Encoding of Speech Spectral Parameters Using Adaptive Quantization Range Method

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Efficient quantization methods of the line spectrum pairs (LSP) which have good performances, low complexity and memory are proposed. The adaptive quantization range method utilizing the ordering property of LSP parameters is used in a scalar quantizer and a vector-scalar hybrid quantizer. As the maximum quantization range of each LSP parameter is varied adaptively on the quantized value of the previous order's LSP parameter, efficient quantization methods can be obtained. The proposed scalar quantization algorithm needs 31 bits/frame, which is 3 bits less per frame than in the conventional scalar quantization method with interframe prediction to maintain the transparent quality of speech. The improved vector-scalar quantizer achieves an average spectral distortion of 1 dB using 26 bits/frame. The performances of proposed quantization methods are also evaluated in the transmission errors.

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I. INTRODUCTION

Most of speech coders including code excited linear prediction (CELP) coders use linear predictive coding (LPC) parameters for transmitting the short-time spectral envelope information of speech. The LPC coefficients can be transformed into mathematically equivalent representation of line spectrum pairs (LSP) that have the favorable properties for quantization and interpolation [1]. For the quantization of LSP parameters, several vector quantization (VQ) methods [2]-[4] were recently developed in order to overcome the performance limit of scalar quantization method. A benchmarked VQ algorithm is the split VQ approach designed by Paliwal and Atal [2]. It can obtain the 1 dB average spectral distortion (SD) by spending 24 bits/frame in which the 10 dimensional vector is split into the smaller 4 dimensional vector and the small 6 dimensional vector. Another approach to exploit advantages offered by vector quantization while reducing the computational complexity and memory usage is to use a vector-scalar quantization algorithm. Grass and Kabal proposed several methods for vector-scalar quantization that achieved 1 dB spectral distortion by spending 30 bits/frame [4]. Recently a vector-scalar quantizer was used in a 4.8kbps CELP coder [5].

Even though vector quantization techniques show good performances in LSP quantization, applications of them have been so far limited due to the increase of computational complexity and large storage requirement. As the scalar quantization algorithm, the adaptive quantization with the backward sequence (AQBW) using the ordering property of the LSP parameters was designed by Sugamura and Farvardin [6]. They achieved a 1 dB SD using a 32 bits/frame AQBW nonuniform quantizer. The strong interframe correlation of LSP parameters was utilized in several quantization methods so that the amount of bit assignment for LSP parameters transmission can be reduced [7]-[9]. The variable rate QCELP coder which was adopted as the standard vocoder (IS-96) in the North American code division multiple access (CDMA) digital cellular system quantizes the residuals of LSP parameters in the differential pulse code modulation (DPCM) system with a uniform scalar quantizer [9], [10]. The DPCM scheme utilizes the strong interframe correlation of LSP parameters.

In this paper, an efficient scalar quantization method which has good performance, very low complexity and memory is proposed. A new scheme utilizing the ordering property in the DPCM algorithm of QCELP coder is developed so that the quantization ranges of LSP parameters can be reduced significantly. The maximum quantization range of each LSP parameter is varied adaptively on the quantized value of the previous order's LSP parameter. Moreover, the adaptive quantization scheme utilizing the ordering property of LSP parameters is applied in the vector-scalar hybrid quantization method. The performances of proposed methods were evaluated in the presence of transmission errors as well as in noise free channel. This paper is organized as follows. In section II, the DPCM scheme utilizing the ordering property of LSP parameters is presented. In section III, the vector-scalar hybrid quantizer using the adaptive quantization range method is described. In section IV, the performances of proposed algorithms are evaluated in the noisy channel. Finally, a conclusion is given in section V.

II. ADAPTIVE SCALAR QUANTIZATION OF LSP PARAMETERS

In a speech coder, the parameters for the short-term prediction are generally updated every 20 ms to 30 ms at the 8 kHz sampling rate. The coefficients of the short-term prediction error filter (STP) are extracted by the autocorrelation method from the bias-removed input speech signal. The short-term synthesis filter is given as follows:

$$H(z)=\frac{1}{A(z)} = \frac{1}{1+a_1 z^{-1}+\cdots+a_p z^{-p}} \tag{1}$$

where a_i, $i=1,\cdots,p$, are the LPC coefficients and p is the order of the filter. In the low bit rate CELP coder, the order of STP filter is usually 10.

These coefficients are then transformed into the LSP parameters which have the excellent properties for quantization such as the boundness of the parameters, easy stability checking condition of synthesis filter, and the ordering relation among the parameters. For extracting the LSP parameters from the STP coefficients, two polynomial $P(z)$ and $Q(z)$ are introduced as follows:

$$\begin{aligned} Q(z) &= A(z) - z^{-(p+1)} A(z^{-1}) \\ &= (1-z^{-1}) \prod_{i=2,4,\cdots,p} (1-2z^{1}\cos\omega_i + z^{-2}) \end{aligned} \tag{2}$$

$$\begin{aligned} Q(z) &= A(z) + z^{-(p+1)} A(z^{-1}) \\ &= (1+z^{-1}) \prod_{i=1,3,\cdots,p-1} (1-2z^{-1}\cos\omega_i + z^{-2}). \end{aligned} \tag{3}$$

Note that ω_i are the LSP parameters and $e^{j\omega_i}$ are the roots of the $P(z)$ and $Q(z)$. Moreover, the roots of $P(z)$ and $Q(z)$ have very important properties [6]. Firstly, all roots of $P(z)$ and $Q(z)$ are located on the unit circle. Secondly, the roots of $P(z)$ and $Q(z)$ are interlaced with each other on the unit circle. From the second property the following specific relationship among the LSP parameters is obtained;

$$0=\omega_0<\omega_1<\cdots<\omega_p<\omega_{p+1}=0.5 \tag{4}$$

If the ordering property in (4) is satisfied, the stability of the short-term synthesis filter is guaranteed. The ordering property is also very useful for effectively quantizing the LSP parameters [6], [11].

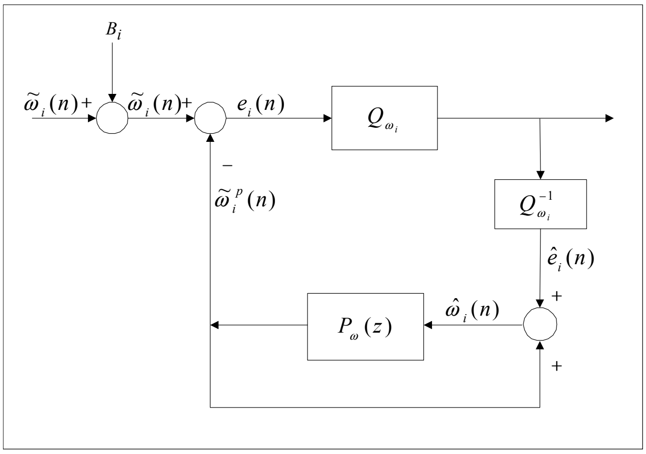

Fig. 1. The DPCM encoding process for the LSP parameters.

The encoding process of DPCM system utilizing the interframe correlation is shown in Fig. 1. The following equations describe the encoding process of LSP quantization. Each of LSP parameter centers roughly around bias value. The bias used for each LSP parameter is as follows:

$$B_i=\frac{0.5i}{p+1} \quad \text{where} \quad p=10. \tag{5}$$

Each bias removed LSP parameter is given by

$$\tilde{\omega}_i(n) = \omega_i(n) - B_i \,. \tag{6}$$

The each input to qunatizer is the prediction error of LSP parameter, given by

$$e_i(n) = \tilde{\omega}_i(n) - 0.90625\hat{\omega}_i(n-1)\,. \tag{7}$$

The quantizer output is

$$\hat{e}_i(n) = Q_{\omega i}^{-1}\left[Q_{\omega i}\left[e_i(n)\right]\right]. \tag{8}$$

The reconstructed LSP parameter is

$$\hat{\omega}_i(n) = \tilde{\omega}_i^p(n) + \hat{e}_i(n)\,. \tag{9}$$

The quantizer for the i th LSP frequency, $Q_{\omega i}$ is a linear quantizer with uniform step size. Each LSP frequency is quantized as follows:

$$Q_{\omega i}(x) = \max\left[0, \min\left(2^N - 1, Q_{ti}(x)\right)\right] \tag{10}$$

where $Q_{ti}(x) = round\left(\frac{2^N-1}{2}\frac{x+e_{i\max}}{e_{i\max}}\right)$, N is the number of quantization bits, $e_{i\max}$ is the maximum quantization level, and $round(x)$ is the function rounding to the closest integer. The maximum quantization range $e_{i\max}$ is given in Table 1. The adaptive quantization range method considering the ordering property of LSP parameters is applied in the DPCM system so that the maximum quantization range of LSP parameters can be adaptively varied. The scalar quantization of a LSP parameter starts from the ω_{10} and then proceeds to the quantization of the lower order LSP parameters. This approach is called the DPCM with backward sequence (DPCM-BW). In this algorithm, $\omega_{10}(n)$ is quantized with the usual maximum quantization range. From $\omega_9(n)$, check if the maximum quanti-zation range could be shrunken by using the ordering property of the LSP parameters.

Define a checking variable by

$$z = \omega_{i+1}^q(n) - \omega_i^p(n)\,. \tag{11}$$

Here, $\omega_{i+1}^q(n)$ is the quantized value of the $(i+1)$ th order LSP parameter, and the predicted value of the i th order LSP parameter adding the bias term, $\omega_i^p(n)$, is given by $\tilde{\omega}_i^p(n) + B_i$. The quantized value of $e_i(n)$ cannot be greater than z because the $\omega_i^q(n)$ should be less than $\omega_{i+1}^q(n)$ to guarantee the stability of short-term synthesis filter. Therefore, if $|z| < e_{i\max}$, it is not necessary to assign the normal quantization range of $e_i(n)$ that covers from $-e_{i\max}$ to $+e_{i\max}$. The maximum quantization range of $e_i(n)$ can be reduced to $[-e_{i\max}, z]$. The reduced maximum quantization range is shown in Fig. 2.

Table 1. Maximum LSP quantization level in DPCM scheme.

LSP Frequency	Max. Range ($e_{i\max}$)
ω_1	0.03
ω_2	0.04
ω_3	0.07
ω_4	0.07
ω_5	0.06
ω_6	0.06
ω_7	0.05
ω_8	0.05
ω_9	0.04
ω_{10}	0.04

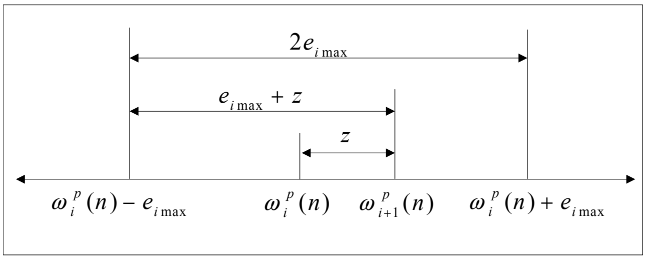

Fig. 2. Maximum quantization range for DPCM-BW.

The quantization algorithm of DPCM-BW is as follows:

1. *Quantize* $\omega_{10}(n)$ *with DPCM quantizer which has a normal quantization range* $[-e_{i\max}, e_{i\max}]$ *and set* $i = 9$.
2. *Compute a checking variable* $z = \omega_{i+1}^q(n) - \omega_i^p(n)$.
3. *(a) If* $|z| < e_{i\max}$, *then quantize* $e_i(n)$ *using the uniform quantizer with a shrunken quantization range* $[-e_{i\max}, z]$ *in DPCM system.*
   (b) If $|z| > e_{i\max}$, *then quantize* $e_i(n)$ *using the uniform quantizer with a normal quantization range* $[-e_{i\max}, +e_{i\max}]$ *in DPCM system.*
4. *If* $i = 1$, *stop; otherwise, set* $i = i - 1$ *and go back to step 2.*

While the DPCM-BW method is adaptive quantization scheme, no additional information bits are needed. This is because of the fact that the quantization range used for encoding the i th LSP parameter is uniquely determined by the quantized value of ω_{i+1}, which is also available in the decoder. The idea used in the DPCM with backward sequence can be also applied in the forward sequence. This method is called the DPCM with forward sequence (DPCM-FW) and described below.

1. Quantize $\omega_1(n)$ with DPCM quantizer which has a normal quantization range $[-e_{i\max}, e_{i\max}]$ and set $i = 2$.

2. Compute a checking variable $z = \omega_{i-1}^q(n) - \omega_i^p(n)$.

3. (a) If $|z| < e_{i\max}$, then quantize $e_i(n)$ using the uniform quantizer with a shrunken quantization range $[z, e_{i\max}]$ in DPCM system.

(b) If $|z| > e_{i\max}$, then quantize $e_i(n)$ using the uniform quantizer with a normal quantization range $[-e_{i\max}, +e_{i\max}]$ in DPCM system.

4. If $i = 10$, stop; otherwise, set $i = i + 1$ and go back to step 2.

Note that the quantizer used in DPCM scheme without the adaptive quantization range is at least as coarse as the corresponding quantizer in the DPCM-BW or DPCM-FW. Therefore, the average spectral distortion of DPCM-FW or DPCM-BW should be less than that of DPCM scheme. Another advantage of the proposed adaptive quantization method is that it does not need to have the stability checking routine in the decoding process since the new algorithm already satisfies the ordering condition of LSP parameters.

III. VECTOR-SCALAR HYBRID QUANTIZER USING ADAPTIVE QUANTIZATION RANGE METHOD

The advantage of vector-scalar hybrid quantization(VQ-SQ) is that a codebook size in vector quantization stage is much smaller than that in a single stage vector quantization with the same number of bits. Thus, the complexity for codebook search, and memory requirement can be drastically reduced by using the vector-scalar hybrid quantizer of LSP parameters. The block diagram is shown in Fig. 3.

At first, input LSP parameters are vector quantized using a codebook with the moderate number of entries. Multiple candidate code vectors are selected in the VQ stage. In the second stage of quantization, the components of residual vector are individually quantized by the scalar quantizer. The following equations describe the encoding process of vector-scalar hybrid quantizer.

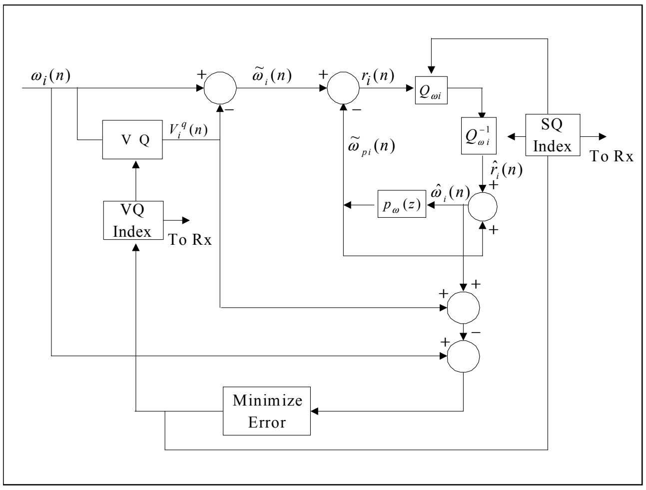

Fig. 3. The encoding process of vector-scalar hybrid quantizer.

$$V_i^q(n) = V_i(index_{VQ}(n)) \tag{12}$$

$$\tilde{\omega}_i(n) = \omega_i(n) - V_i^q(n) \tag{13}$$

$$r_i(n) = \tilde{\omega}_i(n) - \tilde{\omega}_{pi}(n) \tag{14}$$

$$\hat{r}_i(n) = Q_{\omega i}^{-1}[Q_{\omega i}[r_i(n)]] \tag{15}$$

$$\hat{\omega}_i(n) = \tilde{\omega}_{pi}(n) + \hat{r}_i(n) \tag{16}$$

$$\tilde{\omega}_{pi}(n) = 0.3625\hat{\omega}_i(n-1) \tag{17}$$

where $i = 1, 2, \cdots, p$.

The quantizer, $Q_{\omega i}$, for the i th LSP frequency is a linear quantizer with uniform step size. The maximum quantization range $\gamma_{i\max}$ is given in Table 2. The final optimal VQ-SQ code vector which has a minimum distortion is selected among multiple VQ-SQ quantized value combinations.

The adaptive quantization range method utilizing the ordering property of LSP parameters is applied in the vector-scalar hybrid quantization of LSP parameters. The improved vector-scalar hybrid quantization method is called as vector-scalar hybrid quantization with forward sequence (VQ-SQ-FW) or vector-scalar hybrid quantization with backward sequence (VQ-SQ-BW).

The checking variable in VQ-SQ-BW that decides the maximum quantization range in the scalar quantizer is given by

$$x = \omega_{i+}^q(n) - [\omega_{pi}(n) + V_i^q(n)], \tag{18}$$

where $\omega_{i+1}^q(n)$ is the quantized value of the $(i+1)$ th order LSP parameter, $\omega_{pi}(n)$ is the predicted value of the i th order LSP parameter in the scalar quantizer, and $V_i^q(n)$ is the i th order quantized output of the first stage vector quantizer.

Table 2. Maximum LSP quantization level in vector-scalar hybrid quantizer.

LSP Frequency	Max. Range ($e_{i\max}$)
γ_1	0.0125
γ_2	0.0175
γ_3	0.025
γ_4	0.025
γ_5	0.025
γ_6	0.025
γ_7	0.0225
γ_8	0.0225
γ_9	0.0175
γ_{10}	0.0175

The maximum quantization range of each scalar quantizer in the VQ-SQ-BW may be shrunk by checking the relation of the $(i+1)$ th order quantized value and the sum of i th element of vector quantizer output, $V_i^q(n)$ and the predicted value $\omega_{pi}(n)$ of the ith LSP parameter. The nominal maximum quantization level of each parameter is given by $\gamma_{i\max}$. If $|x| < \gamma_{i\max}$, quantize $\gamma_i(n)$ with the shrunk quantization range, *i.e.*, $-\gamma_{i\max} \sim x$ and otherwise, quantize $\gamma_i(n)$ with the nominal quantization range, *i.e.*, $-\gamma_{i\max} \sim +\gamma_{i\max}$. The final optimal VQ-SQ code vector which has a minimum distortion is selected among multiple VQ-SQ quantized value combinations.

IV. SIMULATION RESULTS

The performances of new algorithms are evaluated in terms of the average spectral distortion. The SD is defined as follows:

$$SD(dB) = \frac{1}{NF}\sum_{n=1}^{NF} [(\frac{1}{\pi}\int_0^{\pi} [10\log_{10}\left|A_n(e^{j\omega})\right|^2 - 10\log_{10}\left|\hat{A}_n(e^{j\omega})\right|^2]^2 d\omega)]^{\frac{1}{2}}$$

where NF is the number of total frames, and $A_n(e^{j\omega})$ and $\hat{A}_n(e^{j\omega})$ are the spectra of the n th speech frame without quantization and with quantization. Two kinds of Korean speech database are used for the experiment. One consists of 6 male and 5 female speeches which are recorded from an FM radio station. Another consists of 3 male and 3 female speeches recorded in two anechoic rooms with different recording conditions. The test database of about 15,000 LSP frames with 20 ms duration is used for evaluation.

Table 3. The performance of three scalar quantization schemes.

bit/frame	AQBW		DPCM		DPCM-BW	
	SD>2*dB*		SD>2*dB*		SD>2*dB*	
	dB	%	dB	%	dB	%
27	1.45	4.6	1.59	12.6	1.32	2.6
29	1.26	2.4	1.42	6.3	1.17	1.7
31	1.18	2.0	1.21	2.0	1.02	1.0
33	1.06	1.8	1.07	1.4	0.92	0.8
36	0.86	1.6	0.89	1.1	0.77	0.7

Table 3 shows the performance comparisons among the AQBW, the DPCM scheme of the QCELP, and the proposed DPCM-BW at different bit rates in terms of the average spectral distortion and the number of outlier frames greater than 2 dB. The DPCM-BW algorithm which utilizes an ordering property achieves a 2-3 bits/frame saving over the conventional DPCM and the AQBW in terms of the SD. The DPCM-BW needs 31 bits/frame to maintain the transparent speech quality that the average distortion is less than 1 dB and the percent of frames with over 2 dB distortion is less than 2%. The ratio of speech frame that is quantized with the shrunken quantization range in DPCM-BW quantization process is shown in Table 4. The simulation results show that the shrunken quantization range is used in over 50 % of total speech frames.

Table 4. The ratio of frame used the shrunken quantization range in DPCM-BW.

LSP Parameter	1	2	3	4	5	6	7	8	9	10
Percent	71.8	60.7	87.0	73.0	79.6	66.7	63.6	57.5	42.0	0

The proposed scalar quantizer was applied to the LSP quantization method of a speech coder. The QCELP coder is used as a platform for the performance evaluation of LSP quantization method. The QCELP coder is a variable rate speech coder in which the transmission rate is decided depending on input speech energy level. The input speech is sampled at 8 kHz and speech is broken down into 20 ms frames. Each frame may have one of the four different basic rate, 8, 4, 2, 1 kbps. The QCELP coder assigns 40 bits/frame, 20 bits/frame, 10 bits /frame, 10 bits/frame for LSP parameters transmission in the rate of 8 kbps, 4 kbps, 2 kbps, 1 kbps, respectively. The objective

Table 5. The SEGSNR (dB) values of the QCELP coder according to LSP quantization methods.

	DPCM	DPCM-BW
8 kbps	14.84	15.11
Variable Rate	13.79	14.02

Table 6. Performance of 27 bits/frame vector-scalar quantization according to the number of candidate vectors.

Can Vec	1	4	8	12	16	20	24
SD	1.51	1.14	1.00	0.97	0.95	0.94	0.94

Table 7. The SD (dB) values in vector-scalar quantization schemes.

bit/frame	VQ-SQ		VQ-SQ-BW	
	SD>2*dB*		SD>2*dB*	
	dB	%	dB	%
24	1.25	3.0	1.15	1.0
25	1.20	2.8	1.10	0.5
26	1.10	1.5	1.01	0.4
27	1.05	1.2	0.95	0.3
28	0.98	1.0	0.88	0.2

performance comparison between the conventional DPCM scheme and DPCM-BW method is shown in Table 5 for Korean speech database. The average segmental signal to noise ratio (SEGSNR) of the QCELP coder using the DPCM-BW at 40 bits/frame is about 0.3 dB higher than that of the QCELP coder using the DPCM at 40 bits/frame. In an informal listening test, the performance improvement was small but not significant because the QCELP coder assigned sufficiently enough bits (40 bits/frame) for the LPC parameter transmission. But the speech quality improvement may be evident if 3 or 4 bits obtained by introducing the new LSP quantizer are assigned to other speech parameters such as pitch periods, gain parameters.

In the vector-scalar hybrid quantization, the codebook size of the first stage vector quantizer is selected as 256 (8 bits). The codebook of vector quantizer is designed by using the LBG algorithm [12] on the training speech data. The performance of vector-scalar quantizer is improved as the number of candidate vectors in the first stage increase. The performance of the quantizer according to the number of candidate vectors is shown in Table VI. Sixteen candidate vectors in the first stage are adequate. The performance comparisons between the conventional VQ-SQ and the VQ-SQ using adaptive quantization range method are shown in Table 7. The simulation results show that the adaptive VQ-SQ provides 1-2 bits/frame saving over the conventional VQ-SQ. The VQ-SQ-BW quantization algorithm needs 26 bits/frame to achieve 1 dB average spectral distortion. Note that the proportion of outlier frame with distortion greater than 2 dB is significantly low. Both DCPM-BW and VQ-SQ-BW showed the same performance improvement in the other language (English) because the performance gain is not obtained by use of codebook trained by special speech database, but by the adaptive use of reduced maximum quantization range.

Table 8. Channel error performance of DPCM-BW at 32 bits/frame and VQ-SQ-BW at 27 bits/frame.

BER	DPCM-BW		VQ-SQ-BW	
	SD>2*dB*		SD>2*dB*	
	dB	%	dB	%
0	0.98	0.8	0.95	0.3
10^{-4}	0.99	1.2	0.96	0.4
5×10^{-4}	1.05	3.0	0.99	1.0
10^{-3}	1.13	5.6	1.03	1.8
5×10^{-3}	1.66	22.0	1.34	8.1
10^{-2}	2.28	40.1	1.69	15.4

The performances of proposed quantization methods are evaluated in the presence of channel errors and the structure of quantizers is modified to have the robustness to the channel errors. The scalar quantizer with interframe prediction may be very sensitive on the effect of channel errors due to autoregressive structure of predictor. Thus, the predictor in the DPCM-BW is changed to moving average predictor from the autoregressive predictor. The order of moving average predictor is given by 3. The use of the moving average predictor in proposed adaptive scalar quantizer showed the 0.02 dB performance degradation in spectral distortion in the error free channel. But the use of a moving average predictor provided 0.22 dB gain at bit error rate (BER) of 10^{-3} and 2.06 dB gain at BER of 10^{-2}.

In the adaptive vector-scalar quantizer, the choice of predictor structure did not affect the performance of quantizer in the noisy channel. The performances of DPCM-BW at 32 bits /frame and VQ-SQ-BW at 27 bits/frame in the presence of channel errors are shown in Table 8. The simulation results show that the adaptive quantizers using the ordering property of LSP parameters perform as well as other robust LSP quantizer [13] in the presence of channel errors. The proposed adaptive quantization methods did not show the severe performance degradation at the BER below 10^{-3}.

V. CONCLUSIONS

The DPCM quantization algorithm of the LSP parameters is improved by investigating the important ordering property of the LSP parameters. The simulation results show that the DPCM scheme using an adaptive quantization range method (DPCM-BW) provides about 3 bits/frame saving over the conventional DPCM scheme without increasing complexity. The DPCM-BW quantization algorithm needs 31 bits/frame to achieve the transparent quality of speech. Also, the new quantization method considering the ordering property of LSP parameters has been applied in the vector-scalar hybrid quantizer. The proposed vector-scalar quantizer achieves an average spectral distortion of 1 dB using 26 bits/frame. Moreover, new adaptive quantization range algorithm is robust to channel errors. Even though the vector-scalar hybrid quantizer and scalar quantizer using adaptive quantization range method showed a significant bit saving over conventional methods, they are not the proper LSP quantization methods for low rate speech coders below the rate of 4 kbps. The LSP quantizer should show an average spectral distortion of 1 dB at 20 bits/frame or less. The new structure for LSP quantizer must be developed for low rate speech coder. The vector-vector hybrid quantizer or quantizer using transformation of LSP parameters can be a solution for low rate speech coders.

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