Computerized Tools in Digital Signal Processing

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INTRODUCTION Digital Signal Processing (DSP) is an important and growing area in electrical/computer engineering with applications in electronics, communication/control systems and biomedical instrumentation. The author has taught undergraduate and graduate courses in DSP where students usually have difficulty understanding and visualizing such abstract concepts as spectrum, impulse response, convolution and difference equations. The subject of DSP is a highly mathematical one which is based on the student prior work in engineering mathematics, electric circuits, linear systems, microprocessors, probability and random processes. The availability of computerized tools and software packages has given the engineering educators a flexibility in covering principles and fundamentals of the subject matter and, at the same time, illustrate the difficult concepts by means of numerical examples and high resolution graphics. We will concentrate on the use of MATHCAD and MATLAB in this paper, although other packages such as MATHEMATICAL and LABVIEW can also be used in certain cases.

THE COURSE The DSP course topics include: (see text reference, [1])

> A description of digital (discrete-time) signals and systems
> The sampling process (A/D, D/A), quantization and coding
> Convolution/correlation functions and difference equation modeling
> Analysis of linear time-invariant systems in the z-domain
> Frequency analysis of discrete-time signals and systems
> The discrete Fourier transform (DFT) and FFT algorithm
> Implementation issues, structures for realization of DT systems
> Digital filter design methods, FIR and IIR filter design strategies
> Quantization of filter coefficients and roundoff effects
> Optimal filtering in the presence of noise

Our students take the DSP course in their senior year after completing courses in linear systems and probability/random processes. The problem is that they may not be very familiar with classical methods of analog filter design (a very detailed and matured subject in electrical engineering) and this is a place where tools like MATLAB and MATHCAD can be very useful and allow students to experiment with different filter approximations like Butterworth, Chebyshev and elliptic design methods.

MATHCAD EXAMPLE With the availability of computer tools, it is important that the students can formulate and define the problem first and also have some understanding of the outcome of the problem and its physical interpretation. In this regard, we have found Mathcad to be a useful tool and students, with the availability of the classroom

version, are generally very pleased with the user friendly nature of this package. The professional version of Mathcad (see reference [2]) is available in our Computer Aided Engineering (CAE) center so that all engineering students can use it. Students can use Mathcad for solving homework problems or for their term papers. Here we present one example of its use.

Example. We want to design an FIR digital filter with the ideal (desired) frequency response (transfer) function given by

$$Hd\ (\omega) = 1;\ 0.8\pi < \omega < 0.9\pi, \qquad H_d\ (\omega) = 2\ ;\ 0.9\pi < \omega < \pi$$
$$Hd\ (\omega) = O\ ;\ \text{otherwise}$$

Note that ω represents digital frequency in radian. With this information the impulse response of the filter can be derived as follows
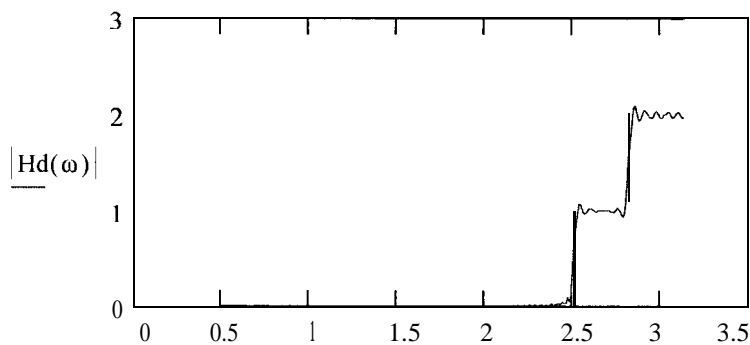
$$I(\omega) = H_d(\omega)\exp(j\omega)$$

$$H_d(n) = 1/2\pi \int_{-\pi}^{\pi} I(\omega)\ d$$

$$n := 1, 2 .. 100$$
$$\omega := 0, 0.01 .. \pi$$
$$hd(n) := \left(\frac{1}{n \cdot \pi}\right) \cdot (\sin(0.8 \cdot \pi \cdot n) + \sin(0.9 \cdot \pi \cdot n))$$
$$Hd(\omega) := 0.3 + 2 \cdot \sum_{n} hd(n) \cdot \cos(n \cdot \omega)$$
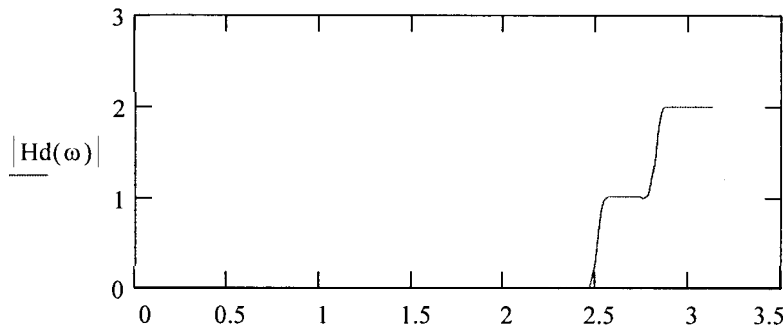

Here we have used the so-called rectangular window in our design. We can use other types of windows as well. Next we present the results using the Hamming window function.

$$n := 1,2..100$$

$$\omega := 0,0.01..\pi$$

$$hd(n) := -\left(\frac{1}{n\cdot\pi}\right)\cdot(\sin(0.8\cdot\pi\cdot n) + \sin(0.9\cdot\pi\cdot n))\cdot\left(.54 + .46\cdot\cos\left(n\cdot\frac{\pi}{100}\right)\right)$$

$$Hd(\omega) := 0.3 + 2\cdot\sum_n hd(n)\cdot\cos(n\cdot\omega)$$

MATLAB EXAMPLE With the availability of the classroom (student edition) of the MATLAB and its companion, SIMULINK, and also because signal processing toolbox (see reference [3]) has many useful functions, this is a widely used package in this subject area. The professional version is also available in the CAE center of our school for student use. We now present an example of its use in designing filters.

Example. Here we design an analog Chebyshev filter, type I with normalized cutoff frequency of 1 rad./sec. It is a seventh order filter with a ripple of 1 dB in the passband (PB). Using matlab command of [z,p,k]=CHEB1AP(7,1) we obtain the midband gain of 0.03075434 and poles (in the left half plane or LHP) of

$$s_0 := -0.2054$$

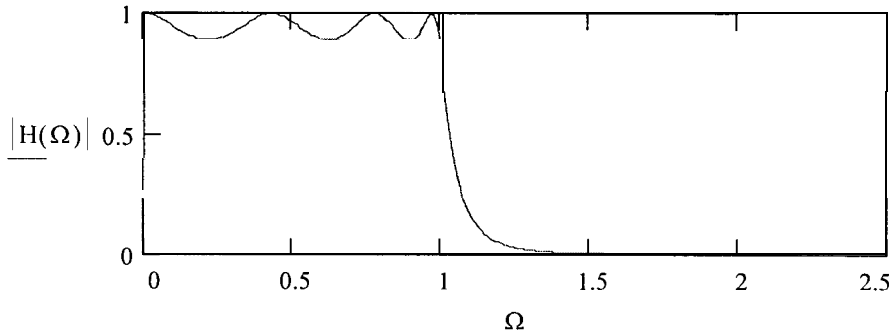$$s_1 := -.1851 + 0.4429j$$

$$s_2 := -.1281 + 0.7982j$$

$$s_3 := -0.0457 + 0.9953j$$

$$k := 0.0307$$

$$\Omega := 0,0.01..2$$

$$j := \sqrt{-1}$$

$$H(\Omega) := \frac{k}{\left(\Omega\cdot j - s_0\right)\cdot\left(\Omega\cdot j - s_1\right)\cdot\left(\Omega\cdot j - \overline{s_1}\right)\cdot\left(\Omega\cdot j - s_2\right)\cdot\left(\Omega\cdot j - \overline{s_2}\right)\cdot\left(\Omega\cdot j - s_3\right)\cdot\left(\Omega\cdot j - \overline{s_3}\right)}$$

The design of the digital filter can then be completed using several known transformation methods, e.g., bilinear transformation. In this method (assuming that students have done pre-warping in analog design) one can simply replace the "s" in the transfer function of the analog filter by the expression $2(z-1)/(T(z+1))$ where T denotes the sampling interval (period) and z is the complex frequency variable. More examples will be presented at the ASEE annual conference in Washington, D.C.

CONCLUSIONS In the subject area of Digital Signal Processing and indeed in many courses in electrical/computer engineering field, there are a variety of "powerful" software packages available for use in the classroom. Our experience has shown that there is a right place and right time for using a specific package. In DSP, we have found MATHCAD to be very useful and student friendly. We also recommend the use of MATLAB and its signal processing toolbox. The important thing is that students need to experiment with these tools, but also they need to have an understanding of the principles behind the rich subject of signal processing. They should be able to interpret the results of their computer simulation.

REFERENCES

[1]. "Digital Signal Processing", 3rd ed., by Proakis and Manolakis, Prentice Hall, 1996
[2]. Mathcad User's Guide, MathSoft, Inc., 1995
[3]. Signal Processing TOOLBOX User's Guide for use with MATLAB, by Little and Shure, MathWorks, Inc., 1992