EE6133:Multirate Signal Processing

Computer Assignment

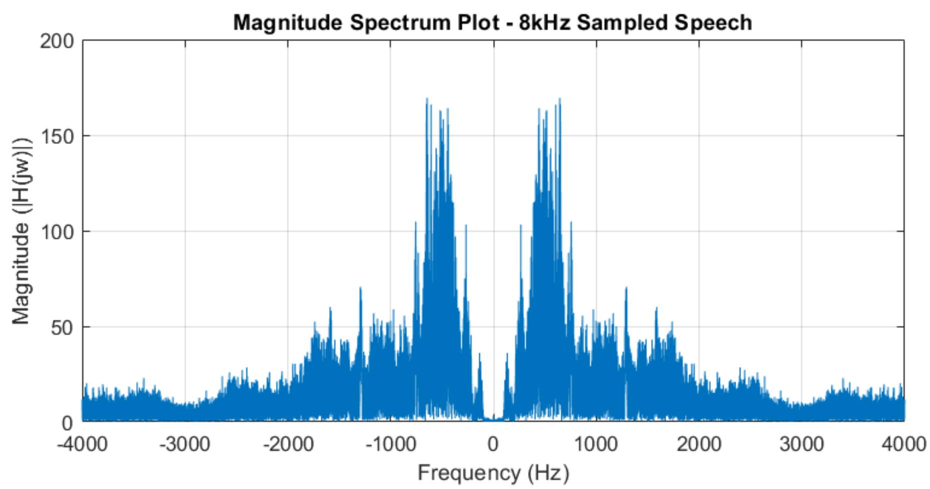
K.R.Srinivas (EE18B136)

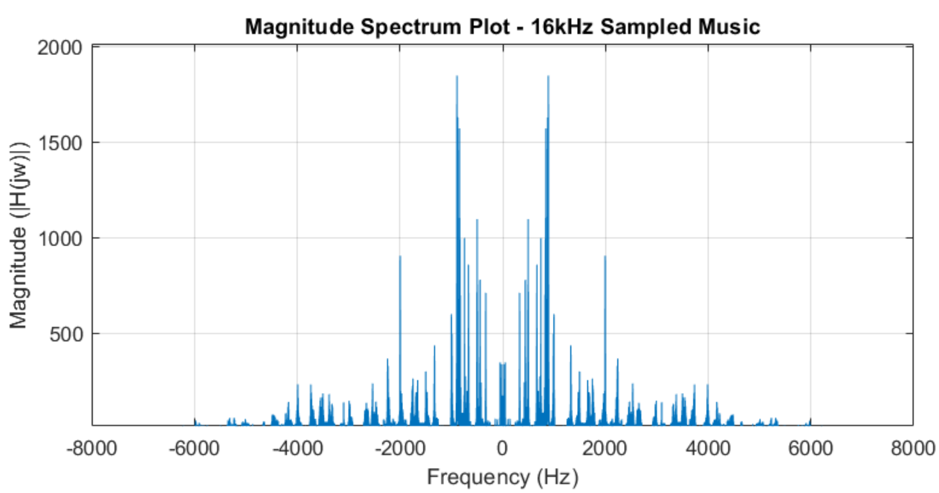
Abstract :

This assignment focuses on experimenting Up/Downsampling, Decimation and Interpolation on 2 sets of audio files sampled at 8kHz & 16kHz and understanding their effect on the audio quality as well as their magnitude spectrum.

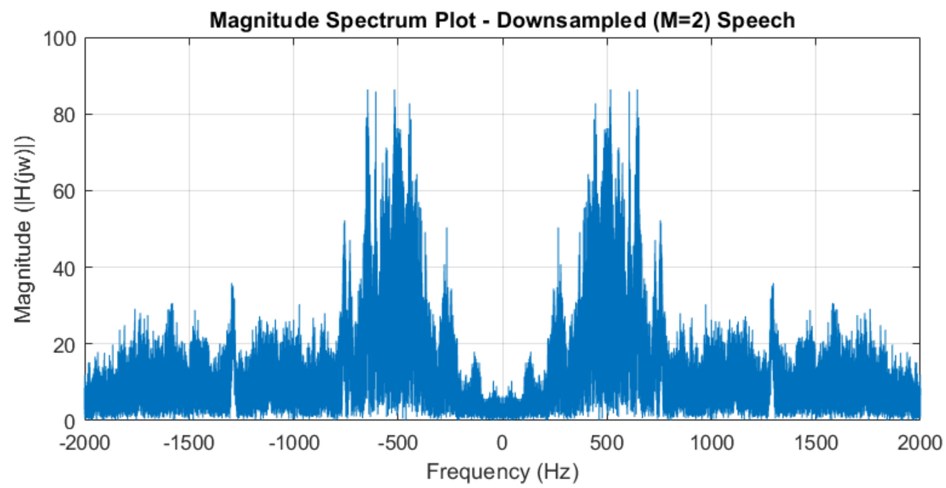
Question 1 :

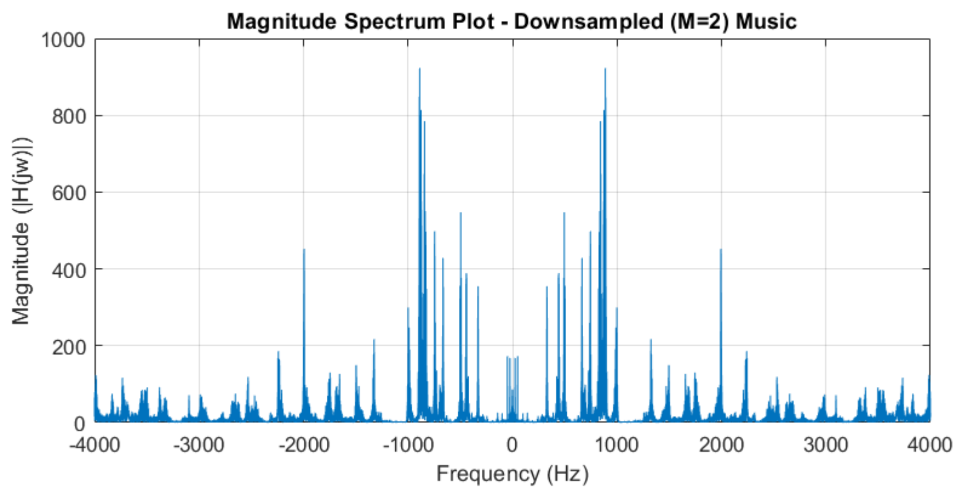
<u>Magnitude Spectrum of the Original Signal</u>





Magnitude Spectrum of the Downsampled (M = 2) Signal





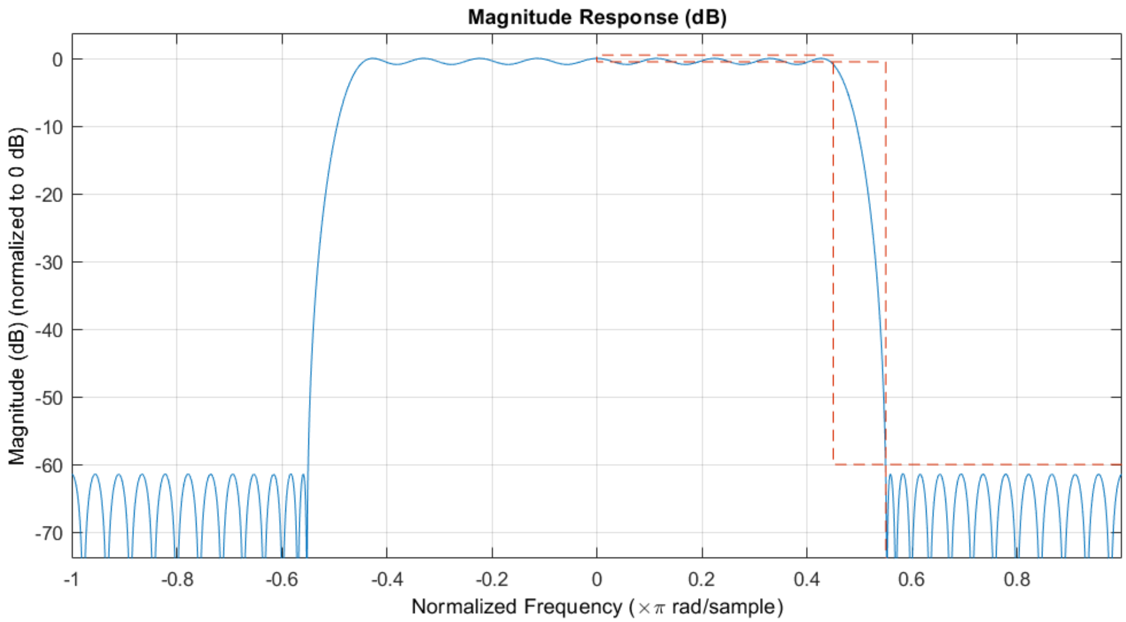
The Magnitude Spectrum of the original signal shows that the speech signal (Fs = 8 kHz) has significant information content in the spectrum upto Fs/2 ($\omega = \pi$),while the music signal (Fs = 16 kHz) has most of the information in the spectrum upto Fs/4 (digital frequency = $\pi/2$).

Downsampling (M = 2) both the signals lead to aliasing, which is more pronounced in the speech signal compared to music (since the former has significant spectrum content upto $\omega = \pi$, while downsampling without aliasing requires information to be contained within $\omega \leq \pi/M$) In the case of music, there is aliasing due to the leak of spectrum beyond ($\omega = \pi/2$) but doesn't affect the spectrum or audio significantly due to their relatively lower strength. The new sampling rate is R = Fs/2.

Downsampled (M = 2) speech8khz.wav : DS_speech8khz.wav
Downsampled (M = 2) music16khz.wav: DS_music16khz.wav

Question 2 :

<u>Magnitude Spectrum of the Equiripple Filter ($\omega_p = 0.45\pi$, $\omega_s = 0.55\pi$)</u>

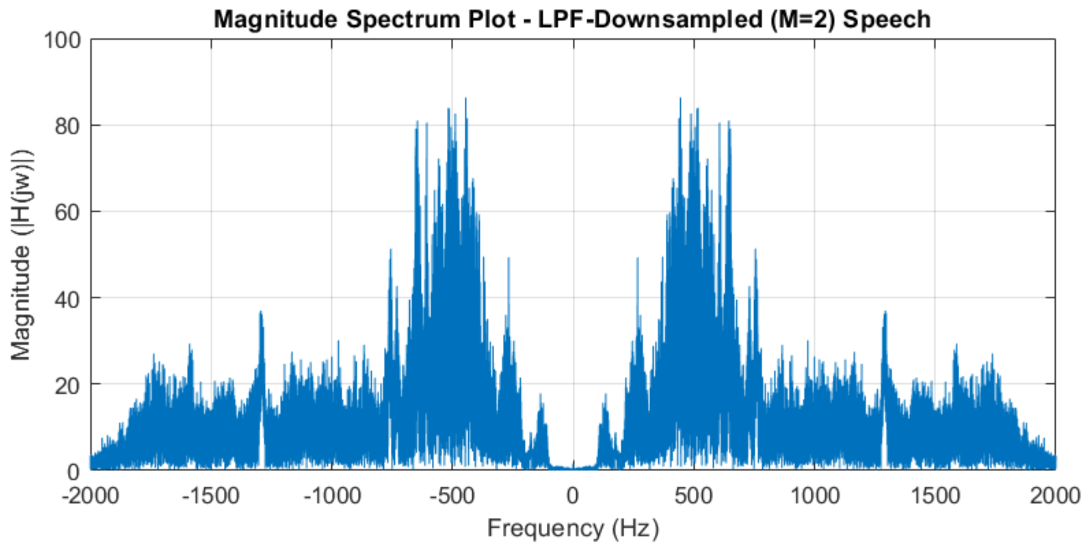


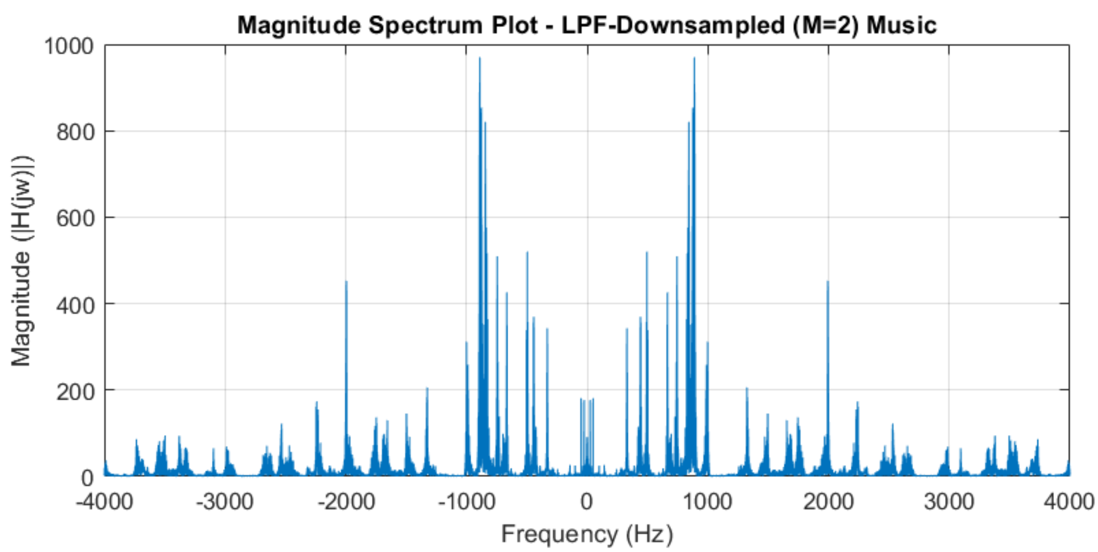
In order to overcome the corruption of audio due to aliasing caused by downsampling (M = 2) , we design an equiripple LPF (a type of linear FIR digital filter) that limits spectrum content to less than Fs/4 ($\omega \leq \pi/M$).

Filter Specifications :

- Passband frequency (ωp) = 0.45π
- Stopband frequency (ωs) = 0.55π
- Passband ripple (δ) = 1 dB
- Attenuation = 60 dB

Magnitude Spectrum after Anti-Aliasing and Downsampling (M = 2)



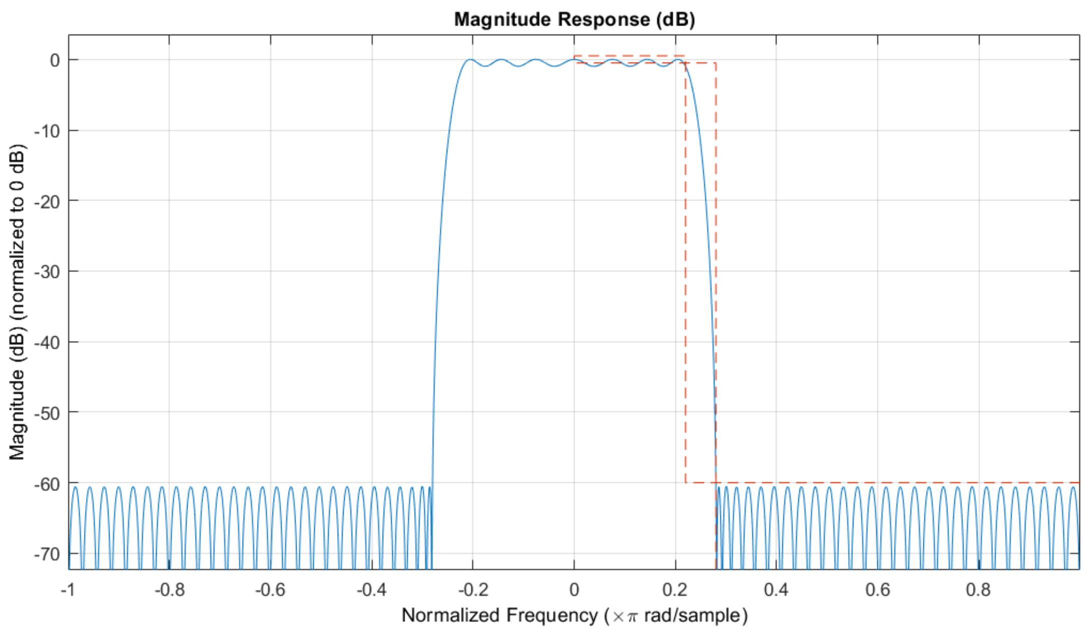


Filtering the audio signals before downsampling has removed the spectrum content beyond $\omega = \pi/2$ (Fs/4). Clearly this leads to loss of high frequency information, which can distort the audio, nevertheless it has managed to avoid aliasing significantly which was present in Q1. The loss of high frequency content is not very clear in the output audio because of their low strength compared to low frequency content. The output audio quality in Q2 is better than the audio quality obtained by direct downsampling as in Q1. The new sampling rate is R = Fs/2.

AA Filtered Downsampling speech8khz.wav : LPF_DS_speech8khz.wav
AA Filtered Downsampling music16khz.wav : LPF_DS_music16khz.wav

Question 3:

<u>Magnitude Spectrum of the Equiripple Filter ($\omega_p = 0.22\pi$, $\omega_s = 0.28\pi$)</u>



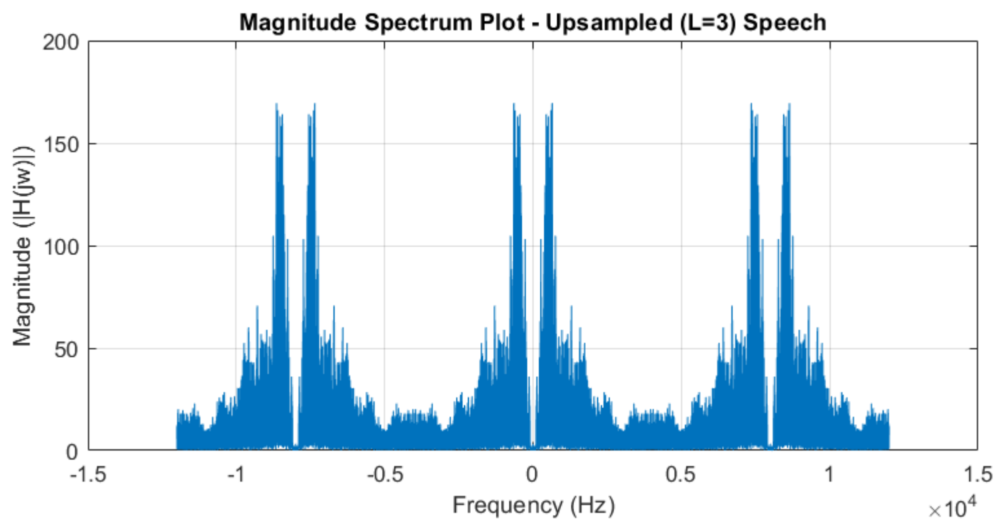
In order to overcome the corruption of audio due to aliasing caused by downsampling (M = 4) , we design an equiripple LPF (a type of linear FIR digital filter) that limits spectrum content to less than $\omega \leq \pi/M$ (0.25π).

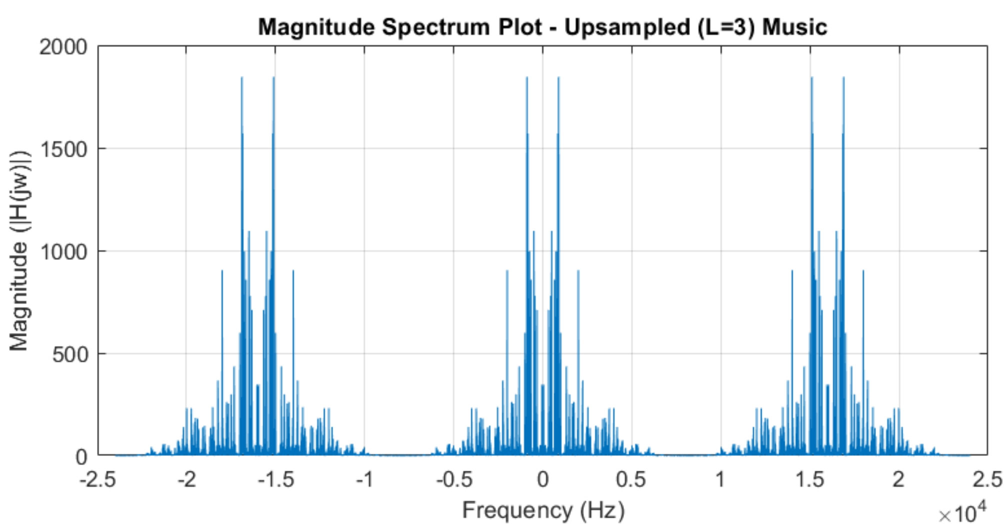
Filter Specifications :

- Passband frequency (ωp) = 0.22π
- Stopband frequency (ωs) = 0.28π
- Passband ripple (δ) = 1 dB
- Attenuation = 60 dB

Question 4 :

Magnitude Spectrum of the Upsampled Signal (L = 3)

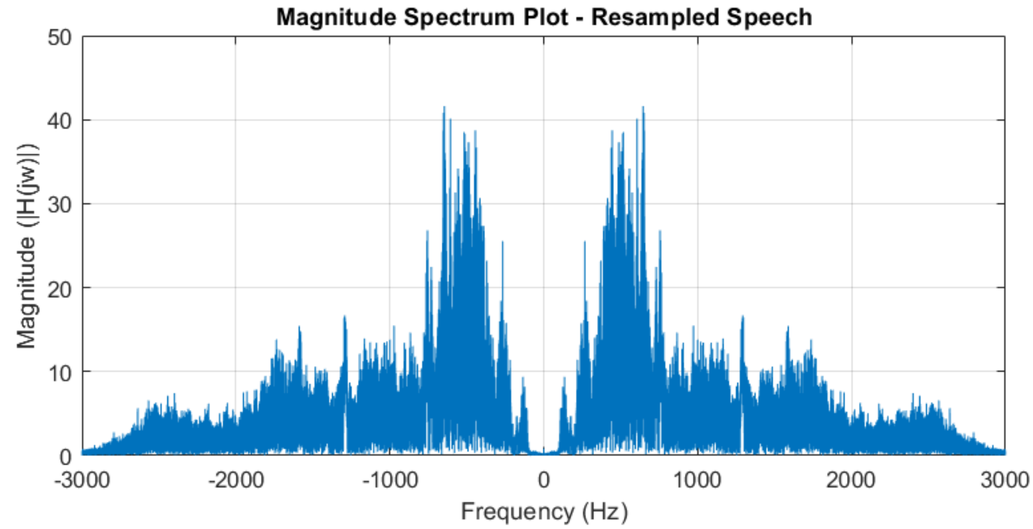


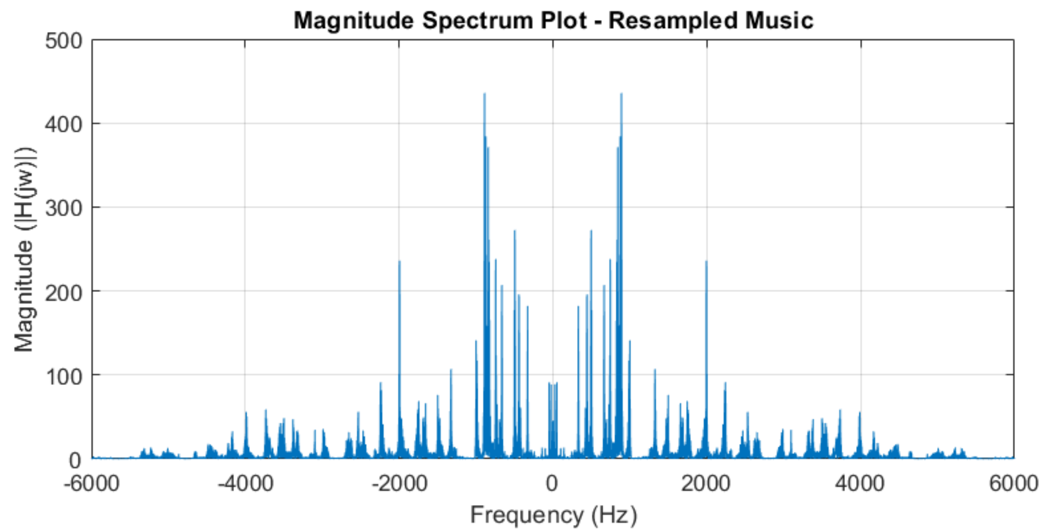


Here as the first step we upsample the input audio signal by L = 3 (which means the new sampling rate Fs_up = 3 x Fs) and hence Fs/2 which earlier corresponded to $\omega = \pi$, now corresponds to $\omega = \pi/3$. Clearly there is no loss of information while upsampling unlike the decimation done before. The actual spectrum content of the original signal is now contained in the digital band $|\omega| \leq \pi/3$.

Upsampled speech8khz.wav : US_speech8khz.wav
Upsampled music16khz.wav : US_music16khz.wav

Magnitude Spectrum of the Re-sampled Signal (M↓ = 4, L ↑= 3)





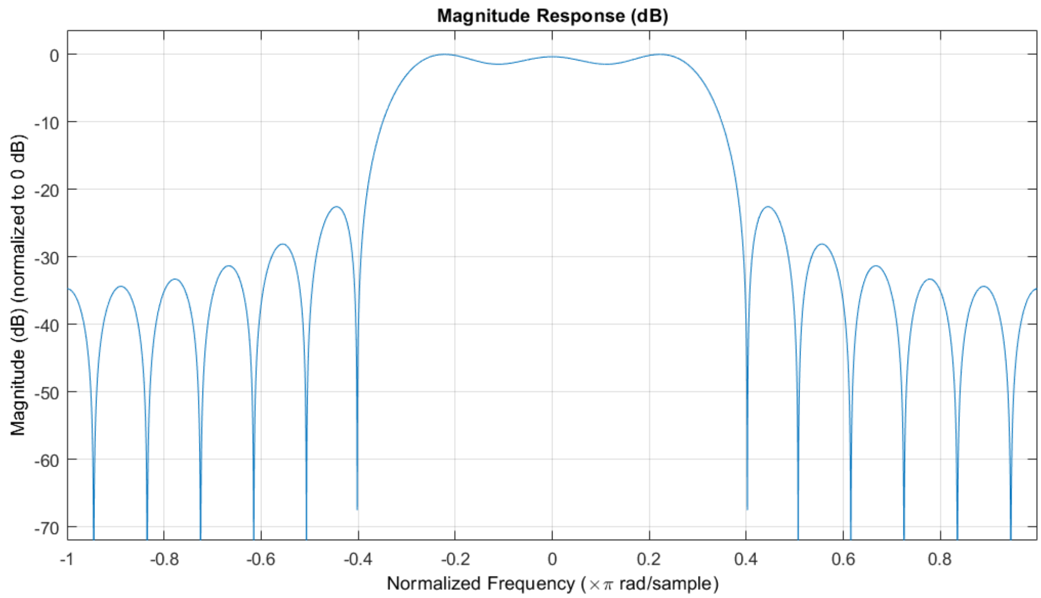
The upsampled (L = 3) signal is now decimated using a cascade of equiripple-LPF ($\omega c = 0.25\pi$) and downsampler (M = 4). On applying the anti-aliasing filter we lose the information content in the band $\pi/4 \leq |\omega| \leq \pi/3$ (25% loss of spectrum content), but then can downsample (M = 4) to fewer samples without aliasing effect. However notice that the information contained in this band are low-magnitude high frequency components and hence doesn't cause much distortion to the output (evident from the output audio files as well). The new sampling rate of the signal is R = 3Fs/4 .

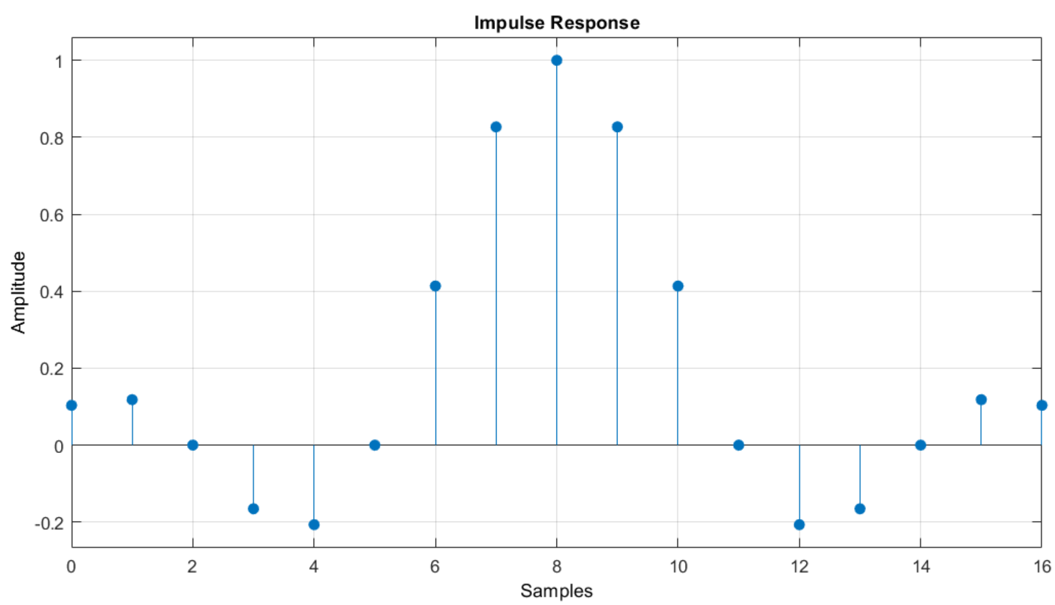
For input signal amplitude of A, the output signal has amplitude LA/M (0.75A).

Resampled speech8khz.wav : RS_speech8khz.wav
Resampled music16khz.wav : RS_music16khz.wav

Question 5 :

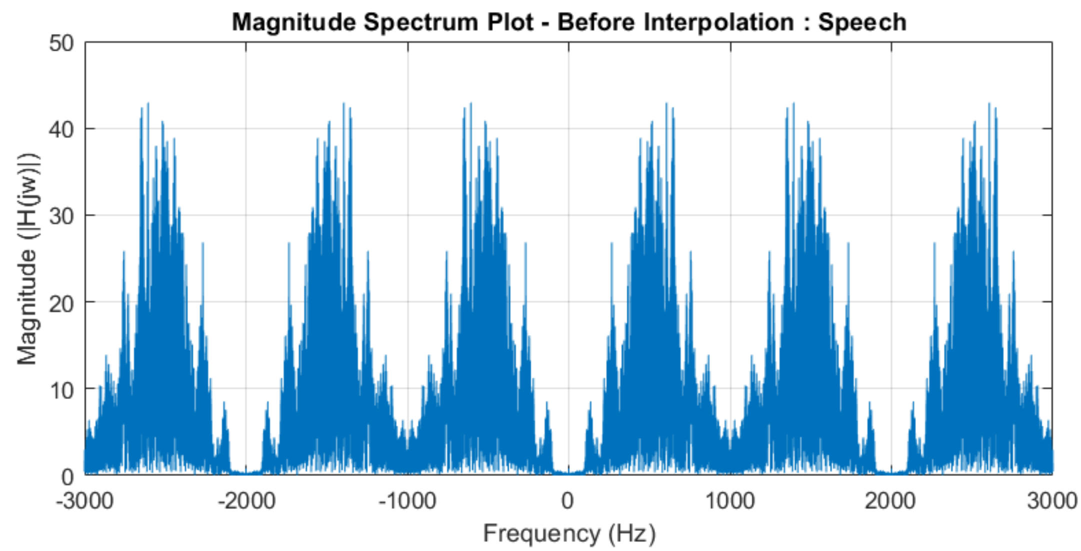
Magnitude Spectrum & Impulse Response of the Interpolator

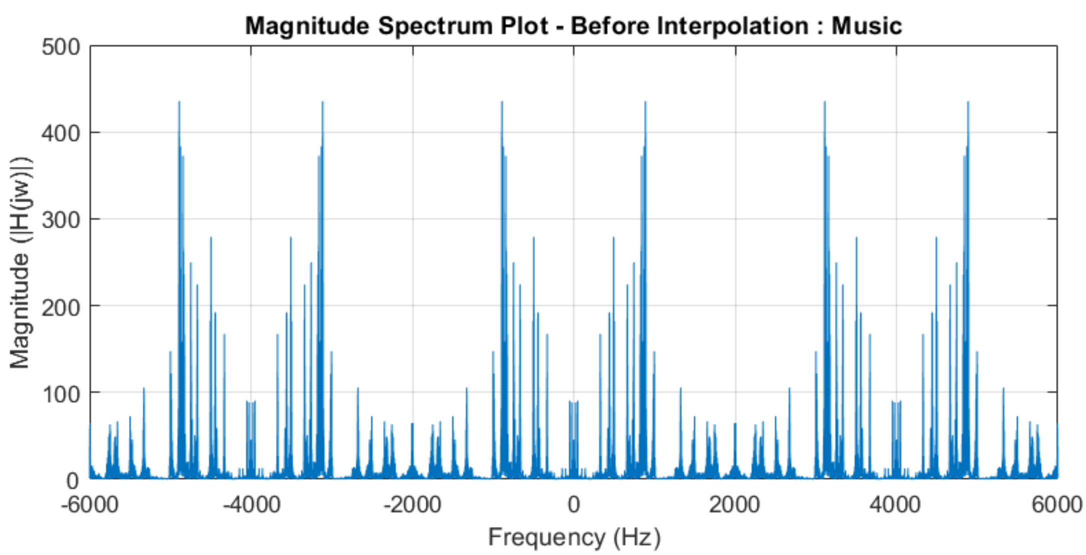




It is asked to construct an FIR interpolator for an upsampled (L = 3) sequence, with support [-8,8], basically 17 coefficient FIR filter. We use the intfilt() tool available in Signal Processing toolbox to construct a linear phase FIR bandlimited interpolator setting α = 1. We set p = 3 to get 17 coefficients.

<u>Magnitude Spectrum before Interpolation</u>

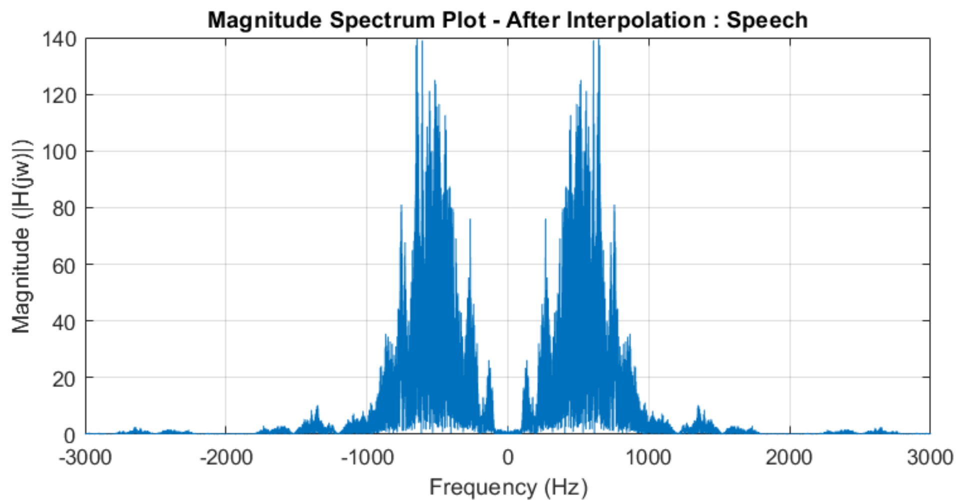


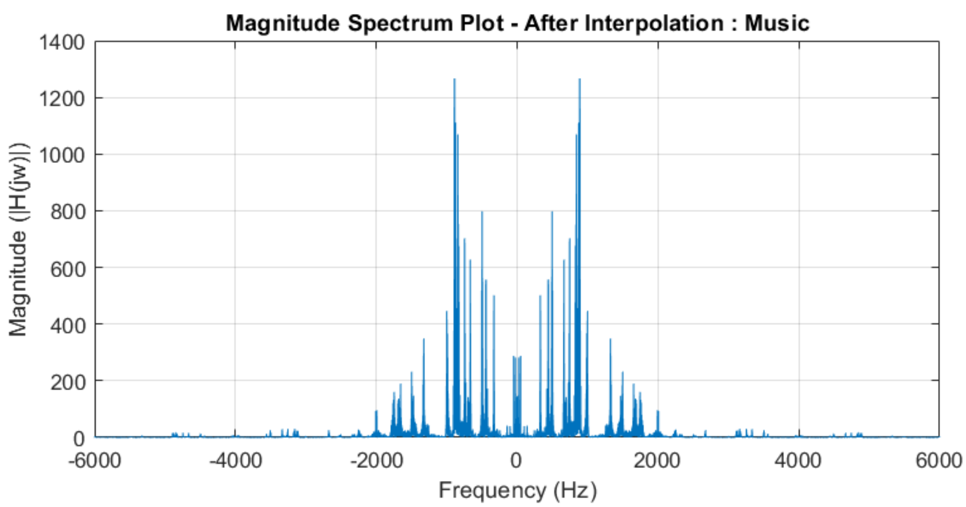


In this question, we first band limit the input signal with the AA filter (ωc = 0.25π) followed by downsampling (M = 4). Note that the original signal contained information in the band |ω| ≤ π, of which 75% is lost because of the AA filtering only. The resultant signal is downsampled without aliasing and subsequently upsampled (L = 3). Clearly, This hints the reason as to why the quality of even the interpolated signal in Q5 is worse than Q4, where the information loss was lower.

For input signal amplitude of A, the output signal has amplitude LA/M (0.75A).

Magnitude Spectrum after Interpolation





Thus the resampled signal is now interpolated (kinda LPF) with the Linear Phase Interpolator constructed initially taking into consideration the upsampling factor. Thus in the resultant spectrum we can clearly see the image rejection (additional spectrum copies created by upsampling). As mentioned in the previous page the order of AA filtering has affected the signal quality. Q4 audio sounds clearer than Q5. The new sampling rate of the signal is R = 3Fs/4 .

Interpolated speech8khz.wav : INT_speech8khz.wav
Interpolated music16khz.wav: INT_speech16khz.wav

The MATLAB code for this experiment can be found here .