Outline miniCOM.P.A.S.S.

Testing a line array element

By Pat Brown

The subject of this loudspeaker profile is the Outline miniCOM.P.A.S.S. line array element. The objective is to quantify the loudspeaker with a series of tests that provide the data necessary for a sound system designer to implement the product using some available predictive tools of the trade. As always, the design principles are applicable to other loudspeakers and software tools. The test data has been published in the Common Loudspeaker File format – a platform-independent data format supported by four major room-modeling programs. The CLF data file for the miniCOM.P.A.S.S. is available from www.clfgroup.org, along with a free data viewer. It is not a review per se, and I will not draw any conclusions regarding the quality of the unit or how it compares to similar products. Think of this as a tutorial on specifications and line array design that uses a specific product.

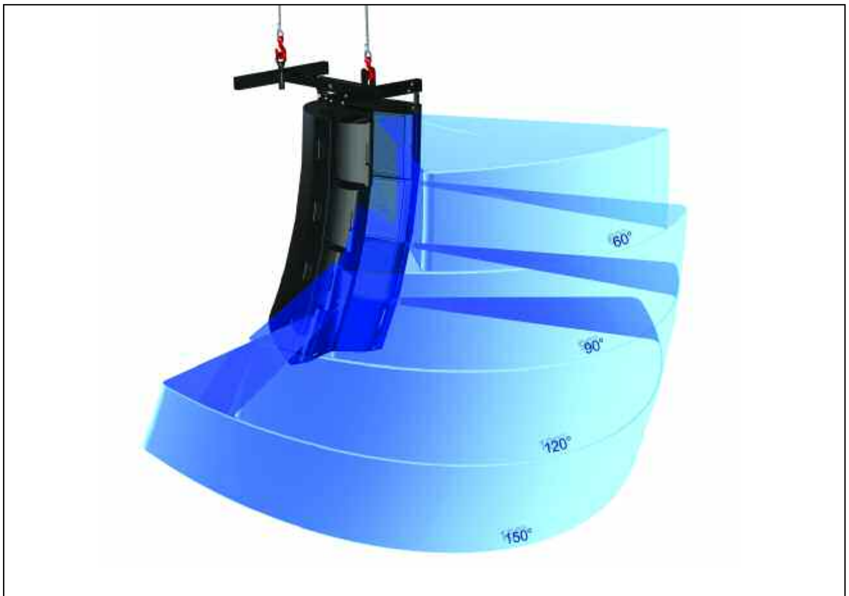

Figure 1 – The results of the maximum SPL test. I drove the MiniCOMP.A.S.S with a gradually increasing voltage until its response changed by 3 dB. The microphone was 1 meter from the box.

I will first give an overview of the relevant specifications of a single miniCOM.P.A.S.S. line array box. Armed with this information I will move on to predicting the response of several different array configurations using computer modeling. The examples will include arrays that work as expected and some that don't.

OVERVIEW

The miniCOM.P.A.S.S. is a compact powered loudspeaker that is designed to be simple to implement. An XLR female input connector is provided, as is an XLR male output for driving an adjacent device (presumably paralleled with the female connector). Since this is an active, high impedance input, a large number of devices can be paralleled without loading down the source (usually a signal processor). As such, arrays of just about any practical length can be formed without concern for signal degradation due to loading, and with no special circuitry required for splitting the signal. The utility power input is via Neutrik PowerCon, and both inputs and outputs are provided – again for daisy-chaining additional devices. Four equalization settings are provided for the various configurations (I'll get to that shortly) and are easily selected via a push-button switch/LED on the back of the unit. An 8P8C (RJ-45) connector is present. The manufacturer states that this is currently for mfg-upload of DSP presets, but in the future will allow user-adjustment of selected parameters. An interesting feature of this loudspeaker is the mechanically-adjustable horizontal pattern control. Adjustments are made by moving the hinged baffle elements to one of four stops – an interesting implementation of an old idea.

SOME KEY SPECIFICATIONS

There are many line array elements in the audio marketplace. These high-tech building blocks are designed to be used in multiples to form a system that provides vertical coverage which can be optimized for an auditorium. Usually only vertical pattern control is adjustable (by array length and curvature). The miniCOM.P.A.S.S. adds control in the horizontal plane. But before I get to that, I will examine some of the one-number specifications needed by the system designer.

I measured the input impedance of the miniCOM.P.A.S.S. by connecting my TOA ZM-104 impedance meter to pins 2 and 3 on the input XLR jack. The impedance to a 1 kHz square wave is 20K ohms. If 10 units were driven from a single source, the source would see a 2K ohm load – no problem for modern DSPs. I then connected the meter between chassis and pin 1 to check for a "pin 1 problem" – a condition that makes a device susceptible to noise currents flowing on cable shields. The 1 kHz tone was slightly audible during this test, which means that the shield should be disconnected in the cable. Alternately, non-shielded twisted-pair could be used – an option that is growing in popularity among sound system designers. If pin 1 was bonded directly to the chassis at the connector then lifting the cable shield would probably not be necessary.

MAX SPL AND SENSITIVITY

To check the maximum SPL, I used the "difference" mode of SmaartLive. I drove the loudspeaker with 0.13 Vrms (-15 dBu) of IEC noise and saved this as a reference. This drive level is well within the range of linear operation and should not produce any thermal problems in the loudspeaker. It is an appropriate level for warming things up and establishing a reference transfer function. The difference mode compares the reference to the current, live measurement, so we start with a straight, horizontal line on the analyzer screen (**Figure 1**). I then increased the drive level in 3 dB increments until the response began to change, and then 1 dB increments until it changed by 3 dB relative to the reference measurement (also Figure 1). The drive level that produced a 3 dB response change was 1.3 Vrms (+4 dBu) and the SPL at this drive level was 118 dBC – slow. I put on some hearing protectors and went into the chamber to check the indicators on the loudspeaker. The signal present LED was continuously "on" and the overdrive LED barely flickered. Since this is a closed system, it is not necessary to know the internal amplifier ratings, nor the impedance of the transducers. One can assume they are appropriate and the performance can be based on external measurements only.

The published specifications from Outline are in good agreement with the power test. The claimed maximum of 120 dB SPL continuous (126 dB SPL short term) is 2 dB higher than what I measured, but the fact that I used C-weighting can account for the difference. If the response change were due to a limiter engaging rather than power compression, the system may have been in no danger of thermal damage at the maximum level. One can't know without additional tests (i.e. THD), but the numbers given should be adequate for what a designers needs to know at the drawing board. The rating claimed for four units is 132 dB SPL. This obviously assumes a straight line with coherent summation of all four devices (apparent in the predictions shown later). If the line is curved a lower SPL could be expected. This can all be verified in the computer model, which I will get to shortly.

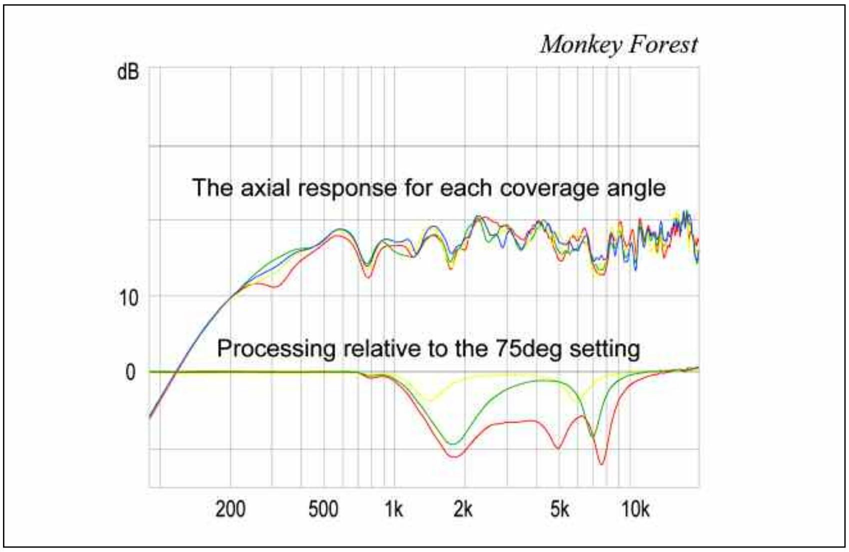

Figure 2 – The axial response for each sidewall and switch setting. I tested symmetrical configurations only.

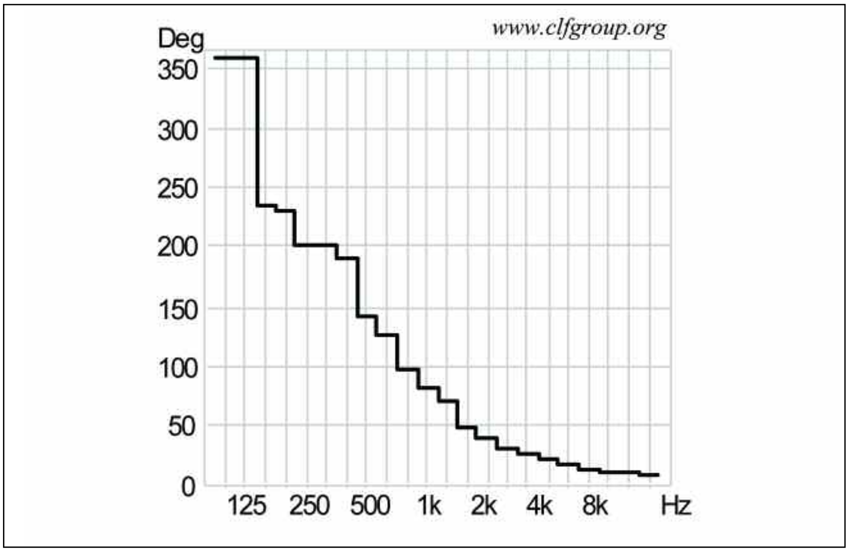

Figure 3 – The vertical beamwidth vs. frequency for a single MiniCOMP.A.S.S.

I'm normally not a big fan of powered loudspeakers that don't have sensitivity controls. Many of these devices are excessively sensitive, and require a big level drop from the preceding device to avoid being overdriven. The miniCOM.P.A.S.S. is lacking a level control but at least its designers picked a sensible sensitivity, quoted at 3.2 dBu (1.12 Vrms) on the spec sheet. This correlated well with the level that concluded the power test. This sensitivity will allow the average console to be operated at optimum (meter zero on a volume indicator), and the resultant level will allow the maximum SPL to be produced from the loudspeaker for typical broadband program material. The DSP (if present) can be operated at unity, which is the default setting for most of these devices. That's a no-brainer, but many powered loudspeakers don't get it right. One can also understand the lack of a sensitivity control on a device that will likely be flown and inaccessible during a show – a different scenario entirely that a portable system that might be placed on a stage or stick. The noise floor of the miniCOM.P.A.S.S. was barely audible, even at close range. Since the LED indicators on the loudspeaker will not be visible during a show, the FOH operator can know they are approaching maximum level as they approach meter zero on the console. If lower-than-maximum sound pressure levels are needed (most applications!) then the device preceding the miniCOM.P.A.S.S. can be used to attenuate the drive signal so that the console can operate at meter zero. If manufacturers are going to remove the sensitivity controls from amplifiers and powered loudspeakers (or recommend that they be operated "wide open"), then this is the way to do it. Kudos to Outline for getting it right. An optimum system gain structure should be easily achieved with the miniCOM.P.A.S.S.

ON-AXIS RESPONSE

I measured the axial response of a single miniCOM.P.A.S.S. at 8 meters – well into its far field. As expected with a powered loudspeaker with processing, the response was relatively flat and no additional equalization should be necessary for a single unit. A switch on the back of the loudspeaker changes the signal processing for each setting of the side walls, allowing the axial sensitivity to remain flat. **Figure 2** shows the difference between the settings (75 degrees as reference), and the axial transfer function magnitude for each setting of the sidewalls and processing switches. This is a good illustration the extreme sensitivity of frequency response to physical factors at or near a loudspeaker, and of the use of corrective equalization. If the sidewall feature were implemented without the processing switches, the system tech would be tasked with re-equalizing for each setting.

RADIATION PATTERN

Since the "special sauce" of this box is the variable horizontal pattern, I did spherical measurements for each setting of the mechanical sidewalls using the appropriate setting of the integral processing switches. There is a separate switch for each sidewall. The settings were easy to change – no tools required – and could easily be set in situ by a tech on a lift or ladder. Spherical data was gathered for all four configurations. There are separate CLF data files for each horizontal configuration.

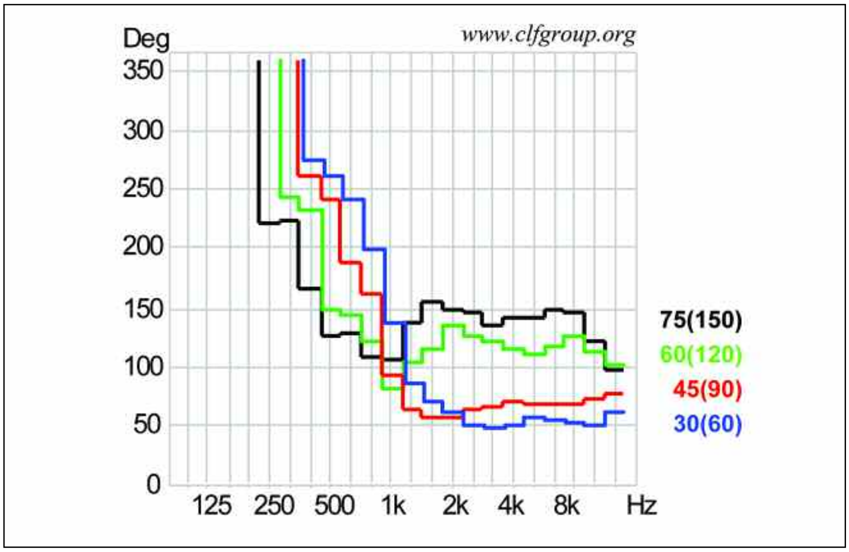

Figure 4 – The horizontal beamwidth vs. frequency for each setting of the mechanical sidewalls (and processing switches) of a single MiniCOM.P.A.S.S.

The vertical pattern is typical of line array boxes in general – broad at the LF end of the spectrum with increased narrowing with increasing frequency. **Figure 3** shows the vertical angle vs. frequency for a single element. A line array achieves its vertical radiation pattern control through interaction of the loudspeakers. By design this interaction is minimized for a curved array at high frequencies by the very tight pattern of each device (allowing the pattern to remain broad), and maximized for both curved and straight arrays at low frequencies by the very broad pattern of each device. Long arrays are capable of achieving LF pattern control when a sufficient array length has been established. In short, at high frequencies straight arrays encourage acoustical coupling and curved arrays discourage it. Straight arrays produce the highest SPL and the most narrow vertical radiation pattern. Curved arrays produce a broader vertical coverage pattern. It's the system designer's job to determine whether "curved" or "straight" (or a combination) is best suited for an application, and computer modeling is the only practical way to come up with what will happen with a given configuration. The miniCOM.P.A.S.S. is contoured on the top and bottom (10 degrees) to allow either straight or curved arrays to be formed, and I assume that the rigging hardware allows finer angular increments. I will show the pattern interaction in the vertical plane later in this profile.

HORIZONTAL PATTERN

The horizontal pattern is where it gets interesting. The sidewalls that control the HF radiation are hinged and pivot to four different stops – 75, 60, 45 and 30 degrees. The left and right side walls can be adjusted individually allowing asymmetrical patterns (I did not test for these). The adjustment can be made on-site and without tools.

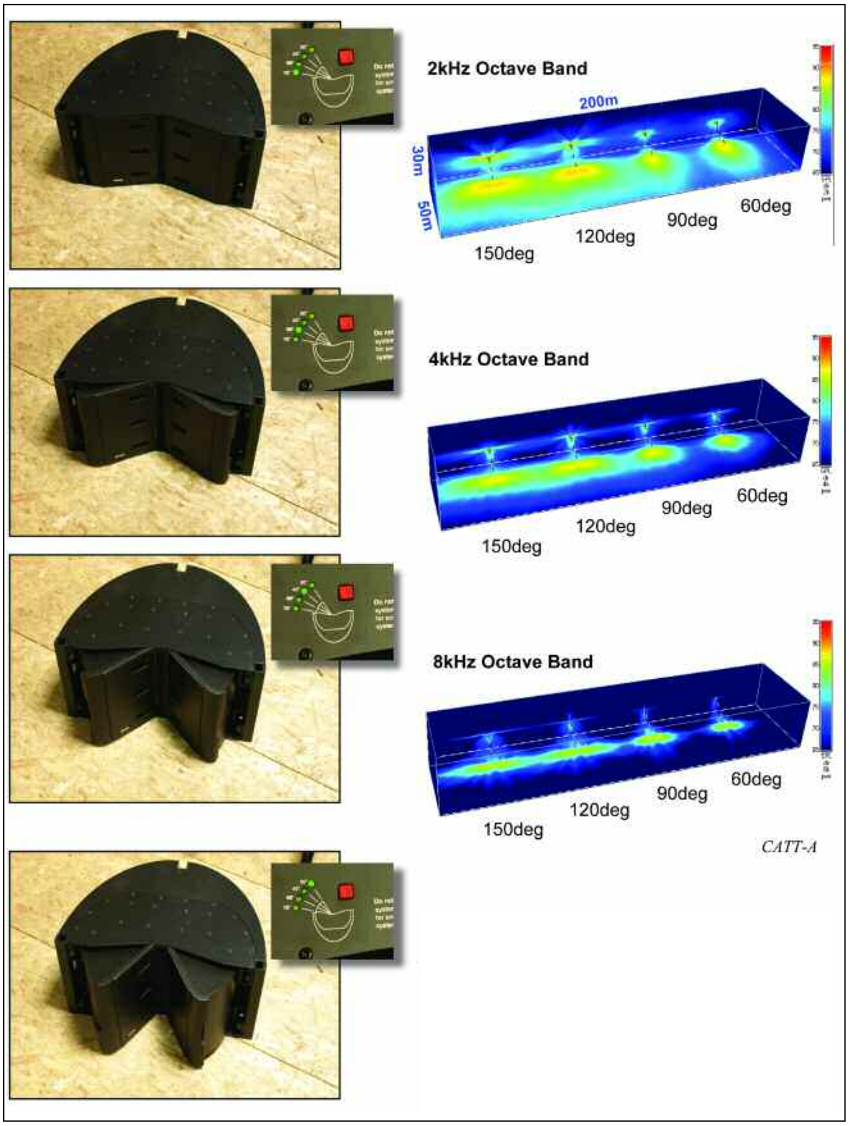

Figure 5 – Above: Coverage maps of each angle setting modeled side-by-side in a very large room. Note that the actual coverage angle is twice the angle selected on the loudspeaker.CATT-ALeft: The mechanical angle adjustment and corresponding switch setting. There are two switches – one for eachsidewall (one shown)

Figure 4 shows the horizontal angle vs. frequency for each of the settings, overlaid on a single plot (works as advertised!). **Figure 5** shows some coverage maps for each setting for mid and high frequencies made by placing a miniCOM.P.A.S.S. data file for each horizontal setting side-by-side in a large virtual space (200 x 50 x 30 m) at a 10 m trim height (try THAT without a computer model!). The horizontal pattern shaping works well, and the pattern produced for each setting holds up for all frequencies whose wavelengths are short enough to be affected by the sidewalls (> 1 kHz). Nice. **Figure 2** makes this very clear. The manufacturer states that in the next upgrade the system will automatically recognize the position of the sidewall, making the push-button switch unnecessary.

That concludes the overview of a single miniCOM.P.A.S.S. There's a lot more that can be known about this loudspeaker, but nothing else that we need to know to complete a system design. Computer modeling will now be used to examine some possible array configurations. There can be good confidence in the predicted results since we are starting with good data for the building block.

ARRAY MODELING

I modeled some potential array configurations using the generic ARRAY0 model of CATT-Acoustic. This allows the array to be tweaked in a virtual environment by looking at polars and balloons at various distances, frequencies and resolutions. When the designer gets a "keeper" the file can be saved and dropped into a computer room model. It should always be remembered that measurements and predictions are not the same thing, and that there will be factors that affect the actual response that are not accounted for in the predictions. I've done enough comparisons of measured vs. prediction in the past to get a feel for the expected correlation. My experience is that it can be quite good if one doesn't get too fancy or resolute in what they try to predict.

I created the cabinet wire frame and laid out an array in Google Sketchup,

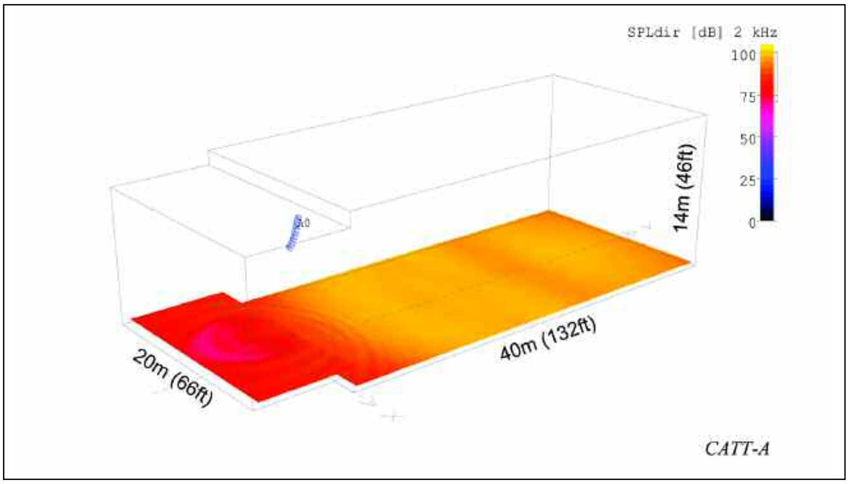

Figure 7 – Array3 wire frame shown in simple room

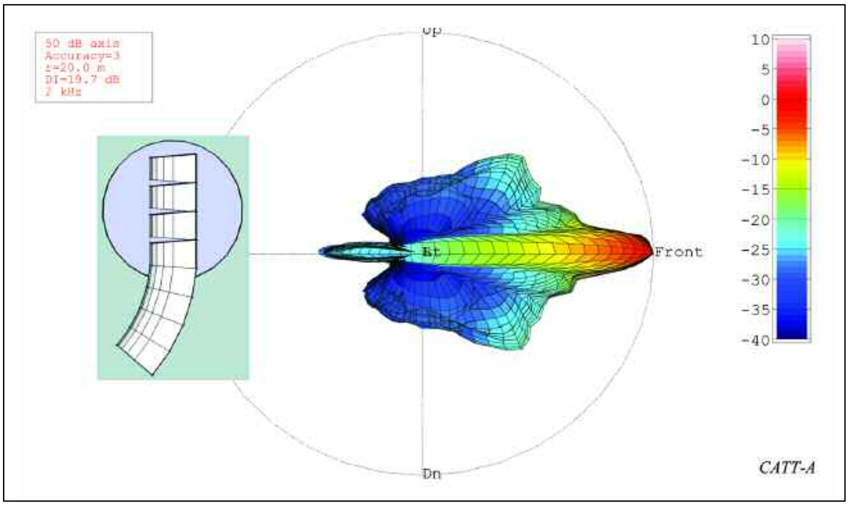

Figure 8 – 2 kHz balloon of the straight segment of Array1.

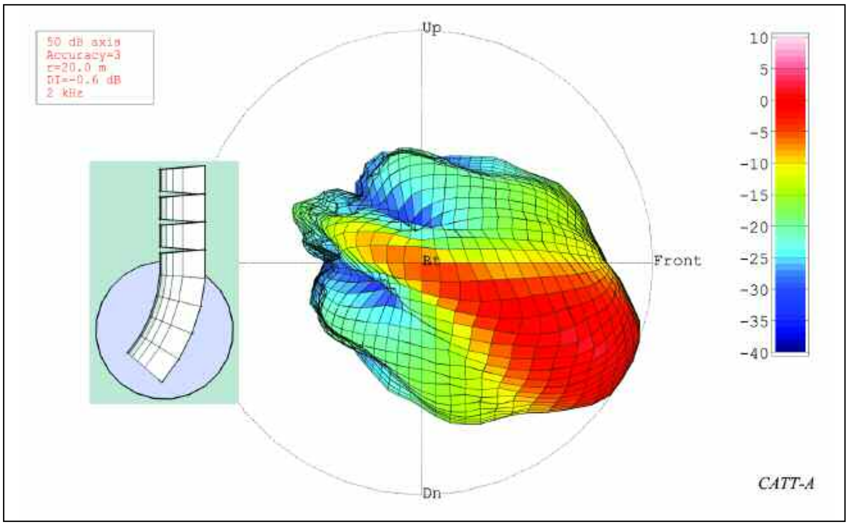

Figure 9 – 2 kHz balloon of the curved segment of Array1.

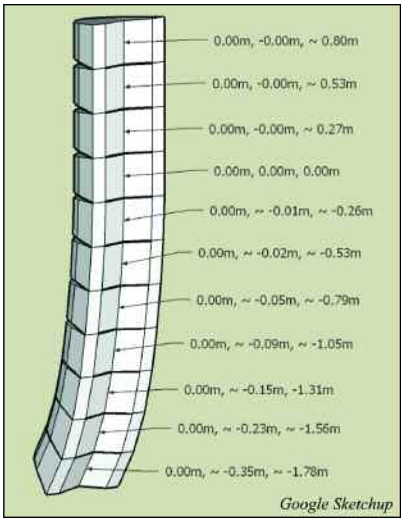

Figure 6 – A Sketchup model gives the box coordinates and angles for CATT-A (Array3 shown).

which gave me the coordinates for import into CATT-Acoustic (**Figure 6**). I also used Sketchup to create a simple room with a large, flat audience plane (**Figure 7**). Room modeling programs consider each array element to be a "point source with directivity" that is located at the specified XYZ coordinate. This is a nice example of how programs can work together to capitalize on the strengths of each. Sketchup has become a universal room modeling and array creation environment, thanks to a plug-in from Rahe-Kraft (www.rahe-kraft.de) that allows model export to the major room modeling packages.

Outline will release their D.I.V.A software later in the year to help the user correctly configure a miniCOM.P.A.S.S. array.

SOME EXAMPLE ARRAYS

To illustrate the sensitivity of line array variables I created three arrays of the MiniCOMP.A.S.S. The first (Array 1) is an array that someone might produce by intuition alone, followed by a Array 2, which fixes some of the problems of Array 1. Lastly I show Array 3, which produces the response that most expect from a large line array.

Array 1 combines a four-box straight section (using the 45-degree miniCOM.P.A.S.S. horizontal setting) and a four-box curved section (using the 75-degree setting). To better

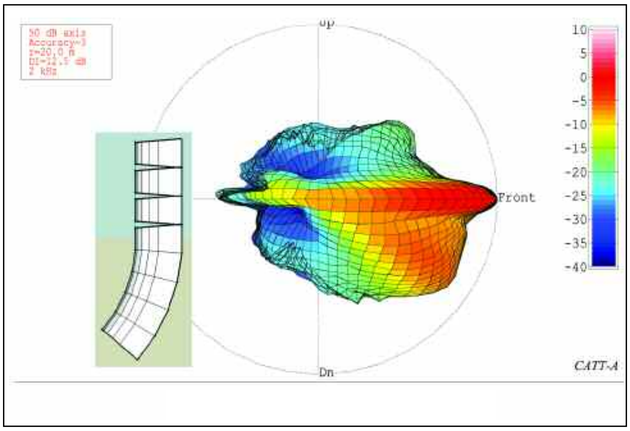

Figure 10 – 2 kHz balloon of Array1 – all elements on.

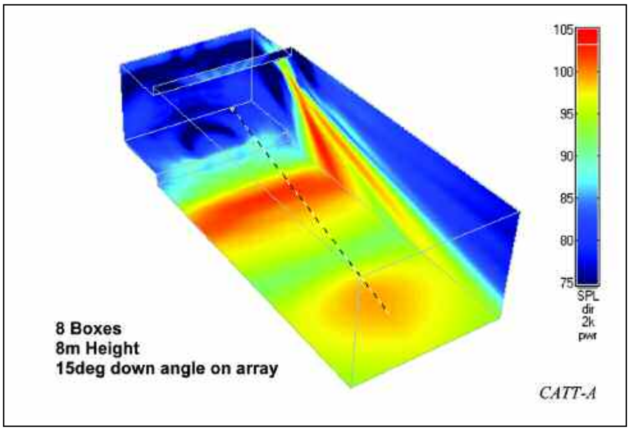

Figure 11 – 2 kHz balloon projected into a room.

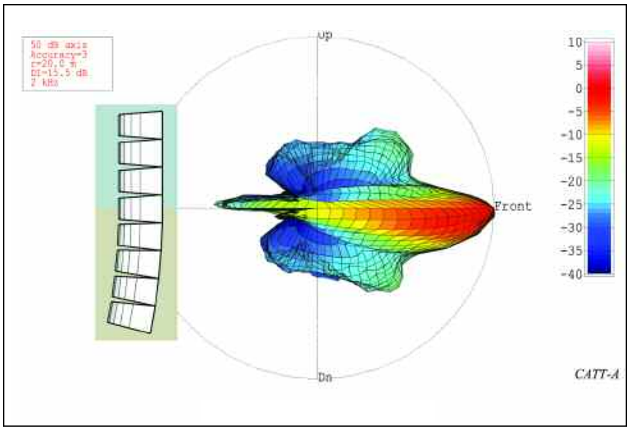

Figure 12 – 2 kHz balloon of Array 2.

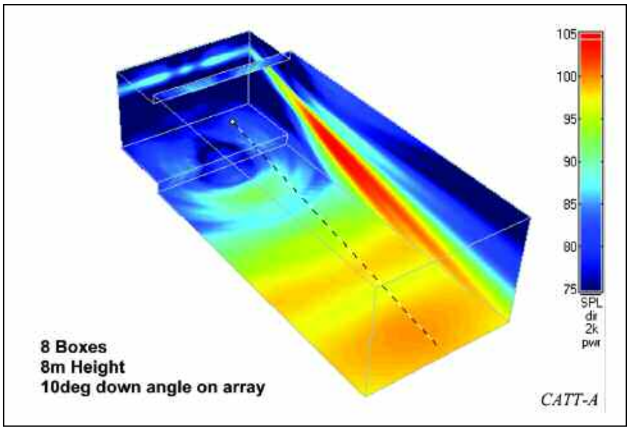

Figure 13 – Array 2 at 8m height.

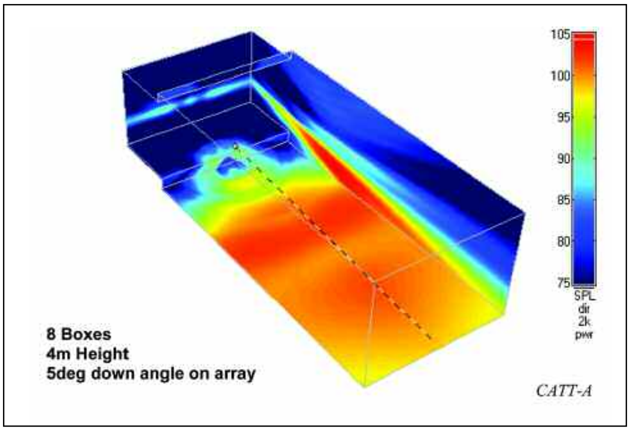

Figure 14 – Array 2 at 4m height.

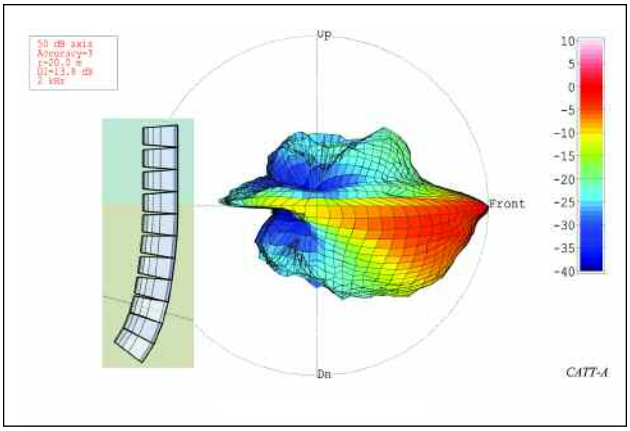

Figure 15 – 2 kHz balloon of Array 3.

illustrate this I produced a 2 kHz balloon of each of these sections (**Figures 8** and **9**). I then plotted the combined response. **Figure 10** shows the side view of the 2 kHz octave-band balloon of Array 1 (all of it) array at 5-degree angular resolution. The array response is determined by calculating the magnitude response from the relative level and time-of-flight from each array element at some angular resolution and distance – a huge mathematical task that is a practical impossibility without a computer. I initially plotted it at 2 degrees, and found this resolution to be excessive for the intended purpose. Backing off to a lower resolution can speed up the calculations and plotting considerably, without sacrificing any practical detail about the array's performance. The sound system designer must consider the balloon for each octave band during the design process, but space constraints only allowed one octave band to be shown in this article. I picked 2 kHz due to its importance to speech intelligibility and music clarity. The straight section does indeed sum to produce a very hot lobe that can be used to cover the back of the room (flash back to the quoted SPL for a four-box array earlier in this article). The curved bottom section has less coupling and the pattern spreads out more – useful for covering the front rows. Array1 has a big problem though. There is a hole in the coverage between the straight and curved sections, and much of the audience will be sitting in this problem area (**Figure 11**). Ironically, these are the expensive seats in many venues. By curving the array too sharply we have produced a discontinuity in the radiation balloon.

Array 2 corrects the problem by reducing the angle between the boxes making up the curved section of the array. The radiation balloon more closely resembles the array shape (**Figure 12**). But, if this array is flown too high, it will miss much of the audience (**Figure 13**). I had to drop it to just above stage height to get even coverage (**Figure 14**). So, this one is a "keeper" if it can be placed in the proper position. Unfortunately this height would be a major problem for a center array, as this puts it right in the middle of the action. It could, though be considered for the left and right arrays of a left-center-right system.

Array 3 adds three boxes to Array 2, with a gradually increasing downward angle (**Figure 15**). This array can be flown much higher than Array 2 and still provide even coverage to the front rows (**Figure 16**). This is the "holy grail" response of flown line arrays, but it is also the most expensive of the three arrays. The octave-centered polars of Array 3 are shown in **Figure 17**.

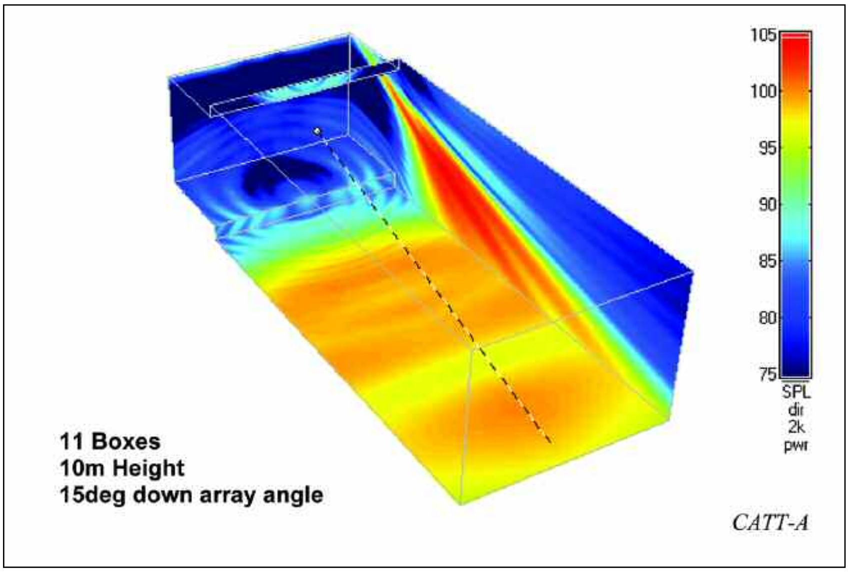

Figure 16 – Array 3 at 10m height.

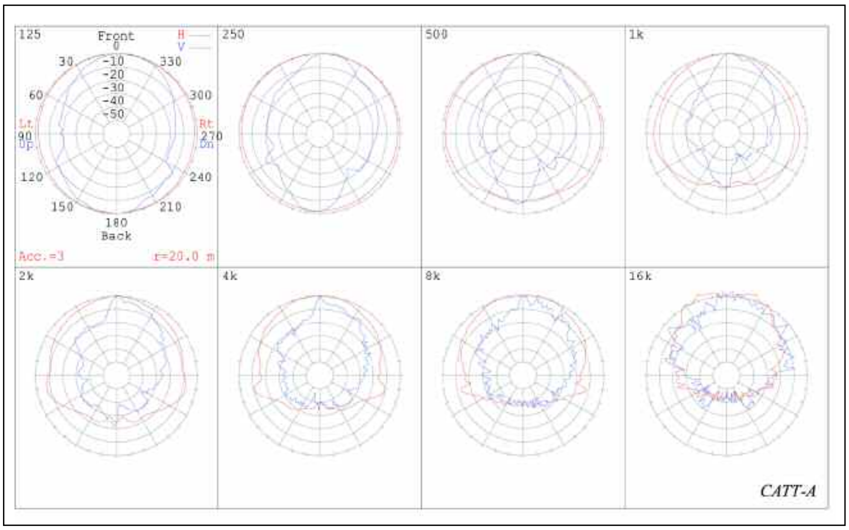

Figure 17 – Octave-centered horizontal and vertical polars of Array3.

My example illustrates some of the caveats regarding line arrays. When configured and placed properly, they can produce superb results. But Array 1 shows that intuition alone can lead to very poor coverage at a very high price. Line arrays are no place to cut cost or implement value engineering. Changing the number of boxes changes the pattern, and there's a lot more to it that just creating nicely curved stacks of boxes.

One big misconception about line arrays is that radiation pattern looks like the physical shape of the array. As I have shown it may not, and even if it did such a pattern may not be appropriate for the room geometry. The listeners on the axis of the straight part of the array are likely at a much greater distance than the listeners on the axis of the curved part. The acoustical behavior depends on more than just the physical shape. The relative levels, arrival times and element coupling must also be considered. There is no "one shape fits all" approach regarding line arrays, and something that looks aesthetically pleasing and uniform can result in very poor coverage if the pattern doesn't fit the audience area. It is incumbent on the sound system designer to make sure that the array design is appropriate for the space. Software tools are essential for dealing with the interactions, whether provided by the array manufacturer or a thid party.

The modeling showed that any of the configurations can produce well in excess of 100 dBC in the audience area, and the power testing used to determine the maximum SPL of a single miniCOM.P.A.S.S. proves that the projections are based on reality. This is a far-field anechoic response, so it represents the worst case. There will be additional "room gain" indoors, but this is venue-dependent. 6-10 dB is typical in auditoriums.

CONCLUSION

The marketplace has become crowded with line array boxes in recent years. Each manufacturer tries to innovate an advantage in terms of performance, size, rigging, price and ease-of-use. The miniCOM.P.A.S.S. integrates processing, amplification and mechanical pattern control into a single, compact system. It serves as a building block for a system designer who needs to exploit the strengths of this popular array configuration.

I started by collecting some needed data on a single miniCOM.P.A.S.S. loudspeaker. The specifications provided by Outline correlated well with my own testing with regard to maximum SPL. The rated coverage angles proved to be accurate, but coverage angles alone are not sufficient to design a sound system. Spherical data for the miniCOM.P.A.S.S. is available in CLF and EASE data formats. Armed with this data, some configurations were "built and tested" in a virtual environment.

Line arrays are complicated devices that are tricky to implement by intuition alone. I have shown some of the considerations and software tools that are required to determine the coverage of a given configuration. It is incumbent on the system designer to model the array interactions at all octaves-of-interest and come up with a design that is appropriate for the intended application. The miniCOM.P.A.S.S. has served as a vehicle for demonstrating the required data and computational tools for line array design. ■

Pat *and* **Brenda Brown** *own and operate Syn-Aud-Con, conducting training seminars around the world. For more information, go to www.synaudcon.com.*