# Digital Signal Processing I ECE538 <br> Final Exam Fall 2008 <br> 17 Dec.. 2008 

## Cover Sheet

Test Duration: 120 minutes.
Open Book but Closed Notes.
Calculators NOT allowed.
This test contains FIVE problems.
All work should be done on blank $8.5 " \times 11 "$ white sheets of paper (NOT provided).
Do not return this test sheet, just return your answer sheets.

## Digital Signal Processing I ECE538 DSP I

Final Exam
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Problem 1. [20 pts]
You are given the matrix inversion lemma:

$$
\left[\begin{array}{ll}
A & B \\
C & D
\end{array}\right]^{-1}=\left[\begin{array}{cc}
A^{-1}+A^{-1} B\left(D-C A^{-1} B\right)^{-1} C A^{-1} & -A^{-1} B\left(D-C A^{-1} B\right)^{-1} \\
-\left(D-C A^{-1} B\right)^{-1} C A^{-1} & \left(D-C A^{-1} B\right)^{-1}
\end{array}\right]
$$

A symmetric-Toeplitz matrix is formed from the first three autocorrelation values of an AR process as $r_{x x}[0]=1 / 3, r_{x x}[1]=1 / 6$, and $r_{x x}[2]=1 / 12$ as

$$
\mathbf{R}_{3}=\left[\begin{array}{ccc}
1 / 3 & 1 / 6 & 1 / 12 \\
1 / 6 & 1 / 3 & 1 / 6 \\
1 / 12 & 1 / 6 & 1 / 3
\end{array}\right]
$$

(a) Use the matrix inversion lemma to compute the inverse of $\mathbf{R}_{3}$ in terms of the inverse of

$$
\begin{aligned}
& \mathbf{R}_{2}=\left[\begin{array}{ll}
1 / 3 & 1 / 6 \\
1 / 6 & 1 / 3
\end{array}\right] \\
& \mathbf{R}_{2}^{-1}=\left[\begin{array}{cc}
4 & -2 \\
-2 & 4
\end{array}\right]
\end{aligned}
$$

(b) Use the matrix inversion lemma to compute the solution

$$
\mathbf{a}_{3}=-\mathbf{R}_{3}^{-1} \mathbf{r}_{3} \quad \text { where: } \mathbf{r}_{3}=\left[\begin{array}{c}
1 / 6 \\
1 / 12 \\
1 / 24
\end{array}\right]
$$

in terms of

$$
\mathbf{a}_{2}=-\mathbf{R}_{2}^{-1} \mathbf{r}_{2}=\left[\begin{array}{c}
-1 / 2 \\
0
\end{array}\right] \quad \text { where: } \mathbf{r}_{2}=\left[\begin{array}{c}
1 / 6 \\
1 / 12
\end{array}\right]
$$

(c) What is the model order $p$ of the AR process?

Problem 2. [20 points]
A signal $x[n]$ of length 12 is broken up into two overlapping blocks of length 8 , denoted $x_{1}[n]$ and $x_{2}[n]$, respectively, for the purposes of filtering with $h[n]=\{-1,2,-1\}$ of length $M=3$ via the overlap-SAVE method. The last $M-1=2$ points of $x_{1}[n]$ are the first two points of $x_{2}[n]$. The first $M-1=2$ zeros in $x_{1}[n]$ were added to get the overlap-save method started, as done in Fig. 7.3.1 in the text.

$$
x_{1}[n]=\{0,0,1,1,1,1,1,1\}
$$

and

$$
x_{2}[n]=\{1,1,-1,-1,-1,-1,-1,-1\}
$$

(a) $y_{1}[n]$ is formed by computing $X_{1}(k)$ as an 8 -pt DFT of $x_{1}[n], H(k)$ as an 8 -pt DFT of $h[n]$, and then $y_{1}[n]$ as the 8 -pt inverse DFT of $Y_{1}(k)=X_{1}(k) H(k)$. Write out the values of $y_{1}[n]$ in sequence form (similar to how $x_{1}[n]$ and $x_{2}[n]$ are written out above.)
(b) $y_{2}[n]$ is formed by computing $X_{2}(k)$ as an 8 -pt DFT of $x_{2}[n], H(k)$ as an 8 -pt DFT of $h[n]$, and then $y_{2}[n]$ as the 8 -pt inverse DFT of $Y_{2}(k)=X_{2}(k) H(k)$. Write out the 8 values of $y_{2}[n]$ in sequence form.
(c) Show how $y_{1}[n]$ and $y_{2}[n]$ are combined to form the full linear convolution $y[n]=$ $x[n] * h[n]$, via the overlap-save method.

Problem 3. [20 points]
A second-order digital filter is to be designed from an analog filter having two poles in the s-plane at $-0.2+0.4 j$ and $-0.2-0.4 j$ and two zeros at $j \sqrt{3}$ and $-j \sqrt{3}$, via the bilinear transformation method characterized by the mapping

$$
s=\frac{z-1}{z+1}
$$

Note that $-0.2+0.4 j=-(1 / 5)+j(2 / 5)$ and $j \sqrt{3}=j \tan (\pi / 3)$
(a) Is the resulting digital filter (BIBO) stable? Briefly explain why or why not.
(b) Denote the frequency response of the resulting digital filter as $H(\omega)$ (the DTFT of its impulse response). You are given that in the range $0<\omega<\pi$, there is only one value of $\omega$ for which $H(\omega)=0$. Determine that value of $\omega$.
(c) Draw a pole-zero diagram for the resulting digital filter. Give the exact locations of the poles and zeros of the digital filter in the z-plane.
(d) Plot the magnitude of the DTFT of the resulting digital filter, $|H(\omega)|$, over $-\pi<$ $\omega<\pi$. You are given that $H(0)=6$. Be sure to indicate any frequency for which $|H(\omega)|=0$. Also, specifically note the numerical value of $|H(\omega)|$ for $\omega=\frac{\pi}{2}$ and $\omega=\pi$.
(e) Determine the difference equation for the resulting digital filter.

Problem 4. [20 points]
(a) Consider a causal FIR filter of length $M=14$ with impulse response

$$
h[n]=u[n]-u[n-14]
$$

Provide a closed-form expression for the 16 -pt DFT of $h[n]$, denoted $H_{16}(k)$, as a function of $k$. Simplify as much as possible. You are given the following four values:

$$
H_{16}(0)=14 \quad H_{16}(4)=1-j \quad H_{16}(8)=0 \quad H_{16}(12)=1+j
$$

(b) Consider the sequence $x[n]$ of length $L=16$ below, equal to a sum of several finitelength sinewaves.

$$
x[n]=1+2 \cos \left(\frac{\pi}{2} n\right)+\cos (\pi n), \quad n=0,1, \ldots, 15 .
$$

$y_{16}[n]$ is formed by computing $X_{16}(k)$ as a 16 -pt DFT of $x[n], H_{16}(k)$ as a 16 -pt DFT of $h[n]$, and then $y_{16}[n]$ as the 16 -pt inverse DFT of $Y_{16}(k)=X_{16}(k) H_{16}(k)$. Express the result $y_{16}[n]$ as a weighted sum of finite-length sinewaves similar to how $x[n]$ is written above.
(c) For the remaining parts of this problem, $h[n]$ is now defined as the causal FIR filter of length $M=14$ below.

$$
h[n]=(-1)^{n}\{u[n]-u[n-14]\}
$$

Provide a closed-form expression for the 16 -pt DFT of $h[n]$, denoted $H_{16}(k)$, as a function of $k$. Simplify as much as possible.
(d) Consider again the sequence $x[n]$ of length $L=16$ below.

$$
x[n]=1+2 \cos \left(\frac{\pi}{2} n\right)+\cos (\pi n), \quad n=0,1, \ldots, 15 .
$$

$y_{16}[n]$ is formed by computing $X_{16}(k)$ as an 16-pt DFT of $x[n], H_{16}(k)$ as a $16-\mathrm{pt}$ DFT of $h[n]$, and then $y_{16}[n]$ as the 16 -pt inverse DFT of $Y_{16}(k)=X_{16}(k) H_{16}(k)$. Express the result $y_{16}[n]$ as a weighted sum of finite-length sinewaves.
(e) Consider the sequence $p[n]$ of length $L=16$ below.

$$
p[n]=4 \delta[n]+4 \delta[n-4]+4 \delta[n-8]+4 \delta[n-12]
$$

Is $p[n]=x[n]$ ? If they are equal, provide an explanation as to why they are equal.

Problem 5. [20 points]


Figure 1(a). Analysis Section of Two-Stage Tree-Structured Filter Bank.


Figure 1(b). Synthesis Section of Two-Stage Tree-Structured Filter Bank.


Figure 2(a). Analysis Filter Bank, $M=4$.


Figure 2(b). Synthesis Filter Bank, $M=4$.

This problem is about synthesizing an $M=3$ channel nonuniform $P R$ filter bank from a two stage tree-structured PR filter bank.

$$
\begin{gathered}
h_{L P}[n]=\frac{\sin \left(\frac{\pi}{2}(n+0.5)\right)}{\frac{\pi}{2}(n+0.5)} \\
h_{H P}[n]=(-1)^{n} h_{L P}[n]
\end{gathered}
$$

Also, $g_{L P}[n]=h_{L P}[n]$ and $g_{H P}[n]=-h_{H P}[n]$. The combination of the analysis filter pair, $\left\{H_{L P}(\omega), H_{H P}(\omega)\right\}$, and synthesis filter pair $\left\{G_{L P}(\omega), G_{H P}(\omega)\right\}$, form a two-channel PR filter bank.
(a) Use Noble's First Identity to determine the analysis filters $H_{m}(\omega), m=0,1,2$, so that the system in Fig. 2(a) is equivalent to the system in Fig. 1(a).
(i) Plot the frequency response for each of the three filters $H_{m}(\omega), m=0,1,2$. You can put all three plots on the same graph to save space.
(ii) Write a time-domain expression for EACH of the three filters $h_{m}[n], m=0,1,2$.
(b) Next, use Noble's Second Identity to express each synthesis filter, $G_{m}(\omega), m=0,1,2$, in terms of $G_{L P}(\omega)$ and $G_{H P}(\omega)$.
(c) Plot $F(\omega)=\sum_{i=0}^{2} H_{i}(\omega) G_{i}(\omega)$ over $-\pi<\omega<\pi$.

