PART B

EEE509: DSP ALGORITHMS & SOFTWARE

FINAL - Due Friday midnight 6/26 (AZ time)

READ BEFORE YOU START

- Start each question on a separate page and label them with page numbers.
- Give your full name and total number of pages written on the first page of your test.
- Number all the pages and put them in sequence.
- Show all your work clearly and label figures, Fig. 1, 2 etc
- Be precise and give all the details.
- Circle your answers.
- Attach your MATLAB programs (when applicable) at the end and cite them from the pertinent problem.
- You can handwrite equations clearly and entire test (no need to type).
- SCAN YOUR TEST IN PDF AND UPLOAD. NAME THE File: (E509Sum20-FINAL - YOURNAME-DATE.PDF)

1. Design a custom FIR band-stop filter using the Fourier series and the Hamming window. The filter should be of order 8. We need to attenuate the signal within 1600-4000Hz Hz and attenuate it elsewhere

. The sampling frequency is 16 kHz.

a) Calculate with pencil and paper the impulse response of the filter and the numerical values of the coefficients. (work out in detail all the equations)

b) Plot the impulse response of the filter using MATLAB

c) Determine the group delay.

d) Compare against a similar filter with the exception that it uses a rectangular window. Show in MATLAB plots of the frequency response magnitude and overlay the two designs on one plot (give magnitude in dB)

(10 pts)

Please make sure you use figure captions

2. The following causal system is excited by white noise ($x(n)=w(n)$) of zero mean and unit variance. The output is $y(n)$.

$$q(n)=x(n) - 0.9\, q(n-1)$$
$$y(n)=0.1\, q(n)$$

a) Determine the autocorrelation of the output $y(n)$ in closed form for all m. Give numerical values for $r_{yy}(0)$, $r_{yy}(1)$, $r_{yy}(2)$. Show your work

b) Find the variance of $y(n)$. Give a numerical value and show all your work (use pencil and paper).

c) Find the poles and zeros of the power spectral density (PSD) of $y(n)$ and sketch them carefully on the z-plane.

d) Use MATLAB (on this part only) and plot the PSD in dB.

(10 pts)

3. You are to design and implement in MATLAB an FFT-based system for signal compression or speech coding. Use the provided clean speech file. The system should process speech data frame-by-frame with 256-sample frames. The system should select the first M components from the FFT which will be implemented for N=256. You need to select the first M components of the FFT and their conjugate symmetric FFT components such that a real valued signal can be reconstructed. For M=8, 32, 128reconstruct and assess the entire reconstructed speech record. After you reconstruct the speech record you need to evaluate the *signal to noise ratio* in dB. (16 pts)

Use the file "New_clean_male.wav" to define s(n). The SNR can be calculated using:

SNR_{dB}=10 $\log_{10}$ [SUM($(s(n))^2$) / SUM($s(n) - s'(n)$ $)^2$)]

where SUM is the sum across **all** speech samples. s(n) input speech, s'(n) reconstruction speech

After you form and run the program, fill in the following table:

M	SNR (dB)	Execution Time FFT	Execution Time DFT
8			
32			
128			

The execution time will have to be computed for the entire program after processing all frames with the FFT. Then repeat with the DFT (you will have to develop a DFT/IDFT with nested loops and use it instead of the FFT and IFFT).

Deliverables on this problem:

a) Report dB values for SNR and fill up the table.
b) Develop also MATLAB code based on a loop for the DFT and report CPU time
c) Give remarks (address them as questions)
   How and why SNR changes
   How does CPU change from FFT and DFT.
d) Give your program for the compression system.
e) Give the program for the DFT and IDFT

(Hint: see also example in Ch. 7)
Frame by frame processing was covered in project 1. You can use in this exercise as well.