Transmission of Audio Signal through Different Filters to reduce AWGN from Channel and Recode there Digital Response

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Abstract: **Mobile communication has become an important part of our daily lives for voice communication and data sharing and access over the Internet. Mobile communication is an open network, so maintaining the privacy and reliability of data has always been anxiety. The reliability of the data against channel noise can be achieved by various error correction codes. The purpose of the channel coding process is to reduce the effect of some disruptive elements that the information is influenced in the transmission phase as much as possible, which is to ensure that the data is delivered to the receiver with minimum error. For that purpose, we use filters, which can be an analog filter and digital filter. From the today scenario, Digital filters have many advantages, such as higher precision, higher stability. In this paper, the signal passes through various types of digital filters and finds their different response which describes the filter accuracy with respect to the input audio signal.**

MATLAB is the best software of numerical analysis, providing user-friendly coding commands. Through the MATLAB design part of filter done by some procedures and given the frequency characteristics curve, comparing the results.

Keywords: Digital Filters, AWGN channel, Channel coding process, audio signals.

I.   INTRODUCTION

The procedure of transfer information from transmitter end to receiver end, there are several processes took place such as Analog to digital converter (ADC), Filters, Digital to analog converter (DAC) and more. In that process, converter and filters depend on their inputs if it is analog then we go through the analog filters and DAC and if it is digital then use of digital filters and ADC needed. In this paper, communication is in the form of digital communication so filters should be digital filters. In the digital communication, there is a way to move information or transfer the signal along the channel, to get digital information first we converted an audio

information into the digital information with the help of sampling with a finite frequency fx and this is called as the sampling frequency. When, information transfer from the channel than some noise add-on in the original information by the external environment and to get again original information filter is used.
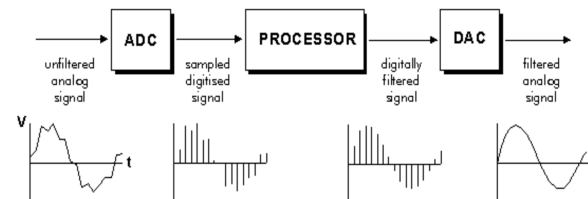

Fig. 1 Basic block diagram of Digital Filter

Filters play an important role in extracting meaningful data from the signal Filters can be classified into different groups based on the requirement. Finite impulse response (FIR) and Infinite impulse response (IIR) filters are used. Both types of filters have their own advantages and disadvantages, which play a vital role while designing a filter. FIR filters provide a linear phase, always stable and can be used for more complex circuits. But, the IIR filter provides non-linear phase characteristics, unstable and they are used for less complexity. [1]
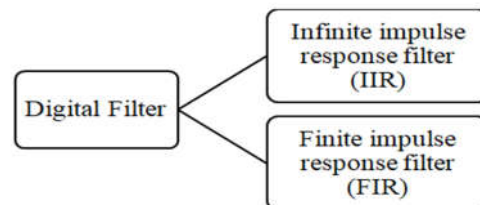

Fig.2. Types of Digital Filter

We know that the ideal filter cannot use of the practical purpose, therefore, the approximation design required. There are four types approximation design of filters are used such as Low pass filter

(LPF), High pass filter (HPF), Band-pass filter, Notch or Band-stop filter. All, the simulation can be done by using the MATLAB.

II. FILTERING CONCEPT

The analog filters are implemented by the analog circuit elements like resistors (R), inductors (L), Capacitors (C). And, the digital filters are implemented by difference equation.

The differential equation can be designed using software in form of algorithm so, we can easily change the filter parameter and characteristics according to the desired output.

III. ADVANTAGES OF USING DIGITAL FILTERS

➢ A digital filter is programmable.
➢ Digital filters are easily designed, tested and implemented.
➢ Digital filters are stable with respect both to time and temperature.
➢ Digital filters can handle both low and high frequency signals accurately.

IV. INFINITE IMPULSE RESPONSE FILTER (IIR)

The digital filters are classified into two types, first one depends on the number of sample points used to determine the unit sample response of linear time invariant (LTI) discrete-time system. And, second one depends on the infinite number of sample points used to determine the unit sample response of LTI System.

The unit sample of any response is also known as Impulse. So, the infinite numbers of samples are used to determine the impulse then that filter is known as Infinite- duration Impulse Response (IIR) digital filters.

On the other words IIR filters are commonly used to replace existing analog filters. IIR filter consists of present and past values which is described in the form of equations,

$$y(n) = \sum_{i=0}^{N} b_i x(n-i) - \sum_{i=0}^{M} a_i y(n-i)$$

…….(i)

$$H(z) = \frac{y(z)}{x(z)} = \frac{b_0 + b_1 z^{-1} + \cdots + b_N z^{-N}}{1 + a_1 z^{-1} + \cdots + a_M z^{-M}}$$

……(ii)

Where,
        y(n) is output and x(n) is input.
        b and a coefficients.

V. FINITE IMPULSE RESPONSE FILTER (FIR)

A Finite impulse response (FIR) filter can be define as a filter whose impulse response (or response to any finite length input) is of finite duration, because it settles to zero in finite time.

Finite Impulse Response (FIR) filter have some properties for which the designers prefer FIR filter over the IIR filter. The impulse response of FIR filter is of finite duration but it will be of infinite duration for IIR filter. [5]

There are some types of FIR filter just same as the IIR filter like:- Low pass filter, High pass filter, Band-pass filter, Band-stop filter, All-pass filter.

VI. DESIGN OF FILTERS

Ideal Filters are usually such that they admit a gain of 1 in a given pass-band (where signal is passed) and 0 in their stop-band (where signal is removed).
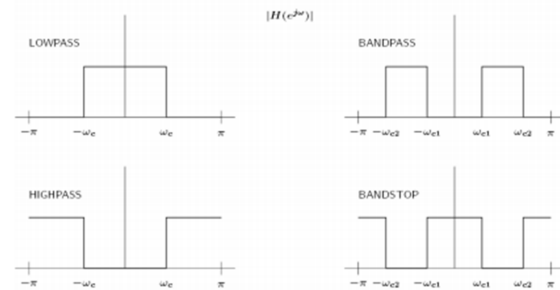

Fig.3. Magnitude Response of Ideal Filters

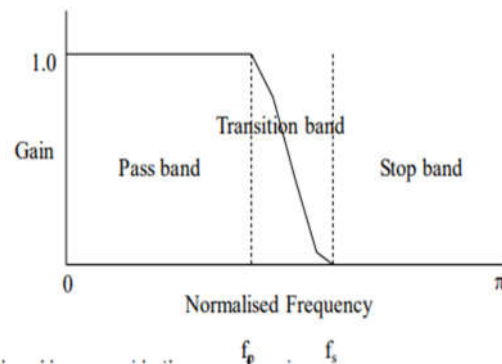The four classical standard frequency magnitude responses are:

A. Low pass Filter :-

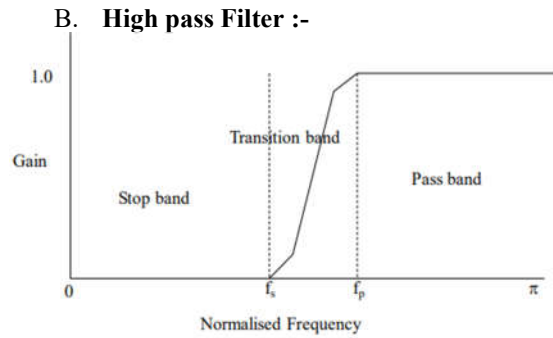

Fig.4. Magnitude Response of Practical LPF

B. **High pass Filter :-**

Fig.5. Magnitude Response of Practical HPF
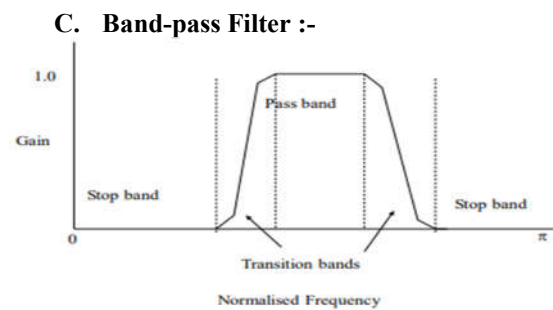
C. **Band-pass Filter :-**

Fig.6. Magnitude Response of Practical Band-pass Filter
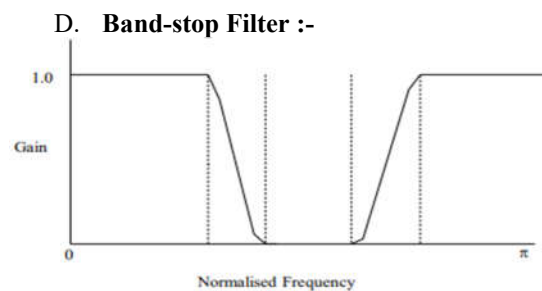
D. **Band-stop Filter :-**

Fig.7. Magnitude Response of Practical Band-stop Filter

VII.   SIMULATION RESULTS

The three second length audio signals were recorded in MATLAB environment with a fs = 8 kHz sampling frequency, and 8 bits resolution.
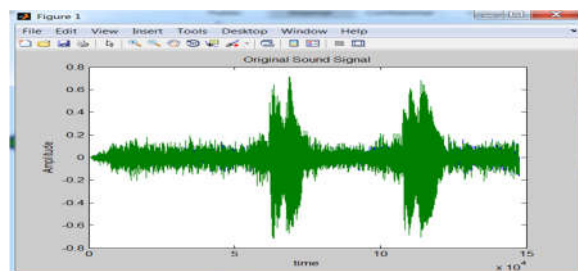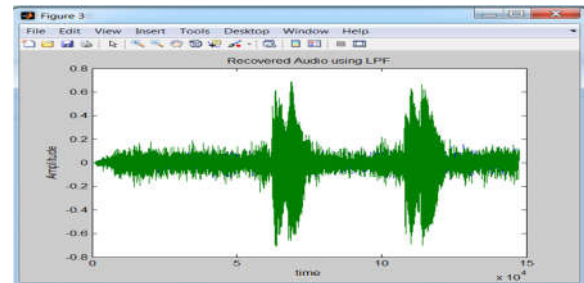

Fig.8. Original Sound Signal

Fig.9. Recovered Audio using LPF
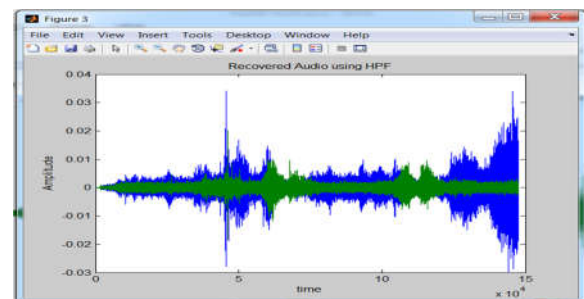

Fig.10. Magnitude response of LPF

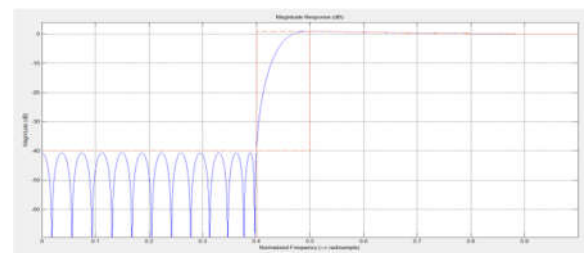Fig.11. Recovered Audio using HPF

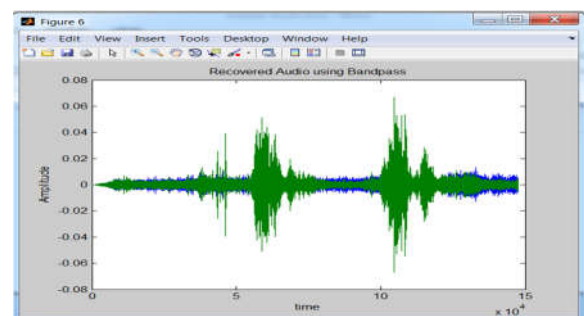Fig.12. Magnitude response of HPF

Fig.13. Recovered Audio using Band-pass Filter
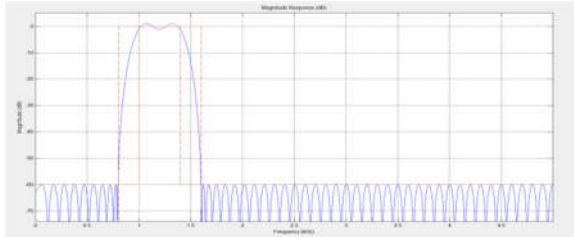
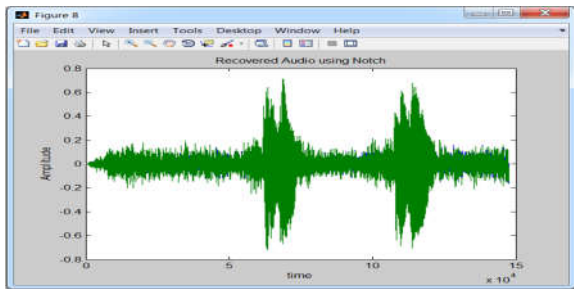Fig.14. Magnitude response of Band-pass Filter

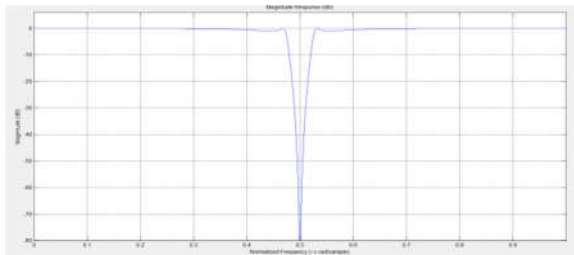Fig.15. Recovered Audio using Notch Filter

Fig.16. Magnitude response of Notch Filter

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VIII.   CONCLUSION

According to the obtained result, the notch filter removed a large amount of noise from the corrupted signal as compared to the others filters. If we Give the rank to the filter accordingly their result then first is notch filter after that low pass filter then band-pass filter. Design filter on MATLAB is used in digital signal processing, and has wide application and development prospect.

IX.   REFERENCES

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