

## <<数字信号处理>>

### 图书基本信息

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### 内容概要

本书介绍了MATLAB在数字信号处理中的应用，包括时分信号与系统、时分傅里叶变换、 $z$ 变换、离散傅里叶变换、离散时间滤波器实现、FIR滤波器设计、IIR滤波器设计以及采样率转换等内容。本书通过使用MATLAB这一“动态实验室”帮助读者提高解决问题的能力 and 严谨思维的能力。

《数字信号处理--应用MATLAB(第3版英文影印版)》可作为电子工程、通信专业的双语教材或教学参考书，也可供工程技术人员参考。

## <<数字信号处理>>

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## 章节摘录

版权页：插图： These applications and products require many interconnected complex steps, such as collection, processing, transmission, analysis, audio/display of real-world information in near real time. DSP technology has made it possible to incorporate these steps into devices that are innovative, affordable, and of high quality (for example, iPhone from Apple, Inc.). A typical application to music is now considered as a motivation for the study of DSP. Musical sound processing In the music industry, almost all musical products (songs, albums, etc.) are produced in basically two stages. First, the sound from an individual instrument or performer is recorded in an acoustically inert studio on a single track of a multitrack recording device. Then, stored signals from each track are digitally processed by the sound engineer by adding special effects and combined into a stereo recording, which is then made available either on a CD or as an audio file. The audio effects are artificially generated using various signal processing techniques. These effects include echo generation, reverberation (concert hall effect), flanging (in which audio playback is slowed down by placing DJ's thumb on the flange of the feed reel), chorus effect (when several musicians play the same instrument with small changes in amplitudes and delays), and phasing (aka phase shifting, in which an audio effect takes advantage of how sound waves interact with each other when they are out of phase). These effects are now generated using digital-signal-processing techniques. We now discuss a few of these sound effects in some detail. Echo Generation The most basic of all audio effects is that of time delay, or echoes. It is used as the building block of more complicated effects such as reverb or flanging. In a listening space such as a room, sound waves arriving at our ears consist of direct sound from the source as well as reflected off the walls, arriving with different amounts of attenuation and delays. Echoes are delayed signals, and as such are generated using delay units. For example, the combination of the direct sound represented by discrete signal  $y(n)$  and a single echo appearing  $D$  samples later (which is related to delay in seconds) can be generated by the equation of the form (called a difference equation)

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