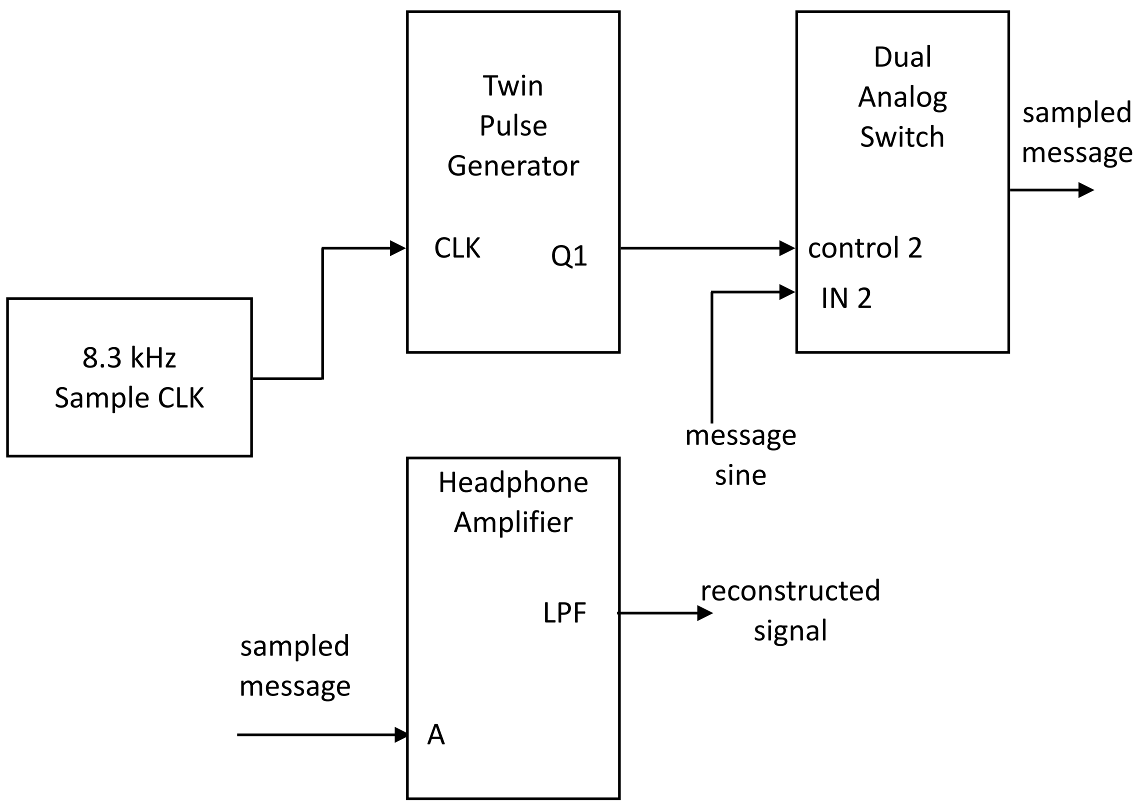
EE 458L

Lab 10: Nyquist Sampling Theorem

Sampling is the first step to convert a continuous analog signal to a digital signal. An analog signal can be sampled at the rate of F_s samples per second. It can be reconstructed perfectly if the following criterion is met:

$$F_S \geq 2B,$$

where, B is the maximum frequency of the signal. The above equation is called the Nyquist Criterion.



Lab Tasks

1. Display the 2 kHz message signal (use the message on the MASTER SIGNALS module) on Channel A of the oscilloscope.
2. Connect the 8.3 kHz Sample CLK signal (also on the MASTER SIGNALS module) to the CLK input of the TWIN PULSE GENERATOR. Display the Q1 output on Channel B. Tune the pulse width knob and observe the effect on the scope. This will be the ***sampling signal***.
3. Connect the message to the IN 2 input of the DUAL ANALOG SWITCH and the sampling signal (from step 2) to the CONTROL 2 input. Display the output on Channel B. This is the ***sampled message signal***.

4. To reconstruct the sampled message, connect the sampled message signal from step 3 to the input A of the HEADPHONE AMPLIFIER module. This module contains a 3 kHz LPF. Display the LPF output on Channel B. Compare the reconstructed signal with the message on Channel A.

5. Now, use the sine wave output of the AUDIO OSCILLATOR as the message (instead of the 2 kHz from the MASTER SIGNALS). Vary the frequency (tuning knob) of the AUDIO OSCILLATOR sine wave and compare the message and reconstructed waveforms on Channel A and Channel B, respectively.

6. At what frequency does the reconstructed signal (on Channel B) **not resemble** the message (on Channel A)? Can you explain why this occurs?

<u>Matlab Tasks</u>

1. Generate a 2 kHz sine wave *x(t)* for 10 periods. Use the sampling frequency of 48 kHz, which is well above Nyquist rate. Since the sample rate is well above Nyquist rate, we will consider this signal \`unsampled.'

2. Next, \`resample' the above signal at 8 kHz to get the signal *y(t)*. Since 8 kHz is one-sixth of 48 kHz, we just retain every sixth sample and set the others to zero. That is, retain only x(T), x(7T), x(13T), … and set all other samples to zero.

3. Next, low pass filter this signal *y(t)* to generate *z(t)*. Use 48 kHz as the sampling frequency for the filter and 4 kHz for the LPF bandwidth. Plot *z(t)* and obtain its frequency from the graph. Is the frequency equal to 2 kHz, the frequency of *x(t)*?

4. To demonstrate the effect of aliasing, repeat steps 1—3 above with 6 kHz as the frequency of *x(t)*. Is the frequency of *z(t)* equal to 6 kHz, the frequency of *x(t)*?