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THE ANALYSIS OF FILTRATION INFLUENCE AT PRIMARY PULSED-CODE CONVERSION ON DISTORTION OF INPUT SIGNALS OF CODERS WITH A COMPRESSION OF AUDIO DATA

The influence of selective circuits of digital-analogue converters at primary pulsed-code conversion on distortion of input signal of coders with audio data compression is considered.

Keywords: compression of audio data, distortion of signal, digital-analogue converters, analogue-digital converters, group delay time.

In psychoacoustic models of MPEG standards the mechanisms of time masking of signals, spatial dismasking the sources of a sound producing a stereo panorama on front, depth and feature of perception of reverberation components of the stereophonic signals are not considered. These mechanisms of spatial hearing play the most important role for stereo reproduction; they define perception of the basic features of quality of the stereophonic sounding, such as spatial perception, sounding transparency, naturalness and wealth of timbres of instruments and voices, perception of acoustic atmosphere of a primary room (a concert hall, studio), etc. Occurrence of this set of distortions leads to decrease in quality of sounding which is distinctly observing by listeners.

However it is necessary to consider, that digital audio signals arrive on an input of coder with a compression after pulsed-code (PSM) conversion. Thus quality of conversion is meant ideal. In a number of works, for example [1], an influence of the errors of quantization on sounding the audio signals written down or transferred with a compression of audio data is shown.

The selective analogue-digital and digit-analogue (ADC and DAC) PSM conversion provide an essential influence on producing the high quality indicators at using the digital methods of sound recording as well as at organizing the digital sound broadcasting.

The low-frequency filters (LPF), limiting a spectrum of input frequencies and eliminating high-frequency components of a output signal accordingly are located at the input of ADC and at the exit of DAC.

Signal suppression of LPF on the frequency equal to half of frequency of digitization, should be not less than 60 dB. In this case the steepness of slope of LPF should be very high (120 dB\octave). For achievement of such values of steepness the high order LPF should be created. Such filters have considerable disadvantages and the main essentially nonlinear phase characteristic that leads to distortions of audio signals appreciable by ear as loss of "transparency" of the sound. Besides, such filters becomes rather difficult in manufacturing and adjustment, and, hence, expensive. In audio equipment the greatest distribution was received by Butterworth and Chebyshev filters.

A dependences of an order of the filter (N_b, N_c) from demanded attenuation (A_{min}) on boundary frequency of a leakless strip (f_{gtsll}) at admissible non-uniformity in pass-band A_{max} = 0.5dB for typical ADC cases are calculated for Butterworth and Chebyshev LPFs and shown in figure 1:

– for signals of a sound broadcasting (3V) for the higher class of quality (f_b = 15,000 Hz, f_{gtsll} = 16,000 Hz);

– for audio signals at a sound recording (f_b = 20,000 Hz, f_{gtsll} = 20,000 Hz).

All calculations in the given work were carried out in software MathCAD.

The figures shows, that for the reception of the required attenuation 60 dB at the boundary frequency of a leakless band the filter order of approximately 124 in the first case, and more than 42 in the second case is required. Such filters in analogue circuitry cannot been realized.

Calculations shows that, for the case of sound recording with the aid of Chebyshev or Butterworth filters, the orders of such filters of 12 and 42 are required respectively. In analogue circuitry the filters with orders 6 or 8, but not above, can be realized. As a criterion of linearity of the phase characteristic a change of group delay time (τ_{gt}) which is normalized the domestic State Standard (GOST) 11515–91 for the channels of sound broadcasting can be considered. It is obvious, that the requirements shown to the sound recording and sound reproducing equipment should be more rigid, than to the channels.

In figures 2 and 3 the received dependences of group delay time for Butterworth filters with order 42 and Chebyshev filter with order 12 are shown. The normalized frequency (w) is postponed along X axis, and w = 1 corresponds to the boundary frequency of a pass band.

Along with impossibility to realize analogue filters with considered orders, the non-uniformity of group delay time in a pass band is 40 ms for the Butterworth filter and 73 ms for the Chebyshev filter. Such non-uniformity of group delay time essentially deforms a primary audio signal (an input signal of the filter).

One of known methods for the solution of the given problem is the using of the PSM coders (decoders) in which

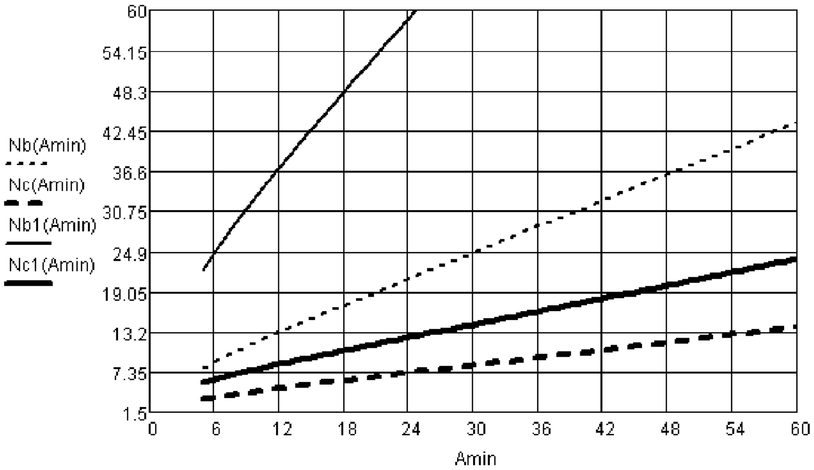

Fig. 1. Dependences of an order of Butterworth (N_b) and Chebyshev (N_c) filters from A_{min} (— — for signals 3V, ———for a sound recording case)

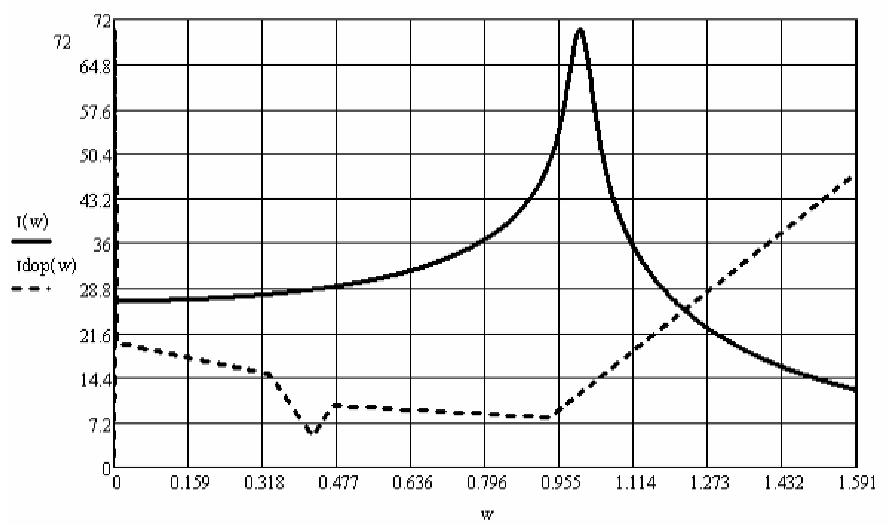

Fig. 2. The dependence of group delay time of Butterworth filter from normalized frequencies ($\tau(w)$) as well as standard ($\tau_{dop}(w)$) admissible delay time

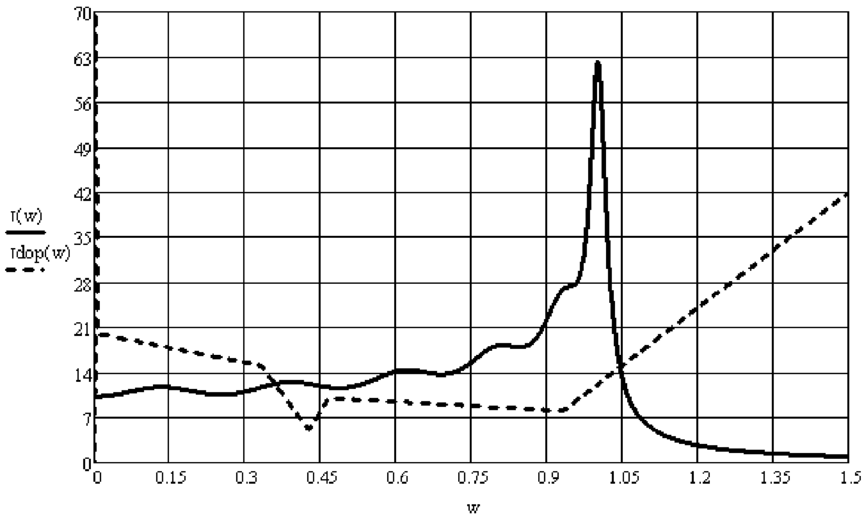

Fig. 3. The dependence of group delay time of Chebyshev filter from normalized frequencies ($\tau(w)$) as well as standard ($\tau_{dop}(w)$) admissible delay time

ADC (DAC) work at the raised frequency. It allows to make considerably lower the requirements to the slope steepness and to the order of analogue LPF, which provide a preliminary filtration of the primary signal. Then the basic attenuation at the boundary frequency of a leakless strip will be provided with a digital filter.

At the choice of structure of the digital filter the non-recursive filters have preference in this case.

The decisive advantage of such filters is possibility to receipt the linear phase-frequency characteristics. In this case, the definition of the requirements to analogue LPF and the order of such LPF should be focused on preliminary filtration and providing the demanded non-uniformity of group delay time in a filter pass band.

In figure 4 the calculated dependences of group delay time from normalized frequencies ($w_{\text{в}}$ a signal it is equal 1) are shown for Butterworth LPF. Such dependences for Chebyshev LPF (with 2nd, 4th, 6th and 8th orders) are shown in figure 5. For Butterworth filters with 8th order the non-uniformity in a pass band does not exceed 3.5 ms [2]. Working attenuation on the boundary frequency of leakless strip (A_{pmin}) of 5 dB for the case of digital transmission of the signals of sound broadcasting described above, and of 13 dB for the case of digital sound recording is thus provided.

The Chebyshev filter seems acceptable with an order not above than 6. In this case the non-uniformity on group delay time in a pass band does not exceed 11 ms, that seems acceptable with taking into account the properties of hearing. Besides the requirements of State Standard 11515–91 are practically satisfied. Thus LPF provides A_{pmin} of 28 dB (for the sound recording conditions given above) and 10 dB (for the case of transferring the signals of sound broadcasting).

At the using PSM coders (and decoders) with redigitization in codec's and compression of audio data for digital transmitting the signals of sound broadcasting and for digital sound recording the minimal distortion (appreciable on hearing) occur under the best linearity of phase characteristics of selective coders and decoders.

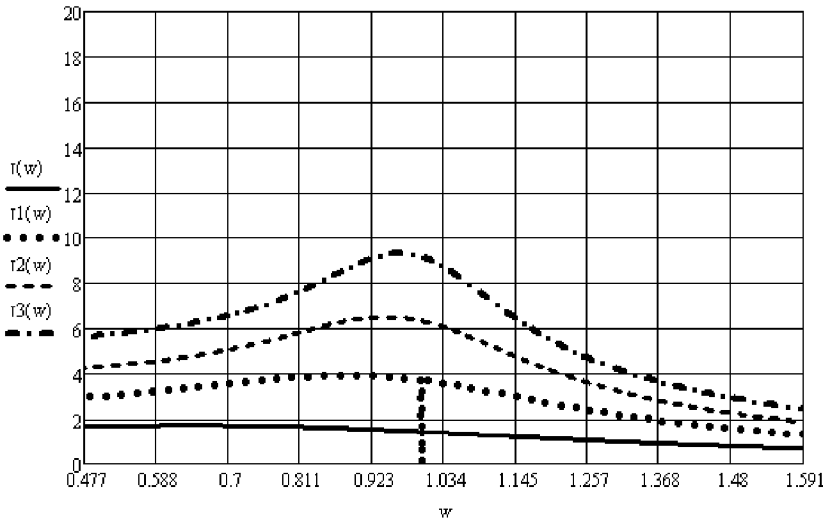

Fig. 4. The dependence of group delay time from normalized frequencies for Butterworth LPF 2nd (τ(*w*)), 4th (τ1(*w*)), 6th (τ2(*w*)) and 8th (τ3(*w*) orders

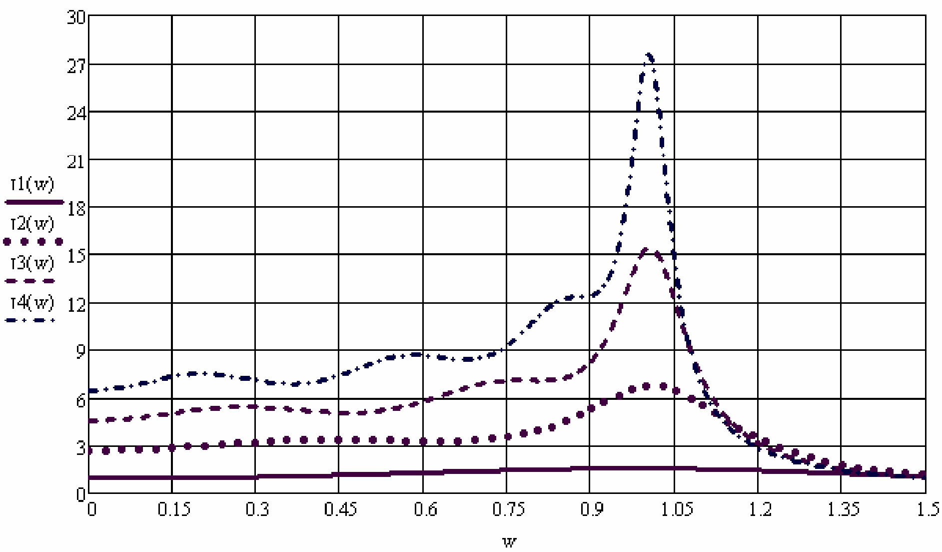

Fig. 5. The dependence of group delay time from normalized frequencies for Chebyshev LPF (τ1(*w*)), 4th (τ2(*w*)), 6th (τ3(*w*)) and 8th (τ4(*w*)) orders

For the filters realized in analogue circuitry the preliminary filtration is preferable to carry out with the help of Chebyshev filters which provide acceptable non-uniformity of group delay time from the point of view of acoustical perception of admissible distortions and provide rather big attenuation on boundary frequency of a leakless strip.

For receiving the linear phase-frequency characteristics in ADC and DAC with redigitization it is necessary to use non-recursive digital filters.

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TWO-LEVEL GENETIC ALGORITHM FOR X-RAY POWDER DIFFRACTION STRUCTURE ANALYSIS

A new evolutionary approach for crystal structure determination of powders based on X-ray diffraction full-profile analysis and genetic algorithm of global optimization is suggested. An investigation of efficiency of given algorithm is carried out on test real-world problems of structure determination.

Keywords: evolutionary algorithm, X-ray powder diffraction analysis, Rietveld method.

Crystal structure information is essential for explanation and prediction of physical and chemical properties of investigated materials. Many materials, multi-phase mixtures in particular, are available in form of powder only, thus severely impeding a research. In such cases X-ray powder diffraction methods, which are being intensively developed during last two decades, are used. They are based on analysis of a whole X-ray diffraction profile of powder pattern, which is a monochromatic X-ray radiation intensity function of polycrystalline sample diffraction angle. By now, in general, crystal cell parameters search problem (indexing methods) and structure model refinement problem (Rietveld method) have been solved. Primary mathematical means used for these problem solutions is a non-linear least-squares method (LSM). Plausible structure model determination in case of powder samples is still a problem even in case of relatively simple structures [1].

In recent years, for this problem solving so-called “direct-space” methods [1] have become of use. They are based on probabilistic generation of trial crystal structure models, their assessment through weighted difference of calculated and observed patterns (profile R-factor) and search for global minimum over corresponding parametric hypersurface in order to find an adequate structure model. An example of this approach is evolutionary algorithm, mimicking processes of natural selection in search of an optimal structure solution [2]. Several implementations of this concept have already been used for structure determination, demonstrating promising prospects [1; 3]. Here a two-level hybrid genetic algorithm is suggested for that purpose and its approval results are described on real patterns of single- and multi-phase polycrystalline samples with well-known crystal structure.

Full-profile crystal structure model refinement. As a tool for crystal structure model refining multi-phase Rietveld method [4] was used. An essence of Rietveld method is a modeling of experimental pattern by complex multi-parametric function:

$$Y^{\mathrm{mod}}(\boldsymbol{P}, \theta_j) =$$
$$= \sum_i^n \Omega_i(\boldsymbol{P}_L, \theta_j) * I_i^{\mathrm{calc}}(\boldsymbol{P}_S, \theta_j) + B(\boldsymbol{P}_B, \theta_j), \qquad (1)$$

where θ_j – diffraction angle; $\boldsymbol{\Omega}_i$ – profile functions for diffraction lines i, dependent on profile parameters set $\boldsymbol{P}_L$, (positions, half-width, form, asymmetry of lines, etc.); I_i^{calc} – calculated integral intensities of lines, dependent on structure parameters set $\boldsymbol{P}_S$ (atomic coordinates, thermal motion parameters, etc.); B – function for background, dependent on background profile parameters set $\boldsymbol{P}_B$.

Firstly, pattern model is calculated from the approximate (initial) values of parameters $\boldsymbol{P}$, including model atomic coordinates in crystal cell. Exact coordinates and other parameters (including quantitative phase composition in case of multi-phase sample) are determined as a result of mathematical fitting of model pattern to observed pattern by structure and profile parameters least-squares method variation.

Formalizing the approach, we get a following mathematical optimization problem. Experimental data (powder pattern) represent a discrete sequence $\{\theta_j, Y_j\}$ of size m, sorted by ascending of θ_j. Some class of parametric function $Y(\boldsymbol{P}, \theta)$ (Rietveld method functions) is given, $\boldsymbol{P}$ is a set of profile and structural parameters (a vector of size N), θ – independent argument. Peculiarities of the problem are large dimensionality (can exceed 100 parameters) and non-polynomiality of functions.