

## E2.5 Signals & Linear Systems

### Tutorial Sheet 7 – Sampling

1.\* By applying the Parseval's theorem, show that

$$\int_{-\infty}^{\infty} \text{sinc}^2(kx) dx = \frac{\pi}{k}.$$

2.\* Fig. Q2 (a) and (b) shows Fourier spectra of signals  $f_1(t)$  and  $f_2(t)$ . Determine the Nyquist sampling rates for the following signals. (Hint: Use the frequency convolution and the width property of the convolution.)

- a)  $f_1(t)$                       b)  $f_2(t)$                       c)  $f_1^2(t)$   
 d)  $f_2^3(t)$                       e)  $f_1(t)f_2(t)$

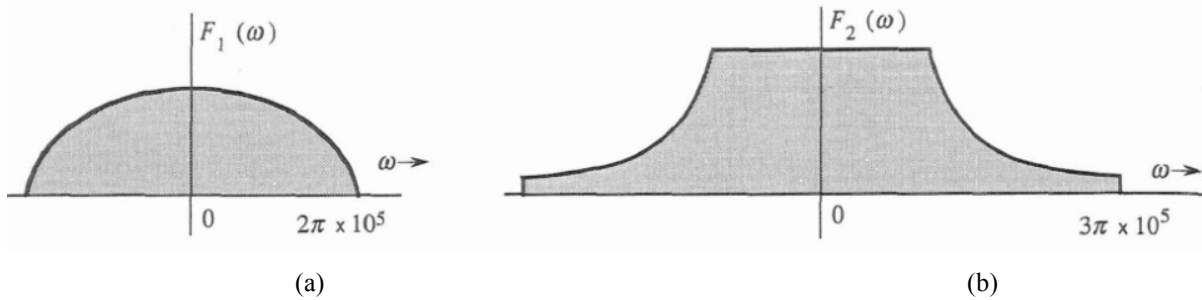


Figure Q2

3.\* Signals  $f_1(t) = 10^4 \text{rect}(10^4 t)$  and  $f_2(t) = \delta(t)$  are applied at the inputs of ideal lowpass filters  $H_1(\omega) = \text{rect}(\frac{\omega}{40,000\pi})$  and  $H_2(\omega) = \text{rect}(\frac{\omega}{20,000\pi})$ . The outputs  $y_1(t)$  and  $y_2(t)$  of these filters are multiplied to obtain the signal  $y(t) = y_1(t)y_2(t)$  as shown in Figure Q3.

- a) Sketch  $F_1(\omega)$  and  $F_2(\omega)$ .  
 b) Sketch  $H_1(\omega)$  and  $H_2(\omega)$ .  
 c) Sketch  $Y_1(\omega)$  and  $Y_2(\omega)$ .  
 d) Find the Nyquist sampling rate of  $y_1(t)$ ,  $y_2(t)$  and  $y(t)$ .

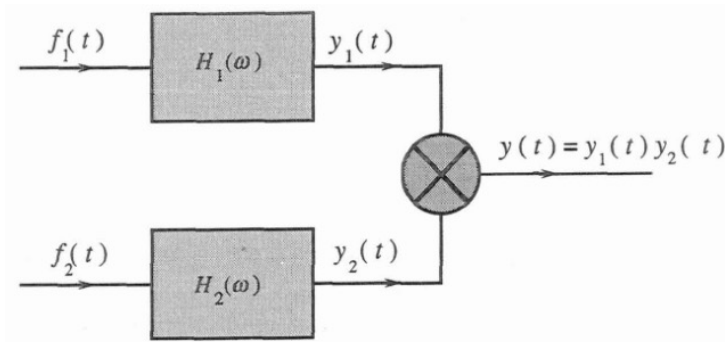


Figure Q3

4.\*\* For the signal  $e^{-at}u(t)$ , determine the bandwidth of an anti-aliasing filter if the essential bandwidth of the signal contains 99% of the signal energy.

5.\*\* A zero-order hold circuit shown in Fig. Q5 is often used to reconstruct a signal  $f(t)$  from its samples.

- Find the unit impulse response of this circuit.
- Find the transfer function  $H(\omega)$ , and sketch  $|H(\omega)|$ .
- Sketch the output of this circuit for an input  $f[n]$  which is the sampled version of  $f(t)$  which is  $\frac{1}{4}$  cycle of a sinewave. The sampling period is  $T$ .

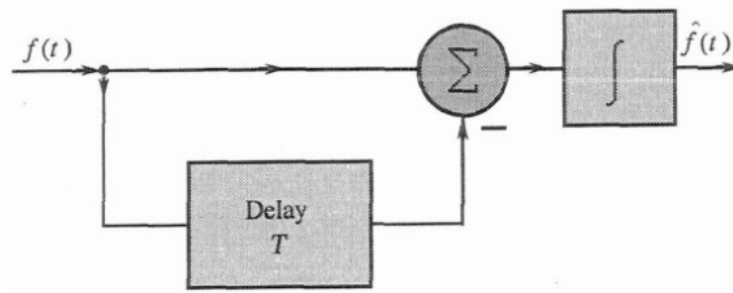


Figure Q5

6. MATLAB exercise: Write a matlab function of the form:

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function [ t, sinewave ] = sinegen( fsig,fsamp,T )
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that is able to generate a sampled sinewave of frequency  $f_{sig}$  sampled at  $f_{samp}$  on the interval  $[0,T)$ . Using this function reproduce the figures in slides 14 of lecture 13 which shows aliasing when sampling sinewaves of frequency 1Hz and 6Hz using the sampling rate  $f_s=5\text{Hz}$ . Plot the sampled signals using the command 'stem'.