The Analog Filter Design and Interactive Analog Signal Processing by PC Control

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Abstract: - This paper presents a new complex system for the cascade frequency filter design and physical frequency filter modeling. It is composed from universal digitally controlled analog frequency filter and a new software - "frequency filter design studio"- product. Our contribution briefly prescribes the principle of system, methods of network solution, advantage of used conception and proposed the effective method of parameter controlling by superior computer. There are shortly discussed in many examples explained wide possibilities and practical use of new developed integral frequency filter design system.

Key-Words: - Frequency filter, cascade synthesis, filter design, digitally control, control software

1 Introduction

In present time there are rapidly increasing requirements of analog signal pre-processing in many electronic systems. To set some desirable factors in signal processing like noise conditions, there are now used systems with dynamically variable parameters as various adaptive frequency filters.

Frequency filters can essentially to improve resulting parameters of designed transfer system in many cases. Increasing complexity of analog signal preprocessing is leading also to more demanding control of a signal filtration realised for example not only using low – pas and high – pass filters, but also narrow band – pass filters or band – pass reject filters with various types of approximation functions. Thereby our work has been focussed to design and to construct general - purpose integral system of analog frequency filter as digitally controlled compact measuring equipment with aim to create a fast adaptive universal system, wide usable in signal preprocessing development, design and education.

2 Universal analog frequency filter

Presented new general – purpose digitally controlled system (Fig. 1) consists of three main parts -
- universal analog filter,
- control unit,
- filter design studio – software .

All resulting parameters of universal analog filter (filter type, filter order, type of function approximation, frequency tolerance area) can be controlled through control unit manually (by mini key – board) or using controlling PC. In this case it is possible take advantage to control universal analog filter by use special developed software - "frequency filter design studio"- product.
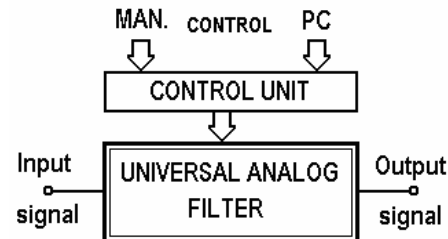

Fig. 1 Block diagram of integral system

The basic structure of universal analog frequency filter is derived from cascade method of filter synthesis. Combining one block of the 1st filter order and m blocks of the 2nd filter order can be realised resulting transfer function from first to m+1 order:

$$H(s) = \frac{\pm B_{(m+1)}s + C_{(m+1)}}{s + \omega_{0(m+1)}} \cdot \prod_{i=1}^{m} \frac{A_i s^2 \pm B_i s + C_i}{s^2 + s\omega_{0i}/Q_i + \omega_{0i}^2} , \quad (1)$$

where s is complex variable, ω_{0i} angel resonant frequency, Q_i quality factor and A_i, B_i, C_i transfer coefficients of each filter block. These filter parameters determine resulting type of transfer function (low – pass, high – pass, band – pass, all – pass) as well as basic function of approximation. By digitally controlled parameters there is possibility to construct universal

controlled analog filter with any required filter type and any required function of approximation (as Bessel, Butterworth, Chebyshev, Inverse Chebyshev, Cauer). The resulting block diagram of main part of the system is constructed according philosophy of cascade architecture and we can se in Fig.2.

Input signal is divided (or amplified) in the input amplifier of block. Then it is formed passing through determined number of selective blocks, what is defined by adjustment of switching pole. It is controlled by controller bus from control microprocessor. Concurrently are directed using control bus all variable parameters (ω_{0i} , Q_i , A_i, B_i, C_i) of each selective block of 1st and 2nd order transfer function. All required parameters it is possible to set also manually through control microprocessor using owner miniature keyboard block, parameters are displayed on the display placed parallel with keyboard in front panel of equipment.
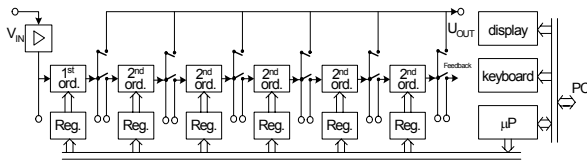

Fig. 2 Block diagram of frequency filter

3 Design of main filter blocks

The transfer parameters of main filter blocks (of 1st and 2nd order) have to fulfil all requirements of transfer function expressed in eq. (1). The block topology of main filter block of 2nd order to realisation all above mentioned requirements is shown in Fig.3. The filter block of 1st order was solved similarly (with rather simpler connection). The modified biquadratic Akerberg - Mossberg network with 2nd pole compensation was used as main 2nd order block (see Fig. 4).
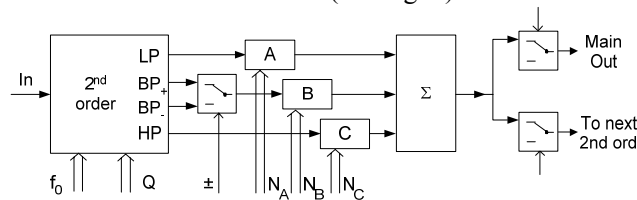

Fig. 3. The 2nd order unit block diagram

Basic coefficients ω_{0i} and Q_i are controlled directly in the 2nd block by means of tuning elements R_f, C_f, and R_q. The numerator coefficients A_i, B_i, C_i are established by gain setting of auxiliary amplifier blocks. Required polarity of the B coefficient is controlled using switching block connected to band-pass outputs. Final output

signal from sum - amplifier block can be connected to central output bus or to next 2nd order filter through controlled switching blocks.
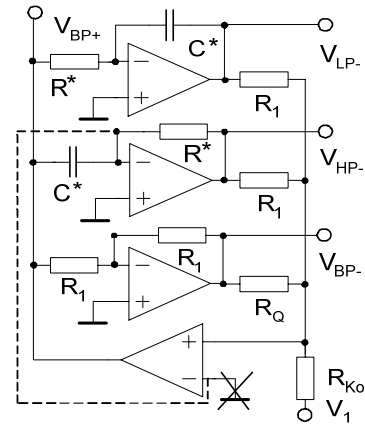

Fig. 4 Final 2nd order filter with 2nd pole compensation .

The resulting circuit diagram of 2nd order filter block we can see in Fig. 4. To limit influence of real properties of operational amplifiers, the 2nd pole compensation of amplifier (dotted line) was here used [3]. By means of capacity C* adjustment (through switching blocks) it is controlled resonant frequency roughly. The fine frequency setting is controlled by R* resistor value.

The 1st order filter block (Fig.5) was designed similarly like previously prescribed 2nd block. Required frequency setting of 1st order filter block is controlled through tuning elements R* and C* .
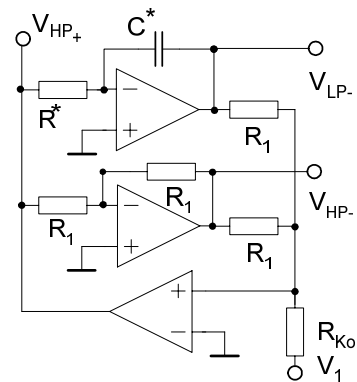

Fig. 5 The 1st order filter block

Thanks independency all parameters (F_0, Q, A, B, C) may be proceed the controlling f and Q independently too. For frequency and quality factor driving, like as for numerators transfers gain controlling, it is use the R-2R four-quadrant DA converter AD7533 [4], whose present equivalent current

$$I_{out} = \frac{N}{10^6} U_{in}\,, \qquad N \in \langle 0,1023 \rangle\,. \qquad (2)$$

By fitting resistor and capacitor value should be frequency and quality factor expressed as

$$f_0 = \frac{N}{2\pi \cdot 10^6 \cdot C_n} \quad (3), \qquad Q = \frac{R_Q}{R_1} = \frac{10^6}{N \cdot R_1}\,. \qquad (4)$$

The capacitor values determine the frequency sub - ranges and they are switched by electronic switch. The DA converter connection is shown in Fig. 6.
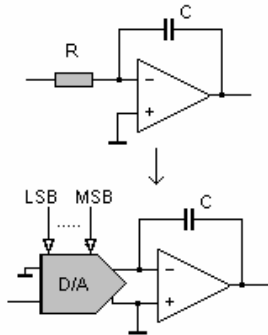

Fig. 6 The idea of frequency control
using DA converter connection

The design and optimization of DA converter is very important, this circuit part essentially determines resulting control parameters and upper frequency limits of each selective filter block. First of all, the undesirable parasitic capacitances of internal CMOS switch must be reflected in case of high equivalent resistance of converter. These capacitances transformed on input and outputs are modifying control function of N (2).

Using proper network compensation these control deviations were diminish to "virtual zero". Discussed problem can be partly solved by gain amplifier increasing by higher frequency area however that means necessity to use the operational amplifier with higher cut-off frequency. The further problem (of required driving function of frequency through whole sub-range) was solved using special common correction algorithm.. The developed algorithms take part of calibration option in system menu and there are implemented in the control processor. Typical curve of the frequency setting error as dependence on control number N for one frequency sub-range used to calibration process is shown in Fig. 7.
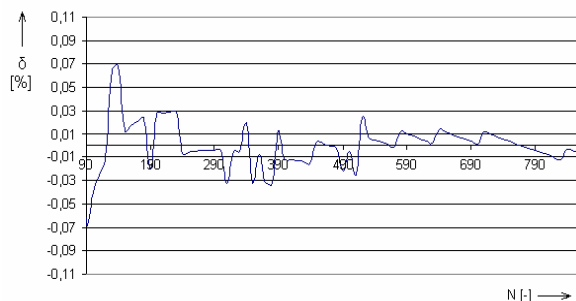

Fig. 7 An example of frequency setting error vs
control number N

4 Microprocessor control system

Whole integral system of filter is controlled by central microprocessor. The user interface is realized in direct mode with internal keyboard and LCD display and in remote control mode form the superior computer. There are these main tasks of the numeric control:

- Create user interface
- Communication with superior PC
- Communication wit 1st and 2nd order units
- All parameters of unit and configuration driving
- Parameters of transfer function controlling

As main control microprocessor the type Motorola MC68HC912B32 was chosen. A communication between the microprocessor and frequency filter blocks is based on I^{2}C bus, as computer interface serves RS232/USB bus.

5 Frequency filter control software

The new software product was developed for the purpose of user-friendly system driving. It allows control of all important parameters of filter by using an external computer. The principal parts of this system are:

- Block parameters direct access (direct access to transfers function parameters)
- Final frequency characteristic representation including zero/pole location,
- Basic filter type (LP, HP, BP, BR) design
- Interactive parameters tuning interface

A matrix of block parameters allows access directly to the actual block parameters and they are used for display of the results of design. The outputs switch configuration is also available in this matrix.

Part of characteristic plotting allows to display the final filter characteristics. It also allows to operate with all particular 1st and 2nd order "filter sub characteristics" (Fig. 8). The part of configurable ranging of blocks is also available.
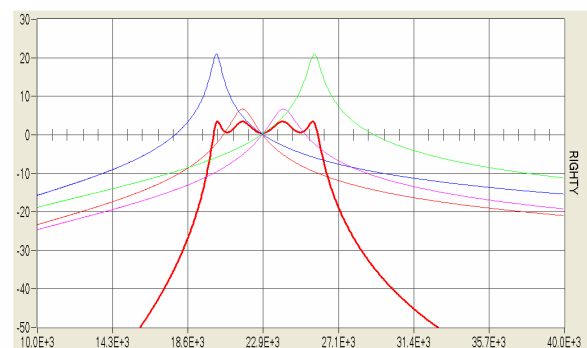

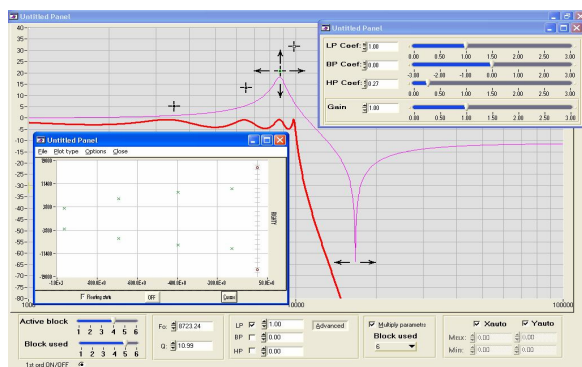Fig. 8 Principe of particular transfer function addition

Fig. 9 Coefficient tuning window with zero/pole sub-chart

The interactive coefficient tuning presents a new easy method in area of non-analytic transfer function design. In the window of PC screen (see Fig.9) it is shown the principle of filter design using particular 1st and 2nd order characteristics, where on each characteristic can be set a special mark (in cross form) corresponding in X-axis to possible resonant frequency shift and in Y axis to quality block establishment. By moving this mark in the graph area we can change directly the parameters of particular "sub-characteristic" the system.
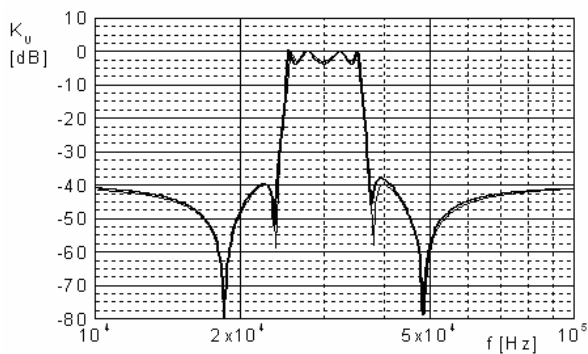

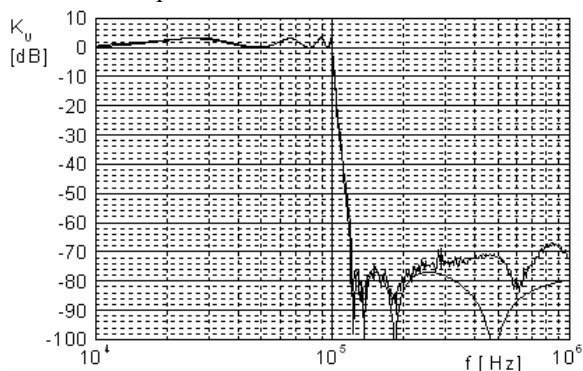Fig.10 Theoretical and measured magnitude responses of the 8th order BP filter

Fig. 11 Theoretical and measured magnitude responses of the 8th order LP filter

This system presents the new effective development support for cascade frequency filter design and optimisation as well as cascade filter design tutorial.

Fig.12 Resulting realization of digitally controlled analog filter unit

6 Conclusion

The new developed integral system presents a powerful system in area of physical filter modelling and realization. It allows fast-adaptation to the condition in exanimate system. High accuracy of filter parameter setting of each filter block makes developed filter system fully comparable to digital signal processing methods. The range of frequency and quality factor step adjustment enable to cover practice filter requirements in frequency range to $2 .10^{5}$ Hz and attenuation requirements in range about 100 dB. It determines presented system as ideal tool for analog signal pre-processing. This system is also advantageous for education process. Examples of theoretical and measured characteristic (Fig. 10, Fig. 11) confirm high quality of realised filters.

References
[1] MICHAEL G. ELLIS: , Sr." Electronic filter analysis and synthesis". Artech House. Boston, London,1996
[2] HÁJEK, K., SEDLÁČEK, J., SVIEZENY, B, MICHAL,V.: Universal digitally controlled analog filter, May 2003, *Radioelektronika* 2003, Brno
[3] HÁJEK, K., SEDLÁČEK, J., SVIEZENY, B.: The New Universal Tunable Biquad with optimization for the High Frequency Band. *In the Proceedings of the XI. International Symposium on Theoretical Electrical Engineering ISTET´01*. Linz 2001, pp. 139 – 142.
[3] MICHAL,V.: Transceiver control unit, EEICT 2002, Brno, http://jaja.kn.vutbr.cz/~michal/