THE JOURNAL OF THE ACOUSTICAL SOCIETY OF KOREA Vol.21, No.1E 2002. 3. pp. 18~24

Vector Channel Simulator Design for Underwater Acoustic-based Communications

Duk-Yung Kim*, Yong-Deak Kim*, Yong-Kon Lim**

*Dept. of Electronics Engineering, Ajou Univ.
**Korea Research Institute of Ships and Ocean Engineering, KORDI
(Received July 19 2001; accepted September 7 2001)

Abstract

This paper discusses the development of an acoustic vector channel simulator for the performance analysis of an acoustic digital communication system. The channel simulator consists of transmission module, acoustic channel model, receiver, beamformer, and adaptive equalizer. The source signal (QPSK) is generated by the specified parameters. The transmitted signal generates multipath signals which have a different delay, amplitude and doppler frequency. The paper presents in details the approach to the performance analysis of an acoustic digital communication system according to the antenna structure and the various baseband signal processing techniques.

Keywords: *Multipath, Nonhomogeneity, Doppler effect, Beamformer, Vector channel*

I. Introduction

The underwater acoustic telemetry channel is influenced by the propagation velocity, multipaths, and nonhomogeneity[2,5]. Therefore the amplitude, phase, and frequency of the received signal is severely influenced by the time-varying aspects of the underwater acoustic channel. There are many popular channel models about the commercial radio communication systems such as PCS or CDMA cellular system, but there is no modest model in the case of underwater acoustic channel because of the above characteristics of it. In the advanced nations with WHOI as leader, they only measured the channel characteristics through the real field test and extracted the channel parameters which are the radio communication channel parameters[1]. The underwater acoustic channel characteristics like this naturally derived the underwater acoustic channel simulations.

Corresponding author: Duk-Yung Kim (dykim@lginnotek.com)
Ajou University, Suwon 442-749, Korea

In the beginning, the underwater acoustic digital communication system used noncoherent FSK (Frequency Shift Keying) or DPSK (Differential Phase Shift Keying) modulation techniques. These modulation method are highly reliable and modem is simpler than PSK modulation method. But their bandwidth efficiency is poor, thus they are inappropriate for a high speed data communication systems such as image transmission. Since Stojanovic presents the signal processing techniques based on coherent PSK modulation method[1], many researchers have begun to study the various coherent modulation techniques which are applicable to the underwater acoustic communication systems. Because the field test of the acoustic-based underwater communication system costs very much, channel simulators have been developed in attempt to increase the success of field experiments.

The purpose of them is primarily on the performance analysis of the various acoustic modem signal processing algorithms. Another field of the channel simulations is the closest generation of the real time-varying ocean environment. The telemetry community has focused almost exclusively on ray theory but it is inadequate, yet[2]. In this paper, we present a vector channel simulator for the acoustic-based underwater communications.

This paper organized as follows. In section II we describe the constituent elements, functions and aims of the simulator. Section III presents the structure of the QPSK modulator. In section IV we turn to underwater acoustic channel modeling in detail. Section V presents receiver structures and baseband signal processing techniques such as beamforming and equalization. In Section VI we describe the channel simulator GUI. Conclusions are finally summarized in Section VII.

II. Simulator Summary

Figure 1 is the block diagram of the designed simulator. It consists of transmission module, channel model and receiver. Transmitter generates QPSK modulated source signal which has 4 khz bandwidth with 32 khz center frequency. Channel is modeled by attenuation, multipath, doppler effect and fading which are originated from the acoustic wave propagation in the ocean environments. We adopted array antenna receiver structure. We have used beamforming and equalization as the baseband signal processing techniques. The GUI for the simulator assigns a function to simulation environment parameters and displays results such as beampattern, signal constellation and SNR (Signal to Noise Ratio).

The simulator parameters are specified as follows.

1. Array antenna structure: ULA (Uniform Linear Array), planar, circular and hexagon.
2. No. of array elements (It is fixed in hexagon structure).
3. Array antenna spacing (It is fixed for a half of the wavelength in ULA and planar structure).
4. Depth of the seabed
5. Depth of the Tx.
6. Range between the Rx and Tx.
7. No. of multipath.
8. Additive noise power
9. Velocity of the Tx.
10. No. of the frames
11. No. of taps and parameters in adaptive equalizer.

Also, the following parameters are fixed.

1. Carrier frequency = 32 kHz
2. Signal bandwidth = 4 kHz
3. Modulation method : QPSK

III. Transmitter: QPSK Modulator

We have adopted QPSK modulation method. Among the family of M-ary PSK signals, QPSK offers the best trade-off between power and bandwidth requirements. Figure 2 is the QPSK modulator structure.

Raw data is compressed and coded with FEC technique such as convolutional code. In Figure 2 the data is the processed raw data. It is periodically multiplexed with the given training sequences in the MUX, where training

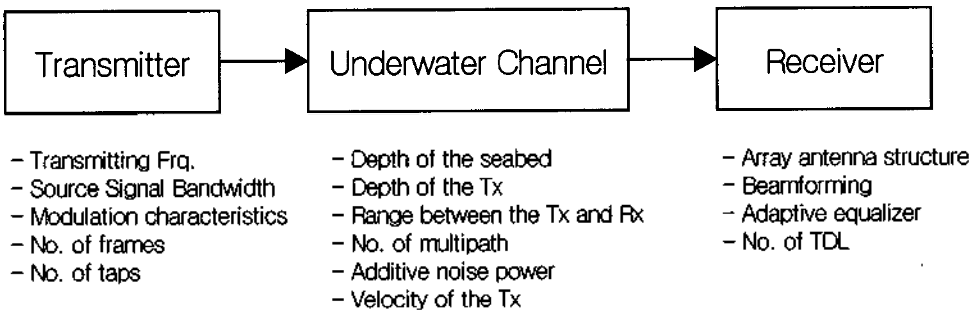

Figure 1. Block diagram of the designed simulator.

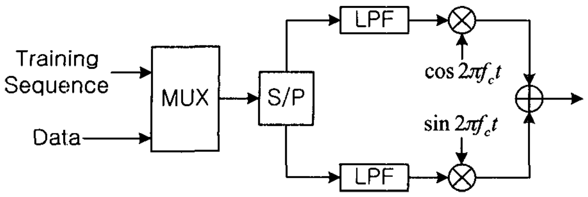

Figure 2. QPSK modulator structure.

Traning Sequence	Data	Traning Sequence	Data	

Figure 3. Time multiplexed data frame structure.

sequence is known to receiver. It is used for DOA (Direction Of Arrival) estimation and channel estimation needed for adaptive equalizer and beamformer. Figure 3 is the time multiplexed data frame structrue.

We have adopted the simplest data frame structure since it is adequate for algorithm performance analysis. Time multiplexed data is separated into in-phase (I) channel signal and quadrature-phase (Q) channel signal by the serial-to-parallel converter. They are filtered and multiplied by $\cos 2\pi f_c t$ and $\sin 2\pi f_c t$, respectively. The transmitted Signal can be formulated as follows.

$$s(t) = b_{ILPF}(t)\cos(2\pi f_c t) + b_{QLPF}(t)\sin(2\pi f_c t)$$

, where f_c is the carrier frequency and $b_{ILPF}(t)$, $b_{QLPF}(t)$ are the LPF output of the I-channel and Q-channel, respectively.

Figure 4 is the frequency spectrum of the transmitted signal. It illustrates QPSK modulated source signal with carrier frequency 32 kHz and bandwidth 4 kHz.

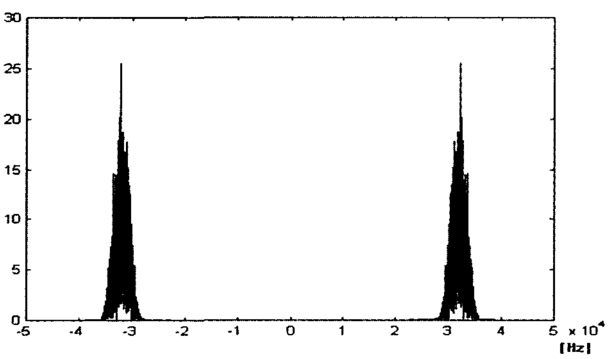

Figure 4. Frequency spectrum of the transmitted signal.

IV. Acoustic Channel Modeling

Path loss is basically caused by geometric dispersion and absorption in the ocean. It can be considered as follows[3].

$$\rho = 20\log r + \alpha(f) r$$

, where $\alpha(f)$ is the acoustic absorption coefficient and r is the range between the transmitter and receiver.

Underwater transmitted signal is reflected in the sea surface and the seabed and so forth the receiver also collects the reflected signals. This phenomenon is termed multipath[5]. Underwater acoustic channel model with multipaths and path loss can be described by

$$h(t) = \sum_{l=0}^{L} \rho_l \delta(t-\tau_l)$$

, where $h(t)$ is the channel impulse response, L is the number of multipaths and ρ_l and τ_l is the path loss and time delay of the l th multipath, respectively.

The seawater movement and the relative velocity between the receiver and transmitter cause the dopper effects. The channel model with doppler effects can be formulated as follows.

$$h(t) = \sum_{l=0}^{L} \rho_l \delta(t-\tau_l)\, e^{-j2\pi\Delta f t}$$

, where Δf is the doppler spread and can be described by

$$\Delta f = \frac{v}{c}\cos\theta$$

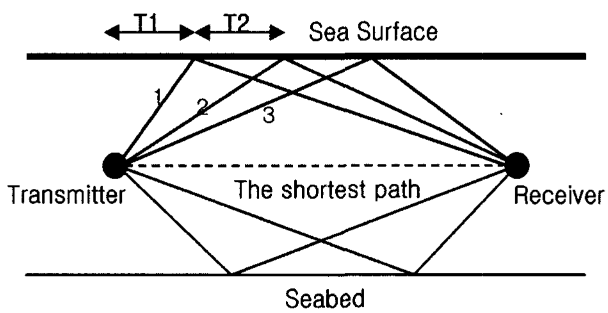

Figure 5. Multipath diagram.

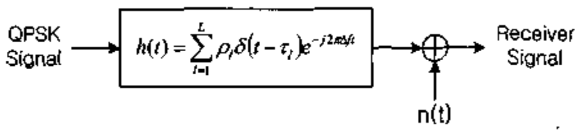

Figure 6. Final channel model (Underwater acoustic propagation model).

, where θ and v is the angle and relative speed between the transmitter and receiver, respectively and is the speed of sound in the ocean. We don' t consider the seawater speed in the doppler effects. We fixed $c = 1500m/s$, where we ignore salinity and temperature difference caused by the sea depth profile.

Figure 6 is the final channel model used in the simulator, where $n(t)$ is the AWGN (Additive White Gaussian Noise). Receiver velocity vector is inputed, then channel parameters such as path loss, time delay, doppler frequency and multipath are varied according to the receiver movement. They are updated each frame.

V. Receiver Structure

We utilize array antenna structure in receiving the propagated signal through the channel model. Received signal vector is determined by the array sensor structure. We modeled the received signal vector using various array structure such as ULA, circular array, planar array, hexagon array[6,9,10]. Hexagon structure can be interpreted as the sum of circular structures. It is illustrated in Figure 7.

We take quadrature receiver structure as following Figure 8 The receiver output is the baseband signal which has no carrier frequency components. The trasmitted signal can be recovered from the receiver output by the various signal processing techniques which compensate the channel distortions.

Underwater acoustic channel is characterized by the nature of the seawater which is the medium and the characteristics of the floor and surface of the sea. The channel parameter is explained well in section IV. They are varied very fast and their standard deviation is very

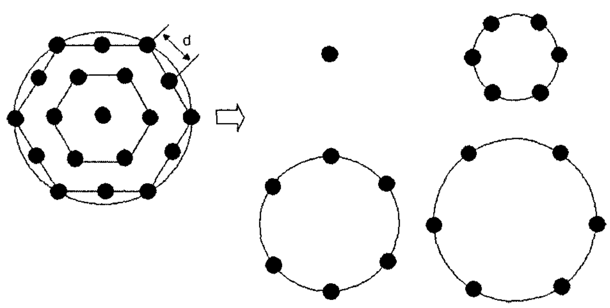

Figure 7. Analysis of hexagon structure.

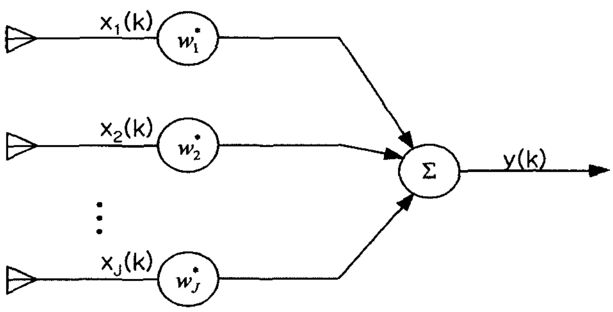

Figure 9. Narrowband beamformer.

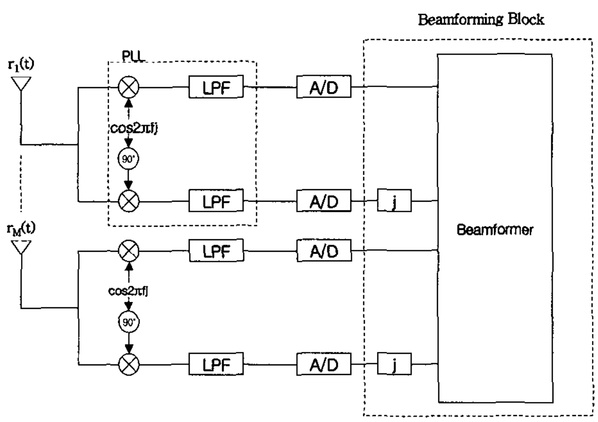

Figure 8. Block diagram of quadrature receiver.

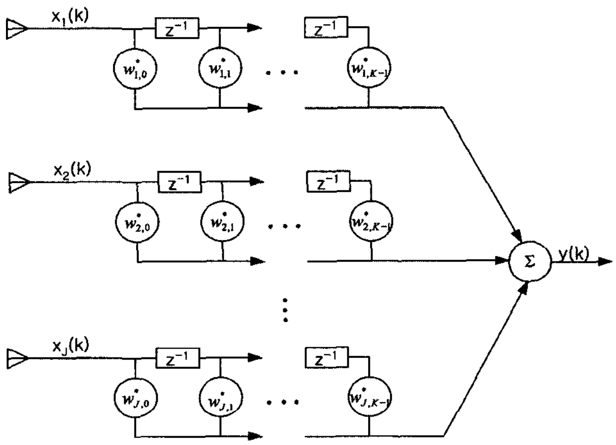

Figure 10. Wideband beamformer.

large in comparison with air. Therefore the compenstation methods for channel distortions are inevitably needed in underwater acoustic communications. We adopt beamforming and equalization as the baseband signal processing techniques. The designed simulator can analyze and compare the above methods according to the weight updating algorithms.

Beamforming is the spartial diversity technique which use array antenna[6,10]. Since we have employed wideband signal, we must take wideband beamforming structure[6-10]. Figure 9 and Figure 10 shows the narrowband beamformer and wideband beamformer, respectively.

Narrowband beamformer is simple. It samples the propagating wave in space. In Figure 9, sensor outputs are each multiplied by a complex weight and then summed. Wideband beamformer is more complex. In Figure 10, it samples the propagating wave in both space and time. Wideband beamformer has a FIR filter in each sensor. It determines a frequency characteristics of the wideband beamformer. How to determine the optimal number of taps per sensor for a given bandwidth is well described in[7,8], but we must verify the validity of it in ocean environments. Since underwater acoustic channel is time-varying, we must use adaptive weight updating algorithm in each frame. We have to designe the signal frame such that channel characteristics remain constant in each frame. Then we can make use of adaptive weight update algorithms which need the training sequence. It is the most improtant problem for underwater communication system design. There are many adaptive beamforming algorithms such as LMS (Least Mean Squres), RLS (Recursive Least Squares), fast RLS and etc., Their characteristics are well known [4,10].

Adaptive Equalizer estimates channel using the known training sequence and implements the inverse transfer function of the channel, so it compensates channel distortions [4]. Adaptive equalizer algorithms are similar to adaptive beamforming.

When we use both adaptive equalizer and beamformer, there is one important fact. Since we adopted wideband beamformer structure and it possessed a FIR filter in each sensor, a tap addition of it increases the total number of taps as much as the number of array sensors. In the case of equalizer, a tap addition of it increases the total number of taps as much since the input to the equalizer is scalar. Therefore we must consider the trade-off between the system complexity and the tap numbers.

VI. Channel Simulator GUI

Figure 11 shows the underwater acoustic vector channel simulator GUI. It is designed such that the position of the both transmitter and receiver, channel model parameters

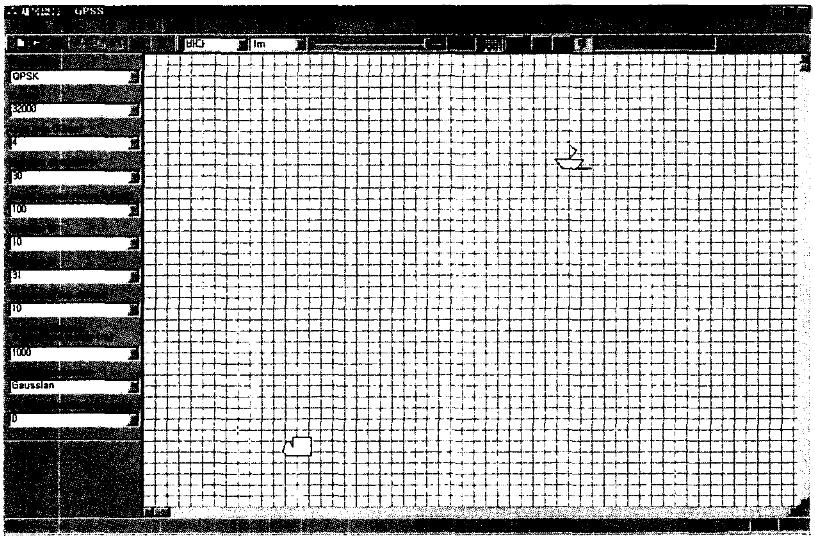

Figure 11. GUI for the underwater acoustic vector channel simulator.

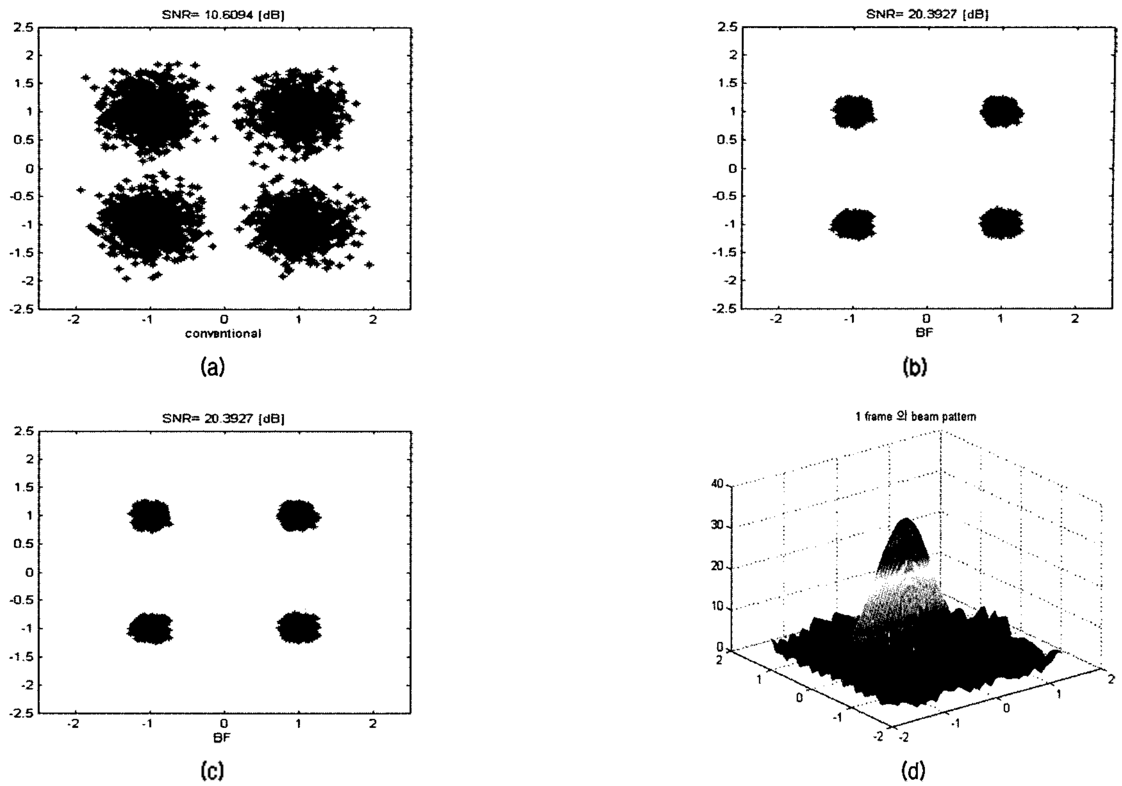

Figure 12. Simulator output.

and various algorithm parameters can be inputed arbitrarily. GUI is coded with C++. Signal processing algorithms are implemented with Matlab and linked to the GUI.

Figure 12 shows the simulator outputs. Figure 12 (a) shows the output signal constellation and SNR with single antenna. In Figure 12 (b), LCMV beamforming algorithm (without equalizer) is used in Figure 12 (b).We have used the Self-optimized adaptive equalizer in Figure 12 (c). Figure 12 (d) shows the beampattern of the hexagon array antenna.

VII. Conclusions

In this paper we have designed underwater acoustic vector channel simulator for the performance analysis of an acoustic digital communication system. It consists of transmitter, channel model and receiver. According to the specified input parameters QPSK modulated signal is generated. It is transmitted through the channel which is modeled by doppler effect, multipath, path loss, fading and AWGN. We adopted quadrature receiver structure with array sensors. It is designed for the system performance analysis according to the array structure and various signal processing algorithms.

We comment on the future works as follows. It is necessary underwater acoustic channel model which is considered environmental and oceanic effects such as acoustic ray propagation although we have designed simplified underwater acoustic channel which is modeled by doppler and multipath effects. And also it is needed to carry out a performance analysis of various beamforming and adaptive equalizer algorithms in order to optimize a system performance.

References

1. M. Stojanovic, J. Catipovic, and J. Proakis, "Phase-coherent digital communications for underwater acoustic channels," *IEEE J. Oceanic Eng.*, vol. **19**, pp. 100-111, Jan. 1994.
2. D. Kilfoyle and A. Baggeroer, "The state of the Art in Underwater Acoustic Telemetry," *IEEE J. Oceanic Eng.*,

vol. 25, pp. 4-27, Jan. 2000.
3. W. Burdic, *Underwater Acoustic System Analysis*, 2nd Edition, Prentice Hall, 1991.
4. J. G. Proakis, *Digital Communications*, McGraw-Hill, 1983.
5. Y. K. Lim, J. W. Park, "A Study of Multipath Analysis of the Acoustic-based Underwater Digital Communication System for run AUV," *KISTI*, May. 1997.
6. J. Litva and T. Lo, *Digital Deamforming in Wireless Communications*, Artech House, 1996.
7. F. Vook and R. Compton, "Bandwidth performance of linear adaptive arrays with tapped delay-line Processing," *IEEE Transactions on Aerospace and Electronic Systems*, vol. 28, no. 3, pp. 901-908, July 1992.
8. W. Rodgers and R. Compton, "Adaptive array bandwidth with tapped delay-line processing," *IEEE Transactions on Aerospace and Electronic Systems*, vol. 15, no. 1, pp. 21-27, Jan. 1979.
9. L. Godara, "Application of antenna arrays to mobile communications, Part II: Beam-forming and direction-of-arrival considerations," *Proceedings of the IEEE*, vol. 85, no. 8, pp. 1195-1245, Aug. 1997.
10. B. Van Veen and K. Buckley, "Beamforming: A versatile approach to spatial filtering," *IEEE ASSP Magazine*, pp. 4-24, April 1988.

[Profile]

• Duk-Yung Kim

He received the B.S. degree in Telecommunication Engineering from Kwangwoon University, Seoul, in 1982, and the M.S. degree in Electronics Engineering from Ajou University, Suwon, in 1994. He is currently a graduate student for Ph. D. degree in Electronics Engineering Ajou University. His current research interests are Digital Communication and Underwater Acoustic-based Communication.

• Yong-Kon Lim

He received the B.S. and M.S. degrees in Electrical Engineering in 1979 and 1984 respectively, both from Chungnam national University, Yusung, Korea, and the Ph. D. degree from Ajou University, Suwon, Korea, in 1994. From July 1980 to March 1999 he worked as a principal Researcher in KIMM, and from April 1990 to now he is working as a principal Researcher in KRISO. He is a manager of the Korean Institute of Maritime Information and Communication Sciences.
His current reseach interests are DSP & Microprocecessor System and Integrated Information Networking of Research Vessel.

• Yong-Deak Kim

He received the B.S. and M.S. and Ph. D. from Department of Electronics Engineering at Yonsei University, in 1971, 1973 to 1976 respectively. From member at Department of Electronics Ecole Superieur D'electricite in France. From 1979 to 1980 , he was a visiting professor of Department of Electronics Engineering at Stanford University. Since 1974, he has been a professor of Department of Electronics Engineering at Ajou University., Korea. His current research interests include ATM Network, Multimedia System, and Intelligent Transportation System.