Hybrid Method Using Wavelets and Predictive Method for Compression of Speech Signal

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Abstract—The development of the signal compression algorithms is having compressive progress. These algorithms are continuously improved by new tools and aim to reduce, an average, the number of bits necessary to the signal representation by means of minimizing the reconstruction error. The following article proposes the compression of Arabic speech signal by a hybrid method combining the wavelet transform and the linear prediction. The adopted approach rests, on one hand, on the original signal decomposition by ways of analysis filters, which is followed by the compression stage, and on the other hand, on the application of the order 5, as well as, the compression signal coefficients. The aim of this approach is the estimation of the predicted error, which will be coded and transmitted. The decoding operation is then used to reconstitute the original signal. Thus, the adequate choice of the bench of filters is useful to the transform in necessary to increase the compression rate and induce an imperceivable distortion from an audible point of view.

Keywords—Compression, linear prediction analysis, multiresolution analysis, speech signal.

I. INTRODUCTION

The subject of data compression is the object of a particular development, which led a variety of algorithms. The analysis by ways of wavelets is one of the new approaches used in the signal-processing field; the method allows the decomposition of a given signal on concentrated functions both in time and frequency. In coding by linear prediction, the value of each signal coefficient in predicted order 5, as well as, the compression signal coefficients. The aim of this approach is the estimation of the predicted error, which will be coded and transmitted. The decoding operation is then used to reconstitute the original signal. Thus, the adequate choice of the bench of filters is useful to the transform in necessary to increase the compression rate and induce an imperceivable distortion from an audible point of view.

The article is organized in the following manner:

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The information difference, $P_{j+1}(f)-P_j(f)$, which exists between two consecutive approximations re presents the additional detail necessary to pass from the approximation $2^j(P_j(f))$ to a finer approximation with scale $2^{-j+1}(P_{j+1}(f))$. According to the theory of Daubechies-Mallat, a set of functions $\{\psi_{j,k}\}$ is then introduced; these functions from the wavelet $\psi(x)$ and form an orthogonal base of the space $V_j$.

Finally, the decomposition of the signal to be analysed $S$, presumably approximated $j=0$, is carried out according to the following steps:

- The approximation of the signal $S^1$ to the $j=1$ resolution is obtained by the convolution of the $S^0$ signal by the analysis filter $h(x)$ and by decimation of order 2, such as:

$$\phi_{j,k}(x) = 2^{-j/2} \phi(2^j x-k)$$

obtained from the scale function $\phi(x)$.

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where $h$ is a low pass filter whose impulse response:

$$h(k) = \frac{1}{\sqrt{2}} \langle \phi_k^0, \phi_k^0 \rangle$$  \hspace{1cm} (3)

- The $D_j$ details are obtained starting from $S_j$ by convolution by the filter $g(x)$ and by retaining only a sample over 2:

$$g(x) = g(-x)$$  \hspace{1cm} (4)

Where $g$ is a high pass filter, of impulse response:

$$g(k) = \frac{1}{\sqrt{2}} \langle \phi_k^0, \phi_k^0 \rangle$$  \hspace{1cm} (5)

Thus, the signal to be analysed will be represented by the approximated signal $S_j$ of low resolution and the details signals with the $2^j$ resolutions for $j<J<-1$. Then the schemes of discrete wavelet transform decomposition and reconstruction of a signal $x$ is shown in Fig. 1 and Fig. 2 respectively.

![Fig. 1 Banc of analysis at two levels](image1)

![Fig. 2 Reconstruction scheme](image2)

The adopted strategy in compression consists in coding the last signal with the coefficients of the details signals, with various resolutions, which exceed a certain fixed threshold [6], [7].

III. ANALYSIS BY THE LINEAR PREDICTION

The linear prediction is another manner to reduce the redundancies between the close coefficients; it is based on the assumption that each sample of the original signal $x(n)$ can be approximated by linear combination of samples which precede it [5], such as:

$$\hat{x}(n) = \sum_{i=1}^{p} a(i) x(n-i)$$  \hspace{1cm} (6)

$$\hat{x}(n) = -a(1) x(n-1) - a(2) x(n-2) - ... - a(p) x(n-p)$$  \hspace{1cm} (7)

where $a(i)$ represents the coefficients of the prediction of order $p$, $x(n)$ is the predicted value.

The prediction error of order $p$ is given by:

$$e(n) = x(n) - \hat{x}(n)$$  \hspace{1cm} (8)

$$e(n) = \sum_{i=0}^{p} a(i) x(n - i) ; \ a(0) = 1$$  \hspace{1cm} (9)

The remaining prediction energy is:

$$E = \sum_n e^2(n)$$  \hspace{1cm} (10)

Also the calculation of the gain is often used to measure the performance of the predictive modelling:

$$G_p = \frac{\sigma_x^2}{\sigma_e^2}$$  \hspace{1cm} (11)

where $\sigma_x^2$ is the variance of the predictor input signal and $\sigma_e^2$ is the variance of the prediction error.

The samples of residue of prediction $e(n)$ have a level generally lower; they require less bits to be represented. The compressed signal is then rebuilt starting from the parameters of the predictor and the samples of the error signal. The optimal predictor is that which minimizes the power of the prediction error, $E\{e(n)^2\}$, which leads to an adequate choice of the coefficients $a(i)$, by the resolution of the equations of Yule-Walker. These latter can, particular, be solved by the Levinson-Durbin algorithm and that of Schur. Let us recall that the coefficients $a(i)$ are those of the denominator of the filter all poles whose transmittance is the spectral envelope of the signal.

IV. IMPLEMENTATION OF THE COMPRESSION OF SPEECH SIGNAL BY THE HYBRID METHOD

The type of signals considered as the object to be treated in our study is the Arabic speech data base (“sin”), illustrated in Fig. 3, presented on 3842 samples at a rate of 16 KHz. The S. Mallat algorithm is applied to this configuration in order to break it up into two signals approximated $S_j$ with $D_j$ details obtained respectively by low pass filtering (the $h$ filter) and high pass filtering (the $g$ filter) according to the values of the analysis filters provided by the works of I. Daubechies [2]. A factor 2 decimation is then produced on the signals $S_j$ and $D_j$ in order to preserve the same information quantity. The compression stage rests on the omission of the coefficients of the signal of detail and the conservation of the obtained approximated signal, Fig. 4, which will be treated thereafter by the linear prediction algorithm.

This algorithm will allow the calculation of the parameters of the order $p=5$ predictor and of the samples of the error signal $e(n)$, Fig. 5. The parameters and the samples of error must be quantified and coded perfectly for their transmission or their storage.

The reconstruction of the speech signal is realized by the follow stages:
i) the calculation of the data, from the parameters and the samples of the error.

ii) and then the application of the inverse discrete wavelet transform.

The reconstructed signal is shown in Fig. 6.

![Fig. 6 Reconstructed signal](image)

**V. RESULTS AND DISCUSSION**

For computing the hybrid method the algorithm are developed on the one hand by choosing the analysing function such as Daubechies2, Daubechies4 and Daubechies8 from levels 1 and 2; on the other hand by varying the prediction order.

The compressed Arabic speech signal were obtained using Daubechies2 and decomposed up to level 2. It is reconstructed only from the approximated coefficients and for this reason the decomposition at a level superior leads at an unacceptable distortion of the compressed signal.

The results of this analysis given in Table I illustrate that as compression ratio and the MSE increase where as the PSNR will decreases and the compression rate CR = 2 were obtained with MSE = 275.89 distributes on 3842 samples.

![Fig. 7 Compression rate versus prediction order](image)

- Mean square error:

\[ \text{MSE} = \frac{1}{N} \sum_{i=0}^{N-1} [S(i) - S'(i)]^2 \]  \hspace{1cm} (13)

where \( S(i) \) is the original speech data and \( S'(i) \) is the compressed signal.

- Root mean square error:

\[ \text{RMSE} = \left[ \frac{1}{N} \sum_{i=0}^{N-1} [S(i) - S'(i)]^2 \right]^{1/2} \]  \hspace{1cm} (14)

- Peak signal to noise ratio:

\[ \text{PSNR} = 10 \log \left( \frac{255}{\text{RMSE}} \right) \]  \hspace{1cm} (15)
The compression ratio obtained combining wavelet and linear prediction attains the value 3.282 with the good speech signal quality like illustrates in Fig. 8, also some of the information is lost. For speech signals this loss is acceptable since it is interested only in recognizing the compressed signal.

VI. CONCLUSION

In this paper the hybrid scheme combining discrete wavelet transform and linear prediction method was proposed for the compression of the Arabic speech signal. The associated both methods are proved its efficiency by an increase of the compression rate with preservation of the quality of speech signal.

The applying a linear prediction only on the approximated coefficients signal with out detail signal was allowed the evaluation of the speech-compressed signal.

It is also observed that the efficient of this technique can be varies with the coding of the prediction error and the adopted method for the choice of the coefficients to truncate.

REFERENCES