LOCALIZED SOUND REPRODUCTION USING A DIRECTIONAL SOURCE ARRAY

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INTRODUCTION

By localized sound reproduction we mean the generation of a sound field that is clearly audible by a few people within a small region and substantially less audible outside this region so as not to cause interference with the surrounding environment. There are many practical applications for such a system. One example is the presentation of information at a museum exhibit where current techniques require the use of earphones. The notion of an "acoustic spotlight" that provides a localized beam of sound is considerably more appealing.

Figure 1a depicts a conventional loudspeaker array arranged in endfire configuration. Each loudspeaker is driven by an appropriately modified signal replica to form a beam that may be steered. The replicas may be simply delayed versions of each other, with delays determined so that coherent summation of the loudspeaker outputs occurs in the desired beam direction. Alternatively, each loudspeaker may receive a filtered replica to control beamwidth and sidelobe levels. The design flexibility of these arrays is obtained at the cost of considerable system complexity, with separate signal processing and amplification required for each array element. Diffraction around the loudspeakers and supporting structure must also be considered.

In principle, a linear end-fire array can form a narrow beam along its axis but the beam becomes conical if steered off axis. A steerable 'spotlight' beam may be generated by a planar array (not illustrated), however planar arrays involve considerably more transducer elements, and associated system complexity, compared to line arrays with the same baseline.

We consider here two alternative approaches to achieving localized sound reproduction (i) an array of point sources, and (ii) a virtual source array [1].

DUCTED POINT-SOURCE ARRAY

A ducted point-source array is represented schematically in figure 1b. The single transducer is coupled to multiple ducts. Each duct terminates at a desired point source location in the array. The delay at each point source is adjusted by altering the length of the ducts, forming a beam in the (fixed) desired direction. For demonstration, an eight-element ducted point-source array, 1.35 meters long, was constructed using flexible hoses.

Acoustic coupling efficiency into the ducts, radiation efficiency at the point source, reflections within the duct system, and diffraction from the hoses and support structure are the principal challenges of this approach. Benefits include requiring only a single transducer and being very simple to construct.

A manifold is required to couple sound from the transducer into each of the ducts. Impedance mismatch at the branches will reduce the coupling efficiency and cause internal reflections. A cylindrical branching manifold was constructed with a 2.5 cm source aperture at one end and eight 1.5 cm branch ports on the circumference. The dimensions of the cylinder were kept as small as possible (about 12 cm long by 2.5 cm diameter). The coupling efficiency of the cylindrical manifold was adequate to achieve substantial output and the individual array elements combined coherently to beamform over a bandwidth from 200 to 5000 Hz.

Internal reflections occur at the end of each duct, where the radiation impedance does not match the duct impedance.
Since each duct has a different length, and a uniform spacing between the ends (i.e. the point sources) was avoided, the resulting reflections sound less like the echo from a series of pipes and more like a 'diffuse' reverberation.

To obtain accurate phase coherence from the eight point sources, the ducts were first cut to their nominal lengths and the position of their ends was then carefully adjusted along the array. Deviations of a millimeter were evident in the arrival time of sound impulses ("clicks"), as measured at the focal point two meters in front of the array. Constructed in this way, the propagation time to the focal point through each duct was identical.

Figure 2 shows the beam pattern of the demonstration array at 250 Hz (dotted curve) and 4000 Hz (dashed curve). The heavy solid line shows the directivity of the total power for white noise over the frequency band from 500 to 5000 Hz. The main beam becomes increasingly wide at lower frequencies due to the short array length relative to the wavelength. Away from the main axis, partially coherent summation of the point sources causes sidelobes. At higher frequencies, random errors in the point source spacing limit the narrowness of the mainlobe and tend to smooth out the sidelobes. For a broadband signal the peaks and troughs of the sidelobes average out. There is some asymmetry in the response due to the ducts and their support structure.

PARAMETRIC ACOUSTIC ARRAY (PAA)

A high intensity sound traveling though air will be distorted due to nonlinear wave propagation. If the signal consists of two frequencies, f1 and f2, then the nonlinear effects will generate, among other distortion products, a signal at the difference frequency $|f1-f2|$. The difference frequency signal is generated locally in the medium along the propagation path of the high intensity sound and so it forms a 'virtual acoustic source'. There has been much interest in using high intensity ultrasound to generate a parametric acoustic array by choosing f1 and f2 in the ultrasonic region with $|f1-f2|$ in the audio region. A narrow ultrasonic beam will produce a virtual linear end-fire array of sources in the audio frequency range [2,3].

A high power, four element ultrasonic transducer with primary frequency 28 kHz was amplitude modulated at audio frequencies to produce a PAA [4]. Figure 3 shows the beam pattern of the PAA at frequencies close to the extremes of its useable audio range (250 and 4000 Hz) as measured 4 m from the transducer. Comparison of Figures 2 and 3 shows that the PAA is capable of generating an extremely narrow beam (localized sound field) even at low frequencies. There are no significant sidelobes in the PAA directivity pattern and no audio output could be detected in the rear hemisphere (i.e., from 90° to 270°). Substantial harmonic distortion was observed that increased with modulation depth.

Although the virtual array generated along the ultrasonic beam has a fixed end-fire direction, the PAA may be steered by physically aiming the ultrasonic transducer. Harmonic distortion due to the nonlinear generation of the audio can be reduced by pre-distorting the modulation signal, however this requires a wide bandwidth ultrasonic source. It is difficult to obtain simultaneously adequate output power and bandwidth from available transducers. Care must be exercised in the development of loundspeakers employing a PAA to ensure no health hazard arises from exposure to the high power ultrasound.

REFERENCES

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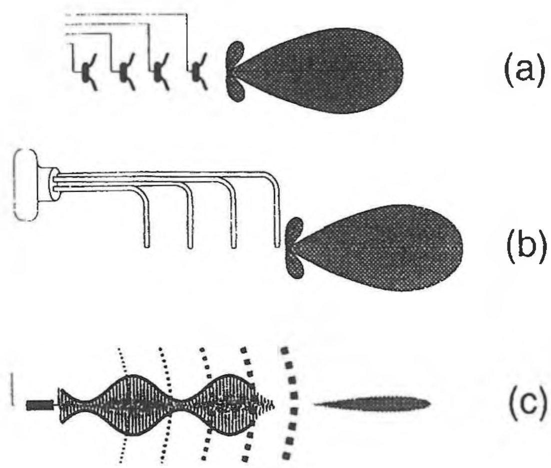

FIGURE 1. End-fire arrays: (a) loudspeakers, (b) ducted point sources, (c) parametric acoustic array.

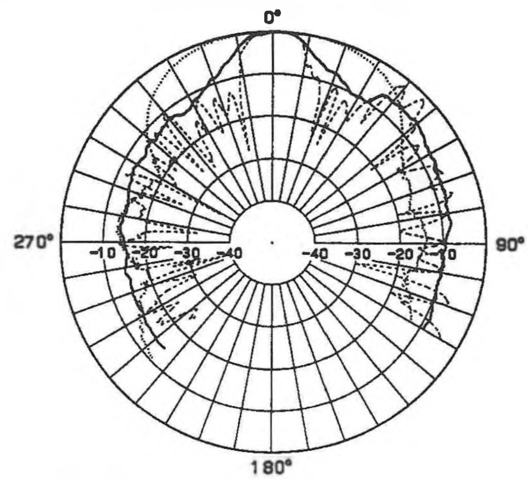

FIGURE 2. Directivity of point source array for signals of 250 Hz (dotted), 4000 Hz (dashed), and 500-5000 Hz white noise (heavy solid).

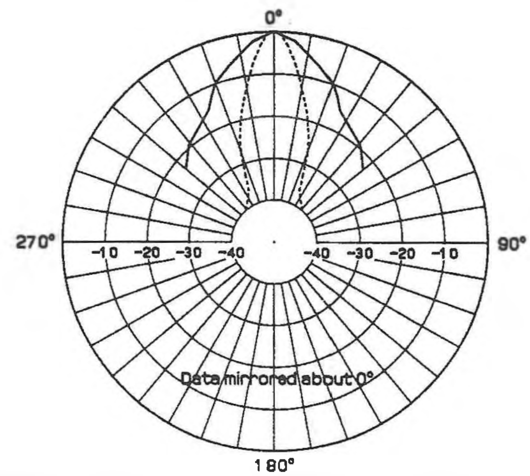

FIGURE 3. Directivity of parametric acoustic array at 250 Hz (solid) and 4000 Hz (dashed).