Written by: Mark Engebretson
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WHITE PAPER

Advanced Loudspeaker Tuning Techniques

QSC Intrinsic Correction™

Sound system equalization has evolved into a body of practice that is universally employed for both fixed and portable sound reinforcement applications. Line arrays have brought the added dimension of array interaction to the process, greatly complicating the task of sound system equalization.

The successful application of preconditioning filters to any loudspeaker system must begin with goals:

- The first goal is acoustical accuracy within the nominal coverage of the loudspeaker.
- The second goal is to rectify any inherent and correctable performance defects of the loudspeakers.
- The third goal is to provide the operator a system that is easy to use and requires a minimum of in-situ manipulation.
- Finally, the user should be given a reasonable degree of adjustment latitude without imperiling the corrections developed for the loudspeakers.

There are limits to what preconditioning filters can accomplish. It is therefore necessary to begin with a well-designed loudspeaker having certain basic characteristics.

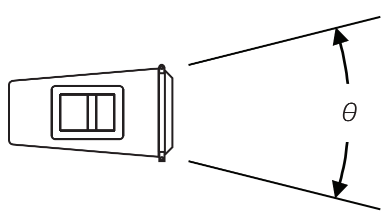
- The loudspeaker's output must be linear with respect to its input over the expected operating dynamic range.
- Line array loudspeakers should have spatially consistent horizontal coverage characteristics and monotonically-narrowing vertical frequency coverage.
- Unit-to-unit consistency is an absolute necessity. Devices that have wide variability in their performance parameters will not array predictably.
- The loudspeaker must perform consistently over time.

Fundamentals

To begin, let's review line array fundamentals because preconditioning filters cannot correct an array that has physical discontinuities – one can't repair a broken array with filters.

Line arrays as we know them today consist of segments, each of which typically comprises a single line array box or element. Each segment has coverage limitations that are frequency dependent. Using the relationship in Figure 1 enables easy prediction of the coverage of a device, or the frequency at which certain coverage obtains. For a 10-inch high line array segment, one can predict the vertical coverage will be 10° between -6 dB points at about 9.6 kHz.

10-in Segment	
2 kHz	48°
4 kHz	24°
8 kHz	12°
16 kHz	6°



$$\theta_{-6} = \frac{24{,}000}{f \; l}$$

where:

θ_{-6} = -6 dB included coverage angle (degrees)
f = frequency (Hz)
l = length of the array segment (m)

Figure 1
Segments and Line Arrays

The formula also enables predicting the coverage limits for a large horn. For example, using this formula, one may calculate that a 60º horn which is 1 meter tall could maintain 60º control down to about 400 Hz. Similarly, a 40º device one meter tall will control to only 600 Hz.

The formula can be used to predict coverage of a direct radiator, a horn device, a line array element, or arrays. It shows that a 10-inch high line array element has about 48º of vertical coverage at 2 kHz, collapsing to the neighborhood of 6º at 16k. At high frequencies, line array vertical coverage will separate into a series of lobes and nulls when the box-to-box splay angles exceed the coverage of the individual segments.

Array Shape

Nothing is more critical to successful line array deployment than the shape of the array. As with waveguides and horns, there are array shapes that work well, are easily deployed and sound good, while others fall short. There are five basic array shapes.

- Straight arrays have no splay between array segments.
- Arcuate arrays are arrays with a constant radius or a constant splay angle between vertically-symmetrical elements.
- J-shaped arrays combine a straight array with an arcuate array to increase the vertical coverage.
- A spiral array is an array that has a decreasing radius of curvature from the top to the bottom of the array.
- Finally, there are convoluted arrays. There may be applications for convoluted arrays somewhere, but discussion of them is beyond the scope of this paper.

Straight Arrays

A straight array narrows in vertical coverage with frequency. Figure 2 shows nested, octave-band polar response traces from 500 to 8 kHz for an array that is nominally 4 meters long. Each of the elements in this case is one-half meter long.

On the upper right side of Figure 2 is a chart illustrating nominal vertical coverage vs. frequency. At high frequencies the coverage narrows monotonically with the product of frequency and the length of the array. This particular array will have approximately three-quarters of a degree vertical coverage between -6 dB points at 8 kHz. For all practical purposes, it is virtually unusable at high frequencies.

Summary of straight array characteristics:

- Monotonic narrowing with f & l
- Far field distance varies with f & l^2
- Directivity = N · segment directivity
- Best (far field) coherence
- Very narrow HF beamwidth

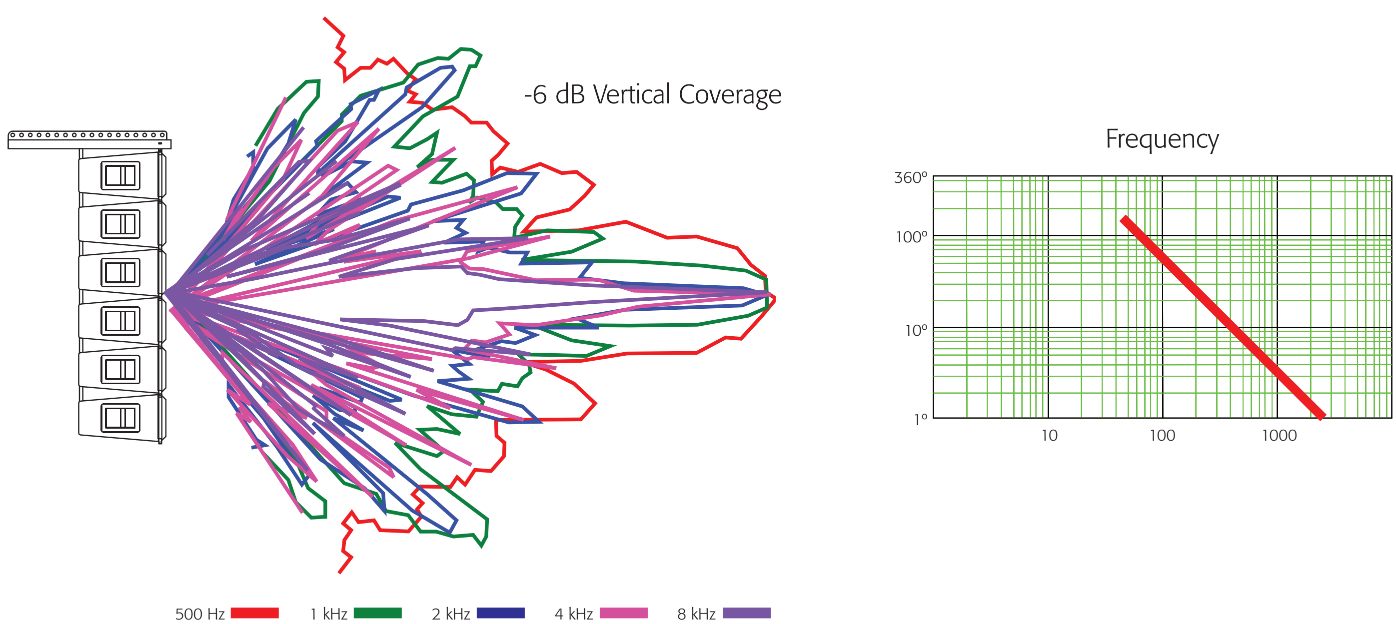


Figure 2
Characteristics of Array Shapes - Straight

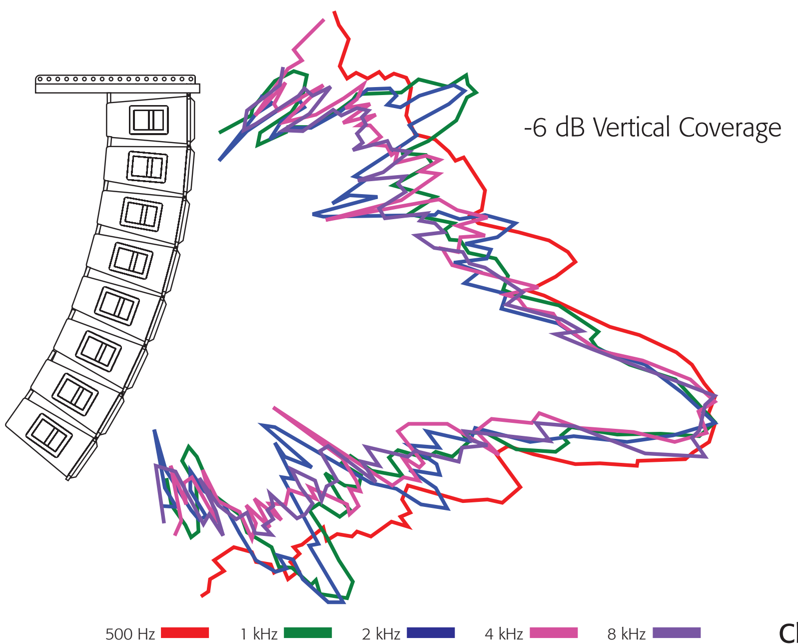
Arcuate Arrays

Figure 3 shows the same array with 1° splay angle between each of the eight elements. Note that 500 Hz (the red trace) hasn't changed significantly, but the rest of the frequency coverage charts describe a nearly constant vertical angle. The total coverage is equal to the sum of the splay angles in the array. Again, the chart on the upper right approximates the coverage versus frequency. Vertical beamwidth narrows until divergence of the elements creates the desired coverage angle.

Figure 4 shows the same array, but now with 5° between each of the elements. The included coverage angle is similar for all frequencies, approximately 35 degrees. Looking at the coverage angle versus frequency chart, the coverage angle has increased, while extending lower in frequency.

Summary of arcuate array characteristics:

- Constant directivity
- Coverage = (N-1) (splay angle˚), Example: 8 @ 1˚ = 7˚
- Uniform far field boundary
- Excellent (far field) coherence



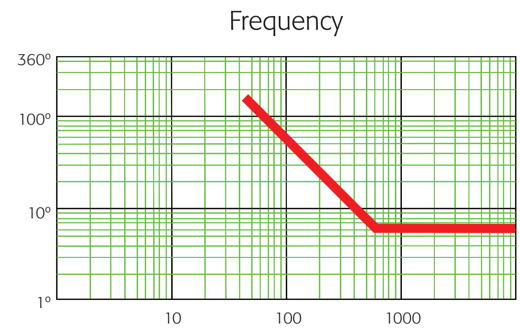
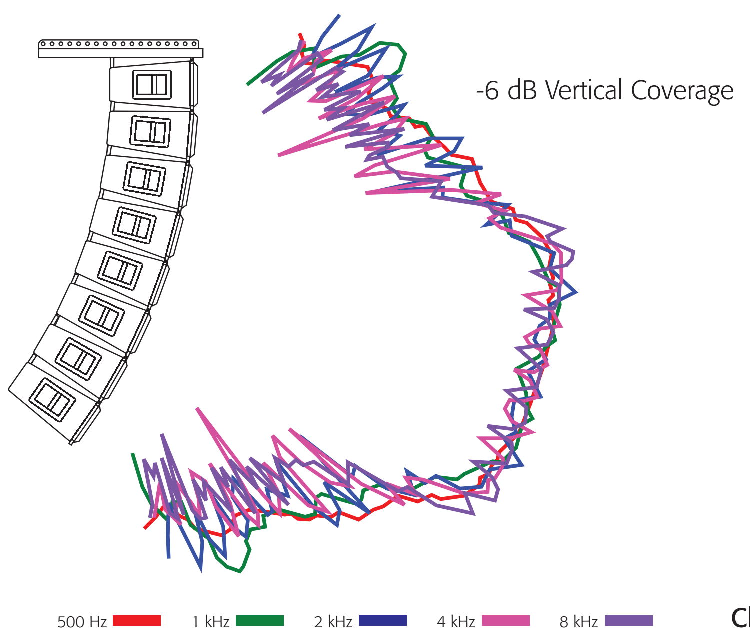


Figure 3

Characteristics of Array Shapes - Curved (Arcuate)



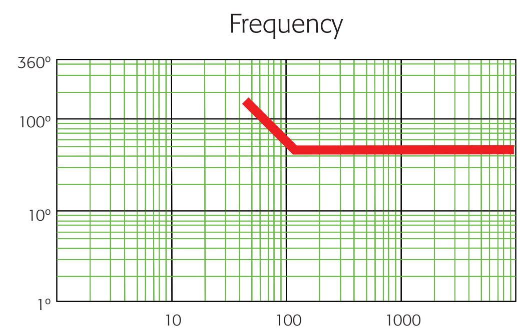


Figure 4

Characteristics of Array Shapes - Curved (Arcuate)

J-Arrays

When articulated line arrays first started appearing on the scene in the early to middle 1990s, the J-array form was widely used. Figure 5 illustrates the polar response characteristics of a typical large J-array. J-arrays were an attempt to increase the vertical coverage while preserving high directivity for coverage at long distances.

As can be seen from the polar response charts, the vertical coverage is very inconsistent with frequency and largely incoherent. J-arrays require separate signal processing for the straight and the curved elements due to the discontinuity where the straight portion meets the curved portion, and are extremely difficult to control and align.

Summary of J-array characteristics

- Inconsistent directivity
- Coverage ≈ ∑ splay angles˚, Example: < 1˚+ (3 • 5˚) ≈ 16˚
- Irregular far field boundary
- Aberrant (far field) coherence

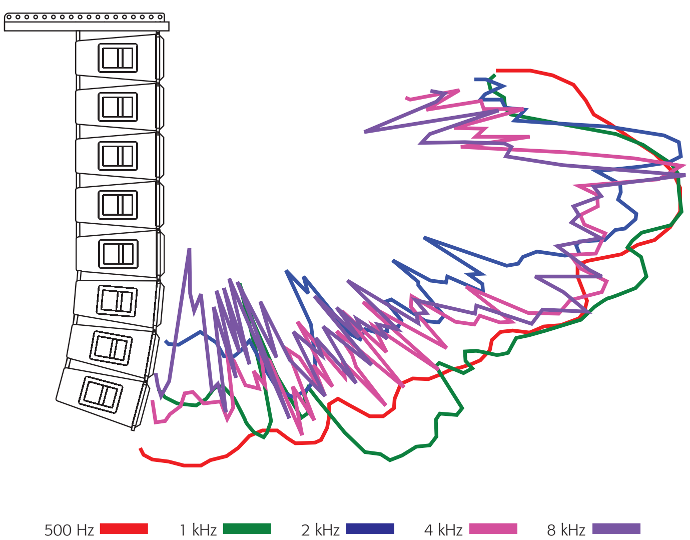


Figure 5
Characteristics of Array Shapes - J-Arrays

Spiral Arrays

A spiral array yields predictable constant coverage, but with a polar shape that is tilted. (Figure 6) This enables more power to be delivered to the more distant seats, while at the same time increasing the vertical coverage angle to include more down front seats. This has replaced the J-array for most concert applications.

Summary of spiral array characteristics

- Constant frequency directivity
- Coverage = ∑ splay angles˚, Example: 1+2+…+N˚ = 28˚
- Far field transitions smoothly
- Good (far field) coherence

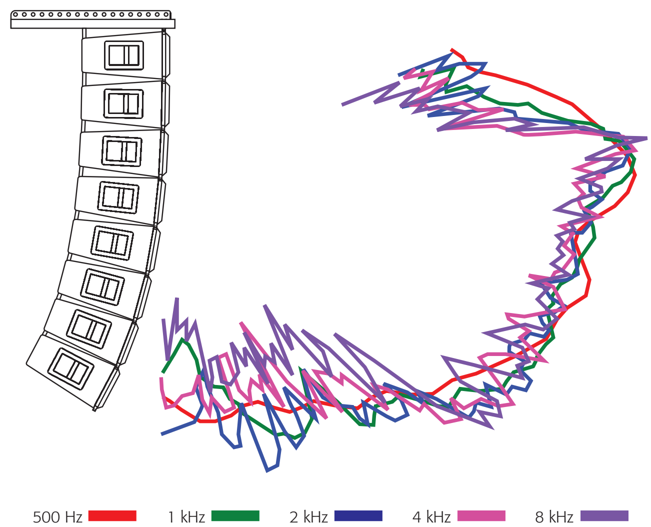


Figure 6
Characteristics of Array Shapes - Spiral Arrays

Shading

So far we have discussed changing the physical shape of the array by changing segment-to-segment splay angles to achieve the desired vertical coverage. But once an array is up, it usually is up for good – or at least for the duration of the show. Array coverage may also be altered in the electrical domain.

Amplitude shading is targeted level adjustment. One turns down the level to the area of the array that needs to be attenuated. But do this with a great deal of caution – the cardinal rule with line arrays is to avoid discontinuities. Because line arrays consist of individual segments – and the segments are really planar shapes – at every box intersection there will be a slight discontinuity by definition. Generally speaking he who shades least shades best. It is good practice to limit amplitude variations between adjacent boxes to 1 or 2 dB per. This can enable an arcuate array to effectively become a spiral array in terms of coverage. The maximum output capability from the upper segments of the array will be reduced compared to a spiral array, but the shape can be emulated with judicious amplitude shading. One can also similarly change or exaggerate the shape of a spiral array using amplitude shading. Successful shading also requires sufficient granularity in the amplifier and signal processing chain.

Delay shading enables reshaping of the virtual array. This is most often used with column type loudspeakers to tilt the coverage downward or upward, so that the array can be integrated with the architecture. But there are some limitations to this practice. One can successfully employ delay shading to tilt the coverage, so long as the tilt remains within the angular coverage of the elements. If the vertical beamwidth of the array element is 10 degrees, there is no hope of tilting the coverage beyond 5 degrees (up or down).

Multi EQ Systems employ split processing, which is another way of saying amplitude shading on a frequency dependant basis.

All of these methods have their place, but in general if you're doing concert sound, you want to get in, get the job done and get out. You can't spend hours or days adjusting the sound system – it should perform right out of the box. Concert sound providers want sound systems that 'plug-and-play'.

For permanent installation, application of these tools may be useful, but the delicate nature of these settings means that they can be easily "broken" should the in-house technical staff or guest operators decide that the system needs "improvement". If you can't put the signal processing gear under lock and key, you're likely to have a different sound system within a few short weeks. Complexity is the enemy of long-term reliability and consistency.

Preconditioning Filter Development

With the preceding line-array fundamentals under our belts, let us turn our attention to the development of preconditioning filters. There are multiple performance shortcomings common to all dynamic sound reinforcement loudspeakers that may be addressed with preconditioning filters.

- Virtually every line array system available uses diffraction as a horizontal pattern control device. Typical high frequency waveguides force expanding wave fronts to abruptly transition from a very tightly controlled, slowly expanding throat into a wide horizontal coverage angle, resulting in a severe acoustic impedance discontinuity at the diffraction slot. A percentage of the energy is reflected at the diffraction slot and returns toward the driver, where it is partially absorbed and partially re-radiated. This produces irregularities in the impedance response, and combing of the frequency response. Usually this is obfuscated by averaging or other methods, but these response peaks and dips are a source of coloration and time smear that require correction.
- All high frequency compression drivers have a mass break roll-off characteristic that stems from using a phase plug to increase the loading. Generally this is a single pole roll-off function beginning between 2.5 and 4 kHz.
- HF diaphragm surround and other resonances are also present and may affect the radiated response of the driver.
- There may be notch filtering caused by an open Helmholtz resonator – either a port tube or a diffraction-type HF horn. Regardless of the causal factors, if these characteristics display minimum phase behavior they can be corrected with minimum-phase filters.
- Another factor is the baffle-step. In small line arrays, such as QSC WideLine™, this occurs horizontally in the region of 300 Hz where the response drops off because the baffle is no longer a boundary.
- There can also be many other correctable device characteristics.

Characterizing Line Array Elements

In order to correct undesirable characteristics, it is first necessary to properly measure the array elements. But line arrays present some unique measurement challenges. There is an inherent limitation to the size of an array that can be measured in the laboratory. The distance to the far-field in a straight array is proportional to the product of frequency and the square of the array length, so with longer arrays, the distance to the far-field can easily exceed the largest dimension of the largest measurement space.

Going into the field is an alternative method of measuring arrays. In the field one encounters unpredictable environmental interferences at the distances needed for far-field measurements. Refraction caused by wind-borne (or other) temperature gradients can be disruptive, and sometimes these have a way of appearing under seemingly zero wind conditions.

There are boundary influences at the low and middle frequencies with which to deal. Free-field conditions are often interrupted because of proximate objects. In theaters or concert venues it can be difficult to obtain good microphone locations free of delayed reflections.

Atmospheric conditions are seldom known or considered. Booking a venue for the purpose of measurement could involve weather, forest fires or any number of disruptions that weren't anticipated. There will always be conflicting venue needs. If one intends to do sound system alignment, the management could decide to paint the proscenium on the same day (this happens).

It is not practical to attempt spatially-averaged measurements in a venue. We typically take about sixty measurements to develop a spatial average from a line array segment. This requires good free-field conditions and the ability to move either the device being measured or the microphone expeditiously. Trying to accomplish this in the field is an exercise in frustration.

Finally, there is an inability to separate loudspeaker, array and environmental influences when making field measurements.

Loudspeaker, Array and Environment

One might assume that causal factors don't matter – that one needs only to correct the observed deficiency with filters without giving thought to what caused the deficiency – but that simply isn't the case. Here's how these effects accumulate in the field...

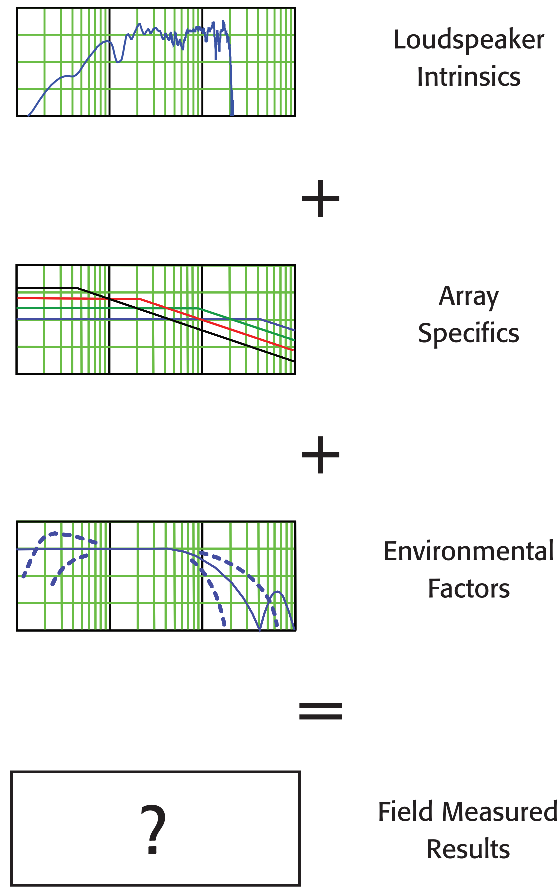


Figure 7
Measuring Line Arrays In-Situ

- Loudspeaker intrinsics are characteristics of the loudspeaker that are inherent in the transducers, enclosure and/or systems of waveguides and transducers.
- Array specifics are the predictable results of combining array elements in a given configuration.
- Finally there are the environmental factors – absorption of high frequency energy at the molecular level, the boundary conditions at low frequencies and customer preferences.

When engineers erect a measuring microphone in the field, all of these combine to yield results that cannot be separated in terms of cause and effect. It's important to understand what causes observed characteristics in order to apply proper corrective measures. Some things are not correctable, as we shall see.

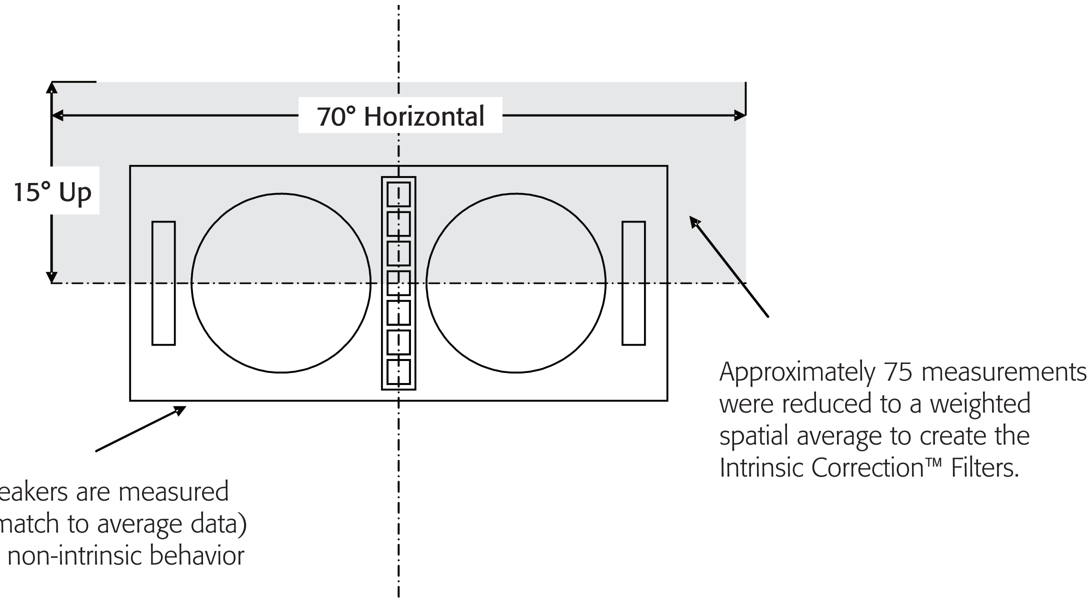


Figure 8

Correcting loudspeaker intrinsics is a laboratory process involving spatial averaging to characterize the loudspeaker response, followed by the appropriate application of preconditioning filters. In order to properly characterize a line array segment in the laboratory, extensive measurement is necessary. We typically make 60 - 75 free-field measurements that are then reduced to a spatial average.

Once the spatially averaged response of the array segment is determined, filters are applied to adjust this response to a maximally flat bandpass target. Successful application of the filters involves a number of requirements:

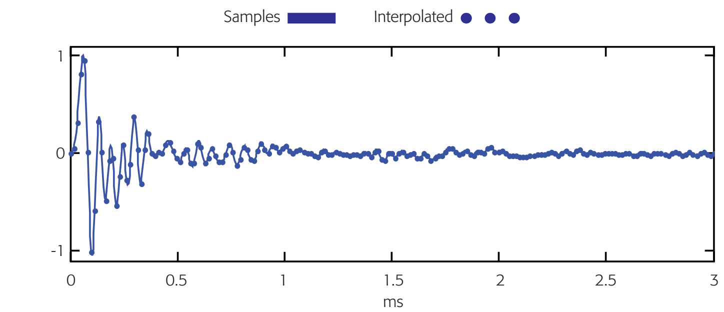
- The filters should impose minimum latency on the overall transfer function of the system.
- There should be zero relative phase difference between corrected transducers with overlapping frequency ranges as well as between corrected loudspeakers that are intended to sum acoustically.
- The minimal boundary condition must also be considered. In free space one 10-inch high box is going to lose all vertical directivity at about 1 kHz. That's too high in frequency for a baffle step, falling directly in the fundamental vocal region, which would be undesirable. Accordingly, we set the minimum boundary condition at four boxes to provide reasonable starting point.

When correcting loudspeaker intrinsics two types of digital filters are used. IIR (Infinite Impulse Response) type filters are used at low and middle frequencies and for bandpass definition. FIR (Finite Impulse Response) filters are employed at high frequencies in order to achieve highly-detailed correction with minimum latency.

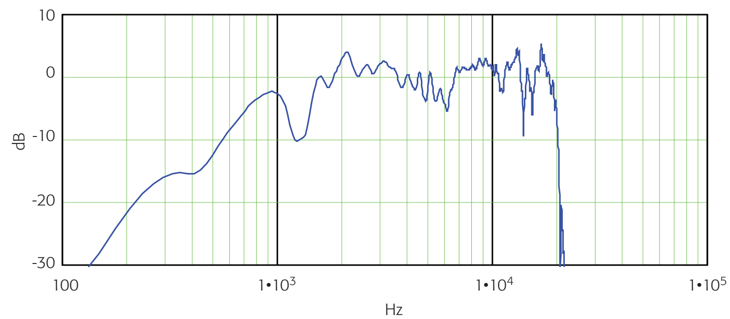
Table 1 shows a comparison of the relative advantages of both filter types based on a number of attributes. FIRs are quite advantageous for high frequency use and their minimum phase attributes are highly desirable. But user adjustment of FIR devices is not practical.

	IIR	FIR
Spectrum of highest efficiency	Low, mid	High
Design methodology	Analog prototype	Arbitrary magnitude response
Penalty for real-time adjustability	Low	High
Phase	Usually minimum	Arbitrary
Implementation complexity	Moderate	Low
Design complexity	High	Moderate

Table 1
IIR vs. FIR Filter Comparison

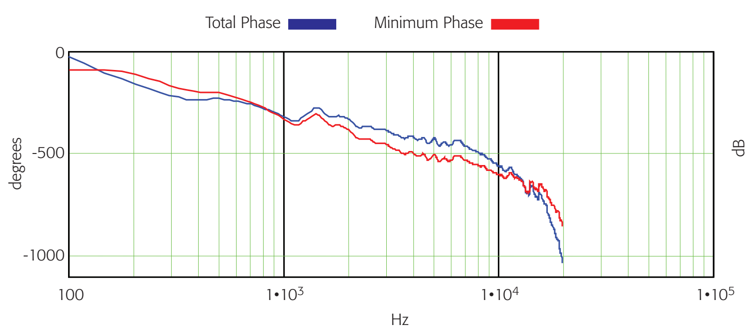


Impulse Response of HF Driver and Waveguide

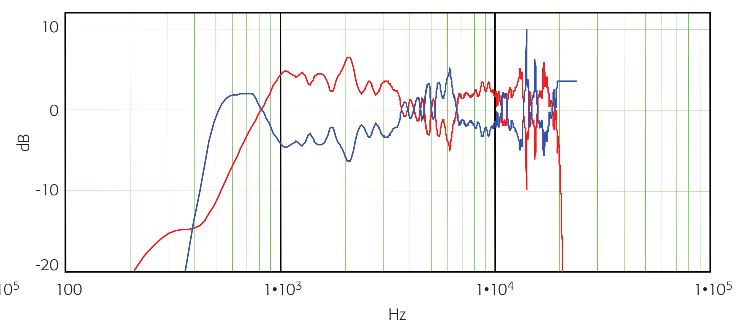


Frequency Response HF Driver and Waveguide

Figure 9

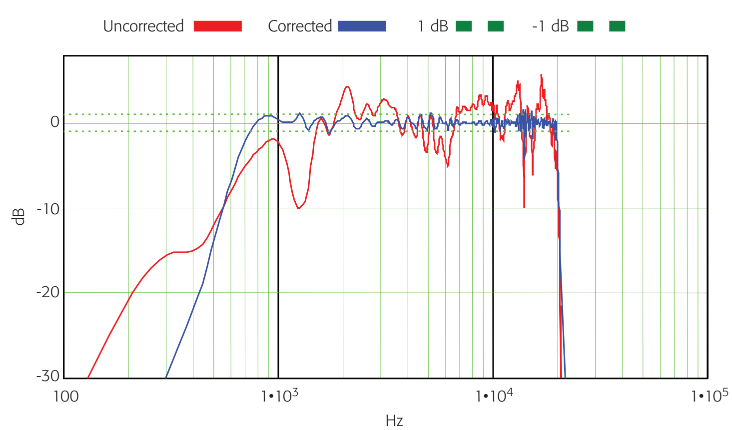


Phase Response of HF Driver and Waveguide



Raw HF and Correction Responses

Figure 10



Corrected and Uncorrected Frequency Responses

Figure 11

Figure 9 shows an impulse response which was derived from the measured data. The frequency response of the high frequency device itself is derived from the impulse.

Figure 10 Illustrates the measured phase response of the device compared to the minimum phase response for the transfer characteristic shown in figure 9. This tells us that minimum phase filters will correct the transfer function.

Figure 11 portrays the corrected and non-corrected responses of the array segment.

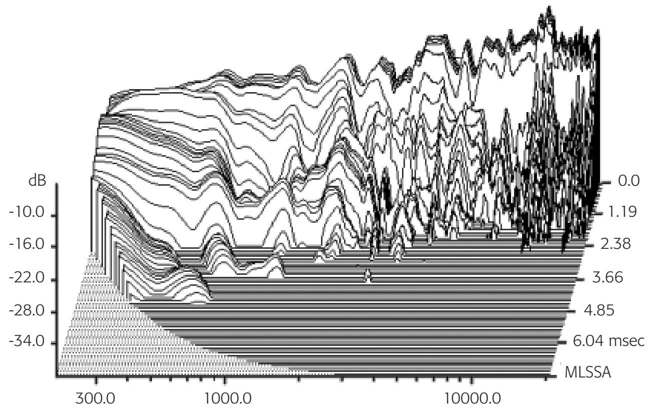


Figure 12

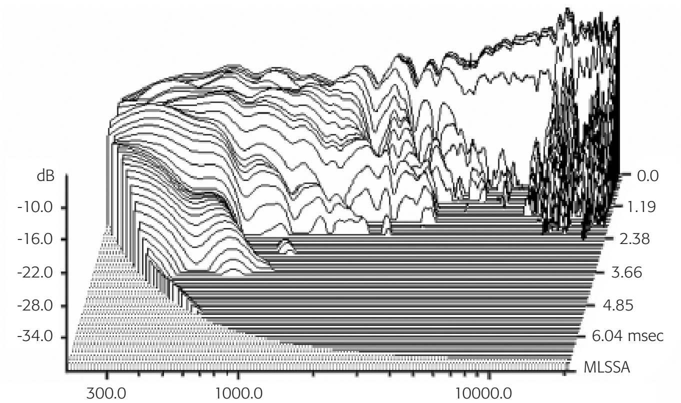
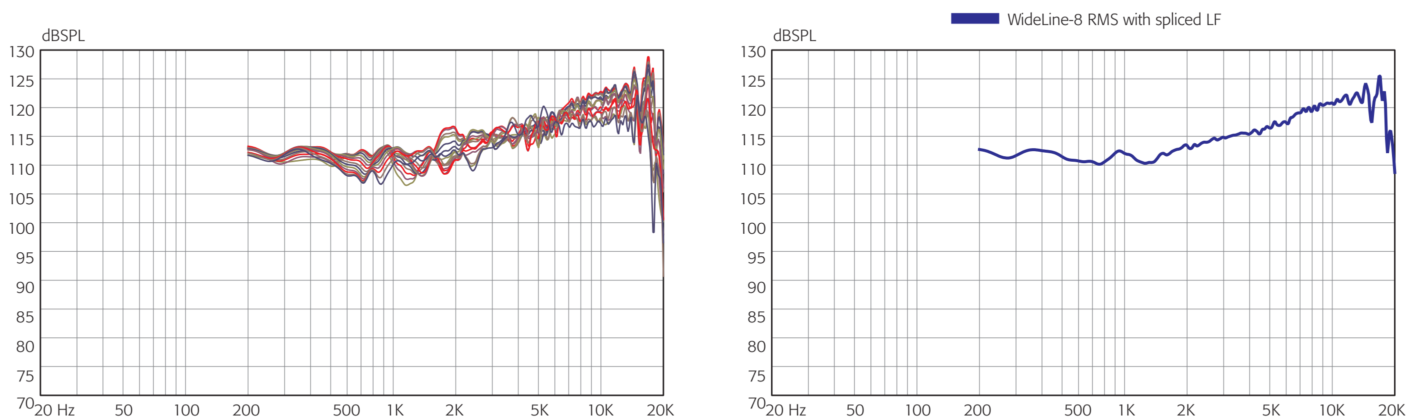


Figure 13



0° to +35° Horizontal

Spatial Average

Figure 14

(Figure 12) The waterfall graph plots amplitude in the vertical scale, frequency along the horizontal, and time along the diagonal. A theoretically perfect device would be one line at the back of this chart. The plot describes a 40 dB window and shows that it takes this loudspeaker x number of milliseconds at a certain frequency to decay 40 dB.

This waterfall portrays a high frequency unit that has been equalized using all-IIR type filtering. The region above 10 kHz shows fairly high density reverberant effects. At very high frequencies typical compression driver diaphragms become quite spurious. This behavior can't realistically be corrected with any kind of pre-conditioning filter, so we just leave it alone. For the record, above 10 kHz, 2 milliseconds to decay 40 dB is exemplary performance. The state of the art is someplace between 10 and 12 milliseconds.

Figure 13 shows the effects of the QSC Intrinsic Correction™ process. Nothing has been attempted at the very high frequencies for the reasons previously discussed. But between 2 kHz and roughly 10 kHz, nearly all the mid-frequency reverberation and time smear have been eliminated. These improvements are dramatic and clearly audible.

Figure 14 shows about one-fifth of the total field of measurements used to create a spatially-averaged response. The derived spatial average response is shown on the right. A climbing response is desirable because each of these segments has a collapsing vertical beamwidth – in order to radiate flat power versus frequency, the frequency response must climb at the same rate as the directivity response narrows.

Figure 15 shows the electrical transfer characteristics of the filters used for the QSC ILA (Installation Line Array). Here, IIR-type filters are used for lows and mids, while FIR filters are used for high frequencies. Notice that there are numerous peaks and dips in the high frequency filter. This corrects for the anomalies created by the diffraction waveguide, the compression driver, etc. These are not characteristics that are easily ascertained or corrected in the field. But, using spatial averaging techniques in the laboratory as described earlier, these characteristics can be measured and accurately corrected.

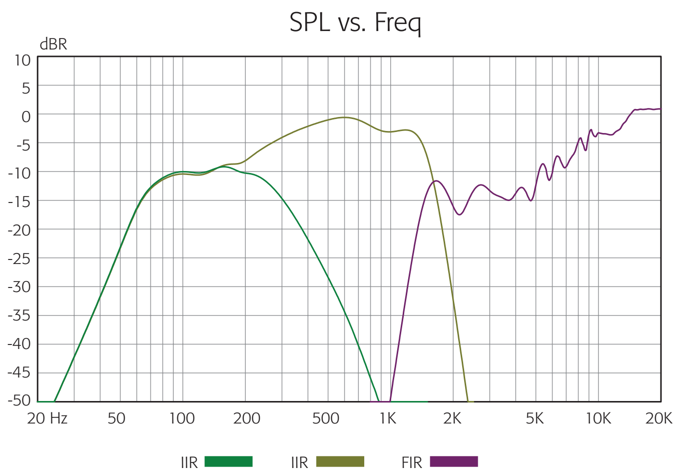


Figure 15
ILA Series WL2082-i Intrinsic Correction™ SC28 Transfer Function

Correction of Array-Specific Response

With the appropriate corrections applied to the individual array segments, we can now turn our attention to adjustments that optimize arrays.

- Straight arrays sum coherently on axis in the far field. But as we move off axis, fewer segments contribute, and eventually the farthest segments will interfere with one another. This increases with frequency and observation angle.
- Curved arrays, either an arcuate or spiral types, have diverging radiation axes. As frequency increases, the effects of divergence become increasingly evident.
- Arcuate and spiral shapes maintain a constant coverage angle with frequency, but there will be a reduction in output at higher frequencies. These effects are governed by the length of the array and the total included angle.

Figure 16 depicts a set of nested octave-band polar measurements for an 8-element, 40°, arcuate array. Note that there's a 7 dB difference at 2 kHz (the blue trace) between observation axes A and B. Because the axes are diverging, along certain axes the summation will be coherent, while along other axes the summation will be complex.

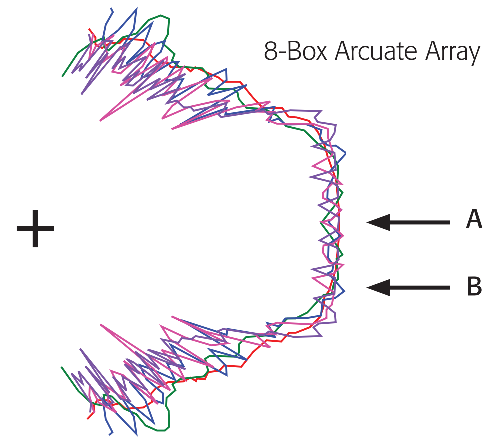


Figure 16
Correction of Array-Specific Response

Figure 17 shows frequency responses along various radiation axes within the nominal coverage of the 40-degree array. Along the 0° axis (green), there is about 7 dB of difference compared to the 15° axis (purple).

If a sound engineer places a measuring microphone right on that 15° axis, what would one suppose he is going to do? He's going to fix it – that's human nature. If he does that, people seated along the 0° axis will experience a broad and unpleasant peak centered at 500 Hz. Remember, all an equalizer can achieve is to modify the power spectrum of the drive signal. Averaging the relative SPL from all of the axes yields the red line, a very nice predictable 3 dB per octave (10 dB per decade) roll off.

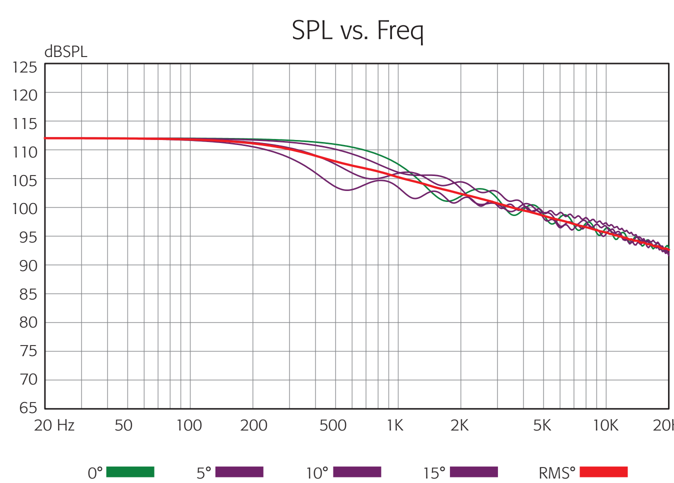


Figure 17
8 Boxes - 40°

In **Figure 18** the frequency responses of eight boxes with 1° splay between each are depicted. The break frequency has changed, but the actual slope characteristic remains the same, 3 dB per octave (or 10 dB per decade). This consistent behavior means that a simple algorithm may be created that enables us to employ the electrical inverse of the slope and apply a correction that corresponds to the array configuration.

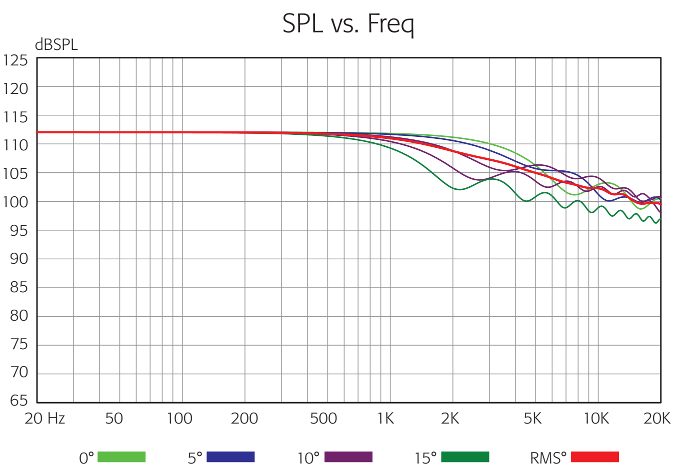


Figure 18
8 Boxes - 8°

In **Figure 19** a comparison of the RMS average of the two 8-box arrays is shown. The blue trace depicts the response of an 8-box, 8° array while the red trace shows the response of an 8-box, 40° system.

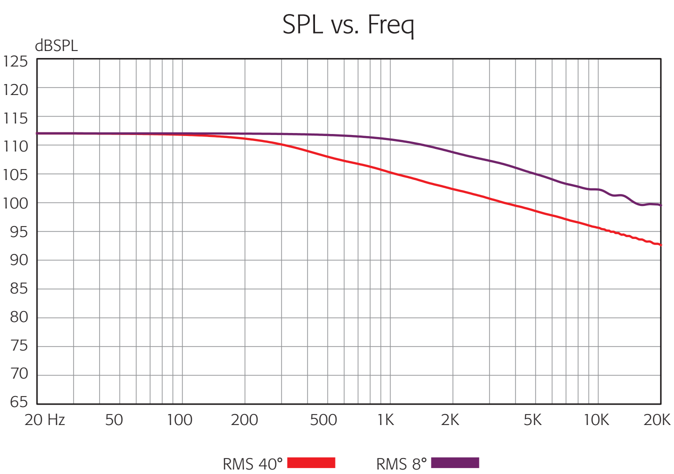


Figure 19
8 Boxes - 8° total coverage vs. 40° total coverage
RMS sum of axial radiation for each

Figure 20 shows at the average frequency responses of 2, 4, 8 and 16 box arrays, with each array describing a 10° coverage angle. Notice that at the lower frequencies – where we have coherent summation – a 6 dB increase per doubling of the array size takes place. At the upper frequencies, however, the breakpoint changes but the slope rate remains the same.

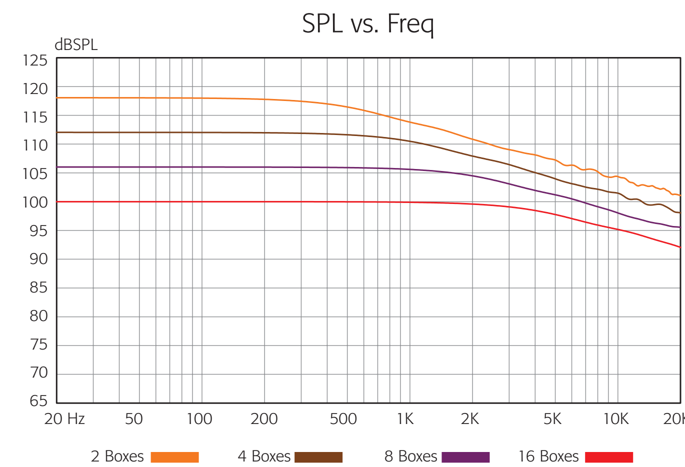


Figure 20
2, 4, 8 and 16 Boxes - 10° total coverage each array
RMS sum of axial radiation for each

This chart **Figure 21** compares two 8-box arrays with 28° of splay. One is spiral. The other is arcuate. In both instances the overall average system response is essentially identical meaning that the average correction will be the same for both. But the spiral array changes in frequency response over its length, due to the ever-increasing divergence from top to bottom, while the arcuate array will be constant.

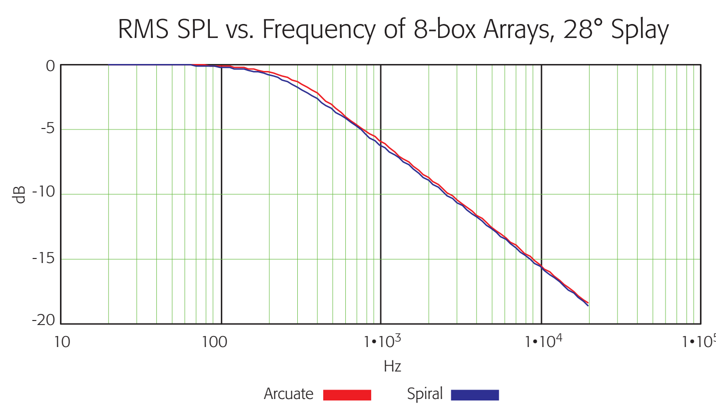


Figure 21
Correction of Array-Specific Response

Figure 22 illustrates the uncorrected frequency responses along observation axes defined by the cabinet intersections of an 8-segment, 28-degree spiral array. The heavy black line shows the RMS average of the total radiation, which tracks closely with the responses nearest the center of the array.

Thus, we see that a spiral array has an average frequency response of X at the vertical mid-coverage point, with progressively elevated high frequency response above that axis and progressively depressed high frequency response below. Arcuate arrays that employ progressive level shading are similar to spiral arrays in this sense.

There are many important decisions to make in deploying line arrays. Perhaps the most important one is where to locate the array and what type of array to specify. For uniform front to back coverage of the audience there will need to be more curvature in the array, and it's going to have to hang higher. Conversely, if one brings the array down, it needs to be straighter. It will be louder in the front than it is in the back, but it also will be more coherent throughout because the sources are less disparate.

With the preceding knowledge at our disposal, it is possible to create a DSP device capable of consistently, quickly and easily providing the proper transfer functions for a known line array system. The QSC SC28 incorporates these concepts into lookup tables that enable one to simply specify the loudspeaker type, how many cabinets are in the spiral or arcuate array, and the total splay angle of the array.

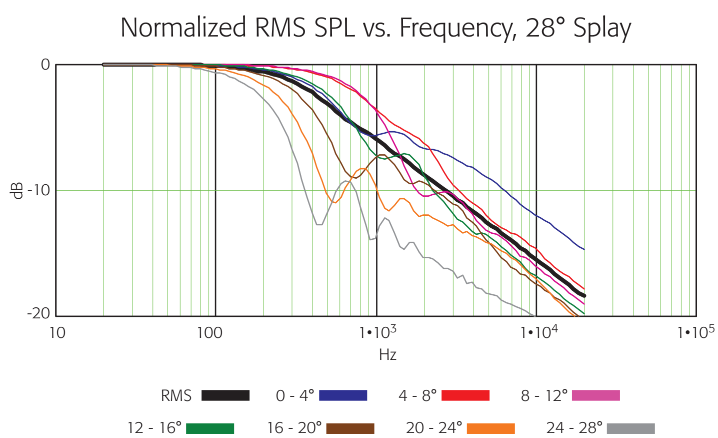


Figure 22
Correction of Array-Specific Response

This makes for corrections that are easy to implement and much more robust and consistent from place to place, while being less subject to upset by unauthorized tampering.

Figure 23 shows SC28 screen captures. The speaker configuration selected is WideLine-8, the array size is specified as four enclosures and the total splay is 15°. It's that easy, you don't have to do anything more.

Summary

In summary, we've corrected the inherent device characteristics by themselves and in isolation. We've identified what the array characteristics are going to be and corrected for that.

That leaves environmental correction for the user, if it's necessary. In many cases, especially if you're indoors, it may not be necessary. One might need to trim the low frequencies a bit, and if there is a very long propagation distance, make correction for high frequency absorption.

Intrinsic Correction has made line array systems faster and easier to deploy and consistent results easier to obtain. As a result, more of the sound engineer's valuable, on-location time, talent and effort can be turned to crafting a mix that enhances the performance.