ECEN610 Lab 1

Department of Electrical & Computer Engineering
Mixed-Signal Interface

Name: Xiaohan Xi
UIN: 236002639
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1 FIR and IIR filter

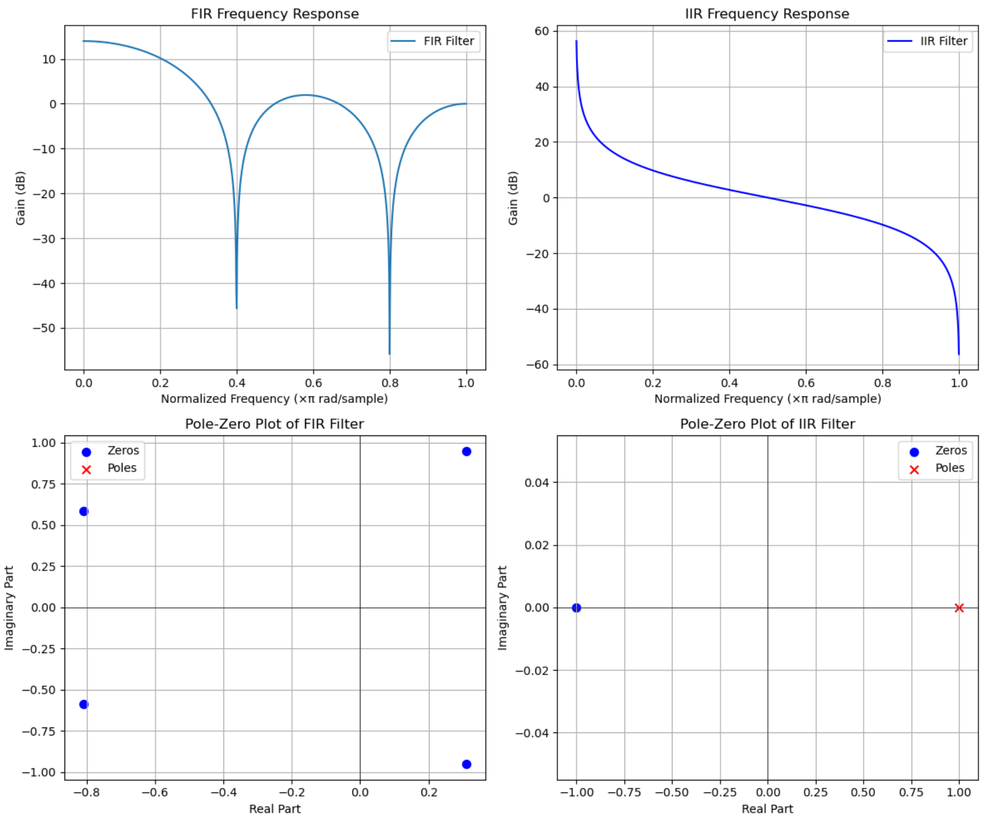

Figure 1.1: Simulation Results

The FIR filter FIR filter has multiple zeros and no poles, making it always stable. Because the denominator is 1. The zeros are on the unit circle. The FIR filter's impulse response decays, confirming its stability. IIR filter has a zero at z= -1 and a pole at z= 1, leading to marginal stability. Only the pole on or in the unit circle.And if the pole lies out of the circle, the system will be unstable.

2 Sampling

- a.

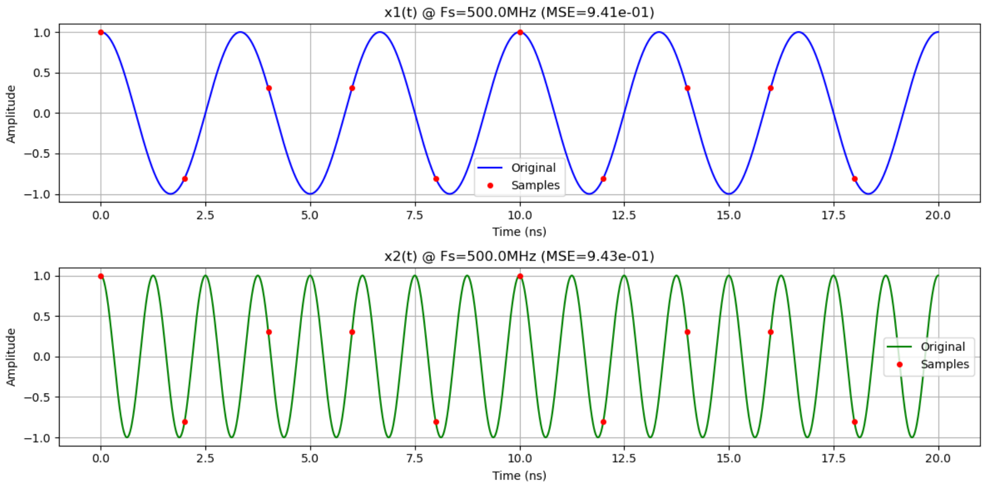

Figure 2.1: Sampling Results

For a signal with $f_1 = 300\,\text{MHz}$, a sampling frequency of $F_s = 500\,\text{MHz}$ will cause aliasing, and the aliased frequency is $200\,\text{MHz}$. Similarly, for a signal with $f_2 = 800\,\text{MHz}$, aliasing will also occur, and the aliased frequency is $200\,\text{MHz}$. The reason for aliasing is that the sampling frequency is lower than the Nyquist frequency.

- b.

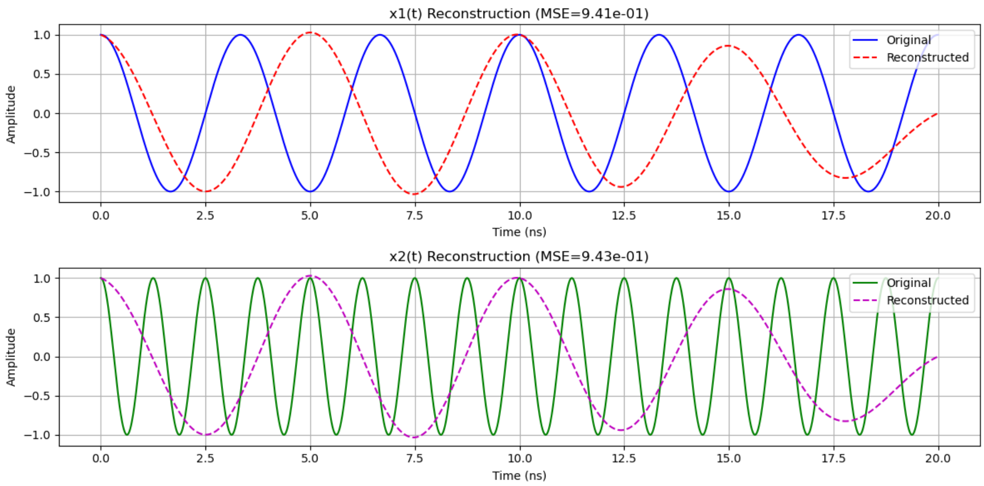

Figure 2.2: Reconstruction Results

We cannot reconstruct the signal by using the sampled signal. And one option we can use is increasing the sampling rate. For example, we can add sampling rate to $1.6GHz$.

- c.
Sampling period T, satisfying Nyquist rate ($F_s = 1/T \geq 2B$, where B is the signal bandwidth). Pulse width $w \leq T$, with sampling points at the end of each pulse. Ideal low-pass filter cutoff frequency $F_c = F_s/2$.

Reconstruction Process:

1. Zero-order hold: Each sample value $x[n]$ is held for duration w.

2. Pass through an ideal low-pass filter $H(f) = \text{rect}\left(\frac{f}{2F_c}\right)$.

Reconstruction Formula:

$$x(t) = \sum_{n=-\infty}^{\infty} x[n] \cdot h(t - nT)$$

where $h(t)$ is the convolution of a rectangular pulse with the ideal low-pass impulse response:

$$h(t) = \left[\text{rect}\left(\frac{t}{w}\right) * \text{sinc}(2F_c t)\right] = \int_{-w/2}^{w/2} \text{sinc}(2F_c(t-\tau))\, d\tau$$

Analytical Solution:

$$h(t) = \frac{1}{2\pi F_c}\left[\text{Si}(2\pi F_c(t + w/2)) - \text{Si}(2\pi F_c(t - w/2))\right]$$

where Si is the sine integral function. When $w = T$, this simplifies to:

$$h(t) = \text{sinc}\left(\frac{t}{T}\right) * \text{rect}\left(\frac{t}{T}\right)$$

- d.

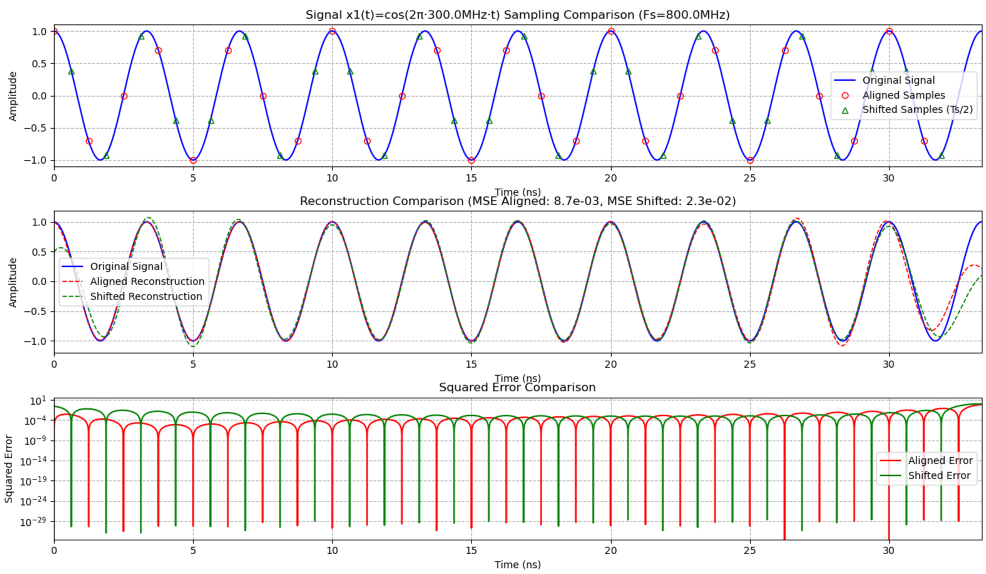

Figure 2.3: Simulation Results

- e.

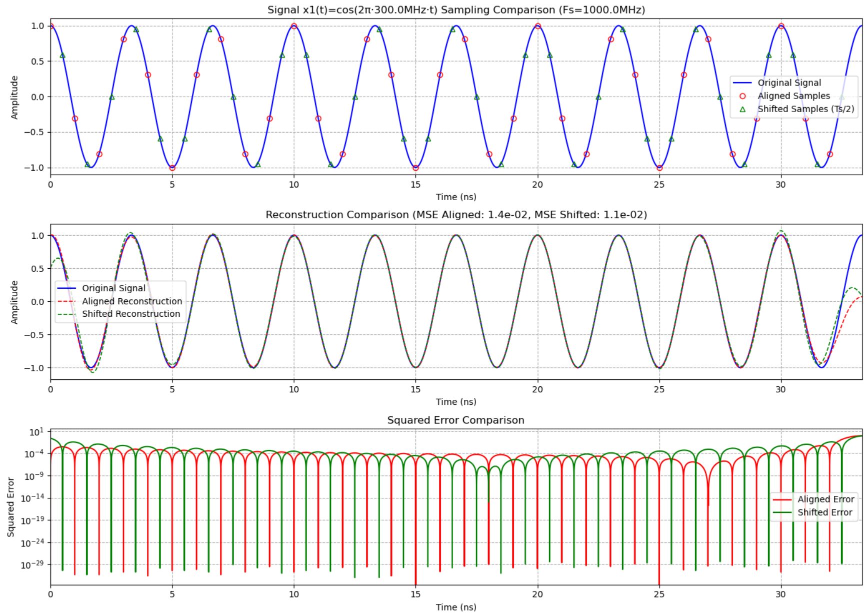

Figure 2.4: Simulation Results

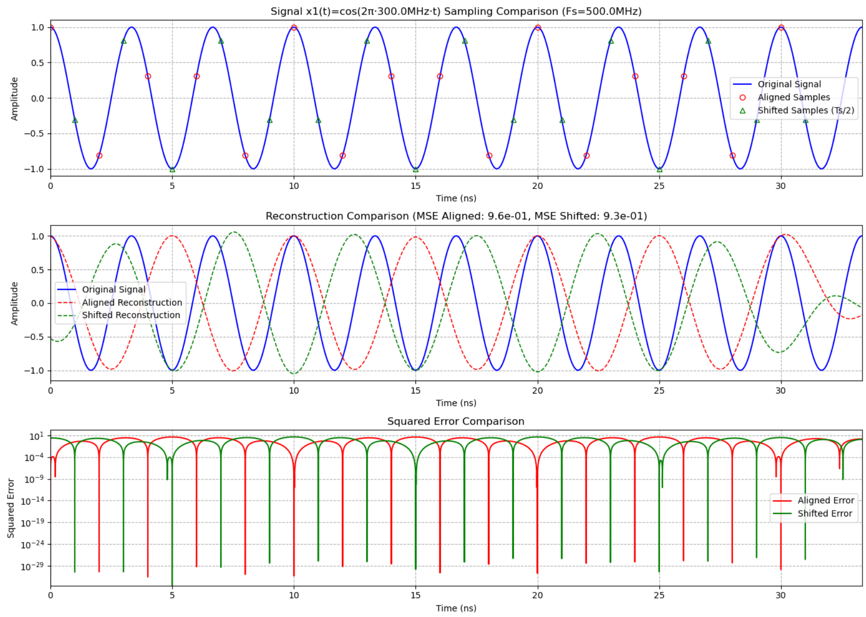

Figure 2.5: Simulation Results

3 Discrete Fourier Transform

- a.Fig. 3.1
- b.Fig. 3.2 I can identify two peaks in the figure. One lies on $200MHz$ and another on $400MHz$.
- c.Fig. 3.3 After adjusting the sampling frequency, the Nyquist frequency is 250 MHz. There-

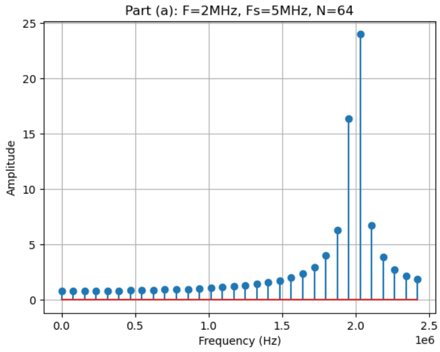

Figure 3.1: Simulation Results

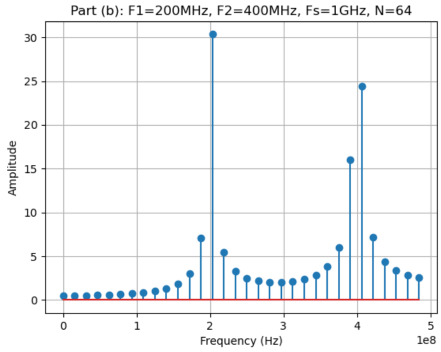

Figure 3.2: Simulation Results

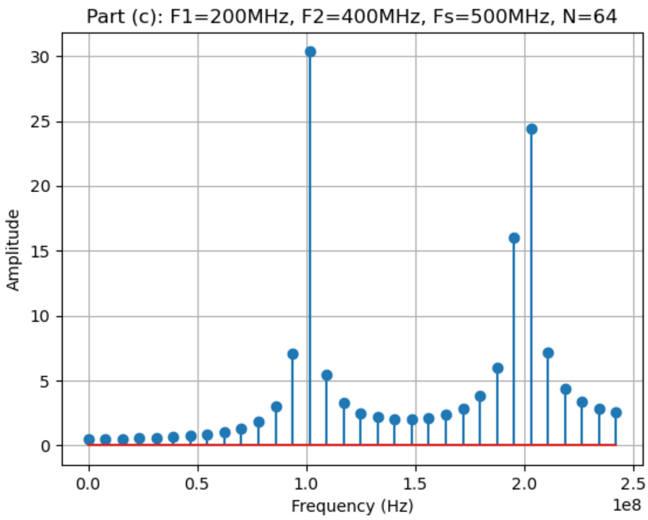

Figure 3.3: Simulation Results

fore, a 200 MHz signal will not experience aliasing, whereas a 400 MHz signal will.

- d.

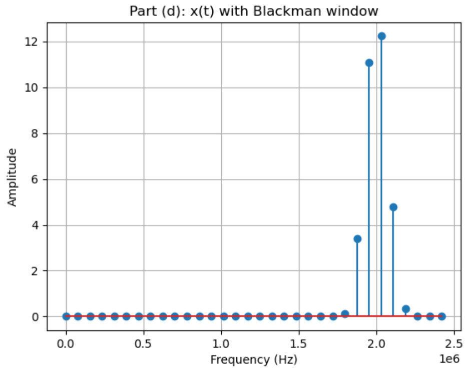

Figure 3.4: Signal with window

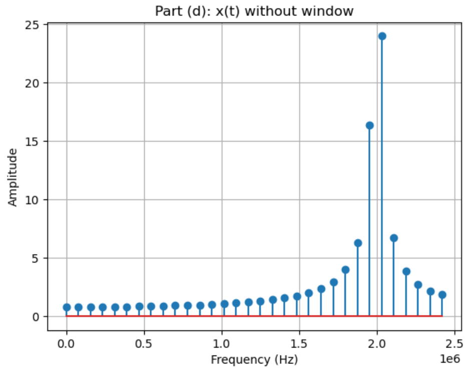

Figure 3.5: Signal without window

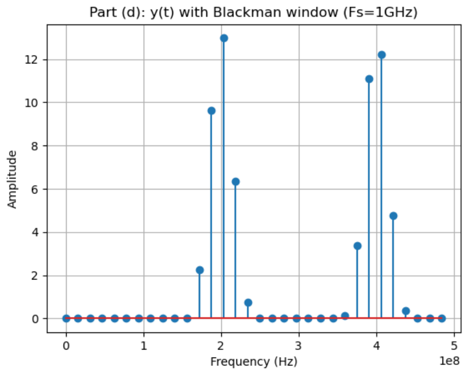

Figure 3.6: Signal with window

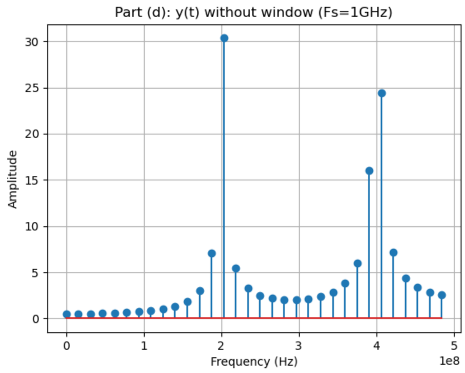

Figure 3.7: Signal without window

When no window is applied, the DFT can produce spectral leakage on finite-length samples, especially when the signal frequency does not precisely align with the FFT frequency bins. The Blackman window is a smoothing window that, when multiplied in the time domain before performing the FFT, significantly reduces the side lobe levels, thereby minimizing leakage from adjacent frequencies. However, this also results in a broader main lobe.
Comparing the magnitude spectra with and without windowing:
Without windowing: The main lobe is relatively narrow, but the side lobes are higher, leading to noticeable leakage.
With a Blackman window: The main lobe is slightly wider, but the side lobes are greatly attenuated, making the spectrum appear much "cleaner."