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Method of Complex Envelope Processing for Signal Edges Detection

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Problem statement. The need of information processing automation in modern radio monitoring systems stimulates development of flexible methods for signal detection and its parameters estimation in time domain. A priori uncertainty of signal time-frequency structure complicates the automatic determination of signals edges. **Purpose**. The purpose of the article is subsequent automation of radio frequency spectrum analysis process by developing and implementing a method for determining signals time edges under conditions of a known noise power and signal-to-noise ratio. **Method**. To determine signal time edges in given frequency channel, square of signals' complex envelope is first calculated, smoothed with moving average window and compared with threshold. Threshold is calculated as a quantile of gamma distribution using Wilson-Hilferty approximation of χ^2 distribution quantiles for a given probability of false alarm. An analytical expression is obtained for calculation length of moving average window depending on signal-to-noise ratio. An algorithm has been developed for determining signals' time parameters and filtering them by duration. Unknown noise power value in frequency channel can be replaced by its estimate under the assumption that frequency channel is not constantly occupied and noise level is estimated on signal-free time intervals. **Conclusions**. Proposed method makes it possible to automatically determine edges of signal with an arbitrary structure at signal-to-noise ratio values from -6 dB. Adjustable length of moving average window makes it possible to reduce the error in determining signal time parameters by 2-4 times with an increase in the signal-to-noise ratio compared to a fixed window length. Prospects for further research in this direction should be focused on development and implementation of methods for detection signal edges under conditions of an unknown noise level.

Keywords: signal edges; complex envelope; moving average; threshold; signal structure; signal-to-noise ratio

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Introduction

Rapid development of wireless radio systems [1] requires automation of information processing in modern radio monitoring systems [2, 3]. In particular, when conducting radio monitoring, it is necessary in real time to determine radio signals edges in order to establish the moment of start and end of recording in given frequency channel. Also, this problem arises when analyzing recordings of wide frequency bands, especially for processing pulse signals or signals transmitted by packets, when it is necessary to remove noise areas between signals to recognize the type of modulation and further processing. A priori uncertainty of signal time-frequency structure makes it difficult to automatically determine signal edges. Therefore, development of flexible methods for detection and estimation signals' time parameters is actual scientific and technical task.

1 Related works

The issue of detection pulse signals in time domain and estimation their parameters are considered in many works by researchers from all over the world. In particular, in [4] it is proposed to determine pulses duration and their repetition period using Haar filter and ratio of sliding sums, and in [5] signals edges are detected using continuous wavelet transform. Estimation of signals time parameters by smoothing its instantaneous energy using window functions is proposed in [6]. In [7] image processing methods were used to detect pulses edges. In [8] it is proposed to estimate signals time parameters by analyzing time-frequency transforms. The method works with a minimum signal-to-noise ratio (SNR) of -5 dB. To determine pulse signals parameters it is proposed in [9] to use a neural network. In [10] sequential analysis in form of a cumulative sum algorithm was used to detect pulse signals. In [11, 12] rough detection of signal begin and end is carried out using a spectrogram with further refinement of its edges using a special filter [13]. To determine signal edges it is proposed in [14] to calculate the smoothed complex envelope of signal.

As a result of latest research and publications analysis, it was established that the general drawback of considered methods is using algorithms with fixed parameters, which in case of known SNR value, does not make it possible to achieve potential accuracy of signal time parameters estimation. Also remains unsolved the task of automatically determining signals edges with an unknown time-frequency structure.

2 Problem statement

The purpose of the article is subsequent automation of radio frequency spectrum analyzing process by developing and implementing a method for determining signals edges under conditions of a known noise power and signal-to-noise ratio.

3 System model

If the noise level is known, its value must be used to calculate threshold for signals edges detection. Noise power can be estimated using known methods [15], even with unknown occupancy of analyzed frequency band. Radio signals to be processed in receiving frequency channels can have coherent (with regular phase structure) or non-coherent (noise-like) structure. Vector of complex received signal in selected frequency channel may be written in following form:

$$\dot{\mathbf{x}} = \dot{\mathbf{s}} + \dot{\xi}. \tag{1}$$

Vector of complex signal samples can be written as vectors sum of in-phase and quadrature signal components:

$$\dot{\mathbf{s}} = \mathbf{s_I} + j\mathbf{s_Q}, \tag{2}$$

where $j = \sqrt{-1}$ – imaginary unit.

By analogy, vector of complex Gaussian noise samples we will write in following form:

$$\dot{\xi} = \xi_\mathbf{I} + j\xi_\mathbf{Q}, \tag{3}$$

where $\xi_\mathbf{I}$ – vector of in-phase noise component; $\xi_\mathbf{Q}$ – vector of quadrature noise component.

Vectors $\xi_\mathbf{I}$ and $\xi_\mathbf{Q}$ are also normally distributed with zero mean and equal standard deviations (STD) $\sigma_{\xi_I} = \sigma_{\xi_Q}$. Then vector $\dot{\xi}$ will also have zero mean and its STD calculates according to next equation:

$$\sigma_\xi = \sqrt{\sigma^2_{\xi_I} + \sigma^2_{\xi_Q}} = \sqrt{2}\sigma_{\xi_I} = \sqrt{2}\sigma_{\xi_Q}. \tag{4}$$

To determine signal edges in given frequency channel, we will calculate signal complex envelope as follows:

$$\mathbf{E_x} = \mathbf{x_I^2} + \mathbf{x_Q^2}. \tag{5}$$

This approach to the calculation of complex envelope will simplify further mathematical calculations, since $\mathbf{E}_\xi$ samples for noise will be subject to exponential probability density function (PDF).

Then threshold for probability of false alarm P_F is calculated according to the following equation:

$$\gamma = \sigma^2_\xi \ln\left(-P_F\right). \tag{6}$$

When smoothing complex envelope $\mathbf{E}_\xi$ with moving average window of L samples length, vector of smoothed samples $\mathbf{C}_\xi$ will be subject to gamma distribution [16] and will be described by the following expression:

$$p_{\mathbf{C}_\xi} = C_\xi^{L-1} \frac{e^{-C_\xi} L^L}{\Gamma\left(L\right)}. \tag{7}$$

Threshold value γ_2 for smoothed complex envelope $\mathbf{C}_\xi$ can be calculated as a quantile of gamma distribution of $p = 1 - P_F$ level. To do this, we will use the quantile approximation of gamma distribution using the chi-square distribution χ^2 [17]:

$$\gamma_p = \chi^2_p \left(2\left(L+1\right)\right), \tag{8}$$

where $\chi^2_p\left(2\left(L+1\right)\right)$ is p level quantile for χ^2 distribution with $K = 2\left(L+1\right)$ levels of freedom.

Using the Wilson-Hilferty approximation of χ^2 distribution quantiles [17], threshold value γ_2 can be calculated using the following expression:

$$\gamma_2 = \sigma^2_\xi \left(1 - \frac{2}{9K} + u\sqrt{\frac{2}{9K}}\right)^3, \tag{9}$$

where

$$u = \frac{1,24 + 0,85H^{0,657}}{1 + 0,0001H^{-3} + \frac{2,38}{H}}, \quad H = -0,96\ln\left(\frac{P_F}{1-P_F}\right). \tag{10}$$

Value of moving average window length L depends on SNR value and should be increased when SNR decreases. Therefore, signal processing in each frequency channel will depend on SNR. This approach will ensure optimal distribution of computing resources, since smoothing signal envelope requires additional calculations.

Consider case when signal has an incoherent structure, rectangular envelope and differs from noise only in power. With low values of SNR, L must be chosen large (hundreds). In this case, PDF of $\mathbf{C}_\xi$ can be considered normal and threshold can be calculated according to the following expression:

$$\gamma_2 = m_{C_\xi} + \alpha\sigma_{C_\xi}, \tag{11}$$

where $m_{C_\xi} = \sigma^2_\xi$ – mean of $\mathbf{C}_\xi$; $\sigma_{C_\xi} = \frac{\sigma^2_\xi}{\sqrt{L}}$ – STD of $\mathbf{C}_\xi$; α – coefficient, which value depends on the false alarm probability.

Then L value will be determined on the condition that the minimum values of smoothed complex envelope of signal and noise mixture will be greater than threshold. Then γ_2 for smoothed complex envelope can be written similarly to expression (11):

$$\gamma_2 = m_{C_x} + \alpha\sigma_{C_x}, \tag{12}$$

where $m_{C_x} = \sigma_s^2 + \sigma_\xi^2$ – mean of signal and noise mixture smoothed complex envelope;

$\sigma_s = \sqrt{\sigma_{s_I}^2 + \sigma_{s_Q}^2}$ – STD of complex signal;

$\sigma_{C_x} = \frac{\sqrt{\sigma_s^2+\sigma_\xi^2}}{\sqrt{L}}$ – STD of signal and noise smoothed complex envelope.

Equating expressions (11) and (12), we get:

$$L = \alpha^2\left(\frac{1}{q} + \sqrt{1+\frac{1}{q}}\right)^2, \tag{13}$$

where q – SNR by power (not in dB).

Figure 1 shows graphs of moving average window length L dependences via SNR for different values of α parameter. From these curves, it can be seen that at high SNR (20 dB and more), value of window length L approaches to α^2. Resulting L value can be used to smooth signals with any structure.

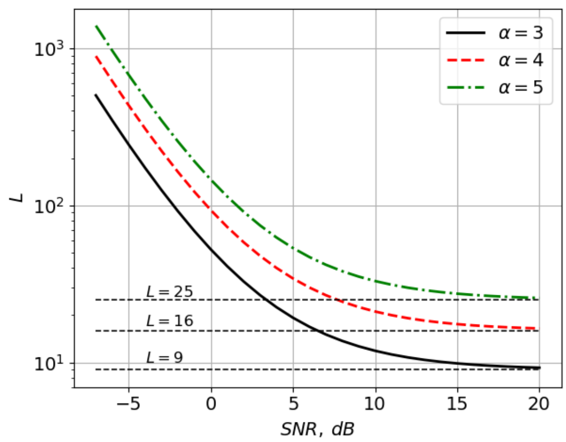


Fig. 1. Moving window length dependence via SNR

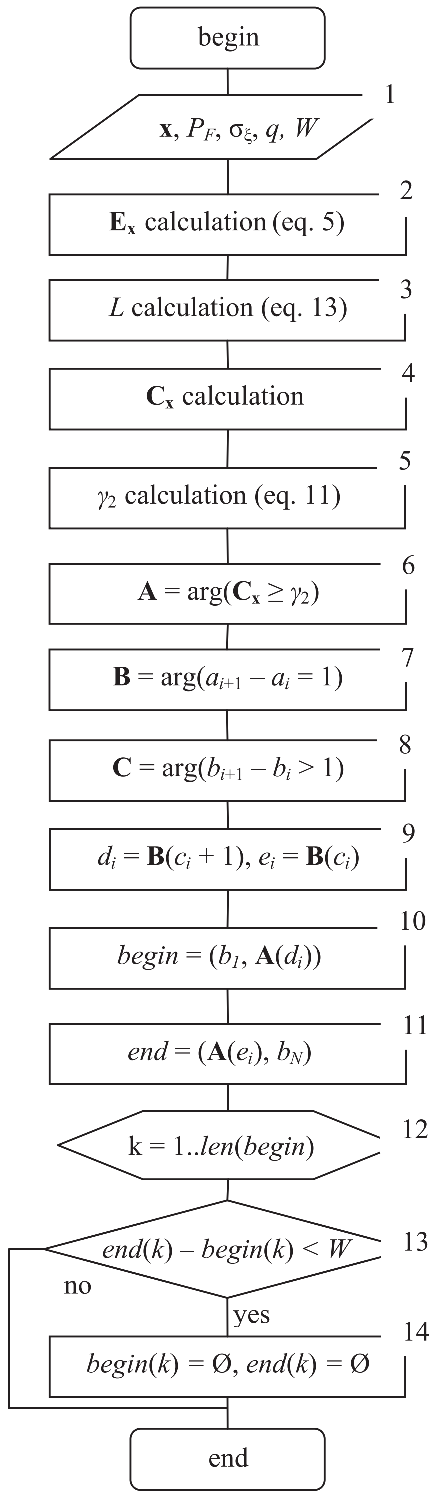


Fig. 2. Block diagram of signal edges detection algorithm

4 Algorithm

Figure 2 shows block diagram of signal edges detection algorithm in frequency channel. For a non-zero P_F value, it is possible to incorrectly determine signal edges. As P_F decreases, probability of signal edge detection will also decrease. Therefore, it is advisable to choose exceeding specified threshold by a certain number W of consecutive samples as a criterion for detection signal edges.

The following input data are required for algorithm implementation: vector of received signal samples $\mathbf{x}$, noise STD σ_ξ, P_F, SNR value q and W (block 1). When calculating length of moving average window L according to expression (13), value of α coefficient is recommended to be chosen depending on SNR: at $q \geq 12$ dB – $\alpha = 3$; at $12\,\text{dB} > q \geq 5$ dB – $\alpha = 4$ and at $q < 5$ dB – $\alpha = 5$. In block 6 is formed a vector of received signal samples numbers **A** that exceeded threshold γ_2. In blocks 7-11 vector **A** is processed, as a result of which two new arrays of samples numbers are formed: *begin* – array of numbers of beginnings and *end* – endings of signal samples sequences. In blocks 12-14, values of *begin* and *end* arrays are filtered and only those signals whose duration is at least W samples remain.

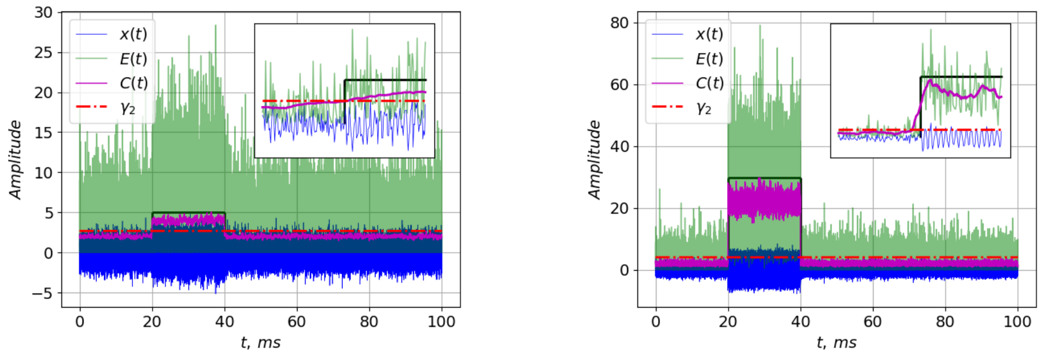
If noise STD is unknown, then in order to detect signal edge in frequency channel, the unknown value of noise power can be replaced by its estimate, assuming that the frequency channel is not constantly occupied and noise power is estimated at signal-free time intervals. Moreover, method of noise power estimation should provide its' reliable estimate with a sample of small size. In this case, the signal may be present or absent in frequency channel at moment of observation. Then time of signal start (end) may be estimated by a sharp change in power level in given frequency channel.

However, only the known value of noise STD will not make it possible to determine SNR and calculate optimal L value. In this case, it is necessary to choose L and α values for the worst expected case, or based on such a minimum value of SNR, at which further processing of signal (recognition, demodulation) will be impractical.

5 Simulation results and discussion

To study developed method were used signals with two types of structure: coherent – a pulse with a rectangular envelope and non-coherent (like OFDM) – a fragment of white Gaussian noise also with a rectangular envelope. During simulation, sampling frequency was 10 MHz, carrier frequency of coherent signal was 1,6 MHz. Duration of studied signal and noise mixture fragment was 100 ms. Signal starts at 20 ms and ends at 40 ms. Probability of a false alarm during simulation was set at the level of 10^{-4}.

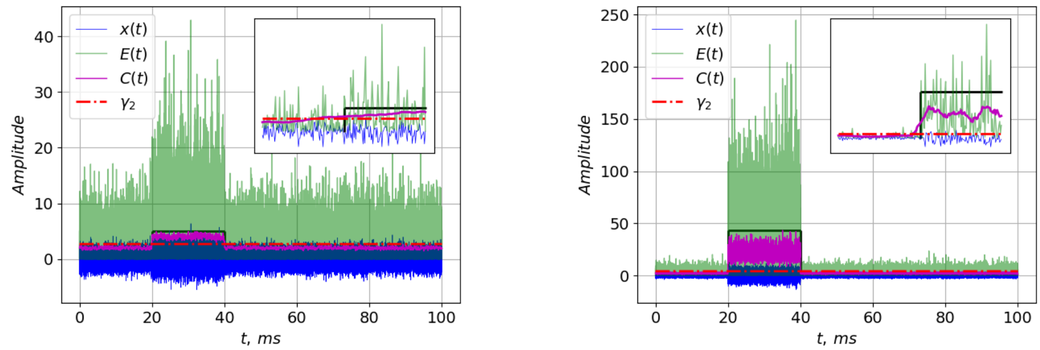
Figure 3 shows results of signal processing with coherent structure in time domain for frequency channels with 0 dB (Fig. 3a) and 10 dB (Fig. 3b) SNR. For the first case, moving average window length was $L = 146$ at $\alpha = 5$, and for the second – $L = 21$ at $\alpha = 4$.



(a) (b)

Fig. 3. Results of coherent signal processing for SNR = 0 dB (a) and 10 dB (b)

Figure 4 shows results of signal processing with a noise-like structure in time domain for frequency channels with SNR of 0 dB (Fig. 4a) and 10 dB (Fig. 4b).



(a) (b)

Fig. 4. Results of noisy signal processing for SNR = 0 dB (a) and 10 dB (b)

Comparing Fig. 3 and Fig. 4 we can conclude that for reliably signal edges determination, it is necessary to smooth complex envelope, regardless of signal structure and SNR. Moreover, for a coherent signal at high SNR (above 10 dB), we can choose value of α parameter less than 2. At the initial stage of signal processing, as a rule, it is not known what structure signal has, therefore it is recommended to choose value of α parameter in accordance with recommendations from previous section.

Figure 5 shows the dependence of signal edges estimation error via SNR, which changed during the experiment from -6 dB to 20 dB with a step of 1 dB.

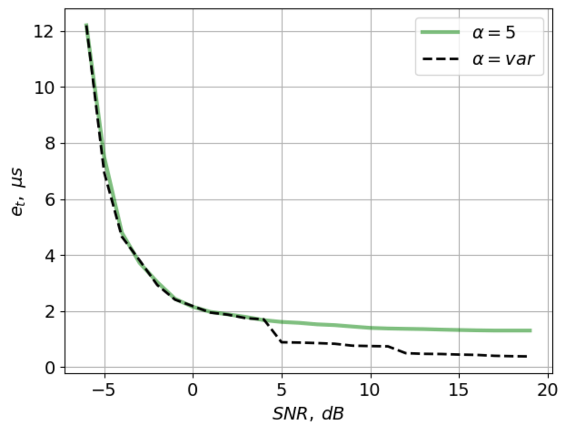


Fig. 5. Dependence of signal edges estimation error via SNR

It can be seen from this figure that the error of signal edges estimation decreases with the increase of SNR. This is explained by decrease of moving average window length L. In addition, at medium and high SNR values due to the adjusting α parameter, which determines L, error of signal edges estimates is reduced by 2-4 times, compared to fixed α.

Conclusions

Proposed method allows automatically determine signal edges with an arbitrary structure under the conditions of known SNR and noise power. Dependency between SNR and parameters of moving average window were established. It gives possibility to reduce by 2-4 times error of signal edges estimates in case of SNR increasing compared to fixed length of window with moving average. Developed algorithm can be implemented in existing and prospective radio monitoring systems.

Prospects for further research in this area should be focused on development and implementation of methods for determining signal edges under conditions of an unknown noise level and the development method for estimation noise level using samples of restricted volume.

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Метод визначення часових меж радіосигналів шляхом аналізу комплексної обвідної

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Постановка задачі. Необхідність автоматизації процесів оброблення інформації в сучасних системах радіомоніторингу стимулює розроблення гнучких методів виявлення та оцінювання часових параметрів сигналів. Апріорна невизначеність щодо частотно-часової структури сигналу ускладнює автоматичне визначення часових меж сигналів. **Мета статті**. Метою статті є подальша автоматизація процесу аналізу радіочастотного спектра за рахунок розроблення та реалізації методу визначення часових меж радіосигналів в умовах відомого значення потужності шуму та відношення сигнал-шум. **Виклад матеріалу дослідження**. Для визначення часових меж сигналів у кожному із частотних каналів розраховується квадрат комплексної обвідної сигналу, згладжується із використанням вікна ковзаючого середнього та порівнюється із порогом. Значення порогу розраховується як квантильгамма-розподілу із використанням апроксимації Вілсона-Хілферті квантилів розподілу χ^2 для заданої ймовірності хибної тривоги. Отримано аналітичний вираз для розрахунку довжини вікна ковзаючого середнього в залежності від відношення сигнал-шум. Розроблено алгоритм визначення часових параметрів сигналів та їх фільтрації за тривалістю. Невідоме значення потужності шуму в частотному каналі можна замінити його оцінкою в припущенні, що частотний канал зайнятий не постійно і на вільних від сигналів інтервалах часу проводиться оцінювання рівня шуму. **Висновки**. Запропонований метод дозволяє автоматично визначати часові межі сигналів з довільною структурою при значеннях відношення сигнал-шум від -6 дБ. Змінна довжина вікна ковзаючого середнього дозволяє у 2-4 рази зменшити помилку визначення часових параметрів сигналу при збільшенні відношення сигнал-шум у порівнянні з фіксованою довжиною вікна. Перспективи подальших досліджень у даному напрямку доцільно зосередити на розробленні та реалізації методів визначення часових меж сигналів в умовах невідомого рівня шуму.

Ключові слова: часові межі сигналу; комплексна обвідна; ковзаюче середнє; поріг; структура сигналу; відношення сигнал-шум