ASIAN INSTITUTE OF TECHNOLOGY
SCHOOL OF ENGINEERING AND TECHNOLOGY
TELECOMMUNICATIONS
AT77.02 - SIGNALS, SYSTEMS AND STOCHASTIC PROCESSES

Sampling Theory

1 Objectives

1. To understand sampling theory
2. To understand aliasing effect

2 Software

Matlab

3 Background and Theory

Sampling is the process of recording the values of a signal at given points in time. The number of samples taken during one second is called the sample rate.
The Nyquist Sampling Rate is the lowest sampling rate that can be used without having aliasing. The sampling rate for an analog signal must be at least two times the bandwidth of the signal.

4 Procedure

1. Using the provided makecos function generate an analog signal with 20 Hz frequency.
2. Sample this signal using a 50 Hz(oversampling) sampling frequency. For this you have to generate an impulse train with frequency 50 Hz with the given makeimp function then use sampleit1 function to sample the signal and finally plot the signal with smpl plot function.
3. Sample this signal using a 30 Hz(under sampling) sampling frequency. Use the same steps as in the previous step.
4. Using the first sampled signal ms1, original signal can be reconstructed while using ms2 aliasing will occur.
5. Reconstruct the signal using the function interpsinc function and plot the waveforms in both time and frequency using recon plot function in order to verify the sampling theory.

6. Notice how the first reconstructed cosine has the same frequency as the original while the second has a different frequency. Clearly in the second reconstructed signal is not a perfect reconstruction. The new, folded frequency is 10 Hz. This is because the original frequency of 20 Hz is being folded over 30/2 = 15 Hz.

7. Calculate the frequency spectrum of original and two reconstructed signals using am spectrum function and plot it using am plot function. Verify the frequencies of signals and the aliasing.

8. Sampling Music - Load the given music sample and look at its spectrum.

9. The spectrum extends up to 16 kHz. This is to be expected since the sampling rate fs of the original signal is 32kHz. If this signal is sampled with anything less than 32 kHz there will be aliasing. Use one eighth of the original sampling frequency that is 32/8=4 kHz. This means that frequencies only up to 2kHz will be represented.

10. The signal now sounds dull, it lost its brightness since the high frequencies are gone. Actually those high frequencies are folded because of aliasing thus creating distortion. Now repeat the procedure but by using a low pass anti aliasing filter.

11. Compare the signals md1 and md2 and listen to the difference with the following simple command: soundsc(md1 - md2,newfs)

12. By comparison of the three spectra it can be seen that in the case of no anti aliasing filter there is more energy on the 1-2 kHz band compared to the case with anti aliasing filter. This is because the anti aliasing filter prevents frequencies from being folded over.

13. Try to change the playback rate of the original sound using the command- soundsc(m,fs/2) for half the speed and the command - soundsc(m,fs*2) for double speed. This is a practical application of the scaling property of the Fourier transform.

5 Discussion

1. Explain all the Matlab functions used in this experiment.

2. Following the procedure described in the first part of the lab check for aliasing and calculate the frequency of the reconstructed cosines in the following cases:

   (a) Cosine: 30Hz, Sampling: 50Hz
   (b) Cosine: 40Hz, Sampling: 15Hz
   (c) Cosine: 10Hz, Sampling: 50Hz
   (d) Cosine: 20Hz, Sampling: 40Hz