

INDIAN INSTITUTE OF TECHNOLOGY,
MANDI

DIGITAL SIGNAL PROCESSING (EE305)

ASSIGNMENT 1

Basic operations on sequences

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1 P1

The sequence, $x[n]$ is taken as input and it is assumed to start from $n = 1$. The sequence is finite and arbitrary.

1.1 Shifting of any given sequence

The sequence is shifted by user given amount by manipulating the x-axis. The example is shown in figure 1.

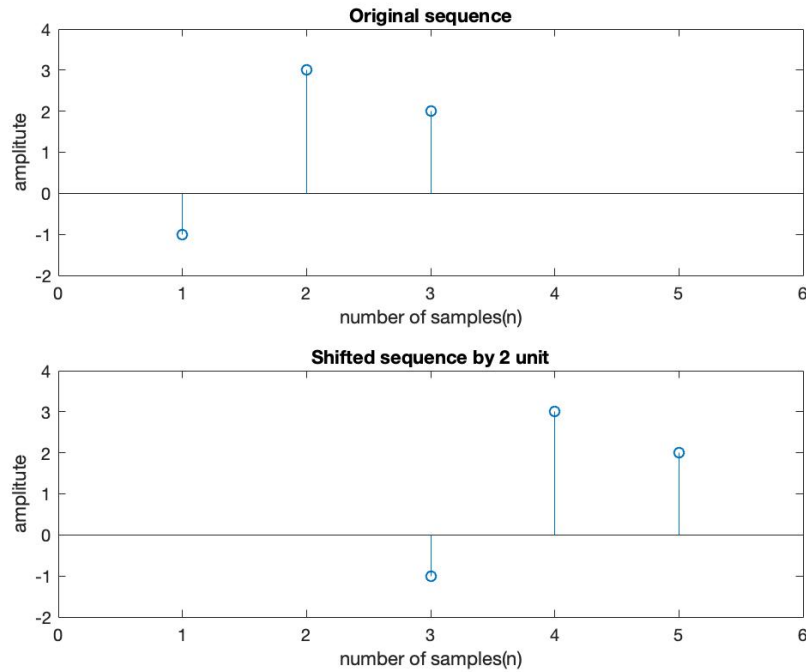


Figure 1: Sequence $x[n] = [-1 \ 3 \ 2]$ and is shifted by 2 units to right

1.2 Folding any given sequence

Folding a sequence means mirroring across the y-axis. An example is shown in figure 2.

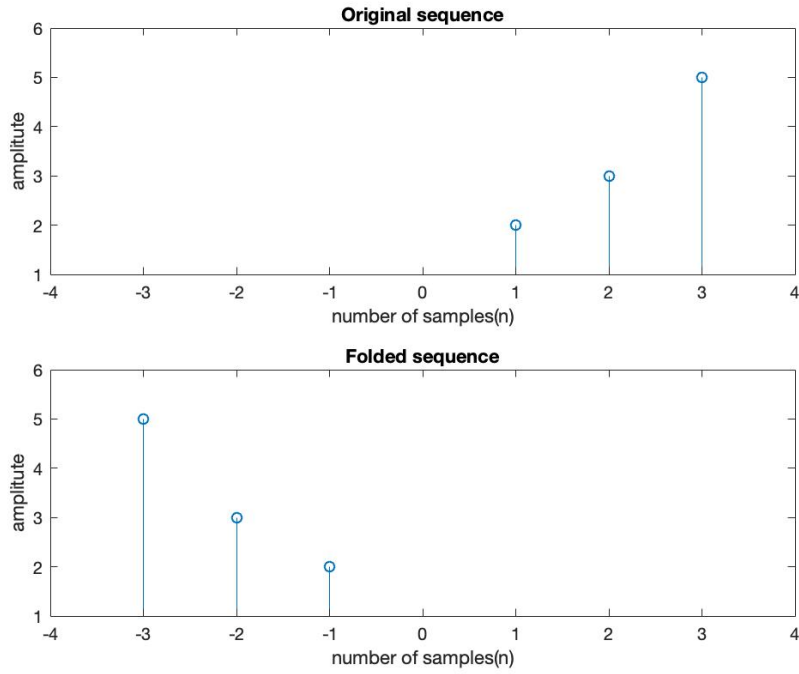


Figure 2: Sequence $x[n] = [2 \ 3 \ 5]$ and its folded version

2 P2

The sequences are finite and arbitrary. x_1 and x_2 are two sequence taken as input. The convolution of two sequences x_1 and x_2 is calculated and plotted. An example is shown in figure 3.

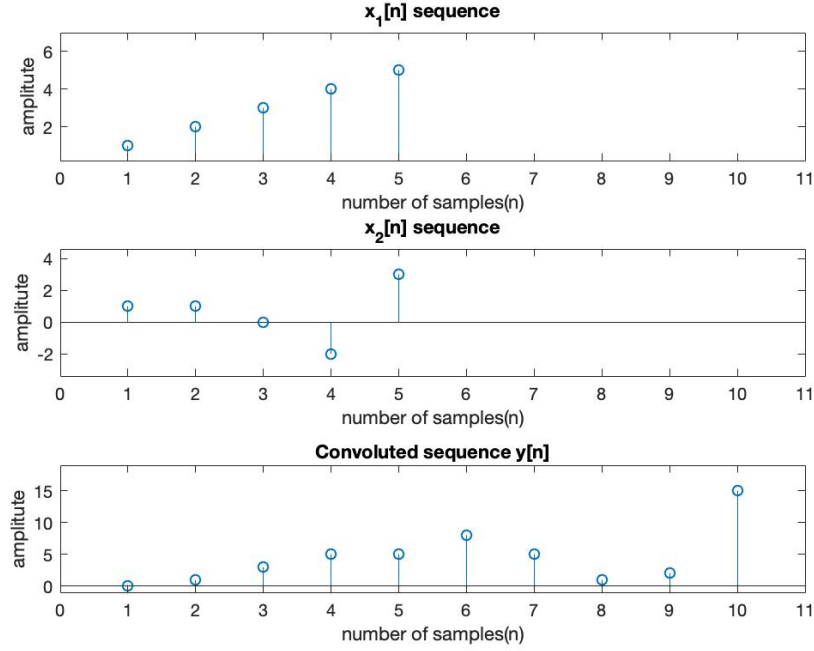


Figure 3: Here, $x_1[n] = [1 \ 2 \ 3 \ 4 \ 5]$ and $x_2 = [1 \ 1 \ 0 \ -2 \ 3]$. The convolution between them is defined as $y[n] = x_1 \otimes x_2 = [0 \ 1 \ 3 \ 5 \ 5 \ 8 \ 5 \ 1 \ 2 \ 15]$

3 P3

Given signal, $x_a(t) = \cos(2000\pi t)$. The signal is continuous, sinusoidal with frequency 1000 Hz. Since the signal exists from $-\infty$ to $+\infty$, it is not possible to represent the given signal in MATLAB with limited memory. Hence, the signal is assumed to be time and band limited and defined at discrete time points. The decimation of time is much smaller as compared to sampling rates defined to later part of the problem.

3.1 Fourier transform of $x(t)$

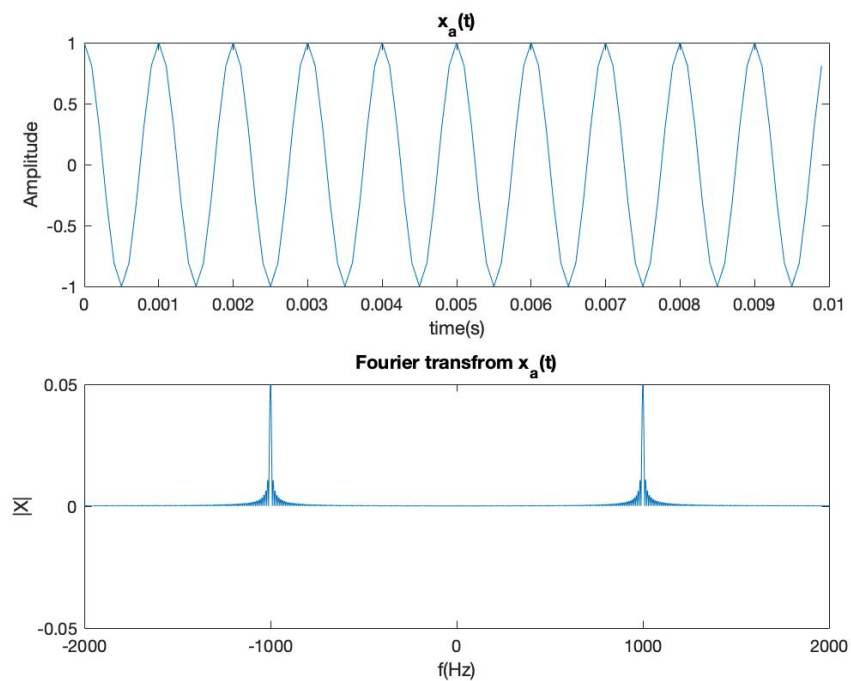


Figure 4: $x_a(t)$ and its Fourier transform

3.2 Sampling

For $T_{s1} = \frac{1}{2500}$, $x_1[n]$ and $X_1(\exp j\omega)$ and for $T_{s2} = \frac{1}{3500}$, $x_2[n]$ and $X_2(\exp j\omega)$ are shown below.

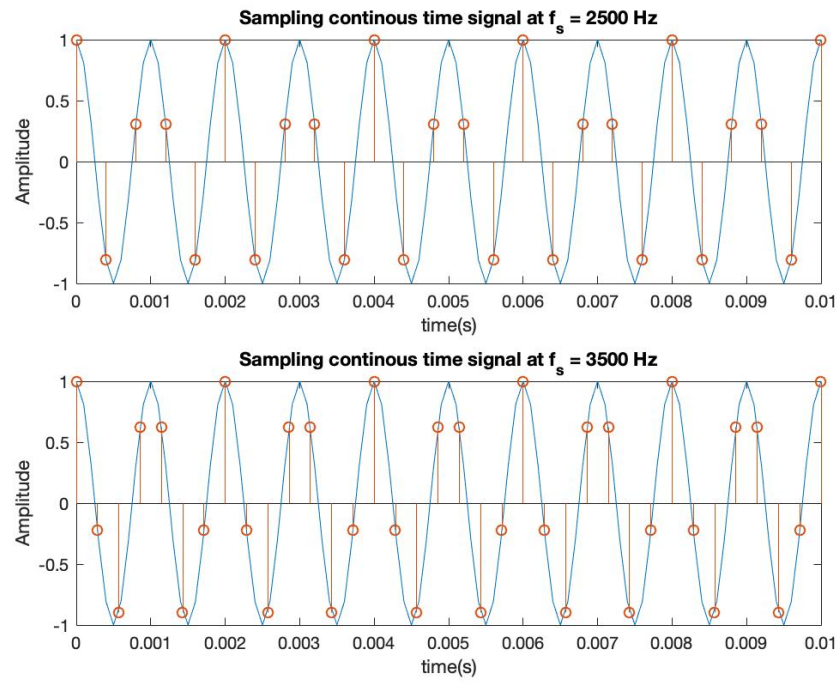


Figure 5: Sampled signals

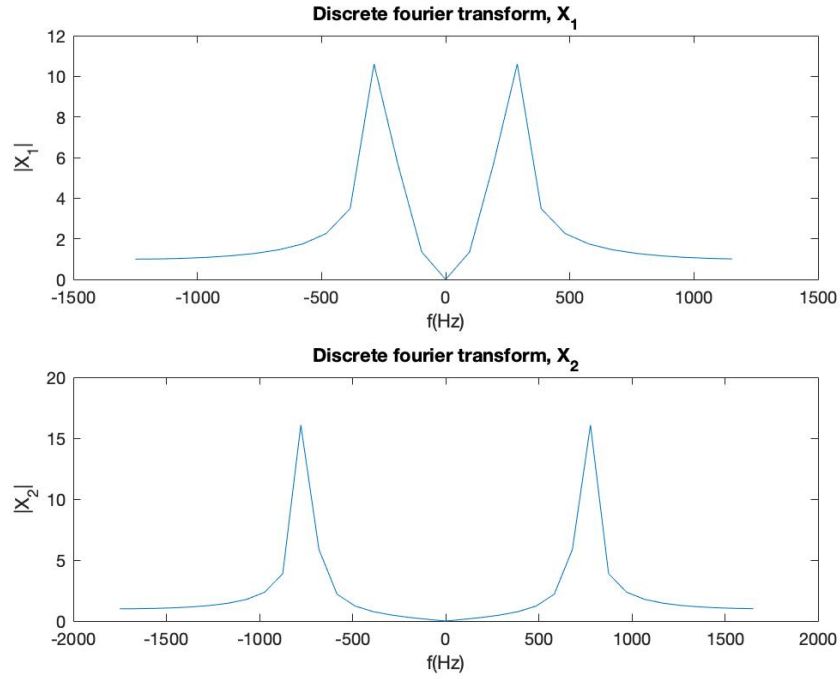


Figure 6: Fourier transform of sampled signals

3.3 Reconstruction of original signal

The original signal is reconstructed from the sampled sequence using linear combination of *sinc* function in time domain which is basically performing low pass filtering in frequency domain. The reconstructed signals are shown in figure abc.

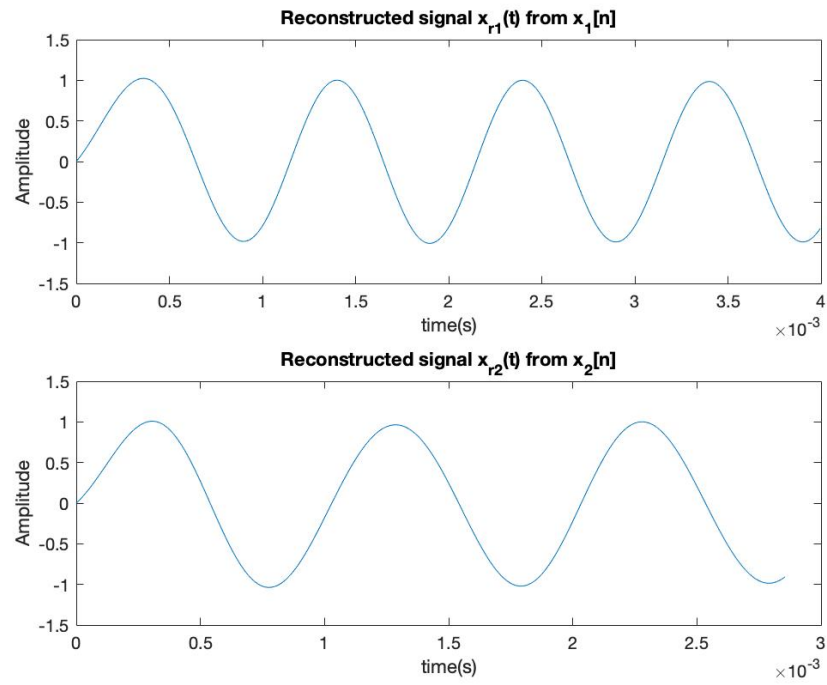


Figure 7: Reconstructed signals