Regular paper

Recognition of narrowband radio signals using autoregressive models and pattern comparison approach

Jerzy Łopatka

Abstract — This paper presents an improved spectral recognition method for digitally modulated radio signals. It is based on a signal autoregressive (AR) model. Model poles are used as signal features for neural network based on recognition process. To reduce an influence of the signal noise and distortions introduced by the digital receiver, a nonlinear *Z* plane transformation is proposed.

Keywords — ***signal models, modulation recognition.***

1. Introduction

Automatic radio signal recognition and classification is an essential problem in adaptive radio links, radio monitoring and electromagnetic compatibility analysis. Many authors focused on this problem during last years, but usually they take a lot of simplifying assumptions. They assume that the center frequency is known [8], data rate is given [3], perfect bit synchronization is performed [3] or data stream is known [1]. Unfortunately, in real system these assumptions are invalid.

In this paper, we propose an asynchronous approach, i.e., we assume that there is no a priori knowledge about signal parameters. The proposed method enables fast and reliable recognition of signals with unknown parameters. It is based on spectral analysis, but the number of parameters for signal description is minimized to reduce computational effort and memory requirements. For this purpose, a parametric model of the signal is used. It is an adaptive filter described by a set of parameters corresponding to the analyzed signal. To achieve a precise description of the signal, both model parameters, order and structure should be selected properly according to the structure of the signal under consideration. For narrowband radio signals, AR models can be used because AR spectral estimates have sharp peaks that correspond with carriers in radio signals. The AR method is computationally effective and has high spectral resolution for short data sequences. To achieve precise carrier frequency estimation and to avoid spectral peaks splitting, the LS method [9] has been chosen.

2. Selection of parameters used for signal recognition

To use model parameters in the recognition process, they must be complete and distinctive, i.e., contain the main part of information concerning the signal and enable signal recognition. The parameters should be also resistant to radio signal distortions and frequency offset caused by inaccurate receiver tuning or the Doppler shift. Algebraic relations between parameters of the original signal and distorted or shifted one should be easy to determine, the shift should be easy to calculate and some simple transformations should enable correcting the shift.

Adaptive AR filter, being a signal model, is usually described by its transfer function parameters, reflection coefficients (for lattice structure) or poles layout:

$$H(z) = \frac{1}{b_1 z^{-n} + .. + b_{n-1} z + b_n} = \frac{A}{\prod_{k=1}^{n} \left(1 - d_k z^{-1}\right)}, \quad (1)$$

where: b_i – filter parameters, d_k – filter poles, A – constant.

In previous works [7] it was proved that the values of transfer function parameters b_i depend strongly on the signal frequency, initial phase and frequency shift. The distances between these parameters for uncorrelated realizations of the same signal are so large, that recognition on their basis is impossible. Dissipation of reflection coefficients for the lattice structure is much smaller, but relations between coefficients change in the case of the frequency shift. These parameters cannot be used for radio signals recognition either. As an alternative it is possible to use the poles d_k layout. This enables control and correction of the filter stability. If any of poles is outside the unit circle, it can be simply reflected inside to correct the problem [5]. In the case of the frequency shift, the poles are simply rotated on the ***Z*** plane. The phases of all poles are changed by the same value, whereas the poles magnitudes remain unchanged [7]. This enables simple correction of poles' locations to eliminate the frequency shift.

3. Model order selection

Some standard methods can be used to select the AR model order [5], but these methods are inaccurate. If the model order is underestimated, some part of information may be lost. In the case of overestimation, additional, false spectral peaks may occur in the spectral estimate. One or two additional poles may cause a considerable change of the spectral estimate. Moreover, the radio signals change continuously and it would be necessary to estimate the model order after each realization.

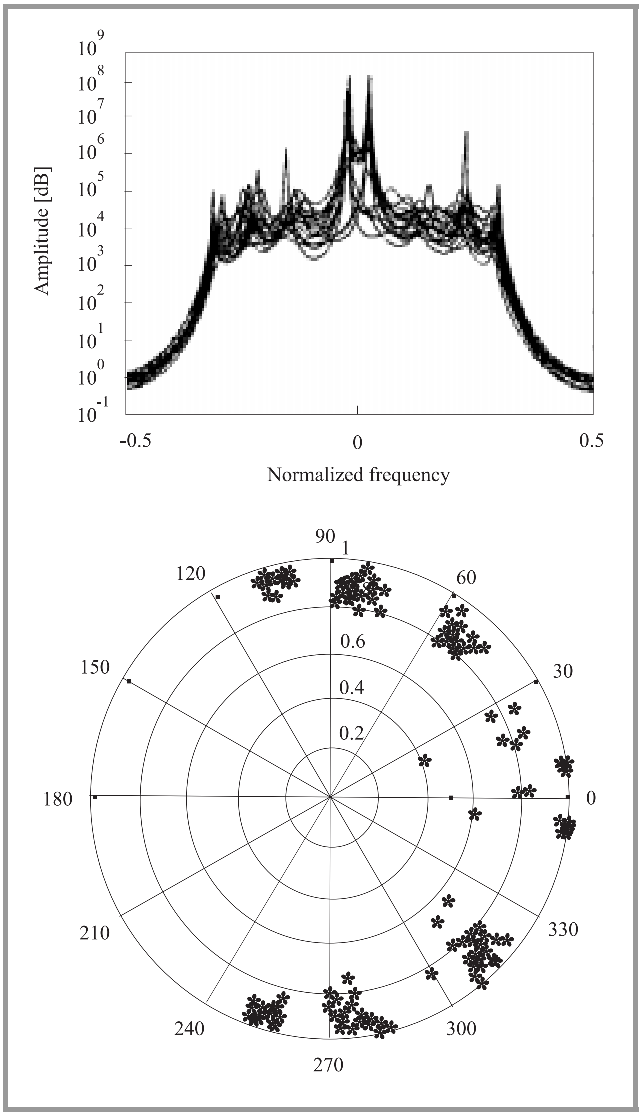

Fig. 1. FSK signal spectrum estimate and poles location for AR model of order 8.

On the other hand, radio signals are usually noisy because of influence of the radio channel, so the received signal consists of determined components and the noisy background. After a number of tests, an overestimated, the constant model order equal 8 was selected. Usually 1, 2 or 3 "signal" poles with magnitudes near to 1 correspond to the analyzed signal, whereas the several additional "noise" poles (lying near to the circle centre) create a quite good model of the noisy, spectrally flat environment of the useful signal (Fig. 1).

4. Correction of the receiver distortions

According to the previous works [4], the noisy environment may be modeled by equally spaced poles located along the unit circle. It means that poles magnitudes should be approximately equal to each other and should not depend on the pole phase. In real digital radio receivers this assump-

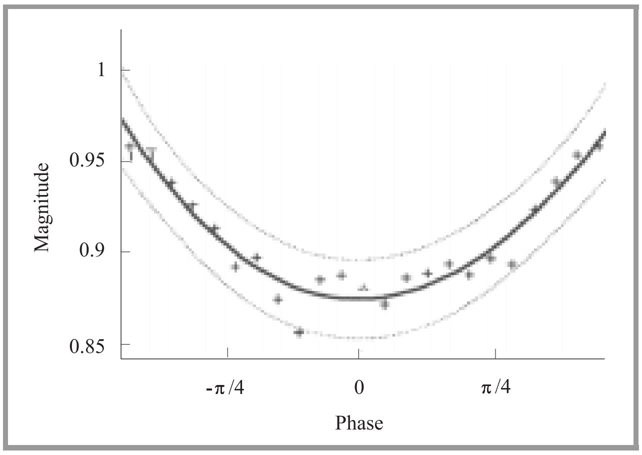

Fig. 2. Location of poles for narrowband noise and interpolating polynomial.

tion is invalid, because they realize complex processing of the signal. They use digital downconversion, decimation and lowpass filtration. To provide good selectivity they also use an oversampling, equal minimum 1.7 for the 3 dB output bandwidth [2]. This causes the channel noise to become a narrowband noise, and magnitudes of "noise" poles depend on their phases (Fig. 2). To compensate the filtration problem, the average magnitudes of poles versus their phases were calculated and interpolated using the 4th order polynomial. The calculated values of the polynomial for phase values $[-\pi;\ \pi>$ were used to create a normalized correction table $C(\varphi)$. Its maximum value equals 1. The calculated magnitudes of poles are multiplied by correction coefficients from this table:

$$\left|\tilde{d}_k\right| = C\left(arg\left(d_k\right)\right)\ \left|d_k\right| . \tag{2}$$

This correction gives approximately the same magnitudes for all "noise" poles.

5. Poles location transformation

The model poles can be divided into "signal" and "noise" poles. The "signal" poles are located on or near to the unit circle and have low dissipation for successive realizations of the signal. It is an important advantage from recognition point of view. The "noise" poles located along the unit circle but nearer its center have much larger dissipation. They depend on the noise components which are not very important. Moreover, the influence of poles on the signal spectral estimate becomes lower for larger distance from the unit circle. Thus the influence of "noise" poles on final spectrum is lower then "signal" poles. Because pattern recognition consists in calculation of distances between stored patterns and the received one, distances between all corresponding poles should be taken into consideration. From this point of view, "noise" poles have too high weight. Distances between them should be smaller to decrease their dissipation and decrease their influence on the final decision. It can be

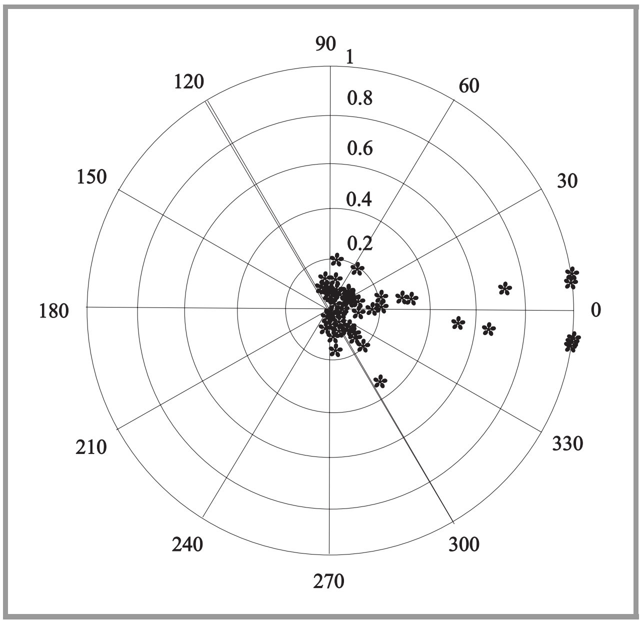

Fig. 3. FSK signal poles after compensation of narrowband noise and transformation.

done by shifting them towards the ***Z*** plane origin, without changing their phases:

$$|\hat{d}_k| = \frac{|\tilde{d}_k|}{1-|\tilde{d}_k|}\left[\max\left(\frac{|\tilde{d}|}{1-|\tilde{d}|}\right)\right]^{-1}. \tag{3}$$

After transformation, the filter poles are shifted towards the ***Z*** plane origin according to the distance of each pole from the unit circle and from another side this decreases the distance between "noise" poles and increases the distance between "signal" and "noise" poles (Fig. 3).

6. Pattern recognition

Determination of distances between received and stored patterns is the next step of the recognition process. The minimum distance indicates the type of the received signal. Its value is a confidence rate defined as:

$$\mathrm{D} = min\left(\sum_{j=1}^{w}\sum_{i=1}^{k}|\hat{d}_j - d_i|^2\right), \tag{4}$$

where: p – number of poles, w – number of classes, $\hat{d}_j$ – received pattern poles, d_j – stored pattern poles.

Distances should be calculated between corresponding poles. It is a nontrivial problem if the errors of receiver tuning occur. To diminish these errors, a 3-step preprocessor is proposed. First the preprocessor performs a shift procedure:

$$d'_k = Re(d_k) - Re(d_{med}) + j(Im(d_k) - Im(d_{med})), \tag{5}$$

where: d_{med} – the gravity center of the pattern. This operation eliminates errors of overlaying of the stored and received patterns. The second and the third step is scaling and rotation:

$$d''_k = \frac{d'_k}{Ke^{j\Phi}}, \tag{6}$$

where: K – scaling factor, Φ – rotation factor.

$$K = \frac{1}{p}\left(\sum_{i=1}^{p}|d'_j|\right); \quad \Phi = arg(d_{med}). \tag{7}$$

Scaling factor K corrects changes of the pattern size, that can be caused e.g. by changes of the SNR. Rotation eliminates the frequency shift.

7. Efficiency analysis

To check the efficiency of the proposed method some measurements were made. Selected digitally modulated signals: 2FSK, 2PSK, 4PSK, 8PSK and 16QAM with known S/N ratios, data rates and deviation (for FSK) were recorded using a digital receiver. The signals were divided into short, 40 sample segments and for each segment the layout of its poles was calculated. On the basis of the transformed poles pattern on the ***Z*** plane, neural network training for pattern recognition was performed. The network consisted of 2 layers and 60 neurons with the log-sigmoid transfer function. It was trained using the back propagation learning algorithm. The training set consisted of 20 records for each type of the signal, with SNR varying from 0 dB to 20 dB. The achieved results are shown in Fig. 4.

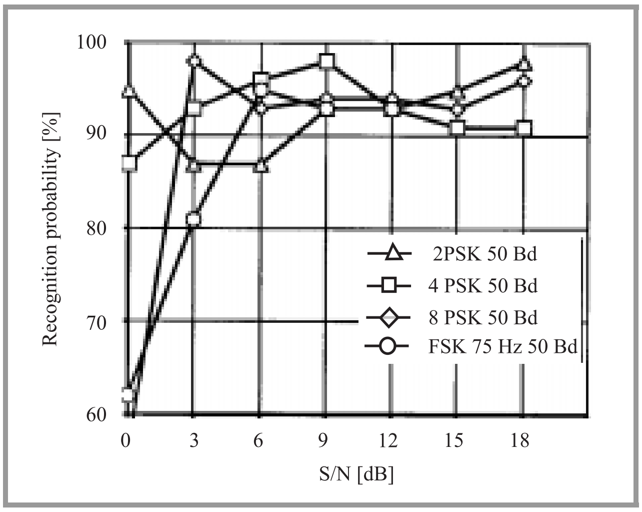

Fig. 4. Percentage efficiency of signal recognition versus S/N ratio for: 2, 4, 8 PSK 50 Bd and FSK 50 Bd with frequency shift 75 Hz.

The computational effort for proposed method is a sum of computations for AR parameters estimation C_{AR} [9], necessary corrections C_C and neural network operation C_N:

$$C = C_{AR} + C_C + C_N = (MN + 6M^2) + \\ +17M + (2MF + 2F + 2FS + 2S), \tag{8}$$

where: N – length of signal record, M – order of the model, F, S – number of neurons in the first and the second layer.

For values of parameters used in the simulation, a single recognition calculation required less than 2000 operations that makes the method suitable for real time applications.

8. Conclusions

The autoregressive models with neural networks can be used for recognition of the digital radio signals. The description of each signal is very concise. The method is useful for discrimination signals with different spectral shapes. It performs well for SNR better than 3 dB. Identification of signals with similar spectral shapes or different data rates can be performed in the next stage using additional distinctive features. This method may be also implemented as the initial step of selection in intelligent multipurpose demodulators and radio monitoring systems.

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Jerzy Łopatka received the M.Sc. and Ph.D. degrees in communications from the Military University of Technology (MUT), Warsaw. At present he is a Head of Radio and Electronic Warfare Section in the Communication Systems Institute (MUT). His main research interests include digital signal processing in wireless systems.
e-mail: jlopatka@wel.wat.waw.pl
Communication Systems Institute
Military University of Technology
Kaliskiego st 2
00-908 Warsaw, Poland