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A Study on Voice Communication Quality Criteria Under Mobile-VoIP Environments

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Abstract

In this paper, we present criteria of objective measurement of speech quality to provide the mobile-VoIP services efficiently over wireless mobile internet. The mobile-VoIP service, which is based on mobility and is error-prone compared to conventional VoIP over wired network, is about to be launched, but there have not been adequate quality indexes and the Quality of Service (QoS) standards for evaluating speech quality of Mobile-VoIP. In addition, there are many factors influencing on the speech quality in packet network of which packet loss contribute directly to the overall voice communication quality. For this reason, we adopt the Gilbert-Elliot Channel Model for modeling packet network based on IP and assess the voice quality through the obejective speech method of ITU-T P. 862 PESQ and ITU-T P. 862.1 MOS-LQO under various packet loss rates in the transmission channel environments. Our simulation results address the specific criteria and QoS for the mobile-VoIP services in terms of the various packet loss environments.

Keywords: *Mobile-VoIP, ITU-T P. 862, ITU-T P. 862. 1 MOS-LQO, Gilbert-Elliot Channel Model, Packet Loss*

I. Introduction

With the rapid advances of the communication network technology, the transportation of audio and video, text data through the wide-band network has been significantly increased. In general, we call this wide-band network as a Broadband-convergence-Network (BcN). Based on the BcN, the networks of wireless-mobile communication and wired internet services have been recently converged. A Voice over Internet Protocol (VoIP) has emerged as an important application under the BcN environments and especially mobile-VoIP services over wireless mobile internet such as mobile WiMAX networks are an area of intense research interest. Figure 1 compares the mobile-VoIP network with the conventional wireless mobile communication network [1]. Among all kinds of services in the mobile communication network such as audio, video, text data, voice, a voice quality is fundamental and important issue.

Since the wireless mobile internet networks employ packet-switched transport based on internet protocols (IP), the mobile-VoIP services in this wireless mobile Internet communication network are affected many factors such as End-to-End delay, jitter, packet losses which lead to a deterioration in voice quality. Although there have been much efforts to evaluate the service quality of the conventional wire-less communication system such as 2G and 3G (WCDMA) and a wired VoIP for consumers, there are not appropriate quality indexes and the QoS standards of

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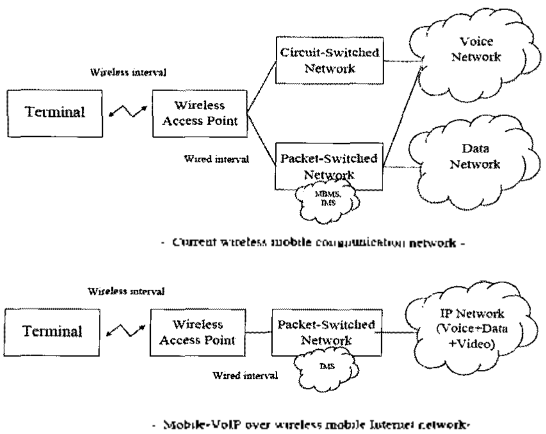

Fig. 1. Comparison between the mobile-VoIP network and the current wireless mobile communication network by Ji-Hyung Han KT (2008) [1].

speech quality for the Mobile-VoIP [2-4]. Therefore, the objective quality criteria and QoS standards for providing the mobile VoIP are inherently required.

Voice quality measurement can be performed using subjective or objective methods. First, mean opinion score (MOS) is usually used in subjective measure of voice quality recommended by the ITU-T P. 800 [5]. The ITU-T P. 800 MOS evaluation is reliable method for assessing the voice quality but difficult to be carried out due to time consuming and expensive cost. Because of the these problems, a variety of objective methods have been proposed to provide the estimation of the objective MOS such as the intrusive objective measurement of Perceptual Analysis /Measurement System (PAMS) [6], Perceptual Evaluation of Speech Quality (PESQ), ITU-T P. 862 [7] or nonintrusive techniques of ITU-T E-model [8] and ITU-T P. 563 [9]. In order to evaluate performances of the public switched telephone and internet network, the ITU-T E-model is mostly used to predict voice quality by computing the system parameters. Also, the E-model as the nonintrusive method is adopted to measure the voice quality for VoIP applications in the internet network. But, the E-model is restricted to the cases which communicate another networks such as public switched telephone systems or current mobile communication networks by a handover. In particular, the E-model can not continuously evaluate the network service quality when the handover occurs to the another networks under the Mobile-VoIP environments.

In this paper, we present the objective speech quality criteria and QoS standards to provide mobile-VoIP services over wireless mobile internet network. To evaluate the voice quality under Mobile-VoIP environments, we investigate impacts of speech codecs such as EVRC (2G), AMR (3G) and G.729 annex A in terms of the various packet loss rates [10-12]. We adopt the Gilbert-Elliot Channel Model for modeling packet network based on IP network and obtain the objective MOS by introducing the ITU-T P. 862 PESQ and ITU-T P. 862.1 MOS-LQO which can linearly compare the PESQ raw data with the subjective MOS scores [13][14]. This paper is organized as follows: Section II briefly compare the PESQ method with the E-model technique for the Mobile-VoIP services. In Section III, we describe the ITU-T P. 862.1 MOS-LQO which is applicable to the Mobile-VoIP. Section IV presents the test results and specific bound lines for reliable voice quality. Finally, the conclusions are drawn in Section V.

II. The overview of objective measurement for the voice quality

In general, the MOS method is broadly used to evaluate the subjective measure of voice quality in the telephone transmission systems. The MOS scores are obtained as an average opinion to denote the test results, where each listener present to the five-point scale from 1 to 5 associated with a definition of voice quality; Excellent (5), Good (4), Fair (3), Poor (2), and Bad (1) [5]. Although the MOS method is internationally a reliable standard measurement of voice quality, the subjective MOS method has the inherent problems which are time consuming, expensive cost, lack of the statistical reliability.

In order to avoid the disadvantages of the MOS method, the objective measurement of voice quality has been developed such as the ITU-T P. 862 PESQ, ITU-T G. 107 E-model, ITU-T P. 563, and PSQM [6-9]. The PESQ and E-model are most commonly used objective measurement method for voice quality in current VoIP systems. The E-model is the relevant technique that can be used for voice quality prediction by computing the system parameters including packet loss, delay, jitter and codec extracted from the network. Since the E-model eventually considers the packet loss, delay and jitter, evaluation of the voice quality for the other networks such as the switched telephone or current mobile communication network is restricted [8]. Also, there is a disadvantage that is impossible to continuously evaluate the voice quality if the handover occurs under the Mobile-VoIP environments.

Contrary to the nonintrusive E-model method, the PESQ of the intrusive objective measurement can be used for evaluating the voice quality criteria and evaluation standard indexes for the speech quality of the end-to-end since the PESQ method compares original reference input signal with degraded speech signal for the end-to-end user. Actually, the PESQ method has been used for measuring the current mobile communication network as voice quality evaluation models based on the actual call quality [15][16]. For this reason, the PESQ is more suitable than the E-model method under the Mobile-VoIP environments. Since the original scores from -0.5 to 4.5 of the PESQ must be identified with the MOS scores of the ITU-T P. 800, the ITU-T P. 862.1 MOS-LQO of the estimated MOS scores from the PESQ raw data is provided. According to the mapping function, the PESQ scores are converted to the estimated MOS (MOS-LQO) which is the values from 1.02 to 4.56. MOS-LQO method is specifically described by the next section. Figure 2 depicts the relationships between MOS-LQS, MOS-LQO and MOS-LQE by the ITU-T P. 800.1 Recommendation [17]. As shown in Fig. 2, the system includes the devices, core network and connection network under the mobile-VoIP service environments. Since the PESQ method objectively measures the voice quality

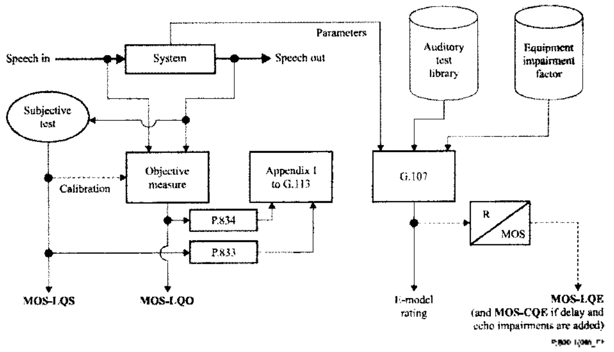

Fig. 2. ITU-T P. 800.1 relationship between some MOS qualifiers.

of the speech in and out in the system, it is called the MOS-LQO. If the speech out results is subjectively evaluated by the listeners, it denotes the MOS-LQS (Listening Quality Subjective). In the E-model method, the estimated MOS scores is obtained through the comparison of speech samples in the database and experimental results based on the ITU-T G. 107 and the extracted system parameters [8]. Note that the E-model does not use the actual speech signals. If the E-model considers the delay and echo, the estimated MOS is the MOS-CQE (Conversational Quality Estimated). Otherwise, the MOS scores of the E-model are classified MOS-LQE (Listening Quality Estimated).

III. The call quality measurement based on the ITU-T P. 862.1 MOS-LQO under the Mobile-VoIP

Contrary to the conventional mobile communication network, where the speech signals were processed in the circuit-switched networks, the Mobile-VoIP are provided with voice over wireless mobile Internet such as mobile WiMAX networks based on IP. The assessment of the voice quality for VoIP applications is conventionally conducted using the intrusive PESQ method or nonintrusive E-model method. Since the handover occurs under the Mobile-VoIP environments, the E-model technique is not suitable for measuring the call quality of the end-to-end.

In order to assess the listening quality of the end-to-end, the PESQ method which is mostly used for conventional VoIP systems based on the fixed internet network must be introduced. Although the PESQ algorithm provides the PESQ-MOS scores from -0.5 to 4.5 as above mentioned in Section II, the PESQ-MOS is identified with the MOS scores by the ITU-T P. 800. Therefore, in this paper, we present the call quality criteria and standard indexes using the PESQ method and MOS-LQO by the ITU-T P. 862.1 Rec so that the estimated MOS scores are compared with the MOS values of the ITU-T P. 800 for the Mobile-VoIP services based on the packet networks in which the voice data is transported with IP. The voice quality in MOS-LQO is measured by converting the PESQ raw data with MOS-LQO (ITU-T P. 800.1) score so that voice assessment of the PESQ method allow a linear comparison with MOS (ITU-T P. 800). The ITU-T P. 862.1 Rec provides a single mapping from raw P. 862 scores to MOS-LQO. The mapping function is given as follows:

$$y = 0.999 + \frac{4.999 - 0.999}{1 + e^{-1.4945 \cdot x + 4.6607}} \tag{1}$$

where y denotes the mapped MOS-LQO score which should be within ± 0.01 absolute of the curve according to the graph of the mapping function and x is the PESQ raw data obtained by the ITU-T P. 862 algorithm. In order for users to compare the PESQ raw data with the MOS-LQO, ITU-T P. 862.1 also provides the graph of the function in Fig. 3 [14].

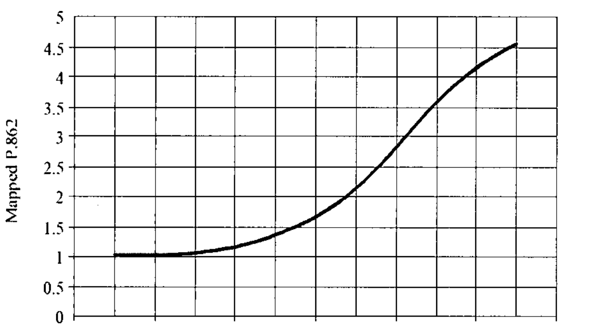

Fig. 3. ITU-T P. 862.1-P. 862 algorithm's mapping function.

IV. Experimental results

The voice data in wireless mobile Internet based on the packet-switching is transported with IP. In order to simulate the packet network, we inherently adopt the Gilbert-Elliot Channel Model based on the two state Markov model where this model has two states, Good (G) and Bad (B). Gilbert packet loss model is illustrated in Fig. 4 [19]. In Fig. 4, state G represents a packet received without a packet loss and state B denotes for the packet loss. The probabilities associated with the transition between G and B states are P and Q where P is the probability of transition from G state to B state and Q is the probability of transition from B state to G state, respectively. According to the above mentioned Gilbert packet model, the error probability BER generated by Gilbert Channel Model is given as follow:

$$BER = \frac{P}{1-\gamma} P_B + \frac{Q}{1-\gamma} P_G \tag{2}$$

where

$$\gamma = 1 - (P + Q) \tag{3}$$

is a coefficient for the correlations of the packet errors and of the burst or random characteristic of the channel. As the above mentioned (3), $\gamma \approx 0$ depicts a nearly random error channel, while $\gamma \approx 1$

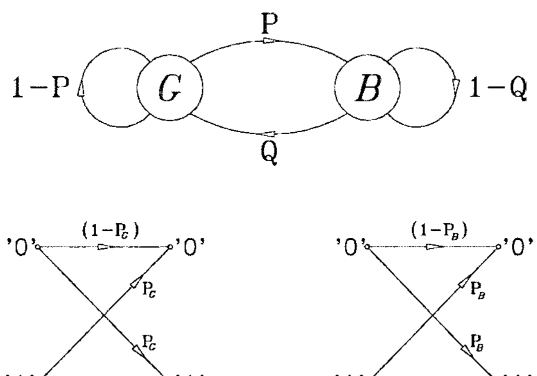

Fig. 4. Gilbert-Elliot Channel Model.

means a totally bursty channel [18].

Combining (2) and (3), the transition probabilities of P and Q is given by:

$$P=(1-\gamma)(1-\frac{P_B-BER}{P_B-P_G}) \tag{4}$$

$$Q=(1-\gamma)\frac{P_B-BER}{P_B-P_G} \tag{5}$$

The channel could be either in good state G or in the bad state B, where the mean error probabilities of P_G and P_B are nearly $P_G=0$ and $P_B=0.5$, respectively. In the simulation environments, we set the $P_G=0$ and $P_B=0.5$. Therefore, The transition probabilities of (4) and (5) is represented such that

$$P=2(1-\gamma)BER \tag{6}$$

$$Q=(1-\gamma)(1-2BER) \tag{7}$$

Based on the Gilbert packet loss model, we evaluate the call quality of the end-to-end for the various speech codecs, where G.729A mostly used for VoIP application, AMR (3G) and EVRC (2G), in terms of the variable packet loss rates using ITU-T G. 191 STL (Software Tool Library) [19]. The PESQ method of objective measurement is conducted and the results are assessed with MOS-LQO by the ITU-T P. 862.1 Recommendation.

In the experiment, the channel is assumed to be performed by either random packet losses or burst packet losses. The random packet loss implies that the random packet loss is generated uniformly under random loss channel characteristic, while the burst packet loss occur continuously for the consecutive frames. In order to conduct the simulation in terms of the variable packet loss rates, the packet loss rates have the values from 1% to 50%. For the simulations, thirty test phrases, spoken by four male and four female speakers from the NTT database, were used. Each phrase was composed with two different 8-second meaningful sentences. To assess the call quality of the speech codecs for the variable packet loss rates under the Gilbert-Elliot Channel Model, we establish the experimental environments as shown in Fig. 5. Firstly, the input speech signal is encoded in the encoder and then the encoding bitstream files were passed through the Gilbert-Elliot Channel Model, where the encoding files with packet losses were generated. Secondly, the bit-stream with or without packet losses was decoded at the near-end speech codecs. Finally, the decoding

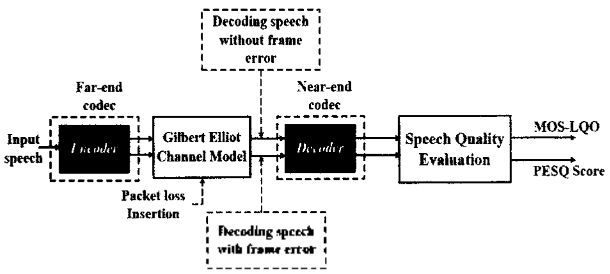

Fig. 5. Block diagram of experimental setup.

Table 1. Relationship between the subjective MOS and user's satisfaction.

MOS	User Satisfaction
4.3	Very Satisfied
4.0	Satisfied
3.6	Some Users Dissatisfied
3.1	Many Users Dissatisfied
2.6	Nearly All Users Dissatisfied
1.0	Not Recommended

Table 2. The PESQ and MOS-LQO results of various speech codecs based on the Gilbert-Elliot Channel Model under random channel envrionments.

	AMR		EVRC		G.729A	
error rates	PESQ	MOS -LQO	PESQ	MOS -LQO	PESQ	MOS -LQO
1%	3.459	3.484	3.364	3.352	3.429	3.450
3%	3.244	3.179	3.014	2.847	3.269	3.220
5%	2.094	2.692	2.814	2.560	3.112	2.989
10%	2.655	2.353	2.162	1.789	2.827	2.576
15%	2.261	1.893	1.894	1.570	2.593	2.268
20%	2.073	1.716	1.732	1.462	2.389	2.025
25%	1.920	1.593	1.585	1.376	2.280	1.905
30%	1.782	1.495	1.387	1.285	2.136	1.765
35%	1.681	1.434	1.229	1.230	2.009	1.658
40%	1.613	1.399	1.150	1.207	1.928	1.597
45%	1.492	1.336	1.082	1.187	1.809	1.513
50%	1.369	1.284	1.011	1.170	1.682	1.431

Table 3. The PESQ and MOS-LQO results of various speech codecs based on the Gilbert-Elliot Channel Model under burst channel envrionments.

	AMR		EVRC		G.729A	
error rates	PESQ	MOS -LQO	PESQ	MOS -LQO	PESQ	MOS -LQO
1%	3.911	4.051	3.667	3.764	3.630	3.717
3%	3.235	3.163	3.584	3.654	3.296	3.255
5%	3.048	2.895	3.342	3.320	3.076	2.936
10%	2.836	2.591	3.342	3.320	2.710	2.418
15%	2.523	2.180	2.158	1.786	2.413	2.048
20%	2.242	1.869	1.938	1.599	2.092	1.724
25%	2.036	1.679	1.355	1.276	2.037	1.676
30%	2.007	1.655	1.356	1.277	2.012	1.657
35%	1.634	1.403	1.317	1.259	1.920	1.587
40%	1.492	1.335	1.177	1.214	1.745	1.469
45%	1.473	1.326	1.177	1.214	1.741	1.470
50%	1.369	1.285	1.014	1.171	1.678	1.432

files with packet losses are compared with the reference speech files without packet losses in the speech quality evaluation block. In the speech quality evaluation block as depicted in Fig. 5, the PESQ scores were obtained and converted to the MOS-LQO using mapping function.

At first, we evaluated the call quality of the representative speech codecs based on the Gilbert-Elliot Channel Model for the packet losses from 1% to 50% under the random channel and the burst channel. The PESQ and MOS-LQO scores can be seen from Table 1 and 2. The MOS-LQO scores for AMR, EVRC and G.729A for the random channel characteristic are summarized in Table 2. In general, if the MOS is larger or equal to 4.0, it commonly represents acceptable satisfaction and means toll quality. However, if MOS is smaller than 3.6 and larger than 3.1, some users are slightly satisfied but certain users could be dissatisfied. In Table 1, it can be seen the user satisfaction relating the subjective MOS. According to Table 1, AMR and G.729A must have be smaller or equal to 3% packet losses to meet the QoS criteria under the Mobile-VoIP for the random channel characteristic environments. Contrary to AMR and G.729A, EVRC should be higher to 3% packet losses.

From Table 3, we can see that MOS-LQO scores for AMR, EVRC and G.729A for the burst channel characteristic are slightly different. EVRC is robust

Table 4. MOS-LQO results of the male versus female for AMR speech codec based on the Gilbert-Elliot Channel Model under random and burst channel envrionments.

	AMR			
	Random channel envrionemts		Burst channel environments	
error rates	Male	Female	Male	Female
1%	3.723	3.246	4.203	3.898
3%	3.453	2.905	3.346	2.981
5%	3.008	2.376	3.059	2.731
10%	2.650	2.055	2.736	2.447
15%	2.104	1.682	2.329	2.030
20%	1.895	1.537	2.019	1.720
25%	1.735	1.450	1.789	1.570
30%	1.608	1.382	1.753	1.557
35%	1.540	1.329	1.476	1.330
40%	1.501	1.296	1.406	1.263
45%	1.417	1.255	1.398	1.255
50%	1.355	1.214	1.356	1.213

Table 5. MOS-LQO results of the male versus female for EVRC speech codec based on the Gilbert-Elliot Channel Model under random and burst channel envrionments.

	EVRC			
	Random channel envrionemts		Burst channel environments	
error rates	Male	Female	Male	Female
1%	3.528	3.175	3.972	3.555
3%	2.956	2.737	3.843	3.466
5%	2.662	2.459	3.488	3.152
10%	1.832	1.746	3.488	3.152
15%	1.616	1.523	1.796	1.775
20%	1.501	1.423	1.601	1.597
25%	1.431	1.321	1.277	1.276
30%	1.318	1.252	1.277	1.277
35%	1.258	1.203	1.264	1.255
40%	1.231	1.183	1.227	1.201
45%	1.206	1.168	1.227	1.201
50%	1.187	1.153	1.189	1.154

in terms of the higher packet losses whose loss rate is ≥10%. We can conclude that MOS-LQO for AMR, EVRC and G.729A should be at least lower or equal to 3% packet loss rates in terms of the QoS for Mobile-VoIP. Also, in Table 4 and 6, we analyzed the characteristic for the MOS-LQO in terms of the male and female, respectively. Since the pitch period

Table 6. MOS-LQO results of the male versus female for G.729A speech codec based on the Gilbert-Elliot Channel Model under random and burst channel envrionments.

	G.729A			
	Random channel envrionemts		Burst channel environments	
error rates	Male	Female	Male	Female
1%	3.616	3.283	3.918	3.516
3%	3.435	3.005	3.525	2.986
5%	3.210	2.769	3.138	2.733
10%	2.764	2.388	2.583	2.253
15%	2.486	2.049	2.190	1.907
20%	2.211	1.837	1.836	1.613
25%	2.082	1.728	1.778	1.574
30%	1.924	1.606	1.759	1.556
35%	1.804	1.512	1.693	1.481
40%	1.741	1.454	1.567	1.370
45%	1.639	1.387	1.579	1.361
50%	1.528	1.334	1.533	1.331

of female is different with the male, the MOS-LQO results are affected by speech codecs which are encoding and decoding with voice data of the male and female. The pitch period of the female is slightly faster than the pitch period of male's voice. Presumable, the female's voice data is determined to be sensitive for the packet losses. Comparing the MOS-LQO of the male with the female, male's MOS-LQO is a difference from 0.1 to 0.4. Therefore, the mobile-VoIP services should be considering the above mentioned results which male's MOS-LQO is more higher than the female's MOS-LQO.

V. Conclusions

In this paper, we have presented a criteria and the QoS standards for objective measurement of speech quality required to provide mobile-VoIP services over wireless mobile Internet such as mobile WiMAX networks in terms of the variable packet loss rates for the various speech codecs. For the simulation of the Mobile-VoIP network based on IP, we adopt the Gilbert-Elliot Channel Model and evaluate the performance of the representative speech codecs which most commonly used AMR (3G), EVRC (2G), G.729A (VoIP applications) by introducing the PESQ of the ITU-T P. 862 and MOS-LQO of the ITU-T P. 862.1.

From the experimental results, it can be seen that the packet loss rates should be at least higher to 3% packet loss under the random and burst channel characteristics to meet the QoS of the Mobile-VoIP. Also, since the MOS-LQO of the male is higher than the female's MOS-LQO, methods to enhance the female's voice quality such as the voice engine should be employed at the network providing the mobile-VoIP services. The PESQ scores and MOS-LQO presented in this paper can be used broadly as the criteria and QoS standards for the Mobile-VoIP services since there are not adequate call quality indexes for the Mobile-VoIP services.

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[Profile]

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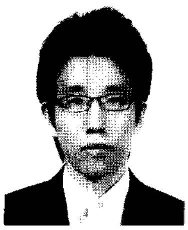

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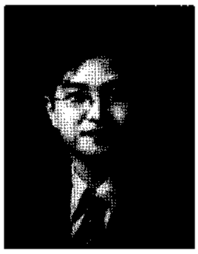
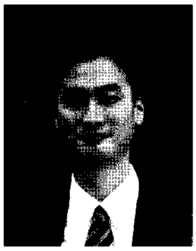
• Soonuk Seol



Soonuk Seol received his B.S. in Information and Communication Engineering from the Korea University of Technology and Education in 1998, and M.S. and Ph.D. in Computer Science from Korea Advanced Institute of Science and Technology, Daejeon, Korea, in 2000 and 2004, respectively. From 1998 he was also a visiting researcher at the Protocol Engineering Center in Electronics and Telecommunications Research Institute (ETRI) until October 1999. Currently, he is with the Central R&D Laboratory at KT, as a senior researcher. His research interests include mobile VoIP/IPTV, mobile computing, QoS network, and protocol engineering.

• Joon-Hyuk Chang



Joon-Hyuk Chang received the BS degree in electronics engineering from Kyungpook National University, Daegu, Korea in 1998 and the MS and PhD degrees in electrical engineering from Seoul National University, Korea, in 2000 and 2004, respectively. From March 2000 to April 2005, he was with Netdus Corp., Seoul, as a chief engineer. From May 2004 to April 2005, he was with the University of California, Santa Barbara, in a postdoctoral position to work on adaptive signal processing and audio coding. In May 2005, he joined Korea Institute of Science and Technology, Seoul, as a Research Scientist to work on speech recognition. Currently, he is an assistant professor in the School of Electronic Engineering at Inha University, Incheon, Korea. His research interests are in speech coding, speech enhancement, speech recognition, audio coding, and adaptive signal processing.