# ThinkDSP

This notebook contains solutions to exercises in Chapter 11: Modulation and sampling



In [1]:
```
from __future__ import print_function, division

import thinkdsp
import thinkplot

import numpy as np

import warnings
warnings.filterwarnings('ignore')

PI2 = 2 * np.pi

np.set_printoptions(precision=3, suppress=True)
%matplotlib inline
```

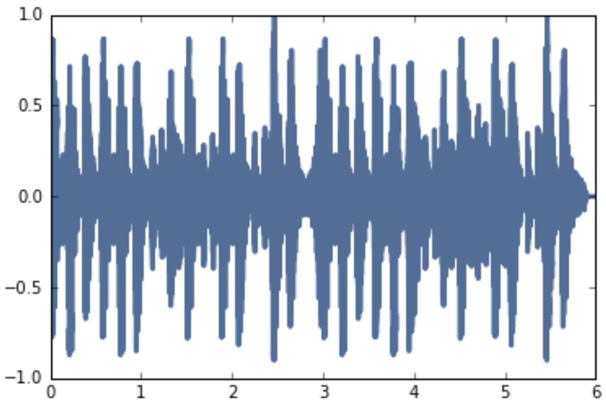
**Exercise:** As we have seen, if you sample a signal at too low a framerate, frequencies above the folding frequency get aliased. Once that happens, it is no longer possible to filter out these components, because they are indistinguishable from lower frequencies.

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.

*Solution:* I'll load the drum solo again.

In [2]:
```
wave = thinkdsp.read_wave('263868__kevcio__amen-break-a-160-bpm.wav')
wave.normalize()
wave.plot()
```




This signal is sampled at 44100 Hz. Here's what it sounds like.

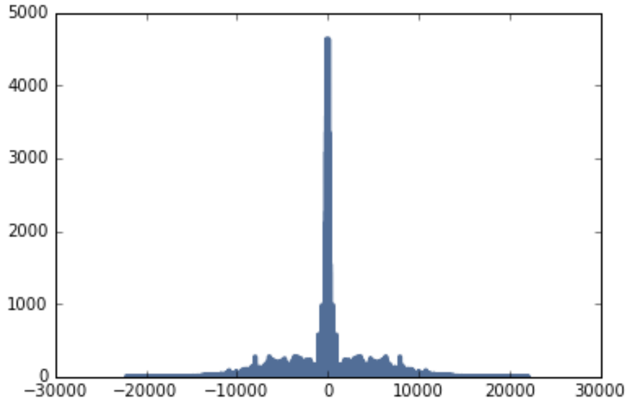
In [3]:
```
wave.make_audio()
```

Out[3]:

0:00:00 / 13:31:36

And here's the spectrum:

In [4]:
```
spectrum = wave.make_spectrum(full=True)
spectrum.plot()
```

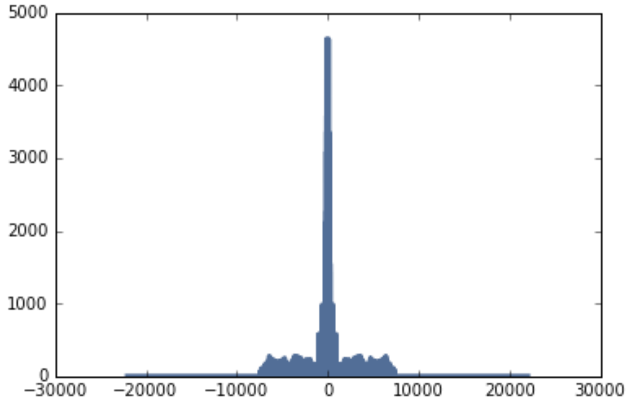



I'll reduce the sampling rate by a factor of 3 (but you can change this to try other values):

In [5]:
```
factor = 3
framerate = wave.framerate / factor
cutoff = framerate / 2 - 1
```

Before sampling we apply an anti-aliasing filter to remove frequencies above the new folding frequency, which is `framerate/2` :

In [6]:
```
spectrum.low_pass(cutoff)
spectrum.plot()
```




Here's what it sounds like after filtering (still pretty good).

In [7]:
```
filtered = spectrum.make_wave()
filtered.make_audio()
```

Out[7]:

0:00:00 / 13:31:36

Here's the function that simulates the sampling process:

In [8]:
```
def sample(wave, factor):
    """Simulates sampling of a wave.

    wave: Wave object
    factor: ratio of the new framerate to the original
    """
    ys = np.zeros(len(wave))
    ys[::factor] = wave.ys[::factor]
    return thinkdsp.Wave(ys, framerate=wave.framerate)
```

The result contains copies of the spectrum near 20 kHz; they are not very noticeable:

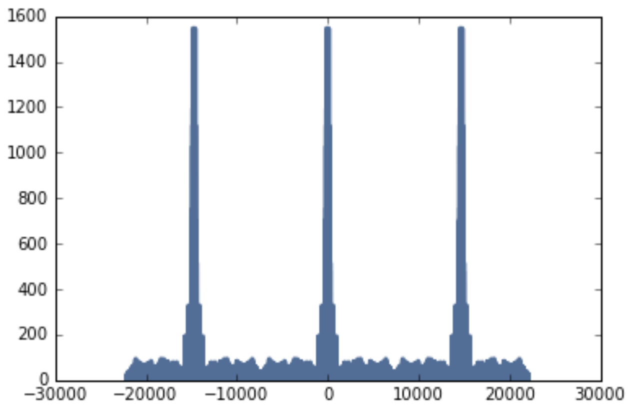
In [9]:
```
sampled = sample(filtered, factor)
sampled.make_audio()
```

Out[9]:

0:00:00 / 13:31:36

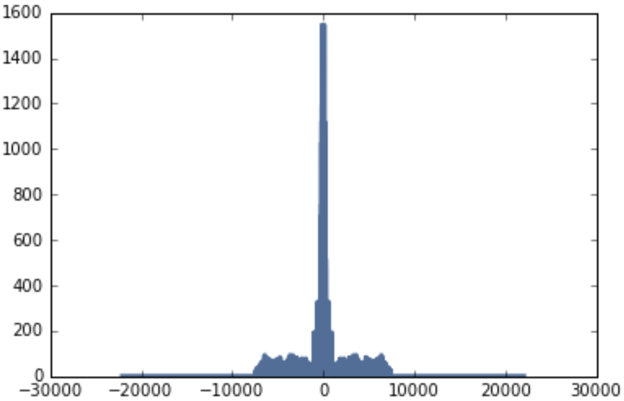
But they show up when we plot the spectrum:

In [10]:
```
sampled_spectrum = sampled.make_spectrum(full=True)
sampled_spectrum.plot()
```



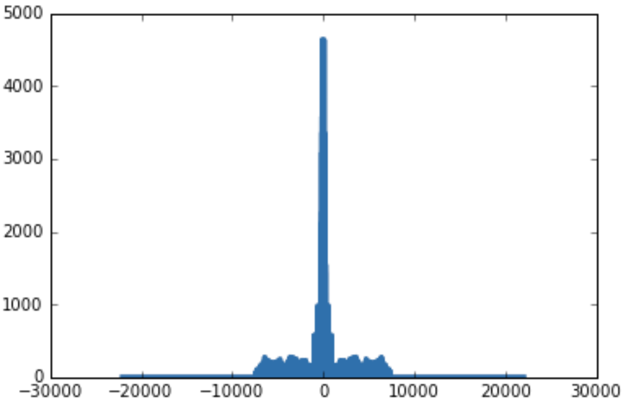

We can get rid of the spectral copies by applying the anti-aliasing filter again:

In [11]:
```
sampled_spectrum.low_pass(cutoff)
sampled_spectrum.plot()
```




We just lost half the energy in the spectrum, but we can scale the result to get it back:

In [12]:
```
sampled_spectrum.scale(factor)
spectrum.plot()
sampled_spectrum.plot()
```




Now the difference between the spectrum before and after sampling should be small.

In [13]:
```
spectrum.max_diff(sampled_spectrum)
```

Out[13]: 9.0949470177292824e-12

After filtering and scaling, we can convert back to a wave:

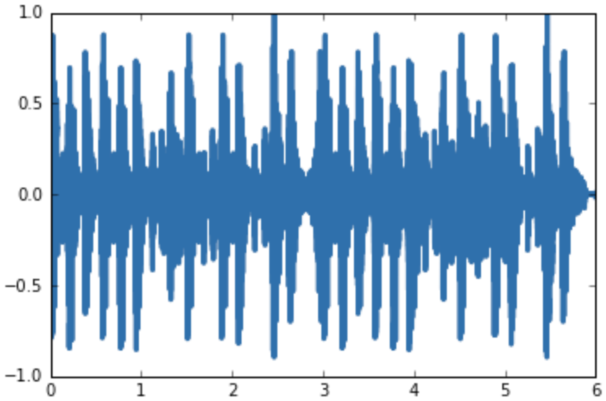
In [14]:
```
interpolated = sampled_spectrum.make_wave()
interpolated.make_audio()
```

Out[14]:

0:00:00 / 13:31:36

And the difference between the interpolated wave and the filtered wave should be small.

In [15]:
```
filtered.plot()
interpolated.plot()
```




Multiplying by `impulses` makes 4 shifted copies of the original spectrum. One of them wraps around from the negative end of the spectrum to the positive, which is why there are 5 peaks in the spectrum off the sampled wave.

In [16]:
```
filtered.max_diff(interpolated)
```

Out[16]: 3.663750800998542e-15

In [ ]:

In [ ]: