Introduction to Audio Content Analysis

Module 2.4: Fundamentals — Blocking

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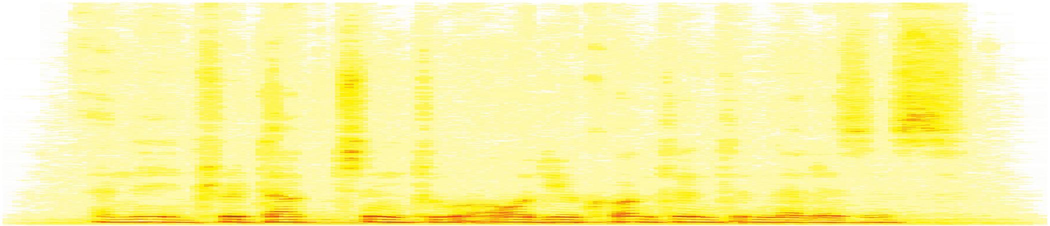

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introduction

overview

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corresponding textbook section

Chapter 2 — Fundamentals: pp. 18–20

- **lecture content**
  - splitting the audio signal into blocks
  - block length and hop size

- **learning objectives**
  - describe the reasons for blocking
  - summarize the principle using the correct terminology

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block based processing
introduction

- typical audio applications **process blocks** of audio data
- instead of having a function called per sample, it is called with block of samples

- **reasons**:
  - block based processing methods such as the Short-Time Fourier Transform
  - quasi-stationary signal properties within block
  - audio hardware characteristics (real-time systems)
  - efficiency (memory allocation, SIMD)

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  - 1. . . thousands of samples
  - often powers of 2

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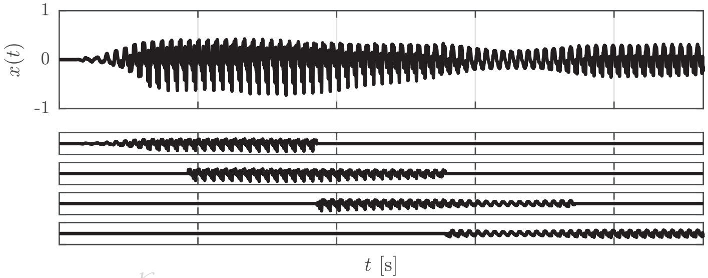

- $\mathcal{K}$: block length
- $\mathcal{H}$: hop size
- n: block index
- i: sample index

matlab source: matlab/displayBlocking.m

block based processing

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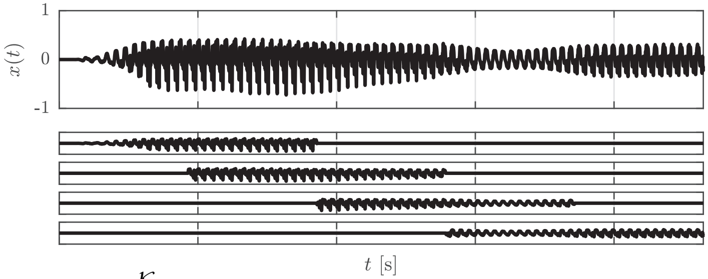

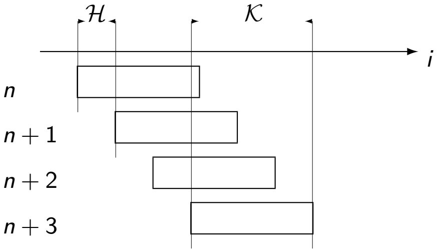

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block based processing

terms and definitions

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- **block boundaries**:

$$\begin{aligned} i_\mathrm{s}(n) &= i_\mathrm{s}(n-1)+\mathcal{H} \\ i_\mathrm{e}(n) &= i_\mathrm{s}(n)+\mathcal{K}-1 \end{aligned}$$

- **overlap ratio**:

$$o_\mathrm{r} = \frac{\mathcal{K}-\mathcal{H}}{\mathcal{K}}$$

- **time stamp**:

$$t_\mathrm{s}(n) = \frac{i_\mathrm{e}(n)-i_\mathrm{s}(n)+1}{2\cdot f_\mathrm{S}} + \frac{i_\mathrm{s}(n)}{f_\mathrm{S}} = \frac{\mathcal{K}}{2\cdot f_\mathrm{S}} + \frac{i_\mathrm{s}(n)}{f_\mathrm{S}}$$

- $\mathcal{K}$: block length
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- n: block index
- i: sample index
- f_S: sample rate

summary
lecture content

- **audio signal is typically split into blocks**
- each block processed individually
- **terms**:
    - *block length*:
        - minimum: 1
        - typical: 256 . . . 16384
    - *hop size*:
        - minimum: 1
        - maximum: block length
        - typical: half of block length
    - block *time stamp*:
        - typically refers to middle of block