## 2-1-1 -15 = 3b

## National Tsing Hua University Department of Electrical Engineering

EE3660 Introduction to Digital Signal Processing, Spring 2020

Note: Detailed derivations are required to obtain a full score for each problem.

1. (15%) Consider a system which is described by

stem which is described by 2y[n] + 3y[n-1] - 2y[n-2] = -2x[n] + 11x[n-1].

- (a) (5%) Determine the region of convergence for this system to be causal and also find the corresponding impulse response h[n].
- (b) (5%) Determine the region of convergence for this system to be stable and also find the corresponding impulse response h[n]
- (c) (5%) Convert this system into a minimum-phase one and determine the system function  $H_{\min}(z)$ .
- 2. (15%) Consider a sinusoidal signal  $x_c(t) = \cos(2\pi F_0 t)$  with  $F_0 = 11$  kHz. It is sampled at different rates of  $F_s$  and then reconstructed as  $y_r(t)$  by the corresponding system

$$G(j\Omega) = \begin{cases} \frac{\sin(\Omega T/2)}{\Omega/2} e^{-j\Omega T/2}, & |\Omega| < \pi/T \\ 0, & \text{otherwise} \end{cases}, \qquad -2 \int_{-2}^{2\pi/2} \frac{d^{2}}{dt} \int_{-2}^{2\pi/2} \frac{dt}{dt} \int_{-2\pi/2}^{2\pi/2} \frac{dt}{dt} \int_{-2\pi/2}^{2\pi/$$

where  $T = 1/F_s$ . Note that this reconstruction system is equivalent to a sample-and-hold DAC followed by an ideal lowpass filter.

- (a) (5%) Determine  $y_r(t)$  if  $F_s = 33$  kHz.
- (b) (5%) Determine  $y_r(t)$  if  $F_s = 6$  kHz.
- (c) (5%) Consider a slightly complex scenario. Use  $x_c(t)$  as a carrier signal to modulate a low-frequency signal  $s(t) = \cos(2\pi F_1 t)$  where  $F_1 = 44$  Hz. And sample the modulated signal  $x_m(t) = s(t) \cdot x_c(t)$  at a rate of  $F_s = 44$  kHz. Then process the sampled signal  $x_m[n]$  by an all-pass digital filter  $H(e^{j\omega}) = e^{-j4\omega^3/\pi^2}$ . Finally, reconstruct  $y_r(t)$  from the filtered signal using the corresponding  $G(j\Omega)$ . Now approximate  $y_r(t)$  as accurate as you can. (Hint: use group delay to approximate the time delay of s(t).)
- 3. (10%) The 8-point DFT of an 8-point sequence x[n] is given by

$$X[k] = \{1, 2+j, 4-j, 8+j, 16, 8-j, 4-j, 2-j\}.$$

Determine the DFT of each of the following sequences.

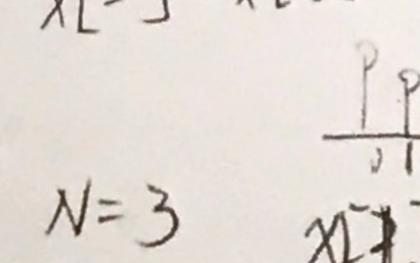
- (a) (2%)  $x_1[n] = x[\langle n-2\rangle_8].$
- (b) (4%)  $x_2[l] = \sum_{n=0}^7 x[n]x^*[\langle n-l\rangle_8]$  (circular autocorrelation).
- (c) (4%) Real-valued  $x_3[n]$  and  $x_4[n]$  such that  $x[n] = x_3[n] + jx_4[n]$ . You may need the symmetry properties as below.

$$X[n] = X_R^{ce}$$

$$X[k] = X_R^{ce}$$

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 $x[n] = x_R^{ce}[n] + x_R^{co}[n] + jx_I^{ce}[n] + jx_I^{co}[n] \quad \chi[] = \chi[]$  $X[k] = X_R^{ce}[k] + X_R^{co}[k] + jX_I^{ce}[k] + jX_I^{ce}[k]$ 



4. (15%) Let X[k] be a real-valued N-point DFT for an N-point sequence x[n].

- (a) (4%) Show that  $x[\frac{N-1}{2}] = x^*[\frac{N+1}{2}]$  if N is odd.
- (b) (5%) Show that  $|x[n]| = |x[\langle -n \rangle_N]|$ .
- (c) (6%) If X[k] satisfies the condition  $X[k] = X[\langle k+M\rangle_N]$  where N = lM and l is an integer, show that x[n] = 0 if  $l \nmid n$ .
- (15%) Complete the following example for computational Fourier analysis.
  - (a) (3%) For a right-sided exponential function  $x_c(t) = 2^{-t/T}u(t)$ , determine its CTFT  $X_c(j\Omega)$ given that  $e^{-at}u(t) \xrightarrow{\mathscr{F}} \frac{1}{a+i\Omega}$ .
  - (b) (4%) Sample  $x_c(t)$  with t = nT to form a discrete-time signal  $x[n] = x_c(nT)$  and then determine its DTFT  $X(e^{j\omega})$ . Also briefly state the relationship between  $X(e^{j\omega})$  and  $X_c(j\Omega)$ .
  - (c) (4%) Sample the frequency components of  $X(e^{j\omega})$  as four-point DFT  $X_1[k] = X(e^{j2\pi k/4})$ where k = 0, 1, ..., 3. Determine its IDFT  $x_1[n]$  which is a finite-length signal now.
  - (d) (4%) Now we want to have a detailed evaluation of the DTFT of  $x_1[n]$ . Explain how to use DFT computation to accurately derive  $X_1(e^{j\omega})$  at  $\omega = 2\pi k/1024$  where k = 0, 1, ..., 1023.
- 6. (10%) Consider a highpass linear-phase FIR filter design using the Kaiser window w[n] = $\frac{I_0\left(\beta\sqrt{1-[(n-\alpha)/\alpha]^2}\right)}{I_0(\beta)}$  where  $0 \le n \le M$  and  $I_0(x)$  is the zeroth-order modified Bessel function. The specifications are  $\omega_p = 0.6\pi$ ,  $\omega_s = 0.5\pi$ , and  $\delta_s = 0.01$ . Determine the following design parameters to minimize the filter length: (a) (3%)  $\beta$ , (b) (3%) cut-off frequency  $\omega_c$ , and (c) (4%) window (filter) length L. You may need the following empirical relations:

$$\beta = \begin{cases} 0.5842(A - 21)^{0.4} + 0.07886(A - 21), & A < 21\\ 0.1102(A - 8.7), & 21 \le A \le 50 \end{cases},$$

$$M = \frac{A - 8}{2.285 \triangle \omega}.$$
(1)

(20%) Consider a MATLAB FFT function myfft512(x) which performs exactly 512-point DFT of a 512-point input vector x. It can be used for efficient computation. For example,

the IDFT y of a given 512-point DFT Y (as a row vector) can be derived by the following MATLAB code.

Yflip=
$$[Y(1) Y(512 : -1 : 2)];$$
  
y= myfft512(Yflip)/512;

Write down efficient MATLAB codes for the following algorithms using only the myfft512 function and simple arithmetic/indexing/padding operations. All 1D vectors are row-wise arranged.

- (a) (10%) Linear convolution. Given a 400-point x1 and a 500-point x2, derive its linear convolution x3 (899-point) through multiplication in DFT domain. You may consider the overlap-save or overlap-add method.
  - (b) (5%) Time-domain resolution scaling-up. Given a real-valued 63-point x4, derive its 8× scaled-up real-valued signal x5 (504-point) by first applying 64-point DFT, then padding zeros, e.g. zeros(1, k), in DFT domain, and finally performing 512-point IDFT.
  - (c) (5%) Efficient DFT for real-valued sequences. Given two 512-point real-valued sequences x6 and x7, derive their 512-point DFTs X6 and X7, respectively, by calling myfft512(...) only once. You may need the real and imag functions to extract the real and imaginary parts. You may also need the symmetry properties in the problem 3.