Simulation of Codec for Adaptive Delta Modulation

Rumen P. Mironov[1]

***Abstract* –Software model of Adaptive Delta Modulation Codec for one dimensional signals is developed. The presented codec is simulated on Simulink for Matlab 6.5 environment and the obtained results for real sound signals are evaluated by the calculation of MSE and SNR for the decoded signals.**

***Keywords* – Digital Signal Processing, Linear Prediction, Adaptive Data Compression, Matlab Simulation.**

I. Introduction

The basic methods for digital signal compression are lossless and lossy compression. Due to the low coefficient of information reduction, the first group of methods is used most frequently in archiving systems and didn't find widespread use in the information transmission systems. The methods for lossy compression are divided into three groups: statistical, psychoacoustic and transforms methods ([1], [2], [3] and [4]). The psychoacoustic compression methods are based on the shortcomings of the human hearing system and are suitable only when the recipient of the recorded signals is a man. The transformation methods provide a high degree of information reduction and work mainly on the basis of unitary transformations ([3], [4]), for which after decompression the specific block distortions are received. Statistical compression methods are based on the reduction of the information redundancy of the transmitted signals and proceed in two stages: decorrelation of communication signals and reduction of binary digits, necessary for transmition of signals ([1], [2]). One of the simplest and most convenient for practical implementation methods for decorrelation is the method for delta modulation ([2], [5]).

An adaptive method of linear prediction coding for one dimensional digital signals, based on delta modulation and adaptation step quantization [6] is presented. From the developed mathematical equations a general block scheme of Adaptive Delta Modulation Codec (ADMC) is synthesized and experimental results from the simulation by Simulink for Matlab 6.5 environment for test signals are given.

II. Mathematical Description

We will assume that the correlation covers n neighborhood elements of the input digital signal, represented by the stationary series $\{x(i)\}$ with $N > n$ values, which have zero average component and correlation function $R_x(r)$ for: $r = \overline{0, n-1}$. Then the basic equation of the linear prediction will be presented by the following way:

$$\hat{x}(i) = a_1 x(i-1) + \dots\dots + a_{n-1} x(i-n+1) = \sum_{k=1}^{n-1} a_k x(i-k), \quad (1)$$

where: $\hat{x}(i)$ is the value of the predicted element from the input signal $x(i)$. The prediction error is described by the equation:

$$e(i) = x(i) - \hat{x}(i), \quad (2)$$

and the quantization error - by the equation:

$$e_q(i) = Q[e(i)]\ . \quad (3)$$

The optimal values of the weighted coefficients can be calculated after the minimization of mean square error of the prediction. From the Eqs. (1) and (2) follows:

$$\overline{e^2(i)} = \frac{1}{N} \sum_{i=0}^{N-1} [x(i) - \hat{x}(i)] = E\left\{ \left[x(i) - \sum_{k=1}^{n-1} a_k x(i-k) \right]^2 \right\}, \quad (4)$$

where E is the averaging operator.

The partial derivative of $\overline{e^2(i)}$ with respect to any weight coefficient a_l can be expressed as:

$$\frac{\partial \overline{e^2(i)}}{\partial a_k} = \frac{\partial E\left\{ [x(i) - (a_1 x(i-1) + \dots . a_{n-1} x(i-n+1))]^2 \right\}}{\partial a_l}$$

$$= -2E\{[x(i) - (a_1 x(i-1) + \dots . a_{n-1} x(i-n+1))] x(i-l)\}$$

.

Once equated to zero and transformed, from the upper expression is obtained:

$$E\{x(i)x(i-l)\} = \sum_{k=1}^{n-1} a_k E\{x(i-k)x(i-l)\}. \quad (5)$$

The autocorrelation function of the digital signal, presented with the series $\{x(i)\}$ is:

$$R_x(r) = \frac{1}{N} \sum_{i=0}^{N-1} x(i)x(i-r) = E\{x(i)x(i-r)\}. \quad (6)$$

From the Eqs. (5) and (6) the following expression is obtained:

$$R_x(l) = \sum_{k=1}^{n-1} a_k R_x(k-l). \quad (7)$$

For $l = \overline{1, n-1}$ equation (7) represents a linear system with $n-1$ unknowns. After calculation of Eq. (7) the optimal values of the prediction coefficients of the delta modulation are:

$$a_1 = R_x(1)/R_x(0); \quad a_k = 0, \text{for } k = \overline{2, n-1}. \quad (8)$$

From Eqs.(1) and (8), the predicted element and the quantization error can be calculated by the following:

$$\hat{x}(i) \approx x(i-1) \quad , \quad (9)$$

$$e_q(i) = \begin{cases} +\xi, & \text{for } e(i) \geq 0 \\ -\xi, & \text{for } e(i) < 0 \end{cases} . \quad (10)$$

In the rapidly changing areas of the signal, where ξ is too small to represent the input's largest changes, a distortion

[1]Rumen P. Mironov is with the Faculty of Telecommunications, Technical University of Sofia, Boul. Kl. Ohridsky 8, Sofia 1000, Bulgaria, E-mail: rmironov@tu-sofia.bg

known as *slope overload* occurs. Moreover, when ξ is too large to represent the input's smallest changes, *granular noise* appear. In one dimensional and two dimensional signals, as is shown in (7), these two phenomena lead to blurred object edges and grainy or noisy surfaces.

To reduce the described errors an algorithm for adaptive computation of the step of the quantized error e_q^a is proposed. This is shown in Tabl.1.

Tabl.1.

$e_q(i)$	$e_q(i-1)$	$e_q(i-2)$	$e_q^a(i)$
1	1	1	+2
0	0	0	-2
1	0	1	+0.5
0	1	0	-0.5
1	1	0	+1
0	1	1	-1
1	0	0	+1
0	0	1	-1

The following equations are used for decoding of signals:

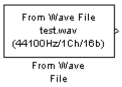
$$\mathbf{x}'(i) = \mathbf{e}'(i) + \hat{\mathbf{x}}'(i), \tag{11}$$

$$\mathbf{e}'(i) = Q^{-1}\left[\boldsymbol{e}_q(i)\right]. \tag{12}$$

The synthesized by the equations from (1) to (10) adaptive delta modulation codec (ADM) is shown on Fig. 1. On Fig. 2 and Fig.3 the synthesized coder and decoder blocks for ADM are shown respectively. The presented schemes are developed through the Simulink package for Matlab 6.5 environment and included the following units:

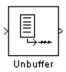
– input unit;

– output unit;

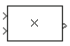


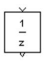
- input unit from wave file;

- output unit to wave file;



- input unit for buffered reading from file;

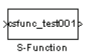

- multiplication unit;


- delay unit for one cycle;


- addition/subtraction unit;

- oscilloscope (GUI output);


- multiplexer (2,3,...*n* inputs, vector output);



- S-function (user defined function in Simulink).

III. Experimental Results

The developed ADM codec is used for simulation on Matlab 6.5 environment of real audio signals (WAV file format, 1 channel (mono), 16 bits, 44.1 KHz sampling rate).

The mean squared error of the transformation is calculated by the equation:

$$\overline{\varepsilon^2} = \frac{1}{N}.\sum_{i=1}^{N} e^2(i).$$

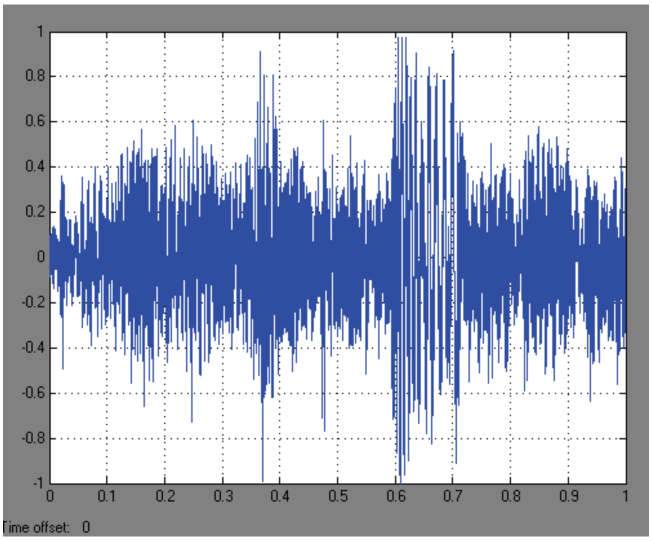


Fig. 4. Input signal $x(i)$

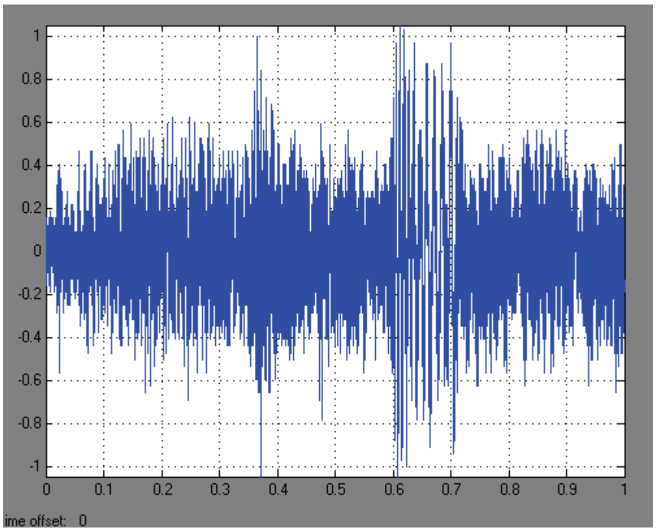


Fig. 5. Output predicted signal $\hat{x}(i)$.

The input test signal and output signal are shown on Fig.4 and Fig.5 respectively. On Fig.6 the input and output signals are shown together (visualized from the oscilloscope Scope 4). On Fig.7 the same signals are visualized by the zooming in horizontal direction (from 0.031s to 0.0324s).

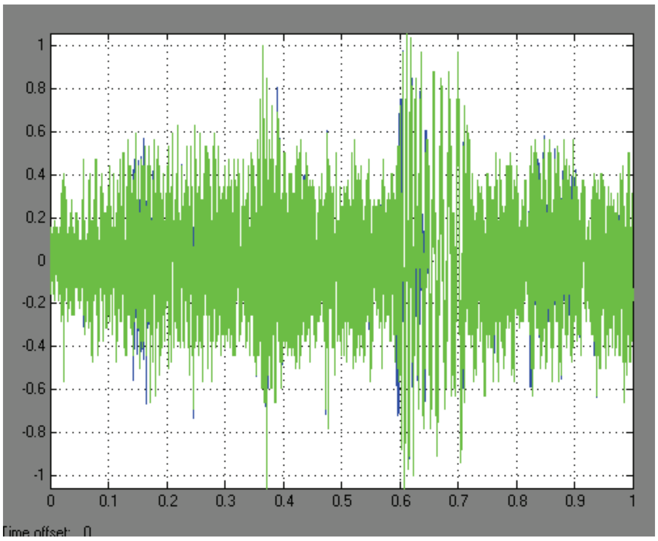


Fig. 6. Input and output predicted signal MUX{ $x(i)$, $\hat{x}(i)$ }.

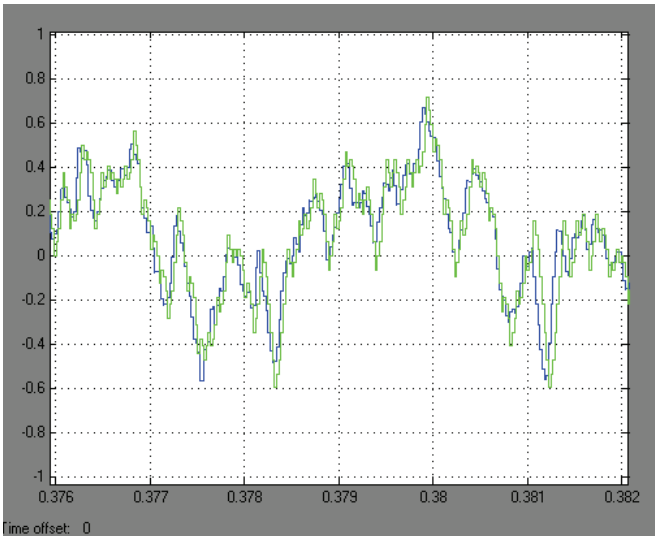


Fig. 7. Input and output signal MUX{ $x(i)$, $\hat{x}(i)$ } (zoomed).

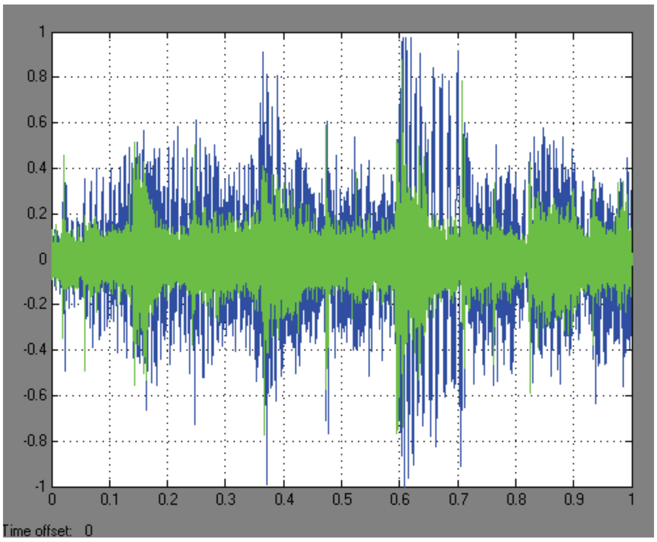


Fig. 8. Input and error signal MUX{ $x(i)$, $e(i)$ }.

On Fig.8 the input signal and the error signal are visualized together and on Fig. 9 the output code is presented.

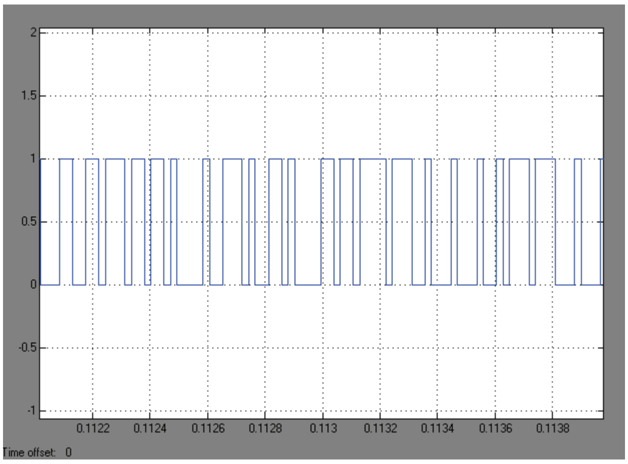


Fig. 9. Output code.

IV. Conclusion

An adaptive method for coding of one-dimensional digital signals, based on the delta modulation techniques and adaptation of the quantization error is presented. From the developed mathematical equations an algorithm and a general block scheme of adaptive delta modulation codec is synthesized and experimental results from the simulation by Simulink for Matlab 6.5 environment for test signals in WAV format are given.

The developed ADM codec provides minimum processing error and lied to increase of PSNR with about 0.3 dB in comparison with other non-adaptive prediction codecs.

The presented simulation model can be used in digital signal processing for spectral analysis, coding and transmission of one-dimensional signals and in distance learning by the using a Matlab Web Server.

The developed codec is used in laboratory work on the disciplines: "Image and Signal Processing" and "Audio and Video Communication on Internet" and in the experimental work in laboratory "Electronic System for Visual Information" in Technical University of Sofia.

V. References

[1] D. Stranneby, W. Walker, *Digital signal processing and applications*, Elsevier, 2004.

[2] S. Mitra, *Digital signal processing. A computer-based approach*, Mc Graw Hill, 2006.

[3] V. Madisetti, D. Williams, Eds. *The Digital Signal Processing Handbook*, CRC Press, 1998.

[4] A. D. Poularikas. *The Transforms and Applications Handbook*, Second Ed., CRC Press, 2000.

[5] G. Blanchet, M. Charbit, *Digital Signal and Image Processing Using MATLAB*, ISTE, 2006.

[6] B. Widrow, S. D. Stearns. *Adaptive Signal Processing*. Engelewood Cliffs, New York, Prentice-Hall, Inc., 1985.

[7] R. Gonzalesz, R. Woods. Digital Image Processing. 3-th Ed., Pearson Prentice Hall, 2008.

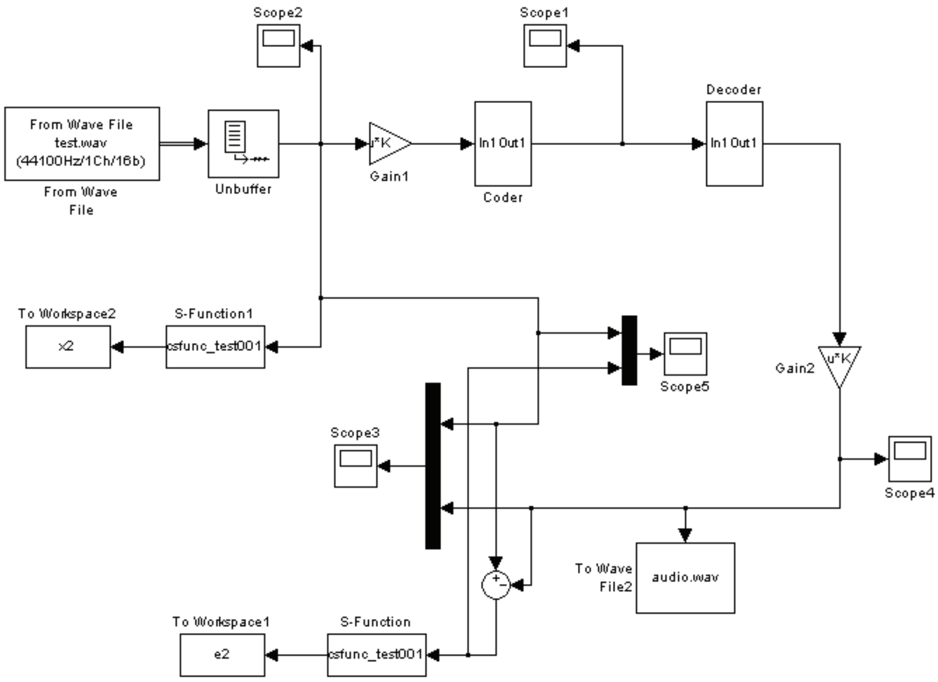


Fig. 1. ADM Codec.

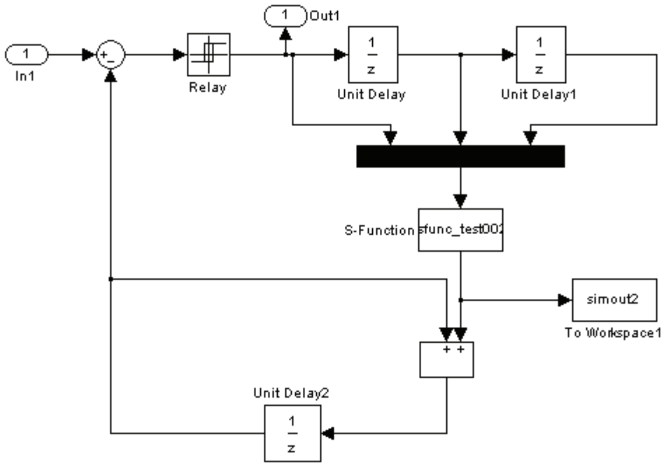


Fig. 2. ADM Coder.

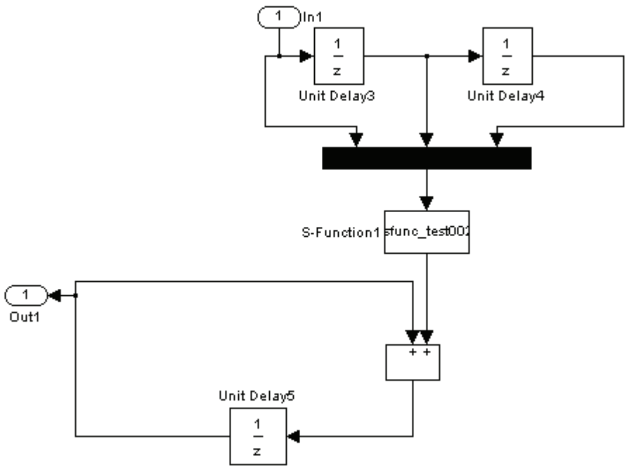


Fig. 3. ADM Decoder.