DESIGN AND IMPLEMENTATION OF DIGITAL FILTERS FOR AUDIO SIGNAL PROCESSING

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1]AUDIO SIGNALS PROCESSING WITH DIGITAL FILTERS IMPLEMENTATION USING MYDSP.

The document contains software application for Digital Signal Processing which is implemented with a device MYDAQ. It has audio signals from MP3 Files which are used as input data. A software tool GUI is developed for this porpoise to visualize frequency spectrum response. Two specific filters as the Finite Impulse Response and Infinite Impulse Response were implemented and compared. The procedure and simulation are designed in Matlab. [1].

2]DESIGN AND ANALYSIS OF DIGITAL FILTERS FOR SPEECH SIGNALS USING MULTIRATE SIGNAL PROCESSING.

Digital filter provide an important role in world of communication. This paper proposes the design of digital-filters for audio application using multi rate signal processing. The main aim of this paper is to analyze various technique for designing digital- filter for speech signal. Additive White Gaussian Noise is added with the input speech-signal. The input speech-signal spectrum is divided into frequency subbands using down sampling by a factor 2. Various transforms like FFT, FWHT,DWT are applied to the signal and its sub-bands. Then the low pass and high pass FIR filters are designed and implemented using windowing technique and IIR filters are designed and implemented using Butterworth,Chebyshev filters. Finally quantization is performed on the filter coefficient of signal and it's sub bands. The performances of digital-filter are measured by calculating Signal to Quantization Noise-Ratio. From the performance measures this paper concludes that which filtering technique is most suitable for designing digital-filter for speech signal. [2].

3]IIR BASED DIGITAL FILTER DESIGN AND PERFORMANCE ANALYSIS.

Digital-filter are mandatory for digital-signal processing. This paper presents digital-filter dispelling the unwanted-signal or noise from the required-signal and enhances the better performances of the signal. The extracted features of the digital-filter has analyzed to acquire the better output of the signal by using IIR Butterworth-filter. It provides different designed parameters of IIR-filter to achieve the desired result. MATLAB FDA-tool is considered to find out the different response of a Phase-response, digital-filter. About eight parameters like Magnitude-response, Magnitude,Phase-response, Step-response, Group-delay, Pole-Zero Plot, Phase-delay, Impulse- response are used to analyze the filter-response. Some selected response of audio-signal are used for observing the empirical high-pass, low-pass, band-stop filter and band-pass filter. [3].

4]IMPLEMENTATION AND PERFORMANCE ESTIMATION OF FIR DIGITAL FILTERS USING MATLAB SIMULINK

In modern communication-system, filtering is the most widespread,extremely important signal processing technology. FIR-filter form the basis of wireless systems in medical-devices, industrial-control, consumer-electronics and cellular-infrastructure. A filter network selectively changes the wave shape of a signal in a desired approach. The most common filtering purpose is to remove noise from signal. FIR-filter are of finite transient duration more stable and efficiently realisable and providing exact linear phase as compared to IIR-filter.FIR- filter structure can be used to realize almost any sort of frequency-response digitally. FIR-filter are invariably used in the situation where linear phase characteristics within the passband of the filter are required. In this paper, implementation and simulation of 1D , 2D FIR-filter is presented in the MATLAB and Simulink environment. The comparison of the output-waveform show that 2D FIR-filter posses higher speed than 1D FIR-filter. [4].

5]DESIGN AND IMPLEMENTATION OF BUTTERWORTH, CHEBYSHEV-I AND ELLIPTIC FILTER FOR SPEECH SIGNAL ANALYSIS

In field of digital-signal-processing, the function of the filter is to remove unwanted part of the signal such as random noise that is undesirable. To remove noise from the speech-signal transmission or to extract useful part of the signal such as component lying within a certain frequency-range. Filters are broadly used in signal-processing and communication-systems in applications such as channel equalization, noise-reduction, radar, audio-processing, speech- signal-processing,video-processing,biomedical-signal-processing that is noisy ECG, EEG, EMG signal-filtering, electrical-circuit analysis and analysis of economic and financial data. In this paper, three types of infinite impulse response-filter that is Butterworth,Chebyshev type I, Elliptical filter have been discussed theoretically and experimentally. Butterworth, Chebyshev type I, elliptic low-pass, high-pass, band-pass and band-stop filter have been designed in this paper using MATLAB-Software. The impulse- responses, magnitude-responses,phase-responses of Butterworth, Chebyshev type I ,Elliptical filter for filtering the speech signal has been observed in this paper. Analyzing the Speech-signal, its sampling-rate and spectrum-response have also been found. [5].

6]A STUDY ON DESIGN AND IMPLEMENTATION OF BUTTERWORTH, CHEBYSHEV and ELLIPTIC FILTER WITH MATLAB

Beside the mathematical comparison between Butterworth Chebyshev-type-I,elliptic-filter, in this paper, these filters have also been encountered with the designed low-pass, high-pass, band-pass and band-stop infinite impulse-response filter with a view to comparing their responses for different parameters like impulse-response, magnitude-response, phase-response and pole-zero characteristics all done using MATLAB simulation. The filter order, pass-band and stop-band ripple are considered for the design of the IIR-filter. It has been found that the Butterworth-filter is the best compromise between attenuation and phase-response. It has no ripple in the pass-band or the stop-band. The speech-signal have also been encountered using MATLAB-simulation, which was the special consideration, and compared the input-output spectrum of the signal. [6]

7]BASE PAPER-

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An analog active-filter do not provide a very sharp cut-off for both higher and lower-frequency component.While a digital-signal-processor using digital-filter effectively reduces the unwanted higher and lower order-frequency components within an audio-signal.The paper has presented the design and implementation of digital-filter for audio-signal. Experimental results show that digital-filter including low pass, high pass,band pass effectively eliminate both low or high frequency components contained in human-voice, providing high- quality voice. [7].

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