Multimedia Signal Processing 1st Module

17/11/2011

Provide a detailed explanation of proposed solutions without skipping steps

Ex.1 (Pt.12)

A filter $h(n)$ is the composed by the cascade of two FIR filters $h_1(n)$ and $h_2(n)$, where:

$$h_1(n) = \left\{1, 0, \frac{1}{4}\right\}, h_2(n) = \left\{1, 0, -\frac{1}{4}\right\}$$

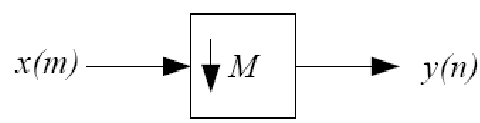
1. Draw the zeros-poles diagram.
2. Define the Impulse response, $h(n)$ of the filter, what can we say concerning its length?
3. Provide an approximated representation of the filter **amplitude** for normalized pulsations from $-\pi$ to π. .
4. The filter is used on a signal $x(n)$ sampled at 20 ksamples/s, the signal before sampling is $x(n) = \cos(2\pi \cdot f \cdot t)$ where f = 5 kHz. Define the first 5 output samples, y(n), of the filter.
5. Define the block diagram for the filter implementation.

Ex.2 (Pt.12)

A low-pass filter has the following z-transform:

$$H(z) = \frac{1 + \frac{1}{2} z^{-1}}{1 - \frac{1}{2} z^{-1}}$$

1. Draw the zeros-poles diagram.
2. Provide an approximated representation of the filter **phase** for normalized pulsations from $-\pi$ to π.



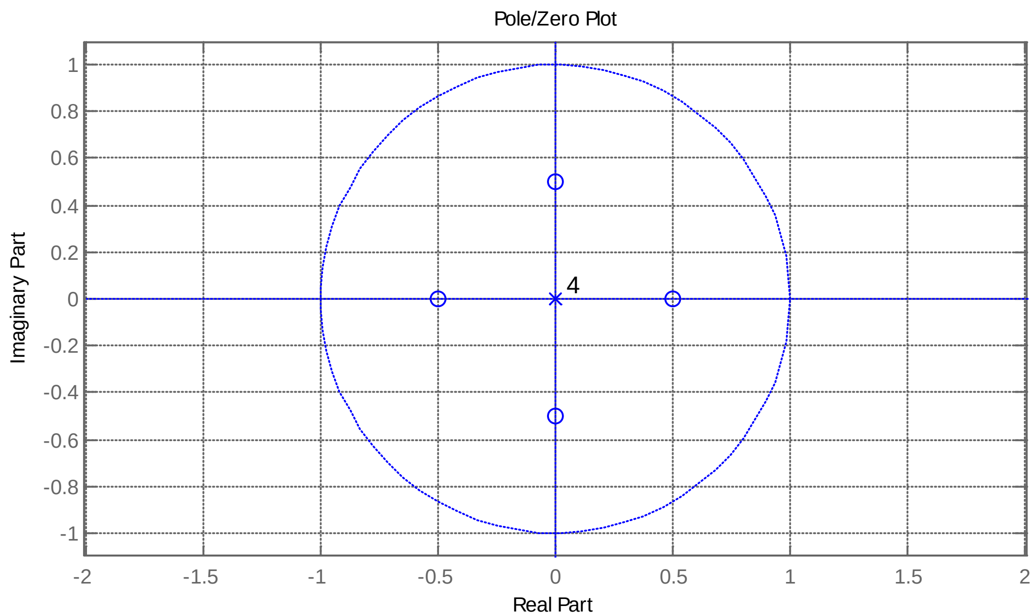
3. The filter is used as a low-pass filter in a decimation 2:1 process for a sinusoidal function $x(n) = \sin(2\pi \cdot f \cdot t)$ where f = 100Hz and the sampling frequency is 800Hz. What will be the SNR after the cascade of the filter and the decimation? [where “noise” are the signal replicas in the base band due to decimation]

Ex.3 (Pt.12)

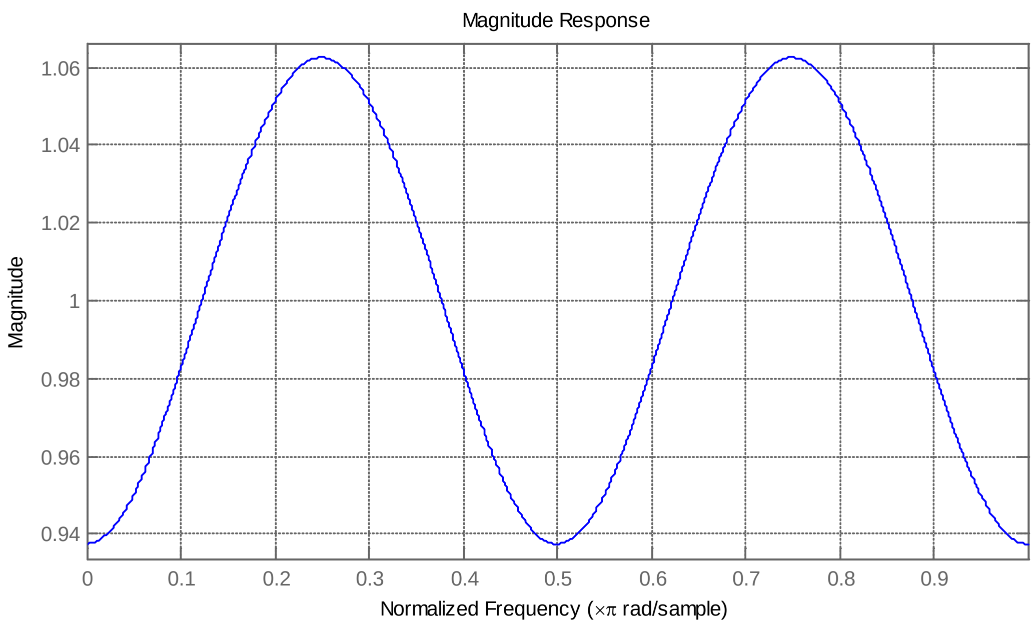
1. Build a signal sum of three different sinusoids sin(2πft) at the radian frequencies w1= π/8, w2=π/10 w3= π/3. The signal is defined over a temporal axis of 512 samples. (Assume that the sampling period T=1).
2. Modify frequency sample rate with a rational factor 11/12. (HINT: Use Matlab function ‘fir1’ to design the filters).
3. Compute and plot the frequency response of both filters.
4. Plot the signals in the time and in the frequency domain (only modula).

Solutions

Ex.1

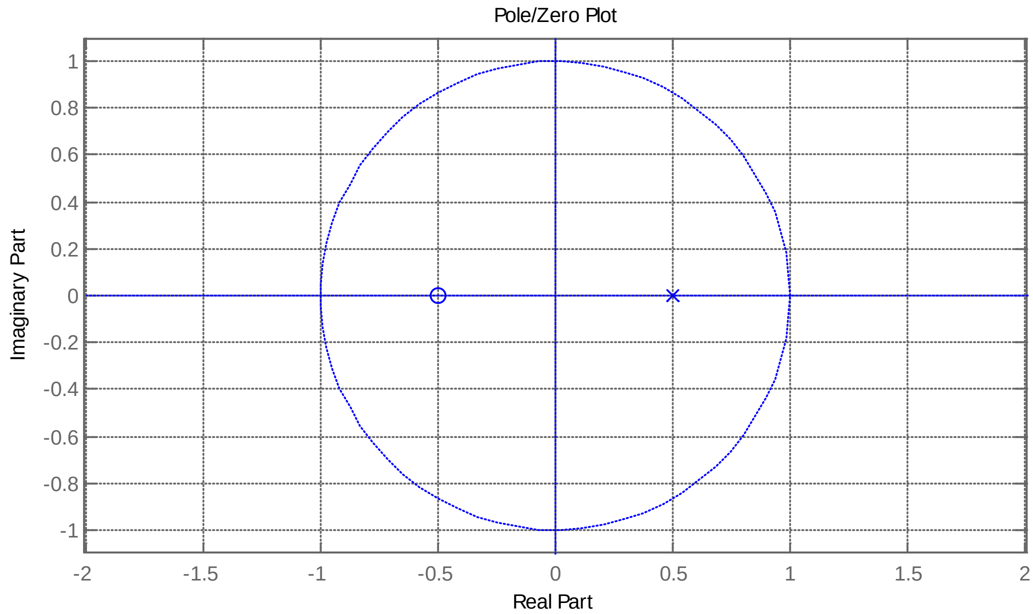


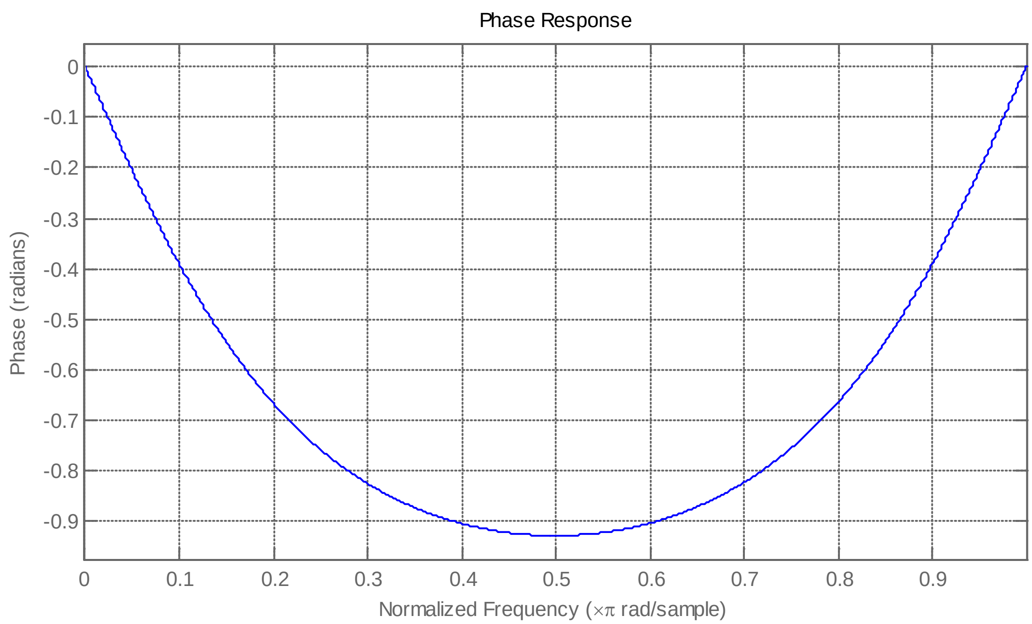
h(n)={1,0,0,0,-1/16}



y(n)={1,0,-1,0,1-1/16}

Ex.2:





After the downsampling , in the base band, there will be a replica of the original signal, in particular, the original DTFT was a couple of impulses (in normalized frequencies) at ±π/4, after the downsampling they will move to ±π/2 but a replicas will generate impulses at ±3π/2.

The SNR can then be evaluated as $SNR = \frac{\left|H\left(\frac{\pi}{2}\right)\right|^2}{\left|H\left(\frac{3\pi}{2}\right)\right|^2} = 1$

Ex.3

```
w1=pi/16
w2=pi/8
w3=pi/2
n=[0:200];
x = cos(w1*n)+0.5*cos(w2*n)+2*cos(w3*n);

upsample_factor = 11;
downsample_factor = 12;

h1 = fir1(30, 1/upsample_factor);
h2 = fir1(30, 1/downsample_factor); %   The new cut-off frequency is now
referred to the new
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                                                % sampling frequency
(fs*upsample_factor)
Nfft=1024;
w=2*pi*[0:Nfft-1]./Nfft;
H1=fft(h1,Nfft);
H2=fft(h2,Nfft);

%   Upsampling (interpolation)
y_up = zeros(length(x)*upsample_factor, 1);
y_up(1:upsample_factor:end) = x;

y_up = filter(upsample_factor*h1, 1, y_up);

%   Dowsample (decimation)
y_down = filter(h2, 1, y_up);
y_down = y_down(1:downsample_factor:end);

X=fft(x,Nfft);
Y=fft(y_down,Nfft);

figure, subplot(211), plot(w,20*log10(abs(X)))
subplot(212), plot(w,20*log10(abs(Y)))

figure, subplot(211), plot(n,x)
subplot(212), plot(n,y_down)
```