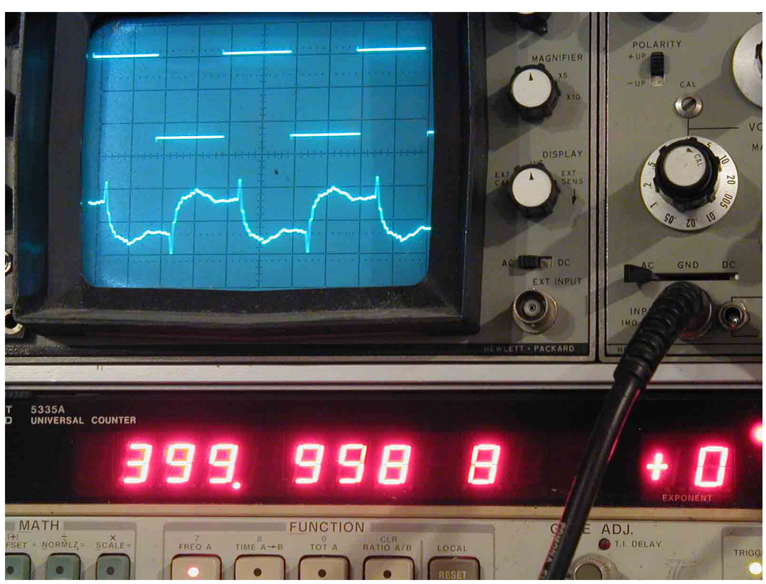
respond from 80 Hz. to upwards of 16kHz in one horn. Danley found that when the drivers are placed in this unconventional orientation, two things happen. First it allows the lower frequencies to enter the horn where the flare rate is faster avoiding the high pressure conditions often found in horn throats. Secondly, if the placement of the drivers on the side of the horn is correct, and the crossovers are correct, the apparent source over the entire range will be the apex of the horn!   By carefully designing the horn and the crossover, a true phase coherent source is produced. The Synergy Horn™ then has all the elements needed to be a close to ideal loudspeaker with no need for any external processing of any kind. Simply hook up a poweramp channel and you are in business.   Because it is a broadband point source controlled by a horn, it is very easy to form arrays. There is virtually no overlap and almost undetectable interference at the transition form one box to the next.

The Proof

At Danley, a core value is honesty and transparency. Our art department does not produce our measurements!

In addition to the conventional displays that buyers have come to expect, we want to include a few more ways to evaluate the Synergy Horn™. A sort of holy grail in loudspeaker building is to build a speaker which will reproduce square waves. Not because square waves are all that fun to listen to, but as an analytic source, the square wave is exceedingly demanding. Of course true square waves cannot exist, at least not in Newtonian physics. Processes always take time! Devices cannot change from one level to another instantaneously. For true square waves to exists the system producing them must have an infinite bandwidth starting at 0 Hz, and must be absolutely phase coherent. We are not claiming that our Synergy Horn™is absolutely flat from DC to blue light! But here are three actual photograph of an oscilloscope screen. We connected an Earthworks 1/8 inch mic connected to the oscilloscope via a broadband preamp. We then sent square waves to the speaker and measured the acoustic output. The top trace is the square wave from the generator, the bottom trace is the signal from the earthworks microphone.





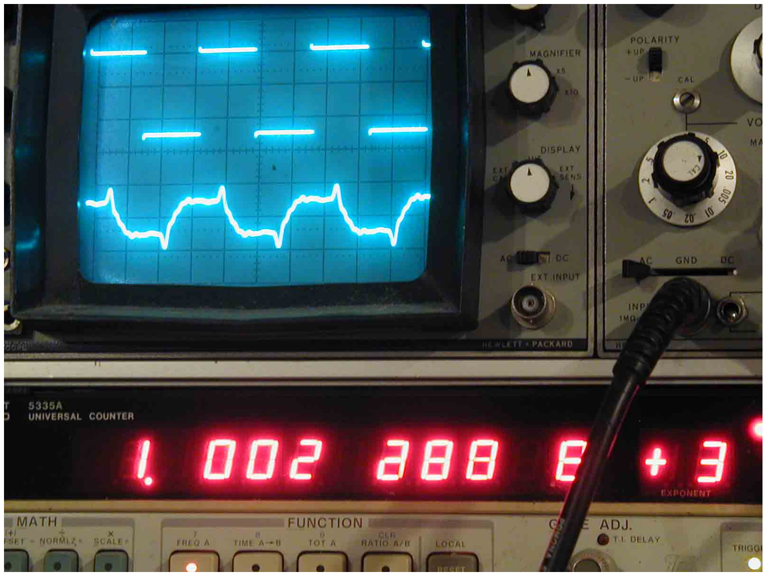
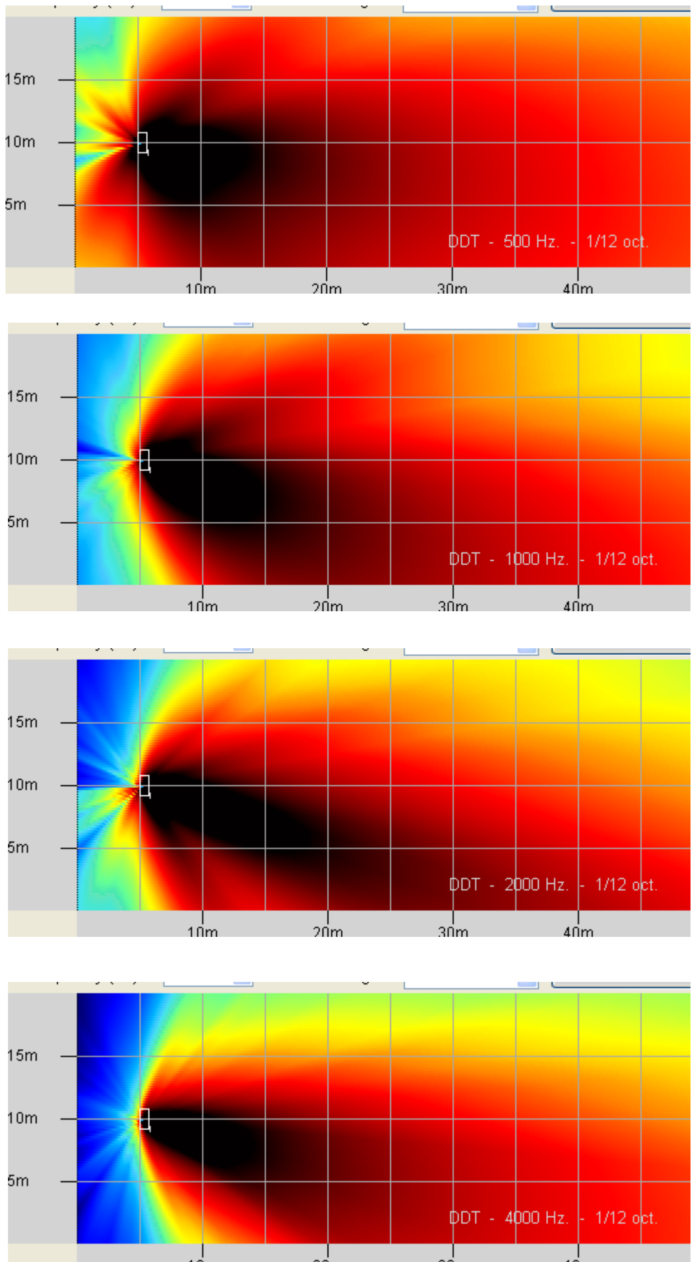


Figure 15; top, 315 Hz, middle 400Hz, bottom 1KHz.

Since we may be the only professional high power PA speaker making this claim, there are not many of these pictures to compare with. So here is our challenge. Try this with your favorite speaker, and get back to us! Finally here is a fantastic subjective test. Get yourself the best digital recorder you can find with the flattest microphones. Play your favorite music into your favorite speaker. Record the output with the digital recorder. Play that back into the speaker. Record that. Repeat until it sounds really bad! If the speaker were truly “flat” i.e. accurate in every way, you could do this many generations deep before the degradation got very bad. We have found that the Synergy Horn™ comes through this test most impressively!

Now some might argue that this is all fine and good, but how can you expect to generate high SPLs without using a line array? Or answer is the Jericho family of horns. The original JH90 or Jericho Horn (™) (the prototype shown here with it’s inventor Tom Danley) is a single box, measuring 60” high by 45” wide by 28”deep. Weighing in a 720 pounds it is not a small box! But compared to a 18 box line array it is decidedly compact! The Jericho consists of 6 high power 18 inch drivers coupled to a conventional folded horn handling the low end. This midrange and high end utilizes 6 6inch midrange drivers and 3 coaxial (two way) compression drivers all coupled to a single 90 by 40 asymmetrical horn. The Jericho is conservatively rated at 148 dBSPL max. continuous, 47 Hz to 18KHz. not by us, but by an independent measuring lab. The J2 (at the time of this writing in advanced prototype stage) is a single 90 by 40 horn with 42 drivers attached to it. Initial tests indicate that the J2 with slightly less bandwidth, will out perform its big brother! Here are a series of coverage maps of a single JH90. You can clearly see the built in down angle of the



asymmetrical horn.

Figure 17. JH90, top to bottom, 500 Hz, !KHz, 2Khz, 4Khz.

In conclusion, we don’t build line arrays because we have something better, the Synergy Horn™!