DSP APPLICATION ASSIGNMENT
DIGITAL FREQUENCY EQUALIZER

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1. Introduction

Digital frequency equalizer (DFE) is an important component of signal conditioning block in communication equipment. The frequency equalizers are essentially a set of band pass filters connected in parallel with a variable gain for each band. A digital filter accepts the discrete data as input performs the desired filtering operations on it and outputs the discrete data which can be converted back to analog form using DAC. The band pass filters in many analog equalizers are only of second order. In general, analog equalizers have the disadvantages of the dependent gain control between adjacent bands, due to the relative large overlap in frequency band and small stop band attenuation, and a large ripple in the pass band. Furthermore analog equalizers inevitably add phase distortion.

2. Implementation of 10-Band Digital frequency equalizer in MATLAB.

In this assignment we have implemented a 10 Band Digital Frequency Equalizer in MATLAB. The schematic of the DFE is shown in Fig.1. The input to the MATLAB program is given by using the command "wavread" which reads the music file and stores the samples in an array. Also we get information about the sampling frequency with which the music signal was sampled from this command. The samples are stored in an array 'x' and the sampling frequency is stored in the variable 'Fs'. Now to implement the frequency equalizer we have used Band Pass Linear Phase FIR filters. For this a direct command 'fir1'is used. It takes the normalized cutoff frequencies and the order of

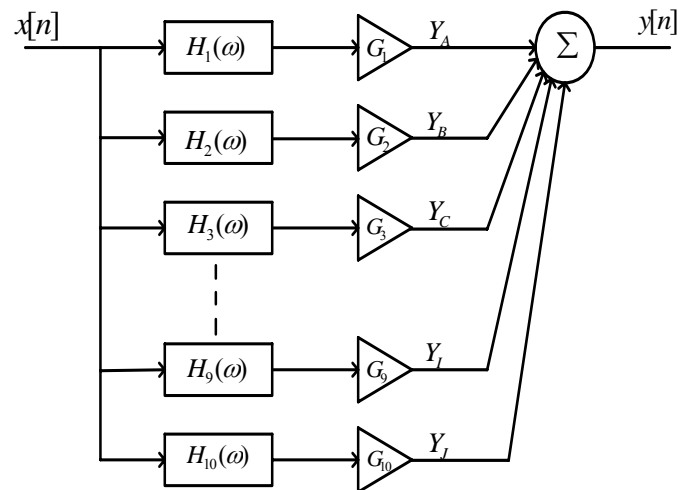the filter as the inputs and gives out the impulse response of the filter.

Fig1. Block diagram representing the parallely connected band pass filters forming a Digital Frequency Equalizer.

The 'fir1' command uses Hamming window for the design of the filter. This impulse response is convolved with the input sequence to give the filtered output. The sequence is then multiplied with the predefined gain, set for a particular band. Similar operation is done on other bands and the outputs of all bands are added and stored in the array 'Y_x'. The array Y_x contains output samples of the equalizer.

3. Design Procedure:

The functional unit of the DFE is a band pass filter (BPF), hence the steps to implement a band pass linear phase FIR filter is described in this section. Fig 2 shows the ideal frequency response of the BPF.

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A BPF with the following specification is described below, f_{c1} = 8 Khz, f_{c2} =16 Khz, order, n =50, f_s = 44100 Khz.
The DTFT of the ideal band pass filter is given by

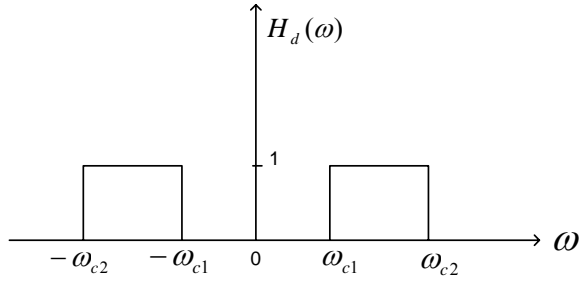$$H_d(\omega) = \sum_{n=0}^{\infty} h_d(n)e^{-j\omega n} \qquad (1)$$

Fig.2 Normalized frequency response of an ideal band pass filter.

By taking inverse DTFT of the frequency response shown in Fig.1 we get,
For $n \neq 0$,

$$h_d(n) = \frac{1}{2\pi} \int_{-\omega_{c2}}^{-\omega_{c1}} 1.e^{j\omega n}d\omega + \int_{\omega_{c1}}^{\omega_{c2}} 1.e^{j\omega n}d\omega \qquad (2)$$

For $n = 0$,

$$h_d(0) = \frac{2(f_{c2} - f_{c1})}{f_s} \qquad (3)$$

Where $\omega_{c1} = \frac{2\pi f_{c1}}{f_s}$, $\omega_{c2} = \frac{2\pi f_{c2}}{f_s}$
From (2) we get,

$$h_d(n) = \frac{2f_{c2}}{f_s}\frac{\sin(n\omega_{c2})}{n\omega_{c2}} - \frac{2f_{c2}}{f_s}\frac{\sin(n\omega_{c1})}{n\omega_{c1}} \qquad (4)$$

As per our specifications for this band, the normalized frequency $\omega_{c1} = 1.1398$, $\omega_{c2} = 1.1398$, order n = 50, length of the filter M = n+1 =51. The value of h_d(n) is calculated for $-25 \leq n \leq 25$ from (3) and (4). The Hamming window function is given by

$$W_{ham} = 0.54 + 0.46\cos\left(\frac{2\pi n}{M-1}\right),$$

$$\text{where, } -\left(\frac{M-1}{2}\right) \leq n \leq \left(\frac{M-1}{2}\right) \qquad (5)$$

The value of W_{ham} is calculated for $-25 \leq n \leq 25$ and the two sequences $h_d(n)$ and $W_{ham}(n)$ are multiplied to get the Finite Impulse Response of the filter.

$$h[n] = h_d(n) \times W_{ham}(n) \qquad (6)$$

To make the system causal the impulse response is right shifted by (M-1) / 2. The sampled input data is convolved with this impulse response to get the desired filtered output of this BPF. Similarly for other BPFs with different cutoff frequencies the filtered output is obtained. To get the frequency equalizer output, the respective gains of the bands are multiplied with the BPF output and added.

Implementation algorithm

1. Read Wave File, getting sampled data (x) and sampling frequency (f_s) using waveread command of MATLAB
2. Use order of filter, n & cut off frequency values in FIR1 function to determine the impulse response of the individual filters.
3. Convolve the output of all filters with x, the sampled data.
4. Multiply the convolved output with their corresponding gain values.
5. Add the output of the filters and store in Yx.
6. Playback using wavewrite command (supply parameter y and Fs).

Advantages of digital filters

The following list gives some of the main advantages of digital over analog filters.
1. A digital filter is programmable, i.e. its operation is determined by a program stored in the processor's memory. This means the digital

filter can easily be changed without affecting the circuitry (hardware). An analog filter can only be changed by redesigning the filter circuit. Digital filters are easily designed, tested and implemented on a general-purpose computer or workstation.

2. The characteristics of analog filter circuits (particularly those containing active components) are subject to drift and are dependent on temperature. Digital filters do not suffer from these problems, and so are extremely stable with respect both to time and temperature.

3. Unlike their analog counterparts, digital filters can handle low frequency signals accurately. As the speed of DSP technology continues to increase, digital filters are being applied to high frequency signals in the RF (radio frequency) domain, which in the past was the exclusive limited to discrete analog devices.

4. Digital filters are more versatile in their ability to process signals in a variety of ways this includes the ability of some types of digital filter to adapt to changes in the characteristics of the signal.

5. Fast DSP processors can handle complex combinations of filters in parallel or cascade (series), making the hardware requirements relatively simple and compact in comparison with the equivalent analog circuitry.

Advantages of FIR Filters

Finite impulse response (FIR) filters are the most popular type of filters implemented in software. The numeric errors that occur when implementing FIR filters in computer arithmetic occur separately with each calculation. In contrast, the feedback aspect of IIR filters can cause numeric errors to compound with each calculation, as numeric errors are fed back.

The practical impact of this is that FIRs can generally implemented using fewer bits of precision than IIRs. For example, FIRs can usually be implemented with 16 bits, but IIRs generally require 32 bits, or even more.

Conclusion

1. Implemented a 10-Band Frequency Equalizer in MATLAB.
2. Band pass linear phase FIR filters of order 50 are used for realization.
3. The Band cutoff frequencies are tabulated in Table 1.
4. Tested the program for various audio files with different gains successfully.
5. Observed the output of the audio for various frequency bands.

Doing this application assignment was a really a lot of fun as we could get our hand on the practical applications of digital signal processing also it was a very good learning experience as a group.

References

1. Digital Signal Processing *principles, algorithms and applications,* John G.Proakis, Dimitris G.Manolakis, D.Sharma.
2. http://www.intel.com/design/mcs96/technote /2318.htm.
3. http://www.circuitsage.com/filter.html
4. http://ccrmawww.stanford.edu/~jos/filters/M athematical_Sine_Wave_Analysis.html.

Appendix

S.No	Band Numbers	Range (in Hz)
1.	Band – 1	1-31
2.	Band – 2	32- 62
3.	Band – 3	63- 125
4.	Band – 4	126-250
5.	Band – 5	251-500
6.	Band – 6	501-1000
7.	Band – 7	1001 – 2000
8.	Band – 8	2001 – 4000
9.	Band – 9	4001 – 8000
10.	Band – 10	8001 – 16000

Table.1 Cutoff frequencies for the 10-bands

Magnitude and Phase response of 2nd Band with wc1 = 32 hz and wc2 = 62 hz
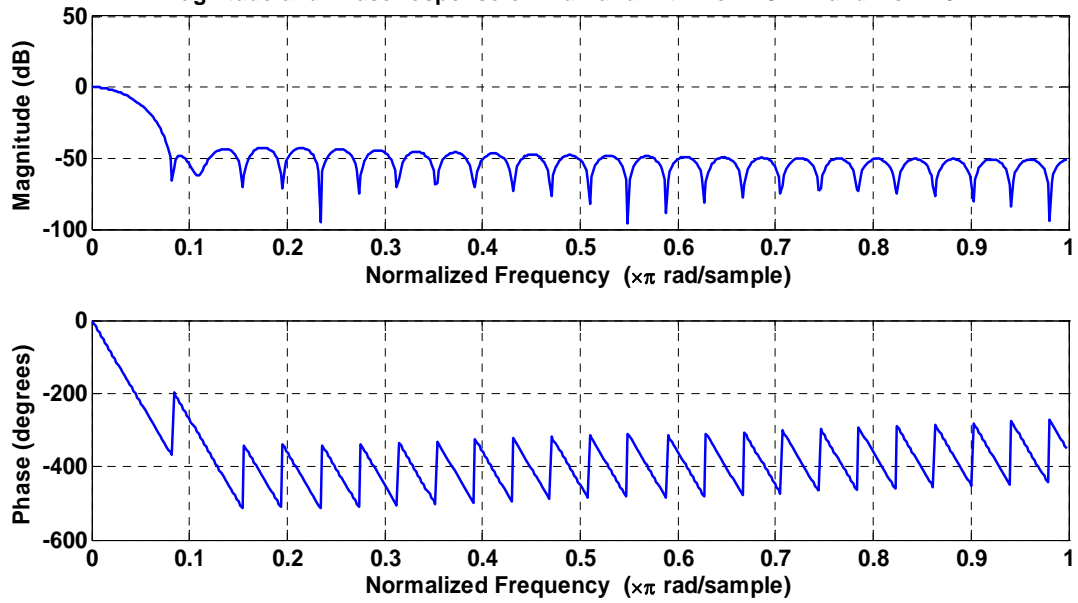
Magnitude and Phase response of 10th Band with wc1 = 8001 Hz and wc2 = 16 Khz