Digital filter design by Micro-Cap tools

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***Abstract*—Digital filter design procedure using Micro-Cap capabilities is proposed. The procedure is illustrated by an example.**

Keywords—digital filter design, circuit simulation, bilinear transform, analog prototype

I. INTRODUCTION

Electronic circuit simulation program Micro-Cap is a popular instrument for analog and digital circuit simulation [1]. The modern versions of the program also allow mixed analog-digital circuits simulation. The program contains a module for analog active and passive filter design and digital filter simulation tool.

Digital technology in general and digital signal processing in particular is a corner stone of modern computer and telecommunication technology. Digital filters are the most frequently used signal processing devices so a circuit simulation program application scope would be significantly enhanced if digital filter design tools were included into it. Unfortunately Micro-Cap does not contain digital filter design module ready for use but it's user can perform the design using available possibilities. Below procedures of infinite impulse response digital filer design and simulation in Micro-Cap are described.

II. DESIGN PROCEDURE

The most commonly used method of infinite impulse response (IIR) digital filter design is the analog prototype method. Usually the normalized low pass analog filter is used as prototype to design all four types of standard magnitude responses (low pass, high pass, band pass and band stop). The transformation from normalized prototype to the IIR digital filter in this case is performed by using bilinear (for low and high pass filters) or biquadratic (for band pass and band stop filters) z-transform. As a result different transform formulas are needed for each of four magnitude response types. This formula diversity can be avoided when using Micro-Cap analog filter design module.

Analog filter design module from Micro-Cap provides transfer function *H(U)* of normalized filter with required type of magnitude response. It is presented as a product of first or second order sections the transfer functions of which $H_j(U)$ are presented in the form shown in Table I (for Butterworth, Chebyshev and Bessel approximations).

TABLE I. TRANSFER FUNCTIONS FORMS

Magnitude response type	Transfer function
Low pass	$H_j(U)=\dfrac{b_{0j}}{U^2+b_{1j}U+b_{0j}}$
High pass	$H_j(U)=\dfrac{U^2}{U^2+b_{1j}U+b_{0j}}$
Band pass	$H_j(U)=\dfrac{b_{0j}U}{U^2+b_{1j}U+b_{0j}}$
Band stop	$H_j(U)=\dfrac{U^2+1}{U^2+b_{1j}U+b_{0j}}$

Due to this fact in all cases the single transform formula can be used [2, 3]:

$$U=c\frac{1-z^{-1}}{1+z^{-1}}$$

Analog filter design module of Micro-Cap provides transfer function of normalized filter so to obtain required value of the frequency f_0 the transformation formula changes to:

$$U=ctg\left(\frac{\pi f_0}{F_s}\right)\frac{1-z^{-1}}{1+z^{-1}}$$

The result of this transformation is digital filter transfer function H(z) as a product of first and second order sections. Second order sections are obtained in the form:

$$H_j(z)=A_0\frac{1+A_{1j}z^{-1}+A_{2j}z^{-2}}{1+B_{1j}z^{-1}+B_{2j}z^{-2}}$$

Formulas for denominator coefficients calculation are the same for all response types. For the second order sections coefficients are calculates as:

$$c = \cos \frac{2\pi f_0}{F_S}$$

$$s = \sin \frac{2\pi f_0}{F_S}$$

$$k_{1j} = (1 + b_{0j}) + c(1 - b_{0j})$$

$$k_{2j} = (1 - b_{0j}) + c(1 + b_{0j})$$

$$B_{2j} = \left(1 - \frac{b_{1j} s}{k_{1j}}\right) / \left(1 + \frac{b_{1j} s}{k_{1j}}\right)$$

$$B_{1j} = -\frac{k_{2j}}{k_{1j}} \left(1 + B_{2j}\right)$$

For the only first order section formulas are following:

$$ct = \frac{1 + c}{s}$$

$$B_{1M} = \frac{1 - ct / b_{0M}}{1 + ct / b_{0M}}$$

In the formulas above f_0 represents characteristic frequency of the filter under design. For low and high pass filters it is the pass band edge frequency. For band pass filters it is central frequency of the pass band. For band stop filters it is central frequency of the stop band.

At the same time numerator coefficients for different response types are calculated differently as shown in Table II, III.

The formulas in the Table II, III may be coded as a number of `.define` directives in the design file. But more convenient way to implement digital filters is to insert both design formulas and implementation in a number of macro definitions. In this case a designer can implement digital filters just by placing the necessary macros into schematics and by specifying parameters of macros in the standard component parameter window. For example macros for low pas second order filter section may be implemented as shown in Fig.1 or in Fig 2.

The digital filter design procedure is comprised of the following steps:

1. Define the required parameter values using .define directive. For band pass and band stop filters recalculate the value of Q-factor taking into account the nonlinear frequency transformation (see Design Example below).
2. Open analog filter design window. As we need only the values of transfer function coefficients and are not interested in implementation both active and passive filter designs are suitable. Choose the simplified style of parameters specification. Fill in the parameters specification fields and calculate the transfer function coefficients.

TABLE II. NUMERATOR COEFFICIENTS FORMULAS FOR SECOND ORDER SECTIONS

Magnitude response type	Transfer function
Low pass	$A_{0j} = a_{0j} \frac{(1-c)(1+B_{2j})}{2k_{1j}}$ $A_{1j} = 2$, $A_{2j} = 1$
High pass	$A_{0j} = a_{0j} \frac{(1+c)(1+B_{2j})}{2k_{1j}}$ $A_{1j} = -2$, $A_{2j} = 1$
Band pass	$A_{0j} = a_{0j} \frac{s(1+B_{2j})}{2k_{1j}}$ $A_{1j} = 0$, $A_{2j} = -1$
Band stop	$A_{0j} = a_{0j} \frac{(1+B_{2j})}{k_{1j}}$ $A_{1j} = -2c$, $A_{2j} = 1$

TABLE III. NUMERATOR COEFFICIENTS FORMULAS FOR FIRST ORDER SECTION

Magnitude response type	Transfer function
Low pass	$A_{0M} = \frac{1+B_{1M}}{2}$ $A_{1M} = 1$
High pass	$A_{0M} = \frac{1+B_{1M}}{2}$ $A_{1M} = -1$

3. Insert into schematic file required number of filter sections macro definitions. First order section must be only one. Usually it is the first or the last section of the filter. While inserting sections set macros parameters (in the automatically opened parameters window) to values of analog prototype coefficients and required frequencies.
4. Perform frequency and transient analysis to obtain frequency responses, impulse and transient responses or reaction to some input signal. Analyze the responses and make sure that the design requirements are met. In the case of failure all the design procedure should be repeated with corrected requirements.

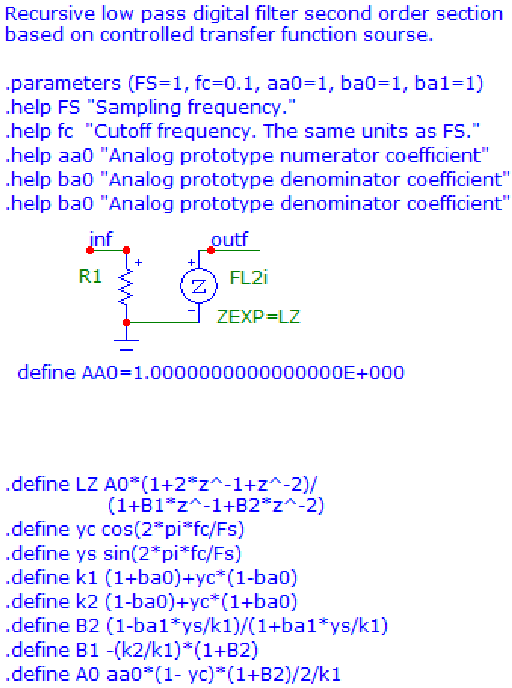


Fig. 1. Macro definition for a low pass filter section using controlled transfer function sourse.

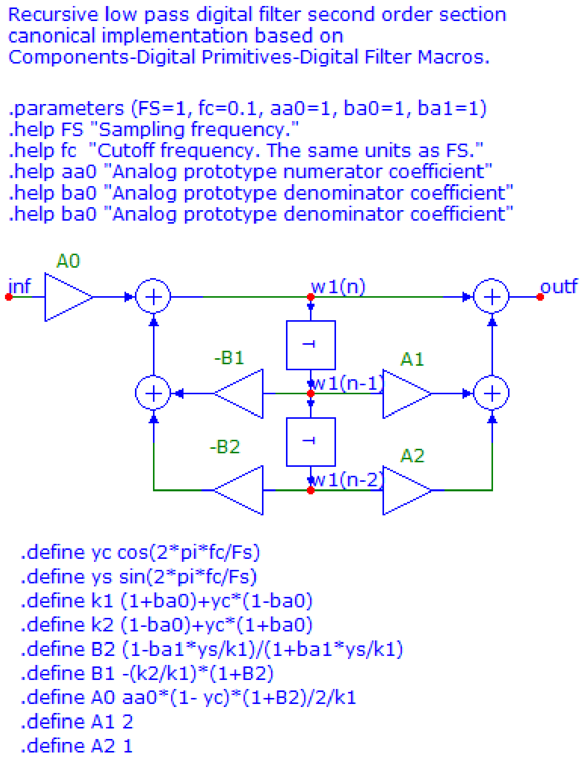


Fig. 2. Macro definition for canonical implementation of second order low pss digital filter using digital filter macros from `Component-Digital Primitives-Digital Filter Macros` supplied with standard Micro-Cap disributive.

Required coefficient representation accuracy depends on the design requirements. In the most practical cases single precision is enough. But in some cases especially in the case of narrow band filters the maximum achievable accuracy may be needed. The accuracy of the displayed calculation results may be changed as necessary through the `Options – User Settings – Format` menu. The maximum display accuracy is 16 decimal digits of the fractional part of a number. If parameter quantisation is required the function `trun` proposed in [4] can be used.

III. DESIGN EXAMPLE

The macro definitions of Fig.1, Fig.2 were tested with Micro-Cap 11 evaluation version.

Design the digital filter meeting the following requirements:

Magnitude response type	- Band pass
Approximation	- Butterworth.
Central frequency	- f_0=455kHz
Pass band ripple	- A_p=3dB
Q-factor	- Q_f=45.5
Filter order	- N=2
Sampling frequency	- F_s=4*f_0.

Insert the requirements into design file (*.cir):

```
.define f0 455k
.define Ap 3
.define Qf 45.5
.define Fs f0*4
```

Calculate Q-factor for analog prototype filter:

```
.define f0a (Fs/pi)*tan(pi*f0/Fs)
.define fp2a
      (Fs/pi)*tan(pi*f0*(1+1/(2*Qf))/Fs)
.define fp1a
      (Fs/pi)*tan(pi*f0*(1-1/(2*Qf))/Fs)
.define Qa f0a/(fp2a-fp1a)
```

Display the Q-factor value (don't forget to check the equation box):

```
.define Q=[Qa]
```

As the result we have

```
Q=2.8963323E+01
```

Open the analog filter design window and enter the analog filter requirements in simplified mode. Press `Enter` to perform calculation. The transfer function coefficients representation accuracy can be changed when necessary through local `Options` menu. Copy transfer function to digital filter design file:

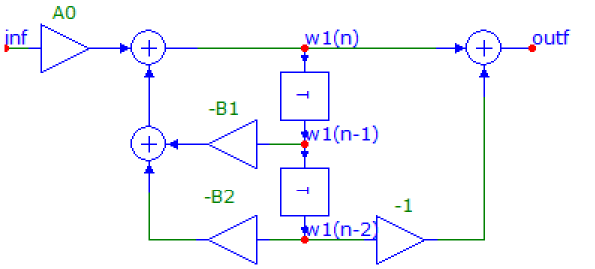
```
.define BP
(34.11639m)*U/
(U*U+24.12213m*U+975.87426m)*
(34.95982)*U/(U*U+24.71848m*U+1.02472)
```

Insert two section of digital filter in series and specify their parameters.

For band pass filter section macro definition must be changed as shown in Fig.3. Coefficient A1 for band pass filters always equals zero so the corresponding multiplier may be omitted. Coefficient A2 always equals -1. Formulas for denominator coefficients are the same for all response types. The only different formula is that for A0. As the order of band pass filter is always even there is no need for first order sections.

```
Recursive pass band digital filter second order section
canonical implementation based on
Components-Digital Primitives-Digital Filter Macros.

.parameters (FS=1, fc=0.1, aa0=1, ba0=1, ba1=1)
.help FS "Sampling frequency."
.help fc  "Central pass band frequency. The same units as FS."
.help aa0 "Analog prototype numerator coefficient"
.help ba0 "Analog prototype denominator coefficient"
```




```
.define yc cos(2*pi*fc/Fs)
.define ys sin(2*pi*fc/Fs)
.define k1 (1+ba0)+yc*(1-ba0)
.define k2 (1-ba0)+yc*(1+ba0)
.define B2 (1-ba1*ys/k1)/(1+ba1*ys/k1)
.define B1 -(k2/k1)*(1+B2)
.define A0 aa0*ys*(1+B2)/2/k1
```

Fig. 3. Macro definition for band pass filter section.

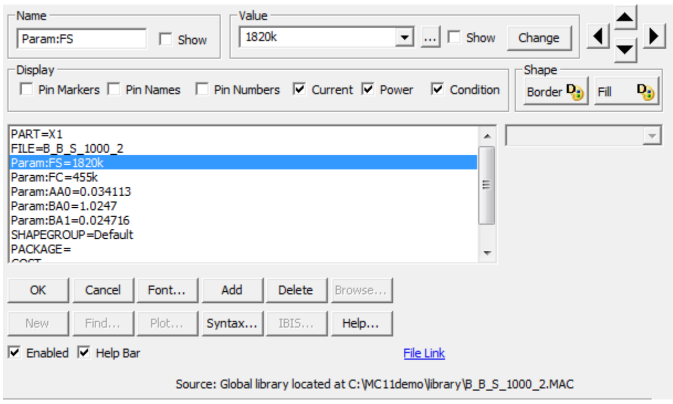
Parameters for each section are specified in component parameter window. Here we have to specify sampling frequency (Fs), central pass band frequency (Fc) and coefficients of the corresponding section of analog prototype obtained from analog filter design module (AA0, BA0,BA1). Their values are displayed in Fig.4 as they appear in the component parameter window during section macros instantiation.

Correctness of the design filter must be verified using frequency analysis. Obtained magnitude response of the filter must meet design requirements specified at the beginning of this design example. For this example Frequency analysis gives the magnitude response meeting the design requirements (Fig.5).

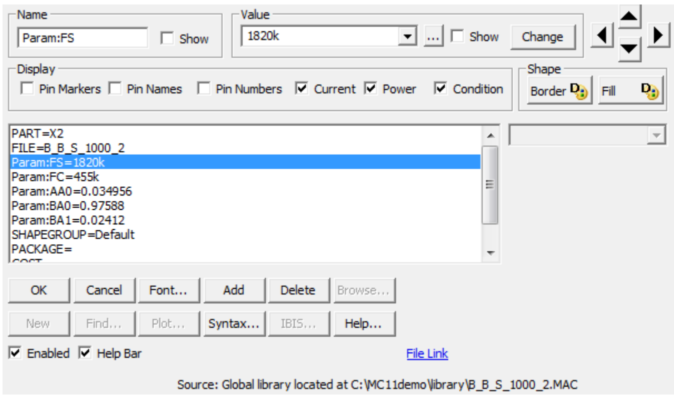
When necessary, transient analysis may be performed to obtain impulse response, step response or response to some arbitrary signal.

IV. Conclusion

Proposed design procedure provides a simple and accurate digital filter design tool in Micro-Cap environment. It enhances an engineer's ability to design and simulate digital and analog-digital systems in a single environment.



a)



b)

Fig. 4. Parameters specifications: a)for the first filter sections; b) for the second filter section..

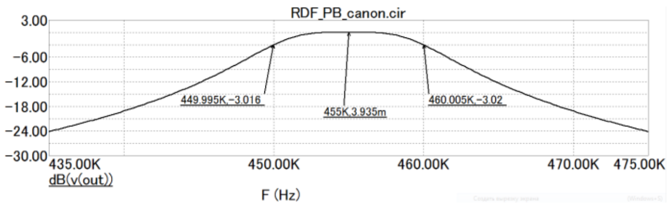


Fig. 5. Resulting magnitude response of the designed digital filter.

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