Application of a Threshold Methods for Compression of Vocal Signals

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***Abstract* – Types of thresholds used for compression of vocal signals are considered. It is shown that quality of the restored speech depends straight on the choice of threshold characteristic. The mathematical description of spectral composition of the signal subjected to threshold is given, its connection with the size of the threshold is revealed and the nonlinear distortion coefficient is calculated. The graphs of revealed dependencies on which it is possible to judge on type selection and threshold level are built.**

***Keywords* – compression, threshold, spectral composition, vocal signal, nonlinear distortion coefficient.**

I. INTRODUCTION

The present work is closely connected to the real problem, arising up at the signal processing in the cellular telephone systems, because of the background noises in them. When a vocal signal is transferred from a cellular telephone to the base station (or in other direction), it should be compressed in order for an operator could transfer as many signals as the real carrying capacity of the channel can allow [1].

For trouble-free work of the compression circuit it is necessary to provide high value of signal/noise relation on its input. Therefore the removal of background noise is made before the compression process. This can be done with an ideal filter which passes only a vocal signal and removes undesirable noises, for example, car noise or people talks. It is obvious that practically this task is enough difficult that's why the attempts to utilize different methods of signal processing, the most successful of which are threshold [2, 3 - 7] are undertaken. Thus, however, there is no data about threshold type influence and the level of threshold on qualitative descriptions of the restored signal.

The purpose of the present work is research of different threshold methods of decreasing the level of background noise while compression of vocal signals and choosing the best one of them on quality of the restored signal criteria.

II. MAIN PART

Threshold methods of noise decrease are based on diminishing of coefficients values of signal transformation while its transfer from one representative domain to another (time-frequency, time-space, and other). It is assumed that the noise component is represented by small coefficients and the threshold method is used for reduction or complete removal of small coefficients. Then the signal is exposed to reverse transformation.

Soft and hard thresholds [2, 3 - 5] are most often used. As it is assumed that the algorithm of processing should operate in a real-time mode, and the length of input signal can be large so the input signal is divided into small segments (shots). The algorithm of processing is used on each segment and the output result is represented as a composition of separate processed segments. Processing of separate segment requires time that all in all leads to the delay of signal in communication network.

Let's designate the size of the threshold θ. Then in case of hard threshold the functional characteristic (FC) of thresholding will be described by the equation:

$$f(x) = \left[1\left(|x| - \theta\right)\right]x, \quad (1)$$

where, $1(x) = \begin{cases} 1, & x \geq 0, \\ 0, & x < 0, \end{cases}$ – unit function.

The graph of FC at hard thresholding is shown on fig. 1,a.

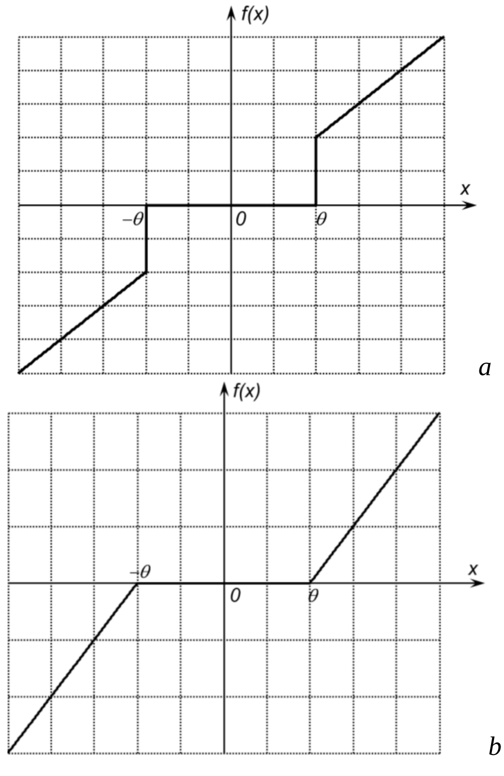

Fig.1 Functional characteristics of thresholding: *a* - hard, *b* - soft

In the case of soft threshold use FC is represented:

$$f(x) = \left[1\left(|x| - \theta\right)\right]\left[x - \theta sign(x)\right], \quad (2)$$

and the graph of such FC is shown on fig. 1,b.

Analysis of the graphs presented on fig.1 shows that the greatest influence on quality of speech transmission, especially with background noises, make initial areas of FC. Therefore we will try to modify these areas so that to provide the least distortions of the restored signal. The simplest decision for this task is realization of FC of threshold types, shown on fig. 2.

We will name them FC of linear supersoft thresholding (fig. 2,a) and FC of quadratic supersoft thresholding (fig. 2,b).

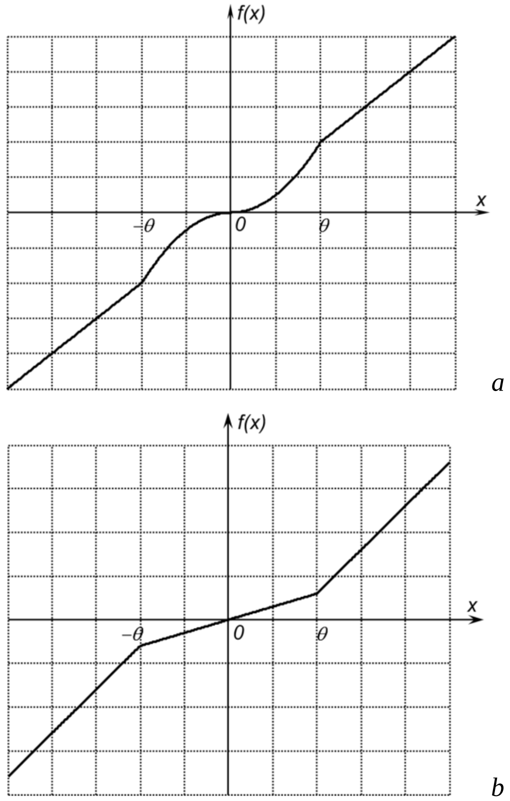

Fig.2 Functional characteristics of supersoft thresholding: *a* - linear, *b* - quadratic

FC at linear and quadratic supersoft thresholding are presented, accordingly:

$$f(x)=\begin{cases} x-sign(x)(1-\beta)\theta, & |x|\ge\theta, \\ \beta x, & |x|<\theta. \end{cases} \qquad (3)$$

$$f(x)=\begin{cases} x, & |x|\ge\theta, \\ \dfrac{x^2}{\theta}sign(x), & |x|<\theta. \end{cases} \qquad (4)$$

In the formula (4), β is an angular coefficient, determining inclination of initial linear area of FC.

At the compression of vocal signal with losses, with growth of compression coefficient (size of threshold θ) signal distortions increase accordingly [2, 3]. It shows up when parasite harmonic components appear in the restored vocal signal.

The number of distortions of signal after thresholding can be defined by the nonlinear distortion coefficient χ as a relation of operating value of output signal without its first harmonic to the operating value of input signal (in case of sinusoidal signal):

$$\chi=\frac{\sqrt{\dfrac{1}{2}\sum_{n=2}^{\infty}b_n^2}}{A/\sqrt{2}}=\frac{\sqrt{P_{эф}^2-\dfrac{1}{2}b_1^2}}{A/\sqrt{2}}. \qquad (5)$$

Using (5), will get the followings formulas for the calculation of vocal signals klirfactors:

$$(\chi)_ж=\frac{1}{\pi}\sqrt{(\pi-2\alpha+\sin 2\alpha)(2\alpha-\sin 2\alpha)}, \quad (6)$$

$$(\chi)_м=\frac{1}{\pi}\sqrt{\pi[2\alpha-\sin 2\alpha+2(\pi-2\alpha)\sin^2\alpha]-(2\alpha+\sin 2\alpha)^2}, \quad (7)$$

$$(\chi)_{лсм}=\left\{\frac{1}{\pi}[(1-\beta)(\beta-3)\sin 2\alpha-2\alpha(1-\beta^2)+\pi+2(\pi-2\alpha)(1-\beta)^2\sin^2\alpha]\left[1-\frac{1-\beta}{\pi}(2\alpha+\sin 2\alpha)^2\right]\right\}^{\frac{1}{2}}, \quad (8)$$

$$(\chi)_{ксм}=\frac{1}{\pi}\left\{\pi\left[\frac{1}{\sin^2\alpha}\left(\frac{3}{2}\alpha-\sin 2\alpha+\frac{1}{8}\sin 4\alpha\right)+\pi-2\alpha+\sin 2\alpha\right]-\left[\frac{1}{\sin\alpha}\left(3-3\cos\alpha-\frac{1-\cos 3\alpha}{3}\right)+\pi-2\alpha+\sin 2\alpha\right]^2\right\}^{\frac{1}{2}}. \quad (9)$$

From these calculations it is evident that the signal suffers most distortions at use of hard thresholding. In addition, depending on the type of threshold the maximum value of nonlinear distortion coefficient corresponds to different values of $\sin\alpha$. Only in case of soft and linear supersoft thresholds these values coincide and equal 0,3.

III. CONCLUSION

1. It was shown that most influence on quality of transmission of vocal signals, especially with background noises, renders the type of initial area of functional characteristic of thresholding.
2. 4 types of functional characteristics of thresholding that have different initial areas were studied. The mathematical description of spectral composition of sinusoidal signal thresholding subjected was given, its connection with the size of threshold was shown and the nonlinear distortion coefficient was calculated. The graphs of shown dependencies which help to choose the type and the size of a threshold were built.

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