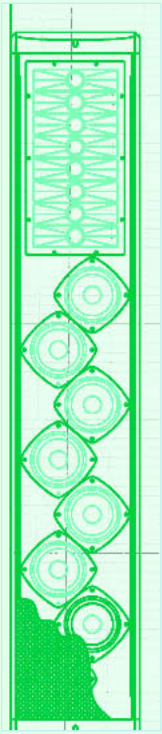

# HISTORICAL PERSPECTIVES AND TECHNOLOGY OVERVIEW OF LOUDSPEAKERS FOR SOUND REINFORCEMENT*

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## INTRODUCTION

Horns and direct-radiating systems have provided the basis for sound reinforcement for more than a century. Both technologies have benefited from engineering and manufacturing improvements as well as demands for pushing the performance envelope. Trends of fashion have often intersected with engineering development, economics, and even marketplace opportunism. A survey tutorial of the significant developments in transduction, signal transmission, and system synthesis is presented here and discussed in historical perspective.

We begin with an overview of sound reinforcement and the technologies that have supported it. This is followed by more detailed technical discussions of both direct radiating and horn systems, leading to a discussion of modern loudspeaker array techniques. The presentation ends with a comprehensive bibliography.

## HISTORICAL PERSPECTIVES

In the early days of transducer development, horn systems offered the only means possible for achieving suitable levels for speech reinforcement. Early power amplifiers were limited to about 10 watts output capability, and horn-driver efficiencies on the order of 20% to 30% were necessary to reach the desired sound pressure levels.

The direct field level referred to a distance of one meter produced by one acoustic watt radiated omnidirectionally from a point source in free space is 109 dB $L_P$ [1, p. 314]. If we use a 10-watt amplifier with a horn–driver combination that is 30% efficient, we can produce three acoustical watts. If the horn has a directivity index (DI) on axis of, say, 10 dB, then we can increase that level to:

$$\text{Level (re 1 meter)} = 109 + 10 \log (3) + \text{DI} = 124 \text{ dB } L_P$$

At a more realistic listening distance of 10 meters, the level would be, by inverse square relationship, 20 dB lower, or 104 dB. If wider coverage is needed, more horns can be added and splayed as required.

There is little documentation of early examples of general speech reinforcement, and that art progressed fairly slowly [9]. The first example of large-scale sound reinforcement occurred on Christmas Eve, 1915, when E. S. Pridham, cofounder of the Magnavox company, played Christmas carols for an audience of 50,000 using Pridham-Jensen rocking armature transducers connected to phonograph horns [12]. Western Electric set up a public address system capable of addressing 12,000 persons through 18 loudspeakers in 1916 [24, p. 24].

The first distributed system was employed in 1919; 113 balanced armature driving units mounted on long horns were strung along New York City's Park Avenue "Victory Way" as a part of a Victory Bond sale [24, p. 25] [2], as shown in Fig. 1. The first successful indoor use of a public address system was at the 1920 Chicago Republican Convention, which also employed the first central cluster configuration [24, p. 25], as shown in Fig. 2. On March 4, 1921, President Harding's inauguration was amplified [24, p. 24], and on November 11, 1921, President Harding's address in Arlington, Virginia, was transmitted by Western Electric, using Edgerton's 1918 design of four-air-gap balanced-armature units. For the first time 150,000 people, at Madison Square Garden in New York, in the adjoining park, and in the Civic Auditorium in San Francisco, simultaneously listened to a person speaking [2].

It was the cinema that paved the way

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Fig. 1. First use of a distributed sound system, New York, Park Avenue (1919). Note trumpet horns in upper left hung above crowd aiming downward. (Photo from Lester Cowan *Recording Sound for Motion Pictures*, McGraw Hill, New York, 1931.)

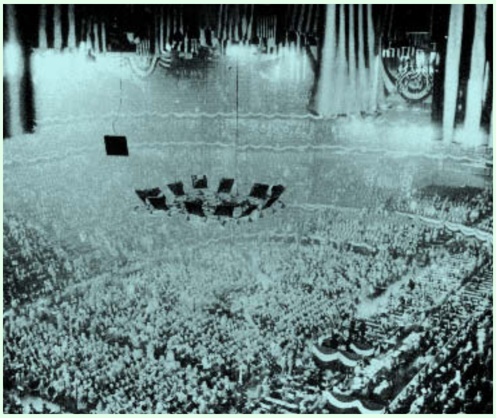


Fig. 2. First central array sound system, 1920 Chicago Republican Convention. (Photo from Frederick Thrasher, ed., *Okay for Sound … How the Screen Found its Voice*, Duell, Sloan, and Pearce, New York, 1946.)

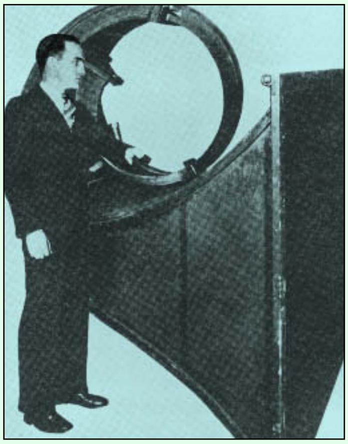


Fig. 3. Western Electric "Roxy" horn type 12A with 555W dynamic driver. (Photo, Cowan, p. 329.)

for rapid development of professional sound reproduction. Talking pictures required more than speech intelligibility, however. The single horns of the day, as shown in Fig. 3, were limited in response to the range from about 125 Hz to 3 kHz. While this was adequate for speech, music required greater bandwidth, especially at lower frequencies [7].

The first widely accepted cinema system used a two-way approach. A multicellular high-frequency horn with a compression driver was coupled to a cone-driven, folded low-frequency horn assembly [3] as shown in Fig. 4. Its high level of performance and approval by the Academy of Motion Picture Arts and Sciences, in the form of a technical achievement award, led to its acceptance as an industry standard [11].

The next step in low frequency (LF) response was to employ multiple large-diameter direct radiators in some sort of "directional baffle" that could both load the drivers for increased efficiency and give them added forward directionality [20]. The best LF direct-radiating transducers of the day were about 10 dB lower in efficiency than horn systems, but their use was mandated due to the size and complexity of traditional bass horns. When employed in multiples, they could provide sufficient level and coverage in large theaters. In time, the various directional baffles evolved into the familiar front-loaded quasi-horns that became universally identified with the cinema [16], as shown in Fig. 5. Field coil energized transducers also gave way to permanent magnets following WWII [15].

The LF transducers chosen for this application had relatively small moving masses and high $(Bl)^2/R_E$ ratios. This enabled them to efficiently handle the acoustical load transformed by the horn and maintain good response up to about 400 Hz, at which point the high frequency (HF) horn took over. The HF horn itself was undergoing modifications aimed at improving its directional response. RCA favored the radial horn, while Bell Laboratories favored the multicellular device [27]. Bell Laboratories later pioneered the use of the acoustical lens in similar applications [8, 14].

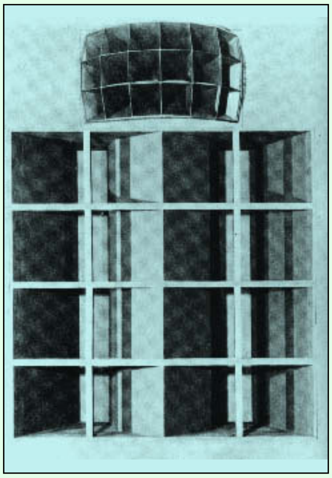


Fig.4. Shearer MGM horn system, which won the Academy of Motion Picture Arts and Sciences scientific award in 1936. Contributors to this program included Douglas Shearer, John Hilliard, James B. Lansing, Harry Olson, John Volkmann, and William Snow. (Photo from Research Council of AMPAS, *Motion Picture Sound Engineering*, Van Nostrand, New York, 1938.)

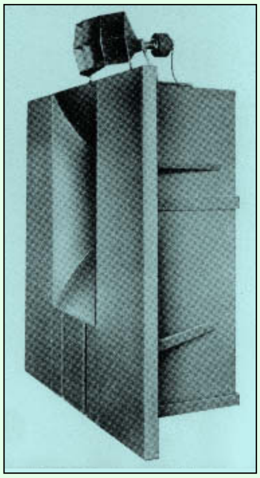


Fig.5. The Hilliard and Lansing designed Altec-Lansing A-4 "Voice of the Theatre" system, 1945. Utilizing a short LF straight exponential horn with an acoustical path length equal to that of the HF horn to avoid delay between the LF and HF sections. (Photo, Altec-Lansing)

By the mid-1940s, HF horns had established their primacy in the frequency range above 500 Hz, while hybrid LF horns, which relied on reflex loading of cones below about 100 Hz, dominated the range below 500 Hz. These families of components formed the basis of postwar sound reinforcement activities and maintained that position for nearly three decades.

By the early 1970s available amplifier power, which had been steadily rising through the introduction of consumer high fidelity during the mid-1950s and 1960s, reached the point where beneficial tradeoffs could be made among the three "eternal" loudspeaker variables of size, efficiency, and LF bandwidth extension. As early as the mid-1950s, consumers began to enjoy relatively small sealed systems that provided substantial LF output [25, 26]. Additional analysis and synthesis of ported enclosure design by Thiele and Small [23], following the earlier work of Locanthi [17], Novak [19], and Benson [31], led to the general

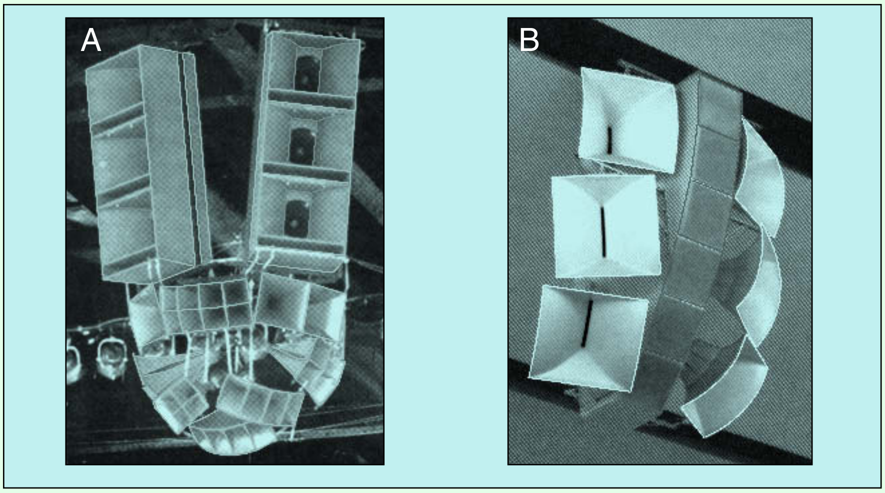



Fig. 6. Large systems for speech reproduction. Design using pre-1970 components, multicellular exponential horns with ported LF horns (A, *J. Audio Eng. Soc.*, vol. 20, pp. 571); design using post-1970 components, constant directivity HF horns and ported LF direct radiators (B, courtesy KMK Ltd.)

adoption of ported enclosures as a LF building block in professional sound system design.

Through the use of Thiele-Small driver parameters, LF systems could be rationally designed for best response and enclosure size. Gone were the days of cutting and trying, and also gone were the days of poorly designed LF transducers. Thiele-Small analysis pointed to the need for identifying the right cone mass, resonance, excursion capability, Bl product, enclosure volume, and tuning to achieve a targeted performance goal.

During the 1970s and early 1980s ported LF systems became the basis for most sound reinforcement LF design. For many music and speech applications the horn HF crossover frequency was raised from 500 Hz to 800 Hz and higher, with cone drivers filling in the midrange [18]. This change from the traditional two-way philosophy was brought on by modern program requirements for generating increased level. Higher power handling transducers were substituted in the LF horns, but these new drivers forced the HF power bandwidth of the horn downward. With the crossover frequency raised to improve HF reliability, the result was uniform on-axis response, but grossly uneven power response [6].

By the mid-1970s ported LF systems were poised to take a commanding lead in the design of speech reinforcement systems. Fig. 6 shows examples of the older approach and the newer one. The system shown in Fig. 6A makes use of the previous generation of hardware (theater-type LF hybrid horns and HF multicellular horns), while the system shown in Fig. 6B makes use of ported LF systems and modern HF horn hardware. In terms of LF response and power handling, both systems have six LF drivers. With horn loading, the output capability of the system shown in Fig. 6A will be higher in the 100 to 200 Hz range than the system shown in Fig. 6B. However, the LF system in Fig. 6B is smaller and has been designed for flat power bandwidth over its operating spectrum in the range from 40 to 400 Hz. It uses high-powered drivers and can accommodate approximately four times the amplifier power as the system Fig. 6A, but by the mid-1970s the cost per watt of amplifier power had diminished significantly.

For large-scale music reinforcement there were other considerations. LF horn systems had developed along lines unique to the music industry, since fairly high mid-bass directivity was a prime concern. The older radial horns were likewise favored for their directionality, which increased at HF. Typical usage is shown in Fig. 7.

The conservative motion picture industry continued with two-way solutions but eventually traded in its hybrid bass horns for standard ported systems and retired the old multicellular HF horns of the 1930s [5]. The concept of flat power response in system design stated that not only uniform direct arrival sound was important; reflected sound (proportional to the total radiated power of the system) should also be uniform. This goal could be more easily met with ported LF systems, along with uniform coverage HF horns. Hilliard [10] was well ahead of his time in bringing this concept to the attention

Fig. 7. Typical early-1970s horn-loaded system using ported LF horns, 90° radial horns with large-format drivers, and 90° radial horns with small-format drivers for augmenting the upper frequencies. A narrow coverage horn with a large format driver is used for long-throw coverage. (Photo, JBL)

of the film industry, but his attempts at commercialization failed.

The well-engineered ported LF system of the 1970s established its dominance at the low end of the frequency spectrum. The combination of low-cost power amplification, low-distortion, high-power LF transducer development, and enclosure simplicity has given the ported system an advantage at low frequencies that all but the very largest horn systems could never match.

Keele [13] made an instructive comparison between LF horns and direct radiators, pointing out the virtual parity between them. For the same LF cutoff, the horn will have highest efficiency, a large complex enclosure, and will require a small number of drive units. By comparison, a direct radiator multiple-driver vented box will have moderate efficiency, a small, relatively simple enclosure, a large number of drivers, and higher power handling capability (because of the multiple drivers). In Keele's words, "This roughly means that if one has a lot of space, not much money to spend on drivers and amplifiers, and lots of cheap labor—build a horn system. If labor is not cheap, and you don't have much space, and if you can afford drivers and amplifier power—build a direct radiator system." In essence, the costs of labor and lumber had far outstripped those of drivers and watts.

Through the late 1980s and 1990s, however, the professional sound industry saw a return to horn systems for covering the range down to about 300 Hz, especially for large-scale speech reinforcement. The reasons have to do not with efficiency per se but rather directional control. These new systems have taken the form of large format compression drivers, or cone transducers designed for horn–driver applications, and large horns optimized for the range from 100 to 300 Hz to 1 to 3 kHz, with ported LF systems handling the range below. Rapid flare HF horns are now employed, and their distortion characteristics, level for level, may be up to 10 dB lower than the older hardware [4]. New digital methods of signal control have made multiway systems more acceptable than they were in the days of passive dividing networks, through time alignment and steep crossover slopes.

As we continue our discussion of direct radiators and horns, we will present a detailed analysis of the engineering fundamentals of both methods of transduction. This discussion will cover basic operating principles, directional control, and distortion mechanisms associated with each method. We will follow this with a discussion of array concepts encompassing both kinds of transducers.

## DIRECT RADIATORS

### Early Development

Ernst Werner Siemens' 1874 U.S. Patent 149,797 was prophetic; he described in detail a radial magnet structure in which a coil of wire was placed. The coil was connected with a radiating surface that Siemens described as the frustum of a cone. He had literally invented the cone loudspeaker—with nothing to play over it except dc tran-

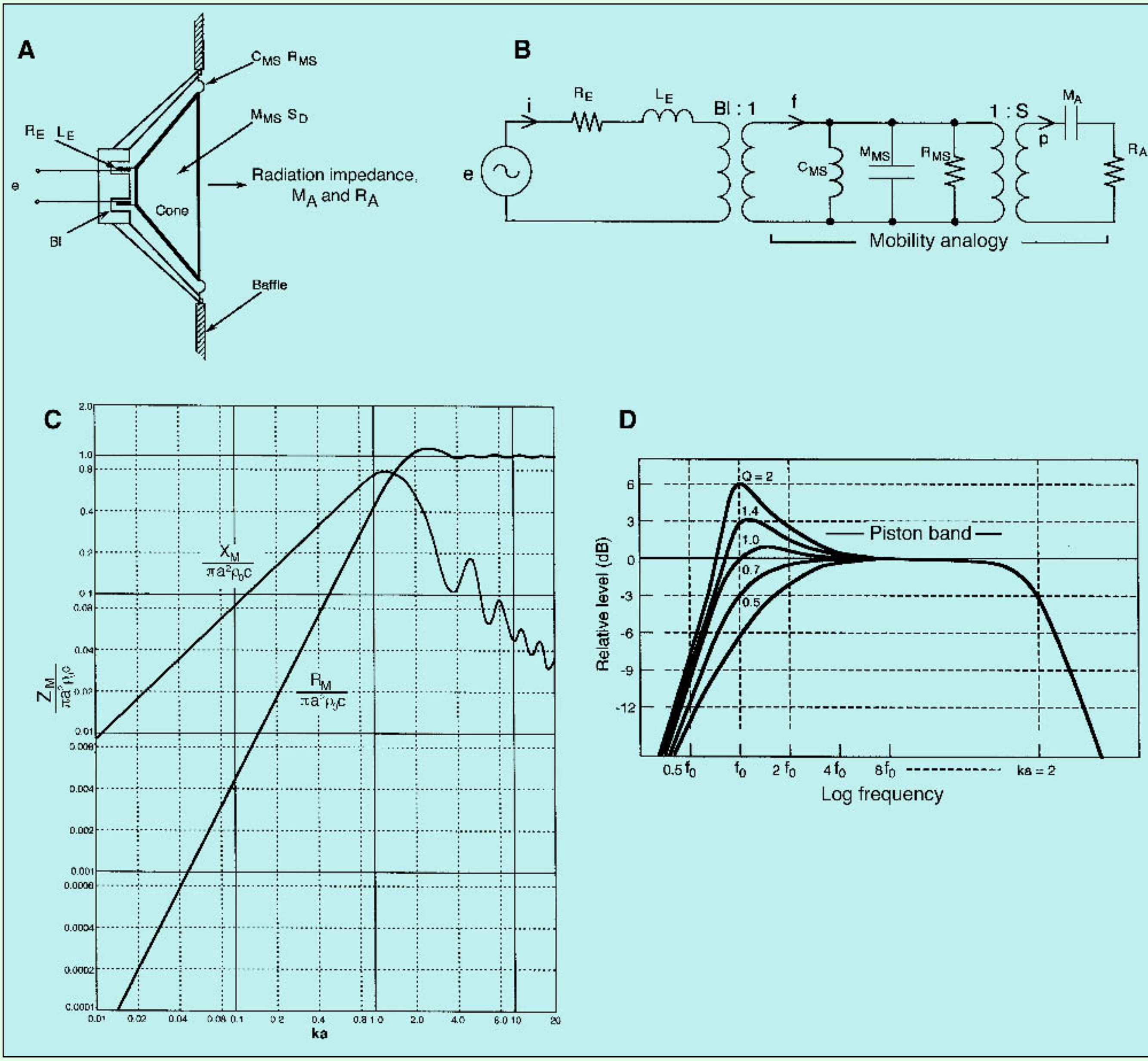



Fig. 8. The cone driver. Section view (A); equivalent circuit (B); radiation impedance of a cone in a large wall (C); power response of system (D).

sients and other telegraphic signals. He remarked at the time that it could be used "for moving visible and audible signals."

Half a century later in 1925, Rice and Kellogg of General Electric described "a new hornless loudspeaker" that resembled that of Siemens—a similarity that prompted Rice to say: "The ancients have stolen our inventions!" [37].

The key difference in the Rice and Kellogg design was the adjustment of mechanical parameters so that the fundamental resonance of the moving system took place at a lower frequency than that at which the cone's radiation impedance had become uniform. Over this range, the motion of the cone was mass controlled, and the cone looked into a rising radiation impedance. This in effect provided a significant frequency region of flat power response for the design. Details of this are shown in Fig. 8.

### Region of Flat Power Response

Fig. 8A shows a section view of the cone loudspeaker with all electrical, mechanical, and acoustical parameters labeled. The equivalent circuit is shown in Fig. 8B; here the mechanical and acoustical parameters are shown in the mobility analogy.

When mounted in a large baffle, the moving system looks into a complex acoustical load as shown in Fig. 8C. The resistive component rises with frequency to approximately $ka = 2$, above which point it is essentially constant ($ka$ is equal to cone circumference divided by wavelength, or, $2\pi a/\lambda$). System response is shown in Fig. 8D, and system efficiency over the so-called piston band is given [49] as:

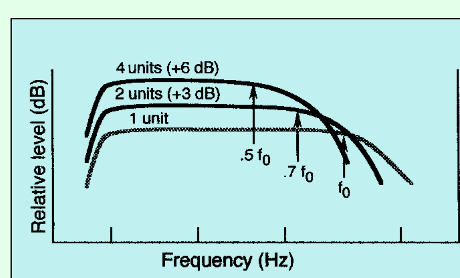



Fig. 9. Illustration of mutual coupling of LF drivers.

$$\eta = [\rho_0 (Bl)^2 S_D{}^2 / R_E] / 2\pi c M_{MS}{}^2 \quad (1)$$

where $\rho_0$ is the density of air (kg/m$^3$), $(Bl)^2/R_E$ is the electromechanical coupling coefficient (N$^2$/W), c is the speed of sound (m/s), $S_D{}^2$ is the area of the cone, and $M_{MS}$ is the mass of the moving system (kg).

The larger the coupling coefficient, the lower the resonant $Q$ at $f_0$ and the higher the piston band efficiency will be. Likewise, the higher the $Q$ at $f_0$, the lower the piston band efficiency. Depending on the application, both kinds of response may be useful to the design engineer.

It can easily be seen that, for maximum extension of the piston band, the lower $f_0$ must be, and the lower the system efficiency will be. The efficiency-bandwidth product, for a given cone diameter and coupling coefficient, thus tends to be constant over a relatively large range.

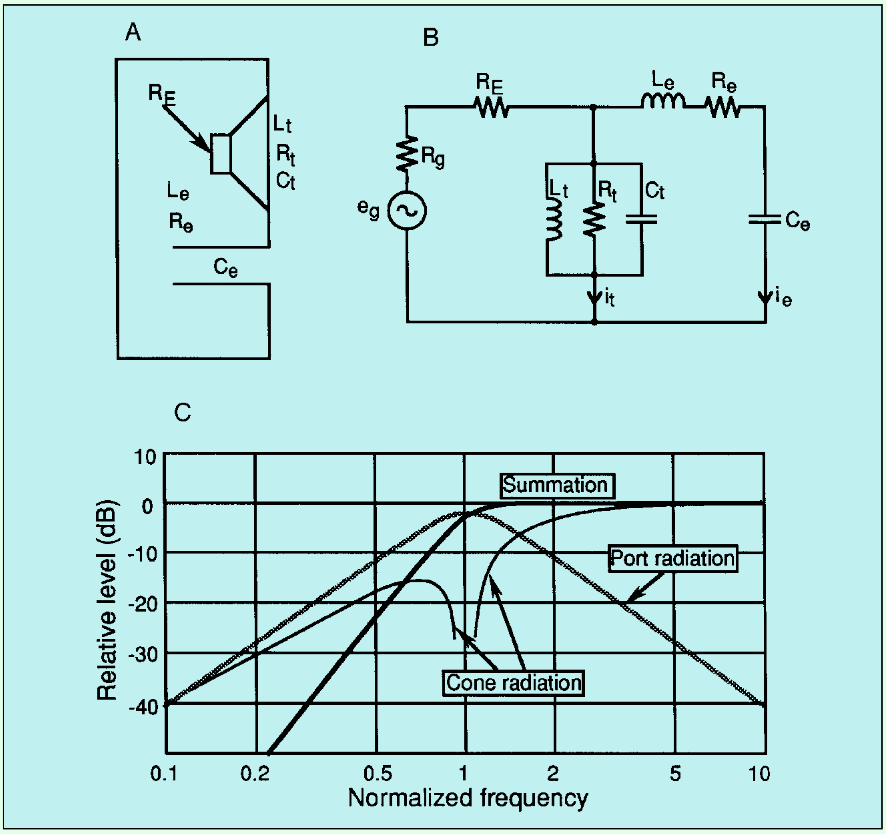



Fig. 10. The ported system. Section view (A); equivalent circuit (B); port and cone contributions to total output (C).

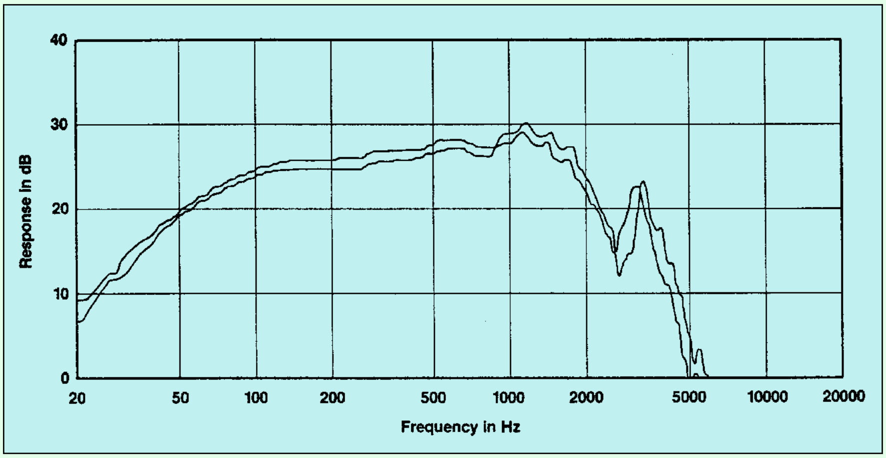



Fig. 11. Power compression in a 380-mm-diameter LF driver. Curves for 1 and 100 watts input are superimposed and displaced by 20 dB. (Data courtesy JBL.)

#### Mutual Coupling

In the LF range over which their response is essentially omnidirectional ($ka$ = 0.2 or lower), a doubling of closely spaced driving units will result in an increase in acoustical output of 3 dB for a fixed input power reference level [39, 48, 52, 53]. The progression in efficiency increase is shown in Fig. 9 for one, two, and four LF transducers, respectively. In each case, the electrical power delivered to each ensemble of drivers is constant. Assume that the reference power fed to the single driver is one watt; then for the set of two drivers, the power per driver is 0.5 watt, and for the set of four, the power per driver is 0.25 watt.

One may imagine that, in the two-driver case with both drivers wired in parallel, those two drivers have, in a sense, coalesced into a new driver—one with twice the cone area, twice the moving mass, and half the value of $R_E$. Thus, by Equation 1, the efficiency will have doubled. For the case where the two drivers are wired in series, the analysis goes as follows: The new driver has twice the cone area, twice the moving mass, four times the $(Bl)^2$ product, and twice the value of $R_E$. Again, by Equation 1, there will be a doubling of efficiency.

Mutual coupling often appears to give something for nothing, but there are clear limits to its effectiveness. With each doubling of cone area, the $ka$ = 0.2 upper response frequency corner moves downward approximately by a factor of 0.7, since this is the reciprocal of the value by which the effective cone radius has increased. As the process of adding drivers is continued, in the limit it can be shown that the efficiency of an ensemble of direct radiators cannot exceed a value of 25% [38]. Because of these constraints, the approximation of power doubling for each two-times increase in drivers is accurate only at very low frequencies and only if the efficiency values are low to begin with.

### Distortion

#### *Mechanical Effects*

The primary distortion mechanism in cone transducers is due to mechanical stress–strain limits. Small identified a practical mechanical displacement limit from rest position in the axial

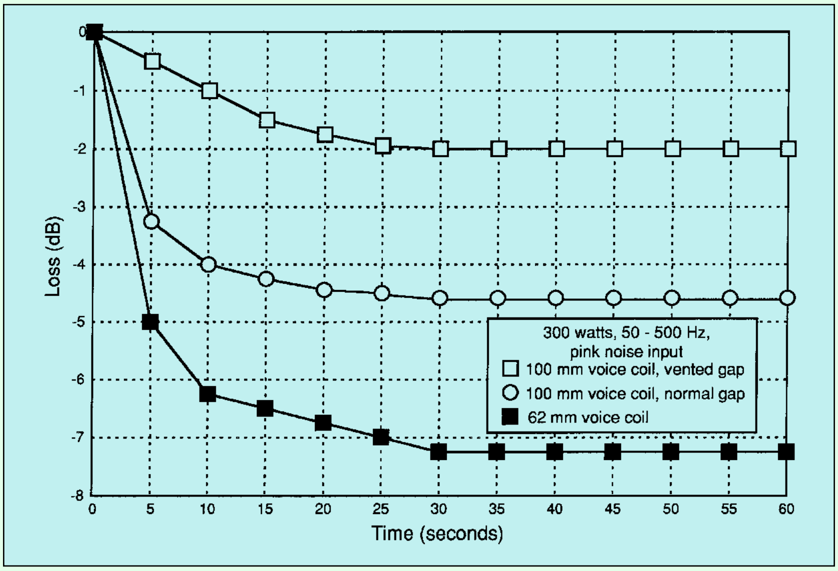



Fig. 12. Power compression. Output level versus time for three LF driver designs. (Data courtesy JBL.)

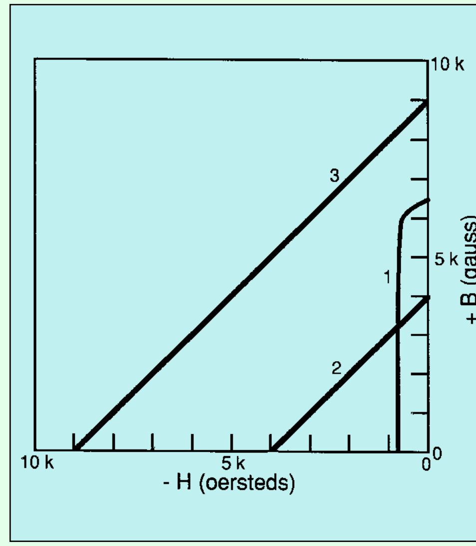



Fig. 13. Demagnetization plots for three loudspeaker magnetic materials. Alnico V (1); ferrite (2); neodymium-iron-boron type (3).

direction as the excursion at which 10% harmonic distortion is reached. This limit is known as $x_{MAX}$. While a loudspeaker may be operated beyond this displacement limit, at least on a momentary basis, the 10% linearity departure is generally recognized as a safe limit for good engineering practice. Since cone displacement tends to increase as the inverse square of frequency down to the $f_0$ region, it is easy to see how the $x_{MAX}$ limitation may easily be encountered in normal operation.

The onset of the cone displacement limit at low frequencies can be alleviated by using ported LF enclosures. The nature of this design is shown in Fig. 10. A section view of a ported system is shown in Fig. 10A, and the equivalent circuit is shown in Fig. 10B. The design relies on controlling the Helmholtz resonance of the enclosure to provide an “assisted output,” via the port, that minimizes cone motion (and thus distortion) at low frequencies, as shown in Fig. 10C. Thiele–Small parameters are universally used today to synthesize these systems.

Virtually all commercial PC design programs for ported systems will indicate transducer displacement limits so that the design engineer will always be aware of whether a system, while still on the drawing board, will go into displacement overload before it reaches its thermal limit. Good engineering practice demands that a ported system remain thermally limited down to $f_0$. Below that frequency the electrical drive signal is generally rolled off to avoid subsonic over-excursions of the cone.

A secondary mechanical distortion effect will be seen when the voice coil driver is far enough out of the gap so that there is a momentary loss of *Bl* product at peak excursion values. The effect is asymmetrical and gives rise to both even and odd distortion components.

### Port Turbulence in Vented Systems

In vented systems the ultimate output at low frequencies may be limited not by considerations of maximum cone excursion, but rather by air turbulence in the enclosure port when the system is operating at the tuning frequency [34]. A tentative limit here is to restrict the port air particle velocity so that its peak value does not exceed about 5% of the speed of sound. In general, ports should be designed with contoured boundaries to minimize turbulence and the noise and losses it often produces. Significant studies of port turbulence and its minimization through tapering the port tube’s cross-section area have been carried out by Vanderkooy [50, 51], Salvatti and Button [45], and Roozen et al. [44].

### Thermal Effects

Modern cone transducers intended for heavy-duty professional applications take advantage of newer materials and adhesives to make them more immune to thermal failure. Thermal failure is reached when the power dissipated in the voice coil as heat cannot be removed at a sufficient rate to maintain a safe operating temperature. A great deal of loudspeaker development has gone into designing structures and moving elements that are not only resistant to heat but aid in its removal from the transducer [32, 36].

For most applications in sound reinforcement, the effects of loudspeaker heating are more likely to result in component failure than those associated with displacement limitations. Dynamic linearity or power compression are terms used to describe the effects of heating on audio performance [34]. The data shown in Fig. 11 presents the frequency response of a single 380-mm LF transducer with inputs of 1 watt and 100 watts. In each case the chart recording of the levels has been adjusted to account for the 20-dB offset between the curves. In this manner the response differences can be clearly seen. If there were no dynamic compression, the two curves would lie one on top of the other. As it is, the progressive heating results in an increased value of $R_E$, which lowers the efficiency. In extreme cases, the increase in $R_E$ can result in changes in the LF alignment, which may be clearly audible as such.

Another way of viewing power compression is shown in Fig. 12. Here, several 380-mm transducers have been driven with a wide-band signal, and the

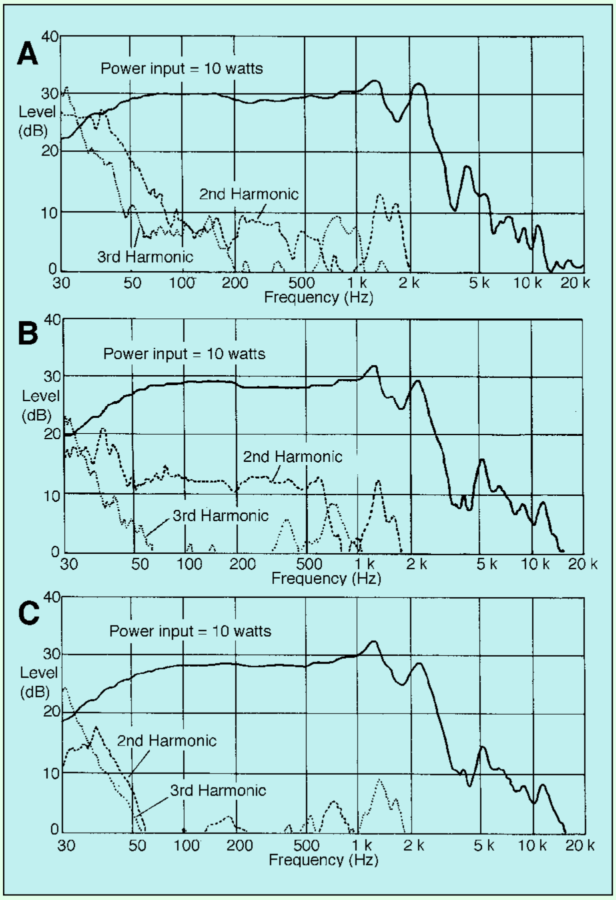



Fig. 14. Performance with same moving system in three different motor assemblies. Alnico V (A); traditional ferrite (B); ferrite with undercut polepiece and aluminum shorting ring (C). (Data courtesy JBL.)

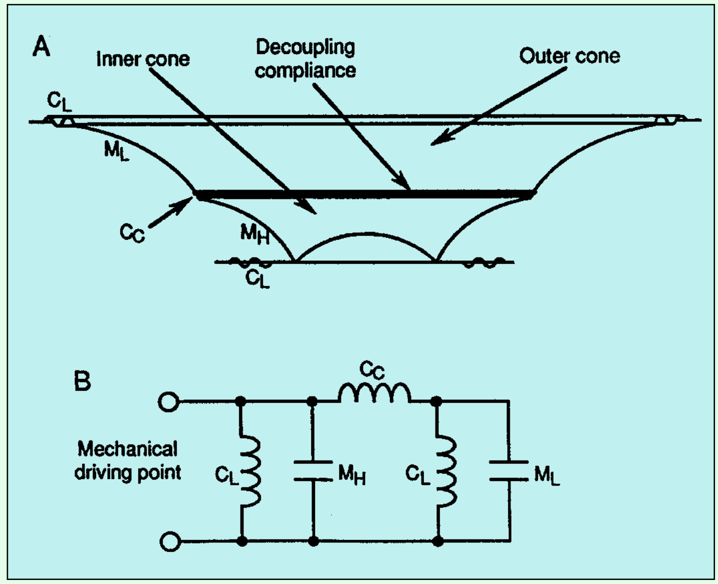



Fig. 15. The Altec-Lansing Duocone. Section view of cone (A); equivalent circuit (B).

effect of temperature rise has been plotted against time. For each transducer design, the reduction in output level eventually reaches an asymptotic value that depends on how effectively heat can be removed from the voice coil and magnet structure. In general, larger diameter voice coils remove heat more efficiently than smaller ones; additional measures, such as increasing convection cooling and increasing radiation to outside mechanical parts of the driver, can be helpful as well.

As the temperature of a ferrite magnetic structure increases, both flux density and efficiency are reduced. The effect is similar to the resistance increase caused by voice coil heating. If the temperature rise in the ferrite material has been moderate, normal performance may be restored upon cooling.

### *Distortion in Transducer Magnetic Systems*

Many aspects of the direct radiator's magnetic system can affect the distortion performance of a transducer [33]. The primary effect is the variation in the magnetic operating point that comes as a result of signal current in the voice coil as it modulates the magnetic structure's operating point. This happens to some degree in all designs, but the effect on acoustical performance depends on the degree of flux modulation of the static magnetic field. The general result of flux modulation is to increase the level of single-ended distortion, which is largely second harmonic.

Fig. 13 shows the demagnetization curves for three magnetic materials, Alnico V, ferrite, and neodymium-based systems. With the Alnico V system, the flux curve has a moderate slope in the operating region, which is near the intersection with the B-axis; at that position there is little tendency for flux modulation to be a problem. However, a strong input signal can result in permanent demagnetization if the operating point is forced downward on the steep portion of the curve to the left of the normal operating point.

With the ferrite magnet there is a certain amount of flux modulation due to the uniform slope of the curve. With the high-energy neodymium materials, the demagnetization curve is normally located high enough in the second quadrant of the B-H graph that the magnetic circuit is very likely to be operating at or near saturation. In this case the degree of flux modulation will be minimal.

Fig. 14 shows how a typical problem was solved. The distortion data shown in Fig. 14A is that of an older 300-mm-diameter LF transducer operating with an Alnico V magnet structure. Keeping the same moving system but changing to a typical ferrite magnet structure yields the data shown in Fig. 14B. Note the significant increase in mid-frequency distortion.

The data in Fig. 14C shows the effect of a ferrite magnet system outfitted with undercut polepiece geometry and a large, low-resistance aluminum flux shorting ring placed at the base of the polepiece. The significant reduction in second-harmonic distortion results from the setting up of an induction current in the shorting ring that counteracts the normal tendency of voice coil current to shift the magnetic operating point.

Other magnetic distortion effects include: the generation of eddy currents in local iron structures, which results in an increase in third-harmonic distortion and $I^2R$ losses; and inductance modulation of the voice coil, due the varying amount iron instantaneously surrounded by the voice coil, which results in increased second-harmonic distortion.

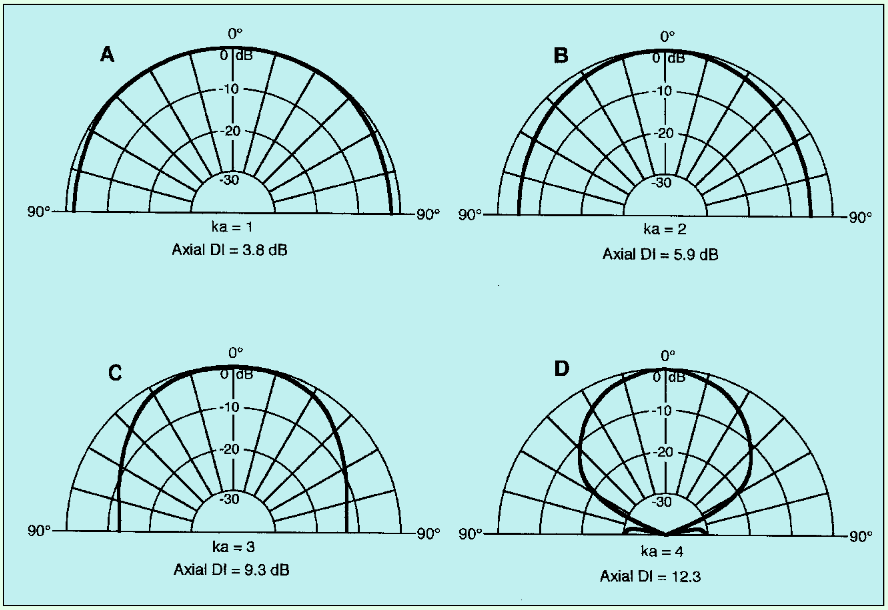



Fig. 16. Polar plots for a piston mounted in a wall. (Data after Beranek, 1954.)

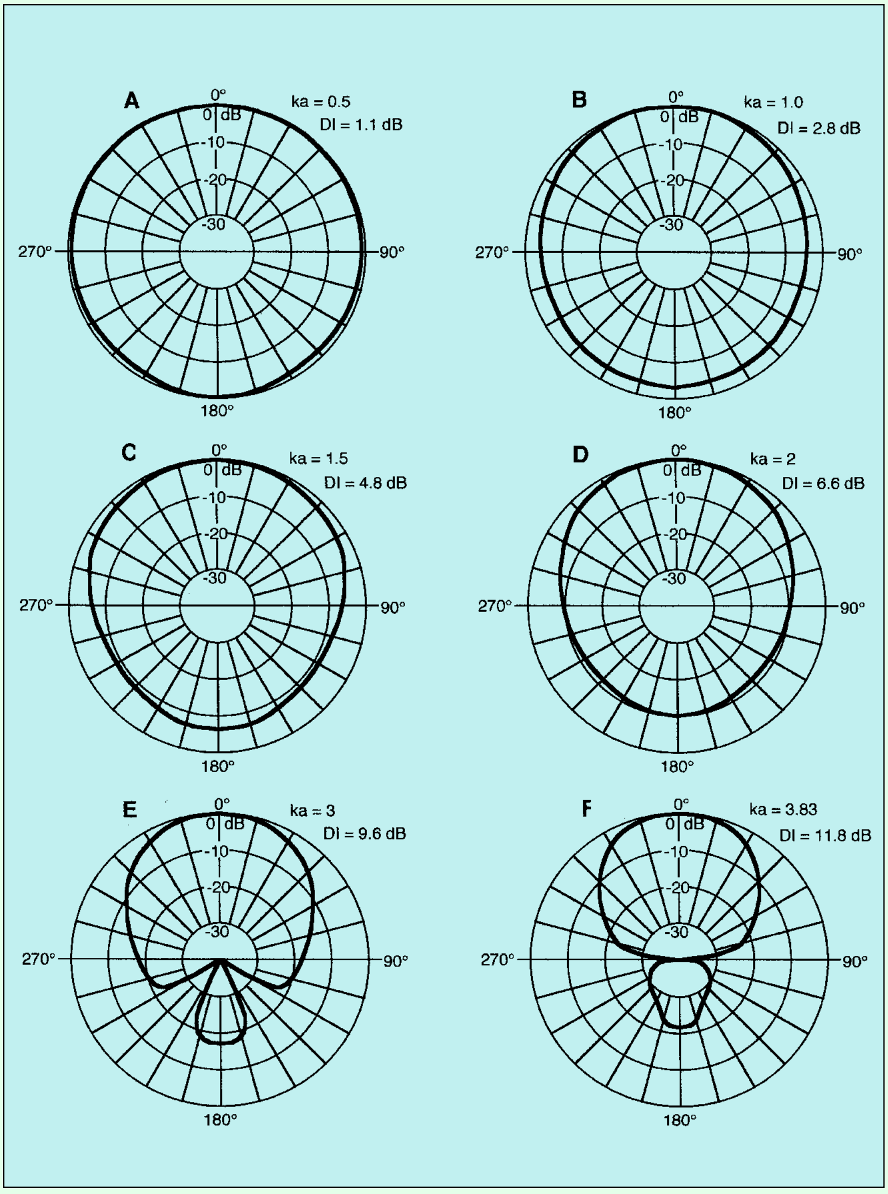



Fig. 17. Polar plots for a piston mounted at the end of a long tube. (Data after Beranek, 1954.)

### *Thermodynamic and FM Distortion Effects*

Thermodynamic distortion, or air overload, is present in relatively small quantities in direct radiator systems and may be disregarded in the normal operation of cones and domes.

Frequency modulation (FM) components are more likely to occur, especially at low to mid frequencies, where high cone excursions at lower frequencies may modulate higher frequencies as the cone's velocity attains a significant percentage of the speed of sound. The effect is quite noticeable in single-cone systems but is minimized in multiway systems. Beers and Belar [30] and Klipsch [40] were among the first to describe this phenomenon in detail.

Another aspect of thermodynamic distortion present in sealed direct radiator systems is due to the nonlinearity of the enclosed air spring. In general, if the maximum instantaneous change in an enclosed volume can be limited to about 0.5%, the effect of the nonlinearity can be ignored [34]. The type and amount of enclosure damping material in a sealed enclosure has the additional effect of increasing the actual enclosed volume. The work of Leach [41] is significant in this area.

## The Decoupled Cone

Over the years loudspeaker designers have observed that, at high frequencies, the cone ceases to move as a single unit, but rather breaks up into more complex motions. These result in an effective lighter moving mass at high frequencies, extending the HF response of the system. This was first commercialized by the Altec Duocone loudspeaker [28]. Fig. 15A shows a section view of the cone profile used in the Duocone loudspeaker, and a mobility mechanical circuit is shown in Fig. 15B. The modern soft-dome HF unit exploits this effect more predictably through high damping of the moving system. Finite element analysis (FEA) provides a means of analyzing in detail the directionality of loudspeaker cones and domes, taking into account multiple breakup effects. Many low-cost drivers intended for ceiling installation in distributed systems make good use of decoupling for attaining wider coverage at HF.

In recent years, the notion of decoupling has resurfaced with the distributed mode loudspeaker (DML). Here, a panel of light, stiff, and fairly lossless material is driven by a mass-loaded transducer. When operated in the frequency range over which the panel exhibits a dense array of higher-order two-dimensional bending modes, the resulting power output and dispersion are uniform over a frequency range defined by:

Upper frequency limit, $f_{max} = R_P/2\pi M_c$ (2A)

Lower frequency limit, $f_{min} = R_P/2\pi M_m$ (2B)

where $R_P$ is the radiation impedance of the panel and $M_c$ and $M_m$ are, respectively, the masses of the driven system and the magnetic actuator.

Due to its random, diffuse nature, the response of a typical DML panel departs from that of a typical dynamic driver in several regards: the modulus of electrical impedance is quite uniform over the normal system passband; the solid-angle radiation pattern is quite uniform over the normal system passband; the impulse response for a typical panel is fairly long, upwards of 25 msec, however it is free of ringing at any specific frequency; commensurate with the long impulse train, the amplitude response will exhibit, in all directions of radiation, many fine response dips on the order of 6 dB [29].

With its unique combination of acoustical properties and relatively low cost, the DML offers great promise in the area of distributed system design.

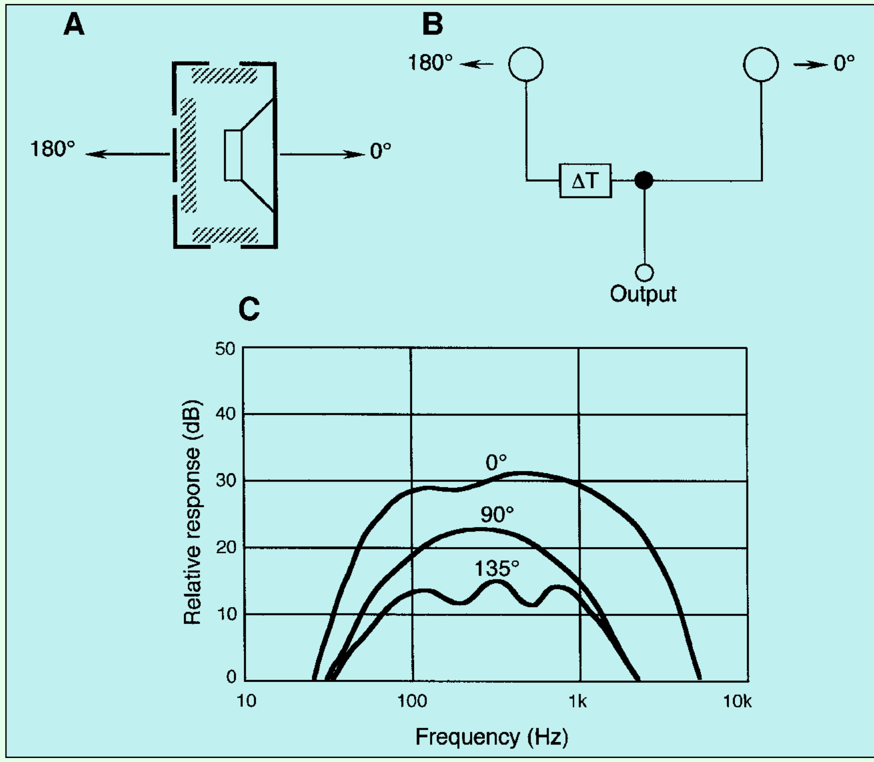



Fig. 18. The Altec Extenda-Voice gradient loudspeaker system. Section view (A); physical circuit (B); nominal off-axis response curves (C).

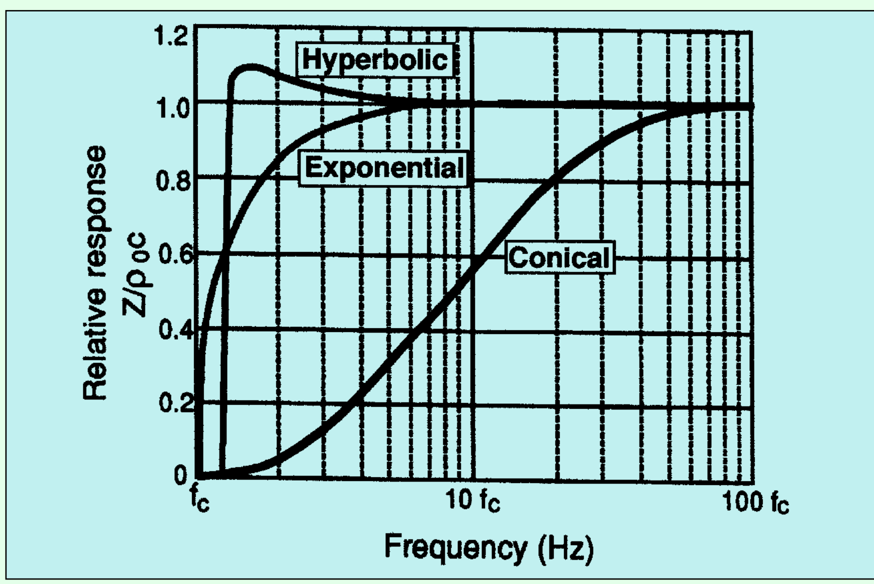



Fig. 19. Radiation resistance for conical, exponential, and hyperbolic horns.

### Directional Properties of Direct Radiators

Assuming that the moving systems have only a single degree of freedom, the polar data in Fig. 16 shows the theoretical directionality of a piston mounted in a large baffle as a function of *ka*. The on-axis directivity index (DI) of each directional pattern is also shown in the figure [1, 42]. Similar data is shown in Fig. 17 for a piston mounted at the end of a long tube. These two conditions simulate normal $2\pi$ and $4\pi$ mounting conditions for loudspeakers. For many routine system design applications, this data is sufficiently accurate.

### Gradient Loudspeakers

The simplest gradient loudspeaker is an unbaffled cone transducer operating as a dipole. The natural directional response for a dipole is a cosine pattern, or a "figure-eight" [43]. Dipoles are very inefficient and have not been widely used in sound reinforcement applications; however, there are certain applications where their specific radiation pattern could be very useful. During the early 1970s, Altec produced a modified gradient loudspeaker in which one side of the dipole operated in series with an acoustical delay path. The delay produced a hypercardioid pattern in much the same manner as a typical dynamic microphone does.

Fig. 18A shows a section view through the hypercardioid loudspeaker. The output from the rear of the transducer is sent through a constant delay, provided by the path length and the resistive elements in that path. The result of the delay is that, at 135°, output from front and back of the transducer will

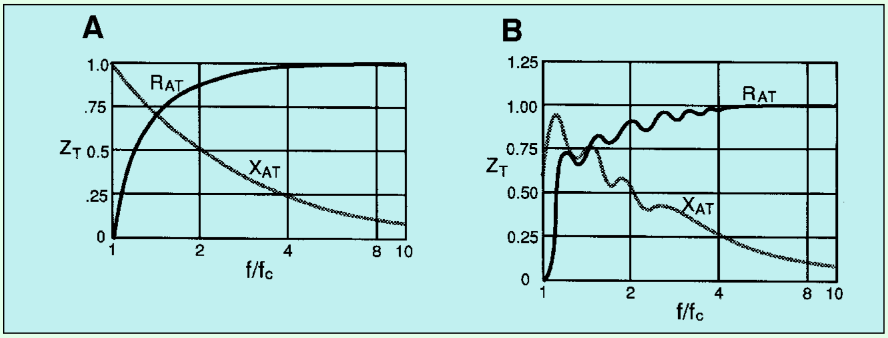



Fig. 20. Real and imaginary components of horn impedance for a long exponential horn (A) and a short exponential horn (B).

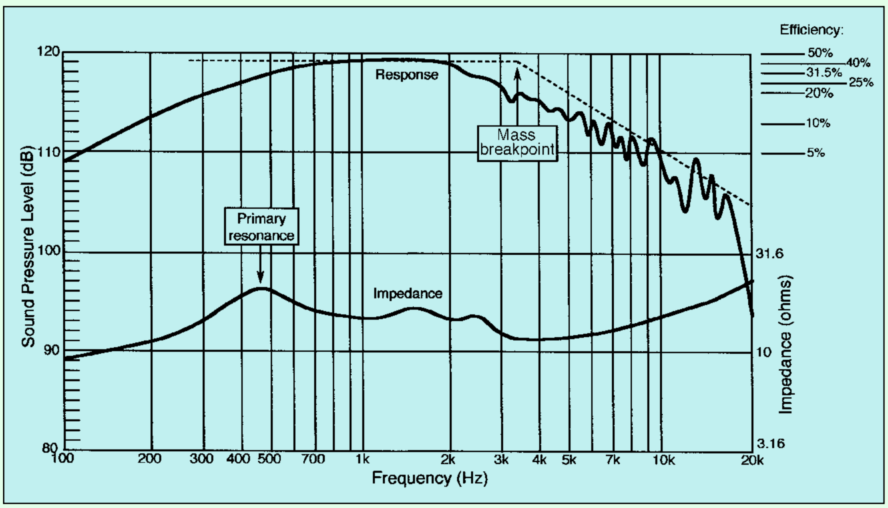



Fig. 21. Plane wave tube measurements of a compression driver showing amplitude response and impedance.

effectively cancel. For useful output from the front of the transducer, the signal fed to the system must be equalized with a 6-dB-per-octave rise for each halving of frequency to compensate for the diminishing gradient. The equivalent physical circuit of the loudspeaker is shown in Fig. 18B, and off-axis polar data is shown in Fig. 18C. A system such as this would normally be used for speech purposes in highly reverberant spaces where the loudspeaker's DI of 6 dB at LF would work to its advantage. Vertical stacks of the device can increase total output capability as well as increase the on-axis DI.

## HORNS AND COMPRESSION DRIVERS

### Early Development

Many engineers and physicists have contributed to horn and compression driver development over the years. Early versions of the horn were used by many tinkerers who basically did not understand how the horn worked—they knew only that somehow the horn increased acoustical output [58]. The first example of thorough engineering was carried out by Bell Telephone Laboratories [83], working from the model of horn impedance described by Webster [81]. Significant later development was carried out by Klipsch [66], who designed a remarkably compact bass horn, and Salmon [76, 77], who described the impedance characteristics of several important horn flares, including the hyperbolic, or Hypex, profile [68]. Geddes [55] sought to position Webster's model as a special case within a broader context.

Fig. 19 shows the real part of the radiation impedance for hyperbolic, exponential, and conical horn profiles. Here, only the exponential and hyperbolic profiles provide useful output at low frequencies. In our discussion we will restrict ourselves to the exponential profile, since it has found almost universal application over the years.

Fig. 20A shows the real and imaginary parts of throat impedance for a long exponential horn. For a horn of practical length, we might observe impedance components such as those shown in Fig. 20B. The slight peaks and dips in response are due to reflections from the mouth of the horn back to the throat. There is an optimum mouth size for a horn of specific cutoff frequency to minimize reflected waves from the horn's mouth [63].

### Theoretical Modeling

The compression driver is designed to match the impedance of the electromechanical system to the throat of the horn, and the radiation impedance, reflected to the electrical side of the circuit, is:

$$R_{ET} = [S_T(Bl)^2]/\rho_0 c S_D^2 \qquad (3)$$

where $S_T$ is the area of the driver throat and $S_D$ is the area of the driver diaphragm. The phasing plug in the driver is the means by which the ratio of the two areas is adjusted for best HF response.

When the driver is attached to the horn, the efficiency in the range where the horn's radiation impedance is essentially resistive is:

$$\eta = (2R_E R_{ET})/(R_E + R_{ET})^2 \qquad (4)$$

where $R_E$ is the voice coil resistance. When the voice coil resistance is made equal to the radiation resistance, the efficiency of the driver over its normal pass-band will in theory be 50%. In practice, efficiencies of the order of 30% can be achieved in the midrange—and this is only about 2 dB below the theoretical maximum.

#### *Region of Flat Power Output*

The data of Fig. 21 shows the normal power response for a compression driver/horn combination when the horn's throat impedance is resistive. The LF limit is due to the primary resonance of the driver; for a typical HF compression driver this may be in the range of 500 Hz.

The principal midband rolloff commences at what is called the mass break point, $f_{HM}$, given by:

$$f_{HM} = (Bl)^2/\pi R_E M_{MS} \qquad (5)$$

where $M_{MS}$ is the mass of the moving system. For most HF compression drivers the mass breakpoint takes place in the range of 2500 to 4500 Hz. It is considered a fundamental limit in HF drivers, inasmuch as today's magnetic flux densities are normally maximized in the range of 2 tesla, and low-moving mass is limited by metals such as titanium and beryllium that are not likely to be replaced in the near future.

Two additional inflection points are often seen in the HF driver response curve: one is due to the volume of the front air chamber in the driver, the space between the diaphragm and the phasing plug. Its effect on response may be seen as low as 8 kHz in some drivers. Voice coil inductance may cause an additional HF rolloff at high frequencies. This may be compensated for through the use of a silver or copper shorting ring plated on the polepiece in the region of the voice coil. (See Distortion in Transducer Magnetic Systems, page 419.)

### *Cone-Driven Horns*

From the earliest days, cone transducers have been employed as horn drivers [71, 72]. The theoretical principles that govern the design parameters for horn drivers apply equally to the adaptation of cone drivers as well as to purpose-designed compression drivers. Keele [60] presented a straightforward and useful analysis of LF horn design using both Thiele–Small and traditional electromechanical parameters. Leach [69] summarized Keele's work, together with Small's approach to the subject [78], and addressed other factors such as reactance annulling.

### *Reactance Annulling*

In some compression driver designs, a mechanical stiffness in the form of a small air chamber is located behind the driver's diaphragm. The mechanical reactance resulting from the stiffness cancels in part the mass reactance portion of the radiation impedance, resulting in a more resistive impedance in the region of the cutoff frequency. The effect of this is greater acoustic output in the horn cut-off frequency range for a given drive signal [73, 74].

Reactance annulling is not normally used in HF compression drivers, but it is used in the design of bass horns, most notably in the case of the Klipschorn, where the normal response associated with a 47-Hz flare rate is extended down to about 40 Hz [66].

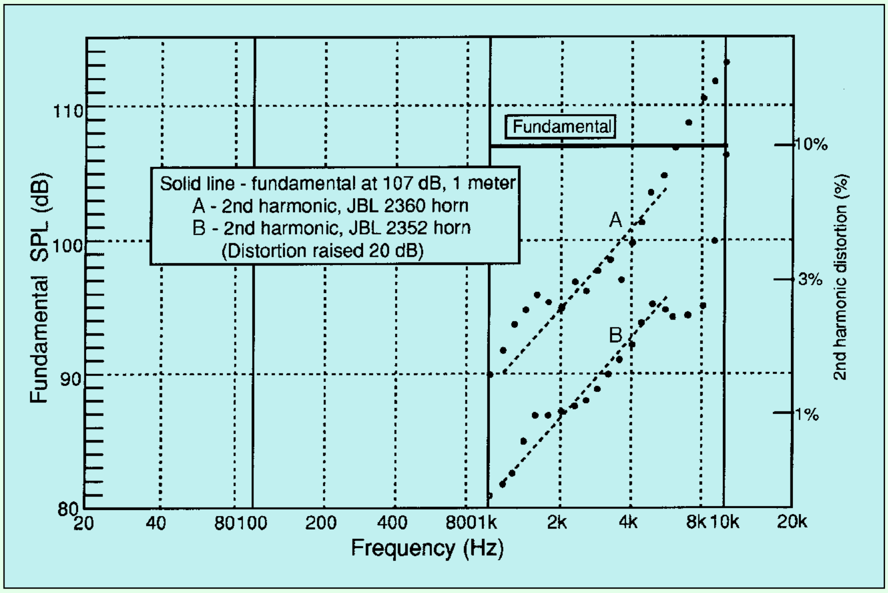



Fig. 22. Second-harmonic distortion in two horn systems, using a compressed fundamental. (Data courtesy JBL.)

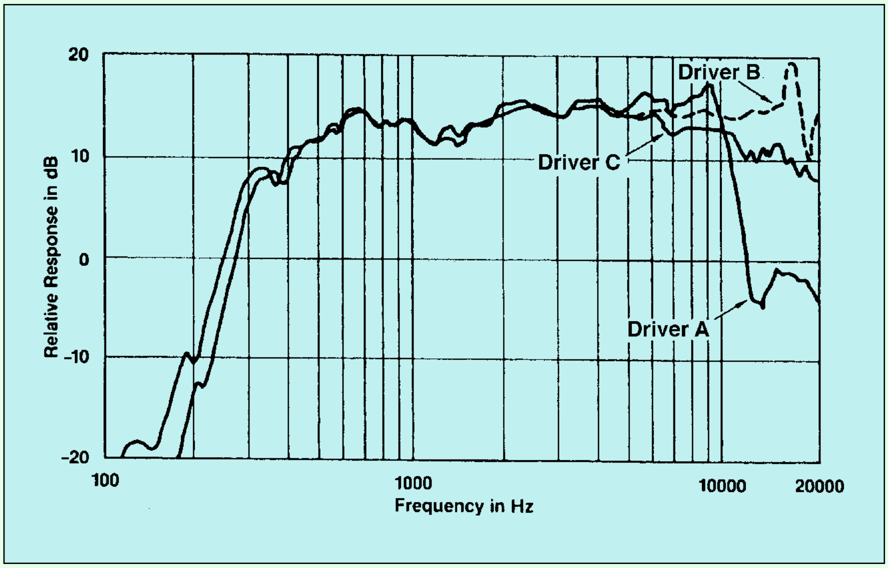



Fig. 23. Plane wave tube amplitude response of three compression drivers.

## Distortion

The dominant cause of distortion in compression driver-horn systems is due to thermodynamic, or air, overload [75]. This comes as a result of extremely high pressures that exists at the horn throat:

$$L_P = 94 + 20 \log (W_A(\rho_0 c)/S_T)^{0.5} \quad (6)$$

where $W_A$ is the acoustical power generated and $S_T$ is the throat area ($m^2$).

For example, in plane wave propagation, an intensity of one watt per square centimeter will produce a sound pressure level of 160 dB $L_P$. For levels in this range, successive pressure peaks are tilted forward as they propagate down the horn due to the increase in sound velocity at elevated temperatures under adiabatic conditions.

Thuras, Jenkins, and O'Neil [80] and Goldstein and McLachlan [57] analyzed the problem, leading to a simplified equation that gives the percent second harmonic distortion in horn systems:

$$\%\ 2\text{nd HD} = 1.73(f/f_C)\sqrt{I_T} \times 10^{-2} \quad (7)$$

where $I_T$ is the intensity in watts per square meter at the horn's throat, $f$ is the driving frequency, and $f_C$ is the cut-off frequency of the horn.

Fig. 22 presents measurements of the second-harmonic distortion produced by two horns of differing flare rates.

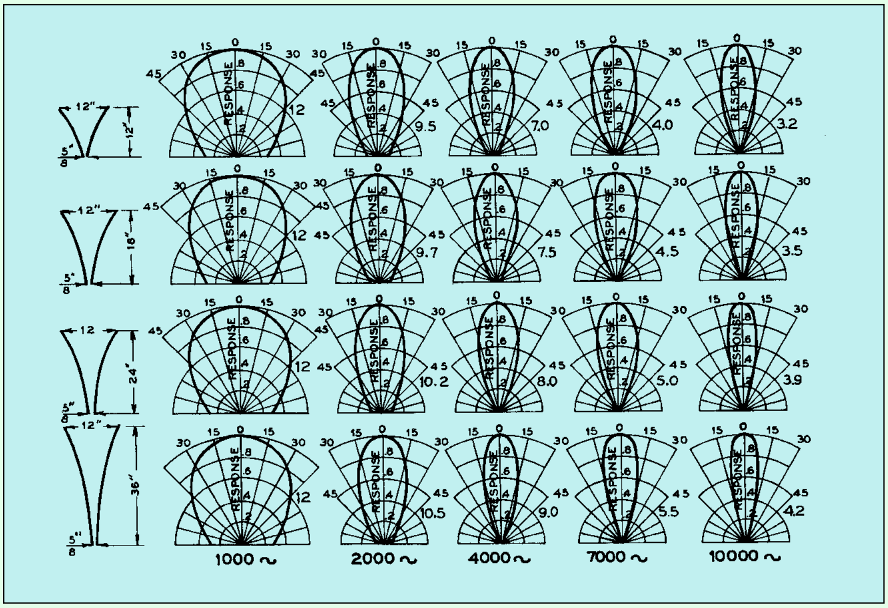



Fig. 24. Directivity of various exponential horns. (Data from Olson, 1957.)

The fundamental output in each case was held constant at a level of 107 dB at a distance of one meter, and the second-harmonic distortion has been raised 20 dB for ease in reading. The scale on the right ordinate indicates the second-harmonic distortion in percentage. The cutoff frequency of the horn used for the data shown in Fig. 22A is 70 Hz, while that of the horn used for the data shown in Fig. 22B is approximately 560 Hz. The average difference in distortion is 8 dB.

A horn with a high cutoff frequency has a rapid flare rate and as such may lack good directional control at low frequencies. This is a tradeoff that the design engineer has to reconcile for a variety of applications. For example, sound-reinforcement applications require specific pattern control in the range from 300 Hz upward, while music-monitoring applications may require horn pattern control no lower than about 800 Hz.

If the exact mechanism for a given kind of distortion can be defined mathematically, a model can be implemented and used to predistort the signal, resulting in reduced distortion in the system's output over a given power operating range. Klippel [65] describes some of the techniques for accomplishing this.

## The Role of Secondary Resonances

As shown earlier in Region of Flat Power Repsonse (page 416), the power response of a horn driver is flat up to its mass break point, above which the response rolls off 6 dB per octave. However, beneficial secondary resonances may be used to increase the driver's output above this point [64, 70]. These resonances generally occur in the surround sections of the diaphragm and are decoupled from the diaphragm itself. As in the case of decoupled resonances in cones discussed earlier, the lowering of moving mass at higher frequencies can result in a considerable increase in useful HF response. Fig. 23 shows the response for three compression drivers, all with 100-mm-diameter diaphragms and mounted on the same horn. Driver A has an aluminum diaphragm and half-roll surround. The secondary resonance is about 9 kHz. Response is maintained fairly flat to that frequency, falling off rapidly above. Driver B has a beryllium

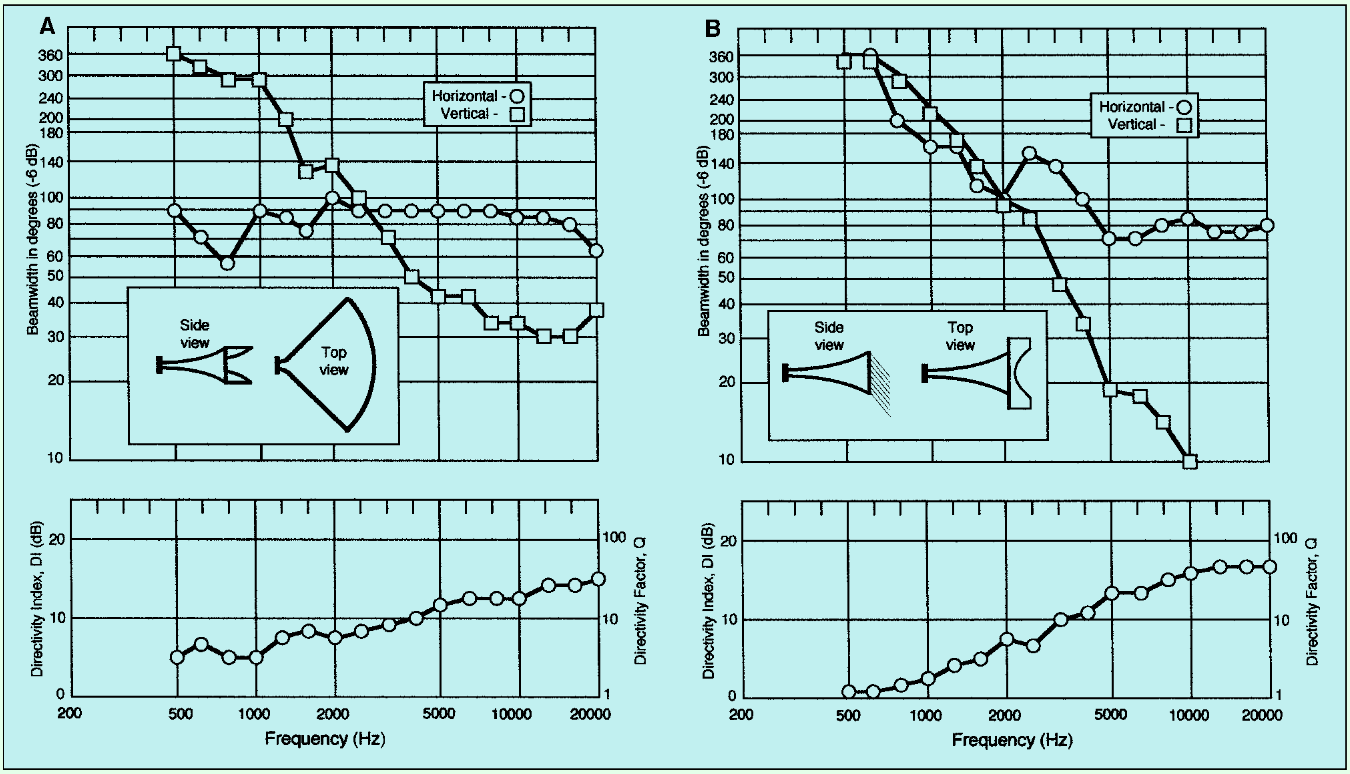



Fig. 25. Beamwidth and directivity data for typical radial horn (A) and slant-plate lens system (B).

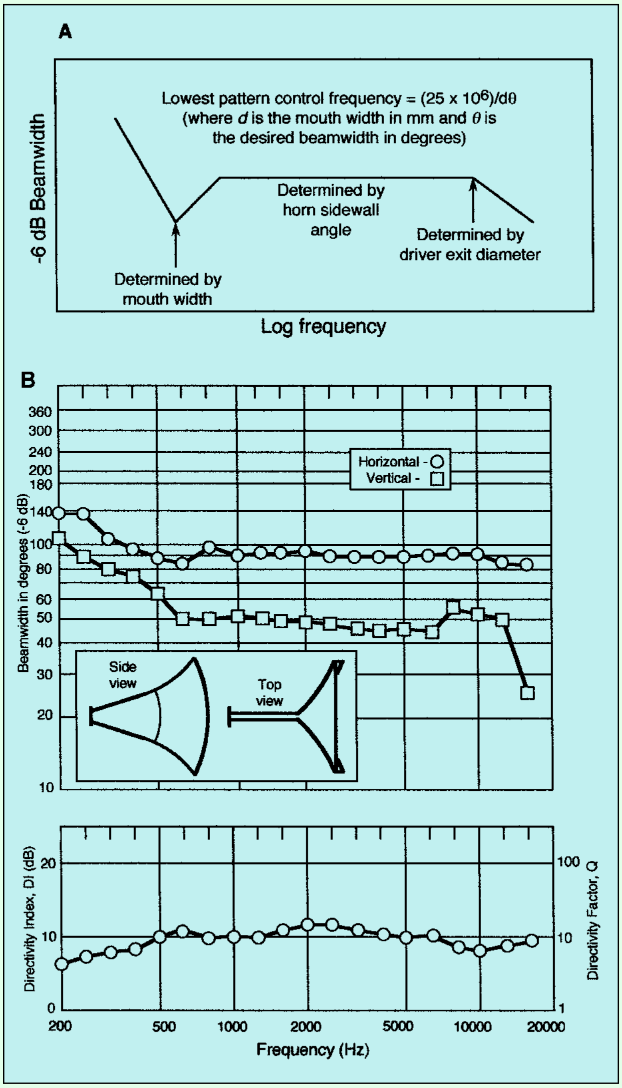



Fig. 26. Directivity of a uniform coverage horn. Basic directivity regimes (A); beamwidth and directivity for a 90° by 40° uniform directivity horn (B).

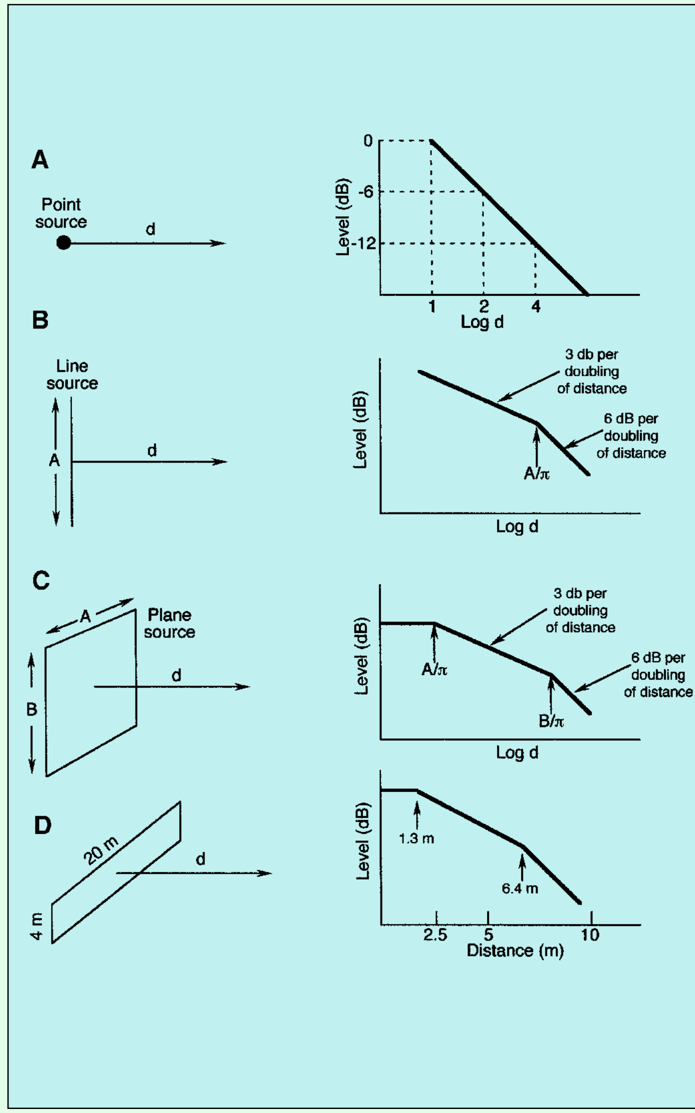



Fig. 27. Attenuation of sound with distance from point, line, and planar sound sources.

diaphragm with a half-roll surround. Note that, due to the greater stiffness and lower mass of the material, the secondary resonance has shifted out to about 17 kHz. Driver C has an aluminum diaphragm with distributed surround geometry that moves the secondary resonance to beyond 20 kHz, resulting in smooth, extended response within the normal audio band with no pronounced peaks [70].

## Directional Response

The basic exponential horn exhibits directional response as shown in Fig. 24. From the earliest days it was recognized that directional characteristics were a key element of loudspeaker performance [84]. Over decades of development, numerous methods have been used to improve directional performance at high frequencies for sound reinforcement applications. For multicellular horns, in the early days [83], groups of exponential cells, each about 15° wide in each plane, were clustered together to define a specific solid radiation angle; this produced excellent results at mid-frequencies, but there was pronounced "fingering" of the response along the cell boundaries at higher frequencies. For radial horns, in this application, the horn's horizontal profile is conical, with straight, radial sides defining a target coverage angle and the vertical profile is tailored to make a net exponential profile along the horn's primary axis; the nominal horizontal and vertical -6 dB beamwidth of a radial horn is shown in Fig. 25A [71]. For acoustical lenses, a slant-plate acoustical lens can be placed at the mouth of an exponential horn to diverge the exiting waves in one dimension, as shown in Fig. 25B [54, 67].

### *Constant Directivity Horns*

Also known as uniform coverage or constant coverage horns, these designs date from the mid-1970s to the early 1980s [59, 62, 79]. The basic design common to a number of manufacturers uses a combination of exponential or conical throat loading, diffraction wave guide principles, and flared terminations to produce uniform nominal coverage angles in the horizontal and vertical planes. The general shape of the beamwidth curve is shown in Fig. 26A, as it applies to the horizontal and vertical planes independently. Fig. 26B shows the measured beamwidth and DI of a typical constant directivity horn with nominal 90°-by-40° pattern control. Within certain limitations,

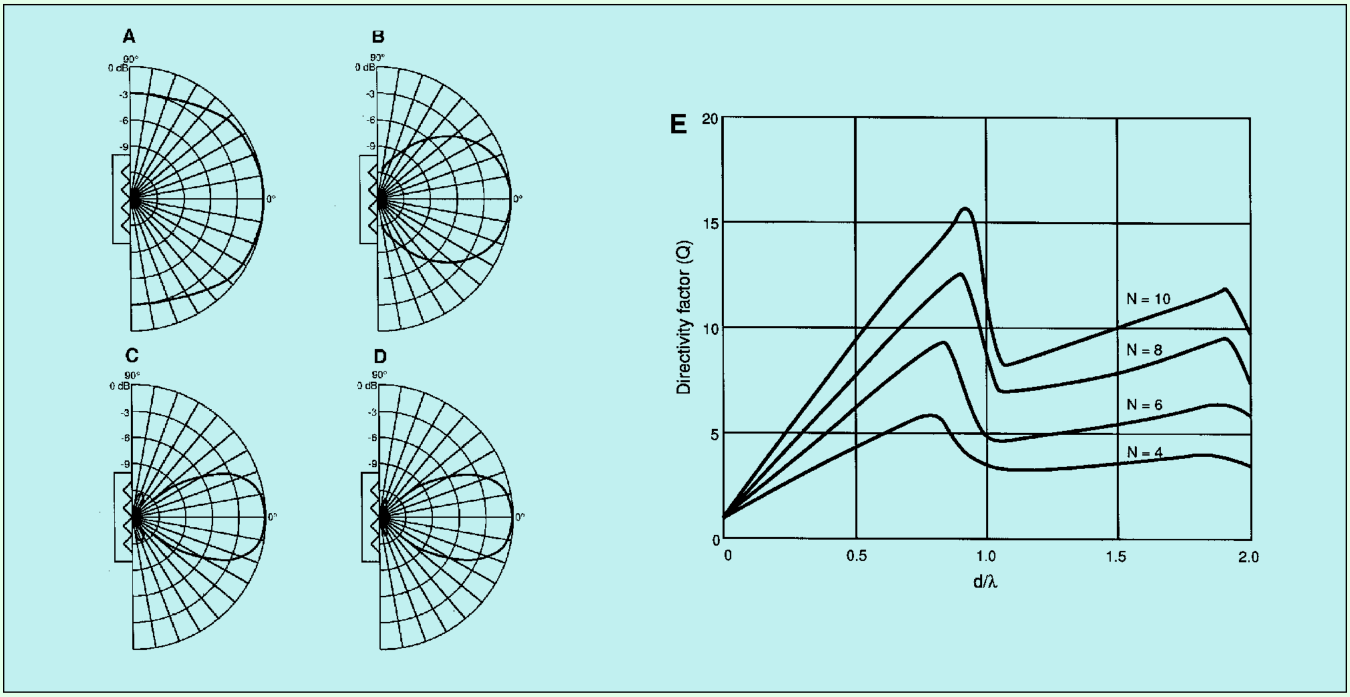



Fig. 28. Directivity of a four-element vertical line array with 0.2 meter separation between driver centers. 200 Hz (A); 350 Hz (B); 500 Hz (C); 1 kHz (D); directivity factor for arrays of 4, 6, 8, and 10 elements (E).

acoustic waveguide theory has proposed an alternate approach to achieving similar goals [55, 56].

## ARRAYS

Both horns and direct radiators may be treated the same in terms of arraying. In this section we will examine some useful concepts. Single-element, line, and planar arrays differ in their radiation characteristics over distance, as shown in Fig. 27. At long wavelengths, the simple inverse square relationship of a point source A is modified by a line array as shown at B, and a very large planar array will show little attenuation with distance up to limits proportional to the array dimensions [98]. A finite planar array will have the characteristics shown at D. Long horizontal line arrays have been placed above prosceniums in performance spaces to extend the range of a high direct-to-reverberant ratio toward the rear of the space; large planar arrays are the mainstay of mega-event music reinforcement [87].

### The Simple Line Array

Kuttruff [95] describes the polar response of a line array of omnidirectional sources in the plane of the array as:

$$R(\theta) = (\sin[1/2\, Nkd \sin\theta])/(N \sin[1/2\, kd \sin\theta]) \qquad (8)$$

where $N$ is the number of elements in the array, $k$ is $2\pi f/c$, $d$ is the spacing of the elements in the array, c is the speed of sound, and $\theta$ is the measurement angle in radians. For four elements as shown in Fig. 28, the polar response

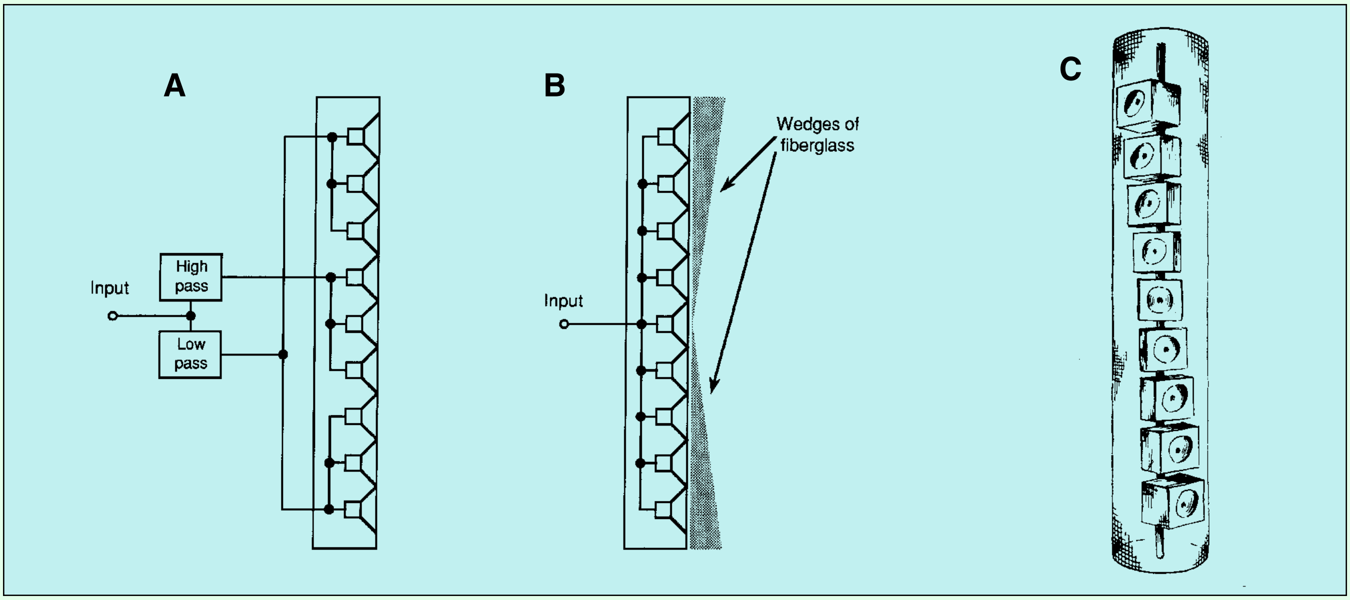



Fig. 29. Tapering of arrays. Electrical tapering (A); acoustical tapering (B); tapering through component rotation (C).

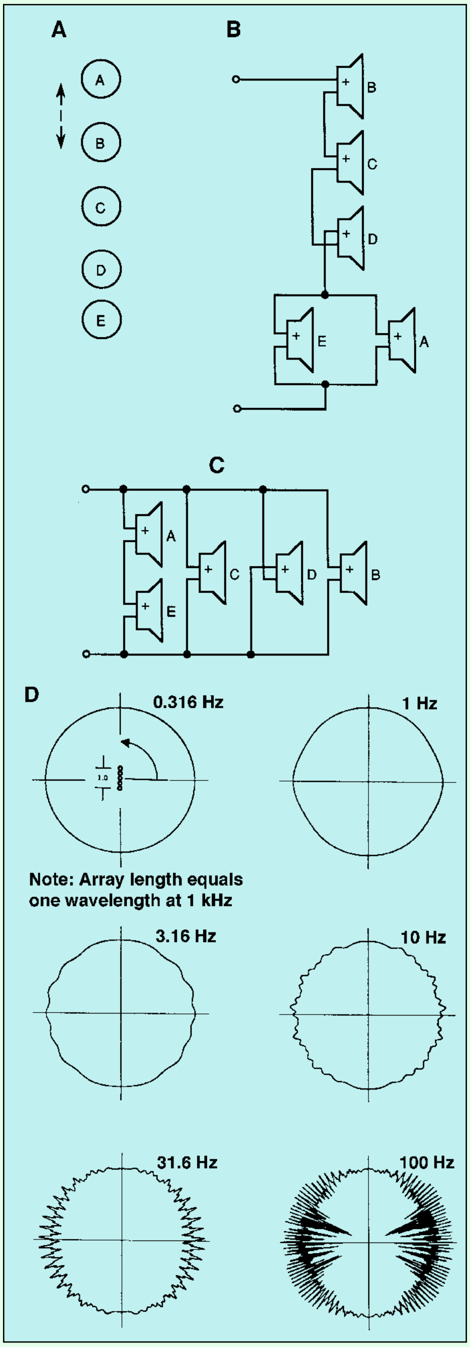



Fig. 30. The Bessel array. Layout (A); series wiring diagram (B); parallel wiring diagram (C); polar response (D). (Data courtesy Keele, 1990.)

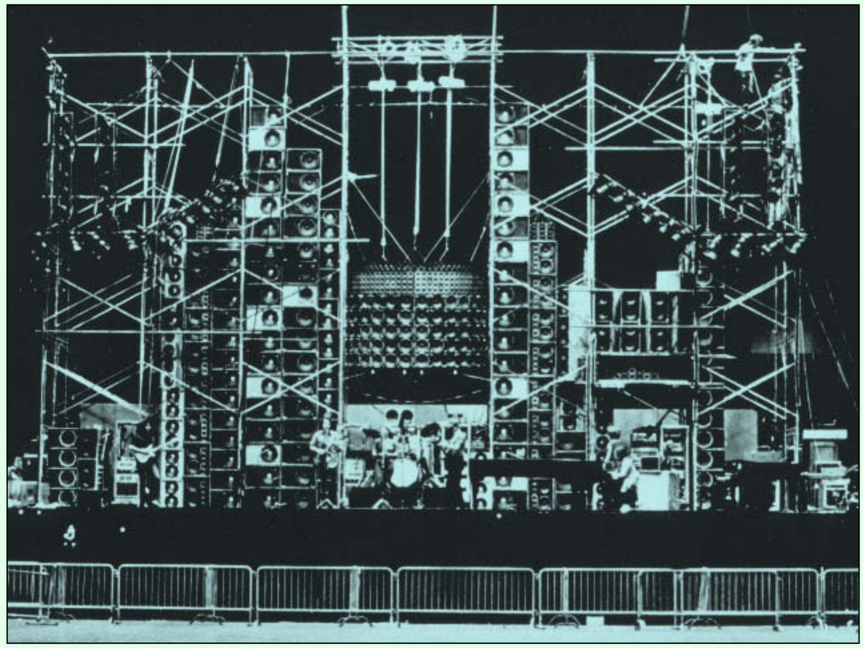


Fig. 31. Grateful Dead "Wall of Sound" direct radiator system, 1974, using line, planar, and arc segment arrays. Individual discrete systems were employed for each instrument separately from the vocal reinforcement. (Photo courtesy Richard Pechner.)

is shown in A through D. The directivity factor is shown at E.

The four-element array will exhibit good pattern control over the range from d/λ 0.5 to 2.0. At higher frequencies the pattern will exhibit narrowing and lobing, and simple arrays of more than six elements will generally have unsatisfactory characteristics.

## Tapering the Line Array

The accepted method of extending the uniform coverage range of a line array is through frequency tapering, or shaping, to allow the array to, in effect, reduce in size with rising frequency. Some techniques are shown in Fig. 29. Electrical frequency tapering is shown at A, and acoustical frequency tapering is shown at B [94]. A unique "barber-pole" array is shown at C [93].

## The Product Theorem

The product theorem [91] states that an array composed of uniform directional elements will exhibit the same far-field response as a like array of omnidirectional elements multiplied by the directional properties of one of the single directional elements. This is another way of stating the principle of superposition, and is useful in estimating the directional response of complex arrays.

### *The Bessel Array*

Franssen [88] describes an array of elements whose amplitude drive characteristics are derived from Bessel coefficients [90, 92]. A simple 5-element array is shown in Fig. 30A, and simplified wiring diagrams to derive the drive coefficients are shown at B and C. The far-field response of the array, modeled with omnidirectional sources, is shown at D. Note that the response is essentially omnidirectional over a 100-to-1 wavelength ratio.

Via the product theorem, each omnidirectional element could be replaced with a directional element, all oriented in the same direction, with the resulting response of the array exhibiting the chosen directional characteristic over the same 100-to-1 wavelength ratio.

The Bessel array has potential for speech reinforcement in live spaces. Its phase response varies with angle and frequency, however, and thus it may be difficult to integrate the concept into a system that includes standard radiating elements.

## Very Large Arrays for Music

Concert sound reinforcement in very large venues, indoors or out, requires large arrays, and the accepted method of assembling these arrays is to use building blocks that are each relatively full-range units. Thus, the assembled system, normally resembling a large plane, or sets of planes with curved sections connecting them, has much in common with the principles described in Fig. 31 [87]. As an example of this we show in Fig. 32 an elevation view of a typical large vertical array (A) along with the on- and off-axis response (B) measured in an arc along the listening plane [89].

The great virtue of these systems is their ability to deliver very high

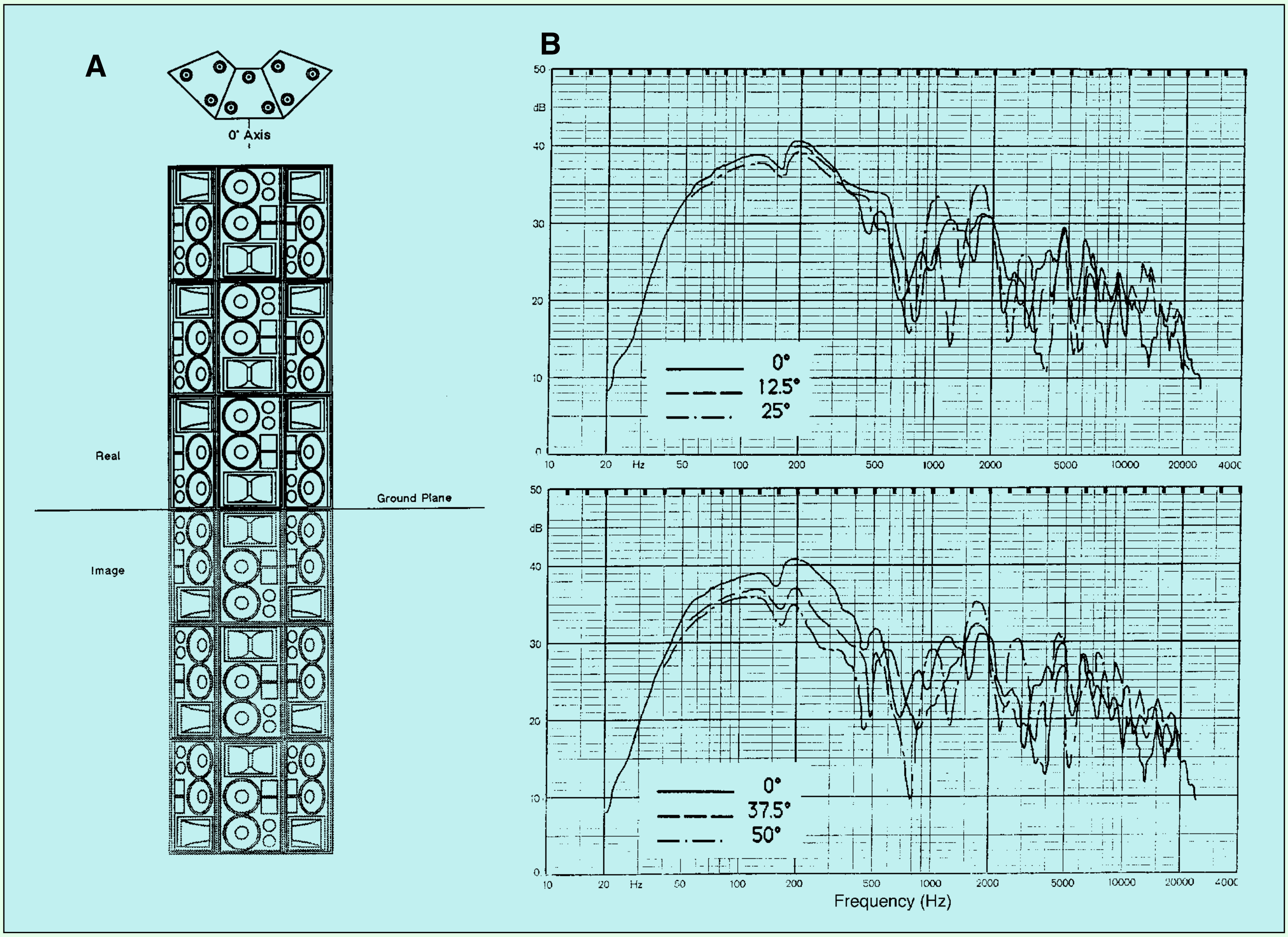



Fig. 32. A large array for music reinforcement. Physical layout (A); off-axis response on the ground plane (B). (Data courtesy Gander and Eargle.)

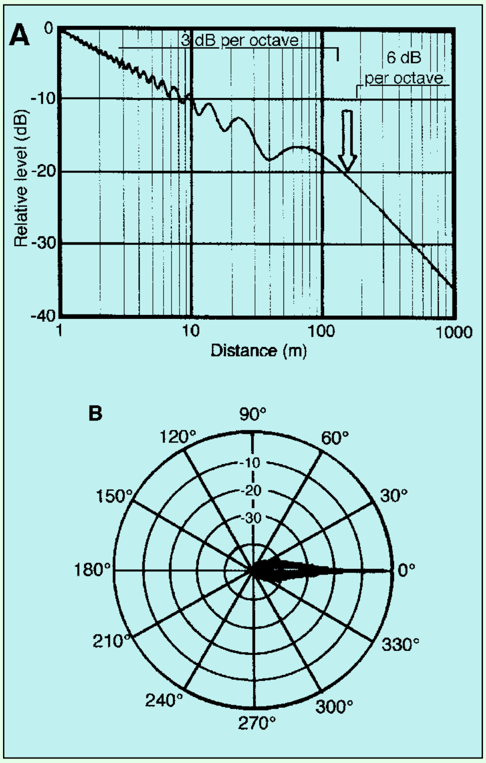



Fig. 33. Performance of a continuous array 3 meters in length. Attenuation with distance at 10 kHz (A); polar response at 10 kHz (B).

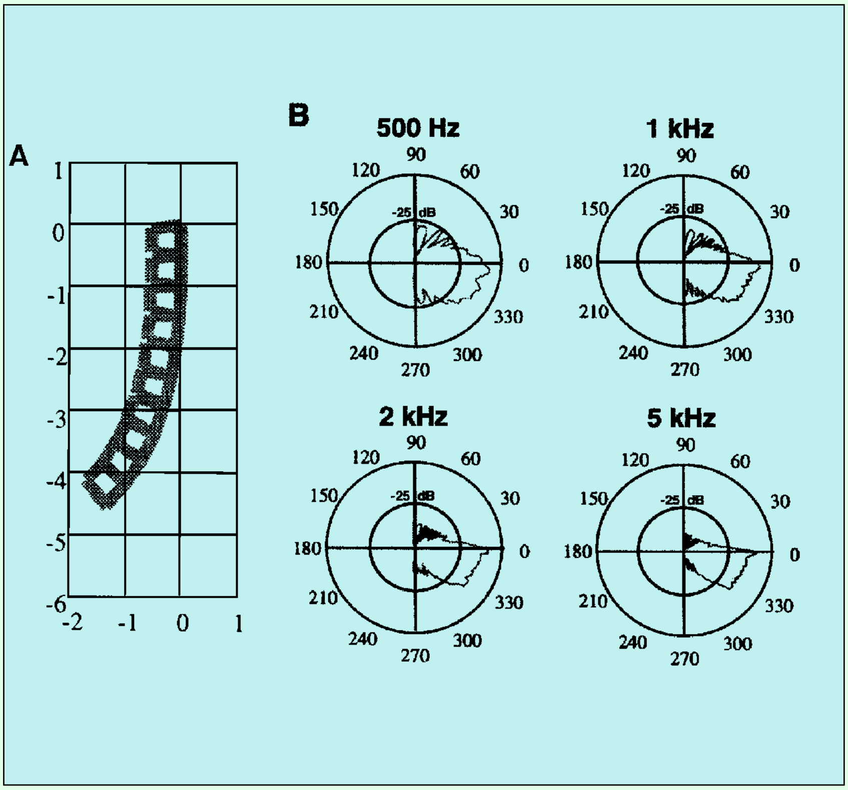
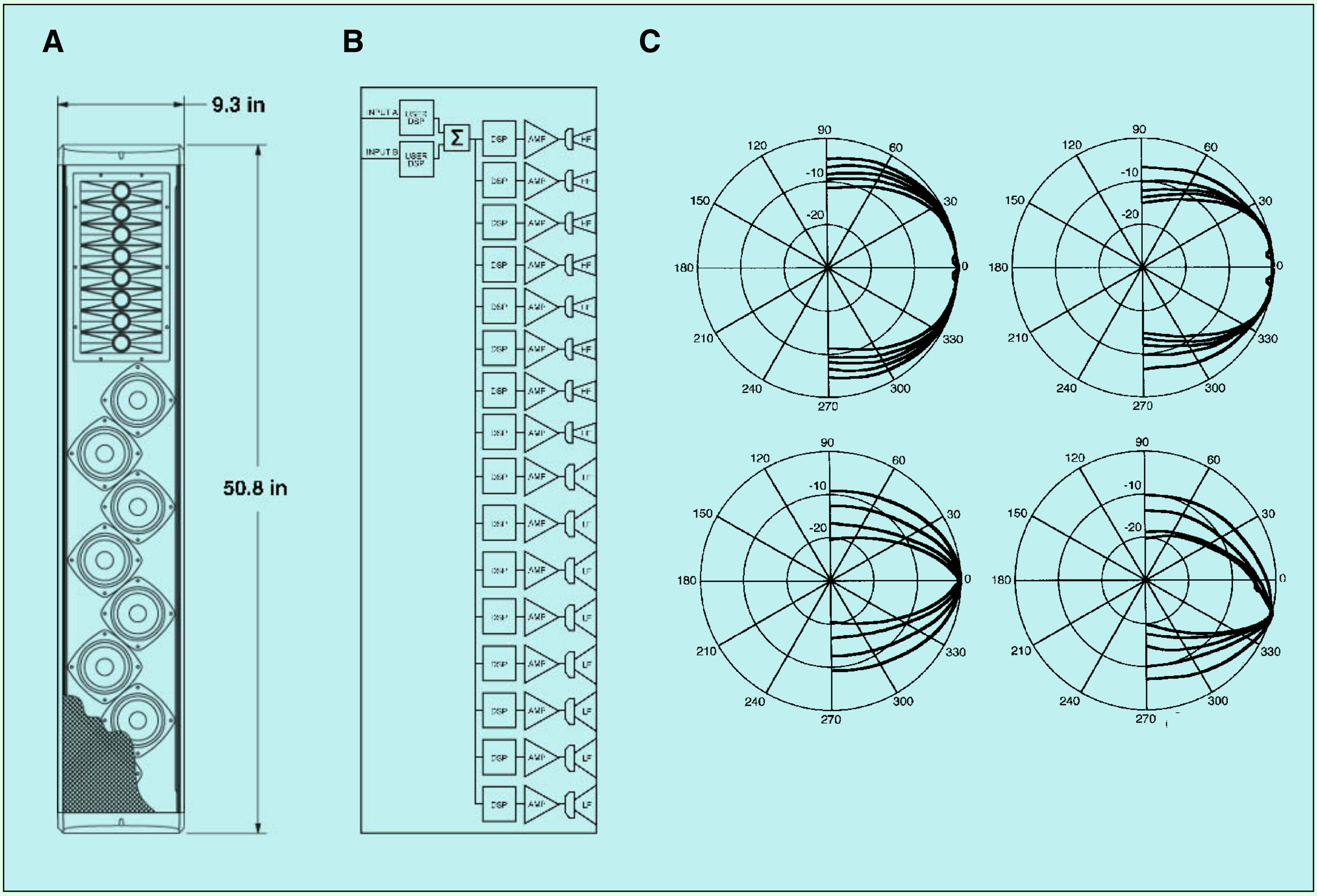



Fig. 34. Performance of a spiral array. Side view of array (A); polar response of array at 500 Hz, 1, 2, and 4 kHz (B). (Data courtesy M. Ureda.)




35. A programmable array. View of array (A); signal flow diagram (B); examples of variable polar response (C). (Data courtesy Eastern Acoustics Works.)

sound pressure levels at considerable distances with relatively low distortion. The primary defect is the dense comb filtering (lobing) and "time smearing" that inevitably results from such a multiplicity of sources covering a given listening position. Actually, since the required acoustic power cannot be achieved by a single source, the aim here should be to keep the coverage at each listening position as uniform as possible. The greater the number of effective sound sources, the finer the lobing patterns and the more uniform the received signal will be. Ideally, we would like the interference peaks and dips among the elements to be well within the ear's critical bandwidths.

## Continuous Line Arrays

A continuous line array is an approximation of a uniformly illuminated "ribbon" of sound, and its directional behavior in the far field can be determined by equation (8). At low frequencies, the far-field for a straight line array begins approximately at a distance equal to the array length divided by π [98]. For progressively shorter wavelengths, this distance increases according to the following equation:

$$r = l^2 f/700 \text{ meters} \qquad (9)$$

where $l$ is the array length, and r and $l$ are in meters [99].

Fig. 33A shows the attenuation pattern with distance from a straight 3-meter array at 10 kHz, and the polar response in the far field of that array is shown at B. Note that the beamwidth (-6-dB) is 0.8°, as given by the equation:

$$\theta_{\text{line array}} = 2 \sin^{-1}(0.6\lambda/l) \qquad (10)$$

where $l$ is the length of the array and λ is the wavelength [99].

The pronounced HF beaming of straight arrays is of course a liability, and articulation of the array is one way of dealing with the problem

### *J and Spiral Arrays*

Ureda [100] describes a J array as having two distinct portions: a straight upper section and a uniformly curved lower section. Each segment of the array will act independently, with the upper section producing a highly directive HF beam for distant coverage and the lower section producing broader radiation for near coverage.

Ureda proposes the spiral array for more uniform overall coverage. The spiral array is continuously curved from the top down, beginning with small angular increments, which increase downward in arithmetic fashion. Fig. 34A shows a side view of such an array with a total length of 5 meters and a terminal angle of 45°.

The directivity function is remarkably uniform with frequency over about a decade. Fig. 34B shows a group of polar plots from 500 Hz to 5 kHz.

## Steerable Arrays

The very large arrays for music discussed earlier consist of elements equally driven in terms of level and bandwidth, and the directional properties are due entirely to the spatial relationships among elements. A steerable array is one in which the elements are fixed in space, with relative drive levels, signal delay, and frequency tapering individually adjustable for each transducer.

Relatively simple arrays can be reconfigured, through sequential timing, to steer their beams as needed [85, 86, 96, 97]. While far-field modeling may be fairly simple, the fact that many listeners are seated in the transition region between near and far fields makes the problems of reconfiguration and uniformity of coverage fairly complex to estimate.

The relatively small system shown in Fig. 35 can be configured via a PC by the user as required. The system profile and signal flow diagram are shown at A and B, and a family of typical far-field polar plots is shown at C. Systems such as these, large or small, are presently used to solve intelligibility problems in a variety of large reverberant spaces.

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