Binaural Directivity Pattern Simulation of the KEMAR Head Model with Two Twin Hearing Aid Microphones by Boundary Element Method

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Abstract

Two twin microphones may produce particular patterns of binaural directivity by time delays between twin microphones. The boundary element method (BEM) was used for the simulation of the sound pressure field around the head model in order to quantify the acoustic head effect. The sound pressure onto the microphone was calculated by the BEM to an incident sound pressure. Then a planar directivity pattern was formed by four sound pressure signals from four microphones. The optimal binaural directivity pattern may be achieved by adjusting time delays at each frequency while maintaining the forward beam pattern is relatively bigger than the backward beam pattern.

Keywords: *Hearing Aid, Planar Binaural Directivity Pattern, Microphone Array, BEM, KEMAR Head Model*

I. Introduction

The hearing impairment becomes the thirdly most popular disease in modern society after high blood pressure and inflammation of the joints. The hearing aid (HA) is a medical device for compensating the hearing loss of hearing impaired persons. Because of improved sophisticated semiconductor IC manufacturing technology, the previously popular analog HAs are being replaced with digital HAs which have much more advantages in electrical signal processing than analog HAs. One of the advanced features of the digital HA chip is its parameter adjusting function[1]. These specific features make digital HAs to progress in their performances such as noise cancellation, feedback control, hearing loss fitting and directivity. One particular feature we consider in this paper is the directivity of the digital HA. Many HA wearers complain about speech perception difficulty particularly in noisy environments. Thus, improving the sound quality in the environmental noise becomes the hottest issue in the HA field. The common but effective solution for increasing S/N ratio is to operate HA with directivity feature[2]. Two microphones are used for making directional HA[3]. The directional HA may have the relatively increased sensitivity to the sound coming from a particular direction. This geometrical feature effectively improves the noise reduction in the presence of the environmental noise.

Audiologists recommend the binaural fitting of HAs. That is to fit each ITE (In-The-Ear) HA into the right and left ears simultaneously. The one-sided fitting is called as monaural fitting which is ineffective in directivity (monaural directivity). Each of the right and the left ears may be fixed by an ITE HA with a pair of microphones. If two microphones are built in an ITE HA, the binaural fitting of HAs means two twin microphones; total 4 microphones. These two twin microphones can produce particular patterns of directivity (binaural directivity) by time delays between twin microphones. Two microphones in an ITE HA can be synchronized, that is, both microphones are connected to a digital amplifier with two input channels. Those two channels of

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input signals may be summed with time delays between two signals. However the other two microphones in the left ear are not synchronized with those two microphones in the right ear. When microphone arrays are placed near to the head, the performance of arrays is modified because of the acoustic head effect. Then, how can binaural directivity be approached systematically? This paper shows the solution by numerical simulation, the boundary element method (BEM).

II. Boundary Element Method (BEM)

The BEM was used for the simulation of the sound pressure field around a head model. The BEM numerical solution for the directional HA is to calculate the sound pressure on the ear of the head model. The sound pressure onto the microphone fitted in the ear was calculated by the BEM to an incident sound pressure. Then a planar binaural directivity pattern was formed by four sound pressures from four microphones. The time delay between twin microphones was changed to produce the most optimal directivity pattern.

The boundary element solution of the sound pressure field simulation is based on the Helmholtz partial differential equation[4]. For sinusoidal steady-state problems, the Helmholtz equation, $\nabla^2\Psi + k^2\Psi = 0$ represents the wave mechanics. Ψ is the acoustic pressure with time variation, $e^{j\omega t}$, and k is the wave number. In order to solve the Helmholtz equation in an infinite air media, a solution to the equation must not only satisfy structural surface boundary condition (BC), $\partial\Psi/\partial n = \rho_f \omega^2 a_n$ but also the radiation condition at infinity, $\lim_{|r|\to\infty} \int_S (\partial\Psi/\partial r + jk\Psi)^2 dS = 0$. a_n is normal displacement and ρ_f is the air density.

The Helmholtz integral equation derived from Green's second theorem provides such a solution for radiating pressure waves;

$$\oint_S \left(\Psi(q)\frac{\partial G_k(p,q)}{\partial n_q} - G_k(p,q)\frac{\partial \Psi(q)}{\partial n_q} \right) dS_q = \beta(p)\,\Psi(p) \quad (1)$$

where $G_k(p,q) = e^{-jkr}/4\pi r$, $r = |p-q|$

p is any point in either the interior or the exterior and q is the surface point of integration. β (p) is the exterior solid angle at p.

The acoustic pressure for the i^{th} global node, $\Psi(p_i)$, is expressed in discrete form [5]: $(1 \le i \le ng)$

$$\beta(p_i)\Psi(p_i) = \oint_S \left(\Psi(q)\frac{\partial G_k(p_i,q)}{\partial n_q} - G_k(p_i,q)\frac{\partial \Psi(q)}{\partial n_q} \right) dS_q \quad (2a)$$

$$= \sum_{m=1}^{nt} \int_{S_m} \left(\Psi(q)\frac{\partial G(p_i,q)}{\partial n_q} - G(p_i,q)\frac{\partial \Psi(q)}{\partial n_q} \right) dS_q \quad q \in S_m \quad (2b)$$

$$= \sum_{m=1}^{nt} \int_{S_m} \left(\sum_{j=1}^{8} N_j(q)\Psi_{m,j}\frac{\partial G(p_i,q)}{\partial n_q} - G(p_i,q)\sum_{j=1}^{8} N_j(q)\frac{\partial \Psi_{m,j}}{\partial n_q} \right) dS_q \quad (2c)$$

$$= \sum_{m=1}^{nt}\sum_{j=1}^{8} \left(\int_{S_m} N_j(q)\frac{\partial G(p_i,q)}{\partial n_q} dS_q \right)\Psi_{m,j} - \rho_f\omega^2 \sum_{m=1}^{nt}\sum_{j=1}^{8} \left(\int_{S_m} N_j(q) G(p_i,q) n_q dS_q \right) a_{m,j} \quad (2d)$$

$$= \sum_{m=1}^{nt}\sum_{j=1}^{8} A^i{}_{m,j}\Psi_{m,j} - \rho_f\omega^2 \sum_{m=1}^{nt}\sum_{j=1}^{8} B^j{}_{m,j} a_{m,j} \quad (2e)$$

where ng is the total number of surface nodes and nt is the total number of surface elements and $a_{m,j}$ are three dimensional displacements.

Equation (2b) is derived from equation (2a) by discretizing integral surface. And equation (2c) is derived from equation (2b) since an acoustic pressure on an integral surface is interpolated from adjacent 8 quadratic nodal acoustic pressures corresponding the integral surface. Then equation (2d) is derived from equation (2c) by swapping integral notations with summing notations. Finally the parentheses of equation (2d) is expressed by upper capital notations for simplicity.

When equation (2c) is globally assembled, the discrete Helmholtz equation can be represented as

$$([A] - \beta[I])\{\Psi\} = +\rho_f\omega^2[B]\{a\} \quad (3)$$

where [A] and [B] are square matrices of (ng by ng) size.

When the impedance matrices of equation (3), [A] and [B], are computed, two types of singularity arise[6]. One is that the Green's function of the equation, $G_k(p_i,q)$, becomes infinite as q approaches to p_i. This problem is solved by mapping such rectangular local coordinates into triangular local coordinates and again into polar local coordinates[7]. The other is that at certain wave number the matrices become ill-conditioned. These wave number are corresponding to eigenvalues of the interior Dirichlet problem[8]. One approach to overcome the matrix singularity is that [A] and [B] of equation (3) are modified to provide a unique solution for the entire frequency range[9-12]. The modified matrix equation referred to as the modified Helmholtz gradient formulation (HGF)[12] is obtained by adding a multiple of an extra integral equation to equation (3).

$$\left([A]-\beta[I]\oplus\alpha[C]\right)\{\Psi\}=+\rho_f\omega^2\left([B]\oplus\alpha[D]\right)\{a\} \tag{4}$$

where $\alpha=-\dfrac{\sqrt{-1}}{k\cdot(\text{Number of surface element adjacent a surface node})}$

[C] and [D] are rectangular matrices of (nt by ng) size. ⊕ symbol indicates that the rows of [C],[D] corresponding to surface elements adjacent a surface node are added to the row of [A],[B] corresponding to the surface node, that is,

$$\sum_{i=1}^{ng}\sum_{j=1}^{ng}A(i,j)=\sum_{i=1}^{ng}\sum_{j=1}^{ng}A(i,j)+\sum_{i=1}^{ng}\sum_{j=1}^{ng}\left(\sum_{m=1}^{s(i)}\alpha\ C(m,j)\right)$$
$$\sum_{i=1}^{ng}\sum_{j=1}^{ng}B(i,j)=\sum_{i=1}^{ng}\sum_{j=1}^{ng}B(i,j)+\sum_{i=1}^{ng}\sum_{j=1}^{ng}\left(\sum_{m=1}^{s(i)}\alpha\ D(m,j)\right) \tag{5}$$

where S(i) is the number of surface element adjacent a surface node. The derivation of the extra matrices [C], [D] are well described by Francis D.T.I. [12]. Equation (6) may be reduced in its formulation using superscript ⊕ for convenience;

$$A^{\oplus}\{\Psi\}=+\rho_f\omega^2B^{\oplus}\{a\} \tag{6}$$

where $\left([A]-\beta[I]\oplus\alpha[C]\right)\equiv A^{\oplus}$, $\left([B]\oplus\alpha[D]\right)\equiv B^{\oplus}$

The discrete BEM formulation of the Helmholtz surface integral equation with incident pressure term can be represented as a matrix formula

$$\{\Psi\}=+\rho_f\omega^2\left[A^{\oplus}\right]^{-1}\left[B^{\oplus}\right]\{a\}-\left[A^{\oplus}\right]^{-1}\{\Psi_{inc}\} \tag{7}$$

where Ψ_{inc} is the incident sound pressure. $\lfloor A^{\oplus}\rfloor$ and $\lfloor B^{\oplus}\rfloor$ are acoustic impedance matrices[5-7].

If we assume a rigid surface boundary condition because the present KEMAR head model has a rigid surface, Eq. (7) becomes

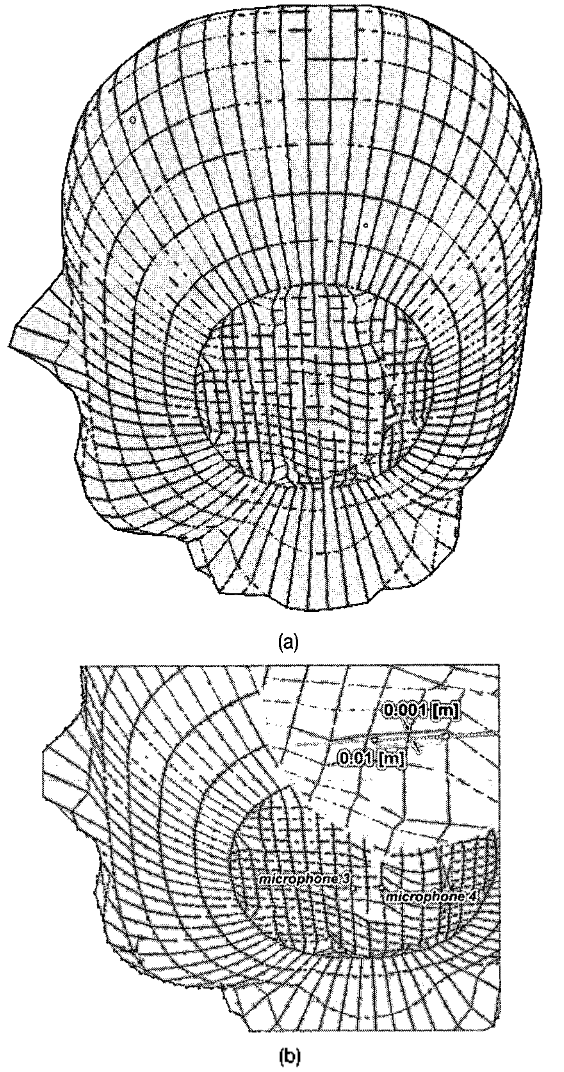
$$\{\Psi\}=-\left[A^{\oplus}\right]^{-1}\{\Psi_{inc}\} \tag{8}$$

Once $\{\Psi\}$ is known, the acoustic pressure in the near or far field is determined by $\beta(p)=1$ of Eq. (1) for given values of surface nodal pressure and zero surface nodal displacement;

$$\Psi(p_i)=\sum_{m=1}^{nt}\sum_{j=1}^{8}A^i{}_{m,j}\Psi_{m,j} \tag{9}$$

3. Boundary Element Mesh Generation

The KEMAR, a dummy head model, was three dimensionally scanned by a laser scanner[13]. The scanned points data of the three dimensional coordinates were then used to be meshed for



(a)

(b)

Fig. 1. (a) The surface element mesh of the KEMAR dummy head. (b) The magnified view of twin microphones on the left ear.

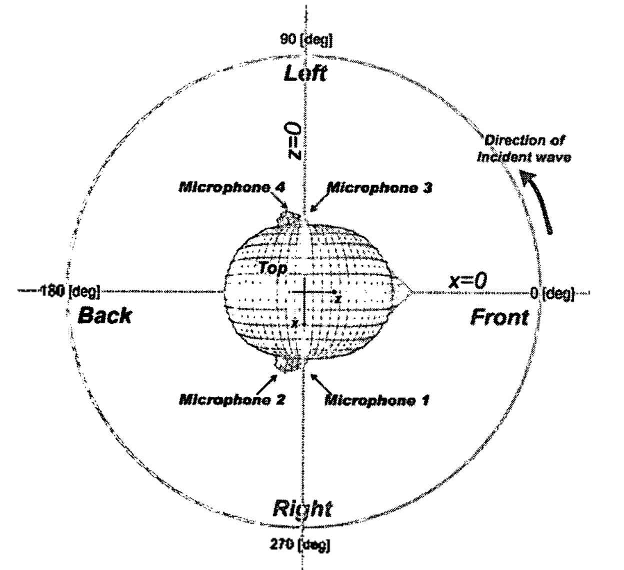


Fig. 2. Polar coordinates for directivity pattern calculation. Mic. 1and Mic.3 are reference microphones.

boundary surface element grids[14]. Each boundary element has 4 corner nodes and 4 mid-side nodes. It is a common practice to have the size of the largest surface element to be at least less than $\lambda/3$(λ=wavelength), so that the numerical approximation might be converged[15]. In this paper, the upper frequency of the acoustic radiation is less than 4 kHz, so that $\lambda/3$ is about 0.28 m. Therefore the present BEM meshing size is much small enough for the BEM calculation. Figure 1 shows the surface element meshes of the KEMAR model. Figure 1(b) shows the magnified view of twin microphones at the left ear. The angle of the directivity pattern starts from the front in anti-clockwise direction (Fig. 2).

VI. Time Delay Methods

Figure 3 shows time delay (Δt) circuits between the front and the rear microphones (Mic. 1 and Mic 2 or Mic. 3 and Mic. 4 respectively). Fig.3 (a) is a summed method in which a signal from the front reference microphone is summed with a time-delayed signal from the rear microphone. Fig.3 (b) is a Knowles method where 50% and 25% time delays mean a half and a quarter phase shifts respectively. The Knowles method was derived for producing the sharpest directivity pattern and its time delays are fixed at π and 0.5π phase shifts[16].

In the summed method, the strength of the resulted sound pressure from the two twin microphones is calculated by equation (10);

$$\left|A^r{}_r\cos\left(2\pi f\cdot(t-\Delta t)+\theta^r{}_r\right)-A^r{}_f\cos\left(2\pi f\cdot(t-0)+\theta^r{}_f\right)\right|+$$
$$\left|A^l{}_r\cos\left(2\pi f\cdot(t-\Delta t)+\theta^l{}_r\right)-A^l{}_f\cos\left(2\pi f\cdot(t-0)+\theta^l{}_f\right)\right| \quad (10)$$

where $A^r{}_f$ and $\theta^r{}_f$ are the amplitude and the phase of the right ear front microphones while $A^l{}_r$ and $\theta^l{}_r$ are those of the left ear rear microphones. In Equation (10), total pressure (incident pressure plus scattered pressure) amplitudes and phases are calculated by the BEM.

In the summed method, the time delay can be independently changed at different frequencies. We consider three ways of time delay variations; The first variation is to produce the similar directivity pattern as the Knowles method (Summed I) at each frequency. The second variation is to increase the receiving sensitivity of the microphone array output at each frequency while maintaining the forward beam pattern is relatively bigger than the backward beam pattern (Summed II). The third variation is simply done by keeping constant time delay (0.7π[Rad] at 1kHz = 0.35 ms. see later Fig. 9) at all frequencies (Summed III).

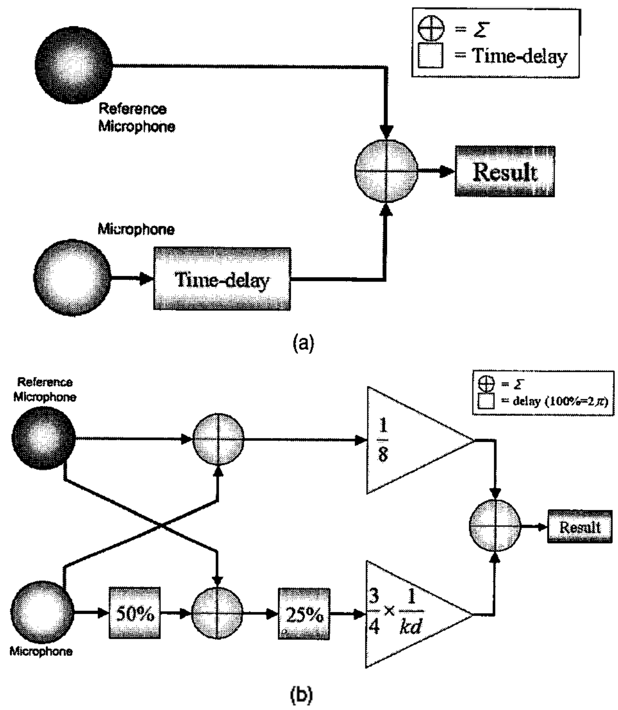


Fig. 3. Time delay (Δt) circuits between front and rear microphones (Mic. 1 and Mic. 2 or Mic. 3 and Mic. 4 respectively). (a) Summed method (b) Knowles method. Reference microphone (0 time delay) = front microphone. k = wave number. d = the distance between twin (the front and the rear) microphones (10mm separation).

In order to quantitatively compare the resulted directivity patterns from the Knowles method with the three summed methods at four frequencies of interest, the factor of the articulation index-directivity index (AI-DI) is used;

$$AJ\text{-}DI = 0.2 \cdot Dp(500) + 0.23 \cdot Dp(1000) + 0.33 \cdot Dp(2000) + 0.24 \cdot Dp(4000) \quad (11)$$

where Dp (f) is planar directivity index [17] and derived as

$$Dp(f) = 10 \times \log_{10}\left(22.92 \Big/ \sum_{j=0}^{355} \left(10^{\frac{SISDR_j(f)}{10}} \Big/ 10^{\frac{SISDR_0(f)}{10}} \right) \cdot \left| \sin\left(\frac{2\pi}{360}\theta_j \right) \right| \right) \quad (12)$$

j is in 5°interval. f is a frequency and Dp(f) is expressed in dB unit. SISDR(f) is a simulated in situ directional response in dB. The SISDR(f) is the difference between the sound pressure level at a given azimuth angle(θ) of sound incidence and the sound pressure level in the reference position(θ=0°), plotted as a function of azimuth angle.

V. Results

Figure 4 shows directivity patterns of the KEMAR head model in the Knowles method. The polar coordinates are scaled in dB. 50% and 25% time delays follow the Knowles method. AI-DI is 4.1

Figure 5 shows directivity patterns of the KEMAR head model in the Summed method I. The time delay is adjusted by the Summed method I. AI-DI is 4.4. Figure 6 shows directivity patterns of the KEMAR head model in the Summed method II. The time delay is adjusted by the Summed method I. AI-DI is 2.7. Figure 7 shows directivity patterns of the KEMAR head model in the Summed method III. The time delay is adjusted by the Summed method I. AI-DI is 2.2.

Table 1 compares directivity Indexes (DI) at four frequencies for each of four time delay methods and the last column indicates the AI-DI for each time delay method.

Figure 8 shows summed sound pressures of the KEMAR head model as a function of frequency for four different time delay methods . The figure shows the receiving sensitivity of the two twin microphone arrays. The sensitivity is below - 10 dB (ref. 10.88 [Pa]) at all frequencies for the Knowles method and the Summed method I. The Summed method II has smaller AI-DI

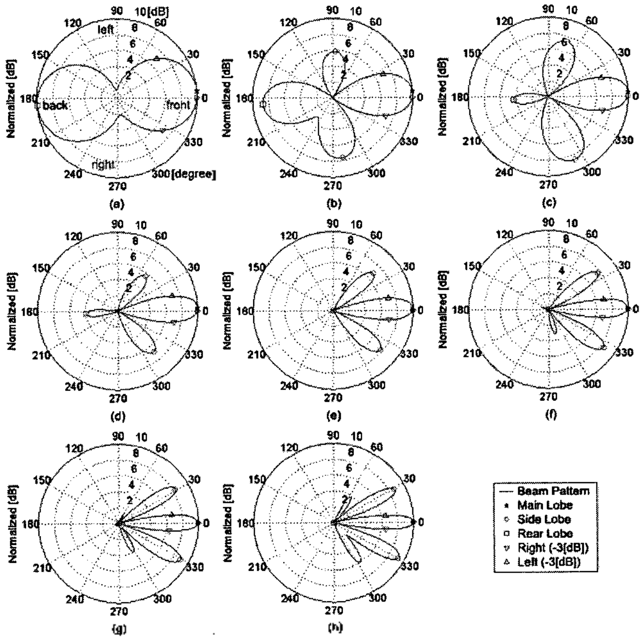


Fig. 4. Directivity patterns of the KEMAR head model. Microphone separation=10mm. 50% and 25% time delays follow the Knowles method. Input frequencies= (a) 500 Hz , (b) 1000 Hz , (c) 1500 Hz , (d) 2000 Hz , (e) 2500 Hz , (f) 3000 Hz , (g) 3500 Hz , (h) 4000 Hz. AI-DI = 4.1

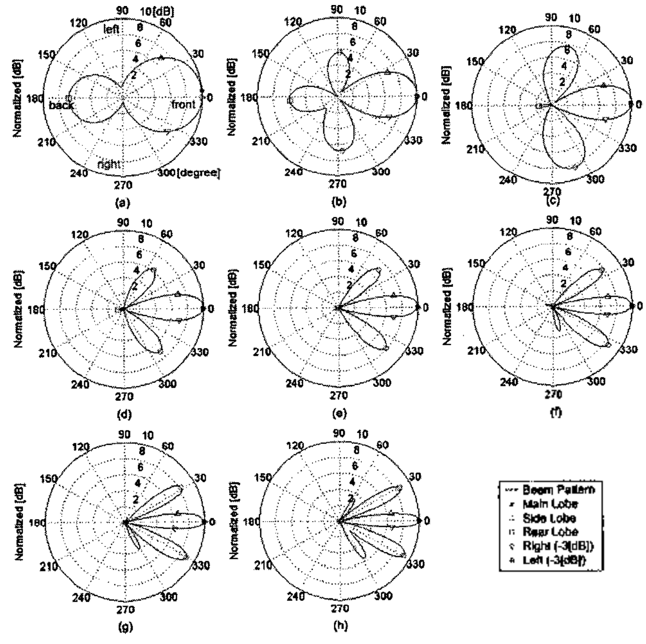


Fig. 5. Directivity patterns of the KEMAR head model. Microphone separation=10mm. Time delays are adjusted by the Summed I method. Input frequencies= (a) 500 Hz , (b) 1000 Hz , (c) 1500 Hz , (d) 2000 Hz , (e) 2500 Hz , (f) 3000 Hz , (g) 3500 Hz , (h) 4000 Hz. AI-DI = 4.4

than the Knowles method, but the receiving sensitivity is much better than that of the Knowles method. Figure 9 shows the time delay in the Summed II method in radian.

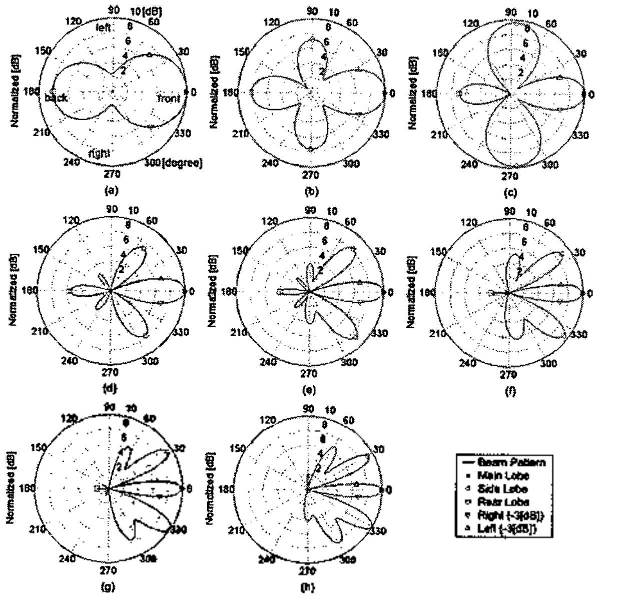


Fig. 6. Directivity patterns of the KEMAR head model. Microphone separation=10mm. Time delays are adjusted by the Summed II method. Input frequencies= (a) 500 Hz , (b) 1000 Hz , (c) 1500 Hz , (d) 2000 Hz , (e) 2500 Hz , (f) 3000 Hz , (g) 3500 Hz , (h) 4000 Hz. AI-DI = 2.7

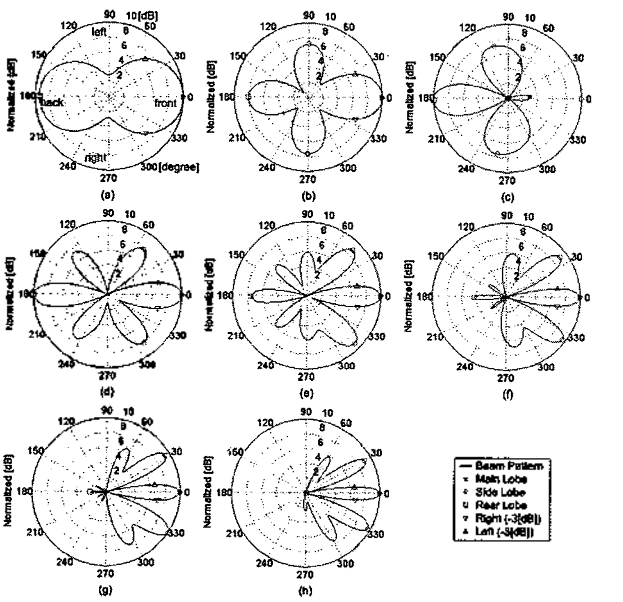


Fig. 7. Directivity patterns of the KEMAR head model. Microphone separation=10mm. The time delay is constant (0.35 ms) in the Summed III method at all frequency. Input frequencies= (a) 500 Hz , (b) 1000 Hz , (c) 1500 Hz , (d) 2000 Hz , (e) 2500 Hz , (f) 3000 Hz , (g) 3500 Hz , (h) 4000 Hz. AI-DI = 2.2

Table 1. Directivity Index at four frequencies for each of four time delay methods

Method	Frequency [Hz]	500	1000	2000	4000	AI-DI
Knowles	DI	1.3	1.9	5.6	6.3	4.1
Summed I	DI	2.3	2.5	5.6	6.5	4.4
Summed II	DI	1.6	1.4	3.8	3.4	2.7
Summed III	DI	1.0	1.4	1.8	4.6	2.2

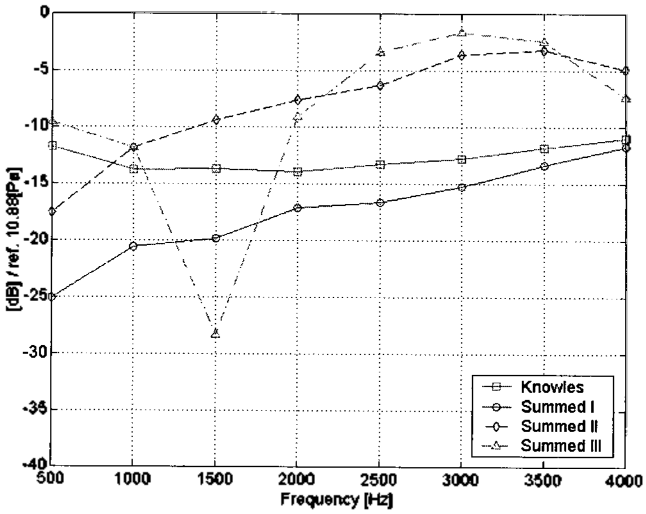


Fig. 8. Receiving sensitivity of the KEMAR head model as a function of frequency. (a) Knowles method. (b) Summed Method I. (c) Summed Method II. (d) Summed Method III. Unit=[dB]/ref. 10.88[Pa].

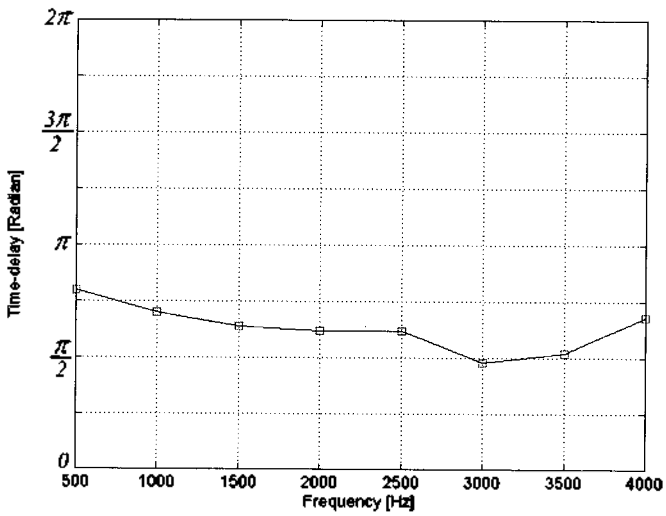


Fig. 9. The time delay in the Summed II method. Unit= [radian]

VI. Conclusion

The present paper dealt firstly the simulated sound pressures onto two twin microphones fixed in the right and the left ears of the KEMAR head model by the BEM. Then the direction of the incident sound pressure was changed from 0 degree to 360 degree around the head model. The complex values of the sound pressures onto the 4 microphones calculated by the BEM represented the input sound pressures of the 4 channels of the binaural HAs. Secondly, the calculated complex sound pressure values were systematically modified in order to represent the time delay effects of the binaural HAs. Thus a binaural directivity was achieved systematically. Then, planar binaural directivity patterns were derived by varying time delays between twin microphones.

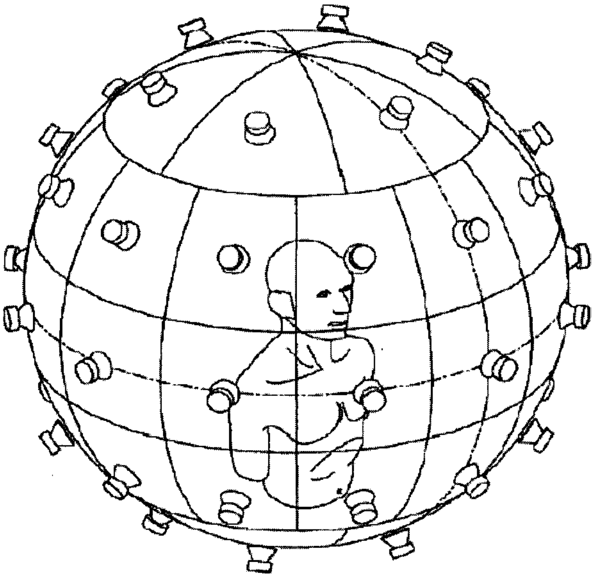


Fig. 10. Incident sound pressure direction into the center for spherical directivity patterns.

Also the directivity index was quantified. Two factors were considered for the optimal directivity pattern. One is the planar binaural directivity index. The other is the receiving sensitivity of the binaural HA.

From the results the Summed II method seems to be the most suitable in practical application for optimal directivity pattern formation because it produces the highest directivity index as well as the highest receiving sensitivity simultaneously. The optimal directivity pattern may be achieved by adjusting time delays at each frequency while maintaining the forward beam pattern is relatively bigger than the backward beam pattern. In Figure 9 it is indicated that the time delay of the digital HA needs to be varied as a function of the frequency. It is an important indication that the HA chip manufacturers must consider the time or phase delay between directional microphones as a function of frequency. The next further study will be done about spherical directivity patterns and their AI-DI (Fig. 10).

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References

1. S.S. Jarng, "Gennum GB3211 ITE Hearing Aid Chip Interface and Electro-Acoustic Testing" , Proc. of Korean Sensor Society, **14**, 59-62, 2003.
2. A. B. Dittberner, A three dimensional instrument-based approach to estimate the directivity index and predicting the directional benefit of directional microphone system in hearing aid" , Ph.D. Thesis, Uni. Of Iowa, 2002.
3. G. J. Frye, Testing Digital and Analog Hearing Instruments: Processing Time Delays and Phase Measurements" , Hearing Review, **8**, 1-8, 2001.
4. D.T.I. Francis, A boundary element method for the analysis of the acoustic field in three dimensional fluid-structure interaction problems" , Proc. Inst. of Acoust., **12** (4), 76-84, 1990.
5. S.S.Jarng, "PZT4 sonar Shell transmitter design using a coupled FE-BE method", J. of Acoust. Soc. of Korea, 17 1(E), 14-19, 1998.
6. S.S.Jarng "Optimal Design of Deep-water 30 kHz Omnidirectional Sonar Transducer using a Coupled FE-BEM", J. of Acoust. Soc. of Korea, **18** (4E), 3-9 , 1999.
7. S.S.Jarng, J.H. Lee, "Tuning fork modal analysis and sound pressure calculation using FEM and BEM", J. of Acoust. Soc. of Korea, 21 (3E), 112-118 , 2002.
8. D.T.I. Francis, "A boundary element method for the analysis of the acoustic field in three dimensional fluid-structure interaction problems", Proc. Inst. of Acoust., 12 (4), 76-84, 1990.
9. H.A. Schenck, "Improved integral formulation for acoustic radiation problems", J. Acoust. Soc. Am. 44, 41-58, 1968.
10. A.J. Burton and G.F. Miller, "The application of integral integration methods to the numerical solutions of some exterior boundary problems", Proc. R. Soc. London, Ser. A 323, 201-210, 1971.
11. R.F. Kleinman and G.F. Roach, "Boundary integral equations for the three dimensional Helmholtz equation", SIAM Rev., 16, 214-236, 1974.
12. D.T.I. Francis, "A gradient formulation of the Helmholtz integral equation for acoustic radiation and scattering", J. Acoust. Soc. Am. 93 (4) Part 1, 1700-1709, 1993.
13. http://www.knowleselectronics.com/images/products/pdf/kemar.pdf, 2004.
14. Jarng S.S., Yang H.J., Lee J.H, "Finite element mesh generation from 3D laser scanned data" , Journal of Korean CAD/CAM Society, 10 (1), 69-74, 2005.
15. S.S. Jarng, Sonar transducer analysis and optimization using the finite element method" , Ph.D. Thesis, University of Birmingham, 1991.
16. S.C. Thompson, Directional Patterns Obtained from Two and Three Microphones" , http://www.knowleselectronics.com/engineering/publications/microphones.asp, 2000.
17. ANSI S3.35, Methods of Measurement of Performance Characteristics of Hearing Aids Under Simulated In-Situ Working Conditions" , 2004.

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