**Department of Communications and Computer Engineering**

**CCE5201 Advanced Digital Signal Processing – Assignment 2011-2012**

## Section A. Wavelet Analysis

Start with a Daubecheis wavelet with 20 coefficients. (Matlab should have this in toolbox). Set up the highpass and lowpass sides of a multiresolution analysis. Initially use speech clips available on my website. Iterate through the lowpass side.
Use approximately 10 seconds of material, with around 20 - 30 ms per frame and one of the overlap techniques to get the composite picture at each level. (again you can use any appropriate MATLAB function directly).
Stick in some noise at a signal-to-noise ratio of about 4 db.s in sections of 3 seconds in 2 parts of the original waveform used. Repeat the above MRA and look at the composite output in time and frequency.
The filters are to be used as lowpass filters. So before beginning reconstruction, retain only the LP coefficients of the last LP stage, and set all other coefficients to 0. This way obtain a successive time and frequency picture of the original waveform, at the different subband levels.

## Section B Using the DCT Transform

Download the lena.bmp monochrome file 512x512.

(a) In Matlab use *imread* to port the bmp file to a 512 x 512 numeric array.
(b) Subdivide the 512 x 512 array into subarrays of 16 x 16 in a proper way. There should be 1024 subarrays.
(c) Perform a DCT on each 16x16 subarray.
(d) Remove from the DCT any element whose magnitude is less than 3, substituting it with 0.
(e) Estimate the number of zeroes inserted as a percentage of the total elements, averaging over the whole picture.
(f) Use the inverse dct, IDCT, on the approximated DCT arrays to recompose the submatrices, and eventually the overall picture array.
(g) Restore picture with *imwrite* to a .bmp file.

(h) Repeat (c) to (g) LEAVING ONLY the first row from each submatrix, putting every other row element to 0.
Discuss the results obtained

## Section C - Use of Polyphase Filters, using Audio

Design a lowpass filter design with cutoff at $\pi/8$ with an attenuation of 40db at $\pi/6$. Use the material in Chapter 10 or directly from Matlab. Check out by trying the filter on a composite sine wave with frequencies at $\pi/20$, $\pi/6$ and $2\pi/3$. Assume a sampling rate of 40,000 samples per second or thereabout (ie also 44,100 if you use software that automatically samples at this frequency).

Hence use an audio waveform and low-pass filter it. Downsample the output by a factor of 4, and a factor of 8, and compare the direct and the downsampled waveforms.

Section D. Polyphase Structures

Using the filter in (a) above downsample the original 40,000 sample per second waveform to 5,000 samples per second. If necessary alter slightly the number of original nonzero coefficients, M, to make it divisible by D, the decimation factor. (Ideally this should be taken care of in section C). If the sampling rate is different then alter the downsampled frequency to $F_S/8$.

Hence implement the decimation using a direct architecture (Fig 11.5.9a from textbook) and repeating with the embedded decimation, Fig 11.5.9b from textbook). Again as input use the original composite sinusoid sampled at 40,000 samples per second. The output should be the filtered waveform now operating (sampled) at 8,000 samples per second. Compare the output obtained from the polyphase filter with the corresponding output obtained from C.