DSP Filter Cookbook (Electronics Cookbook Series)

John Lane
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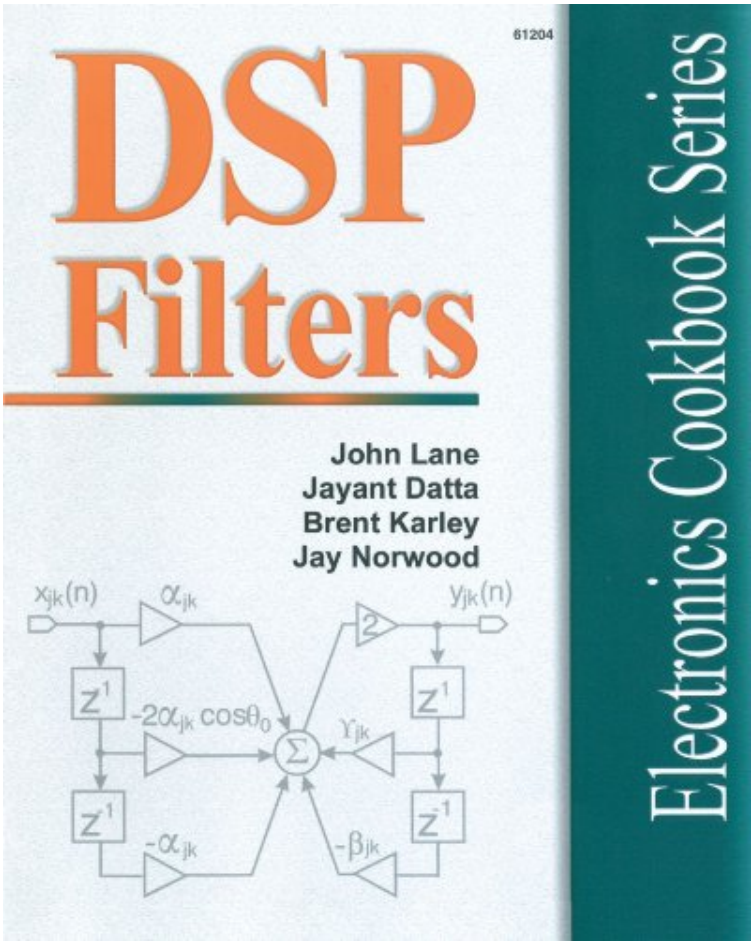

#1362441 in Books Cengage Learning 2000-12-01Ingredients: Example IngredientsOriginal language:EnglishPDF # 1 9.25 x .80 x 7.38l, 1.36 #File Name: 0790612046352 pages | File size: 17.Mb

John Lane : DSP Filter Cookbook (Electronics Cookbook Series) before purchasing it in order to gage whether or not it would be worth my time, and all praised DSP Filter Cookbook (Electronics Cookbook Series):

28 of 28 people found the following review helpful. Good for implementation details, go elsewhere for theoryBy calvinnmeThe title for this book may be misleading. Unlike most books on this subject, "DSP Filters" deals with a very specific class of digital filters, the infinite impulse response (IIR) filter. In addition, the IIR filters described are of the "maximally flat" gender. DSP books in general dedicate no more than a chapter or less to this particular subject. There are two reasons that an entire book on this subject might be a worthwhile read. First, these types of digital filters closely resemble filters which traditionally have been the backbone of audio electronics since the beginning of audio engineering. This is not to say that the maximally flat IIR filter is useful only in audio - there exists a wide range of applications for these filters, from instrumentation to business economics since, after all, an IIR filter is nothing more than a moving recursive average. The second purpose of this book is to fill the void of information on the subject. The details of this particular field of DSP seem to have fallen through the cracks of the university curriculum, DSP

textbooks, as well as semiconductor industry support. There are many reasons for this lack of detailed treatment on the subject. Even though the DSP industry strongly supported digital audio over a decade ago, the present focus is moving rapidly in the direction of wireless products. There may be a return to industry support of digital audio, since high quality audio will very likely become an important part of the future wireless market. A final conjecture as to why there exists a lack of treatment on this subject in other books is the prevailing DSP philosophy that analog filters are inferior due to their non-linear phase characteristics, therefore, there would be no reason to model analog filters in the digital domain. The finite impulse response (FIR) filter is the mainstay of DSP filter technology, since it has the ability to produce linear phase characteristics. In many applications, including audio, this may be very advantageous. However, there are numerous applications (such as fixed band equalizers, tone controls, and parametric equalizers), which do very well with analog filters, and therefore are ideal applications for the maximally flat IIR filter. In this case, the advantage of digital versus analog is not to "improve" the phase or magnitude response characteristics, but rather to add inherent stability to the filter response, as well as the ability to tune the response based on digital control of the filter's cutoff (or center) frequency, bandwidth (or Q), and gain. The table of contents of this book is as follows:SECTION I: IIR FILTER DESIGN FORMULASChapter 1: IntroductionChapter 2: Filter Design BasicsChapter 3: Digital BasicsChapter 4: First-Order Low-Pass FilterChapter 5: First-Order High-Pass FilterChapter 6: Second-Order Low-Pass FilterChapter 7: Second-Order High-Pass FilterChapter 8: Second-Order Bandpass FilterChapter 9: Second-Order Band-Stop FilterChapter 10: Peaking FilterChapter 11: Shelving FilterChapter 12: Cascaded Low-Pass FilterChapter 13: Cascaded High-Pass FilterChapter 14: Cascaded Bandpass FilterChapter 15: Cascaded Band-Stop FilterSECTION II: FILTER PROJECTSChapter 16: IntroductionChapter 17: Tone ControlChapter 18: 60 Hz Hum EliminatorChapter 19: 31-Band Graphic EQ - IChapter 20: 31-Band Graphic EQ - IIChapter 21: 4-Band Parametric EQChapter 22: Digital CrossoverAPPENDIX: Odd-Order FiltersThe code for all of the filters and projects is implemented in C++ and is very easy to understand. There isn't much here in the way of theory, so it would be best if the reader already knew the theory behind DSP and was just coming to this book for the implementations/examples missing in other books. The source code for all of the examples is available at the book's website. Since kicks out reviews with website addresses, suffice it to say all you need to do is type "dspaudiocookbook" into Google and the first address you see will be the book's website complete with example code.0 of 0 people found the following review helpful. easy to code based on the bookBy W BoudvilleLane et al start off with a quick review of the general case of DSP filters and the use of transfer functions to model them. This leads naturally into a bifurcation, yielding Finite Impulse Response and Infinite Impulse Response filters.The theory is presented in concise fashion. And with no problem sets in the chapters. Seems more like a summary of material you perhaps should have learnt in a more leisurely text. Example codes are given in C++. Short subroutines. There is very little need or usage of the object oriented nature of C++. If you need to code from the formulae in the book, the example codes can be used to get you started. But for most engineers, this may not even be necessary. A modicum of acquaintance with whatever language you write in should suffice.2 of 3 people found the following review helpful. Great practical DSP IIR filter design bookBy B. BoltonProbably the best practical how-to books on IIR filters I've found. There is no problem finding theory books. It took a long time to find this one, and I recommend it to anyone who wants to go from the theory to functional difference equations.In a professional environment, you still want to test for stability, impulse response, quantization effects, etc. But, this book will give a great starting point for simple results. If you want to go to the next level you'll design IIR FIR filters in MatLab or the likes. This book however gives a functional starting point to get results.

Digital filters and real-time processing of digital signals have traditionally been beyond the reach of most, due partially to hardware cost as well as complexity of design. In recent years, low-cost digital signal processor (DSP) development boards have put this within reach. This book will break down this design complexity barrier by means of simplified tutorials, step-by-step instructions, along with a collection of audio projects. Design formulas are presented to build the digital equivalent of standard audio filters: lowpass, highpass, and bandpass.

From the Back CoverFinally, low-cost digital signal processor (DSP) development boards have been introduced in a price range that fits the budget constraints of students, engineers, and hobbyists. Digital filters and real-time processing of digital signals have traditionally been beyond the reach of most hobbyists, due to hardware cost and complexity of design. Design complexity continues to be a major hurdle, but in DSP Filters this technology is clearly explained. DSP Filters breaks down the design complexity barrier through the use of simplified tutorials and step-by-step instructions, along with a collection of audio projects. The authors describe digital filter design by presenting the design formulas needed to build the digital equivalent of standard audio filters - low-pass, high-pass, bandpass, and band-stop - and also include the more specialized peaking and shelving filters. In Section 1, the design and analysis formulas of both analog and the digital equivalents of 14 specific filter types are described. Section 2 presents five audio projects and provides implementation examples of the various filter types using the C++ programming language, which further supports the mathematical descriptions. Projects were chosen for potential interest, with special

consideration of the usefulness in illustrating the design and implementation of concepts presented throughout the chapters of Section I.