AMNOR System with Linear Phase Response to the Desired Signal

(원하는 신호에 대해 선형 위상 응답 특성을 갖는 AMNOR 시스템)

Byeong-Gwan Iem,* Byeong-Mo Cho,* Il-Whan Cha,* Dae-Hee Youn*

임 병 관,* 조 병 모,* 차 일 환,* 윤 대 희*

요 약

본 논문에서는 원하는 방향에 대해서 선형 위상 응답 특성을 갖는 적응 빔 형성기에 대해서 설명한다.

본 논문에서 제안하는 빔 형성기는 원하는 신호에 대해서 선형 위상 응답 특성을 얻기 위해 전방 적응 빔 형성기(Forward adaptive beamformer)와*후방 적응 빔 형성기(Backward adaptive beamformer)를 이용한다. 두 빔 형성기의 각각의 출력 신호를 합하여 원하는 신호를 얻는다.

TDL(Tapped-Delay-Line) 필터 계수는 가상신호(fictitious desired signal)와 출력 신호간의 차이 값의 파워가 최소가 되도록 조정된다.

원하는 방향에 대한 빔 형성기의 임펄스 응답이 중심 샘플에 대해서 대칭이 되도록하여 원하는 방향에서 입사되는 신호에 대한 선형 위상 응답 특성을 얻을 수 있다.

출력 잡음 분산을 최소로 하는 반면 원하는 신호 방향에 대한 주파수 응답의 왜곡을 미리 정한 허용치보다 작게 함으로써 음성 신호를 향상시키는 AMNOR(Adaptive Microphone-array system for Noise Reduction) 시스템에 본 논문에서 제안한 빔 형성기를 적용하여 컴퓨터 시뮬레이션 결과를 제시한다.

ABSTRACT

This paper describes an adaptive beamformer with linear phase response to the look-direction. A uniqueness of the proposed beamformer is that it uses a forward adaptive beamformer and a backward adaptive beamformer to obtain the linear phase response to the desired signal. The output singals of the two beamformers are added to yield the final output.

The relevant tapped-delay-line filter coefficients are updated to minimize the power of the difference between a fictitous desired signal and the output. The linear phase response to the look-direction signal is obtained by making the look-directional impulse response of the beamformer even symmetric about its midpoint.

*ASSP Lab. Dept. of Electronic Engineering, Yonsei University

As an application of the beamformer, it is applied to realize the AMNOR(Adaptive Microphone-array system for NOise Reduction) system, which enhances speech signal by minimizing the output noise power, while maintaining the degradation in the frequency response to the desired signal below some permissible level. Simulation results demonstrating the performances of the proposed methods are presented.

I. INTRODUCTION

Adaptive microphone-array systems have been studied to receive a broad band speech signal arriving at the desired direction while minimizing spatially separated interfering noise power[1,2,3]. When a microphone is used to receive a broad band speech signal, ambient noise from other sources often interferes with the speech signal. A possible approach to solve such a problem is the use of the Frost's beamformer[4] which minimizes the noise power while passing the desired signal without distortion. It has been shown[1,2,3] that the SNR improvement using the Frost's beamformer is generally not satisfactory due to the rigidity of the constraint for the frequency response to the look-direction signal. To overcome such a difficulty, adaptive beamforming algorithms with soft constraints have been studied[1,2,3]. These methods try to attain maximum noise reduction while allowing a small degree of degradation in the frequency response to the desired signal. Even though they have been successful in attaining improved noise reduction, they may introduce phase distortion to the look-direction signal. The intent of this paper is to present an adaptive beamformer with linear phase response to the desired signal. The AMNOR (Adaptive Microphone-array system for NOise Reduction) optimization criterion[1,2] is applied to allow permissible degree of amplitude distortion in the frequency response to the desired signal and the relevant multichannel filter coefficients are computed using the least-mean-square(LMS) algorithm[5] To attain the linear phase response, it is proposed to use a forward and a backward adaptive beamformers and the output signals of the two beamformers are added to yield the final output.

After describing the AMNOR system with linear phase response in Section II, computer simulation results will be included in Section III.

II. AMNOR System with Linear Phase Response

Fig. 1 shows the block diagram of the AMNOR system[1] in which the M-input / single-output filter weights represented by H_2 are computed using the LMS algorithm to minimize the power of the difference $e(n)$ between the fictitious desired (FD) signal $A\ s'(n-\tau)$ and the beamformer output $y'(n)$; and the other filters H_1 and H_3 copy the filter coefficients from H_2. The filter coefficients are computed in the absence of the look-direction signal.

To control the response degradation level $D(A)$, the third filter H_3 takes noise free FD signal as its input ; the resulting response degradation $D(A)$ is compared to the pre-set threshold value D ; and the scale factor A controlling the degradation level is computed.

Here, $G_1(Z)$ to $G_m(Z)$ represent the transfer functions characterizing the propagation paths from the desired signal source to the m-th, $m=1, 2, \cdots, M$ microphones. When $D(A)$ is sufficiently close to D, the first filter H_1 copies the coefficients from the second filter H_2 and process the M sensor output signals to yield the enhanced look-direction signal.

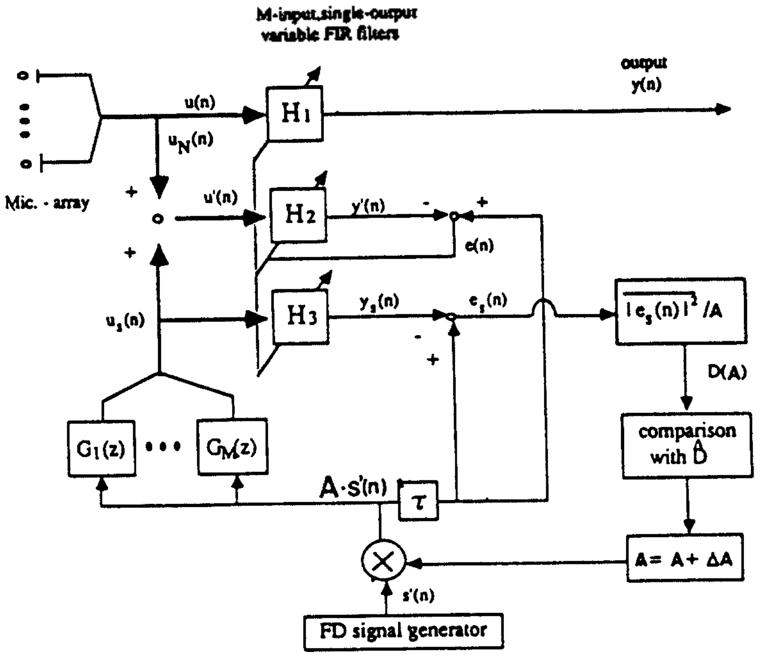

Fig. 1. Block diagram of the AMNOR system

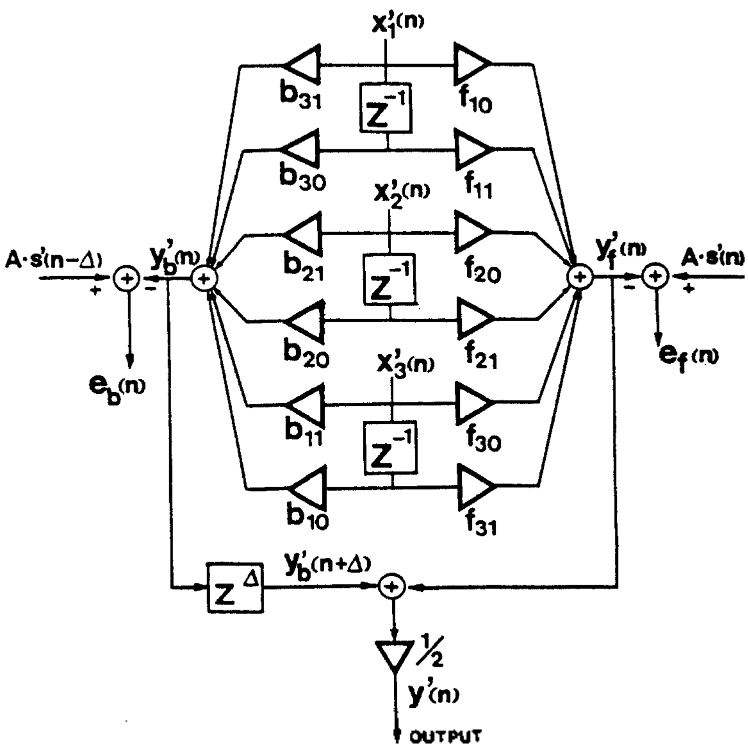

Fig. 2. Block diagram of the forward and backward adaptive filter

Now, consider the noise reduction problem under the assumptions that the microphones are equally and very closely spaced to construct linear array ; the signal source and microphones are far apart to assume plane wave ; and the direct signal propagatoin path is short compared to indirect ones. Under these assumptions, we have.

$$G_m(z)=z^{(1-m)\delta} \quad 1\leq m\leq M \tag{1a}$$

where

$$\delta=\frac{d\ \cos\theta_0}{V\ T_s} \tag{1b}$$

d : distance between microphones

θ_0 : incident angle of the desired signal

V : wave propagation velocity

T_s : sampling interval.

To attain the linear phase response of the AMNOR system to the desired signal, it is proposed to use a forward and a backward adaptive beamformers as illustrated in Fig. 2 for M=3 and N=2.

Here, N denotes the number of coefficients for each TDL filter. From Fig. 2, we have

$$e_f(n)=A\ s'(n)-y_f'(n) : \text{forward estimation error} \tag{2a}$$

$$y_f'(n)=F^T\ X_f(n) \tag{2b}$$

$$e_b(n)=A\ s'(n-\triangle)-y_b'(n) : \text{backward estimation error} \tag{2c}$$

$$y_b'(n)=B^T\ X_b(n) \tag{2d}$$

$$\begin{aligned} X_f(n)=[&x_1'(n)\ x_1'(n-1)\cdots x_1'(n-N+1)\ x_2'(n)\ x_2'(n-1)\cdots \\ &x_2'(n-N+1)\ x_M'(n)\ x_M'(n-1)\cdots x_M'(n-N+1)]^T \end{aligned} \tag{2e}$$

$$\begin{aligned} X_b(n)=[&x_M'(n-N+1)\cdots x_M'(n-1)\ x_M'(n)\cdots \\ &x_2'(n-N+1)\ x_2'(n-1)\ x_2'(n)\ x_1'(n-N+1)\cdots x_1'(n-1)\ x_1'(n)]^T \end{aligned} \tag{2f}$$

$$F=[f_{10}\ f_{11}\cdots f_{1(N-1)}\ f_{20}\cdots f_{M0}\ f_{M1}\cdots f_{M(N-1)}]^T \tag{2g}$$

$$B=[b_{10}\ b_{11}\cdots b_{1(N-1)}\ b_{20}\cdots b_{M0}\ b_{M1}\cdots b_{M(N-1)}]^T \tag{2h}$$

$$\triangle=(M-1)\ \delta+(N-1). \tag{2i}$$

We can see that by computing the coefficient vectors F and B to minimize the sum of the powers of $e_f(n)$ and $e_b(n)$, the multichannel filter output signals $y_f'(n)$ and $y_b'(n)$ estimate A $s'(n)$ and A $s'(n-\triangle)$, respectively. Thus the final output $y'(n)$ is given by

$$y'(n)=[y_f'(n)+y_b'(n+\triangle)]/2. \tag{3}$$

It is well known[6, 7, 8] that an FIR filter with symmetric impulse response function has linear phase response. From Fig. 2, if we let $f_{mn}=b_{mn}$, the impulse response function of the system to the desired signal is symmetric about the center coefficient given by $2f_{10}$ located at the time index of $\triangle$ samples. Computing the transfer function of the system to the desired signal, we have

$$H_s(Z)=\frac{Y'(Z)}{S'(Z)}=\frac{1}{2}\sum_{m=1}^{M}\sum_{n=0}^{N}f_{mn}[Z^{-(m-1)\delta-n}+Z^{(m-1)\delta+n}]. \tag{4}$$

If the signals are stationary, we can see that the optimum coefficient vectors F_{opt} and B_{opt} minimizing $E[e_f^2(n)]$ and $E[e_b^2(n)]$, respectively, are the same. However, applying the LMS algorithm to compute the weight vectors separately, the resulting coefficient vectors are different. Thus it is proposed to find the weight vector minimizing the sum of the mean squared estimation errors. That is

$$\epsilon=E[e_f^2(n)]+E[e_b^2(n)]. \tag{5}$$

The LMS algorithm computing the coefficient vectors can be expressed as

$$F(n)=B(n) \qquad (6a)$$

and

$$F(n+1)=F(n)-\mu\nabla(n) \qquad (6b)$$

where

$$\nabla(n)=\partial\epsilon/\partial F(n)=-2[e_f(n)\ X_f(n)+e_b(n)\ X_b(n)]. \qquad (6c)$$

The AMNOR system equipped with the forward and the backward beamformers will be referred to as the FB-AMNOR (Forward and Backward-AMNOR) system.

Ⅲ. Simulation Results

To demonstrate the performances of the proposed FB-AMNOR system, the following situations are simulated.

number of microphones=6
distance between neighboring microphone=5 cm
sampling frequency=8 KHz
signal−pass band=0.3−3.0 KHz
look-direction=120°
interference noise direction=60°
signal−to−interference noise ratio=−10dB, 0 dB and 10 dB.

Fig. 3 and Fig. 4 display the directivity pattern and the frequency response of the FB-AMNOR system. Fig. 5 shows the frequency response to the desired signal of the noise reduction filter derived at the fixed FD signal levels of −10dB, 0 dB and 10 dB. Fig. 6 shows the measured rel-

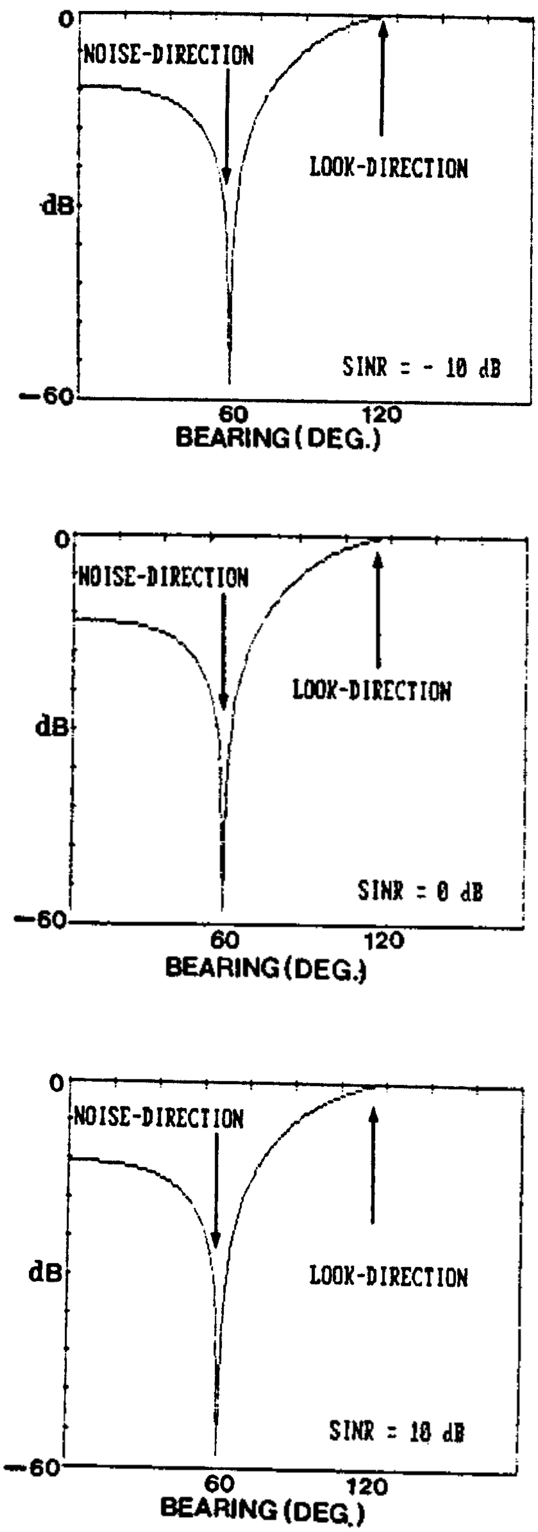

Fig. 3. Directivity patterns
a) −10dB b) 0 dB c) 10 dB

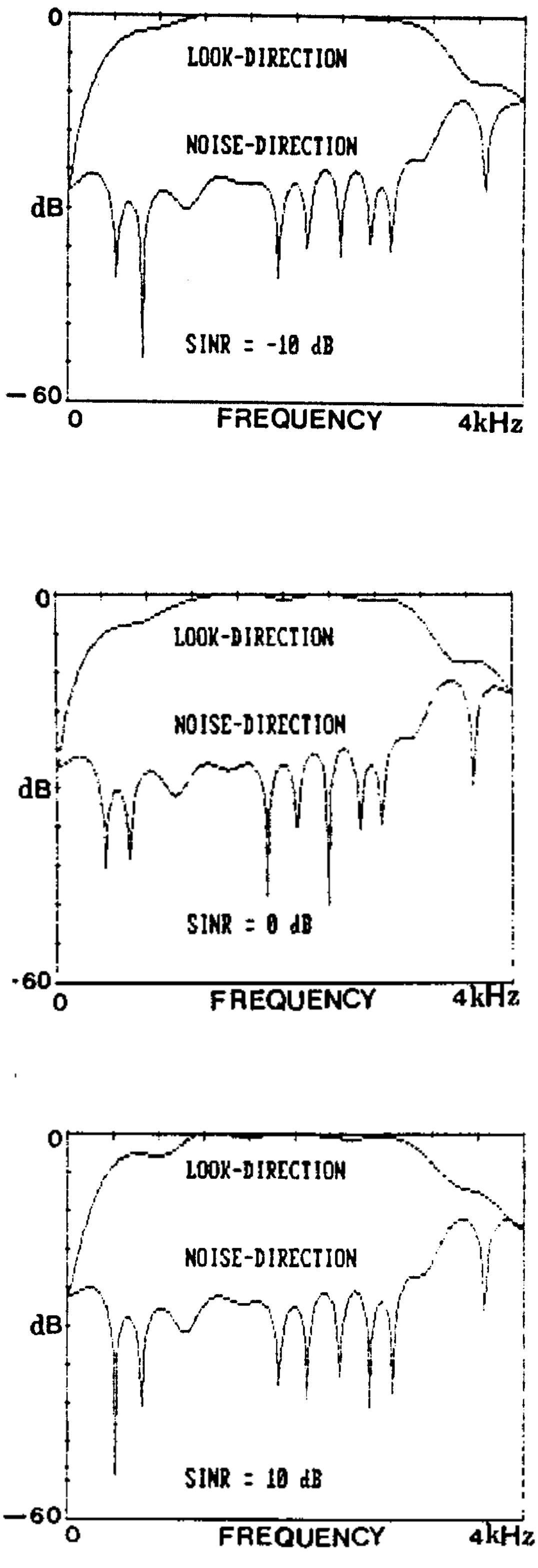

Fig. 4. Frequency responses of the FB AMNOR system
a) -10dB b) 0dB c) 10dB

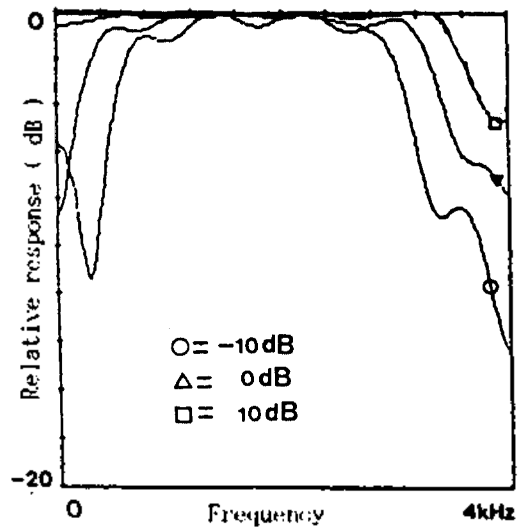

Fig. 5. Frequency response to the desired signal.

ationship between FD signal level A and response degradation. The monotonic relationship between FD signal level A and response degradation can be seen in these curve.

Fig. 7 shows the learning curve for the FB-AMNOR system. The horizontal axis shows time in terms of the number of adaptation steps and the vertical axis shows response degradation and output noise power relative to the input noise power.

They demonstrate that the interfering noise is successfully reduced. Other simulations indicate that the frequency response of the AMNOR system of the desired signal is almost the same as the case of the FB-AMNOR, but the FB-AMNOR system yields more natural sound and less frequency response to the interfering noise than the AMNOR system.

Ⅳ. Summary

FB AMNOR system with linear phase response to the look direction signal is presented. The impulse response function of the system is made symmetric by computing the forward

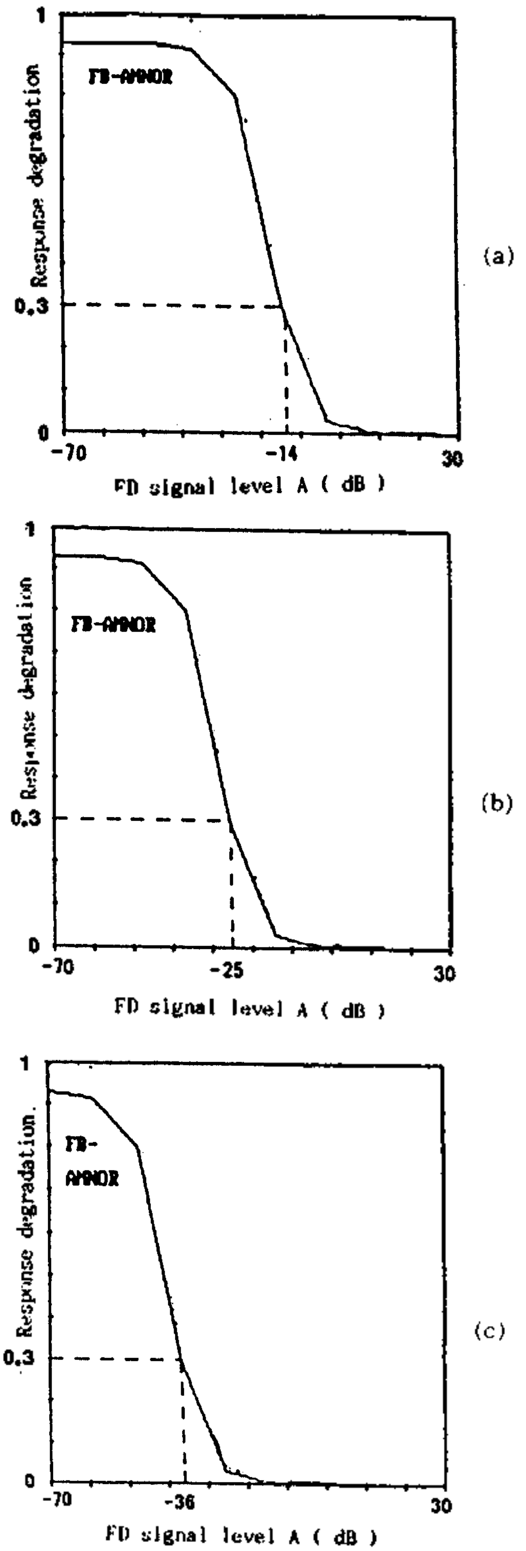

Fig. 6. Relationship between FD signal level A and response degradation.
a) SINR=−10dB b) SINR=0dB c) SINR=10dB

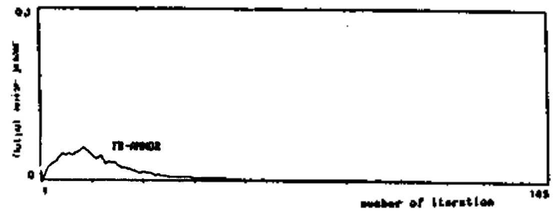

Fig. 7. Learning curve for the FB-AMNOR system.

and the backward coefficients minimizing the sum of the forward and the backward estimation error powers. The coefficients are updated using the LMS algorithm under the constraint that the backward coefficient vector is the same as the forward coefficient vector.

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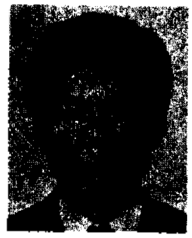
▲B.G. Iem

1988. 2 : Department of Electronic Engineering, Yonsei University (B.S)
1990. 2 : Department of Electronic Engineering, Yonsei University (M.S)

▲B.M. Cho

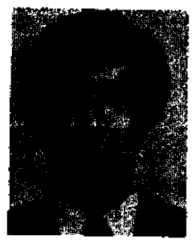


1982. 2 : Department of Electronic Engineering, Inha University (B.S)
1984. 8 : Department of Electronic Engineering, Yonsei University (M.S)
1985. 8～ : Department of Elctronic Engineering, Yonsei University Ph.D course

▲I.W. Cha

1959. 2 : Department of Electrical Engineering, Yonsei University (B.S)
1983. : Department of Electronic Engineering, Yonsei University (Ph.D)
1969～1970 : Institute of Sound and Vibration Research, University of Southampton, England
1959～1972 : Department of Electrical Engineering, Yonsei University
1981～1987 : The president of the Acoustical Society of Korea
1986. 9～1987. 7 : Department of Electrical and Computer Engineering, University of Iowa, Visiting Professor
1973～ : Department of Electronic Engineering, Yonsei University

▲D.H. Youn



1977. 2 : Department of Electronic Engineering, Yonsei University (B.S)
1979. 5 : Department of Electrical Engineering, Kansas State University (M.S)
1982. 5 : Department of Electrical Engineering, Kansas State University (Ph.D)
1982. 8～1985. 6 : Department of Electrical and Computer Engineering, University of Iowa
1985. 9～ : Department of Electronic Engineering, Yonsei University