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Filtration and Synthesis of Different types of Human Voice Signals: An application of digital signal processing

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Abstract

An observation of the effect in audio signal by using digital filter plays an important role in the field of digital signal processing (DSP). Day by day the digital form of signal is becoming more preferable than the analog one which is increasing the need of DSP in the rapidly changing world. Yet, there are many attractive schemes for designing a digital filter; we adopt windowing technique for design a FIR low pass filter in the frequency domain for the short period. However, the main task of our work is to perform filtration of the different types of practical human voice signals by using digital filter and synthesis of those signals to reduce the memory size (kB) by remaining the same quality of the signal. We used MATLAB for the design of digital filter and synthesis of those audio signals. MATLAB provides different options for signal synthesis. Finally, this paper gives an idea about reconstructed signals and filtrated signals.

Keywords: Digital filter, Cutoff frequency, Fourier transform, Inverse Fourier transforms, normalized frequency.

1. Introduction

An audio signal, such like human voice, is a representation of sound waves in a different form. Typically this is an electrical voltage, but these signals can be expressed through alternative mediums such as magnetic particles, when recorded onto analogue tape [1]. A fundamental aspect of signal processing is filtering. Filtering involves the manipulation of the spectrum of a signal by passing or blocking certain portions of the spectrum, depending on the frequency of those portions [1-2]. Filters are designed according to what kind of manipulation of the signal is required for a particular application. Filters are basically classified into two classes: analogue filter and digital filter. Digital filters are a very important part of DSP. In fact, their extraordinary performance is one of the key reasons that DSP has become so popular [3]. The problem of this paper can be attacked with either analogue or digital filters. Which is better? Analogue filters are cheap, fast, and have a large dynamic range in both amplitude and frequency [2]. On the other hand, digital filters, in comparison, are vastly superior in the level of performance that can be achieved. Digital filters are totally flexible, as they are designed by some set of programs. So to change the filters response we just have to make a little change in that program [2-3]. It is also easy to handle and store digital signals. So, digital filters are preferable over analogue filters. In our experiment, we take an analogue audio signal of a particular memory size and finally we show that the memory size of this audio signal has been reduced by remaining the same quality through digital signal processing system. In the starting process of this project, we record the human voice by a recorder. As human voice is analogue, it has been digitized. The digitization process has done automatically when it take in a computer. We designed a low pass FIR filter by MATLAB program. The input audio signal is then passed through the filter. The filtrated signal is synthesized. In the synthesized process, Fourier transform, low amplitude signal suppression, inverse Fourier transform are done respectively. The reconstructed signal and filtrated signal are achieved practically, which almost have the same quality but the reconstructed signal's memory size is almost half of the filtrated signal.

This paper is organized as follows. Section 2 represents the working procedure in details. Section 3 describes the complete system of our work with Simulation results. Summary of the results are shown in Section 4. Research trends and potential for future applications is discussed in Section 5 and followed by a conclusion in Section 6.

2. Working Scheme

A fundamental aspect of signal processing is filtering. Filtering involves the manipulation of the spectrum of a signal by passing or blocking certain portions of the spectrum, depending on the frequency of those portions. Generally, the work of this paper involves two parts which one is to observe the frequency response and the second one is to reconstruct of the original (human voice) signal. These two phenomena, finally, shows that the memory size of the reconstructed signal with respect to filtrated signal is almost half by remaining the same quality. A signal may contain noise. For this reason, filtering has done. For filtering approach, a low pass filter has been designed in the frequency domain by using FIR type window-sinc (Hamming window) technique [1-4]. Windowed-sinc filters are used to separate one band of frequencies from another [3]. They are very stable, produce few surprises and can be pushed to incredible performance levels. To design a windowed-sinc, two parameters must be selected which one is the cutoff frequency, f_c and other is the length of the filter kernel, M [1,3]. The cutoff frequency (f_c) is expressed as a fraction of the sampling rate and the length of the filter kernel (M) which determines the transition bandwidth (BW) of the filter. This is only an approximation since roll-off depends on the particular window being used in [2]. Down sampling has done to reduce the lower amplitude. The down sampling factor is usually an integer or a rational fraction greater than unity. In this paper, we carried down sampling by a factor two. After that an audio signal synthesis is done by taking FFT (Fast Fourier Transform) and IFFT (Inverse Fast Fourier Transform) [5]. FFTs are of great importance in [5] to a wide variety of applications, from digital signal processing and solving partial differential equations to algorithms for quick multiplication of large integers. The following flow chart shows the working scheme of our work of this paper which is shown in Fig. 1.

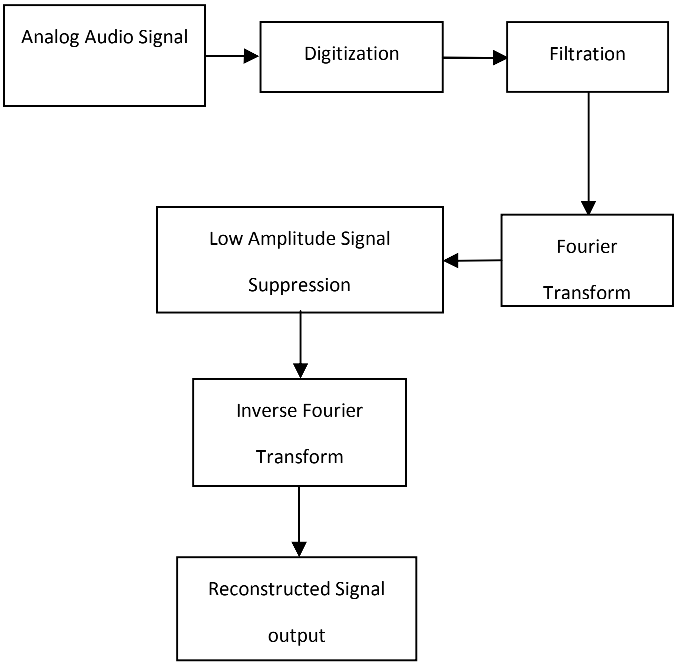


Fig. 1. Flow chart for showing the complete process.

3.1 System Description (Analog Audio signal)

An analogue human audio signal is practically recorded by a recorder in wav format. The human audio frequency range mostly for men is (2-3) kHz and for women (3-4) kHz. However, the audio signal was recorded and collected data after certain period of time with differential interval. From such collected data, the graphical representation of one of the audio signal is shown in Fig. 2 as an example.

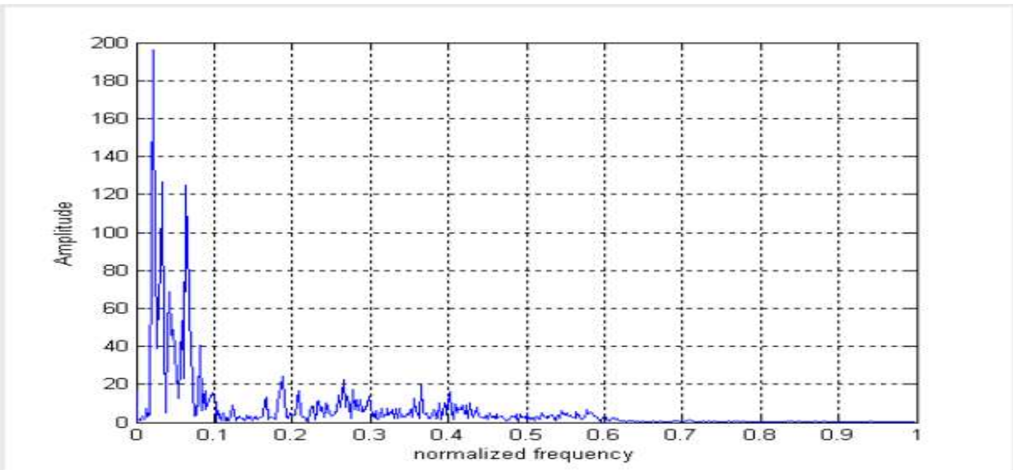


Fig. 2. Recorded Human Audio Signal.

We used normalized frequency in every case due do some beneficial aspects. Normalized frequency has no unit, decrease calculation complexities, good for graphical comparison.

3.2 Signal Digitization

When we work with digital signal processing in computer, the analogue signal has turned digital automatically. So, there was no need of extra arrangement to turn the analogue signal into digital.

3.3 Design of digital filter

This subsection is normally complex compare to other parts. The most straightforward way to implement a digital filter is by convolving the input signal with the digital filter's impulse response [3]. Actually, digital filters are described by their impulse response which is called filter kernel. The impulse response is the output of the filter if the filter is feed by impulse. According to filters impulse response, digital filters are classified into finite impulse response (FIR) filter and infinite impulse response (IIR) filter respectively. There are several methods to design a digital filter [1-6]. We discuss here one simple method called Windowing method because of two main applications which one is to data analysis based on Fast Fourier Transform (FFT) and the other is to design of Finite Impulse Response (FIR) filters from Infinite Impulse Response (IIR) filters. For FFT analysis, windows are employed to suppress the so-called 'leakage effect' and for FIR filter design according to the 'windowing method', Gibbs oscillations are attenuated [1-3]. Desirable characteristics for a window in the frequency domain are small main-lobe width and side-lobe peak [14]. However, these two requirements are contradictory, since for a given length, a window with a narrow main-lobe has a poor attenuation, and vice versa. According to [14], also a third preferred property of window when applied in data spectrum analysis is that, in the time domain, the sum of window function (w[n]) with its shifted version by M/2 samples (M is the window order) would be constant shown in Equation (1).

$$w[n] + w\left[n - \frac{M}{2}\right] = constant, \qquad \frac{M}{2} \le n \le M \tag{1}$$

Unispherical windows [7], which are derived from Gegenbaur polynomials, have the property that the side-lobes roll-off can be controlled by a parameter, independent from the main-lobe width and first side-lobe level. Data windows given by the inverse of the product of two Gamma functions [8], offer the same advantage of controlling side-lobes decay, but in trade-off with the main-lobe width. Another class of windows based on Gegenbaur polynomials introduced in [9], besides of having controllable tradeoff between main-lobe width and side-lobes peak, also have an additional parameter to combine the two following criteria in designing FIR filters by desired ratio: minimizing side-lobes peak, and/or minimizing their energy. A family of windows based on Legendre polynomials has also been reported in [10]. The windows in [7-10] have high computational complexity in deriving the windows coefficients; therefore they may not be used in real-time applications as well as have no property of (1). To overcome this difficulties, a windowing method, the so-called Hamming window, based on the 'Sinc' function have been introduced in [14], which offers the advantage of equation (1). The design of linear FIR low pass filter using Hamming window which is a special case of the important class of windows, named raised cosine window [13], is shown in Fig. 3.

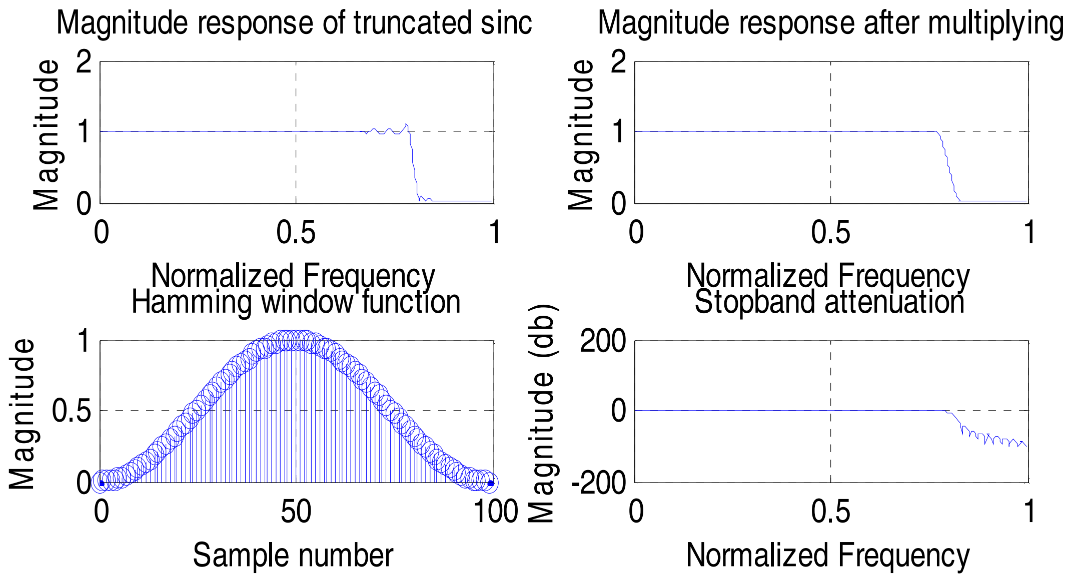


Fig. 3. Design of digital low pass filter using Hamming window at frequency domain.

It is found that in [15], the Hamming window has about a 20% faster roll-off than the Blackman. The Mathematical model of Hamming window is given as follows:

$$w[n] = 0.54 - 0.46\cos\left(\frac{2\pi n}{M-1}\right), \quad 0 \le n \le M-1 \tag{2}$$

Where M is represent the length of filter kernel. When equation (2) is convolving with truncated-sinc function, the output result is found like Fig. 3 after representing in frequency domain based on FFT and dB scale for showing passband ripple (0.2%) and stopband attenuation (-53dB) respectively.

3.4 Filter Response

When the input audio signal (shown in Fig. 2) is passed through the desired low pass filter (shown in Fig. 3), the output signal is found like Fig. 4.

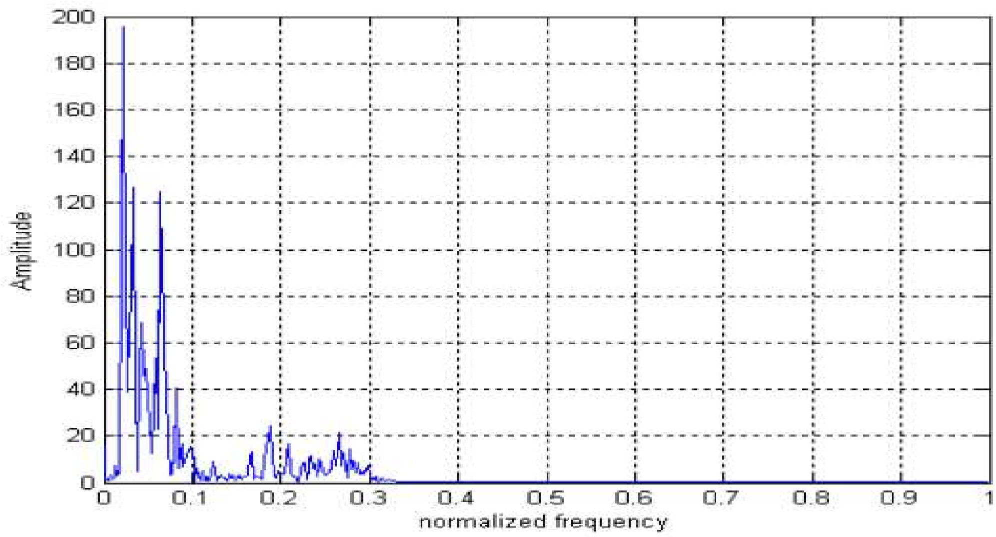
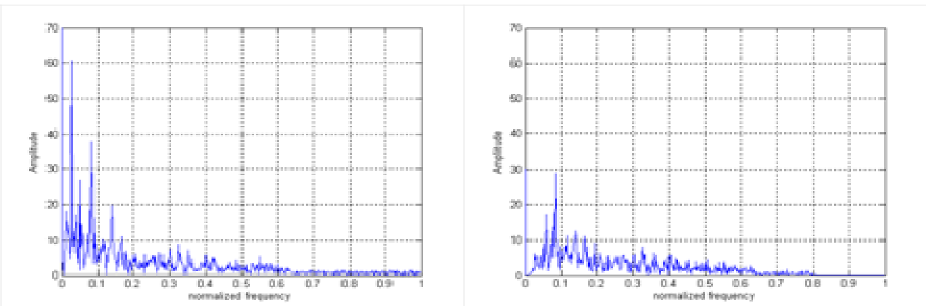


Fig.4. Output filtrated signal

3.5 Signal Synthesis

In the sciences and engineering, the decomposing process of a function into simpler pieces is often called an analysis. The corresponding operation of rebuilding the function from these pieces is known as synthesis. Audio signal synthesis is done by taking FFT (Fast Fourier Transform) and IFFT (Inverse Fast Fourier Transform). After FFT of the filtrated signal, low amplitude suppression has been accomplished. Low amplitude reduction has been done by down sampling method. In signal processing, down sampling or 'sub sampling' is the process of reducing the sampling rate of a signal. This is actually done to reduce the data rate or the size of the data. The signals before reconstruction and after reconstruction are shown in Fig. 5 .



Before Reconstruction After Reconstruction

4. Result and discussion

We took several input signals with different interval of times and analyzed by following above procedure. After reconstruction of the signal, it is found that the sound quality is unchanged whereas the memory size is almost half compare to original signal. Actually the memory size depends upon the down sampling rate. Table 1 shows a comparison between before and after reconstruction.

TABLE I

A COMPARISON BETWEEN BEFORE AND AFTER RECONSTRUCTION OF DIFFERENT TYPES OF AUDIO SIGNALS

Serial No.	Period	Memory(KB) (Before Reconstruction)	Memory(KB) (After Reconstruction)
1	2 s	25.5	12.7
2	5 s	64.8	31.8
3	8 s	102	50.88
4	15 s	191	95.40
5	1.0 min	736.8	381.6
6	1.15 min	954.5	477.0

5. Further study

Many Scientists and Engineers feel guilty how to remove noise from noisy signal using digital low pass filters. This is because every linear filter (including high pass, band pass) can be implemented with the help of low pass filter. However, good performance in the frequency domain results in poor performance in the time domain, and vice versa. Now, it is challenge to implement the mathematical model of window function which shows better performance in time and frequency domain so that it can be easily analyzed not only in signal (including image, speech and biomedical such as ECG, EEG, MRI etc.) processing but also in communication (cellular, radar and satellite etc.).

6. Conclusion

At first, we designed a filter according to our need and came up with a decision that FIR filters have their own advantages over IIR filters. As they need only finite numbers of samples so there is need of finite amount of memory whereas IIR filter needs infinite amount of memory. However, Matlab software provides different options for digital filter design, which include function calls to filter algorithms. A variety of filter design algorithms are available in Matlab for both IIR and FIR filters. We also used Matlab for the FFT & IFFT computation for signal processing. In practical, we have done synthesizing of the filtered signal. By using this technique, it is found that the synthesized signal acquired the memory size around fifty percent less compare to the original input signal. But, the quality of signals has remained the same. This technique can be used to minimize the size of disc drive which is helpful to get a lot of audio signal in a fix disc place. But, if the input signal memory is large enough than there is a problem occurred during matrix array formation which is one of the main drawbacks of this work.

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