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Line Arrays: A Buyer's Guide- Part I

By Rod Falconer

Line arrays have been readily available to the installation market for the past two to three years, and have enjoyed "favored child" status with most of the major audio manufacturers during this period. The question remains, however, whether they represent the appropriate solution when upgrading your existing system or designing for your new facility. The answer, as always, depends on your application. In this issue, we'll look at what's currently available on the market, how arrays operate, and some key purchasing issues to consider. Next issue, we'll dive into some additional concepts to consider, as well as take a glimpse at the next generation.

Let's look first at the goals, and the variety of solutions available in today's market. To set a baseline, let's presume that the objective of any quality system should be to provide clear communication and accurate music reinforcement, and that system performance should be as consistent as possible from seat to seat.

The development of the modern line array has yielded a new set of tools that potentially represent a substantial improvement over traditional systems, as we'll explore further. Their performance, however, hinges greatly on the design of the array elements, and how they are arranged and installed. Line arrays can definitely be created poorly, creating extremely uneven frequency and SPL response, and unwanted power lobes at various frequencies. We'll explore some of these issues, and how to anticipate in advance what a given array will generate. While there are a number of line arrays designed specifically for voice reinforcement, in this article we'll focus on arrays that provide full-bandwidth performance.

In the church market, it's clear that effective communication of the message is the reason the system is there to begin with. For better or worse, the overall image projected by the church is also a key component to consider. Today's audiences expect a fairly high level of audio performance, and a system with poor performance communicates its own message. The bottom line is that systems have to be good enough to keep people satisfied (and returning), while being, in the truest sense of the phrase, cost-effective. With church budgets being typically tight, there is little room for getting it right next time.

Physics
The introduction of full-frequency line-arrays in the 1990's caused a certain amount of confusion because line array performance didn't fit into the neat descriptive boxes we'd used in the past to define system performance. New terms were introduced, many of which attempted to articulate in simple terms the complex performance of the waveforms developed by an array. Thus, terms like "cylindrical wavefront" and "line-source" were used to educate consumers and users. While these terms may have oversimplified many of the concepts, keep in mind that they were all introduced in

an attempt to explain new performance parameters. True array performance was different enough to justify the new terms.

Line arrays can be an elegant and efficient solution to a number of problems that have plagued sound reinforcement systems since the second speaker was placed next to the first. While line arrays are capable of a substantial increase in performance over a traditional system, they must be designed and installed with careful regard to the physics involved with array behavior. Bolting cabinets into a linear arrangement will not create the performance of a true full-bandwidth line array, unless the array elements are specifically designed to create a line-source.

Just about any audio manufacturer can park gear on a stage and make it sound good. The challenge is to make the total system, designed for the correct coverage area, respond so that the performance is consistent no matter where you are in the facility. This, by the way, is where the consultants and design-build contractors earn their money.

To understand the difference between a demo and a well-engineered and installed system, it's extremely important to understand the root physics issues. Remember that when two radiating acoustic sources reproducing the same signal have overlapping coverage patterns, the result includes phase-induced interference. This interference is exhibited in a number of forms, most notably cancellations (comb-filtering) within the primary coverage area, leading to variations in SPL and frequency response and off-axis side-lobes (uncontrolled energy outside of the primary coverage pattern). The amount of interference is directly related to the distance separating the two sources, and wavelengths related to this distance (frequency). If, however, the distance between acoustic sources can be reduced, the negative, or destructive, cancellations within the primary coverage area are minimized. If the acoustic center spacing can be reduced far enough, the result is constructive interaction, where the output of the acoustic sources sum coherently, thereby increasing the total output. Note that these are universal principles and apply to any array or cluster of speakers.

The optimum solution, therefore, would be to create a system that effectively reduces the distance between array elements to zero, thereby creating a phase-coherent single acoustic source. This theoretical result would eliminate phase-induced cancellations, while minimizing off-axis lobing and destructive inter-element interaction. Creating a single acoustic source capable of generating a configurable coverage pattern with sufficient output is the goal of modern array designers.

Line array performance

Please note that a line array, by definition, is an assembly of devices in a line, no more, no less. Since single acoustic devices radiate energy as a point source, expanding both horizontally and vertically, it's not typically a great idea to create an array of devices whose overlapping patterns will interact. Properly constructed, however, purpose-built elements in a line array will have coverage patterns that can couple coherently and combine to form a single acoustic source in the form of a line (line-source), creating phase-coherent summation in the primary coverage area.

A well-designed line array forms a radiating plane or ribbon of energy that expands primarily in one dimension (as opposed to a point source), with destructive interference creating null zones, or areas of cancellations (typically above and below an array). These null areas can be 12 to 15dB below the level of the primary coverage area. While this is also frequency and array-length dependent, it's a huge potential benefit for situations where gain-before-feedback is problematic, and where poor mic technique and extensive lavaliere mic use is prevalent (sound familiar?) The primary advantages of a well-constructed line array remains it's ability to create very even frequency and SPL coverage throughout the seating areas (since the array can act as a single acoustic source), and better stereo imaging, since you've reduced the amount of interference within the primary sources.

Line-arrays are also by nature very efficient. Since array elements do not have to overcome cancellations created by their interaction with neighboring elements, the physical size of the system is frequently smaller than a traditional system. The primary benefits here are better sightlines, and in many instances, lower system costs (fewer cabinets, plus fewer amp channels). Arrays also provide the ability to transmit energy efficiently over quite a distance, particularly in the higher frequencies. While this is again dependent on the size and shape of the array, this can help offset HF air-loss absorption, and minimize or eliminate the need for delay fill speakers (again, fewer speakers and their related amp channels needed).

An additional benefit is the apparent extension of the near field, exhibited as 3dB attenuation per doubling of distance (as opposed to far field and point source's 6dB attenuation per doubling of distance). Please note, however, that near-field extension is highly dependent on the size and shape of the array, frequency, and listener position. For example, an array may transition from near field to far field behavior in a matter of a few feet at the lower end of the frequency spectrum, while high frequency transition from near field to far field may take place hundreds of feet from the array. The immediate impact of extended near field though, is an extended level of immediacy or presence from the mid-frequencies on up.

How it works
What you hear as broad bandwidth line-source array behavior is primarily dictated by three factors: the size (length) of the array, the spacing between the individual components, and listener position. Simply stated, devices will couple coherently if their acoustic sources are within a half-wavelength of each other at their highest operating frequency. Alternately, if the acoustic centers cannot be placed close enough together (due to the size of the wavelength and the size of the driver), devices will also couple coherently if their output’s waveform is relatively flat (curvature less than a 1/4 wavelength) and their total radiating area occupies at least 80% of the face of the array1. Research has shown that the flatter the waveform; first, the closer the performance will be to approximating a perfect line source, and second, the lower the side lobes will be, keeping spurious energy sources at bay. This second set of criteria tightly defines how much an array can be articulated (shaped) before the inter-element coupling breaks, resulting in a series of independent point-sources (with the inherent comb-filtering issues).

To put this in physical terms, if two devices operate up to 200 Hz, their acoustic centers can be placed up to 34” apart (1132 [speed of sound] divided by 200 [frequency], divided by 2) and still generate a coherent wavefront with relatively well-behaved side lobes (at least 12-14dB below the level of the primary lobe). On the other hand, compression drivers operating up to 17kHz would have to be small enough to allow their acoustic centers to be placed within 0.4” in order to maintain the first coupling criteria. Thus, it is relatively easy to build a coupling array in the lower frequencies, up to the point where driver size forces the second criteria to come into play. This dictates both the size of the driver, and the amount of separation available between array elements. It also means that any driver operating above approximately 1kHz will probably need to adhere to the second set of coupling criteria.

Since acoustic sources radiate spherical wavefronts, some manipulation must take place in order to adhere to the second set of criteria, making the wavefront transition from one that is spherical, to one that is relatively flat. The first device to address this aspect of high-frequency waveform manipulation was the DOSC waveguide, introduced by L-ACOUSTICS in the early 1990’s. Later product development included other forms of complex manifold technology as utilized by manufacturers like Adamson, Meyer Sound, Nexo, etc. This is a highly proprietary field, since the success of this device will dictate the success of the coupling performance in the higher frequency range. The effects of a curved wave-front’s negative interaction within an array is well documented, so make sure the devices you’re interested in can support their claims.

One notable exception to maintaining these coupling criteria is EAW, whose arrays incorporate the approach that various technologies should be used for the various frequency bands. In other words, line array behavior should be maintained where inter-device spacing allows, and other relevant technologies used where appropriate for the mid-range and up. Thus, high-frequency coverage is handled via traditional horns. This approach takes advantage of low-frequency coupling and its inherent efficiency gains. The use of horns in the mid-high area allows specific coverage angles to be used for specific coverage areas, which typically increases the total vertical coverage of a given array. This approach, however, should not be considered as an example of line-source behavior in the upper frequency ranges.

It should be noted here that high-frequency manifold developments reflect working with the physical waveform to create seamless coverage over a specific coverage area. Tailored pattern control has also been realized by manipulating phase and SPL between various array elements, though care must be exercised in order to maintain the “single source” performance benefits of the array. This DSP-centric approach represents the next wave of array development, and the new products by EAW, Meyer Sound and others reflect this approach. More on this in the following article.

An interesting side note is that the HF waveform manifolds from Adamson, L-ACOUSTICS, Nexo, Meyer Sound, etc. all seek to generate a seamless ribbon of energy along the face of the array. The question to be raised then, is whether a ribbon driver (with a waveform that inherently meets the coupling criteria) is viable as a driver in line arrays. At this point only three professional audio manufacturers, SLS Loudspeakers, Stage Accompany, and Alcons Audio are actually using HF ribbon driver arrays. While early ribbon drivers could not handle sufficient power levels or generate sufficient SPL, recently developed ribbon drivers take advantage of new materials and technologies to allow greater short-term power handling than was previously available. While generally still a few dB shy of matching the output of systems using traditional HF horns and compression drivers, potential ribbon driver benefits include coherent coupling between HF components (given the nature of their inherently flat waveform), greater transient response, and much lower distortion levels than that which is typical of a compression driver. However, spacing between driver assemblies still must adhere to the 80% rule in order to create coherent summation.

System Design
One of the beauties of line array design is that it can be molded to fit fairly specific performance requirements. Since the elements (theoretically) couple coherently, an array's shape can be straightened to generate additional SPL for long-throw situations. Essentially, the straight portion of an array increases the total number of components covering a given area, thus effectively decreasing the total area being covered by these components, providing denser coverage (and thus higher SPLs). Similarly, if you want to reduce the SPL (to the near-fill areas for example), you can curve the array (within limits) to dissipate the energy. The key issue here is that the array can be articulated as long as the coupling criteria are maintained. If the coupling criteria are not maintained, the array will generate variances in both SPL and frequency response within the primary lobe, and unacceptable off-axis lobes. Thus, the primary consideration is to what degree they can be separated before the cylindrical waveform breaks.

Articulated vs. constant-curvature arrays
Another approach taken by a number of manufacturers is to move the apparent acoustic centers behind the physical device, where they merge to effectively create a single acoustic source. Most arrays using this approach create a constant curvature array (think of the merged acoustic centers being the reference point, and the front of the array elements creating a section of an expanding sphere). This approach, used by L-Acoustics in their ARCS series and by Renkus-Heinz in their RPA arrays for example, yields a theoretical single acoustic source whose output has a curvature that remains constant (hence the name). Nexo also uses a variation of this approach in their Geo-series array elements.

While solving the issue of multiple acoustic sources, systems of this type typically cannot be effectively articulated into a straight line in order to drive energy further into the house (doing so would move the acoustic centers apart). Thus, they generate consistent SPL across the length of the array as opposed to the variable output available from an articulated array. The exception to the model is the Nexo Geo system, which uses a series of internal acoustic reflectors to generate a configurable pattern. Within these limits, constant-coverage arrays do couple effectively, and should be considered in the appropriate application.

What does all this mean to a system designer/user/buyer? Bottom line, do your homework and pay attention to the physics. There is hard science behind the development of many of these products. Unfortunately, the array market has become congested, and some marketing claims may exceed what can be realistically accomplished. Looks can definitely be deceiving, but the legitimate contenders will be able to back up their claims with research, and will support independent predictive software packages such as EASE and CATT.

System Cost Issues
Beyond the individual array element costs, there are three key considerations. First is ancillary hardware costs. With many systems, a given amp channel can drive multiple components. DSP-centric systems may need a DSP channel and individual amp channel per band, per element. Thus, for a simple three-way, six element array, you may need up to 18 DSP and amp channels per side. Pricing of self-powered systems such as those available from EAW, JBL, Meyer, and Renkus-Heinz include DSP and amplification. Each non-powered system can use different DSP processing and amplification schemes, so you'll have to verify each design and its DSP and amp requirements to achieve a true "apples to apples" comparison.

Rigging
Another hotly debated topic has been the various array rigging systems. While primarily the concern of the touring production companies, the type of rigging used can also have an effect on the cost of an installation if it significantly increases (or decreases) assembly time. Early arrays used rigging systems composed of a variety of specific components to dictate the angles between cabinets. Most recent arrays, however, utilize rigging components that can be used at a variety of angles, and which remain attached to the individual array element. These captive rigging systems are generally included in the cost of the array element, while non-captive specific-value rigging components are not (which effectively increases the cost of the system). Additionally, manufacturers are addressing system costs by offering installation versions that eliminate expensive rigging components. For example, the McCauley IN.LINE‘ array elements are purpose-built for permanent installations, and JBL has introduced installation versions of their arrays.

What to watch
Vertical coverage capabilities are probably the biggest issue that affects consistent system

performance and cost. Vertical coverage is traditionally calculated as being a sum of the angles between the array elements. Thus, if a manufacturer allows their arrays to be constructed with greater inter-element angles, they will be at a competitive advantage since fewer elements are required to cover the vertical dimension. Remember however that the amount an array can be articulated is tightly defined by the coupling criteria, and exceeding the theoretical limits results in the system acting, again, as an assembly of inter-reacting point sources. Maximum inter-element angles are based on math (and as such, are absolutely definable), taking into account such things as the size of the components, size of the array elements, and listener position. Many manufacturers provide rigging hardware that will accommodate angles that exceed the theoretical maximum allowable values, so pay attention. When in doubt, have the arrays modeled in an independent predictive program, with the aim of keeping frequency and SPL coverage as consistent as possible.

Predictive software

Virtually every manufacturer has created design support software to calculate the number of elements required for a given system. Some are more extensive than others, providing information such as aim angles, center of gravity, horizontal coverage, weights, heights etc. Some of the newer tools also indicate SPL variation over distance at various frequencies, which is extremely useful in determining the effects of various array configurations, transition to far-field effects, etc. One of the most cost-effective methods to explore array behavior before you buy is to examine the array's performance in modeling software that includes phase information (since this is the basis for line array performance.)

Both the CATT and EASE 4.0 programs now utilize phase information to calculate array coverage in a room and inter-element interference, shown as coverage balloons at a given distance from the array. Using information provided by the manufacturers (EAW, Electro-Voice, JBL, L-Acoustics, Renkus-Heinz, and others support EASE array models), these predictions illustrate the interaction between elements at various frequencies and inter-element splay angles.

Since the response of any array changes with distance (again, based on the transition from near field to far field and contingent on the size and shape of the array), these balloons should be examined at various distances. However, they do show what happens to system response as the arrays are articulated. Request these predictions (based on the specific array you'll need) since they represent a relatively objective perspective on different arrays and their behavior at various inter-element angles and frequencies. Again, it's easier to get an array to couple nicely at lower frequencies, so make sure you see plots reflecting the higher frequencies as well.

The next arena of battle is likely to be steerable patterns and horizontal coverage. Over the years, the industry developed an extensive set of tools in the form of a variety of constant-directivity horns to tightly control coverage within specific portions of a coverage pattern. Most line array offerings are not yet as extensive, offering a limited palette from which to attempt to construct exact coverage patterns as the horizontal pattern transitions from the front of the house to the back. To date, only the McCauley offers array elements in a full range of horizontal coverage angles (60o, 90 o, and 120 o versions), which can be used as needed within an array. Other remaining issues include the aspect of smooth interaction from adjacent arrays, as would be found in systems employing left-left/left/right/right-right, LCR, and matrix-fed configurations.

Conclusion

Line arrays present the capability of providing significant performance increases over traditional designs, if constructed accurately. It remains the buyer's responsibility to examine particular manufacturer's claims in the light of the available research and objective modeling programs, especially if the claims yield suspiciously advantageous bids. In the next issue, we'll explore the potential of DSP-based systems, as well as some other alternate solutions available on the market today.

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Line Arrays Buyer's Guide Part II

By Rod Falconer

In the last issue, we reviewed the basic physics concepts behind performance-grade line arrays. As we explore the upcoming crop of new technologies, please keep in mind that the basic concepts – driver spacing, inter-element spacing, coupling criteria etc. - still apply. In this article, we'll explore how various companies approach working within these criteria to create new levels of performance. In particular, we'll investigate the EAW Digitally Steerable Array (DSA) series, the Duran Audio Target series, and the Magnetic Audio Devices (MAD) Surface Array technology.

As before, the systems provide bandwidth (within some restrictions) that supports both music and speech. Each also represents years of development, and foreshadow the direction the professional audio industry will take. At this writing, the EAW DSA should start shipping within the month, the Duran Audio Target series should be available by the end of the year, and the MAD systems are available now.

As we discussed earlier, line array theory has been explored for decades, though physical limitations in driver spacing and the resultant interaction within the coverage area limited bandwidth to speech reinforcement. The advent of affordable signal processing in the form of digital processors, however, opened new avenues of performance for small line arrays. Introduced in the mid-nineties, Duran Audio's Intellivox system is probably the best known initial example of the potential benefits of DSP in line arrays. Designed as a speech bandwidth system, the Intellivox system provides individual DSP processing and amplification for each driver within the columnar array. By manipulating the respective phase and amplitude between respective drivers, the system provided the ability to steer the primary and secondary coverage lobes in the vertical domain, and control of their angle opening and focus distance. This allows the system to be mounted flush to a wall, providing a downward-aimed coverage pattern without physically tilting the enclosure itself.

EAW expanded on this approach with their new DSA series of self-powered systems. By providing individual amplification and on-board DSP, the DSA series is capable of impressive vertical steering of the primary coverage lobe, as well as internal high-pass and low-pass filters, parametric EQ, delay, and limiting functions. EAW employed a modular approach, providing a broad-bandwidth system (DSA250) with a sister low-frequency unit (DSA230, essentially the LF section of a DSA250 in a separate enclosure), packed in an extruded aluminum column. The aluminum case also acts as a heat sink for the sixteen amplifiers in the DSA250 (eight amps in the DSA230). [It should also be noted that JBL used a similar form and function for their EVO system though the EVO is not a line array. The EVO system provides other interesting functionality, but does not address beam steering]. The packaging allows the DSA-series units to be mounted either flush to a wall or flown, with the vertical coverage angle dictated by software programming via EAW's DSAPilot™ software.

EAW states that the intent for the DSA250 was to provide a stand-alone system with the ability to provide steerable pattern control down to about 300Hz, suitable for high output, high definition speech and music reinforcement (the DSA230 is used to provide additional low frequency power and pattern-control extension, or as a stand-alone speech-only system). Based on research developed for the very large-scale EAW KF900-series, the DSA uses similar principles to optimize system performance and provide vertical lobe steering. The DSA250 consists of a total of eight 4" drivers and a tightly –packed array of eight soft-dome tweeters loaded on a common multi-cell horn providing 120-degrees of horizontal coverage. EAW states that the system provides the ability to both shape and steer the primary coverage lobe, varying the vertical coverage pattern anywhere from 15o to 120o,within a frequency range of 200Hz to 15kHz. The tight-pack arrangement of both high frequency and low-frequency drivers minimizes inter-element lobing at the upper-end of their respective frequency ranges.

The DSA250 has a stated output of between 120 and 125dB, depending on the shape of the vertical lobe. As with other arrays, if the total surface area being covered is reduced, the available SPL is higher, and if the coverage area is larger, the available SPL is reduced. Again, the vertical coverage angle and aiming is dictated by the DSAPilot software, which also provides system monitoring and multi-level password protection. It should be noted that there are limits to the amount of vertical beam steering that should be implemented, though a more extreme angle can be obtained if the upper frequency ranges are bandwidth-limited.

As mentioned above, the DSA250 can be supplemented with the DSA230 low-frequency unit if additional low-frequency bandwidth, power, and pattern control are desired. Again, as with other arrays, the longer the array, the more control is applied to low frequencies. For example, a single DSA250 with two DSA230s (effectively three DSA230 arrays) provides frequency response down to about 100Hz(-3dB), and pattern control down to approximately 100-200Hz.

The DSAPilot software provides a number of design and operational features. After entering the coverage area parameters and enclosure mounting height, the program can pre-configure the necessary coverage and steering angles, and upload the parameters via RS485 link (Cobranet is also available as an option). System performance parameters can also be manually configured, and exported to EASE 4.0 for further study. In addition, the program provides the ability to select and highlight individual units via LED indicators on each enclosure, a nice feature when troubleshooting the installation.

The result is an elegant solution for problem environments where pattern control and intelligibility is required. While not a concert-grade line array, the DSA system is an interesting answer, utilizing the latest developments in self-contained systems.

Since Duran Audio was one of the first major manufacturers to utilize integrated DSP and amplification for every driver in a system, it was an obvious extension to develop their Target series of concert-grade line arrays. Rather than being restricted to voice-band reinforcement, the Target series is essentially a full-size line array with onboard DSP and amplification. Consisting of the T-2820 (dual 1.4"-exit neodymium drivers on individual constant-directivity horns and a horn-loaded pair of 10" cone transducers) mid-high unit and the B-215 (two direct-radiating 15's) low-frequency unit, the array intended to be primarily configured as a straight-hung array with the coverage lobe shaped and steered to cover the designated seating area. The system utilizes Duran's Digital Directivity Synthesis (DDS) processing and Digital Directivity Analysis (DDA) software to create the coverage pattern required for a given venue. The software allows the coverage pattern to be asymmetrical, providing a varying coverage pattern provided that the array configuration (horizontal and vertical, curved or straight) ands length are appropriate to the task.

The DDA software automates the system implementation process to a great degree. Room specifications (dimensions and reverberation data) are entered, along with the desired speaker mounting location. The software then calculates the phase and amplitude settings required for each driver in the array, and uploads them to the system. Duran Audio states that the resulting waveform optimizes both near and far-field coverage, and can cover complex asymmetrically shaped listening areas, while minimizing energy on undesirable areas, such as walls and balcony fascia.

The on-board amplification consists of hot-swappable four-channel (mid-high) and two-channel (woofer) units. In addition to amplitude and delay control, the on-board DSP also provides equalization and compression/limiting functions, as well as amplifier monitoring via an RS485 link. Please note that while the DSP functions allow manipulation of the coverage area, the array elements themselves are still governed by standard physics coupling criteria, as we explored in Part 1. (July/August '03 issue -KRC)

The Magnetic Audio Devices (MAD) Surface Array technology is a different animal altogether. The heart of this technology lies in the application and marriage of line physics with planar magnetic transducers. For those unfamiliar with the terminology, planar magnetic transducers are essentially large-format ribbon drivers, which traditionally have been precluded from professional audio applications because of inherent limitations in power handling, efficiency, and repeatable manufacturing procedures. However, the inherent advantages of the technology led HPV Technologies of Costa Mesa, CA to pursue methods of overcoming these obstacles.

Marketed as MAD Surface Array systems, the planar magnetic drivers used in these systems were specifically engineered to address the traditional limitations. The result is a very lightweight transducer with excellent transient and phase response, which, when arranged in an appropriate array, yields impressive power handling, efficiency, and frequency response. Since these transducers are by nature full-range, a number of other benefits are obtained. First, the only crossover in the system is for optional subwoofers. Therefore, traditional issues of crossover-related phase anomalies and intelligibility problems throughout the vocal range are eliminated. This also means that the system requires far fewer amplifiers than a large three-way line array. Next, the transducers replace all the horns, compression drivers, and low frequency cone transducers (except for subwoofers, if desired), which eliminates all the traditional issues of throat distortion, compression-driver distortion, and power compression. In addition, since the array starts as a dipole, no cabinets are needed. The result is a very lightweight, full-bandwidth system with excellent intelligibility. Frequency response for the basic array is approximately 80Hz to 20kHz, and can achieve up to approximately 145dB (1W/1m).

Beyond essential system performance, an intriguing feature of the MAD array is its performance as an array. Since the transducers can be arranged to adhere to the surface area coupling criteria (see previous article for further details), the array acts as a single device. In addition, and this is a key point, the array acts to provide pattern control behavior in both the horizontal and vertical dimensions. This means that the primary coverage lobe can be designed to keep the energy only on the seats, and off of the sidewalls, stage, and ceiling. As Colonel Klink would say, “Very interesting”.

The array is available as a standard 90o horizontal “rib”, but can be ordered to fit specific custom applications. The number of ribs required is defined by both bandwidth and power output requirements, since the total surface area of the array dictates overall system performance. Minimum requirements are typically eight ribs per side to achieve concert-level performance. However, MAD also offers a portable self-contained (self-powered/processed) system called the MuzikBox that consists of four rows of six transducers with a series of four 12” cone transducers providing low-frequency extension. The MuzikBox is designed to provide 60o of horizontal coverage, and at the time of this writing, a pair is being used to provide reinforcement for a large jazz festival in New York’s Lincoln Center.

In each of the MAD systems, the rows of transducers can be articulated to change the vertical coverage angle. The system was designed to quickly rig and fly, and so is appropriate for both fixed and touring systems. The first installation of the technology took place late last year in Sacramento’s ARCO Arena for the NBA season. ARCO Arena has been notorious for the high levels of crowd noise, and reports are that the four new arrays that were installed are quite capable of overcoming the highest ambient levels. Side-by-side comparisons with other standard concert-grade line array systems have been conducted throughout the country, generating a considerable amount of discussion in the pro-audio industry.

An interesting side note on the MAD systems is system pricing. While the initial array costs are apparently high, the overall cost for a complete system are competitive. First, less cable and fewer amplifiers are required, since the transducers are full-range. Next, because the arrays are so light, addition savings can be realized in infrastructure. For example, because each of the four arrays weighed less than 350 pounds, ARCO Arena saved approximately $70,000 by being able to delete the steel support sub structure that would have been required by any other system.

The MAD arrays represent a paradigm shift in array technology, and the Duran and EAW systems represent milestones in loudspeaker development in their own arenas. Because the MAD system uses specifically engineered drivers that can be manufactured consistently, though, other manufacturers may find it difficult to generate a competitive system. The DSP-based answer is more easily appropriated, so expect to see versions of large-format DSP-based self-powered systems like the Duran Target system from each of the major manufacturers in the near future. As other manufacturers head in this direction, the apparent advantages (and disadvantages) of the various line array elements will remain a part of the equation, so again, pay attention to the basics, and don’t get sidetracked by sexy software technology. DSP-based systems are a cool tool and have definite performance advantages in many applications, but the bottom line is that any properly designed system should sound as consistent as possible from seat to seat. It still depends on both the array element design and system designer's skills, and how physics are utilized to accomplish specific results.

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