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A NEW APPROACH TO SIMULATING THE LESLIE SPEAKER

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INTRODUCTION

The Leslie speaker [1] is an electromechanical device which imparts some of the spatial characteristics of a pipe organ to the tone of an electronic organ. This is achieved by rotating horn - like mechanisms which deflect the sound emanating from enclosed loudspeakers. In effect a directional sound source rotates about a vertical axis, and the organ tone is modulated in both phase and amplitude. In the current terminology animation is added. In order to cut the cost, improve reliability and produce a more compact device, there have been many attempts to emulate this modulation electronically. Some of those described in the popular electronics press [2,3] tend to oversimplify the problem, whilst some more sophisticated examples may be found amongst recent patents [4,5], but generally it is the author's opinion that the resultant sound lacks depth when compared with a real Leslie speaker, especially when heard 'live' as opposed to through a p.a. system or via a recording.

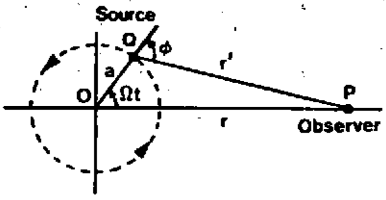

Fig.1 Configuration used to analyse the Leslie speaker.

In this paper an expression is given for the acoustic field of a Leslie speaker in free - field conditions, and then the effects of echoes and reverberation are considered. A phenomenon is found which suggests that to reproduce the Leslie sound convincingly a physically rotating sound beam must be employed. The basis of a solution is proposed in which a small number of loudspeakers is arranged as an array. By applying suitable time varying weighting to the phase and amplitude of the drive to each loudspeaker the beampattern of the Leslie speaker can be closely approximated and made to rotate.

FREE - FIELD ANALYSIS

Schematically the Leslie speaker may be represented as an acoustic source with directional properties described by $D(\phi)$, as shown at position Q in fig.1. This source follows a circular path, radius a, about centre O, and is constrained to point radially away from O at all times. The rate of rotation is Ω rad/sec. The observer is at P, distance r from O and r' from Q. For radiation at frequency ω the acoustic pressure p(t) at P is

$$p(t) = \frac{S(\omega)D(\phi)}{r'} e^{i(\omega t - kr')} \qquad (1)$$

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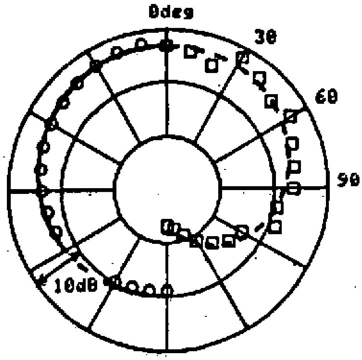

Fig.2 Measured beampattern of LX-7 rotor at 250Hz (circles) and 1kHz (squares) compared with predicted patterns at 250Hz (solid line) and 1kHz (dashed line).

$k=\omega/c$ is the wavenumber and c is the sound speed. $S(\omega)$ is a (possibly complex) function describing the strength of the source.

The directivity function D may be determined by, for instance, modelling the rotor as a piston radiator set in a sphere [6]. Rotary speakers, however, may be found in a variety of shapes and sizes, so it seems more appropriate to use a reasonable but generally applicable approximation. Inspection of measured beampatterns suggests that D might be represented by a cardioid, of the form

$$D(\phi) = 1 + m\cos\phi \qquad (2)$$

Fig.2 shows the cardioid beampattern compared with the measured directivity of a Leslie model LX-7 rotor at 250Hz and 1kHz, towards the extremes of the frequency range of this unit, m being given the empirically estimated value $0.68\log\omega - 3.1$

Eq(1) may be simplified if a far - field condition, $r \gg a$, is specified. Then $r' = r - a\cos\Omega t$, $\phi = \Omega t$ and the real part of the observed pressure is found to be

$$p(t) = A(1 + m\cos\Omega t)\cos(\omega t + ka\cos\Omega t) \qquad (3)$$

Where $A = S(\omega)\exp(-ikr)/r$ is a normalising factor. This expression will be recognised as simultaneous phase and amplitude modulation, where the instantaneous amplitude depends upon the directivity of the source and the instantaneous phase depends upon the radius of the rotor and the signal frequency.

Fig.3a shows the time waveform at 250Hz measured 2.5m from an LX-7 rotor turning at 6.2 rev/sec. The amplitude modulation is clearly visible, but it is not possible to distinguish the individual cycles, so the phase modulation cannot be seen. Fig.3b shows the magnitude of the spectrum of this signal. The multiple sidebands, which are more or less symmetrical about the carrier and spaced at intervals of 6.2Hz, confirm the presence of both phase and amplitude modulation [7].

REVERBERATION

Clearly, echoes and reflections are not necessary for the operation of the Leslie speaker, but reverberation does enhance the sound and the reason for this will now be discussed.

If there is a reflecting surface near the rotary speaker then the observed sound will be a combination of the direct sound and another, similar, signal delayed by some time Δt depending on the difference between the lengths of the direct and reflected paths. If the direct signal is written as $\cos\omega t$ then, neglecting possible amplitude differences, the magnitude of the combined direct and reflected signals is

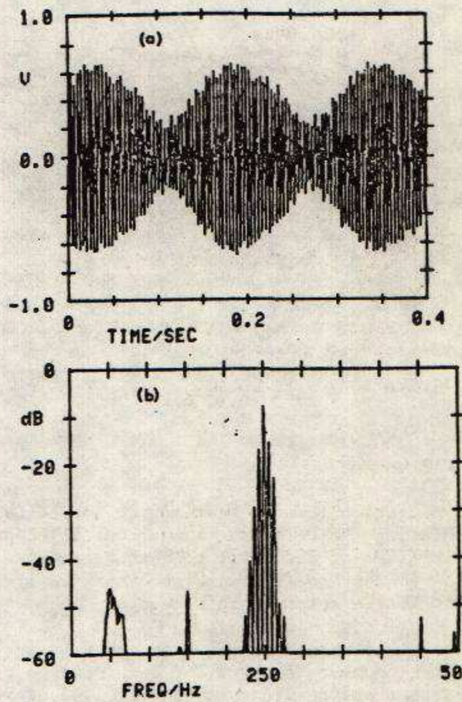

Fig.3 Waveform (a) and spectrum (b) of 250Hz signal measured 2.5m from rotor.

$$|p(t)| = 2|\cos(\Delta t/2)| \tag{4}$$

This represents a comb filter response, and has equally spaced nulls at frequencies $f=n/2\Delta t$, $n=1, 3, 5$ etc. The real situation is rather more complex because the path difference varies with the rotation of the source. Fig.4 shows the response as a function of both time and frequency calculated using eq(3), with the effect of a nearby reflecting surface included, and it can be seen that the dips in the response move in the frequency domain as the rotor turns. The effect is similar to phasing or flanging [8], and imparts an ethereal swirling quality to the sound. In practice, with many reflected paths, the situation may be even more complicated, but it is clear that an effect exists that relies on the physical motion of the sound beam, and that a convincing simulation of the rotary speaker requires more than the basic phase and amplitude modulation.

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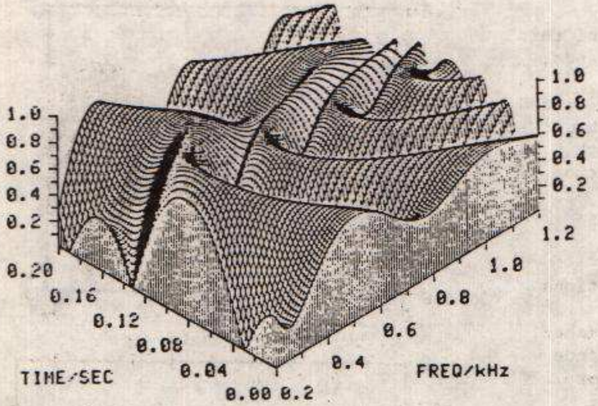

Fig.4 Predicted variation of frequency response with time when rotor is close to a reflecting surface.

THE ARRAY

A directional sound beam may be formed using an array of transducers, and its pointing direction and shape controlled by weighting (shading) the phase and amplitude of the drive to each element. A method has been developed [9] to find the shading coefficients necessary to simulate a particular beampattern, that of the Leslie rotor in this case.

The architecture of the beamformer is shown in fig.5a. The input signal is split into a number of channels, one for each transducer. Each of these channels is further split into two channels, one of which is phase shifted by 90deg, so that the cophase and quadrature components of the shading coefficients may be applied

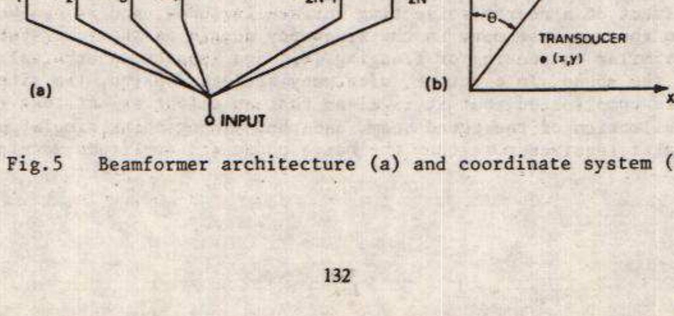

Fig.5 Beamformer architecture (a) and coordinate system (b).

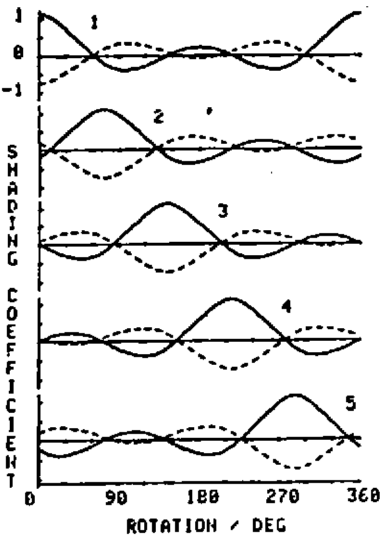

Fig.6 Real (solid lines) and imaginary (dashed lines) parts of drive to 5 element array plotted against beam rotation.

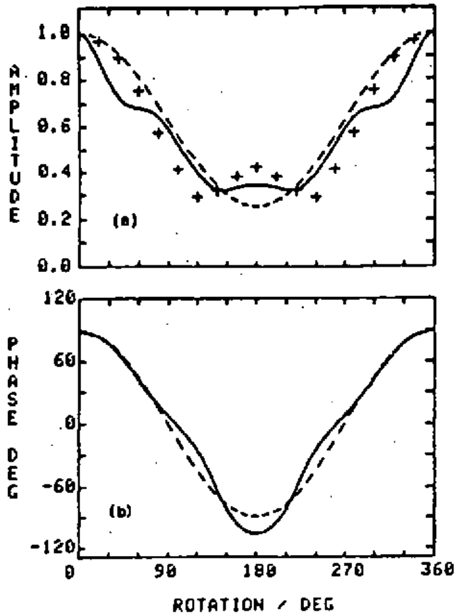

Fig.7 Amplitude (a) and phase (b) modulation of acoustic field of 5 element array (solid lines) compared with eq(3) (dashes) and measured response (crosses) of LX-7 rotor.

separately by the variable attenuators. The outputs from each cophase/quadrature pair are then recombined and passed to the power amplifiers and transducers.

Fig.5b shows a transducer located at an arbitrary point (x,y) in cartesian coordinates. The sound pressure due to an array of N elements, all in the XY plane, observed in the far field in direction θ is

$$Q = e^{i\omega t} \sum_{n=1}^{2N} G_n(\theta,\omega) A_n (\cos u_n + i \sin u_n) \tag{5}$$

where $G_n(\theta,\omega)$ is the directivity of the nth element, the A_n are the required shading coefficients and

$$u_n = \begin{cases} k(x_n \sin\theta + y_n \cos\theta) & ; \text{ n odd} \\ u_{n-1} - \pi/2 & ; \text{ n even} \end{cases} \tag{6}$$

The array output Q may be separated into its real and imaginary parts and each separately equated to the real and imaginary parts of directivity function of the rotary speaker. This will form a system of linear equations which must be solved to find the shading coefficients.

To demonstrate the principle this procedure was carried out for an array of 5 loudspeakers equally spaced around a circle of diameter 18cm. For this demonstration the directivity of the speakers was taken to be a simple cardioid. The required shading coefficients are shown plotted against beam rotation in fig.6, where the solid lines are the cophase components and the dashed lines the quadrature components. Note that the curves for each element are identical except for a phase difference corresponding to the angular separation of the speakers.

The amplitude and phase of the output from this array are plotted against beam rotation at a frequency of 500Hz, about the middle of the range of interest, in figs.7a and 7b respectively. In each plot the solid line is the calculated array output and the dashed line is the required response from eq(3). The crosses show the measured results for the LX-7 rotor. The simulated array output closely approximates the desired phase and amplitude modulation, but would probably be improved by using a greater number of loudspeakers and the problems and compromises involved in realising a working system will now be discussed.

PRACTICAL PROBLEMS

A number of problems arise when trying to design a practical system, firstly in selecting the size of the array and the number of elements, and secondly in implementing the beamformer electronics.

A circular array with equally spaced elements was chosen in the example above for symmetry, the same beampattern being desired in any direction. If the array diameter were the same as that of the Leslie rotor (33cm for the LX-7) then the required beampattern could be reproduced exactly and the imaginary parts of the shading coefficients would vanish, simplifying the beamformer. To prevent steps appearing in the pattern, as in the example, the elements must be closely spaced, certainly within $\lambda/2$ at the highest frequency of interest ($\sim$2kHz), and a 33cm array would require 12 speakers, with 12 power amplifiers and 12 channels in the beamformer.

A smaller array is desirable to reduce both the bulk and cost of the device, but neither the phase nor amplitude response would exactly match the desired response over the entire frequency range. Whether or not this degradation in performance would be noticeable to a listener still requires investigation.

The beamforming system includes a unity gain frequency invariant 90 deg phase shift. Such a circuit does not exist in practice, and the choice is between an all pass filter with constant gain but frequency varying phase, or else an integrator or differentiator with the desired 90 deg phase shift but a gain that changes with frequency. Such responses can be included in eqs.(5-6) but the ability to produce the desired beampattern over a broad frequency range may be impaired, and this too requires further investigation.

Other problems may arise, such as acoustic interaction between the elements of the array, and these points are the subject of continuing work. It is hoped to publish a more comprehensive paper describing the performance of a prototype shortly.

CONCLUSIONS

The acoustic field of a rotary speaker has been examined and, when reverberation was considered, it was found that an effect exists which relies on the physical motion of the sound beam and that a convincing simulation of the Leslie speaker requires more than the basic phase and amplitude modulation.

A technique for forming a rotating sound beam using an array of fixed loudspeakers has been suggested and the feasibility of such a system demonstrated, subject to the solution of a number of practical problems.

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