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Software for teaching design and analysis of analog and digital filters

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ABSTRACT

We describe custom designed software oriented for teaching design and analysis of analog and digital filters. The tools described are used as teaching aids and are not intended as substitutes for lectures on the topics covered. The analog filter design covers approximation of the filter functions as well as the design of the active-RC filters stages. The digital filter design part is able to design IIR using transformations from the s-plane. FIR filters are designed using the Remez exchange algorithm. Another part of the software package is in charge of simulating an arbitrary digital filter topology for different wordlengths allowing students to observe the limitations inherent to each topology.

INTRODUCTION

The teaching of filter design is present in almost every electrical engineering undergraduate curriculum. This fact is due to the presence of at least a filter circuit in every electronic system. Examples of systems which make extensive filter usage are telecommunications networks, audio/video systems, medical diagnosis systems, etc. However, if we wish to teach to students as many topics on filter design as possible in a limited amount of time(usually two semesters for teaching analog and digital filters) instructors have to use computer aids to help themselves in the teaching process. In our university, we have designed several software packages to aid in the teaching-learning process. This software aids have as common basis

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that they are small enough in order to not requiring to be loaded in a hard disk and that they are specifically designed as teaching-learning aids. Although there are already available a great deal of fine software packages which could be used in teaching, none of them satisfied the requirement to be used as teaching aids, and therefore, could not be used for our purpose.

MOTIVATION FOR A SOFTWARE ENVIRONMENT

Commercial software packages used for the design of digital and analog filters have several important drawbacks [1]. Two of the most significant are 1) their large size and thus needing a hard-drive for installation; this fact not being very practical when there are several students having access to it , 2) in addition, they are very costly and need a hardware device allowing usage in only one computer at a time [1].

On the other hand,the University designed packages can be very friendly (if well designed), small and can be used in several computers at a time. Furthermore, they can be designed to do only the work the instructor wishes to study in the classroom and thus the students can check their results in the computer. In this way, students are now seeing the development and implementation of algorithms which convert theory to practice. Any misconceptions they might have about the theoretical material are dispelled by this procedure. [2].

With the above concepts in mind, we decided to apply them to the teaching of filter theory and to design an environment with a teaching approach of the topics involved.

ANALOG AND DIGITAL FILTER DESIGN

The analog filter design software is called FILTROS III [3] and it consists of three parts: a) input stage, where the input data is requested, b) the output stage, where the output data and plots, are given by the computer to the user, and c) the network realizations, where active-RC networks and FIR and IIR implementations for digital filters realizing the requested approximation function are obtained. A brief description of each stage follows.

Input Stage

The input is menu oriented so it is easier for the student entering the requested input data. Menus at this stage include promptings regarding which kind of filter the students wish to design: digital or analog; type of filter: LP, HP, BP, or BR; type of characteristic : Butterworth, Chebyshev, Inverse-Chebyshev, and Elliptic, (this option is not used to design FIR digital filters); frequency (Hz or rad/seg), and attenuation specifications in passband and stopband (dB). For digital filters, students can choose between FIR and IIR digital filters, and additionally a sampling frequency has to be specified. For the case of IIR filters, if is possible to use either the bilinear, backward, or forward transformations to

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obtain the digital filter transfer function from an analog one. For FIR filters, the user specifies the number of bands, the cut off frequencies, and the attenuations required for each band. Also, the user can input an arbitrary transfer function to obtain plots of magnitude, phase, or group delay responses, and network realization.

Output Stage

In this stage, we can obtain the data calculated by the software package. This information includes the filter transfer function in any one of three forms: poles and zeros, numerator and denominator polynomial coefficient, and quadratic factors. The user can obtain also plots of magnitude, phase, and group delay responses. A perturbation option is provided for user provided functions or for the ones obtained with FILTROS III. For FIR digital filters, the output data obtained is the set of coefficients corresponding to the impulse response which satisfies the requirements.

Network Realizations

The last stage is the synthesis of a filter network. To this moment, we have passive ladder realization and active-RC cascade realizations. Active-RC realizations include Sallen-Key, Multiple-Feedback, KHN, Tow-Thomas, and GIC topologies. For the case of digital filter realizations, the package provides direct form topologies for FIR and IIR (cascade of 2nd order stages) filter realizations.

FINITE WORDLENGTH SIMULATION FOR DIGITAL FILTERS

Among the many limitations an actual implementation of a digital filter can suffer, one of the most important and that can severely limit the performance of the filters is the finite wordlength. As described above, the design procedure starts form specifications and it proceeds to obtain a transfer function and ends with a circuit realization where the coefficients are implemented by multipliers. However, the multipliers in the realization implement finite wordlength multiplications and this phenomenon gives place to round-off errors which distort the output signal. To observe the effects of the finite wordlength in the filter behavior, we have designed a software package which allows us to observe the output signal of an arbitrary digital filter topology for user different wordlengths. With this tool, students can observe, before actually building the network, how the finite wordlength will modify the output of the network. The input data to this software package is the digital filter topology. They description of the digital filter topology is given in terms of input and output nodes for each branch as well as the function this branch is performing (adding, multiply, and delay). Fig. 1 shows the input data for a direct form second order digital filters. The following section will show examples on the use of the software described.[4]

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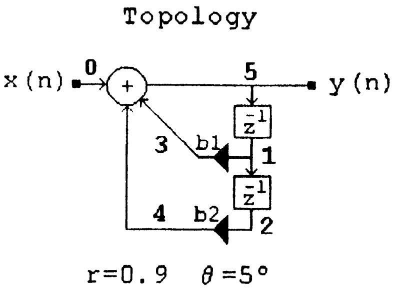



```
{ structure }
/SUM(5,3,4,0)
/ZET(1,5)
/ZET(2,1)
/MUL(3,1,b1)
/MUL(4,2,b2)
```

Fig. 1 Input data for direct form digital filter.

SOME EXAMPLES USEFUL IN THE TEACHING PROCESS

In this section we present some examples representative of the use that can be given to the software described above.

As our first example let us consider the design of an active-RC filter. Here, a number of design issues arise which illustrate some of the theoretical concepts covered in the lectures. To mention the most important ones, once the type of transfer function has been entered and computed, students can observe how the approximation type choice will affect the transfer function magnitude, phase and group delay. After this students can see how the circuit realization is dependent upon the approximation type chosen which will change the number of op-amps needed, as well as the passive component count (resistors and capacitors). As an example of an active-RCA realization, in Fig. 2 it is shown a state-variable second order stage which is part of a 4th order bandpass Butterworth filter. This example will provide students with several options for the active RC topologies. There they will be able to choose number of op-amps, number of capacitors, op-amps with grounded non-inverting input, and many other options for the topology.

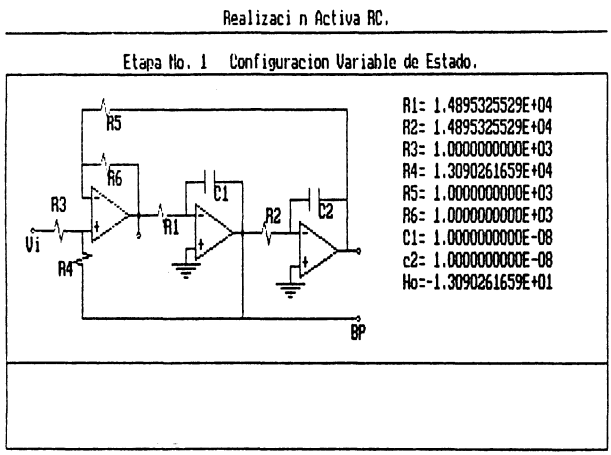
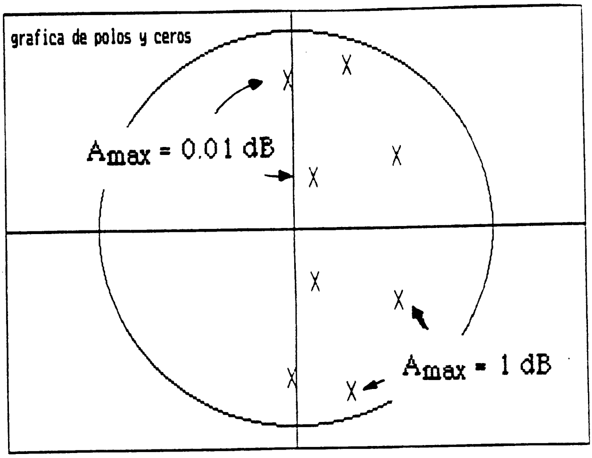
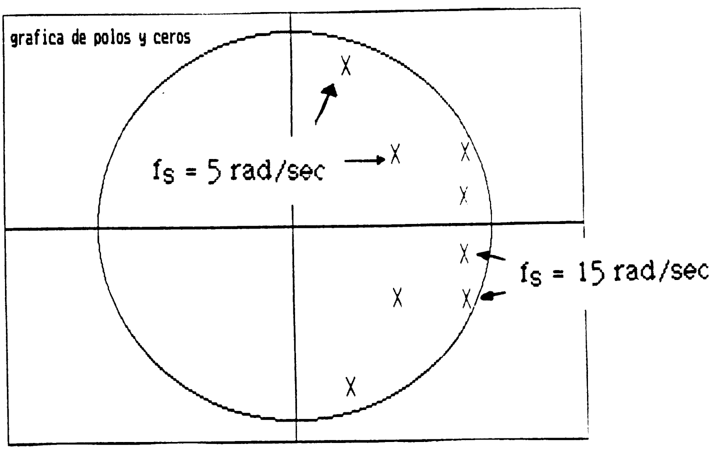


Fig. 2 A state-variable KHN stage. This is the first second-order stage of a 4th order bandpass filter.

Our second example refers to IIR digital filter design. It is well known that each set of specifications gives different pole positions. Our software package has the option to plot pole-zero positions in the z-plane (This option is also available for analog filters in the s-plane.) Plots of pole positions for fourth order Butterworth low-pass digital filters are shown in Fig. 3a for different sampling frequencies. Another important aspect of pole position occurs when the sampling frequency, f_s, is increased. The change in f_s will shift the poles to cluster close to z=1. This is shown in Fig. 3b for the Butterworth filter poles of Fig. 3a. This example can be used to explain some characteristics of filter functions having poles close to the point z=1, such as higher sensitivities to wordlength and noise. [5]



(a)



(b)

Fig. 3 a) Pole positions for Butterworth digital filters for different passband attenuation, b) Pole position shift with sampling frequency change.

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As the last example we show how the output signal of a digital filter is improved as the wordlenght is increased. The filter topology is the one shown in Fig.1 and the input signal is a sum of signal with frequencies 1KHz, 2KHz, and 4 KHz with a 10 KHz sampling frequency Hz. The output is shown for 4 bits, 6 bits, 8 bits, and infinite wordlength. Fig. 4 shows these output signals. This example shows students how to select an optimal wordlength for a digital filter. Here they will see that 8 bits in some cases are enough while some other topologies might require a longer wordlength.

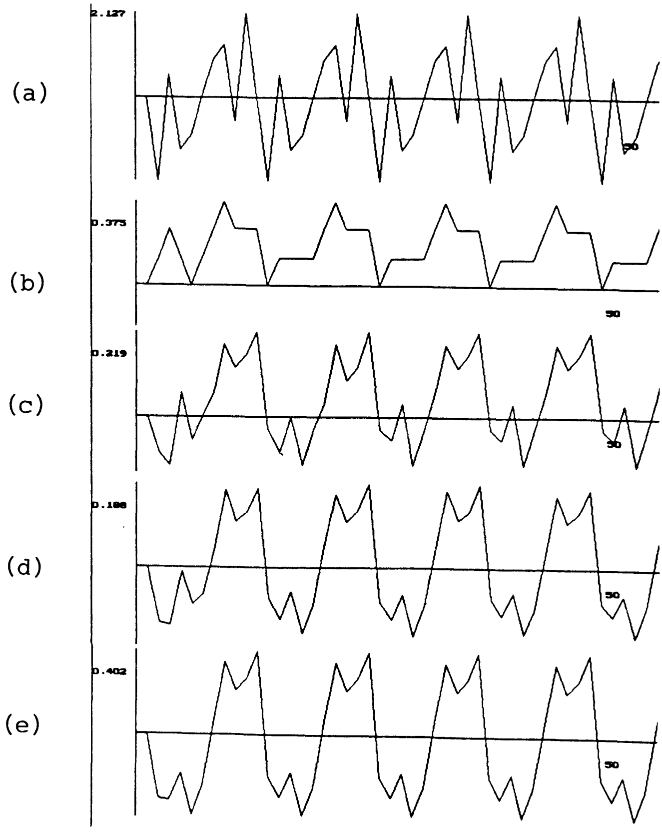

Fig. 4 a) Input signal, b) 4 output for a 4 bit wordlength, c) Output for 6 bit wordlength, d) Output for 8 bit wordlenght, e) Output for infinite wordlength.

CHARACTERISTICS OF THE SOFTWARE

The software packages described above are written in Turbo Pascal to run on IBM or compatible PC. There are about 10,000 lines of code and require 520 K bytes of RAM to run requiring overlays to be executed.

CONCLUSIONS

We have described software for teaching analog and digital filters design and realizations. These packages have proved to be valuable in the teaching-learning process. The software described runs on IBM or compatible PC.

ACKNOWLEDGMENT

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