**DIGITAL SIGNAL PROCESSING**

**LAB**

**ADDITIONAL CYCLE – 1**

**SUBMITTED BY:**

**BATCH B-6**

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1. Sampling: Program below illustrates the sampling of a continuous time sinusoid of frequency 13 Hz. Since Matlab cannot strictly generate a continuous time signal, we simulate a continuous time signal by sampling it at a very high rate Th=0.0001 sec. A plot of the samples using plot command will then look like a continuous time signal. Since the frequency of the continuous time signal is 13Hz, the Nyquist rate is T=1/26. Run the program for several values of T, below and above the Nyquist rate such as 1/(8\*13), 1/(6\*13), 1/(4\*13), 2/(7\*13), 1/(3\*13), 2/(5\*13), 4/(9\*13), 1/(2\*13), 2/(3\*13), 4/(5\*13), 1/13, etc. . (all rational multiples of 1/13 so that the resulting discrete time signal is periodic) Which of the discrete time signals will give correct reconstruction of the continuous time signal? Mathematically determine the angular frequency in rad/sample and the fundamental period N of each of the discrete time signals and verify from the plots.

* Sampling Theorem states, a continuous time signal can be represented in its samples and can be recovered back/reconstructed when sampling frequency fs is greater than or equal to the twice the highest frequency component of message signal.

i.e., Nyquist frequency of sampling is (2\*13)Hz, fn= 26Hz

The Nyquist rate here is T=1/26.

* All those discrete signals obtained by sampling the given sinusoid with sampling frequency greater than or equal to the nyquist frequency can be used to reconstruct the original sinusoid.

The discrete-time signals when sampled with Ts= 1/(8\*13), 1/(6\*13), 1/(4\*13), 2/(7\*13), 1/(3\*13), 2/(5\*13), 4/(9\*13), 1/(2\*13); will give correct reconstruction of the given sinusoid.

* N(samples) – Fundamental Period is the smallest positive integer for which the signal repeats.

Discrete-time angular frequency (Ω) = 2/N (radians/sample)

|  |  |  |
| --- | --- | --- |
| Ts(Sampling Time) | Ω (Angular Frequency) | N (Fundamental Period) |
| 1/(8\*13) | /4 | 8 |
| 1/(6\*13) | /3 | 6 |
| 1/(4\*13) | /2 | 4 |
| 2/(7\*13) | /7 | 7 |
| 1/(3\*13) | 2/3 | 3 |
| 2/(5\*13) | 25 | 5 |
| 4/(9\*13) | 29 | 9 |
| 1/(2\*13) |  | 2 |
| 2/(3\*13) | /3 | 3 |
| 4/(5\*13) | 2/5 | 5 |
| 1/13 |  | 1 |

Inference:

Fundamental periods of each of the discretized signals are verified from the plots. From the plots, it can be verified that the signal repeats after the mathematically calculated fundamental period.

1. The family of continuous time sinusoids cos(Ω0+kΩs)t , k=0,±1, ±2..... where Ωs=2π/T leads to identical sampled sequences.( Prove) This phenomenon of a continuous time sinusoid of a higher frequency acquiring the identity of a sinusoidal sequence of lower frequency after sampling is called aliasing. Consider the continuous time sinusoid g1(t)=cos(6πt). When sampled with T=0.1sec, it will lead to cos[0.6πn]. When sampled with T=0.1sec, g2(t)=cos(26πt) will also lead to cos[0.6πn]. (Verify mathematically). Run the program below to see aliasing in this case.

* Let x(t)= cos(Ω0+kΩs)t , k=0,±1, ±2.....where Ωs=2π/T

x(t)= cos(Ω0t+k t2π/T) = cos(2π(fs\*k\*t)+ Ω0t) = cos(Ω0t)

This implies that the family of continuous time sinusoids cos(Ω0+kΩs)t , k=0,±1, ±2.....where Ωs=2π/T leads to identical sampled sequences

* When g1(t)=cos(6πt) is sampled with T=0.1 sec, i.e., t=nT this implies, discretized signal g1[n] = cos(6πn\*0.1) = cos(0.6πn);

Nyquist frequency for g1(t) [cos(2π\*3\*t)] is 6Hz, i.e., Fs=10Hz >nyquist frequency. So, there will be no aliasing and the original signal can be reconstructed from the obtained discretized signal.

* When g2(t)=cos(26πt) is sampled with T=0.1 sec, i.e., t=nT this implies, discretized signal is g2[n] = cos(26πn\*0.1) = cos(2.6πn) = cos(2π+0.6πn) = cos(0.6πn) (cosine is periodic with the period 2πn).

Nyquist frequency for g1(t) [cos(2π\*13\*t)] is 26Hz, i.e., Fs=10Hz <nyquist frequency. So, there will be aliasing effect and the original signal cannot be retrieved from the obtained discretized signal.

Verification to see aliasing:



* The sampled values of cos(26πt) are same as that of cos(6πt), when these signals are reconstructed back cos(26πt) will be wrongly retrieved.

1. The filter() function : The filter() command recursively computes the output y(n) of an LTI system described by a difference equation from the input x(n) and initial conditions. b=[b0,b1,.....bM ]; a=[ a0,a1,......aN]; y=filter(b,a,x) . The number of output values in y correspond to the number of input values in x. (See help on filter for more details)
2. Run the code fragment below to determine the first 50 values of the output of the system described by y(n)-1.143y(n-1)+.4128y(n-2)= 0.0675x(n) +0.1349x(n-1)+0.675x(n-2) if the initial conditions are zero and x(n)=.2u(n).

a=[1 -1.143 0.4128 ]; b=[0.0675 0.1349 0.675];   
y=filter(b,a,2\*ones(1,50)); stem(0:49,y)

* The first 50 values of the output of the system are:

0.0135, 0.056, 0.233, 0.42, 0.558, 0.641, 0.677, 0.685, 0.679, 0.669, 0.659, 0.6533, ……..

The Plot is as follows:



1. Using filter(), determine and stem the first 41 samples of the impulse and step response of the system described by y(n)-ay(n-1)=x(n) for a=0.8 and -0.8, Verify that the step response is the running sum of the impulse response.

* First 41 values of the Impulse and Step response of the given system for **a=0.8**, using filter function-



* First 41 values of the Impulse and Step response of the given system for

**a= -0.8,** using filter function-



1. Run the following program to generate output using both conv() and filter():

h = [3 2 1 -2 1 0 -4 0 3]; % impulse response

x = [1 -2 3 -4 3 2 1]; % input sequence

y = conv(h,x);

n = 0:14;

subplot(2,1,1);stem(n,y);

xlabel('Time index n'); ylabel('Amplitude');



Inferences:

The values are verified to be the same in both the subplots generated using convolution function and filter function.