Lec. 11 Application of Digital Filters

11.1 Speech Enhancement and Filtering

This section investigates applications of speech enhancement using a pre-emphasis filter and speech filtering using a bandpass filter.

11.1.1 Pre-Emphasis of Speech

A speech signal may have frequency components that fall off at high frequencies. In some applications such as speech coding, to avoid overlooking the high frequencies, the high-frequency components are compensated using pre-emphasis filtering. A simple digital filter used for such compensation is given as:

$$y(n) = x(n) - \alpha x(n-1), \qquad (11.1)$$

Where, α is the positive parameter to control the degree of pre-emphasis filtering and usually is chosen to be less than 1. The filter described in Equation (11.1) is essentially a highpass filter. Applying z-transform on both sides of Equation (11.1) and solving for the transfer function, we have:

$$H(z) = 1 - \alpha z^{-1}. \qquad (11.2)$$

The magnitude and phase responses adopting the pre-emphasis parameter $\alpha = 0.9$ and the sampling rate $f_s = 8,000$ Hz are plotted in Fig. (11.1 a) using MATLAB.

Figure (11.1 b) compares the original speech waveform and the pre-emphasized speech using the filter in Equation (11.2).

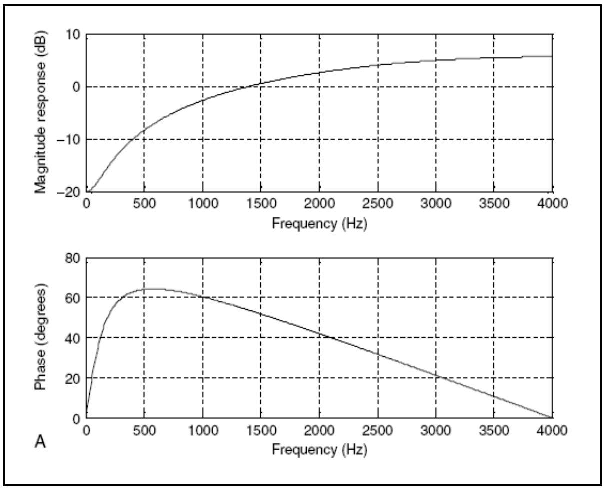


Fig.(11.1a) Frequency responses of the pre-emphasis filter.

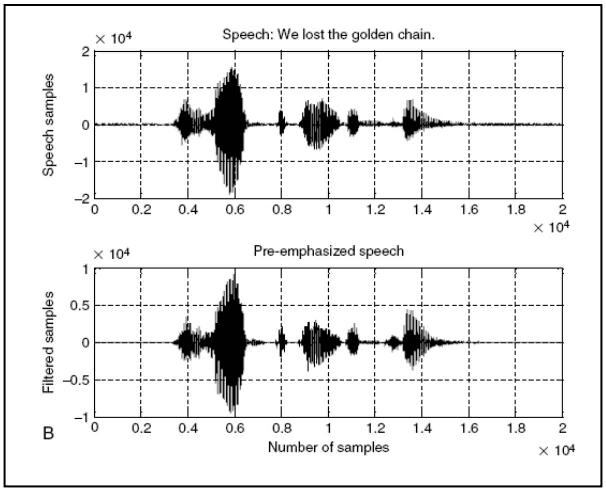


Fig.(11.b) Original speech and pre-emphasized speech waveforms.

The fast Fourier transform (FFT) is applied to estimate the spectrum of the original speech and the spectrum of the pre-emphasized speech. The plots are displayed in Fig. (11.2).

From Fig.(11.2), we can conclude that the filter does its job to boost the high-frequency components and attenuate the low-frequency components.

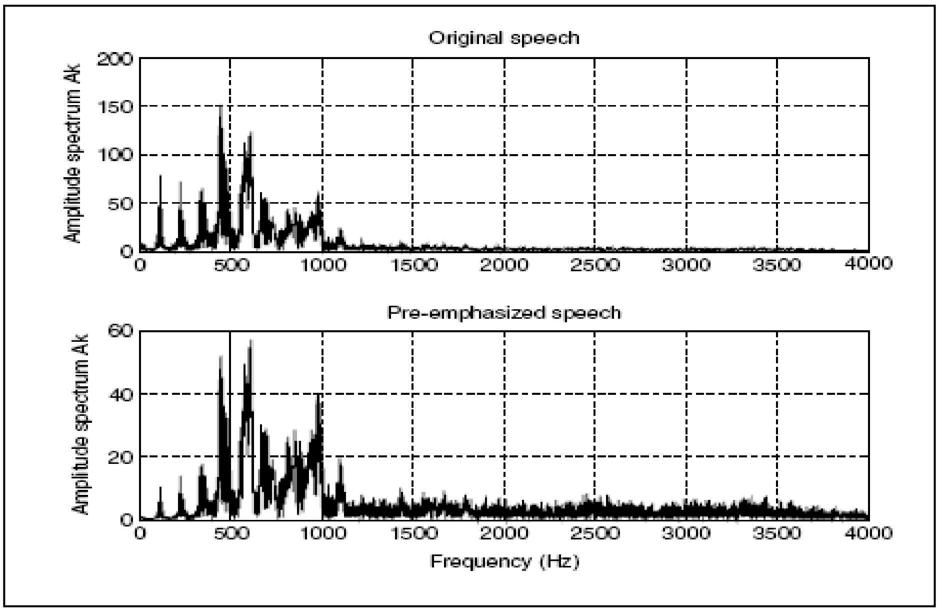


Fig. (11.2) Amplitude spectral plots for the original speech and pre-emphasized speech.

11.1.2 Bandpass Filtering of Speech

Bandpass filtering plays an important role in DSP applications. It can be used to pass the signals according to the specified frequency passband and reject the frequency other than the passband specification. Then the filtered signal can be further used for the signal feature extraction. Filtering can also be applied to perform applications such as noise reduction, frequency boosting, digital audio equalizing, and digital crossover, among others.

consider the following digital fourth-order bandpass Butterworth filter with a lower cutoff frequency of 1,000 Hz, an upper cutoff frequency of 1,400 Hz (that is, the bandwidth is 400 Hz), and a sampling rate of 8,000 Hz:

$$H(z) = \frac{0.0201 - 0.0402z^{-2} + 0.0201z^{-4}}{1 - 2.1192z^{-1} + 2.6952z^{-2} - 1.6924z^{-3} + 0.6414z^{-4}}. \tag{11.3}$$

Then

$$\begin{aligned} y(n) = {} & 0.0201x(n) - 0.0402x(n-2) + 0.0201x(n-4) \\ & + 2.1192y(n-1) - 2.6952y(n-2) + 1.6924y(n-3) - 0.6414y(n-4). \end{aligned} \tag{11.4}$$

The filter frequency responses are computed and plotted in Fig. (11.3 a) using MATLAB. Figure (11.3b) shows the original speech and filtered speech, while Fig.(11.3c) displays the spectral plots for the original speech and filtered speech. As shown in Fig.(11.3c) the designed bandpass filter significantly reduces low-frequency components, which are less than 1,000 Hz, and high-frequency components, above 1,400 Hz, while letting the signals with the frequencies ranging from 1000 to 1400 Hz pass through the filter.

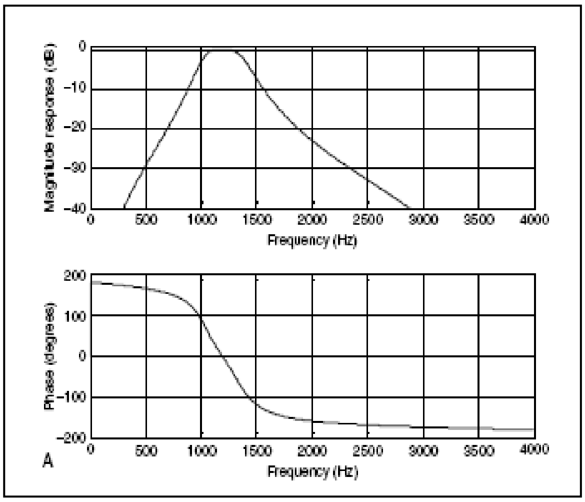


Fig.(11.3 a) Frequency responses of the designed bandpass filter.

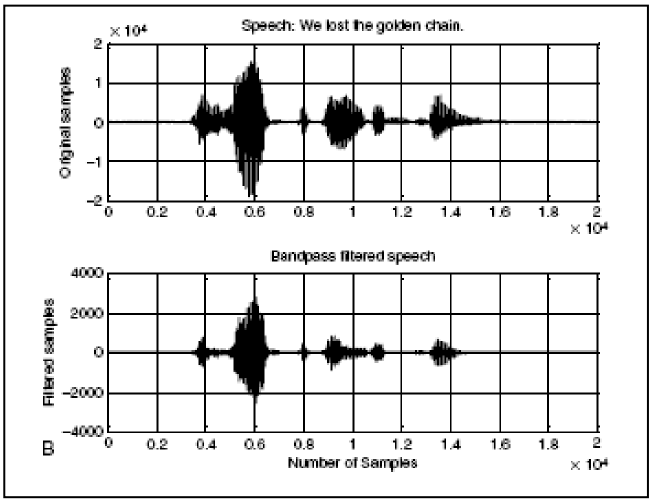


Fig.(11.3 b) Plots of the original speech and filtered speech.

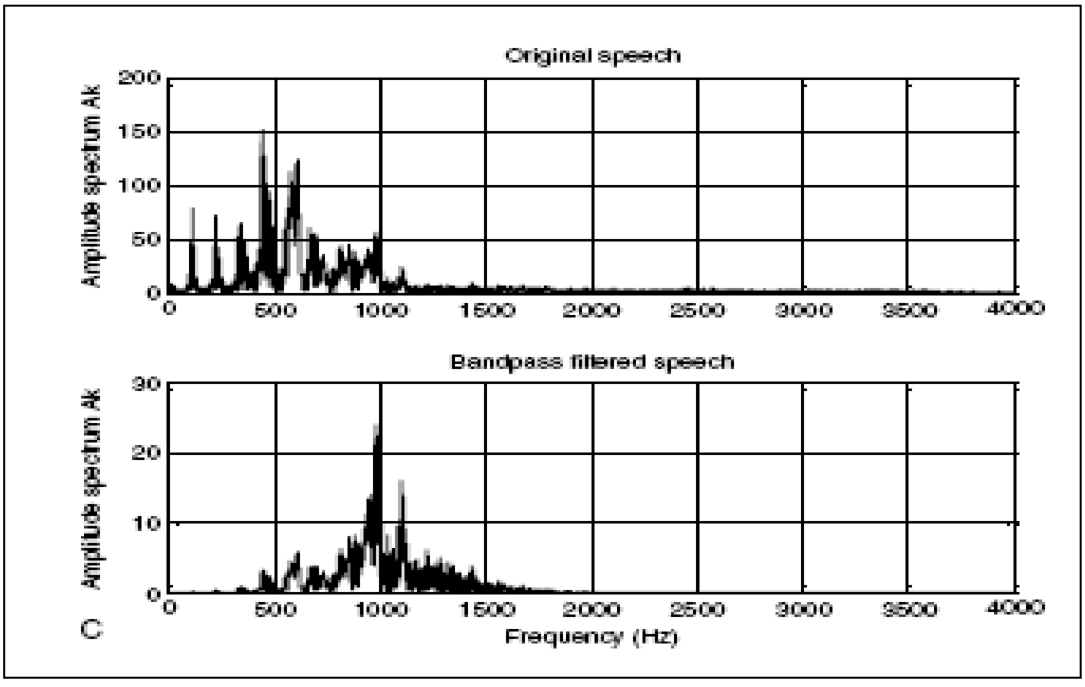


Fig.(11.3c) Amplitude spectra of the original speech and bandpass filtered speech.

MATLAB program for bandpass filtering of speech.

```
fs=8000;            % Sampling rate
freqz[(0.0201 0.00 -0.0402 0 0.0201],[1 -2.1192 2.6952 -1.6924 0.6414],512,fs);
axis([0 fs/2 -40 1]);% Frequency responses of the bandpass filter
load speech.dat
y=filter([0.0201 0.00 -0.0402 0.0201],[1 -2.1192 2.6952 -1.6924 0.6414], speech);
subplot(2,1,1),plot(speech);grid;          % Filtering speech
ylabel('Origibal Samples')

title('Speech: We lost the golden chain.')
subplot(2,1,2),plot(y);grid
xlabel('Number of Samples');ylabel('Filtered Samples')
title('Bandpass filtered speech.')
figure
N=length(speech);
Axk=abs(fft(speech.*hamming(N)'))/N;        % One-sided spectrum of speech
Ayk=abs(fft(y.*hamming(N)'))/N;    % One-sided spectrum of filtered speech
f=[0:N/2]*fs/N;
Axk(2:N) = 2*Axk(2:N);Ayk(2:N) = 2*Ayk(2:N);           % One-sided spectra
subplot(2,1,1),plot(f,Axk(1:N/2+1));grid
ylabel('Amplitude spectrum Ak')
title('Original speech');
subplot(2,1,2),plot(f,Ayk(1:N/2+1),'w');grid
ylabel('Amplitude spectrum Ak');xlabel('Frequency (Hz)');
title('Bandpass filtered speech');
```

H.W Write MATLAB program for pre-emphasis of speech.

11.2 Applications: Noise Reduction and Two-Band Digital Crossover

In this section, we will investigate noise reduction and digital crossover design using the FIR filters.

11.2.1 Noise Reduction

One of the key digital signal processing (DSP) applications is noise reduction. In this application, a digital FIR filter removes noise in the signal that is contaminated by noise existing in the broad frequency range. For example, such noise often appears during the data acquisition process. In real-world applications, the desired signal usually occupies a certain frequency range. We can design a digital filter to remove frequency components other than the desired frequency range.

In a data acquisition system, we record a 500 Hz sine wave at a sampling rate of 8,000 Hz. The signal is corrupted by broadband noise v(n):

$$x(n) = 1.4141 \cdot \sin(2\pi \cdot 500n/8000) + v(n).$$

The 500 Hz signal with noise and its spectrum are plotted in Fig.(11.4), from which it is obvious that the digital sine wave contains noise. The spectrum is also displayed to give better understanding of the noise frequency level. We can see that noise is broadband, existing from 0 Hz to the folding frequency of 4,000 Hz. Assuming that the desired signal has a frequency range of only 0 to 800 Hz, we can filter noise from 800 Hz and beyond. A lowpass filter would complete such a task. Then we develop the filter specifications:

Passband frequency range: 0 Hz to 800 Hz with passband ripple less than 0.02 dB.

Stopband frequency range: 1 kHz to 4 kHz with 50 dB attenuation.

The lowpass filtering will remove the noise ranging from 1,000 to 4,000 Hz, and hence the signal-to-noise power ratio will be improved.

Based on the specifications, we design the FIR filter with a Hamming window, a cutoff frequency of 900 Hz, and an estimated filter length of 133 taps. The enhanced signal is depicted in Fig.(11.5), where the clean signal can be observed. The amplitude spectrum for the enhanced signal is also plotted. As shown in the spectral plot, the noise level is almost neglected between 1 and 4 kHz. Notice that since we use the higher-order FIR filter, the signal experiences a linear phase delay of 66 samples, as is expected.

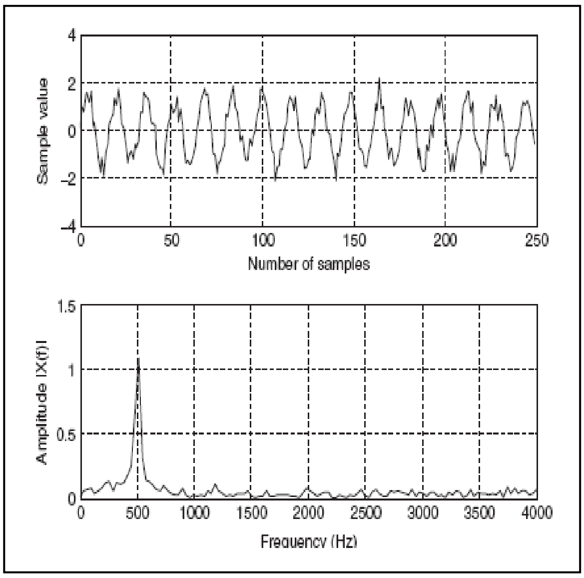


Fig.(11.4) Signal with noise and its spectrum.

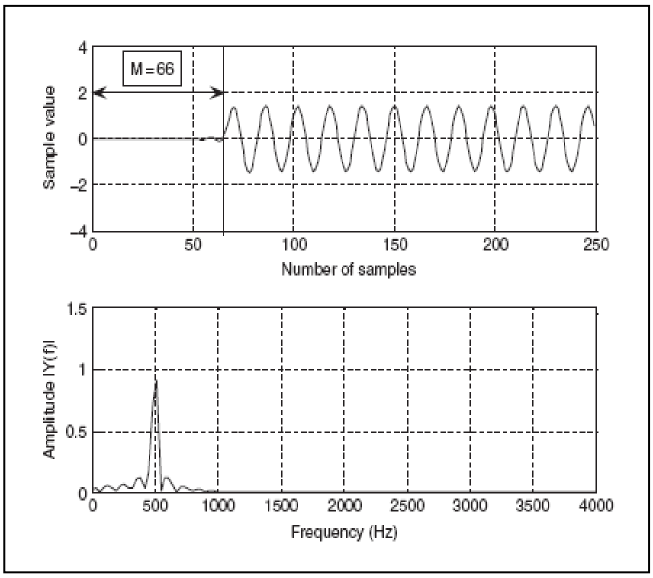


Fig.(11.5) The noise-removed clean signal and spectrum.

MATLAB program for the application of noise filtering.

```
close all; clear all
fs = 8000;           % Sampling rate
T = 1/fs;            % Sampling period
v=sqrt(0.1)*randn(1,250);          % Generate the Gaussian random noise
n = 0:1:249;            % Indexes
x = sqrt(2)* sin (2*pi*500*n*T) + v; % Generate the 500-Hz sinusoid plus noise
subplot(2,1,1);plot(t,x);
xlabel('Number of samples');ylabel('Sample value');grid;
N=length(x);
f=[0:N/2]*fs/N;
Axk= 2*abs(fft(x))/N;Axk(1)=Axk(1)/2; % Calculate the single-sided spectrum
subplot(2,1,2);plot(f,Axk(1:N/2+1));
xlabel('Frequency (Hz)');ylabel('Amplitude (f)|');grid;
figure
Wnc= 2*pi*900/fs;      % Determine the normalized digital cutoff frequency
B=firwd(133,1,Wnc,0,4);          % Design the FIR filter
y=filter(B,1,x);          % Perform digital filtering
Ayk= 2*abs(fft(y))/N;Ayk(1)=Ayk(1)/2;% Single-sided spectrum of the filtered data
subplot(2,1,1);plot(t,y);
xlabel('Number of samples');ylabel('Sample value');grid;
subplot(2,1,2);plot(f,Ayk(1:N/2+1));axis([0 fs/2 0 1.5]);
xlabel('Frequency (Hz)');ylabel('Amplitude |Y(f)|');grid;
```

11.2.2 Speech Noise Reduction

In a speech recording system, we digitally record speech in a noisy environment at a sampling rate of 8,000 Hz. Assuming that the recorded speech contains information within 1,800 Hz, we can design a lowpass filter to remove the noise between 1,800 Hz and the Nyquist limit (the folding frequency of 4,000 Hz). Therefore, we have the following filter specifications:

Filter type = lowpass FIR

Passband frequency range= 0–1,800 Hz

Passband ripple = 0.02 dB

Stopband frequency range =2,000–4,000 Hz

Stopband attenuation = 50 dB.

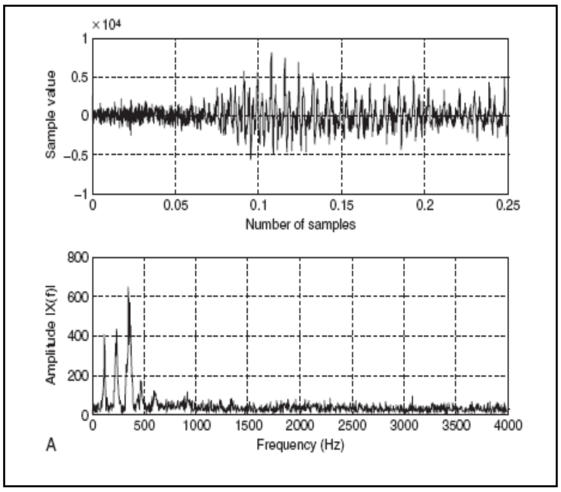
According to these specifications, we can determine the following parameters for filter design:

Window type = Hamming window

Number of filter taps = 133

Lowpass cutoff frequency = 1,900 Hz.

Figure (11.6a) shows the plots of the recorded noisy speech and its spectrum. As we can see in the noisy spectrum, the noise level is high and broadband. After applying the designed lowpass filter, we plot the filtered speech and its spectrum shown in Fig.(11.6b), where the clean speech is clearly identified, while the spectrum shows that the noise components above 2 kHz have been completely removed.



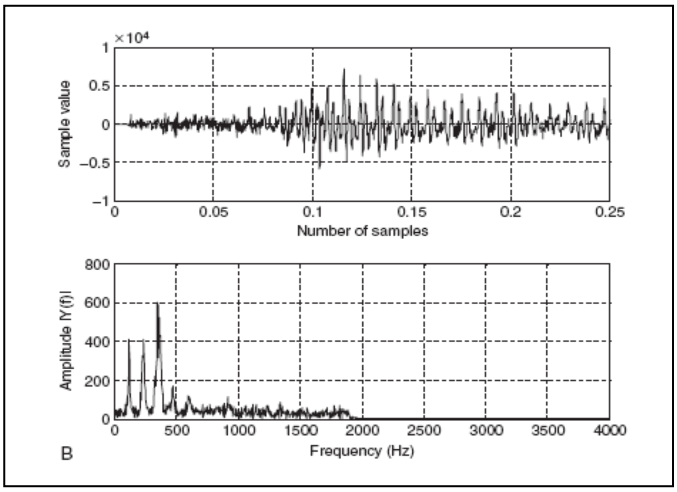


Fig.(11.6a) Noisy speech and its spectrum.

Fig.(11.6b) Enhanced speech and its spectrum.

<u>11.2.3 Two-Band Digital Crossover</u>

In audio systems, there is often a situation where the application requires the entire audible range of frequencies, but this is beyond the capability of any single speaker driver. So, we combine several drivers, such as the speaker cones and horns, each covering different frequency range, to reproduce the full audio frequency range.

A typical two-band digital crossover can be designed as shown in Fig.(11.7). There are two speaker drivers. The woofer responds to low frequencies, and the tweeter responds to high frequencies. The incoming digital audio signal is split into two bands by using a lowpass filter and a highpass filter in parallel. We then amplify the separated audio signals and send them to their respective corresponding speaker drivers. Hence, the objective is to design the lowpass filter and the highpass filter so that their combined frequency response is flat, while keeping transition as sharp as possible to prevent audio signal distortion in the transition frequency range. Although traditional crossover systems are designed using active circuits (analog systems) or passive circuits, the digital crossover system provides a cost-effective solution with programmable ability, flexibility, and high quality.

A crossover system has the following specifications:

Sampling rate = 44,100 Hz

Crossover frequency = 1,000 Hz (cutoff frequency)

Transition band = 600 to 1,400 Hz

Lowpass filter = passband frequency range from 0 to 600 Hz with a ripple of 0.02 dB and stopband edge at 1,400 Hz with attenuation of 50 dB.

Highpass filter = passband frequency range from 1.4 to 44.1 kHz with ripple of 0.02 dB and stopband edge at 600 Hz with attenuation of 50 dB.

In the design of this crossover system, one possibility is to use an FIR filter, since it provides a linear phase for the audio system. However, an infinite impulse response (IIR) filter can be an alternative. Based on the transition band of 800 Hz and the passband ripple and stopband attenuation requirements, the Hamming window is chosen for both lowpass and highpass filters. We can determine the number of filter taps as 183, each with a cutoff frequency of 1,000 Hz.

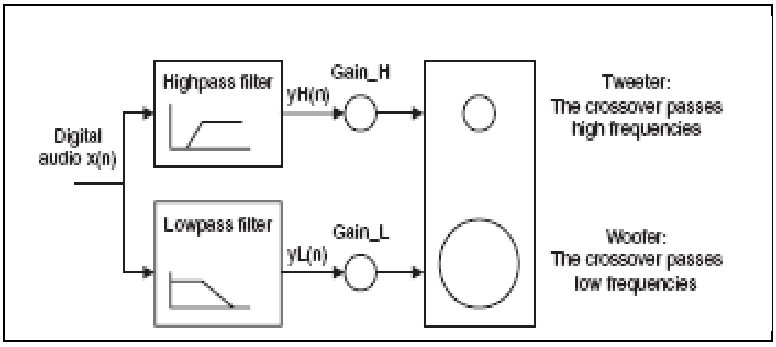
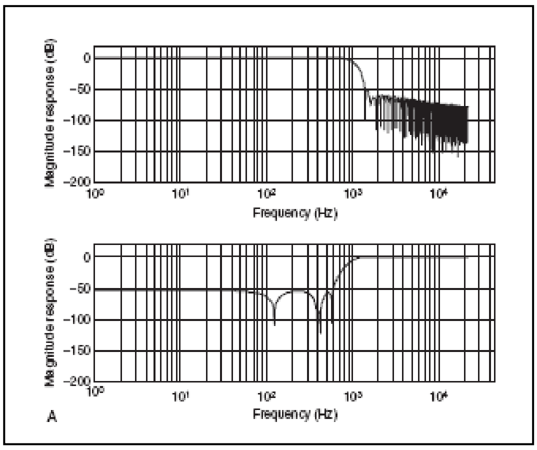


Fig. (11.7) Two-band digital crossover.



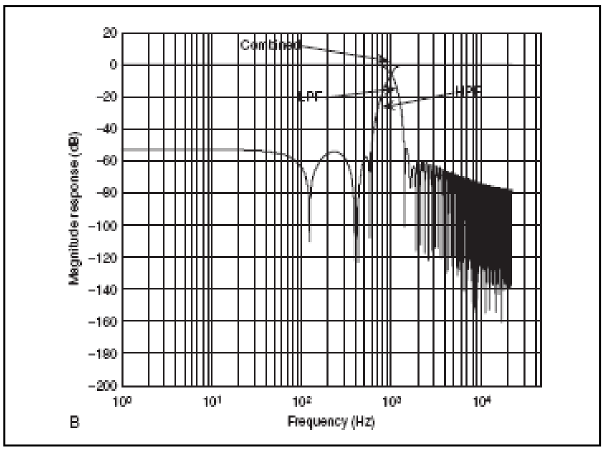


Fig.(11.8a) Magnitude frequency responses for lowpass filter and highpass filter.

Fig.(11.8b) Magnitude frequency responses for both lowpass filter and highpass filter, and the combined magnitude frequency response for the digital audio crossover system.

The frequency responses for the designed lowpass filter and highpass filter are given in Fig.(11.8a), and for the lowpass filter, highpass filter, and combined responses in Fig.(11.8b). As we can see, the crossover frequency for both filters is at 1,000 Hz, and the combined frequency response is perfectly flat. The impulse responses (filter coefficients) for lowpass and highpass filters are plotted in Fig.(11.8c).

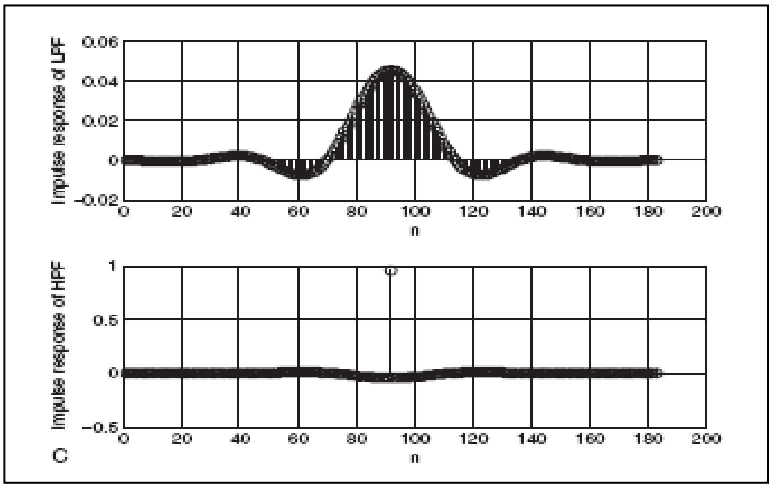


Fig. (11.8c) Impulse responses of both the FIR lowpass filter and the FIR highpass filter for the digital audio crossover system.