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Speech Processing

Volume II of V

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A LONG HISTORY QUANTIZATION APPROACH TO SCALAR AND VECTOR QUANTIZATION OF LSP COEFFICIENTS

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ABSTRACT

It is essential that in low bit rate speech coding systems, codec parameters are quantized with a minimal number of bits without a corresponding reduction in the quality of the decoded signal. In an effort to increase coding efficiency, a new and general methodology for the quantization of codec parameters, called Long History Quantization (LHQ), has recently been proposed. LHQ, when applied in conjunction with scalar quantization for the coding of LSP coefficients in a CELP system, offers "transparent" quantization at an average bit rate of 25 bits per frame. This paper presents certain improvements to the above LHQ-Scalar quantization scheme, which further reduce the average LSP bit rate to about 22 bits per frame. In addition, a new LHQ-Vector quantization scheme is proposed which allows the transparent quantization of LPC coefficients with only 19 bits per analysis frame.

1. INTRODUCTION

The line spectrum pair (LSP) coefficients have been extensively used as an equivalent representation of the LPC coefficients in medium and low bit rate speech coding systems. As a consequence, several LSP quantization schemes have been proposed in the past which exploit the interframe [1,2,3] and intraframe [4,5] correlation of LSP parameters. More recently, a general approach for the adaptive quantization of codec parameters, called Long History Quantization (LHQ), has been proposed. LHQ exploits the constraints imposed on the signal by i) the speech production mechanism characteristics of individuals and ii) language and phonetic considerations [6].

This paper presents first certain improvements in the LHQ-Scalar quantization (LHSQ) of the LSP coefficients scheme reported in [6] and then combines LHQ with vector quantization in a new and efficient LHVQ coding of the LSP coefficients. The general LHQ approach is briefly reviewed in Section 2. The LHSQ and LHVQ schemes are described in Section 3 and their compression characteristics are discussed in Section 4.

2. LONG HISTORY QUANTIZATION (LHQ)

Conventional low bit rate coders operate on successive speech frames and derive a set of parameters for each frame. These parameters are quantized and used to drive a speech synthesis model and thus produce an approximation of the corresponding speech frame. Optimized scalar [5] and vector [7] quantization is usually applied to model parameters. In this case, quantizers are designed and optimized according to long term statistics and are therefore fixed rather than adaptive. Fixed quantization ignores the fact that for each "signal event", in a speech frame, it is likely that another previous frame contains a very similar or exactly the same signal event. LHQ exploits this long-term redundancy present in the model parameters to obtain further compression.

In particular, consider that E_k represents the kth speech frame to be encoded and $\{P_k\}_i$, $i=1, 2, ..., M$ are M sets of model parameters "describing" E_k. In addition, let E'_{k-j} and $\{P'_{k-j}\}_i$, $j=1, 2, ..., N$ be the N previously decoded speech frames and the corresponding quantized parameter sets respectively. A parameter set $\{P_k\}_i$ can be LHQ-quantized to one of the N $\{P'_{k-j}\}_i$ parameter sets representing part of the signal's history. In this way, the information transmitted regarding $\{P_k\}_i$ is the index j of the entry $\{P'_{k-j}\}_i$ of what is effectively an adaptive codebook of size N. The LHQ approach can be applied to a single parameter, individual sets of model parameters or combinations of them. It can lead to fixed or variable bit rate coding schemes and can considerably reduce the bit rate of the underlying codec. LHQ can be used together with conventional fixed scalar or vector quantization and the choice between LHQ and fixed quantization is transmitted in the form of a binary flag, $F_{k,i}$, on a frame by frame basis. Thus, $\{P_k\}_i$ is first quantized, using a fixed quantizer, to $\{P''_k\}_i$ which is then compared to the quantized entries $\{P'_{k-j}\}_i$ of the LHQ codebook. The system then decides whether to transmit $\{P'_k\}_i=\{P''_k\}_i$ with $F_{k,i}=0$ or the index j of the $\{P'_k\}_i=\{P'_{k-j}\}_i$ assignment with $F_{k,i}=1$. In this case, the LHQ codebook search is based on quantized parameter values and involves only the indices of the fixed quantizer output levels. This leads to an effective codebook search strategy which is discussed in the next section as a specific application of LHQ to LSP parameters.

The performance of an LHQ scheme is determined by the accuracy of the fixed scalar or vector quantizer employed by the system, the size of the LHQ adaptive codebook and the threshold values used in the LHQ codebook search. The LHQ codebook is updated, in our experiments, using the simple Least Recently Used (LRU) caching technique [8] which maximizes the coding relevance of the information stored in the codebook.

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LSP coeff. p	Bits B_p	LSP coeff. p	Bits B_p
1	3	6	4
2	4	7	4
3	4	8	4
4	4	9	4
5	4	10	3

Table 1
Bit allocation for 38-bit fixed scalar quantization

LSP sub-vector p	Dimension	Bits B_p
1	3	10
2	3	10
3	4	10

Table 2
Bit allocation for 30-bit fixed split-VQ

3. APPLICATION OF LHQ TO LSP COEFFICIENTS

LHQ has been applied, in conjuction with scalar (LHSQ) or vector quantization (LHVQ), for coding 10 LSP coefficients. These parameters are estimated every 20ms and they are initially quantized using a total of 38 bits(scalar) or 30 bits(vector) per frame respectively, see Tables 1 and 2.

In general, the LHQ search algorithm "compares" and "attempts to match" the output index of the fixed quantizer to those stored in the LHQ codebook. The LHSQ search algorithm is described first and a discussion on the LHVQ search procedure then follows.

Consider that $SQ(O_p)_k$ represents the index (quantization output level number, $SQ(O_p)_k = 1, ..., 2^{B_p}$) of the pth scalar quantized LSP coefficient, $C''_{k,p}$, in the kth LPC analysis frame. In the same way, $SQ(O_p)_{k-j}$ represents the index of the pth element of the jth vector, $C'_{k-j,p}$, stored in the LHQ codebook.

The LHSQ search algorithm compares sequentially $\{C''_{k,p}\}$, $p=1, 2, ..., 10$ to the N codebook vectors $\{C'_{k-j,p}\}$, $j=1, 2, ..., N$. This means that the process starts by comparing $SQ(O_1)_k$ of $C''_{k,1}$ to the index $SQ(O_1)_{k-1}$ of $C'_{k-1,1}$ element of the first codebook vector. If $|SQ(O_1)_k - SQ(O_1)_{k-1}| \leq T_1$, where T_1 is a fixed integer threshold, then $|SQ(O_2)_k - SQ(O_2)_{k-1}| \leq T_2$ is examined. Thus the algorithm examines the mth, m=2, 3, ..., 10, inequality,

$$|SQ(O_m)_k - SQ(O_m)_{k-1}| \leq T_m \tag{1}$$

only if the (m-1)th inequality is satisfied. When (1) is not satisfied, the search on the current codebook vector is aborted and the next vector is tested. Furthermore, a codebook entry is identified as a possible candidate to represent $\{C''_{k,p}\}$ if (1) is satisfied for all m=1, 2, ..., 10. When none of the LHQ codebook vectors can be identified to be useful, $\{C''_{k,p}\}$ is assigned to represent $\{C'_{k,p}\}$ and F_k is set to zero. On the other hand, one or several codebook entries may satisfy the search criterion, and a spectral distortion (SD) is formed for each candidate entry. In general,

$$SD = \sqrt{\frac{1}{N_\omega}\sum_{l=1}^{N_\omega}\left[\log_{10}S_k(\omega_l) - \log_{10}\hat{S}_k(\omega_l)\right]^2} \text{ (dB)} \tag{2}$$

where $S_k(\omega_l)$ and $\hat{S}_k(\omega_l)$ are the spectral values at frequency ω_l of the original and "quantized" LPC spectra respectively. N_ω denotes the number of spectral values used from 0 to π. When several candidate vectors are available at the end of the search process, the vector with minimum SD is selected. This gives a single candidate codebook vector, and its SD value is compared to a threshold value TSD which effectively controls the "transparent" quantization of the input LSP vectors. F_k is set to 1 only when

$$SD \leq TSD \tag{3}$$

A similar search procedure is performed in the LHVQ coding of the LSP coefficients. Let $VQ(O_p)_k$ represent the index of the pth vector-quantized LSP sub-vector, $C''_{k,p}$, in the kth LPC analysis frame, p=1, 2, 3. In the same way, $VQ(O_p)_{k-j}$ represents the index of the pth sub-vector of the jth vector, $C'_{k-j,p}$, stored in the LHQ codebook. Unfortunately, (1) cannot be used as the search criterion in this case because the absolute difference between the indices of two LSP sub-vectors does not usually provide any information on their spectral discrepancy. In order to utilize the $\{VQ(O_p)_{k-j}\}$ indices in the LHVQ search process, a new search method has been devised. During the fixed vector quantization of $\{C_{k,p}\}$; p=1, 2, 3, N_s best candidate indices, $\{VQ(O_{p,q})_k ; q=1, 2, ..., N_s\}$, for each p sub-vector are selected. The search commences with the possible matching of any of the N_s candidate indices, $\{VQ(O_{1,q})_k\}$, produced from the quantization of the first sub-vector, $C_{1,p}$, to the $\{VQ(O_{1,q})_{k-j}\}$ elements of the LHVQ codebook. The same search process is then repeated for the second and third sets of N_s candidate indices with the corresponding $\{VQ(O_{2,q})_{k-j}\}$ and $\{VQ(O_{3,q})_{k-j}\}$ subsets of the LHVQ codebook. The entries of the LHVQ codebook for which

$$\{VQ(O_p)_{k-j}\} \in \{VQ(O_{p,q})_k ; q = 1, 2, ..., N_s\} \tag{4}$$

is satisfied for all 3 sub-vectors are considered further, in terms of their SD and the one with minimum SD is subsequently tested for satisfying (3). It is only then F_k is set to 1 indicating the use of LHVQ in the quantization of the input frame.

4. RESULTS AND DISCUSSION

The proposed LHSQ and LHVQ schemes have been evaluated using computer simulation and their compression potential examined. When coding 10 LSP coefficients obtained from a large database, the performance of a conventional scalar LSP quantizer and a three-way split vector quantizer is shown in Figure 1, in terms of Average Spectral Distortion (ASD) measure and the number of bits allocated every 20ms, where

$$ASD = \frac{1}{N_f}\sum_{k=1}^{N_f} SD \text{ (dB)} \tag{5}$$

and the ASD is measured over N_f=12800 frames. Transparent quantization is defined to have been obtained when i) the ASD $\leq$ 1dB, ii) less than 2% of decoded frames satisfy $2dB \leq SD \leq 4dB$ and iii) none of the decoded frames has $SD > 4dB$. This is achieved with 32 or more bits per frame in the case of scalar quantization and 24 or more bits if vector quantization is to be used. With bit allocations of 38 bits and 30 bits for fixed scalar and vector quantization respectively, ASD values of 0.631 dB and 0.647 dB are obtained.

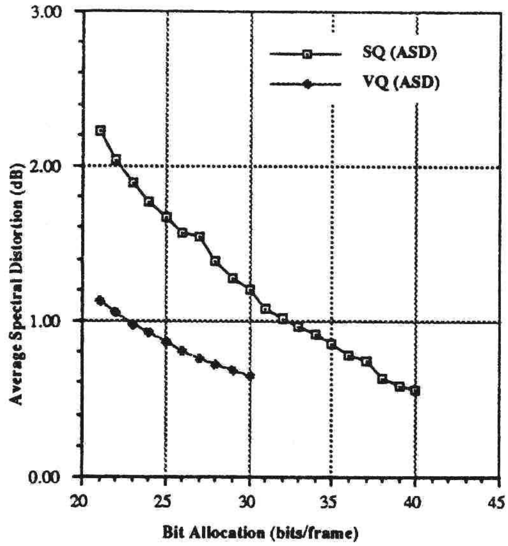

Figure 1

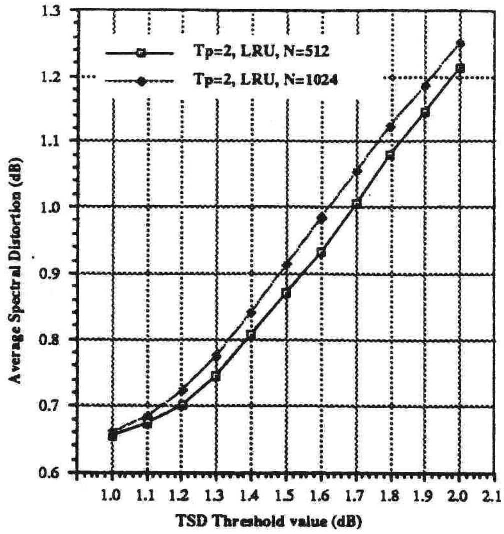

Figure 2

Figures 2, 3, 4 and 5 refer to the LHSQ coding of the LSP coefficients. Figure 2 shows the ASD performance of the system for different values of the TSD threshold used in the codebook search and for two codebook sizes, i.e. 512 and 1024. T_p is set to 2 and the LRU caching technique is used to update the codebook. As expected, higher TSD values relax the distortion constraint, imposed by the the quantization algorithm, on selecting previous codebook entries, and allow the ASD of the system to increase. A TSD value of 1.6dB, however, ensures that ASD is less than 1dB. Using this TSD value and also TSD=∞, which represents the system reported in [6], Figures 3, 4 and 5 illustrate the LHSQ performance, for different codebook sizes, in terms of ASD, Average Bit Rate (AvBR) and Percentage of Frames (%) with =1, respectively. Transparent LSP quantization is achieved with TSD=1.6 dB at an average bit rate of 22.4 bits per frame. Notice that the TSD assumes different "optimum" values when the fixed quantization bit allocation given in Table 1 is modified.

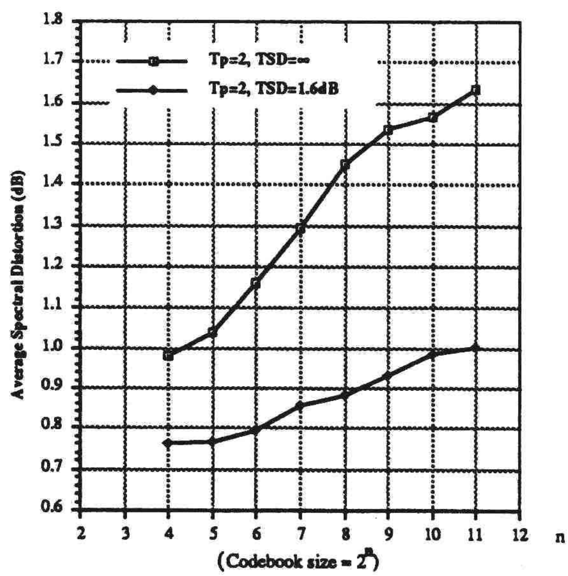

Figure 3

The performance of the LHVQ scheme described in the previous section is summarized in Table 3. Again, TSD is set to 1.6dB, while N_s = 60. It has been found that LHVQ achieves transparent LSP coding with an average bit rate of 19 bits per frame.

Notice that both the LHSQ and LHVQ algorithms described in this paper represent a particular variable bit rate implementation of the LHQ approach, which can easily be integrated within the framework of an LPC based constant bit rate codec. Furthermore, the codebook search in both algorithms is mainly based on the comparison of codeword indices and may therefore be implemented efficiently.

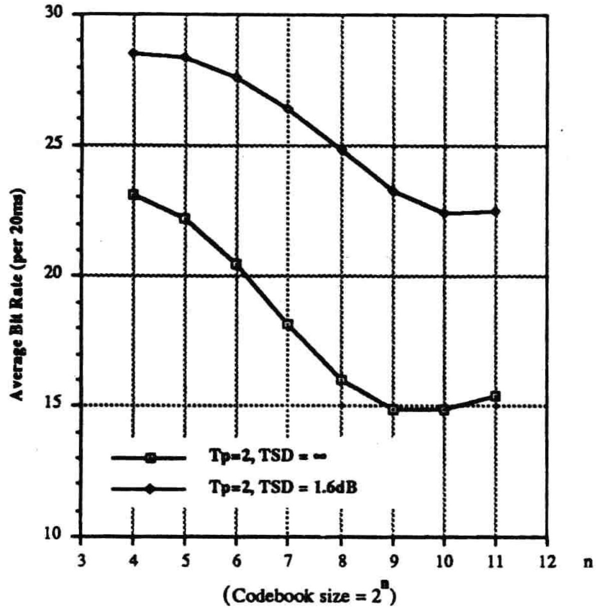

Figure 4

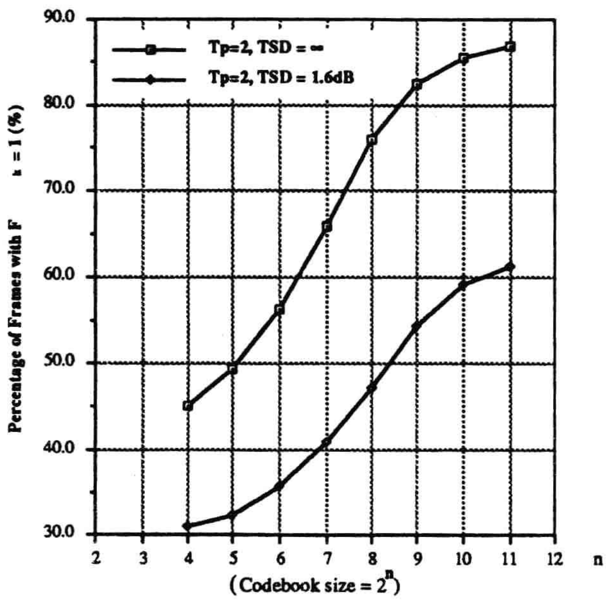

Figure 5

Codebook size, N	ASD (dB)	% of $F_k = 1$	Average Bit Rate
16	0.791	34.49	22.03
32	0.804	36.25	21.94
64	0.829	39.37	21.55
128	0.866	44.19	20.84
256	0.916	50.12	19.97
512	0.968	56.37	19.16
1024	1.019	61.81	18.64

Table 3

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EFFICIENT CODING OF LPC PARAMETERS USING ADAPTIVE PREFILTERING AND MSVQ WITH PARTIALLY ADAPTIVE CODEBOOK

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ABSTRACT

In this paper, we propose an efficient vector quantization scheme and a new LPC analysis scheme, both of which exploit interframe correlation in the successive spectrum envelope of speech signals. The first quantization scheme proposed is a multi-stage vector quantization of LSP parameters with a partially adaptive codebook (MSVQ-AC). The second new algorithm is an LPC analysis scheme, with closed-loop adaptive prefiltering (LPC-PF), which realizes temporary higher order analysis than the standard LPC with a few additional transmission bits. A combined system of the LPC-PF and 2-split, 2-stage VQ with the adaptive codebook can quantize 10-th order LSP parameters at around 23 bits/frame, realizing sufficient quality and reasonable complexity.

1. INTRODUCTION

Linear predictive coding is widely used as a short time spectral envelope estimation in various speech processing applications. For low bit rate speech coding, it is important to quantize LPC parameters using as few bits as possible without sacrificing the speech quality and within a reasonable complexity. Various quantization schemes have been proposed for such objectives. Scalar quantization of individual coefficients results in acceptable levels of spectral distortion at 32 to 36 bits/frame. Vector quantization is a more efficient scheme which utilizes intraframe correlation among the LPC parameters [1]. For higher efficiency, it is necessary to exploit the interframe correlation of successive parameter sets. Shoham studied VQ with vector predictive quantization (VPQ) [2]. In their VQ, the current spectral envelope is predicted using past quantized spectra and its prediction residual spectrum is quantized. Matrix quantization (MQ) is a direct scheme to quantize a set of parameter vectors [3]. MQ is a very expensive scheme, however, in terms of computation and memory and in that it introduces a large encoding delay. Farvardin et al. studied two-dimensional DCT coding of LSP vectors [4]. Encoding is possible at 21 bits/frame by using appropriate bit allocation based on the distribution of each DCT coefficient. However, this scheme also introduces a large encoding delay. Grass et al. studied vector-scalar quantization with an adaptive codebook, which does not require extra transmission information or encoding delays [5]. We have extended this idea for a multi-stage VQ framework as described in Section 2.

On the other hand, an LPC analysis itself is usually performed in a memoryless fashion, and no one has yet considered using the interframe correlation of spectral envelope in the analysis stage.

In this paper, we propose a new efficient vector quantization scheme and a new LPC analysis scheme, both of which exploit interframe correlation of the spectrum. The new quantization scheme is a multi-stage vector quantization with a partially adaptive codebook (MSVQ-AC). The first stage codebook has a small adaptive part as well as a large fixed part, which stores the quantized parameters of the past frames.

To exploit the interframe correlation in the analysis stage, a new LPC analysis scheme called LPC-PF, which employs closed-loop adaptive prefiltering prior to LPC analysis, is also proposed.

This paper is organized as follows. In Section 2, as a practical implementation of MSVQ-AC, we present a 2-split, 2-stage VQ with an adaptive codebook for 10-th order LSP parameters. In Section 3, we present a detailed algorithm and do a performance evaluation of LPC-PF.

2. MULTI-STAGE VQ WITH PARTIALLY ADAPTIVE CODEBOOK (MSVQ-AC)

2.1 Configuration

A multi-stage quantization structure can make use of a partially adaptive codebook [5] to exploit interframe correlation. A multi-stage VQ configuration, with this partially adaptive codebook, is shown in Fig. 1. The codebook of first stage VQ has fixed and adaptive parts. The adaptive part stores the quantized vectors of the past (n-J)-th frames, where J is the adaptive codebook size. It is sufficient to add a small size adaptive part to a fixed codebook to efficiently exploit interframe correlation, because a coefficient vector with the highest correlation to that of the current frame can be usually found within the parameter sets of the most recent frames.

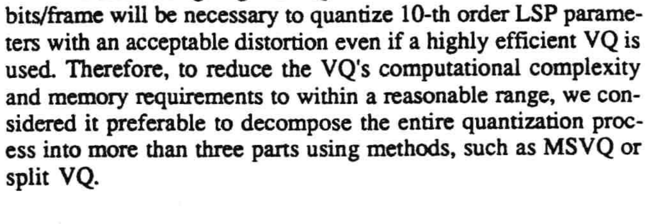

Figure 1. MSVQ with partially adaptive codebook

Before designing the VQ, we assumed that more than 20 bits/frame will be necessary to quantize 10-th order LSP parameters with an acceptable distortion even if a highly efficient VQ is used. Therefore, to reduce the VQ's computational complexity and memory requirements to within a reasonable range, we considered it preferable to decompose the entire quantization process into more than three parts using methods, such as MSVQ or split VQ.

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In our study, we have developed a 2-split, 2-stage VQ (Fig. 2). At the first stage, the 10-dimensional LSP vector is split into 2 subvectors, one of length 4 (lower band) and the other of length 6 (higher band). Each subvector is separately quantized. Split VQ has been reported as an efficient scheme for LSP quantization due to its localized spectral sensitivity to quantization error [6]. At the second stage, the 10-dimensional error vector between the input vector and the reconstructed vector from the first stage VQ is quantized. The distance measure used in the search of this VQ is the same as in reference [6].

In the first stage split VQ, both the lower band (CB1-L) and higher band (CB1-H) codebooks have small independent adaptive parts. This configuration allows separate searches of the adaptive codebooks in the lower and higher bands.

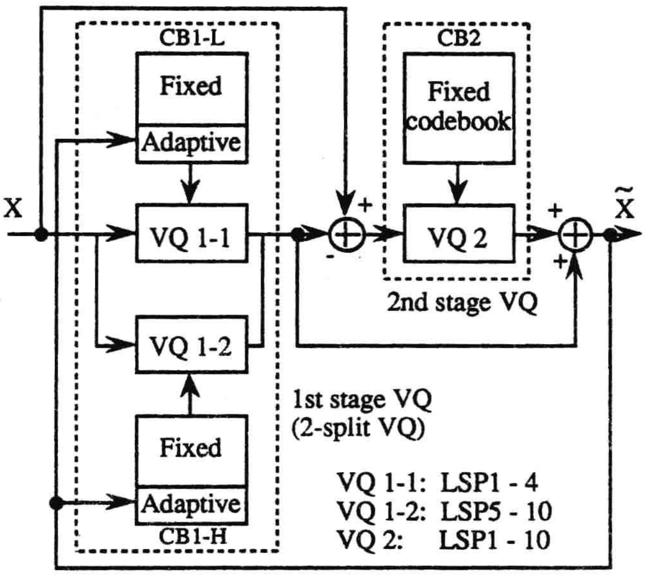

Figure 2. 2-split, 2-stage LSP-VQ with partially adaptive codebook

2.2 Codebook design

Fixed part of each codebook is designed by using the LBG algorithm on the training data. A weighted Euclidian distance is used for the design as well as for the codebook search. For training the second stage codebook, we found that the quantization error vectors of the first stage, whose spectral distortion is greater than a certain threshold value, are useful for the training data to reduce the percentage of outliers. We used a simulation to determine the appropriate threshold of 1.5 dB.

2.3 Codevector reconstruction in split VQ

In a sequential search of CB1-L and CB1-H, invalid unstable combinations of codevectors are present which should be excluded in the CB1-H search. To prevent the loss of valid codebook size from this, CB1-H codevectors are reconstructed. In this method, unstable codewords are modified to satisfy the LSP parameter stability conditions by nonlinearly transforming each coefficient according to the selected CB1-L codevector.

2.4 Evaluation

Thirty minutes of Japanese speech was used as the database in this study. The first 25 minutes were used for codebook training and the last 5 minutes were used for evaluation. 10-th order LPC analysis, based on the autocorrelation method with high frequency compensation, was performed every 20 ms using a 20 ms Hamming window.

The performance is summarized in Table 1. Average spectral distortion (SD) and outlier percentage (calculated for frames having SD in the range of 2-4 dB and for SD above 4 dB) are used for evaluation. The VQ with second stage codebook training using first stage outliers reduces the outliers percentage with slight SD degradation. Average spectral distortion equivalent to 30 bits/frame SQ was achieved at 24 bits/frame (8 bits for each codebook) for the proposed VQ.

The appropriate adaptive codebook size was determined experimentally. Improvement in performance saturated when the size was larger than 4. So, we set the size to 4 in the following simulation. Using the adaptive codebook significantly reduces outliers as well as average spectral distortion. The proposed 24 bits/frame VQ seems to be sufficient for speech coding applications.

Table 1. Performance of various LSP quantizers

Quantization scheme	Bits/20 ms	SD (dB)	Outlier (%)	
			2dB<SD<4dB	4dB<SD
SQ	34	0.97	1.87	0.0
	32	1.12	4.26	0.0
	30	1.23	5.63	0.01
	28	1.38	9.36	0.04
3-MSVQ	24	1.21	5.81	0.0
2-split VQ	24	1.25	5.83	0.0
2-split, 2-stage VQ	24	1.23	3.81	0.0
2-split, 2-stage VQ with O.T.	24	1.25	3.53	0.0
2-split, 2-stage VQ with O.T. & A.C. (M=4)	23	1.26	3.19	0.0
	24	1.19	1.90	0.0
	26	1.10	0.81	0.0

O.T.: 2nd stage codebook is trained by outliers from the 1st stage VQ
A.C.: Adaptive codebook

2.5 Complexity

The complexity of the proposed 24 bits/frame VQ (with three 8-bit codebooks) is 8 times smaller than that of the conventional 2-split VQ, since it only requires three 8-bit codebook searches rather than two 12-bit codebook searches. Moreover, the memory required for codebook storage is also reduced from 40 K words to 5 K words.

3. LPC ANALYSIS WITH ADAPTIVE PREFILTERING (LPC-PF)

3.1 LPC-PF configuration

This section presents an efficient LPC analysis scheme, LPC-PF, which is an alternate scheme for exploiting interframe correlation. The LPC-PF configuration is shown in Fig. 3. In this scheme, LPC analysis and quantization are included in an adaptive feedback loop. The input speech signal is first fed into a low order prefilter, which is a kind of an inverse filter whose coefficients are determined from past quantized LPC parameters. This prefilter preceding LPC analysis performs rough spectral flattening of the input signal. The overall spectrum envelope, H(z), estimated by this system is represented by the cascade characteristics of the prefilter P(z) and the following LPC analysis filter A(z) as follows.