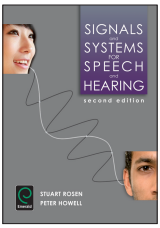

Signals and Systems for Speech and Hearing (Second Edition)

Authors: Stuart Rosen and Peter Howell
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Synopsis

New for the 2nd edition

In the nearly 20 years since the book originally appeared, little has changed in the basic concepts. (Fourier's theorem is still as valid as it was when first proposed at the beginning of the nineteenth century!) What has changed the field immeasurably is the nearly complete replacement of analogue for digital techniques in the recording, manipulation, storage and transmission of signals. This has been reflected by changes throughout the book in the kind of instrumentation described. Furthermore, two chapters have been heavily revised. Chapter 11, dealing with spectrograms, has been much extended and describes the two different ways in which spectrograms can be constructed – through filter banks and time windowing – and the relationship between them. Chapter 14, dealing explicitly with digital signals and systems, has been expanded greatly to give concrete examples of digital systems and digital signal processing, including the notion of infinite impulse response (IIR) and finite impulse response (FIR) filters. Finally, Chapter 12 now focuses on the notion of the auditory periphery as a set of systems, showing how its function is analogous to that of making a spectrogram.

Table of Contents

Chapter 1 – Introduction – The relevance of studying the properties of signals and systems for work in the speech and hearing sciences; examples of signals and systems; input and output signals.

Chapter 2 – Signals in the real world – The similarity of acoustic, mechanical and electrical signals; transduction.

Chapter 3 – Introduction to signals – The specification and construction of sinusoids; the concepts of frequency, period, amplitude and phase; periodic and aperiodic sounds; measures of amplitude – peak-to-peak, rms; dB scales.

Chapter 4 – Introduction to systems – Linearity, additivity, homogeneity and time-invariance. Saturating non-linearities. Application to middle ear and basilar membrane vibration, and to audio recording.

Chapter 5 – A preview – If we know what a linear time-invariant system does to sinusoids, we know what it will do to any signal.

Chapter 6 – The frequency response of systems – Amplitude responses as the ratio of output levels to input levels at particular frequencies. Low-pass, high-pass, bandpass and band-stop filters. The use of logarithmic axes (dB vs log Hz). The amplitude response of a cascade of systems. Application to middle ear vibrations and vocal tracts of varying shapes. Formants as resonances. Phase responses, linear and otherwise.

Chapter 7 – The frequency characterization of signals – Fourier analysis and synthesis. The amplitude and phase spectra of periodic signals: sinusoids, sawtooths, square waves, triangle waves and pulse trains. The spectra of aperiodic signals: transients and noise.

Chapter 8 – Signals through systems – Determining the output of systems to specified signals in the frequency domain, for both amplitude and phase. A sawtooth through ideal and realistic low-pass filters. Noise through a filter. Distortion.

Chapter 9 – The time characterization of systems – The notion of an impulse, and the impulse response. The relationship between the frequency response of a system and its impulse response. Determining the frequency response of a set of headphones in three ways: via the impulse response, a sinusoid swept in frequency, and white noise.

Chapter 10 – The relationship between the time and frequency domains – Signals short in time tend to be wide in spectrum, and vice versa. The trade-off between temporal resolution and frequency resolution in band-pass filters. The relationship between a system's impulse response and its frequency response.

Chapter 11 – Spectrograms: practical short-term spectral analysis – Determining dynamic spectral changes; the notion of a filter bank; rectification and smoothing; short-term spectra; the use of wide- and narrow-band filters; making spectrograms in the time domain; windowing.

Chapter 12 – Applications to hearing – The measurement of frequency responses, and notions of linearity and non-linearity in the peripheral auditory system: head and pinna, the ear canal resonance, middle ear vibration, basilar membrane motion; modelling the auditory periphery as a set of systems.

Chapter 13 – Applications to speech production – The source-filter theory of speech production and its application to vowels, diphthongs and fricatives.

Chapter 14 – Digital signals and systems – sampling and quantization; digital-to-analog and analog-to-digital conversion; aliasing; simple digital systems; infinite impulse response (IIR) and finite impulse response (FIR) filters.