

On Compensating Nonlinear Distortions of an OFDM System Using an Efficient Adaptive Predistorter

Hyun Woo Kang, Yong Soo Cho, and Dae Hee Youn

Abstract—This paper presents an efficient adaptive predistortion technique compensating for nonlinear distortions caused by a high-power amplifier (HPA) cascaded with a linear filter in an OFDM system. In the proposed approach, the memoryless HPA, preceded by a linear filter with memory in OFDM systems, is modeled by the Wiener system, which is then precompensated by the proposed adaptive predistorter with a minimum number of filter taps. It is confirmed by computer simulation that the proposed approach produces a faster convergence speed than the previous adaptive predistortion technique, and provides a small output backoff as low as 5.5 dB for an OFDM system employing an HPA with a linear filter.

Index Terms— Adaptive predistortion, nonlinear distortion, Wiener system.

I. INTRODUCTION

ORTHOGONAL frequency division multiplexing (OFDM) is currently under significant investigation due to a high spectral efficiency and immunity to multipath fading and impulse noise. OFDM is known as an effective technique for high bit rate applications, such as terrestrial television broadcasting [1], [2], where the delay spread of the channel extends over many symbol periods. Usage of an appropriate guard interval in OFDM can easily prevent intersymbol interference (ISI) and interchannel interference (ICI), whereas powerful equalization techniques are required for single carrier modulation. However, OFDM-based terrestrial broadcast systems are known to be significantly more sensitive to nonlinear distortion, usually caused by a high power amplifier (HPA), than are single carrier systems. Therefore, in order to reduce performance degradation in OFDM systems, compensation of nonlinear distortion is required.

Although several adaptive data predistortion techniques have been proposed by modifying the input constellation efficiently using the least hardware requirements (RAM and

look-up encoder) for single carrier systems [3], they cannot be applied to OFDM systems. This is because possible input levels for HPA in OFDM systems are infinite, whereas finite input levels exist in single carrier systems. Recently an analytical solution and an adaptive solution for predistorters to reduce nonlinear effects in OFDM systems were proposed [4]. However, in order to implement the adaptive predistortion technique in OFDM systems, a large amount of RAM is required, whose contents are updated with low convergence speeds.

Several prefiltering techniques for a memoryless nonlinear system, preceded by a linear system, have been reported [3], [5]. Some of these techniques, based on Volterra series modeling, are not practical for real-time implementation due to the high computational complexity for the large number of filter coefficients. The block-oriented model is another approach for modeling nonlinear systems without requiring large filter coefficients. This approach is based on the assumption that the nonlinear system consists of relatively simple subsystems, the structures of which are known. In particular, the class of systems which can be represented by a linear dynamic subsystem followed by a zero-memory nonlinear subsystem is called the Wiener system [6]. The Hammerstein system consists of the same subsystems connected in the reverse order.

In this paper, the memoryless HPA preceded by a linear filter in OFDM systems is modeled by the Wiener system. The linear and nonlinear distortions present in the Wiener system are then compensated for by an adaptive predistorter structured by the Hammerstein model. An adaptive algorithm for adjusting the parameters of the predistorter is then derived using a stochastic gradient method. Finally, it is demonstrated by computer simulation that the performance of an OFDM system suffering from nonlinear distortion can be greatly improved by the proposed efficient adaptive predistorter using a minimum number of filter taps.

II. AN ADAPTIVE NONLINEAR PREDISTORTER IN A OFDM SYSTEM

A simplified block diagram of the baseband-equivalent system for an OFDM transmission system is shown in Fig. 1. The serial-to-parallel block converts a QAM input data stream to a block of N symbols, which modulates the corresponding subcarrier. After the OFDM modulation is performed by the IFFT, the guard interval that is longer than the largest delay spread is inserted to remove the ISI and the ICI. The linear filter in Fig. 1 represents the transmitter pulse shaping filter usually placed at the baseband or at the IF stage. The

Paper approved by J. H. Winters, the Editor for Equalization of the IEEE Communications Society. Manuscript received October 29, 1996; revised November 4, 1997 and June 16, 1998. This work was supported by the KOSEF (Korea Science and Engineering Foundation) in 1995. This paper was presented in part at the IEEE International Symposium on Circuits and Systems, Hong Kong, June 1997.

H. W. Kang was with the Information and Telecommunication R&D Center, Hyundai Electronics Industries Company, Ltd., Kangnam-Ku, Seoul 135-280, Korea. He is now with the Department of Computer Science, Kangnam University, Yongin-City, Kyungki-Do 449-702, Korea.

Y. S. Cho is with the Department of Electronics Engineering, Chung-Ang University, Dongjak-Ku, Seoul 156-756, Korea.

D. H. Youn is with the Department of Electronics Engineering, Yonsei University, Sudaemoon-Ku, Seoul 120-749, Korea.

Publisher Item Identifier S 0090-6778(99)03319-X.

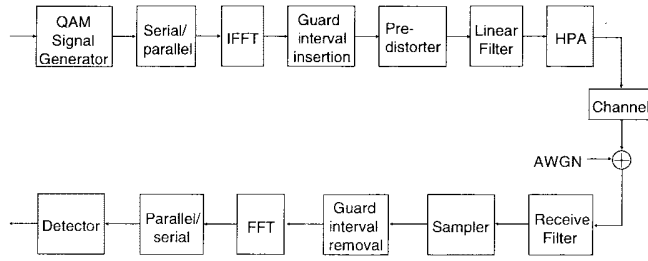


Fig. 1. A simplified block diagram of the baseband-equivalent system for an OFDM transmission system.

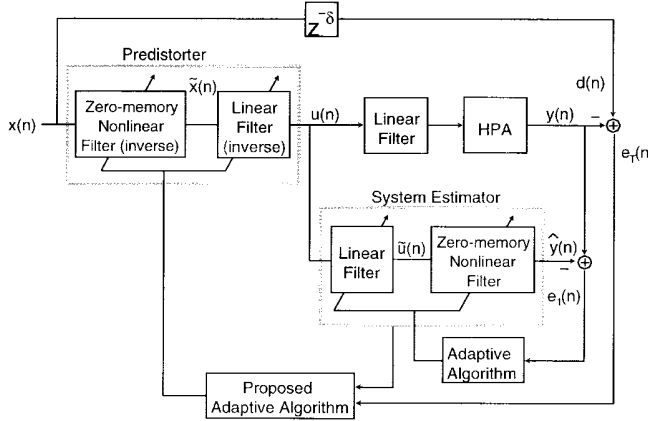


Fig. 2. An adaptive predistorter for the HPA preceded by a linear filter.

HPA widely used for high capacity satellite transmission is traveling-wave tube (TWT). The TWT, which is known to have a frequency-independent memoryless nonlinearity, can be completely characterized by its AM/AM conversion and AM/PM conversion [7].

From Fig. 1, one can see that a TWT preceded by a linear filter can be well modeled by the Wiener system, which consists of a memoryless nonlinear subsystem preceded by a linear dynamic subsystem [6]. The block diagram of a proposed adaptive predistorter, which can compensate the Wiener system, is shown in Fig. 2. The proposed scheme is composed of a system estimator that estimates the parameters of the Wiener system using an adaptive algorithm, and an adaptive predistorter that compensates for the constellation warping and reduces the ISI effects using a proposed adaptive algorithm. Samples of OFDM signal, $x(n)$, at the input of the predistorter is given by

$$x(n) = \sum_{i_2=0}^{N-1} X(i_1, i_2) e^{j2\pi i_2 n/N}, \quad n = -N_G, -N_G + 1, \dots, -1, 0, 1, \dots, N-1 \quad (1)$$

where $X(i_1, i_2)$ is a QAM symbol, i_2 refers to subcarrier number, i_1 is time index, N is the length of IFFT, and N_G is the length of guard interval. In contrast to the single carrier system where the input levels of HPA are finite, the OFDM signal, $x(n)$, is a complex Gaussian random process with a zero mean. The probability density functions of the amplitude and phase of $x(n)$ are Rayleigh-distributed and uniformly distributed in the interval $[-\pi, \pi]$, respectively.

If the TWT can be approximated by a polynomial form of finite order, the input and output relationship of the system estimator is given by

$$\hat{y}(n) = \sum_{l=1}^{N_a} a_l \left(\sum_{k=1}^{N_h} h_k u(n-k) \right)^l \quad (2)$$

where N_h and N_a , respectively, denote the memory length of the linear filter h_k , and the order of the nonlinear filter a_l . Also, $u(n)$ denotes the input signal of the linear filter. We assume that all coefficients used here are complex. Using only the input and output signals of the system, the coefficients of the system estimator, h_k and a_l , are adjusted to minimize the mean square error, $E\{|e_1(n)|^2\}$ between $y(n)$ and $\hat{y}(n)$.

After the correct parameters of the Wiener system have been estimated, an adaptive predistorter, which is ideally the inverse of the Wiener system, can be designed. An appropriate structure of the predistorter is easily shown to be the Hammerstein model [8]. The predistorter in Fig. 2 is constructed by a memoryless nonlinear inverse filter followed by a linear inverse filter. By using a polynomial form of finite order as the memoryless nonlinear inverse filter, the predistorter can be expressed as

$$u(n) = \sum_{i=1}^{N_p} p_i \sum_{j=1}^{N_s} s_j x^j(n-i) \quad (3)$$

where N_p denotes the memory length of the linear inverse filter p_i , and N_s denotes the order of the nonlinear inverse filter s_j . The error of the total system is defined by

$$e_T(n) = d(n) - y(n) \quad (4)$$

where the desired signal $d(n)$ is the delayed version of the input signal $x(n)$ by δ samples to account for causality of the predistorter. The coefficients of the predistorter are obtained by minimizing the mean square error, $E\{|e_T(n)|^2\}$, of the total system. An adaptive algorithm for updating the coefficients of the predistorter is given by applying the stochastic gradient method as follows:

$$p_m(n+1) = p_m(n) - \alpha_p \hat{\nabla}_{p_m}(n), \quad m = 1, 2, \dots, N_p \quad (5)$$

$$s_m(n+1) = s_m(n) - \alpha_s \hat{\nabla}_{s_m}(n), \quad m = 1, 2, \dots, N_s \quad (6)$$

where α_p and α_s , respectively, represent the step-size constants of the linear inverse filter p_m and the nonlinear inverse filter s_m . The step-size constant controls stability and the convergence speed of the algorithm. The term $\hat{\nabla}_{p_m}(n)$ in (5) represents the instantaneous estimate of the gradient of $E\{|e_T(n)|^2\}$ with respect to coefficients of the linear inverse filter p_m defined by

$$\hat{\nabla}_{p_m}(n) \equiv \frac{\partial |e_T(n)|^2}{\partial p_m(n)} \quad (7)$$

$$= -e_T^*(n) \frac{\partial y(n)}{\partial p_m(n)}. \quad (8)$$

After the coefficients of the system estimator have converged to the correct values for the Wiener system, the output of the system estimator $\hat{y}(n)$ will approximate the output of the

Wiener system, $y(n)$ in (8). Since $\hat{y}(n)$ is a function of input signals, $u(n-1), u(n-2), \dots, u(n-N_h)$, (8) can be rewritten as

$$\hat{\nabla}_{p_m}(n) = -e_T^*(n) \sum_{r=1}^{N_h} \frac{\partial \hat{y}(n)}{\partial u(n-r)} \frac{\partial u(n-r)}{\partial p_m(n)}. \quad (9)$$

For simpler notation, we define the derivative of the Wiener system as

$$g(r; n) \equiv \frac{\partial \hat{y}(n)}{\partial u(n-r)}. \quad (10)$$

The above equation can be rewritten by substituting (2) into (10) as

$$g(r; n) = \sum_{l=1}^{N_a} l a_l h_r \left(\sum_{k=1}^{N_h} h_k u(n-k) \right)^{l-1}. \quad (11)$$

The second part of the summation term in (9), i.e., the derivative of the predistorter output with respect to p_i , is defined as

$$b_m(r; n) \equiv \frac{\partial u(n-r)}{\partial p_m(n)}. \quad (12)$$

Then, (12) can be rewritten by substituting (3) into (12) as

$$b_m(r; n) = \sum_{j=1}^{N_s} s_j x^j(n-r-m). \quad (13)$$

Finally, the adaptive algorithm for the linear inverse part of the predistorter can be obtained by substituting (9), (11), and (13) into (5) as follows:

$$\begin{aligned} p_m(n+1) &= p_m(n) + \alpha_p e_T^*(n) \sum_{r=1}^{N_h} \sum_{l=1}^{N_a} l a_l h_r \\ &\times \left(\sum_{k=1}^{N_h} h_k u(n-k) \right)^{l-1} \sum_{j=1}^{N_s} s_j x^j(n-r-m). \end{aligned} \quad (14)$$

The adaptive algorithm for the nonlinear inverse part of the predistorter can be derived in a similar manner as follows:

$$\begin{aligned} s_m(n+1) &= s_m(n) + \alpha_s e_T^*(n) \sum_{r=1}^{N_h} \sum_{l=1}^{N_a} l a_l h_r \\ &\times \left(\sum_{k=1}^{N_h} h_k u(n-k) \right)^{l-1} \sum_{i=1}^{N_p} p_i x^m(n-r-i). \end{aligned} \quad (15)$$

Note that in order to obtain the coefficients of the nonlinear inverse filter s_m in the predistorter, both the coefficients of the linear inverse filter p_m of the predistorter, and the estimates h_k and a_l of the system estimator, are required. Consequently, (14) and (15) constitute the adaptive algorithm for the predistorter.

III. COMPUTER SIMULATIONS

In this section, a computer simulation is used to demonstrate the validity of the proposed predistortion technique for compensation of the TWT preceded by a linear filter in an OFDM system using the block-oriented model. The serial-to-parallel converter transfers a block of 1024 16-QAM symbols to the OFDM modulator, which uses an 800-point IFFT to modulate them on to the subchannels. The first and the last 112 subcarriers of the block are set to zero to avoid spectrum overlapping. For close examination of the effectiveness of the predistorter in compensating the nonlinear distortion, it was assumed that the multipath fading effect is completely compensated, resulting in an additive white Gaussian noise (AWGN) channel. The TWT model given in [7] was used for HPA, and the linear filter is approximated by the FIR filter whose coefficients are given by [5].

The parameters of the Wiener system, the TWT preceded by the linear filter, were first estimated by the system estimator. The order of the nonlinear filter N_a in the system estimator was set to five in order to model only significant nonlinear distortions of TWT with a minimal number of parameters. The memory length of the linear filter N_h was set to three, the same number as in [5]. Fig. 3(a) shows the learning curve of the system estimator, obtained by averaging 200 independent trials. The linear filter h_k , and the nonlinear filter a_l , of the system estimator were updated simultaneously using the stochastic gradient method. The step-size constants, which determine convergence speed and MSE after convergence, were set to 0.0067 for the linear filter and 0.27 for the nonlinear filter. With zero initial condition, about 45 dB in the MSE was obtained, resulting in a fairly accurate estimation of filter coefficients for the Wiener system under test.

After having checked the convergence of the MSE in the system estimator, the adaptive algorithm described in Section II was applied to obtain filter coefficients of the predistorter. The order of the nonlinear inverse filter N_s and the memory length of the linear inverse filter N_p in the predistorter were set to numbers large enough to model inverse functions, seven and ten, respectively. Fig. 3(b) shows the learning curve of the predistorter obtained by averaging 200 independent trials. The step-size constants for the nonlinear inverse filter and linear inverse filter were set to 0.5 and 0.1, respectively. The consequent convergence of the learning curve to 45 dB signifies that the proposed adaptive predistorter is effective in compensating the linear and nonlinear distortion present in the Wiener system even with a small number of filter taps. Note that the existence of two different slopes of the learning curves in Fig. 3 is due to the different convergence characteristics of the linear filter and the nonlinear filter. Also shown in Fig. 3(b) are the results obtained by using the previous adaptive predistortion technique [4]. The input amplitude, output amplitude, and phase-shift are all quantized over 6 bits. The required memory used for this approach is 64 words of 12 bits. By comparing these two results, one can see that the proposed approach has faster convergence speed and small fluctuations after convergence.

A typical performance measure for quantifying the influence of nonlinear distortion in HPA is total degradation (TD_{dB}).

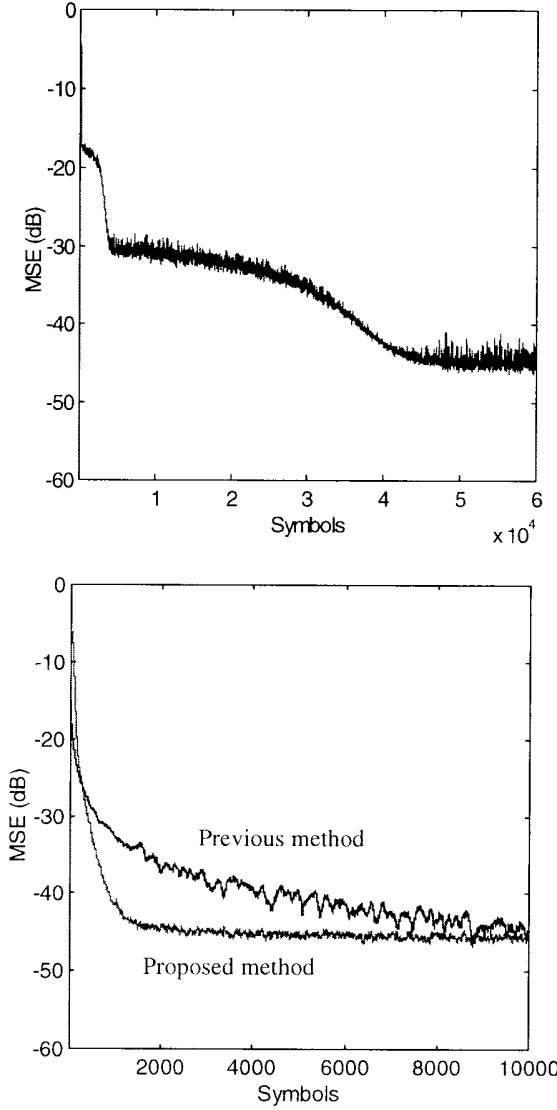


Fig. 3. Learning curve (a) system estimator ($N_h = 3, N_a = 5$) and (b) predistorter ($N_s = 7, N_p = 10$; output backoff ≈ 6.0 dB).

The TD_{dB} is defined as follows [4]:

$$TD_{dB} = B_{dB} + \Delta_{dB}(SNR) \quad (16)$$

where B_{dB} represents the TWT output backoff, which is defined as the ratio of saturation power to the average transmitted signal power, and where $\Delta_{dB}(SNR)$ represents the SNR degradation due to nonlinearity of TWT, at the BER of 10^{-4} for our uncoded system. In Fig. 4, TD_{dB} is shown for different values of the output backoff, which provides the optimum TWT backoff. In Fig. 4, the solid line represents the case where the proposed predistorter is used, whereas the dashed line represents the case without any predistorter. Note that, when the proposed predistortion technique is applied, the optimum output backoff can be reduced to as much as 5.5 dB. Fig. 5 shows the BER comparison of the proposed approach for the case without any predistortion when the output backoff was set to 6.0 dB. The effect of quantization on the BER performance is also shown in Fig. 5.

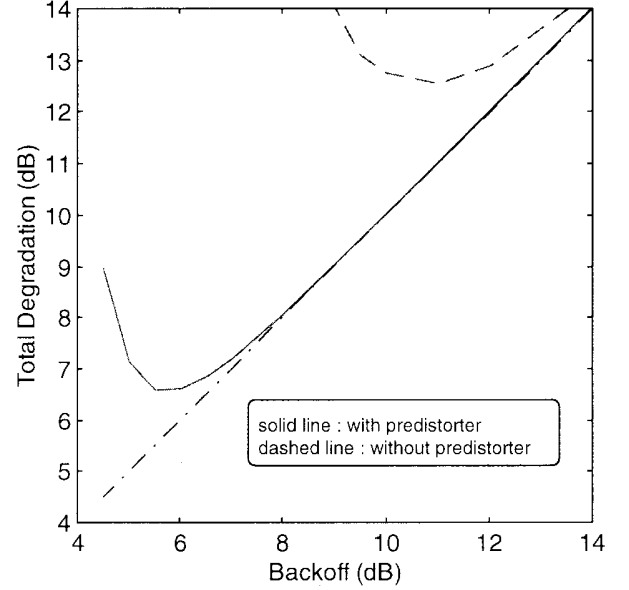


Fig. 4. Total degradation versus TWT output backoff (BER = 10^{-4}).

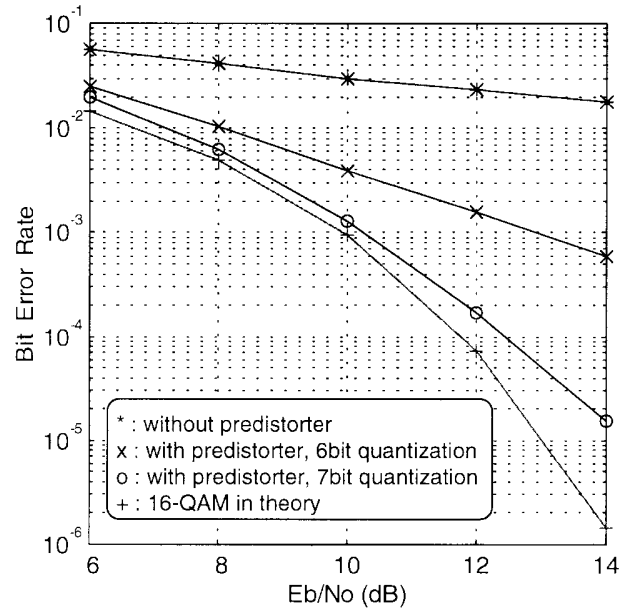


Fig. 5. Bit error rate versus E_b/N_0 (output backoff ≈ 6.0 dB).

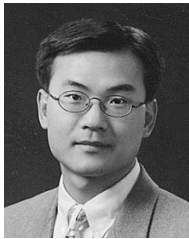
IV. CONCLUSION

The OFDM scheme is known to be significantly more sensitive to nonlinear distortion than the single carrier scheme since the peak amplitude of OFDM signals is more important than the peak amplitude of single carrier signals. In this paper, an efficient adaptive predistortion technique is proposed which can compensate for the distortion caused by an HPA with a linear filter, in an OFDM system using a minimum number of filter taps. From simulation results, it is confirmed that the proposed adaptive predistorter is very effective in reducing the linear and nonlinear distortion of an HPA with a linear filter in OFDM systems, allowing small output backoff as low as 5.5 dB. Stability analyses of the system estimator and the

predistorter in the proposed adaptive predistorter need to be performed in the future.

REFERENCES

- [1] H. Sari, G. Karam, and I. Jeanclaude, "Transmission techniques for digital terrestrial TV broadcasting," *IEEE Commun. Mag.*, vol. 33, pp. 100–109, Feb. 1995.
- [2] Y. Wu and B. Caron, "Digital television terrestrial broadcasting," *IEEE Commun. Mag.*, vol. 32, pp. 46–52, May 1994.
- [3] G. Karam and H. Sari, "A data predistortion technique with memory for QAM radio systems," *IEEE Trans. Commun.*, vol. 39, pp. 336–344, Feb. 1991.
- [4] A. Brajal and A. Chouly, "Compensation of nonlinear distortions for orthogonal multicarrier schemes using predistortion," in *Proc. GLOBECOM*, San Francisco, CA, Nov. 1994, vol. 3, pp. 1909–1914.
- [5] C. S. Eun and E. J. Powers, "A predistorter design for a memoryless nonlinearity preceded by a dynamic linear system," in *Proc. GLOBECOM*, 1995, pp. 152–156.
- [6] W. Greblicki, "Nonparametric identification of wiener systems," *IEEE Trans. Inform. Theory*, vol. 38, pp. 1487–1493, Sept. 1992.
- [7] A. A. M. Saleh, "Frequency-independent and frequency-dependent nonlinear models of TWT amplifiers," *IEEE Trans. Commun.*, vol. COM-29, pp. 1715–1720, Nov. 1981.
- [8] S. Y. Mao and P. X. Lin, "A test of nonlinear autoregressive models," in *Proc. Int. Conf. Acoustics, Speech, and Signal Processing*, New York, 1988, pp. 2276–2279.



Hyun Woo Kang was born in Jeonju City, Korea, in 1969. He received the B.S., M.S., and Ph.D. degrees in electronics engineering from the Yonsei University, Seoul, Korea, in 1991, 1993, and 1997, respectively.

From 1997 to 1998, he was a Research Engineer at Hyundai Electronics Company, Ltd., Seoul, Korea. Since 1999, he has been with the Department of Computer Science, Kangnam University, Kyungki-Do, Korea. His research interests include adaptive signal processing, audio and speech processing, and

applications of signal processing in multimedia communication systems.



Yong Soo Cho received the B.S. degree in electronics engineering from Chung-Ang University, Seoul, Korea, the M.S. degree in electronics engineering from Yonsei University, Seoul, Korea, and the Ph.D. degree in electrical and computer engineering from the University of Texas at Austin, in 1984, 1987, and 1991, respectively.

During 1984, he was a Research Engineer at Goldstar Electrical Company, Osan, Korea. In 1992, he joined Chung-Ang University, where he is presently an Associate Professor of the School of Electrical and Electronic Engineering. His research interests are in the development of digital signal processing techniques for use in digital communication area.

Dae Hee Youn was born in Chungju City, Korea, in 1951. He received the B.S. degree in electronic engineering from Yonsei University, Seoul, Korea, in 1977 and the M.S. and Ph.D. degrees from Kansas State University, Manhattan, in 1979 and 1982, respectively.

From 1979 to 1982, he served as a Research Associate at Kansas State University. From 1982 to 1985, he served as an Assistant Professor in the Department of Electrical and Computer Engineering, University of Iowa, Iowa City. Since 1985, he has been with the Department of Electronic Engineering, Yonsei University. His research interests include audio and speech signal processing and adaptive digital filters.