EE 451

Homework #14

Due December 3, 2001

1. Problem 10.8 from the Text.

2. Consider the system below:

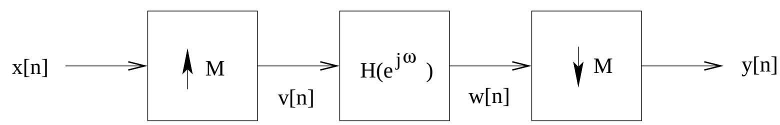


Figure 2−1

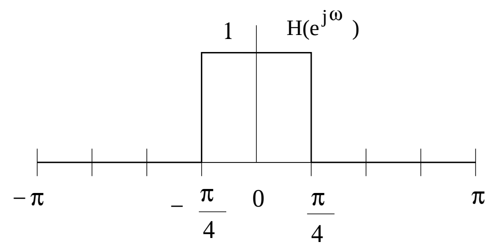


Figure 2−2

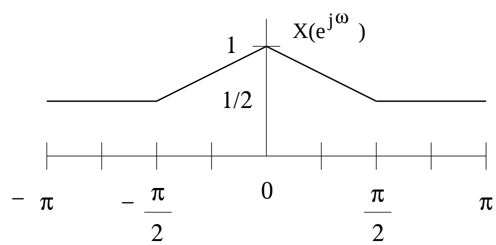


Figure 2−3

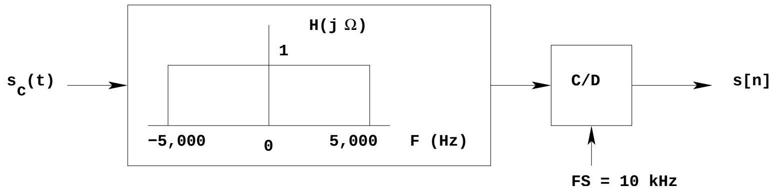
(a) For $M = 2$, make an appropriately labeled sketch of $V(e^{j\omega})$, $W(e^{j\omega})$, and $Y(e^{j\omega})$. Be sure to label significant amplitudes and frequencies.

(b) For $M = 2$, determine and sketch the magnitude of the frequency response for the overall system, $|H_{eff}(e^{j\omega})|$.

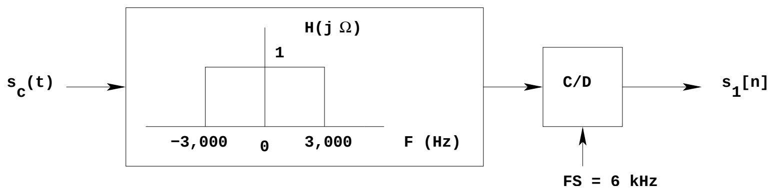
(c) For $M = 6$, make an appropriately labeled sketch of $V(e^{j\omega})$, $W(e^{j\omega})$, and $Y(e^{j\omega})$. Be sure to label significant amplitudes and frequencies.

(d) For $M = 6$, determine and sketch the magnitude of the frequency response for the overall system, $|H_{eff}(e^{j\omega})|$.

3. Suppose you obtained a sequence $s[n]$ by filtering a speech signal $s_c[t]$ with a continuous-time low-pass filter with a cutoff frequency of 5 kHz and then sampled it at a 10 kHz rate, as shown below:



Later you find that what you should have done was to filter the signal at 3 kHz and sample it at 6 kHz:



Develop a method to obtain $s_1[n]$ from $s[n]$ using discrete-time processing. If your method uses a discrete time filter, you should specify the frequency response of the filter. (You may use an ideal filter.)

4. You are given a digital audio tape (DAT) of some music, and are asked to transfer the music to a CD. You want to preserve the information on the DAT tape with frequencies from 0 to 20 kHz, with a passband ripple of at most 0.5 dB. Any high-frequency alias into a low-frequency signal should be attenuated by at least 80 dB. Design a system to do this.

   (a) Draw a single-stage implementation (one up-sampler followed by a LPF followed by a down-sampler) of the system, like the one discussed in class.

   (b) What are the specifications of the low pass filter – gain, passband frequency, stopband frequency?

   (c) Suppose you want to implement this filter with an FIR filter using a Kaiser window. What order of filter do you need?

   (d) Use MATLAB to find the coefficients for your filter. (If the order of your filter is much larger than 3,000, limit the order to order 3,000 so the next part will not take an excessive amount of computer time.) Plot the impulse response of the filter.

   (e) Use MATLAB to generate a signal with a frequency of 5 kHz, sampled the DAT frequency:

```
FS_DAT = 48e3;                   % DAT Sampling Frequency
F1 = 5e3;                        % Frequency of input signal
t_DAT=0:1/FS_DAT:0.01;           % Time vector at DAT frequency
n_DAT = 1:length(t_DAT);         % Indices of times
x1_DAT = cos(2*pi*F1*t_DAT);     % Input signal sampled at DAT freq
```

   Process this signal with the system you designed in Part a, using the filter you designed in Part d. To put L-1 zeros between each sample you can do something like this:

```
n_up = 1:(length(n_DAT)*L);   % Indices of up-sampled signal
x1_up = zeros(size(n_up));    % Start with all samples 0
ii = 1:L:length(n_up);        % Find index of samples which will have data
x1_up(ii) = x1_DAT;           % Put data into every L'th sample
```

You can filter your signal with MATLAB's `filter()` function, where `h` is the impulse response of your filter:

```
y1_up = filter(h,1,x1_up);     % Filter up-sampled signal with FIR filter
```

You can then downsample to the CD rate by taking every M'th sample:

```
FS_CD = 44.1e3;                    % CD sampling frequency
y1_CD = y1_up(1:M:length(y1_up));  % Take every M'th sample to downsample
n_CD = 1:length(y1_CD);            % Indices of down-sampled signals
t_CD = n_CD/FS_CD;                 % Time vector for down-sampled signal
```

Plot the signal $y_{1CD}[n]$ (vs. t_{CD}) and compare it with $x_{1DAT}[n]$ (plotted vs. t_{DAT}). Do the two signals look similar?

(f) Repeat the above part for an input signal of frequency 20 kHz.