

Design IIR Notch Filter and Comb Filter to Eliminate Unwanted Frequencies for Environment

Dr. Kuldeep Kaur¹ and Mandeep Singh Saini²

¹Ph. D (Asst. Prof.), Faculty Member of B.A.M. Khalsa College, Garhshankar. (Punjab)

²M.Tech. (E.C.E), Guru Nanak Dev Engineering College, Ludhiana .P.T.U Regional Centre, (Punjab) India.

ABSTRACT—The Digital notch filter and Digital comb filter most useful in eliminating annoying frequency components. In this paper the cost analysis and resource analysis of both the filters has been done with different interpolation factor. The low pass filter is used to recover original signal from its samples. This is also known as interpolation filter. The result shows that Comb filter provides an 85% reduction in multipliers as compared to Notch filter.

Keywords:-Annoyance, Interpolators, Notch, Comb, Environment.

[A] INTRODUCTION

1. Need of unwanted frequency removed

Unwanted frequency generates a noise in environment system. Noise is annoying. Generally, louder the noise the greater the annoyance.

Factors affecting Annoyance

- Primary Acoustic: Sound level, frequency and duration
- Secondary Acoustic: Spectral complexity, fluctuations in frequency, level, localisation of noise source
- Non-acoustic: Adaptation and past experience, listener's activity interference, predictability of noise, individual personality

The term filter is commonly used to describe a device that discriminates, according to some distribute of the object applied at the input, what passes through it, e.g., an air filter that allows air to pass through it but prevents dust particular that are present in the air from passing through. An oil filter performs a similar function, with the exception that oil is a substance allowed to pass through the filter,

while particles of dirt are collected at the input of filter and prevented from passing through it. In photography, as ultraviolet filter is often used to prevent ultraviolet light, which is present in sun light and which is not a part of visible light, from passing through and affecting the chemical on the film.

In digital audio, the different sampling rates used are 32 KHz for broadcasting, 44.1 KHz for compact disc and 48 KHz for audio tape. In digital video, the sampling rates for composite video signal are 14.3 MHz and 17.7 MHz for NTSC and PAL respectively. But the sampling rates for digital component of video signals are 13.5 MHz and 6.75 MHz for luminance and colour difference signal. Different sampling rates can be obtained using as up sampler and down sampler using Filter.

So the filter is used in digital signal processing in a variety of ways, such as removal of undesirable noise from desired signal, spectral shaping such as equalization of communication channels, signal detection in radar, sonar and communication and for performing spectral of signal.

2. Digital notch filter

The direct complement of a band pass filter is called a "band stop filter". A notch filter is essentially a very narrow band stop filter. Figure 2.1 shown the frequency response characteristics of a notching filter with null at frequency ω_0 and ω_1 . Notching filter are used many application where specific frequency component must be eliminate. For example instruments and recording system required that the power line frequency of 50 Hz and its harmonic be eliminated (Proakis et al., 2006).

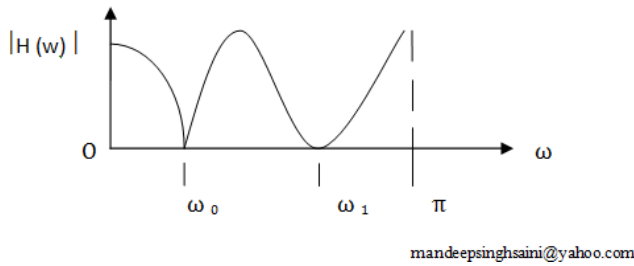


Fig.2.1 Frequency response characteristics of a notch filter

To create in the null in the frequency at a frequency ω_0 , we simply introduce a pair of complex-conjugate zero on the unit circle at an angle ω_0 . That is

$$z_{1,2} = e^{\pm j\omega_0}$$

Thus the system function for an FIR notch filter is simply

$$H(z) = b_0(1 - e^{j\omega_0} z^{-1})(1 - e^{-j\omega_0} z^{-1}) \quad 1.1$$

$$H(z) = b_0 \left(1 - \frac{2(e^{j\omega_0} + e^{-j\omega_0})}{2} z^{-1} + z^{-2} \right) \quad 1.2$$

$$H(z) = b_0(1 - 2 \cos \omega_0 z^{-1} + z^{-2}) \quad 1.3$$

The problem with FIR notch filter is that the notch has a relatively large bandwidth, which means that other frequency components around the desired null are severely attenuated.

To reduced the bandwidth using IIR Digital filter where

$$H(z) = b_0 \frac{(z - e^{j\omega_0})(z - e^{-j\omega_0})}{(z - re^{j\omega_0})(z - re^{-j\omega_0})} \quad 1.4$$

Suppose that we place a pair of complex-conjugate pole at

$$p_{1,2} = re^{\pm j\omega_0}$$

The effect of the poles is to introduce a resonance in the vicinity of the null and thus to reduce the bandwidth of the notch, the system function for the resulting filter is

$$H(z) = b_0 \frac{(1 - 2 \cos \omega_0 z^{-1} + z^{-2})}{(1 - 2 r \cos \omega_0 z^{-1} + r^2 z^{-2})} \quad 1.5$$

Frequency response of the IIR filter effect of poles is to reduce the bandwidth of the notch.

3. IIR notch filter Simulation

Notching filter to remove interference at 50 Hz and its harmonics. Therefore IIR digital notch filter, AC power supply 50 Hz power interference eliminated does not affect the effective audio signal transmission.

In this paper IIR notching filter eliminate $F_0 = 50$ Hz frequency centre frequency with quality factor $Q=100$. Where filter order or filter coefficient is $N = 2,4,6,8$. And $BW = F_0/Q$. Figure 3.1 shown Amplitude Frequency characteristics.

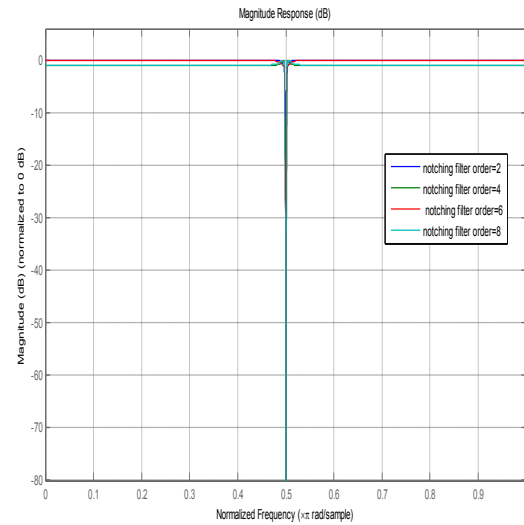


Fig. 3.1 Notching filter Amplitude Frequency characteristics

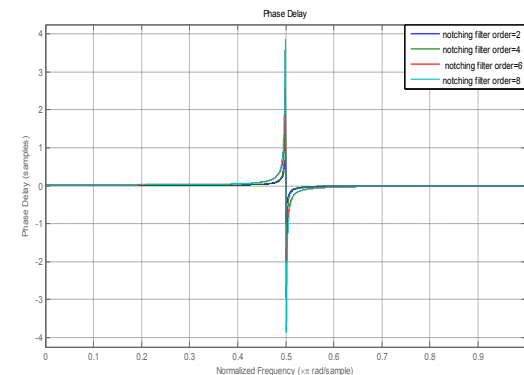


Fig. 3.2 Notching filter Phase delay characteristics

Table 3.1: cost analysis IIR notch filter

FILTER ORDER	MUL.	ADDER	STATES	MPIS	APIS
2	3	2	2	3	2
4	10	8	4	10	8
6	13	10	6	13	10
8	20	16	8	20	16

```

MATLAB R2011b
File Edit Debug Parallel Desktop Window Help
Shortcuts How to Add What's New
ans =
Number of Multipliers      : 3
Number of Adders          : 2
Number of States          : 2
Multiplications per Input Sample : 3
Additions per Input Sample : 2
ans =
2
ans =
Number of Multipliers      : 10
Number of Adders          : 8
Number of States          : 4
Multiplications per Input Sample : 10
Additions per Input Sample : 8
ans =
4

```

```

MATLAB R2011b
File Edit Debug Parallel Desktop Window Help
Shortcuts How to Add What's New
ans =
4
ans =
Number of Multipliers      : 13
Number of Adders          : 10
Number of States          : 6
Multiplications per Input Sample : 13
Additions per Input Sample : 10
ans =
6
ans =
Number of Multipliers      : 20
Number of Adders          : 16
Number of States          : 8
Multiplications per Input Sample : 20

```

4. Digital comb filter

"Comb Filters", as their name implies, look like a hair comb. They have many "teeth", which in essence are notches in the transfer function where information is removed. These notches are spaced evenly across the spectrum, so they are only useful for removing noise that appears at regular frequency intervals.

Comb filters are used in a variety of signal processing applications. These include:

- Cascaded Integrator Comb filters, commonly used for anti-aliasing during interpolation and decimation operation that change the sample rate of a discrete-time system.

- 2D and 3D comb filters implemented in hardware for PAL and NTSC television decoders. The filters work to reduce artifacts such as dot crawl.
- Audio effects, including echo, flanging, and digital waveguide synthesis. For instance, if the delay is set to a few milliseconds, a comb filter can be used to model the effect of acoustic standing waves in a cylindrical cavity or in a vibrating string.
- In astronomy the astro-comb promises to increase the precision of existing spectrographs by nearly a hundredfold.

In acoustics, comb filtering can arise in some unwanted ways. For instance, when two loudspeakers are playing the same signal at different distances from the listener, there is a comb filtering effect on the signal. In any enclosed space, listeners hear a mixture of direct sound and reflected sound. Because the reflected sound takes a longer path, it constitutes a delayed version of the direct sound and a comb filter is created where the two combine at the listener.

Comb filters exist in two different forms, Feed forward and Feedback, the names refer to the direction in which signals are delays before they are added to the input.

Comb filters may be implemented in discrete time or continuous time, this article will focus on discrete time implementations, the properties of continuous time comb are very similar.

Feed forward form

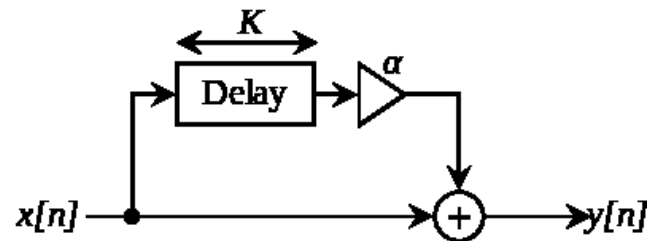


Fig. 4.1 Feed forward comb filter structure

$$y[n] = x[n] + \alpha x[n - k] \tag{4.1}$$

Where k is the delay length (measured in samples), and α is a scaling factor applied to the delayed signal.

$$Y(z) = (1 + \alpha z^{-k}) X(z) \tag{4.2}$$

We define the transfer function as:

$$H(z) = \frac{Y(z)}{X(z)} = 1 + \alpha z^{-k} = \frac{z^k + \alpha}{z^k} \tag{4.3}$$

For positive value of α , the first minimum occurs at half the delay period and repeat at even multiples of the delay frequency thereafter:

$$f = \frac{1}{2K}, \frac{3}{2K}, \frac{5}{2K}$$

Feedback word form

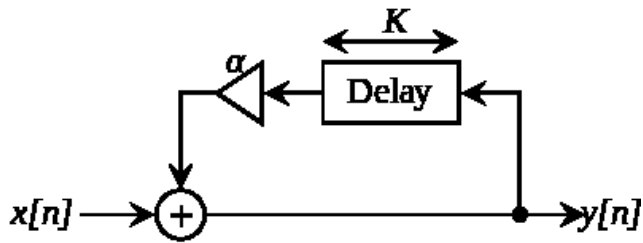


Fig.4.2 Feedback comb filter structure

$$y[n] = x[n] + \alpha y[n - k] \tag{4.4}$$

Take the Z transform, we obtain:

$$1 + \alpha z^{-k}) Y(z) = (X(z) \tag{4.5}$$

The transfer function is therefore

$$H(z) = \frac{Y(z)}{X(z)} = \frac{1}{1 + \alpha z^{-k}} = \frac{z^k}{z^k - \alpha} \tag{4.6}$$

For positive value of α , the first minimum occurs at 0 and repeat at even multiples of the delay frequency thereafter:

$$f = 0, \frac{1}{K}, \frac{2}{K}$$

In this paper IIR comb filter eliminate $F_0 = 50$ Hz frequency centre frequency with quality factor $Q=100$. Where filter order or filter coefficient is $N = 2,4,6,8$. And $BW = F_0/Q$. Figure 3.1 shown Amplitude Frequency characteristics.

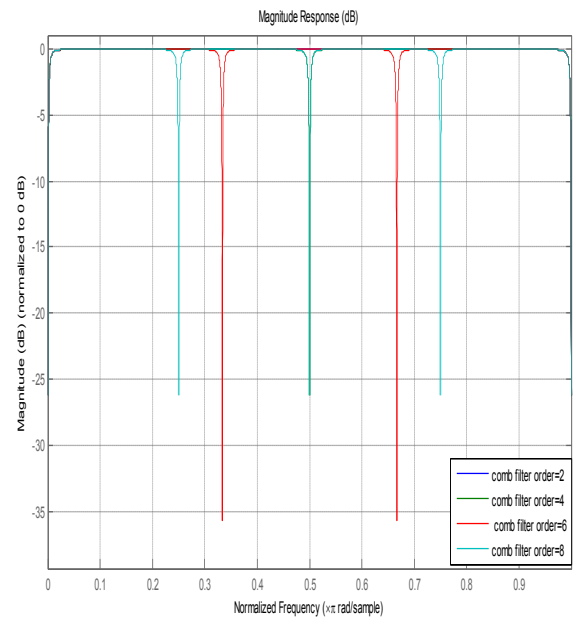


Fig. 4.3 Comb filter Amplitude Frequency characteristics

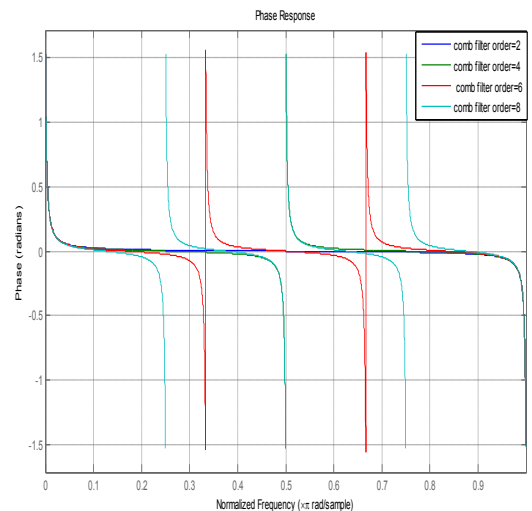


Fig.4.4 comb filter phase response

Table 4.1: cost analysis IIR Comb filter

FILTER ORDER	MUL.	ADDER	STATES	MPIS	APIS
2	3	2	2	3	2
4	3	2	4	3	2
6	3	2	6	3	2
8	3	2	8	3	2

```

MATLAB R2011b
File Edit Debug Parallel Desktop Window Help
Shortcuts How to Add What's New
ans =
Number of Multipliers      : 3
Number of Adders          : 2
Number of States           : 2
Multiplications per Input Sample : 3
Additions per Input Sample  : 2

ans =
2

ans =
Number of Multipliers      : 3
Number of Adders          : 2
Number of States           : 4
Multiplications per Input Sample : 3
Additions per Input Sample  : 2

ans =
4
    
```

```

MATLAB R2011b
File Edit Debug Parallel Desktop Window Help
Shortcuts How to Add What's New
Number of Multipliers      : 3
Number of Adders          : 2
Number of States           : 6
Multiplications per Input Sample : 3
Additions per Input Sample  : 2

ans =
6

ans =
Number of Multipliers      : 3
Number of Adders          : 2
Number of States           : 8
Multiplications per Input Sample : 3
Additions per Input Sample  : 2

ans =
8
    
```

5. Problems with Digital Notching filter and Digital comb filter

Digital Notching Filter	Digital Comb Filter
Eliminate the sinusoidal interference while leaving the broad-band signal unchanged. If the sinusoidal frequencies are known and fixed, then a fixed notch filter can be used. However, if these frequencies are unknown or time-varying,	Eliminate the noise from the on-going signal. In digital color TV systems, where an IIR frame comb can be used to eliminate transmission noise.

then adaptive notch filters are needed.	
The notch filters Remove the periodic noises in the image and Removing Power line or other interference in the ECG recording system but notching filter is not Distortion less.	The comb filters are used to separate the luminance (black & white) and chrominance (color) signals from the composite video signal, and also to reduce noise.
The larger the filter order, the longer the transient response. This causes problems when particularly short signals are filtered. Also the results shown Hardware requirement is Large.	The larger the filter order, the longer the transient response. This causes problems when particularly short signals are filtered. Also results shown Hardware requirement is Less.

[B] PROPOSED FILTER DESIGN

A Notching/Comb filter is used to shape and oversample a symbol stream before modulation/transmission as well as after modulation and demodulation. It is used to reduce the bandwidth of the oversampled symbol stream without introducing inter symbol interference.

In this proposed work Notching filter has been designed using filter order 2,4,6,8 shown in Table 3.1 and Comb filter has been designed using filter order 2,4,6,8 shown in Table 4.1

The cost analysis results shown multi per input sample is fixed in comb filter If filter order is increased or decreased. Also we find out if filter order or no. of coefficient is large in comb filter then Hardware requirement is less or constant.

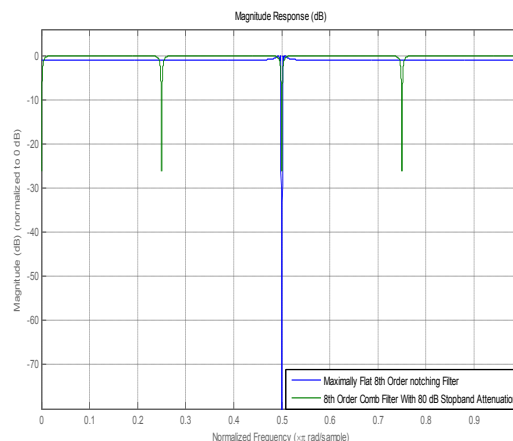


Fig. 5.1 all filter design performance

[C] RSEULTS & DISCUSSION

The performance and cost of all the all designs have been analyzed in Fig. 6.1.

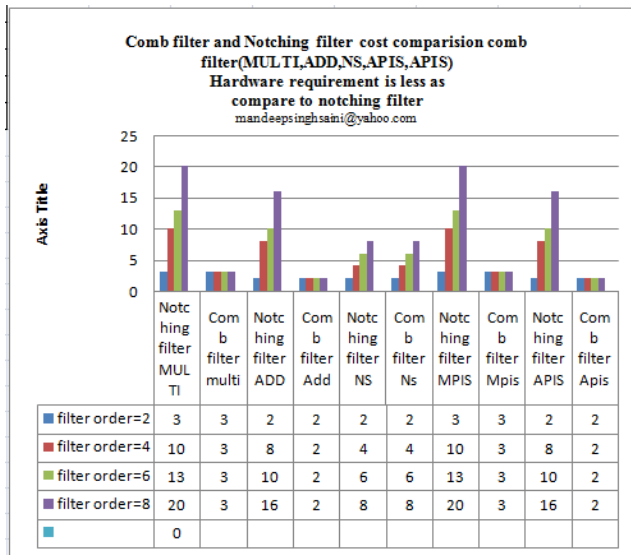


Fig. 6.1 comb filter is less hardware requirement (MULTI, ADD, NS, MPIS, APIS)

This method achieves computational costs with filter order 8 (more filter order more computer complexity and more hardware or component requirements) in Comb filter only 3 MPIS on average compared to 20 MPIS that of Notching Filter Design.

[D] CONCLUSION

Theory and practice prove, digital audio signal Processing system using IIR digital notch filter/Comb filter, AC power supply 50 Hz power interference eliminated does not affect the effective audio signal transmission, performance improvement, the signal-to-noise ratio can be improved greatly.

But the results showed that Comb filter 85% saving the hardware requirement as compare to notching filter. So proposed alternative designs comb filter may be used to provide cost effective solution for the different sampling rates.

REFERENCES

[1]. Chun Mei WANG, Wei Cai XIAO, "Second-order IIR Notch Filter Design and implementation of digital signal processing system," *Proc. of the 2nd Int. Symp. on Comput., Commun., Control and Automation ISCCA-2013*.
 [2]. http://dij.ee.ntu.edu.tw/Notch_Filter.pdf.
 [3]. <http://tripp.iitd.ernet.in/delhibrbs/metro/Metro/Noise%20effects%202.ppt>.

[4]. J. G. Proakis and D. G. Manolakis, "Digital Signal Processing Principle, Algorithms, and Applications", *Prentice Hall*, 1996.
 [5]. Mikel Gainza, Bob Lawlor and Eugene Coyle, "Harmonic Sound Source Separation using FIR Comb Filters," *117th AES (Audio Eng. Soc.) Conv. , Oct. 28-31, San Francisco, 2004*.
 [6]. Monika Singh, Saroj Dogra and Rajesh Mehra, "IIR filter design and analysis using notch and comb filter," *Int. J. of Scientific Research Eng. & Technology (IJSRET) vol. 2 Issue 6 pp. 358-361. Sep. 2013*.
 [7]. S. K. Mitra, "Digital Signal Processing A Computer-Base Approach", *McGraw-Hill*, 2001.
 [8]. www.mathwork.com



MANDEEP SINGH SAINI was born in Punjab, India on 26th January 1975. He completed his B. Tech in Electronics Engineering from Nagpur University (Maharashtra State) in 2001. He has also completed Master of Technology (M. Tech) from Guru Nanak Dev Eng. College Ludhiana (Punjab) in 2012. His current research and teaching interests are in Digital Signal Processing and Very Large Scale Integration Design. He has also worked as M. Tech Research guide with more than 22 Journal and Conferences.