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This work was done to devise a new type of loudspeaker. The theory for sound reproduction of this loudspeaker is based on nonlinear acoustics of sound wave interaction in air. A finite amplitude ultrasound wave that can be amplitude modulated by any audio signal is radiated from a transducer array into air as the primary wave. As a result, an audio signal is produced in the air because of the self-demodulation effect of the AM sound wave due to the nonlinearity of the air. It is possible to get a flat characteristic of reproduced sound pressure by using an equalizer. In some fundamental experiments the characteristic of the reproduced sound pressure is not quite flat due to an imperfect transducer array. Improvement of the transducer makes it possible to get a flat characteristic. A special feature of this loudspeaker is its very sharp directivity pattern, which makes it possible to realize a sound spotlight.

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INTRODUCTION

The nonlinear interaction of finite amplitude ultrasonic waves in the air can be applied to a loudspeaker.¹ This paper describes the fundamental concept of a loudspeaker based on the nonlinear interaction of sound waves in air. Also, some of the experimental results from the operation of a prototype loudspeaker are presented.

When two finite amplitude sound waves (primary waves), having different frequencies, interact with one another in a fluid, new sound waves (secondary waves) whose frequencies correspond to the sum and the difference of the primary waves may be produced as the result.

This phenomenon was first analyzed by Westervelt² and is well known as "nonlinear interaction of sound waves," or the "scattering of sound by sound."³ Based on Lighthill's arbitrary fluid motion equation⁴ as shown in Eq. (1), Westervelt derived an inhomogeneous wave equation which is satisfied by the sound pressure of secondary waves produced by the nonlinear interaction [Eq. (2)].

$$\frac{\partial^2 \rho}{\partial t^2} - c_0^2 \nabla^2 \rho = \frac{\partial^2 T_{ij}}{\partial x_i \partial x_j}, \quad (1)$$

ρ : density of fluid, T_{ij} : stress tensor,

$$\nabla^2 p_s - \frac{1}{c_0^2} \frac{\partial^2 p_s}{\partial t^2} = -\rho_0 \frac{\partial q}{\partial t}, \quad (2)$$

$$q = \frac{\beta}{\rho_0^2 c_0^4} \frac{\partial}{\partial t} p_1^2.$$

In Eq. (2), p_s is the secondary wave sound pressure, p_1 is the primary wave sound pressure, β is the nonlinear fluid parameter, and c_0 is the small signal sound velocity.

The solution for Eq. (2) may be expressed by the superposition integral of the Green's function and the virtual second source [right side of Eq. (2)] as shown in Eq. (3).

$$p_s = \frac{\rho}{4\pi} \iiint_v \frac{1}{|\mathbf{r} - \mathbf{r}'|} \frac{\partial}{\partial t} q\left(\mathbf{r}, t - \frac{|\mathbf{r} - \mathbf{r}'|}{c_0}\right) d\mathbf{r}', \quad (3)$$

where \mathbf{r} is the observation point position vector, \mathbf{r}' is the source position vector and v is the nonlinear interaction space.

When the primary wave consists of two continuous sinusoidal waves and both are planar and well collimated, the integral of Eq. (3) is calculated in the same manner as in previous papers.^{2,5} When the directivity of a circular piston is taken into consideration, however, Eq. (3) must be used with the expression of Muir *et al.*⁶

A new type of loudspeaker has been developed on the basis of the nonlinear interaction of sound waves mentioned above. In this type of loudspeaker, ultrasound is amplitude modulated by an audio signal and radiated from a transducer array as finite amplitude waves. When the amplitude-modulated ultrasound wave interacts in a nonlinear fashion in air, the modulated signal (the audio signal) can be demodulated in the air.¹

In the following section, the principle underlying this type of loudspeaker is described.

I. THEORY

A. Acoustic reproduction by nonlinear interaction of AM ultrasound in air

When two sinusoidal sound waves are radiated in the air, two new waves with angular frequencies of $\omega_1 \pm \omega_2$ arise by nonlinear interaction of the two original sinusoidal waves, whose angular frequencies are ω_1 and ω_2 .

Therefore one might expect the secondary wave which corresponds to the modulation signal, to appear in the air as a result of the nonlinear interaction between the carrier ultrasound and the lower and upper sideband waves, provided that a finite amplitude AM ultrasound wave is radiated into

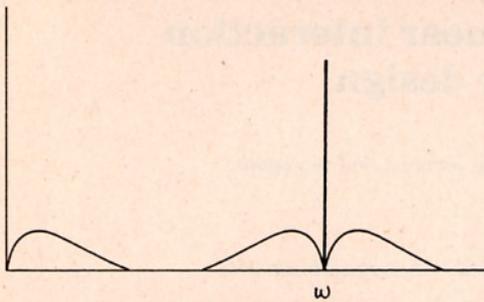


FIG. 1. Frequency spectra of an AM wave and demodulated wave.

the air. That is, the AM ultrasound is self-demodulated by the nonlinear interaction.

Figure 1 shows the spectra for both an AM wave and a demodulated wave. In this case, since the modulation wave is reproduced in the air, a new type of loudspeaker can be devised if the modulation signal is selected as the program audio signal.

If a finite amplitude ultrasound beam, modulated by an audio signal $g(t)$, is radiated into the air from a transducer array, the sound pressure p_1 of the primary wave (AM wave) at a distance x from the array on axis may be represented by Eq. (4)

$$p_1 = p_0 [1 + mg(t - x/c_0)] e^{-\alpha x} \sin \omega_0(t - x/c_0), \quad (4)$$

where p_0 is the initial sound pressure of the ultrasound, m is the parameter indicating modulation index, and α is the absorption coefficient of carrier sound.

A virtual audio signal source occurs in the primary sound beam because of the nonlinearity of the acoustic interaction in air. This sound source may be represented by Eq. (5) using Eq. (2) and Eq. (4)

$$q = \frac{\beta p_0^2}{\rho_0^2 c_0^4} e^{-2\alpha x} \frac{\partial}{\partial t} \left[mg \left(t - \frac{x}{c_0} \right) + \frac{1}{2} m^2 g^2 \left(t - \frac{x}{c_0} \right) \right]. \quad (5)$$

In the above equation, the second term on the right side implies a harmonic distortion component arising from the interaction between the lower and upper sideband waves. If the primary sound beam cross section is assumed to be circular with radius a , then the demodulated audio sound pressure p_s at the point r from the array, on axis, can be calculated analytically using Eqs. (3) and (5) in the form

$$p_s = \frac{\beta p_0^2 a^2 m}{8 \rho_0 c_0^4 \alpha r} \frac{\partial^2}{\partial t^2} g \left(t - \frac{r}{c_0} \right). \quad (6)$$

On the other hand, the sound pressure of a harmonic distortion component may be expressed as

$$p_d = \frac{\beta p_0^2 a^2 m^2}{16 \rho_0 c_0^4 \alpha r} \frac{\partial^2}{\partial t^2} g^2 \left(t - \frac{r}{c_0} \right). \quad (7)$$

The Fourier transform of Eq. (6) can be expressed as

$$P_s(\omega) = -(\beta p_0^2 a^2 m / 8 \rho_0 c_0^4 \alpha r) \omega^2 \exp[-j(r/c_0)\omega] G_s(\omega), \quad (8)$$

where $P_s(\omega)$ is the Fourier transform of $p_s(t)$, and $G_s(\omega)$ is the Fourier transform of $g(t)$. As evident from Eq. (8), $P_s(\omega)$ is proportional to ω^2 and thus the frequency characteristics of the reproduced sound show a 12 dB/oct dependence. Consequently, the audio signal (modulation signal) must be pro-

cessed by an equalizer having -12 dB/oct frequency characteristics before the audio signal is introduced into the AM modulator.

B. Harmonic distortion

In the case of pure-tone modulation, $g(t) = \sin \omega t$, the sound pressures arising from both the signal secondary wave and the second harmonic distortion signal are calculated from Eqs. (6) and (7), respectively,

$$p_s(t) = -(\beta p_0^2 a^2 m \omega^2 / 8 \rho_0 c_0^4 \alpha r) \sin \omega(t - r/c_0), \quad (9)$$

$$p_d(t) = (\beta p_0^2 a^2 m^2 \omega^2 / 8 \rho_0 c_0^4 \alpha r) \cos 2\omega(t - r/c_0). \quad (10)$$

From these equations, it is possible to define the second harmonic distortion ratio as follows

$$\epsilon = [|p_d(t)| / |p_s(t)|] \times 100 = m \times 100\%. \quad (11)$$

Because the second harmonic distortion ratio is proportional to m , a good distortion ratio requires a very small modulation depth to prevent cross interaction between the lower and upper sideband waves. The signal and distortion sound waves are represented by the first and the second term on the right side of Eq. (5), respectively. The sound pressure of the signal is proportional to m , while the distortion is proportional to m^2 . In accordance with this relation, if m is selected less than 1, the distortion sound pressure will be much less than the signal sound pressure.

If the equalizer of -12 dB/oct is used, the modulation depth m varies with the frequency of the modulation signal, as expressed in Eq. (12)

$$m = m_0 / \omega^2, \quad m_0 \text{ is constant.}$$

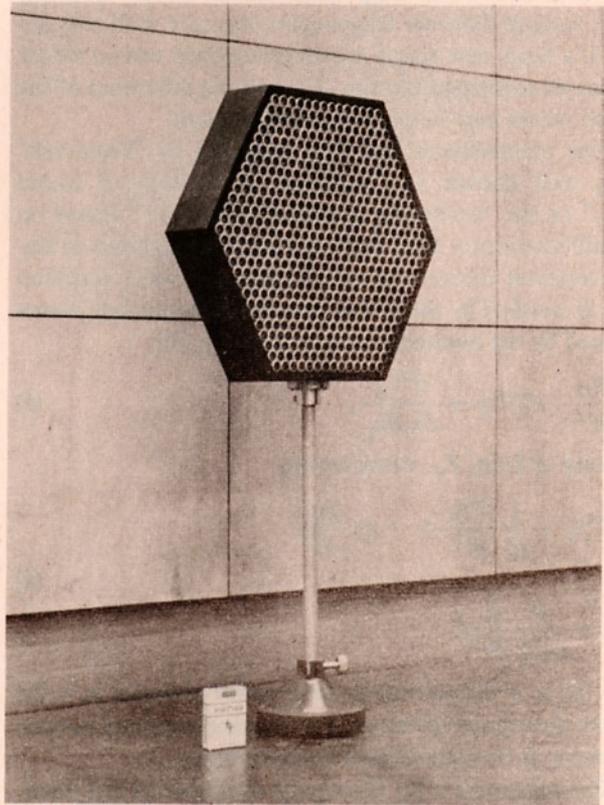


FIG. 2. Front view of the loudspeaker.

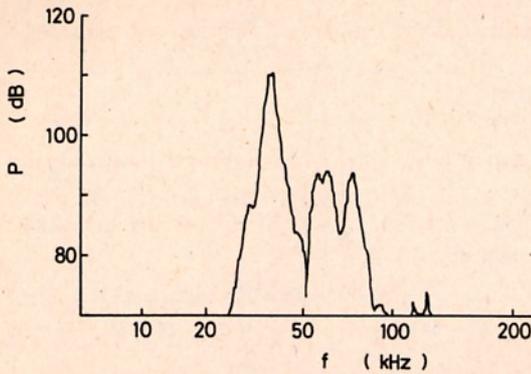


FIG. 3. Sound pressure-frequency response characteristics of the transducer array, for a point 4 m from the transducer. The input voltage is 0.5 V.

In this case, since the second harmonic distortion ratio ϵ is proportional to $1/\omega^2$, distortion in low-frequency regions increases markedly.

That ϵ is proportional to $1/\omega^2$, in spite of the flatness of the signal frequency characteristics, is due to the fact that $p_d(t)$ is proportional to m^2 even though $p_s(t)$ is proportional to m . If m is kept small to make distortion low, the sound pressure $p_s(t)$ also decreases.

Therefore, either the initial sound pressure p_0 of the carrier wave or the radius of the primary beam cross section should increase to maintain the expected $p_s(t)$.

II. EXPERIMENT

A loudspeaker using a finite amplitude AM ultrasound radiated from a transducer array was developed and put to practical use. This array consisted of 547 PZT bimorph transducers. The fundamental resonant frequency of each transducer was about 40 kHz. A front view of the array appears in Fig. 2.

The sound pressure frequency response characteristics

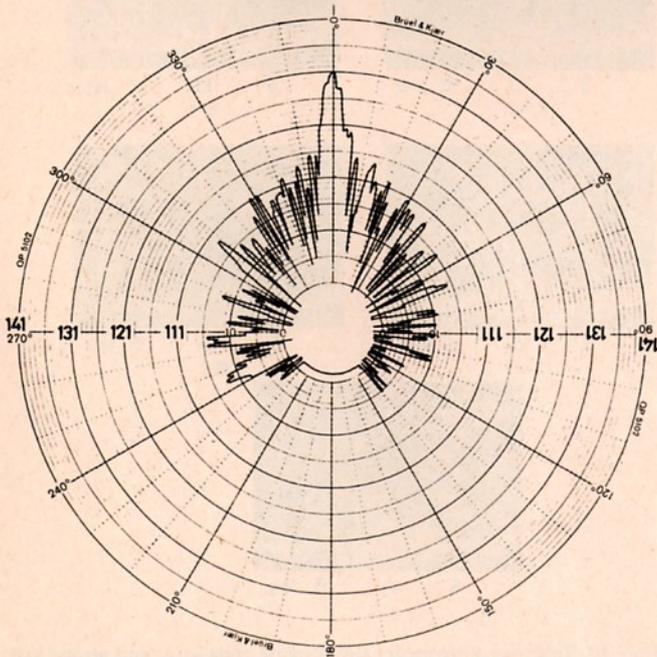


FIG. 4. Directivity at 40 kHz of the transducer array, for a point 4 m from the transducer. The input voltage is 10 V.

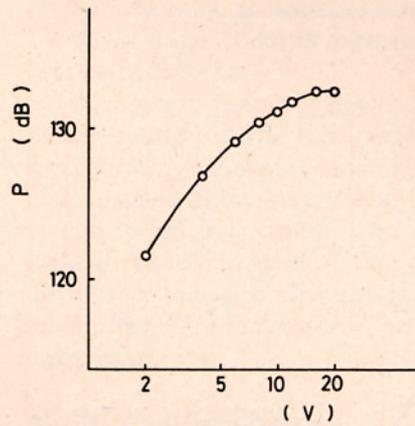


FIG. 5. Sound pressure versus input voltage at 40 kHz, for a point 4 m from the transducer.

and the directivity at 40 kHz (the primary wave) of the array are shown in Figs. 3 and 4, respectively. As can be seen from Fig. 3, the frequency response characteristics of the array are not symmetrical for 40 kHz. Moreover, there are many harmonic resonances and antiresonances. The frequency response characteristics of the secondary sound wave are distributed by the resonances and antiresonances.

Figure 5 shows the sound pressure at 40 kHz, at a point 4 m from the array, plotted against input voltage.

The sound pressure frequency response characteristics of the secondary wave produced by the nonlinear self-interaction of the finite amplitude AM ultrasound radiated from the array, are shown in Fig. 6. The characteristics were measured with modulation depth $m = 0.5$ at a point 4 m from the array in an anechoic chamber. The 12 dB/oct equalizer was not used. In the frequency region below 1.5 kHz, the characteristics almost follow the 12 dB/oct curve. The sound pressure characteristics of the primary wave have a flat region within the frequencies of 40 ± 1.5 kHz as shown in Fig. 3. When the sideband spectra of the modulated ultrasound deviates from the flat range, the sound pressure of the secondary wave decreases. The peak of the primary sound pressure curve at 60 kHz produces the peak of the secondary wave at 20 kHz. All of these phenomena can be predicted

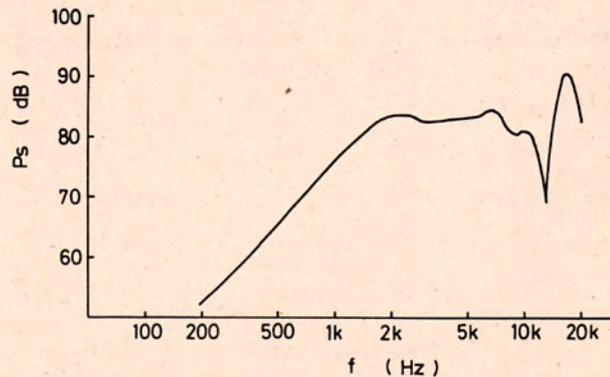


FIG. 6. Sound pressure-frequency response characteristics of secondary wave, for a point of 4 m, $m = 0.5$, and input voltage of 10 V.

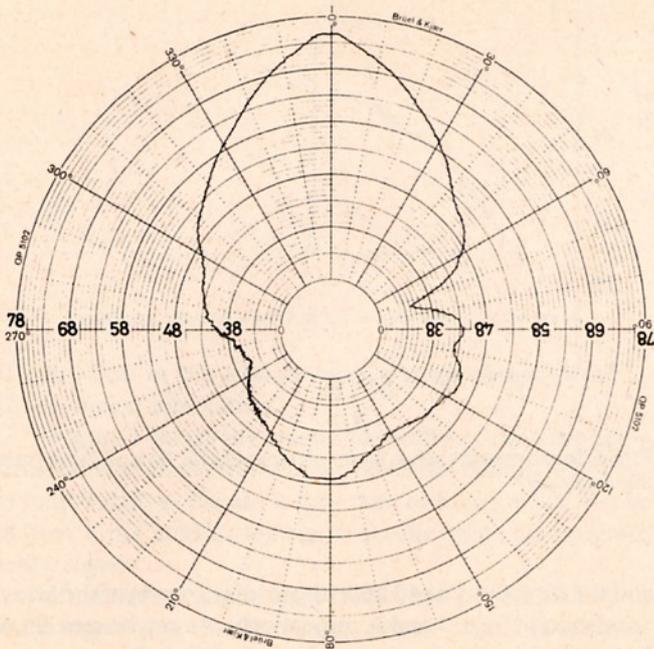


FIG. 7. Directivity of secondary wave at 1.0 kHz, for a point of 4 m, $m = 0.5$, and input voltage of 10 V.

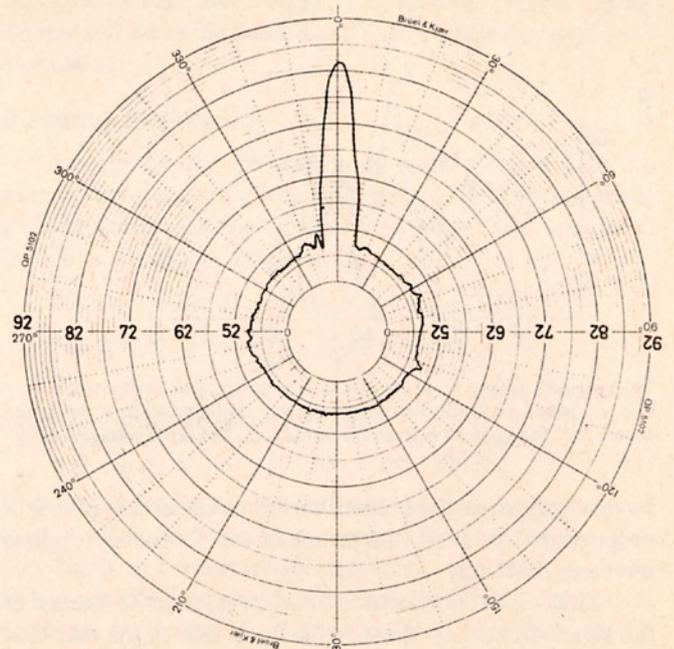


FIG. 9. Directivity of secondary wave at 10.0 kHz, for a point of 4 m, $m = 0.5$, and input voltage of 10 V.

from Eq. (9) and the characteristics of the primary wave.

The measured directivities of the secondary signal waves at 1.0, 5.0, and 10.0 kHz are shown in Figs. 7, 8, and 9, respectively.

To check the relation between the secondary signal sound pressure $p_s(t)$ and second harmonic component sound pressure $p_d(t)$ of the secondary wave, the secondary wave picked up by audio microphone was analyzed by a spectrum analyzer for various values of m . Figure 10 shows the mea-

surement results at $f_s = 5.0$ kHz. These results show that the relation of the sound pressure level between signal and distortion are predicted by Eqs. (9) and (10). For example, if the results of $m = 1.0$ and $m = 0.5$ are compared, it is clear that the signal level (i.e., 5 kHz) decreases 6 dB and the second

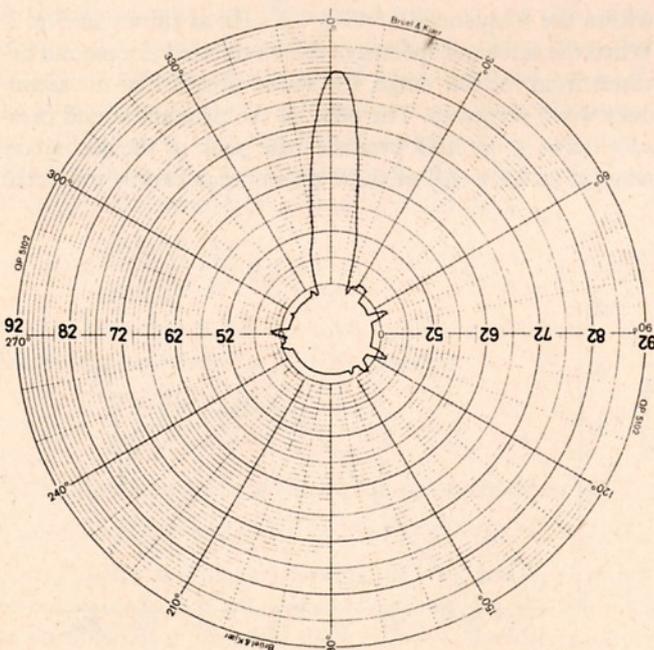


FIG. 8. Directivity of secondary wave at 5.0 kHz, for a point of 4 m, $m = 0.5$, and input voltage of 10 V.

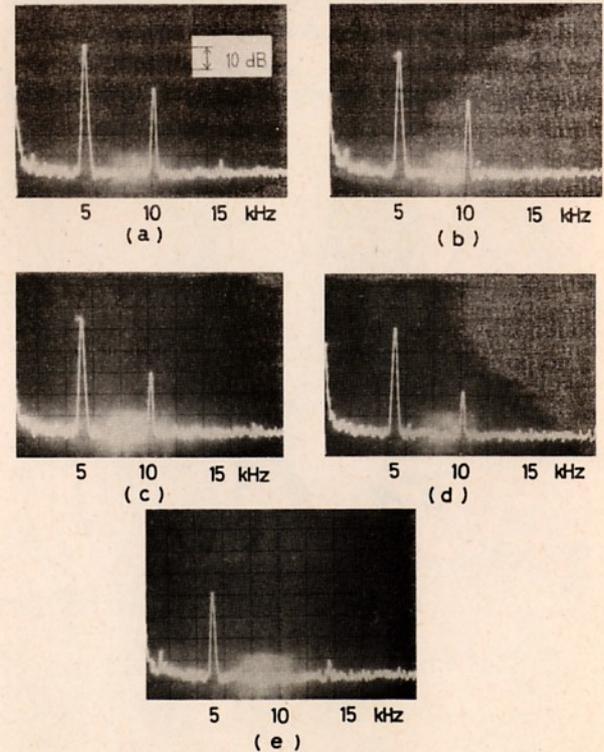


FIG. 10. Relations of secondary signal sound pressure p_s and second harmonic sound pressure p_d , (a) $m = 1.0$, (b) $m = 0.7$, (c) $m = 0.5$, (d) $m = 0.3$, and (e) $m = 0.1$.

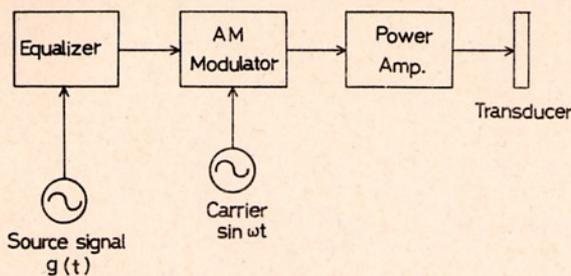


FIG. 11. Construction of the loudspeaker.

harmonic distortion level (10 kHz) decreases 12 dB. Accordingly, the signal sound pressure is proportional to m and that of the distortion is proportional to m^2 .

III. DISCUSSION

An entirely new type of loudspeaker has been developed. This research is based on the phenomenon of the nonlinear interaction of sound waves. That is, the self-modulation effect of finite amplitude AM ultrasound by the nonlinearity of the air has been applied in the construction of the loudspeaker. This loudspeaker consists of an ultrasound transducer array, a driving amplifier for the array, an AM modulator, a pure-tone oscillator for the carrier frequency and equalizer as shown in Fig. 11.

The sound pressure obtained from the loudspeaker is proportional to the depth m of the modulation. However, m should be as small as possible because the second harmonic distortion ratio ϵ is equal to m . The sound pressure of the secondary wave is also proportional to the square of the initial sound pressure p_0 of the carrier sound and the square of the beam radius a . These values must be as large as possible to obtain adequate sound pressure for practical use.

Since the frequency response of the secondary wave has

ω^2 characteristics, an equalizer is required for flat response. Usually, it is quite difficult to produce low-frequency sound because of distortion.

One special feature of this loudspeaker is its very sharp directivity pattern. This loudspeaker can be used as a sound spotlight. Since an acoustic spotlight has never existed in an audible sound region, various uses for this loudspeaker may be anticipated. For example, the sharp directivity would make it possible to speak to one group of people without disturbance to neighboring groups. In a museum or an exhibit, expensive sound barriers between exhibits would be unnecessary.

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¹M. Yoneyama, Y. Kawamo, J. Fujimoto, and S. Sasabe, "An application of nonlinear parametric interaction to loudspeaker," Meeting of Institute of Electronics and Communication Engineers of Japan, Paper EA81-65 (1982).

²P. J. Westervelt, "Parametric Acoustic Array," *J. Acoust. Soc. Am.* **35**, 535-537 (1963).

³R. T. Beyer, "Nonlinear Acoustics," *Navel Ship Command* (1974).

⁴M. J. Lighthill, "On sound generated aerodynamically, I," *Proc. R. Soc. London A211*, 564-587 (1952).

⁵H. O. Berkta, "Possible exploitation of nonlinear acoustics in underwater transmitting applications," *J. Sound Vib.* **2**, 435-461 (1965).

⁶T. G. Muir and J. G. Wille, "Parametric acoustic transmitting arrays," *J. Acoust. Soc. Am.* **52**, 1481-1486 (1972).

