1branchFBdirectImpl

May 3, 2017

0.1 Python Example:

Demonstration of 1 branch of a filter bank in direct implementation for N=16 subbands, and subband k=1 #Gerald Schuller, October 2016.

0.2 Import the modules and define the variable:

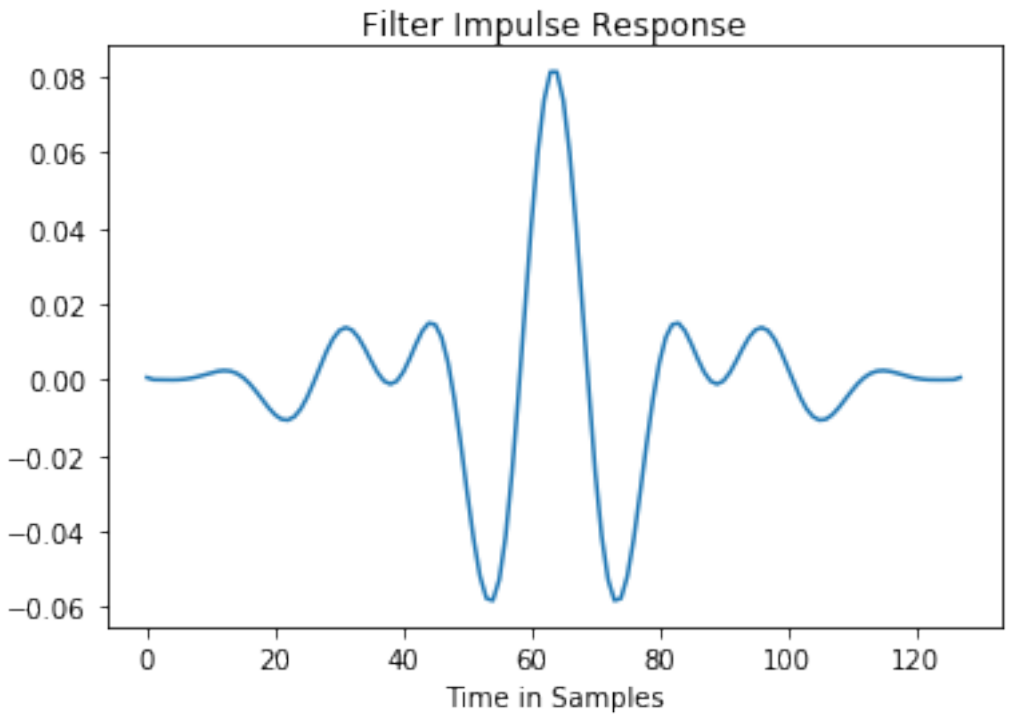
```
In [1]: %matplotlib inline
        import scipy.signal as signal
        import matplotlib.pyplot as plt
        import numpy as np
        import sound as snd
        import os

        N = 16
```

0.3 FIlter Design:

```
In [2]: b = signal.remez(8*N, [0,500,1000,2000,2500,16000], [0,1,0], [100,1,100], Hz=32000, type
```

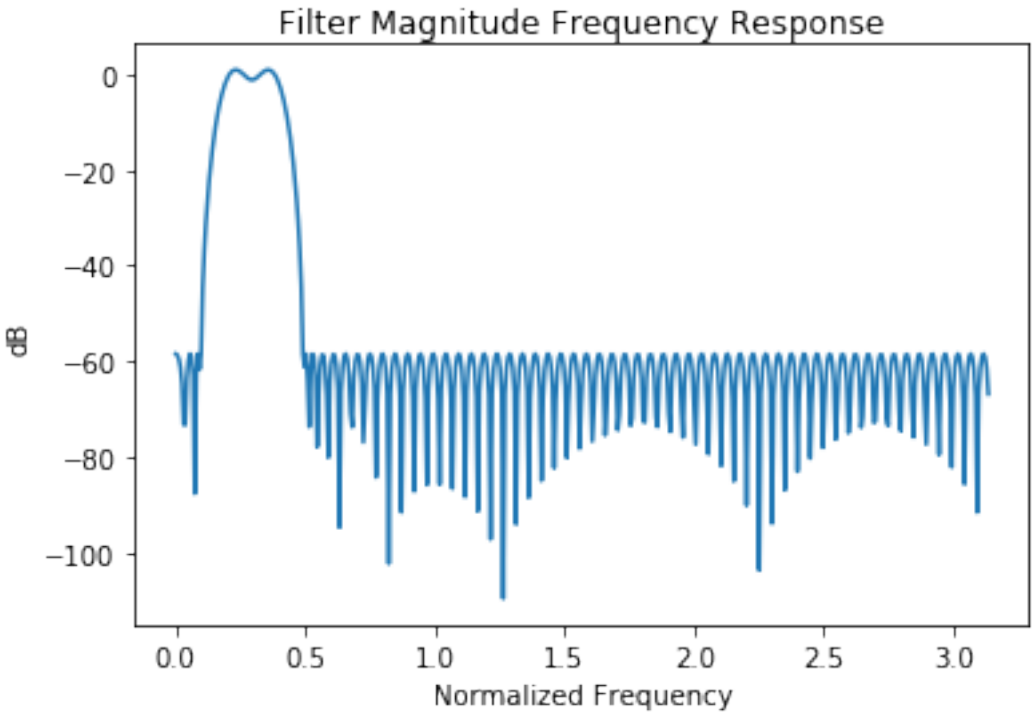
0.4 Check the design and plot the impulse response:

```
In [3]: plt.plot(b)
        plt.title('Filter Impulse Response')
        plt.xlabel('Time in Samples')
        plt.show()
```



1 Plot the frequency response:

```
In [4]: w, H=signal.freqz(b)
        plt.plot(w, 20*np.log10(np.abs(H)+1e-6))
        plt.title('Filter Magnitude Frequency Response')
        plt.xlabel('Normalized Frequency')
        plt.ylabel('dB')
        plt.show()
```

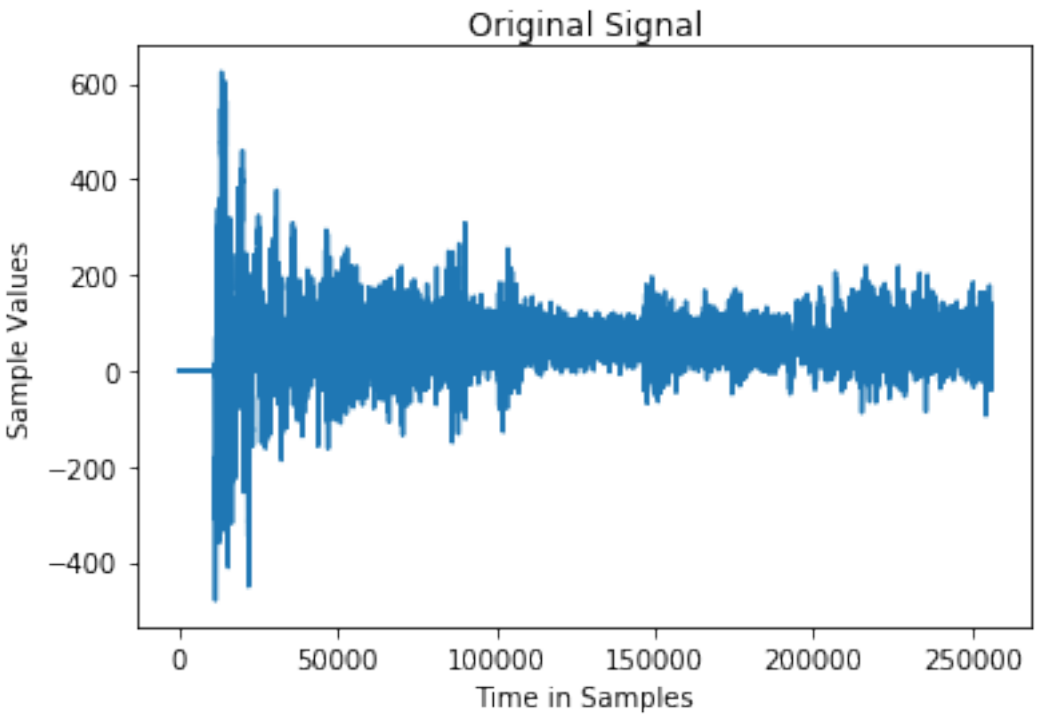


1.1 Analysis Filtering:

```
In [5]: print("Attention: Recording for 8s!")
        os.system('espeak -ven -s 140 '+'"Attention: Recording for 8 seconds!"')
        s=snd.record(8, 32000)
        print("length of sound in samples: ", len(s))

Attention: Recording for 8s!
('device count=', 12L)
('i = ', 0)
('max Input Channels=', 2L)
('default Sample Rate=', 44100.0)
('i = ', 1)
('max Input Channels=', 2L)
('default Sample Rate=', 44100.0)
('i = ', 2)
('max Input Channels=', 0L)
('default Sample Rate=', 44100.0)
('i = ', 3)
('max Input Channels=', 0L)
('default Sample Rate=', 44100.0)
('i = ', 4)
('max Input Channels=', 2L)
('default Sample Rate=', 44100.0)
```

```
('i = ', 5)
('max Input Channels=', 2L)
('default Sample Rate=', 44100.0)
('i = ', 6)
('max Input Channels=', 0L)
('default Sample Rate=', 44100.0)
('i = ', 7)
('max Input Channels=', 0L)
('default Sample Rate=', 44100.0)
('i = ', 8)
('max Input Channels=', 0L)
('default Sample Rate=', 44100.0)
('i = ', 9)
('max Input Channels=', 2L)
('default Sample Rate=', 44100.0)
('i = ', 10)
('max Input Channels=', 2L)
('default Sample Rate=', 44100.0)
('i = ', 11)
('max Input Channels=', 0L)
('default Sample Rate=', 44100.0)
* recording
('length of sound in samples: ', 256000)
```

```
In [6]: plt.plot(s)
        plt.title('Original Signal')
        plt.xlabel('Time in Samples')
        plt.ylabel('Sample Values')
        plt.show()
```




```
In [7]: os.system('espeak -ven -s 140 '+'"Now hear the Original Signal"')
        snd.sound(s / np.max(s) * 30000, 32000)
```

```
* done
```

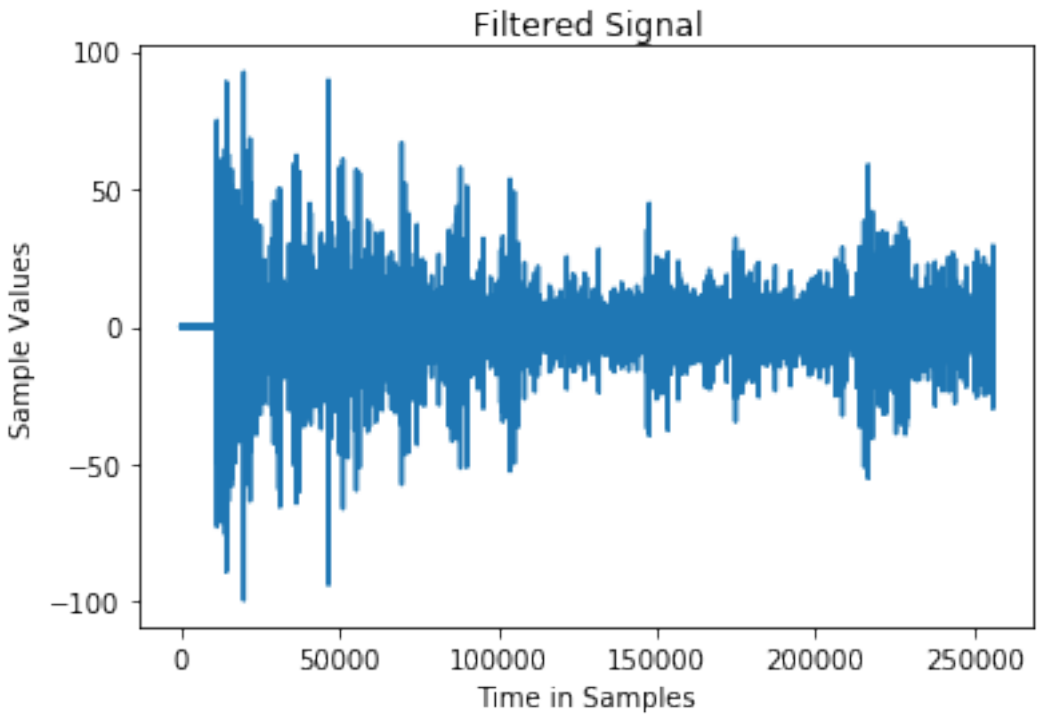
1.2 Filter Implementation:

```
In [8]: filtered=signal.lfilter(b,1,s)
        print("length of filtered sound in samples: ", len(filtered))
```

```
('length of filtered sound in samples: ', 256000)
```

1.3 Plot the Filtered Signal:

```
In [9]: plt.plot(filtered)
        plt.title('Filtered Signal')
        plt.xlabel('Time in Samples')
        plt.ylabel('Sample Values')
        plt.show()
```



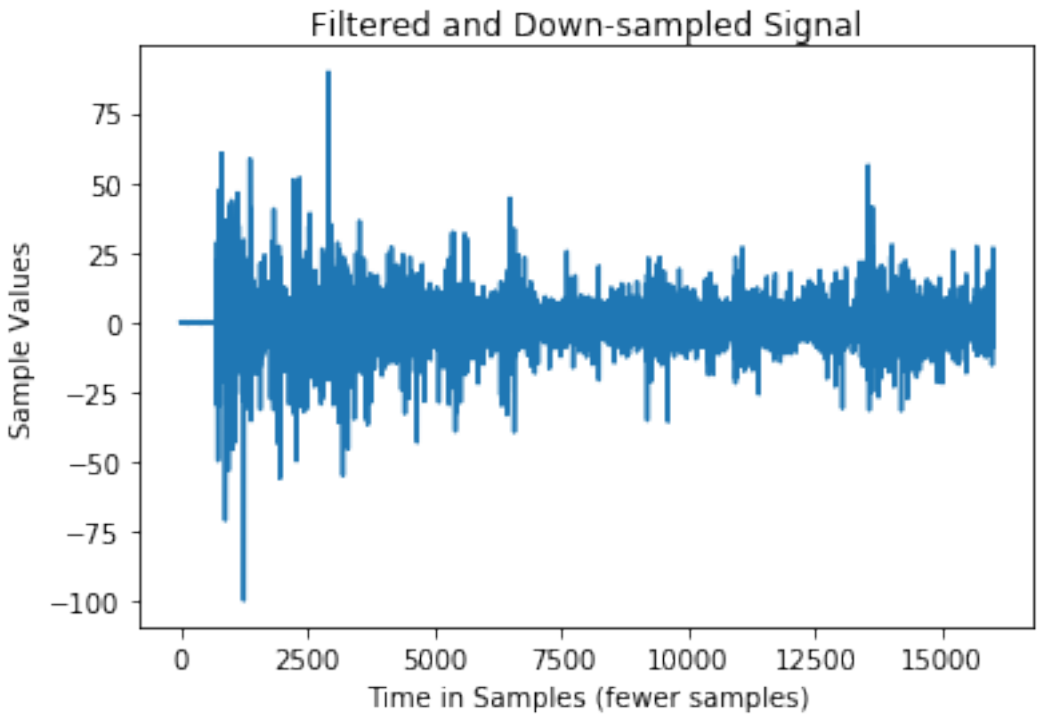
1.4 Play the filtered sound:

```
In [10]: os.system('espeak -ven -s 140 '+'"Now hear the filtered Signal"')
         snd.sound(filtered/np.max(filtered)*30000, 32000)
```

```
* done
```

1.5 Now Down-sampling with factor N:

```
In [11]: N=16
         filteredds=filtered[::N]
         plt.plot(filteredds)
         plt.title('Filtered and Down-sampled Signal')
         plt.xlabel('Time in Samples (fewer samples)')
         plt.ylabel('Sample Values')
         plt.show()
```



1.6 Listen to it at `1/N'th` sampling rate:

```
In [12]: os.system('espeak -ven -s 140 '+'"Now hear the filtered and down-sampled Signal at the
         snd.sound(filteredds/np.max(filteredds)*30000, 2000)
```

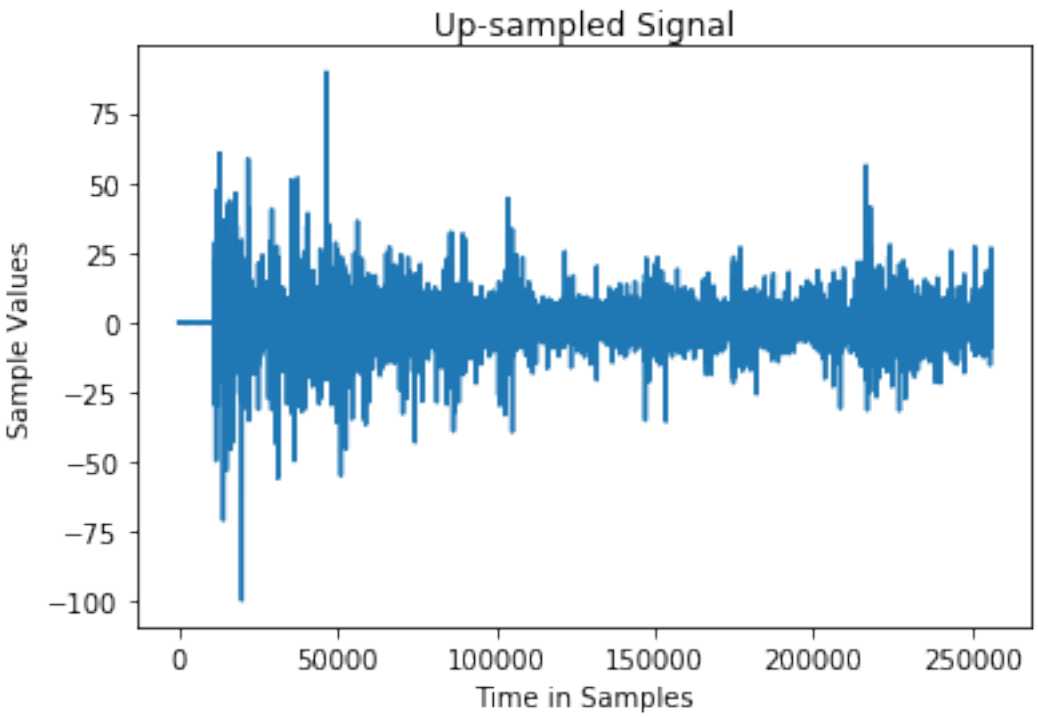
```
* done
```

1.7 Upsampling:

```
In [13]: filteredus=np.zeros(len(filteredds)*N)
         filteredus[::N]=filteredds
```

1.8 Plot the upsampled signal:

```
In [14]: plt.plot(filteredus)
         plt.title('Up-sampled Signal')
         plt.xlabel('Time in Samples')
         plt.ylabel('Sample Values')
         plt.show()
```



1.9 Listen to the up-sampled sound:

```
In [15]: os.system('espeak -ven -s 140 '+'"Now hear the up-sampled Signal back at the higher sam
         snd.sound(filteredus/np.max(filteredus)*30000, 32000)
```

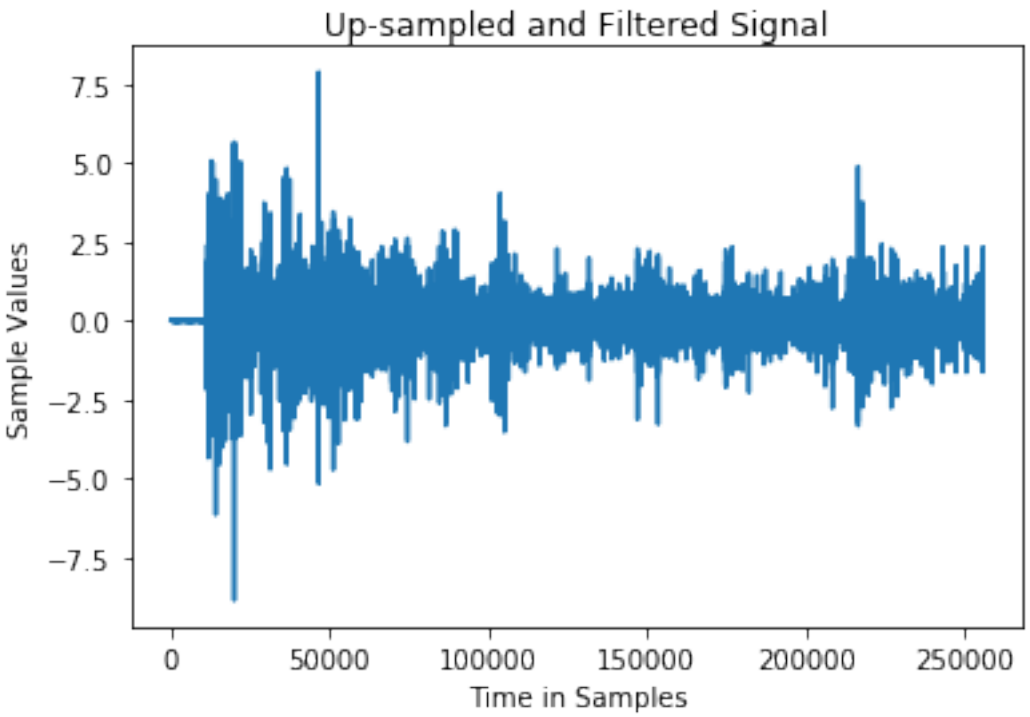
```
* done
```

- Synthesis Filtering: Bandpass Synthesis Filter implementation to attenuate the spectral copies:

```
In [16]: filteredsyn=signal.lfilter(b,1,filteredus)
```

1.10 Plot the upsampled and filtered version of the signal:

```
In [17]: plt.plot(filteredsyn)
         plt.title('Up-sampled and Filtered Signal')
         plt.xlabel('Time in Samples')
         plt.ylabel('Sample Values')
         plt.show()
```




```
In [18]: os.system('espeak -ven -s 140 '+'"Now hear the up-sampled and filtered signal"')
         snd.sound(filteredsyn/np.max(filteredsyn)*30000, 32000)
```

```
* done
```