Single and Double Frame Coding of Speech LPC Parameters Using a Lattice-Based Quantization Scheme

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***Abstract*—A lattice-based scheme for the single-frame and the double-frame quantization of the speech line spectral frequency parameters is proposed. The lattice structure provides a low-complexity vector quantization framework, which is implemented using a trellis structure. In the single-frame scheme, the intraframe dependencies are exploited using a linear predictor. In the double-frame scheme, the parameters of two consecutive frames are jointly quantized and hence the interframe dependencies are also exploited. A switched scheme is also considered in which, lattice-based double-frame and single-frame quantization is performed for each two frame and the one which results in a lower distortion is chosen. Comparisons to the Split-VQ [13], the Multi-Stage VQ [12], the Trellis Coded Quantization [15], the interframe Block-Based Trellis Quantizer [17], and the interframe scheme used in IS-641 EFRC [26] and the GSM AMR codec [28] are provided. These results demonstrate the effectiveness of the proposed lattice-based quantization schemes, while maintaining a very low complexity. Finally, the issue of the robustness to channel errors is investigated.**

***Index Terms*—CELP, interframe coding, intraframe coding, lattice-based quantization, LPC, LSF, quantization, speech coding.**

I. Introduction

THE short-term spectral information of the speech signal is often modeled by the frequency response of an all-pole filter in speech coding applications. The filter coefficients, also known as the linear predictive coding (LPC) coefficients, are derived from the input signal through linear prediction analysis of each frame of speech, which is typically 10 to 30 ms long.[1] The quantized LPC coefficients play a major role in the overall bit-rate and the quality of the encoded speech. For practical speech codec deployments, the challenge in the quantization of the LPC parameters is to achieve *transparent quantization* quality [1], with the minimum bit-rate while maintaining the memory and computational complexity at a low level. This article focuses on the quantization of LPC parameters for narrowband speech coding applications.

Several equivalent representations of LPC coefficients have been suggested in the literature [2]–[5], which are more suitable for quantization than the LPC coefficients in their direct form. This is due to certain interesting properties of these representations including improved control over the effect of quantization errors in the frequency domain [1]. Among them, the line spectral frequency (LSF) is a widely accepted representation of LPC coefficients [5]. In the case of narrowband speech sampled at 8 k-samples/s, a tenth-order LPC filter is considered which is represented by ten LSF parameters.

Various schemes based on scalar quantization have been suggested for the quantization of the LSF parameters. These schemes are interesting due to their low level of complexity; however, they often require high bit-rates to achieve the transparent quality. Direct scalar quantization of the LSF parameters at 34 bits per frame (bpf) is used for the U.S. federal standard FS-1016 [7]. Differential scalar quantization of LSF parameters is considered in [6]. To improve the coding efficiency, a hybrid vector-scalar quantization scheme is proposed in [8].

Vector quantizers achieve the transparent quantization quality at lower bit-rates. However, they are more computationally complex and have higher memory requirements. A full search VQ is estimated to achieve the transparent quality at about 18 bpf [1], but it requires 10 Megabytes of memory for codebook storage and a very large number of operations for the codebook search. A more recent study suggests higher estimates [9]. To reduce the complexity, various forms of structured vector quantizers have been proposed [10]–[15]. The multistage vector quantization of the LSF parameters is proposed in [12]. The transparent quantization quality is reported at 22–30 bpf with high to moderate levels of complexity. In [13], split vector quantization of LSF parameters at 24 bpf is suggested. An algebraic vector quantization algorithm for the transparent quantization of the LSF parameters at 28 bpf with a small complexity is proposed in [14]. In [15], the trellis coded quantization (TCQ) [16] of the LSF parameters is presented. More recently, a quantization scheme based on a trellis structure, that models the statistical properties of the LSF parameters, is proposed in [17].

All the schemes mentioned above attempt to efficiently quantize the LSF parameters of one frame using only the dependencies among the parameters of the same frame, hence they are categorized as the *Intraframe LSF Quantizers*. However, since the speech spectrum varies slowly with time, there is a substantial dependency between the parameters of the nearby frames as

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[1]In this work, the frame size is considered to be 20 ms.

well. The *Interframe LSF Quantizers* exploit these dependencies to reduce the bit-rate further. But, this comes at different prices of increased delay, increased complexity and, increased vulnerability to channel errors. In [25], the parameters of up to four consecutive frames form a matrix and are jointly quantized using a split matrix quantizer.

The interframe *predictive quantizers* are designed based on the fact that the LSF parameters of a given frame can be predicted from the parameters of the previous frames [18]–[23]. Moving average (MA) prediction is used in [18], and the ITU-T Rec. G.729 8-kb/s speech coding standard [19]. In [20], an autoregressive predictive scheme is suggested in which intraframe and interframe coded frames are interlaced. This limits error propagation to, at most, one adjacent frame. Along the same direction, an interframe block-based trellis quantizer is proposed in [17] and, switched-predictive quantization schemes are proposed in [21], [22]. Nonlinear prediction has also been considered for predictive interframe quantization of the LSF parameters [23]. For a comprehensive review of the interframe schemes, refer to [22].

In this work, we propose a lattice-based quantization (LBQ) scheme for the quantization of the LSF parameters in the intraframe and the interframe modes. Three methods are presented based on the same lattice structure, and their software implementation is available at [29]. In the first method, the intraframe dependency of the LSF parameters are exploited using a first-order scalar linear predictor, and the LSF differences are quantized. In the second method, two consecutive frames are jointly encoded and hence, the interframe dependencies are also exploited. The third method considers a switched approach in which, lattice-based double frame and single-frame quantization is performed for each two frame and the one which results in a lower distortion is transmitted. One additional bit is transmitted to indicate the selected method of quantization. Numerical results are provided, which indicate an improved performance compared to some well-known methods from the literature, while significantly reducing the computational complexity and memory requirements. The proposed double-frame and switched coding approaches achieve substantial gains, over the single-frame coding scheme, at the cost of an additional frame delay (here 20 ms). However, this delay is acceptable for real-time speech communications. The North American standard IS-641 [26], and the 3GPP specification [27] based on the GSM-AMR [28] codec, require the encoded bitstream of two consecutive frames to be stored and interleaved prior to transmission. As a result, the delay requirement of the proposed quantization schemes is already provisioned in these popular standards.

The organization of the rest of this article is as follows. Section II, provides a brief introduction on LSF parameters and the distortion measure used. Section III describes the proposed lattice-based quantization schemes. Section IV presents the numerical results. We conclude this article in Section V.

II. Preliminaries

In this section, we present a brief review of the properties of LSF parameters and proceed with the description of the distance measure used in this work.

A. Line Spectral Frequencies

A 10th-order LPC analysis results in an all-pole filter with 10 poles whose transfer function is denoted by $H(z) = 1/A(z)$, in which $A(z) = 1 + a_1 z^{-1} + \ldots + a_{10} z^{-10}$, and $[a_1, a_2, \ldots, a_{10}]$ are the LPC coefficients. These coefficients are equivalently represented by the LSF parameters which are related to the zeros of a function of the polynomial $A(z)$ [1]. The LSF parameters are denoted by

$$\mathbf{l} = [l_1, l_2, \ldots, l_{10}]^T \tag{1}$$

and they are in fact a scaled version of the angular frequencies known as Line Spectral Pairs, which are located between 0 and π. The ordering property of the LSF parameters states that these parameters are ordered and bounded within a range, i.e., $0 < l_1 < l_2 < \ldots < l_{10} < 0.5$. Provided that this property is preserved for the quantized LSF's, it is known that the reconstructed LPC filter is stable [1]. Since the LSF representation is a frequency domain representation, it can be used to exploit certain properties of the human perception system.

B. Distance Measure

The simplest metric, usually used in quantization, is the Euclidean distance. In order to incorporate the characteristics of the human auditory system, different weighted Euclidean distance measures have been proposed in the literature for the quantization of the LSF parameters. These distance functions are generally of the form

$$d_i(l_i, \hat{l}_i) = w_i c_i (l_i - \hat{l}_i)^2 \tag{2}$$

$$D_{k=10}(\mathbf{l}, \hat{\mathbf{l}}) = \sum_{i=1}^{k=10} d_i(l_i, \hat{l}_i). \tag{3}$$

The vector $\mathbf{c} = [c_1, c_2, \ldots, c_{10}]$ is a constant weight vector which prioritizes the LSF parameters. These weights are meant to emphasize the lower frequency components which are more important to the perceptual quality of speech. The vector $\mathbf{w} = [w_1, w_2, \ldots, w_{10}]$ is a variable weight, which is derived from the LSF vector in each frame, and is meant to provide a better quantization of LSF parameters in the formant regions. Paliwal and Atal [13] suggested assigning a variable weight w_i to the ith LSF, which is proportional to the value of the LPC power spectrum at this frequency. In [10], a simpler weight function was proposed which takes advantage of the fact that formant frequencies are located at the position of two or three closely located LSF parameters. Gardner and Rao in [31] analyzed high-rate (full-search) vector quantization of the LPC parameters and presented a weighted distance function which at high rates approximates the spectral distortion measure.

Equation (2) is the definition of the metric used in this work. We employ a nonlinear weight function to determine the variable weights. This weight for a sample LSF vector $\mathbf{l}$ is given by

$$w_1 = \begin{cases} 1.0, & \text{if } (2\pi(l_2 - 0.02) - 1) > 0 \\ 10\,(2\pi(l_2 - 0.02) - 1)^2 + 1, & \text{otherwise.} \end{cases}$$

$$w_i = \begin{cases} 1.0, & \text{if } 2\pi(l_{i+1} - l_{i-1}) - 1 > 0 \\ 10\,(2\pi(l_{i+1} - l_{i-1}) - 1)^2 + 1, & \text{otherwise.} \end{cases} \quad 2 \leqslant i \leqslant 9$$

$$w_{10} = \begin{cases} 1.0, & \text{if } (2\pi(0.471 - l_9) - 1) > 0 \\ 10\,(2\pi(0.471 - l_9) - 1)^2 + 1, & \text{otherwise.} \end{cases} \tag{4}$$

which has been designed based on the same idea of emphasizing the closely positioned LSF parameters. The constant weights c_i in (2) are all set to one, except c_4 and c_5 which are set to 1.2. This weight function is the same as that used in the ITU-T G.729 standard [19]. The values 0.02 and 0.471 used in (4) are, respectively, the minimum value of the first LSF and the maximum value of the tenth LSF for the codec for which our LSF quantizer has been designed. The experiments in [17] indicate the effectiveness of the weight function of (4).

III. Lattice-Based Quantization

A lattice is a discrete set of N-dimensional vectors which forms a group. The fundamental region of a lattice is a building block, which fills the whole space with just one lattice point in each copy. Lattice quantizers are based on using the points of the lattice to partition the space into the quantizer regions. We can achieve a reduction in the quantization noise by using the points of lattice instead of an N-dimensional rectangular array of points (quantization granular gain). For quantization, given a point $\mathbf{l} = (l_1, \ldots, l_N)$, we find a point $\hat{\mathbf{l}}$ in the lattice which has the minimum Euclidean distance to $\mathbf{l}$. The objective is to choose a lattice with a low search complexity and with a high quantization gain. In this work, we use the lattice D_n, $n = 10$, since it provides a reasonable quantization gain [37] and, as we will see in the following, facilitates a quantizer with low search complexity. The lattice D_{10} comprises of the points with the coordinates $(j_1, j_2, \ldots, j_{10})$, for which

$$\sum_{i=1}^{10} j_i \equiv 0 \pmod 2. \tag{5}$$

Assuming independent quantization of each dimension of the vector $\mathbf{l}$, e.g., using a scalar quantizer, the indices j_i, $1 \leq i \leq 10$, correspond to the codewords of the ith quantizer. Therefore, we have $0 \leq j_i < 2^{r_i}$, where r_i is the bitrate of the ith quantizer. The D_{10} lattice expressed in (5), constrains the sum of the quantizer indices to be even. In fact, the LBQ can simply drop (do not transmit) the least significant bit of one of the ten indices, say the last index, leading to a rate of

$$r = \sum_{i=1}^{10} r_i - 1 \quad \text{bpf}. \tag{6}$$

The decoder easily identifies the missing bit, using the lattice constraint of (5). More importantly, this transforms the independent quantization of different dimensions (LSF parameters), to a low-complexity vector quantization process. As described below, we use a trellis diagram with two states and ten stages to implement this quantizer. The Viterbi algorithm is used to find the path with the minimum distortion.

A. Trellis Representation

The structure of the lattice D_{10} is represented by a trellis structure. Fig. 1 depicts an example of such a trellis diagram for the case of a 19-bpf quantizer. Each stage in the trellis diagram is associated with one dimension of the LSF vector; hence, there are ten stages in the trellis, plus an initial stage 0. At each stage of the trellis, there are two states identified by $(stage, state) = (i, s)$, where

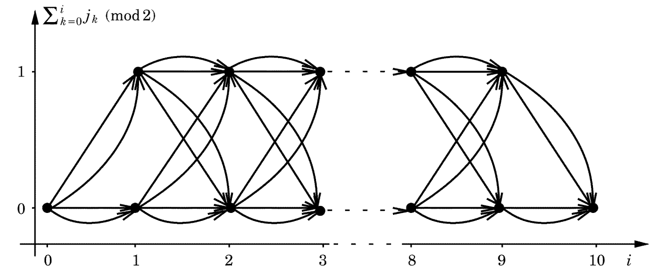

Fig. 1. Trellis representation of the lattice structure for the example of a 19-bpf LBQ with two bits allocated to each dimension (LSF parameter).

$$\begin{aligned} &s \in \{0, 1\}, \quad 1 \leq i < 10 \\ &s = 0, \quad i = 0, 10. \end{aligned} \tag{7}$$

The motivation for this is explained shortly. Each branch is identified by $(stage, state, branch) = (i, s, j)$. Associated with each branch $(i-1, s, j)$, going out of the state $(i-1, s)$, is the codeword $C_i(s, j)$. We assume

$$C_i(0, j) = C_i(1, j) = C_i(., j), \quad 1 \leq i \leq 10, \ \ 0 \leq j < 2^{r_i}. \tag{8}$$

The set of codewords $C_i(., j)$ form the codebook $\mathcal{C}_i$, which is related to the ith LSF parameter. The LBQ codebook is composed of ten such sets as follows:,

$$\mathcal{C} = \{\mathcal{C}_1, \mathcal{C}_2, \ldots, \mathcal{C}_{10}\}. \tag{9}$$

A sequence of k branches (and their associated codewords), connecting a state in the initial stage 0 to another state in the kth stage, provides candidate quantized values for the first k LSF parameters. Therefore, each state represents $\sum_{k=0}^{i} j_k \pmod 2$, which reflects the lattice structure. This, in fact, categorizes the codewords of the ith stage $\mathcal{C}_i$ to two groups of odd and even indexed codewords. The branch $(i-1, s, j)$ connects the state $(i-1, s)$ to state $(i, s + j \pmod 2)$: Depending on j being an even or an odd number, the branch arrives at the same state or the alternative state in the next stage, respectively. The collection of the paths of the trellis starting at state (0,0), and ending at state (10,0) determine the set of LBQ codevectors, all of which meet the constraint of (5).

B. Trellis Search

The ultimate goal of the LBQ search algorithm is to find the path, which results in the minimum distortion to quantize a particular sample LSF vector. The LBQ search algorithm starts from the first stage and performs a set of operations in each stage until reaching the last stage. These operations include calculating a metric for each branch and assigning a cost to each state, similar to the Viterbi algorithm [30]. This specifies one surviving path reaching each state (i, s). The surviving path reaching the last stage (state (10,0)) will determine the quantizer output. The metric corresponding to the branch $(i-1, s, j)$, is the distortion (2) introduced in the ith reconstructed LSF, $\hat{l}_i$, if this branch is taken.

In the following, we will present three different schemes, which are all based on the same lattice structure. In the first and simplest algorithm, introduced in Section III-C as LBQ-LSFD, the branches (codewords) correspond to the LSF differences. Subsequently, in Sections III-D and III-E, we extend this to more complex constructions.

It is noteworthy that, the LBQ search may also be conducted by the so-called Wagner rule [38], which is a popular approach in the decoding of the single parity codes. Based on this approach, after independent quantization of the LSF parameters, if the lattice constraint of (5) is not satisfied, one of the codewords is replaced with another codeword from the opposite category (odd indexed or even indexed), such that it results in a minimum increase in distortion. In this work, we choose to use the trellis representation and the Viterbi search, which as described below accommodates the closed-loop quantization of the LSF differences. The search remains suboptimum, since it does not follow a dynamic programming approach in a differential quantization scheme. However as seen in Section IV, it provides an acceptable performance.

C. Single Frame Quantization

We set up a single-frame scheme for coding of LSF parameters using the lattice-based quantization approach described. To exploit the intraframe dependencies of the LSF parameters, we incorporate a closed-loop first-order DPCM [35] scheme within the lattice structure. In this case, the branches or codewords correspond to the LSF parameter differences. The candidate quantized value for $\hat{l}_i$, associated with the state $(i, s+j \pmod 2)$, provided by the branch $(i-1, s, j)$, is given by

$$\hat{l}_i\,(s+j \pmod 2)) = C_i(.,j) + G_i(s), \quad 1 \le i \le 10 \qquad (10)$$

where $G_i(s)$ provides a prediction of the value l_i, and is associated with the state (i,s), and the codeword $C_i(.,j)$ compensates for the prediction error. We have

$$G_1(s) \triangleq 0$$
$$G_i(s) \triangleq \alpha_{i-1}\hat{l}_{i-1}(s) + \beta_{i-1}, \quad 1 < i \le 10. \qquad (11)$$

In (11), the term $\hat{l}_{i-1}(s)$ is the quantized LSF parameter for the surviving path reaching state $(i-1, s)$. Equation (11) indicates a first-order linear Auto Regressive prediction, whereby, each LSF parameter is predicted from the previous parameter. It is straight forward to see that the coefficients α_i and β_i minimizing the mean squared prediction error is given by

$$\alpha_i = \frac{COV(l_i, l_{i+1})}{VAR(l_i)}$$
$$\beta_i = E[l_{i+1}] - \alpha_i E[l_i], \quad 1 \le i \le 9 \qquad (12)$$

where E denotes the expectation and VAR and COV denote the variance and the covariance, respectively. These parameters are calculated using our training database (described in Section IV) and are presented in Table I.

The lattice-based quantization approach described is referred to as the LBQ-LSFD, and is evaluated in Section IV. For an r bpf LBQ-LSFD, a total of $r+1$ bits are allocated to the codebooks $\mathcal{C}_i$, $1 \le i \le 10$, as follows. First, a maximum equal number of bits is assigned to each codebook. Next, starting from the lower indexed ones, one additional bit is allocated to each codebook, until all the bits are exhausted. The emphasis on lower indexed LSF parameters are motivated by the facts that: 1) they are perceptually more important and 2) a finer quantization of these parameters also contribute to better quantization of higher indexed LSF parameters through the employed DPCM scheme. For example, consider a $r = 22$ bpf LBQ-LSFD. The bit allocation across different dimensions are given by (3,3,3,2,2,2,2,2,2,2). We verified the effectiveness of this approach through extensive simulations.

TABLE I
PARAMETERS FOR THE INTRAFRAME FIRST-ORDER LINEAR PREDICTION OF LSF PARAMETERS

i	1	2	3	4	5	6	7	8	9
α_i	1.211	1.284	0.653	0.800	0.779	0.598	0.707	0.581	0.641
β_i	0.015	0.024	0.091	0.080	0.087	0.154	0.129	0.192	0.172

D. Double-Frame Quantization

In this scheme, the LSF parameters of two consecutive frames $\mathbf{l}^{(n-1)}$ and, $\mathbf{l}^{(n)}$, $n = 2m$, $m \in \mathcal{N}$ are jointly quantized. The same trellis structure, and linear prediction as used in LBQ-LSFD is used, however, the branches correspond to two-dimensional (2-D) codevectors instead. The candidate quantized value associated with the state $(i, s+j \pmod 2)$, provided by the branch $(i-1, s, j)$, is given by

$$\begin{bmatrix} \hat{l}_i^{(n-1)}\,(s+j \pmod 2)) \\ \hat{l}_i^{(n)}\,(s+j \pmod 2)) \end{bmatrix} = \begin{bmatrix} C_{1,i}(.,j) \\ C_{2,i}(.,j) \end{bmatrix} + \begin{bmatrix} G_i^{(n-1)}(s) \\ G_i^{(n)}(s) \end{bmatrix}, \quad 1 \le i \le 10 \qquad (13)$$

where $\begin{bmatrix} G_1^{(n-1)}(s) \\ G_1^{(n)}(s) \end{bmatrix}$ provides a prediction of a vector $\begin{bmatrix} l_i^{(n-1)} \\ l_i^{(n)} \end{bmatrix}$ and is associated with the state (i,s). Next, the codevector $\begin{bmatrix} C_{1,i}(.,j) \\ C_{2,i}(.,j) \end{bmatrix}$ compensates for the prediction error. We have

$$G_1^{(k)}(s) \triangleq 0$$
$$G_i^{(k)}(s) \triangleq \alpha_{i-1}\hat{l}_{i-1}^{(k)}(s) + \beta_{i-1},$$
$$k = n-1, n, \quad n = 2m$$
$$m \in \mathcal{N};\ 1 < i \le 10 \qquad (14)$$

which still indicates a first-order autoregressive scalar linear predictor. Since each two consecutive frames are jointly quantized, an extra delay equivalent to the duration of one frame (20 ms) is imposed, which is justified as discussed in Section I. The bit allocation for the LBQ-2LSFD is performed in the same way as described for the LBQ-LSFD, except that the quantizer now uses $2r+1$ bits to jointly code the parameters of two consecutive frames at the overall bitrate of r bpf. The performance of this scheme, that is referred to as the LBQ-2LSFD, is reported in Section IV.

E. Switched Quantization

Two successive frames are quantized once with the single frame and once with the double-frame lattice-based quantizers, LBQ-LSFD and LBQ-2LSFD. Subsequently, the quantized set which achieves a lower distortion, in terms of the weighted distance of (3), is transmitted to the receiver. One bit is used as

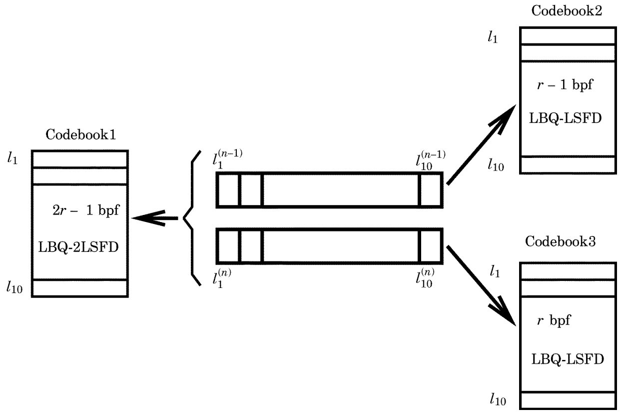

Fig. 2. Switched single and double-frame quantization of LSF parameters: The frames n and $n-1$ are jointly considered, $n = 2m, m \in \mathcal{N}$.

the *switch bit* to indicate the single-frame or the double-frame quantization.

Fig. 2 depicts this method for the case of switched quantization at r bits per frame, and demonstrates how the bits are allocated to different quantizers. A total of $2r - 1$ bits are used for the quantization of the two frames, and considering the switch bit, the bitrate remains fixed at r bpf. This method is referred to as the LBQ-SWCH, and is evaluated in Section IV. Our experiments with the LBQ-SWCH scheme show that about 20% of the frames are quantized in the single-frame mode by LBQ-LSFD and the rest are quantized in the double-frame mode by LBQ-2LSFD.

F. Quantizer Design

In order to design the lattice-based quantizers described above, we use a modified Linde-Buzo-Gray [32] algorithm. The design process is performed in a stage by stage manner, starting from the first stage with the LBG quantizer design for the first LSF parameter, and finishing with the last stage, i.e., the LBG quantizer design for the tenth LSF parameter. The weighted distance of (2) is also used in the design process. In fact, the codeword (centroid) update rule for the LBQ-LSFD scheme is given by

$$C_i(.,j) = \frac{1}{\sum\limits_{S_i(j)} w_i c_i} . \sum_{S_i(j)} w_i c_i \left(l_i - G_i(s)\right), \quad 1 \le i \le 10. \tag{15}$$

In (15), $S_i(j)$ is the set of all training values l_i, which are selected for encoding by $C_i(.,j)$. This is simply extended to the case of the LBQ-2LSFD. For the switched scheme, our experiments show that the design of the codebooks, while the LBQ-SWCH algorithm is used for quantization, as opposed to simply using the codebooks trained for LBQ-LSFD and LBQ-2LSFD slightly improves the performance, and therefore, we take this approach here. As discussed in the next section, an established measure for the performance evaluation of different LPC quantization schemes is the average spectral distortion [1]. Obviously, in this context, the quantizer design based on the weighted distance of (2) is subtpimal. However, due to its simplicity and effectiveness, using weighted Euclidean distance for LPC quantizer design is a popular approach.

IV. Performance Evaluation

The proposed lattice-based schemes for the quantization of LSF parameters are examined for two important attributes of every LPC quantization scheme, i.e., the quality of the encoded parameters and the encoding/decoding complexity. The complexity considerations consist of the computational complexity and the memory requirements.[2] Also, various performance comparisons with several other methods presented in the literature are provided.

A. Experiment Setup

In assessing the performance of different quantization schemes of the LSF parameters, the experimental setup is of vital importance. The factors that affect this setup and hence the simulation results include the training and test speech databases, speech preprocessing, LP analysis and objective measurement. In the literature, various objective measures of speech quality have been proposed [36]. The most popular approach for the evaluation of quantization quality of the LSF parameters is the spectral distortion [1]. However, the definition of the desired quality based on this measure still varies and depends on the frequency range over which this measure is calculated. In order to evaluate and compare the performance of different LSF quantizers, we need to simulate and test the system using a common experimental setup, and that is the approach taken in this work.

We use a training database of 175 726 LSF vectors derived from a 58.57 min long recorded speech (20 ms frame). This database contains a combination of clean speech and speech with background noise from a number of male and female speakers. Another outside test database of 102 400 LSF vectors derived from a 34.13-min-long recorded clean speech is used to test the performance of the quantizers.[3] The spectral distortion measure measured in the frequency range of $f_1 = 60$ Hz to $f_2 = 3500$ Hz) is employed to measure the objective quality of the quantized LPC coefficients. This criterion is a function of the distortion introduced in the power spectral density of speech in each particular frame. The spectral distortion in the nth frame is given by (16), shown at the bottom of the next page, in which

$$P^{(n)}(f) = \frac{1}{\left|A^{(n)}\left(\exp\left(\frac{j2\pi f}{F_s}\right)\right)\right|^2} \tag{17}$$

and

$$\hat{P}^{(n)}(f) = \frac{1}{\left|\hat{A}^{(n)}\left(\exp\left(\frac{j2\pi f}{F_s}\right)\right)\right|^2} \tag{18}$$

are the original and quantized power spectral density of the nth frame, respectively. The terms $A^{(n)}(z)$ and $\hat{A}^{(n)}(z)$ are the corresponding original and quantized LPC filters as described in Section II-A.

[2]In this work, the computational complexity is measured in number of floating point operations (flops). Each addition, multiplication or comparison is considered as one flop. The memory unit considered here is float. The number of codewords is equivalent to the number of floating point numbers needed to be stored.

[3]The speech databases used in this work are provided by Nortel Networks.

TABLE II
AVERAGE SPECTRAL DISTORTION, 2-dB OUTLIERS, CODEBOOK SIZE, AND COMPUTATIONAL COMPLEXITY FOR SCALAR QUANTIZATION OF LSF PARAMETERS

bit-rate	SD (dB)	outliers >2dB (%)	codebook (floats)	comp. (kflops/f)
26	1.75	28.83	64	0.19
30	1.40	9.39	80	0.24
34	1.06	2.47	112	0.34
40	0.75	0.24	160	0.48

We consider transparent quality to be achieved when the average spectral distortion is about 1 dB, and the fraction of 2-dB outliers is less than 2%. In our experiments, when this condition was valid the 4-dB outliers percentage was zero or negligible. Since our objective is to compare the performance of different *quantization* schemes, we used the same weights as described in (4) for all the systems considered. Nevertheless, our experiments showed that the proposed weight function results in lower spectral distortion than those of IS-641 [26] and Paliwal *et al.* [13].

B. Systems for Comparison: Scalar, Differential, Split-VQ, MSVQ, TCQ, BTQ, IS-641 EFRC, and GSM AMR

We consider eight schemes for comparison with our proposed lattice-based quantization schemes. Scalar quantization of LSF parameters is used as a baseline for comparison. We have also simulated the differential scalar quantization with a first-order linear prediction, the SVQ [13], MSVQ [12], TCQ [15], the interframe Block-based Trellis Quantization (BTQ) [17] and, the 3-part interframe Split-VQ as employed in the IS-641 EFRC [26] and the GSM AMR codec [28].

- Table II depicts the results of our simulation for the nonuniform scalar quantization of LSF parameters at different bit-rates. This simple approach is used in the federal standard FS-1016 [7] at the high rate of 34 bpf.
- Table III depicts the results of our simulation for the intraframe differential scalar quantization of LSF parameters at different bit-rates. A first-order prediction as described in Section III-C is used. This is similar to the works of Soong and Juang [6].
- Table IV presents the results of our simulation for the intraframe 2-part Split Vector Quantization of LSF parameters [13]. In this scheme, each LSF vector is split into two parts of (4,6) dimensions. Next, each part is quantized by using a full search vector quantizer. The bits are divided equally between the two parts, and for odd rates, the first part is given an extra bit. Although the transparent coding quality is achieved at 24 bpf, the complexity of Split-VQ is very high. At 24 bpf, it requires 164 000 floating point operations per frame to locate the appropriate codeword in a codebook of 40 960 codewords.

TABLE III
AVERAGE SPECTRAL DISTORTION, 2-dB OUTLIERS, CODEBOOK SIZE AND COMPUTATIONAL COMPLEXITY FOR DIFFERENTIAL SCALAR QUANTIZATION OF LSF PARAMETERS

bit-rate	SD (dB)	outliers >2dB (%)	codebook (floats)	comp. (kflops/f)
20	1.72	24.63	40	0.18
22	1.62	19.97	48	0.20
24	1.44	13.34	56	0.22
26	1.21	6.36	64	0.25
28	1.02	2.75	72	0.27

TABLE IV
AVERAGE SPECTRAL DISTORTION, 2-dB OUTLIERS, CODEBOOK SIZE, AND COMPUTATIONAL COMPLEXITY FOR 2-PART SPLIT-VQ OF LSF PARAMETERS

bit-rate	SD (dB)	outliers >2dB (%)	codebook (floats)	comp. (kflops/f)
22	1.16	3.00	20480	82
23	1.13	2.69	28672	114
24	1.05	1.27	40960	164
25	1.02	1.17	57344	229
26	0.95	0.59	81920	328
27	0.91	0.45	114688	459

TABLE V
CODEBOOK SIZE (ROM) AND COMPUTATIONAL COMPLEXITY OF TCQ WITH NONLINEAR PREDICTION

bit-rate	24
ROM (floats)	2560
comp. (kflops/f)	16.2

- Several LSF quantization schemes based on TCQ [16] were proposed in [15]. The best performance was achieved by a scheme denoted by TCQ-NLP which utilizes a 5-stage trellis with 2-D codebooks and nonlinear intraframe prediction. This scheme was reported to have a performance comparable with the 2-part Split-VQ. At 24 bpf, a 16-state trellis is used, and 4 bits are allocated to each stage [39]. Table V presents the complexity of the TCQ-NLP method at this rate. We observe that, this scheme is much less complex than the Split-VQ both in terms of the codebook size and the number of computations. As we will see in the next part, the proposed LBQ schemes offer an improved performance with a further substantial reduction of complexity.
- Table VI presents the performance results of the tree-searched MSVQ proposed in [12] for quantization of LSF parameters. The MSVQ is constructed with a number of full-search VQ stages each of which has a dimension of 10 and L_i quantization levels (VQ points). The VQ of the first stage quantizes the signal and those of the subsequent stages quantize the error corresponding to the previous

$$SD^{(n)} = \sqrt{\frac{1}{f_2 - f_1}\int_{f_1}^{f_2}\left[10\log_{10}\left(P^{(n)}(f)\right) - 10\log_{10}\left(\hat{P}^{(n)}\ (f)\right)\right]^2 df} \quad (16)$$

TABLE VI
AVERAGE SPECTRAL DISTORTION, 2-dB OUTLIERS, CODEBOOK SIZE (ROM), AND COMPUTATIONAL COMPLEXITY FOR TREE-SEARCHED MSVQ OF LSF PARAMETERS

MSVQ bit-rate (type)	M	SD (dB)	outliers >2dB (%)	ROM (floats)	comp. (kflops/f)
22 (2048-2)	1	1.05	1.70	40960	164
	2	0.99	0.98	40960	246
	4	0.97	0.80	40960	410
24 (64-4)	1	1.09	3.10	2560	10
	2	1.01	1.61	2560	18
	4	0.97	1.05	2560	33
	8	0.95	0.93	2560	64

TABLE VII
AVERAGE SPECTRAL DISTORTION, 2-dB OUTLIERS, CODEBOOK SIZE, AND COMPUTATIONAL COMPLEXITY FOR IS-641 OF LSF PARAMETERS

bit-rate	SD (dB)	outliers >2dB (%)	codebook (floats)	comp. (kflops/f)
22 (7,8,7)	1.20	4.06	1664	6.6
23 (7,8,8)	1.12	2.96	2176	8.7
24 (8,8,8)	1.07	2.37	2560	10.2
25 (8,9,8)	1.01	1.68	3328	13.3
26 (8,9,9)	0.95	1.20	4352	17.4
27 (9,9,9)	0.90	0.96	5120	20.5

stage. An $M - L$ tree-search is used to locate the appropriate codewords in the codebooks of different stages, where M is the number of preserved paths in each stage. The 22-bpf MSVQ achieves a similar performance to the 24-bpf Split-VQ with the same (high) level of complexity.

- Table VII presents the performance of the 3-part interframe Split-VQ of LSF parameters as employed in IS-641 and the GSM AMR codec. In this scheme, the LSF vector is split into three parts with the dimensions 3, 3, and 4. Also, a first-order moving average scalar linear predictor is employed whose coefficients have been recomputed with our training database. The selected bit-rate in IS-641 is 26 bpf, distributed as (8,9,9) bits among the three parts [26]. The same scheme is used in the GSM AMR codec [28] at several modes. Also a 27 bpf scheme with bit distribution (9,9,9) is used in the AMR codec for the 7.95-kb/s mode.
- In [17], based on a trellis modeling of the LSF parameters several interframe and intraframe coding schemes are proposed. The approach, referred to as the BTQ, is shown to outperform several intraframe and interframe schemes from the literature, including 2-Split VQ, interframe 3-Split VQ, MSVQ, and TCQ. The performance of the interframe BTQ scheme using our training and test database is provided in Table VIII.

C. Numerical Results

Tables IX–XI present the numerical results of respectively, the single frame, the double frame and, the switched lattice-based quantization of the LSF parameters at different bit-rates. Table XII provides the corresponding bit allocations.

TABLE VIII
AVERAGE SPECTRAL DISTORTION, 2-dB OUTLIERS, CODEBOOK SIZE, AND COMPUTATIONAL COMPLEXITY FOR INTERFRAME BTQ OF LSF PARAMETERS

average bit-rate	SD (dB)	outliers >2dB (%)	codebook (floats)	comp. (kflops/f)
21	1.21	5.10	1731	5.7
22	1.17	4.00	2463	8.0
23	1.07	2.72	2289	7.4
24	1.02	1.96	3468	11.2
25	0.96	1.54	3843	12.3
26	0.90	1.21	5050	16.3

TABLE IX
AVERAGE SPECTRAL DISTORTION, 2-dB OUTLIERS, CODEBOOK SIZE, AND COMPUTATIONAL COMPLEXITY FOR SINGLE-FRAME LBQ-LSFD OF LSF PARAMETERS

bit-rate	SD (dB)	outliers >2dB (%)	codebook (floats)	comp. (kflops/f)
20	1.58	17.39	44	0.30
21	1.56	16.57	48	0.31
22	1.51	15.49	52	0.33
23	1.40	11.57	56	0.34
24	1.27	7.17	60	0.36
25	1.19	5.54	64	0.38
26	1.08	3.32	68	0.39
27	1.00	2.25	72	0.41

TABLE X
AVERAGE SPECTRAL DISTORTION, 2-dB OUTLIERS, CODEBOOK SIZE, AND COMPUTATIONAL COMPLEXITY FOR DOUBLE-FRAME LBQ-2LSFD OF LSF PARAMETERS

bit-rate	SD (dB)	outliers >2dB (%)	codebook (floats)	comp. (kflops/f)
20	1.24	6.25	352	0.85
21	1.19	5.25	416	0.98
22	1.09	3.15	480	1.10
23	1.01	1.93	544	1.23
24	0.95	1.37	608	1.36

TABLE XI
AVERAGE SPECTRAL DISTORTION, 2-dB OUTLIERS, CODEBOOK SIZE, AND COMPUTATIONAL COMPLEXITY FOR LBQ-SWCH OF LSF PARAMETERS. THE BIT ALLOCATION TO THE THREE CODEBOOKS (CB1, CB2, CB3) AS SHOWN IN FIG. 2 IS ALSO INDICATED

bit-rate (CB1; CB2,CB3)	SD (dB)	outliers >2dB (%)	codebook (floats)	comp. (kflops/f)
20 (39; 19,20)	1.24	5.05	404	1.07
21 (41; 20,21)	1.19	4.20	476	1.22
22 (43; 21,22)	1.12	3.08	548	1.36
23 (45; 22.23)	1.04	1.86	620	1.50
24 (47; 23,24)	0.96	1.13	692	1.65

Our experiments with the LBQ-SWCH scheme show that, using spectral distortion as opposed to weighted distance to choose (switch) between the double-frame and single-frame quantization slightly improves the performance at the cost of an increase of computational complexity.

TABLE XII
BIT ALLOCATIONS TO DIFFERENT LBQ-LSFD AND LBQ-2LSFD DIMENSIONS AT DIFFERENT BIT RATES

bit-rate (bpf)	LBQ-LSFD (bpf/dim)	LBQ-2LSFD (bits/2frames/dim)
20	(2 2 2 2 2 2 2 2 2 2)	(4 4 4 4 4 4 4 4 4 4)
21	(3 2 2 2 2 2 2 2 2 2)	(5 5 4 4 4 4 4 4 4 4)
22	(3 3 2 2 2 2 2 2 2 2)	(5 5 5 5 4 4 4 4 4 4)
23	(3 3 3 2 2 2 2 2 2 2)	(5 5 5 5 5 5 4 4 4 4)
24	(3 3 3 3 2 2 2 2 2 2)	(5 5 5 5 5 5 5 5 4 4)

Compared to the differential scalar quantization of LSF parameters, the LBQ-LSFD scheme achieves comparable performance with two bpf reduction of the rate. The memory requirement is the same and the computational complexity is only slightly higher. By exploiting the interframe memory, the double-frame lattice-based quantization scheme (LBQ-2LSFD) saves an additional 4 bpf. The switched scheme still improves the performance by reducing the percentage of outliers.

At 24 bpf, the proposed LBQ-SWCH achieves an improved performance compared to the 2-part Split-VQ (Table IV), at significantly lower level of complexity. It requires only 1650 floating point operations/frame to search for the corresponding codevector in a codebook of 692 floating point codewords. This denotes almost 100 times reduction of the computational complexity and 60 times reduction of the memory requirement (codebook size).

Compared to the 24 bpf TCQ-NLP [15], as discussed above, the proposed LBQ-SWCH algorithm achieves an improved performance, while reducing the search complexity by a factor of ten and the codebook size by almost 70% (Table V). Also, compared to the 24-bpf MSVQ $(M = 4)$ (Table VI), the LBQ-SWCH achieves a comparable performance with almost 20 times smaller complexity and 70% smaller codebook size. It is worth mentioning that this configuration of MSVQ (24 bpf with four stages of 64 codevector each) was recognized in [12] as "one of the best [MSVQ] configurations in terms of the trade-off between complexity and performance".

Comparing to the 26-bpf interframe 3-part Split-VQ used in IS-641 and the AMR (Table VII), the proposed switched LBQ achieves a comparable performance at 24 bpf, while reducing the computational complexity and the memory requirement by a factor of ten and six, respectively.

Comparing the performance of the LBQ-SWCH scheme in Table XI, with that of the interframe BTQ scheme in Table VIII, it is observed that at each bitrate an improved performance is achieved with a noticeable reduction of the computational complexity and a smaller codebook size. The performance improvement at higher bitrates (24–25 bpf) amounts to one bpf.

Tables XIII–XV present the performance of the proposed lattice-based quantizers over a binary symmetric channel. The results indicate that they compare favorably to the 2-Split VQ, when transmitted over a noisy channel. Note that this performance is achieved using a binary index assignment, where due to the lattice constraint the least significant bit of the LSF10 quantizer index is not transmitted (see Section III). In these simulations, no form of unequal error protection or error concealment is employed.

TABLE XIII
AVERAGE SPECTRAL DISTORTION AND 2-dB OUTLIERS FOR QUANTIZATION OF LSF PARAMETERS USING DIFFERENT SCHEMES FOR $P_e = 0.05$

bit-rate	LBQ-2LSFD		LBQ-SWCH		Split-VQ	
	SD	OL	SD	OL	SD	OL
20	3.47	68.34	3.35	65.69	3.93	62.99
21	3.56	69.47	3.45	66.92	3.99	63.50
22	3.57	68.86	3.50	67.45	4.04	63.82
23	3.58	68.58	3.52	67.44	4.10	64.30
24	3.56	68.31	3.51	66.83	4.11	64.78

TABLE XIV
AVERAGE SPECTRAL DISTORTION AND 2-dB OUTLIERS FOR QUANTIZATION OF LSF PARAMETERS USING DIFFERENT SCHEMES FOR $P_e = 0.01$

bit-rate	LBQ-2LSFD		LBQ-SWCH		Split-VQ	
	SD	OL	SD	OL	SD	OL
20	1.80	24.86	1.78	23.35	1.92	22.32
21	1.78	24.44	1.76	23.09	1.91	22.00
22	1.72	22.79	1.73	22.50	1.85	20.16
23	1.67	21.83	1.67	21.50	1.85	20.35
24	1.63	21.49	1.63	20.94	1.75	18.45

TABLE XV
AVERAGE SPECTRAL DISTORTION AND 2-dB OUTLIERS FOR QUANTIZATION OF LSF PARAMETERS USING DIFFERENT SCHEMES FOR $P_e = 0.005$

bit-rate	LBQ-2LSFD		LBQ-SWCH		Split-VQ	
	SD	OL	SD	OL	SD	OL
20	1.52	15.83	1.48	14.39	1.61	14.75
21	1.49	15.07	1.44	13.77	1.58	14.22
22	1.40	13.30	1.41	13.25	1.51	11.94
23	1.35	12.32	1.34	12.08	1.50	11.94
24	1.30	11.77	1.28	11.21	1.43	10.89

V. CONCLUSIONS

A lattice-based scheme for the single-frame and double-frame quantization of the speech LSF parameters is proposed. The intraframe and interframe dependencies are exploited using a linear predictor, and through vector quantization of the parameters of two consecutive frames, respectively. The lattice structure provides a low-complexity vector quantization framework. A switched scheme is also considered in which lattice-based double-frame and single-frame quantization is performed for each two frame and the one which results in a lower distortion is chosen. Numerical results demonstrate an excellent performance with very low complexity, compared to some of the previously known methods from the literature. A possible next step is to devise a more sophisticated bit allocation algorithm for the proposed lattice-based quantizers that takes into account the statistics of different parameters and their corresponding weights.

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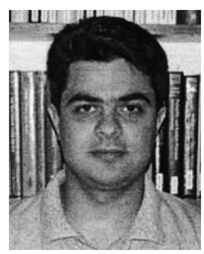

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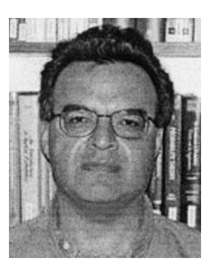

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