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Analysis of Sampling Rate and Aliasing on MATLAB

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ABSTRACT: Sampling is the process of converting continuous time signal into discrete time signal. Sufficient number of samples must be taken in order to properly reconstruct the signal from its sampled version. The sufficient number of samples needed is determined by the Nyquist-Shannon theorem. The theorem states that the perfect reconstruction of an analog signal is possible when the sampling frequency is greater than twice the maximum frequency of the signal being sampled. Without any spectral overlap or aliasing a DSP system can sample a analog signal with its highest frequency up to $f_s/2$. In this paper $f_s=5000$ Hz and input sine waves are of 500,2000,3000 and 4500 Hz frequencies. These signals are sampled and aliasing effect can be seen in 3rd and 4th signal i.e. 3000 and 4500 Hz , they are above $f_s/2$ and they will be changed by aliasing effect to -2000 and -500 Hz.

I. INTRODUCTION

Prior to sampling we should pass the input signal through a low pass filter(cutoff frequency f_{max}) in order to restrict it to a maximum frequency f_{max} .of input signal. Aliasing is the appearance of phantom frequencies when a signal is not sampled at a high enough rate. $-\pi \leq w_i \leq +\pi$ that is $-1/2 \leq f_i/f_s \leq +1/2$ since $|f|=1/2$ or $|w| = \pi$ is the highest frequency in a discrete time signal. The condition $f_s > 2f_{max}$, where f_{max} is the largest frequency component in the analog signal, make sure all sinusoidal components in the analog signal are mapped into discrete time frequency components with frequencies in the fundamental interval. Let us take an example we have $f_s=5000$ samples/sec and input signals are 500, 2000,3000 and 4500 Hz. We have to discretized the input signals with length of samples $N=100$.The outputs are shown below in Fig.-1

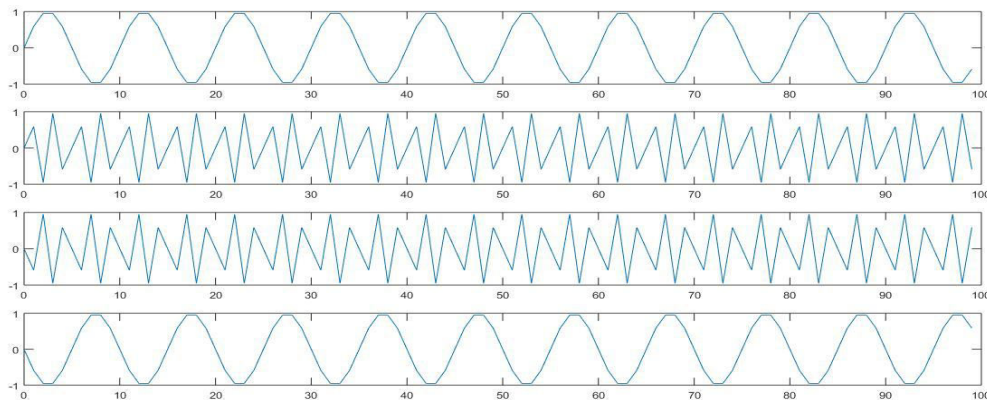


Fig.-1 shows discretized outputs with input signals having frequencies of 500,2000,3000 and 4500Hz and $f_s=5000$ samples/sec length $N=100$.3rd and 4th signal are changed due to aliasing effect.

we observe that first signal is having digital frequency of 0.1 that is having 10 cycles, similarly digital frequency of second signal is 0.4 that is 40 cycles. We note that digital frequency of third signal is 0.6, that is means 40 cycles but



exactly opposite phase of second signal. Digital frequency of fourth signal is 0.9, that is 10 cycles but exactly opposite phase of first signal. Third and fourth Signal are changed due to aliasing effect to -2000 and -500 Hz. As shown in Fig.-2 spectrum of signals clearly show that 3rd and 4th signal are alias of 1st and 2nd signal.

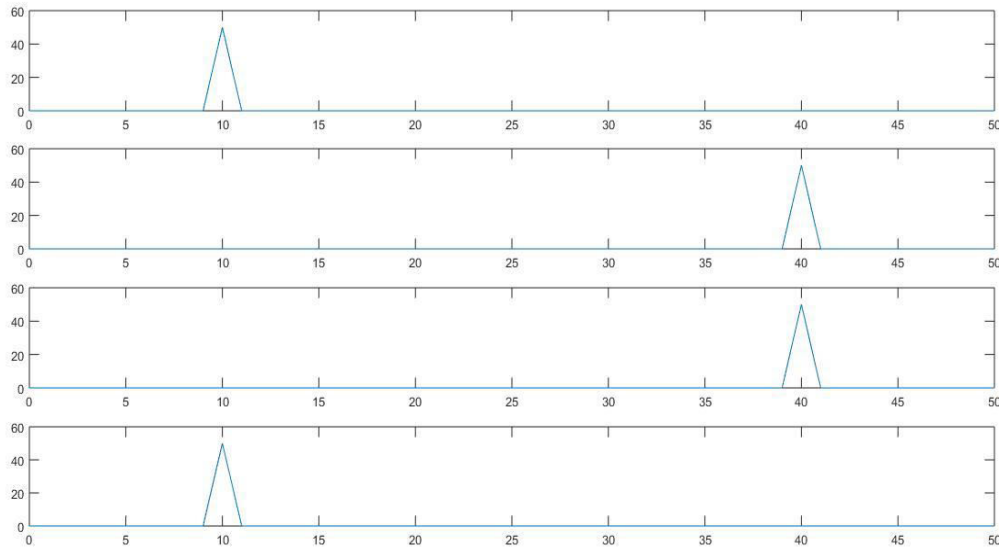


Fig.-2 shows spectrum of discretized output with input signals having frequencies of 500,2000,3000 and 4500Hz and $f_s=5000$ samples/sec length $N=100$.

II. METHODOLOGY

Consider sinusoidal signals of frequencies from 1 Hz to 10 Hz with sampling frequency of 10 Hz. Nyquist frequency or folding frequency is 5 Hz i.e. $f_s/2$. Now signals having frequencies less than 5 Hz can be recovered from their sampled version since $f_s > 2f_{max}$. Similarly signals having frequencies greater than 5 Hz can not be recovered from their sampled version since $f_s < 2f_{max}$.

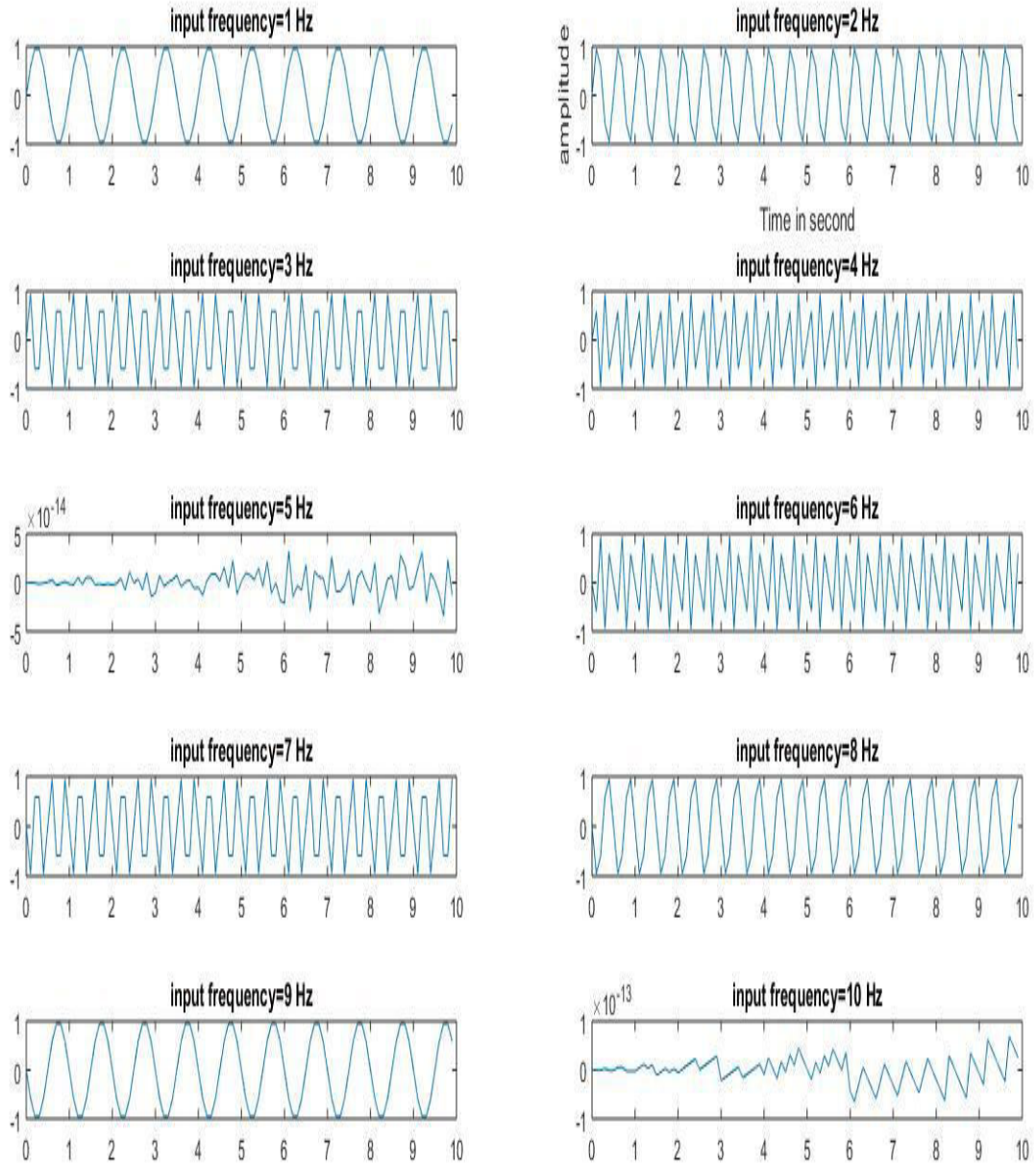


Fig. 3 shows discretized outputs with input signals having frequencies of 1,2,3—10 Hz and $f_s=10$ samples/sec length $N=100$.

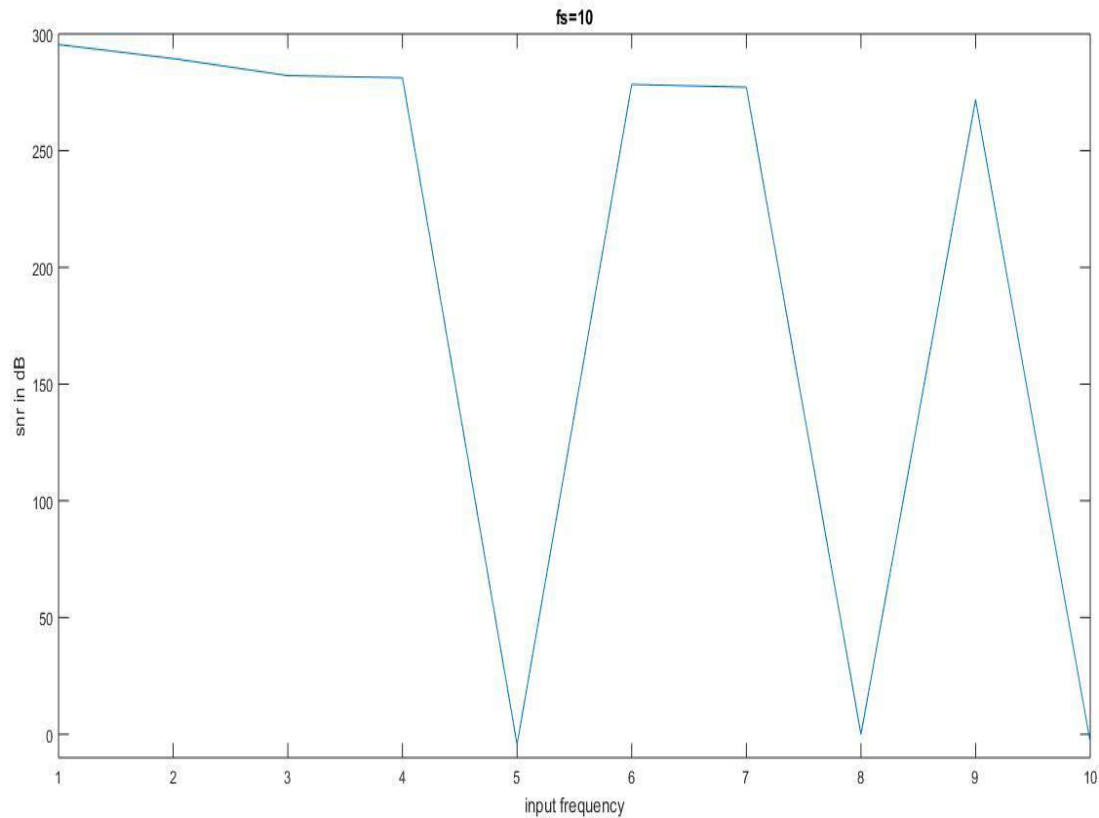


Fig. 4 shows plot between input signals having frequencies of 1,2,3..10 with $f_s=10$ samples/sec length $N=100$ and SNR.

III. OBSERVATION

We observe in figure 3 that at $f_i=5,8,10$ signals shows attenuation and distortions. Signal at $f_i=1,2,3,4$ Hz and at $f_i=9,8,7,6$ Hz are opposite in phase. It means signals at 6,7,8 and 9 Hz are above $f_s/2$ and due to aliasing changed to -4,-3,-2 and -1 Hz

IV. RESULT

For proper recovery of signal after sampling it is necessary to have $f_i < f_s/2$ and aliasing changes signals above $f_s/2$ to -4,-3,-2 and -1 Hz.

V. CONCLUSIONS

Sampling must be done with $f_i < f_s/2$. Due to aliasing signals are distorted at $f_i=5,8$ and 10 Hz. signal having digital frequency of 0.1 is having 10 cycles, similarly digital frequency of second signal is 0.2 that is 20 cycles. It means signals at 6,7,8 and 9 Hz are above $f_s/2$ and due to aliasing changed to -4,-3,-2 and -1 Hz



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