

Does elevating the L1™ change the line array performance?**L1® Users Forum**

This topic can be found at:

<http://bose.infopop.cc/eve/forums/a/tpc/f/3976055944/m/3841041115>

Dr. Dad**Sat December 15 2007, 11:29 AM****Does elevating the L1™ change the line array performance?**

Chris,

It's funny you mention Q.E.D. in one of your posts, because this issue reminds me of the book by Richard Feynman of the same name.

Although he is talking about Quantum ElectroDynamics, the principle of phase cancellation and unexpected, counterintuitive results based on interfering waves is very clear and elegant. I recommend it as an eye-opening treatise for anyone who ever took physics or optics and was confused.

Light does not travel in straight lines. It just appears that it does. Sound does not travel in straight lines either; using the idea of a pie shaped wedge of sound, as Chris pointed out, is just a convenient, easily understandable way of showing where most of the sound is going and where the sound is best.

It is possible for phase cancellation to strengthen the perceived sound. Our ears perceive sound in not only in intensity terms, but also in temporal terms. If a sound is not in phase, but is delayed enough, we perceive it as an echo. This can be pleasing to the ear, or it can muddy the sound. It will increase the sound pressure level that we hear, but will not necessarily add to the music.

I consider sound pressure levels to be something misunderstood and often misused. They should be measured to protect listener's hearing, not as a selling point for sound reinforcement equipment, IMHO.

Phill Space**Sun December 16 2007, 01:58 PM**

Although it's in an old post, I'd like to reiterate a bit of a safety concern when elevating the L1. Beware the B1! We played Wednesdays at a long, very narrow restaurant with a high ceiling as a trio (elec.gtr, bass, and drums). Only three vocal mics went thru the L1. We were at one end of the room about 6 feet in front of the L1 on a "stage" that was elevated about 2 feet with half-walls that concealed us from the waist down. Talk about a sound engineer's nightmare! The owner was happy with more vocals and less instruments but raising the L1 volume began to cause feedback, perhaps because of our "boxy" environment. The answer was to place the L1 on a table, raising it perhaps 3 ft. above the stage. The L1 was now elevated 5 ft. above the audience and it was a success, solving both the feedback and the audience coverage. I believe this is approaching the "bare L1 throw" that Chris has just discussed earlier in this thread. Safety? The B1 could not reach the floor with the short cable we had (I think the floor is the preferred location), so the B1 sat on the table also. The B1 vibrated happily until it reached the edge of the table, when I turned around in time to see it bounce nicely off the floor and then hang suspended, still working, while we finished the song. After that song, upon saving the B1 I found the cable to be shorted and interrupting the L1 so we had to finish the night without a sub. Thanks to the engineers at Bose that the L1 kept it's cool and didn't blow anything.

Dan Cornett**Mon December 17 2007, 01:43 PM**

quote:

Originally posted by Phill Space:

Although it's in an old post, I'd like to reiterate a bit of a safety concern when elevating the L1. Beware the B1...The B1 vibrated happily until it reached the edge of the table...

B1's bounce! That is, they put out a lot of sound relative to their weight, so they can "move".

Quick fixes: Tie them in place (rope, belt, velcro, rough carpet) and stack 'em between other heavy objects (e.g.: between two L1's).

Better solution (in many ways!): get longer 4-wire cables.

Thanks for the reminder, Phill!!

[L1 Wiki](#)

Jazzman888**Mon December 17 2007, 01:47 PM**

Thank you for your answer, Chris. You wrote:

.....
 "There is increased level of anywhere from 0 to +6 dB within the near field layer when the floor is added. It has to be exactly +6 dB at the floor where the direct and reflected are the same magnitude and phase. The interference is constructive there, not a cancellation. As you move up from the floor at any distance, the reflection magnitude is decreasing and its phase is shifting slowly, so eventually you go from constructive to destructive interference and it works out nicely that this happens at the top of the layer. But there is not a pronounced volume increase at ear height at normal distances, I think. (I've never made the floor come and go with an A/B switch, so I have some uncertainty on this.)

However if you were really in the far field of a geometrically perfect L1 system (a long way away), then adding the floor would narrow the polar pattern by a factor of 2 and add +6 dB throughout the new main beam of that pattern. All speakers get this +6 db in their far fields, if they are right on the floor and you stand far from them, but on that floor.

So, now that you bring it up, I guess I'm going to say that the sound pressure level really does increase in general when the floor is added. So "throw" increases in both senses. But I wasn't referring to the level effect when I wrote about throw above. I was referring to the shape of the near field."

.....

If I may summarize: You are saying that in the near field (with the L1 on a floor) there may be an increase in SPL of up 6db near the floor, but there probably isn't a pronounced increase in SPL at ear level. And also, that you used the word "throw" to describe the "shape of the near field", which Bose describes as a "cylindrical wave". (I have to say that I think that your use of the word "throw" in that way leaves something to be desired in the way of clarity. I'm glad that I asked my question.)

So you have answered my question, and I can conclude from your posts that the near field (where a cylindrical wave maintains its unique properties) is extended when the L1 is placed on a floor. And also, the presence of the floor doesn't significantly increase SPL at ear level in the near field.

You also went a little further by saying that in the far field (where sound from line arrays have the same properties as point source speakers), there is an increase in SPL of +6 db if the speaker is on the floor, and the listener is standing on that floor.

So it looks like the benefits of placing the L1 on a floor would be most noticeable in large venues with high ceilings, with improved clarity in an extended near field and also db gain out around 100 ft and beyond being the primary benefits. Conversely, it looks like using an L1 on a raised stage (as long as it wasn't so high that the listener's ears were below the bottom of the near field wave) in a small venue with a low ceiling wouldn't present a significant negative.

I await the rest of your answer, Chris. I imagine that if it snows, I'll get it sooner than later!

Also, so reader's of this forum might not be confused by references to "Quantum Electrodynamics" after your use of the salutation "QED" (plus a smiley face) on one of your posts:

The abbreviation, "Q.E.D." (from the Latin phrase, "quod erat demonstrandum" which translated literally means "which was to be demonstrated") was used in antiquity to signify the completion of a philosophical argument or mathematical proof. These days, it more or less has the connotation of "inarguable truth", usually in a humorous way. ☺

W.A.

Chris-at-Bose

Tue December 18 2007, 07:12 PM

Hi Jazzman,

I agree with your summary above, with one new piece of information: I checked my computer model of an ideal line source to see if my sense that "there isn't a pronounced increase in SPL at ear level" was corroborated by the model. It is basically true in the model, except that there is an increase of 4-6 dB in the lower octaves of the L1, starting at 50 feet or so. As one increases one's distance beyond that, the boosted region extends upward another octave for each distance doubling. The effect arises at slightly greater distances if one is standing rather than sitting. In retrospect, this would have to happen, as there needs to be some way for us to get the full bandwidth extra 6 dB when we are very far away, but near the floor. So there is no substantial gain at ear level nearby, but this progressively rises to a gain of 6 dB far away.

I also agree that "throw" is misleading, since it is commonly used to describe only the "maximum usable distance to a listener, based on an SPL requirement". As usual, line arrays behave differently. I don't know if there is a succinct term for what I could more precisely call "the transition distance from cylindrical coverage to far-field coverage."

Here's my second answer: no, the transition distance is theoretically well beyond 25 feet for an L1 with no floor (even for the midrange) and that agrees with many listening judgments at 25-50 feet. The Bose MA12 (2 units high) is often used well above floor level and it certainly maintains its cylindrical effect well beyond 25 feet. (Maybe one of our lurking Field Engineers can comment on this, Thomas.) So, if the transition distance is, in practice, well beyond 25 feet with no floor, why do I hear a softening of the edges of the cylindrical pattern at only 100 feet, even with a floor? I've always assumed that this was the theoretical transition I was hearing, but the models refute that.

I'm not certain of the true cause, but my primary suspect is non-identical drivers. To focus a sharp angular cut-off very far away requires extremely high precision in matching transducers in magnitude and phase. The drivers in the L1 are very consistent, but not identical. So my best guess is that the pattern may be getting "fuzzy" at 100 feet and starting to spread. The floor would neither enhance or degrade this effect, so in practice we may possibly have a similar transition distance with and without the floor. But this is speculation. What I am sure of, based on theory and experience is that the floor reflection helps confine the lower frequencies of the L1 into a layer more completely, it adds 6 dB at the floor everywhere, and it adds 6 dB far away at ear height.

Oh, and I've heard it said that QED is the mathematicians' way of saying "so there". This is illustrated in the following mathematical proof that 17 is the "most random number":

Pick a random number, say "17".
QED

Chris

Edit 10/19: added "(2 units high)" for clarification.

This message has been edited. Last edited by: Chris-at-Bose, Wed December 19 2007 04:35 AM

Jazzman888**Thu December 20 2007, 04:10 PM**

Hi Chris,

Ha! I like your mathematical proof. I imagine that there is a similar one defining "the least random number" as well! 😊

Thank you for the second part of your answer. While I was waiting, I did take some time to explore a paper presented at an AES convention by Urban, Heil and Bauman ("Wavefront Sculpture Technology"), which I found most informative.

From what I've gleaned from your posts on this forum and other sources, I see that the "the transition distance from cylindrical coverage to far-field coverage" is a function of frequency as well as the height of the array and reflections from a floor, with the transition distance being roughly proportional to the frequency. So we are really simplifying by talking about a single distance as representing this point. I think that your ladder analogy pointed this out. But wouldn't this phenomenon contribute much more to the appearance of mids and some of the highs that you might hear at the "top of the ladder" at longer distances, than small differences in the speakers? Or are you only talking about the highest highs?

I think the good news regarding this phenomenon is that the transition distance for the highs extends a long, long way out there. So, since, in general, highs tend to fade faster than lower frequencies over long distances, this is a good thing and contributes much to sound clarity at longer distances.

But it looks like the bad news is that raising the L1 off the floor, using it on a stage, or depriving it of a floor courtesy of a packed crowd, would result in shortening the already shorter transition distances for the lower mids to the point that the frequencies in the 200 to 400 hz range would behave as if they emanated from a point source speaker, and frequencies of around 1000 hz would have a transition distance of no more than 25 ft. or so.

Again, this doesn't look like a significant negative if a venue is on the small side, but I think that this underlines your recommendation to use the L1 on a floor whenever possible.

Always wanting a perfect world, I'm sitting here trying to conjure up a possible solution. Might there be an "extendable L1" out there in the future? I could see a little accessory 2 or 2 ½ foot speaker array inserted under (or positioned on top of) the "stick" serving as a much better alternative to raising the L1 by putting something under the whole unit. A lot of venues have ceilings at least 10 feet high. Also, wouldn't something like this positioned on the floor in front of a stage, aligned with the L1 sitting behind it on the stage (and, if necessary, incorporating a little delay) extend the theoretical height of the line array and also possibly maintain a mirror image of floor reflections?

W.A.

Oldghm**Thu December 20 2007, 07:04 PM**

With all it's flaws, many duly noted by the engineers among us, it still performs better than anything else I have in my arsenal.

O..

Drumr**Thu December 20 2007, 07:42 PM**

Now there ya go...

As I mentioned in Big Sur...and I might have been wrong or selfish to say it, but I don't care about "throw", or any of the other complaints. The L1 sounds good *to ME*, on stage, every time. That is what I really care about.

ThomasS-at-Bose**Fri December 21 2007, 10:50 AM**

Hi everybody.

They got me now – here's a sign of life of the 'lurking field engineer', ThomasS-at-Bose. Sorry for not being able to chime in earlier, but we all have to get a lot of stuff done before our vacations.

I stumbled over this post more or less by accident. I was naturally attracted by the issue and sent some comments to Ken in a private mail. He then asked me to contribute something to this problem from my perspective as a field engineer that has been using (stacks of) MA12 from the very beginning. I will try to do that in this post.

First, let me say that the pictures painted here by Chris are as visual as can be in written words and for most of that part, there's not much I feel I could add to that. I think I'm very familiar with most aspects of arrays but I can certainly say that Chris (again) opened new ways for me to look at things: I learned something. Good thing.

Neither would I like to try to add any more detail to the discussion of exact array performance that recently peaked, since I think that - without having a good simulation model to commonly look at - this just gets a little too fuzzy once a specific level of accuracy is achieved in the analysis. In my humble opinion, I think this point is almost reached. Note that we haven't yet included the directivity of the individual drivers in our view of things and that we also didn't yet touch the issue of the property of sound that enables it to travel "around" obstacles. So much more things to take care of to get the "full" picture.

Nonetheless, I would like to comment on a couple of specific points that were raised throughout this whole series of posts, starting with one of the latest, say

Near-to-Farfield-Transition

At Bose Professional, we apply arrays of MA12s for both voice and music reproduction in a variety of applications. For those who are not

that familiar, an MA12 element is – from a radiation perspective - pretty equivalent to half an L1 Model 1. Of course, we use subwoofers for LF extension if necessary.

The arrays of MA12s are typically between one or three meters in height, but some are even six or eight meters tall, depending on the application. Rooms we have to work with can be relatively small or really large (especially: tall). They can be pretty dry or have reverberation times up to as much as six to ten seconds. Yes, churches in Europe are significantly more reverberant than the typical modern US type.

Everybody (also outside of Bose) loves the advantages of a tight vertical dispersion pattern of line arrays in such spaces since they improve clarity and speech intelligibility.

For our practical work in Pro, we have a couple of rule of thumb rules to design – what we would call – “throw distance” of an array versus length of the array versus reverberation time (RT). (A detailed analysis is typically performed in our simulation program, Modeler.)

And yes, I can confirm the near-to-farfield transition ranges that Chris has mentioned. The thing is a little more complex in typical enclosed spaces as discussed here, but I can say that we use 2m arrays for distances of up to 12-15m (40 to 50 ft.), depending on the RT and mounting options. For longer throws, we either elongate the array or use one or several lines of delayed systems in the rear, which is very typical for churches over here. The higher the RT, the taller we try to build the arrays (we typically have a lot of architectural constraints), so 3m tall arrays are not uncommon in some churches.

Floor Reflections (The mysterious mirror image)

Yes, they are important. I would like to repeat that – in real rooms, not in a (semi-) free-field (aka outdoors) - we have to deal with two different aspects of the floor reflections. One is the extension of the nearfield and the other is the improved directivity at low mid frequencies that is so beneficial in reverberant spaces where we want to keep energy as confined to the height of the audience's ears as possible. For a typical club-setting, the first effect is probably the most pronounced, but for full effect it requires that everybody can actually see the reflected image: maybe you can draw a mental cross-section of the arrangement including the image and note who in the audience actually can see the image, especially regarding the potential obstruction by people or gear in front.

This is more or less the same in our typical Pro applications, so we strive to get the arrays a little higher above the heads and aim them like 2 degrees downwards if possible. The basic idea is just to have as many people see as much of the array as possible and also point the main axis of the array towards the audience.

Now, where do we use the super long arrays ? Do we use 6m MA12 in churches ? No, we typically don't. The standard use case for the very long arrays is an audience plane that extends significantly in the z-direction (height). For many auditoria, a raked seating is used (more on that below) and may span several meters in z. Sometimes there's an additional balcony above that needs to be covered as well. In these situations, we build the array so long that its pancake covers all ears distributed in z.

The coherent floor reflection

Another one that I'd like to point out again: You always (for all frequencies and for all distances) get a coherent arrival from the mirror image (at the exact same time as the direct arrival) if your observation location is directly on the mirror plane (aka floor). If you would get down on your knees and right on the floor with your ear (Unfortunately, I can only get one ear on the floor...), you get the extra 6 dB for all frequencies.

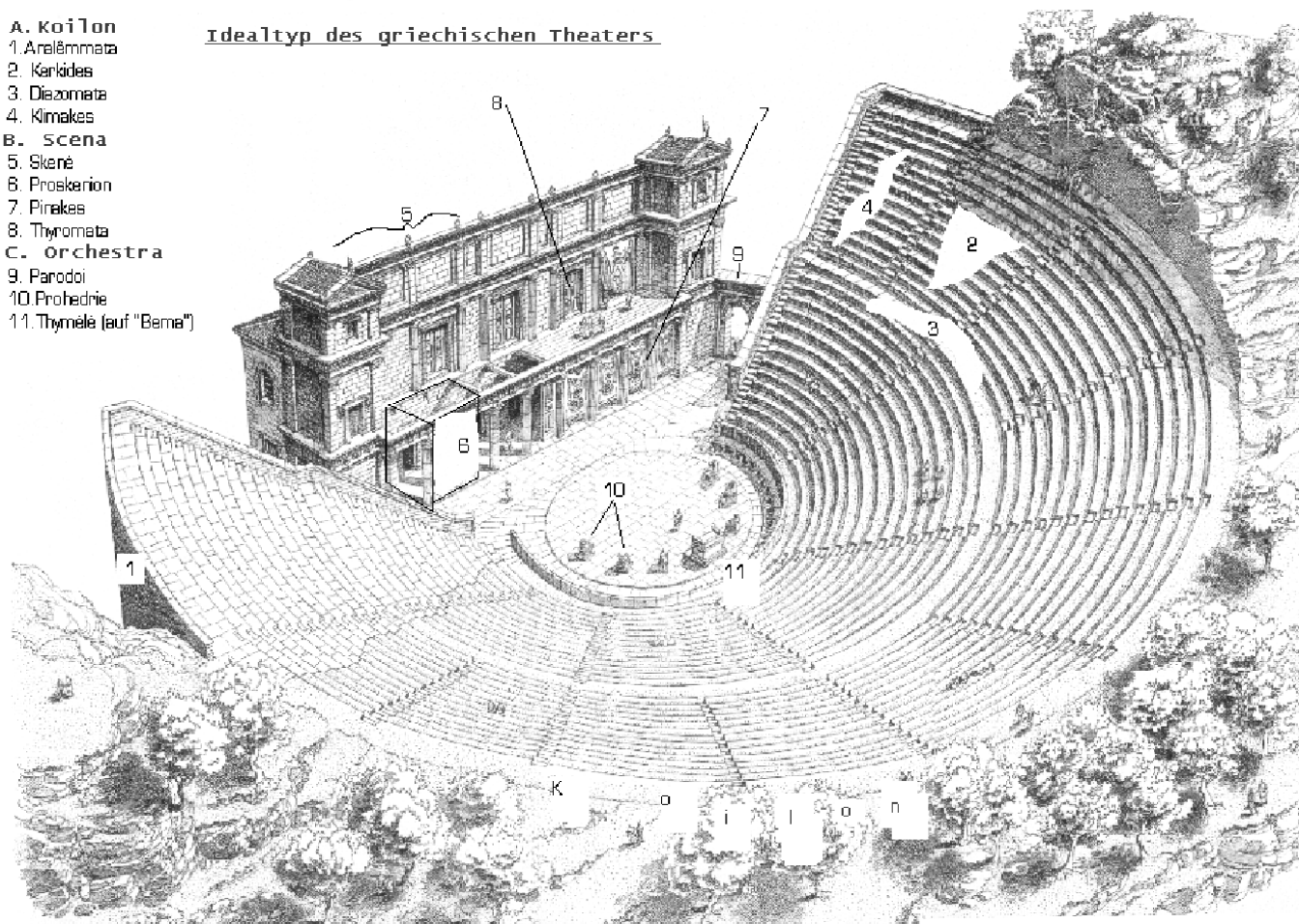
This effect is actually used in a loudspeaker measurement technique called “groundplane” measurement. Here, the measurement microphone is placed directly on the floor during the measurement. One gets the same result as if the floor would be gone, just 6 dB more level at all frequencies, which is easy to correct for. As everybody can imagine, it's much easier to mount something to a floor than to suspend it in free air, so this technique is often used, e.g. in semi-anechoic rooms.

More Room Acoustics

One thing that came to my mind early when I read about the importance of floor reflections in this thread was the design of ancient Greek theatres, large outdoor spaces designed for the un-amplified reproduction of voice and music for a large audience. Although the thing is a little more complex, as also described in a recent JASA paper (e.g. [Livescience](#)), I think it's worthwhile to point out that the stage geometry was designed in conjunction with large reflecting surfaces in the front (floor) and back (wall) to improve the sound level in the audience area by providing early reflections. The picture should give a pretty good indication of what I'm trying to describe:

- A. Koilon**
 1. Anelämmata
 2. Kerkides
 3. Diezomata
 4. Klimakes
B. scena
 5. Skenē
 6. Proskenion
 7. Pirakes
 8. Thyromata
C. orchestra
 9. Parodoi
 10. Prohedrie
 11. Thymelē (auf "Bema")

Idealtyp des griechischen Theaters



(Picture taken from Wikipedia: [Wikipedia EN](#))

Chris also talked about the effect of sound waves being bent downwards when they "flow" above and parallel to a highly absorptive surface. In the classical room acoustical design theory for auditoria, the audience is therefore located on a progressively raked seating area. This is not only for better views and better direct sound, but also specifically tackles the problem with the "grazing sound absorption" as well.

All of the above naturally leads me the question:

What is the optimum shape of a performance venue to work well with L1 systems ? What seating rake, what ceiling height, where to provide floor space, what to do with side and rear walls ?

OK. I think I have to come to an end now. Last thing: 17, the magic number. It has always been used by my math professor at university as the standard "dummy" if any exemplary number was needed. I kinda copied that good tradition for my lectures, but it's one of those things I do without knowing exactly why... Chris, do you have further insight into this secret ?

So far for today. Have a good time, everybody.

Best rgds,

Thomas

This message has been edited. Last edited by: [ThomasS-at-Bose](#), Fri December 21 2007 09:14 AM

JD1

Fri December 21 2007, 01:08 PM

As luck would have it, I just booked a gig this morning that involves setting up on an elevated stage in an auditorium.

I'm not embarrassed to say that I have no idea what you Boys-at-Bose are talking about but it sure is impressive to say the least.

Suffice it to say that I ain't going anywhere near that stage and am setting up on the main floor right in front of it. If anybody asks why, I'm gonna print out Chris and Tom's response and say "Here's why!!!"

I'm just a blind chicken pokin' round for a grain of corn.

www.jdsnydermusic.com

Drumr

Fri December 21 2007, 03:20 PM

quote:

and am setting up on the main floor right in front of it.

That's what we do.

quote:

originally posted by ThomasS-at-Bose
Have a good time, everybody.

Words of wisdom Thomas. That's what the L1 was made for.

Col. Andy

Fri December 21 2007, 04:15 PM

I regularly work on a stage elevated about 2 feet. Using my L1 or my L2, I've never noticed any sound problems anywhere in the room. I thought that as long as you were covering "ear level" everything was fine. Am I misunderstanding?

Respect

ST

Fri December 21 2007, 05:22 PM

Hello ThomasS-at-Bose

Thank you for your wonderful post.

quote:

Neither would I like to try to add any more detail to the discussion of exact array performance that recently peaked, since I think that - without having a good simulation model to commonly look at - this just gets a little too fuzzy once a specific level of accuracy is achieved in the analysis. In my humble opinion, I think this point is almost reached.

I wish that it was always so clear when discussions have reached this point.

I really appreciated the insights into the MA12.

When you wrote about the RT (reverberation time) it reminded me of an old question - is the Critical Distance calculated differently with a line array?

Perhaps this is not the real question. I should be asking if there is any scientific support for my general observation that using the L1™ tends to tame highly reverberant spaces. That is, compared to a point source, I seem to be hearing more direct sound from the L1™ at a greater distance, before the reverberant sound matches and then overcomes it.

quote:

For longer throws, we either elongate the array or use one or several lines of delayed systems in the rear, which is very typical for churches over here.

When you do this, do you typically place the delayed systems at the rear of the space? Is this preferable to having delayed systems placed at intervals in the space.

If you can direct me to some references online for either of these ideas, I would be glad to do the reading.

Thanks again Thomas.

ST

JD1

Fri December 21 2007, 09:06 PM

Andy...if you've never noticed any sound problems than that's all that matters, right?

When I walk into a dark room I flip the wall switch on and suddenly light appears. I don't give a rat's patoot about what the protons and neutrons and electrons are doing inside the wall. All that matters is light. Let the experts figure out the rest. I know light when I see it and I know how to use it.

www.jdsnydermusic.com

Jazzman888

Sun December 23 2007, 11:26 AM

Oldghm,

I think the word "faults" is way too strong a word to use here. Really, the discussion is about the inherent characteristics of line arrays, which, since it is one, the "stick" on the L1 shares. A closer examination of these characteristics may reveal that all is not lost, for instance, if the L1 is used on a stage. Read on, and you'll see what I mean.

Thomas,

Thank you so much for bringing the MA12 out of the closet and presenting its qualities as a reasonable proxy for the line array on the L1 (since they are so similar). I think that doing this will help demonstrate and simplify things a lot.

You mentioned not having "a good simulation model to commonly look at". But there is one! It's available on Bose's Pro-sound site on page 6 of a white paper entitled "Bose® Panaray® MA12 Modular Line Array: Technical Information and Polar Data". Here's a link:

http://pro.bose.com/pdf/pro/app_notes/panaray_ma12/an_paranaray_ma12.pdf

I remember reading this paper before, some time ago. But recently, having a "senior moment" of sorts, I completely forgot about its applicability to this discussion until I read your post. I recall that at my first read of this paper way back, after digesting the information on page 6, my first thought was, "Wow, what a cool piece of gear!".

Interested readers,

You might benefit from taking a look at that paper right now, with special attention being paid to page 6. (Those who are just interested in the bottom line may just want to just skip this and read on.) Even though I'm not an engineer or sound professional, I found this information to be easy enough to understand (at least a little easier than what I see written in this thread by guys-from Bose! 😊). And while the analysis of sound waves incorporating all of the factors that can affect them can be exceedingly complex, this information, particularly the graphs on page 6, goes a long, long way towards explaining in a reasonably concise way why the MA12 (or L1) "works as it does".

On page 6 of the white paper, you'll notice two colorful columns of graphs which beautifully illustrate the "vertical radiation patterns" of a single MA12 on the left and a stacked pair of MA12s on the right, as a function of distance (visualize a side view, with the sound directed to the right). The "stick" on the L1 most closely resembles a stacked pair of MA12's, so for now, pay particular attention to the column of graphs on the right.

That small vertical black line on the left side of each graph represents the MA12's. (Remember, you are looking at this from the side.) The graphs were created with the Bose Modeler® Design Program, so it's modeled simulation of radiation characteristics in an anechoic environment. But for our purposes here, I think that these graphs bear a close enough resemblance to the reality of the phenomena that we are trying to understand. In terms of real life situations, I think that these graphs (the ones in the right hand column) most closely represent sound coming from an L1 that was raised off the ground, outdoors, in still air.

Each graph represents the shape of the wave (viewed from the side) at the indicated frequency, with the colors representing the decrease in SPL (-2 db "per color"). The left /right scale is marked in meters, which goes out to 32 meters (about 100 ft.).

When looking at these graphs, what is immediately apparent is that the shape of wave varies from one frequency to another. Also, the reduction in SPL over distance, as represented by the colored bands, vary over distance for each frequency as well.

Now right here, I hope everybody can take a deep breath and RELAX.....

These variations DO NOT represent bad sound, faults, or anything like that. They just represent "what is", and when you look at it in some detail, you can see that even if the L1 is raised off the ground, or used on a stage (as long as "ear level" is somewhere within the height of the array), that the L1 (or stacked MA12's) offers significant advantages over point source speakers.

Let's first look at SPL over distance. In terms of "throw", it's very apparent that the higher frequencies benefit the most from the line array configuration. With 2 stacked MA 12's, at 32 meters (about 100 feet), there is only about a -14 db drop in SPL for 8 kHz sound. For comparison purposes, sound from a point source would be down by -30 db at that distance. That represents a very significant improvement in throw.

Now lets look at one of the less advantaged frequencies lower down. From the graph we can see that at 1 kHz, the sound would be down by about -17 db at 32 meters. That's still a HUGE improvement. And this improvement comes despite the fact that for this frequency, beyond the first 25 feet or so, the sound pressure level drops at -6 per doubling of distance, which is the same as sound from a point source. All of the improvement occurs in those first 25 feet.

But some of you might be thinking, "What happened to flat frequency response? Wouldn't the highs then be too dominant?"

Consider that we're looking for optimum clarity in live sound situations. A lot of good vocal microphones have peaks in the higher mids to the highs. For instance, the frequency response of a Shure SM58 microphone shows +5 db bulge in this area that peaks at about 5.5 kHz. They are engineered that way for a reason: That "bulge" makes them sound better! Also, Bose uses EQ and presets to optimize the overall sound, which takes into account the characteristics of line array behavior, and with the presets, the specific type of sound being

amplified. And there are other factors that occur in real world situations, which Chris-from-Bose got into earlier in this thread, that aid in boosting the lows and lower mids at longer distance. I'll spare you a rehash of these!

Let's continue: I'm imagining some of you looking at those strange looking "jaggies" close to the line array source on that 8 kHz graph?

This "aliasing" is present in higher frequencies only. I'm guessing a little here, but I think that a chunk of what Chris has talked about with regard to the benefits of using the L1 on a floor deals with handling the aliasing in a good way, where reflections from the aliasing leaking out the bottom ultimately cancel out those leaking out the top.

In any case, the aliasing that may occur in a floorless situation appear to be of little consequence when the frequencies generated top out at around 5 or 6 kHz, as they do with electric guitars, electric basses, Rhodes pianos and Hammond organ clones. I think I'll leave it to the Guys-from-Bose to comment further on this, since I see no other relevant comment in the Bose white paper.

I'd say the bottom line here is: Use the L1 on a floor whenever possible, as it will considerably enhance already good performance.

I want to be clear that all of this is my interpretation of the data presented by Bose. I believe it to be an objective, accurate interpretation, and it is consistent with my experience with my L1 Classic, which included a little bit of informal testing. But I'm not a sound engineer or sound professional, and I wrote this while in a bit of a hurry.

I've seen a few inaccuracies (unintentional, but inaccurate nevertheless) pass without comment on this forum. As readers use this forum to glean accurate information, it is my hope that guys-from-Bose address any errors in interpretation that I may have presented.

So Oldghm, looking at this information, I can see that there are some good reasons why your L1 is your favorite piece of amplifying gear. And Col Andy, it looks like continuing to use your L1 on a stage will work as well as it always has!

Happy Holidays!

Sincerely,

W.A.

This message has been edited. Last edited by: [Jazzman888](#), Mon December 24 2007 10:07 AM

Oldghm

Sun December 23 2007, 08:31 PM

Even though the L1 is something less than perfect, as duly noted by the engineers among us, it still performs better than anything else I have in my arsenal.

O..

Edit to remove "way too strong a word" and more accurately represent my intended message.

KC

Mon December 24 2007, 02:18 PM

wow, a lot of verbage and some long posts here. But the topic caught my eye because on several occassions where there is a stage approx 1-ft off the ground, I have elected to set my system on a couple chairs just off the stage, usually with the B1's on the stage, to get a bit of distance between me and the Bose although, no complaints, now I am concerned about the floor reflection thing...

since Bands generally are on a stage, does this interrupt the floor-relection-design plan?

usually the guys on the other side of the stage have a bunch of crap IN FRONT of their L1's, anything from B1s to chairs with amps, to drums and props. I try to keep mine clear so as to get the entire array effect with an unfettered column of sound. How much will the sound "go around" objects?

Dan Cornett

Wed December 26 2007, 03:17 PM

quote:

Originally posted by KC:

...How much will the sound "go around" objects?

Simple answer: very easily.

A bit longer answer: As long as the 'object' is not real close to the speakers, and/or is not too large, the sound will easily 'bend' around the object.

For example, a person standing as close to the L1 as the base allows, will actually block a fair amount of the sound ... particularly the sounds in the vocal range and above.

However, if that person moves 4 feet away (in any direction), then their presence will have almost no affect on the sound for any listener that is more than, say, 4 feet beyond that 'blocking person'.

Sounds "bounce and bend" (*sort of*); so, the more any 'blocking objects' either absorb the sound (because they are soft and squishy) or they are big ... or there are many of them, such as a crowd of people ..., then the less sound will be able to "go around" them.

So, in rooms with 'wall-to-wall' crowds of people, particularly close to the L1's, elevating the L1's so a portion of the column is above

them is a real good thing to do.

L1 Wiki

L1 Lover

Fri July 01 2011, 09:19 AM

Chris: Having read through this discussion with great interest, why does the posted reflection application data and explanation seem to dramatically contradict the various videos showing how the L1 propagates its sound waves (e.g 90 degrees top and bottom, about 170 degrees left and right)

It was previously understood that the L1 takes the floor composition and ceiling height almost completely out of the acoustical equation.

Would you please provide a layman's answer to a possible inquiry that the words doesn't match the pictures?

Cap Capello
Bose L1 Family of Products

Chris-at-Bose

Fri July 01 2011, 04:37 PM

Hi Cap,
"Long time no chat." Good to hear from you.

I'm not sure I understand what contradicts what yet, but it has been 3.5 years since I read this thread in detail, so maybe that's not surprising. The one thing you said that seems to suggest a contradiction to me is "90 degrees top and bottom". Is there something you've seen that shows or describes the L1 as radiating into a wide, 90 degree vertical angle? Or are you saying that vertically the sound is all moving at a 90 degree angle to the line of drivers? That is what I remember seeing in the videos.

So before I offer a layman's explanation of the wrong thing, I'd still like a bit more detail from you to clarify the problem to me. I think you are referring to the fact that the sound is mostly confined to a vertical layer, but not perfectly confined. I can go into more detail on that. But I'm not sure I'm on the right track yet.

I will be away for the long weekend, but I will try to answer your question as thoroughly as I can when I get back on Wednesday.
(Caught that second discrepancy there? Yes, I'm taking an extra day off.)

Cheers,
Chris