ADVANCED SIGNAL ANALYSIS AND PROCESSING LAB

Lab sheet. No: 09

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Q1. Let the continuous signal xc(t) = cos(2π50t) + cos(2π500t) is sampled at 8000 Hz to obtain x(n). use fir1 function to generate low-pass and high-pass filters of order 100. Use the filter function to separate the two sinusoidal signals from the composite signal by applying the filters.

AIM: To generate the discrete signal by sampling the given signal with a sampling frequency 8000hz.and generate the low pass filter and high pass filter of order 100 and separate the two sinusoidal signals from the composite signal.

Short Theory:

The discrete-time signal is obtained by "taking-samples" of the Analog signal every T second.

$$X(n)=x(nT)$$

The time interval T is called the sampling period or sampling interval . The sampling rate or the sampling frequency is found as **Fs =1/T.** After sampling the continuous signal we pass the signal through low pass filter and high pass filter to extract the low frequency signal and high frequency signal.

Key Commands:

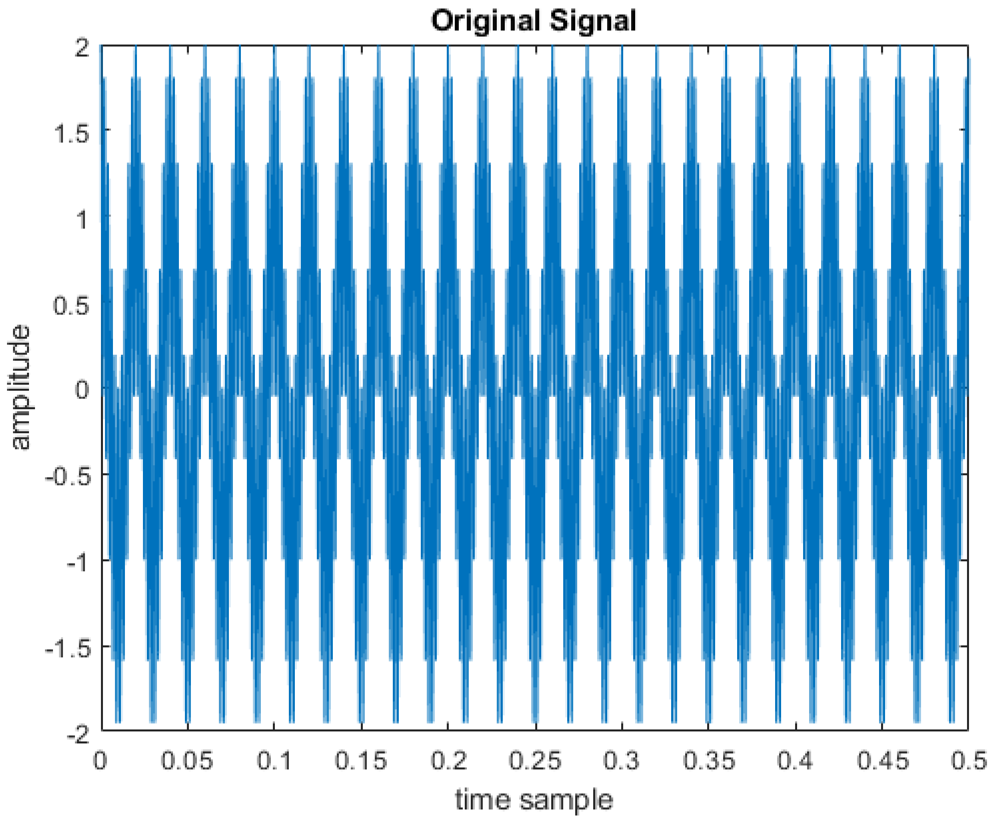
filter % filter(b,a,x) filters the input data x using a rational transfer function defined by the numerator and denominator coefficients b and a.

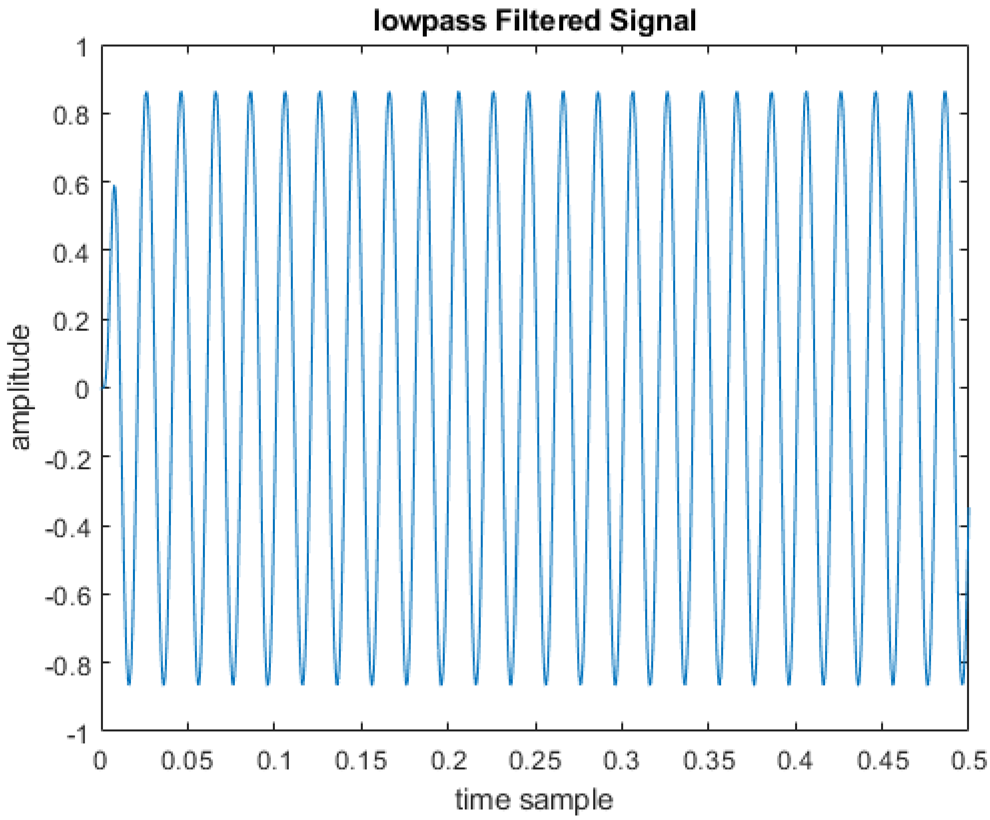
fir1 %fir1(n,Wn) uses a Hamming window to design an nth-order lowpass, bandpass, or multiband FIR filter with linear phase.

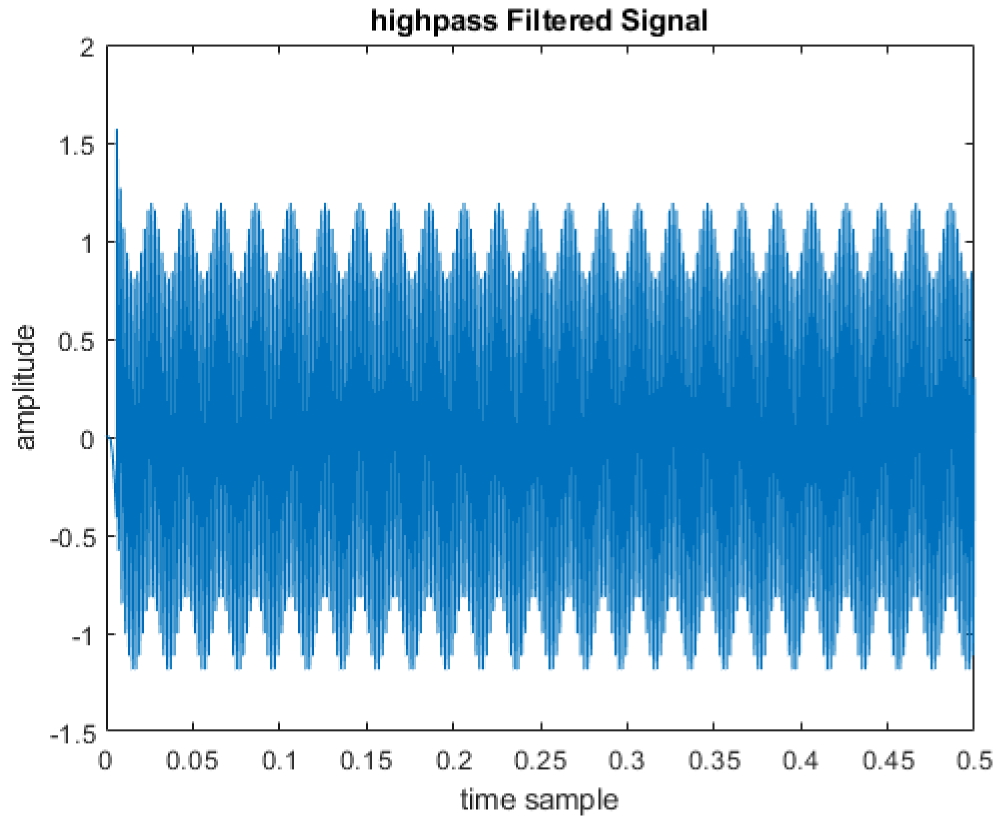
cos %cos(X) returns the cosine for each element of X

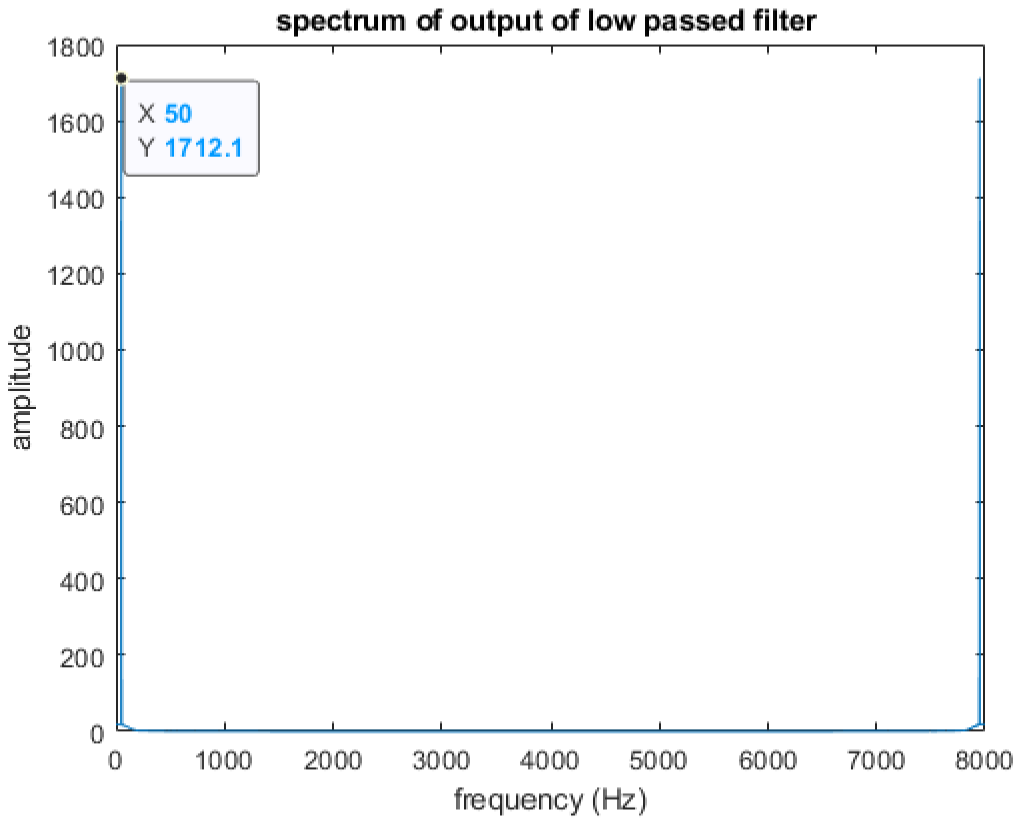
fft %fft(X) computes the discrete Fourier transform (DFT) of X using a fast Fourier transform (FFT) algorithm.

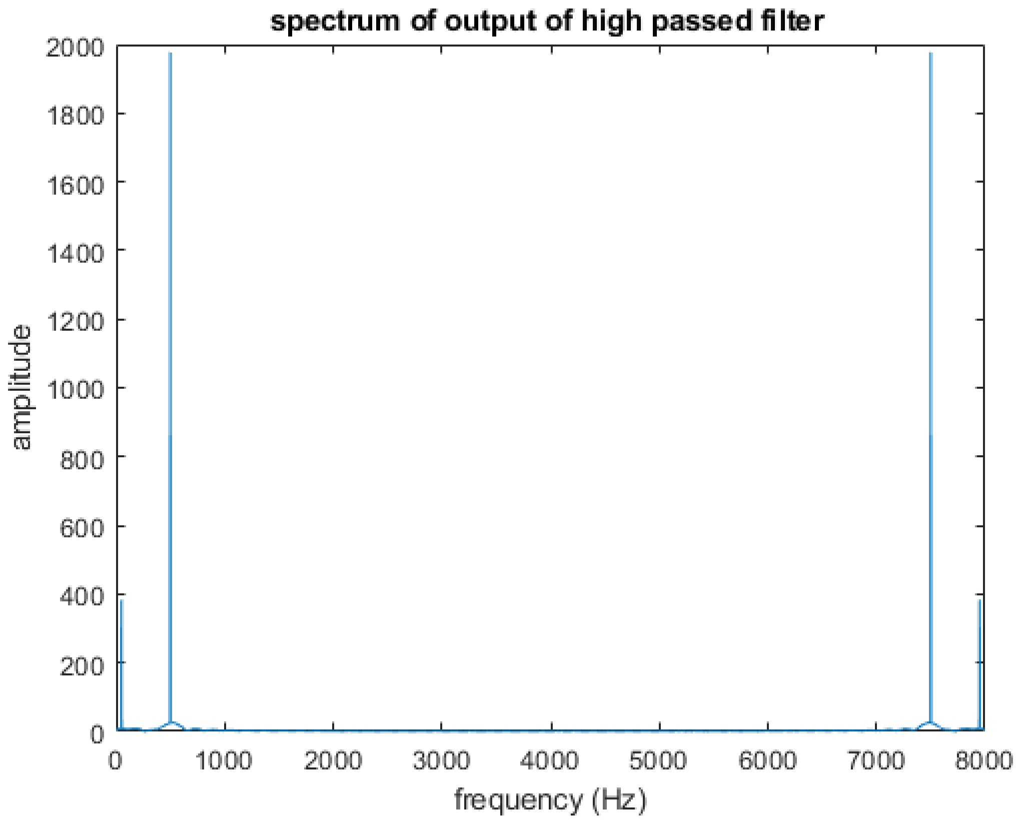
Plots:


Original Signal
amplitude
time sample
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
0
0.05
0.1
0.15
0.2
0.25
0.3
0.35
0.4
0.45
0.5


lowpass Filtered Signal
amplitude
time sample
1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
0
0.05
0.1
0.15
0.2
0.25
0.3
0.35
0.4
0.45
0.5


highpass Filtered Signal
amplitude
time sample
2
1.5
1
0.5
0
-0.5
-1
-1.5
0
0.05
0.1
0.15
0.2
0.25
0.3
0.35
0.4
0.45
0.5


spectrum of output of low passed filter
X 50
Y 1712.1
amplitude
frequency (Hz)
1800
1600
1400
1200
1000
800
600
400
200
0
0
1000
2000
3000
4000
5000
6000
7000
8000



Inferences/comments:

1) From a given composite signal we can separate the signals by passing through the appropriate filter with proper cut off frequency.
2) We can recover the given signal after sampling only when sampling frequency is greater than the 2 times the maximum of the given frequency by using proper filtering technique.
3) If the Fs<2*fm, we cannot recover the original signal.

Q2.

(a) Let the continuous signal xc(t) = cos(10πt) + cos(20πt) + cos(30πt) is sampled at 40 Hz to obtain x(n). Perform down sampling by 4 to obtain xd(n) and compare x(n) and xd(n) and their spectrum. What is sampling frequency after down sampling. Does aliasing occur in this case?

(b) Next let our aim be to avoid aliasing and retain the 5Hz signal alone in xd(n). For that perform low pass filtering of x(n) with cut of frequency 5 (order 100) and then down sample by 4. Again compare the spectrum and the signals before and after down sampling.

(c) Perform low pass filtering in frequency domain, instead of using filter command as above. Which filtering is preferable?

AIM: To generate the discrete signal by sampling the given signal with a sampling frequency 40Hz. Perform down sampling by 4 to obtain xd(n) and compare x(n) and xd(n) and their spectrum. Perform the appropriate filtering to get the original signal and compare the spectrums.

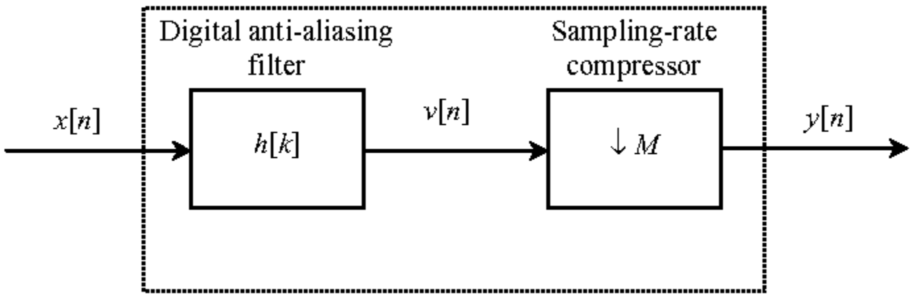
Short Theory:

Down-sampling: The process of reducing the sampling rate by an integer factor(D) is called decimation of the sampling rate. It is also called down sampling by factor(D).Decimator consists of decimation filter to band limit the signal and down sampler to decrease the sampling rate by an integer factor (D).

Decimation

– Reduce the sampling rate of a discrete-time signal.

– Low sampling rate reduces storage and computation requirements.



Aliasing can be avoided if x(n) is low-pass signal band-limited to the region |w| < p / M.. In most applications, the down-sampler is preceded by a low-pass digital filter called "decimation filter". To prevent aliasing at a lower rate, the digital filter h[k] is used to band-limit the input signal to less than Fs /2M beforehand. Sampling rate reduction is achieved by discarding M-1 samples for every M samples of the filtered signal w[n].

Key Commands:

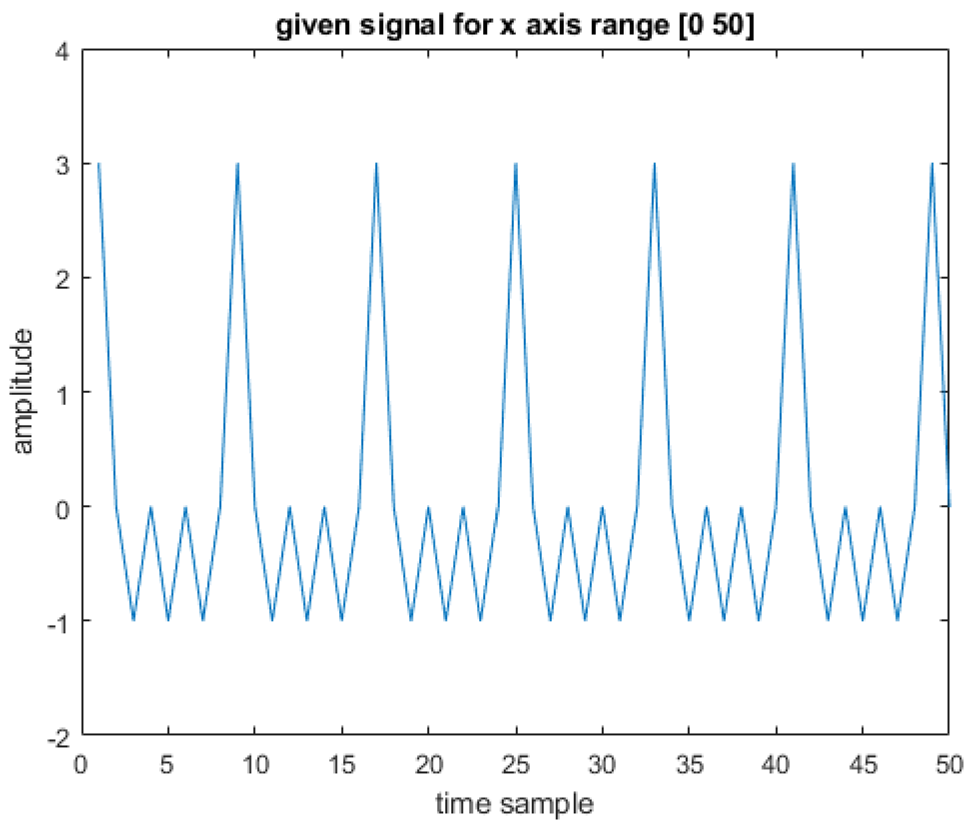
filter % filter(b,a,x) filters the input data x using a rational transfer function defined by the numerator and denominator coefficients b and a.

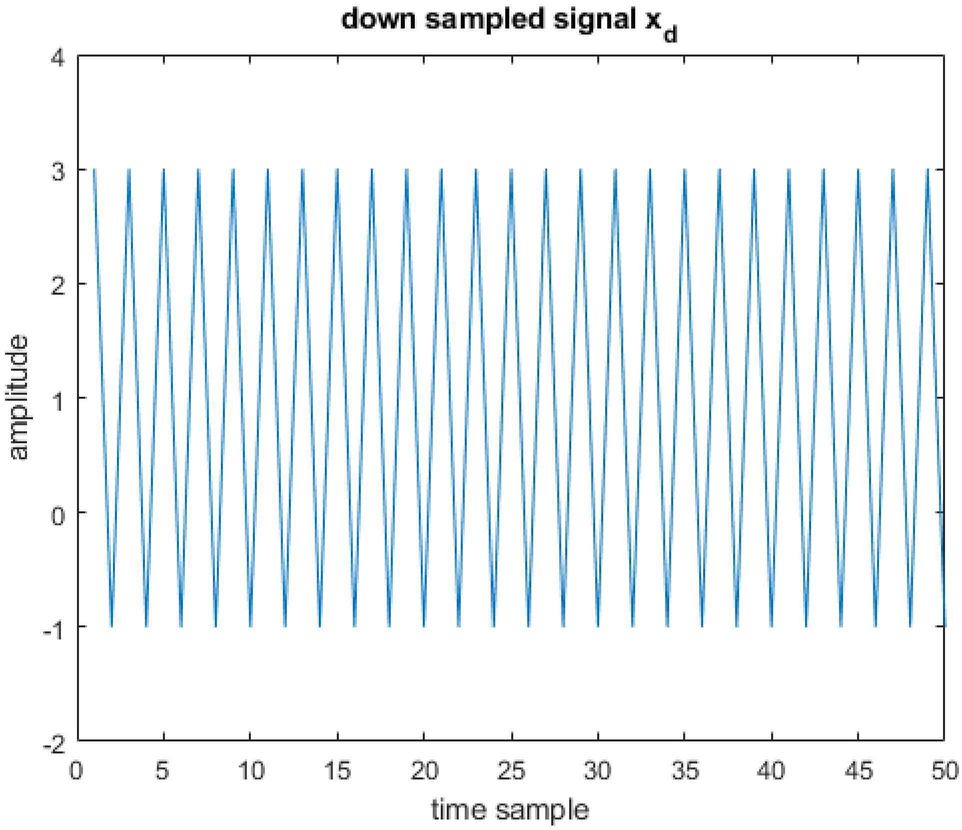
fir1 %fir1(n,Wn) uses a Hamming window to design an nth-order lowpass, bandpass, or multiband FIR filter with linear phase.

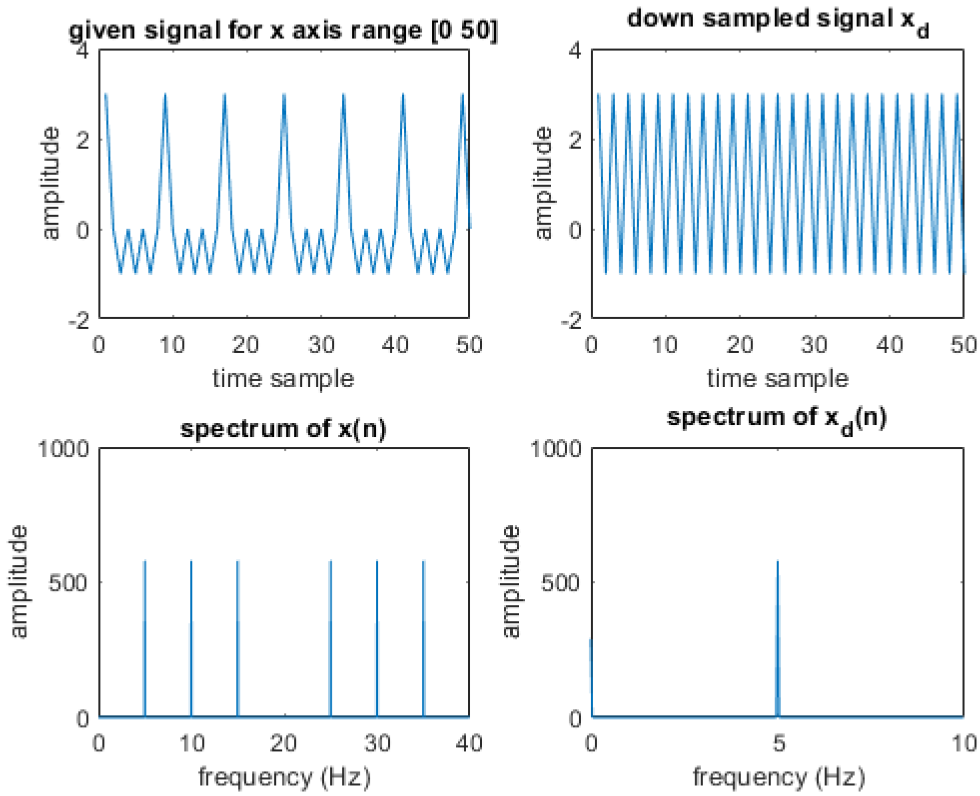
cos %cos(X) returns the cosine for each element of X

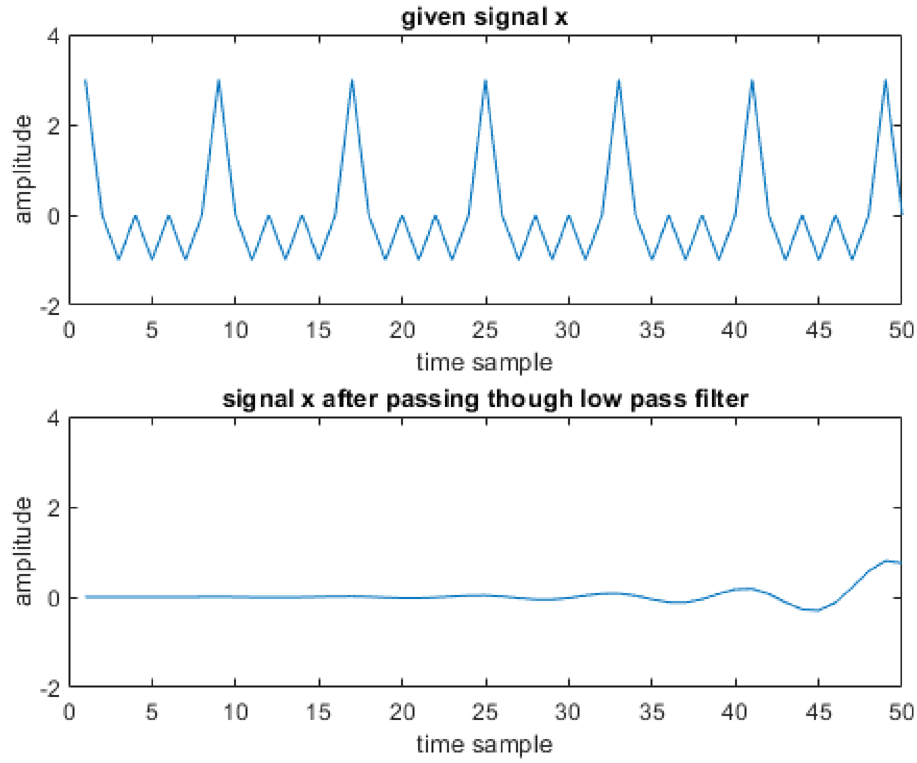
fft %fft(X) computes the discrete Fourier transform (DFT) of X using a fast Fourier transform (FFT) algorithm.

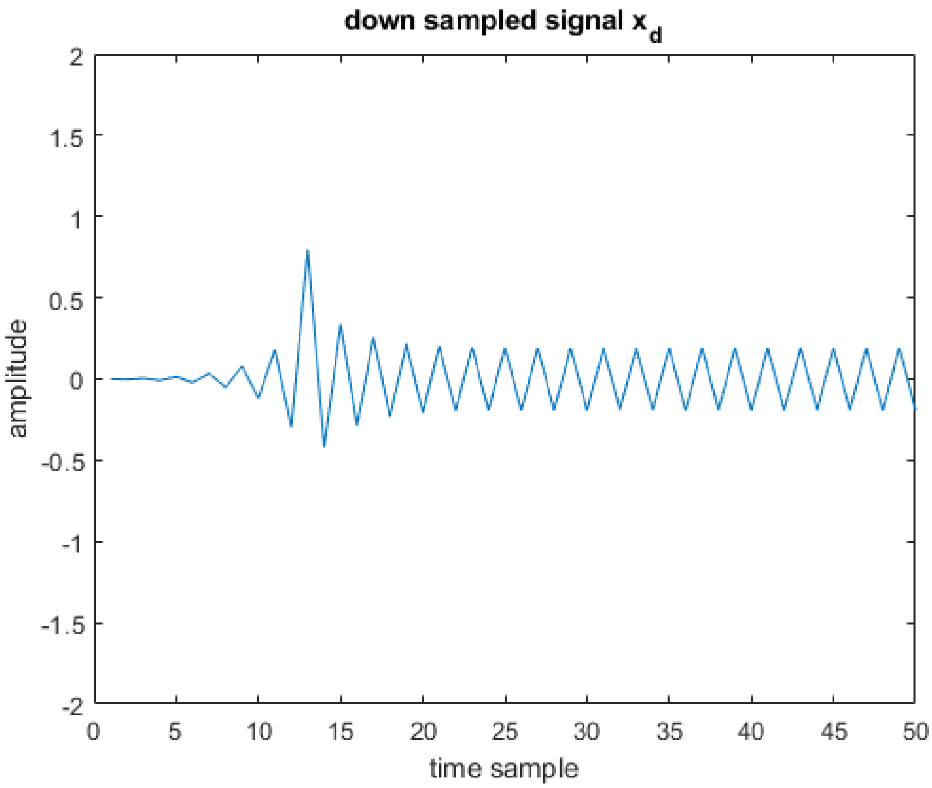
Plots:

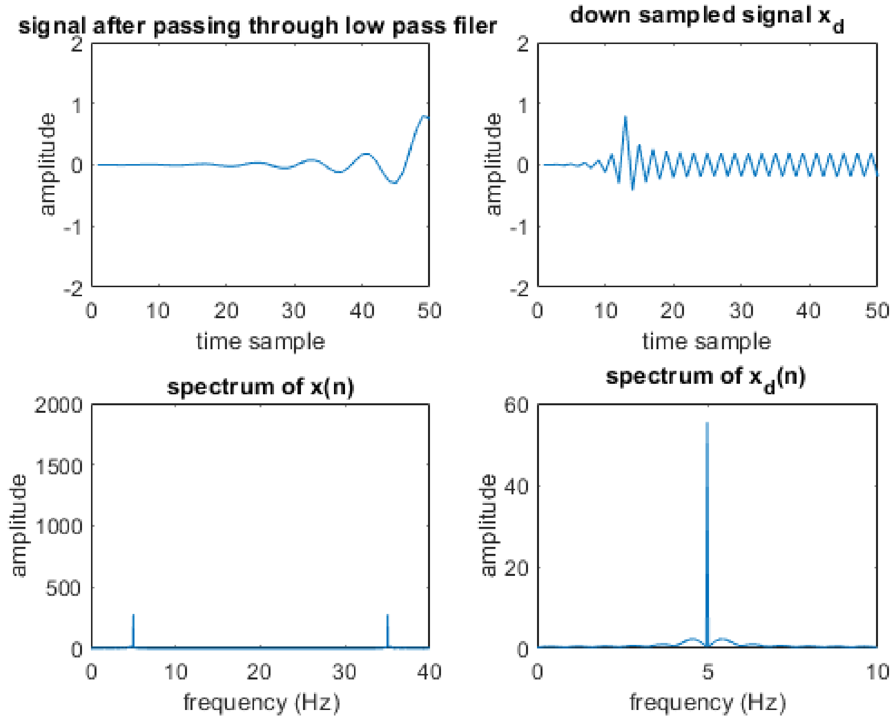



down sampled signal x_d
amplitude
time sample


given signal for x axis range [0 50]
down sampled signal x_d
spectrum of x(n)
spectrum of x_d(n)
amplitude
time sample
frequency (Hz)


given signal x
amplitude
time sample
signal x after passing though low pass filter
amplitude
time sample


down sampled signal x_d
amplitude
time sample



Inferences/comments:

1) sampling frequency after down sampling is Fs/4=10Hz.
2) aliasing occur after the down sampling by 4.
3) We can recover the low frequency 5 kHz signal only when low pass filtering of x(n) with cut of frequency 5 (order 100) and then down sample by 4.
4) Low pass filtering in frequency domain is more accurate because we can exactly select the window with a suitable cut off frequency.

Q3.

(a) Let the continuous signal xc(t) = cos(10πt) is sampled at 15 Hz to obtain x(n). Perform up sampling by 5 to obtain xi(n) and compare x(n) and xi(n) and their spectrum. What is sampling frequency after up sampling.

(b) Now perform low pass filtering with cut of frequency 7.5 to get back the original signal spectrum.

(c) Perform low pass filtering in frequency domain, instead of using filter command as above. Which filtering is preferable?

AIM: To generate the discrete signal by sampling the given signal with a sampling frequency 15Hz. Perform up sampling by 5 to obtain xi(n) and compare x(n) and xi(n) and their spectrum. Perform the appropriate filtering using to get the original signal and compare the spectrums.

Short Theory:

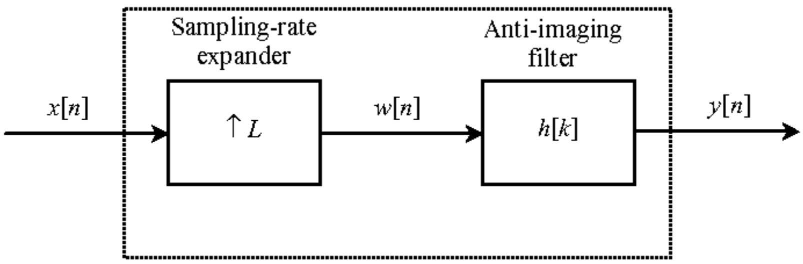
Multi-rate signal processing: The process of converting a signal from a given rate to a different rate is called sampling rate conversion. Systems that employ multiple sampling rates in the processing of digital signals are called multi rate digital signal processing.

up-sampling: Increasing sampling rate of a signal by an integer factor I is known as Interpolation or up-sampling. An increase in the sampling rate by an integer factor I may be done by interpolating (I-1) new samples between successive values of the signals.

- **Interpolation**

– Increase the sampling rate of a discrete-time signal.

– Higher sampling rate preserves fidelity



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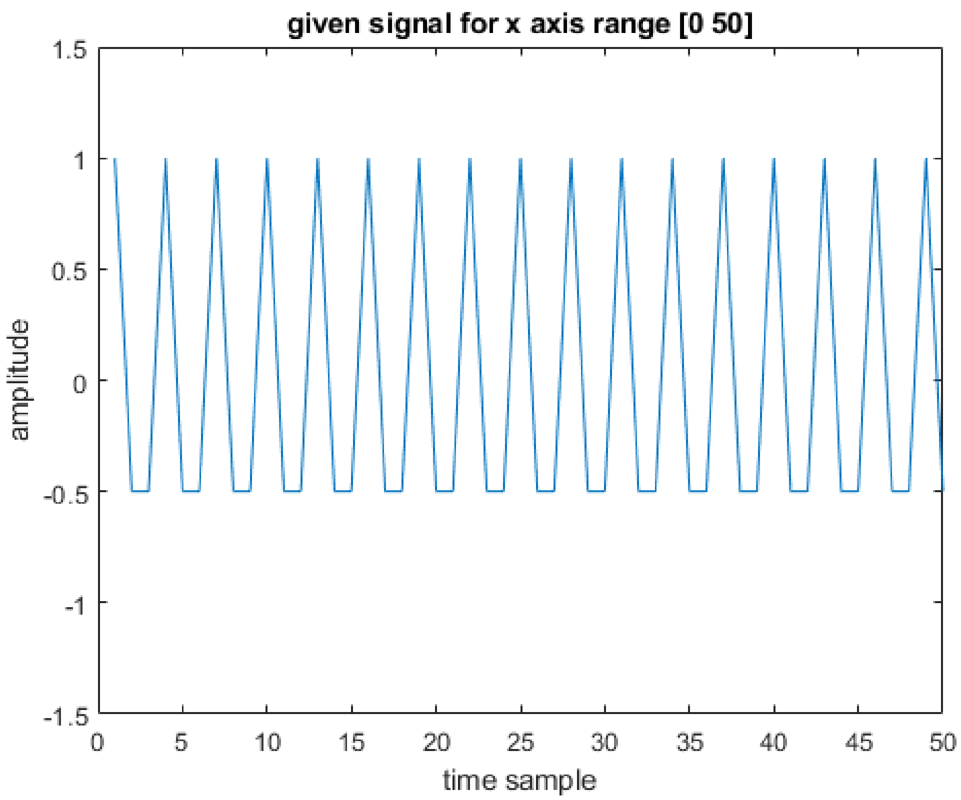
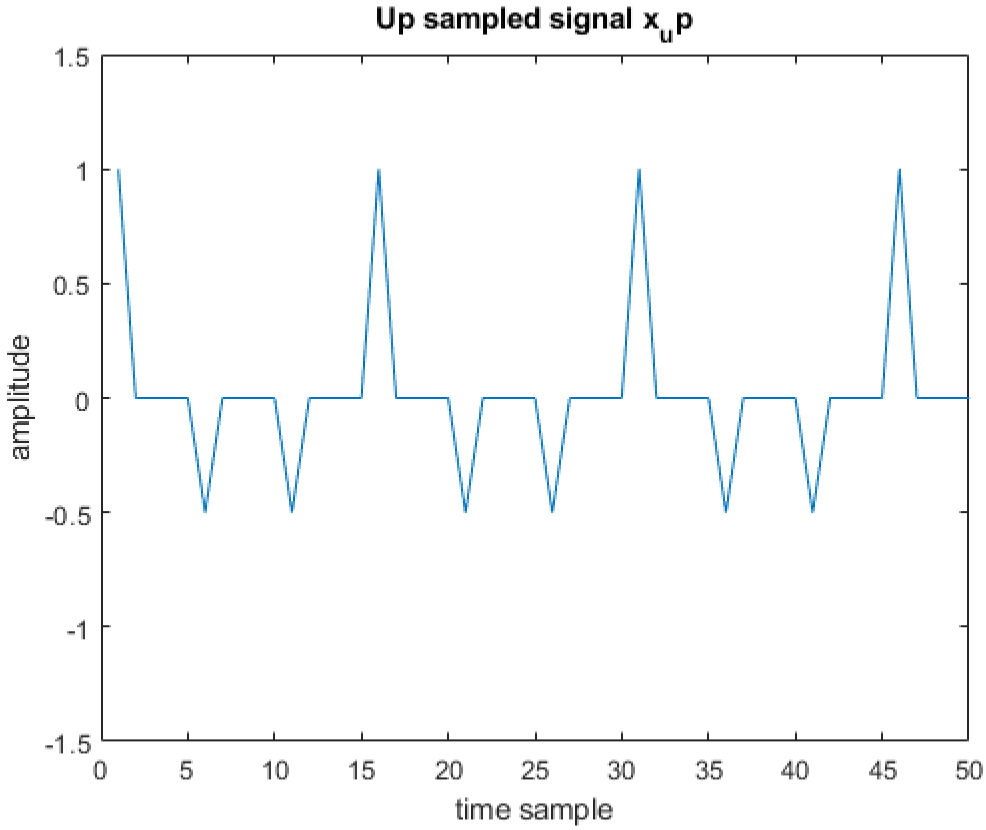
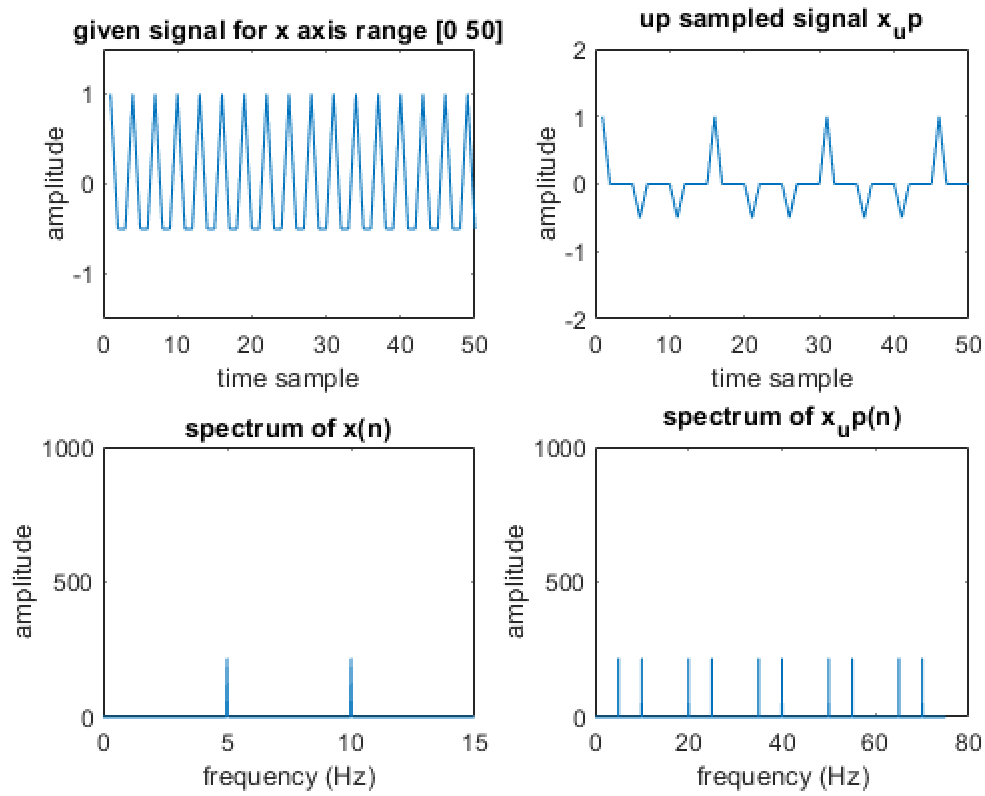
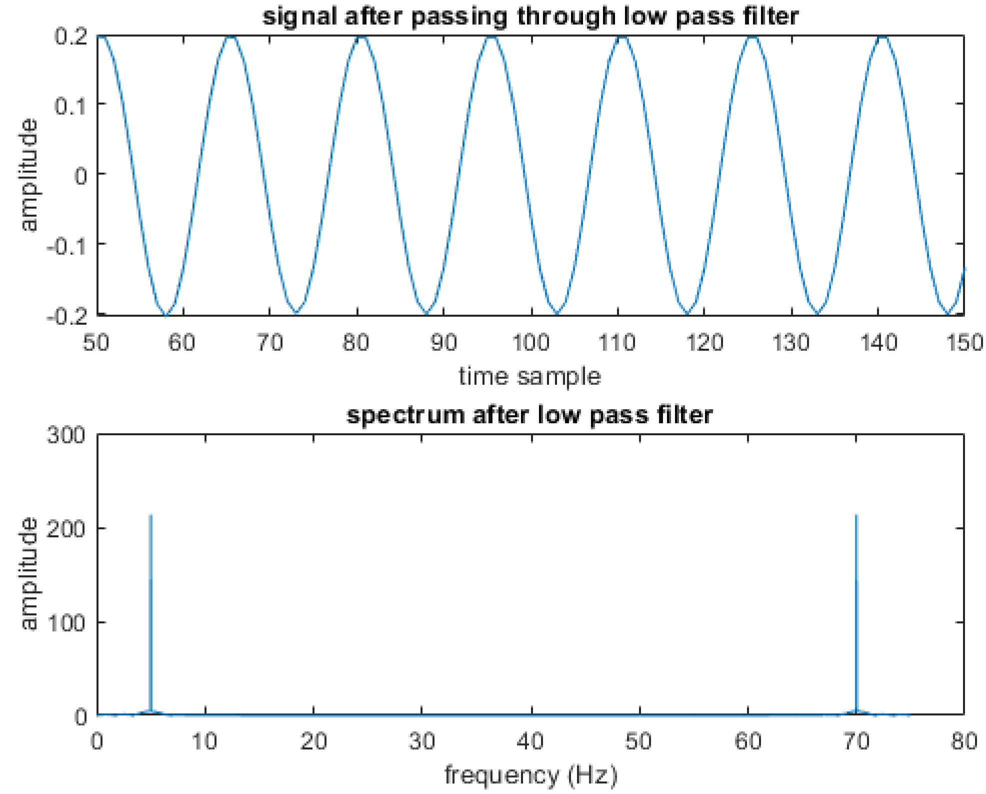
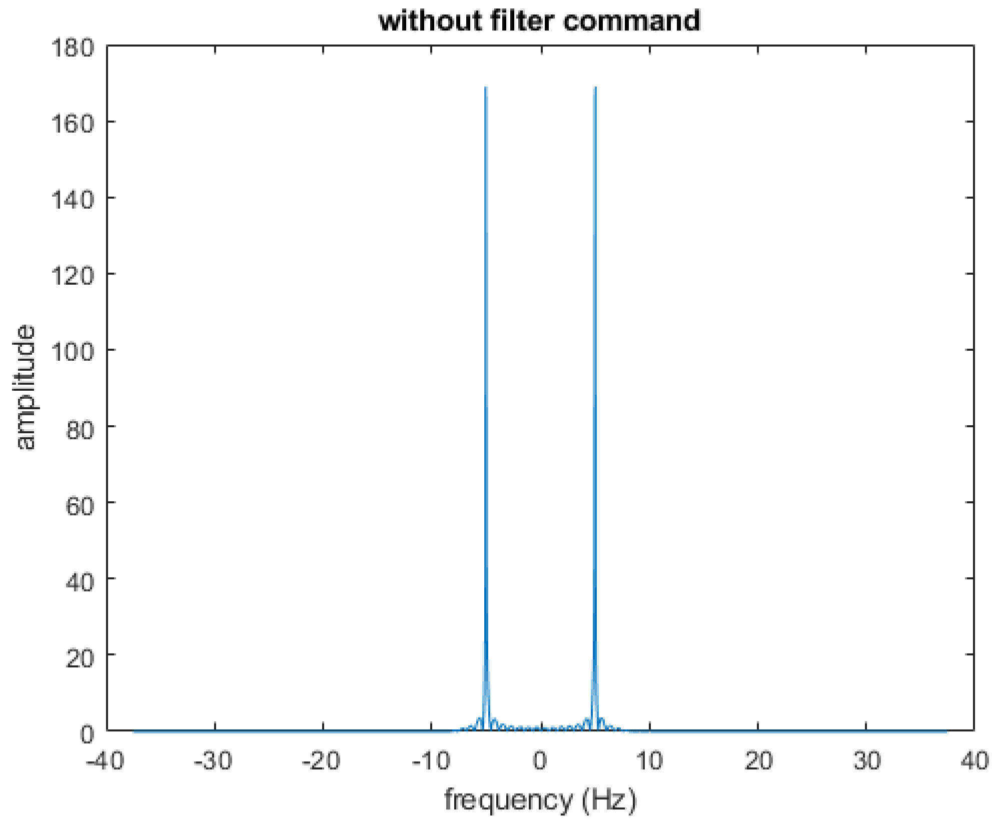
Plots:

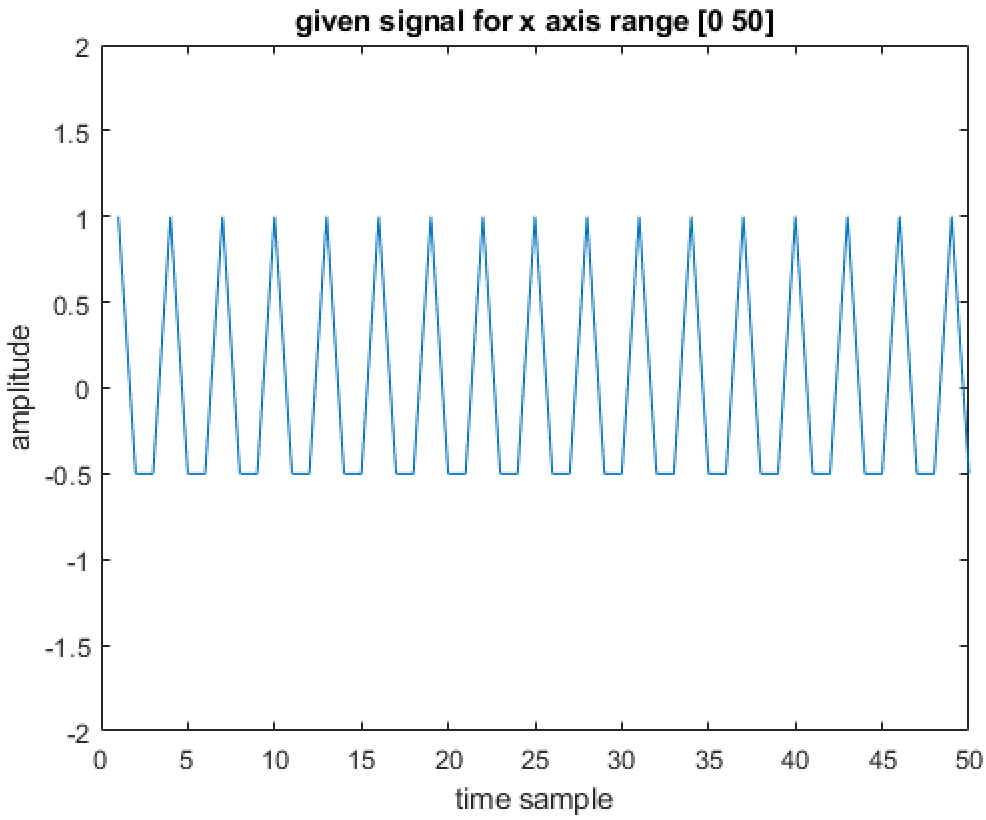


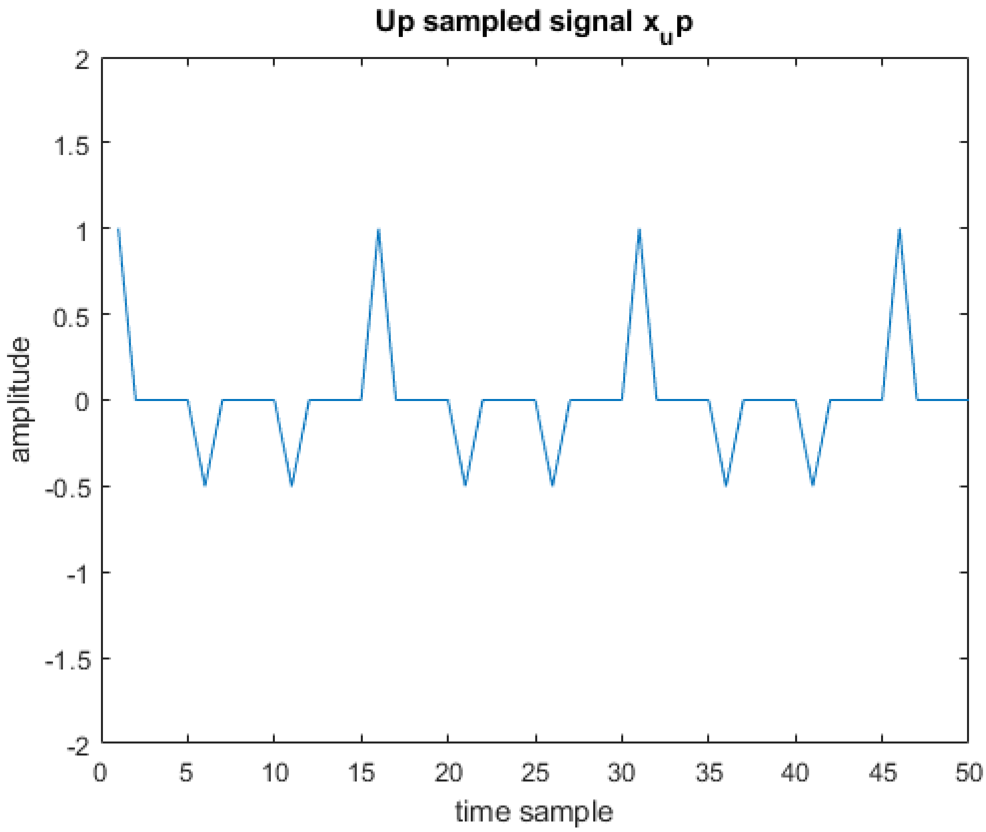

Up sampled signal x_u p
amplitude
time sample

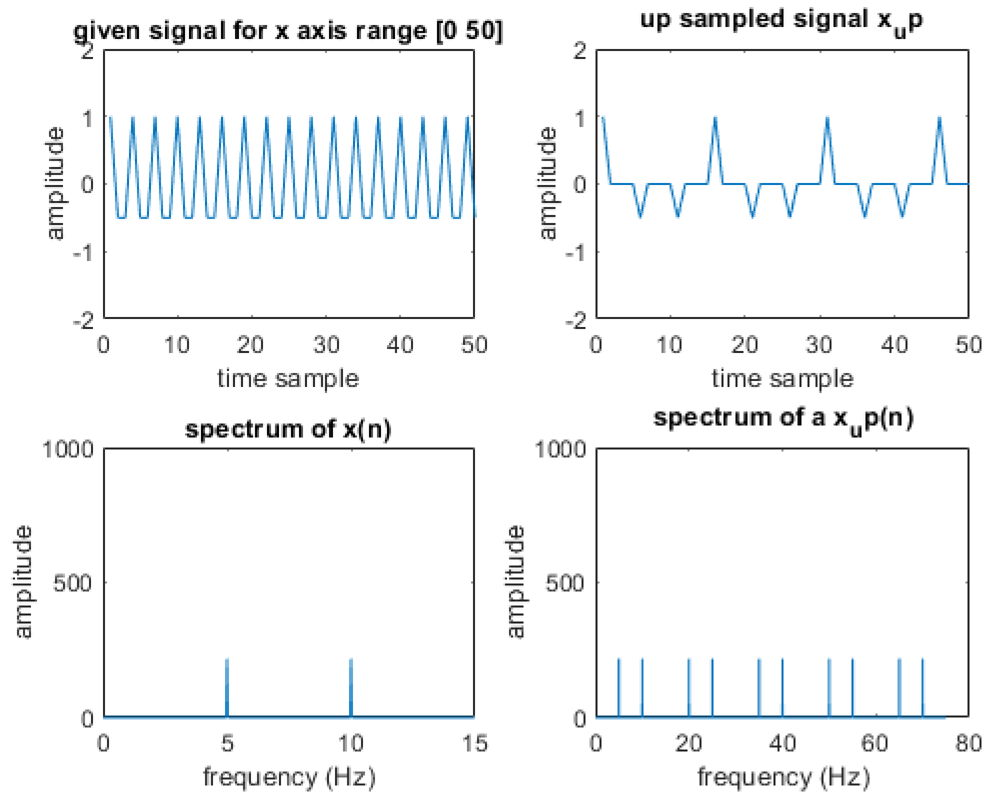

given signal for x axis range [0 50]
up sampled signal x_u p
spectrum of x(n)
spectrum of x_u p(n)
amplitude
time sample
frequency (Hz)


signal after passing through low pass filter
amplitude
time sample
0.2
0.1
0
-0.1
-0.2
50
60
70
80
90
100
110
120
130
140
150
spectrum after low pass filter
amplitude
frequency (Hz)
300
200
100
0
0
10
20
30
40
50
60
70
80


without filter command
amplitude
frequency (Hz)
180
160
140
120
100
80
60
40
20
0
-40
-30
-20
-10
0
10
20
30
40


given signal for x axis range [0 50]
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
amplitude
0
5
10
15
20
25
30
35
40
45
50
time sample


Up sampled signal x_u p
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
amplitude
0
5
10
15
20
25
30
35
40
45
50
time sample



Inferences/comments:

1) sampling frequency after up sampling is Fs*5=75Hz.
2) After the up sampling the spectrum of original signal repeats in frequency domain because of interpolation in time domain.
3) Low pass filtering in frequency domain is more accurate because we can exactly select the window with a suitable cut off frequency.