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ECE251 Assignment 2 explanation

1. I chose a sampling frequency of 16kHz because it allows the PSD to contain several sidelobes of the 4kHz modulated signal.
2. To ensure the PSD was averaged over multiple data blocks, I have a variable to set the number of symbols per PSD block. Adjusting this variable has the effect of visually varying the noise present in the PSD due to the random bits.
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4. The main difference I’m noticing between s, z1 and z2 is that z1 and z2 are ‘rectangular pulses’ with a lower number of harmonics than the ideally generated pulse train p(t-nT). For z2, this is because its real and imaginary parts u(t) and v(t) are LP filtered as a part of the quadrature demodulation, which both cuts out the negative signal and some of the harmonics of each signal. My LPF had a cutoff of 2kHz, which only retained one sidelobe, visualized in figure 2 b. For z1, this is because finding the analytical signal x1+ of the received signal x1 neutralizes any negative frequency components, or any components that were aliased effectively bandlimiting the signal when it is shifted back down to baseband, as seen in Figure 1 c/d. This bandlimiting can be asymmetrical depending on the sampling frequency and the carrier frequency. In this instance, I chose a sampling frequency of 16kHz which causes the bandlimiting effect to be symmetrical. In this case, the PSD of z1 was ‘bandlimited’ to 4 sidelobes, which makes the signal z1(t) appear to be a square pulse with more harmonics than signal z2(t).