Lower-order ARMA Modeling of Head-Related Transfer Functions for Sound-Field Synthesis Systems

*Jeong-Bin Yim, **Chun-Duck Kim, and ***Seong-Hoon Kang

Abstract

A new method for efficient modeling of the Head-Related Transfer Functions(HRTF's) without loss of any directional information is proposed. In this paper, the HRTF's were empirically measured in a real room and modeled as the ARMA models with common AR coefficients and different MA coefficients. To assess the validity of the proposed ARMA model, psychophysical tests by subjects were conducted over headphones with appropriate ITD's introduced in the model HRTF's. The results from tests show that the proposed ARMA model, in comparison with the conventional MA model, requires a small number of parameters to represent empirical HRTF's and improves the back-to-front confusions in sound-field localization. Thus, significant simplifications in the implementations of sound-field synthesis systems could be obtained by using the proposed ARMA model.

I. Introduction

The Head-Related Transfer Function(HRTF) characterizes the acoustical transformation accompanying a sound source from a given location in a space to a listener's ear-drum. The HRTF is empirically measured at the ear-canal of a subject or a Dummy-Head(DH) which is designed to duplicate lifelike head of a listener.

The model HRTF with a small number of parameters can greatly reduce the computational demands for a sound-field synthesis, and the shorter latencies associated with the shorter model HRTF filters can be beneficial for simulating dynamic sound-field synthesis system. Hence the efficient modeling of the HRTF's with lower-order filters is important for a sound-field synthesis system.

Generally, the HRTF, empirically measured in an ordinary room, have a non-minimum-phase property. This property causes a complex relationship between the HRTF magnitude and phase functions. Additionally, when complex valued HRTF vectors are averaged due to multiple delays from each contributing HRTF, linear interpolation of HRTF has spurious consequences [1].

To solve that problems, mentioned above, we made the assumption that HRTF's are minimum-phase sequences and Interaural Time Difference(ITD) for a given position is modeled as a pure delay. This allows us to compute a fit to the magnitude function alone, the phase function calculated as the Hilbert transform of the log-magnitude spectrum being unique. Insertion of an appropriate pure delay in the phase function at the lagging ear, in order to equate model ITD with the overall empirical ITD, has been shown to be an appropriate representation as in [2].

Furthermore, in the case of a Moving Average(MA) model, a large number of parameters are required to represent HRTF when a space has a long reverberation time. Moreover, in the case of a AutoRegressive-Moving Average(ARMA) model, all the coefficients are estimated as variable parameters for the HRTF variations. Therefore, all the parameters for the ARMA model are required to be renewed when the HRTF varies. These results are thus required a large memory for a sound field simulator.

However, it is can be thought that all the HRTF's for different azimuths have common resonance systems which contain the features that do not have directional dependence, like pinna and ear-canal. These resonance systems can be considered as the common characteristics based on physical approaches [3], [4].

With these solution cues for the modeling problems, mentioned above, HRTF's were modeled as ARMA models with common AR coefficients which remain constants despite HRTF variations. In this paper, these common AR coefficients were estimated by averaging each sets of different AR coefficients.

The evaluation for the proposed ARMA model was done by the model fits to the magnitude spectra of empirical HRTF's, and the validity of the model HRTF's

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was verified by the psychophysical test to assess the ability of sound localization for the sound-field synthesis system. All the results for the proposed models are compared with those of the conventional MA models. From the test results, the proposed ARMA model is shown to require far fewer parameters to represent HRTF's than the conventional MA model, and directional information was preserved for the model orders tested.

II. Background

2.1 SOUND-FIELD SYNTHESIS SYSTEM

Fig. 1 shows the sound-field synthesis system in a space. The HRTF's are empirically measured by Dummy-Head Microphone(DHM) [5].

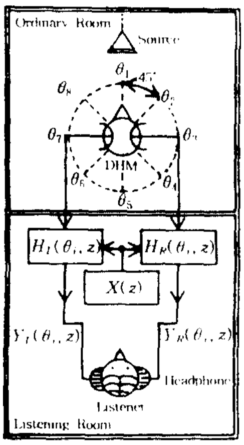


Figure 1. Block diagram of sound-field synthesis system.

If a pair of the HRTF's are denoted by $H_L(\theta_j, z)$, $H_R(\theta_j, z)$, then a pair of the outputs $Y_L(\theta_j, z)$, $Y_R(\theta_j, z)$ to an input signal $X(z)$ are expressed as

$$Y_L(\theta_j, z) = H_L(\theta_j, z)\, X(z), \qquad (1)$$

$$Y_R(\theta_j, z) = H_R(\theta_j, z)\, X(z), \qquad (2)$$

where $\theta_j (j = 1, 2, \cdots, M)$ are M azimuths of DHM to a source at fixed position, and subscripts L and R are indicate Left and Right, respectively.

Thus, the convolution of a sound stream $X(z)$ with a pair of HRTF's, $H_L(\theta_j, z)$, $H_R(\theta_j, z)$, should result in a pair of pressure signals, $Y_L(\theta_j, z)$, $Y_R(\theta_j, z)$, at the listener's two ears, having the attributes of a natural sound source at the location of HRTF measurement.

2.2 RATIONAL TRANSFER-FUNCTION MODELS

The most general formulation of rational transfer functions for Linear-Time-Invariant(LTI) systems, relates an driving input sequence, u_n, and the output sequences, $x_n(\theta_j)$ $(j = 1, 2, \cdots, M)$, by following linear difference equation [6]:

$$x_n(\theta_j) = -\sum_{k=1}^{p} a_k(\theta_j)\, x_{n-k}(\theta_j) + \sum_{k=0}^{q} b_k(\theta_j)\, u_{n-k}, \qquad (3)$$

where n is sample number, k is time index, p is AR model order, q is MA model order, and $a_k(\theta_j)$ and $b_k(\theta_j)$ represent AR(p) coefficients and MA(q) coefficients at k for $\theta_j (j = 1, 2, \cdots M)$, respectively.

Then, the system function $H(\theta_j, z)$ describing the model is given by

$$H(\theta_j, z) = \frac{B(\theta_j, z)}{A(\theta_j, z)} = \frac{1 + \sum_{k=1}^{q} b_k(\theta_j) z^{-k}}{1 + \sum_{k=1}^{p} a_k(\theta_j) z^{-k}} \qquad (4)$$

where $A(\theta_j, z)$ and $B(\theta_j, z)$ are z-transforms of the model coefficients $a_k(\theta_j)$ and $b_k(\theta_j)$ $(b_k(\theta_j) = 1$ for $k = 0)$, respectively. Here, note that the empirical HRTF takes the form of an all-zero(MA) model.

2.3 CONVENTIONAL MODELS

For the empirical HRTF's, the conventional MA (all-zero) models have the form:

$$H(\theta_j, z) = 1 + \sum_{k=1}^{q} b_k(\theta_j) z^{-k}, \qquad (5)$$

and the conventional ARMA(pole-zero) models have the form:

$$H(\theta_j, z) = \frac{1 + \sum_{k=1}^{q} b_k(\theta_j) z^{-k}}{1 + \sum_{k=1}^{p} a_k(\theta_j) z^{-k}}. \qquad (6)$$

To model M HRTF's, the MA(q) models require M FIR filters with q coefficients, and the ARMA(p, q) models require also M IIR filters with p and q coefficients. Therefore, the total number of different coefficients needed to construct the MA(q) models and the ARMA(p, q) models are $M \times q$ and $M \times (p + q)$, respectively.

III. Modeling Methods

3.1 MINIMUM-PHASE APPROXIMATION OF HRTF PHASE

The empirical HRTF, $H(z)$, measured in an ordinary room, has a non-minimum-phase property. This non-minimum-phase HRTF can be always written as the product of a minimum-phase transfer function and an all-pass transfer function [7]:

$$H(z) = H_{min}(z)\, H_{all}(z), \tag{7}$$

where $H_{min}(z)$ is minimum-phase transfer function and $H_{all}(z)$ is all-pass transfer function.

The minimum-phase impulse response for the $H(z)$ was thus obtained by Eq.(7). A minimum-phase impulse response is most compactly supported about the zero lag, compared to any other impulse response realization having the same magnitude spectrum.

3.2 COMPUTATION OF ITD

The overall ITD, k_e, was estimated by computing the peak value of the cross-correlation function for a pair of measured HRTF's. In the frequency domain, for a pair of left and right HRTF's, $L(j\omega)$ and $R(j\omega)$, the peak of the cross-correlation function is given by

$$k_e = \arg\max_t F^{-1}\,[L(j\omega)\, R^*(j\omega)], \tag{8}$$

where F^{-1} is Inverse Discrete Fourier Transform(IDFT) and $*$ is conjugate complex.

Because a pair of model HRTF's may already contain an ITD before any shift is introduced, it needs to estimated and accounted for. This delay, k_m, is estimated from the cross-correlation function for a pair of model HRTF's, $L_m(j\omega)$ and $R_m(j\omega)$. Specifically,

$$k_m = \arg\max_t F^{-1}\,[L_m(j\omega)\, R_m^*(j\omega)], \tag{9}$$

Hence the appropriate ITD to be introduced in the model HRTF's is:

$$ITD = k_e - k_m. \tag{10}$$

3.3 PROPOSED ARMA MODELS

All the HRTF's have common resonance systems which contain the features that do not have directional dependence, like pinna and ear-canal. These common resonance systems can be interpreted as common poles based on physical approaches. Since there is one-to-one correspondence between poles and AR coefficients, these poles can be considered as common AR coefficients.

The ARMA models with common AR coefficients and variable MA coefficients are given by

$$H(\theta_j, z) = \frac{1 + \sum_{k=1}^{q} b_k(\theta_j) z^{-k}}{1 + \sum_{k=1}^{p} a_k z^{-k}}, \tag{11}$$

where common AR coefficients, a_k, differ from $a_k(\theta_j)$ $(j = 1, 2, \cdots, M)$ in Eq.(6).

The total number of parameters for the proposed ARMA (p, q) models to represent M HRTF's require $p + (M \times q)$ coefficients. Hence it is clear that the proposed models require fewer parameters than the conventional ARMA models(Eq.6) with $M \times (p+q)$ coefficients to represent M HRTF's for $\theta_j (j = 1, 2, \cdots, M)$.

The block diagram of the proposed ARMA models is shown in Fig. 2. The ARMA models are representing the HRTF's with one recursive filter having a_k's and with one nonrecursive filter having a unique set of $b_k(\theta_j)$.

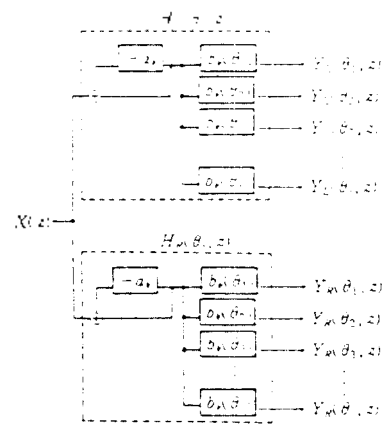


Figure 2. Block diagram of the proposed ARMA model for the multiple HRTF's. The coefficients, a_k's, are common parameters and the coefficients, $b_k(\theta_j)$, are variable parameters.

3.4 ESTIMATION OF MODEL PARAMETERS

If the MA(Q) model of a sufficiently large and finite number of Q parameters ($Q \gg p + q$) is given, then it is possible to express that model in terms of an ARMA(p, q) model. Here, the parameters of MA(Q) model corresponds to the minimum-phased impulse response with Q sequences for the HRTF's in Eq.(7) [11].

The numerator polynomials of the MA(Q) models for $\theta_j (j = 1, 2, \cdots, M)$ respresented as

$$D(\theta_j, z) = 1 + \sum_{k=1}^{Q} d_k(\theta_j) z^{-k}, \tag{12}$$

then the $d_k(\theta_j)$ parameters of the MA(Q) models that are equivalent to the ARMA(p, q) models are obtained by equating,

$$\frac{B(\theta_j, z)}{A(\theta_j, z)} = D(\theta_j, z). \tag{13}$$

This will yield for $\theta_j (j = 1, 2, \cdots, M)$,

$$d_n(\theta_j) = \begin{cases} 1 & \text{for } n = 0 \\ -\sum_{k=1}^{p} a_k(\theta_j)\, d_{n-k}(\theta_j) + b_n(\theta_j) & \text{for } 1 \le n \le q, \\ -\sum_{k=1}^{p} a_k(\theta_j)\, d_{n-k}(\theta_j) & \text{for } n > q \end{cases} \tag{14}$$

with initial conditions $d_{-1}(\theta_j) = \cdots = d_{-p}(\theta_j) = 0$.

Since the matrix of d_n parameters in Eq.(14) is Toeplitz, we made use of the Toeplitz algorithm to solve Eq.(14) for the $a_k(\theta_j)$ terms in the AR(p) parameters of the ARMA(p, q) models over the range $q + 1 \le n \le p + q$[12].

With the estimated $a_k(\theta_j)$ parameters of the AR(p) models, the common AR coefficients, a_k's, are estimated by averaging each sets of the AR coefficients, $a_k(\theta_j)$ $(j = 1, 2, \cdots, M)$, as follows:

$$a_k = \frac{1}{M} \sum_{j=1}^{M} a_k(\theta_j), \ (j = 1, 2, \cdots, M). \tag{15}$$

Previous investigations for the averaging method was reported in [8~10].

The $b_k(\theta_j)$ terms in the MA(q) parameters of the ARMA (p, q) models, then, are recovered by the convolution,

$$b_n(\theta_j) = d_n(\theta_j) + \sum_{k=1}^{p} a_k(\theta_j)\, d_{n-k}(\theta_j), \text{ for } 1 \le n \le q. \tag{16}$$

and the MA($p + q$) models that are equivalent to the ARMA(p, q) models are obtained by truncating the parameters $d_k(\theta_j)$ for $k > p + q$ to zero.

The orders p and q of the ARMA(p, q) models are determined to minimize the following quadratic expression:

$$\varepsilon^2 = \frac{1}{2\pi} \int_{-\pi}^{+\pi} |H(e^{j\omega}) A(e^{j\omega}) - B(e^{j\omega})|^2 \, d\omega, \tag{17}$$

where $H(e^{j\omega})$ is the magnitude spectrum of empirical HRTF, $A(e^{j\omega})$ and $B(e^{j\omega})$ are the magnitude spectra for the z-transforms of a_k and $b_k(\theta_j)$, respectively.

The measurement of modeling error E were obtained in terms of a normalized root-mean-square(RMS) error by

$$E = \frac{\|H(e^{j\omega}) - \hat{H}(e^{j\omega})\|^2}{\|H(e^{j\omega})\|^2} \times 100\%, \tag{18}$$

where $H(e^{j\omega})$ and $\hat{H}(e^{j\omega})$ are the magnitude spectrum of empirical HRTF and that of model HRTF, respectively.

IV. Experiment

4.1 EXPERIMENTAL SET-UP

The schematic configuration of the experimental set-up for HRTF measurement is shown in Fig. 3. The experimental work was done in a real room(19m^3: 3.0 × 2.5 × 2.5m), and its reverberation time was 100msec.

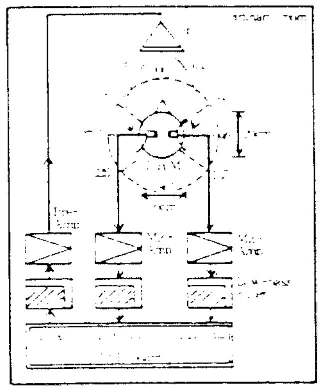


Figure 3. Configuration of experimental set-up in the real room(3.0 × 2.5 × 2.5m). The sampling frequency was 50kHz, and the sized of samples was set to 4096 points.

The responses at the inner-ear of Dummy-Head(DH) were measured by sub-miniatured condenser microphones (Audio-Technica type AT 803d), placed about 5mm inside from the center of the ear-canal entrance of DH. The measurements of HRTF's were performed by Digital-Signal-Processing board(Ariel type DSP16+) in conjunction with PC and Low Pass Filter(LPF) with cut-off frequency of 10kHz. The sampling frequency was 50kHz, and the sized of samples was set to 4096 points.

The stimulus used in the measurement was a maximum-length-sequence(MLS) signal which makes the signal-to-noise ratio(SNR) high in all frequency range. To improve the SNR, synchronized averages were taken over 100 pulses.

4.2 MODELING RESULTS

Fig. 4 shows an example of the empirical left-HRTF impulse response (top panel) at θ_1(00° azimuth) and its minimum-phase realization(bottom panel) by Eq.(7). These results show only the initial 300 sample points in the total 4096 sample points.

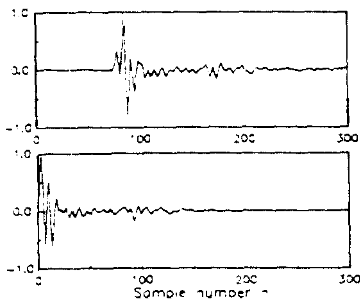


Figure 4. Example of an empirical left-HRTF impulse response (top panel) and its minimum-phase realization(bottom panel). The empirical HRTF was measured at θ_1(00° azimuth) with respects to a source at fixed position, and the results show only the initial 300 points in the total 4096 sample points.

Here, we observed that the minimum-phase sequence appears to have larger samples at its left-hand end than the sequence of the empirical left-HRTF impulse response. These minimum-phase sequences for each of empirical HRTF's, were used for the estimation of model parameters in Eqs.(12~16).

Using the estimated model parameters, each of empirical HRTF's for 8 azimuths was modeled as the ARMA model with 50 common AR coefficients and 50 MA coefficients. Here, the number of coefficients was predetermined to minimize modeling errors by Eq.(17). To show the superior efficiency of the proposed model, conventional MA model were also constructed with 100 variable MA coefficients. Therefore, the total number of coefficients used in tests are 450 coefficients($50 + 50 \times 8$) in the proposed ARMA(50, 50) models and 800 coefficients(100×8) in the conventional MA(100) models. Results from the measurement of modeling error E by Eq.(18) are shown in Fig. 5.

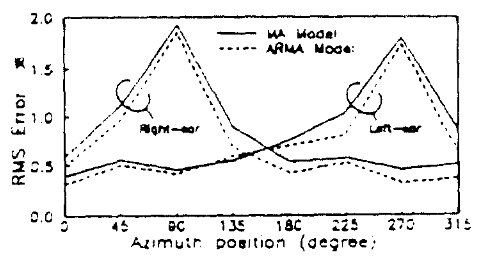


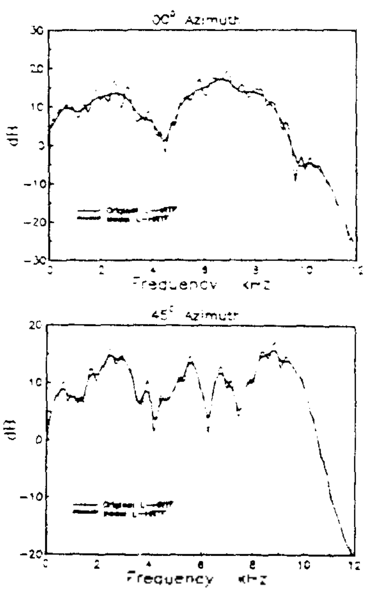
Figure 5. RMS errors of model HRTF's for 8 azimuths in horizontal plane. The results of the proposed ARMA(50, 50) model and the conventional MA(100) model are denoted by dashed lines and solid lines, respectively.

In overall, the modeling errors of proposed ARMA(50, 50) models are somewhat less than those of the MA(100) models over all the azimuth positions, despite the proposed ARMA(50, 50) models used only about half as many filter taps as the conventional MA(100) models.

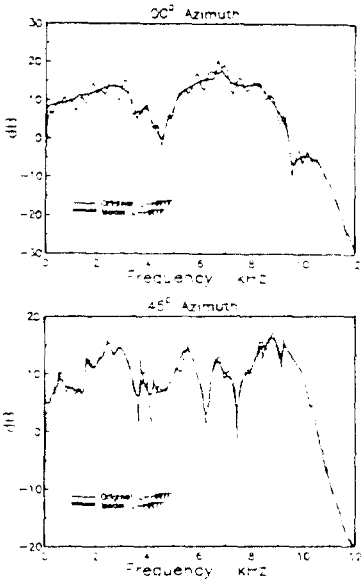
The both modeling errors are worse for the empirical HRTF's at the ear maximally shadowed from the source (around 90° azimuth in the right-ear and 270° azimuth in the left-ear), because of poor SNR. Furthermore, the modeling errors of the right-and left-model HRTF's show that the both are asymmetric. It is typical to assume that HRTF measurements are symmetric about the vertical median plane because of symmetric feature of DHM. However, actual measurements show that the HRTF's are not symmetric. The asymmetries are perceptually important, especially for localizing sources as reported in [13].

Fig. 6 shows the examples of model fits to the empirical left-HRTF's for the ARMA(50, 50) model and the MA(100) model in 00° and 45° azimuths.

Results from model fits show that the ARMA(50, 50) models are better fits to empirical HRTF's than those of the MA(100) models. Moreover, it is clear that the features of deep peaks and dips in magnitude spectra of the ARMA(50, 50) models are coincident with those of the empirical HRTF's. These results confirm that the proposed ARMA models with common AR coefficients are successful representations of the empirical HRTF's

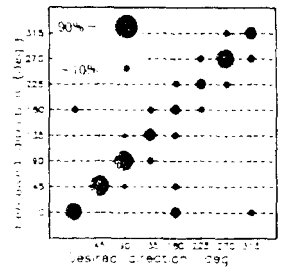


(a)

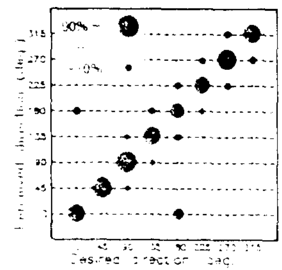


(b)

Figure 6. Examples of model fits to the empirical left-HRTF's in 00° and 45°azimuths : (a) MA(100) model ; (b) ARMA(50, 50) model. The magnitude spectra of the empirical HRTF and the model HRTF are denoted by a fine solid line and a thick solid line, respectively.



(a)



(b)

Figure 7. Results from psychophysical tests using the model HRTF's : (a) MA(100) model ; (b) ARMA(50, 50) model. The results were obtained by averaging ten subjects for each azimuths.

V. Psychophysical Testing

With appropriate ITD introduced in the model HRTF's, psychophysical experiments were conducted over headphones. The sensitivity of subjects to discriminate between perceived and target directions using the model HRTF's was measured. Ten subjects, three female and seven male, aged 20 to 35 years, were participated in tests. All subjects had assessed normal localization ability and thus were well practised in the task. Tests were performed with conditions as following,

- White noise stimuli synthesized with model HRTF's
- 500 msec burst ; 200 msec silence ; 10msec linear on/off ramp
- Comfortable listening levels ~65dB SPL(10dB rove)
- Total of 5 tests per a subject with random order stimulus :
- Test 1 : 45-270-135-90-225-180-315-00 (degree)

  . . . .

  Test 5 : 135-315-00-90-270-225-180-45 (degree)

Fig. 7 shows the correct answer rate of the perceived directions. This correct answer rate was averaged by ten subjects for each azimuths, and denoted by 9 different area of circles from the largest one for the rate above 90% to the smallest one for the rate below 10%. Thus the sum over all the answer(%) gives 100% along with the axe of perceived direction.

Results from psychophysical tests show that directional information, in general, is preserved in the both models except for the back-to-front azimuths(00° and 180°). In the back-to-front azimuths, the perceived localization for the ARMA(50, 50) models are improved than the MA (100) models. These results are consistent with the model fits to the empirical HRTF magnitude in Fig. 6. The back-to-front confusions are already a known fact which occurs in the reproducing system by two loud-speakers or heddphones as reported in [14].

VI. Conclusions

The HRTF's were modeled as the ARMA models with common AR coefficients and different MA coefficients. The common AR coefficients were estimated by averaging each sets of different AR coefficients with respect to HRTF's. The modeling method is based on the assumption that HRTF's are minimum-phase sequences and Interaural Time Difference(ITD) for a given position is modeled as a pure delay.

From the test results, one can see that the modeling errors of ARMA(50, 50) models are somewhat less than those of the conventional MA(100) models over all azimuths, despite the ARMA(50, 50) models used only about half as many filter taps as the MA(100) models. In the spectral features for the ARMA(50, 50) models, the deep peaks and dips of magnitude spectra are coincident with those of the empirical HRTF's. Thus, the results yielded the improvement of back-to-front confusion in a sound-field localization, confirming the efficiency of the proposed model.

Thus, significant simplifications in the implementations of sound-field synthesis systems could be obtained by using the proposed model. Studying the theoretical background of averaging method for the estimation of common AR coefficients is left for future direction of research.

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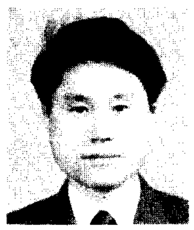

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