3/02/2014. EE 5480 Lab Assignment - 3 Name: Alfred Ajay Aureate . R.

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Exercises:

Designing an FIR filter in Python.

FIR Filters:

Qn: If a digital low-pass filter is designed for a sampling frequency of $f_s = 1000Hz$ and cutoff frequency $f_c = 200Hz$, then if we use the same coefficients on a sampling frequency of $f_s = 4000Hz$, what cutoff frequency does that correspond to?

Ans: The normalized frequency should be the same for the cutoff frequency,

$$\therefore \frac{200}{1000} = \frac{f_c}{4000} \Rightarrow f_c = 800Hz.$$

Qn: Why 20 log and not 10 log?

Ans: Power $\propto V^2$ $\Rightarrow \frac{\text{output Power}}{\text{input Power}} = \left(\frac{V_{output}}{V_{input}}\right)^2 = \left(\frac{V_o}{V_i}\right)^2$.

$$\Rightarrow 10 \log\left(\frac{\text{output power}}{\text{input power}}\right) = 20 \log\left(\frac{V_o}{V_i}\right).$$

Qn: Why -40 and not +40?

Ans: The power attenuated or decreases.

$$\therefore P_o < P_i \quad \& \quad V_o < V_i \Rightarrow \cancel{20 \log\left(\frac{V_o}{V_i}\right) < 0}$$

$$\Rightarrow \frac{V_o}{V_i} < 1 \Rightarrow 20 \log\left(\frac{V_o}{V_i}\right) < 0.$$

FIR Design and Simulation with Python:

Frequency Content:

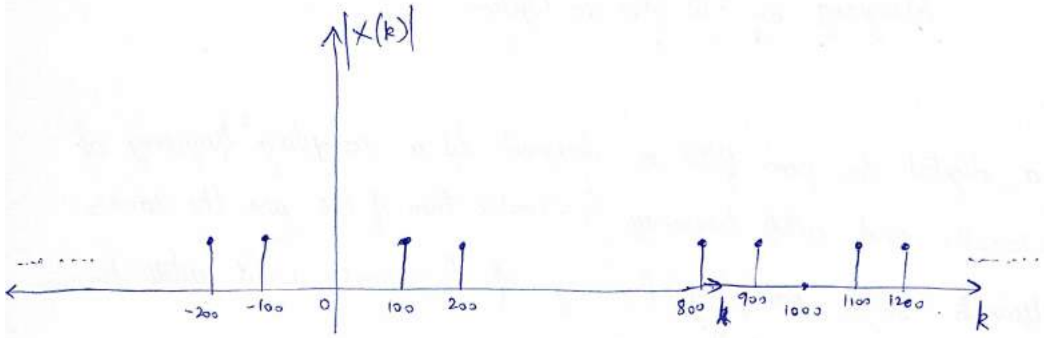
Qn: Why is the peak amplitude the value that you see?

Ans: $x[n] = s = \sin\left(2\pi \times \frac{100n}{1000}\right) + \sin\left(2\pi \times \frac{200n}{1000}\right) = e^{j2\pi \times \frac{100n}{1000}} - e^{-j2\pi \times \frac{100n}{1000}} + e^{j2\pi \times \frac{200n}{1000}} - e^{-j2\pi \times \frac{200n}{1000}}$

$$X(k) = FFT(s) = \sum_{n=0}^{999} x_n e^{-j2\pi k \frac{n}{N}} = \frac{1}{2i}\left[\delta(k-100) - \delta(k-900) + \delta(k-200) - \delta(k-800)\right]$$

②

$\because$ in frequency domain, we have,



$\therefore$ we see peaks at $k = 100, 200, 800$ & 900,

$|X(k)| = \frac{1}{2}$ at those k values.

$20 \times \log_{10}\left(\frac{1}{2}\right) \simeq -6.02$.

Qn: Why is there a noisy floor?

Ans: In python, the plot of sine had been of an interpolation of points sampled at ~~0.001~~ 1ms. So, though, it resembles, a sine wave, it has slightly abrupt slopes which are discontinuous. This rises to higher frequency noise in addition to the desired signal.

Qn: Change the time base to the following: ~~and too~~

$t = \text{arange}(0, 1.001, 1.0/1000)$;

and repeat the above steps. Why does the FFT look so different.

Ans: In this case, $x[n] = S = \sin\left(2\pi \times \frac{100n}{1000}\right) + \sin\left(2\pi \times \frac{200n}{1000}\right)$.

but $X(k) = \sum_{n=0}^{1000} x_n \, e^{-i2\pi \frac{kn}{1001}}$, $k = 0, \ldots, 1000$.

$\therefore$ This imbalance between $\frac{100n}{1000}$ & $\frac{200n}{1000}$ with $\frac{kn}{1001}$ gives rise to such a different FFT.

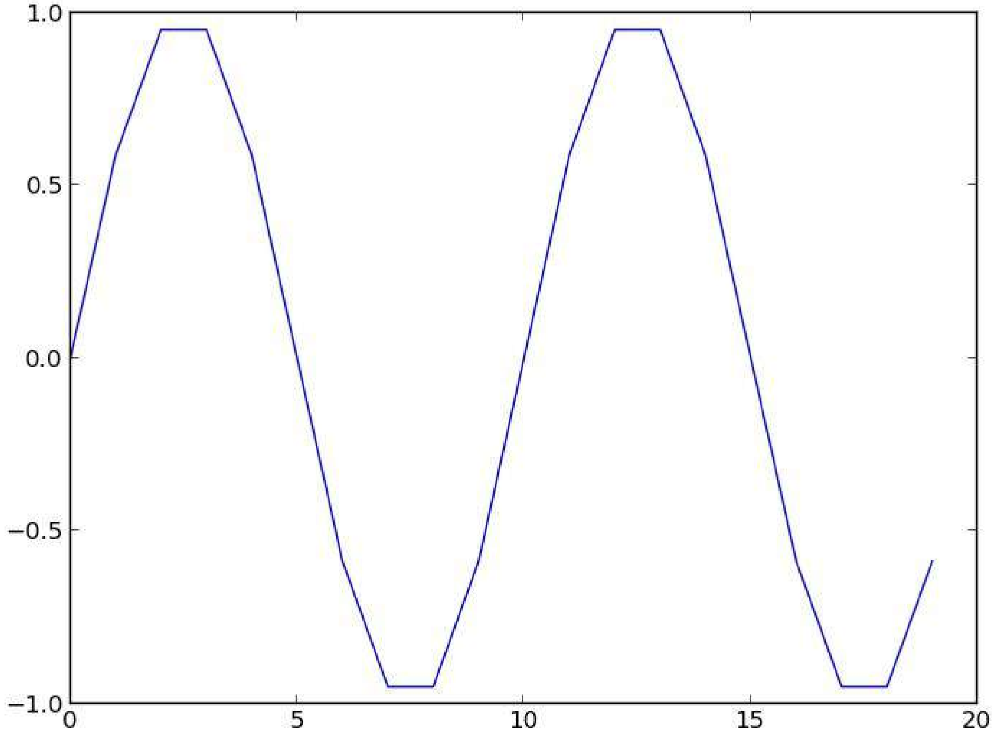


Figure: Sin wave function with irregularities due to interpolation

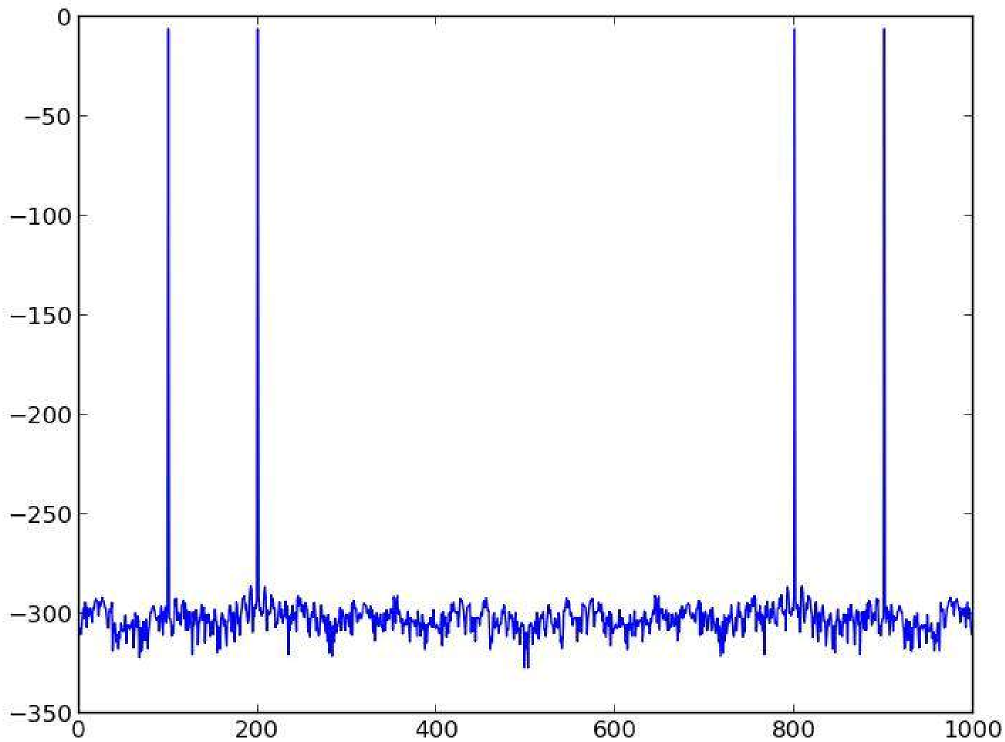


Figure: fft of the given signal

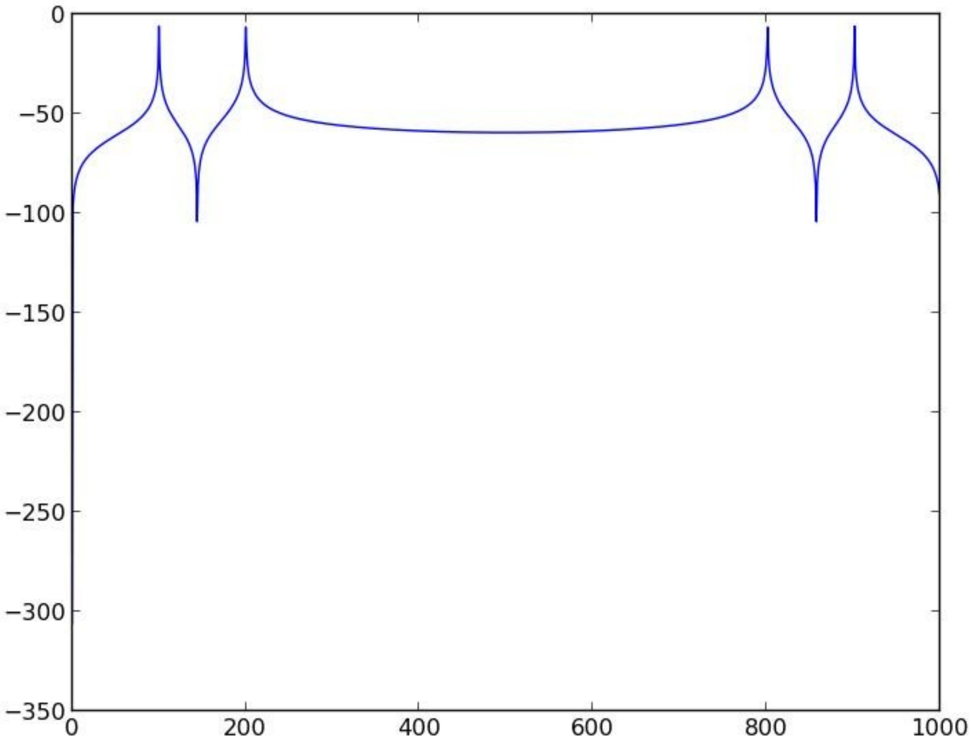


Figure: fft of the signal with time range changed

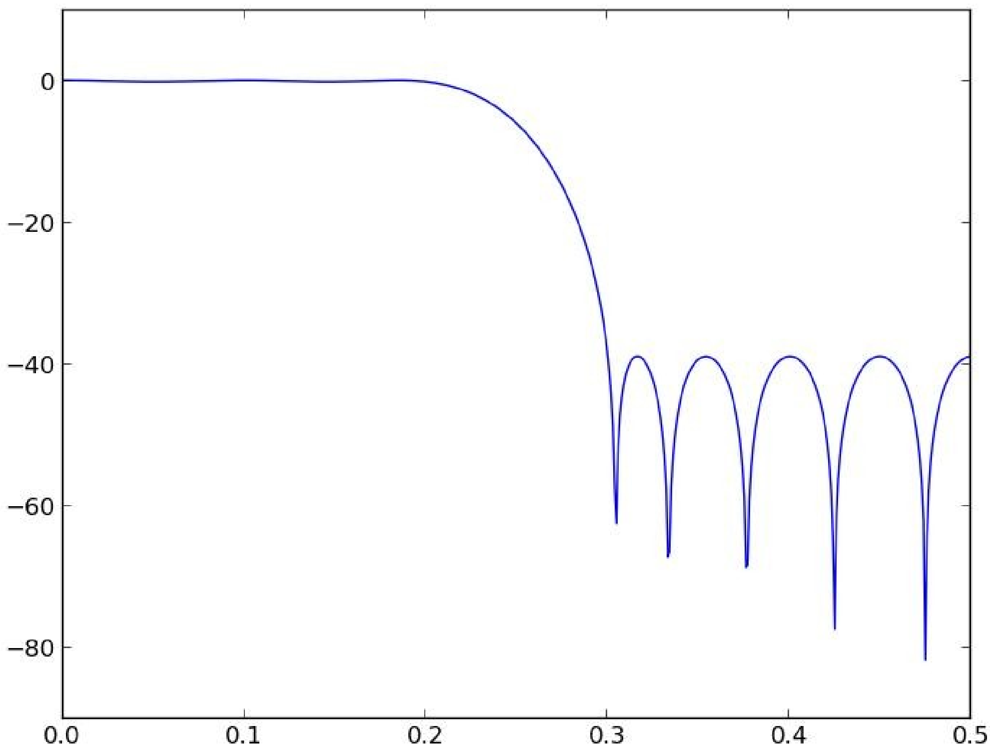


Figure: LPF with 21 taps

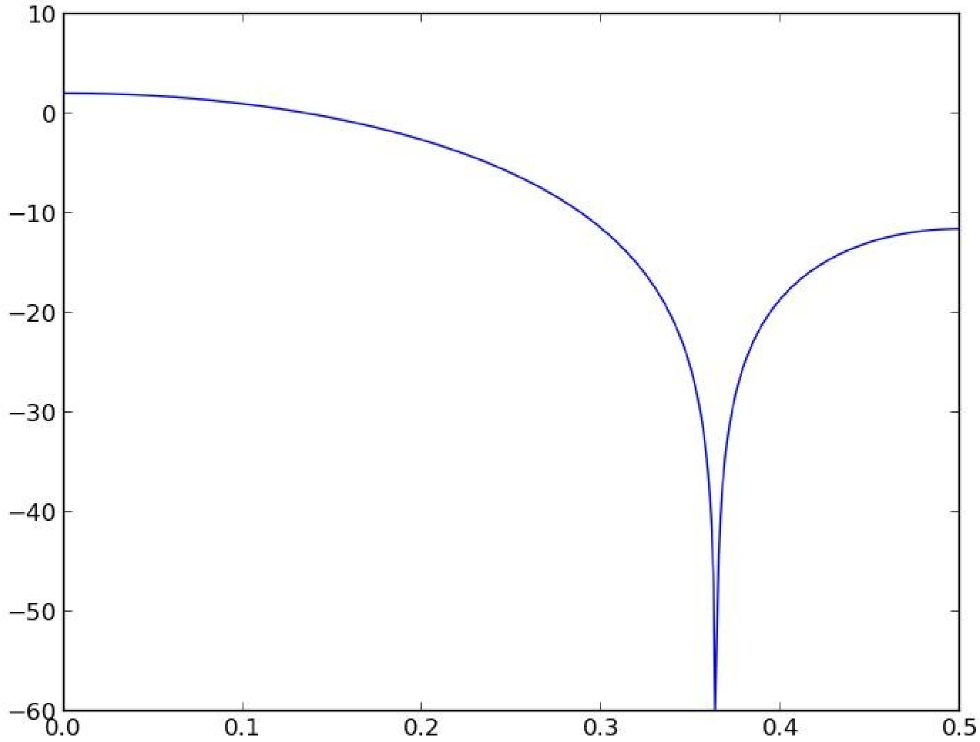


Figure: LPF with 5 taps

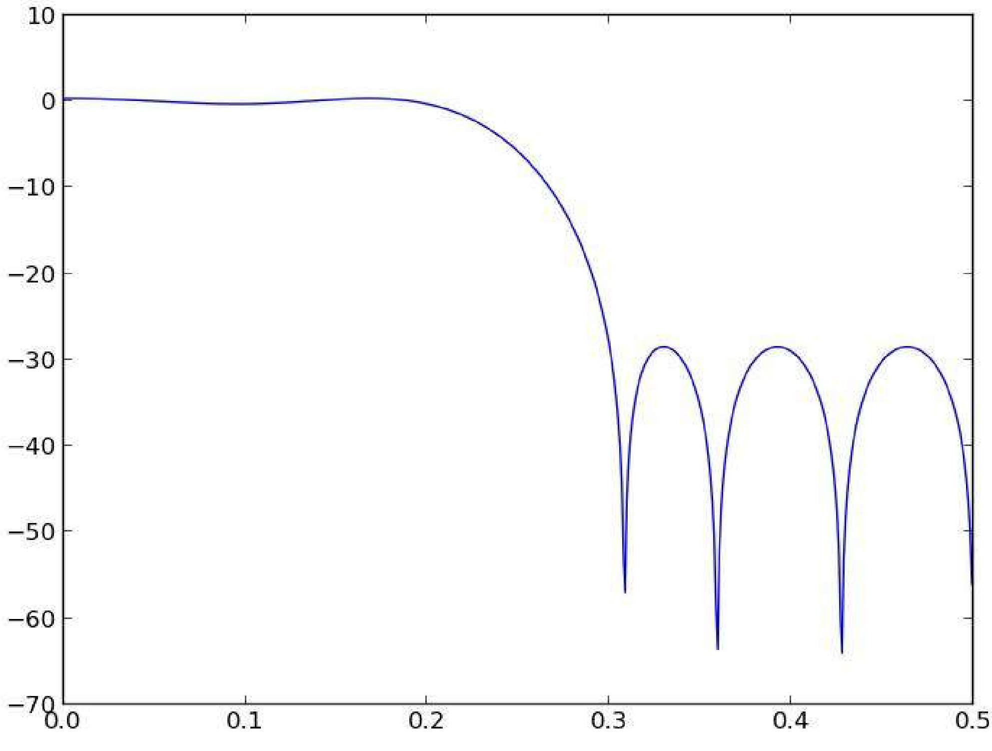


Figure: lpf with 14 taps where 30 dB attenuation happens at stop band – 300Hz

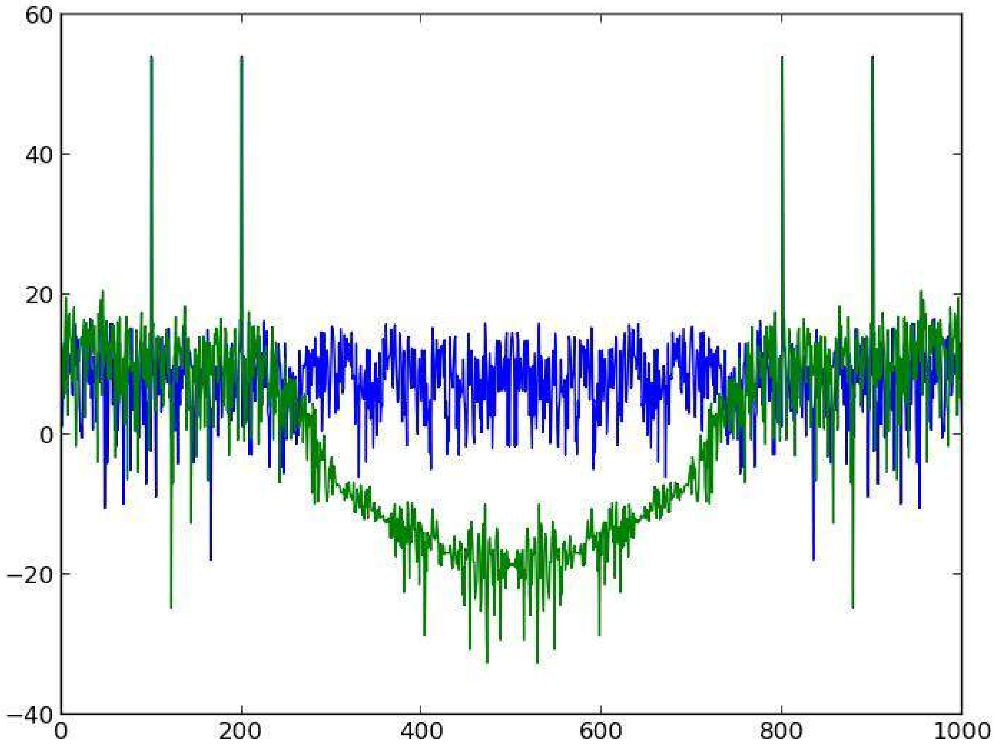


Figure: Input and filtered output

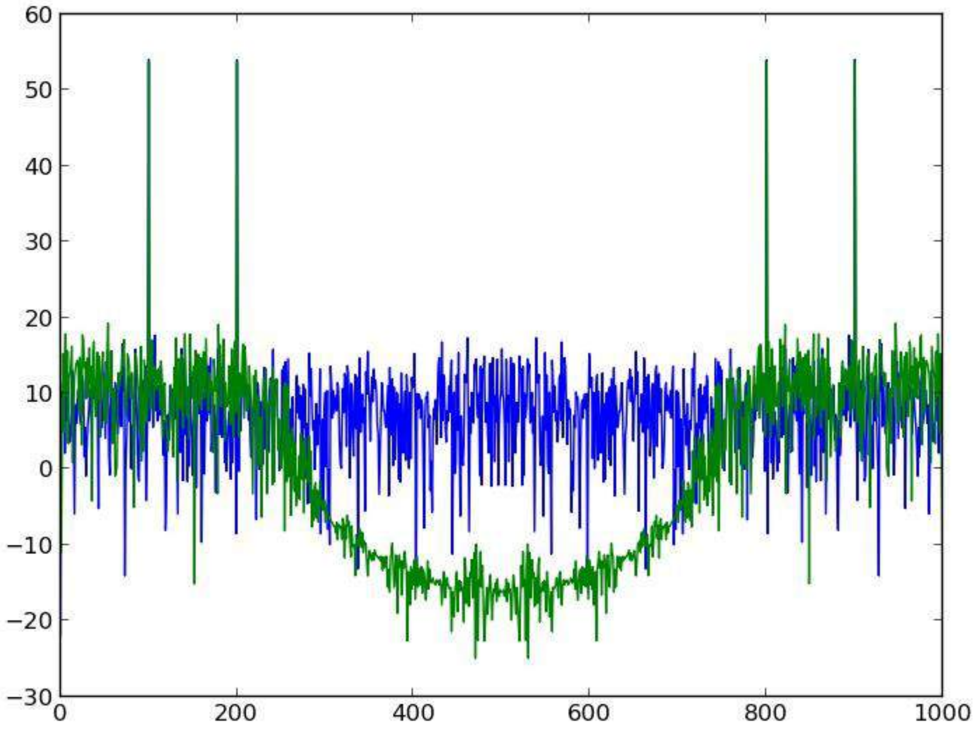


Figure: Input and filtered output, when output is got from input that is reduced to only 8 bits

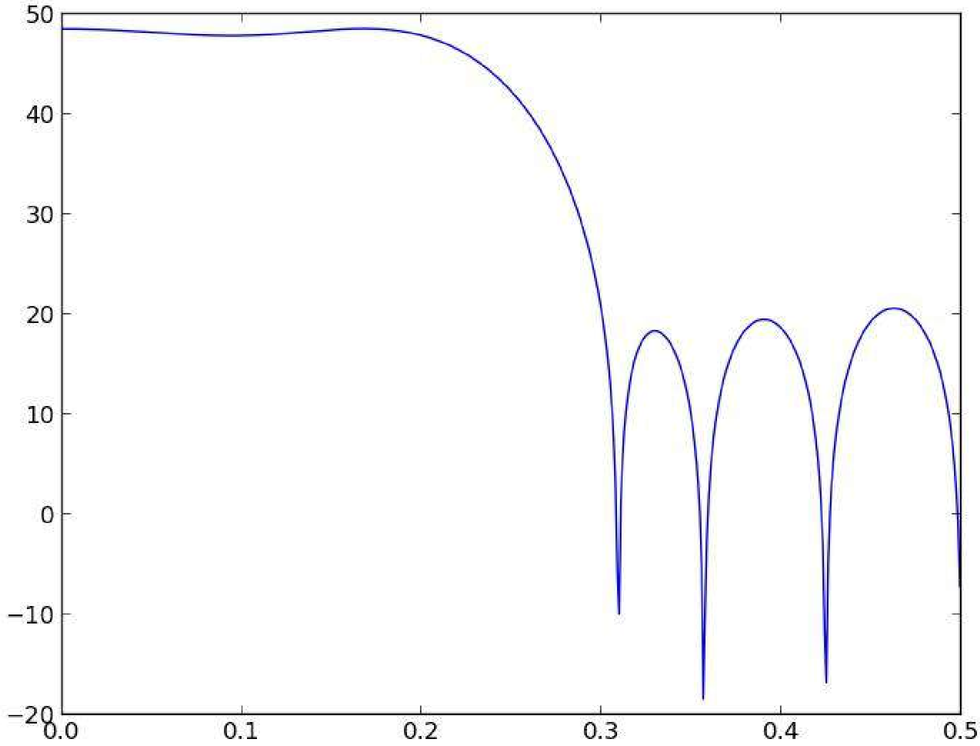


Figure: LPF with 14 taps and coefficients reduced to only 8 bits

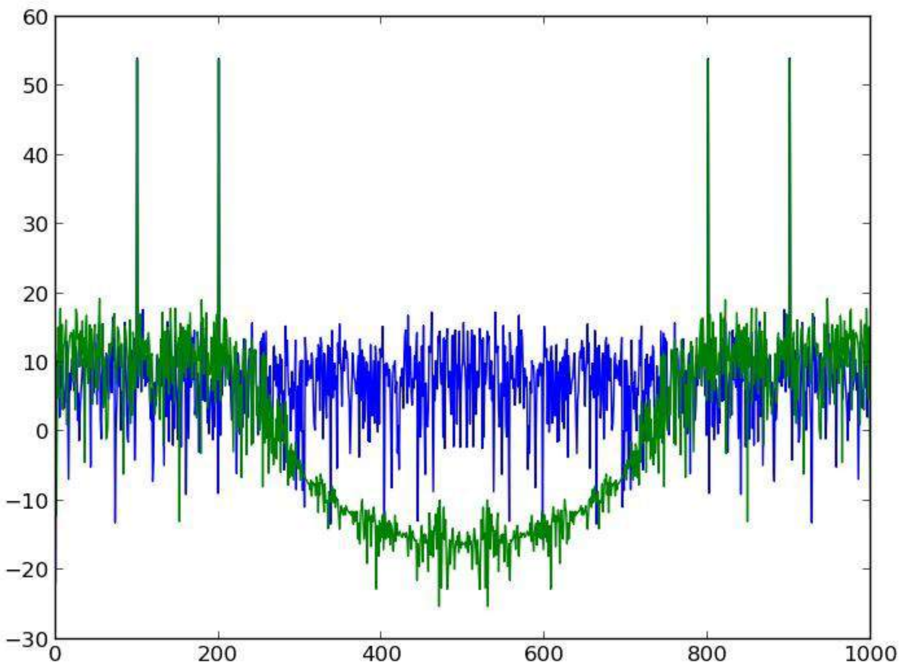


Figure: Input and filtered output, where output is filtered using the above described LPF

Filter coefficients:

Qn: What is the attenuation of the above figure in the stop band?
Ans: ~~about -76.5 dB~~ ~~about -46.37dB~~, about -37.19 dB.

Qn: What happens if you reduce the number of taps to 5?
Ans: The attenuation of the figure in the stop band increases to about ~~-20 dB~~
-11.61 dB

Qn: How many taps do you need for your filter for a 30dB stopband attenuation?
Ans: ~~about 15 taps~~ about 14 taps.

Qn: What happens to signals in the range 200-300Hz with the above filter?
Ans: The signals gets attenuated at variable degrees ranging from ~~0dB to -46.37dB~~
0dB to -37.19 dB

Filtering:
Qn: What does the filtered signal look like?
Ans: Ref figure - "Filtered Signal".

Qn: When input is 8 bits, how many bits required before decimal place for output?
Ans: ~~sine fun~~ Function "S" runs from $-\sqrt{2}$ to $\sqrt{2}$. So, we just need one bit for magnitude (0 or 1) and one bit for the sign.
Hence we need only 2 bits.

Qn: How many bits in the coefficients of the LPF, for 30dB attenuation.
Ans: We need 8 bits. Because, then LPF becomes 20dB attenuated at stop band.

~~LPF coefficients~~