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

Seminar 3

WS 2014/2015

Homework assignment

1. Download and use Seminar3_template.m from the moodle in unit Seminar 3
2. Upsample the given speech signal (8kHz) to 32 kHz
   - use the .wav-file which is uploaded at moodle in unit „Seminar 4“

   a) Upsampling by the factor 4

      Hint: insert 3 zeros after each sample

   b) Plot the spectra of the original and upsampled signal and compare

      Hint: use freqz(…) for creating the spectrum

   c) Listen to both signals

      Hint: use wavwrite(...)

Homework assignment

3. FIR lowpass filtering
   a) Implement the FIR lowpass filter with the following difference equation:

$$y(n) = 0.3235 * x(n) + 0.2665 * x(n-1) + 0.2940 * x(n-2) + 0.2655 * x(n-3) + 0.3235 * x(n-4)$$

   b) Plot the impulse response (first 50 samples)
   c) Plot the frequency response
   d) Filter the upsampled speech signal with the given FIR filter

Hint: use filter(...)

Homework assignment

3. IIR lowpass filtering

   a) Implement the IIR lowpass filter with the following difference equation:

$$y(n) = 0.256 * x(n) + 0.0512 * x(n-1) + 0.256 * x(n-2) + \\ (\ 1.3547 * y(n-1) - 0.6125 * y(n-2))$$

   b) Plot the impulse response (first 50 samples)

   c) Plot the frequency response

   d) Filter the upsampled speech signal with the given IIR filter

Hint: use filter(...)

Homework assignment

4. Downsampling
   a) Filter the given speech signal (8kHz) with the given IIR filter
   b) Downsample the signal by the factor 2

      Hint: means taking every second speech sample
   c) Plot the spectra of the original and downsampled signal and compare

      Hint: use freqz(…) for creating the spectrum
   d) Listen to both signals

      Hint: use wavwrite(...)

Homework assignment

5. Noble Identities
   a) Reverse the order of upsampling and filtering according to the Noble Identities (only for FIR case)
   b) Compare the resulting signals (plot and listen)

6. Compare FIR and IIR lowpass filters:
   - Transfer functions
   - Signals after upsampling and filtering
   - Listen to signals before and after lowpass filtering (aliasing)
   - Make your own conclusions about comparison
   - Which filter is better? Why?