

Adaptive Forward-Backward Quantizer for Low Bit Rate High Quality Speech Coding

Jozsef Vass *Yunxin Zhao*[†] *Xinhua Zhuang*

*Department of Computer Engineering & Computer Science
University of Missouri-Columbia
Columbia, MO 65211*

[†]*Beckman Institute and Department of Electrical & Computer Engineering
University of Illinois
Urbana, IL 61801*

Abstract

A novel variable rate linear predictive coding (LPC) parameter quantization scheme is proposed in which linear prediction is done by using either the current (forward LPC) or previously decoded (backward LPC) speech blocks. The proposed LPC quantization scheme was integrated into the FS1016 Federal Standard CELP coder. Significant LPC bit rate reduction is achieved without compromising the decoded speech quality.

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EDICS Category: SA 1.4.2 Analysis-by-Synthesis Coding
Tel. (573) 882-2382
Fax. (573) 882-8318
E-mail: zhuang@ece.missouri.edu

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1 Introduction

Linear prediction plays a central role in various low and intermediate bit rate speech coding algorithms [1]. Usually, a new set of linear predictive coding (LPC) coefficients is determined every 20 to 30 ms and, after quantization, transmitted to the decoder as side information. To reduce the degradation of the speech quality caused by a direct quantization of LPC coefficients, Line Spectral Pairs (LSP) parameters are used for an indirect quantization and interpolation of predictor coefficients. Traditionally, scalar quantization of LSP coefficients was used. For example, in FS1016 Federal Standard Code Excited Linear Predictive (CELP) coder [2] a total of ten LSP coefficients are scalarly quantized to 34 bits-per-frame (bpf). Since the predictor coefficients are updated every 30 ms, the side information required for transmitting LSP parameters needs 1133.3 bits-per-second (BPS). The overall bit rate of FS1016 coder is 4.8 kbps, so more than 23% of the required bandwidth is spent on transmission of LSP coefficients.

One possibility to reduce the bit rate spent on transmission of LPC coefficients is the application of vector quantization schemes (instead of scalar quantization) for quantization of LSP coefficients. By applying split VQ method, Paliwal and Atal [3] reported 24 bpf for quantization of LSP parameters and their results were superior to the 34 bpf scalar LSP quantizer used in FS1016.

In virtually all published CELP algorithms, predictor coefficients are determined based on the current speech frame by using the so-called forward linear prediction [4]. The disadvantages of forward linear prediction include: i) exclusive transmission of predictor coefficients, increasing the required bandwidth; and ii) extensive data buffering, yielding a large coding delay. In the Low-Delay CELP coder (G.728) [5], backward linear prediction is used to overcome these disadvantages. In this scheme, the predictor coefficients are determined by using previously decoded speech samples, available at both the encoder and decoder. The main advantages of the scheme are that neither buffering of speech samples (the overall coding delay by the Low-Delay CELP coder is less than 2 ms compared to a 50 to 60 ms delay by a conventional CELP coder [5]), nor transmission of LPC parameters are needed. However, the quality of forward linear prediction is usually superior to backward linear prediction [4]. In the Low-Delay CELP coder, in order to ensure high performance backward linear prediction, good waveform matching is needed, requiring the excitation sequences to be updated every 0.625 ms (thus, the LPC coefficients are updated every 2.5 ms). This is about ten times faster than the conventional CELP coder, resulting in an undesirable increase of the bit rate to 16 kbps.

As is well known, the speech signal is often slowly time-varying and nonstationary. The statistics between the current block and some temporally close previous blocks may often be similar,

leading to close sets of predictor coefficients. A method termed Long History Quantization (LHQ) was proposed based on this idea (see Xydeas and So [6]). By allowing previous blocks to be overlapped, the chance for statistical matching between the current block, and one of the so constructed temporally close previous blocks, will surely increase. By adapting the quantizer design to this new strategy, the “global” statistical correlation of speech signals will be more thoroughly exploited and a significant bit rate decrease is expected. To exploit the advantages of both forward and backward linear prediction, we propose the following adaptive forward-backward coding scheme: A previously decoded and temporally close speech signal is segmented into overlapping blocks. If, and only if, the LPC coefficients calculated from one of those synthetic blocks is sufficiently “close” in some sense to the *unquantized* LPC coefficients calculated from the current speech block, the backward LPC scheme shall be applied, i.e., the LPC coefficients based on the previously decoded optimal speech block are used to encode the current block and only the time delay shall be transmitted to the decoder.

In this paper, logarithmical spectral distortion (LSD) measure is used to evaluate the similarity between two different sets of predictor coefficients. Depending on the LSD measure between the current and previous block, either forward or backward linear prediction is selected. In the case of forward linear prediction, a new set of predictor coefficients still needs to be transmitted. But, in the case of backward linear prediction, a significant reduction of the required bandwidth for transmission of LSP coefficients is attained because only the time index is to be transmitted. This leads to a variable rate coding scheme. By applying the proposed method to the FS1016 CELP coder, only about 10% of the required bandwidth would be used for transmission of predictor coefficients, resulting in a much lower average bit rate of 4.08 kbps.

The organization of the paper is as follows. Section 2 introduces our quantization scheme of LSP coefficients. Performance evaluation of the new algorithm is given in Section 3. Conclusions and further research directions are given in Section 4.

2 Adaptive Forward-Backward Quantization of Predictor Coefficients

As usual, the input speech is divided into non-overlapping blocks of L samples. For each block, the LPC coefficients are determined by using, e.g., the Levinson-Durbin algorithm. These LPC coefficients, i.e., a_1, \dots, a_p , are optimal for the current speech block in the sense that the energy of the prediction residual signal is minimized. In traditional CELP coders, the LPC coefficients based solely on the current block are quantized by using either scalar or vector quantization scheme. In the following, we describe our *adaptive forward-backward quantizer*.

We start by defining the adaptive forward-backward LPC codebook, which consists of S code vectors each having p entries, where p represents the order of linear predictor. The i th code vector is determined by calculating LPC coefficients, i.e., $\hat{a}_1^{(i)}, \dots, \hat{a}_p^{(i)}$, based upon the previously decoded (synthetic) speech block $[y_{n-iK-L}, y_{n-iK-L+1}, \dots, y_{n-iK-1}]$ that is available at both the encoder and decoder (see Fig. 1), where L is the length of the LPC block and K is the time delay unit chosen to be equal to the length of the sub-block, i.e., $K = N$.

We then use logarithmical spectral distortion (LSD) measure to evaluate the similarity between the previous and current set of LPC coefficients defined above. The LSD introduced by the i th code vector of the adaptive forward-backward LPC codebook is given by

$$\text{LSD}^{(i)}[\text{dB}] = \frac{10}{\ln 10} \sqrt{\frac{1}{2\pi} \int_{-\pi}^{\pi} |V^{(i)}(\omega)|^2 d\omega}, \quad i = 0, \dots, S-1, \quad (1)$$

with $V^{(i)}(\omega)$ being defined by

$$V^{(i)}(\omega) = \ln \frac{1}{|A(\omega)|^2} - \ln \frac{1}{|\hat{A}^{(i)}(\omega)|^2}, \quad i = 0, \dots, S-1, \quad (2)$$

where $A(z) = 1 + \sum_{l=1}^p a_l z^{-l}$ and $\hat{A}^{(i)}(z) = 1 + \sum_{l=1}^p \hat{a}_l^{(i)} z^{-l}$. As usual, a_1, \dots, a_p denote the LPC coefficients based solely on the current speech block.

As seen, the LSD measure is determined for every candidate code vector. Then the one that has the smallest spectral distortion, i.e., $\text{LSD}^{(index)}$ with $index = \arg \min_i \text{LSD}^{(i)}$, is selected from the adaptive LPC codebook. If $\text{LSD}^{(index)} > T$, a predefined threshold, then the current LPC coefficients, i.e., a_1, \dots, a_p , are used in speech coding and, after quantization, transmitted to the decoder. If $\text{LSD}^{(index)} \leq T$, then the corresponding LPC coefficients, i.e., $\hat{a}_1^{(index)}, \dots, \hat{a}_p^{(index)}$, are used in speech coding and only the $index$ to the adaptive LPC codebook needs to be transmitted to the decoder. An additional flag bit is required to notify the decoder whether forward or backward linear prediction is applied at the encoder.

The application of the proposed algorithm slightly increases the computational complexity at both the encoder and decoder. At the encoder, the increase in computational complexity is two-fold. First, for every new block, the code vectors in the adaptive LPC codebook need to be updated by the use of the newly decoded (synthetic) speech samples. However, each time, only $L/K = 4$ code vectors which involve the most recently determined synthetic speech samples need to be calculated and added to the adaptive LPC codebook to replace the oldest code vectors (see Fig. 1). Second, the optimal code vector, which has the smallest LSD, needs to be selected from the adaptive LPC codebook. However, the computational complexity raised by the proposed algorithm at the encoder is negligible compared to the closed-loop excitation sequence generation of the CELP algorithm. This is because i) each time only the four newest LPC code vectors are calculated to replace the

four oldest ones and ii) instead of the LSD measure we have used the computationally less expensive COSH measure [7], which is an upper bound of the LSD measure. At the decoder, if backward linear prediction is applied, the LPC coefficients are determined based on the previously decoded speech samples $[y_{n-indexK-L}, y_{n-indexK-L+1}, \dots, y_{n-indexK-1}]$.

The adaptive forward-backward quantization of the LPC coefficients is summarized as follows. At the encoder:

- Step 1.** Calculate the LPC coefficients, i.e., a_1, \dots, a_p , based on the current speech block.
- Step 2.** Update the adaptive LPC codebook by replacing the $L/K = 4$ oldest code vectors by the four newest ones, i.e., $\hat{a}_1^{(i)}, \dots, \hat{a}_p^{(i)}$ with $i = 0, \dots, L/K - 1$, that are based on the most recently decoded speech block $[y_{n-iK-L}, y_{n-iK-L+1}, \dots, y_{n-iK-1}]$ as shown in Fig. 1.
- Step 3.** Calculate the logarithmical spectral distortion measure for each code vector in the adaptive LPC codebook. Instead of directly calculating the LSD measure by using expressions (1) and (2), use the COSH measure.
- Step 4.** Select the code vector from the adaptive LPC codebook that has the minimal spectral distortion, i.e., $LSD^{(index)}$ with $index = \arg \min_i LSD^{(i)}$.
- Step 5.** If $LSD^{(index)} \leq T$, a predefined threshold, then the LPC coefficients determined in **Step 4**, i.e., $\hat{a}_1^{(index)}, \dots, \hat{a}_p^{(index)}$, are used for coding the current speech block and the *index* is encoded and transmitted to the decoder. Set the flag bit to 0 to inform the decoder that backward linear prediction is applied at the encoder. Go to **Step 7**.
- Step 6.** If $LSD^{(index)} > T$, then the LPC coefficients based on the current block (determined in **Step 1**) are used for coding the current speech block and, after scalar or vector quantization, transmitted to the decoder. The flag bit is set to 1 to inform the decoder that forward linear prediction is applied at the decoder.
- Step 7.** Encode the speech by using the LPC coefficients calculated for the current speech block in either **Step 1** or **Step 4**.

At the decoder:

- Step 1.** If backward linear prediction is applied at the encoder (the received flag bit is 0), determine the LPC coefficients based on the previously decoded speech samples $[y_{n-indexK-L}, y_{n-indexK-L+1}, \dots, y_{n-indexK-1}]$. Go to **Step 3**.

Step 2. If the flag bit shows that forward linear prediction is applied at the encoder, receive the current LPC coefficients.

Step 3. Decode the speech by using the LPC coefficients determined in either **Step 1** or **Step 2**.

3 Performance Evaluation

Extensive computer experiments are conducted to evaluate both the objective and subjective performance of the proposed algorithm. The speech database used for objective performance evaluation contains 600 seconds of speech spoken by both male and female speakers [8]. Subjective performance evaluation is done on eight sentences spoken by both male and female speakers.

We used segmental signal-to-noise ratio (segSNR) and logarithmical spectral distortion to evaluate the objective performance of the proposed algorithm. Other objective performance measures are the bandwidth used for transmitting predictor coefficients, the resulting overall bit rate of the coder, and the percentage of the bandwidth used for transmitting LSP coefficients.

Experiments are conducted by integrating the proposed adaptive forward-backward LSP quantization scheme into the FS1016 Federal Standard CELP coder [2]. Performance evaluation is done by varying the size of the adaptive LPC codebook S , the block length L , and the threshold T . Fig. 2 shows the distribution of the codebook indices to be sent when the adaptive LPC codebook has $S = 64$ entries. In most cases, the LPC coefficients based on several of the most recently decoded speech blocks are more likely to be selected. This implies that the adaptive LPC codebook should consist of a relatively small number of code vectors. Thus, the size of the adaptive LPC codebook varies from $S = 1$ (0 bit is required to specify the time delay) to $S = 128$. Increasing the size of the adaptive LPC codebook for a given threshold T increases the frequency of applying backward LPC analysis versus forward LPC analysis (see Fig. 4) resulting in reduction of bandwidth used for transmission of LPC coefficients. However, it also increases the spectral distortion (see Fig. 3). Fig. 5 and 6 show the LSD and segSNR as the average LPC bit rate changes. In terms of the LSD measure, codebooks with a large number of entries are preferable since for a given bit rate an adaptive LPC codebook with $S = 128$ entries gives the smallest spectral distortion and an adaptive LPC codebook with $S = 1$ entries gives the largest spectral distortion (see Fig. 5). In terms of segSNR, an adaptive LPC codebook with $S = 1$ code vector performs the best. Adaptive LPC codebooks with $S > 2$ entries have similar performance. Since, at low bit rates, LSD measure is a more meaningful measure of the decoded speech quality than the segSNR, the adaptive LPC codebook should contain from $S = 16$ to $S = 128$ entries. On the other hand, adaptive LPC codebooks with $S > 128$ are impractical due to computational complexity.

Block lengths of both 30 ms and 20 ms are used, which correspond to $L = 240$ and $L = 160$ samples, respectively, when the sampling rate is $f_s = 8$ kHz. As is well known, the performance of the CELP coding technique depends on the block length. As a matter of fact, both the segSNR and the decoded speech quality improve as the coder parameters are updated more frequently. The results shown in Table 1 and Table 2 show 1–1.5 dB improvement in the segSNR when the block length is reduced from 30 ms to 20 ms.

The threshold varies from $T = 3$ dB to $T = 6$ dB in 0.5 dB increments. As the threshold increases, more spectral distortion is tolerated resulting in the reduction of bandwidth used for transmitting LSP coefficients (see Table 1, Table 2, and Fig. 4). It is clear that $T \rightarrow \infty$ corresponds to the case when exclusively backward linear prediction is applied and $T = 0$ corresponds to the case when solely forward linear prediction is used. Since exclusive application of backward linear prediction results in unacceptably low speech quality, this case is not considered any more. In the following, forward linear prediction ($T = 0$) serves as a baseline to evaluate the performance of the proposed adaptive forward-backward quantization scheme of the predictor coefficients.

The results of objective performance evaluation are shown in Tables 1 and 2 for block lengths 20 ms and 30 ms, from $T = 3.0$ to $T = 6.0$ dB, when the adaptive LPC codebook contain $S = 1$ and $S = 128$ entries, respectively.

Subjective performance evaluation is performed by informal comparison tests using six listeners. Eight sentences spoken by both male and female speakers are chosen from the TIMIT database. The size of the LPC block is 20 ms, i.e., $L = 160$ samples. The adaptive forward-backward LPC codebook contains $S = 16$ code vectors. The threshold varies from $T = 4$ dB to $T = 6$ dB in 0.5 dB increments. Listeners were asked to compare the decoded speech obtained by the FS1016 CELP coder ($T = 0$) and by the application of adaptive forward-backward quantization scheme. The test sentence and the threshold were randomly selected, so it is possible that listeners had to compare the same decoded sentences more than once. The results are summarized in Tables 3 and 4. Table 3 shows for each given sentence and threshold T , in what percentage of cases listeners preferred the decoded results obtained by FS1016 CELP coder, the proposed adaptive forward-backward quantization scheme, or judged that the two decoded sentences are indistinguishable. Table 4 summarizes the results of the subjective listening tests over the eight sentences. In all the cases, listeners judged that the decoded sentences obtained by FS1016 CELP and adaptive forward-backward quantization have the same subjective quality. As a result, we can state that the decoded speech obtained by the application of the proposed adaptive forward-backward LPC quantization scheme is statistically indistinguishable from the decoded speech generated by the FS1016 CELP coder. Thus, substantial bit rate reduction is achieved without compromising the

decoded speech quality.

Finally, Fig. 7 shows the comparison of the proposed adaptive forward-backward quantization scheme with long history quantization (LHQ) [6]. The proposed quantization scheme slightly outperforms LHQ. Another advantage of the proposed quantization scheme over LHQ is that the order of linear prediction can be higher when backward linear prediction is applied. Applying $p = 12$ order backward LPC analysis increases the segSNR by 0.2 dB.

Integrating the proposed adaptive forward-backward quantization scheme of predictor coefficients into the FS1016 Federal Standard CELP coder results in a significant reduction of the bandwidth required for transmitting LSP coefficients from 34 bpf (1133 BPS) to 12.4 bpf (413 BPS) maintaining high decoded speech quality. This means that the overall bit rate of the coder is reduced from 4.8 kBPS to 4.08 kBPS. As shown in Table 2, 10.1% of the overall bit rate is spent on transmission of predictor coefficients compared to 23.6% of the traditional FS1016 CELP coder.

4 Conclusions and Further Research Directions

In this paper, we have introduced an adaptive forward-backward LSP quantization scheme. The proposed variable rate quantization technique adapts to the local statistics of the signal resulting in a significant reduction of bandwidth required for transmitting LSP coefficients. The algorithm has been integrated into the FS1016 Federal Standard CELP coder. Extensive computer experiments showed that the bandwidth required for transmission of predictor coefficients was reduced by a factor of 2.7 with less than 1 dB drop in the segmental SNR and virtually no degradation in the perceived speech quality. Currently, we are combining the adaptive forward-backward quantization scheme with vector quantization. As our primary interest is to apply the proposed quantization scheme to mobile communication, special emphasis will be placed on investigation of the effects of channel errors and/or lost packets.

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References

- [1] A. Gersho, “Advances in speech and audio compression,” *Proceedings of IEEE*, vol. 82, pp. 900–918, 1994.
- [2] J.P. Campbell, V.C. Welch, and T.E. Tremain, “An expandable error-protected 4800 BPS CELP coder (U.S. Federal Standard 4800 BPS voice coder),” in *Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing*, 1989, pp. 735–738.
- [3] K.K. Paliwal and B.S. Atal, “Efficient vector quantization of LPC parameters at 24 bits/frame,” *IEEE Transactions on Speech and Audio Processing*, vol. 1, no. 1, pp. 3–14, Jan. 1993.
- [4] N.S. Jayant and P. Noll, *Digital Coding of Waveforms*, Prentice-Hall, Englewood Cliffs, NJ, 1984.
- [5] J. Chen, R. Cox, Y. Lin, N. Jayant, and M. Melchner, “A low-delay CELP coder for the CCITT 16kb/s speech coding standard,” *IEEE Journal on Selected Areas Communications*, vol. 10, pp. 830–849, June 1992.
- [6] C.S. Xydeas and K.K.M. So, “A long history quantization approach to scalar and vector quantization of LSP coefficients,” in *Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing*, 1993, vol. 2, pp. 1–4.
- [7] A.H. Gray and J.D. Markel, “Distance measures for speech processing,” *IEEE Transactions on Acoustics, Speech, and Signal Processing*, vol. 24, pp. 380–391, Oct. 1976.
- [8] R.C. Cole, M. Fanty, M. Noel, and T. Lander, “Telephone speech corpus development at CSLU,” in *Proceedings of IEEE International Conference on Spoken Language Processing*, Sept. 1994, pp. 1–3.

T [dB]	Block size=20 ms					Block size=30 ms				
	LPC Rate [bpf]	Overall Rate [BPS]	LPC %	segSNR [dB]	LSD [dB]	LPC Rate [bpf]	Overall Rate [BPS]	LPC %	segSNR [dB]	LSD [dB]
0	34.0	7200	23.6	11.83	1.59	34.0	4800	23.6	10.41	1.56
3.0	29.1	6956	20.9	11.45	1.73	30.5	4685	21.7	10.28	1.67
3.5	25.7	6788	18.9	11.22	1.91	28.0	4600	20.2	10.22	1.81
4.0	22.4	6620	16.9	11.01	2.13	24.9	4498	18.4	10.10	2.03
4.5	19.2	6460	14.8	10.87	2.41	22.1	4406	16.7	10.01	2.28
5.0	16.3	6319	12.9	10.75	2.72	19.4	4315	15.0	9.96	2.57
5.5	14.0	6202	11.3	10.67	3.02	16.7	4226	13.2	9.94	2.91
6.0	12.0	6102	9.8	10.61	3.33	14.3	4145	11.5	9.90	3.27

Table 1. Objective performance measures with adaptive LPC codebook size of $S = 1$.

T [dB]	Block size=20 ms					Block size=30 ms				
	LPC Rate [bpf]	Overall Rate [BPS]	LPC %	segSNR [dB]	LSD [dB]	LPC Rate [bpf]	Overall Rate [BPS]	LPC %	segSNR [dB]	LSD [dB]
0	34.0	7200	23.6	11.83	1.59	34.0	4800	23.6	10.41	1.56
3.0	21.4	6571	16.2	10.90	1.95	22.3	4412	16.8	9.65	1.92
3.5	17.3	6364	13.5	10.82	2.22	18.2	4275	14.2	9.52	2.20
4.0	13.8	6193	11.1	10.76	2.51	14.8	4162	11.9	9.49	2.48
4.5	11.5	6078	9.5	10.70	2.75	12.4	4082	10.1	9.45	2.75
5.0	9.9	5997	8.2	10.67	2.96	10.6	4023	8.8	9.38	2.98
5.5	8.9	5945	7.4	10.63	3.11	9.4	3983	7.9	9.39	3.17
6.0	8.1	5909	6.9	10.61	3.24	8.7	3958	7.35	9.40	3.33

Table 2. Objective performance evaluation with adaptive LPC codebook size $S = 128$.

Sentence	Preference	Threshold T in [dB]				
		4	4.5	5	5.5	6
sx78	FS1016 CELP	36%	40%	25%	9%	55%
	AFBQ-CELP	64%	20%	25%	36%	11%
	No difference	0%	40%	50%	55%	34%
sx65	FS1016 CELP	36%	16%	14%	30%	14%
	AFBQ-CELP	64%	0%	28%	10%	14%
	No difference	0%	84%	58%	60%	72%
sx198	FS1016 CELP	50%	50%	50%	50%	66%
	AFBQ-CELP	0%	33%	0%	0%	16%
	No difference	50%	17%	50%	50%	18%
sx204	FS1016 CELP	33%	50%	25%	34%	25%
	AFBQ-CELP	16%	0%	25%	0%	25%
	No difference	51%	50%	50%	66%	50%
sx57	FS1016 CELP	20%	40%	12%	33%	42%
	AFBQ-CELP	40%	40%	25%	22%	16%
	No difference	40%	20%	63%	45%	42%
sx221	FS1016 CELP	80%	60%	50%	33%	50%
	AFBQ-CELP	0%	0%	38%	25%	38%
	No difference	20%	40%	12%	42%	12%
sx308	FS1016 CELP	50%	50%	20%	37%	37%
	AFBQ-CELP	0%	0%	30%	13%	0%
	No difference	50%	50%	50%	50%	63%
sx93	FS1016 CELP	0%	0%	12%	12%	12%
	AFBQ-CELP	50%	50%	25%	25%	25%
	No difference	50%	50%	63%	63%	63%

Table 3. Result of subjective listening tests for each test sentence.

Preference	Threshold T in [dB]				
	4	4.5	5	5.5	6
FS1016 CELP	35%	38%	27%	31%	37%
AFBQ-CELP	29%	18%	25%	17%	18%
No difference	36%	44%	48%	52%	45%

Table 4. Summary of subjective listening tests over the eight sentences.

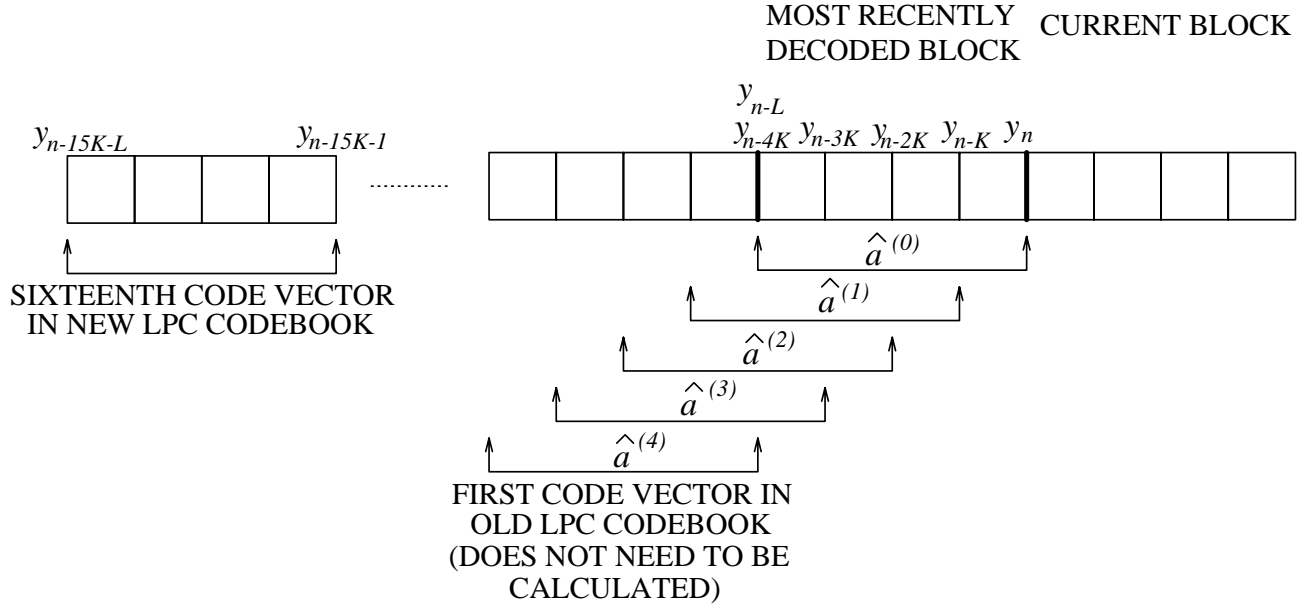


Figure 1. Algorithm for updating the adaptive LPC codebook.

$$\hat{a}^{(i)} = [\hat{a}_1^{(i)}, \dots, \hat{a}_p^{(i)}].$$

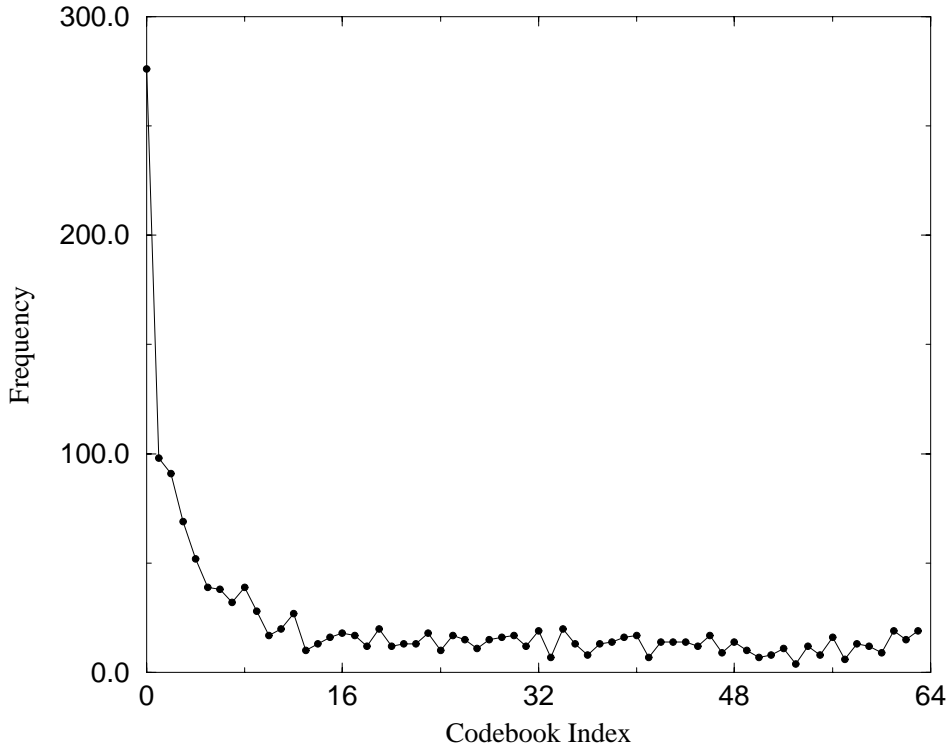


Figure 2. Distribution of the adaptive codebook indices. The threshold was $T = 5$ dB. The LPC coefficients were sent 677 times (31%), the codebook index was sent 1498 times (69%).

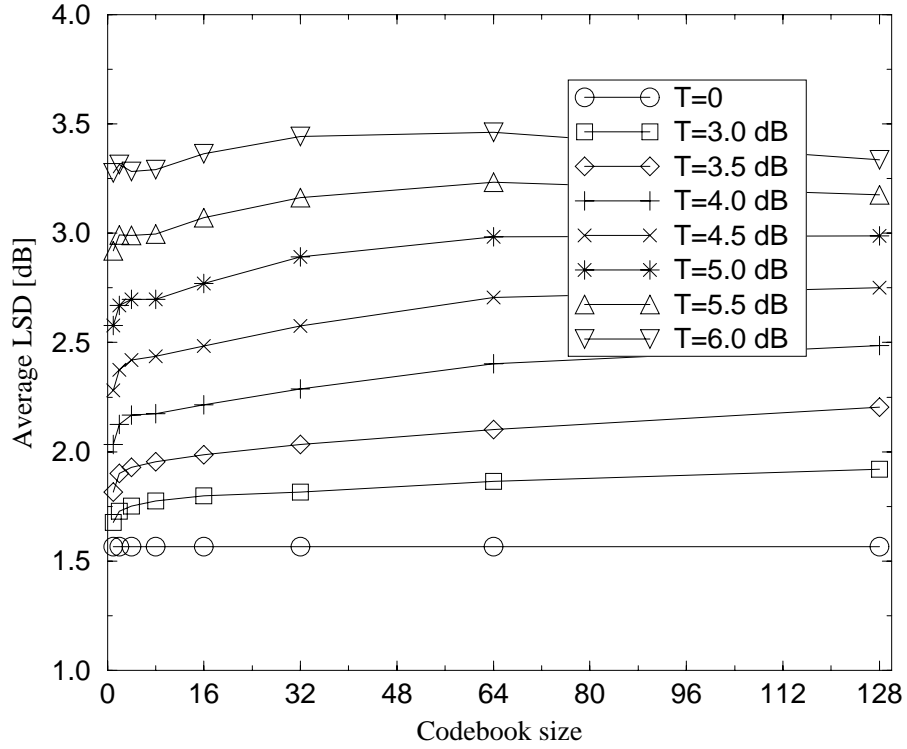


Figure 3. Change of average LSD due to different size of adaptive LPC codebook, $L = 240$.

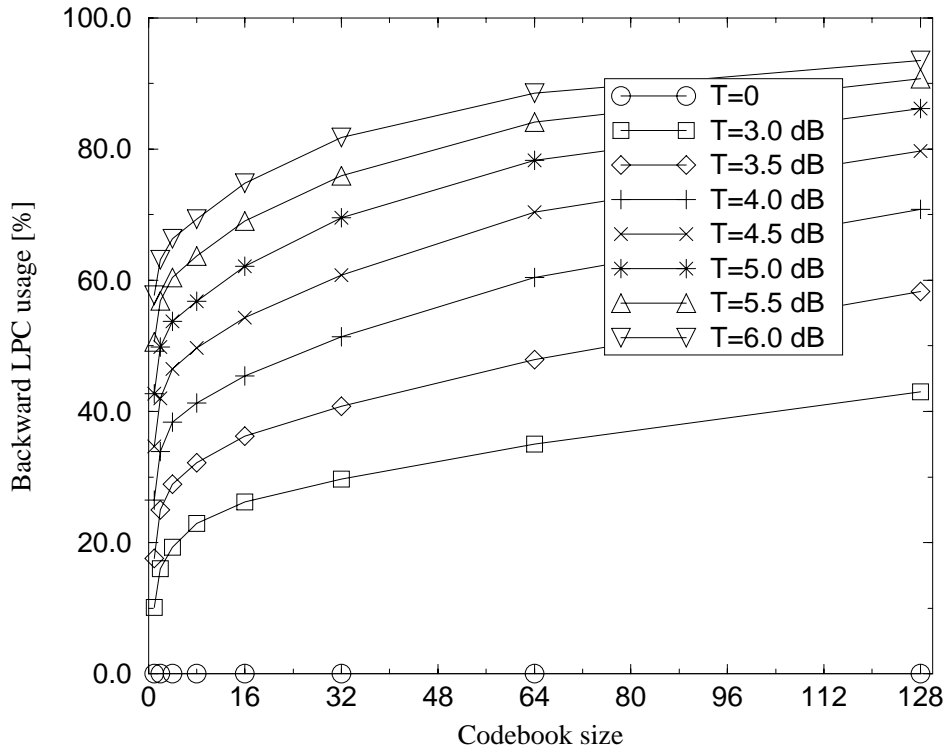


Figure 4. Percentage of backward LPC due to different size of adaptive LPC codebook, $L = 240$.

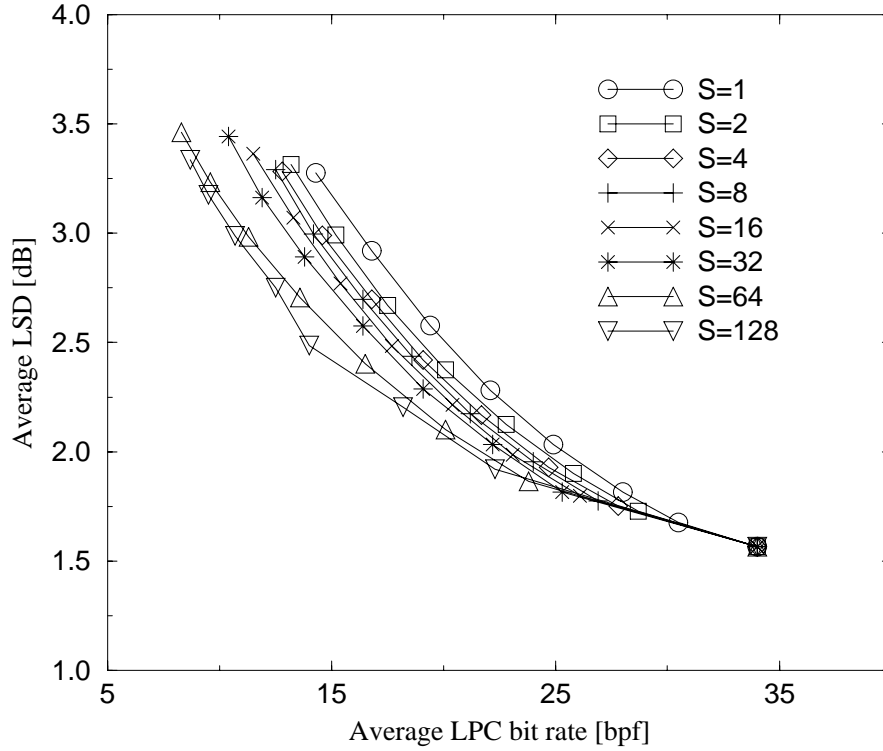


Figure 5. Average LSD introduced at different bit rates with different size of adaptive LPC codebook, $L = 240$.

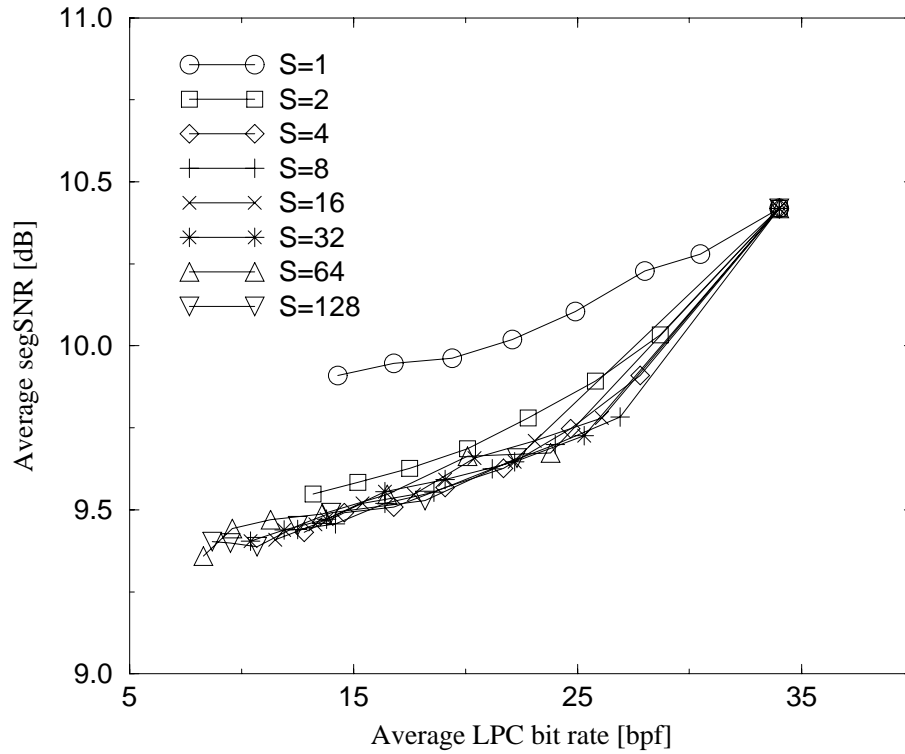


Figure 6. Average segSNR at different bit rates with different size of adaptive LPC codebook, $L = 240$.

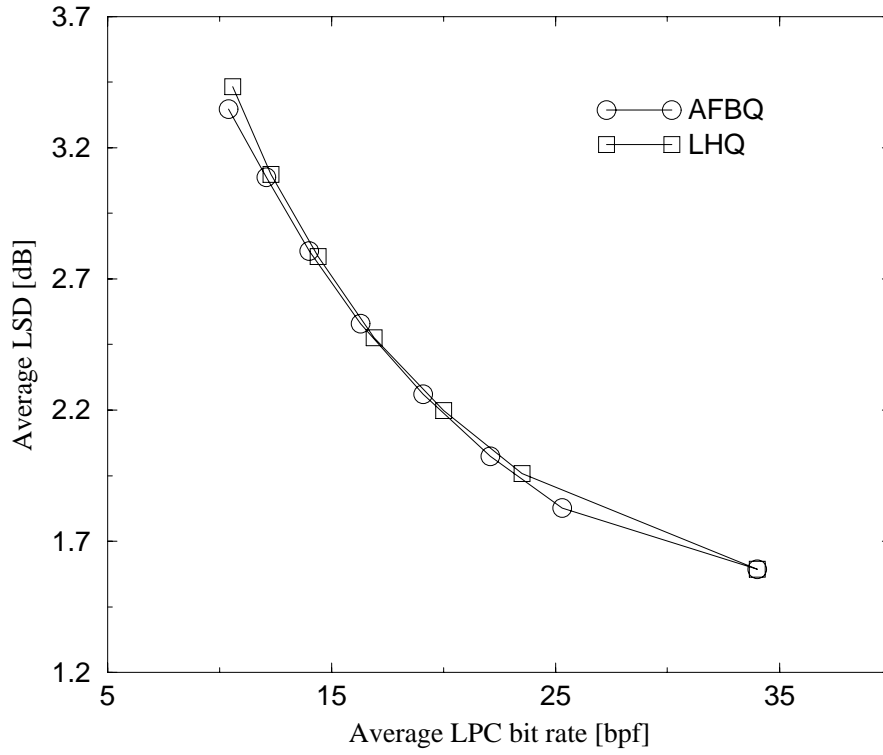


Figure 7. Average LSD introduced at different bit rates by adaptive forward-backward quantization scheme and long history quantization, when $S = 16$, and $L = 160$.