Exercises for **Audio and Video Signal Processing**: FH OÖ Hagenberg 2011
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Exercise 3: **Filter Bank, Psychoacoustic and Audio compression**

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1. Multi-Complementary Filter Bank

a) What is an octave-spaced frequency splitting? How can we design a filter bank for that task?

b) How can we perform an aliasing free subband processing?

c) How can we achieve narrow transition bands for a filter bank?

d) What is the computational complexity of an octave-spaced filter bank?

2. Modified Octave Filter Bank in MATLAB

a) Implement a modified octave filter bank (analysis and synthesis part) according to Figures 5.63 and 5.64 in the book "Digitale Audiosignalverarbeitung". Realize the lowpass and highpass filters directly without complementary technics.

b) Replace the lowpass and highpass filters in the analysis part by filters in complementary technics according to Figure 5.67 in the book "Digitale Audiosignalverarbeitung". The "Kernfilter" should be realized directly without complementary technics. How do the delays D_{Hx} have to be chosen?

c) Realize also the "Kernfilter" in complementary technics according to Figure 5.67 in the book "Digitale Audiosignalverarbeitung".

3. Psychoacoustic

a) Human hearing and voice

- **(i)** In which frequency range is human being perceiving sounds?
- **(ii)** What is the frequency range of the speech?
- **(iii)** In the above specified range where is the human hearing the most sensitive?
- **(iv)** How do we call low and high frequencies in case of normal voice?
- **(v)** Use the equation of absolute threshold hearing and plot it with Matlab for $f_S = 44.1$kHz. Explain in few words how this threshold has been obtained?

b) Frequency masking

- **(i)** What is frequency masking?
- **(ii)** How can we explain the temporal masking and what is its duration after stopping the active masker?
- **(iii)** What is a critical band and why is it needed for frequency masking phenomena?

(iv) Consider a partial signal at index P. a_P and f_P are the amplitude and frequency respectively. $V(a_P)$ is the corresponding volume in dB. The difference between the level of the masker and the masking threshold M is $\Delta = -10$dB. The masking curves towards lower and higher frequency are described by left slope (27 dB/Bark) and right slope (15 dB/Bark) respectively. Explain the main steps of the frequency masking in this case and show with plots how this masking phenomena is achieved.

4. Audio compression

a) Since compression could be a synonym of reduction of data size, explain the basic approaches of compression?

b) What are the two kinds of data compression and how do they work?

c) What is MPEG and which algorithm does it use for audio compression?

d) Which MPEG layers do you know and what are their principles?

e) A 16 bit Nyquist-rate sampled signal of size 4MB has to be compressed with MP3 at 128kb/s. Make a statement about the compression rate.