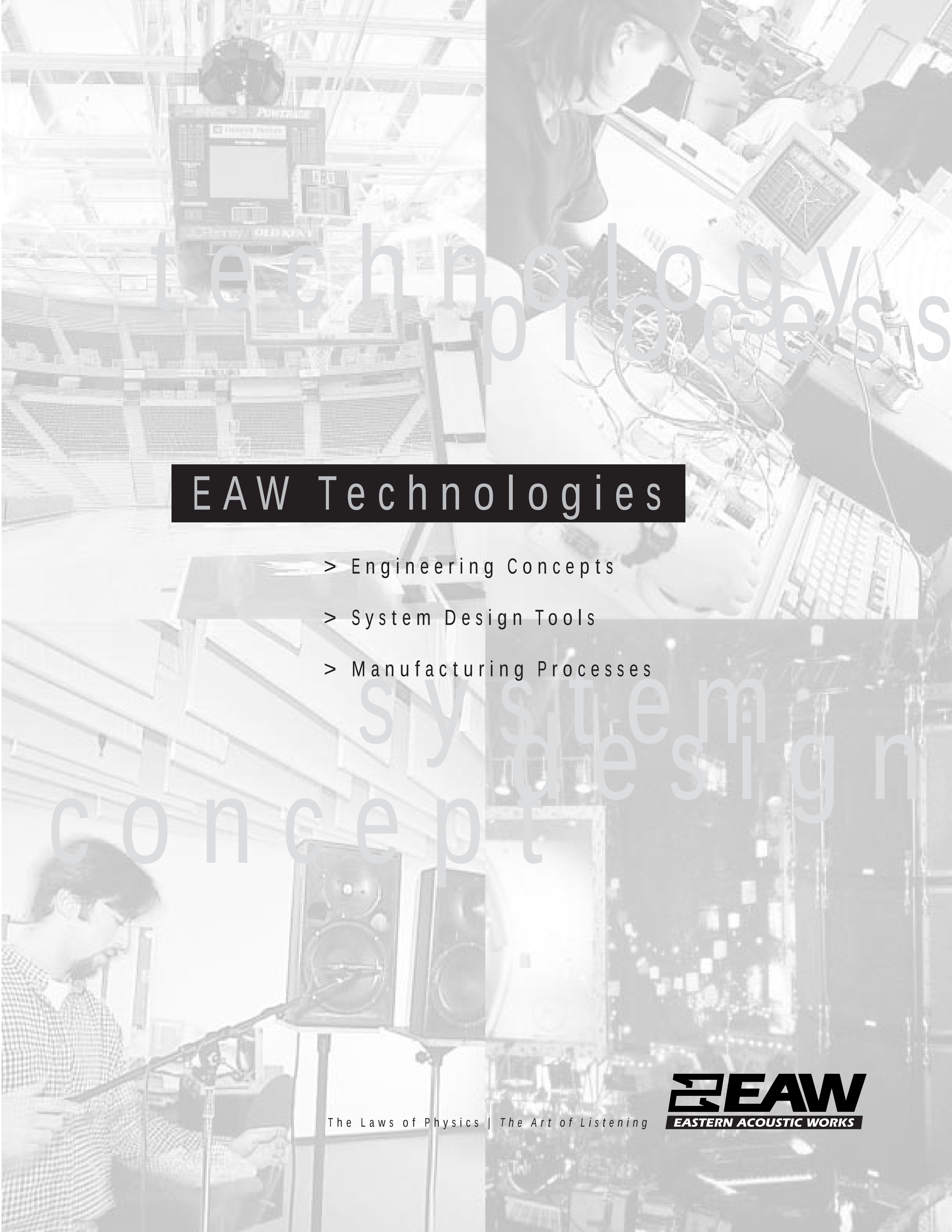

EAW Technologies

> Engineering Concepts
> System Design Tools
> Manufacturing Processes

The Laws of Physics | *The Art of Listening*

EASTERN ACOUSTIC WORKS




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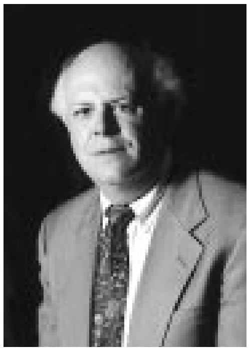
I – EAW Engineering



EAW co-founder
and Executive VP,
Product Development
Kenton Forsythe.

Introduction

Throughout more than two decades of innovation, EAW co-founder and Executive Vice President, Strategic Product Development Kenton G. Forsythe has integrated the laws of physics with the latest audio technology to design loudspeaker systems that give audio professionals the most accurate tools possible.

Over the past two decades, Kenton and the team of engineers he has built at EAW have developed several new approaches to loudspeaker system design. Yet advanced technology is never more than a means to an end - loudspeaker systems that deliver unparalleled performance.

While the results are often revolutionary, EAW engineering is an evolutionary process that proceeds from the fundamentals of acoustics. Listening to customers and responding to the specialized needs of the professional audio community is a key in applying innovative technology to the development of products that lead the industry in new directions.

The EAW Engineering Team

EAW's engineering direction remains under the guidance of Kenton Forsythe, but most of the department's day-to-day operation are overseen by VP, Engineering Stephen Siegel. Originally the ASG Manager, Stephen has played a key role in the creation and execution of many notable sound reinforcement projects including the PPST™/KF900 installation at Athens Olympic Stadium.

Long before products reach market, design engineers work with market managers to define the performance and utility parameters the product needs to meet.

Virtually all of the actual design work is carried out by Senior Design Engineers David Gunness and Jeffrey Rocha. The Senior Design Engineers are assisted by a staff of technicians specializing in system testing and measurement, mechanical engineering, manufacturing engineering, CAD and other support roles.

At EAW, we regard accurate, reliable engineering data as an essential element of the loudspeaker system. Without this information, it is difficult to use any sound reinforcement device properly. Our CAD Documentation and Quality Control teams are charged with making sure that our products have accurate and complete specifications.

Total System Engineering

To achieve the right balance of performance factors for each particular application, every aspect of the loudspeaker system - transducers, enclosures, horns, crossover design and manufacturing, exterior finishes, electronic signal processing - is optimized and integrated into a functional whole. By creating an engineering ethic that emphasizes total system performance rather than any one component or technology, Kenton Forsythe has redefined the standard of excellence as it applies to professional loudspeaker systems.

While pursuing overall performance, EAW engineers have developed many important design approaches and technologies: What follows are brief explanations of several of the most important design tools in use today at EAW.

II – Virtual Array™ and VA⁴ Technology™

Virtual Array Design Concepts

At the core of Virtual Array Technology™ is a simple realization: a single loudspeaker enclosure rarely provides adequate coverage and output for professional applications. So how well multiple systems work together is at least as important as how a single system behaves.

Simply designing an enclosure with trapezoidal sides does not make the system arrayable. Conventional loudspeakers designed for stand-alone performance array poorly, creating coverage gaps and areas of comb filtering that degrade audio quality.

The Horn-Loaded, Three-Way System Solution

Virtual Array Technology employs advanced horn-loading techniques to achieve broadband pattern control — the key to arrayability.

Since they were designed specifically to array, VA loudspeakers provide:

- optimized array coverage
- minimal interference between adjacent loudspeaker systems
- accurate, consistent, high-powered coverage
- steep roll-off beyond the defined coverage area.

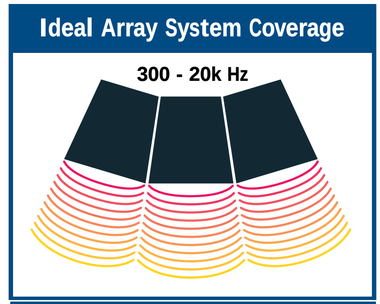
VA Technology loudspeakers are three-way systems built around Kenton Forsythe's midrange horn and displacement plug. Early in his career, Kenton realized that a minimum of three subsystems was required to achieve realistic reproduction of either speech or music. He was most concerned with the frequency range 250 Hz to 3k Hz - the human vocal range.

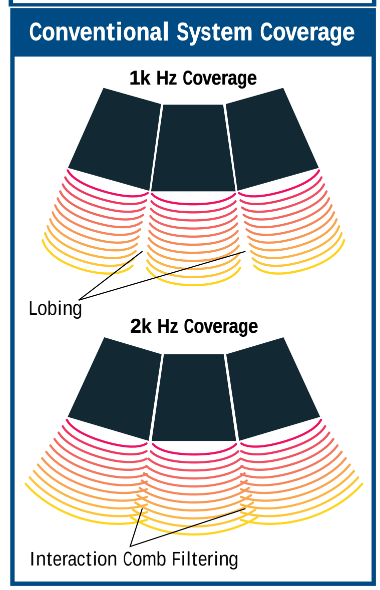
Dedicating a subsystem to the midrange solves many of the problems associated with two-way designs that place the crossover in the vocal region. Dividing the vocal range between creates power response gaps (holes in off-axis frequency response) as well as excessive distortion in just the frequency range where the ear is most sensitive to acoustical anomalies of any kind.

By dedicating a horn-loaded subsystem to cover the midband, power response problems in the crossover area are minimized. The large horn throat allows the wavefront to develop without distortion yet be efficiently directed to the defined coverage area. In addition, the horn serves as an impedance-matching device, increasing the driver's efficiency by an order of magnitude.

KF850/Stadium Array Series: The World Touring Standard

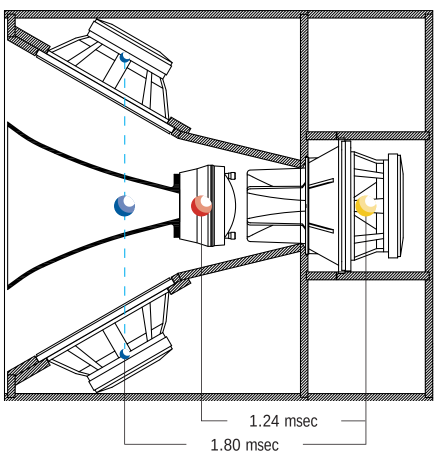
Built around the KF850 – the most accepted touring loudspeaker on the planet - the Stadium Array Series provides a variety of specialized loudspeaker systems that create custom-configured touring arrays.



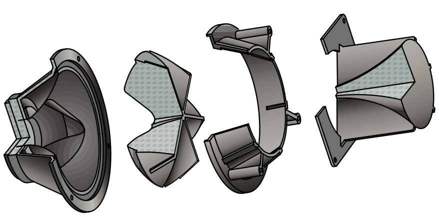


Virtual Array Technology provides optimized arrayability (above) through the use of sophisticated manufacturing techniques that let us develop mid frequency horns with complex, mathematically correct flares.

This KF850 Series array uses each member of the KF850 family of loudspeaker systems. KF853's and BH853's provide long throw, KF850's provide general coverage and KF855's provide downfill coverage.



The KF750's Acoustic Singularity design makes it the first three-way array module that acts as a single point source.



HOW SOON IS NOW? Until now, all mid frequency cone drivers created temporal smearing of critical vocal intelligibility information because the energy from the cone, dustcap and surround arrived at different times. EAW engineers developed the cone and the new generation phase plug as a single unit that creates a unified arrival of all mid frequency energy at the throat of the horn.

The Series includes:

- KF850 Full Range System
- KF853/BH853 Long Throw Systems
- KF855 Downfill System
- SB1000e Subwoofer System

Road-worthy construction and efficient truck packing are essential qualities for touring loudspeakers. That means making the systems as compact and sturdy as possible.

The KF850 coaxial three-way design places the HF driver and horn in the LF cone's Wave Guide Cavity to minimize the enclosure volume. Acoustic Refraction Control prevents interaction between the two subsystems.

EAW builds touring loudspeakers exclusively from 15mm cross-grain laminated 18-ply-to-the-inch Baltic birch plywood. This enclosure material is then coated with proprietary urethanes to achieve a hard, durable finish that has survived up to a decade on the road.

Finally, we ergonomically engineered the systems for easy load-in. Easy-to-use fly-tracks accept industry standard clips. Handles are placed on centers of gravity and only heavy-duty casters are specified.

VA^4 Technology: Extending VA into the Fourth Dimension

In the 1980's the KF850's Virtual Array Technology significantly reduced interaction between array modules in the horizontal plane. Today, VA^4 Technology extends that same level of array control to the vertical plane.

And it goes one step beyond, solving problems in the fourth dimension: Time.

For the first time ever, the KF750's Acoustic Singularity™ design creates a unified space/time origin over the entire audible spectrum, eliminating the temporal smearing that has plagued all previous loudspeaker designs — even our own KF850 , the most accepted touring array module in the world. A broadband unified arrival enhances the clarity and impact of any sonic event.

KF750 Full Range Loudspeaker Array Module is essentially a very large mid frequency horn with the low- and high frequency subsystems mounted within its flare.

Since all three subsystems' acoustic centers are on the same axis, appropriate signal delay creates a unified space/time origin for the entire system making it the only three-way array system that acts as a true point source.

While the Dipolar Array LF section's dual woofers are positioned above and below the mid/high axis, their acoustic center is exactly on that axis. Since all acoustic centers are on the same line, simple signal delay makes the entire broadband audio image originate from one specific location at one specific point in time.

Acoustic Singularity design makes this the first three-way concert touring array module that acts as a true point source.

IT'S ABOUT TIME

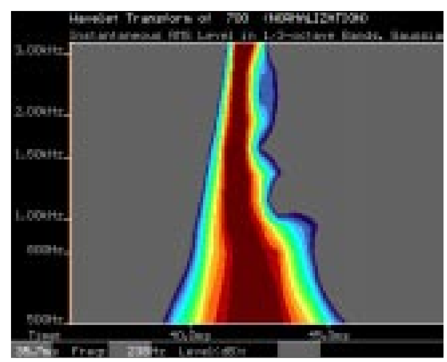
Unlike the relatively simple geometry of a compression driver's diaphragm, there is a slight but noticeable difference in the point of origin of a cone driver's dustcap, cone, and surround. Particularly in the upper midrange, these differences create a "smearing" of arrival times at the listener that degrades the clarity and impact of mid-frequency sonic events: most notably vocal reproduction. Because they are what the ear hears first, early arrivals out of the passband can affect overall fidelity even though they are substantially lower in level.

The distance from a cone driver's voice coil to its dustcap is shorter than the distance from the voice coil to either the cone or surround. Therefore, the energy radiating from the dustcap most often leads the energy from the rest of the system. Traditional phase plug designs have isolated this energy and routed it through a longer path than that which faces the energy from the cone or surround. In so doing, the phase plug attempts to equalize the arrival smear.

Cutaway views of the KF750 and KF755.

Conventional phase plug designs use a circular entrance and exit to the phase plug, converting the output from a point source into a ring radiator. A ring radiator, however, exhibits a more dramatic narrowing of beamwidth with increasing frequency than a cone transducer. Thus the mid frequency device's directivity narrows to the point where it no longer fills the bell of the horn, leaving upper mid holes in the frequency response on the seams of an array.

The KF750 mid/phase plug assembly attacks the problem at the source, precisely aligning the cone/dustcap/surround geometry to maintain temporal unity. The phase plug, whose geometry is matched to the cone, then serves to leave this unity intact. Expanding radial slots within a compressing frame lower the mechanical reactance of the load facing the transducer without modifying the directivity associated with the source. This allows for faithful reproduction of upper mid frequencies without any narrowing of beamwidth.



This wavelet image of a KF750's mid frequency cone shows the uniform arrival of acoustic energy over a very wide passband.

The example above of a KF750's response shows a smooth, steep, tightly-packed slope through the entire measured range clearly which indicates the unified arrivals of all sonic events.

KF700 Series: VA[4] for General Purpose Applications

KF750

The KF750 is the high "Q" full range loudspeaker system that serves as the primary array building block. Even with its exceptional long throw capabilities, its smooth power response and coherent tonal quality allow it to work in nearfield applications as well.

At its core, a KF750 is a very large mid frequency horn. Allowing the horn to fill nearly the entire front of the enclosure enhances pattern control in the lower midrange, minimizing interaction between adjacent modules.

A KF750 array.

The 10-in mid frequency cone transducer and phase plug assembly was developed specifically to eliminate the temporal smearing that degrades vocal intelligibility in all other array systems. It is so unique that US and international patents are pending.

The high- and low-frequency subsystems fit within the mid horn, minimizing both size and weight.

KF755

The KF755 provides dedicated downfill coverage in array configurations where trim height requires additional nearfield output. The KF755 can be inverted to provide upfill to auditorium balconies and can even be used as a stand-alone system in applications requiring that coverage angle. In addition to the single 12-in woofer, it contains the same mid and high frequency transducers as the KF750 loaded on vertically asymmetrical horns.

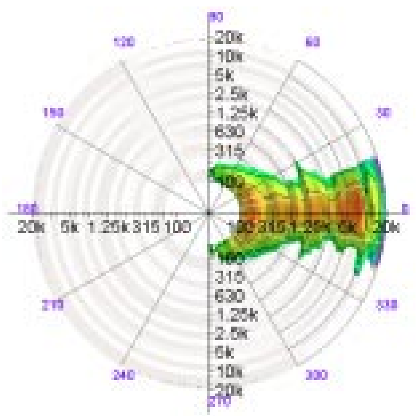
The MQ Series: VA^4 for Permanent Installation

EAW's MQ Series of mid/high VA^4 modules is engineered solely for optimized performance in permanent installations. Portability played no part in its design.

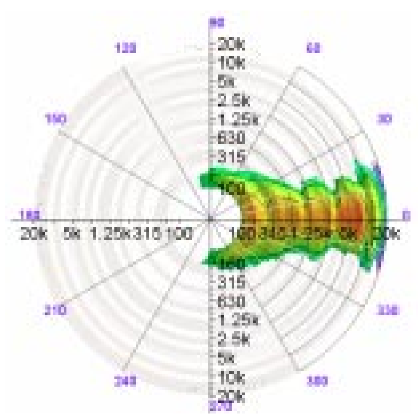
By allowing the mid- and high-frequency horns to expand to their optimal sizes achieved two important goals of VA^4 Technology:

- expanded the frequency range of the system's pattern control
- smoothed power response to a lower frequency

The MQ Series uses the patented VA^4 10-in mid frequency cone/radial phase plug assembly. This design uses a more logical cone geometry than previous designs to achieve a significantly more unified arrival over the upper portion of the mid frequency band.



These topographical response images demonstrate the KF750's exceptional pattern control in both the horizontal and vertical planes.



Low frequency output is provided by dedicated MQ Series LF modules or TD Series Tuned Dipolar Array™ systems. Sound system designers use these systems to create precision arrays for arenas, stadiums, theme parks or any large space where oratory or musical performance is given.

KF860/Virtual Line Array™

Compared to traditional horizontal arrays, a KF860/KF861 Virtual Line Array (VLA) offer the sound system designer a number of advantages when designing systems for large spaces:

- fewer enclosures provides adequate coverage
- reduced the signal to noise ratio
- reduced set up time
- enhanced broadband pattern control
- minimal "spill" onto the stage
- additional gain before feed-back

In addition to the advanced horn-loading techniques developed for Virtual Array loudspeakers, VLA's achieve unprecedented LF pattern control by employing Tuned Dipolar Array Technology (TDA). The woofers of each KF860 or KF861 VLA module are separated in the horizontal plane to create off-axis rejection over much of the LF range. A beneficial side effect of TDA: an increase of on-axis response over the same frequency range.

In the vertical plane, all similar subsystems in adjacent modules couple acoustically to create line arrays. Like the TDA effect, line arrays create off-axis rejection as well as additional axial output. Vertical coverage angle is determined by the height of an array and any applied signal processing. As a rule of thumb, the taller the array, the narrower the the vertical pattern. Appropriate digital signal processing will "open" a tall array's pattern to achieve the necessary coverage.

Television production crews were the first to embrace VLA technology for their large-scale televised live events. The low profile arrays did not limit the camera angles they could use and the broadband pattern control minimized "spill" from the sound system into the broad-cast audio feed.

KF860 arrays used for the Juno Awards (Canadian music industry event).

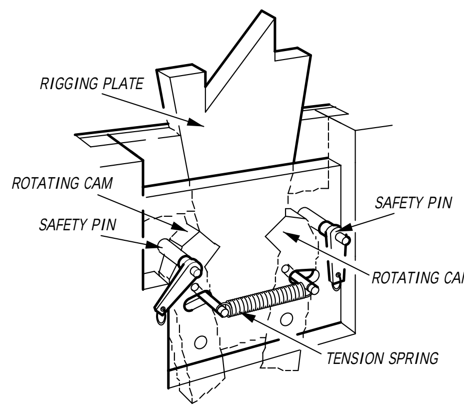
Better Mechanics, Less Elbow Grease

EAW's new KF860/KF861 rigging system substantially reduces the time associated with hanging arrays without sacrificing the accuracy or safety of the original steel-to-steel design.

Coded holes on both the rigging plates and the cabinet end brackets speed the selection of the six angles at which VLA modules can be arrayed.

Angle selection is accomplished when setting the rigging plate into the bottom end bracket of a module already hanging. Simply line up the clearly marked holes on the rigging plate with the appropriate holes on the bottom cabinet end bracket and insert the quick release pins.

The hanging module and rigging plate are then lowered onto the next module. The rigging plate is held in place with spring-loaded cam which are locked in place with quick release pins. The system includes a customized cart which allows each module to be pre-positioned at its intended array angle. The need for human muscle-power to align the modules has been all but eliminated from the arraying process.

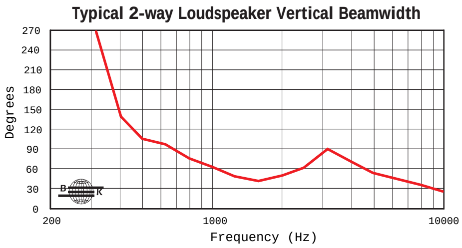


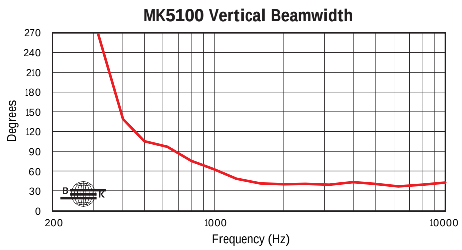
The ergonomics of a load-in helped define the requirements the KF860's rigging system needed to meet.

III – Horns and Waveguides

Horns and Waveguide: A brief primer

In the early days of sound reinforcement, amplifiers supplied only a few Watts of power and loudspeakers were not very loud at all. But like cupping your hands to your mouth to shout, an audio horn increases the amount of air a driver can move by matching its impedance to a larger surface area. So horn-loading was developed to increase the output of drivers and became a required part of every loudspeaker system.





Note that the beamwidth discontinuity in the crossover region of the traditional two-way system (top) is absent in the MK5100 Series system.

Horns also provided another benefit — they helped direct the driver's sound to a particular area and prevented it from going to other areas. This ability to control a loudspeaker's dispersion pattern is the reason for most modern horn-loading.

In the 1960s, more powerful tube and solid state amplifiers made the efficiency-improving benefits of horn-loading became less important than its directivity enhancement. It was at this time that the distinction between horns (impedance matchers) and waveguides (sound directors) was first made. The distinction is more semantic than practical; either will, to a certain extent, achieve both aims.

Even though modern amplifiers can power cone and compression drivers to very high output levels, increasing a driver's efficiency through horn-loading reduces strain and minimizes distortion. At the high output levels demanded by today's professional loudspeaker, harmonic distortion remains a problem for poorly designed systems.

EAW's advanced horn-loading techniques maximize both the directivity-enhancing and efficiency-improving aspects of this basic loudspeaker design tool.

Constant Directivity Horns

The earliest horns — designed strictly to increase output — suffered from poor audio quality and uneven coverage. In addition to creating substantial amounts of harmonic distortion, frequencies in the upper third of most horn's range never "saw the sides of the horn." Since they never filled the horn, these higher frequencies beamed straight ahead, creating uneven power response.

EAW Cinema loudspeaker systems use the latest constant directivity horn technology.

In response, John Gilliam and Don Keele developed the constant directivity flare which smoothed the directivity across the waveguide's entire range.

But this design only affected output in one plane. For a constant directivity horn to produce separate coverage patterns in the vertical and horizontal planes, the flare for the narrower of the two angles must start farther back, creating a throat section where the wider-angled plane's walls are straight for some period before flaring out rapidly. This is the design Cliff Henrickson and Mark Ureda developed.

As is so often the case, the solution created a new problem. Here, the long throat was resonant at lower frequencies, creating an acoustical anomaly commonly referred to as a "honk." For the typical high frequency horn, this honk appears in the 600 - 800 Hz range where the ear is most sensitive to acoustical anomalies of any kind. The result is speech which sounds as if it is coming through a megaphone, which in a sense it is.

Radical equalization using both passive filters and active filters proved ineffective in solving the problem, which is caused by cancellations in the throat section of the horn.

In recent years, EAW's David Gunness has developed CD horns with shorter throats and flares that follow complex mathematical functions, dramatically improving the naturalness of reproduction. Noteworthy among these is the HK259 high frequency horn developed for use in two- and three-way screen channel cinema loudspeakers. The improved naturalness of speech in CB259 two-way systems using the HK259 has made it increasingly popular for cinema applications.

Waveguides and the EAW Wave Guide Plate™ (WGP™)

High frequencies tend to "beam" straight ahead unless the HF driver is properly loaded with a horn. But constant directivity horns are better suited to longer throws and narrower coverage angles. As a result, most nearfield loudspeaker systems that used direct radiating tweeters were engineered for excellent axial response but sounded considerable worse off-axis and provided uneven power response.

The Elliptic Conical Waveguide.

EAW's Wave Guide Plates provide even dispersion of high frequency information over a wide area while providing enhanced efficiency and pattern control. This lets us use powerful 1- and 2-in exit compression drivers without the harsh "horn-loaded" sound that even the best-designed CD horns suffer from when heard in the immediate nearfield.

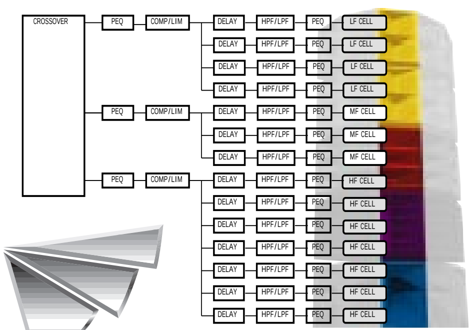
This is particularly important in nearfield applications such as small venue sound reinforcement or cinema surround channels where listeners might be either very close to a loudspeaker or well off to the side. WGP-loaded systems form the LA, JF or Cinema Series fill broad listening areas with HF information without taking up a lot of space.

For an aperture/flare type HF horn to achieve these wide dispersion patterns, it would have to be unmanageably large. EAW engineers discovered that the throat section could be removed entirely, placing the driver right at the mouth. Very shallow waveguides with "wide by wide" dispersion characteristics could then effectively handle frequencies up to and above 15k Hz.

The WGP is a very shallow waveguide which employs an axis-symmetrical, constant directivity flare to achieve a 100° x 100° dispersion pattern. EAW currently specifies the WGP for use on a wide variety of nearfield systems such as the MM and JF series. While other manufacturers have loaded dome tweeters with shallow waveguides, EAW is the first to load a high power 2-in compression driver with such a device.

Elliptic Conical Waveguide

For applications that require a wide asymmetrical dispersion pattern, such as small venue live performance, the Elliptic Conical Waveguide offers an optimized solution. Like its cousin, the WGP, the ECW is exceptionally shallow. It produces a 70° x 90° dispersion pattern. The ECW is currently used on the LA325 Performance Audio full-range system.



Phased PointSource™ TECHNOLOGY

The KF900 Series leverages the power of DSP to control each transducer in a massive array.

IV – New Loudspeaker Design Concepts

Phase PointSource Technology™

Phased PointSource Technology was developed to solve the essential problem of large-scale sound reinforcement: more drivers means more sound but fewer drivers means better sound.

The essential concept behind the development project was that by generating high resolution measurement data for each separate element of a loudspeaker array at a variety of listening locations, the signal path of each array element can be manipulated to maximize integration and minimize interference.

PPST, then, is the process of measuring, modeling and optimizing loudspeakers for use in specific configurations in specific venues.

A part of the development process was the creation of loudspeaker systems optimized to express PPST principals. These became the KF900 Series of frequency specific array modules.

The PPST process then creates specific DSP configurations for each driver/horn cell in any given KF900 array.

These DSP-controlled modules create arrays custom configured to specific venues ranging from moderately sized sheds to the largest stadiums in the world.

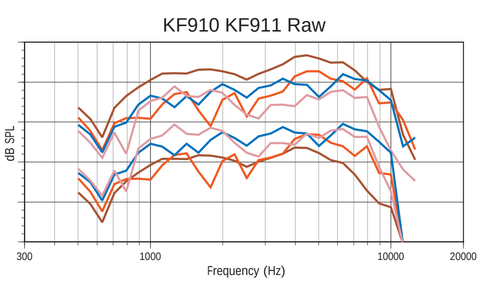
In the case of the largest venues, KF900 arrays can provide even level and frequency response for an audience area from just 25 feet below the array to over 700 feet away.

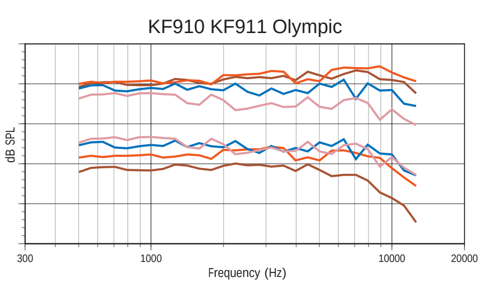
The PPST Process

The PPST measurement/modeling/optimization process creates specific DSP settings for each cell in the array. In the case of the array illustrated above, 14 different DSP channels are required to drive each column.

The process begins with a round of precision measurements of each cell of the loudspeaker array in the exact position the modules would occupy in an array. Measurements are accurate to 1/10th of an inch or 10 kHz.

Using microphones set at fixed positions around our test facility, phase and frequency measurements are taken on-axis as well as at 15, 30, 45, and 60 degrees below the horizontal axis.





Computer-driven optimization routines align the off-axis response of each array element to provide even coverage at all listening positions.

Here, the top graph shows HF response before e optimization and the bottom graph shows after. The colors represent different listening positions measured in 15° increments below the horizontal axis.

After the data are normalized to 1/6th octave resolution and "windowed" to isolate first arrivals, the results of the entire measurement round are fed into a proprietary goal seeking computer program called F-Chart. F-Chart sums the measurements of each cell to create a composite frequency response plot for the entire array and then manipulates delay and amplitude parameters for each cell to align response at each of the five measurement angles.

EAW engineers monitor F-Chart's progress – usually over the course of a few days – until they feel certain that total system response has been optimized and then fine tune the DSP settings for use in a specific venue. Using the venue's actual geometry and intended array location, the vertical beam profile is custom tailored.

Implementing PPST with KF900 Modules

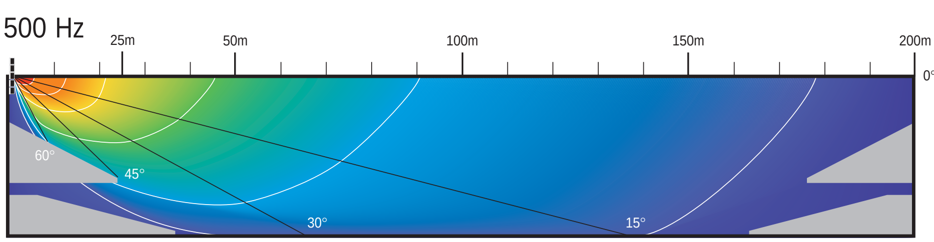
The KF900 Series comprises several frequency specific array modules each of which, in turn, comprise a number of "cells." The modules are KF910, KF911, KF913, KF920 and KF930.

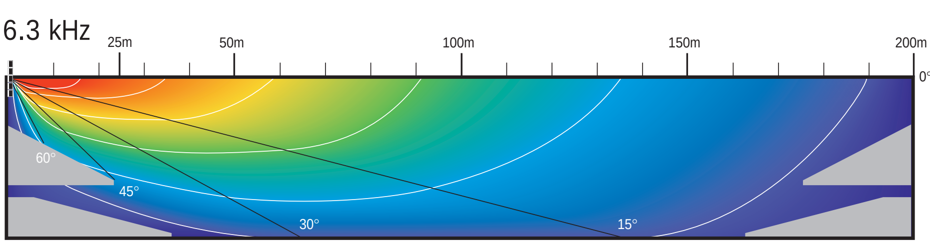
This array uses (from bottom) one KF911, one KF910, one KF920 and two KF930's per column.

Each cell of each module is measured and a separate processor setting is developed for each. The array used for Olympic Stadium at Athens, Greece illustrated at right required 14 distinct signal paths for each array column.

The goal of the optimization program, F-Chart, is to optimize frequency response at a range of specific listening positions in a specific venue with a specific array in a specific location.

These illustrations show predicted average dB SPL at a variety of locations along the central axis of the array (upper left) at Olympic Stadium. Each white arced line represents a 6 dB difference ranging from 97 dB SPL (far right) to 127 dB SPL in front of the array.

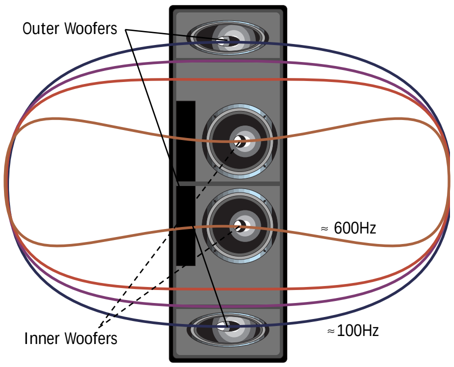




The black lines indicate the measurement angles of 0, 15, 30, 45 and 60 degrees below the horizontal axis. Distances were calculated to provide accurate representation of high frequency losses due to air absorption.

At both 500 Hz and 6.3 kHz, all listening positions receive sound at a level somewhere between 96 and 105 dB or ±4.5 dB. On-site adjustment at installation brought actual response to within ±2 dB.

PPST accepts the well-documented but poorly understood effect known as air absorption as an important parameter to address when designing long throw arrays. Using a technique we call high frequency pre-emphasis PPST arrays project HF information more than 700 feet. Even level and frequency response at distant listening positions results from uneven response at 1m. Note how much “hotter” it is right in front of the array at 6.3 kHz as opposed to 500 Hz.



The Tuned Dipolar Array™ uses well-documented driver interaction to create off-axis rejection over a broad 2 1/2 octave range. The colored shapes indicate approximate coverage pattern at various frequencies.

Tuned Dipolar Array Technology™

Because low frequency sound waves are so large – 22.6 feet @ 50 Hz – effectively controlling their dispersion has baffled audio engineers for decades. Attempts using very large horns or baffles have proven to be unmanageable for most applications.

Recently, however, EAW engineers have developed the Tuned Dipolar Array™ (TDA™), applying the well-documented cancellation effects of spaced drivers. Over a certain frequency range, the cancellation creates a desirable, controlled dispersion pattern along the axis of the drivers. Above a certain point, the axial pattern becomes too narrow and pronounced side lobes occur.

In brief, the farther apart drivers are spaced, the lower the frequency at which beneficial cancellation occurs. The dispersion pattern follow these trends according to the relationship between spacing (D) and wavelength (L):

Relationship	Pattern
D = 1/8 L	~360°
D = 1/4 L	~120°
D = 1/2 L	~60°
D = L	~20° with pronounced lobes @ 90°

The TDA uses this effect only in the range where it is helpful and then “gets out” before lobing becomes problematic. However, this limits the bandwidth over which low frequency drivers could operate to too small a range.

So EAW engineers developed the system of utilizing two pairs of drivers. The outer pair, spaced further apart, operate at a lower range while the inner pair cover a higher range. Given a properly developed crossover between the two pairs, the system demonstrates desirable, consistent off axis rejection over the entire low frequency range.

EAW implements this approach in the TD Series of Tuned Dipolar Array dedicated low frequency loudspeaker systems. The table below summarizes the components and frequency range of off-axis rejection.

System	Outer Drivers	Inner Drivers	-10 dB @ 90°
TD415	2x 15-in	2x 12-in	110 - 300 Hz
TD412	2x 12-in	2x 8-in	160 - 500 Hz

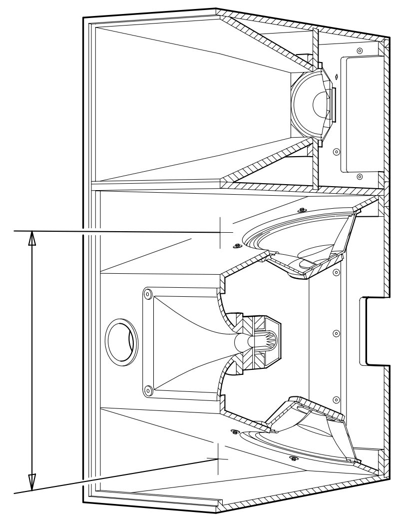
Dipolar Arrays in Full Range Systems

Tuned Dipolar Array Technology low frequency modules solve LF directivity problems for large format arrays, but EAW engineers also looked to apply TDA benefits to smaller format, single enclosure full range system.

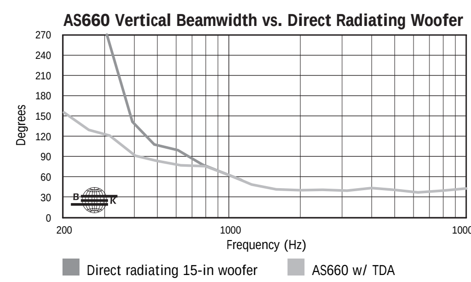
The goal was to combine the horn-loaded benefits of three-way Virtual Array systems with a TDA-type LF section. After testing a variety of prototypes, they arrived at the basic design illustrated at the bottom of the page.

To keep the enclosures manageable in size the system employs only a single pair of LF cones mounted on angled baffles behind the HF driver and horn with the midrange horn positioned above them. This design – which eventually became the AS660i – was contained in an enclosure just 36-in tall.

The LF array is referred to as dipolar – as opposed to tuned dipolar – because the constraints of the single enclosure limited the degree to which the spacing could be adjusted. As they are, the dual LF drivers provide optimal off-axis rejection around 300 Hz but provide quite usable rejection below 200 Hz.



The AS660i's TDA™ low frequency section creates off-axis rejection for better low and low-mid frequency pattern control compared to a direct radiating woofer.



Beamwidth Matching Crossover Design

A typical 2-way loudspeaker consisting of a 15-in woofer crossed over in the 1.5 to 2 kHz range to a compression driver loaded with a small format 90° x 40° constant directivity horn produces a beamwidth plot that first collapses and then immediately expands in the crossover region before ultimately flattening out. The cause is a mismatch of coverage angles at crossover: 90° from the woofer vs. 125° from the HF horn. The beamwidth plot below averages these angles through crossover.

Off-axis listeners will hear noticeably uneven frequency response. If this system were located anywhere near a reflective surface (an auditorium ceiling, for example), late arrivals would noticeably degrade sound quality for the entire venue.

Beamwidth Matched Crossover Points

To produce a 2-way loudspeaker with consistent coverage through and above crossover, EAW engineers studied the collapsing coverage pattern of the direct radiating woofer and then determined optimum crossover points for various coverage patterns.

By designing constant directivity high-frequency horns large enough to match the woofer's coverage at that crossover frequency, coverage through crossover is remarkably consistent. This approach eliminates the uneven off-axis response that plagues typical 2-way designs, producing highly accurate vertical pattern control, substantially reduced late arrivals and a dramatic improvement in fidelity.

Concentric Phase Aligned Array™

CP621 Concentric Phase Aligned Array

Typical coaxial systems consist of a large cone woofer surrounding a compression driver and a small horn flare. This approach creates many performance problems: boomy bass response, narrow midrange dispersion at the top of the woofer's range and excessively wide midrange dispersion at the bottom of the HF horn's range.

The result is uneven frequency response everywhere but directly beneath the loudspeaker. This is why most distributed paging systems using ceiling-mounted coaxial speakers are so ineffective.

CPAA Solves the Problem

The Concentric Phase Aligned Array takes the opposite approach: a large axis-symmetrical horn flare surrounds the low frequency section, which consists of a number of smaller cone transducers mounted within the flare.

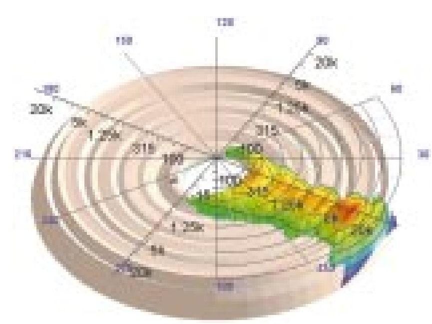
CP621

The larger horn mouth extends the range of consistent pattern control downward by an octave compared to ordinary coaxial horns. Meanwhile, the multiple woofers mounted around the circumference couple acoustically and provide the effective radiating surface of a 21" woofer, enhancing directivity in the lower octaves.

To smooth response in the crossover region, two additional woofers are mounted within the horn throat. They are phase aligned with the circumferential drivers to cancel reflections from the horn throat. Because their output is coupled to the horn, they have enhanced directivity and also serve to alleviate lobing near the crossover frequency.

The results as measured in EAW's automated test facility are clearly superior to conventional coaxial designs in consistent directivity over a wide bandwidth.

V – EAW Engineering Design Tools



EAW engineers create these topographical performance images using the same data set used to generate 1/3 octave polar plots. The ability to view the data of an full set of polar plots helps engineers identify problem areas in both the spectral and spatial domains.

Asymmetrical Crossover/Filter Design

Crossover design is critical to loudspeaker performance. But many loudspeaker manufacturers design crossovers within narrow guidelines of price or performance and expect the end user to equalize the system to compensate for the its built-in failures. This is not the EAW way. We don't expect system operators to make our speakers sound great; we design our loudspeakers to sound great right out of the box.

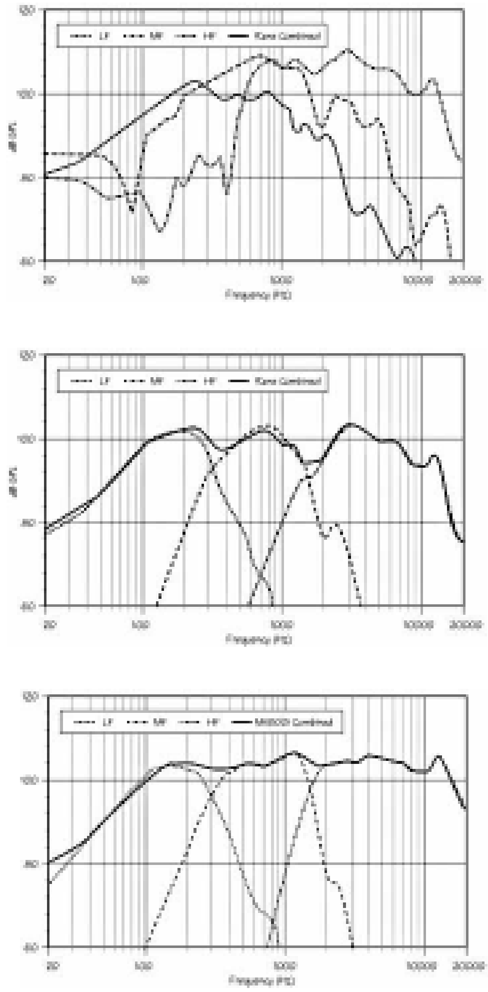
The Iterative Process

To do this, EAW has invested substantially in the human and technological resources necessary to create complex, asymmetrical passive crossover networks that optimize total system performance. Kenton Forsythe insists that his team of engineers use the time consuming, but superior, iterative process of development.

An iterative processes repeats a cycle of operations, beginning each new cycle with the results of the previous cycle. With each cycle (iteration) the actual result is brought closer and closer to the "ideal" or "model" result.

In the case of an EAW loudspeaker system, the ideal result is perfectly flat on-axis response and perfectly linear power response. Of course, perfection cannot be achieved, but it certainly can be approached.

First, the acoustical and electrical response of the raw drivers and enclosure is measured with a multichannel digital FFT analyzer, and the data are fed into proprietary computer software called Filter Designer. Based on the data obtained, EAW engineers build an "optimal" network. The system is measured again and the new data are fed back into Filter Designer. The network is refined, iteration after iteration, until total system performance is optimized.



Raw drivers in a three-way system sum incoherently (top). "Mathematically correct" crossover slopes provide some improvement (middle) but only complex, asymmetrical slopes provide optimal summation (bottom).

The Importance of Asymmetrical Crossover Slopes

Only asymmetrical crossover slopes can precisely match the characteristics of a specific driver through the crossover transition. Unfortunately, some manufacturers cut corners using mathematical abstractions to design filter networks.

To optimize power response, a system-specific crossover network utilizing asymmetrical slopes must be designed around the actual performance of the raw components and enclosure.

It should be noted that not only do we design around actual drivers, but we also take into account variations which occur during component production and use the iterative process to set limits on the variances we will accept. The result are loudspeaker systems that are consistent across production runs.

Close Coupling™

Close Coupled Processing is an extends EAW's passive crossover/filter design philosophy into the realm of active processing. Where internal passive networks control multiway systems that use only one amplifier channel, Close Coupled Processors (CCP's) control single bi- or triamped systems and integrate full range systems with subwoofers.

Close Coupled Processing applies the same iterative approach to generate unique, asymmetrical electronic filter sets/DSP configurations custom tailored (Close Coupled) to the actual response of the loudspeaker or loudspeakers being controlled.

Because electronic and digital processors are more flexible than passive networks, the level of integration CCP's achieve is even greater than what passive networks can provide. For this reason, CCP's are not interchangeable. Using a CCP to control a system other than that for which it was configured will degrade performance and could damage the loudspeakers.

EAW provides a variety of Close Coupled Electronic Processors and the MX8600 Close Coupled Digital Processor to control or integrate a variety of loudspeaker systems.

**MX8600 Close Coupled Digital Processor is appropriate for any application in which an enhanced level of control is desired.*

System(s) To Be Controlled/Integrated	Recommended CCP
Biamplified Loudspeaker System	MX200i/MX8600*
Triamplified Loudspeaker System	MX300i/MX8600*
Passive Loudspeaker Systems and Subwoofer	MX100/MX200i/MX8600*
Biamplified Loudspeaker System and Subwoofer	MX300i/MX8600*
Triamplified Loudspeaker System and Subwoofer	MX800i/MX8600*

MX800i CLose Coupled Electronic Processor

Rather than rely on dynamic effects to hide a loudspeaker's built in flaws, CCP's simply optimize each loudspeaker system's potential. Because the processing is "transparent," CCP's do not change the loudspeaker's tonal balance or power response at high volumes.

EAW's long range R&D goal for Close Coupled Processing is to create signal processors that integrate systems into arrays with the same degree of precision that today's crossovers integrate the separate subsystems within a multi-way loudspeaker system. Phased PointSource Technology represents the model for this goal.

Close Coupled Electronic Processors™

Close Coupled Electronic Processors (CCEP) control a range of critical parameters including low frequency driver excursion,amplifier clipping protection limiting, phase compensation and individual asymmetrical filters for each frequency band.

MX Series CCEP's include the two-way, two-channel MX100 dedicated subwoofer crossover, the MX200i (two-way, two-channel) the MX300i (three-way, two-channel) and the four-way, two channel MX800i processor.

Close Coupled Digital Processing

The new MX8600 incorporates a number of exclusive EAW authorized parameters and factory presets. In addition, it supplies dozens of programmable memories, protected by a security lock-out function. MIDI capability allows external control and the linking of master and slave MX8600's via a personal computer.

Each MX8600 includes two inputs and four outputs, with one output able to be configured as a direct subwoofer output. Each output has its own 3-band parametric equalization offering a wide range of Q's, with +15 dB and -25dB of gain at any center frequency (20 Hz -20k Hz) controllable in 0.1dB steps.

Each output also includes delay, incremental up to 682 mS, as well as digital level controls and variable high- and low-pass filters that can be set for 12dB, 18dB, or 24dB octave slopes. A choice of Bessel, Butterworth or Linkwitz-Riley responses is available.

MX8600 CLose Coupled Digital Processor

A convenient front panel design features three velocity-sensitive rotary encoders for adjustment, with channel and section "flat" keys allowing A/B comparison or EQ to be pre-set and dropped in. All filter information is displayed on a backlit LCD display, while the selectable headroom is indicated via LED meters.

Close Coupled Power

EAW has developed a reputation as pro audio's leading system integrator. From passive crossover/filter design through Close Coupled Processing, the emphasis has been on developing total systems that meet our design goals of smooth power response with zero distortion.

EAW powered Loudspeaker systems Close Couple the power module to the specific loudspeaker system. Each EAW powered loudspeaker system's Close Coupled Power Module was developed specifically for that loudspeaker system. This allow for complete optimization of the signal path including:

- signal processing
- system protection
- amplification
- drivers
- horns
- enclosure venting

EAW's KF400a Powered Loudspeaker System

The KF400a combines EAW's benchmark Virtual Array Technology® - long associated with high-quality portable sound reinforcement - with the real-world convenience of a powered loudspeaker.

Like all VA® systems, the KF400a uses three-way loudspeaker design to provide high-impact, high-definition output with a tightly controlled dispersion pattern. VA Technology's application of advanced horn-loading techniques on both the mid- and high-frequency subsystems optimizes pattern control, allowing multiple systems to be arrayed with minimal interference.

The KF400a bring the Close Coupling concept to its ultimate conclusion. All aspects of the KF400a's Close Coupled Power Module are optimized to meet the needs of this specific loudspeaker system. This ensures an idealized signal path from the source (CD, mix console, etc.) to the listener.

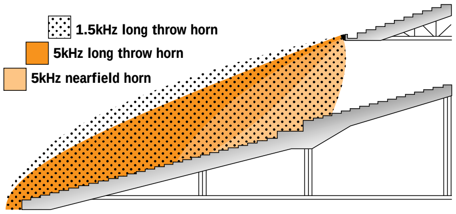
Automated construction techniques allow for the creation of midrange horn flares that follow complex mathematical functions rather than simple straight lines. A proprietary phase/displacement plug ensures coherent summation of mid-frequency energy at the horn-throat.

Developed in partnership with high-end amplification specialists CyberLogic, all amplification components are engineered specifically for the KF400a:

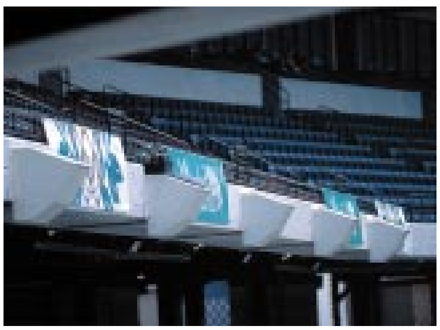
- power supply
- protection circuitry
- output devices

The KF400a features auto-sensing AC power input, allowing the system to accept any AC feed from 95 to 125 Volts or 190 to 250 Volts. Front-end signal processing divides the audio signal into LF/MF and HF passbands for amplification.

Each of the CCP module's dual-power amplification channels is load-tailored to match the demands of the KF400a's drivers. Vertical N-channel MOSFET output devices provide ample power for the drivers creating a powerful, dynamic sound quality with plenty of headroom to handle transient peaks.



The AOS90 (pictured below at Turner Field, Atlanta) uses frequency shading to integrate the output of two very different HF horns. Since the high power long throw horn provides usable mid frequency coverage in the nearfield, the nearfield horn is used only for truly high frequencies coverage of which the long throw horn does not provide.



Power supply for the HF channel features Class H rail switching with two steps at 60 and 102 VDC. The HF amplification channel delivers 200 Watts continuous/320 Watts peak output into an 8 load.

The LF channel features Class H rail switching with three steps at 60, 102 and 143 VDC. The LF amplification channel delivers 320 Watts continuous/800 Watts peak output into an 8 load.

Transparent Protection Circuitry

Unlike other self-powered loudspeakers whose protection circuitry is actuated by a time-sampling sensor, the KF400a's CCPM continuously monitors the LF/MF and HF channels indecently reading both output load and Voltage in real time. This allows for the gradual application of complex compressor/limiters, soft clipping circuitry and sliding high pass filters. Even when driven to peak output, listeners have difficulty determining whether or not protection circuitry is engaged and must rely on indicator lights to be certain.

Frequency Shading

Frequency shading leverages the well documented interaction between spaced drivers in a manner that provides some benefit to the loudspeaker user. It can be achieved in either the passive electrical or digital processing domains.

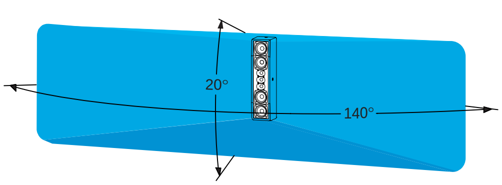
The basic idea is to limit a given driver's passband to that region where its interaction with another driver covering an overlapping passband achieves some aim of the loudspeaker design.

Over several years of use, the technique has provided a workable solution to three different problems. In each case, it allows a loudspeaker to perform within a set of tightly-defined parameters that would otherwise be unobtainable.

AOS90

Altanta's Olympic Stadium, now Turner Field, needed a single, passively powered loudspeaker to cover an area from directly below the enclosure to over 100 feet away – a task usually divided between a nearfield and longthrow loudspeaker. No additional powering or external processing could be employed.

The AOS90 uses frequency shading to integrate a high power compression driver/longthrow HF horn subsystem with a lower power nearfield HF subsystem without problematic interference between the two. A single woofer covers the entire area.



The LS Series uses frequency shading to integrate the output of several drivers forming an arced array. Using well-documented driver interaction, sophisticated passive filtering maximizes beneficial pattern control while virtually eliminating destructive interference.

Since a constant directivity horn provides pattern control only within a certain frequency range, the concept was to allow the off axis radiation from high power longthrow device in the mid frequency area to cover the nearfield area. Both spectral balance and level were appropriate to the application. The nearfield horn was used only where the longthrow horn's pattern begins to narrow. As a result, the AOS90 provided high impact, full range response over the entire defined coverage area using only a single amplifier channel.

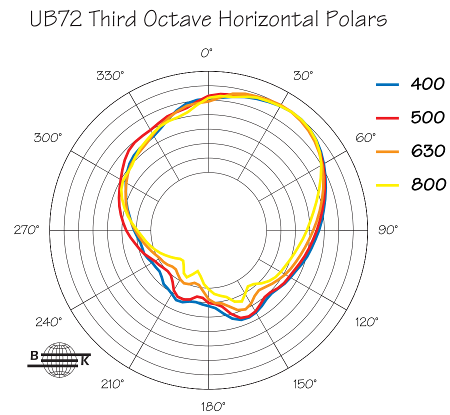
LS Series

EAW's LS Series of line source loudspeaker systems brings the classic column speaker up-to-date. Sophisticated frequency shading and all-pass filtering integrates the drivers, maximizing the benefits of line source coupling while eliminating destructive interference.

The systems provide a well behaved nominal vertical coverage pattern of 20° that opens slowly and holds together into the LF region. With the enclosure baffle defining a gentle arc, the drivers form a curved line source to help prevent the vertical pattern from collapsing in the crossover region.

At the same time, the drivers act as direct radiators in the horizontal plane, giving the systems an extra-wide 140° horizontal coverage pattern with response that meets professional standards for fidelity and intelligibility.

Like the classic column speakers of the '50s and 60's, the LS Series was designed to solve speech-only installation problems in highly reverberant spaces with low ceilings and hard floors. These might include small houses of worship, libraries or other civic spaces, and transportation hubs.



Look what frequency shading does to the UB72. These polars show that even into the midrange, coverage is still angled down 30°.

UB72

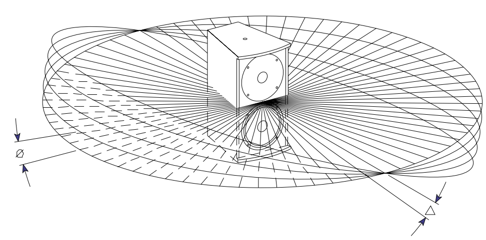
The original design specification for the UB72 required a compact rectangular enclosure with a flat baffle that mounted flush to a wall to provide coverage angled down 30°.

For the high frequency range, a downfiring horn was designed to provide the necessary coverage. But the design constraints precluded creating a similar horn for the midrange, nor could the woofer be mounted at an angle.

The solution was to employ dual woofers and develop frequency shading that would "bend" the combined output of the woofers to the necessary angles.

Power Response and How it is Measured

Some speaker systems have outstanding on-axis response, but sound considerably worse from almost any other angle. This is due to uneven power response. Power response is an average of the system's response measured at incremental locations across the entire listening area. EAW engineer Jeff Rocha has developed a complex and highly accurate method for determining a speaker's power response, called Beamwidth Delimited Power Response.



Determining the Beamwidth Delimited Power Response of a loudspeaker system requires measurement at 5184 (72 x 72) discreet locations.

Previously, power response was measured by manually moving the measurement microphone across a system's coverage area. Using an RTA in "Forever Averaging" mode, the power response measurement was partly a function of the length of time the microphone spent in each sector of the coverage area. If the microphone was in one sector longer than another, data from the former would influence the average more than that from the latter.

Another method, more time consuming but more accurate, is to place the system on a turntable and measure response in 5° increments. After one horizontal circle is completed (72 measurements) the system is vertically rotated 5° and 72 more measurements taken in the horizontal plane. This entire cycle is then repeated in 5° vertical increments until a complete sphere of measurement points is created. The responses are then averaged to produce a power response chart. (Download this .pdf file illustrating the power response measurement process.)

While this method produces accurate data at each measurement point, Jeff was dissatisfied with the method of averaging the data. Like the sectors defined by the longitudes and latitudes of the earth, the sectors near the "equator" of the measurement sphere were much larger than those at the "poles." Thus, there are more measurement points near the poles, and since the poles represented the on-axis position (or 180° opposite on axis), on-axis response carried more weight than the off-axis response which power response is supposed to measure.

His solution was to create a mathematical formula to "weight" each measurement point depending on the size of the sector it represents. This equalized the measurements, returning a more accurate power response measurement. But Jeff was still dissatisfied. He had devised a method for producing accurate power response information. But he questioned its usefulness in the real world because the system's response outside the intended coverage area carried more weight than the response inside the coverage area. Again, he created a formula to weight proportionately the measurement points inside the defined coverage area. Finally, he created a system for measuring power response which he felt was both accurate and useful.

A KF850 undergoing testing and measurement in our automated test facility affectionately known as "The Pit."

Testing and Measurement

Directivity Data

The directivity data gathered on EAW speaker systems for the APP program is collected using the Brüel & Kjær 2012 Audio Analyzer in TSR (Time Selective Response) mode. "TSR mode enables "free-field" measurements without an anechoic chamber, by rejecting the reflections from an ordinary listening room. The Type 2012 incorporates a technique that allows a useful combination of speed, accuracy, and signal/noise ratio for such measurements." (From the B&K 2012 Reference Manual, p.2.)

The test signal generated by the 2012 is a constant amplitude, linear sine sweep: The instantaneous frequency varies directly with time. Figure 1.1 illustrates a typical test setup. The 2012's output is connected a power amplifier (when testing systems that are powered in fullrange passive mode) or to an MX series CCEP (Close Coupled Electronic Processor) when testing bi-, tri- or quad amplified systems. When a processor is part of the system under test, its output signals are then amplified by Crest 8001 power amplifiers set at their maximum levels and the resulting signals are sent to the loudspeaker enclosure.

The 2012 allows the user to enter a single output sensitivity factor. For these tests, the maximum gain of a Crest 8001 amplifier is used. The voltage driving the system under test can be set to any desired value-the 2012 automatically scales the resulting SPL values to a level that would have been produced by a reference voltage of 2.83 Volts (one nominal Watt across an 8 load). It is important to remember that the MX Series processor's gain is not factored into the Type 2012's calculations, so the recorded SPL values result from a voltage other than 2.83 Volts.

Directivity data is collected in a full spherical model using 5° increments along both the horizontal and vertical planes. 77, 256 SPL values are recorded for every system (72 points for each horizontal polar [0° to 355°] x 37 vertical polar angles [0° to 180°] x 29 third-octave centers).

Impedance vs. Frequency

Impedance versus frequency for each speaker system is measured using the Steady State Response mode of the B&K 2012. The output channel of the B&K 2012 is connected to a power amplifier, which is turned into a current generator by placing a 1k resistor in series with its relatively small output impedance. The speaker is connected after the 1k resistor. The voltage across the known resistor can be easily measured, allowing the calculation of current flow through the loop. The unknown speaker impedance is simply calculated using Ohm's Law, R=V/I.

CCEP Response Curves

CCEP response sweeps are obtained using the Steady State Response mode of the B&K 2012 audio analyzer.

Distortion Measurements

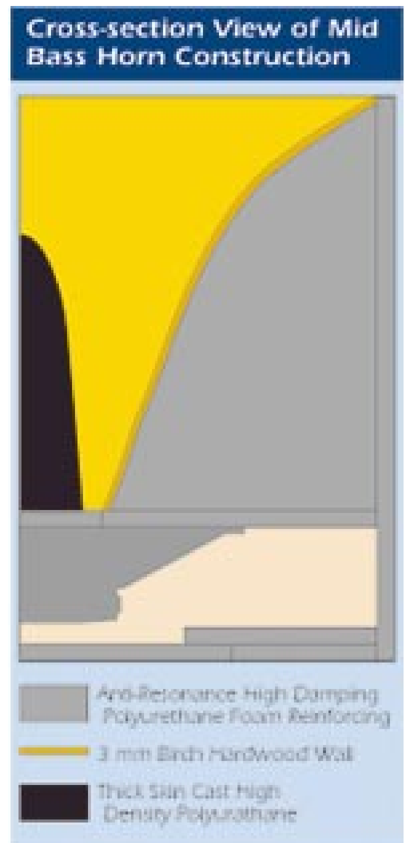
Like the directivity measurements, distortion measurements are made using the B&K 2012's Time Selective Response mode. Six measurements are made of each system: The fundamental, second harmonic, and third harmonic are measured at 1% and 10% of each system's rated long term power handling (the highest 100 hour sine wave value of the various drivers in the system).

ARC: Acoustic Refraction Control

To create a true three-way system like the KF650 that was manageable in size, efficient use of space was critical. Since the woofer need only handle frequencies up to 250 Hz, we were able to mount the high frequency subsystem within the woofer horn.

Testing in our automated measurement facility revealed that small amounts of energy at the bottom of the HF range were refracting or "wrapping around" the HF waveguide and radiating back down the LF horn. This energy would then reflect off the woofer cone and arrive in the listening area significantly later than the original HF sound, blurring staccato sounds like percussion.

To alleviate this problem, the Acoustic Refraction Control (ARC) device was developed using a proprietary material that would be transparent to low frequencies yet absorb high frequencies. The ARC is mounted in the woofer cavity directly behind the HF horn. It absorbs the reflected HF sound waves while allowing LF energy to pass through unobstructed. While testing this new system, a fringe benefit to using ARC was discovered. Not only does it absorb refracted HF energy, it also acoustically filters harmonic distortion from the woofer which cannot be eliminated by active or passive crossovers. Again, since all ARC filtering occurs above the LF subsystem crossover, the woofer's intended response is unaffected.



Our proprietary construction techniques allow us to create complex horn flares in which a 3mm hardwood veneer acts as the mold for high density structural urethane foam that serves to make the horn acoustically opaque.

EAW weatherproofing helps loudspeakers – like these in Barcelona, Spain – survive years of exposure to sun, wind and rain, often in locations where they cannot easily be reached for repair.

Midrange Horn Construction

To achieve truly flat response and consistent directivity across a horn's passband, the flare of the horn must conform precisely to an exponential curve. The curvature must be directly proportional to distance from the driver; the farther from the driver, the greater the rate of expansion.

Modern manufacturing technology allows such designs to be fabricated relatively easily using molded plastics or fiberglass, materials EAW uses to make high frequency waveguides. But those materials are resonant within the bass and midrange horns' passbands and are therefore unacceptable for this purpose.
The 3/4-in Baltic birch ply used by EAW for cabinet construction is acoustically "dead" but very difficult to shape into the complex curves required for an accurate horn. Conversely, a thinner, more bendable birch ply would be acoustically transparent at low frequencies and could not function as a waveguide.

The solution EAW has developed is to use a 3-mm sheet of birch ply fitted into a track cut by a computer-numerically-controlled router. This allows the horn flare to conform precisely to the mathematical formula.

To create an acoustical barrier, the space behind the flare is filled with a high density polyurethane foam. After the foam hardens, it, in effect, becomes the horn flare and the thin birch ply sheet is merely a mold which defines its shape.

WP: Weatherproofing

The single most important requirement of any loudspeaker specified for outdoor use is that it survive Mother Nature's continual onslaught. From the constant humidity and daily thunderstorms of south Florida to the searing heat of southern California, EAW's unique WP weatherproofing process has proven itself the master of the elements. Still, EAW engineers are continually researching new materials and methods in order to provide increased reliability and cost-effectiveness.

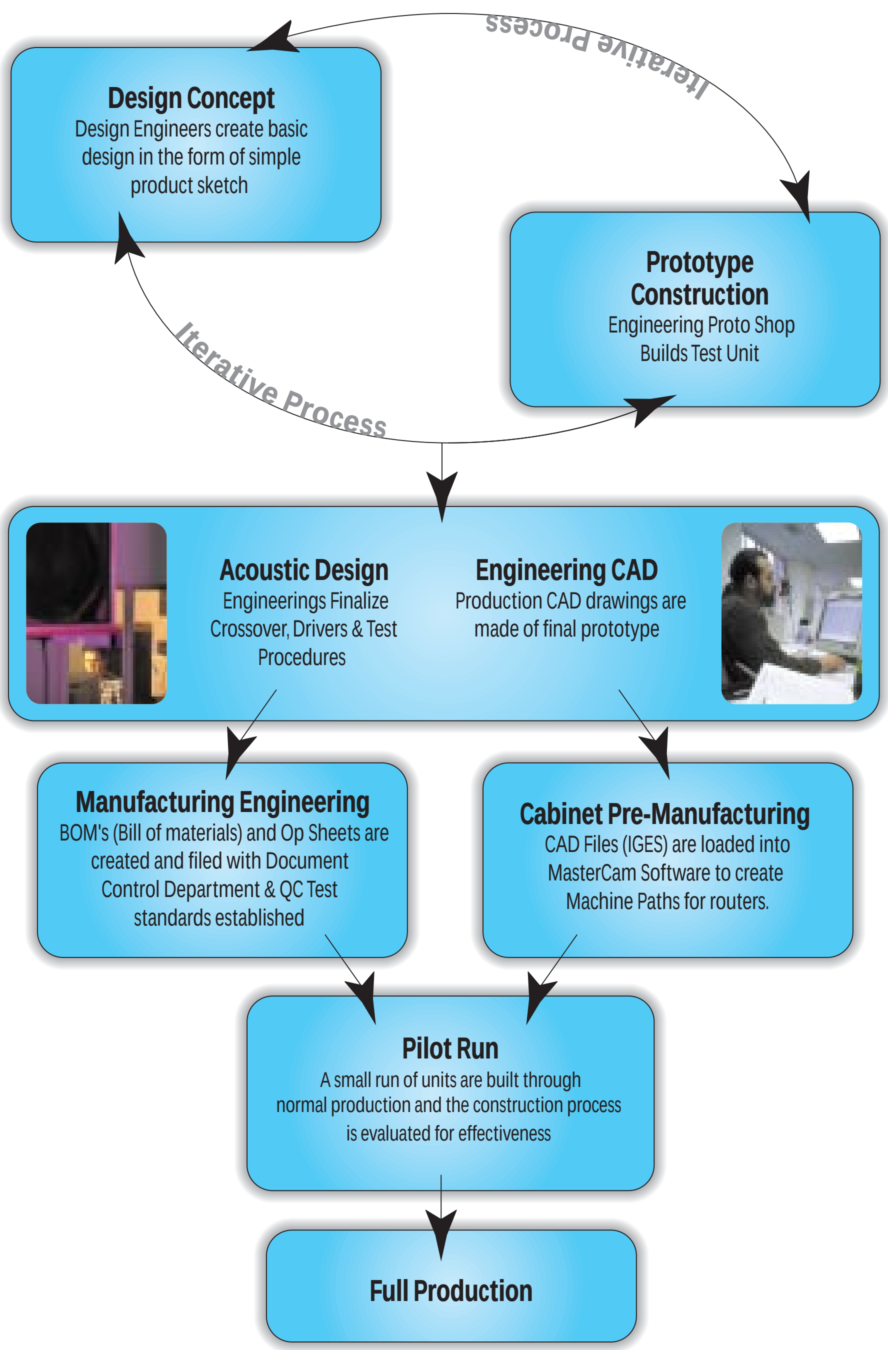
WP versions of all EAW systems are available as a special order option. EAW's proprietary waterproofing technique includes a fiberglass resin coating on all exterior cabinet surfaces and a waterproof epoxy coating on all interior cabinet surfaces. Transducers are protected from the elements by a three-layer weatherproof grill consisting of a powder-coated perforated stainless steel outer grill, an ozone-resistant, fungus proof open cell foam middle grill as a dust and water barrier, and an acoustically transparent stainless steel mesh water barrier with 10,000 pores to the inch.

The standard input configuration for weatherproof systems consists of a barrier strip mounted to a phenolic panel which is recessed into the rear of the enclosure. For extreme conditions, this panel can be covered by a 1/8-in thick aluminum plate. Stainless steel machine screws attach the plate to the enclosure: a weatherproof gasket makes good the seal. Water proof gland-seal nuts allow the input cables to pass through without introducing moisture.

CAD/CAM – Rapid Prototyping

EAW's use of CAD/CAM (Computer-Aided Design/Computer-Aided Manufacture) technology enables a remarkably rapid prototyping cycle, greatly condensing each iteration within a product's development.

A prototype design for a new loudspeaker enclosure can be created from a CAD drawing in days, not weeks as used to be the case. After the prototype enclosure is loaded with components in our dedicated prototyping shop, the total system is ready for testing and measurement in one of our labs. Data from these test are then used to refine the next generation design and the cycle begins anew.



EAW's unique rapid prototyping capabilities allow us to create the best loudspeaker systems in the world in a relatively short product development cycle. This allows us to be responsive to our customer's needs.

One Main Street, Whitinsville, MA 01588 ***tel:*** 800 992 5013 / 508 234 6158 ***fax:*** 508 234 8251 **web:** www.eaw.com

EUROPE: EAW International Ltd., ***tel:*** +44 1494 539090 ***fax:*** +44 1494 539091

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