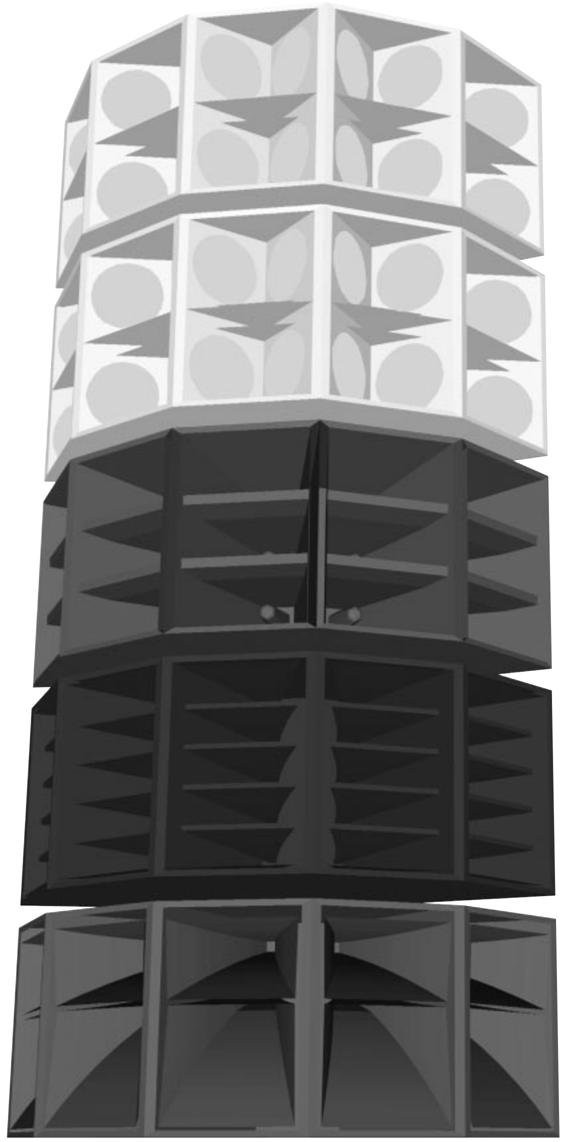

Phased PointSource Technology and the Resultant KF900 Series

EASTERN ACOUSTIC WORKS

The Laws of Physics/*The Art of Listening*

Phased PointSource™
TECHNOLOGY

EAW
EASTERN ACOUSTIC WORKS

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Preface

When this project was in the theoretical state, I had my doubts that signal processing could accomplish the things my simplified models said it could. But it was too promising not too pursue. A few experiments verified that the predictions were very good, indeed.

Once we had constructed a full system, I had my doubts that the PPST technique would be able to solve the directional problem through the whole vertical range. But it worked – even better than I had hoped.

Once we had a complete system with impressive technical specs, I had my doubts that this new approach would sound beautiful. But it does. It is more aesthetically pleasing than I ever would have hoped.

We can document the technical performance through measurements and published data, but to fully appreciate this system, you have to hear it. Its sound is just as unique as the technology it employs.

David Gunness
Senior Design Engineer

I KF900 Series And PPST: An Introduction

The requirements of large scale sound reinforcement present the sound system designer with a conundrum of the first order. On one hand, covering the entire audience area with the necessary sound pressure levels (SPL's) requires many drivers and/or multiway loudspeaker systems. On the other hand, creating a coherent wavefront providing intelligibility and musicality dictates that the number of transducers be kept as low as possible.

Experiments in the 1970's such as the Grateful Dead's "Wall of Sound" yielded poor results but pointed the way toward the array-optimized loudspeakers that have held sway from the 1980's to the present. Yet even the best integrated arrays project only a few hundred feet and are plagued by high frequency attenuation and dubious intelligibility.

The largest applications go beyond the scope of conventional single-source arrays. For football and soccer stadiums, often 600 ft (181.8m) or more end-to-end, sound system designers have been forced to use a more complex and expensive distributed approach. Single source arrays historically produced inadequate coverage at the far end of the stadium and/or excessive SPL's under the array. Similarly, music events held in stadiums required the use of delay systems to supplement the primary arrays, another costly necessity. (Apparently, these events can never be too loud.)

THE KF900 SERIES IS ENGINEERED FOR USE IN THE LARGEST VENUES.

A loudspeaker technology producing a coherent impulse that sounds both intelligible and musical at distances up to and beyond 600 ft (181.8m) has become a sort of Holy Grail for high-end professional loudspeaker manufacturers. Unfortunately, they seem to have enjoyed about as much success in the quest as the Knights of the Round Table did in theirs.

Despite the considerable difficulty of the task, Eastern Acoustic Works (EAW) set out to create a full range loudspeaker array that could provide highly intelligible yet musical sound, with the necessary SPL's and minimal high frequency attenuation at distances in excess of 600 ft (181.8m) - and closer than 50 ft (15.2m).

Phased PointSource™
TECHNOLOGY

THE PPST LOGO ILLUSTRATES THE BASIC CONCEPT: INTEGRATING SUBSYSTEMS THAT COVER VERTICALLY DISSIMILAR AREAS

The Entire Equation

By approaching the total problem, the result has been a total solution. Rather than simply creating a specialized tool to achieve extreme long throw with existing arrays, our approach instead solves the true problem: incoherent sound wave summation from multiple drivers.

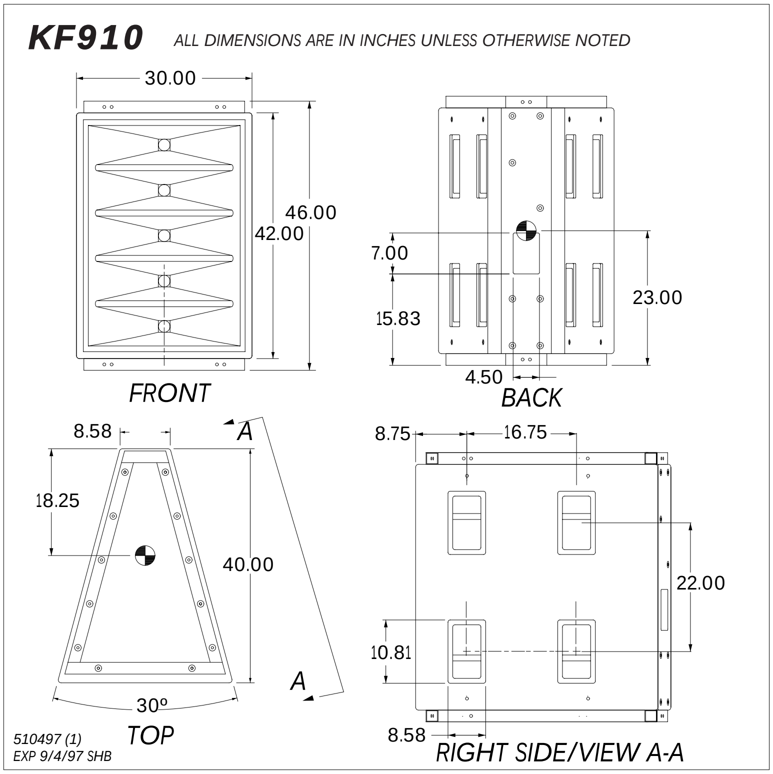
To solve the problem we first optimized all well-documented electro-acoustic elements that would apply to the loudspeaker array. Transducers would need to be very efficient and capable of very high output and waveguides would need large mouths and long throats to achieve the highly controlled pattern required.

Next came vigorous pursuit and refinement of new or overlooked concepts such as high frequency pre-emphasis to combat air loss, and the benefits of vertical versus horizontal audience segmentation.

Finally, the issue of coherent summation was addressed via scientific methodology and a commitment to genuine, useful results.

Enter PPST

Phased Point Source Technology (PPST) and the resultant KF900 Series of loudspeakers leverage the ever-increasing power of digital processing to create a unified source sound impulse at all points within the coverage area. KF900 loudspeaker modules use newly engineered mid- and high-frequency drivers packed into the smallest possible space and loaded on new SimplePhase™ horns, optimized to accommodate the PPST process. Building on EAW's previous use of phase and frequency "shading" techniques to manipulate beam profiles and to blend vertically dissimilar subsystems, PPST integrates the KF900 loudspeaker modules into a single acoustical element the beam profile of which can be adjusted and even steered in the vertical plane.

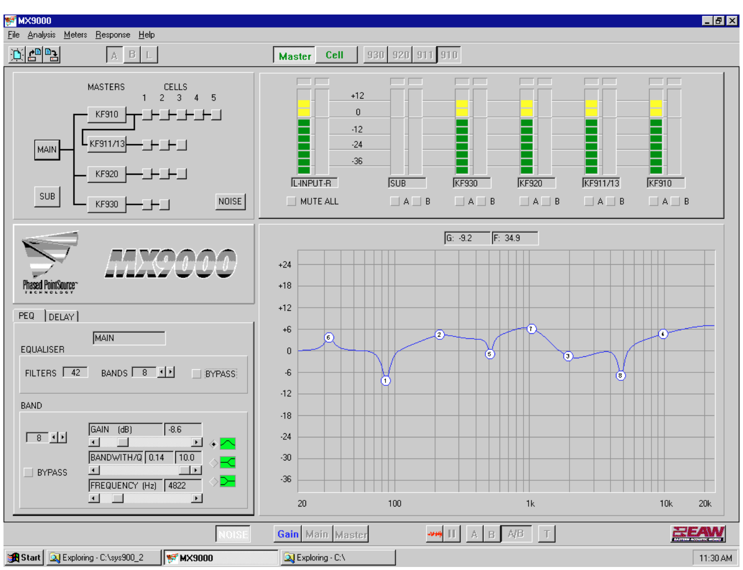


ALL KF900 SERIES ENCLOSURES SHARE IDENTICAL EXTERIOR DIMENSIONS.

A KF900 array is comprised of a variety of modules, each optimized for a particular need. Modules include:

- KF910 long throw HF module
- KF911 downfill HF module
- KF913 medium throw/downfill HF module
- KF920 long throw/downfill MF module
- KF930 PPST LF module
- And, the KF900 "placeholder"

THE MX9000 FEATURES A USER FRIENDLY INTERFACE.



A modest KF900 array produces 151 dB SPL average (163 dB SPL peak) at 1m (equivalent) with flat on-axis frequency response (±3 dB to 8 kHz) far as 600 ft (181.8m). Attenuation is 15-20 dB at 60° below vertical, with acceptably flat frequency response. Most impressive: the acoustical quality is studio-like, with impulsive high frequencies, excellent intelligibility and a distinct absence of the mid-bass "growl" typical of large arrays.

Due Processing

While currently available DSP systems can accomplish much of the necessary signal processing, several of each are required to do so. In response to the lack of an adequate control system EAW has developed the MX9000, a dedicated digital processing system for the KF900 Series. This powerful digital package permits the array to deliver high fidelity sound with far greater precision, and it also enables the user to optimize every array configuration.

THE KF900 SERIES ARRAY INSTALLED AT OLYMPIC STADIUM, ATHENS, GREECE.

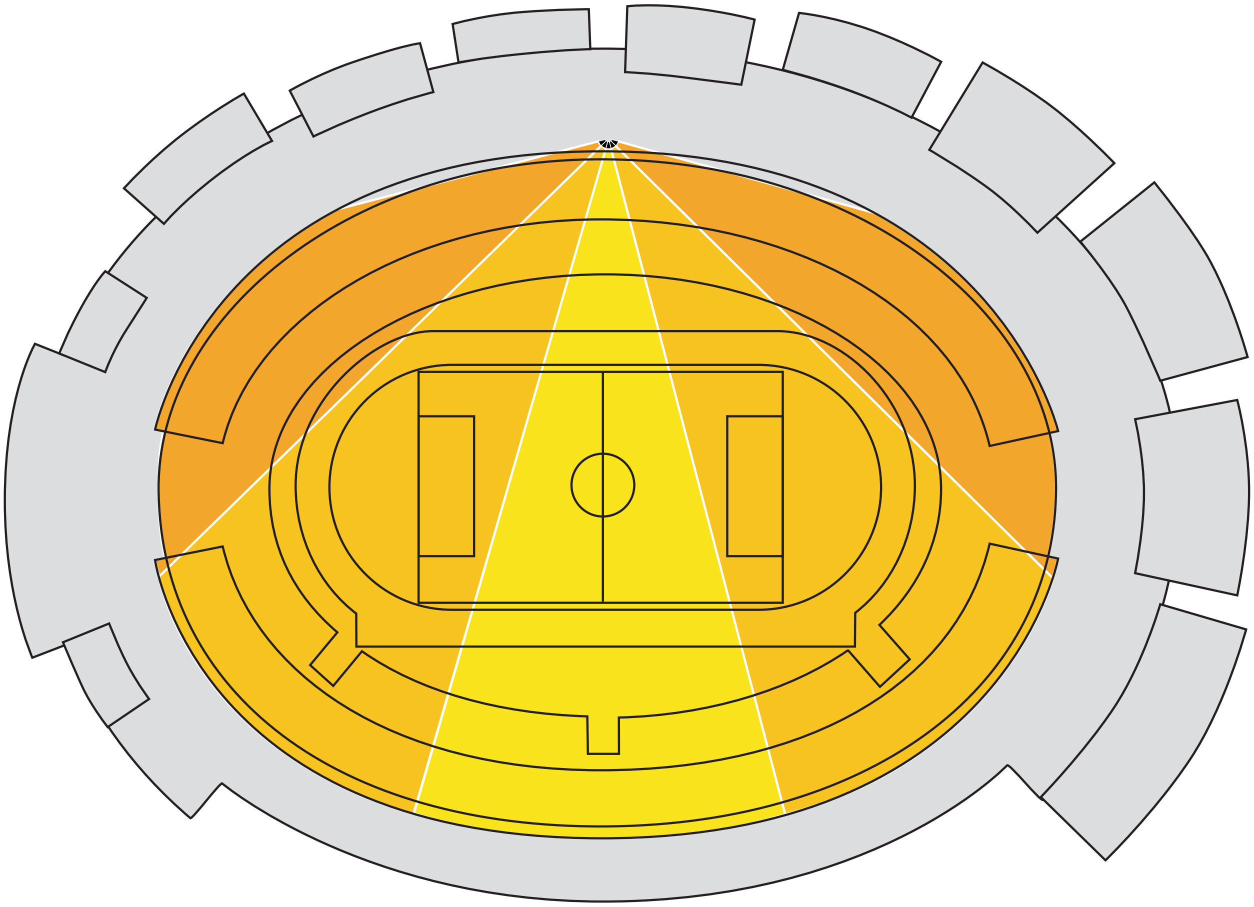
The MX9000 is an extraordinarily sophisticated custom hardware/ software package designed from the ground up. It includes a 32-bit floating point machine that co-processes four high-speed DSP's per channel with a 166 MHz Pentium processor - enough horsepower to provide 150 dB of internal dynamic range and over 200 million operations per second.

Combining this hardware capability with user-friendly, graphic oriented software, the MX9000 precisely steers and adjusts the KF900 loudspeaker system's directionality by internally making hundreds of adjustments at the single "click" of a computer mouse. As a result, KF900 arrays deliver much greater accuracy and precision than ever before possible from any array.

Raising The Bar

The KF900 has been refined for real world applications through an aggressive refinement process conducted over several months in 1997 on the Promise Keepers tour. Held in stadiums throughout the US, this tour provided EAW engineering with vital performance data that has resulted in a loudspeaker series proven in a variety of live applications for tens of thousands of people at each venue.

Further proof that PPST and the KF900 Series raise the performance bar for large-scale sound reinforcement has also recently been confirmed. A single KF900 Series array, installed by Alpha Sound at the Olympic Stadium in Athens, Greece prior to the IAAF World Track & Field Championships, provides coverage to the entire 85,000-plus seat venue. The 25-loudspeaker array has been measured to provide broadband uniformity, with variation of just ± 2 dB at every measurement point, even at more than 500 ft (151.2m) from the array. Only small areas completely masked by the upper deck require supplemental reinforcement from a distributed underbalcony system.



CONSISTENT HORIZONTAL DIRECTIVITY FORM 300 HZ TO 20 KHZ LETS US GIVE EACH COLUMN OF THIS 5 COLUMN ARRAY ITS OWN DISTINCT PROCESSING FINE-TUNED TO ITS COVERAGE AREA. MINIMAL INTERACTION BETWEEN COLUMNS MEANS THAT EACH HORIZONTAL COVERAGE ZONE ONLY RECEIVES SOUND FROM ITS ASSIGNED COLUMN.

II Application Requirements

KF900 SERIES ARRAYS USED FOR THE 1997 PROMISE KEEPERS TOUR.

The first requirement of large-scale sound reinforcement is generating high levels of clear, undistorted sound across the full frequency spectrum. Attaining the desired SPL's is easy enough; developing broad-band, undistorted, coherent sound at high SPL's has proven significantly more difficult.

The Inverse Square Law states that sound levels generated by a point source fall 6 dB per doubling of distance. Thus a sound system must produce 145 dB at 1m in order to have a chance of attaining 100 dB at 600 ft (181.8m).

Common practice using today's technology involves tight-packing full-range loudspeakers into arrays until the necessary SPL's are achieved. Typically, these loudspeakers employ horn-loading to maximize driver sensitivity and to help focus as much acoustic energy as possible on the audience area.

However, this approach introduces several problems:

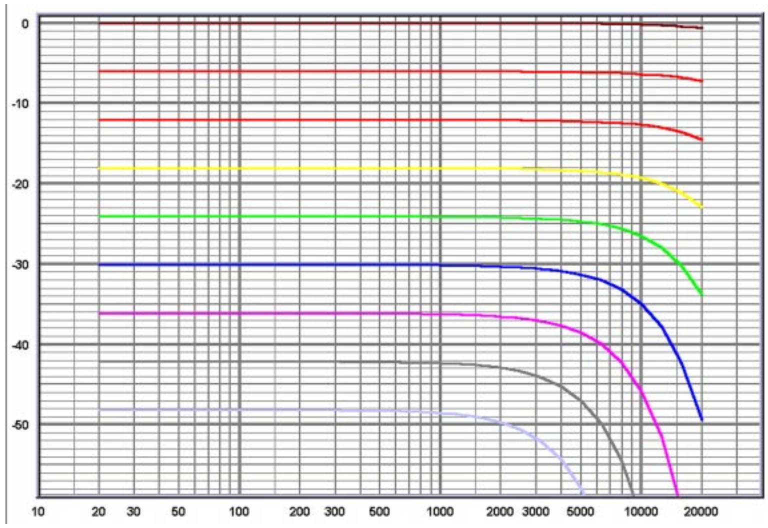
1) Driver spacing results in inefficient summation;
2) Multiple arrivals "smear" attacks and garble articulation;
3) Acoustical coupling creates powerful lobes at certain frequencies.

To solve these problems, all of which result from driver interaction, each element or subsystem of the array must either be isolated from the others or (somehow) integrated.

Impact Of Air Absorption

As distances increase, high frequencies fade due to a phenomenon known as air absorption that seriously impacts intelligibility and musicality. Briefly, air molecules absorb more acoustic energy from shorter wavelength high frequencies than they do from longer wavelength low frequencies. While air absorption has been clearly documented for quite some time, its full impact on sound reinforcement has largely been ignored by loudspeaker designers.

The following example provides a point of reference regarding the detriments of air absorption. For an array to supply high frequency level (4 kHz) at a distance of 400 ft (121.2m), an additional 6 dB is required relative to 500 Hz, which is not impacted by air absorption until 1000 ft (303m) or more. Therefore, the array must produce 151 dB at 4 kHz at 1m.



THIS CHART ILLUSTRATES THE EFFECTS OF THE INVERSE SQUARE LAW AS WELL AS AIR ABSORPTION. THE BROWN LINE AT 0 DB REPRESENTS SOUND AT 1M. EACH SUCCESSIVE LINE REPRESENTS SOUND AT A DOUBLING OF DISTANCE (2M, 4M, 8M, ETC.) NOTE THE SUBSTANTIAL 10 KHZ ATTENUATION AT 64M (VIOLET).

However, generating sufficient sound pressure levels can lead to another problem. Listeners close to the array, perhaps as close as 50 ft (15.2m), only 45° degrees below vertical, are likely to be assaulted by excessive SPL's, possibly dangerously high ones. Audience members could be subjected to 121 dB of 500 Hz energy, and 124 dB in the 4 kHz range where the ear is most sensitive to pain. (Ouch!)

Principles Of Higher "Q"

To be truly useful, the array would need very high "Q" characteristics to provide substantial off-axis attenuation. Of course, additional loudspeakers would then need to be employed to cover the areas under the array. That is, unless the array provided off-axis attenuation in such a way that frequency response remained acceptable at 45° below the array.

Finally, the array would need to provide truly consistent coverage throughout the audience area. For the sake of argument, an array is placed in a hypothetical stadium with a horizontal coverage pattern of 120 degrees and a maximum throw of 600 ft (181.8m) directly on-axis, and with the nearest listeners only 33 ft (10m) away, 60° below the array.

Frankly, conventional loudspeaker array technology will no longer suffice in this situation. Many systems will generate the

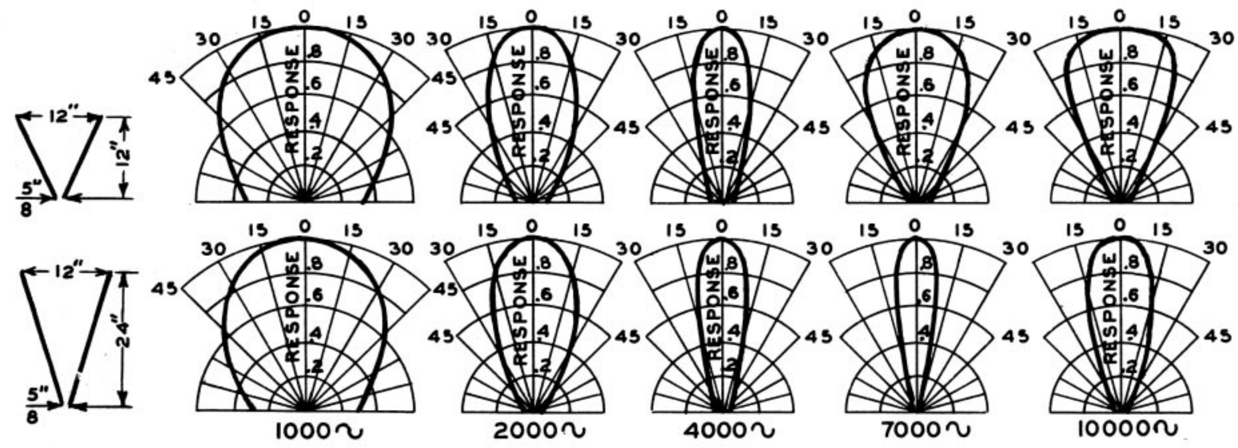


FIG. 2.18. The directional characteristics of two conical horns with mouth diameters of 12 inches and throat diameters of $\frac{5}{8}$ inch and lengths of 12 inches and 24 inches. The polar graph depicts the sound pressure, at a fixed distance, as a function of the angle. The sound pressure for the angle 0° is arbitrarily chosen as unity. The direction corresponding to 0° is the axis of the horn. The directional characteristics in three dimensions are surfaces of revolution about the horn axis.

MOUTH SIZE, THROAT LENGTH AND FLARE SHAPE DETERMINE THE FREQUENCY RANGE OVER WHICH A HORN DEMONSTRATES PATTERN CONTROL. TO ACHIEVE CONSTANT HORIZONTAL DIRECTIVITY, SIMPLEPHASE HORNS FEATURE A LONG THROAT AND A WIDE MOUTH FOR MAXIMUM PATTERN CONTROL. THEIR UNCOMPLICATED FLARES – MUCH LIKE THESE HORNS FROM OLSON'S ACOUSTICAL ENGINEERING – BEST SUPPORT THE PPST PROCESS.

appropriate SPL's. Some may supply the necessary "Q" to achieve both long throw and off-axis attenuation. But no system currently available provides acceptably flat frequency response at -60°; no full range system achieves acceptable HF response at distances over 400 ft (121.2m), let alone 600 ft (181.8m); and no loudspeaker technology produces a truly coherent wavefront from multiple sources.

With Phased PointSource Technology (PPST) and the KF900 Series, EAW has addressed all of the above criteria. The pursuit of these solutions began with an element we've become quite familiar with through the years: horn design.

The laws of physics clearly indicate that the size of a horn's throat and mouth determine it's coverage pattern at a given frequency. As a rule, the larger the horn, the better control it exhibits at lower frequencies. However, when multiple large-format horns are arrayed, the drivers are typically spaced so far apart that coherent summation does not occur. Interference between the sources creates comb filtering, phase alignment anomalies, and other detrimental effects.

The acoustical physics of source coupling and interaction are well-documented. The closer together sources are placed, the more they act like a single source (or a point source). The key is for the wavefronts of horizontally adjacent horns to sum smoothly at the mouth creating a single arc. The closer the drivers are placed, the more their horn's outputs will behave as if they were generated by a single source.

The goal, then, was to integrate these two mutually exclusive needs: tight-packing drivers so they could act like a point source and loading them with large horns that all fire in the same direction. The larger the horn, the farther apart the drivers; if the drivers were pulled close together, the horns would then be firing in different directions.

Early in the PPST development process, it was realized that both horn-based high "Q" performance **and** point-source-like coupling in both the horizontal and vertical planes would be impossible. A choice was made to design horns providing consistent broadband directivity in the horizontal plane. If horns could provide seamless horizontal coverage in tight packed arrays, research and development energies could then be focused on the much more difficult problem of integrating multiple drivers in the vertical plane to produce a coherent wavefront.

Each horn's vertical dimension became small relative to the horizontal so that drivers stacked vertically in a single enclosure sat quite close - almost touching. Virtually no vertical pattern control was demanded from a single horn "cell" as they came to be called. The theory was that source coupling and signal processing would provide both pattern control and coherent integration. Not only did this approach turn out to be possible, a variety of side benefits presented themselves, encouraging the pursuit of the idea.

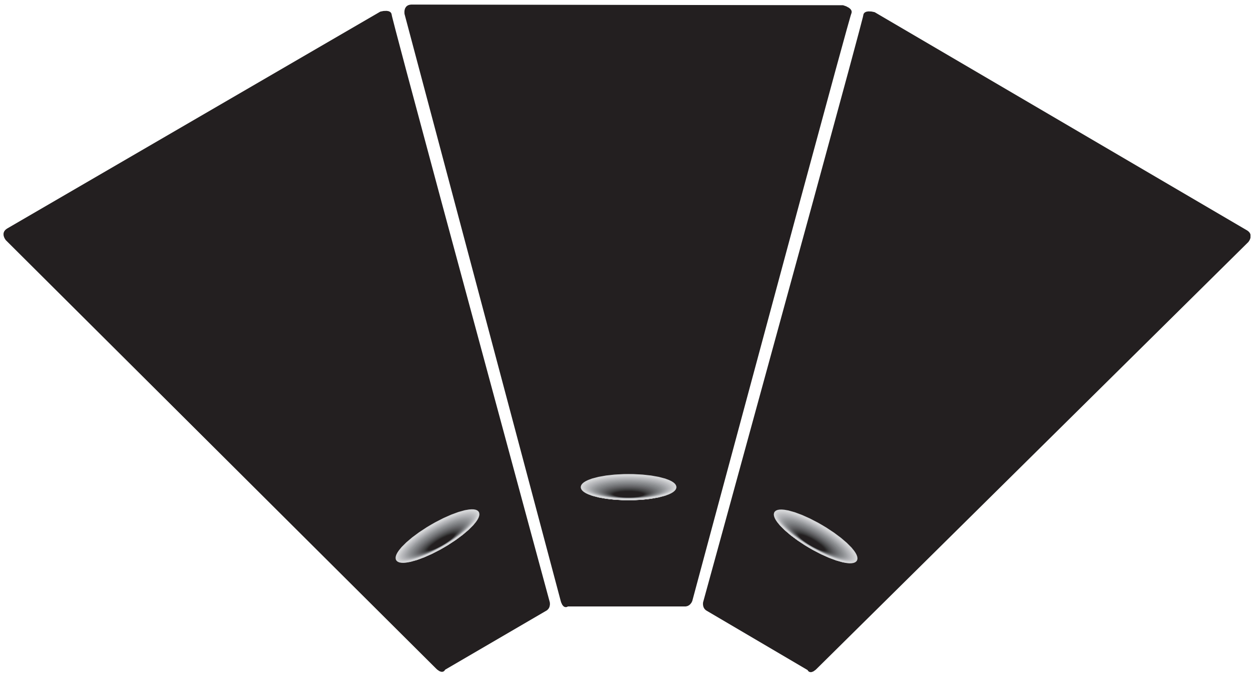
Put It In Reverse

To solve the horizontal coverage dilemma, a bit of reverse thinking was applied to the problem of "growing" the horn. Theoretical work had shown that for signal processing to integrate horns in the vertical plane, the horns would need exceptionally simple geometry to minimize phase effects and diffraction, enhancing coherent summation. This dictated a longer horn throat than usual.

Typically, when seeking to expand a horn, the approach is usually to "push" the mouth of the horn farther from the driver. Instead, EAW engineers "pulled." The mouth of the horn was left untouched, while the driver was "pulled" back.

The drivers were drawn back as far as physically possible, until only two thicknesses of enclosure material (wood) separated the drivers of two tight packed enclosures. Not only was the horn's throat extended to the necessary length to achieve the desired directivity, but the drivers of adjacent enclosures were now as close together as physically possible so that the wavefront arcs they produced were essentially continuous.

The result: true 30° horizontal pattern control and seamless horizontal transitions between enclosures in both the mid and high frequency ranges.



BY DRAWING THE DRIVERS BACK AS FAR AS POSSIBLE, SIMPLEPHASE HORNS COUPLE ACOUSTICALLY, ACTING AS IF THEY WERE DRIVEN BY A POINT SOURCE IN THE HORIZONTAL PLANE.

III Technology Precedents

KF860/KF861 VIRTUAL LINE ARRAY USED FOR THE HALFTIME SHOW AT SUPER BOWL XXXI IN NEW ORLEANS, 1997.

The next step was integrating the horn cells in the vertical plane. Rather than start from scratch, several existing EAW technologies were re-examined to see if they could provide some answers.

The KF860 Virtual Line Array (VLA) system, also designed for larger sound reinforcement applications, had shown the benefits of segmenting the audience area vertically rather than horizontally. VLA technology was developed in conjunction with a well-known audio consultant who was interested in pursuing the idea of vertical - not horizontal - audience segmentation.

The thinking was that an audience member in the front row has nothing in common sonically with an audience member toward the rear, even though they may be share the same horizontal “section” (i.e. one is seated directly behind the other separated by some distance). Remarkably, however, this is usually the manner in which sound systems have segmented audiences. Arrays are designed by determining the horizontal coverage angle that’s required, with loudspeakers then tight-packed into an arced array that is flown or stacked with enough cabinets to throw sound to the rear of the audience area.

VLA technology demonstrated that audience members positioned a given distance from the stage had virtually identical acoustic needs whether they sat to the left, right, or center. KF860/KF861 modules were designed to stack vertically such that each could cover a discreet vertical audience segment. By tailoring EQ for these vertical segments (near-, mid-, and far-field) the total audience receives a much higher quality of sound reinforcement.

Similarly, two recent additions to the KF850 family of Stadium Array concert touring loudspeakers pointed toward PPST. The KF853 mid/high and BH853 LF specialized long-throw (higher

"Q") loudspeakers as well as the KF855 downfill loudspeaker helped prove that building array columns with loudspeakers tailored to cover specific vertical segments improved overall sound quality for the entire audience.

With the addition of these specialized systems, a KF850 Stadium Array became a structure of columns several of which would be tight packed to achieve the necessary horizontal coverage pattern. A typical array column would feature a KF853 and BH853 on the top two rows to "shoot" sound to the back of the venue, two or three KF850's for the bulk of the house and a KF855 on the bottom to fill the front rows.

It was this idea of array columns made up of specialized loudspeakers that would play such an important role in PPST/KF900 development.

The KF853 also represents an early attempt to address air absorption and its development led to a break from traditional methods of developing loudspeakers around standard measurement criteria. The measurement position was moved back from the traditional 1m to 50 ft (15.2m), more accurately showing the impact of high frequency attenuation, even over relatively short distances.

The solution for the KF853D was what is now called "high frequency pre-emphasis." That is, adding additional high frequency level so that frequency response would be flat (± 3 dB to 17 kHz) at 50 ft - not at 1m. This idea that a loudspeaker should be designed to sound good at the listening position - not at an arbitrary nearfield position - played an important role in the success of the KF900 Series.

The Shading Of Frequencies

The KF855 downfill loudspeaker incorporated another proprietary concept that had previously been used in distributed loudspeakers: frequency shading. The concept was first employed by EAW in the AOS90, a loudspeaker custom engineered for the Olympic Stadium in Atlanta.

AOS90 LOUDSPEAKERS INSTALLED IN ATLANTA OLYMPIC STADIUM. THEY WERE INSTALLED IN THE PORTION OF THE STADIUM WHICH TODAY IS CALLED TURNER FIELD AND SERVES AS THE HOME TO MAJOR LEAGUE BASEBALL'S ATLANTA BRAVES.

The stadium's system design called for a two-way loudspeaker, powered by a single amplifier channel, to cover an area ranging from 15 feet to over 100 feet from the loudspeaker. The loudspeaker incorporated a long throw high-frequency horn and compression driver combination to reach the farthest seats. The horn had a very tight coverage pattern to "shoot" energy over the heads of audience members in the nearfield.

Because any horn is effective only down to a certain frequency, depending upon its size, some off-axis radiation just above crossover point reached the nearfield at "useable" levels. Rather than fight to contain this energy, it was instead left intact to provide coverage at those frequencies. A second wide dispersion high frequency horn was developed to cover the nearfield, emphasizing higher frequencies that didn't reach the nearfield.

The output of the second horn was carefully integrated so that the output of both horns together provided smooth frequency response in the nearfield. The technique was dubbed "frequency shading."

Adapted for the KF855, the technique proved successful within the scope of a high-output array. And, it has proven two key concepts that would play a role in PPST/KF900 development:

1) Complex signal processing could integrate subsystems that covered vertically dissimilar audience areas, and

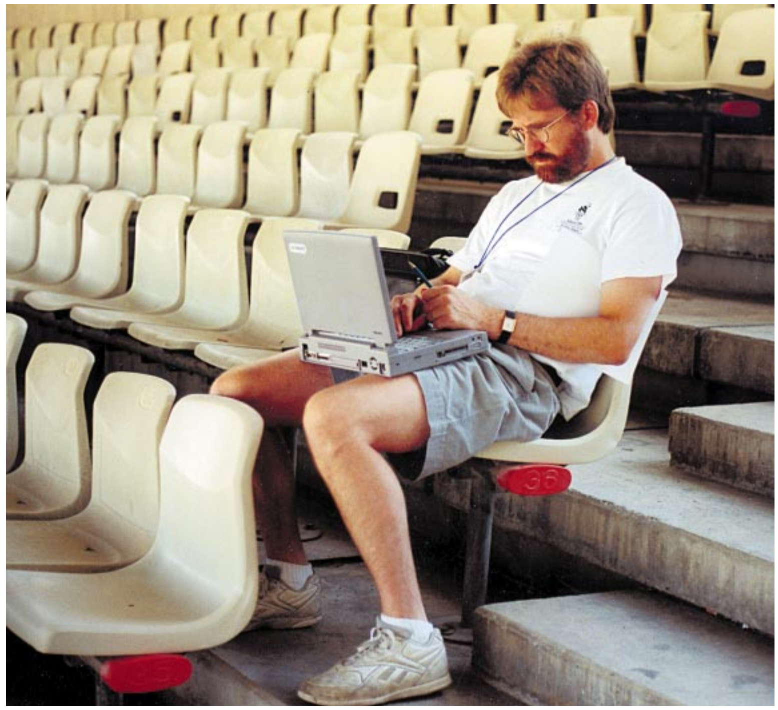
2) Off-axis radiation from a subsystem can provide usable off-axis coverage, provided it is integrated with other subsystems to "fill in" the holes in frequency response.

Putting It All Together

From the beginning, the KF900/PPST project sought to develop a total solution to the problem of large-scale sound reinforcement designed from the ground up. Rather than just develop a complex processing system to control an array made up of existing loudspeakers, completely new loudspeaker array modules were created.

The loudspeakers were conceived as modules that would be built up in array columns each of which covered a 30° horizontal audience section. These modules would need to be integrated in the vertical plane with extraordinary precision.

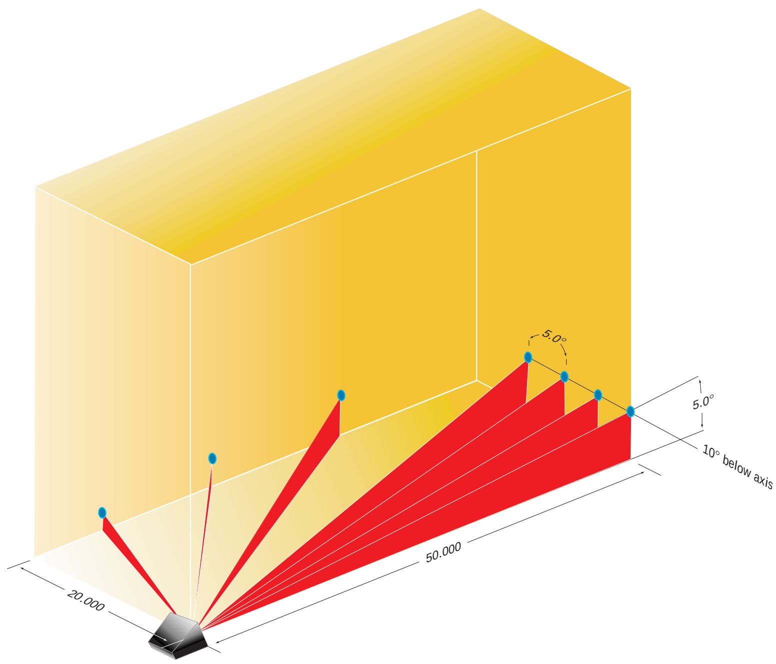
Whatever this method of integration would be, it would need to be based on very accurate component measurements. Traditional methods of measuring loudspeakers proved inadequate for the task, so a completely new method of measuring individual loudspeaker subsystems in multiple loudspeakers configuration was devised.



EAW SENIOR DESIGN ENGINEER DAVID GUNNESS WORKING ON THE OLYMPIC STADIUM INSTALLATION IN ATHENS, GREECE. FINE TUNING THE PROCESSING "IN SITU" BROUGHT CONFORMITY TO WITHIN ±2 DB ACROSS THE ENTIRE AUDIENCE AREA.

As a result, the EAW automated testing and measurement facility was modified to accommodate the new testing regimen. The measurement, optimization and signal processing regimen developed there - today called Phased PointSource Technology (PPST) - has allowed the creation of large scale loudspeaker arrays capable of exceeding the performance expectations not only of the audio industry in general, but even of the very engineers who created the technology.

IV PPST Measurement and Calculation Process



TO DEVELOP THE PPST PROCESS, EAW'S AUTOMATED TESTING FACILITY WAS MODIFIED, PLACING MICROPHONES AT FIXED LOCATIONS ON THE WALL. METAL BARS BOLTED TO THE FLOOR PROVIDED A FIXED LOCATION FROM WHICH TO MEASURE THE LOUDSPEAKERS.

The fundamental concept of PPST is quite simple. By measuring the performance of each driver "cell" across its intended coverage area and then applying separate signal processing to each cell, those cells can be forged into a single acoustical unit the performance of which can be optimized for specific venues and environments.

More specifically, if a particular point in space is to receive output from a number of different speaker elements, the complex sound pressure produced by each individual element can be measured and the total sound pressure when all elements are turned on at once can then be predicted. This is achieved by adding the complex pressures.

Of course, all the elements could be measured simultaneously, but nothing that would help control or integrate the elements will have been learned.

Once a separate measurement for each speaker has been taken, separate signal processing can be applied to each one. It is a relatively simple matter to calculate the complex response of the filters provided by digital signal processors. The filters are mathematically derived and mathematically implemented, so they can be mathematically characterized *exactly*. Multiply each measured speaker response by its corresponding filter response, and then add those results. Once again the result at a particular point in space matches exactly the response that will be measured with all of the processed speakers turned on at once.

Now that the combined result at that point in space is known, the response at that point can be made coherent by adjusting filter settings for each of the processor channels. This approach would provide one good seat at that point in space.

However, most venues have many more than one seat. Therefore, to assure even response throughout the venue, measurements must be taken at a variety of location (henceforth referred to as “mic positions”). Then, if we study the complex response at several mic positions as the processor settings are adjusted, response for the venue as a whole can be optimized. This last step of optimization is the key to achieving positive performance improvements.

While the PPST process is simple, the optimization step is complicated. The keys to its success are operator knowledge, judgment and experience, powerful purpose-built computer tools, and most importantly, listening. Each setting must be evaluated to ensure that it is aesthetically pleasing, while also satisfying all technical criteria.

One of the many benefits of the complex and exacting PPST measurement, optimization and modeling process is that a successful PPST design is perfectly repeatable. For a particular combination of elements, the operator can select a *known good* configuration as his starting point. Then he can adjust the parameters to achieve optimal performance in the new situation.

Precision Required

Before discussing specifics of the measurement process, it is important to have in mind an idea of the level of precision that is required to allow the process to work. For high level reproduction in large spaces, which is currently the object of the PPST process, significant constructive interference (driver coupling) must be achieved up to at least 8 kHz. In order to accomplish this goal, the level of precision in the measurement process that provides phase data must be within about ± 30 degrees at 8 kHz. This translates to a physical requirement of ± 0.14 inches (3.5 mm).

This requirement presents obvious physical challenges, as far as placing loudspeakers and microphones accurately and repeatably. If the goal is to apply the PPST process to loudspeaker elements in separate boxes, the boxes must be placed exactly as they will be in the target situation. Even the boxes themselves will need to be constructed very accurately for the process to work.

A less obvious challenge is presented by the characteristics of air itself. In order to avoid angular distortions in the measurements, the mic must be placed as far from the loudspeakers as practically possible. The measurements made to date have been conducted at distances of 75 ft (23 m) and 50 ft (16 m). The 50 ft measurements were done in a specially constructed facility, with hard points for the microphone positions as well as hard points in the floor for accurately positioning loudspeaker modules. It was felt that if the microphones and speakers were positioned accurately enough, measurements of different devices could be made at different times without requiring some measurements to be repeated.

Unfortunately, this approach was not successful at all. Despite our best efforts to set up the measurements with care, variations equivalent to as much as 6 inches were found from one measurement session to the next. The problem is in the temperature sensitivity of the speed of sound. From Beranek[1], the speed of sound can be represented as a function of temperature:

$$c = 331.4 \times \sqrt{\frac{T}{273}} \text{ meters per second}$$

where T is temperature in Kelvin, and (is the temperature in Celsius.

Near room temperature, the approximate speed of sound is:

$$c = 1052 + 1.106 \times F \text{ feet per second}$$

where F is temperature in degrees Fahrenheit.

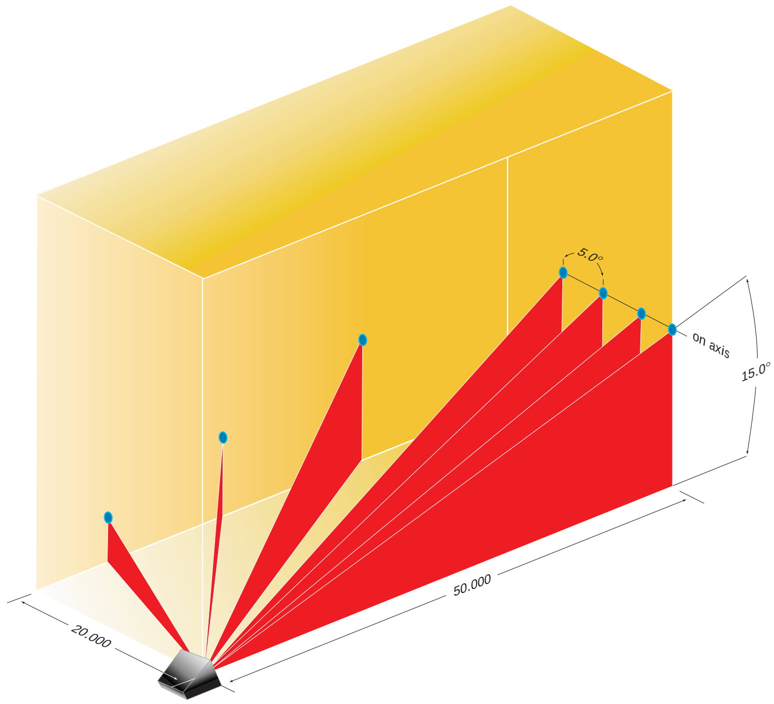
At a room temperature of 72 degrees, the speed of sound varies by a fraction of 1.106/1135 (or 0.1%) per degree. Applying this to a 50-ft measurement setup (such as was used in PPST development), the propagation delay varies by an amount equivalent to 0.58-in per degree of temperature change. The 6 inches of variability can be attributed to a 10° variation in temperature. In order to guarantee the required precision of ± 0.14-in, the testing facilities temperature would need to be controlled to ± 0.2°, which would not be practical. A practical alternative would be to incorporate a correction factor in the measurement process which adds or subtracts delay, according to the measured temperature in the laboratory. For the moment, the decision was taken to measure all the elements of a particular model in the same setup at the same time, thereby minimizing temperature fluctuation.

Measurement Setup

The first step in the process, of course, is to take acoustical measurements of each of the sources at each mic position. On the surface, it seems that the best approach would be to take all the measurements in situ. If microphones were placed at carefully selected positions in an actual venue, the actual results could then be optimized via signal processing thereby avoiding the step of translating the results from the measurement setup into the venue. However, there are serious practical problems with preserving the required precision at long distance.

Even if a calm day were selected on which to take measurements, there would still be slight air currents, as well as changing temperature gradients, which would prevent the achievement of the required precision of ± 0.14-in. Real-time two-channel analyzers such as SIMM and SMAART are very effective at displaying the variability in arrival time and frequency response shape that occurs in long-distance measurements. It is not uncommon in a concert setting to observe arrival time excursions of one or more milliseconds in the presence of a light breeze (0.14 inches is .009 ms).

Obviously, then, it is futile to attempt PPST measurements in situ in large spaces. This conclusion raises the question, “If the measurement precision is inadequate at distance, doesn’t that mean the acoustic summation itself will fall apart at distance?”



THE PPST PROCESS MEASURES EACH LOUDSPEAKER "CELL" AT EACH MIC LOCATION. THE DATA GENERATED BY THIS PROCESS ARE USED TO PREDICT THE PERFORMANCE OF ARRAYS IN SPECIFIC VENUES.

Fortunately, of course, the answer is no - or this paper would not have been written. As long as the sources are close enough together to be coherently integrated, each of their contributions will encounter almost identical acoustical environments enroute to the listener.

In effect, a single source will have been created which has the same susceptibility to environmental changes as any other single source. It was simply found impractical to measure the subcomponents of the source in situ.

Obtaining Valid Summation Calculations

The complex frequency response measurements are obtained using a time-windowing measurement system. The measurements performed to date have been taken using a Bruel & Kjaer 2012 Audio Analyzer and Model 4135 free-field measurement microphone.

In the case of the KF900-Series, the cabinets would rarely be used in single columns, since the horizontal coverage angle is a fixed 30 degrees. It is appropriate then, that all PPST measurements were taken with the cabinet's virtual twin next to it. That is, it was laid on its side on the floor so that, acoustically, a two-wide cluster was represented. The microphone was then placed at various points along the vertical bisector of the cabinet which is, of course, angled 15 degrees away from the floor (see figure x).

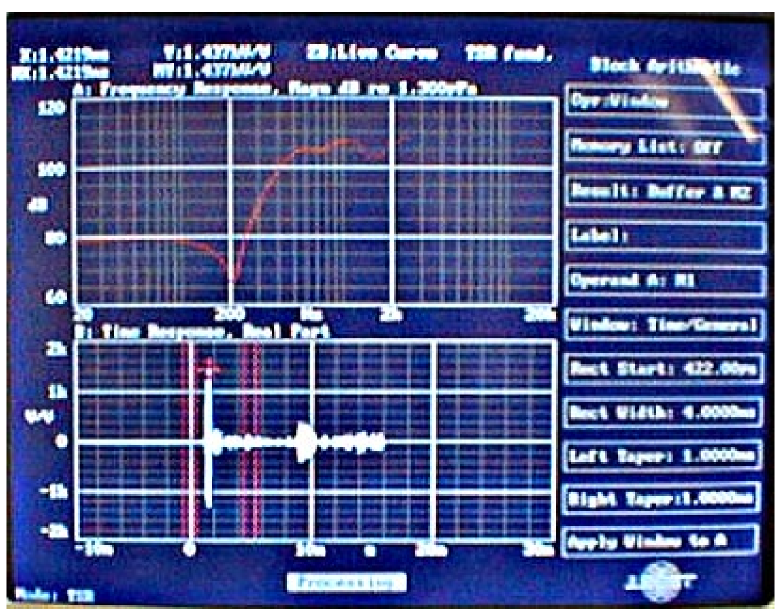
The selection of the time window is a decision which is critical to the process. With numerous sources in close proximity, there are diffractions and reflections off of adjacent horn mouths and cabinet edges, sympathetic horn resonance, as well as the arrivals from the adjacent virtual cabinet (the floor reflection).

One of the goals of PPST is to accurately predict the response produced at a distant mic position. However, this goal is not primary. The primary goal is to maximize the quality of the response at a distant mic position. If predictive accuracy were primary, the decision would have been made to use a very wide time window. All of the late arrivals would then be incorporated into the response, and the correlation to measured results would be maximized. However, it was felt that the coherence of the system could be improved by running the optimization process on just the primary arrival from each source.

The late arrivals from one cell are not very similar to the late arrivals from other cells, because each cell has a slightly different environment (different neighbors and boundary conditions). The primary arrival, on the other hand, is almost identical to nearby identical cells. Another way to state this is to say that the primary arrivals are highly correlated, while the late arrivals are highly uncorrelated.

When the contributions from numerous cells are added, the primary arrivals sum in a correlated fashion. Eight correlated sources sum to +18 dB. Late arrivals add in random fashion. Eight uncorrelated sources sum to +9 dB, and are smeared in time, which reduces their perceived level at high frequencies. Consequently, it was thought that the summation of numerous sources could be used to increase the level of perceived coherence. This is an open-ended judgment, since there has been no attempt to verify, mathematically, that the coherence is measurably higher as a result of excluding the late-arriving energy. On the other hand, many observers have characterized the transient attack of the KF900 system as startlingly impulsive, which tends to support the contention.

DATA FROM MEASUREMENTS ARE FED INTO THE BRUEL & KJAER FFT ANALYZER WHICH FIRST "WINDOWS" THE FIRST ARRIVAL TO ISOLATE IT FROM REFLECTIONS (BOTTOM GRAPH).



Another reason for using a small time window is that the frequency domain results can then be represented with relatively low resolution (i.e., 1/6 octave). Late arrivals

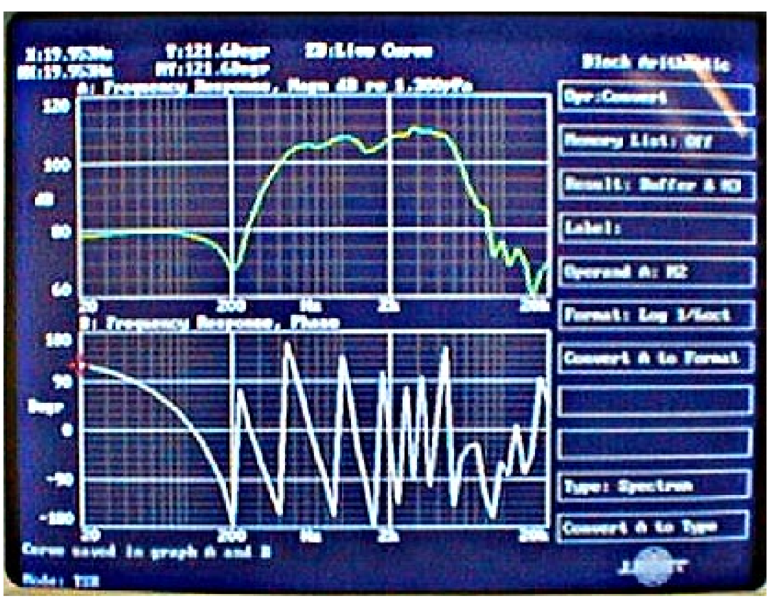
would necessitate densely-spaced frequency points in order to correctly represent their phase response.

Reducing the complex frequency response data to low-resolution is essential to the optimization process. At 1/6-octave resolution, the calculation and optimization processes can be accomplished within a reasonable length of time (several days' runtime for a new configuration, several minutes to several hours for a derivative configuration starting from a "known good" configuration).

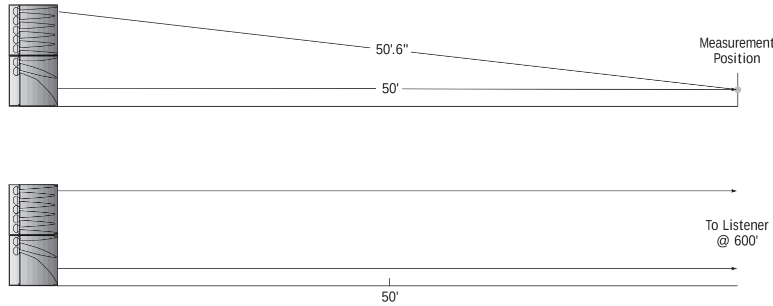
THE WINDOWED FREQUENCY/PHASE DATA (ABOVE) ARE THEN "NORMALIZED" TO A 1/6 OCTAVE RESOLUTION (BELOW). DATA IN THIS FORMAT ARE THEN NAMED AND SAVED FOR FUTURE USE IN THE PPST MODELING AND OPTIMIZATION PROCESSES.

Validity and Conditions for Superposition

There are some prevalent misconceptions, stemming from the sometimes mysterious interactions that occur between nearby loudspeakers. If two direct-radiating low-frequency loudspeakers are placed in close proximity, a phenomenon known as mutual coupling occurs. It is common knowledge that if the two loudspeakers are close enough together, their efficiency increases. The acoustical power doubles, for a given electrical power.



Consider a test case. Two different low-frequency loudspeakers are measured independently at a single microphone position. The question could be posed, "What will the response be with both turned on at once?" Many would respond, "That depends on how close together they are." They would be wrong. As long as the conditions for superposition are satisfied, the result is simply the complex sum of the two measurements. Mutual coupling is not a separate issue from superposition. In fact, mutual coupling is predicted via the principle of superposition. If the pressure over a complete sphere could be measured and the power produced integrated, the pressure map predicted by superposition would predict the efficiency increase that is usually treated separately.



THE UPPER GRAPHIC ILLUSTRATES THE DIFFERENCE IN DISTANCE BETWEEN A MICROPHONE PLACED AT 50 FT AND THE CLOSEST AND FARTHEST CELLS IN A KF911/KF910 CONFIGURATION. WHILE AT THE LISTENING POSITION, SOME 600 FT AWAY, THE DISTANCES ARE ESSENTIALLY EQUAL, AT THE 50 FT MEASUREMENT POSITION THE DIFFERENCE IS 6-IN, WELL OUT OF OUR ±0.14-IN MARGIN OF ERROR.

The condition for superposition is simply that all legs in a circuit (whether electrical or acoustical) must retain the same impedance, whether the sources in that leg are active or not. In electrical circuit analysis, this means that inactive current sources must be replaced by open circuits, and inactive voltage sources must be replaced by short circuits. In acoustics, it means that inactive sources (woofers & compression drivers) must have their terminals shorted, either by an amplifier output or a wire.

For the purpose of PPST, when a cell is measured, all neighboring drivers of the same type are present with shorted terminals.

Geometric Focus

As discussed above, individual sources should be measured from as far away as possible, in order to minimize angular errors. However, the distances must be small enough to satisfy the precision and repeatability requirements. The current standard measurement setup employs a 50 ft microphone distance on the axis of the speaker. Intuitively, it would seem that 50 ft would be adequate to predict the results at long distance (400+ ft). However, a geometric analysis shows that the relative arrival time between the first and last element in a KF900 high-frequency array is over 5 inches different at 50 ft than it is at infinite distance (see figure x). To achieve optimum results at long distances, these errors must be corrected in the summation model. This is accomplished by adding phase delay to the response of the nearer sources.

The focal length should be corrected separately for each mic position, since the geometry and ideal focal length is different for each. The desired focal distance at -60 degrees may be only 30 ft, while at 0 degrees it may be 600 ft or more.

Performance Representation

The traditional approach to loudspeaker coverage has relied on polar response plots and beamwidth specifications. In the large venue applications under discussion, these traditional measures are perfectly applicable in the horizontal plane, but have very little utility in the vertical plane. The nearest listeners are typically 40 ft from the nearest speaker while the farthest might be 600 ft or more. That is a difference of 24 dB in inverse square law attenuation and 18 dB in air absorption at 8 kHz.

Fortunately distance and, therefore, air absorption can be mapped directly to vertical angle. The distance to the listener directly in front of the speaker can be determined and both the attenuation and air absorption calculated. Likewise, the distance and air absorption can be calculated for each angle below horizontal and the attenuation curve produced may be applied to the one-meter-equivalent frequency response. The result is a family of curves representing the frequency response at the listening positions.

Importantly, not only the overall level but the shape of the frequency response must be different for each direction below horizontal. Consequently, the polar response must be different for each frequency, and the part of the plot that is 30 dB down is just as important to PPST as on-axis response is. The response 60 degrees below horizontal may be 30 dB lower than it is on the axis, but it will encounter over 30 dB less attenuation, so it will actually produce the same or higher SPL at the listener.

A much more enlightening presentation of the vertical directional information is the “family of curves”. If the response is displayed at each of a standard set of below-horizontal angles, the characteristics and quality of the directional response can be assessed at a glance. It is also helpful to present both the “attenuated with air absorption” and the “one-meter equivalent” families. The former indicates the net performance attained, while the latter gives an experienced operator an instant reading as to whether any improvement is possible.

For large-scale arrays, focal errors result in a radically different frequency response at 50 feet from the array than they do at 400 feet. In addition, air absorption has a radical effect on the frequency response for distances greater than 50 feet. So, it should be obvious that there is very little that can be learned from a polar response measured at 20 feet.

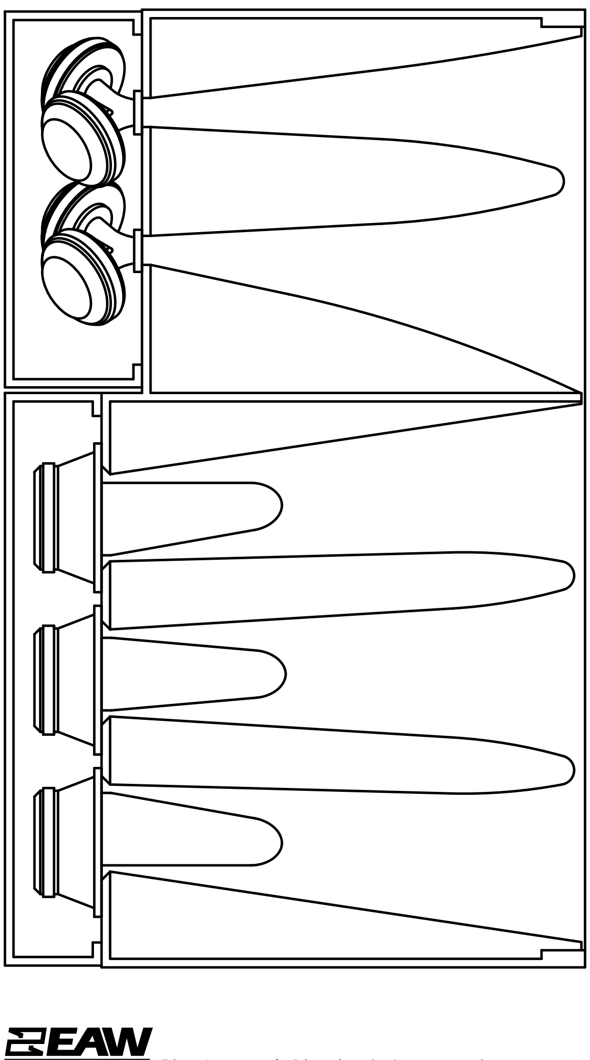
Given this considerable theoretical background and methodology for its implementation, the next challenge was to create the loudspeaker modules that would bring these concepts to reality.

V The KF900 System - A clean sheet design with PPST in mind

Historical Genesis - The MH433

Phased PointSource Technology was developed in response to the challenges encountered in the design of a speaker system that was specifically targeted for use in stadiums and other very large venues. The first product developed through the course of this program was the MH433 mid/high speaker system. It employed a new, three-celled mid-range section powered by three phase-plug-loaded 10-inch cone transducers, and a two-celled, vertically asymmetrical high-frequency section powered by four compression drivers mounted on Y-throat adapters.

CROSS-SECTION OF AN MH433. NOTE THE MANIFOLD LOADING OF THE HF DRIVERS.



One of the goals for the MH433 had been to provide a “soft” pattern edge underneath, in order to minimize the number of additional components required for coverage below the main system. The compound horn employed a vertical profile with a naturally “soft” pattern underneath and a hard cutoff above horizontal. However, the two cells produced significant interference, which somewhat negated the soft underside. Consequently, a passive shading circuit was developed to smooth the transition. The PPST concept evolved through the development of this circuit.

The passive shading circuit effected significant improvement in the measured response of the system, providing a response with maximum sensitivity on axis, 12 dB of attenuation at 30 degrees below horizontal, and yet, exhibited generally flat response in both directions. Furthermore, listening tests showed conclusively that the technique was very successful, aesthetically, as well.

Buoyed by the success of the MH433, the PPST process then became the driving force behind an all-new system. The physical positioning of the sources, the characteristics of the drivers, the design of the horns, and the physical details of the cabinets were all designed specifically to work well in a PPST-controlled system. This system that resulted is what has now been named the KF900 Series.

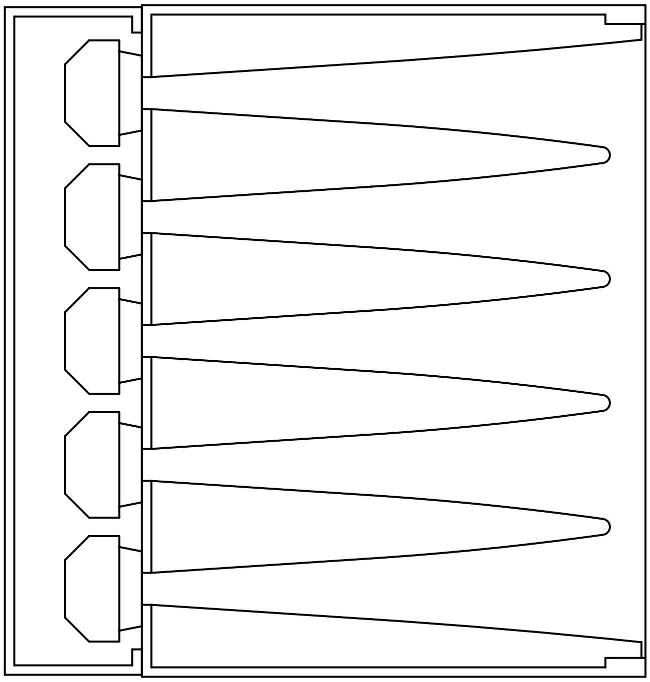
The Challenges

The MH433 had shown that PPST could be an extremely powerful tool - so powerful as to drive a re-assessment of what could be accomplished in very large spaces. It was entirely possible that performance parameters could be achieved which had never before seemed possible.

An accounting had to be taken of the primary challenges involved with very-large venue sound reinforcement. To bolster the input that was solicited from various customers and in-house experts, sound-system design consultants were employed to clearly identify these challenges - with special emphasis on the challenges that had not yielded to current state-of-the-art approaches.

From the outset, it was apparent that the primary challenge associated with very large venues is high-frequency projection. Because of the tremendous high-frequency absorption that occurs over distances of 400 ft or more, it is extremely difficult to achieve meaningful output at 8 kHz or higher. In fact, over the 600-foot throws that are frequently encountered in stadiums, it is rare to achieve flat frequency response past 5 kHz. Extending the upper frequency limit at the farthest seats would become the first focus for the system.

The next most important challenge was to prevent the SPL near the array from getting too high. Producing enough power to achieve high levels at the farthest seats is a futile exercise, if the buildup below the array exceeds comfortable levels for the listeners there. Conventional concert-style arrays do a good job of projecting mid-to-high frequencies to miss the front rows, but produce excessive low-mid and low frequency levels in the front rows. Preventing excessive SPL near the array would become the second focus of the design.



CROSS-SECTION OF A KF910 LONG THROW HF MODULE. NOTE THE RADICAL VERTICAL DRIVER PACKING.

Conventional approaches to high level sound reinforcement have typically treated long-throw coverage and front-row coverage as two separate issues, to be addressed with two separate sound systems. The result is the familiar “flying junkyard” that results, with additional loudspeakers suspended below the main system and aimed down into the front rows. As long as PPST is being applied to the challenge of SPL reduction under the array, every attempt should be made to make the coverage in that region approach the same quality produced in the main beam. The third focus, then, was to smoothly transition to high-quality coverage below the array.

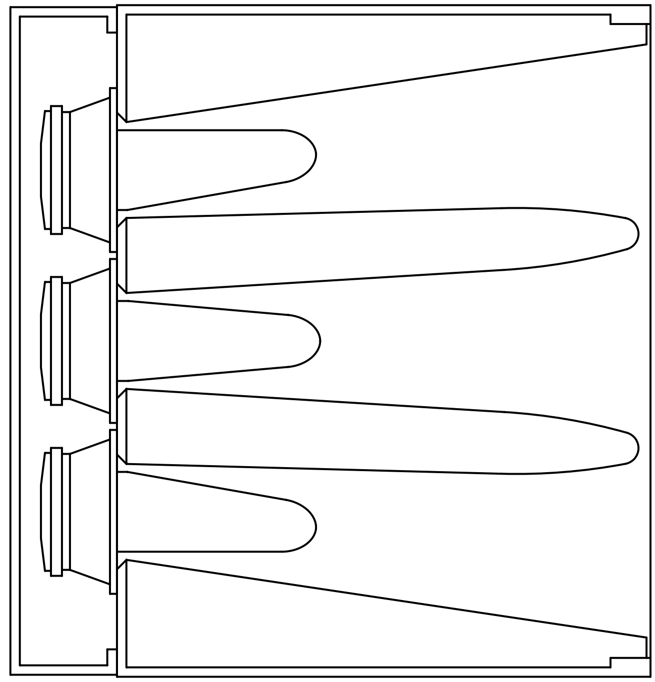
Previous generations of trapezoidal cabinet-based loudspeakers made great strides in smoothing the horizontal transitions from cabinet to cabinet. The new system would be expected to perform no worse than the best existing systems. Some new theory regarding horizontal arraying had emerged, and a very simple application of PPST could be applied. So, the fourth focus became the establishment of a new standard of horizontal arrayability.

Finally, there needed to be a recognition of the characteristics of existing systems that were successful, and ensure that the new system did not fall short in any of those. The primary concern, here, was with aesthetic issues. The solutions to the enumerated technical challenges also had to meet the most stringent criteria for aesthetic results. The fifth focus was to use listening and field verification as the final measure of success.

This fourth challenge will be considered first because, in actuality, it was met before the others were approached.

Horizontal Arraying

The horizontal arrayability was addressed in the same way in the midrange modules and in the high frequency modules. The goal was to produce, as nearly as possible, a continuous con-



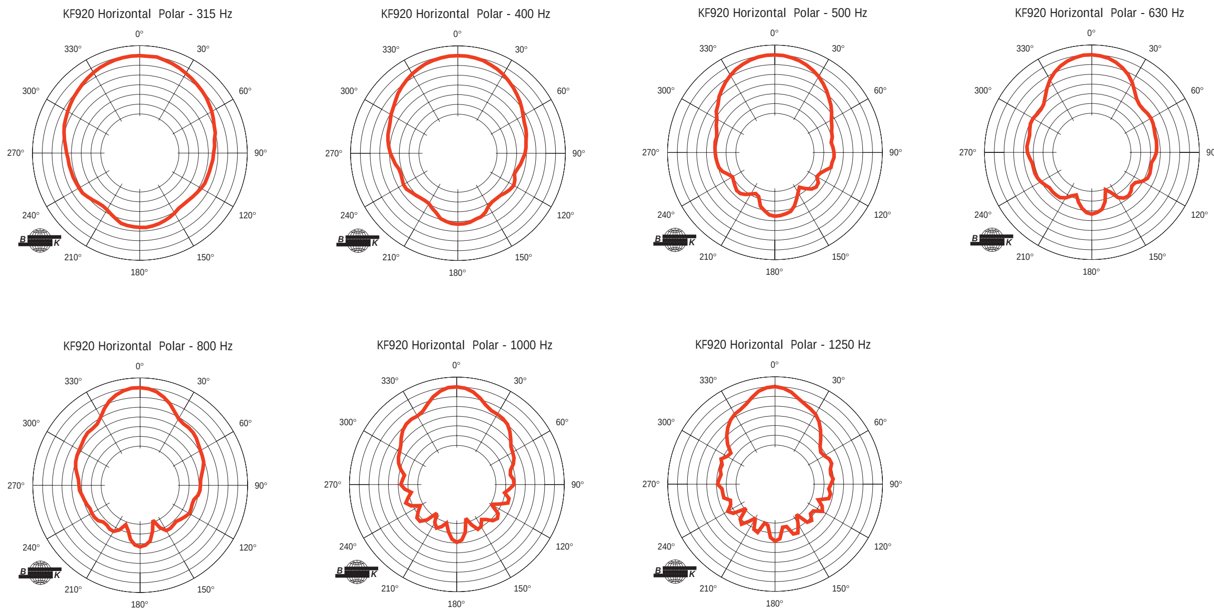
CROSS-SECTION OF A KF920. LIKE ALL SIMPLEPHASE HORNS, THE MF HORNS' WALLS ARE VIRTUALLY STRAIGHT.

stant-radius arc. In order to accomplish this, all horn mouths would occupy the full width of the cabinet, and would be so deep that the drivers were physically as close together as possible. The horns would not employ a constant beamwidth profile, but would use a modified-conical profile with an initial angle essentially matching the 30-degree angle of the cabinet walls. As in the vertical profile, a slight, diffraction-reducing flare would be employed.

Simplified modeling illuminates the reasoning behind the selection of horizontal horn profiles. A truncated arc source produces a beamwidth which dips to 2/3 of the nominal angle, before widening again to approach the nominal angle[2]. A 45 degree arc, for example, narrows to approximately 30 degrees before widening again to approach 45 degrees at high frequencies. Similarly, a 90 degree arc narrows to approximately 60 degrees before widening again to approach 90 degrees.

A constant beamwidth horn uses a bent or curved sidewall to present a path delay to the portions of the arc near the ends. The source shape produced is not an arc but an arc with lagging ends. This source shape produces a beamwidth which approaches the nominal angle without narrowing first, but it cannot be "spliced" with others to create a continuous arc. Rather, the source that is produced by an array of constant-beamwidth horns has a scalloped shape, which must exhibit lobing in its polar response.

The apparent drawback of producing a pure arc-shaped source is that the resulting array will exhibit beamwidth narrowing at some frequency. In the case of a large-scale array, however, the frequency of narrowing will be quite low, and the crossover may be set above it. From Olson[3], the narrowest beamwidth produced by a 90-degree arc occurs when the wavelength is equal to the radius. In the case of the KF920 mid frequency module, the arc radius is 56 inches, for a narrowing frequency of 243 Hz. For a 120-degree array, the frequency is 121 Hz.

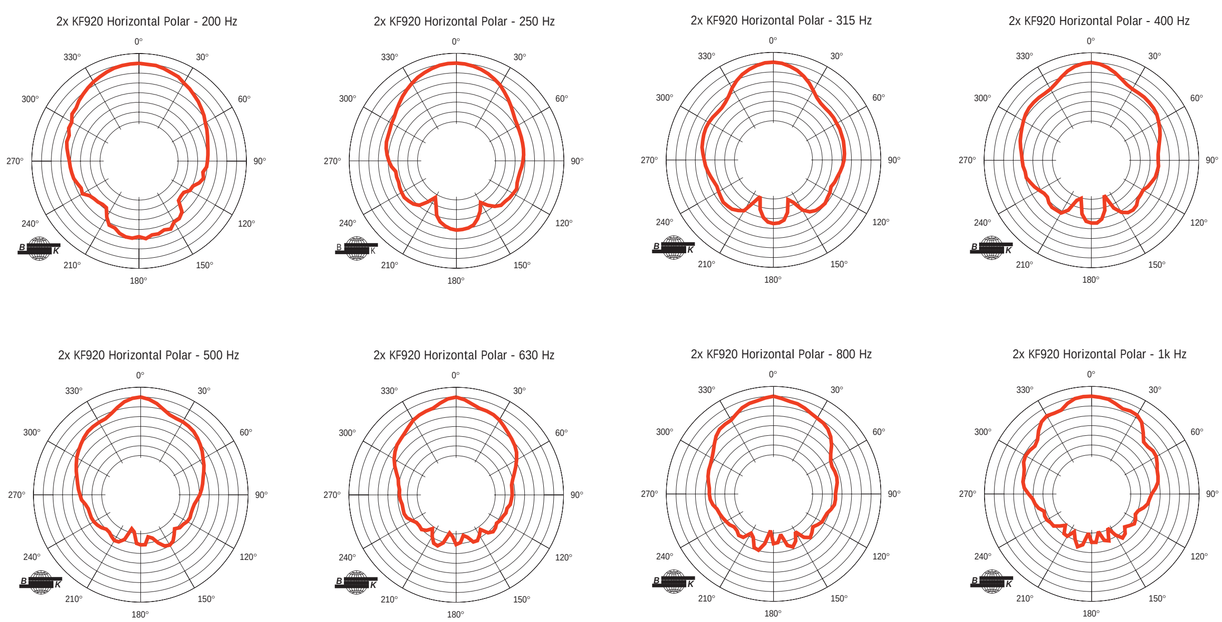


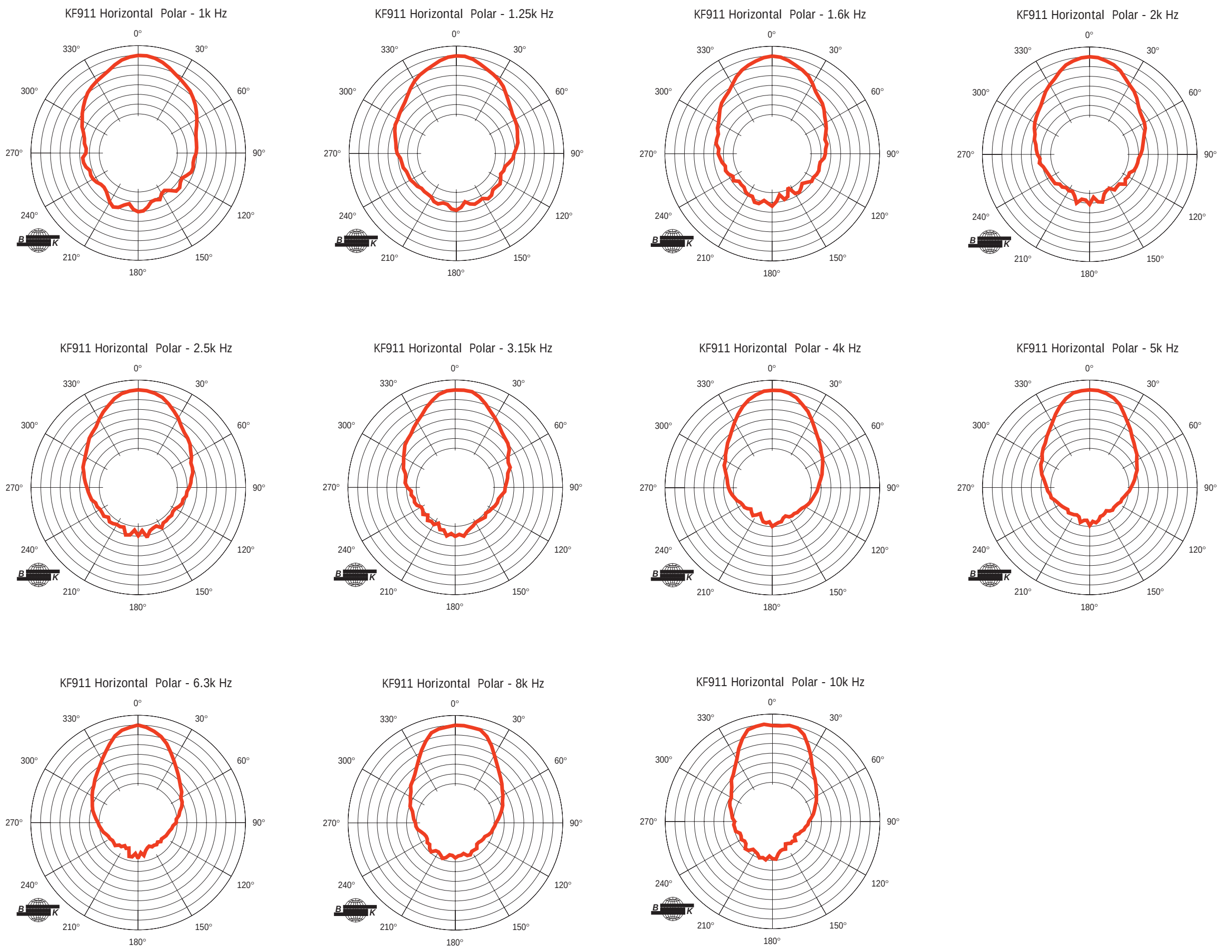
HORIZONTAL POLAR PLOTS OF ONE KF920. NOTE THE WIDENING OF THE PATTERN BELOW 630 HZ, THE NATURAL CONTROL LIMIT OF A SINGLE SIMPLEPHASE HORN.

It is apparent, then, that given the typical 160 to 350 Hz crossovers employed in the KF900 Series, beamwidth narrowing must be dealt with in the low-frequency cabinets only. A simple, inherently arrayable horn profile could be employed in both the high frequency and midrange cabinets. The result is extremely smooth transitions from cabinet to cabinet, as evidenced in the polar measurements below.

HORIZONTAL POLAR PLOTS OF TWO KF920'S. COUPLING OF ADJACENT HORNS EXTEND PATTERN CONTROL TO 300 HZ.

The narrowing of coverage that occurs in tight-packed arc arrays is easily corrected by applying delay to the outside cabinets in the arc (the first and last cabinet in a row). In



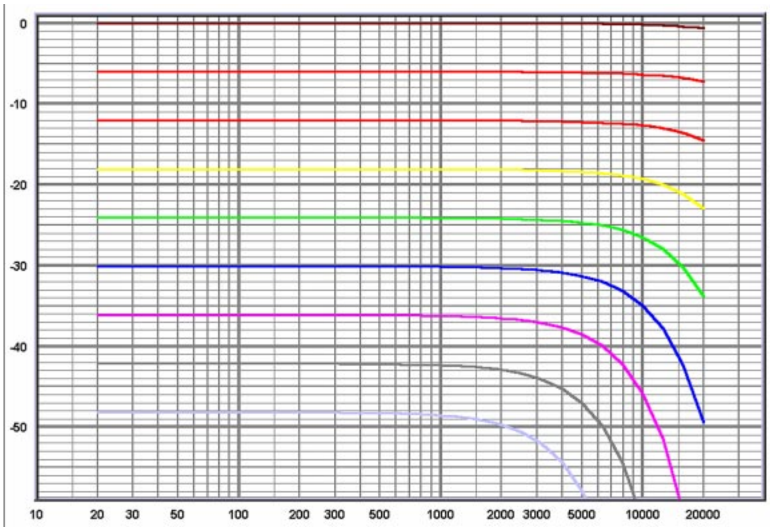


HORIZONTAL POLAR PLOTS OF ONE KF910. PATTERN CONTROL IS CONSISTENT UP TO 10 KHZ.

order to limit the number of processor channels required for large arrays, the required delay has been provided in the form of a switchable passive filter in both the KF920 and KF930 cabinets. An alternate way to deal with this problem is to deploy the KF930s in columns, rather than arcs. A single column of KF930s will have wider horizontal coverage than the rest of the array, which may be acceptable in many cases.

High Frequency Projection

The attenuation of sound over distance has two main components, inverse square law attenuation & air absorption. These two mechanisms are fundamentally different in that the inverse-square law expresses the dilution of pressure that occurs as a wave front diverges, while air absorption refers to the conversion of acoustic power to heat. There are other important differences:



THIS CHART ILLUSTRATES THE EFFECTS OF THE INVERSE SQUARE LAW AS WELL AS AIR ABSORPTION. THE BROWN LINE AT 0 DB REPRESENTS SOUND AT 1M. EACH SUCCESSIVE LINE REPRESENTS SOUND AT A DOUBLING OF DISTANCE (2M, 4M, 8M, ETC.) NOTE THE SUBSTANTIAL 10 KHZ ATTENUATION AT 64M (VIOLET).

Inverse-square law attenuation is independent of frequency. Air absorption is strongly dependent on frequency, having the greatest effect on high frequencies. For most atmospheric conditions, air absorption can be approximated very closely by a Bessel low pass filter.

Inverse-square law attenuation produces 6 dB of attenuation per doubling of distance. Air absorption produces attenuation at a rate expressed as decibels per unit distance for a given frequency (for example, 6 dB per 100m at x Hz).

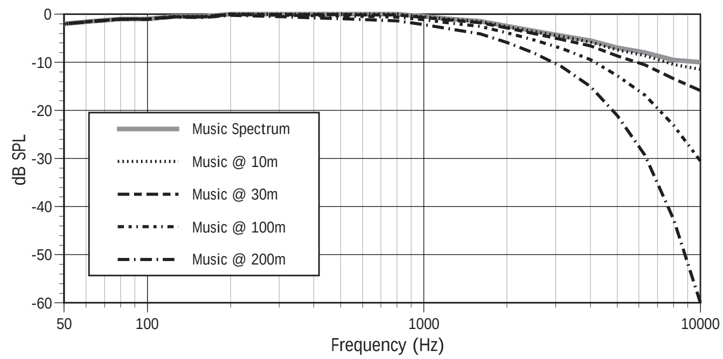
The family of curves representing the attenuation at various distances from an ideal source is shown in figure X. This family represents the calculated behavior for 72 degrees F, and 50% humidity. Several interesting observations can be extracted from this figure.

The response at 128m and 256m illuminate the challenge of projecting high frequency sound over long distances. In order to project 8 kHz over a distance of 256m, 24 dB of air absorption must be overcome, in addition to 48 dB of dilution. To achieve an 8 kHz level of 90 dB SPL at 256m would require a 1-meter equivalent output of 162 dB.

Another interesting aspect of figure X is that absorption becomes significant at surprisingly short distances. For 3 dB of attenuation at 10 kHz, the listener must only be 20 m from the source. Whereas air absorption is often dismissed as a “super-long-distance” phenomenon, it is actually significant for any distance over 10 m.

The consequence of absorption is that the power-handling of a system is thrown severely out of balance when the system is equalized for long-distance use. Most loudspeaker systems adhere to a sensible balance of low-frequency to mid-frequency to high-frequency power handling. The spectrum of “normally balanced” music has much less high frequency energy than pink noise, for example, so a typical system might combine a 600W woofer with a 300W midrange and a 150W compression driver. If a system incorporates 12 dB of pre-emphasis at 8 kHz to combat air absorption, then at least 6 dB more high frequency capability will be required. The previously mentioned typical system would need at least 600W of power handling in the high frequency section.

Chart 4 - Air Absorption Effects on Typical Music Spectrum



THIS GRAPH ILLUSTRATES THE EFFECTS OF AIR ABSORPTION ON THE TYPICAL MUSIC SPECTRUM AT VARIOUS DISTANCES.

High Frequency Compression Driver Selection - CD6001

The ideal compression driver for such a system should satisfy a very different set of criteria than the ones established for drivers used in more conventional systems. Significant pre-emphasis (12 dB or more) will be required to extend the response as high as 8 kHz. There will be no detectable energy above 10 kHz at distance. So, the driver should be optimized for maximum 5 kHz to 8 kHz sensitivity, even at the expense of 10 kHz to 20 kHz sensitivity.

A system such as this will have no “headroom”, per se. All available output can be expected to be used whenever the system is operational. Consequently, the driver must be capable of reliable full-power operation essentially full time. Also, the magnitude & phase response must be stable under full-power operation. And, the sonic character must be assessed at full power.

All of these criteria were met admirably in the CD6001 compression driver. It has a 4" titanium diaphragm, a neodymium magnetic structure, and a simple half-roll suspension. Unlike competitive 4" drivers, the first suspension breakup mode occurs well above 10 kHz, so it does not affect the phase response in the critical 5 kHz to 10 kHz region. It also contributes to the driver’s exceptional unit-to-unit consistency.

High Frequency Horns

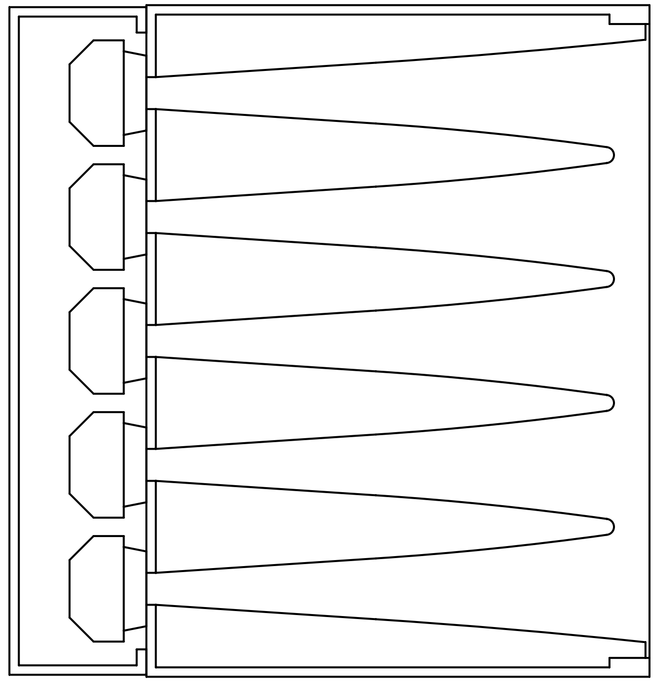
The system's high frequency section is based upon a psycho-acoustic observation. It has been observed that adding high frequency sources to a system does not add to the perception of high-frequency level unless the added sources are almost perfectly synchronous with the other sources. Hence, adding full-range cabinets to an array of similar full-range cabinets does not necessarily increase the perception of high-frequency content in the audio image (the voice or musical instrument), even though high frequency power is being added to the system. The spacing between high frequency sources is too great for ear integration to take place. So perceptually, the high frequency energy seems to become part of the reverberant field, without becoming part of the primary audio source.

In order to integrate the high frequency energy as constructively as possible, the high frequency horns in the KF910 long throw high frequency module are packed as tight together as possible, vertically. The spacing between horns is essentially set by the driver diameter. The horn mouths, at 7 in. tall, are only slightly taller than the diameter of the compression driver. As a result of the small vertical mouth dimension, the individual horn cells exhibit very little vertical pattern control individually. However, when operated together as a line source, the module exhibits extremely tight vertical directionality.

In fact without processing, the KF910 exhibits an unusably narrow vertical pattern. Above 6 kHz the vertical beamwidth is less than 5 degrees.

The first generation of PPST processing (PPST-1) was employed to broaden the pattern of the KF910. With a simple scheme of delays and parametric EQ settings, the beamwidth was converted to a constant 8-degree pattern with a smooth, MH433-like broadband roll-off below horizontal. With other settings, a usable 15 degree beam was created, and experiments showed that +/-5 degrees of beam steering could be accomplished, though the parametric sections had to be varied also, in order to preserve the smooth pattern edge.

The horns themselves have a deceptively simple shape. The horizontal profiles will be addressed below, when horizontal



CROSS-SECTION OF A KF910. WITHOUT PPST PROCESSING, A KF910'S VERTICAL PATTERN WOULD BE LESS THAN 5° AT 6 KHZ. WITH PPST PROCESSING, ITS PATTERN IS VARIABLE FROM 8° TO 15° AND CAN BE STEERED ±5°.

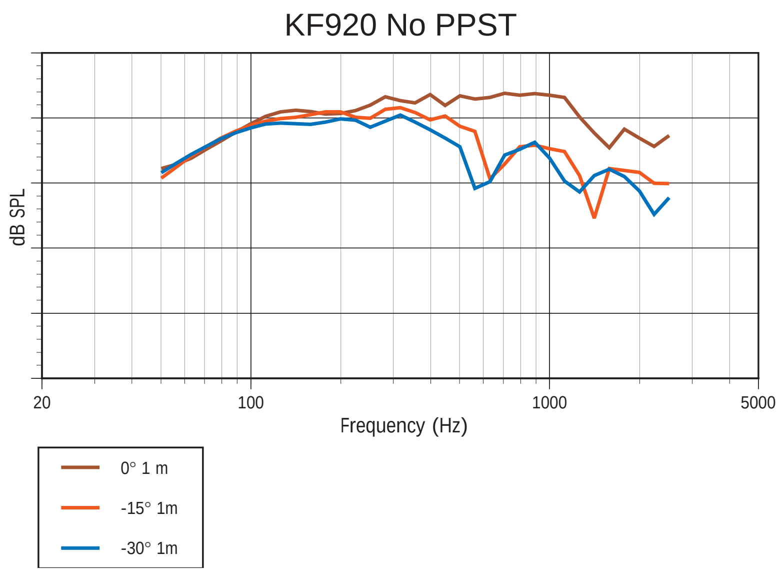
arraying is discussed. The walls which establish the vertical profile are almost parallel initially (near the throat). Out toward the mouth, the walls flare slightly before blending into a rounded separator, which separates one horn from the next. This profile is essentially a diffraction slot up to 3 kHz or so. Above 3 kHz, it is most similar to an exponential horn - narrowing with increasing frequency. The slight flare simply minimizes diffraction of high frequencies from the mouth edges.

Rather than attempting to produce a constant-directivity beam or other class of directional behavior, the horn simply seeks to produce a simple phase-vs.-frequency characteristic, so that it will engage in the PPST process with minimal anomalies. This is accomplished by avoiding diffraction by eliminating sharp wall bends & mouth edges, and by providing an exceptionally resistive throat impedance. A side effect of the resistive throat impedance is almost complete freedom from horn resonance, a character which is immediately apparent upon listening to the system.

Broadband Attenuation Below the Array

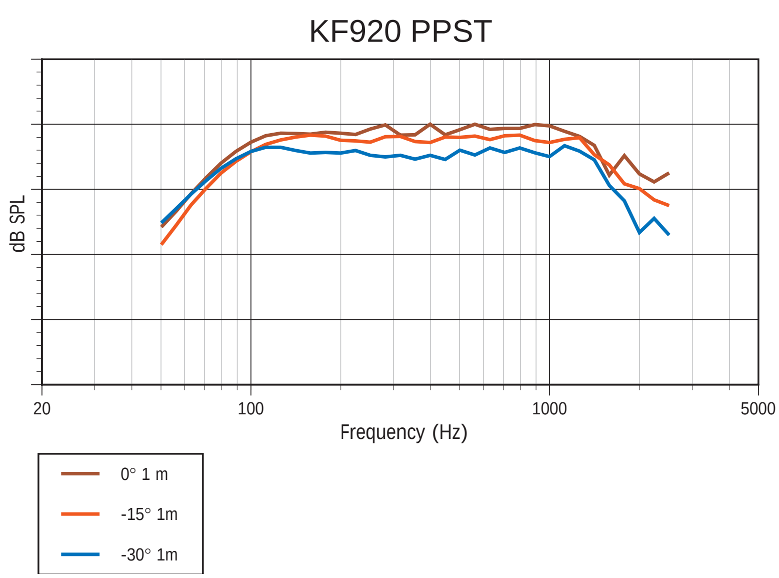
The first generation of PPST was spreadsheet based, and used simple point and line source models. It accurately predicted what results could be obtained, but the filters that accomplished those results had to be somewhat different than the method predicted. Nevertheless, the predictions were very valuable in that they established that the MH433 midrange section was capable of full angular coverage down to 60 degrees below horizontal, with appropriate PPST processing. Consequently, the downfill function would only have to be supported in the high-frequency section.

Furthermore, the processed midrange section, later named the KF920, produced essentially flat response, even 60 degrees below horizontal. Both the broadband attenuation & "soft edge" goals could be met with a single KF920. In fact with



RAW KF920 RESPONSE IS UNACCEPTABLE AT ONLY 15° BELOW THE ARRAY.

PPST PROCESSING ALLOWS THE KF920 TO PROVIDE ADMIRABLY FLAT RESPONSE AT 30° BELOW THE ARRAY.



carefully selected filters, the response below the stack can exhibit more attenuation to a lower frequency than an unprocessed, naturally “beamy” source.

At low frequencies, PPST-1 predicted that a stack of five low-frequency modules could be processed for broadband control with flat response down to 50 Hz. A processing channel for each cabinet was sufficient. However, it seemed apparent that the system would be more flexible if the required control could be accomplished with fewer vertical rows of cabinets. Tuned Dipole Technology proved to be the ideal approach, since useful control could be accomplished within the height of two modules. The KF930 low-frequency module was developed to produce Tuned-Dipole spacing when two modules are put together. This configuration provides over 10 dB of broadband attenuation directly below the array, down to about 80 Hz.

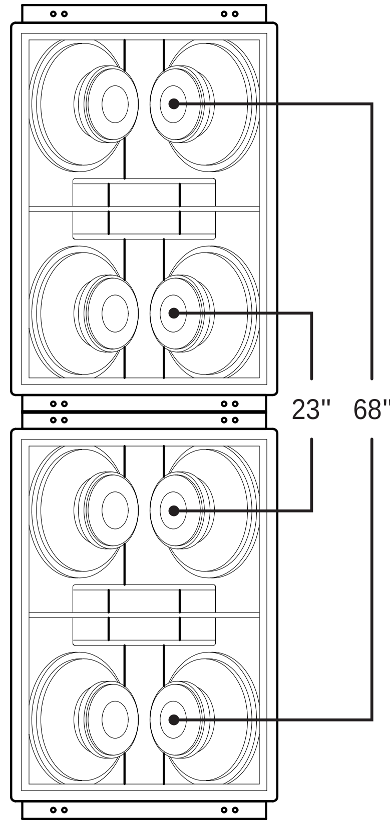
Each KF930 module houses four 600W, 15-inch woofers. When used in configurations other than tuned dipole, all of the woofers can be driven equally. PPST-2 has been used to produce configurations ranging from maximum attenuation (tuned dipole mode) to maximum low frequency output (with less attenuation below the array). Configurations have also been created for three and four-high KF930s.

Soft Lower Pattern Edge

With broadband attenuation accomplished below the array, the goal of providing a soft lower pattern edge was then reduced to the task of adding high frequency coverage. Unlike the KF920 midrange unit, the KF910 high frequency module could not be processed for adequate coverage to 60 degrees below horizontal. The individual horn cells are too directive at high frequencies to be used 60 degrees off axis. It became obvious that a downfill module would be required - the KF911.

The KF911 downfill module contains two CD6001 compression drivers. The upper driver is loaded by a horn which covers from 0 degrees (horizontal, or on axis) down to -15 degrees. The lower cell covers from -15 degrees to - 45 degrees, with a soft edge that is usable down to - 60 degrees.

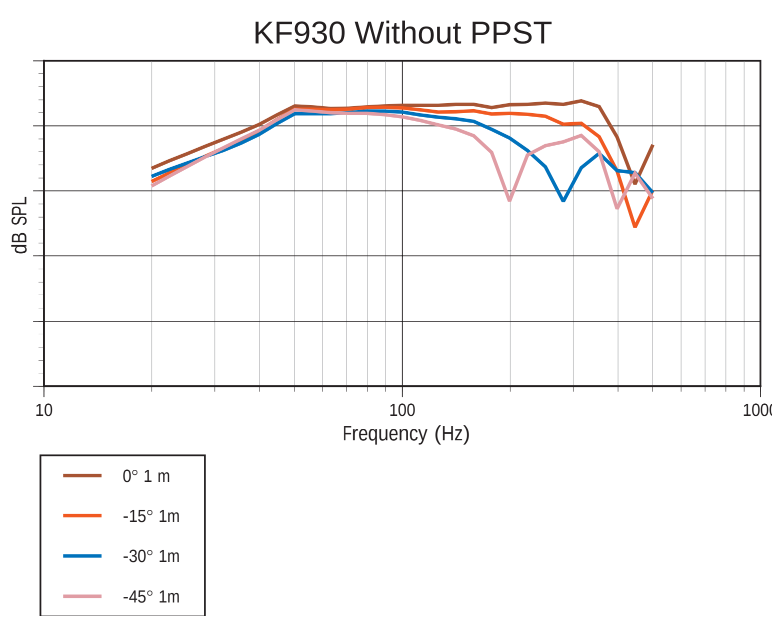
The lower cell is remarkable in that it doesn't break the bottom panel of the cabinet. Geometrically, to accomplish the angular coverage that it does, the lower compression driver would have to be located within the mouth of the upper horn. Instead the lower cell uses its upper wall surface as a mirror surface, so that the virtual image of the driver is located within the upper horn. Like the long-throw cells, the mouths of the KF911 are immediately adjacent, spaced only by a narrower separator. Consequently, the sources can be very intimately combined, using PPST processing.



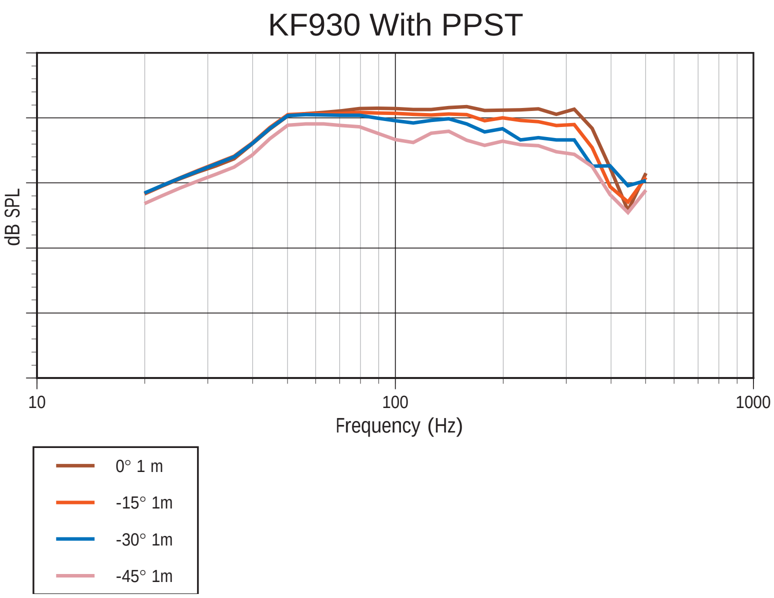
FROM THE RESPONSE CHART BELOW, IT IS OBVIOUS THAT SOME FORM OF PROCESSING MUST BE USED WITH KF930 LF MODULES. THE TDA APPROACH (SPACING ILLUSTRATED ABOVE) IS ONE OPTION.

At this point in the development of the high frequency section, the primitive version of PPST had met its match. The downfill cells do not match any simplified source model closely enough to be useful, so the only way to approach the problem was by way of the full, measurement-based approach. Furthermore, several attempts to process a KF910/KF911 combination by trial & error only proved that the results without a PPST-based process would be unacceptable. It looked as though the KF910 and KF911 might have to be separated in the array by mid & low frequency cabinets, in order to reduce the effect of the interactions.

It was felt that it would be far preferable to integrate the two high frequency modules in adjacent positions, because that would minimize spectral variability over the vertical pattern. Development was accelerated on the more ad-



KF930 LF MODULES CAN BE USED AS TUNED DIPOLAR ARRAYS, ... PROCESSING CAN PROVIDE EVEN GREATER LF CONTROL.

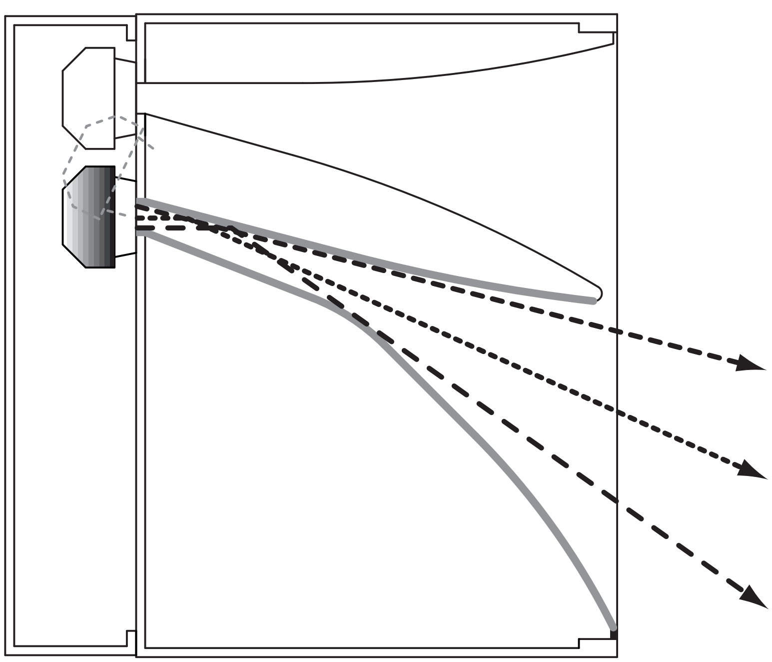


PPST PROCESSING PROVIDES OPTIMUM CONTROL OF KF930 LF MODULES.

vanced PPST program, which is a 32-bit, multithreaded, Visual C++ application.

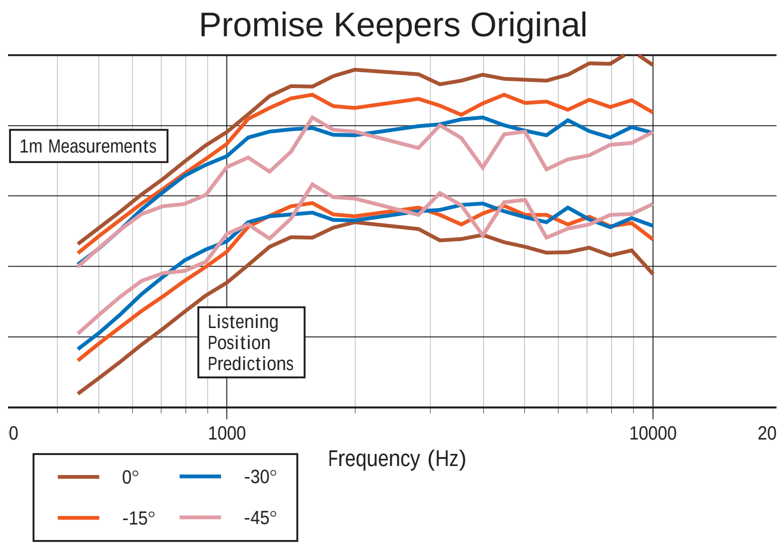
Along with the prediction and optimization program, the measurement process described earlier (in the PPST section) had to be developed. Finally, all of the required elements were brought together and a full PPST high frequency model was developed. After several days of adjusting processor settings for each of the seven high frequency channels, a setting was found which successfully integrated the KF910 and KF911 in adjacent position. In fact, the model predicted a remarkably smooth transition from the long-throw response to the downfill response. If the processing scheme proved to be aesthetically effective, the cabinets would not have to be split, and PPST would have produced a revolutionary result - smoothly combining seven individual horns into a cohesive whole.

REFLECTING THE LOWER DRIVER'S OUTPUT OFF THE UPPER WALL OF THE HORN, CREATES A “VIRTUAL” DRIVER INSIDE THE UPPER HORN.



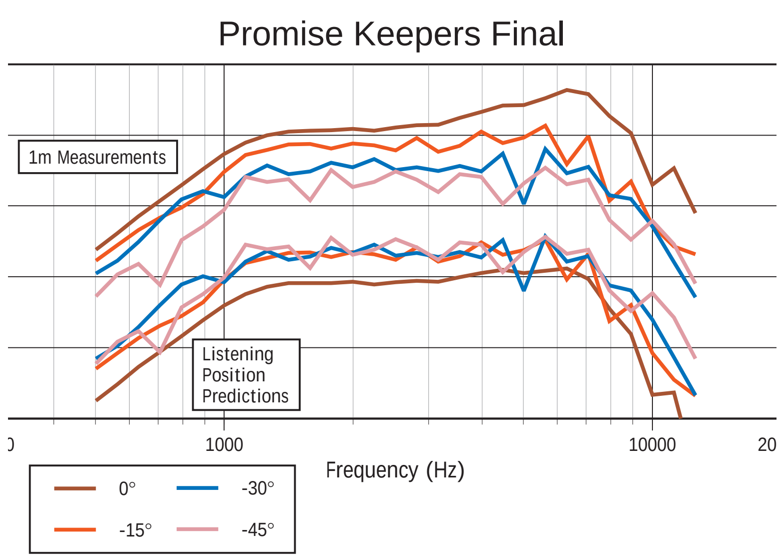
Aesthetic testing and refinement was carried out in a series of real-world settings, as a KF900 system had been contracted for 11 dates of The Promise Keepers, a religious conference series featuring high-level, live music held in stadiums throughout the United States, for audiences of 25,000 to 60,000 attendees, each. Before the first event, the system underwent preliminary testing in a mid-sized arena in Nashville, Tennessee. Then, it was on to the first date in the Detroit Silverdome.

Over the course of eleven Promise Keepers Conferences, both the target characteristic for the high frequency directionality, and the means of attaining it evolved. The air absorption pre-emphasis was increased, and extended to angles further below horizontal. The downfill high-frequency target response was changed from nominally flat to a more natural-sounding setting with the top end rolled off. The processing scheme was simplified and adjusted to rely more on delays, less on low-pass filters. This resulted in a smoother, less fatiguing high end. Measurement and equalization techniques were also refined, as the SIMM measurement system seemed to give unreliable results above 5 kHz, due to its non-adjustable high frequency window.



THE ORIGINAL PROCESSING FOR THE PROMISE KEEPERS ARRAY (ABOVE) PROVIDED ACCEPTABLE RESPONSE ACROSS THE LISTENING AREA. ADVANCES IN PPST PROCESSING OVER THE SUMMER (BELOW) IMPROVED THE ON-AXIS REPSONSE AT 400 FT. (BROWN LINE, LOWER SET OF CURVES), EXTENDING HF RESPONSE BEYOND 5 KHZ.

Ultimately, the PPST high-frequency process exceeded all expectations, on both the technical and aesthetic fronts. The system provided perfect intelligibility and startling intimacy at distances of up to 600 ft. Frequent SPL peaks of 118 dB were measured at 450 ft, during the musical sets. The spectral balance was constant from the nearest seat to the farthest, and the SPL did not vary more than +/- 3 dB at any point in the house.



VI MX9000 Dedicated PPST Array Function Generator

The Optimized Array Function

Based on the precise measurements of each KF900 module subsystem, the PPST modeling/optimization process calculates a unique transfer function for each of those array elements. The aggregate of all these transfer functions could be considered the "optimized array function" which takes into account both the array configuration as well as the venue in which that array will operate.

DSP Requirements

Each individual transfer function is quite complex, incorporating delay, multiple bands of parametric equalization, high pass and low pass filters with slopes of up to 48 dB/octave, gain and compression/limiting. A PPST array requires 120 or more of these functions, simultaneously. Clearly, this level of processing can only be achieved in the digital domain.

For any given digital signal processor, the more operations it can perform in a given sample time, the more functionality it will have. Generating the optimized array function for a modest KF900/PPST array requires a chip (or chips) that can execute more than 400 million operations per second. Development and testing of the KF900 Series modules and array was carried out using racks of MX8600 Close Coupled Digital Processors, but the actual process of system set up and installation made it clear that a new level of DSP power was required if the performance advantages of PPST were to become a practical reality.

The goal, then, was to develop an optimized array function generator with enough horsepower to control a KF900 array and put it into a user-friendly package with an intuitive, mouse-driven interface.

MX9000: Enhanced Functionality

The MX9000 PPST Workstation has been designed to generate highly complex optimized transfer functions for large arrays.

These functions can then be reliably and repeatedly applied to the array elements, allowing the array to deliver predictable performance that accurately reflects the predictive model on which the functions were based.

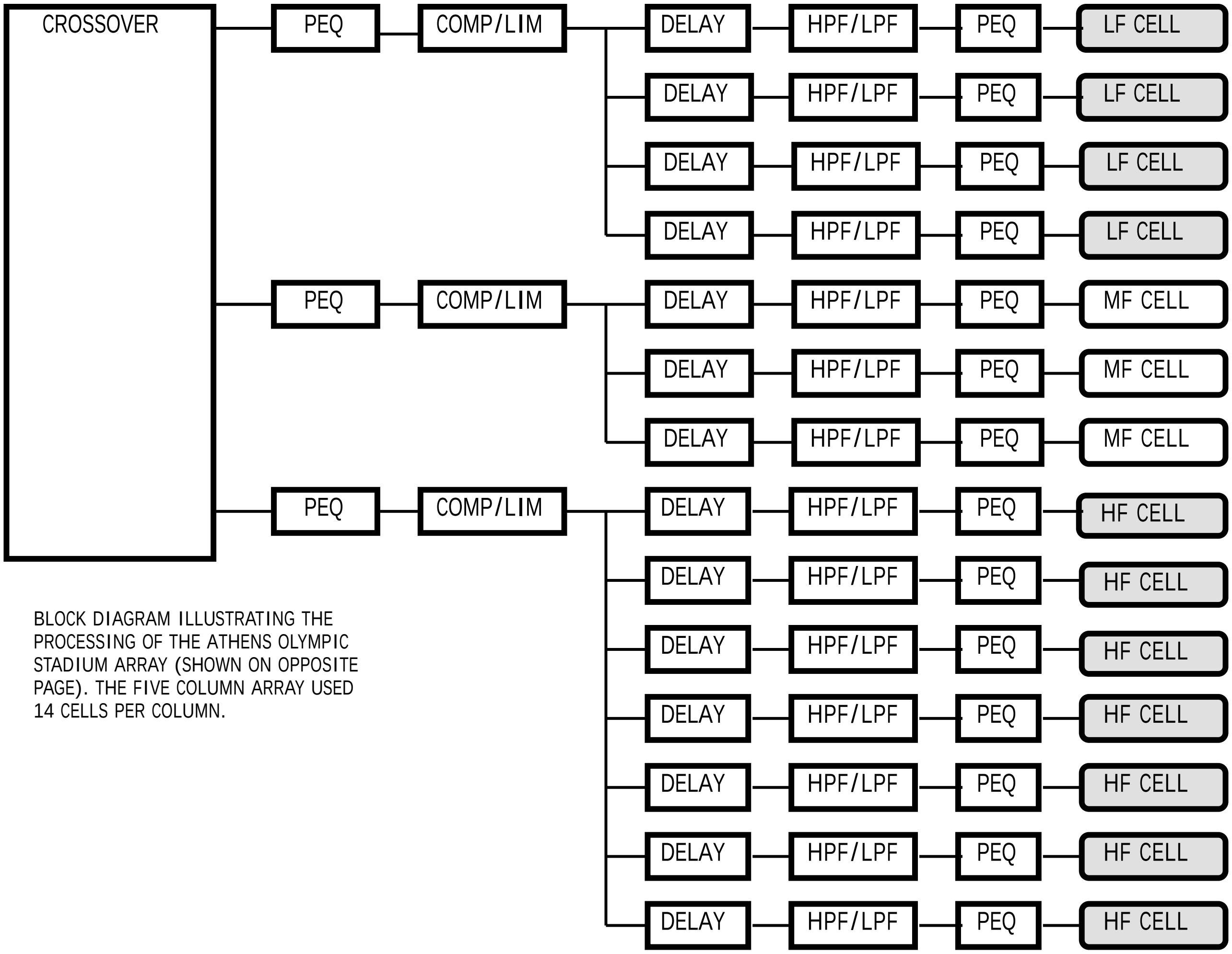
By generating the complete optimized array function, the MX9000 makes system setup much simpler and faster. Issues that complicate temporary sound reinforcement setup such as delay configuration, speaker limiting and processing, and the extensive system EQ associated with these arrays can be virtually eliminated. The MX9000 will save a “library” of optimized array functions that the user can access on demand. Provided that signal routing from the amplifiers to the speakers is correct, the system can be EQ’d to suit the users aesthetic taste.

But additional capabilities will unleash yet more of the power of PPST in practical applications. The MX9000 is designed to integrate real time control of parametric “house EQ,” limiting and “bulk” delay functions that are typically adjusted by the system operator to satisfy aesthetic considerations. To assist the adjustment process, the MX9000 will measure the system’s electrical performance and compare it in real time with the acoustical performance. This information will be displayed on the same screen as the user-controlled real time parametric equalizer.

Planned software enhancements to the MX9000 PPST Workstation include upgraded measurement capabilities, as well as a PPST modeling program that can use summed measurement data to simulate array configurations, model acoustical performance, and calculate an optimized array function. “Goal-seeking” EQ and optimization are also on the drawing board, along with amplifier and driver monitoring, on-screen trouble-shooting and performance reporting capabilities.

A Next Generation DSP Engine

The MX9000’s present and planned future capabilities depend on a significant amount of DSP power. Fortunately, recent advances in technology have made that power not only available, but affordable. The basic DSP engine used by the MX9000



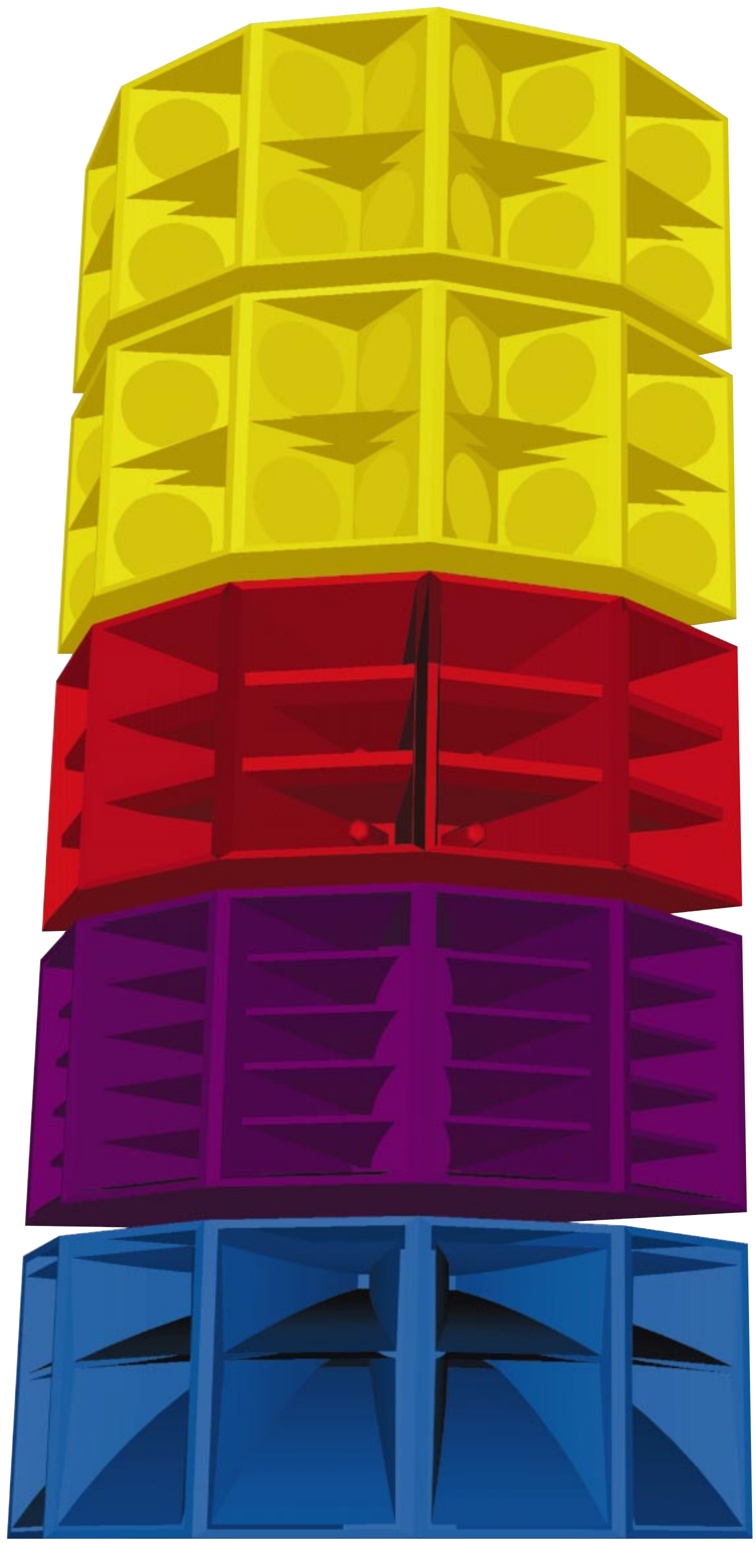
BLOCK DIAGRAM ILLUSTRATING THE PROCESSING OF THE ATHENS OLYMPIC STADIUM ARRAY (SHOWN ON OPPOSITE PAGE). THE FIVE COLUMN ARRAY USED 14 CELLS PER COLUMN.

is a recent entry from Analog Devices, the SHARC processor. This high speed 32-bit floating point chip can perform billions of operations per second. In addition, it contains high speed on-chip memory (important for fast filter calculations) and is easy to use in multi-processor applications. After a comprehensive review of existing DSP topologies, the SHARC was chosen for its functionality, speed, and overall performance. This is clearly a next generation audio DSP device, and one that will not quickly become obsolete.

Optimized Array Function Generation

Both the optimized array function and the real time, operator-controlled parameters, are generated by proprietary (patent pending) digital algorithms which implement the PPST-derived optimized array function for a KF900 Series system. The block diagram illustrates basic functionality:

Certainly other signal processors (Peavey MediaMatrix, for instance) are capable of this level of processing. However, only the MX9000 has sufficient power to consolidate processing, modeling and measurement in one unit.



The MX9000 PPST Workstation: Hardware

Initially the MX9000 was conceived as a machine to control and manipulate large arrays and facilitate proper show operation. Very early in the design process, we realized that the complexities of calculating and generating the optimized array functions and pattern control beam steering would be best performed in the background with little user assistance. But we also wanted a full-function real-world live-show machine with the ability to measure system performance in both the acoustic and electrical realms, allowing the user to gauge performance against a simulated model and adjust it in real time.

To perform this complex set of tasks, the MX9000 uses a Pentium based platform for display and 'housekeeping' functions, coupled with a number of SHARC processors. The minimum number of SHARC's required for a basic array is four, three for audio processing and one for measurement and display manipula-

GARY HARDESTY (LEFT) AND DENIS LAMBERT HAVE WORKED COLLABORATIVELY ON THE DEVELOPMENT OF THE NEW MX9000.

tion. Adding a second channel (stereo arrays, for example) requires the addition of another DSP card with three more SHARC processors on board.

The complete workstation includes a computer display, keyboard, and mouse to produce a functional, show-ready machine. Initial releases of the hardware/software allow the MX9000 to act as an overall system controller, replacing all front-of-house EQ, delay and speaker limiting and processing. With an MX9000, the signal path can run directly from the console to the MX9000 and on to the amplifier systems.

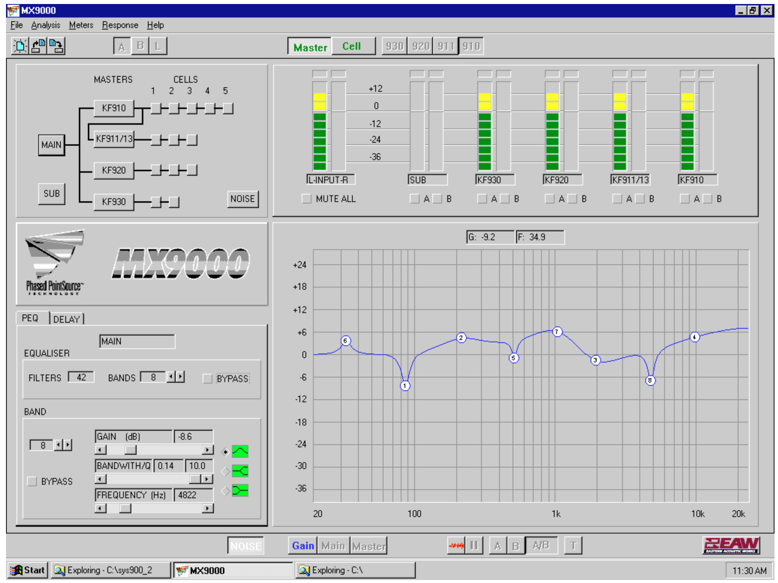
Digital Audio Transmission via CobraNet

Perhaps the greatest advance embodied in the MX9000 is its ability to transmit digital audio and communication via the Peak Audio CobraNet Ethernet interface. This hardware/software solution takes the non-deterministic, packet-based, Ethernet protocol which cannot support live audio routing and makes it audio-friendly by allowing the packets to be routed deterministically. Standard Ethernet hardware can them be used from the console outputs all the way to the amplifier inputs. This opens the door to the creation of exceptionally large sound systems spread over great physical distances that can be integrated into a single, easy to use network.

At this time, EAW recommends the use of QSC RAVE CobraNet devices for conversion from analog to digital and back. These devices will act as the converter system, transferring digital audio over a simple fiber optic based Ethernet system (the MX9000 has a built-in fiber optic interface). The advantages of this signal transmission system include:

- Dozens of audio channels being carried over a single pair of fiber optic cables
- The complete elimination of interference or ground loops between FOH and stage amplifier systems

- Add/drop capability for the network (simplifies cable systems to amplifiers as well as other systems that may communicate over the network in the future)
- Communication back to the MX9000 is possible for future applications and products



THE MX9000'S USER-FRIENDLY INTERFACE SIMPLIFIES THE PROCESS OF MONITORING ARRAY ELEMENTS.

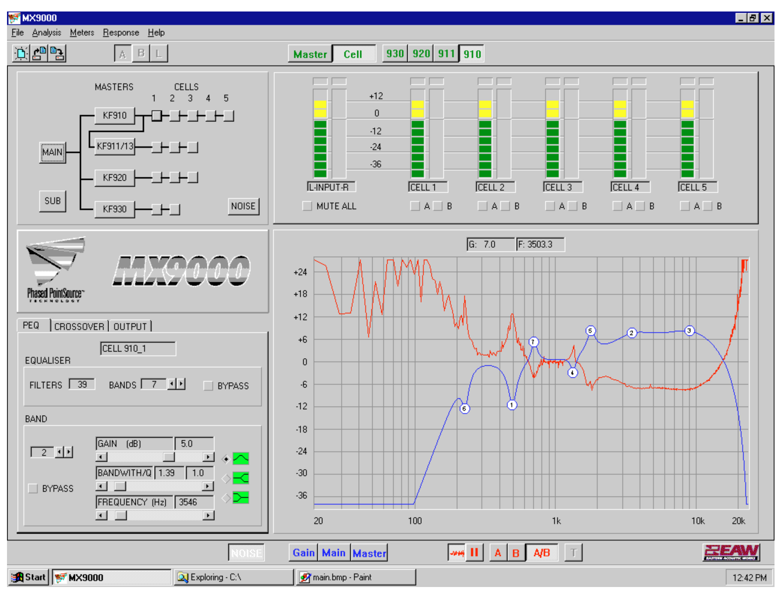
User Interface

The user interface combines an analog look and feel with a fully functional dual FFT measurement system. The MX9000 displays both the real FFT trace and the user controlled frequency response on screen, at the same time. With this functionality, the user can easily measure and EQ the system, either to a pre-set 'goal', or to personal preference.

Planned Developments

With the release of version 1.0 software early net year, the user will be able to:

- Recall optimized array functions for a variety of KF900 Series array configurations
- Manipulate and change overall system EQ, limiting and bulk delay in real-time
- Compare measured electrical performance against measured acoustic performance. This information will be displayed on the same graph as the user controller parametric EQ curve.



Software release 2.0 will include:

- An upgraded real-time measurement system
- An integrated modeling program for simulating array configurations, accurately predicting acoustical performance, and calculating the optimized transfer function
- ‘Goal-seeking’ EQ and optimization

Future software releases are already under development, they include:

- System amplifier and driver monitoring, with on-screen troubleshooting and performance report
- System modeling and optimization, based on ‘what-if’ scenarios

Enhanced Performance for Large Arrays

Although it was designed initially as a part of the KF900 Series system, the DSP power of the MX9000 PPST Workstation can be integrated with any large array. With properly designed signal flow and amplification, it can deliver substantial performance improvements relative to previous generations of simple “loudspeaker processors.”

As this paper has pointed out, the full benefit of Phased PointSource Technology is derived from a highly precise measurement and summation system, sophisticated modeling and optimization software, advanced DSP control and specialized array elements. Conventional full range arrayable enclosures do not allow as much control over total system performance: an array function can be calculated, but it cannot be optimized to the extent possible when the array is composed of specialized modules. Moreover, delivering the appropriate transfer functions to the correct enclosures within the array requires a major rethinking of signal flow from FOH through amp racks and speaker cables. However, significant performance benefits should result from the application of PPST concepts to conventional arrays of full range enclosures. A parallel development effort is under way to collect the required measurement data from which optimized transfer functions for arrays of various sizes can be calculated.

Conclusions

By combining everything EAW has learned about optimizing large loudspeaker arrays into a single project, the PPST/KF900/ MX9000 represents a genuine breakthrough in sound reinforcement technology. Despite its seeming impossibility, a loudspeaker technology producing a coherent impulse that sounds both intelligible and musical at 600 ft has been created. The effects of air absorption have been reduced to produce impulsive high frequency response at great distances without producing dangerously high SPL's in the nearfield.

New transducers have been created capable of producing the necessary SPL. New SimplePhase horns specifically designed to integrate and support the PPST process have been developed. And, most importantly, a method for accurately predicting the performance of various frequency-specific array modules in various configurations has been proven to be accurate. This allows EAW to create signal processing configurations for specific arrays that will produce coherent, musical, intelligible sound impulses across great distance with even level and flat frequency response.

Finally, by creating a dedicated digital array control system (MX9000) EAW has given audio professionals the tools to create and implement loudspeaker arrays that can satisfy the needs of sound reinforcement into the 21st century.

1 Beranek, Leo L.; Acoustics; pp. 307-311; McGraw-Hill; 1954

2 Olson, Harry F.; Elements of Acoustical Engineering; pp. 34-37; Van Nostrand, 1947

3 Ibid.