Acoustic characteristics of an electrodynamic planar digital loudspeaker[a)]

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In this paper, an electrodynamic planar loudspeaker driven by a digital signal is experimentally discussed. The digital loudspeaker consists of 22 voice coils, 11 permanent magnets, a diaphragm with streamlined sections molded in plastic, and a suspension made of handmade Japanese paper between the diaphragm and the frame. First, the acoustic responses are affected by the arrangement of the voice coils, so an asymmetric arrangement is studied. This asymmetric arrangement is designed to obtain as flat a frequency response to an analog signal as possible. This arrangement is compared with a symmetric one and results show that the flatness of the frequency response around 1 kHz and 4 kHz is improved and that the sound reproduction band is from 40 Hz to 10 kHz. Second, to evaluate the acoustic responses to a digital signal, the digital loudspeaker is driven with a pulse code modulation signal. Results show that the digital loudspeaker can reproduce pure sound with a total harmonic distortion of less than 5% from 40 Hz to 10 kHz, exceeding this value only in a narrow frequency band near 4 kHz. This digital loudspeaker was demonstrated to have good linearity over its dynamic range of 84 dB.
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I. INTRODUCTION

For decades, various kinds of audio and visual media, such as voice used in cellular phones and movies recorded on DVDs, have been digitized thanks to advancements in signal processing. For these media, digital electro-acoustic transducers that can be connected directly with many kinds of digital equipment are desirable. However, digital transducers have not advanced enough for practical use yet. The reasons are (1) it is very difficult to sample and quantize directly the mechanical vibrations from a microphone because a microphone's displacement is extremely small; (2) it is not easy to control the driving forces in a loudspeaker on a bit by bit basis.

The first digital loudspeaker was made by Flanagan.[1] This loudspeaker consisted of a round piece of electret foil, which was 46 mm in diameter. The area of the fixed electrode was divided into concentric circles and driven by 4-, 5-, and 6-bit linear PCM (pulse code modulation) signals with a constant DC voltage ($E = 20$ V). After that, Hohm *et al.*[2] made a loudspeaker with a piezoelectric polymer film and experimentally studied its acoustic properties. By using a logarithmic quantized signal (A-law), higher performance and speech clarity was obtained. Also, Yanagisawa *et al.*[3] made a piezoelectric type 16-bit loudspeaker, i.e., the same number of bits used in CD and DAT recording. However, the sound pressure level was low and the frequency response was poor.

Compared with the above-mentioned loudspeakers, electro-dynamic types are expected either to be easier to make or to have a larger sound pressure level. Inanaga *et al.*[4] used D-A conversion with an electrodynamic loudspeaker, which had a conical shape with a diameter of 38 cm. Driven directly with weighted voltages, an 8-bit digital loudspeaker could achieve a frequency range of 300 Hz to 3 kHz. However, either the dynamic range or the sound reproduction band was narrow, so it did not have enough capacity for reproducing music.

In this paper, an electrodynamic planar digital loudspeaker is presented which takes into account the above limitations. In order to achieve an adequate sound pressure level and bit resolution, the loudspeaker was made with a diaphragm composed of streamlined sections and 22 voice coils. However, the acoustic responses, such as frequency and distortion, are affected by the selection of the voice coils. Therefore, in order to improve the responses, (a) the combination and (b) the arrangement of the voice coils are discussed. As for (a), pairs of voice coils are chosen by analyzing the results of each acoustic response when the loudspeaker is driven by an analog signal. In regards to (b), each bit is assigned to a voice coil based on the result of the best combination in (a).

In order to compare a digital loudspeaker with the above asymmetric arrangement to one with a symmetric arrangement, experiments were performed in which the loudspeakers were driven by a weighted discrete voltage with a maximum amplitude of 16 V_{p-p} and a resolution of 16 bits. In these experiments, the output waveform, frequency response,

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linearity, total harmonic distortion, distribution of sound intensity level, and directional pattern of both arrangements were evaluated.

II. ELECTRODYNAMIC PLANAR LOUDSPEAKER

In order to accomplish $D-A$ conversion at the loudspeaker directly, the binary numbers representing quantized samples of the original signal need to be decoded to PAM (pulse amplitude modulation) form and then filtered to recover the analog acoustic output. In this section, the operating principles of the digital loudspeaker and its system are described.

To increase the sound pressure level and achieve multibit resolution, an electrodynamic loudspeaker was proposed where one voice coil corresponds to one bit. However, as several voice coils were integrated into a single bobbin, a trade-off between the efficiency and resolution results because of the gap in the magnetic circuit. It is possible to achieve a resolution of up to 8 bits using a concentrically driven loudspeaker.[4,5] On the other hand, it has been reported in theory that a multiple drive panel loudspeaker has a flat frequency and omni-directional response even at high frequencies.[6] Hence, a planar type loudspeaker with multiple magnets was chosen for our digital loudspeaker.

A. Operating principles

To reduce distortion and quantization noise each sample must be represented by at least a 16-bit number giving $\pm 32\,768$ values per sample. An electrodynamic digital loudspeaker driven by a sign magnitude binary (SMB) signal is effective.[1–5] SMB is the simplest and one of the most obvious methods of encoding positive and negative numbers. The most significant bit (MSB: b_{16}) is assigned to be the sign bit. If the sign bit is 0, this means the number is positive. If the sign bit is 1, then the number is negative. The remaining 15 bits $(b_{15}, b_{14}, b_{13}, \ldots, b_2, b_1)$ are used to represent the magnitude of the binary number in unsigned binary notation. Therefore, the formula to convert the 16-bit input signal with a length of 16 bits is

$$V_p(t) \propto (1-2b_{16}) E_0 \sum_{i=1}^{15} 2^{(i-15)} b_i(t), \tag{1}$$

where V_p is the total velocity amplitude of the panel surface.

Figure 1 shows the schematic of an electrodynamic digital loudspeaker consisting of a panel, a suspension, a frame and parallel exciters with magnets and voice coils. Its operating principle is based on Fleming's left-hand rule. Provided that the velocity of the diaphragm $V_i(t)$ is controlled to be in proportion to the driving force $F_i(t) = BlI_i(t)$ of the ith exciter, the volume velocity is proportional to the input voltage $(1-2b_{16}) E_0 \cdot 2^{(i-15)} b_i(t)$.

The linear dynamic properties are described by two operating equations: an electrical one and a mechanical one given by

$$(1-2b_{16}) E_0 \cdot 2^{(i-15)} b_i(t) = (Z_{0i} + Z_{ei}) I_i(t) + BlV_i(t), \tag{2}$$

$$0 = (Z_r + Z_p) V_i(t) - BlI_i(t). \tag{3}$$

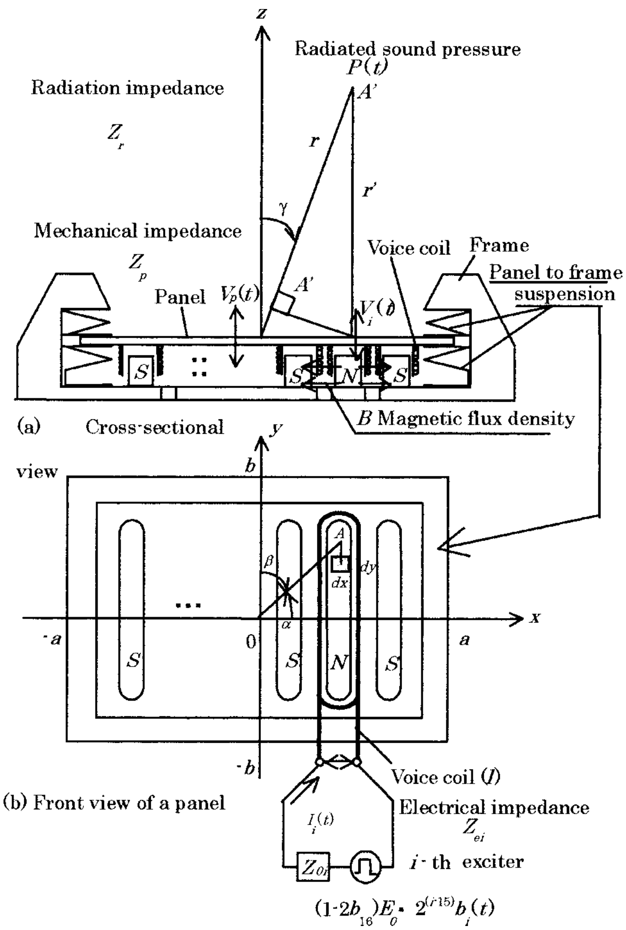

FIG. 1. Schematic of an electrodynamic planar digital loudspeaker consisting of a panel, a suspension, a frame and parallel exciters with magnets and voice coils.

The symbols used are

$(1-2b_{16}) E_0 \cdot 2^{(i-15)} b_i(t)$	pulse power supply voltage applied the ith exciter (V)
E_0	voltage source (V)
Z_{0i}	internal electrical source impedance (Ω)
Z_{ei}	electrical impedance of the ith voice coil (Ω)
$I_i(t)$	current in the ith voice coil (A)
B	magnetic flux density (Wb/m^2)
l	length of the ith voice coil (m)
Bl	force factor (N/A)
$V_i(t)$	velocity amplitude of the surface of the panel driven by the ith exciter (m/s)
Z_r	radiation impedance (N s/m)
Z_p	mechanical impedance of the load including the panel and the exciter assembly (N s/m)

From the electrical Eq. (2), the current in the ith voice coil is

$$I_i(t) = \frac{(1-2b_{16}) E_0 \cdot 2^{(i-15)} b_i(t)}{Z_{0i} + Z_{ei} + \dfrac{(Bl)^2}{Z_r + Z_p}}. \tag{4}$$

From the mechanical Eq. (3), the velocity of the panel driven by the ith exciter is

$$V_i(t)=\frac{Bl}{Z_r+Z_p}I_i(t). \quad (5)$$

The total velocity amplitude of the surface of the panel driven by 15 exciters, which are assumed to be a rectangular plane rigid piston mounted flush in an infinite plane baffle, is

$$V_p(t)=\sum_{i=1}^{15} V_i(t)=\frac{Bl}{Z_r+Z_p}\sum_{i=1}^{15} I_i(t). \quad (6)$$

Therefore, when the signal defined in Eq. (6) is passed through a mechanical low-pass filter, the analog drive velocity $V_p(t)$ is

$$V_p(t)=\sum_{n=-\infty}^{\infty}\frac{Bl}{Z_r+Z_p}\sum_{i=1}^{15} I_i(n\pi/\omega)\frac{\sin\omega(t-n\pi/\omega)}{\omega(t-n\pi/\omega)}, \quad (7)$$

where ω is the angular frequency.

In general, the radiation produced by the vibration of the surface of the rectangular panel [$2a\times 2b$ (m^2) as shown in Fig. 1] does not have symmetric spherical radiation patterns characteristic of a simple source. The radiation produced by the panel can, however, be found by considering them to be assemblages of simple sources whose pressure at a point is given by

$$P(t)=\left(\frac{\rho c\kappa V_{p\,\max}}{4\pi r'}\right)e^{j(\omega t-\kappa r')}, \quad (8)$$

where r' is the distance from a point A to the source, ρ is the density of air, c is the speed of sound, $\kappa=\omega/c$ is the wave number of a sound wave, and $V_{p\,\max}$ is the maximum amplitude of the analog drive velocity $V_p(t)$.

An elementary area of the surface of the panel, $dx\,dy$, can be considered to be a simple point source radiating into the infinite half-space on the upper part of the baffle. This amounts to twice the effect of the same source radiating into free space. Then

$$dP=\left(\frac{\rho c\kappa V_{p\,\max}}{2\pi r'}\right)e^{j(\omega t-\kappa r')}dx\,dy, \quad (9)$$

where r' is now the distance from point A to the $dx\,dy$ element. The total pressure at A due to the vibration of the entire piston is therefore found by integrating the above expression over the surface of the piston. Now

$$OA'=x\cos\alpha+y\cos\beta$$

$$r'=r-OA'=r-(x\cos\alpha+y\cos\beta)$$

and at a great distance from the piston, $r=r'$, so we have

$$P(t)=\frac{j\rho c\kappa V_{p\,\max}}{2\pi r}e^{j(\omega t-\kappa r)}\times\int_{-b}^{b}dy\int_{-a}^{a}e^{j\kappa(x\cos\alpha+y\cos\beta)}\,dx \quad (10)$$

from which

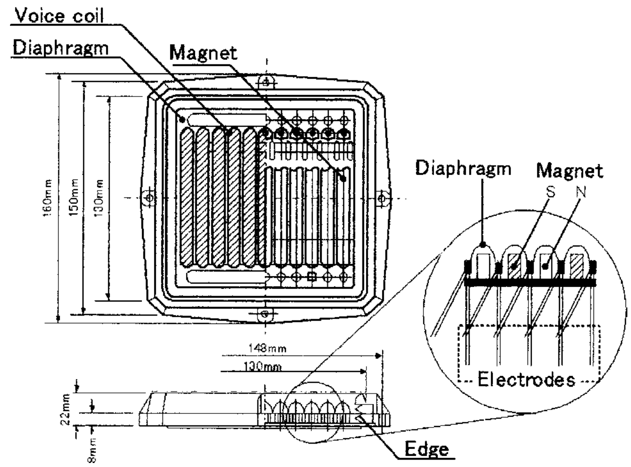

FIG. 2. Construction of our digital loudspeaker.

$$P(t)=\frac{4j\rho c\kappa abV_{p\,\max}}{2\pi r}\cdot\frac{\sin(\kappa a\cos\alpha)}{\kappa a\cos\alpha}\cdot\frac{\sin(\kappa b\cos\beta)}{\kappa b\cos\beta}e^{j(\omega t-\kappa r)}. \quad (11)$$

The radiation in the yz plane can be determined by putting $\cos\beta=\sin\gamma$, and $\sin(\kappa a\cos\alpha)/\kappa a\cos\alpha=1$. As α approaches 90°,

$$P(t)=\frac{2j\rho c\kappa abV_{p\,\max}}{\pi r}\cdot\frac{\sin(\kappa b\sin\gamma)}{\kappa b\sin\gamma}e^{j(\omega t-\kappa r)}, \quad (12)$$

where $\sin(\kappa b\sin\gamma)/\kappa b\sin\gamma$ is known as the directivity function which determines the directional characteristics of the radiation of the source. It is clear that the directional pattern becomes more pronounced at high frequencies. In other words, the greater the line dimensions of the radiator, the more pronounced the directivity will be. At the same time, minor lobes develop in addition to major lobes as the dimensions of the piston are increased.

B. Design and construction of the loudspeaker

Figure 2 shows the construction of our electrodynamic planar digital loudspeaker. The actual active part of the diaphragm is 13 cm wide, 13 cm high, and 1.1 cm deep. The diaphragm is molded plastic (total mass $m_p=0.0344$ kg, additional mass $m_{ad}=0.000\,88$ kg) and has streamlined sections to suppress the divided vibration. Two voice coils are rolled together in one section of the diaphragm, resulting in a total of 22 voice coils, each of which has a DC impedance of 4 Ω, 22 turns, and $l=4.142$ m. Also, 11 permanent magnets (each magnet is 7.5 mm wide, 87.25 mm long, and 6.0 mm thick, and each gap is 2.0 mm) are arranged under the diaphragm so that the poles of adjacent magnets are opposite each other. The suspension between the diaphragm and the frame is made of a piece of handmade Japanese paper (0.09 mm thick). The lowest resonant frequency f_0 was measured and found to be 22.7 Hz. Therefore, the stiffness s_p of the suspension is 736 N/m.

The free moving diaphragm generates sound in both the front and back of the element. The normal use of the element is to place it in an enclosure. In our test of the digital loud-

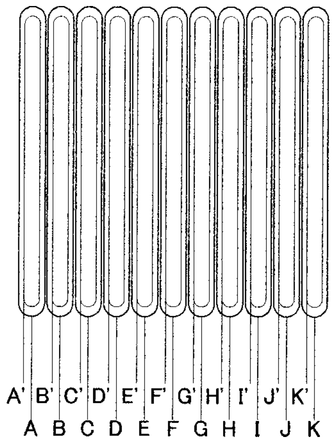

FIG. 3. The 22 voice coils in our electrodynamic loudspeaker.

speaker we used an enclosure ($V_c = 0.041\,94\ \mathrm{m}^3$) which was 28.5 cm wide, 54.5 cm high, and 27.0 cm deep. Therefore, the equivalent stiffness $s_c = \rho c^2 (ab)^2 / V_c$ is 549 N/m (20 °C).

Our planar type loudspeaker with 22 exciters has many acoustically interesting qualities, but the following technical problems must be solved:

(a) The radiating area vibrates as one unit only at lower frequencies. At higher frequencies, the radiating area exhibits more or less strong partial vibrations. Therefore, how can a flat frequency response be achieved in order to satisfy the piston motion of Eq. (6)?
(b) How can the interference be reduced within the motional electrical impedance $(Bl)^2/Z_r + Z_p$ of Eq. (4) due to motion and each exciter?
(c) In the high frequency limit, is a mechanical low-pass filter [Eq. (7)] formed as a result of the stiffness of the joint between the diaphragm and the 22 voice coils?
(d) What is the relationship between the timing precision of $D-A$ conversion and the directional responses of our electrodynamic planar digital loudspeaker?

III. ACOUSTIC RESPONSES WHEN DRIVEN BY AN ANALOG SIGNAL

The acoustic response and vibration mode are affected by the points where the diaphragm is driven by the voice coils. Therefore, the acoustic responses when driven by an analog signal are studied and improvements are discussed.

A. Measurement method

Figure 3 shows the voice coils in our electrodynamic planar loudspeaker. Each letter denotes a voice coil. Coils denoted by the same letter, such as A and A', mean that these two voice coils are rolled together in one section of the diaphragm. Experiments were carried out in an anechoic room (capacity: 60 m^3).

The fundamental test signal in the time domain is the impulse response, i.e., the output of the component when presented with a narrow rectangular voltage pulse. The impulse response carries within it a complete characterization of a component's linear performance. However, there are practical difficulties with using pulses to test loudspeakers.

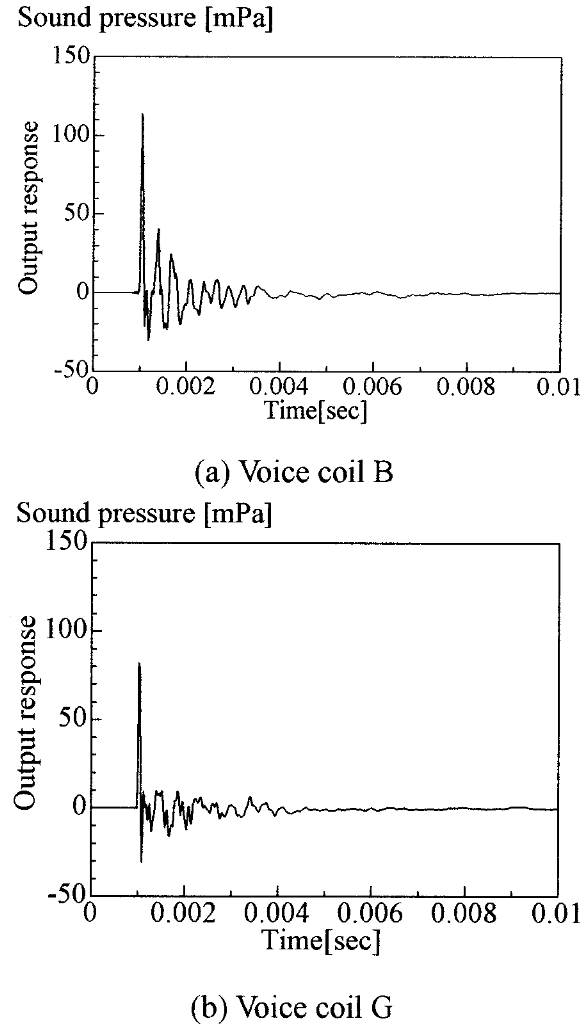

FIG. 4. Impulse responses of two voice coils (B and G).

The very high dynamic range leads to a compromised signal/noise ratio, so we used a method based on maximum-length sequence test signals (sampling frequency: 48 kHz, 16th order, and average response number: 10), realized in a commercial piece of test equipment: the CTX1010 and DOCKING STATION measuring system from CORETEX Corp.

The CTX1010 and DOCKING STATION feed each voice coil a pseudorandom sequence of rectangular voltage pulses with an electrical input power of 1 W. By performing a cross correlation between the test signal and the signal picked up by a microphone with a sound level meter (RION: NL-14) at a distance of 1 m from the diaphragm, the host notebook computer (TAIWAN, R.O.C.: 7600 Series) can calculate the system's impulse response.

The sensitivity of a speaker is defined as the free-field sound pressure level (SPL) produced by a 1 W electrical input signal, measured at an on-axis distance of 1 m. To calibrate the SPL for the y axis of the frequency response, a pure tone with a frequency of 1 kHz was used.

B. Results

First, as an example, the relative impulse responses of coils B and G are shown in Figs. 4(a) and (b), respectively. The sharp up-spike and down-spike of each impulse re-

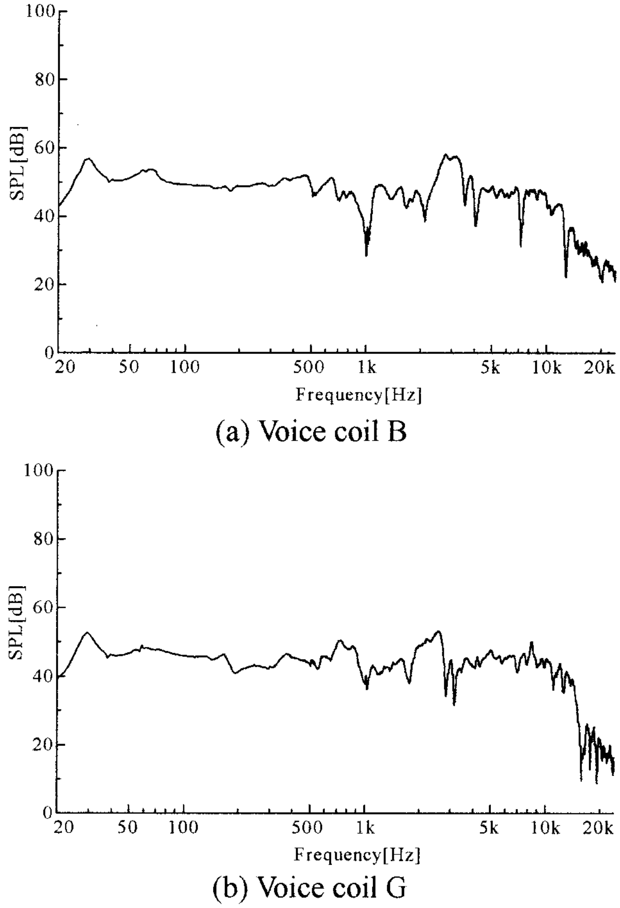

FIG. 5. Frequency responses of two voice coils (B and G). The vertical axis is calibrated to the free-field sound pressure level produced by a 1 W electrical input signal, measured at an on-axis distance of 1 m.

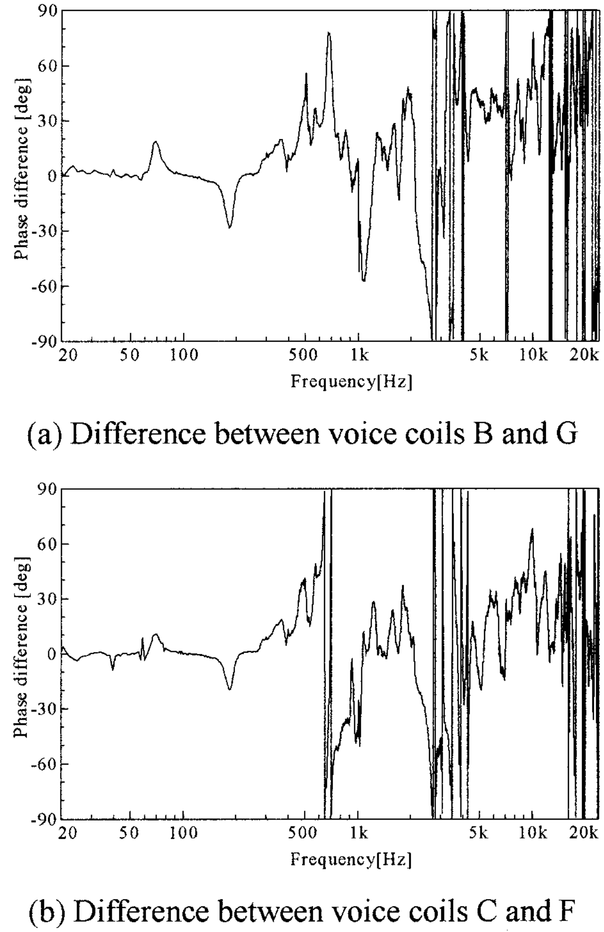

FIG. 6. Phase difference characteristics.

sponse is the electrodynamic planar loudspeaker's output, which is followed by lower-frequency information, a few ripples (about 0.33 ms) and reflections of the sound wave from the speaker's baffle and cabinet. The results show not only that the amplitudes are different but also that the transient responses decay slowly. Therefore, these responses may affect the cause of intersymbol interference of pulses when the loudspeaker is driven by a digital signal.

Second, the frequency responses of coils B and G are shown in Figs. 5(a) and (b), respectively. At 30 Hz a small peak appears due to the closed enclosure and the lowest resonance frequency (f_0). From 30 Hz to about 500 Hz, by acting as a piston, the diaphragm moves as a whole. Such a loudspeaker is mass-controlled over most of its pass-band. However, when it comes to higher frequencies, the radiating area exhibits more or less strong partial vibrations. Especially, at about 3 kHz, a peak is found in Fig. 5(a) corresponding to voice coil B, while a valley is found in (b) corresponding to voice coil G. A significant gap (maximum 15 dB re: 20 μPa at 1 m W) can be seen between the response levels. Similar phenomena are reported in the case of a panel loudspeaker.[6]

Last, Fig. 6(a) shows the phase difference between coils B and G. The phase does not deviate more than $\pm 35°$ in the region where the diaphragm acts as a piston, but there are large phase differences, especially at frequencies larger than 500 Hz. The same result (voice coils C and F) can be found in Fig. 6(b). Consequently, timing jitter and the distortion may result.

The measured sensitivity of our speaker (voice coil F only) is 56.6 dB at 1 kHz and 58.0 dB when a random noise signal of 2 V rms (band-limited to 16 kHz) is input. Because of the achievement of a flat frequency response from 40 Hz to 10 kHz, this result indicates our electrodynamic planar loudspeaker has a sensitivity problem in comparison with a conventional voice-coil cone speaker.

Similar acoustic responses are found in other voice coils, while coils rolled in the same section show almost the same response. The cause of these phenomena is most likely the divided vibration in the resonance or antiresonance frequencies, which vary with position.

C. Discussion

As mentioned in the preceding section, electrodynamic planar loudspeakers have some problems that may cause large distortion when driven by a digital signal because each voice coil's response is not the same. Therefore, in order to accomplish the $D-A$ conversion at the loudspeaker, it is necessary to flatten each response. One solution is to connect two voice coils in series if one coil shows a peak in its frequency response, while the other shows a dip.

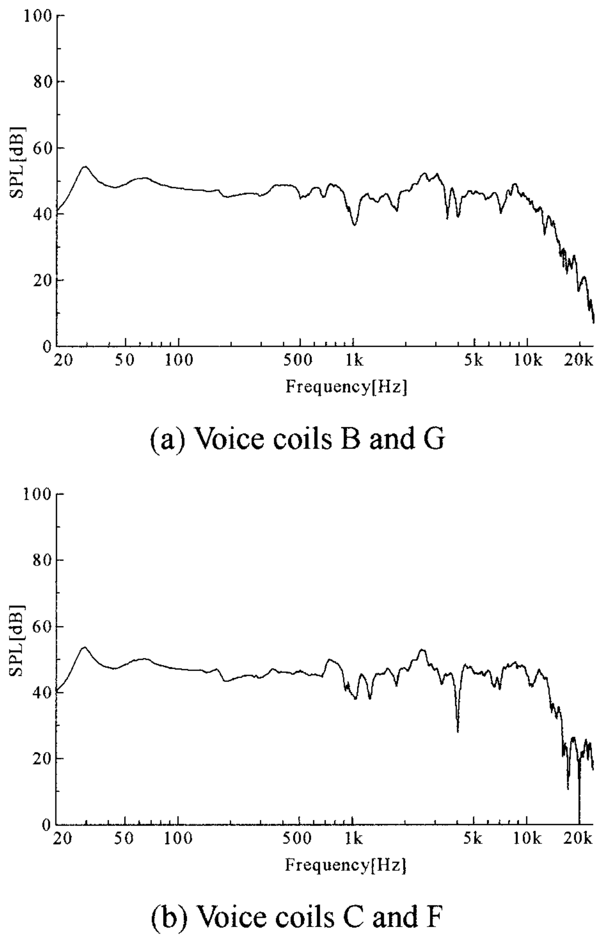

FIG. 7. Frequency response of electrodes connected in series.

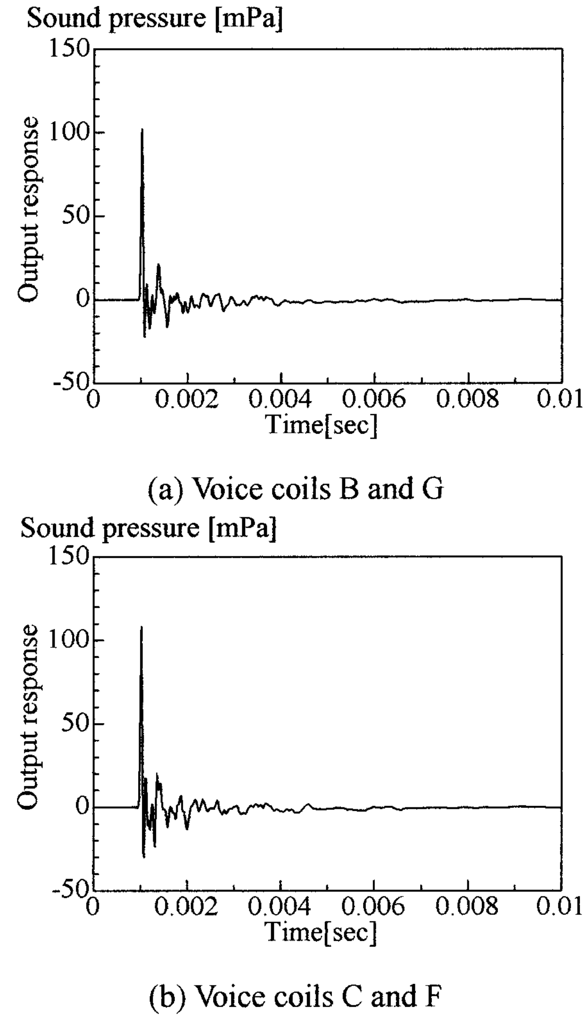

FIG. 8. Impulse response of each pair of voice coils.

1. Acoustic response improvement

First, Fig. 7(a) shows the frequency response when electrodes B and G are driven simultaneously. Comparing Fig. 7(a) with Figs. 5(a) and (b), we find that the dip in the frequency response at around 3 kHz is reduced by 10 dB. The same result (in the case of the electrodes C and F) can be seen in Fig. 7(b). The results show that the loudspeaker acts as a low-pass filter with a cutoff frequency of 10 kHz and a slope of about -18 dB/octave.

Second, Fig. 8(a) shows the relative impulse response of electrodes B and G connected in series. Compared with Figs. 4(a) and (b), an improvement in transient response can be seen. The same result (in the case of electrodes C and F) can be seen in Fig. 8(b). This result is due to the enlarged effective area of vibration.

Finally, Fig. 9(a) shows the phase difference response between electrodes $B+G$ and $C+F$. The phase only deviates from $+55°$ to $-20°$ in the operational band. The same result (in the case of electrodes $E+J$ and $B+G$) can be seen in Fig. 9(b). The results show that the phase difference is diminished and a little improvement in flatness can be found at frequencies less than 10 kHz when compared with Figs. 6(a) and (b).

2. Combination of voice coils

From the above results, we found that the acoustic responses can be improved by an appropriate combination of voice coils. When a 16-bit digital signal is used with an SMB code, 1 bit is used to select the direction of movement of the diaphragm and the other 15 bits are used to select the amount of diaphragm movement. Since two voice coils are required for each bit, a total of $15\times2=30$ coils are necessary. However, it is not possible at present to make 30 coils, so the combination and arrangement of the 22 voice coils in our loudspeaker are decided as follows:

(a) Two voice coils are used with each of the upper 7 bits. For each of the lower 8 bits only one coil is used.
(b) The coil positions for the upper 7 bits are chosen to make the frequency response as flat as possible.

Figure 10(b) shows the relationship between the bit numbers and voice coils that was decided according to the above methods. Coils connected in series are denoted by the same circled number. As bit numbers are not assigned symmetrically with respect to the center of the diaphragm, we call this arrangement of voice coils the ''asymmetric arrangement.'' On the other hand, the arrangement in Fig. 10(a) is called the ''symmetric arrangement'' because bit numbers are assigned symmetrically with respect to the center of the diaphragm. However, this arrangement does not follow the above-mentioned methods. Next, we discuss the difference in acoustic responses between these two arrangements.

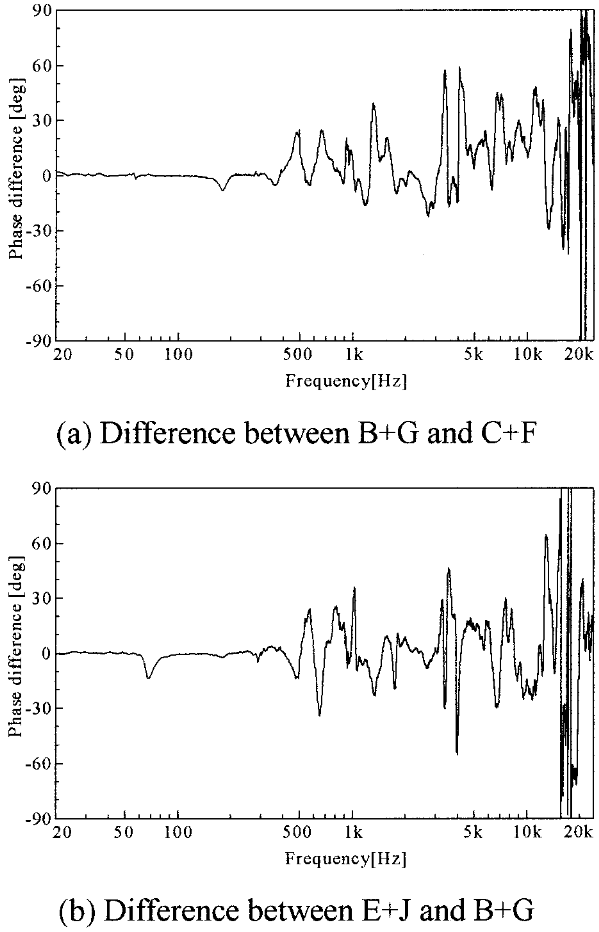

FIG. 9. Phase difference characteristics.

IV. RESPONSES TO A DIGITAL SIGNAL

A. PCM driving circuit

Figure 11 shows the block diagram of the PCM driving circuit for our digital loudspeaker. The computer (NEC: LW500J/2) generates the digital audio signal, which is sent to the audio data interface (NITTOBO: AD216). Next, the digital signal is converted from a serial code into a parallel one, and then changed from Offset Binary (OB) code into SMB code in order to drive the loudspeaker. Therefore, the

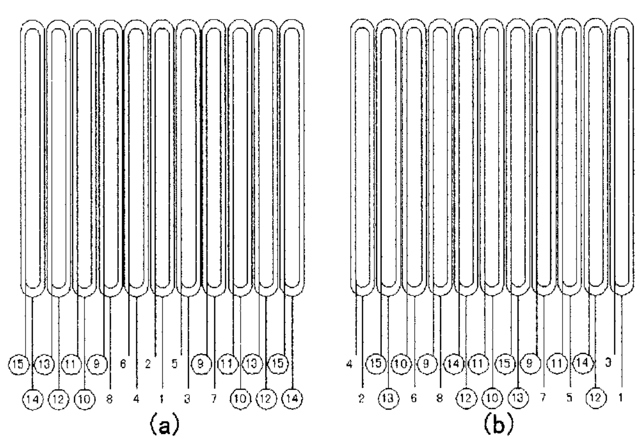

FIG. 10. The arrangement of the electrodes. (a) Symmetric arrangement. (b) Asymmetric arrangement. Circled numbers that are the same indicate a series connection.

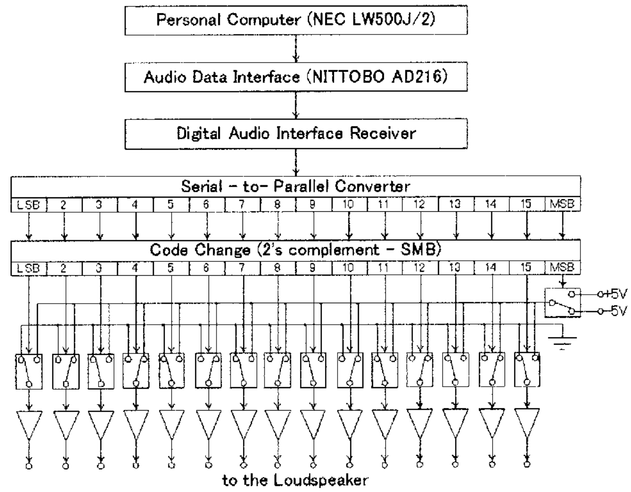

FIG. 11. Block diagram of the PCM driving circuit.

digital audio signal is changed from two levels into three, which enables the loudspeaker to vibrate in both backward and forward directions. Figure 12 shows the output waveforms of the 15th, 14th, and 13th bits of the PCM driving circuit. Table I shows the relationship between the bit number and the peak-to-peak voltage of each bit waveform.

B. Measurement method

The experiments were carried out in an anechoic room.

First, the input signal to the loudspeaker was generated by a computer (NEC: LW500J/2), which was conveyed to the PCM driving circuit. The digital audio signal had a sampling frequency of 48 kHz, and was quantized to 16 bits. Pure tones from 20 Hz to 20 kHz were used every 1/3 octave. The peak level of the input signal was $2^{15}-1=37\,767$ which corresponds to 0 dB. This level was decreased in 6 dB steps and the acoustic responses were measured at each step. Moreover, a rectangular tone-burst covering a fairly wide frequency band was used to evaluate the dynamic behavior of the digital speaker. The transient behavior of the loudspeaker is indicated by a change in the envelope of the burst signal. The on-axis response was measured at 10 cm from the front of the diaphragm.

Second, it is difficult to measure directly the mode pattern on the uneven diaphragm. To help identify where the

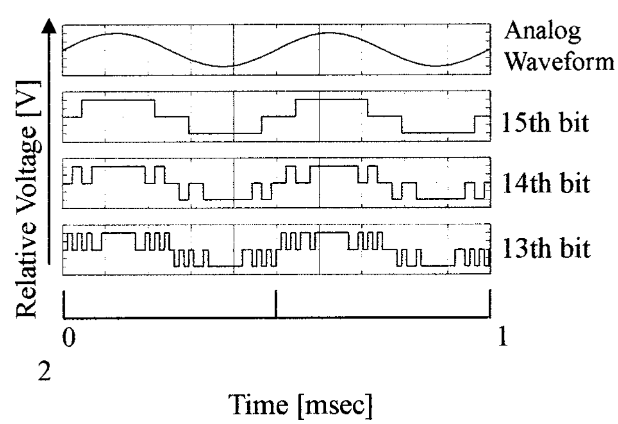

FIG. 12. Output waveforms of our PCM driving circuit.

TABLE I. Relationship between the bit number and peak-to-peak voltage.

BIT	Voltage (V_{p-p})	BIT	Voltage (V_{p-p})
1	0.001	9	0.25
2	0.002	10	0.5
3	0.004	11	1.0
4	0.008	12	2.0
5	0.015	13	4.0
6	0.031	14	8.0
7	0.062	15	16.0
8	0.125		

radiating area exhibits more or less strong partial vibrations, sound intensity distributions were measured on grids (at intervals of 13 mm) at a distance of 10 cm from the front of the diaphragm. The sound intensity analyzer used here is made by RION Co. Ltd. (sound intensity probe: SI-21, FFT analyzer: SA-74).

Third, using a microphone (RION NL-14) positioned at a distance of 1 m and at angles from 0° to 360° in 5° increments, the directional patterns of the loudspeaker were measured.

C. Results

The performance was measured using the following indices.

1. Output waveforms

Example output waveforms (pure tone, 15 bit: 0 dB, and 13 bit: −12 dB) from our electrodynamic planar digital loudspeaker with (a) the symmetric arrangement and (b) the asymmetric arrangement at 100 Hz are shown in Fig. 13. The following results were obtained.

(a) Our digital loudspeaker can reproduce pure sounds.
(b) There is a little interference between the motional and electrical components [Eq. (4)] due to the motion of the diaphragm and the exciters which drive the diaphragm.
(c) The timing precision of $D-A$ conversion on the diaphragm is good.

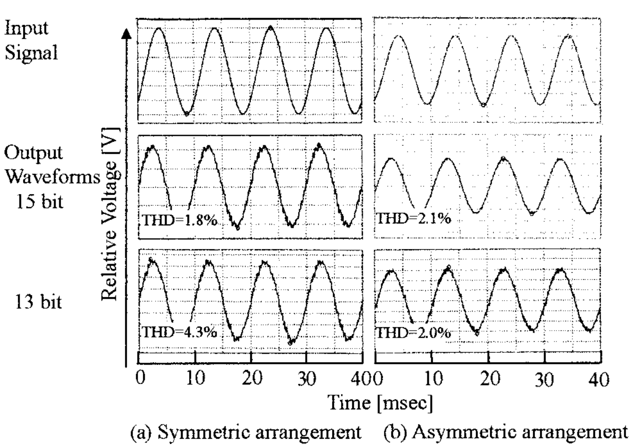

FIG. 13. Output waveforms from the electrodynamic planar digital loudspeaker at 100 Hz.

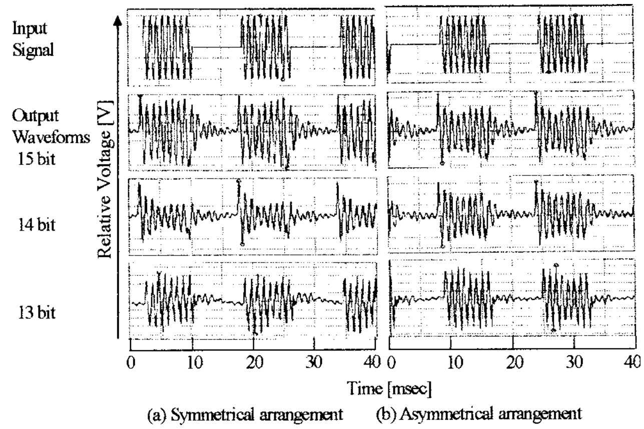

FIG. 14. Output waveforms driven by a tone burst signal at 1 kHz.

Figure 14 shows the output waveforms from the digital loudspeaker using (a) the symmetric arrangement and (b) the asymmetric arrangement driven by a tone burst signal at 1 kHz. From this figure, there is evidence of a switching transient, and seems to be about as good as is possible. Overhang (ringing after the signal is gone) is present to a small degree.

2. Frequency response

Figure 15 shows the frequency responses to 16-bit digital signals with various peak levels. We found that the asym-

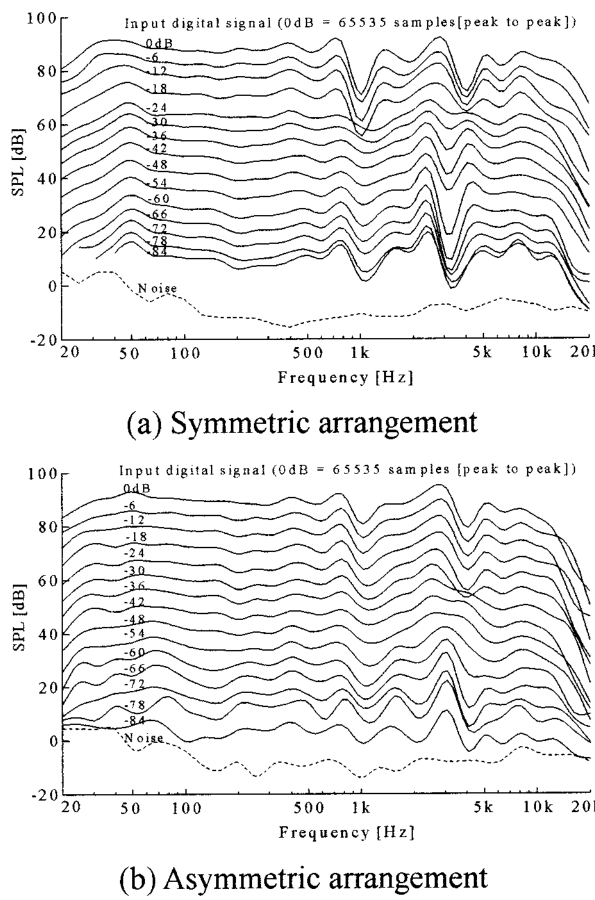

FIG. 15. Frequency responses of the digital loudspeaker at a distance of 10 cm from the center of the diaphragm. The dashed lines show the ambient noise level.

TABLE II. Relationship between SPL responses and input level from 0 dB to -84 dB.

	Symmetric arrangement			Asymmetric arrangement		
	Regression coefficient		Coefficient of determination	Regression coefficient		Coefficient of determination
Frequency	Slope	Intercept		Slope	Intercept	
50 Hz	0.912	92.6	0.9976	1.034	92.7	0.9995
100 Hz	0.928	88.0	0.9985	1.034	91.0	0.9986
1 kHz	0.840	74.9	0.9798	0.968	82.1	0.9958
4 kHz	0.871	80.7	0.9718	0.987	83.0	0.9596
10 kHz	0.892	83.0	0.9974	0.997	84.5	0.9984

metric arrangement (b) can reduce the peak and dip at around 1 kHz and 3 kHz, and improve the flatness of the responses compared to the symmetric arrangement (a). Here the sound reproduction band is from 40 Hz to 10 kHz.

3. Linearity

Table II shows the linearity characteristics at 50 Hz, 100 Hz, 1 kHz, 4 kHz, and 10 kHz. For the asymmetric arrangement, the slopes from 50 Hz to 10 kHz are improved to be about 1.0. At the same time, because the linearity range is 84 dB, it can be said that 16-bit $D-A$ conversion is performed on the diaphragm.

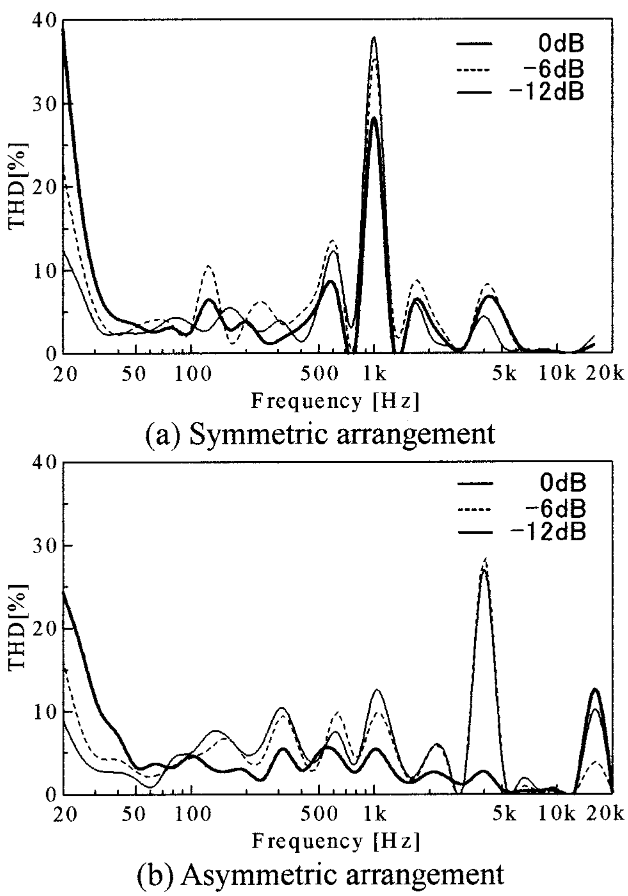

FIG. 16. Total harmonic distortion (THD) vs frequency when the input signal was changed from 0 dB to -12 dB in 6 dB steps.

4. Total harmonic distortion

THD stands for ''total harmonic distortion.'' The word ''total'' refers to the fact that the number shown represents the total of all harmonics. This total is a geometric total, formed by taking the square root of the sum of the squares of the amplitude of each of the harmonics. The THD of the digital loudspeaker using the asymmetric arrangement is about 2% at 100 Hz as shown in Fig. 13.

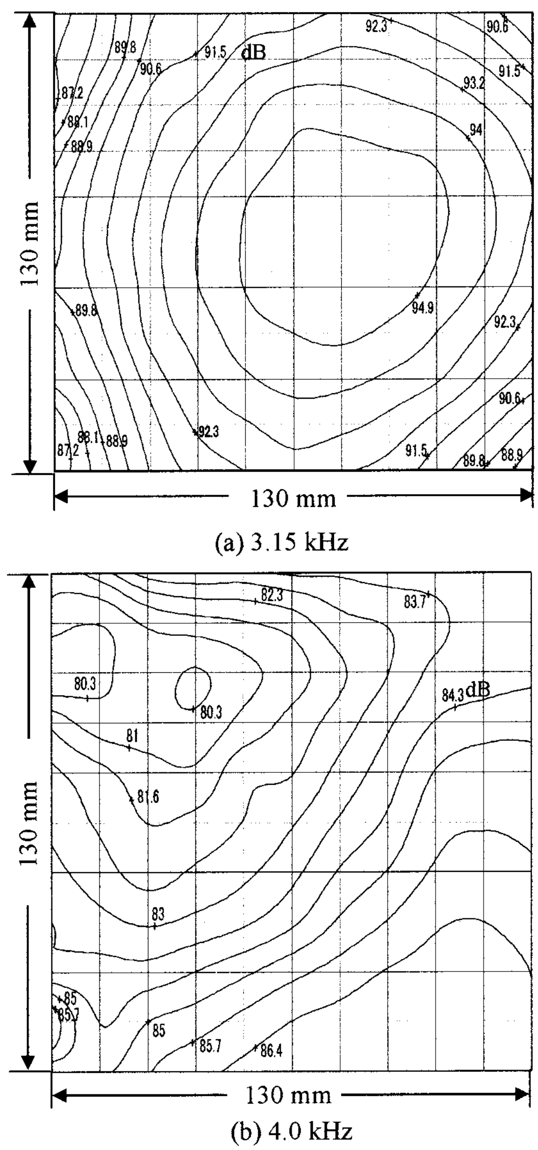

FIG. 17. Distribution of the sound intensity level (dB re: 10^{-12} W/m^2) at a distance of 10 cm from the front of the diaphragm.

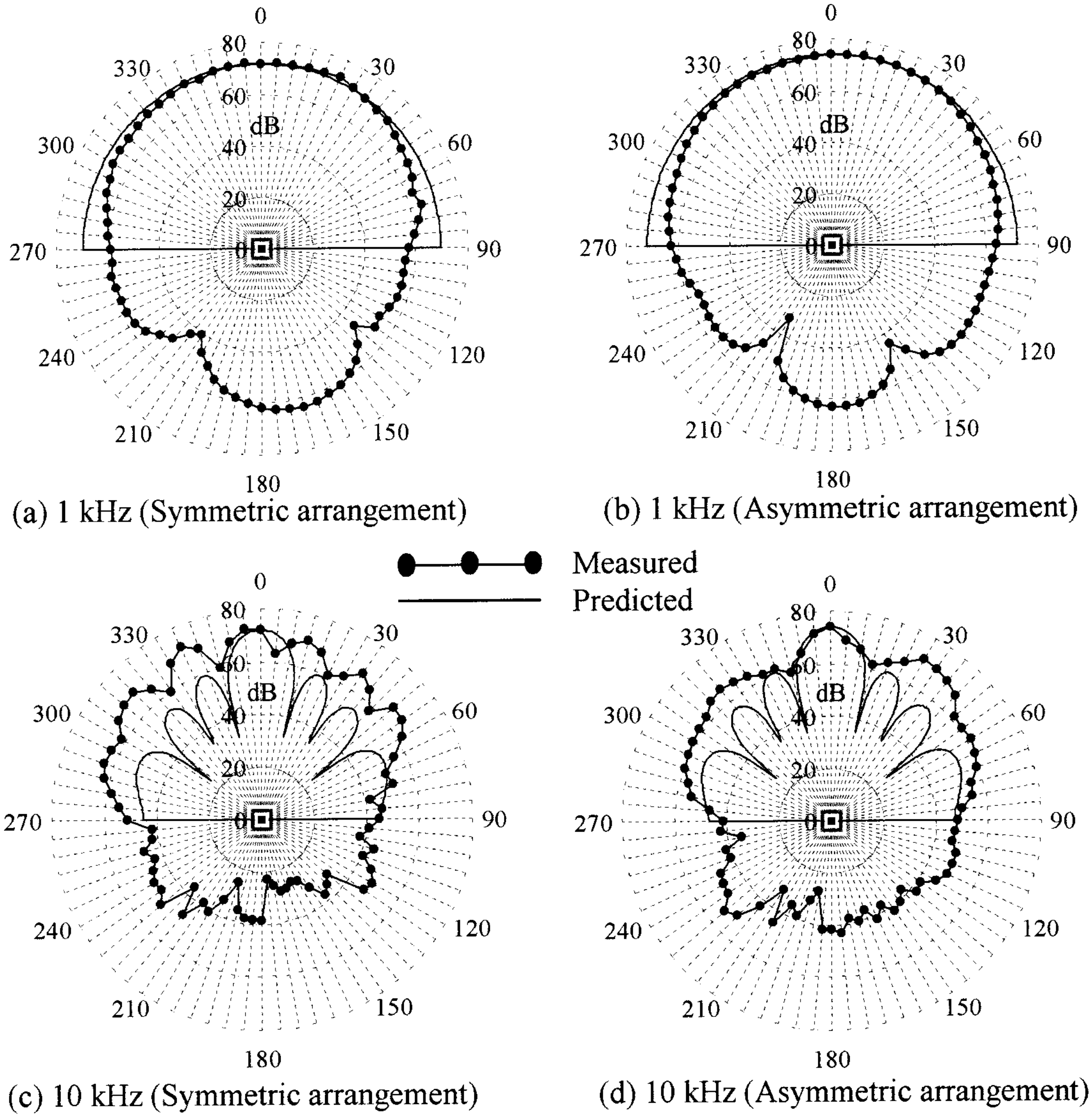

FIG. 18. Directional patterns of the loudspeaker at 1 kHz and 10 kHz. The radial scales are in SPL (dB) with the outermost circle corresponding to 80 dB *re*: 20 μPa.

Figure 16 shows the distortion factors when the input signal is changed from 0 dB to -12 dB in 6 dB steps. The reason why the THD is inferior at lower frequencies is that the voice coils separate from the magnetic circuit. The asymmetric arrangement results in an improvement in the characteristics around 1 kHz, which is relatively important for the sense of hearing. However, a new distortion at 4 kHz occurs in the asymmetric arrangement. These changes in the distortion are thought to be caused by mode transitions which depend on the arrangement of the voice coils.

5. Distribution of sound intensity level

As is evident from the contour maps of sound intensity level shown in Fig. 17, the diaphragm movement changes from (a) piston motion at 3.15 kHz to (b) divided vibration mode motion at 4 kHz. Especially, from Fig. 17(b), it can be seen that the sound intensity is highest, 86.4 dB, on the lower right surface, and lowest, 80.3 dB, on the upper left surface.

6. Directional pattern

Figure 18 shows the measured directional response (●) of the digital loudspeaker versus the theoretical data (—) based on the directivity function of Eq. (12) for a rectangular plane rigid mounted flush in an infinite plane baffle. Only theoretical data in half-space are shown because the diaphragm is embedded in the baffle. As can be seen from Fig. 18, it is clear that the directional pattern becomes more pronounced at high frequencies. It can be seen that both the measured and theoretical patterns have about six minor lobes in addition to a major lobe in half-space at 10 kHz. Because the diaphragm molded in plastic has streamlined sections as shown in Fig. 2, the measured result indicates that the asymmetric arrangement can more effectively suppress the appearance of the side lobe at 10 kHz than the predicted result based on Eq. (12).

Consequently, it can be said that the baffled-plated idealization as mentioned in the operating principles in Sec. II A is useful for our digital loudspeaker.

7. Auditory impressions

We evaluated auditory impressions with a short listening experiment. When many subjects listened to the music produced by our digital loudspeaker in a normal room (68 m^3, reverberation time of 0.8 s at 500 Hz), their auditory impressions were good for the perception of loudness, naturalness, and clearness. However, there was an impression of a little distortion peculiar to the digital speaker.

V. CONCLUSIONS

In this paper, an electrodynamic planar loudspeaker driven by a digital signal was experimentally discussed. The diaphragm was molded in plastic to have streamlined sections in order to suppress divided vibrations. Two voice coils were rolled together in one section of the diaphragm, resulting in a total of 22 voice coils. Also, 11 permanent magnets were arranged under the diaphragm so that the poles of adjacent magnets were opposite each other. The suspension between the diaphragm and the frame was made of a piece of handmade Japanese paper. The acoustic responses were affected by the arrangement of the voice coils on the diaphragm.

Therefore, first, the combination and arrangement of the voice coils were studied in order to improve the acoustic responses and achieve a 16-bit digital loudspeaker. The results are as follows:

(1) Our hand-picked pairs (asymmetric arrangement) are better than arbitrarily selecting them (symmetric arrangement).
(2) Hand-picking can improve the frequency response and suppress distortion.

Second, in order to compare the asymmetric and symmetric arrangements, experiments were performed where the digital loudspeaker was driven by a weighted discrete voltage with a maximum amplitude of 16 V_{p-p} and a resolution of 16 bits. The results are as follows:

(3) Our digital loudspeaker can reproduce pure sound. The sound reproduction band is from 40 Hz to 10 kHz.
(4) A different asymmetrical arrangement can improve the acoustic responses. Specifically, the peaks and dips in the frequency response around 1 kHz and 4 kHz are diminished and the distortion factor at 1 kHz is reduced from 28% to 2.3%.
(5) The loudspeaker has a linearity range of 84 dB in the above sound reproduction band.
(6) The baffled-plated idealization is useful to determine directional responses for the electrodynamic planar loudspeaker driven by a digital signal.

To summarize, we have constructed and demonstrated a 16-bit digital loudspeaker with good performance, wide bandwidth, and a linear dynamic range of 84 dB.

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