EE451 LAB REPORT

LAB 1 – Sampling & Quantization

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1.1 Sampling

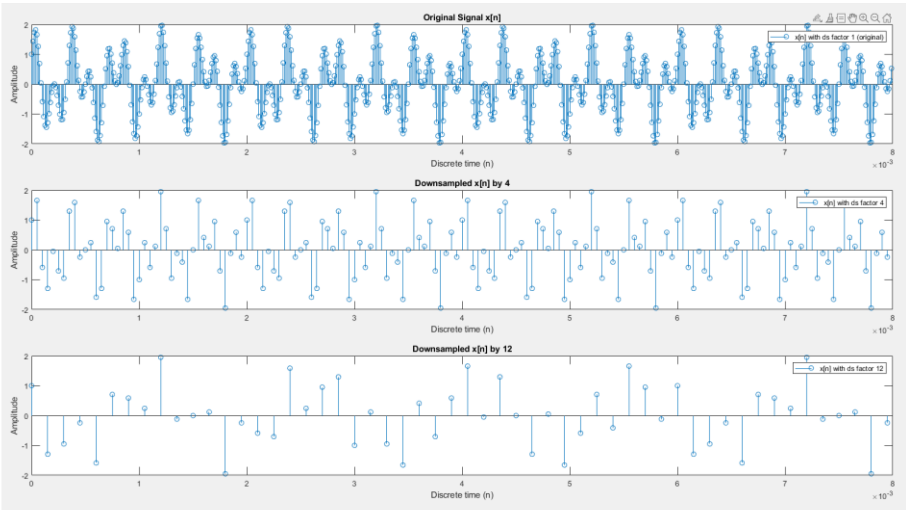

Figure 1: Original Signal and the Downsampled Signals in Time Domain

- Increasing downsampling factor effects sample period directly propotionally.

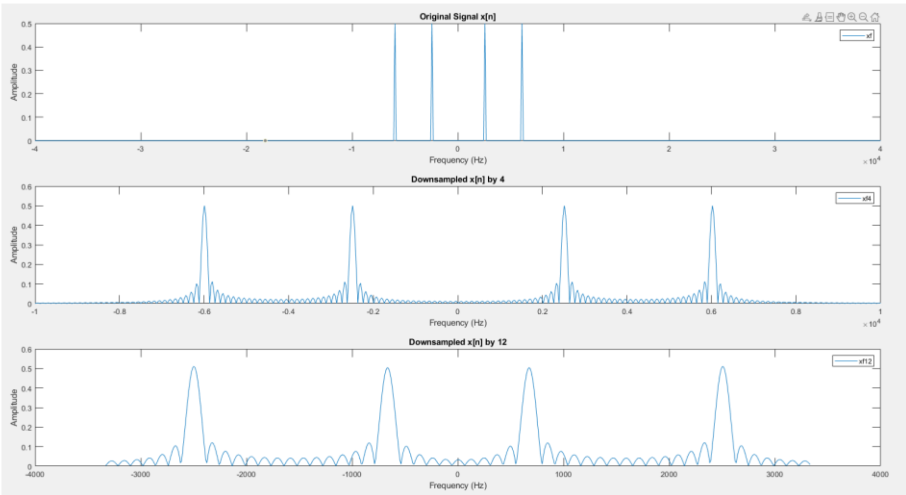

Figure 2: Original Signal and the Downsampled Signals in Frequency Domain

- It is observed that sample frequency value is decreased by increasing downsampling factor.

Comment

- According to the relation:
- Ts = 1 / Fs,
  sample period and sample frequency are inversely proporitonal to each other.

```
nvec = 0:Ts:(Td-Ts);
nvec4 = 0:(Ts*4):(Td-Ts);
nvec12 = 0:(Ts*12):(Td-Ts);
```

The code lines above is for downsampling operation. The size of an time array is rearranged for true figuration. (Td = Duration time, Ts = Sample period)

```
x4_n = x_n(mod((1:length(nvec)), 4) == 1);
x12_n = x_n(mod((1:length(nvec)), 12) == 1);
```

Every value of original vector time array by the modulo of corresponding downsampling factor number is taken and written in the new array for downsampled signals.

Code Fix

- The operation in linspace(.) and magnitude response is changed to display frequency domain in a right way. This difference provided rearranged frequency ranges in each subfigure in figure 2. The fixed code provides true frequency ranges and normalization.

```
FVec = linspace(-Fs/2, Fs/2, N);
FVec4 = linspace((-Fs/2)/4, (Fs/2)/4, N);
FVec12 = linspace((-Fs/2)/12, (Fs/2)/12, N);
xf = abs(fftshift(fft(x_n,N)))/N;
xf4 = abs(fftshift(fft(x4_n,N)))/N_4;
xf12 = abs(fftshift(fft(x12_n,N)))/N_12;
```

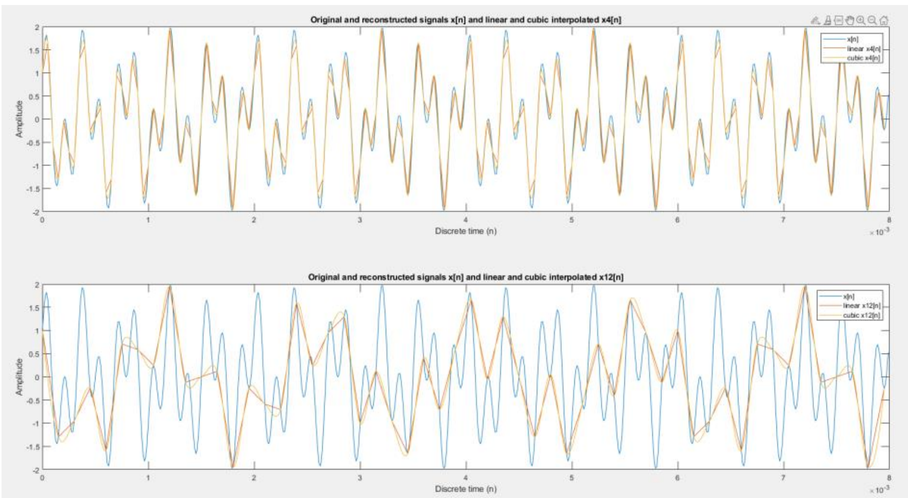

Figure 3: Linearly and Cubic Interpolations of Downsampled Signals

- It is observed that higher downsampling factor is not effective while reconstructing the original signal. Also it can be understood from the first subfigure that cubic interpolation is more accurate than linear interpolation.

1.2.1 Quantization

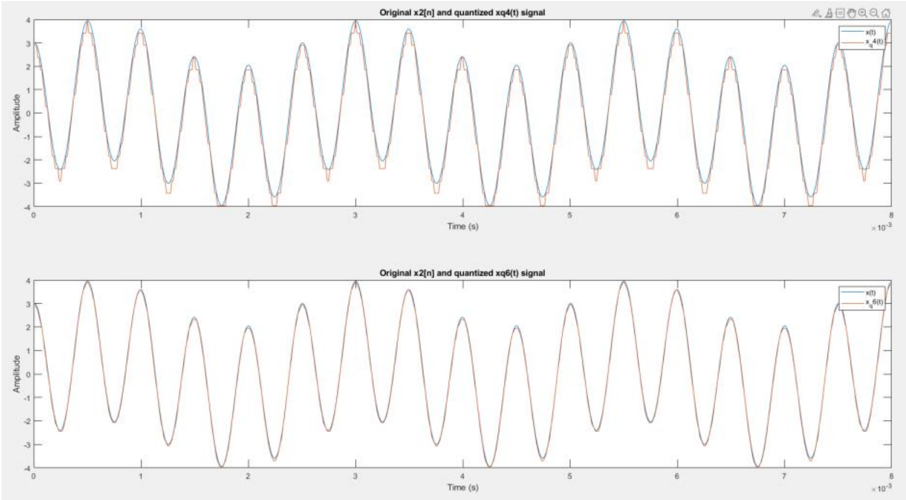

Figure 4: Quantization of x(t) with two different quantization numbers 4 and 6

- Since the number of bits that is used for representation of the original signal as a pulse waveform is increased from 4 to 6 (2^4 to 2^6), accuracy between original and quantized signals also increases.

1.2.2 SQNR$_{dB}$ Calculation

```
SQNR4_dB =

   22.8350

SQNR6_dB =

   35.4057
```

- From the figure from the command window above is Signal-to-Quantization-Noise Ratio is obtained by the following formula:
- SQNR = var(x(t)) / var(x(t) – x_Q(t)),

  where the noise component represents the difference between original signal and quantized signal.

$$\mathrm{SQNR}_{dB} \triangleq 10\log_{10}\mathrm{SQNR}$$

Figure 5: dB Conversion

Comment

- Signal-to-Quantizaton-Noise Ratio shows the power of the signal compared to error. From the figure 4, it is possible to observe the error difference between two quantization numbers visually and SQNR values from the command window verify it. So, higher SQNR value means higher accuracy in quantization. Quantization with k=6 is more effective while reconstructing the original signal compared to k=4.

1.3 Remaining Questions for the Report

Q1. What is the Nyquist frequency, explain and give the formula.

A1. The Nyquist frequency is half of the sampling frequency and represents the maximum frequency that can be accurately captured in a digital signal. It ensures that frequencies in the original analog signal up to the Nyquist frequency can be faithfully reconstructed from the digital samples without distortion or aliasing. In formula:

- Nyquist Frequency ($F_{Nyquist}$) = Sampling Frequency (F_s) / 2

Q2. Explain the sampling theorem, in which case(s) the perfect reconstruciton of the signal from its sampled one is not possible.

A1. The sampling theorem states that to accurately reconstruct an analog signal from its digital samples, you need to sample it at a rate at least twice the highest frequency in the signal. This is known as the Nyquist rate. Perfect reconstruction is not possible when you sample below this rate, as it can result in distortion and loss of information due to a phenomenon called aliasing. In summary, to avoid loss of information and distortion, adhere to the Nyquist rate when sampling analog signals.

Conclusion

- In this lab session, we explored key concepts in signal processing, including the Nyquist frequency, downsampling, quantization, and Signal-to-Quantization-Noise Ratio (SQNR). We learned that adhering to the Nyquist rate is crucial for accurate signal reconstruction. Additionally, we observed the trade-offs between quantization precision and data representation. These fundamental principles provide a solid foundation for digital signal processing in electrical engineering and related fields.