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Voice Collection under Different Spectrum

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Abstract

According to the short-time Fourier transform theory and principle of digital filtering, this paper established a mathematical model called collection of voice signal collection at different spectrum. The voice signal was a non-stationary process, while the standard Fourier transform only applied to the periodic signal, transient signals or stationary random signal. Therefore, the standard Fourier transform could not be directly used for the speech signal. By controlling the input different types and parameters, this paper analyzed the collected original voice signal spectrum with the use of MATLAB software platform. At the same time, it realized the extraction, recording and playback of the speech signal at different frequencies. Therefore, the waveforms could be displayed obviously on the graphic user interface and voice effect could be more clearly. Meanwhile, the result was verified by the hardware platforms, which consisted of TMS320VC5509A [1] chip and TLV320AIC23 voice chip. The results showed that the extraction of voice signal under different spectrum model was scientific, rational and effective.

Keywords: *DSP, speech signal, Graphic User Interface, digital filter*

1. Introduction

As the hot research topics in the field of high-tech application, voice signal has a more closely relationship with many fields. Such as telephones in the areas of industrial production, automatic dial of telecommunication systems, speech recognition [2], speech consultation and management, healthcare and assisted query in the areas of living . Speech signal processing can be more efficient to produce, transport, store and get speech message, which has a great significant for promoting social development.

Digital signal processor (DSP) is specially designed for digital signal processing algorithm of real-time and rapid implementation [3]. It involves many subjects and is widely used in many areas of emerging disciplines. It has more advantages than simulation system, such as predictability, programmable, high precision, good stability, reliability and repeatability, easy to realize the adaptive algorithms, large-scale integration etc. This paper uses TMS320VC5509A chip launched by TI, this chip has the characteristics of efficiency, portable and low consumption. It also adopts the uniform addressing ways to partition the storage space, which is convenient to a program optimization, as well as the realization data processing. It provides a convenience for the speech signal processing by using the IIC, McBSP [4] or RTC peripheral interface.

The design of the speech signal system interface is based on MATLAB GUI environment. It designs a digital signal filter with the characteristics of handling low frequency signals, no drift and the ideal frequency response, and finally complete speech signal acquisition, extraction and playback. Users can directly loading and listen to the speech resource files. By clicking on the corresponding button, we can complete the data results storage and waveform display after the extraction.

2. Research Method

2.1. Short-time Fourier Transform theory

Standard Fourier transform only applies to transient signals, periodic signal or stationary random signal. The speech signal belongs to non-stationary process, with the characteristics of time-varying. But within a short time, its characteristics remain relatively

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stable. Therefore it could be seen as a quasi-steady-state process [5]. Short-time Fourier transform (STFT) is a mathematics transform related to the Fourier transform, which is to select a time-frequency localization window function. Assumes that the window function g(t) is smooth within a short time interval (pseudo smooth), movable window function, makes f (t)g(t) a smooth signal within different time, and calculates out power spectrum at different moments. Short-time Fourier transform window functions can't be changed, once selected, its shape will not change, and the resolution is determined accordingly. To change the resolution, we need to reselect the window function.

2.2. Design of FIR filters

The paper adapts window functions method to design FIR filter, it can be realized arithmetic by differential equations $y(n)=\sum_{k=0}^{N-1} h(k)x(n-k)$ [6]. In the formula, $x(n-k)$ is the k sampling period delays of input signal, N is the number of filter order, $h(k)$ is the k time-delay weighted value (filter coefficient), $y(n)$ is the filter's output signal of $t=nT$.

3. MATLAB Model Design and Simulation

This design acquires a .wav format's audio file through the computer sound card. First, conduct a short-time Fourier transform, then design low-pass, high-pass, band-pass and band-stop filter for the transform results. We can playback the speech after extraction of the high frequency, low frequency, medium frequency through the broadcast function, therefore the effect after filter extraction is more clearly.

3.1. Design of FIR Filters

The paper uses the function $fir1$ in the MATLAB signal processing Toolbox to design FIR filter. The $fir1$ calls format: b = $fir1(n, Wn, 'ftype', window)$ [7].

The n is FIR filter order, which is a even for high-pass and band-pass filter; Wn is cutoff frequency: For band-pass and band-stop filter: $Wn=[W1, W2]$, $W1$ and $W2$ are respectively the lower cut-off frequency and the upper cut-off frequency. ftype is the filter type, it is a low-pass or band-pass filter without emphasized. The $'high'$ is high-pass filter and the $'stop'$ is band-stop filter; window is the type of window function.

3.2. Main Functions for MATLAB Interface Design

1) [y, FS, bits] = wavread ('filename')

'wavread' supports multichannel data, with up to 32 bits per sample, and supports reading 24- and 32-bit .wav files. The .wav extension is appended if no extension is given. Amplitude values are in the range [-1,+1]. Returns the sample rate (Fs) in Hertz and the number of bits per sample (bits) used to encode the data in the file. 'y' represents a string of data, in which stores the sample values. 'fs' is .wav format file 's sampling frequency (Hz); 'bits' is the quantification length in the A/D converter, that is sampling bits [8].

2) sound (x, fs, bits)

It plays the sound using bits number of bits/sample, if possible. Most platforms support bits = 8 or bits = 16.

wavplay is a synchronous operation, MATLAB will stop other tasks until it has finished playing. If you require other operations while playing the speech, sound or soundsc function can be used for non-synchronous play. The difference between them is that soundsc calibrates the values in the x ranging from -1 to 1, for formalizing processing, in order to achieve the best performance.

3) wavreord (n,fs)

It records n samples of an audio signal, sampled at a rate of Fs Hz (samples per second). The default value for Fs is 11025 Hz.

4) wavwrite (x, fs, n, 'filename')

It writes the data stored in the variable y to a WAVE file called filename. The data has a sample rate of Fs Hz and is N-bit, where N is 8, 16, 24, or 32. For N < 32, amplitude values outside the range [-1,+1] are clipped.

5) wavplay (y,Fs)

It plays the audio signal stored in the vector y on a PC-based audio output device. You specify the audio signal sampling rate with the integer Fs in samples per second. The default value for Fs is 11025 Hz (samples per second). wavplay supports only 1- or 2-channel (mono or stereo) audio signals.

3.3. Use of Graphic User Interface

GUI [9], [10] can be used to create controls needed in this paper. Three dynamic text box FH, FL, Fs are used, they are respectively the upper cut-off frequency, lower cut-off frequency and the sampling frequency. A pop-up menu is used to select the FIR filter type. Add five axes: two waveforms on the left are respectively for the original signal time-domain and frequency-domain, waveforms on the right show the time-domain and frequency-domain waveforms after filtering. The Middle is the amplitude-frequency for filtering. Buttons on the right are function selection keys.

3.4 Simulation Results and Analysis

(1) Design of high-pass filter

If the high frequency part of speech signal is to be extracted, put the signal in the high-pass filter. Fs is set for 400Hz, FH is set for 100Hz. At this time, high frequency is selected while low frequency is filtered. Compared with the original signal, the sound becomes more acute after high-pass filter.

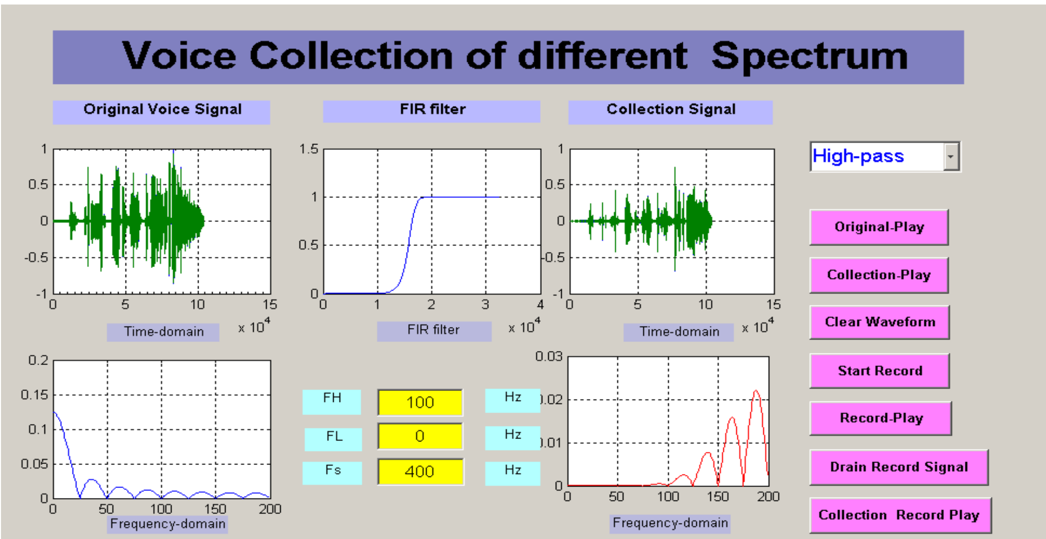

Figure 1. High-pass filter

(2) Design of band-stop filter

Reset the upper and lower cut-off frequency, low frequency and high frequency band is reserved, intermediate frequency is filtered through the band stop filter. The spectrum changes of the speech before and after filtering shows: To some extent, speech after filter becomes lower, but it is close to the original speech signal.

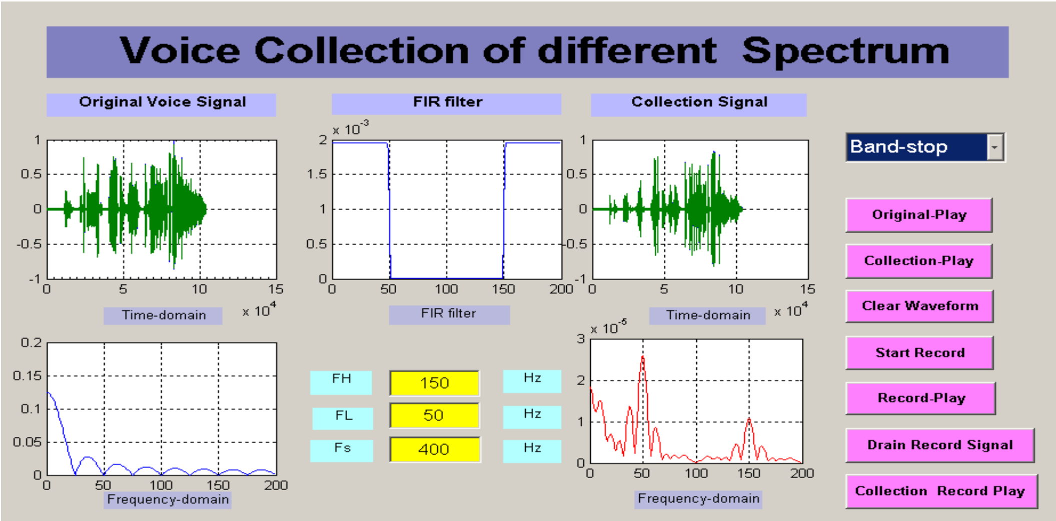

Figure 2. Band-stop filter

(3) Design of low pass filter

Select low-pass filter, modify the parameters of high frequency and low frequency. The waveform shows that: the low frequency is reserved while the high frequency is filtered, the speech after low-pass filter becomes low and boring.

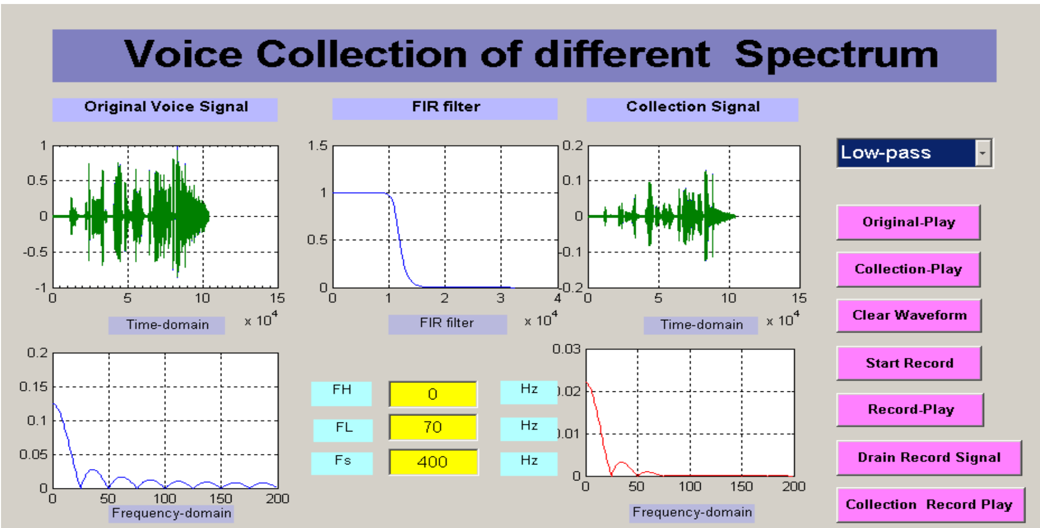

Figure 3. Low-pass filter

(4) Real time recording of speech signal

The paper increases the real-time recording speech signal function. Click the start-record button, speech signal can be recorded and saved. Select filter type, modify the parameters, we can extract the certain frequency range by the function of real-time recording speech signal. This paper adopts band-pass filter, extracting the frequency from 50Hz to 150Hz. The speech becomes acute after the band pass filter, but is lower than high pass filter.

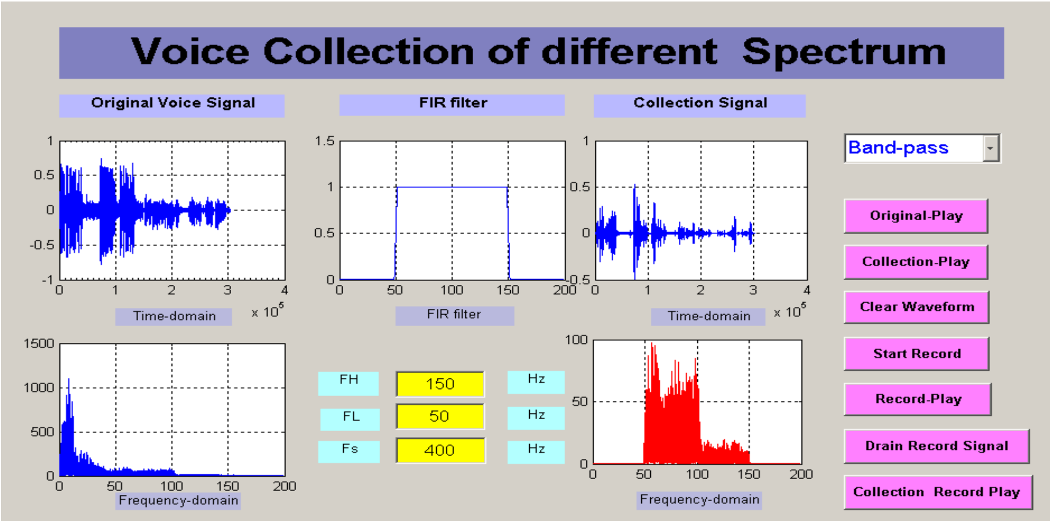

Figure 4. Recording speech signal

(5) Sound effects of different FIR filters

With the comparison of waveforms and sound effects, we can see: The speech sounds lower if the low frequency is extracted .In the contrast, the speech becomes sharp. The results of the comparison among the four filters are shown in the table 1.

Table1. Sound effects of different FIR filters

Filter type	Frequency selection	Sound effects
High pass	High frequency	pitch、sharp、harsh
Band stop	Band frequency	low、but close to original signal
Low pass	Low frequency	deep、low、boring
Band pass	Middle frequency	sharp、but close to original signal

4. System Implementation

4.1. Hardware Design Scheme

The system hardware design scheme is shown in figure 5, it mainly consists of DSP processing module and speech signal acquisition module. Its working process is as follows: mike acquires a speech signal, put it into the signal conditioning circuit. Speech signal acquisition and processing module (TLV320AIC23) [11] mainly completes speech signal A/D and D/A conversion [12]; TMS320VC5509A (DSP) processing module realizes the communication with audio chip, and completes the speech signal processing function [13]. The speech signal after processing is filtered and carried on power amplifier, then playback the speech by the headphone.

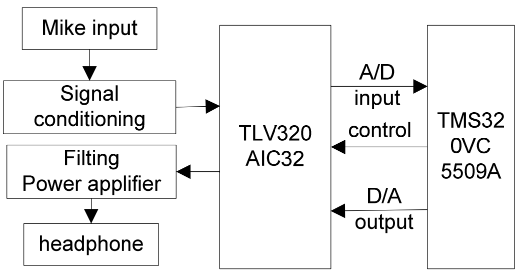

Figure 5. Hardware design scheme

4.2. Software Flow Chart

System software contains the main program and interruption subroutine. The main program completes the system hardware initialization, key scanning and key processing. While interrupt program mainly completes data acquisition, software filtering, speech recovery processing.

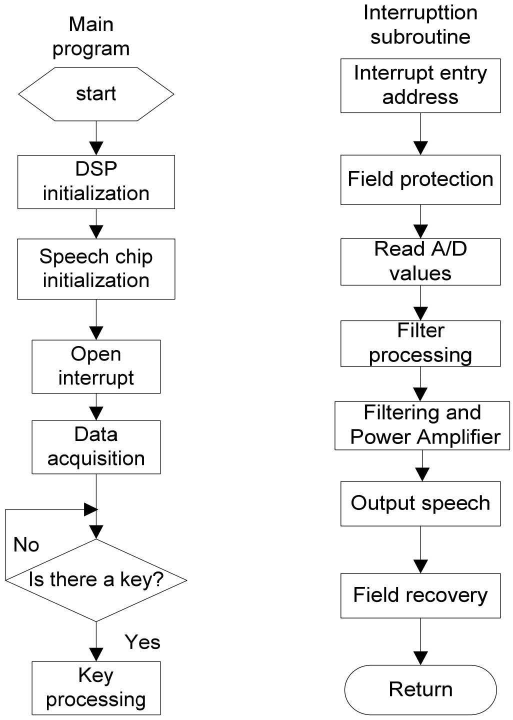

Figure 6. Software flow chart

Parts of the software program code are as follows:

```
void main()
{
  uint16 AIC23data = 0 ;
  // Initialization CSL library
  CSL_init();
  // Set system frequency : 144 MHZ
  PLL_config(&myConfig);
  hMcbsp = MCBSP_open(MCBSP_PORT1,MCBSP_OPEN_RESET);
  //setMcBSP1
  MCBSP_config(hMcbsp, &Mcbsp1Config);
  //start McBSP1
  MCBSP_start(hMcbsp,MCBSP_RCB_START| MCBSP_XMIT_START,0);
  ......
  //set AIC23 digital interface
  I2C_write(digital_audio_inteface_format, //pointer to data arrary
            2,                //length of data to be transmitted
            1,                //master or slaver
            CODEC_ADDR,       //slave address
            1,                //transfer mode
            30000             //time out for busy bus
            );
```

```
//setAIC23sampling frequency
I2C_write(sample_rate_control,2,1,CODEC_ADDR,1,30000);
//start AIC23
I2C_write(digital_interface_activation,2,1,CODEC_ADDR,1,30000);
while(TRUE)
{
   //speech signal processing program
}
```

5. Conclusion

The extraction of speech signal of different frequency band has many extensive applications, such as speech recognition, cartoon sound synthesis, etc. This paper combines MATLAB GUI and M documents , establishes speech signal extraction model; realizes the speech signal acquisition, different frequency band extraction, playback functions based on MATLAB software platform . It has been verified by the hardware platform, which contains TLV320AIC23 and TMS320VC5509A .The results show that the algorithm is efficient and feasible, thus realize the inspection of theoretical knowledge. At the same time, it provides a necessary premise for new development and validations.

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