Audio- / Videosignalverarbeitung
Advanced Digital Signal Processing
Digital Signal Processing 2

Seminar 4

WS 2014/2015

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Task

1. Upsample the speech signal by the factor of N = 4, using **Noble identities (polyphase decomposition)**
   - Hint: Use the same .wav file as in the Homework 3

2. Design filter for the anti-alias-filtering
   - FIR with 32 filter coefficients
   - Use: Parks-McClellan-Algorithm (remez filter design function)
   - Plot impulse and frequency response

Task

- Reasonable filter design, i.e. consider:
  - passband, stopband, transition band
  - stopband attenuation
  - weights
  - normalization of frequency
  - stopband should start where aliasing components appear

3. Listen to and compare signals before and after upsampling and filtering

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Task

4. Design a frequency warped filter with the following parameters:
   - Sampling frequency: 44.1 kHz
   - Cutoff frequency: 0.15*pi
   - No. of filter coefficients: 6

5. Plot the filter's:
   a) impulse response
   b) frequency response
   c) z-plane (poles,zeroes)

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Task

6. Implement the minimum phase version of a linear phase filter
   - Create a FIR filter with the help of *remez()* function with the passband of 0,25
   - Make the minimum phase version out of it

7. Plot the minimum phase filter‘s:

   a) impulse response

   b) frequency response

   c) z-plane