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A New Side-lobe Canceller with Adaptive Compensator

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Abstract

In the conventional generalized side-lobe canceller (GSC), the output is an estimated error signal that causes the adaptive filter weights do not converge to the optimal value. This paper presents a new side-lobe canceller with adaptive compensator that reduces the misadjustment of the adaptive filter coefficients for the structural problem in the GSC. The adaptive compensator separates the output signal from the estimated error. The newly estimated error signal converges to the zero while the output signal tracks the target signal. This paper shows improvement of the performance by comparing the computer simulation of the output signal of GSC with the output signal of the proposed algorithm.

Keywords: *Array beam-forming, Side-lobe canceller, Robust adaptive algorithm*

I. Introduction

An array signal processing deals with the processing of signals carried by propagating wave phenomena. The received signal is obtained by means of an array of sensors located at different points in space in the field of interest. The aim of array processing is to extract useful characteristics of the received signal field (e.g. its signature, direction, the speed of propagating wave)[1].

In most array signal processing, beam-forming algorithm is used. In RADAR application, beam-forming algorithm steers the direction of the wanted signal in receiving multiple signals and interferences. Conventional delay-and-sum beam-forming makes specific assumptions on the form of the propagation signal. Therefore an adaptive beam-forming has become more important in acoustical application for spatial filtering. Beam-forming can be applied to the situation where a receiver receives multiple signal (sound) sources. The objective of the adaptive beam-forming is attenuating the jammer or the interference signals while maintaining the target signals[2].

Adaptive beam-forming algorithm is proposed in the 1970's, the researches and applications are excited in the 1990's. The time-varying processing to the array sensors makes it possible to approach to the speech signal, acoustic signal and digital communication system, and so on. The receiver does not know about the characteristics of the transmitted signal, a blind detection method is used to obtain the hiding informations[3]. Frost et al. proposed the concept of the side-lobe canceller. The idea is to eliminate the unwanted signal which impinges to the side-lobe while not damaging the target signal by giving an unity gain simultaneously[4]. Griffiths-Jim adaptive beam-former is generally used because of the simplicity of structure and implementation by the block matrix.

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The block matrix presents a simple update rule without constraint in Frost's adaptive beam-forming algorithm. Griffiths-Jim adaptive beam-former generalized a side-lobe canceller, so we can call it a generalized side-lobe canceller; GSC. In this paper we will use the term 'GSC' and 'Griffith-Jim adaptive beam-former' interchangeably. The adaptive filter commonly adopts least-mean-square (LMS) algorithm in updating the filter coefficients. The adaptive side-lobe canceller is viewed spatially as an adaptive noise cancelation (ANC) concept. For example, ANC is used to eliminate the noise from a specific direction in the application of 'own-ship' noise canceller with a towed array[1,4,5].

Adaptive filtering is a common solution to many signal processing applications in which the environments are time varying. Such applications include an ANC, line enhancement, channel equalization, etc[2,6,10]. In most cases, the output signal of the ANC should be used to track the target signal by reducing the noise components. When the filter coefficients are updated, the LMS adaptive algorithm uses the output signal as an estimated error signal. Therefore the fluctuating output components cause the filter coefficients to misadjust in adapting to the optimum values. To overcome this problem, K. S. Son[10] and K. C. Ho [11] proposed a new ANC with two adaptive components: model filter and a noise whitening filter. The convergence study reveals that there is only one minimum in the error surface, and global convergence is guaranteed. Analysis of this method can reach to stable asymptotical convergence and obtain minimum parameter variance is the best linear unbiased estimator (BLUE)[11]. ANC and GSC have the same structure conceptually. So, GSC being proposed by Griffith and Jim has the same problem in the conventional ANC.

This paper proposes a new structure with an adaptive compensator (cascade-connected to the conventional GSC). A new estimated error is produced by the compensator. The linear predictor is used as an adaptive compensator. The new estimated error generated this adaptive compensator will converge to zero, then the adaptive filter coefficients will be adjusted to the optimal values. In the first computer simulation, we used the mono-chromatic wave as the target signal and the random white Gaussian signal as the jammer signal. In the second, the random colored Gaussian signal is used as the target signal and the random white Gaussian signal as the jammer signal. Both simulations show the improvement of the performance to the problem in the GSC.

II. Griffiths-jim Adaptive Beam-former[5]

This paper is based on the Griffiths-Jim adaptive beam-former. Fig. 1 shows the GSC with N - sensors. The fixed vector $W_f = [w_{f1}, w_{f2}, \ldots, w_{fN}]^T$ is the weight vector that is used in conventional beam-former. It determines a beam-width and terminates the priori known interferences.

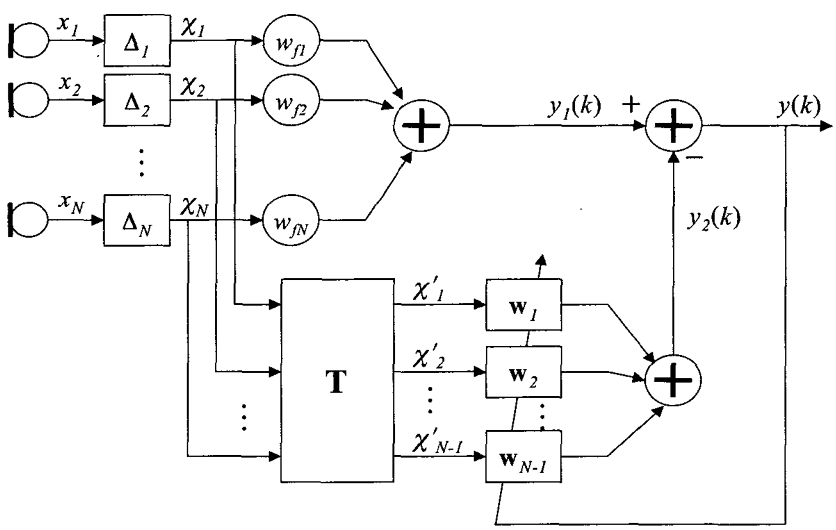


Figure 1. The structure of Griffiths-Jim side-lobe canceller.

The delay $\triangle$ at each sensor plays a role in steering the sensors to the look-direction without physical rotation. This processing is more efficiently in cost and simple than physically movement[1,2,6]. An estimate of the desired signal direction is required, then control the delay to get the maximum SNR. The statistical signal processing is used to catch the direction of the wanted signal in communication system[8] or in under-water acoustic applications[9].

If the array is steered to the target signal direction, each signal component of the sensor is the same, i. e. all signals at the array are in-phase signal. The block matrix has the form each sum of the row is zero, so the block matrix prevents the signal which is incident to the broadside. For example, the simple block matrix is like this

$$T=\begin{bmatrix} 1 & -1 & 0 & 0 & \dots & 0 \\ 0 & 1 & -1 & 0 & \dots & 0 \\ & & \dots & & & \\ 0 & 0 & \dots & 0 & 1 & -1 \end{bmatrix} \tag{1}$$

The succeeding adaptive filter section, which estimates signals that do pass through by considering the total output, can freely adjust the optimal weights (filter coefficients) to emphasize these signals so that they can be subtracted from the main beam's output. After removing the constant wavefront signals, the adaptive section emphasizes the remainder based on what's in the output and how the conventional beam former shades the desired signal. The general mathematical framework for the GSC relies on unconstrained optimization. Because the constrained algorithm proposed by Frost. et. al. needs a cost function at every updates, so there are many limitation in adopting to the real application[4,5].

Now, consider the update equation using LMS algorithm [2,6].

$$W(k+1)= W(k)+\mu(k)\,e(k)X(k) \tag{2}$$

The filter coefficients with length L are defined by weight vector,

$$W(k)=[\ w_1^T,\ w_2^T,\ \dots\ ,\ w_L^T]^T. \tag{3}$$

The received signal vector is

$$X(k)=[x'_1(k),x'_2(k),\dots,x'_{N-1}(k),x'_1(k-1),\dots,x'_{N-1}(k-L+1)]^T \tag{4}$$

This vector has the length $N\times L$.

Let the target signal as $s(k)$ at broadside and the jammer as $j(k)$ at arbitrary direction ($\theta\neq 0^o$), and the direction is estimated perfectly, and the fixed weight vector is set all elements '1'. Then deduce the LMS equation,

$$\begin{aligned} W(k+1) &= W(k)+\mu(k)y(k)J_d(k) \\ &= W(k)+\mu(k)\{y_1(k)-y_2(k)\}J_d(k) \\ &= W(k)+\mu(k)s(k)J_d(k)+\mu(k)\,j_s(k)J_d(k) \\ &\quad -\mu(k)\,W^T(k)J_d(k)J_d(k) \end{aligned} \tag{5}$$

In this equation $J_d(k)$ is the jammer component to the adaptive filter via a block matrix. And $j_s(k)$ is the also another jammer component that is just the sum of the jammer of each sensor. In equation (5), the term $\mu(k)s(k)J_d(k)$ makes filter misadjust because of the fluctuating effect of the target signal $s(k)$. Adaptive noise cancelling (ANC) filter is one of the application of adaptive signal processing. The goal of a ANC is to estimate a signal of interest from a noise corrupted observation. The performance of an adaptive filter is limited by the misadjustment resulting from the variance of adapting parameters. K. C. Ho proposed a method of minimizing misadjustment adaptive FIR filter. The conventional ANC with adaptive compensator (noise whitening filter) is approaching the best linear unbiased estimator asymptotically. To make this possible, there should be only one minimum in the error surface and global asymptotical convergence should be guaranteed[11].

III. Proposed Side-lobe Canceller

In many applications with adaptive array beam-forming, the stable convergence is important. However, as described above, conventional GSC has a problem of unstable convergence. To overcome the problem of the GSC, this paper proposes a new side-lobe canceller with adaptive compensator. The linear predictor is used as an adaptive compensator as seen in Fig. 2.

An adaptive compensator is cascade-connected to the GSC, then the output of the compensator is used as an estimated error signal to the adaptive filter. This new estimated error signal is converged to zero, so the fluctuation of the estimated error signal is reduced. The target signal is continually tracked by the adaptive output signal. The estimated signal is separated from the conventional GSC output signal by using the adaptive compensator.

The proposed technique is represented as the following equation.

$$v(k) = y(k) - y_c(k) = y(k) - \boldsymbol{W}_c^T(k)\,\boldsymbol{y}(k-1) \tag{6}$$

$$\boldsymbol{W}_c(k+1) = \boldsymbol{W}_c(k) + \mu(k)\,v(k)\,\boldsymbol{y}(k) \tag{7}$$

The compensator output signal is denoted by $v(k)$, and the weight vector $\boldsymbol{W}_c(k)$ represents a linear predictor filter coefficients. This output $v(k)$ of the compensator is used as an estimated error signal in LMS adaptive algorithm. The object of GSC is to minimize the side-lobe input signal while maintaining the target signal. All in all, this scheme uses the output signal $y(k)$ to track the target signal and computes a new estimated error signal $v(k)$. Here, the signal $y(k)$ is used as an primary input signal to the adaptive compensator.

$$y(k) = y_1(k) - y_2(k) \tag{8}$$

$$\boldsymbol{y}(k) = [y(k), y(k-1), \ldots, y(k-L+1)]^T \tag{9}$$

The update equation of the side-lobe canceller uses the following LMS algorithm.

$$\begin{aligned}
\boldsymbol{W}(k+1) &= \boldsymbol{W}(k) + \mu(k)\,v(k)\,\boldsymbol{J}_d(k)\\
&= \boldsymbol{W}(k) + \mu(k)\,\{y(k) - y_c(k)\}\,\boldsymbol{J}_d(k)\\
&= \boldsymbol{W}(k) + \mu(k)\,(y_1(k) - y_2(k) - y_c(k))\,\boldsymbol{J}_d(k)\\
&= \boldsymbol{W}(k) + \mu(k)[\,s(k) - \boldsymbol{W}_c^T(k)\,\boldsymbol{s}(k-p)\,]\,\boldsymbol{J}_d(k)\\
&\quad + [\,e(k) - \boldsymbol{W}_c^T(k)\,\boldsymbol{e}(k-p)\,]\,\boldsymbol{J}_d(k)
\end{aligned} \tag{10}$$

In this equation, we can observe that the target signal component $s(k)$ is reduced by $\boldsymbol{W}_c^T(k)\,\boldsymbol{s}(k-1)$ component to the zero asymptotically. The vector term $\boldsymbol{e}(k)$ represents the error between the upper side jammer component (in the primary signal) and the lower side jammer component (in the reference signal). The target signal data vector and estimated error signal of the compensator is described in (11).

$$\boldsymbol{s}(k) = [s(k), s(k-1), \ldots, s(k-L+1)]^T \tag{11}$$

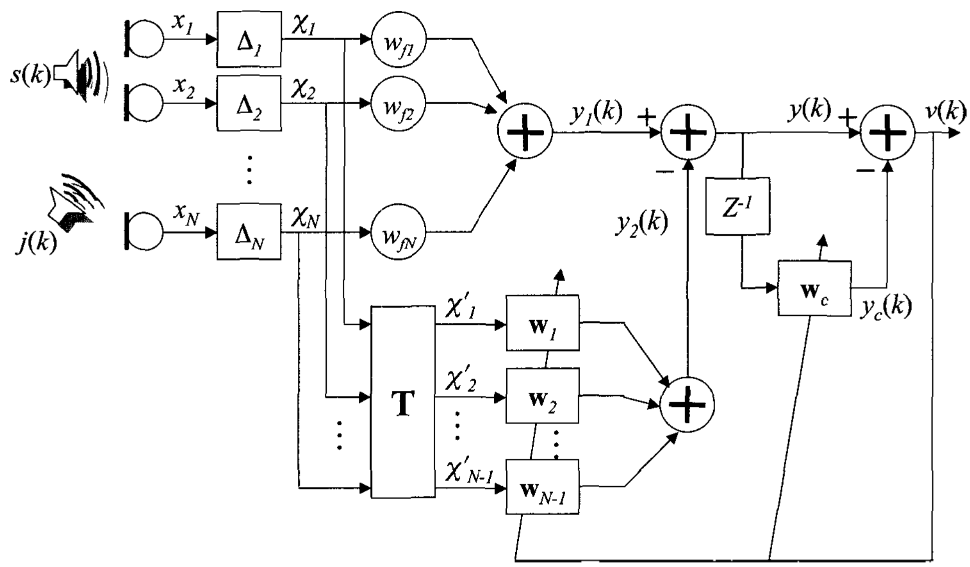


Figure 2. The structure of the proposed side-lobe canceller with adaptive compensator.

$$e(k) = j_s(k) - W^T(k)\, J_d(k) \tag{12}$$

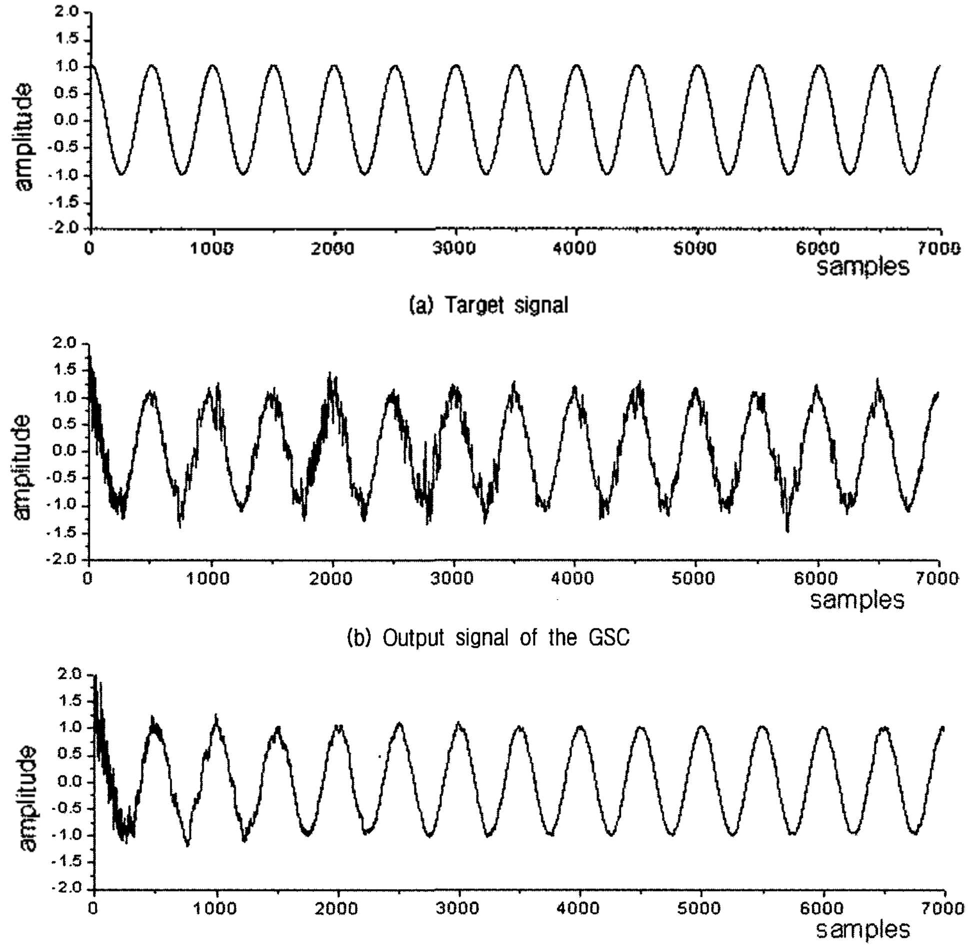
To use in computer calculation, equation (12) is formatted as a vector form.

$$\underline{e}(k) = [\, e(k), e(k-1), \ldots, e(k-L+1)]^T \tag{13}$$

The ' z^{-p}' means p-samples delay, the weight vector $W_c(k)$ is the linear predictor coefficient set. The term $[\, s(k) - W_c^T(k)\, s(k)]\, J_d(k)$ reduces the effect of the fluctuating target signal $s(k)$, causing the filter to have unstable convergence. The proposed adaptive compensator separates the estimated error signal from the output signal with which non-converging target signal is generally computed. Even though the adaptive compensator which converges to zero decreases the speed of convergence, stable convergence and right adjustment of the adaptive filter are more critical. Our experimentation shows that the proposed algorithm works more efficiently than the conventional GSC.

IV. Simulations and Results

To compare the performance of the proposed method with that of the Griffiths-Jim adaptive algorithm, this paper used a computer simulation. As seen at Fig. 3 the target signal $s(k)$ is a mono-chromatic wave signal with a 0.16 m wave length. Its amplitude is '1', and the propagation velocity in underwater environment is 1.5 km/s. The jammer signal is white Gaussian noise with a '0' mean, '1' variance, and the same velocity. The inter-sensor



(a) Target signal

(b) Output signal of the GSC

(c) Output signal of the GSC with compensator

Figure 3. Output signals of GSC and the proposed GSC with compensator.

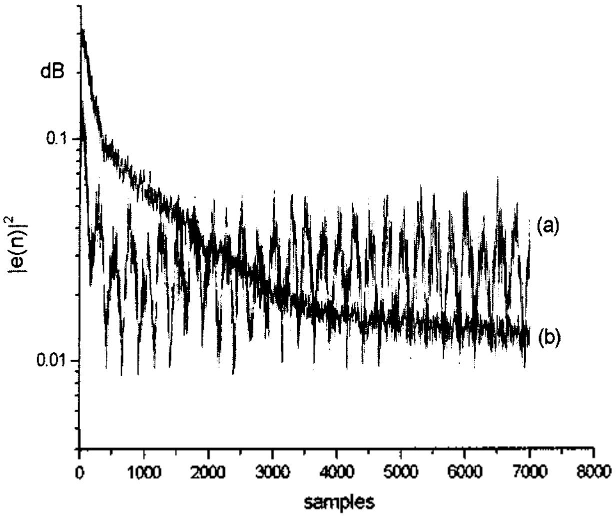


Figure 4. Learning curves of error signals with a sinusoidal signal as a target signal and with a white Gaussian random signal as a jammer.
(a) learning curve of GSC, (b) learning curve of GSC with compensator

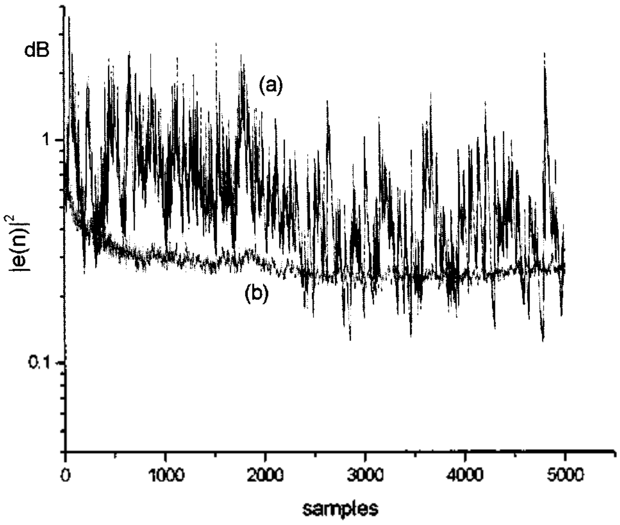


Figure 5. Learning curves of the error signals with a colored Gaussian random signal as a target signal and with a white Gaussian signals a jammer.
(a) GSC, (b) GSC with compensator

distance is 0.08 m to avoid a grating lobe effect.

Fig. 3 illustrates results of side lobe cancelling with conventional GSC and the proposed GSC with compensator. The target signal is showed in (a), then (b) represents the conventional GSC output signal and the lowest waveform, (c) is the output signal of the proposed GSC with compensator. It is shown that the output of conventional GSC is noisy and the proposed GSC yields similar waveform of target sinusoidal signal. Learning curves of GSC are illustrated in Fig. 4. (a) and (b) are the learning curves of a conventional GSC and the proposed GSC. The error signal component of the conventional GSC has a fluctuation, then the convergence of the filter is not exact to the optimum value. But the proposed algorithm shows a stable convergence. Although the convergence speed is slower than that of the GSC or sum algorithm, its performance in the tracking of the target signal and the reduction of the misadjustment of the exact optimal value works well.

Another simulation used two independent Gaussian random signals. One signal is a colored random signal for the target signal, generated by filtering Gaussian noise through the IIR filter $y_k = 0.95y_{k-1} + 0.19y_{k-2} + 0.09y_{k-3} - 0.5y_{k-4} + x_k$ described in[11]. Another is a white random signal for the jammer signal. The Learning curves of both algorithms are depicted in Fig. 5. It is certain that the proposed GSC has a better performance. However, the average error power of the GSC does not clearly guarantee for the improvement of the performance to the proposed one. Fig. 6 represents the power spectral density of the error signal. This shows that the proposed structure has a less error power density. In this figure, the power density of the error signal in the GSC decreased at maximum 3 dB in the range of 100 ~ 300 Hz, and about 1 dB in the other range. The very high values in the low frequency range is due to the structure of the array sensor[4].

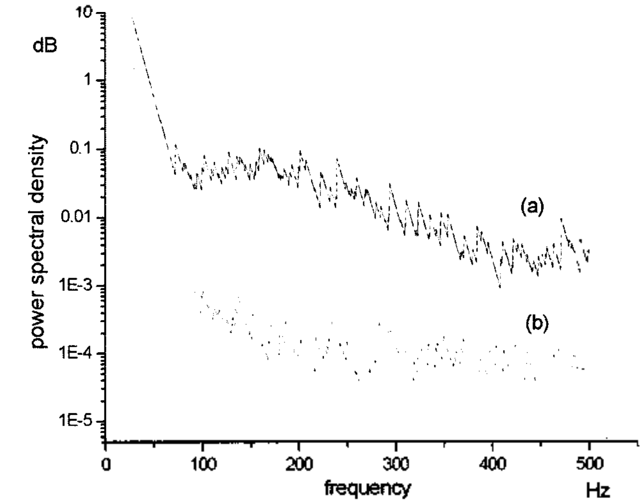


Figure 6. Power spectral densities of the error signals.
(a) GSC, (b) GSC with compensator

V. Conclusion

The simulation results illustrate the effects of the adaptive compensator cascade-connected to GSC. We showed the performance improvement by comparing the GSC with the proposed side-lobe canceller for the narrow-band signal and colored noise signal. The PSD of the proposed algorithm is about maximum 3 dB improved to the GSC, and about 1dB improved in average.

However, the result for the wide-band signals and highly correlated signals between the target signal and the jammer signal is not satisfied. The study for this limitation is necessary and will be investigated. And in the real application, the more study for the non-linearity of array and the calibration of the array should be considered carefully.

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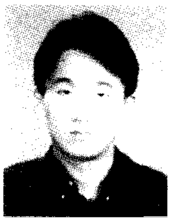

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