
Audio Coding - Practice Lessons

Seminar 2

Polyphase MDCT Filterbank

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General information

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Website:

<http://www.tu-ilmenau.de/mt/lehrveranstaltungen/lehre-fuer-master-mt/audio-coding/>

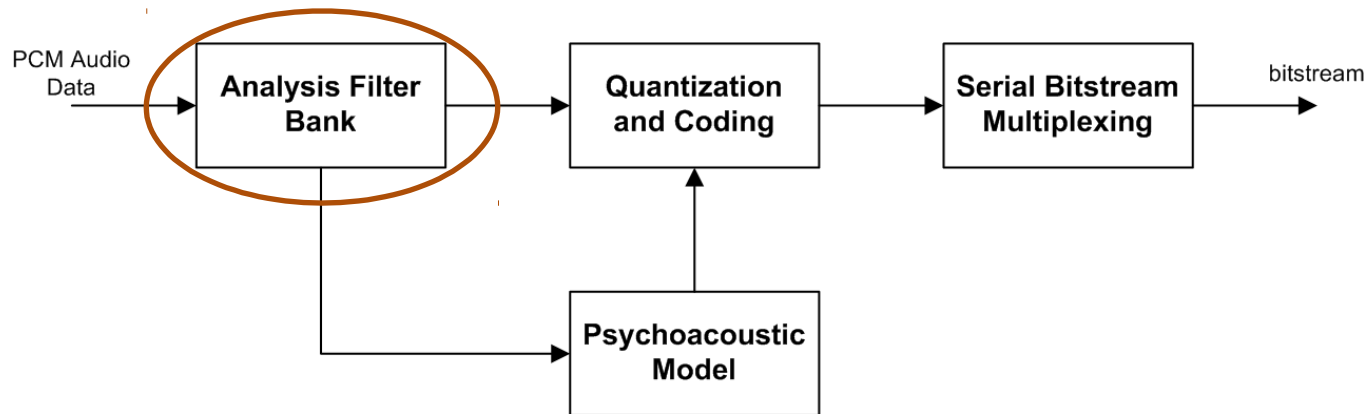
Moodle: moodle2.tu-ilmenau.de

- Check for updates
- News (schedule changes, etc.)
- Homework tasks
- Lecture slides

Homework Assignment 2

Goal:

- Subband analysis and synthesis of an audio signal
→ achieve perfect reconstruction (recover input signal perfectly, but with a delay)
- How to achieve that:
Implement the MDCT via polyphase description
→ Lecture: FilterBanks 1



Homework Assignment 2

Task 1:

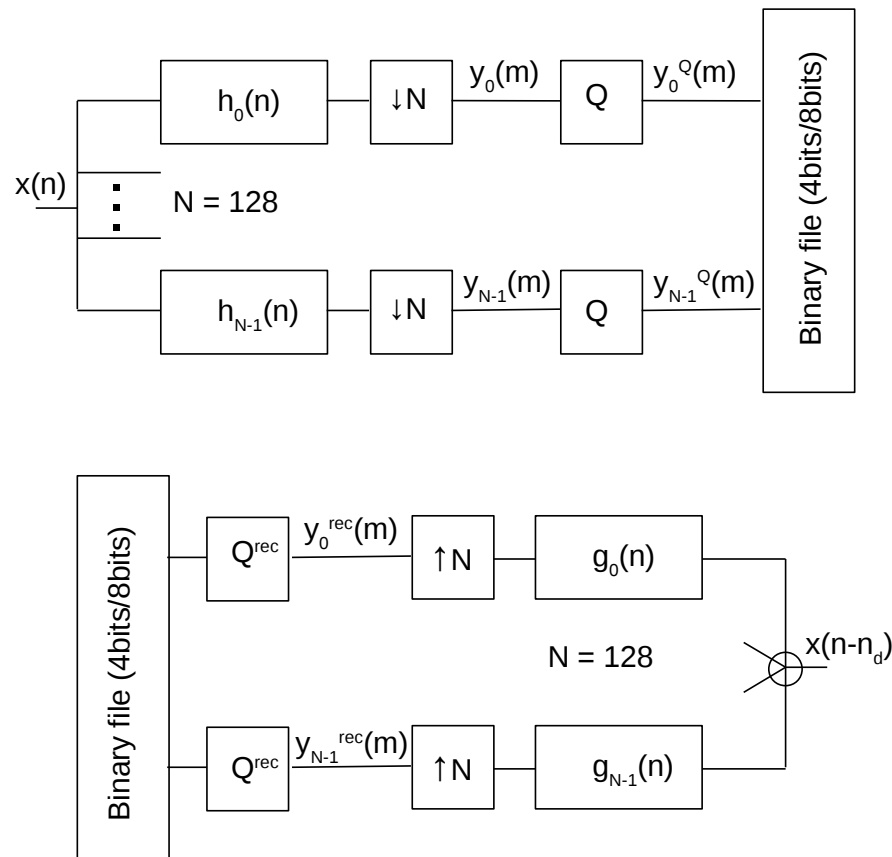
- Use the **direct implementation of the MDCT** analysis and synthesis filter bank with $N=128$ subbands, using its definition of the impulse response and downsamplers after the analysis filters and upsamplers before the synthesis filter bank
- Hint 1:
Have a look at the lectures *Basics of Multirate Signal Processing*, *FilterBanks 1 & 2* and the lecture slides of *Multirate Signal Processing*, if necessary
- Hint 2 – MDCT: „modulated filter“ is described by following function
$$h_k(L-1-n) = h(n) \cdot \cos\left(\frac{\pi}{N} \cdot \left(k + \frac{1}{2}\right) \left(n + \frac{1}{2} - \frac{N}{2}\right)\right),$$
 where „window“ function
$$h(n) \text{ is } h(n) = \sin\left(\frac{\pi}{2N}(n+0.5)\right), \text{ for } n=0, \dots, 2N-1 \text{ (see also: lecture 3).}$$

Homework Assignment 2

Task 2:

- Use your implementation from the first seminar to quantize each subband. Define your quantization steps in order to match the amplitude range of your subband.
- Quantize your subband with both 8bit and 4bit

Homework Assignment 2



Homework Assignment 2

Task 3:

- Test perfect reconstruction with a ramp function
→ is it reconstructed after the synthesis filter bank?
- Compare 8bit and 4bit quantization with original 16bit quantization

Task 4:

- Test the filter impulse responses by inputting a 1 followed by zeros as input to one synthesis filter in the synthesis filter bank
- Does it look okay?
- Check its frequency response with `freqz()`
- Again: compare quantizations