Rajs Arkadiusz, Maszewski Mirosław. Low-pass filteration using adaptive delta modulation. Journal of Education, Health and Sport. 2017;7(12):133-138. eISSN 2391-8306. DOI http://dx.doi.org/10.5281/zenodo.1112371
http://ojs.ukw.edu.pl/index.php/johs/article/view/5112

The journal has had 7 points in Ministry of Science and Higher Education parametric evaluation. Part B item 1223 (26.01.2017).
1223 Journal of Education, Health and Sport eISSN 2391-8306 7

Received: 10.11.2017. Revised: 15.11.2017. Accepted: 12.12.2017.

LOW-PASS FILTERATION USING ADAPTIVE DELTA MODULATION

FILTRACJA DOLNOPRZEPUSTOWA WYKORZYSTUJĄCA ADAPTACYJNĄ MODULACJĘ DELTA

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Streszczenie

Praca przedstawia metodykę wykorzystania adaptacyjnej modulacji delta do realizacji filtracji dolno przepustowej oraz omawia zalety takiego rozwiązania.

Słowa kluczowe: filtracja cyfrowa, modulacje różnicowe, filtr dolnoprzepustowy

Abstract

Linear Delta Modulation (DM) is applicable to processing of signals in digital systems of telecommunication. One of most interesting kind of LDM is Adaptative DM (ADM). This Work presents method of using ADM for low-pass digital filtration.

Key words: digital filtration, differential modulation , low-pass filter

1. Introduction

Digital filtering is used in telecommunication systems as one of the basic methods of digital signal processing. Traditionally, the specialized processors for digital filtering, multi-position binary codes, which is not conducive to the development of efficient algorithms and structures of digital filters. To speed up the operation of digital filters is necessary to use other, more economical way of digital signal processing - differential modulation. The most famous such modulation is linear delta modulation (LDM). It uses the quantization of the difference between the input signal and aproksymującym, which leads to less positions output codes. The consequence of this is profit compression coding data stream and resistance to interference which makes it an interesting alternative to traditional ways of coding.

Variant of LDM is adaptive delta modulation (ADM). It combines the advantages of linear DM (LDM) - single position code, and Differential PCM (DPCM) - different values of quantization steps. Values are determined on the basis of the current and previous code combinations ADM [1].

The systems transmit voice signals, image and others found the broadest use of filtering with finite impulse response functions based on the strand. Such filtration is characterized by a linear phase-frequency characteristics, and an absolute stability of the filter and therefore this type of filtering is considered in this work.

2. Methodology filtration

On the basis of the difference on increasing convolution function [2]:

$$\nabla y_n = y_n - y_{n-1} ,$$

where: $y_n = \sum_{m=0}^{M-1} x_{n-m} h_m$

$\{x_i\}_{i=0}^{N-1}, \{y_i\}_{i=0}^{N-1}, \{h_m\}_{m=0}^{M-1}$ - respectively the values of input output area subject to filtering the digital code PCM,

h_m - weight coefficients of the digital filter pulse characteristics,

N, M - the size of the input area and the number of weight coefficients of the filter

can be obtained by the differential function convolution codes PCM - Differential PCM [1], and wherein the differential PCM input signal will be described, but the pulse response of the filter (ie. the weighting coefficients $\{h_m\}, m=\overline{0,\ M-1}$) will be in PCM codes.

If there is no overload due to signal's steepness it takes place

$$\nabla x_i = s_i^{(x)} \pm \psi_i ,$$

where: $\nabla x_i = x_i - x_{i-1}$; $\{x_i\}$ - samples the input signal, $\{s_i^{(x)}\}$ - ADM - quantization steps (ADM - code) of input signal; $\{\psi_i\}$ - quantization errors.

then we can write:

$$y_n = \sum_{i=0}^{n} \sum_{m=0}^{M_D-1} s_{i-m}^{(x)} h_m \tag{1}$$

where: $\forall s^{(x)} \in \{-\varepsilon, -2\varepsilon, -4\varepsilon, \varepsilon, 2\varepsilon, 4\varepsilon\}$ - quantization steps ADM,

ε - absolute value of the minimum step,

M_D - number of weight coefficients characteristic pulse by code PCM, (digitalised with the frequency of adaptive delta modulation f_D).

If present ADM - steps $\{s_r^{(x)}\}$ as $\{\varepsilon \ \mathrm{sgn} s_r^{(x)} 2^{l_r}\}$,

where: $\forall l \in \{0,1,2\}, \quad \mathrm{sgn} s_r^{(x)} \Rightarrow B_{s.r}^{(x)}$,

and weighting coefficients $\{h_m\}$ as $\{\mathrm{sgn} h_a \cdot |h_a|\}$, which corresponds to the binary code:

$$\mathrm{sgn} h_a \Rightarrow B_{s.a}^{(h)}, \quad |h_a| = \sum_{k=0}^{c-1} B_{a.k}^{(h)} 2^k ; \forall B \in \{0,1\}$$

then from [1] the product of the characters can be represented in the form of logical operations on binary codes:

$$\mathrm{sgn} s_r^{(x)} \cdot \mathrm{sgn} h_a = 2\left(\overline{B_{s.r}^{(x)} \oplus B_{s.a}^{(h)}}\right) - 1$$

Taking into account the value of adaptive change ε (quantization steps) convolution function (1) can be represented as a function of convolution codes for PCM-ADM:

$$y_n = \varepsilon(x)\sum_{r=0}^{m}\sum_{m=0}^{M_D-1}\left[2\left(\overline{B_{s.n-m}^{(x)} \oplus B_{s.r}^{(h)}}\right)-1\right]\sum_{k=0}^{c-1} B_{n-m.k}^{(x)} 2^{k+l_r} = \varepsilon(x)\sum_{r=0}^{m}\sum_{m=0}^{M_D-1} 2^{l_r}\sum_{k=0}^{c-1} B_{n-m.k}^{(x)} 2^{k} \quad (2)$$

which means that the operation of multiplication is converted to:

- logical modulo-2 summation operation and negation
- shift by left for bits of weight coefficient on l_{i-m} positions

3. The low pass filter uses an Adaptive Delta Modulation

Economic work algorithm for specialized processor of adaptive digital filtering format ADM-based PCM (2) has the form:

$$y_k = \varepsilon(x)\sum_{k=1}^{n} \mathbf{S}_k^{(x)}\mathbf{H}_M \quad (3)$$

where:

$$\mathbf{S}_k^{(x)} = \left\| s_k^{(x)} \ldots s_{k-M_D+1}^{(x)} \right\| = \left\| \operatorname{sgn} s_k^{(x)} \cdot 2^{l_k} \ldots \operatorname{sgn} s_{k-M_D+1}^{(x)} \cdot 2^{l_{k-M_D+1}} \right\|$$

$$\mathbf{H}_\mathbf{M}^\mathbf{t} = \left\| h_0 \ldots h_{M_D-1} \right\|$$

Computer simulations using low-pass filtering codes ADM-PCM was carried out on the basis of (2). As the input stream used pink noise, All of harmonics were equal, and the phase shifts of each of them were random and subjected to a uniform probability distribution rule.
In filtration PCM PCM-defined 41 weighting coefficients, and 100 samples of the input signal. The results of filtration ADM-PCM compared to filtration PCM-PCM. It shows Fig. 1

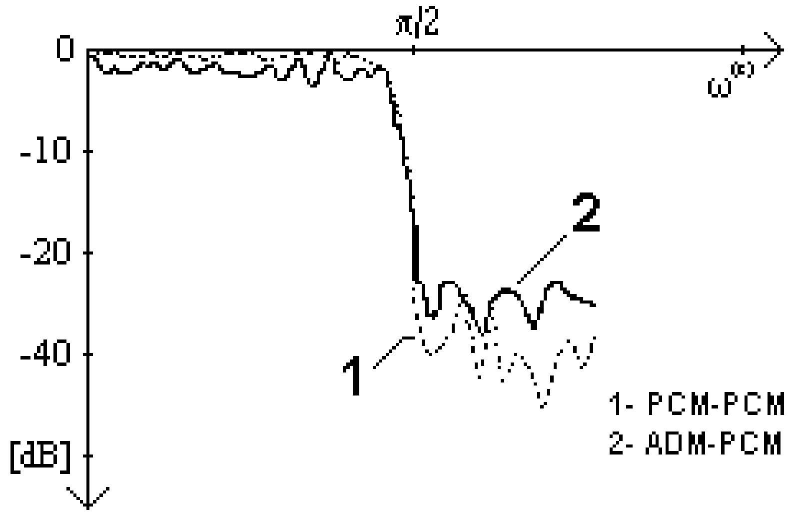

Fig. 1 Comparison of the results of filtration using the codes DP PCM-PCM and PCM-ADM

Performed computer simulations indicate a small difference between the results of filtration using codes ADM-PCM and the PCM-coded (PCM such filtering is treated as a reference). Errors indicated filtration can easily be reduced by a corresponding change in the quantization step ε and the sampling frequency f_D. It is also important to choose the ADM algorithm to encode the weighting coefficients. The ADM-PCM digital filtration processor based on the presented algorithm, due to the lack of multibite multiplication operations and the use of logical operations on the appropriate codes, is economical and suitable for a large degree of integration, making it suitable for use in telecommunications systems

4. Conclusions

Use of the adaptive delta modulation to the digital filter is interesting solution and can find wide range of applications in the processing of any digital signals, in particular, various types of digital images. This type of delta modulation is more resistant to distortion than the PCM, and at the same single position code ADM is more convenient and faster processors for processing in real time.

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