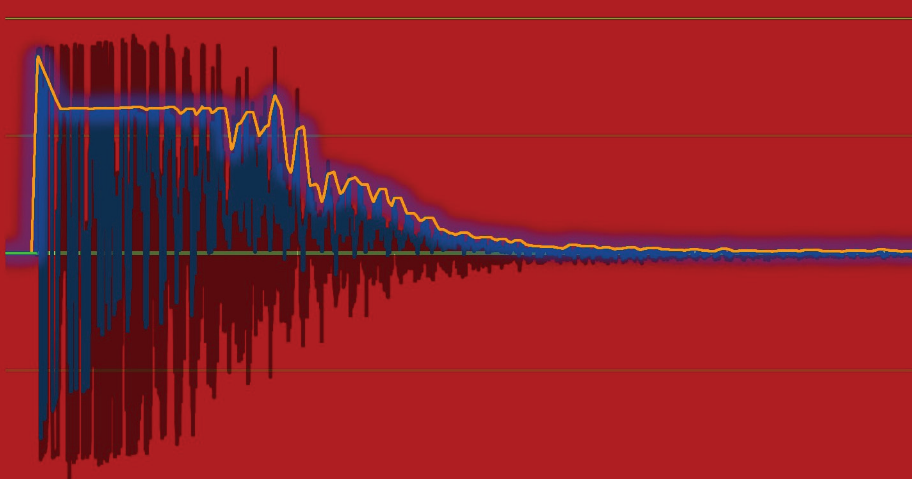
Digital Signal Processing for Audio Applications

Volume 2 - Code
Third Edition

Anton Kamenov



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Foreword to the First Edition of Digital Signal Processing for Audio Applications

In the summer of 2003 we began designing multi-track recording and mixing software – Orinj at RecordingBlogs.com – a software application that will take digitally recorded audio tracks and will mix them into a complete song with all the needed audio production effects. Manipulating digital sound, as it turned out, was not easy. We had to find the answers of many questions, including what digital audio was, how we could mix audio tracks, how we could track the amplitude of digital sound so that we could apply compression, how we could track frequencies so that we could equalize, what a good model of artificial reverb would be, and many others. Bits of relevant information were available, albeit not always well organized and not always intuitive.

"Digital Signal Processing for Audio Applications" provides much of the needed information. It is a simple structured approach to understanding how digitally recorded sound can be manipulated. It presents and explains, and sometimes derives, the mathematical theory that the DSP user can employ in designing sound manipulating applications.

Although this book introduces much mathematics, we have designed it not for mathematicians, but for the engineers and hobbyists, who would be interested in the practical applications of DSP and not in its theoretical derivations. If properly explained, much of the practical DSP applications reduce to simple algebra. This said, we have included a sufficient amount of theory to provide an explanation of why DSP works the way it does. It is important for practitioners to have a good understanding of how DSP concepts come about. Much of the available DSP information has too much theory and not enough examples. Much of it has too many practical examples and not enough theoretical backing. We hope to have found the proper balance.

We hope you enjoy this book and make use of its definitions, explanations, and numerous examples.

The author and the administrators of www.recordingblogs.com

Foreword to The Code

Explaining the mathematics behind digital signal processing – DSP – is the task volume 1. It is a start, but there is more. It is not always straightforward to translate the mathematics into code. The purpose of volume 2 is just that. It translates the mathematical formulae in volume 1 into practical algorithms. It does so with actual DSP effects, including distortion, delay, chorus, equalizer, compressor, reverb, wah wah, and others.

Volume 1 of this book makes the argument that much of DSP can be reduced to simple algebraic and trigonometric manipulations. We hope that this volume shows that coding DSP is similarly not complex. In contemporary audio recording and mixing software, storing audio data, managing audio files, and designing an intuitive but functional user interface could be much more intricate than modifying the audio data themselves.

We hope you make use of this book and design some of your own DSP effects. They may just sound better than anyone else's. Audio production is as much an art, as it is science.

Many thanks to Mic of RecordingBlogs.com for providing access to the Orinj source code.

The author and the administrators of www.recordingblogs.com

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