**Guided Solution for Tutorial 3**

Q1-8



Figure 1.38 A certain DSP system

Figure 1.38 shows the block diagram of a certain DSP system. Given that *fs* is the sampling frequency used and: *x*(*t*) = cos(2000*t*) + 2cos(4000*t*) + 3cos(10000*t*)

(a) In Figure 1.38, if *fc* = *fs* = 8 kHz, sketch the spectrum of the signal *y*(*t*) for 0   *f*  8 kHz. Briefly point out the possible problems or limitations for processing *x*(*t*), if any.

(b) If *fc* = 4 kHz and *fs* = 8 kHz, sketch the spectrum of the signal *y*(*t*) for 0   *f*  8 kHz. Briefly point out the possible problems or limitations for processing *x*(*t*), if any.

(c) If *fc* = 6 kHz and *fs* = 12 kHz, sketch the spectrum of the signal *y*(*t*) for 0   *f*   8 kHz. Briefly point out the possible problems or limitations for processing *x*(*t*), if any.

(d) If *fc* = ∞ (no filters) and *fs* = 12 kHz, briefly point out the possible problems or limitations for processing *x*(*t*), if any.

x(t) is an analog signal and went through an ideal low pass filter with cutoff frequency, fc. There are 3 frequency components in x(t), f1, f2 and f3. Using the example in guided solution for Tutorial 1, the frequency f1, 1000Hz for cos(2000*t*). Similarly, f2 is 2000Hz for cos(4000*t*) and f3, 5000 Hz.

For part (a), all the frequencies are not affected by the low pass filter with cutoff frequency, fc, 8 kHz, that is none of the input frequency component is removed.

However, we can see that the 5 kHz frequency component is affected by the sampling frequency, fs of 8 kHz as it does not satisfy Nyquist critiera, hence aliasing occurs and the sampled frequeny spectrum will be affected.

For part (b), the 5 kHz frequency component is being removed by the cut off frequency, 4 kHz leaving only f1 and f2. Note that f1 and f2 satisfy the Nyquist criteria for sampling frequency of 8kHz. However, now the input signal is being modifed to 2 frequency components.

For part (c) and (d), similar reasonings are carried out.

**Hint:** See Figure 1.21, page 14 - Processing analog signals using DSP techniques (textbook)

Q1-9\* An analog signal *x*(*t*) is given as  and it is sampled with 8 kHz sampling frequency. If the first sample is taken at *t* = 0, what is the magnitude of the 1000th sample?

The analog signal, x(t) is now sampled at 8kHz, substitute t by nTs where Ts = 1/fs. fs is 8kHz. Substitute, n=0 to find out the first sample and 999 for the 1000th sample.

**Hint:** Page 11- 1.5 Converting an analog signal into a digital signal (textbook)