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Different Approaches To Learning And Applying Digital Signal Processing In Power Electronics

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A few decades ago digital signal processing (DSP) was a relatively obscure field associated with the mathematics of numerical analysis. However, as microcomputers have become cheaper and faster, DSP has become commonplace in cell phones, motor drives, medical instruments, and many other applications, often mixed in with analog circuits. Power electronic systems are no exception. Microcomputers (µCs, also known as microcontrollers) are fast enough for real-time control of converter feedback loops.

DSP is so common that it has moved into the mainstream of what an electronics engineer needs to know nowadays. Yet the emergence of DSP has happened well within the career lifetime of most engineers, and it has required that we all learn the subject to some extent, or else hope to get around it.

One of the early confusion factors in learning DSP is the rather different emphases placed upon it in DSP books. If you learn DSP from a book like the Prentice-Hall classic, *Digital Signal Processing*, by Oppenheim and Schafer (or from Alan Oppenheim's excellent MIT video course, which follows his book to a great extent), or even more so, from another Prentice-Hall book, *Digital Filters*, by R.W. Hamming (the "Hamming window" Hamming), you will get a filter-oriented view of DSP.

On the other hand, if you learn it by reading digital control books, you will acquire a control-oriented view. And more basically, if you learn it by reading numerical analysis books of mathematics, you might wonder how it is related to electronics (or engineering). What then is the best approach to an efficient acquisition of the subject?

In this article, I'll explore this issue from a power electronics perspective. I'll briefly explain the differences in how DSP is explained by the different math and engineering disciplines, which source materials I consider the most effective in teaching DSP within each of these disciplines, which key DSP concepts must be grasped by power electronics engineers, and how the different approaches and methods serve the varied design tasks encountered in power electronics.

Numerical Methods Vs. Controls Vs. Filters

I know of no single DSP book that explains the different perspectives on studying and applying DSP at the outset so that the reader knows of the different emphases and styles of presentation of what is essentially the same subject-matter. Consequently, each learner of DSP must go through the somewhat bewildering experience of finding this out and of making the correspondences between the different styles.

For instance, in numerical-methods mathematics there are different methods of integration. Yet in control DSP, essentially the same topic is covered by expressing these different methods in different domains of analysis (s, w, z) as poles, and emphasizing how they affect feedback loop stability. In filter DSP the emphasis is instead on steady-state frequency response and how sharp the filter rolls off to provide frequency band separation.

Numerical methods tend to relate difference and differential equations more readily while the differing engineering DSP styles can roughly be categorized as time- (control) and frequency-domain (filter) presentations of DSP. For communications and pure signal processing, such as FFTs, Oppenheim and Schafer's book offers the optimal approach. For microcomputer real-time control, Phillips and Harbor, *Feedback Control Systems*, Prentice-Hall, 1988 might be better.

For control DSP, my favorite is an older book—the best control theory book I have found—by Roberto Saucedo and Earl Schiring: *Introduction to Continuous and Digital Control Systems*, published by Macmillan in 1968. More recent books have shown an increasingly refined presentation of DSP for control, as *digital* or *discrete-time control*, such as Tim Wescott, *Applied Control Theory for Embedded Systems*, Newnes, 2006.

DSP is categorized in control theory as discrete-time control. The quantity represented in discrete time either has discrete (digital) value representation (fully digital) or the range of values is continuous (analog). Sample-and-hold circuits, for instance, produce discrete-time samples with continuous (analog) values while DACs input and output discrete values in both time and quantity. In dual-slope DVMs and power electronics with switching converters, time is discrete in measurement or switching cycles while voltages and currents remain continuous or *piecewise-continuous*—that is, continuous in successive intervals (see Fig. 1.)

Whether a quantity has discrete or continuous values, the sampling behavior in time applies to both. The piecewise-linear waveform is continuous between discontinuities at the steps. Although drawn to look continuous, they are like step functions; the steps are mathematically discontinuous, but as electrical waveforms are continuous with a quick change.

These quick steps contain frequencies outside the bounds of the sampling frequency, $f_s = 1/T_s$. A perfect low-pass filter can recover what in Fig. 1 is the dotted waveform. It is delayed by half a sampling period, $T_s/2$. The frequency cutoff of the perfect reconstruction filter is at the *Nyquist frequency* of $f_s/2$. The mathematics required to derive the equations for these waveforms is no different than those found in continuous, complex-frequency *s*-domain theory.

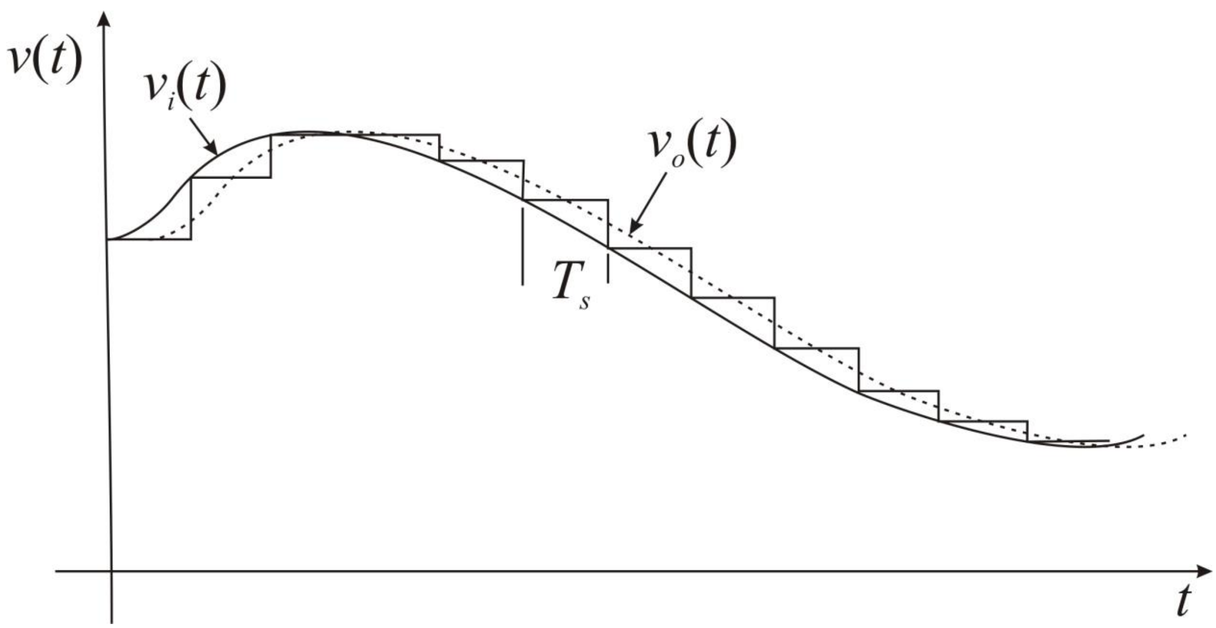

Fig. 1. Discrete-time waveforms result from sampling at regular time intervals. The result is piecewise-linear or "stairstepped" waveforms, as shown for this sample-hold waveform. The dotted waveform is the reconstructed output, a half a sampling interval ($T_s/2$) later, intersecting the center of the steps.

For circuit design, the best of the three ways to learn the same topics—numerical-analysis math, DSP, or discrete-time control—depends on whether you work largely in the frequency or time domain. In power electronics, discrete-time control is preferred, with numerical analysis math as a contributing source. For both DSP and discrete-time control, having a numerical-analysis book or two is a recommended supplement to either DSP books for frequency-domain work or control theory books for time-domain work. Although this distinction is not perfect, filter implementation is typically emphasized in DSP books while sampling circuits and feedback stability with sampling in the loop is the domain of control theory.

DSP Building Blocks

Sampling, as in sample-and-hold or track-and-hold circuits, causes continuous (analog) waveforms to be made into discrete-time, continuous-value waveforms. The discrete-time aspect can be either an impulse train or piecewise-continuous (stair-stepped) waveforms—usually the latter.

Sampling oscilloscopes have front-end samplers that sample at an instant in time to result in a series of dots on a graph of the sampled waveform. More commonly, the instant in time that the sample occurs is the active edge of a clocking waveform that captures a digital value into an A-D converter's digital hold circuit, which is a register or latch.

A-D converters (ADCs) perform the inverse function of D-A converters (DACs). Both convert between an analog ratio of an analog quantity, v_X to full-scale reference quantity, V_R, as v_X/V_R. The other side of the converter represents this ratio as a digital quantity, w_X to the full-scale value for N bits of representation, as $w_X/2^N$. Then

$$\frac{v_X}{V_R} = \frac{w_X}{2^N}$$

ADCs and DACs are sample-and-hold (S&H) functions. The sampling occurs on a digital clock edge and the hold is performed by a digital register. Most analog S&Hs also have a digital sampling waveform output in that sampling occurs during an active level and holding is performed by a capacitor as a voltage. (An inductor can hold a current as an electrical dual for holding an analog value, and is found most commonly in peak-current loop converter circuits.) In actual circuits, the S&H functions are found together though it is conceptually useful in block diagrams to separate them and place them in cascade for analysis.

In analog design, op amps are common for amplification and also integration. An open-loop op amp is essentially an integrator. Amplification is easier in the digital domain as multiplication by a constant. Analog multipliers are typically faster though less precise than multiplication in µCs, which is often time-consuming. A good design lets the analog circuits and µC or digital circuits each do what they do best.

DSP In The Control Loop

Electronic circuits with both analog and digital waveforms, especially in sampled feedback loops as shown in Fig. 2, can at first be confusing. ADCs and DACs are sample-hold components.

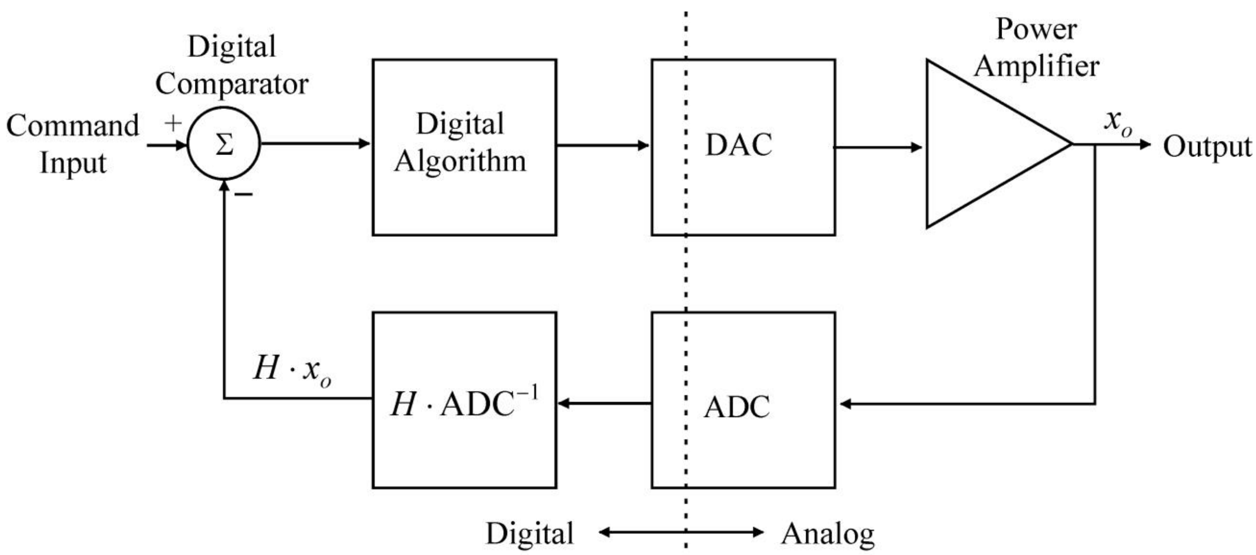

Fig. 2. A "mixed-waveform" analog and digital system with a sampled feedback loop such as might be found in a digitally controlled power supply. ADC and DAC are sample-hold components that convert between ratios of digital counts, $w/2^n$ and voltage ratios, v_X/V_R.

The sampling instants occur at the clock edges. Both convert between ratios of digital counts and voltages so that

$$\frac{w}{2^n} = \frac{v}{V_R}$$

where *w* is the digital count and the ratio is the fraction of the full-scale (fs) count, 2^n. V_R, the reference voltage, is the fs voltage for *v*, which forms a fractional ratio with V_R as the output voltage for a DAC or input voltage for an ADC. DACs and ADCs have inverse functions. To recover the input voltage to an ADC, a digital function in software inputs the ADC output, *w*, and calculates

$$v = \left(\frac{V_R}{2^n}\right) \cdot w$$

where $V_R/2^n$ is the mV/LSB scale factor of the A-D converter. The same equation applies to DACs. To output a voltage, *v* from the DAC, software applies the previous equation, solved for *w*;

$$w = \left(\frac{v}{V_R}\right) \cdot 2^n$$

where *n* is the number of bits of the converter. This formula and the previous equivalent formula are functions found on the digital side of the converter that convert to or from the analog quantities in software to or from counts of the converter. This scaling equation is static and does not include any of the *s*-domain dynamics that characterize, for instance, the half-interval delay of a waveform output from a DAC. Yet DSP literature derives the transfer functions of ADCs and DACs as an extension of continuous *s*-domain theory.

In closing, discrete and continuous waveforms are intermixed in circuits nowadays. By understanding sampling theory, in whatever form it can be learned, and some of its applications, the behavior and even the design of *mixed-signal* circuits, including sampling in power converters, can be best understood.

About The Author

Dennis Feucht has been involved in power electronics for 35 years, designing motor-drives and power converters. He has an instrument background from Tektronix, where he designed test and measurement equipment and did research in Tek Labs. He has lately been working on projects in theoretical magnetics and power converter research.

For more information on digital control in power electronics, see How2Power's Design Guide, locate the "Popular Topics" category and select "Digital Power".