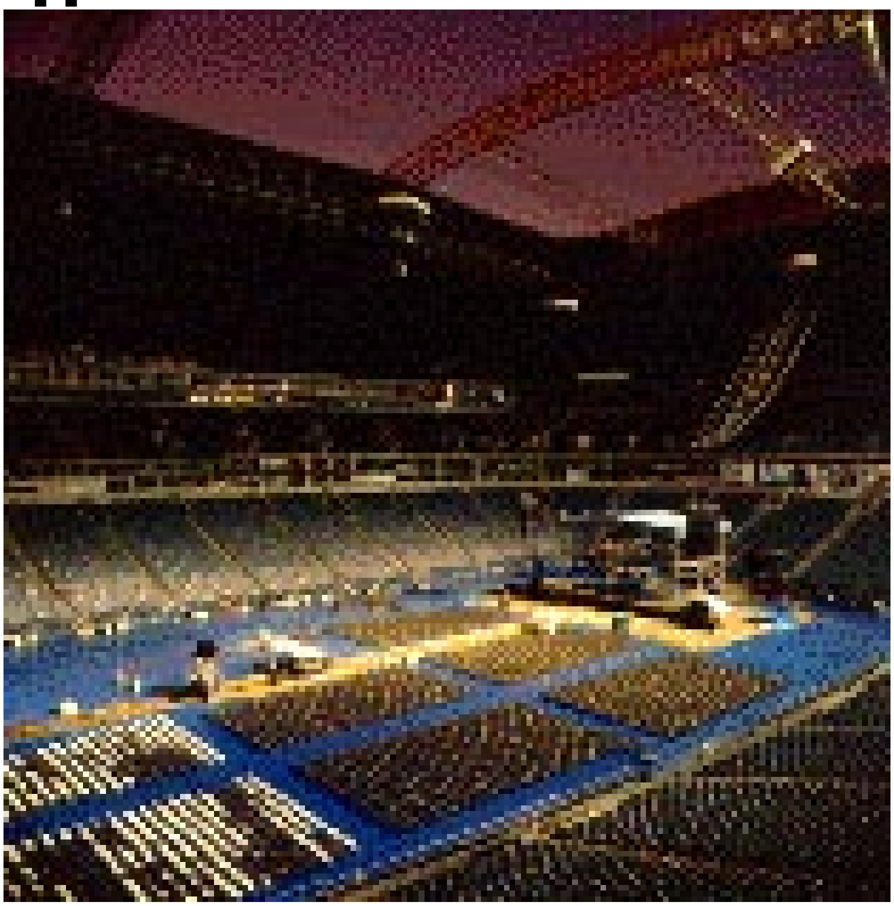
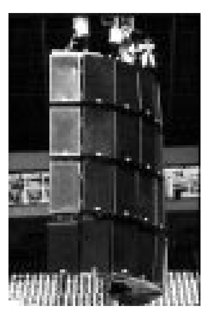
Real World High Q
Definitions
Design Issues
Applications





An Eastern Acoustic Works Engineering White Paper

Q: CONCEPTS & DEFINITIONS

Pat Brown, Synergetic Audio Concepts

Introduction

Energy is all around us. We shape and form it in useful ways, to make our lives easier and more convenient. We transmit it, radiate it, absorb and dissipate it. In this modern era, we experience the radiation of energy in all facets of our lives. Television and radio waves are simply electromagnetic energy shaped into something useful. The common light bulb is a transducer of electrical energy into light energy. Man has found many ways to efficiently propagate energy and utilized it to perform useful tasks.

What is Sound?

Sound is energy. When an air particle is disturbed from its restful, ambient state, it causes a very small change in the static atmospheric pressure around it. A "domino effect" occurs, with one particle striking another, radiating energy outward from the point of the initial disturbance. The process continues until this energy is dissipated in the form of heat into the environment. If a human were to intercept the path of this phenomena, the disruptions of atmospheric pressure would cause movement of the timpanic membrane, or eardrum, and be perceived as sound. Sound is a part of everyday life, whether it be the sound of an engine, the rustling of leaves, or the roar of thunder.

Sound, like other forms of energy, occurs randomly in nature seemingly with little purpose. But also like other forms of energy, it has been harnessed by man to do work. Work is the act of an object being moved a distance by a force, and the eardrum is an object that must be moved for sound to occur. Just as the potter molds and shapes the clay to form a vessel, audio and acoustics engineers mold and shape sound. When a person speaks, a small amount of sound energy is generated and propagates through the environment. Energy, when radiating though a liquid (which includes air) is called "acoustic" energy. When acoustic energy is intercepted by a microphone, it is "transduced" into the electromagnetic spectrum. The term "audio" is used to describe the sound signal when it is in this realm. This includes its path through the mixer, equalizer, amplifier, and all of the wires that connect them together. In the electromagnetic domain man can exhibit an extraordinary degree of control over this energy. It can be amplified, attenuated, equalized, compressed, expanded and literally "molded" into whatever shape we find useful. But this energy is of little use while still in the audio realm. It must be reintroduced into the environment to be of use to humans, whose built-in "pressure transducers" are sensitive only to acoustic energy. This is the job of the loudspeaker. Like the microphone, the loudspeaker changes, or transduces energy from one realm to another. The loudspeaker designer, like the potter, is a sculptor of acoustic energy.

A-1 Audio arrayed KF1000's, KF850's, SB850's and JF260's to transduce electrical energy into acoustic energy on Van Halen's 1995 Balance World Tour.

Sound Power and Directivity

Man learned long ago that energy could be used more efficiently if it were confined to a smaller unit area. This principle is inescapable in everyday life. Whether it be the radiation of a light bulb, the blow of a hammer, or the spray of a hose, the increase of energy per unit area is a valuable tool.

It is time to construct some fundamental identities upon which to expand this idea. Consider that there existed a perfectly omnidirectional radiator of sound, one that when placed in a free field (no nearby objects) would radiate energy equally in every direction. Since we have already imagined something impossible, let's make it just a little more preposterous and imagine that this were a true point source, a pin-head sized transducer. If the transducer were perfectly efficient, we could feed it one electrical watt (through a very small cable!) and it would "transduce" or convert this into one watt of acoustical energy. How loud is one acoustic watt? At this point it becomes necessary to convert our numbers into a system that correlates better with how the ear works. Watts are fine for motors and light bulbs, but when dealing with sound and human perception, watts must be converted into decibels. The decibel system is used to compare two numbers with each other, and to "compress" the answer to a range of numbers that is more convenient to work with. So just how much sound power is one acoustic watt?

$$L_W = 10\log\frac{1\,Watt}{10^{-12}\,Watts} = 120\,dB$$

The L_W is the sound power of the source. It is important to note that this is just raw acoustic power, without consideration for directivity or distance.

Sound Intensity

If the energy were to radiate in a spherical manner, at a distance of .282 meters from our tiny source, the energy will be passing though a sphere with one square meter of surface area. At this point the energy can be described in terms of power per unit area, better known as sound intensity. What is the sound intensity at the surface of the sphere? Using the decibel:

$$L_I = 10\log\frac{1\,Watt/m^2}{10^{-12}\,Watts/m^2} = 120\,dB$$

The sound intensity is 1 Watt per square meter, and expressed as a level in decibels is 120 dB.

When a single watt of acoustical power is radiating in an omnidirectional pattern, the maximum level that can exist on the surface the one square meter sphere is 120dB. Can this be improved upon? If more power per unit area is needed, and the power from the source cannot be increased, the last recourse is to confine the power to a smaller unit of area. Suppose the transducer were placed on an infinitely large, reflective surface. The one watt now radiates into half of the area that it did previously, and the power per unit area is effectively doubled. So now an L_W of 120dB becomes an L_I of 123dB.

$$L_I = 10\log\frac{2\,W/m^2}{10^{-12}\,W/m^2} = 123\,dB$$

The sound intensity is 2 Watts per square meter, and expressed as a level in decibels is 123 dB.

Suppose that the transducer were placed where the floor and wall meet. It still radiates its one watt, but now the area of radiation is restricted to one-fourth sphere, effectively quadrupling the power per unit area. In mathematical terms this becomes:

$$L_W = 10\log\frac{4\,W/m^2}{10^{-12}\,W/m^2} = 126\,dB$$

The sound intensity is 4 Watts per square meter, and expressed as a level in decibels is 126 dB.

As long as the radiated power is confined to a smaller and smaller unit of area, the intensity will continue to increase. It is this very principle that allows a nail to pierce wood, or a laser to cut metal. While we typically don't confine sound to such small areas, the utility of the concept is becoming apparent.

Sound Pressure

Thus far, the descriptors that have been used for the sound energy have not included the most common one, sound pressure level or SPL. This was intentional, since an understanding of SPL must begin with an understanding of sound power and sound intensity. Remember that sound is energy, and power is the term that we use to describe energy. Sound pressure is what results when power exists. While power can be used to describe the total energy or the energy per unit area, sound pressure is always a measurement at a single point in space. If you will, sound power is the cause and sound pressure is the effect. It always follows that as the sound intensity is increased or decreased over a given unit of area, the resultant sound pressure will also increase or decrease. Referring to the spherical model previously described, a sound pressure measurement could be obtained by placing a microphone at the surface of the sphere and reading the pressure caused by the power at that point. Sound pressure is measured in Pascals, and as with sound power it is usually converted into a level in decibels so that changes in it can be better correlated with the loudness that a listener will experience. The difference in loudness between two sound pressures can be determined by the relationship:

$$L_P = 20\log\frac{p_1}{p_2} = dB\,differenc$$

where p1 and p2 are two pressures that are being compared.

Rather than compare pressures to each other, there is great utility in choosing a reference pressure to compare all pressures to, which will also then allow the resultant levels to be compared with each other. The standard reference pressure is 20 mPascals, which happens to be about the smallest change in pressure that the human auditory system can perceive. The formula looks like this:

$$L_P = 20\log\frac{p_x}{20(10^{-6})}\,dB$$

Sound Pressure Level

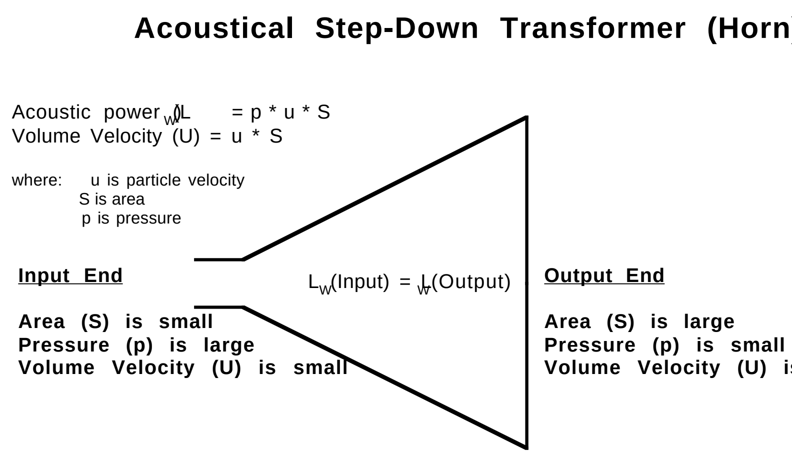
When sound pressures are mathematically manipulated in this manner, the resultant is called a sound pressure level.

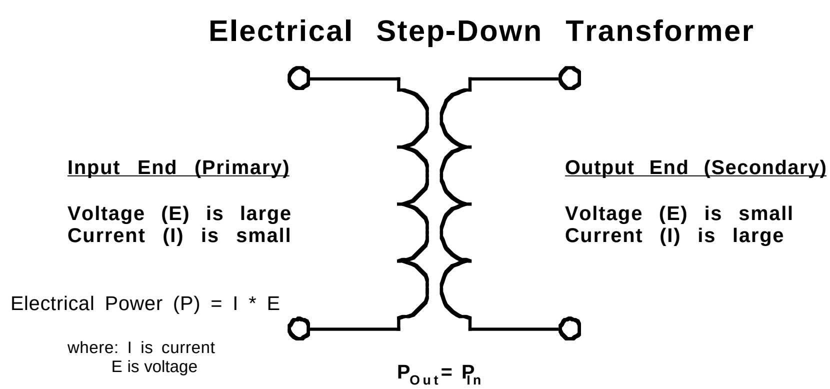
Notice that the multiplier used is 20,

rather than 10, since acoustical sound pressure is analogous to electrical voltage. Ohm's law establishes that power (or wattage) is proportional to the square of pressure (or voltage), and 20 times the log of a number is the same thing as taking 10 times the log of a number squared.

Compression Drivers and Horns

The loudspeaker or compression driver converts energy from electrical to acoustical by means of a motor mechanism, a moving coil, and a diaphragm. The movement of the diaphragm causes the disturbances to the atmosphere that we call sound. As in any circuit, each subsequent stage must be coupled to the previous one. In the case of the electrical signal, this is accomplished with wire, cables, and connectors. Since the compression driver is a transducer, at this component the circuit model changes from an electrical circuit to an acoustical circuit. Upon the transduction of energy from electrical to acoustical, the first device placed in the acoustic circuit is a transformer. The horn is essentially an acoustical transformer. Its surfaces define a bounded region whose cross-sectional area increases from the input end to the output end. The energy flow though this bounded region can be described as the product of the pressure and the volume velocity, or pU. At the input end of a horn, where the cross-sectional area (S) is small, the pressure is large and the volume velocity is small for a given acoustic power. At the output end of the horn, the volume velocity is large, the cross-sectional area is large, and the pressure is small for a given acoustic power. As the sound energy propagates through the horn,





the flare rate of the horn transforms the impedance of the "acoustical circuit" from high to an impedance that more closely matches the air around us. A driver without a horn has a poor impedance match with the environment, and you simply don't get much sound. The horn then, can be thought of as a device of transition, providing the proper impedance transformation and serving to effectively couple the compression driver to the environment.

Efficiency & Power Compression

The term "efficiency" is used to describe a transducers ability to convert electrical energy into acoustical energy. It is usually expressed as a percentage, and can range from as low as 1% for a woofer to as high as 50% for some compression drivers. Whatever energy is not converted into sound is converted into heat. If you have ever accidentally touched a 100 watt light bulb, you have experienced the amount of heat that this power can produce. Transducer engineers must deal with this heat, since it can effect the performance of a transducer and even shorten its life. Power compression is a phenomenon that occurs when the heat build-up causes a rise in the impedance of the transducer. As impedance increases, less power is drawn by the device and the resultant sound level no longer reaches the level expected from the driver.

Sensitivity

As previously described, baffles or horns can confine sound power to a smaller unit area, producing an increase in sound intensity and sound pressure. A sensitivity rating takes this confinement into account, and reflects the increase level caused by increasing Q beyond unity. It is typically measured by placing a microphone on-axis with the device at some nominal distance (usually 30'), with one electrical Watt applied to the device. The measured sound pressure level is extrapolated back to a distance of one meter using inverse-square law. This becomes the one Watt/one meter sensitivity rating for the device. On-axis sensitivity measurements are a valuable design tool, but should be considered as only a partial describer of a devices performance. Polar plots reveal the sensitivity at various points around the device, and give a better picture of how the device might actually do in providing coverage to an audience.

Q and Coverage Angles

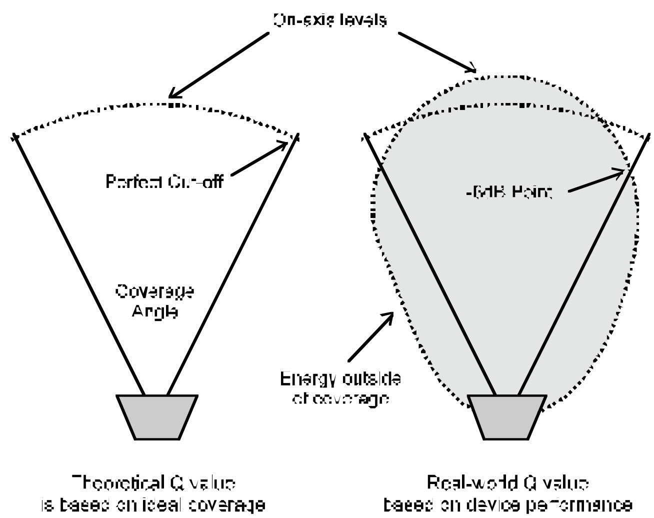
How do real-world Q values compare to theoretical ones? An ideal device would have all of its energy passing through its coverage angles. If such a device existed, its Q could be described by

$$Q = \frac{180}{\sin^{-1}\left[\left(\sin\frac{\theta}{2}\right)\left(\sin\frac{\phi}{2}\right)\right]}$$

where f and q represent the horizontal and vertical coverage angles, respectively. At first glance it would seem that a devices Q could never exceed this theoretical value. Keep in mind some of the assumptions of this ideal case. First, it says that the level is the same at every point included within the two angles, and that it drops infinitely outside of the area of coverage. In real-world devices, the limits of coverage are defined to be the 6dB down points. This means that the level (and the Q) at the edge of coverage is at least one-fourth that of the on-axis angle. Also, real-world devices tend to focus most of the energy on-axis. While the formula gives the highest average Q that a device can have, the Q of real devices will typically exceed this value on-axis and be less than the ideal at the limits of coverage.

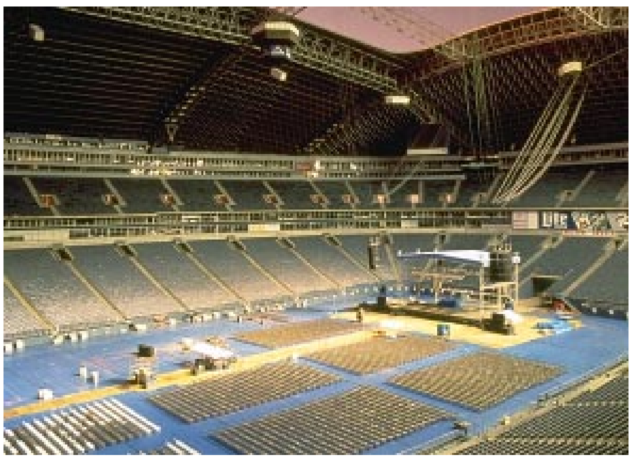
Directivity Index

Actual usable values of Q will range from 1 (omnidirectional) to about 50 (large pattern control horn). The most common values found for mid and high frequency devices will range from 10 to 20. Q can be turned into dB, which is called the directivity index. The DI will yield the increase in sound pressure level at a point compared to what would exist had the radiation been omnidirectional. For instance, a Q of 10 would yield a DI of



$$DI = 10\log Q = 10\log 10 = 10dB$$

This means that 10dB more sound pressure level is achieved by confining the sound radiation to a smaller unit area. This can be quite useful when trying to produce high SPL's on-axis. High Q values are difficult to realize at low frequencies, due to the longer wavelengths of the sound. One should consider that a 100Hz sound wave has a physical size in space of about 11 feet, and that an appropriate waveguide (horn) would need to approach this size at the mouth to exhibit directional control over the wave. Such horns, while they do exist, can be prohibitively large for many auditoria. Directional control of low frequencies is often limited to device placement. Placing an omnidirectional low-frequency device on the floor yields a Q of 2, placing it at the juncture of a floor and wall yields a Q of 4, and placing it in a corner yields a Q of 8.



KF853's, KF852's, BH852's and KF850E's arrayed for the Promise Keepers at Texas Stadium – where RT_{60}'s are in the 10 second range – by Spectrum Sound of Nashville.

Q can also be increased by arraying techniques. Stacking loudspeakers horizontally or vertically will result in focusing of sound, and an increase of Q. Line arrays, which utilize this concept, are gaining in popularity among system designers.

The prudent system designer will try to keep the Q of the system as consistent as possible throughout the spectrum, resulting in equal energy per octave band being delivered to the acoustic environment and reverberant sound field. Such a system could be called "constant Q."

Recycled Sound

We humans tend to be wasteful creatures, giving little thought to what happens to the "extra" that results from the use of a resource. For instance, the extra water from taking a shower goes down the drain, and we seldom give it another thought. If the drain were plugged and that "extra" water could not leave, we would be more concerned with how much we used. The vast majority of the sound power emitted by a sound system is wasted. It only takes about 35 mWatts of energy at the eardrum to produce a level consistent with normal conversation. If one could inject this amount of sound power directly into each ear canal, the needed acoustic power of a sound reinforcement system could be calculated by counting the ears occupying the space and multiplying by .000035 Watts. For an audience of 1000 people, this would be

$$W = 1000(2)(.000035) = 35\,mW$$

If a 1000 watt power amplifier were used to drive a loudspeaker that was 10% efficient, there would be 100 acoustic Watts emitted into the space. This is nearly 3000 times the energy needed to produce an average listening level at the eardrum. Where does all of this acoustic power go? It goes into the environment, and is experienced repeatedly in the form of reflections and reverberation, until it is dissipated as heat by the absorption of the air and wall surfaces of the space.

In an enclosed space, the only "acoustic drains" that exist are open doors and windows, whatever absorption is present in the furnishings, and the air itself. Of course if all of the doors and windows are closed, and all of the surface materials are hard, the sound is going to hang around for awhile. This "recycled" sound is sometimes friend and sometimes foe, and we must learn how to deal with it.

The direct sound field by definition is that sound energy that arrives at the listener before encountering a boundary. A little thought reveals that this sound field, at a given distance, should be the same in any environment for a given loudspeaker. The very predictable nature of the direct sound is useful to the sound system engineer. This is the sound field that we design with, using the required levels at listener seats to determine the needed loudspeaker sensitivity and required amplifier power.

A reflection occurs when sound arrives at a listener after bouncing off of something. There are can be many reflections in an enclosed space, and the pattern of reflections will be different at every listener seat, since each seat has a unique distance relationship to the boundaries in the space. If, due to the lack of absorption in the space, the reflections are allowed to continue unhindered for a long enough period, another sound field emerges. This is the reverberant sound field and it has some interesting characteristics of its own.

The large amount of glass in the Crystal Cathedral of Garden Grove, CA necessitates directing sound to the listeners and away from the walls and ceilings. This difficult acoustic environment has been tamed with a combination of MH Series and KF Series Virtual Array systems installed by Audiowest.

The level of the reverberant field is uniform throughout the enclosed space. It can be described as a random and chaotic distribution of energy that impinges upon the listener equally from every direction. The direct sound field carries the music or speech information, and the early reflected field determines the "sonic signature" of the room. The reverberant sound field is just a noise, and does not contribute to the intelligibility of speech nor the clarity of music. Reverberation is not necessarily bad, but as with any noise source it must be controlled for the sound system to work properly. It is important to remember that the sound system is supplying the energy for the reverberant field. It is therefore useful in such spaces to "manage" the energy emitted by the system, which brings us back to Q. Reverberation can be reduced by using a directional loudspeaker to confine the radiated energy to the audience area. People are excellent sound absorbers, and in some venues about the only usable absorption present besides the air. Directivity factor (Q) allows the sound power from the loudspeaker to be used efficiently in providing more sound to the listener and less to the room.

Critical Distance

Suppose you were to place a low-Q loudspeaker in a reverberant space, and use a noise source to energize the loudspeaker and excite the room. As long as you are standing very near the loudspeaker, the level of the direct sound will predominate over the level of the reverberant sound field. As the distance from the source is increased, the direct sound drops in level because it is spreading (inverse-square law), yet the reverberant sound field remains constant. The distance from the source at which the direct sound and reverberant sound are equal is called Critical Distance (D_{C}). Critical distance is a parameter that exists in reverberant spaces, and it is a distance that can be adjusted by the system designer. Critical distance, for the case of a single loudspeaker in a space, can be estimated from

$$D_c = .141\sqrt{QSa}$$

where Sa is the absorption present in Sabins

Q is the directivity factor of the loudspeaker

It is important to note that Q is an adjuster of D_C. Higher Q loudspeakers allow D_C to be extended further from the sound source. The goal of the system designer, for a speech reproduction system, is to choose the loudspeaker type and location that places as many listeners as possible inside of or near D_C.

It becomes apparent that in reverberant spaces, there is a direct/reverberant energy ratio that exists for each seat. The integration properties of the ear/brain system allow some early reflections to be included as part of the direct energy component of this ratio. That is because the brain "averages" sound arrivals in approximately 35ms increments. It should be pointed out that this is a sliding scale, and is dependent upon program material. The breakpoint for music is later and is usually set at about 80ms.

The direct sound is typically the first major sound arrival, including any reflections that are within 35 – 80ms of it, depending on the program material. The reverberant or late energy part of the ratio will then be whatever lies beyond that time interval. Using these concepts, it becomes possible to qualify a listener seat for speech or music by examining the E_D/E_R (direct-to-reflected) energy ratio that exists at that position. The sound system designer can specify a Q that provides the desired ratio. This usually means higher Q devices for speech and lower Q for music. Yes, speech and music reproduction have opposite goals, and prudent system designers often implement discrete systems for each.

If we were outdoors, we could afford to be more wasteful with our energy. While the sound may bother the neighbors, it won't bother the listeners, assuming that some of it isn't returned in the form of a late reflection. Modern drivers are robust and can handle lots of power, and modern amplifiers can certainly produce it, so the concern for efficient use of acoustic energy is typically limited to the indoor case.

Directivity and Acoustic Gain

Another reason for controlling the radiation of acoustic energy is to maximize acoustic gain. When a loudspeaker is placed in room, the sound emitted is typically audible at some level everywhere in the room. Microphones, like humans, are pressure-sensitive transducers, and when placed in the room with a loudspeaker are able to "hear" the pressure waves emitted by it. The same loudspeaker that was installed to provide "input" to the audience is also providing "input" to the microphone. Of course, when the microphone receives a pressure wave, it goes about its task of converting the diaphragm move-

Outdoors, less acoustic energy is "recycled" by reflective surfaces. Signal Perfection Ltd. installed ASR7952's at Garden State Arts Center. Solstice Company, consultants. Acoustic Dimensions, SIA Acoustics, design services.

ment into electrical energy, which then proceeds through the signal chain and is eventually re-emitted by the loudspeaker into the environment. It takes only a few milliseconds for the audio signal to traverse this electronic path, and once re-emitted from the loudspeaker, sound propagates towards the mic at about one foot per millisecond. What does all of this mean? Suppose that due to placement, orientation, or level the loudspeaker becomes the *predominant* source of acoustical input to the microphone. The loop formed will be executed over and over until some steady state condition is reached. This "feedback" will continue to build until something in the electronic audio signal chain clips, limiting a further build-up of energy, or a device burns up in an attempt to sustain it. Feedback is the system designer's curse, and becomes the limiting parameter on performance in most sound reinforcement systems.

When loudspeakers are placed to the side fo the stage (as in the typical touring concert rig), horizontal directivity is the main determinant of gain before feedback. Phillips Pro Audio system for Restless Heart.

How can feedback be avoided? The answer is a simple one; prevent the microphone from hearing the loudspeaker. Let's look at some possible ways to accomplish this:

a. Put the mic and loudspeaker in different rooms. While not always practical to do, it does make a good object lesson for what we are trying to accomplish, namely, acoustic isolation. Try it with a wireless mic and experience acoustic gain that approaches infinity, limited only by the capabilities of the sound system.

b. Place a barrier between loudspeaker and mic. This is a little more practical, and can be used to get some extra gain when circumstances permit using a barrier.

c. Utilize directional loudspeakers and microphones to increase acoustic isolation. This is the most practical solution, and shall be the topic our continuing discussion.

The M_E Factor

Previously directivity factor (Q) was suggested as a means of increasing the sound per unit area (intensity) of an omnidirectional radiating source. The waveguide used to accomplish this is not creating energy, just shaping it. If the sound per unit area is increased at some listening angle, the additional energy had to come from somewhere. An increase in intensity at one angle must be accompanied by a decrease in energy at another angle, and it is this principle that can allow a microphone and loudspeaker to co-exist in the same room. If the sound heard by the microphone is reduced 6dB by using a directional device, the effective acoustic distance between the microphone and loudspeaker has been doubled. Quite a neat trick, and one that the system designer can use to dramatically increase acoustic gain. At the drawing board stage, this increased gain is termed the M_E factor. Its very existence is the reason that many sound systems can work at all. It should be remembered that the acoustic isolation afforded by directivity is frequency dependent, and tends to go away as frequency gets lower. This is another reason why the loudspeaker designer tries to achieve pattern control to as low a frequency as possible.

If the loudspeakers are located overhead, then vertical pattern control becomes the prime objective. If located to the side, then horizontal control must be maximized. Given the requirements of most venues and the logistics involved in loudspeaker placement and proper imaging, it would seem that vertical pattern control most often affords the greatest improvement in gain-before-feedback.

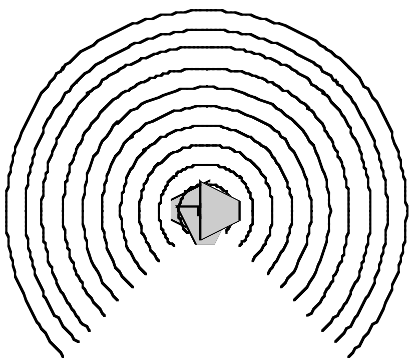
Arrayability

It is desirous at a given listener position, for all of the energy that forms the direct sound field to arrive at the listener's ear simultaneously. This can be tricky for multi-way systems, and to accomplish this, the loudspeaker designer must determine some key parameters. The first question is "From what point in space is the sound emitted?" This is called the acoustic origin. The second is, "From what point in space does the sound spread?" This is called the acoustic center. With knowledge of

Concert Sound's array for Eric Clapton's 100th Royal Albert Hall concert included KF1000's, KF850E's and BH852's.

these two parameters, a multi-way loudspeaker system can be designed that reconciles these points and propagates a coherent wavefront into the acoustic environment.

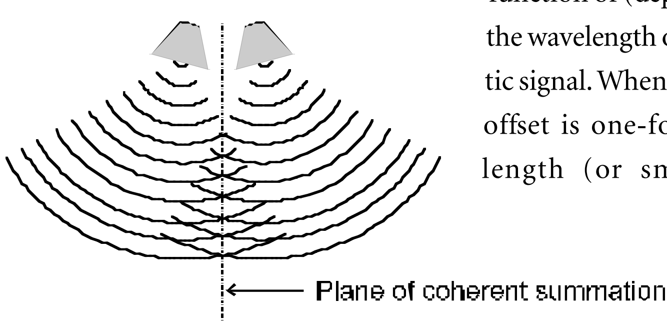
The Theoretical Model



A perfectly arrayable loudspeaker array would have the acoustic origins and acoustic centers of all devices reconciled to a single point in space

A more difficult problem to address is that of propagatinge a coherent wavefront using an array of loudspeakers. For two loudspeakers offset in space, there only exists one plane in which the sound emitted from both can arrive simultaneously. At other vantage points various degrees of cancellation occur due to the time/distance offset of the energy arrivals. This is termed *phase cancellation* since it is a phenomenon that is a function of (depends upon) the wavelength of the acoustic signal. When the distance offset is one-fourth wavelength (or smaller) the acoustic signals will add at the listener seat, even though they don't quite get there at the same time. This is called mutual coupling, and it is a useful tool for the designer. Mutual coupling is not difficult to realize at low frequencies, due to the longer wavelengths involved. In fact, there is no way to avoid it. As frequency increases and wavelengths get shorter, it becomes increasingly difficult to achieve mutual coupling between devices in an array. One solution is to minimize the overlap of coverage patterns at higher frequencies. While the term "arrayable" is one that is used loosely (initially, it meant only that the boxes would fit together physically), a loudspeaker that could described by this term would have the characteristic that two or more of them could be placed in close proximity, and their coverage patterns would either mutually couple or overlap in a minimal fashion.

Non-arrayable Loudspeaker System



Excessive overlap in coverage places much of the audience in the pattern of both loudspeakers, resulting in "hot spots" and "dead spots" in the coverage.

Great steps have been made in the arrayability of loudspeaker systems as prudent designers develop better methods to control the dispersion of their devices. Remember that it is the polar pattern of the array that is of interest, not just the polar pattern of a single loudspeaker. As performance conscious manufacturers measure and publish this data, higher levels of performance from multiple-enclosure arrays are being achieved.

Multi-way Loudspeaker

It can be demonstrated that the useable passband of a transducer is about one decade, which represents a ten to one frequency ratio. The audible spectrum for humans extends from about 20 Hz to 20,000 Hz, which divides nicely into three decades as follows:

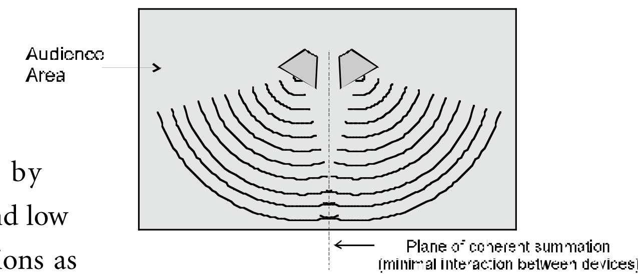
20 Hz to 200 Hz

200 Hz to 2000 Hz

2000 Hz to 20,000 Hz

The logical conclusion from this information is that full-range loudspeaker systems should be three-way, with separate sections for the low frequencies, mid frequencies, and high frequencies. Of course, this is a rule of thumb, and there are many successful systems based on single or 2-way design. Many applications do not require reproduction of the full audible passband, and the loudspeaker designer is afforded liberty to compromise on the "one device per decade" guideline. It is interesting to note that the human voice is a single-way system, and is arguably the most effective communication device in existence. Effective voice reproduction can be achieved by using a single transducer to cover the critical midrange decade, where most of the human voice frequencies fall. This assures a good communication loudspeaker system, whose bandpass can be extended by adding high and low frequency sections as needed for music reproduction.

Arrayable Loudspeaker System



Overlap between loudspeakers is minimized, and each listener seat is primarily in the pattern of a single loudspeaker system. Coverage is even, except in the narrow overlap areas.

Summary

Loudspeaker design is a process of reconciling all the parameters outlined above, and doing so in such a manner as to produce a loudspeaker system that sounds good. If it were merely a scientific process, it could be done by computers with little intervention by humans. The involvement of the human listener takes loudspeaker design beyond science and into the realm of art, because human perception has not been fully quantified by physics and mathematics. While this may change in the future, at least for now the design of good loudspeaker systems requires that listening and good judgment be an important part of the process.

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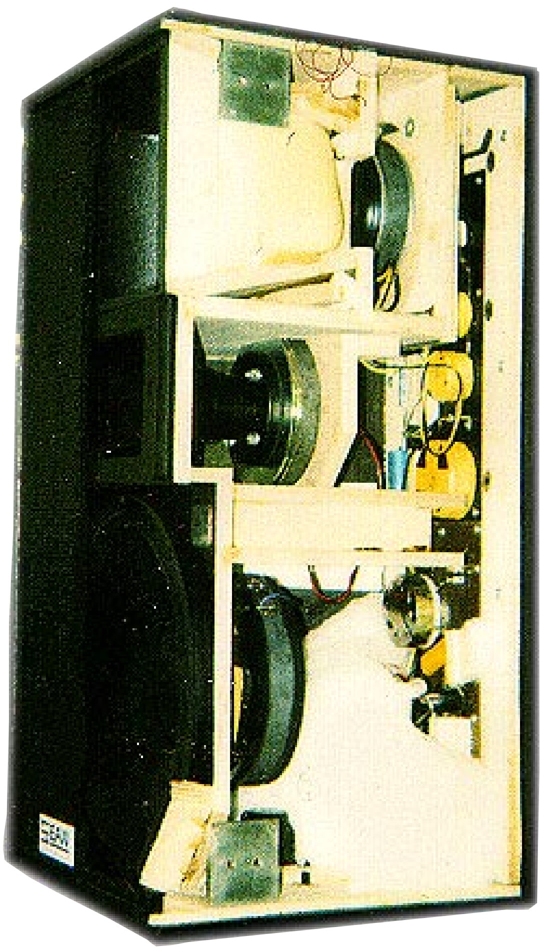
Sound System Engineering Second Edition Don and Carolyn Davis 1987 Howard W. Sams & Co.
"Compression Drivers and Horns" from the Theory and Design of Professional Loudspeakers Lab Manual Dr. Eugene Patronis June 1995



EAW's KF300i is a true three-way design, engineered along the "one decade per driver" concept.

DESIGN ISSUES IN HIGH-Q LOUDSPEAKER SYSTEMS

Kenton G. Forsythe, Executive Vice President, Engineering, EAW

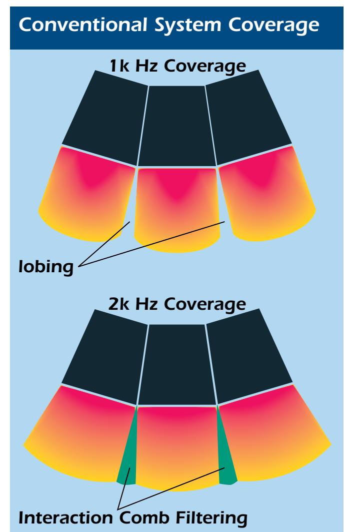
As Don & Carolyn Davis, among others, have pointed out, there exists in some respects an inherent conflict between natural-sounding speech and high Q devices. When speech is part of the program material to be amplified by a sound system, its sonic character tends to be used as the basis for evaluating the "naturalness" of the audio experience produced by the system and the room. This is only to be expected, since typical humans spend most of their time listening to the voices of other people. Since the human vocal apparatus is an inherently low-Q system, high-Q loudspeaker systems do a poor job of imitating it.

However, in large enclosed spaces the demand for intelligibility outweighs the preference for "natural" sound. As Pat Brown has explained in the previous section, the relationship between critical distance and intelligibility usually mandates the use of high-Q devices in these environments. One of our design goals at EAW is to produce loudspeaker systems that combine a high degree of intelligibility with a sound quality that listeners appreciate as "natural."

A Bit of History

The earliest attempts to build high-Q horns compounded the inherent "unnaturalness" of a narrow-beamwidth source with other departures from linearity and natural-sounding reproduction. This is not surprising, because the main purpose of those devices was to develop the maximum possible SPL from of a fragile transducer powered by a small, distortion-prone amplifier. Without sufficient SPL, the critical distance would be so short that most of the audience could not understand what was being said by lecturers on stage or actors in a film.

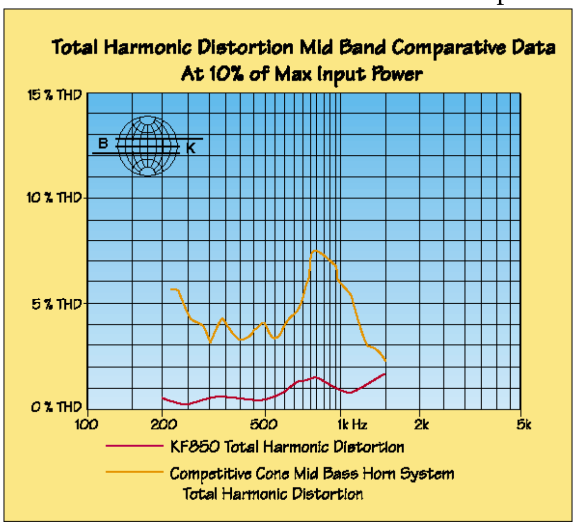
Because they were looking first and foremost for efficient "acoustical transformers," the designers of these devices attempted to maximize efficiency by focusing on parameters such as flare rate. Distortion, uneven amplitude vs. frequency caused by changes in loading factors at different frequencies, throat reflections that produced cancellations – all of these problems, if they were noticed, were



If pattern control is not consistent, high Q does not prevent destructive interaction when loudspeaker systems are arrayed adjacent to one another.

considered acceptable tradeoffs against the overriding need for maximum sensitivity.

As techniques of horn design became more sophisticated, matters improved, at least within the middle and upper octaves of the horn's operating band. Amplifiers developed more output and drivers were developed to handle more energy – this allowed designers to produce constant directivity high-frequency horns that require substantial equalization to compensate for their inherent 6 dB/octave drop-off.



Because EAW horn flares are not designed as "acoustic transformers," our engineers can focus on minimizing distortion and maximizing pattern control consistency within the device's passband.

Limitations of the Two-Way

Sound system performance was clearly improved by these horns, but no one would describe the typical "horn/box" array of the 70's and early 80's as a "natural-sounding" system. The two-way design of these systems forced the crossover point between the direct radiating woofer and the horn/driver to be rather low. most often in the 500 – 800 Hz region established by the first cinema stage systems. At those frequencies, a 15-in or 12-in woofer is already fairly directional but even "large-format" horns are not able to maintain pattern control. The result is that even when amplitude vs. frequency on axis is flat, the amount of energy radiated into the room varies with frequency. This becomes especially troublesome in large spaces, because the reverberant field is excited to different degrees at different frequencies. Thus the spectral balance of the reflected sound does not match that of the direct sound. The problem is often exacerbated by spurious radiation from the sides and rear of the exposed horns. This mismatch between direct and reflected sound degrades both intelligibility and "naturalness" at the same time.

Advantages of True Three-Way

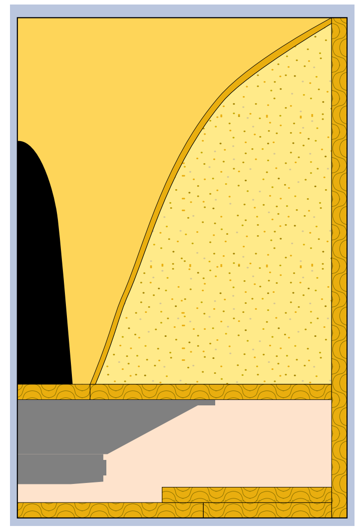
True three way loudspeaker systems use a single subsystem to cover the entire midrange. True three way designs exhibit lower distortion as compared to two-way or quasi-three-way systems in which the midrange is divided between a cone woofer and a compression driver/horn. Both devices in a two-way system are forced to operate at the extremes of their performance range - the woofer at or beyond the upper limit of its capabilities, the compression driver at or beyond its lower limit. Distortion is increased by overtaxing the drivers. Consistent pattern control is far more elusive: the woofer will tend to beam the upper midrange frequencies, whose wavelengths are factions of its large diameter radiating surface. The compression driver and its horn will not be able to maintain directivity in the crossover regions, since midrange wavelengths are too long to be controlled by typical high frequency horns.

Unique Pattern Control Devices

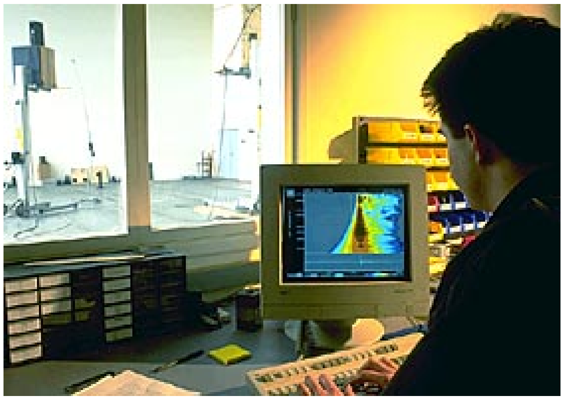
At frequencies of 5 kHz and above, achieving high Q or narrow dispersion is easy: in fact, it is more difficult to produce a wide pattern than a narrow one in the top two octaves of the audible spectrum because the wavelengths involved are very short in relation to the radiating device. A loudspeaker system with a collapsing pattern at high frequencies can be called a high Q device. However, it will not array well. Loudspeakers with collapsing high frequency dispersion but weak and inconsistent pattern control in the midrange fail to solve the sound reinforcement problems that created the demand for a high Q device in the first place.

Although superior in some ways to the "acoustic transformers" that preceded them, these systems introduce new problems. When coverage collapses at high frequencies, arrays of multiple systems produce high frequency lobes that project excessive sibilance at certain seats while leaving others with poor intelligibility and definition. Low frequency radiation is poorly controlled, so that reflective surfaces are excessively excited in the midbass region. Direct sound output is not consistent with the "muddy" reverberant field, further confusing the ears of listeners who are trying to understand spoken or sung words, or to follow musical rhythms over the midbass "buildup" in the room.

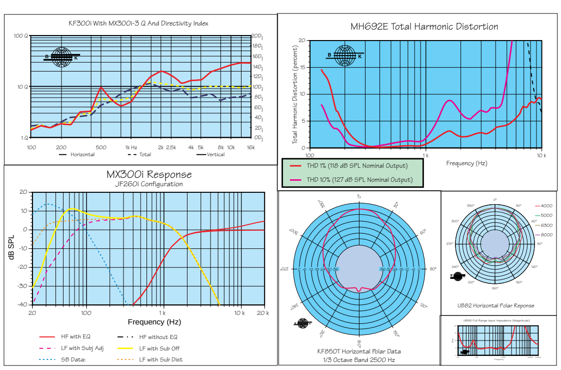
EAW engineers rely on two types of exponential horn flares to provide the consistent wideband pattern control that is the foundation of Virtual Array Technology. High frequency horns are molded of fiberglass. A proprietary collapsible mold technique al-



EAW's proprietary construction techniques combine high density polyurethane foam and thin, flexible sheets of laminated birch to produce large complex horn flares that are acoustically rigid.



An automated test and measurement system is coupled to our multiplatform computer network to integrate actual performance data into the design process.



A wide range of data, defining all aspects of system performance, can be acquired quickly and accurately.

lows these horns to be designed with an undercut throat that minimizes the tradeoffs between distortion (which demands a wide throat) and consistent coverage at high frequencies (which requires a narrow throat).

In the midbass region, fiberglass horn flares resonate within their passband, introducing output anomalies that are difficult to tame. The problems only become more severe as the size of the horn mouth and the associated structure increase. To avoid these problems, EAW builds large midbass horns using proprietary construction techniques to produce complex flares that are acoustically rigid. Numerically controlled routers cut the complex horn flare into the top and bottom of the enclosure. Thin sheets of laminated birch are glued into the tracks and reinforced with heavy wood formers. High density polyurethane foam is injected into the space between the flare walls and the enclosure sides. As the material expands and hardens, it reinforces the structure while damping any passband resonances out of the system.

Iterative Design Process

The design of these horn flares is guided by a combination of automated test and measurement equipment and years of experience. Real world acoustic performance rather than a mathematical construct is the goal of this process. We enjoy one overriding advantage over our predecessors – the availability of both extremely high power amplifiers and transducers robust enough to turn thousands of electrical Watts into tens of acoustic Watts. R&D by both amplifier and driver manufacturers continues to pack more power into fewer rack spaces, and to reach higher levels of mechanical and thermal reliability. This in turn affords us the luxury of designing horns as waveguides, not as “acoustic transformers.” We don’t bother to measure flare rates, because we are confident that the final system will reach acceptably high sensitivity levels no matter how the horn flares are designed. Thus our designs for both foam-reinforced wood veneer midbass horns and fiberglass or molded plastic high frequency flares can focus on constant directivity and low distortion rather than efficiency factors. Prototypes are molded of fiberglass or constructed using the foam-reinforced wood laminate technique. Our in-house computer network and advanced test equipment allow design engineers to gather real world performance on horn prototypes quickly and accurately. Agile manufacturing speeds the construction of further prototype generations until the desired performance specifications are achieved.

High Q Design Evolution

The Performance/Engineered KF1000 VA System, introduced in 1990, was designed to function as the top row of large KF850 arrays: its first use was in the system assembled by touring sound company TASCO for a worldwide arena tour by Kiss. KF1000's are used today by large touring companies such as Concert Sound of London and A-1 Audio of Los Angeles to project high frequencies into the upper rows of indoor arenas.

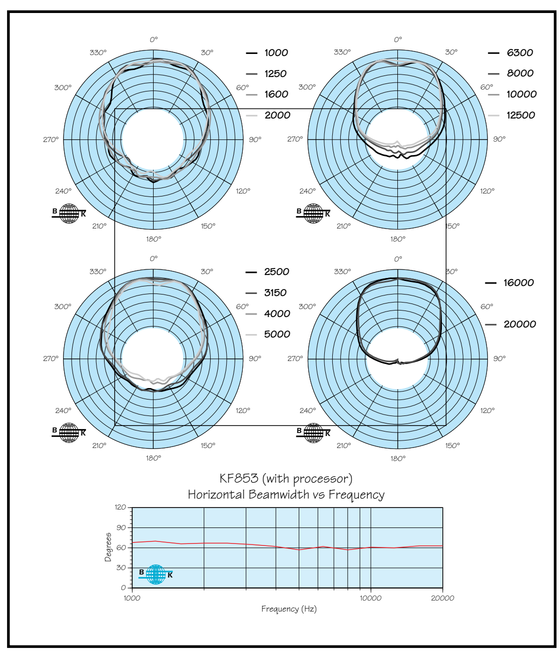
Permanent installations for large crowds lend themselves to EAW's Project/Engineering process, in which loudspeaker systems are developed for a particular site. The MH242 midbass horn is a Project/Engineered system developed about the same time as the KF1000 for Fiesta Texas, a theme park near San Antonio. Installed at the top of a large cliff, MH242's are used nightly in the park's pyrotechnic spectacular.

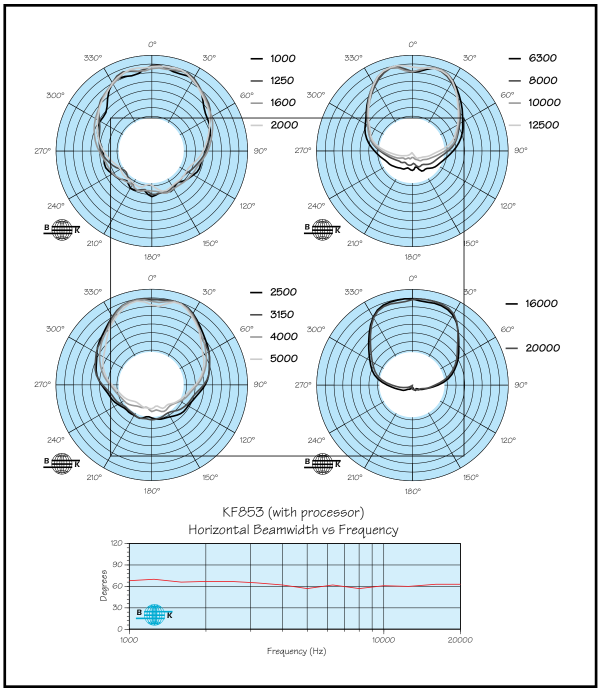
The MH242 midbass horn became the center of the AS943, another Project/Engineered system that is installed at a leading theme park in Orlando, Florida. The AS943 and the MH242 are practical demonstrations of the effectiveness of high Q devices in outdoor applications.

Recent revisions of the MH242 horn seem to indicate that the depth of the flare as well as the mouth size is crucial for maintaining consistent pattern control over a wide operating band. In addition, our measurements show that deeper horns tend to have a steeper drop-off of emitted energy outside the -6 dB points. These systems will have the same nominal Q as

The KF853 is engineered to project midrange and high frequencies over long distances.

KF853 polar response measurements show consistent High Q performance.





Polar response of two KF853's shows minimal interaction between adjacent enclosures.

any system with the same -6 dB points. Their actual measured Q, however, is much higher. In fact, the revised MH242 horn has a Q of 20 above 1250 Hz. To our knowledge, this is the first horn to achieve this kind of performance. As an acoustic transformer it is relatively inefficient. However, we expect highly superior results in large indoor installations using this horn.

KF853 High Q Stadium Array System

The KF853 illustrates this point well. The KF853 maintains the enclosure angles of the KF850 for easy arrayability, but is six inches deeper than the KF1000 it replaces. The longer enclosure accommodates a deeper midbass horn and a newly designed fiberglass high frequency flare that is slightly larger at the mouth and also deeper than its predecessor. EAW's unique iterative design process and proprietary construction techniques have produced a high Q device that also behaves well in arrays. Test data measured by our Brüel & Kjær 2012 Audio Analyzer show that the KF853 is a real breakthrough in high Q design and performance. Beamwidth and directivity are remarkably consistent. Dispersion is 30° x 40° from 700 Hz - 18 kHz. Horizontal array polars show effective acoustic coupling with minimal interaction between adjacent devices.

KF860 Virtual Line Array

The KF860 applies the techniques of high Q design in a new direction. The KF860 is designed to form vertical line arrays. The KF860 Vertical Stadium Array System is only 18 inches high, yet has the same driver complement as two KF850 Stadium Array Systems. Each system covers an extremely narrow arc in the vertical plane – less than 20°. This is necessary to maintain relatively uniform listening distances within the coverage area of each system in the array. This combination of high output capability and narrow dispersion allows true long throw performance from a system with a very low profile. For example, at the opening and lcosing ceremonies of the Special Olympics 1995 Summer Games, two KF860's were used to cover vertical sections of the Yale Bowl, from the front rows to the top of the upper tier.

Ongoing High Q Evolution

Although its design was constrained by the need to pack, fly and array with Stadium Array systems such as the KF850E and KF852/BH852, the KF853 can be considered a true high-Q loudspeaker system. Systems designed for permanent installation can operate without these restrictions. EAW's MH Series of Mid/High Virtual Array Modules have proven that pattern control bandwidth can be effectively extended using larger horns, without compromising arrayability and ease of installation. The AS Series of Application/Engineered systems build on MH Series and KF Series horn designs to create integrated full range systems that address the problems of specific applications. Both of these product ranges will be expanded in the near future with true high Q systems that extend the design breakthroughs of the KF853 in new directions.

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