Sampling, Aliasing and Anti aliasing filter

Sampling

- f_s – Sampling frequency.
- f_{BW} – Band width of the sampled signal.
- f_N – Nyquist frequency = $0.5 * f_s$

The Aliasing Phenomenon

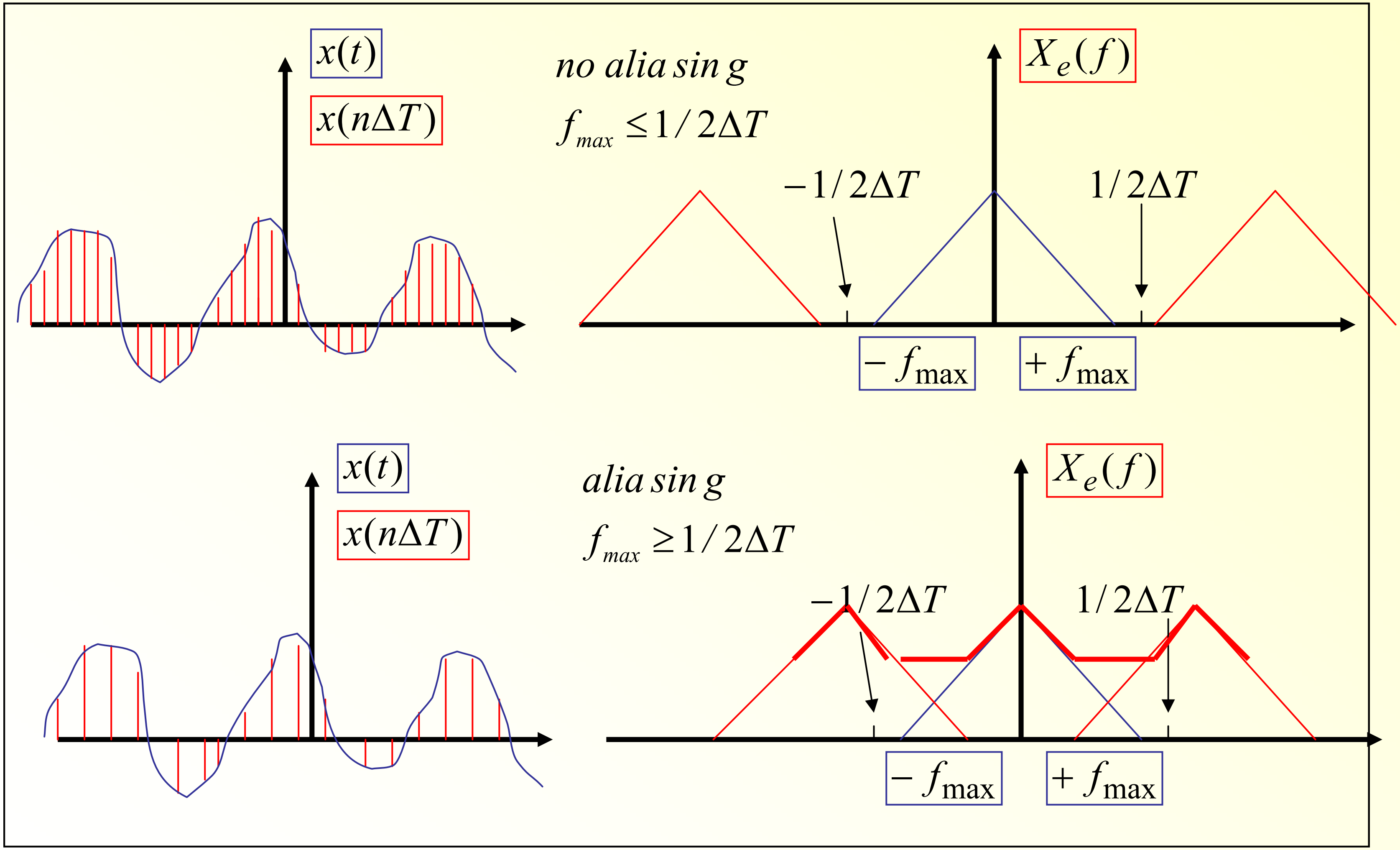
- The aliasing phenomenon appears when

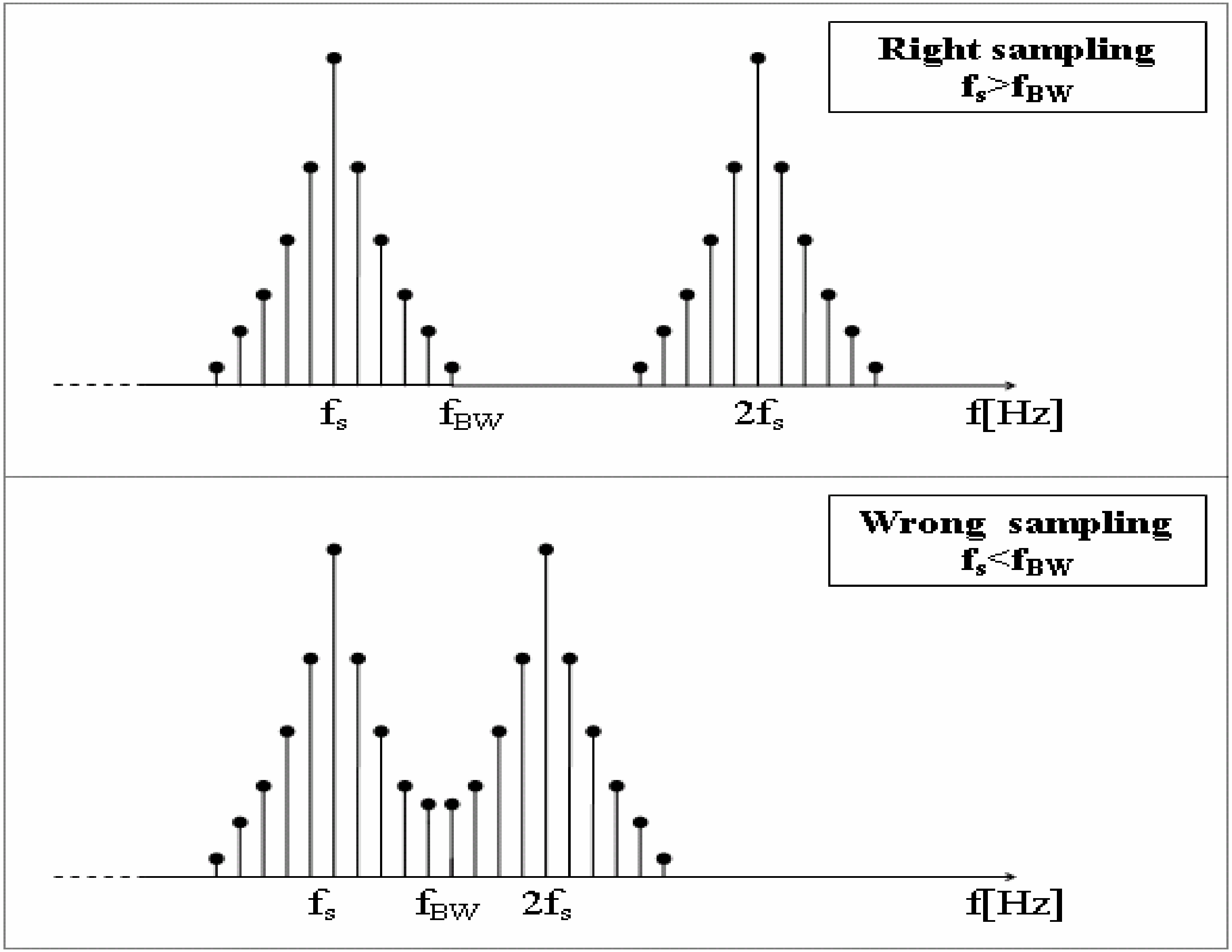
$$f_s < 2 * f_{BW}$$

- All frequencies that are larger than half the sampling frequency, can be interpreted after sampling, as frequencies in the interval (0,fs/2) [Hz].

DFT

Aliasing in the frequency domain

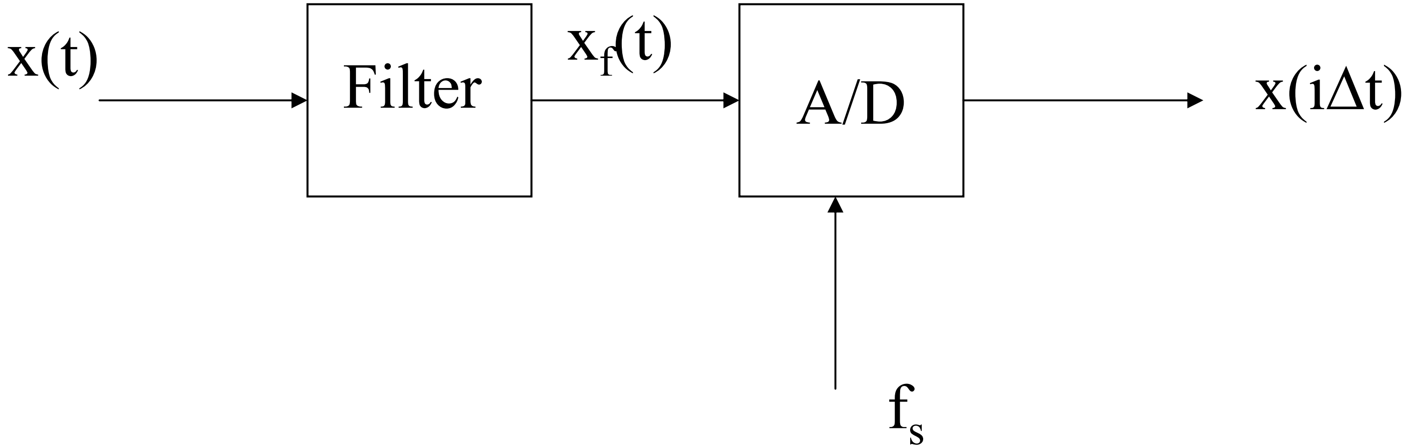



Right sampling
$f_s > f_{BW}$
f_s
f_{BW}
$2f_s$
f[Hz]
Wrong sampling
$f_s < f_{BW}$
f_s
f_{BW}
$2f_s$
f[Hz]

Solution for the aliasing problem

- Best solution: Sampling at $\mathbf{f_s \geq 2.5 * f_{BW}}$
- If our sampling hardware is not fast enough or we don't know the band width of the signal, we should use **anti aliasing filter**.

Anti-aliasing filter



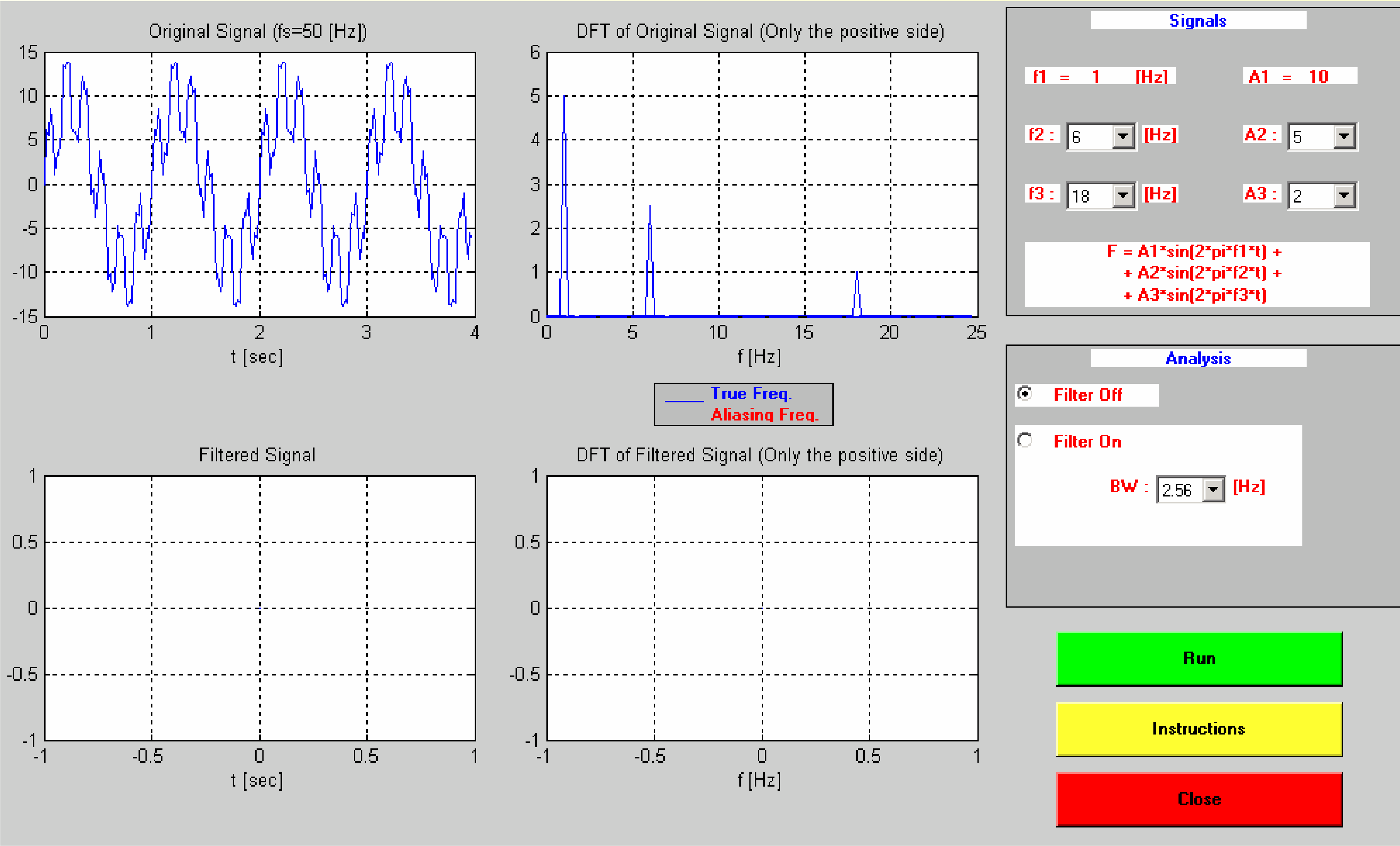
Anti aliasing filter

- The anti aliasing filter is a **LPF**.
- Its goal is to eliminate, <u>before sampling</u>, all frequencies in the signal that are, at least , above the Nyquist frequency and therefore avoid aliasing.
- Note that filtering the original signal cause, of course, losing data from the original signal, but it ensures good reconstruction of the <u>filtered signal</u>.

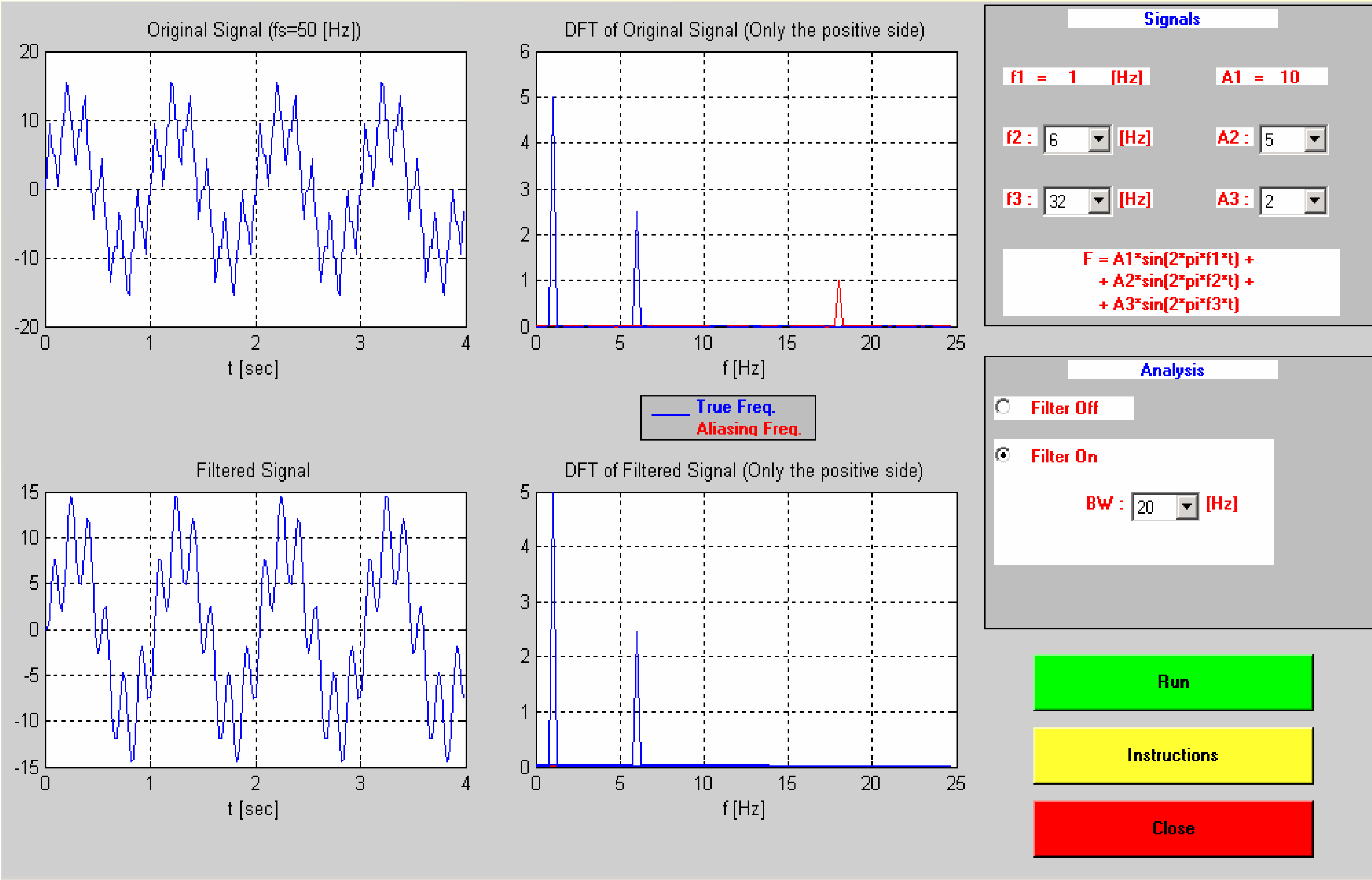
Illustrations

Ex6_2.mat – Aliasing of sine signals

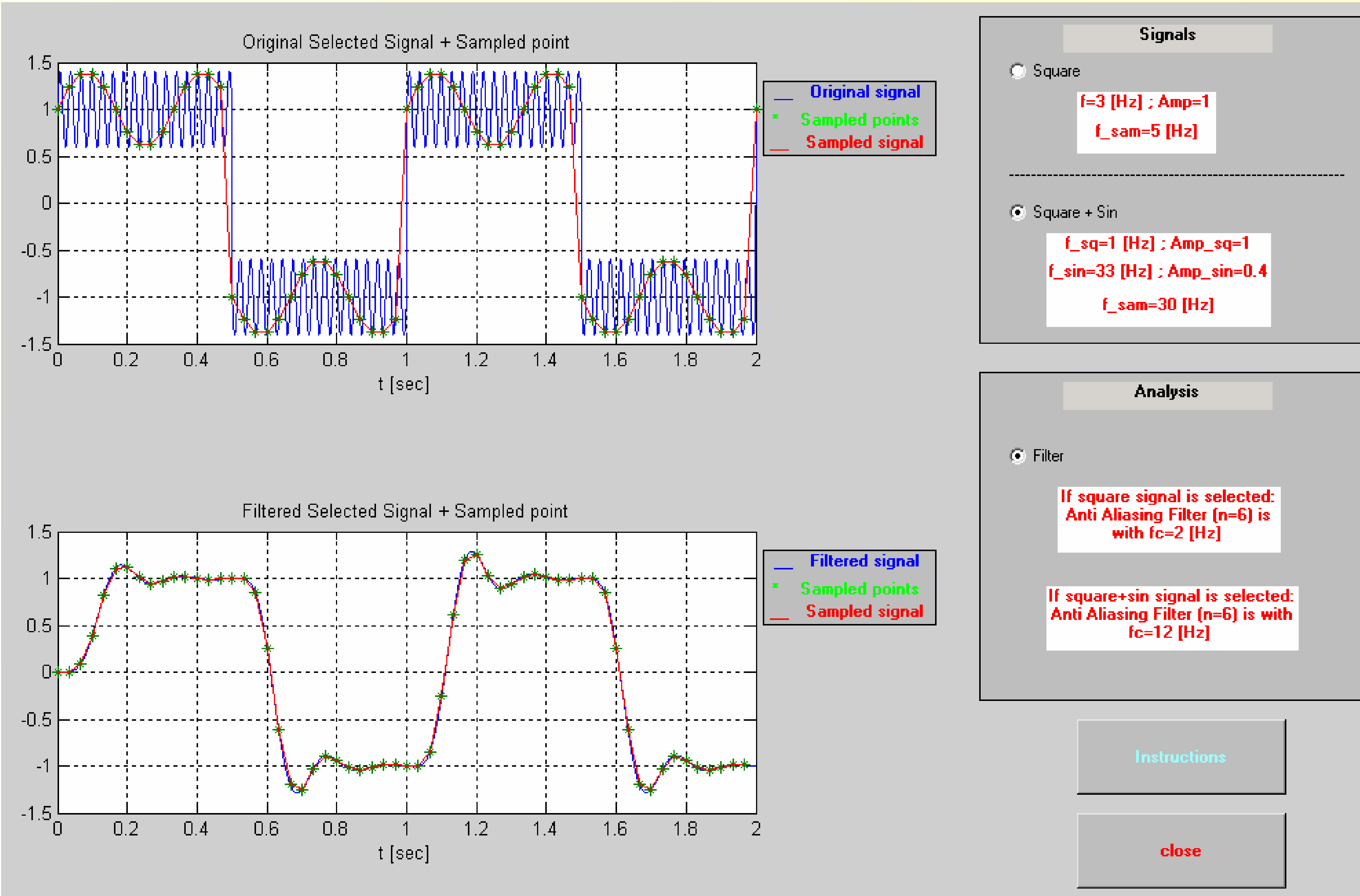
1) Right sampling, **f_s**=50 [Hz] **> 2.5*f_{BW}**=45 [Hz] $\longrightarrow$ No aliasing:



2) Wrong sampling, **fs** = 50 [Hz] **<2.5fBW** = 80 [Hz] , ⟶ aliasing:



Ex6_3.mat – Anti aliasing filter



Conclusions

Whenever you are sampling, always make sure that:

- The sampling frequency is high enough so that the sampled signal in the computer will be sufficiently true to the original.
- Frequencies at least above the Nyquist frequency will be eliminate before sampling, in order to avoid aliasing.