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Computer Modeling of Second-Order Recursive Digital Filters of the Automated Design Signaling System

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In the article the analytical method of modeling software recursive digital filters of the second order with zeros on the circle of the single radius is presented. The corresponding algorithm of scaling of this composition of filters for signal CAD is developed.

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Introduction

The task of the proposed method of processing the signal CAD will be to show the need for the development of recursive digital filters (RTSF) with a redesigned structure. The values contained in the main calculation and synthesis of programmable digital filters are mainly determined by the kind of reconfiguration of the parameters. Here, respectively, distinguish large-scale and non-large-scale reconfiguration of the parameters of amplitude-frequency characteristics (AFC, or hodograph). Often used in the development of programmed as transversal and recursive digital filters large-scale reconfiguration, which changes the zoom settings:
- cut-off frequency $\psi_3$ for low-pass filter (LPF) filters and high-frequency filters (high-frequency);
- the central frequency $\psi_0$ and the bandwidth $\Delta \psi$ for bandpass filters (SF), herewith Q remains constant;
- Analog filters are the basic basis for digital filters.

I. Theory and methodology

Programmable recursive digital filters of the first and second order assign the role of base units in cascade and parallel structures of higher orders ($\geq 3$). Here the most versatility is the two-square block, which allows to be realized on the basis of one special calculator of LPF, high-frequency, and band or recursive filter. The transfer function (characteristic) of the second order is defined as follows:

$$H(z) = B_0 \frac{1+ A_1 z^{-1} + A_2 z^{-2}}{1+ A_1' z^{-1} + A_2' z^{-2}}, \quad (1)$$

where $B_0$ is the scaling factor, $A_i, B_i (i = 0, 1, 2)$ is the Z-position of the z-transformation variables.

In this paper, analytical expressions for calculating (simulating) the coefficients of the transfer function of programmable RTSF are the exact method based on analog filters. This is due to the fact that today very well-designed and well-studied methods for the synthesis of analogue filters and there is an important reference material for different types of approximation. In this case, the calculations of the coefficients of the function $H(z)$ from the transfer function of the analog prototype from the rational form of the transformation operator of the p-region to the z-domain.

II. Practical application

The use of analog prototype allows the basis of bilinear z-transform to obtain the following expressions for analytical modeling: - for digital LPF $B_1 = 2$.

$$A_1 = A_0 \left( \frac{2}{2^2 + 1} - 1 \right), \quad (2)$$

$$A_2 = \frac{A_0}{A_0 \left( 2^2 + 1 \right)} \left( x - \frac{2}{2^2 + 1} \right)$$

where

- for digital FBCH $B_1 = -2$

$$A_1 = \frac{-2}{2^2 + 1} \left( x - \frac{2}{2^2 + 1} \right), \quad (3)$$

$$A_2 = \frac{2}{2^2 + 1} \left( x - \frac{2}{2^2 + 1} \right)$$

here $\Delta$ is the inequality of the AFC of the RSH in the bandwidth $\psi_B$, $\psi_A, \psi_B$ - cut-off frequencies of digital LPF and HF, respectively $\psi_D$ - sampling frequency;
- for bandpass filters $B_0 = 0$

$$A_1 = \frac{2}{\left( \frac{2^2 + 1}{2^2 + 1} \right)} \left( x - \frac{2}{2^2 + 1} \right), \quad (4)$$

484
Gibbs Grand Thermodynamic Potential in the Theory of Kinetic…

\[ A_2 = \frac{1-\frac{\omega}{\omega_0} + \frac{\psi_0}{\psi_1}}{1 + \frac{\omega}{\omega_0} + \frac{\psi_0}{\psi_1}} \]

where \( d = \frac{[10]}{[\Delta \omega / 20]} \) is the approximation plane

\[ \varepsilon = \frac{1-\frac{d^2}{B^2}}{\frac{\omega_0}{\omega} - t(\frac{\psi_0}{\psi_1} + \frac{\psi_0}{\psi_1})}, \]

\[ \omega = t(\frac{\psi_0}{\psi_1} + \frac{\psi_0}{\psi_1}) + t(\frac{\psi_0}{\psi_1} + \frac{\psi_0}{\psi_1}) + \Delta \varphi. \]

Solving the equation (1) with respect to \( \Delta n, \psi_A, \psi_V \), we obtain the following calculation expressions:

\[ \Delta n = 10 \log \left( 2 \arcsinh \left( \frac{1-A_2}{\sqrt{A_2-A_1-A_2}} \right) + 1 \right), \]

\[ \psi_A' = \frac{1}{\pi} \arctan \left( \frac{6A_2-A_2^2-A_2-1}{2A_2} \right), \]

\[ \psi_B' = \frac{1}{\pi} \arctan \left( \frac{6A_2-A_2^2-A_2-1}{2A_2} \right). \]

Since \( \Delta n, \psi_A, \psi_B \) are positive numbers, then from equation (1) and (2) the restrictions on the quantities \( A_1 \) and \( A_2 \) are executed.

\[ A_1 - A_1 > 0, A_2 < A_2 < 0, 1-A_2 > 0, (A_1 + A_1 + 1) > 0 \]

The domain of constraints (6) is interpreted here as an area on a plane with rectangular coordinates \( A_1 \) and \( A_2 \), which is limited by a triangle whose vertices are placed at \((-2.1), (2.1), (0, -1)\). This region of stability of the filter is divided into two parts by the parabola \( A_2^2 = 4 \) (weak damping region \((-2 < A_2 < 0.12)\)) and \( A_2 = 0.5 - 0.6\) (region of strong damping \((-1 < A_2 < 0.1)\)).

In the case of the implementation of LPF and high-frequency control of the frequency of growth it is expedient to perform in the first area close to the parabola, that is, when fast damped self-oscillations of the filter are allowed. Thus, the region of the analysis of the frequency response (or the hodograph) is limited by the inequalities: \(-2 < A_1 < 2, 0 < A_2 < 1\).

The structure of the program of automated analysis of ACL (FCH) is considered by the example of the LPF. It requires the following blocks of the algorithm:

a) Enter the value of the sampling frequency \( \psi \) and the initial frequency of the analyzing area \( \psi_0 \), the analysis \( \Delta \psi \), the value of the coefficients \( A_1 \) and \( A_2 \);

b) calculation of the values of the frequency response \( H(\psi) = 2\pi (\psi_0 + \Delta \psi) / \psi_0 \), i.e., \( i = 1, 2, \ldots, n \);

c) the frequency response on the existence of extrema in the selected frequency range;

d) search in the array \( H(\psi_0) \) of the point \( A(\psi_0 - 3) \) in the interval \( D > \psi > \psi_0 \);

e) calculation of the unevenness of the frequency response;

g) output to the screen of output data and analysis results.

The analysis shows that programmable digital filters that are implemented as a base line with zeros of the transfer function located on the circle of the single radius, retain sufficiently wide possibilities for reordering the frequency response by changing the coefficients \( A_1 \) and \( A_2 \). The cutoff frequency of the LPF can be adjusted, for example, with \( \psi = \text{const} \) and \( \Delta H < 3 \text{dB} \), if you select \( A_2 = 0.5 - 0.55 \), and the coefficient \( A_1 \) varies within \(-1, \leq A_1 < 0\). In doing so, you must fulfill the condition of scaling \( H_{\text{max}} = 0 \) by changing the multiplier \( B_0 \) within \((4, 30) \text{dB} \). Adjusting the frequency of the high-frequency cut-off at the same opportunity on \( \psi D \) and \( \Delta H \) is possible between the limits and for this purpose it is necessary to choose the coefficient \( A_2 = 0.4 - 0.45 \) and to find the second coefficient within \(-1, \leq A_1 < 1 \), choosing for each pair \((A_1, A_2)\) the scaling factor \( B_0 \), due to which the \( H_{\text{max}} \) varies from magnitude from 14.4 dB to 21.0 dB.

The performed algorithm of calculation was used in hardware implementation of the program RTSP, where the 10-bit coefficients of the transfer function were used, and the number of operators within the calculator was equal to 12. Comparing the results of the experiment with theoretical calculations shows the high adequacy of this computer simulation.

**Conclusions**

1. An algorithm for computer modeling of recursive digital filters (LPF, HF, SF) is developed.
2. Using the scale factor \( B_0 \), an algorithm for scaling RTSP is executed.
3. The developed algorithm of modeling determines the signal of CAD digital filtering.

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Комп’ютерне моделювання рекурсивних цифрових фільтрів другого порядку сигнальної системи автоматизованого проектування

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В статті викладений аналітичний метод моделювання програмних рекурсивних цифрових фільтрів другого порядку з нулями на колі одиничного радіуса. Розроблено відповідний алгоритм масштабування даного складу фільтрів для сигнальної системи автоматизованого проектування (САПР).