at a higher frequency, or even if the original had been a continuous analog signal: if we sample at framerate f, we can recover the original signal exactly, as long as it contains no energy at frequencies above $f/2$.

11.6 Exercises

Solutions to these exercises are in `chap11soln.ipynb`.

Exercise 11.1 The code in this chapter is in `chap11.ipynb`. Read through it and listen to the examples.

Exercise 11.2 Chris "Monty" Montgomery has an excellent video called "D/A and A/D — Digital Show and Tell"; it demonstrates the Sampling Theorem in action, and presents lots of other excellent information about sampling. Watch it at `https://www.youtube.com/watch?v=cIQ9IXSUzuM`.

Exercise 11.3 As we have seen, if you sample a signal at too low a framerate, frequencies above the folding frequency get aliased. Once that happens, it is no longer possible to filter out these components, because they are indistinguishable from lower frequencies.

It is a good idea to filter out these frequencies *before* sampling; a low-pass filter used for this purpose is called an **anti-aliasing filter**.

Returning to the drum solo example, apply a low-pass filter before sampling, then apply the low-pass filter again to remove the spectral copies introduced by sampling. The result should be identical to the filtered signal.