AUTOMATIC MODULATION CLASSIFICATION AND MEASUREMENT OF DIGITALLY MODULATED SIGNALS

D. Grimaldi [1], A. Palumbo [1], S. Rapuano [2]

(1) Dip. di Elettronica, Informatica e Sistemistica, Univ. della Calabria, 87036 Rende (CS), Italy.
Ph.:++39 0984 494712, Fax: ++39 0984 494713, E-mail: grimaldi@deis.unical.it.
(2) Facoltà di Ingegneria, Università del Sannio, Piazza Roma, 82100 Benevento, Italy.
Ph.: ++39 0824 305817, Fax: ++39 0824 21866, E-mail: rapuano@unisannio.it.

Abstract - *In the paper the Zero-Crossing-Sequence Shape method is described. The method effectively classifies the M-ary PSK and M-ary FSK modulated signals detecting both the phase and frequency shifts in the incoming signal by using the zero-crossing and instantaneous frequency variation sequences. Moreover, from these sequences, independently from the modulation classification, the Carrier-to-Noise-Ratio (CNR), the carrier frequency and the symbol rate are also determined. In order to make the method able to operate at lower CNR, pre-processing of the instantaneous frequency sequence and statistic hypothesis are assumed. Experimental results confirm both the high accuracy and the performance of the method.*

Keywords - PSK, FSK, modulation measurement, zero crossing.

1. INTRODUCTION

Modulation classification consists of extracting particular parameters from the modulated signal. Some of these parameters, as the modulation type and the carrier frequency, are used for demodulating the incoming signal.

The automatic modulation classification plays an important role in electronic surveillance systems, in military communications, in emitters intercepting, in signal verification and in interference identification.

In particular, great interest is devoted to the development of an automatic instrument able to characterise the signalling quality whichever digital modulation is used. In that direction, the first step is the development of an universal decoder able to recognise the modulation type. The great diversity of modulation scheme and the increasing activity in the frequency spectrum require an identifier able (i) to operate rapidly and automatically without any a priori knowledge of the modulated signal, and (ii) to correctly estimate in noise environment.

Many efforts have been made and several identifiers have been proposed in literature. They are based on (i) the frequency spectrum analysis, (ii) the Wavelet Transform, (iii) the neural tree network classification, and (iv) the zero crossing technique.

The identifier based on the spectrum analysis is formed via auto-regressive spectrum modelling [1] and the estimated parameters are the modulation type, the carrier frequency and the bit rate. This identifier does not operate correctly in estimation of the modulation type and of the carrier frequency, if the Carrier to Noise Ratio (CNR) is below 15 dB.

The identifier based on Wavelet Transform (WT) [2], [3] computes the WT magnitude function of the signal. Under the hypothesis that the ratio between the symbol period and sampling period is integer, it identifies the modulation by detecting the number of steps characterizing the magnitude function. This method is computationally intensive and requires the a priori knowledge of some modulation parameters at relatively high CNR.

The use of the neural network [4]-[6] permits to estimate the modulation type, the instantaneous frequency and the bandwidth better than the traditional methods at CNR higher than 15 dB.

The classification strategy followed in [7], [8] by using the zero crossing technique consists in computing the instantaneous frequency and its variation. These are used to classify and to explore the properties of the modulated signal.

The zero crossing sampling is an attractive method [9], [10] and it is used also in this paper to classify the digital modulated signals. The method proposed is based on extracting and analysing the properties of the Zero-Crossing-Sequence-Shape (ZCSS). According to [7], [8] the zero crossing sequence is obtained by processing the incoming signal and is used for distinguishing from single tone (MPSK) and multi-tone (MFSK) modulation. Differently from [7], [8] the successive classification on the base of the modulation parameters is developed by computing the instantaneous frequency variance and the levels and peaks in the instantaneous frequency sequence shape. In order to make the proposed identifier able to operate at low CNR, the instantaneous frequency sequence is opportunely pre-processed by means of a numerical technique including the integration and interpolation phases. Moreover, by assuming equally likely symbol changes in the signal, the correct classification is obtained at CNR as low as 11 dB.

The proposed identifier is computationally simple, robust and shows high accuracy. It is able to work correctly with the sampling frequency half of that one used in [2], [3]. In the paper the method is presented, the simulation and experimental results are discussed and the method performance is investigated.

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2. ZCSS METHOD

The incoming signal r(t) considered is modelled as:

$$r(t) = \tilde{s}(t)e^{j(2\pi f_c t+\theta_c)} + n(t) \qquad (1)$$

where $\tilde{s}(t)$ is a constant-envelope modulated signal, n(t) an Additive White Gaussian Noise (AWGN) source of power $E\left[\left|n(t)^2\right|\right] = 2\sigma_n^2$, and f_c and θ_c the frequency and phase of the carrier signal, respectively. In the following, $\tilde{s}(t)$ is one of the common digital modulated signals M-ary PSK and M-ary FSK, with M the number of different values of the parameter θ_c in the case of PSK modulation and of the different values of f_c in the case of FSK.

For PSK is:

$$\tilde{s}(t) = \sqrt{S}\sum_{i=1}^{M} e^{j\theta_i} u_T(t-iT), \qquad \theta_i \in \left[\frac{2\pi}{M}(i-1)\right], i=1,..., M \quad (2)$$

where S is the signal power and $\sqrt{S}$ the signal amplitude, u_T a unit height rectangular function in the time interval [0, T[, T the symbol duration.

For FSK is:

$$\tilde{s}(t) = \sqrt{S}\sum_{i=1}^{M} e^{j[2\pi[2_c+f_i)+\theta_i]} u_T(t-iT), \qquad (3)$$
$$f_i \in (f_1,...,f_M), \qquad \theta_i \in (0,2\pi)$$

where f_i is the frequency deviation.

Information for modulation classification are conveniently extracted by recording from the r(t) the zero-crossing sequence points x(i), i=1,…,N. This sequence is influenced by the noise level added to r(t). Indeed, very close zero-crossing points can occur as a consequence of the noise effect and not of the signal trend. Therefore, the effective zero-crossing points are hidden in the noise and erroneous information are extracted. An effective solution consists in the noise filtering, by means of numerical integration, in the time interval in which the zero-crossing point density is increasing. Denoting with r'(k) the values of the numerical integral of r(t), the effective zero crossing points can be detected by interpolating the values of r'(k), corresponding to the slope change.

In order to extract useful information from the zero-crossing sequence x(i), two others sequences are considered:

- the Zero-Crossing-Sequence-Shape (ZCSS) y(i)

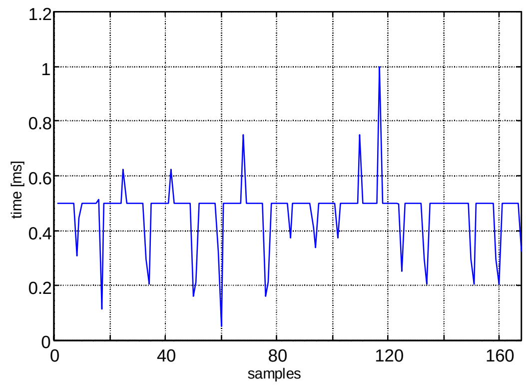
  y(i) = x(i+1)-x(i), i=1,…N-1; (4)

- the instantaneous frequency variation sequence z(i)

  z(i) = y(i+1)-y(i), i=1,…N-2 (5)

The ZCSS is also a measure of the instantaneous frequency and represents the time interval variation between two successive zero-crossing points as a consequence of the modulation.

The ZCSS is different according to the different modulations. Fig.1(upper) shows the ZCSS for PSK modulation in the ideal case without noise. The phase is constant during each symbol, and each peak corresponds to the phase change in the carrier signal occurring at each symbol change. The peak amplitude depends on the value of the phase change. Therefore, the different number of peak amplitudes gives information on the particular type of M-ary PSK modulation. Fig.1(lower) shows the ZCSS for FSK modulation in the ideal case without noise. In this modulation the frequency is constant during each symbol and changes from one symbol to another. Therefore, ZCSS is characterised by different levels, one for each frequency, and all these levels form a staircase characteristic. The number of different levels gives information on the particular type of M-ary FSK modulation.



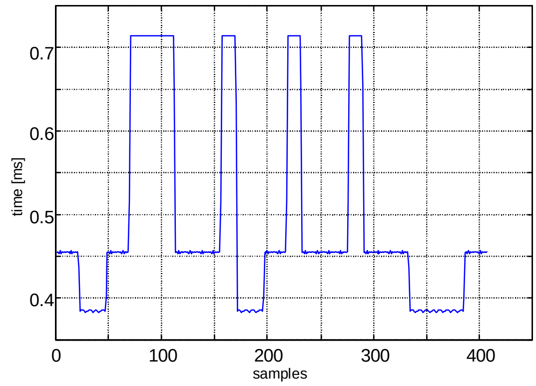


Fig.1 - ZCSS for PSK (upper) and FSK (lower) modulation in the ideal case without noise.

2.1 Classification between PSK and FSK modulation

As shown in the previous section, the ZCSSs are considerably different for the considered modulations. In the case of PSK the ZCSS is constituted by variable amplitude peaks, in the case of FSK by a staircase function with different levels. These differences are useful in order to distinguish and to classify between these two modulations.

The noise added to signal causes difficulties in the ZCSS identification and then a proper procedure is necessary for a correct identification. The z(i) sequence, defined by (5), is a measure of the variations of y(i) and it is constituted by spikes when variations occur in the y(i) sequence. In order to identify these spikes and distinguish them from spikes caused by noise, the new sequence $z_a(i)$ is defined. The sequence $z_a(i)$ is constituted by the dense portion of the density histogram of z(i).

From $z_a(i)$ sequence the variance σ^2_{za} is estimated so to select the spikes in z(i). The spikes, whose amplitude is lower than $3.034\,\sigma^2_{za}$, are discarding and those greater are selected to constitute the new sequence $y_a(i)$. The points of $y_a(i)$ correspond to variation caused by the modulated signal. In particular, if $z(i) > 3.034\,\sigma^2_{za}$, then y(i+1) corresponds to variation of the modulated signal r(t) and belongs to $y_a(i)$. Once determined the sequence $y_a(i)$, the mean value of the points on the left and right of the spikes of y(i), common to $y_a(i)$, are evaluated. In this manner, different levels can be detected. From this research PSK and FSK modulation are distinguishable.

2.2 *M-ary PSK modulation classification*

After the received signal is classified as M-ary PSK modulation, further classification is necessary. ZCSS is still used and both the amplitude and shape of the spikes are considered.

The different amplitude of the spikes in the sequence y(i) is a consequence of the fact that the time interval between two successive zero-crossing points increases or decreases respect to mean value according to the phase shift. Fig.2a shows the r(t) signal in the case of the 270° phase shift and Fig.2b shows the corresponding spike. In this case the time interval between two successive zero-crossing points increases. Fig.2c shows the r(t) signal in the case of the 315° phase shift and Fig.2d shows the corresponding spike. In this case different spike shape occurs. As a consequence of the fact that the time interval between two successive zero-crossing points decreases, two spikes 1 and 2 in Fig. 2d, very close, are determined.

The spike shape in Fig.2b indicates that the phase shift is proportional to its amplitude, differently, the spike shape in Fig.2d indicates that the phase shift is proportional to the sum of the amplitude of the first and second spike.

If CNR is low, difficulties arise in detecting spikes due to the noise presence. The spikes corresponding to phase shift 45° and 135° have high probability to be hidden in noise and, consequently, they are very difficult to detect. In order to overcome these difficulties, a long time duration of the incoming signal must to be examined. In this case is acceptable the statistic assumption of equal probability to detect all the symbols. Consequently, it is sufficient to detect any spikes with occurrence different from the occurrence of the others spikes to obtain correct classification.

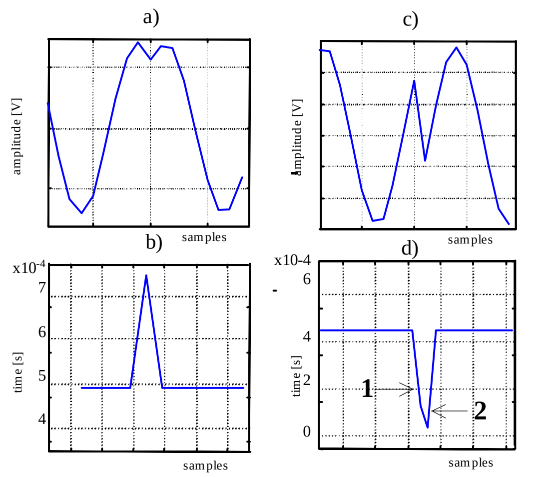


Fig.2 - Correspondence between phase shift and spike. a) 270°phase shift, b) corresponding spike, c) 315° phase shift and d) corresponding spike. In the latter case two partially overlapped spikes are determined and marked as 1 and 2 in (d).

2.3 *M-ary FSK modulation classification*

After the received signal is classified as M-ary FSK, further classification is necessary. The ZCSS is still used and only the different levels of the staircase function are considered.

Fig.3 shows the ZCSS in the case the levels are hidden in noise. In this case, the different levels can be detected by computing the mean value of the samples included in two successive symbol change.

If the CNR is low difficulties arise in detecting the levels. The critera followed in this case are similar to that considered in the previous section. Obviously, for M-ary FSK classification is necessary to detect the different levels of the staircase characteristic and not spikes.

2.4 *Carrier frequency and CNR estimation*

The carrier frequency f_c can be estimated by considering the $y_a(i)$ sequence. The N_{ya} terms of this sequence are the instantaneous frequencies, $f_i=1/(2y_a(i))$, $i=1,\ldots,N_{ya}$. Therefore, the carrier frequency can be estimated by averaging the instantaneous frequencies [7]:

$$f_c = \frac{N_{ya}}{2\sum_{i=1}^{N_{ya}} y_a(i)} \tag{6}$$

As demonstrated in [7], the CNR is estimated by using:

$$CNR = \frac{1}{(2\pi f_c \sigma_{za})^2}\left[3 + \frac{6\rho}{2f_c}\right] \tag{7}$$

where ρ is the normalised autocorrelation function of the noise.

2.5 *Symbol rate estimation*

The symbol rate can be estimated by considering the lower value of the time interval between two successive symbol transitions (T_s). The symbol rate is equal to the inverse of T_s. This estimation falls if the same symbol repeats twice successively.

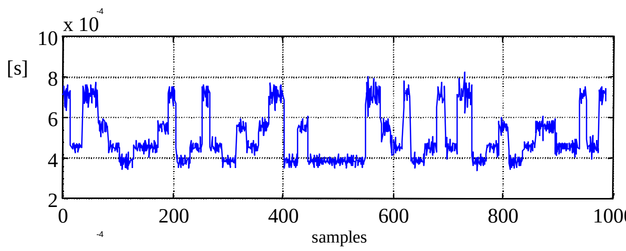


Fig.3 - ZCSS for FSK modulation in the case the levels are hidden by noise.

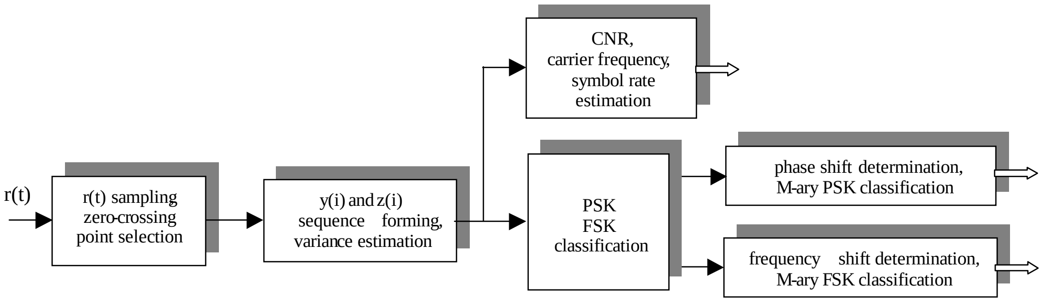


Fig.4 - Block scheme of the ZCSS method. The modulation classification and parameter estimation are organised as parallel processes.

2.6 Block scheme of the ZCSS method

Fig.4 shows the block scheme of the method. As the signal r(t) is sampled, the zero-crossing points are selected to form the sequence y(i) and z(i). The sequences $z_a(i)$ and $y_a(i)$ are formed once the variance $\sigma^2_{z_a}$ is estimated. PSK and FSK modulations are separated on the basis of the different levels checked by using $y_a(i)$. If PSK modulation is detected, all the spikes are taken into account to determine the set of the phase shift and to classify the M-ary PSK modulation. If FSK modulation is detected, all the levels are taken into account to determine the set of the frequency shift and to classify the M-ary FSK modulation. Moreover, in parallel way the signal parameters, including carrier frequency, SNR and symbol rate, are estimated.

3. EXPERIMENTAL RESULTS

The ZCSS method has been implemented in Matlab software environment. Fig.5 shows the Graphical User Interface (GUI) that permits the classification tests to be quickly and easily executed. The button *LOAD* permits to load the acquired and stored signal to be examined. The button *START* runs the classification method. The button *TEST* is used to run the procedure that generates test signals. These signals can be used (i) to verify the correct execution of the classification procedure, and (ii) to evaluate the performance of the classification procedure. Fig.6 shows the GUI of *TEST* procedure. The test signal is generated once fixed (i) modulation, (ii) number of symbol, (iii) time of symbol, (iv) CNR, (v) carrier frequency, and (vi) sampling frequency f_s.

3.1 Evaluation of the ZCSS method performance

The performance of the ZCSS method was evaluated by using the *TEST* procedure. Test signals were generated for three different PSK modulations (BPSK, QPSK and 8PSK) and for three different FSK modulations (BFSK, QFSK and 8FSK). The characteristic parameters were: f_c/f_s=0.1, T_s=100 samples, symbol number in each signal record equal to 100.

The symbols were random generated. The noise was white Gaussian with zero mean. The frequency deviation f_d, in the FSK modulation, was f_d/f_c=0.5 for BFSK, f_d/f_c=0.25 for QFSK and f_d/f_c=0.15 for 8FSK.

In all the tests no a priori knowledge was assumed for the classification.

The results of tests carried out by generating 1000 signals for each different modulation demonstrate correct modulation classification for CNR≥11dB (Fig.7). At CNR=10dB 100% correct classification was observed for BPSK, BFSK, QFSK and 8FSK, error in QPSK and 8PSK was equal to 4%.

Numerous tests were carried out for parameter estimation, CNR and carrier frequency. Tab.1. shows the results generating 1000 signals for each of the different modulations, in the case: CNR=15dB, f_c=1kHz and f_c/f_s=0.2. On the basis of all the tests, it can be noted that a higher sampling frequency and consequently values of f_c/f_s<0.2 do not modify the accuracy and the Standard Deviation (SD).

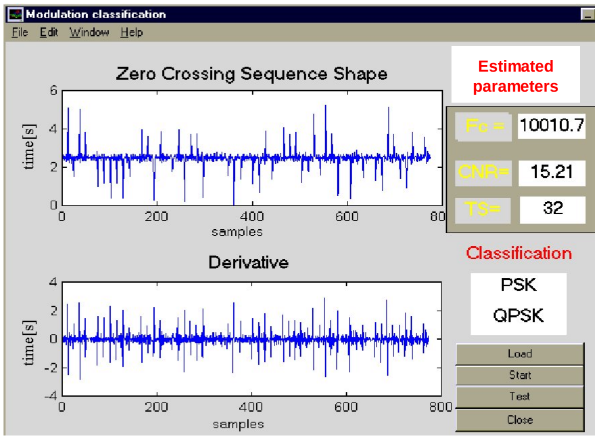


Fig.5 - Graphical User Interface for modulation classification.

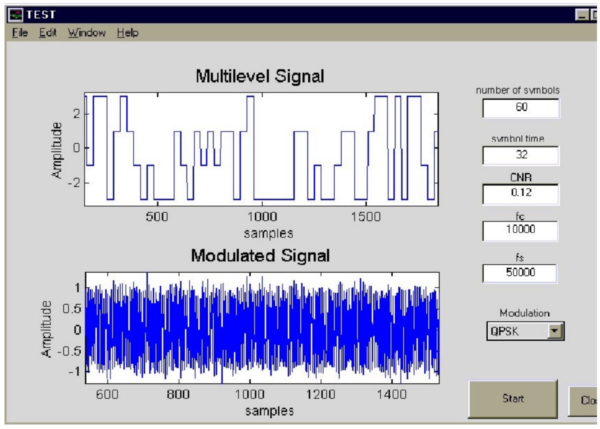


Fig.6 - Graphical User Interface for test signal generation.

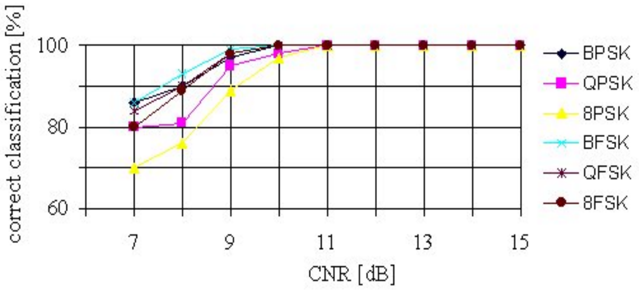


Fig.7 - Percentage of correct classification versus the CNR for three different PSK and three different FSK modulations.

3.2 Modulation classification in the telephone band

In order to investigate the influence of the reduced transmission band on the correct classification several tests were carried out. The signals taken into account were in the telephone band [300 Hz, 3.4 kHz]. The modulated signals were generated by means of a DSP board TMS320C31 by Texas Instruments [11]. The measurement station, organised to acquire the modulated signals, was equipped by means of the DAQ Lab-PC-1200 by National Instruments [12], [13].

The characteristic parameters were: f_c=1.8kHz, f_s=18kHz, number of symbols in each record equal to 100, f_d/f_c=0.25 for BFSK, f_d/f_c=0.20 for QFSK, and f_d/f_c=0.15 for 8FSK.

The results of the tests demonstrate correct classification for CNR$\geq$17dB. In this particular situation the increase of the CNR is caused by the increased deformations of the modulated signal.

4. CONCLUSIONS

A classification method of M-ary PSK and M-ary FSK modulations without requiring any a priori knowledge of the incoming signal has been presented. The method is based on the analysis of the Zero-Crossing-Sequence-Shape.

The experimental results are encouraging. As a matter of fact, the tests, carried out in simulation, show correct classification for CNR greater than 11dB with an increase of the CNR for correct classification in band limited systems.

Further research topics have been scheduled as follows: (i) method extension to the case of few acquired points per signal period; (ii) method extension to other digital modulations as MSK, QAM, xDSL; (iii) improvement of the method performance for band limited transmissions; and (iv) method implementation on a DSP.

Table 1 - Mean values of the CNR and carrier frequency, by considering 1000 signals for each modulation, and standard deviation (SD).

	CNR [dB]	SD [dB]	f_c [Hz]	SD [Hz]
BPSK	15.53	0.65	999.3	7.33
QPSK	15.30	0.67	1001.4	3.29
8PSK	15.02	0.65	1007.5	1.71
BFSK	15.35	0.32	1011.7	1.95
QFSK	15.39	0.24	1014.6	2.34
8FSK	15.43	0.26	1020.5	2.24

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REFERENCES

[1] K. Assaleh, K. Farrel, R.J. Mammone, “A new method of modulation classification for digitally modulated signals”, Military Communications Conference, MILCOM '92, vol.2, 1992, pp. 712-716.

[2] Y.T. Chan, J.W. Plews, K.C. Ho, “Symbol rate estimation by the wavelet transform”, Proc. of IEEE ISCAS'97, Hong Kong, 1997, pp. 177-181.

[3] K.C. Ho, W. Prokopiw, Y.T. Chan, “Modulation identification of digital signals by wavelet trasform”, Military Communications Conference, 1998, pp. 325-330.

[4] K.R. Farrel, R.J. Mammone, “Modulation classification using a neural tree network”, Military Communications Conference, MILCOM '93, vol. 3, 1993, pp. 1028-1032.

[5] Y. Yang, C.H. Liu, “An asymptotic optimal algorithm for modulation classification”, IEEE Transactions on Communication Letters, vol.2, N° 5, 1998, pp. 98-100.

[6] D. Boiteau, C. Le Martrel, “A general maximum likelihood framework for modulation classification”, IEEE Trans. on Acoustics, Speech and Signal Processing, 1998, vol. 4, 1998, pp. 2165-2168.

[7] S.Z. Hsue, S.S. Soliman, “Automatic modulation using zero crossing”, IEE Proceedings, vol. 137, part F, December 1990, pp. 459-464.

[8] B. M. Sadler, S. D. Casey, “Frequency estimation via sparse zero crossings”, IEEE Trans. on Acoustics, Speech and Signal Processing, 1996, ICASSP-96, vol. 5, 1996, pp. 2990-2993.

[9] P. Daponte, D. Grimaldi, A. Molinaro, Ya.D. Sergeyev, “An algorithm for finding the zero crossing of time signals with Lipschitzean Derivatives”, Measurement, vol.16, 1995, pp. 37-49.

[10] P. Daponte, D. Grimaldi, A. Molinaro, Ya.D. Sergeyev, “Fast detection of the first zero-crossing in a measurement signal set”, Measurement, vol.19, No.1, June 1996, pp. 29-39.

[11] TMS320C3X, User’s Guide, Texas Instruments, 1994.

[12] Lab-PC1200/AT, User Manual, National Instruments, 1999.

[13] A. Aiello, D. Grimaldi, ”A virtual instrument for the testing of DTMF signal decoder”, Measurement, in press.