Exercises for **Audio and Video Signal Processing**: FH OÖ Hagenberg 2011
Dr.-Ing. Oomke Weikert
Exercise 2: **Oversampling, Delta-Sigma Conversion, Fast Convolution**

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1. Oversampling

a) How do we define the power spectral density PSD $S_{XX}(e^{j\Omega})$ of a signal $x(n)$?

b) What is the relationship between signal power σ_X^2 (variance) and power spectral density $S_{XX}(e^{j\Omega})$?

c) Why do we need to oversample a time-domain signal?

d) Explain why an oversampled PCM A/D converter has lower quantization noise power in the baseband than Nyquist-rate sampled PCM A/D converter.

e) How do we perform oversampling by a factor of L in the time domain?

f) Explain the frequency domain interpretation of the oversampling operation.

g) What is the passband and stopband frequency of the analog anti-aliasing filter?

h) What is the passband and stopband frequency of the digital anti-aliasing filter before downsampling?

i) How is the downsampling operation performed (time-domain and frequency-domain explanation)?

2. Delta-Sigma Conversion

a) Why can we apply noise shaping in an oversampled AD converter?

b) Show how the delta-sigma AD converter (DSC) has a lower quantization error power in the useful audio band than an oversampled PCM A/D converter ?

c) How do the power spectral density and variance of the total quantization error in the useful audio band change related to the order of DSC?

d) How is noise shaping achieved in an oversampled delta-sigma AD converter?

e) With Matlab plots show the noise shaping effect of the delta-sigma modulator and how the improvement of the SNR for pure oversampling and delta-sigma modulator is achieved.

f) Using the previous plots specify which order and oversampling factor L will be needed for our 1-bit delta-sigma converter if we need a gain of 100 dB.

g) What is the difference between the delta-sigma modulator in the delta-sigma AD converter and the delta-sigma DA converter?

h) How do we achieve a w-bit signal representation at Nyquist sampling frequency from an oversampled 1-bit signal?

i) Why do we need to oversample a w-bit signal for a 1-bit delta-sigma DA converter?

3. Fast Convolution

a) What is the length of an output signal $y(n)$ after convolution between an input signal $x(n)$ of length N_1 and an impulse response $h(n)$ of length N_2?

b) How can we perform IDFT using the DFT algorithm?

c) How can we perform DFT of two real sequences?

d) Explain the main steps of fast convolution if the spectral functions $Z(k)$ and $H(k)$ are known?

e) How do we perform fast convolution of long sequences?

4. Fast Convolution with Matlab

For an input sequence $x(n)$ of length $N_x = 500$ and an impulse response $h(n)$ of length $N_h = 31$, perform the discrete time convolution.

a) Give the discrete time convolution sum formula and the corresponding signal flow graph.

b) Define in Matlab $x(n)$ as sum of two sinusoids and derive $h(n)$ using the Matlab function "remez".

c) Realize the filter operation with Matlab using:

- The Matlab function "conv"
- The sample by sample summation method
- The FFT method
- The FFT with overlap-add method

5. Filter Design by Frequency Sampling

a) Give the general form of frequency response for a linear phase system.

b) Prove the linearity of the phase and derive the group delay.

c) How do we perform frequency sampling?