# University of Illinois at Urbana-Champaign ECE 310: Digital Signal Processing 

PROBLEM SET 5: SOLUTIONS
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## Problem 1

To derive $x_{a}(t)\left(X_{a}(\Omega)\right)$ from $X_{d}(\omega)$, we first need to get rid of the repeated frequency component in $X_{d}(\omega)$. Thus, multiply an ideal LPF on both sides.

$$
\begin{aligned}
G_{a}(\Omega) & =\left\{\begin{array}{cc}
T & |\Omega|<\frac{\pi}{T} \\
0 & \text { else }
\end{array}\right. \\
\Rightarrow g_{a}(t) & =\operatorname{sinc}\left(\frac{\pi}{T} t\right)
\end{aligned}
$$

Therefore,

$$
\begin{array}{r}
\Rightarrow X_{a}\left(\frac{\omega}{T}\right)=X_{d}(\omega) G_{a}(\Omega) \\
\Rightarrow X_{a}(\Omega)=X_{d}(\omega) G_{a}(\Omega) \\
\Rightarrow x_{a}(t)=x(n T) * g_{a}(t) \\
\Rightarrow x_{a}(t)=\sum_{n=-\infty}^{\infty} x[n] \operatorname{sinc}\left(\frac{\pi}{T}(t-n T)\right)
\end{array}
$$

## Problem 2

(a) The Nyquist sampling rate is given by,

$$
\begin{aligned}
T_{\text {Nyquist }} & =\frac{1}{\left(12 \cdot 10^{3}\right)(2)}=0.000042 \mathrm{sec}=0.042 \mathrm{~ms} \\
\therefore \quad f_{\text {Nyquist }} & =\frac{1}{T_{\text {Nyquist }}}=24 \mathrm{kHz}
\end{aligned}
$$

(b)

$$
\begin{aligned}
\omega_{\max } & =T_{\text {Nyquist }} \cdot(2 \pi)(6000)=\frac{\pi}{2} \\
\omega_{\min } & =T_{\text {Nyquist }} \cdot(2 \pi)(300)=\frac{\pi}{40}
\end{aligned}
$$

The sketch of the frequency response of the discrete-time filter, when sampling at the Nyquist rate is shown in Fig. 1
(c) Some aliasing of the input signal is allowed with the condition that the minimum aliasing frequency is greater than the cutoff frequency of the filter.

$$
\begin{gathered}
2 \pi-2 \pi \cdot 12000 T_{\max 1} \geq 2 \pi \cdot 6000 T_{\max 1} \\
1 \geq 18000 T_{\max 1} \\
T_{\max 1}=\frac{1}{18000} \mathrm{sec}
\end{gathered}
$$

Also, the maximum frequency of $H_{d, 2}(\omega)$ should be less than or equal to $\pi$.

$$
\begin{gathered}
\omega_{\max }=T_{\max 2}(2 \pi)(6000) \leq \pi \\
T_{\max 2}=\frac{1}{12000} \mathrm{sec}
\end{gathered}
$$

Therefore, $T=\min \left(T_{\max 1}, T_{\max 2}\right)=T_{\max 1}=\frac{1}{18000} \mathrm{sec}$


Figure 1: Figure for Problem 2(b)
(d)

$$
\begin{aligned}
& \omega_{\max }=\frac{1}{18000} \cdot(2 \pi)(6000)=\frac{2 \pi}{3} \\
& \omega_{\min }=\frac{1}{18000} \cdot(2 \pi)(300)=\frac{\pi}{30}
\end{aligned}
$$

The sketch of the frequency response of the discrete-time filter, when sampling at the maximum rate is shown in Fig. 2


Figure 2: Figure for Problem 2(d)

## Problem 3

(a) The Nyquist rate is twice the highest frequency component, $f_{s} \geq 10 \mathrm{kHz}$.

Therefore, $T_{\max }=\frac{1}{10000} \mathrm{sec}$.
(b)

$$
\begin{aligned}
\omega & =\Omega T \\
\frac{\pi}{8} & =\frac{1}{10000} \Omega \\
\Omega & =2 \pi \cdot 625 \\
f & =625 \mathrm{~Hz}
\end{aligned}
$$

(c)

$$
\begin{aligned}
\omega & =\Omega T \\
\frac{\pi}{8} & =\frac{1}{20000} \Omega \\
\Omega & =2 \pi \cdot 1250 \\
f & =1250 \mathrm{~Hz}
\end{aligned}
$$

## Problem 4

(a) The sketches for $X_{d}(\omega), Y_{d}(\omega)$, and $Y_{a}(\Omega)$ are given below




Figure 3: Figure for Problem 4(a)
(b)

$$
\begin{aligned}
P_{a}(\Omega) & =\int_{0}^{T} 1 \cdot e^{-j \Omega t} d t \\
& =T \cdot e^{-j \frac{\Omega T}{2}} \operatorname{sinc}\left(\frac{\Omega T}{2}\right) \\
\therefore \quad Y_{a}(\Omega) & =T e^{-j \frac{\Omega T}{2}} \operatorname{sinc}\left(\frac{\Omega T}{2}\right) Y_{d}(\Omega T)
\end{aligned}
$$

The sketch of the magnitude of $Y_{a}(\Omega)$ is given in Fig. 4: The component of $Y_{a}(\Omega)$ for $|\Omega|>\frac{\pi}{T}=2000 \pi$ is due to the nonideal $\mathrm{D} / \mathrm{A}$. The highest amplitude of this unwanted component of $Y_{a}(\Omega)$ is at $\Omega=2500 \pi \mathrm{rad} / \mathrm{sec}$ and

$$
\left|Y_{a}(2500 \pi)\right|=Y_{d}\left(\frac{5 \pi}{4}\right) \cdot T \cdot \operatorname{sinc}\left(\frac{2500 \pi T}{2}\right)=0.2353
$$

as shown in the figure above.
Problem 5


Figure 4: Figure for Problem 4(b)
(a)

$$
\begin{array}{rcr} 
& X_{0}(\Omega) & = \\
& X_{1}(\Omega) & P_{0}(\Omega) X_{d}(\Omega T) \\
\therefore H(\Omega) & =\frac{X_{1}(\Omega)}{X_{0}(\Omega)}=\frac{P_{1}(\Omega)}{P_{0}(\Omega)} &
\end{array}
$$

For zero-order hold:

$$
P_{0}(\Omega)=\int_{0}^{T} 1 \cdot e^{-j \Omega t} d t=T e^{-j \frac{\Omega T}{2}} \operatorname{sinc}\left(\frac{\Omega T}{2}\right)
$$

For first-order hold:

$$
\begin{aligned}
p_{1}(t) & =\frac{1}{T} r e c t\left(\frac{t}{T}\right) * \operatorname{rect}\left(\frac{t}{T}\right)=\frac{1}{\sqrt{T}} p_{0}\left(t+\frac{T}{2}\right) * \frac{1}{\sqrt{T}} p_{0}\left(t+\frac{T}{2}\right) \\
\therefore P_{1}(\Omega) & =\left(\frac{1}{\sqrt{T}} T e^{-j \frac{\Omega T}{2}} \operatorname{sinc}\left(\frac{\Omega T}{2}\right) e^{j \frac{\Omega T}{2}}\right)^{2}=T \operatorname{sinc}^{2}\left(\frac{\Omega T}{2}\right) \\
\therefore H(\Omega) & =\frac{T \operatorname{sinc} c^{2}\left(\frac{\Omega T}{2}\right)}{T e^{-j \frac{\Omega T}{2}} \operatorname{sinc}\left(\frac{\Omega T}{2}\right)}=e^{j \frac{\Omega T}{2}} \operatorname{sinc}\left(\frac{\Omega T}{2}\right)
\end{aligned}
$$

(b) For an ideal D/A, $Y_{a}(\Omega)=G_{\text {ideal }}(\Omega) Y_{d}(\Omega T)$ where:
$G_{\text {ideal }}(\Omega)=\left\{\begin{array}{ll}T, & |\Omega| \leq \frac{\pi}{T} \\ 0, & \text { otherwise }\end{array}\right.$ For the FOH, to form an ideal D/A, $Y_{a}(\Omega)=F_{1}(\Omega) P_{1}(\Omega) Y_{d}(\Omega T)$ where $F_{1}(\Omega)$ is an analog filter that follows the FOH and

$$
F_{1}(\Omega)= \begin{cases}\frac{T}{P_{1}(\Omega)}=\frac{1}{\operatorname{sinc}^{2}\left(\frac{\Omega T}{2}\right)}, & |\Omega| \leq \frac{\pi}{T} \\ 0, & \text { otherwise }\end{cases}
$$

(c) Suppose $F_{0}(\Omega)$ and $F_{1}(\Omega)$ are the analog filters that follow the ZOH and FOH , respectively. For reference, the frequency responses of the ZOH and FOH and their subsequent filters are given in Fig. 5. The cutoff frequency for both $F_{0}(\Omega)$ and $F_{1}(\Omega)$ is $\Omega=\frac{\pi}{T}$. The magnitudes of these LPFs are

$$
\begin{aligned}
\left|F_{0}(\Omega)\right| & =\left|\frac{1}{\operatorname{sinc}\left(\frac{\Omega T}{2}\right)}\right| \\
\left|F_{1}(\Omega)\right| & =\left|\frac{1}{\operatorname{sinc}^{2}\left(\frac{\Omega T}{2}\right)}\right|
\end{aligned}
$$



Figure 5: Figure for Problem 5(c)
$\left|F_{1}(\Omega)\right|$ has a frequency response which is steeper that $\left|F_{0}(\Omega)\right|$. Therefore, the FOH (linear interpolation) might interpolate $y[n]$ more precisely than a ZOH (piecewise constant interpolation) at a cost of a more complicated/expensive analog filter that follows the FOH.

## Problem 6

(a) It is given that

$$
y_{a}(t)=x_{a}(t)+\alpha x_{a}\left(t-\tau_{d}\right)+\beta x_{a}\left(t-2 \tau_{d}\right)
$$

Applying the Fourier transform on both sides, we have:

$$
\begin{gathered}
Y_{a}(\Omega)=X_{a}(\Omega)+\alpha X_{a}(\Omega) e^{-j \Omega \tau_{d}}+\beta X_{a}(\Omega) e^{-j 2 \Omega \tau_{d}} \\
H_{a}(\Omega)=\frac{Y_{a}(\Omega)}{X_{a}(\Omega)}=1+\alpha e^{-j \Omega \tau_{d}}+\beta e^{-j 2 \Omega \tau_{d}}
\end{gathered}
$$

(b) Taking appropriate sampling period to mean the Nyquist sampling period:

$$
T_{0}=\frac{1}{2 f_{\max }}=\frac{1}{2 \times 20 \times 10^{3}}=25 \mu \mathrm{~s}
$$

(c) With the sampling period $T_{0}=25 \mu s$ there is no aliasing in the system. With an ideal $\mathrm{D} / \mathrm{C}$, the digital filter we need is:

$$
\begin{align*}
H_{d}(\omega) & =H_{a}\left(\frac{\omega}{T_{0}}\right)=1+\alpha e^{-j \omega \tau_{d} / T_{0}}+\beta e^{-j 2 \omega \tau_{d} / T_{0}}  \tag{*}\\
& =1+\alpha e^{-j 40000 \omega \tau_{d}}+\beta e^{-j 80000 \omega \tau_{d}}
\end{align*}
$$

(d) With $\tau_{d}=100 T_{0}$, Equation $(*)$ above simplifies as:

$$
H_{d}(\omega)=1+\alpha e^{-100 j \omega}+\beta e^{-200 j \omega}
$$

By substituting $z=e^{j \omega}$, we obtain the following transfer function:

$$
H(z)=1+\alpha z^{-100}+\beta z^{-200}
$$

which can be implemented using the block diagram below.


Figure 6: Figure for Problem 6(d)

Problem 7 The Fourier transform of $y_{c}(t)$ is sketched in Fig. 7,


Figure 7: Figure for Problem 7

## Problem 8

(a) $H(z)$ is not a polynomial in $z$ or $z^{-1}$ the system is IIR.
(b) $H(z)$ is not a polynomial in $z$ or $z^{-1}$ the system is IIR.
(c) $H(z)$ is a polynomial of $z^{-1}$ the system is an FIR filter.

