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# STOCHASTIC ALGORITHM FOR CHANNEL OPTIMIZED VECTOR QUANTIZATION: APPLICATION TO ROBUST NARROW-BAND SPEECH CODING

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**ABSTRACT**

In this paper, we propose a stochastic joint source-channel scheme developed for efficient and robust encoding of spectral speech LSF parameters. The encoding system, named LSF-SSCOVQ-RC, is an LSF encoding scheme based on a reduced complexity stochastic split vector quantizer optimized for noisy channel. For transmissions over noisy channel, we will show first that our LSF-SSCOVQ-RC encoder outperforms the conventional LSF encoder designed by the split vector quantizer. After that, we applied the LSF-SSCOVQ-RC encoder (with weighted distance) for the robust encoding of LSF parameters of the 2.4 Kbits/s MELP speech coder operating over a noisy/noiseless channel. The simulation results will show that the proposed LSF encoder, incorporated in the MELP, ensure better performances than the original MELP MSVQ of 25 bits/frame; especially when the transmission channel is highly disturbed. Indeed, we will show that the LSF-SSCOVQ-RC yields significant improvement to the LSFs encoding performances by ensuring reliable transmissions over noisy channel.

***Index Terms*—** Source-channel coding, robust speech coding, LSF parameters, MELP coder.

## 1. INTRODUCTION

In low bit-rates speech coding systems, the short-term spectral information of the speech signal is often modelled by the frequency response of an all-pole filter whose transfer function is denoted by $H(z) = 1/A(z)$ in which $A(z) = 1 + a_1 z^{-1} + \ldots + a_{10} z^{-10}$ [1]. The 10 parameters $\{a_i\}_{i=1,2,\ldots,10}$, known as the Linear Predictive Coding (LPC) coefficients, play a major role in the overall bandwidth and preserving the quality of the encoded speech. In practice, we don't quantify directly the LPC coefficients because they have poor quantization properties. One of the most efficient representations of the LPCs is the Line Spectral Frequency (LSF) [2].

Exploiting the well-known properties of the LSF parameters (LSFs) [3], [4] various coding schemes based on scalar and vector quantizations were developed in the past for the efficient quantization of these parameters. Several works showed that the vector quantizer (VQ) schemes, such as Split VQ (SVQ) [3] …, can achieve at lower bit-rates the transparent quantization quality of the LSFs compared with those conceived based on scalar quantizer (SQ).

In this paper, we propose a low bit-rate encoding scheme designed for efficient and robust encoding of speech LSF parameters. At the beginning, the LSFs encoding system was designed based on the SVQ technique for ideal transmissions over a noiseless channel. We will show that this system, named LSF-SVQ encoder, can achieve transparent quantization quality at a rate of 24 bits/frame.

After that, our interest was drawn to the improvement of the LSF-SVQ encoder robustness for real transmissions over noisy channel. Conventionally, a redundant channel coding [5] is used to ensure an "explicit" protection to sensitive parameters of speech coders against channel errors but at the cost of an increase of the bit-rate/delay transmissions and the complexity of the coding/decoding.

To overcome these disadvantages, we investigate the joint source-channel coding (JSCC) method in which the overall distortion is minimized by simultaneously considering the impact of the transmission errors and the distortion due to source quantization [6], [7], [8]. Many works have proved the effectiveness of the JSCC to protect implicitly (i.e., without redundancy) source data while maintaining a constant bit rate and a reduced complexity. To implicitly protect the transmission indices of our LSFs encoders, we adopted a JSCC method related to vector quantizers optimized for noisy channels. Indeed, we developed a reduced complexity stochastic SVQ algorithm optimized for noisy channel. A comparative evaluation between the new encoder (named LSF-SSCOVQ-RC) and the conventional LSF-SVQ will be presented. After that, we apply the LSF-SSCOVQ-RC encoder (with weighted distance) for the LSFs robust encoding of the 2.4 Kbits/s MELP speech coder.

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## 2. JOINT CODING BY THE CHANNEL OPTIMIZED VECTOR QUANTIZATION

A channel optimized vector quantizer (COVQ) is a JSCC scheme based on the principle of VQ generalization by taking into account the present noise on the transmission channel [6].

Let's recall first that a $k$-dimensional VQ of rate $R$ bits/sample (bps) is a mapping of $k$-dimensional Euclidean space $\Re^k$ into a finite subset (codebook) $Y = \{y_0,\ldots, y_{L-1}\}$ composed of $L = 2^{kR}$ quantization codevectors. The design principle of a VQ codebook consists of partitioning the $k$-dimensional space of source vectors $x$ into $L$ non overlapping cells (partition) $\{R_0,.., R_{L-1}\}$ and associating with each cell $R_i$ a unique codevector (centroid) $y_i$ such as the total average distortion $D$ is minimized. Various algorithms for VQ design have been developed in the past. The most popular one is certainly the LBG algorithm (LBG-VQ) [9].

For a noisy channel, the design of a COVQ encoder is carried out by an LBG-VQ version extended to the noisy case [6]. The COVQ scheme keeps thus the same VQ block structure. The difference is in the formulation of the necessary optimality conditions to minimise a generalized expression of the total average distortion, formulated by [6], [8]:

$$D = \frac{1}{k}\sum_{i=0}^{L-1}\int_{R_i} p(x)\left[\sum_{j=0}^{L-1} p(j/i)\cdot d(x, y_j)\right] dx\ , \qquad (1)$$

where $p(j/i)$ is the channel transition probability, $p(x)$ is the $k$-fold probability density function of the source and $d(x, y_j) = \|x - y_j\|^2$ is the squared Euclidean distance.

The formulations of optimality necessary conditions of COVQ system are derived in two steps [6]. For a given codebook $Y = \{y_0,..., y_{L-1}\}$, the optimal partition $R_i$ ($i$= 0,..., $L-1$) for a noisy channel is such that:

$$R_i = \left\{x \in R^k : \sum_{j=0}^{L-1} p(j/i)\left\|x - y_j\right\|^2 \le \sum_{j=0}^{L-1} p(j/l)\left\|x - y_j\right\|^2,\ \forall l \ne i\right\} \qquad (2)$$

Similarly, the optimum codebook for a fixed partition is given by:

$$y_j = \frac{\sum_{i=0}^{L-1} p(j/i)\int_{R_i} xp(x).dx}{\sum_{i=0}^{L-1} p(j/i)\int_{R_i} p(x).dx}\ ,\ \ j = 0,\ldots, L-1 \qquad (3)$$

In our applications, we considered that the communication channel is a discrete memoryless binary symmetric channel (BSC) model with bit error (crossover) probability $p$ [5], [8].

The design procedure of the COVQ encoding system is a straightforward extension of the LBG-VQ algorithm by introducing the BSC error probability as an input parameter in the optimization process. This parameter, noted $\varepsilon_c$, is generally called design error probability.

The steps of our version of the COVQ algorithm and its performances are detailed in [8]. In the case of transmissions over noisier channels (higher values of $p$), we showed that COVQ encoder outperforms the conventional VQ designed by the LBG-VQ for noiseless channel (i.e., $\varepsilon_c$ = 0.000). However, when the channel is noiseless ($p$ = 0.000) or slightly disturbed, the performances of the COVQ encoder are suboptimal with the increase of the design parameter $\varepsilon_c$.

### 2.1. Disadvantages of the COVQ algorithm

The COVQ encoding system presents several disadvantages. As we already mentioned, the channel error probability is an input parameter which must be fixed preliminary before running the COVQ optimization procedure. In practice this parameter is difficult to estimate because it can vary according to time making the design according to a specific value rather academic.

In other hand, COVQ algorithm is deterministic and converges to a local minimum according to the choice of the initial codebook. Thus, an inappropriate choice of this last can cause a convergence to locally optimum codebook far from the global optimum. The computational complexity of COVQ algorithm is another disadvantage. Indeed, the design of a COVQ codebook could take an enormous CPU time compared with that of the LBG-VQ. In addition, the size of the training vectors database must be larger than the codebook size.

To alleviate these disadvantages, we studied and implemented a modified version of the COVQ algorithm. It is about a reduced complexity stochastic algorithm of vector quantization optimized for noisy channel (noted SCOVQ-CR).

## 3. REDUCED COMPLEXITY STOCHASTIC VQ OPTIMIZED FOR NOISY CHANNEL

In this section, we present the SCOVQ-RC algorithm to design a VQ optimized for noisy channel. It is about a stochastic version simpler than the COVQ algorithm where the channel error probabilities will be introduced only in the codebook update step in order to reduce the construction time [7].

### 3.1. SCOVQ-RC Algorithm

Our version of the SCOVQ-RC algorithm follows the same COVQ design steps. The difference is in the vectors partitioning step where the vectors will not be classified according to the generalized nearest neighbor optimality rule (Eq. 2). We will use instead a stochastic method to carry out the vector partitioning. However, the codevectors will be updated exactly as in the case of the COVQ (Eq. 3).

First, we introduce the training database and the design error probability $\varepsilon_c$. At the beginning, we suppose a high value of error probability noted $\varepsilon_0$ which decreases gradually after each iteration according to the formula:

$$\varepsilon'_k = \varepsilon_0.\alpha^k + \varepsilon_c \,, \tag{4}$$

where $\varepsilon'_k$ is the error probability at the $k^{\text{th}}$ iteration and $\alpha$ is a constant set generally in range [0.8, 0.95].

To partition the training vectors, we use a certain stochastic method. Initially we calculate the appartenance probabilities of the training vector $x_n$ to classes $R_i$ represented by the codevectors $y_i$. Then, the vector will belong to the class having the largest appartenance probability calculated according to the formula [7]:

$$P(i,n,k) = \frac{e^{-\beta(k)\frac{d(x_n, y_{k,i})}{\overline{d(x_n)}}}}{\sum_{m=0}^{L-1} e^{-\beta(k)\frac{d(x_n, y_{k,m})}{\overline{d(x_n)}}}} \quad , i = 0 \,, \ldots, L\text{ -1} \tag{5}$$

where $k$ is the iteration number and $\beta(k)$ is a function which increases at each iteration such as $\lim_{k\to\infty} \beta(k) = +\infty$. Its role is to decrease the stochastic aspect of the algorithm. Indeed, when the iteration index $k$ increases, the algorithm becomes increasingly deterministic. We used the following linear function:

$$\beta(k) = c(k-1)\,, \tag{6}$$

where $c$ is a constant fixed to $c = 2$. The function $\overline{d(x_n)}$ is the average distance of the training vector $x_n$ with all the codevectors of the actual codebook.

In the first iteration ($k$ =1), the function $\beta(k) = 0$. In this case, all the appartenance probabilities $P\,(i, n, k)$ are identically equal to $1/L$. All the codebook codevectors will be close to the mass center of the training database. After that, the appartenance probabilities $P(i, n, k)$ will vary and the algorithm gradually moves away the codevectors towards their optimal positions.

At the beginning, when the error probability is high, we obtain a well ordered codebook but at the same time compressed. The performance degradations of such a codebook will be large if the communication channel is noiseless. After that, the error probability drops during the iterations and the codebook extends gradually in space to have good performances. The algorithm is run with the initial value $\varepsilon_0 = 0.5$.

### 3.2. SCOVQ-RC encoder performances and its advantages over the COVQ

The performances of SCOVQ-RC encoding systems, designed for various values of parameter $\varepsilon_c$, are given in Table 1. These encoders, whose selected characteristics are: $k = 2$, $R = 2$ bps and $L = 16$, were applied to encode memoryless Gaussian source for transmissions over a BSC channel with variable bit error probability $p$.

**Table 1.** SNR performances of SCOVQ-RC encoding systems operating over BSC channel

| $p$ \ $\varepsilon_c$ | ***0.000 (VQ)*** | **0.001** | **0.005** | **0.01** | **0.05** |
|---|---|---|---|---|---|
| **0.000** | *9.678* | 9.637 | 9.556 | 9.511 | 8.801 |
| **0.001** | *9.564* | 9.577 | 9.515 | 9.480 | 8.771 |
| **0.005** | *9.186* | 9.335 | 9.414 | 9.279 | 8.653 |
| **0.01** | *8.755* | 9.123 | 9.179 | 9.221 | 8.509 |
| **0.05** | *6.687* | 7.024 | 7.198 | 7.234 | 7.543 |
| **0.1** | *4.855* | 5.417 | 6.223 | 6.277 | 7.168 |
| **0.2** | *2.671* | 3.975 | 4. 340 | 4.411 | 6.015 |

In the case of noiseless channel ($p$ = 0.000), these results indicate that the performances of the SCOVQ-RC encoder are slightly lower than those of the conventional LBG-VQ ($\varepsilon_c$ = 0.000). This is due to the compression caused by the design error probability, but they remain acceptable. However, the performances of the SCOVQ-RC become definitely better with the increase of the transmission channel probability $p$.

Figure 1 presents a comparative evaluation between the encoders: the SCOVQ-RC ($\varepsilon_c$ = 0.05), the COVQ ($\varepsilon_c$ = 0.05) and the conventionnal VQ applied for coding memoryless Gaussian source.

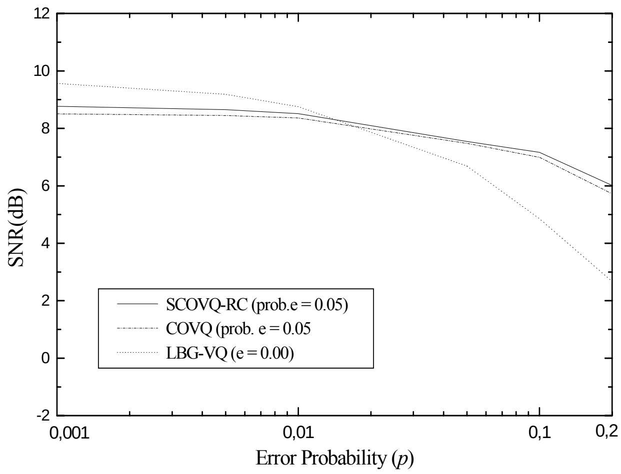


**Figure 1.** Performances comparative evaluation between the SCOVQ-RC, the COVQ and the standard VQ

For all error probability variation range, the SNR results show that SCOVQ-RC performances are comparables with those of the COVQ, and even better.

In addition to the reduction of the computational complexity compared with the COVQ, the SCOVQ-RC algorithm has good convergence properties. Indeed, the SCOVQ-RC algorithm is not dependent of an initial codebook and hence it is able to leave the local minima due to its stochastic aspect. Moreover, the SCOVQ-RC allows constructing a well-balanced optimal codebook even if the number of training vectors is smaller than the size of the codebook.

## 4. EFFICIENT AND ROBUST CODING OF THE LSF PARAMETERS FOR TRANSMISSION OVER NOISY CHANNEL

In this section, we propose a low bit-rate efficient and robust encoding system of speech LSF parameters (LSFs) for transmissions over a noiseless/noisy channel. The direct application of the SCOVQ-RC encoder to implicitly protect the LSFs vectors of size 10 proves to be impractical when a certain bit-rate is required to achieve transparent quantization quality. The major problem encountered in the implementations is the exponential increase of the computational complexity as well as the processing CPU time which becomes extremely excessive as the coding rate is increased.

To overcome this computational complexity problem, we developed an LSFs encoding system based on the SCOVQ-RC principle but modified according to the Split VQ (SVQ) concept. It consists to divide the 10-dimensional SCOVQ-RC codebook into several sub-codebooks of smaller dimensions and sizes.

### 4.1. Low-bit rate efficient encoding of LSF parameters by the SVQ method

At the beginning, we designed a low-bit rate encoding system of LSF parameters by using the SVQ technique. The initial objective of this system, named "LSF-SVQ Encoder", is to achieve a transparent quantization of the LSFs for transmissions over an ideal noiseless channel.

A $k$-dimensional LSF-SVQ encoder of $N$ parts is composed of $N$ classical VQs of smaller sizes and dimensions [3]. Its basic principle consists in partitioning the set of the training base LSFs vectors of dimension $k = 10$ in $N$ sub-sets of sub-vectors of smaller dimension $k_i$ (with $\sum_{i=1}^{N} k_i = k$ ). Then, for each part, the corresponding VQ codebook will be designed by using the well-known LBG-VQ algorithm [9]. Notice that in the design of our LSF-SVQ encoders, the LSF vector of dimension 10 is divided into two parts with (4 – 6) division. In general, the bits are uniformly allocated to individual sub-parts of the same division whereever possible.

To more improve the performances of the LSFs encoders and to get transparent quantization quality at lower bit rate, we used an appropriate distance measure. It's about the weighted Euclidean distance measure which is performed in the frequency LSF domain. The weighted distance used in this work is given by [3], [4], [8]:

$$d(f,\hat{f}) = \sum_{i=1}^{10} c_i w_i (f_i - \hat{f}_i)^2 \ , \tag{7}$$

where $f_i$ and $\hat{f}_i$ are respectively the $i^{\text{th}}$ coefficients of the original $f$ and quantized $\hat{f}$ LSF vectors; $c_i$ and $w_i$ represent respectively the constant and variable weights assigned to the $i^{\text{th}}$ LSF coefficient.

We present below the quantization performances of the LSF-SVQ encoders operating at different bit-rates. The performances are evaluated by the average spectral distortion (SD). When calculated discretely over a limited bandwidth, the spectral distortion for frame $i$ is given, in decibels, by [3], [4]:

$$SD_i = \sqrt{\frac{1}{n_1 - n_0} \sum_{n=n_0}^{n_1 - 1} \left[ 10 \log_{10} \frac{S(e^{j2\pi n/N})}{\hat{S}(e^{j2\pi n/N})} \right]^2} \ , \tag{8}$$

where $S(e^{j2\pi n/N})$ and $\hat{S}(e^{j2\pi n/N})$ are respectively the original and quantized power spectra of the LPC synthesis filter, associated with the $i^{\text{th}}$ speech frame.

Generally, we can get transparent quality if we maintain the following three conditions [3]: -1) The average SD is approximately 1 dB, -2) The percentage of outlier frames having SD between 2–4 dB is less than 2% and - 3) No outlier frame having SD greater than 4 dB.

The LSF vectors database used in the experiments was constructed from approximately 85 minutes of speech taken from the TIMIT speech database [10]. To determine the LSFs vectors, we used the same LPC analysis function of the 4.8 Kbits/s FS1016 speech coder [11]. One part of the LSF database, consisting of 144984 LSF vectors, is used for training and the other part, of 26560 LSF vectors (different from the training set) is used for tests. For different rates $b$ (bits/frame), the performances of the LSF-SVQ encoders using 2 parts (4 – 6) are shown in Table 2.

**Table 2.** Performances of the two parts LSF-SVQ encoders operating over a noiseless channel

| **Bits/frame** $b$ ($b_1$+ $b_2$) | **Average SD (dB)** | **Outliers (in %)** | |
|---|---|---|---|
| | | 2-4 dB | > 4 dB |
| **18** (9+9) | 1.653 | 22.386 | 0.04 |
| **20** (10+10) | 1.464 | 11.36 | 0.006 |
| **22** (11+11) | 1.310 | 5.286 | 0.000 |
| **24** (12+12) | 1.170 | 1.953 | 0.000 |

For transmissions over a noiseless channel, these simulation results indicate that the two parts LSF-SVQ encoder, using the weighted distance, can achieve the transparent quantization at a rate of 24 bits/frame.

### 4.2. Robust encoding of the LSF parameters: Application of the SCOVQ-RC encoder

To implicitly protect the transmission indices of the LSF-SVQ encoder, we developed a JSCC method by SCOVQ-RC modified according to the SVQ basic concept. The new quantizer, applied for the robust encoding of LSFs, is called the Split LSFs SCOVQ-RC (LSF-SSCOVQ-RC encoder). For the design of LSF-SSCOVQ-RC encoder and its performance tests, we used the same training and test LSFs databases. In our applications, the LSF-SSCOVQ-RC codebooks were optimized for a design error probability $\varepsilon_c = 0.01$.

We present in Table 3 examples of simulation results relating to the performances of the 24 bits/frame LSF-SSCOVQ-RC (2 parts (4 – 6), $\varepsilon_c = 0.01$) encoder applied for robust encoding of LSF parameters in a disturbed environment. For comparative evaluations, the performances of the conventional 2-parts LSF-SVQ of 24 bits/frame were also inserted in the table.

These comparative results show that when the channel becomes noisier, the LSF-SSCOVQ-RC yields significant improvement to the LSFs encoding performances by ensuring more reliable transmissions. Without protection, the conventional LSF-SVQ encoder has incurred more severe degradation compared with the protected LSF encoder. This degradation is represented by a brutal increase in the average SD of the LSF-SVQ when the channel error probability $p$ becomes rather high. Under these conditions, the LSF-SSCOVQ-RC ($\varepsilon_c = 0.01$) encoder has permitted to have a good robustness against channel errors by maintaining a reduced and slow increase of the average SD and the number of outliers frames of SD > 4 dB.

## 5. ROBUST ENCODING OF THE MELP SPEECH CODER LSF PARAMETERS: APPLICATION OF THE LSF-SSCOVQ-RC

In this section, we apply our LSF-SSCOVQ-RC encoder (with weighted distance) to robust encoding of LSF parameters of the MELP speech coder operating over a noiseless/noisy channel. The Federal standard MELP (Mixed Excitation Linear Prediction) is a speech coder of 2.4 Kbits/s developed by the US DoD [12]. According to the MELP norm, its LSF parameters are encoded at the origin by a Multi-stage VQ (MSVQ) of 25 bits/frame.

To evaluate the objective quality of the coded speech signals synthesized by the MELP coder, we used the ITU-T recommendation P.862 known under the abbreviation PESQ (Perceptual Evaluation of Quality Speech) [13]. The PESQ algorithm could evaluate the listening quality under many degradation conditions and have a very close correlation with the MOS (Mean Opinion Score) subjective evaluation. The database used in the following evaluations is composed of phonetically equilibrated speech sequences of 8.13s extracted randomly from a different Arabic speech database [14].

We present now the performances of the 25 bits/frame 2-parts LSF-SSCOVQ-RC ($\varepsilon_c = 0.01$) encoder incorporated in the MELP coder. Synthesized speech signals of the Arabic speech sequences were generated by the MELP operating over a BSC channel where the errors affect only the transmission of LSF parameters.

In Tables 4 and 5, we respectively present the average SD-performances of the incorporated LSFs encoders and the performances of the global MELP in terms of average PESQ. The LSFs encoders (the LSF-SSCOVQ-RC and the original MELP MSVQ) are successively incorporated and affected by the same sequences of channel noise.

As the transmission error probability increases, these results show that the incorporated LSF-SSCOVQ-RC encoder yields better SD-performances than those of the original MSVQ MELP. Thus, the LSF-SSCOVQ-RC (25 bits/frame) encoder has ensured certain robustness of MELP LSFs encoding against channel errors by maintaining a reduced increase of the average SD and number of Outliers frames. However, when the transmissions are done over a noiseless channel or slightly disturbed ($p \le 0.01$) the LSF-SSCOVQ-RC performances become sub-optimal.

In terms of PESQ, the objective performances of the global MELP coder (with LSF coded by the LSF-SSCOVQ-RC) are better than those of the non-protected MELP coder. Thus, the global system has a good robustness against the channel errors when the transmission channel is highly disturbed.

## 6. CONCLUSION

In this work, an efficient and robust encoding system for LSF parameters was developed based on the SCOVQ-RC encoder. The comparative evaluations between the new LSF encoder (named LSF-SSCOVQ-RC) and the conventional LSF-SVQ encoder proved that the LSF-SSCOVQ-RC provided a real implicit protection to LSF

**Table 3.** Performance comparisons between LSF-SSCOVQ-RC / LSF-SVQ encoders of 24 bits/frame operating over a BSC channel of variable bit error probability $p$

| | LSF-SSCOVQ-RC Encoder | | | LSF-SVQ Encoder | | |
|---|---|---|---|---|---|---|
| BSC Probability $p$ | Average SD (dB) | Outliers (in %) 2 - 4 dB | Outliers (in %) > 4 dB | Average SD (dB) | Outliers (in %) 2-4 dB | Outliers (in %) > 4 dB |
| **0** | 1.251 | 7.113 | 0.013 | 1.17 | 1.953 | 0.000 |
| **0.001** | 1.288 | 8.641 | 0.266 | 1.323 | 2.213 | 2.020 |
| **0.005** | 1.431 | 13.903 | 1.386 | 1.93 | 3.073 | 9.82 |
| **0.01** | 1.603 | 19.959 | 2.986 | 2.622 | 3.98 | 18.719 |
| **0.05** | 2.751 | 44.073 | 19.313 | 6.639 | 6.846 | 64.353 |
| **0.1** | 3.862 | 43.933 | 41.426 | 9.078 | 5.426 | 86.333 |

**Table 4.** Performance comparisons between LSF-SSCOVQ-CR/ MSVQ encoders (25 bits/frame) incorporated in the MELP coder operating over a BSC channel

| BSC Probability *p* | LSF-SSCOVQ-RC Encoder Average SD (dB) | Outliers (in %) 2 - 4 dB | > 4 dB | Original MSVQ-MELP Average SD (dB) | Outliers (in %) 2 - 4 dB | > 4 dB |
|---|---|---|---|---|---|---|
| **0** | 1.709 | 30.004 | 0.000 | 1.001 | 0.000 | 0.000 |
| **0.01** | 1.995 | 37.722 | 3.87 | 1.639 | 9.816 | 7.001 |
| **0.05** | 3.164 | 48.093 | 27.309 | 3.615 | 28.867 | 35.646 |
| **0.1** | 4.464 | 34.946 | 57.192 | 5.162 | 30.481 | 55.976 |
| **0.2** | 6.072 | 14.669 | 83.921 | 7.052 | 16.817 | 81.207 |

**Table 5.** PESQ objective performances of MELP speech coder

| BSC Probability *p* | PESQ Performances of MELP with : LSF-SSCOVQ-RC (25 bits/frame) | Original MSVQ-MELP (25 bits/frame) |
|---|---|---|
| **0** | 3.041 | 3.259 |
| **0.01** | 2.952 | 2.898 |
| **0.05** | 2.623 | 2.523 |
| **0.1** | 2.350 | 1.995 |
| **0.2** | 1.986 | 1.549 |

parameters for transmissions over disturbed channel. Indeed, the new encoder yielded significant improvement to the LSFs encoding performances by ensuring more reliable transmissions compared with the non-protected system. After that, we applied successfully the LSF-SSCOVQ-RC encoder (with weighted distance) to the robust encoding of LSF parameters of the 2.4 Kbits/s MELP speech coder. The simulation results showed that our incorporated 25 bits/frame LSF-SSCOVQ-RC encoder ensured better performances than the original 25 bits/frame MSVQ, especially when the transmission channel is highly disturbed.

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