Communications and Signal Processing Lab

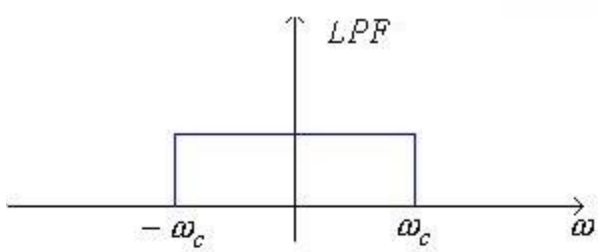
Assignment No.-3

EE21MTECH14002

1. Decimation:

When we use a low pass filter also known as anti-aliasing filter before downsampling a signal than this is known as decimation and the signal that is generated from this process is known as decimated signal.

If we have a input signal let's say **x[n]** and low pass filter with impulse response **h[n]** and cut off frequency equals $\boldsymbol{\pi/M}$**.** Then **xf[n]** is a signal generated at the output of a lowpass filter. Since, **xf[n]** contains more samples than **x[n]** due to convolution we only consider samples in the main lobe of a signal i.e. **sinc** and we discard first and last **(Lh-1/2)** samples where **Lh** is a length of a impulse response array. Then this new **xf[n]** is passed from a down-sampler or compressor that compress **xf[n]** by a factor of **'M'** and generate **xd[n].** This **xd[n]** is nothing but our decimated signal and this complete process is known as **Decimation.**



Anti-Aliasing Filter

Wc= π/M **, Gain= 1**

Designing of Anti-Aliasing Filter:

The simplest method for the designing of the finite impulse response filter is known as windowing method.

Step1: Let the desired ideal frequency response of a low pass filter is $H_d(e^{jw})$. Wc= π/M

Step2: Take IFFT of $H_d(e^{jw})$ to get $h_d[n]$.

Step3: Since $h_d[n]$ has infinite length, truncate it using a finite length window function w[n] to get h[n].

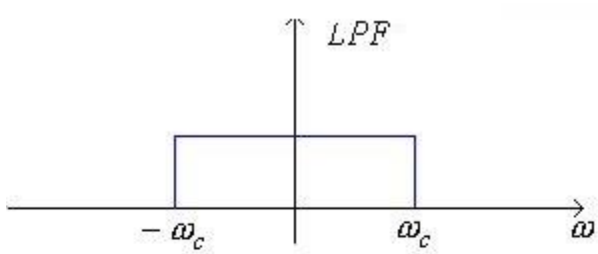
$$\mathbf{h[n] = h_d[n] \times w[n]}$$

Step4: We can see our practical filter frequency response by taking FFT of h[n] which is $H(e^{jw})$ and you can plot magnitude and phase response.

2. Interpolation:

Interpolation is just the opposite of decimation. When we use a low pass filter also known as anti-imaging filter after upsampling of a received signal than this is known as Interpolation and the signal that is generated from this process is known as Interpolated signal.

If we have a received signal let's say **xd[n].** Then we upsample with signal using a upsampler of scaling factor 'L'. Upsampler gives xu[n] as its output then we pass this output from a anti-imaging filter whose cut-off frequency is equals π/L and gain equals **"L"** then we get **y[n]** as a output of anti-imaging filter we discard first and last **(Lh-1/2)** samples from **y[n]** where **Lh** is a length of a impulse response array. This **y[n]** is a interpolated signal and this process is know as **interpolation**.



Anti-Imaging Filter

Wc= π/L , Gain= L

Designing of Anti-Imaging Filter:

The simplest method for the designing of the finite impulse response filter is known as windowing method.

Step1: Let the desired ideal frequency response of a low pass filter is $H_d(e^{jw})$. Wc= $\boldsymbol{\pi/L}$

Step2: Take IFFT of $H_d(e^{jw})$ to get $h_d[n]$.

Step3: Since $h_d[n]$ has infinite length, truncate it using a finite length window function w[n] to get h[n].

$$\mathbf{h[n] = h_d[n] \times w[n]}$$

Step4: We can see our practical filter frequency response by taking FFT of h[n] which is $H(e^{jw})$ and you can plot magnitude and phase response.