Week 2 Pre-Lab Reading

1 Introduction

When searching for radio transmissions from extraterrestrial intelligence, scientists consider several possible types of signals that may be received. One type is continuous wave (CW) transmissions, which are steady signals at a single frequency. These could indicate intentional communication, similar to how radio stations on Earth broadcast continuous signals. Another type is pulsed signals, which are intermittent bursts of energy at specific intervals. Pulsed signals might be used for signaling purposes or as beacons, indicating the presence of intelligent life. Modulated signals, such as amplitude modulation (AM) or frequency modulation (FM), could also be potential signs of deliberate communication, as they carry information in their variations.

Identifying these signals amidst background noise is a significant challenge in the search for extraterrestrial intelligence (SETI). Scientists employ various techniques to distinguish potential alien signals from natural or human-made interference. One approach is to analyze the signal's characteristics, such as its frequency, bandwidth, and modulation pattern, looking for patterns that suggest artificial origin. Advanced signal processing algorithms, including machine learning, can help automate this analysis by identifying signal anomalies that may indicate extraterrestrial origin. Additionally, multiple observations and confirmation from different telescopes or radio arrays are crucial to rule out false positives and confirm the authenticity of detected signals.

To improve signal detection and reduce false alarms, radio astronomers and SETI researchers also rely on sophisticated equipment, such as radio telescopes with high sensitivity and wide bandwidth capabilities. These instruments can scan vast portions of the sky, searching for signals across different frequencies and polarization states. Collaborative efforts, such as the SETI Institute's Allen Telescope Array and international partnerships like the Breakthrough Listen initiative, combine resources and expertise to enhance the chances of detecting potential radio transmissions from intelligent civilizations beyond Earth.

2 Nyquist Sampling Theorem

When an analog signal is converted into digital format, the original signal is sampled (measured) at equally spaced time intervals, and the discrete sequence of samples is used to represent the original signal. The Nyquist theorem says that in order to accurately represent a signal, the sample rate must be at least twice the frequency of the highest frequency component of the signal. For a given sample rate f_s, the highest frequency signal that can be accurately measured is referred to as the Nyquist frequency f_N. For example, at a sample rate of $f_s = 16kHz$, the Nyquist frequency is $f_N = 8kHz$.

$$f_s \geq 2 * f_N \tag{1}$$

If the frequency of a signal is greater than the Nyquist frequency, an effect called aliasing will occur. Aliasing will cause false lower frequency components to appear in the sampled data.

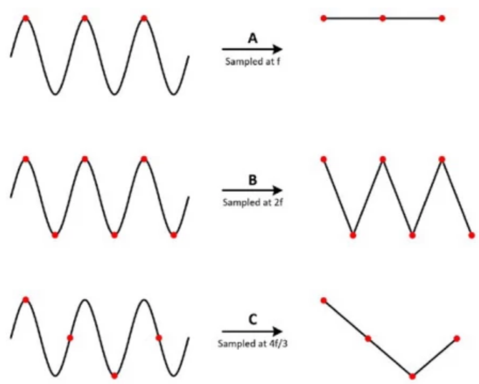

Figure 1: Undersampling a signal leads to an inaccurate reconstruction of the waveform and alias frequencies.

The effect of aliasing can easily be shown using GNU radio. This simple flowgraph demonstrates the relationship between the frequency of a signal and the frequency of the sample rate.

- The "Signal Source" block produces a sine wave with a frequency adjustable by the first "QT GUI Range" block.
- The "Throttle" block samples the signal at a frequency that is adjustable by the second "QT GUI Range" block.
- The "QT GUI Time Sink" block displays the sampled signal waveform.

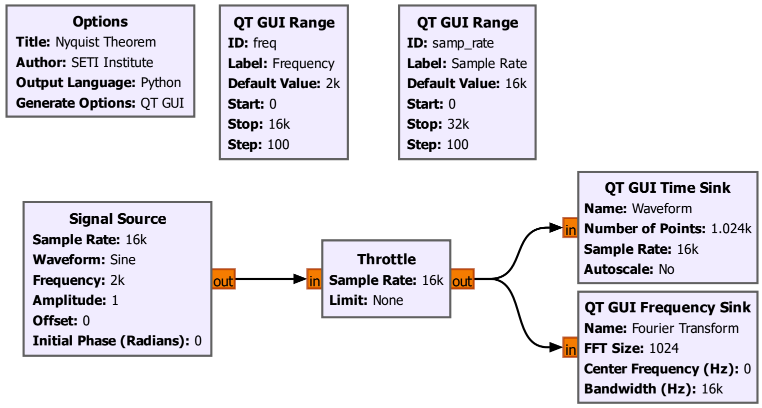

Figure 2: GNU Radio flowgraph demonstrating Nyquist sampling

- The "QT GUI Frequency Sink" block computes and displays the Fast Fourier Transform (FFT).

Using the slider at the top of the window, we can change either the sample rate or the signal frequency. In figure 3, a 2 kHz signal is being sampled at 16 kHz, and the resulting waveform and Fourier transform are displayed. The waveform is clearly a sine wave, and a peak can be seen in the Fourier transform display at the correct incoming frequency of 2 kHz. In figure 4 however, the signal frequency is increased to 10 kHz while the sample rate is kept fixed at 16 kHz. Instead of a peak at 10 kHz, we see an alias frequency of 6 kHz in the Fourier transform display, as well as significant degradation of the waveform.

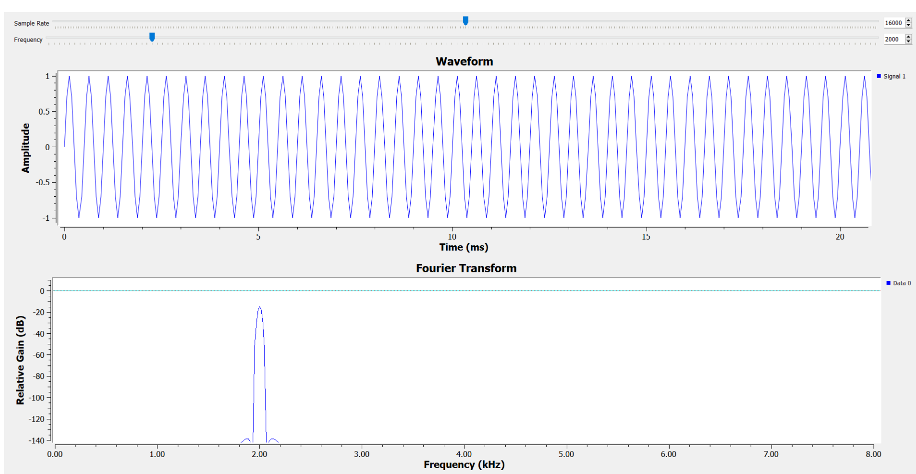

Figure 3: A 2 kHz sine wave sampled at 16 kHz showing an accurate waveform and correct frequency.

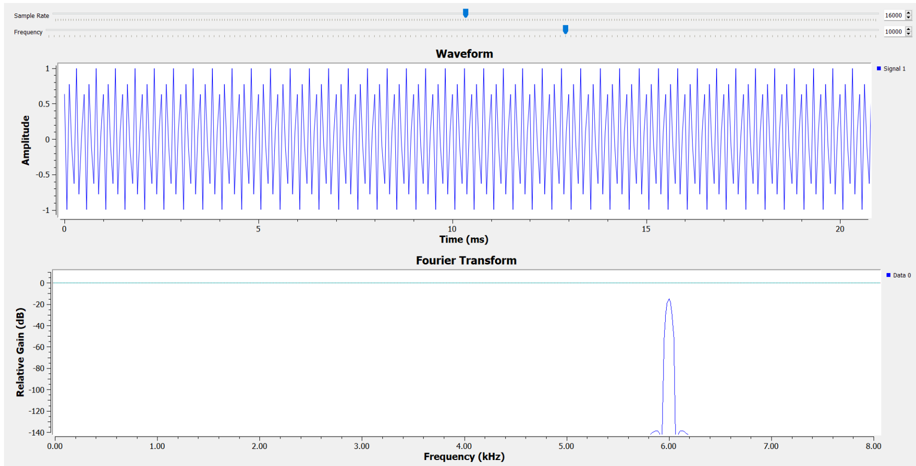

Figure 4: A 10 kHz sine wave sampled at 16 kHz showing a degraded waveform and alias frequency at 6 kHz.

3 Filters

Various kinds of filters are often used in signal processing to separate, modify, or enhance specific frequencies ranges of a signal. Filters can be used to remove noise or other forms of interference, and to extract information from a signal. Here we'll cover the most commonly used types of filters and show how they are used in signal processing applications. First let's define some frequently used terms. A diagram of these terms overlaid on a Band Pass filter can be see in figure 5.

- Attenuate - to decrease the amplitude of a signal.
- Cutoff frequency - the frequency at which a filter's response begins to significantly attenuate or change.
- Passband - the range of frequencies a filter allows to pass through with minimal attenuation.
- Stopband - the range of frequencies a filter will attenuate.
- Transition band - the frequency range between the passband and stopband where the filter transitions from allowing signals to attenuating them, or vice versa.
- Center frequency - the midpoint of the passband or stopband.

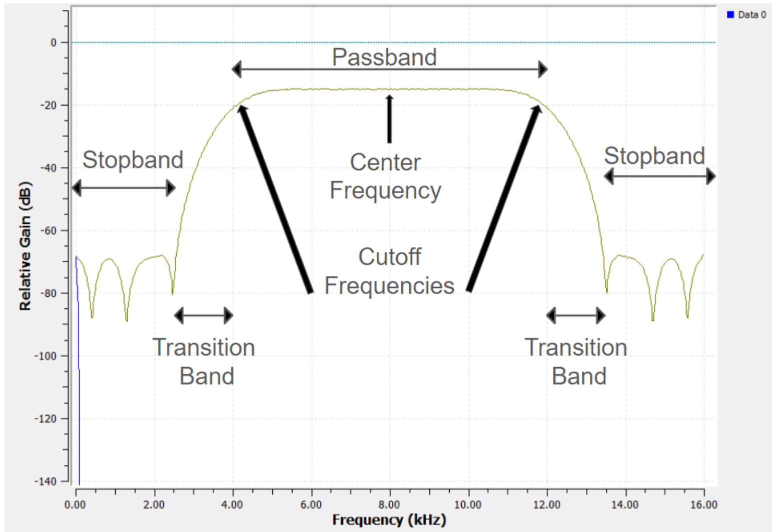

Figure 5: A Band Pass filter with labeled terms.

3.1 Band Pass Filters

Band Pass filters are designed to allow a specific range of frequencies to pass while attenuating frequencies outside this range. This is useful for isolating signals within a specific frequency range of interest and reducing the noise coming from other frequencies. In communication systems, Band Pass filters are used for demodulation, where they extract the baseband signal from the modulated carrier wave.

3.2 Low Pass Filters

As the name implies, Low Pass filters pass lower frequency components of a signal while attenuating higher frequencies. Low Pass filters play a crucial role in signal processing by removing high-frequency components, reducing noise and interference, improving signal quality, and facilitating accurate signal analysis and interpretation. In digital communication applications, Low Pass filters are commonly used for baseband signal processing, channel equalization, and modulation and demodulation.

3.3 High Pass Filters

High Pass filters pass higher frequency components of a signal while attenuating lower frequencies. High Pass filters are used in signal preprocessing stages to remove low-frequency noise and interference from communication signals. This helps in improving the signal-to-noise ratio (SNR) and enhancing the overall quality of transmitted and received signals. They can also be employed for transient detection in radio astronomy, where they highlight rapid changes or events in signal intensity that occur at higher frequencies compared to the background noise and steady signals.