Subject: Is it my imagination .....OR Posted by Pinky on Mon, 28 Aug 2006 12:35:48 GMT View Forum Message <> Reply to Message

Is it my imagination, or is knowledge about how how speakers work in an array painfully minimal by those who use the various point source systems? I mean, I have contacted and asked questions of very knowledgeable people and gotten great answers but when I bring up that I'm using the speakers in a line array, their knowledge of all of the factors that make combining the speakers (spl, smoothing of frequency response, dramatically lowered distortion, need to place the speakers very close together vertically and horizontally regarding comb filtering. etc) seems to disappear. Its like the array domain of speaker design is its own little shangra-la valley that others cannot find their way into. I even had someone tell me that they didn't like arrays because the sound stage was TOO BIG. I guess if you are only listening to single person guitar music, suddenly someone takes you to your first symphony concert with its 40 foot stage, one would find it to be "TOO BIG"? I guess its comforting to know which speaker the sound is coming out of. And then there is the issue of the expense of the individual speakers. I'm not suggesting that its possible to take a some 49 cent woofers, and some 79 cent tweeters and make an array that beats the pants of the expensive woofs and tweets, but when you are only using one woof and one tweet and one mid, then you get a mindset that that says speakers need to be really expensive to get really good sound. As Bill Fitzmaurice says, .....at low volume levels the distortion and frequency response of even the 49 cent mid is pretty good. [and with a little equalization can be made even better]. So when you suggest that you might be using 40 neo tweeters at 5 bucks US a pop, all that is heard is \$5, and kerching-kerching, all that is heard is under \$30 for the whole kerbang. No, No, No....not possible, fingers in ears, closed mind.Just my thoughts....hUH?Pinky

Subject: Re: Is it my imagination .....OR Posted by tuber on Mon, 28 Aug 2006 13:55:20 GMT View Forum Message <> Reply to Message

Comb filter interactions are common knowledge. Any competent speaker designer or DIYer worth his salt is familiar with that. Regarding dirt cheap drivers you get what you pay for. Reducing excursion will not improve breakup modes or magnetic irregularity.

Subject: That's not what I said...but this group is excepted. Posted by Pinky on Mon, 28 Aug 2006 14:08:41 GMT View Forum Message <> Reply to Message

I didn't specifically say that no one knew about comb filtering, or that using dirt cheap drivers would produce high quality arrays. I only used these as small examples. Those DIYers who live

in the high end world of \$250 Revelator tweeters will certainly not agree. I said that there appears to be a general lack of understanding of the elements of array design in the general population of speaker builders. Those with specific EE degrees who specialize in speaker design are usually aware of these issues. Speaker designers whose whole job is designing speakers may or may not be aware of the special characteristics of array speakers---even those with a huge amount of knowledge in the field.Russell is a giant in the field and his interpretation of comb filtering as posted in a previous item here, as well as his 2005 array suggests that all of this is pretty new to him(the array is nice, but it looks like something that was available both commercial and DIY, some time before 2005).It was a generalized comment. Individuals may be in communication with a more enlightened group, I envy them.This group IS EXCEPTED FROM THE OBSERVATION!!!Pinky

Subject: Re: That's not what I said...but this group is excepted. Posted by Bill Fitzmaurice on Mon, 28 Aug 2006 16:22:42 GMT View Forum Message <> Reply to Message

One of the factors that some designers seem to be unaware of, or at the least don't take advantage of, is that the frequency response of grouped drivers is not the same as that of a single driver. The same response flattening effect manifested when you stack multiple cabinets occurs when you stack multiple drivers. It is this fact that explains why an array may be constructed using inexpensive drivers with the the result that the whole greatly exceeds the sum of the parts from which it is constructed, and why the use of expensive drivers has dimishing returns as the number of drivers is increased.

Subject: Two more questions for Bill Posted by Pinky on Mon, 28 Aug 2006 16:52:27 GMT View Forum Message <> Reply to Message

Bill,Measurements are almost always made at 1 watt at one meter, correct?Suppose you have twenty in a row, if you measured them individually at 1/20 of a watt, would you see a flattened frequency response?I'm looking for a reason for the flattening effect of the Frequency Response with multiple drivers.And also, I do have to figure that as volume put out by the speaker(I'm sure there is a technical word for this), the power required to move the cone is not linear, so this is the reason why the sensitivity of the speakers increases as a group? So a dismal 83db sensitivity with just one becomes a respectable 93 db with 10, and maybe even over 100 with 20? Sorry for asking you for reasoning for why, but I'm more of a "why does it work guy". I was once drummed out of an NRI TV repair course in the early 80's for asking "too many questions". They sent me my money back because things they said brought other questions to mind about how things worked. I was supposed to take it on principle, and not ask why. Everybody else did that I suppose. But when one knows why, then one can extrapolate further answers and solve further problems.thanks in advance,Pinky

Pinky,Your thinking is just why I researched and wrote my Near Field Line Array white paper. My goal was to establish a structure to what was, at the time, an unstructured state of affairs for line array design. Hopefully, others have benefitted from the guidelines and observations that I suggested. Bottom line is that many people have either built or previously heard line arrays that were not well designed. Hence, they concluded that all line arrays were bad and not worth the trouble to study or implement. My comment is that you really have to hear a well designed line array to appreciate how well they can sound when well designed and implemented. Finally, I agree with you that a line array is one project that can sound better than the sum of the parts because you can virtually eliminate mechanical and thermal compression via the use of multiple drivers. Hence, you can reduce distortion and increase dynamic range and even achieve sensitivity improvement in the mix. You just have to get people to listen to a good array to convince them.Jim

Subject: SPL measurement Posted by Wayne Parham on Wed, 30 Aug 2006 16:37:48 GMT View Forum Message <> Reply to Message

Measurements are usually made at a distance greater than 1 meter and then calculated back to the equivalent value at 1 meter. If the loudspeaker is large, this is important since otherwise the measurement microphone would be too close. Line arrays and horns are large, so the distances between the microphone and the furthest point and the closest point of the loudspeaker are quite different. For example, in a basshorn the path length from driver to mouth is often over 3 meters. A line array is usually at least a meter tall, and sometimes several sources are lined up spanning several meters. So the measurement microphone is placed further than 1 meter away. A good setup is to measure at 10 meters and add 20dB to obtain the 1 meter figure, but you can use any distance and calculate back.

Subject: SPL measurement Formula? Posted by Marlboro on Wed, 30 Aug 2006 19:08:42 GMT View Forum Message <> Reply to Message

Wayne, Whats the calculation formula? My room isn't 30 meters. In fact my point of listening is only about 4 meters, easily within the near field.M

Point sources fall off at a rate of 6dB per doubling of distance outdoors. This is the inverse of the square of the distance. Line sources falloff at the inverse of the distance, not the inverse squared. This makes falloff of a line source be 3dB per doubling of distance, provided the line source is long enough compared to the distance to the listener. Once you get far away, a finite line source begins to act like a point source. These formulas only work in a free field, such as outdoors. Indoors, the reverberent field becomes charged and retains energy. That will make sound falloff different, because the walls reflect some of the sound back.

Subject: Re: SPL measurement Formula? Posted by Marlboro on Wed, 30 Aug 2006 22:43:11 GMT View Forum Message <> Reply to Message

So then, if you are doing measurement but not of an individual speaker, you might as well just measure where you are seated and get a correct measurement for the whole room, and adjust the equalizer and electronic cross on the basis of the whole effect?Marl

Subject: Re: SPL measurement Formula? Posted by Wayne Parham on Thu, 31 Aug 2006 00:16:18 GMT View Forum Message <> Reply to Message

Sure, you can always measure in-room and if you want to know what to equalize for, that's important information. One extra tidbit to consider is that in-room response is different at different places in the room. Equipment setup, room treatments and EQ all depend on what areas in the room are desired to have best response. When making measurements for loudspeaker comparison, as the 1W/1M SPL figure is generally applied, tests should be made in an anechoic environment. Measurements made outdoors or in an anechoic chamber are used for comparing loudspeakers, since they remove environmental influence.