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VIRTUAL ARRAY LOUDSPEAKER SYSTEMS: DESIGN CONSIDERATIONS FOR "ARRAYABLE" LOUDSPEAKER SYSTEMS

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1) INTRODUCTION

Recent years have seen a plethora of new loudspeaker systems extolling their virtues in being used in an array or creating an array. What are the considerations that go into designing a loudspeaker so that it is suitable for such an application and why do so many of the current designs fall short of this goal?

"Arrayable" loudspeakers have become very much in vogue in recent years. The implied design concept is very useful - that of using several wide-bandwidth systems to be placed in close physical proximity to each other so as to act as one larger system. The design advantages are very real: high output, uniform coverage to any desired angle, good intelligibility to every seat in the house, which in most instances is a highly reverberant space. How well are these design goals really achieved and what are the design considerations that do lead to good realization of them?

2) CHARACTERISTICS OF "GOOD" SPEAKERS

We can start by looking at the factors that make a loudspeaker sound "good". Such a speaker is readily identified by its "natural" sound character. Musical instruments sound as they do in the hands of the musician, voices sound clear, distinct, and easily understood. The full dynamics of the reproduced instrument come through without spurious noise or distortion. Such speakers are apparent upon only cursory listening. In the realm of the hi-fi shops, there are many contenders but the realm of high-level sound reinforcement has few pretenders. What then are the characteristics that provide for natural sound reproduction?

The most important characteristic for a loudspeaker must be its amplitude vs frequency response. This is the fundamental measurement of a loudspeakers quality as an audio transducer. Small departures from linearity , particularly if over bandwidths of an octave or more, are readily apparent as creating a certain sound "character" associated with that speaker. This "character," be it "warmth", "fullness", "brightness" or any of the myriad of terms used to describe the sound of a speaker, only serve to indicate the degree to which the speaker departs from its function of a transducer. Needless to say, large departures from linearity create readily apparent objectionable sound character for the speaker.

Recent emphasis on time and phase performance of a speaker only reinforce the importance of response linearity. Time offsets in multi-way systems invariably produce response anomalies. These usually evidence themselves as a "peak and dip", indicating phase cancellation and reinforcement in the transition from one device to another. Although these anomalies usually are apparent in the axial response characteristic of the speaker, they may only jump out in looking at the response moving off the central axis. It is then the response in the "off-axis" direction that leads to a secondary, but vitally important, factor in the sound of the loudspeaker.

The off-axis response is as vital to the sound of a speaker as the axial (on-axis) response. Ideally, it must track the on-axis response over the full bandwidth of the speaker. The degree to which it does so may be called the "power response" of the speaker. This term has come to be used with increasing frequency in recent years, as the importance of it has become more fully recognized. Basically, the power response is a measure of the total radiated energy of the speaker as a function of frequency. The amount by which it is less than the axial response is a measure of the directionality of the speaker. A very wide coverage speaker, such as may be found in the home or, to an increasing extent, in studios, will exhibit a power response only slightly less than the axial response. On the other hand, a speaker intended for use in reverberant environments, or to achieve long "projection" of sound, may have the power response 10 or more db below the axial response.

The power response is as vital to the sound character of the speaker as the axial response, as it defines the amount of energy coming back to the listener from the reflective surfaces within a room.
A "hollow" or "harsh" sound character may often be attributed to variations in the power response.
Unlike the axial response, it cannot be corrected by electronic equalization but must be designed in to the speaker from the beginning. This is often a very difficult exercise, leading to many tradeoffs on the part of the designed. One aspect of these tradeoffs may be noted - that the power response should depart from linearity only in a decreasing direction as frequency rises. Axial response peaks can be addressed with filters but power response peaks cannot, and are usually much more objectionable.

The distortion characteristic of the speaker is the third most vital aspect. It is usually evidence in non-linearities in the mechanical systems of the loudspeaker, particularly in diaphragm and surround break-up modes. These tend to define the usable bandwidth of the device, thus leading to the inevitability of multi-band loudspeaker systems being required to cover a full audio spectrum. It will later be seen how these distortion characteristics affect the design of "arrayable" speaker systems.

3) REQUIREMENTS FOR LINEAR POWER RESPONSE

Uniform power response is relatively easy to achieve at high frequencies. The short wavelengths make it possible to control of the directional characteristics of the speaker through the use of waveguides. These are usually manifest in the form of horns, which current designers have well mastered techniques for maintaining uniform coverage over a fairly wide frequency band. The extent of this band is determined by the physical size constraints of the horn. While a device providing good pattern control to perhaps 1500 hz may have reasonable size limits of 12-20 inches in frontal dimensions, depending on the desired coverage angle, one that offers the same control to an octave lower must increase in frontal size by a factor of four.

Even if we assume that this size ballooning is not a problem, we run into the issue of diaphragm excursion required to produce the lower frequencies. As excursion increases, so will distortion, so that the sound produced will no longer be of sufficient quality or level. We can address this issue by increasing the size of the diaphragm, but this will lead to reducing the high frequency response. Some balance must be struck in this process.

The obvious solution to this dilemma is to create a crossover transition to a diaphragm of significantly larger size so that usable frequency response may be extended downward by a useful amount. The "useful amount" might be a minimum of one and a half octaves, as there may not be much point in creating a multi-way system for a lesser bandwidth extension. In order to maintain our requirement of linear power response, the new device must exhibit the same power response (or coverage angle) at crossover. Furthermore it should maintain this coverage to its lower limit. The bandwidth for full pattern control might be considered as three octaves wide. Increased bandwidths will run into the distortion limits that made us create a multi-way system in the first place. This will require another waveguide of significantly larger dimensions than used for the high frequencies. Invariably, limits will have to be placed on the either the size of this device or the bandwidth we will require it to operate over. There will be some instances where we can accept the 60" x 60" dimension that may be required to maintain uniform coverage to near 200 Hz. These instances will be relatively infrequent however.

4) INTERFERENCE PATTERNS

One obvious means of handling the size problem has been to employ a direct-radiator woofer (or midrange) crossing over to the high frequency waveguide at a frequency where the power response of the devices match. There are numerous examples of speakers with a woofer, of 12", 15" or even now 18" dimensions, crossing to a horn/driver at some frequency where the power response of the devices presumably match. Many of these are designed with considerable success in this regard. Unfortunately, there are also many designs that are less than successful. This design approach does violate the fundamental requirement for uniform power response over as wide a band as possible however. It furthermore introduces an entirely new set of problems when speakers of this design are used together to create an array.

Let us consider the problem starting from the low end of the frequency scale. Assuming that we are dealing with direct-radiator cone loudspeakers, the radiation pattern is basically omnidirectional at low frequencies. These may be considered as frequencies whose wavelength is considerably longer, by a factor of at least four, than the diameter of the speaker creating them. This works out to approximately 350 hz for a 12" woofer or 250 hz for a 15" woofer. Above these frequencies the cones become progressively more directional, until presumably they reach a pattern that will match the high frequency device. Directivity then is a linear function of frequency and the slope of the directivity curve is linear, not the curve itself as would be required for uniform power response.

If we then take this system and place it adjacent to another identical system, in an effort to provide coverage for a wider area than allowed by a single system, we have created a new problem. The multiple cone loudspeaker now act as a single larger radiating surface. Unfortunately, this surface now has multiple point sources of sound over much of their bandwidth. Only those frequencies with a wavelength longer than four time the spacing of the cones will be reasonably coherent, in the sense that they will sum together uniformly. Above this frequency, which is perhaps 200 hz in the case of two 12" woofers, a series of interference patterns is created. This may be an interesting phaser effect but is a significant departure from our requirement for linear frequency response. If the devices are in the same plane, the interference patterns will become deeper with increasing frequency and will create sever response aberrations that are reflected clearly in the "sound" of the speaker. This effect may be lessened by placing the speakers on planes offset to each other. As the cones become more directional, the interference effects will become lessened as there will be less energy in the "off-axis" area to create the interference patterns.

This effect can be used to advantage by applying frequency tapering to certain elements in the array. If the low frequency response of certain elements is reduced, the interference patterns these elements create will be lessened. Adjusting the time relationship of these elements may also be useful to adjust the directionality of the array to optimize its performance for a certain application. Both of these techniques have been used with great success in the design of home loudspeakers. They do not lend themselves to creating arrays of uniformly defined, incrementally increasing coverage however.
Whenever elements are added to an array to increase the area that it covers, the interference pattern effects will take over. These will manifest themselves as increasingly ragged frequency response plots, as evidenced in several papers that have been done on the effects of combining direct-radiator speakers into arrays. They will also manifest themselves as increasing loss of articulation, clarity, and definition in the sound character of the array.

5) THE WAVEGUIDE SOLUTION

If we now go back to the fundamental requirement of uniform power response, we can find a way of lessening the interference effects. Applying waveguide (horn) design, to as low a frequency as physically practical, gives us a means of creating arrays that exhibit less comb-filtering effects and thus improved clarity, articulation and definition.The requirement for the waveguide then becomes one of precisely defined coverage over some useful angle and rapid attenuation of response outside this angle. It then is possible to place these waveguides in close physical proximity, and splay their planes relative to each other, in a manner that will allow coherent summing between them and minimal interference effects outside the coverage angle of a device. The wider the coverage of the devices, the greater must be the splay angle between them. The greater also must be the total energy radiated by each device. For large concert events, each device must be allowed to cover only a small sector of space so that sufficient energy may be radiated into that space.

In creating arrays that make use of this concept, it is most useful to design each element as a wide-bandwidth device and allow that element to cover its unique sector of space. To the extent that different frequency bands emanate from different sectors of space, there will be a lessening of the design goals of the system - clarity, articulation, definition, etc. A useful rule-of thumb for this bandwidth may be the vocal range, from the 250 hz octave center up. This will require good pattern control to 200 hz. The success of the array will be based on the degree to which this goal is achieved.

The upper frequency limit of the array deserves consideration. Too often the vocal range has been slighted in the upward direction. The overtones of speech are vital in achieving good articulation. The array must then avoid beaming effects with increasing frequency and each element must have extended high frequency response. A reasonable minimal upper limit might be 10k hz, as this will allow full bandwidth to the 8k hz octave band. A desirable upper limit would be 16k hz. Musical instruments will then come through with full clarity.

The size requirement for each element of the array will depend on the coverage angle of that element. Long coverage distances and reverberant spaces will require devices of limited coverage. Using these elements as a universal solution, while possible, will result in arrays of ungainly size for smaller spaces. It therefore is useful to have a range of coverage elements available to the designer, several of which may be used in one system. The "nearfield" coverage can be done with wider coverage devices than may be required for the "farfield" areas. The main idea to keep in mind is that one design cannot do all jobs.

6) GENERALIZED DESIGN SOLUTIONS

The design requirements for effective array elements can then be readily identified:

1) Wide bandwidth for each array element with uniform power response. This will provide the basic building block for a coherent array.

2) Limited bandwidth for each transducer in the array element. The considerations for uniform axial and power response and low distortion (and high reliability) tend to limit the bandwidth of a transducer element to three octaves. This then necessitates three-way elements to achieve 40-16k hz response. As a minimum, the array element should be two-way operating from 200-250 hz upwards. The lower frequencies can be handled by woofer elements placed adjacent to the main array elements.

3) Packaging becomes an important but neglected part of the design solution. Each element must be designed to work in close physical proximity to its neighbor. Spaces left between array elements should be filled to eliminate diffraction phase effects. Spurious vibrations, or rear-radiating energy, must be contained within the cabinet construction. This invariably leads to placing the horns with enclosures that may in turn be placed adjacent to each other.

With careful design, these individual elements may be joined physically to create one large speaker system with the directionality and response appropriate to provide high fidelity sound, in the literal sense of the word, to all areas of interest within the facility.