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Research on DTMF signal detection algorithm for FFT

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Abstract. As a signaling method widely used in communication systems (such as telephone buttons and remote control), the core of Dual-tone multi-frequency (DTMF) signal is to represent a key information through the combination of two specific frequency sine waves. In this paper, a detection algorithm based on fast Fourier transform (FFT) is proposed to meet the detection requirements of DTMF signals. The algorithm uses FFT to convert the time-domain signal into a frequency-domain spectrum after preprocessing (filtering and sampling) the acquired DTMF signal, and identifies the row frequency and column frequency components contained in the signal by analyzing the peak frequency in the spectrum, and then maps it to the corresponding key information. Experimental results show that the algorithm can effectively extract the frequency characteristics of DTMF signals, with high detection accuracy and good real-time performance, and can be applied to signal analysis in telephone communication, industrial control and other scenarios.

Keywords: FFT; DTMF; Signal detection; algorithm.

1. Introduction

DTMF (Double Tone Multi Frequency) is a type of audio telephone dialing signal, most of which are used for audio dialing. Inaddition to being used in program-controlled telephone switching systems, DTMF technology is also widely used in interactive control, such as voice menus, caller IDs, phone banking, email, ATM terminals, smart home remote control, automatic control systems, and in-vehicle navigation terminals [1]. Double tone, mainly each button is composed of two frequency audio signals; multi-frequency, mainly composed of 4×4 different frequencies for each key, forming a total of 16 different frequency signal combinations [2]. DTMF signals are encoded by selecting 2 of the 8 frequencies in combination, and taking a frequency combination from eachofthe high and low frequency groups represents 0~9 ten digits andothernumbers, and the DTMF coding table is shown in Table 1.

Table 1. DTMF coding table

	1209Hz	1336Hz	1477Hz	1633Hz
697Hz	1	2	3	A
770Hz	4	5	6	B
852Hz	7	8	9	C
941Hz	2. *	0	#	D

According to Consultative Committee of International Telegraph and Telephone regulations, DTMF signaling has a maximum of 10 digits per second, that is, the period is 100ms, of which the duration of the DTMF signal is not less than 45ms and not more than 55ms; Therest of the time is a silent signal to distinguish between the two DTMF digital signals [3].

3. The oretical basis of FFT

At the heart of FFT transform lies the FFT time-domain to frequency-domain conversion formula, which enables the decomposition of time-domain signals into superpositions of multiple sine waves, each corresponding to a specific frequency, amplitude, and phase [4]. The FFT calculation process involves the following steps: Ordinal Rearrangement: Rearrange the data in the input sequence in a specific order, usually in the order of binary inversion. Butterfly operations: In each iteration, the

number of butterfly structures is multiplied by merging data through the butterfly structure. In-situ Computation: An important feature of FFT algorithms is in-situ computation, where the results of each iteration are stordin the storage location of the original data without additional storage space.

When performing FFT, the sampling frequency, the number of sampling points, and the frequency resolution are the three key parameters. The sampling frequency should be at least twice the highest frequency component in the signal, and the number of sampling points is usually taken to the power of 2 for easy calculation, and the frequency resolution is inversely proportional to the sampling frequency and the number of sampling points.

The advantage of FFT over direct calculation of DFT is its efficient computational performance. For large-scale datasets, FFT can significantly reduce computational time, enabling real-time signal processing. In addition, the popularity of FFT algorithm also benefits from its implementation in various programming languages and computing platforms, which realizes the drawing of signal shape and spectrum through MATLAB's plot plotting command, and vividly displays the effect of Fohn spectrum vibration or signal approximation after signal superposition with the help of MATLAB's powerful mathematical functions and drawing capabilities, so asto solve the difficulties of Fohn spectrum theory analysis [5].

4. DTMF signal detection based on FFT

4.1. The role of Matlab in signal analysis detection

If the function f(x) is studied directly, this time-domain analysis method directly explores the law from the development process ofthings. The Fourier series expansion or Fourier transform off(x) is to study the influence of various factors affecting the development of things on the development and change of things, which corresponds to frequency domain analysis [6].

In the MATLAB environment, it has excellent performance in processing digital signal-related tasks by skillfully using multiple basic syntax and programming skills, such as command window operation, script writing, and function file design [7].

MATLAB transforms transform from mathematical theory to actionable engineering tools, lowering the barrier to entry for algorithms, improving analysis efficiency, and expanding application boundaries.

4.2. Detection and identification of DTMF

During frequency detection, the fundamental wave and second harmonic of the DTMF signal aredetected, the DTMF signal has a higher energy on the fundamental wave, while the voice signal has a strong second harmonic superimposed on the fundamental wave, and the DTMF signal and the voice signal are distinguished by detecting the second harmonic [8]. The sampling theorem is in the field of signal processing a basic principle. It describes how discrete samples can be taken from continuous-time signals so that the original signal can be accurately reconstructed when needed. This theory is of great significance for the design and implementation of digital audio, video, and communication systems. To convert analog speech signals into digital signals, they are first sampled and quantized to obtain digital speech signals that are discrete in time and amplitude [9]. In digital signal processing, sampling frequency (i.e., sampling rate) and quantization accuracy have an important impact on signal quality. The sample rate determines how dispersed the signal is in time, while the quantization accuracy determines how dispersed the signal is in amplitude. Proper sampling rate and quantization accuracy selection are key to ensuring the quality of digital voice signals [10]. Therefore, a GUI program is designed to complete the voice signal collection. The waveform diagram of the speech signal in the frequency domain is obtained by fast Fourier transform, as shown in Figure 1.

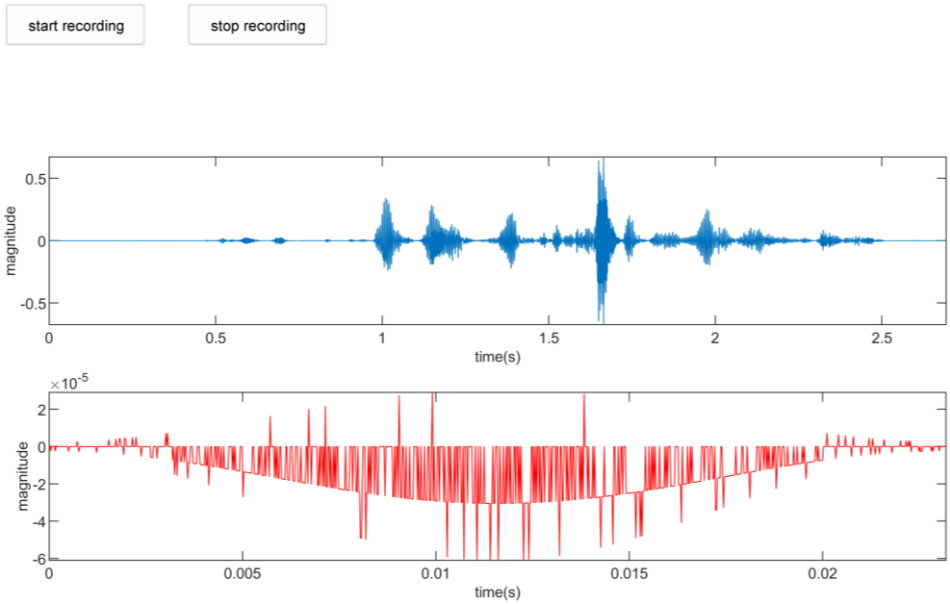

Figure 1. Time-domain waveforms and frequency-domain Wave forms of speech signals

4.3. DTMF signal detection algorithm based on FFT

With previous research, the telephone dialing system was further explored, and the corresponding program was designed, and the graphical interface included each button, the time-domain waveform graph and frequency-domain amplitude spectrum, and the recognition button, which could recognize the voice signal sampled by the file. Combined with the functions of MATLAB, the interface of the telephone dialing system as shown in the figure is designed, which can not only analyze the signals corresponding to the buttons in the time and frequency domains, but also complete the recognition of the signals in the text file.

According to the functional requirements of the program, some important code is displayed. The button that creates the dial pad is the building block for the implementation of the function.

Then, in order to realize the analysis of the time and frequency domains, the DTMF signal is generated, the corresponding frequency is matched, and FFT is performed, and the following is the code to achieve this purpose.

```
[f1, f2] = get_frequencies (key);
t = 0:1/fs: duration-1/fs;
signal = sin (2*pi*f1*t) + sin (2*pi*f2*t);
```

```
function [f, P1] = compute_fft (signal, fs)
nfft = 4096;
Y = fft (signal, nfft);
P2 = abs (Y/nfft);
P1 = P2 (1: nfft/2+1);
P1 (2: end-1) = 2*P1 (2: end-1);
f = fs*(0:(nfft/2))/nfft;
```

Next, you can use simple built-in functions to display the time-domain waveform and

frequency-domain waveform. Finally, the most important thing is the identification of text files. This requires the file's recognition callback and matching to the standard frequency, the code is as follows

```
fullpath = fullfile (path, file);
data = load (fullpath);
key = analyze_signal (data);
axes (ax_result);
cla;
text (0.5, 0.5, sprintf ('Recognition result: %s', key), ...
'FontSize', 24, 'Horizontal Alignment', 'center');
axis off;
```

```
row_freqs = [697, 770, 852, 941];
col_freqs = [1209, 1336, 1477];
[~, row_idx] = min (abs (detected (1) - row_freqs));
[~, col_idx] = min (abs (detected (2) - col_freqs));
buttons = {'1','2','3';
  '4','5','6'; '7','8','9'; '*','0','#'};
key = buttons {row_idx, col_idx};
```

Through the implementation of code requirements and other basic MTLAB functions, the construction of the telephone dialing system is completed. Through the MATLAB simulation results, buttons 0~9 and *, # 's time-domain waveforms and frequency-domain waveforms can be observed. Take button 7 as an example, as shown in Figure 2.

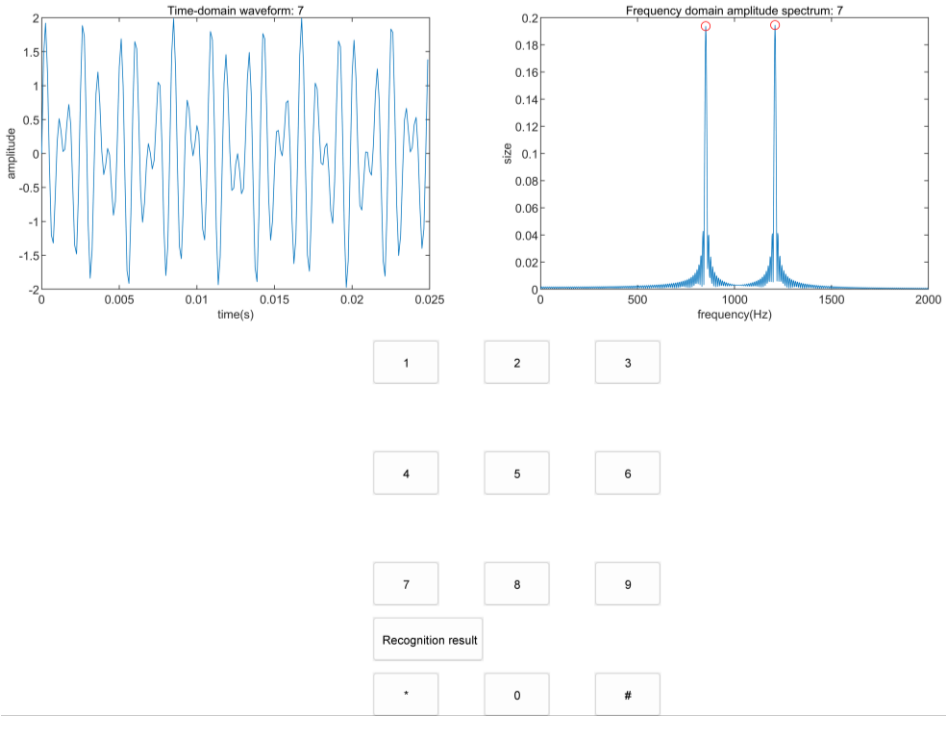

Figure 2. Button 7's time-domain waveform and frequency-domain waveform

It is intuitively clear that the button 7 is actually composed of two single-frequency signals (852Hz and 1209Hz). Not only button 7, but each button is composed of two single-frequency signals, which are composed of two parts: low frequency and high frequency. DTMF signals have a particularly large amplitude at high and low frequencies, and are basically zero at other frequencies.

For a given signal, the corresponding button can be recognized, as shown in Figure 3.

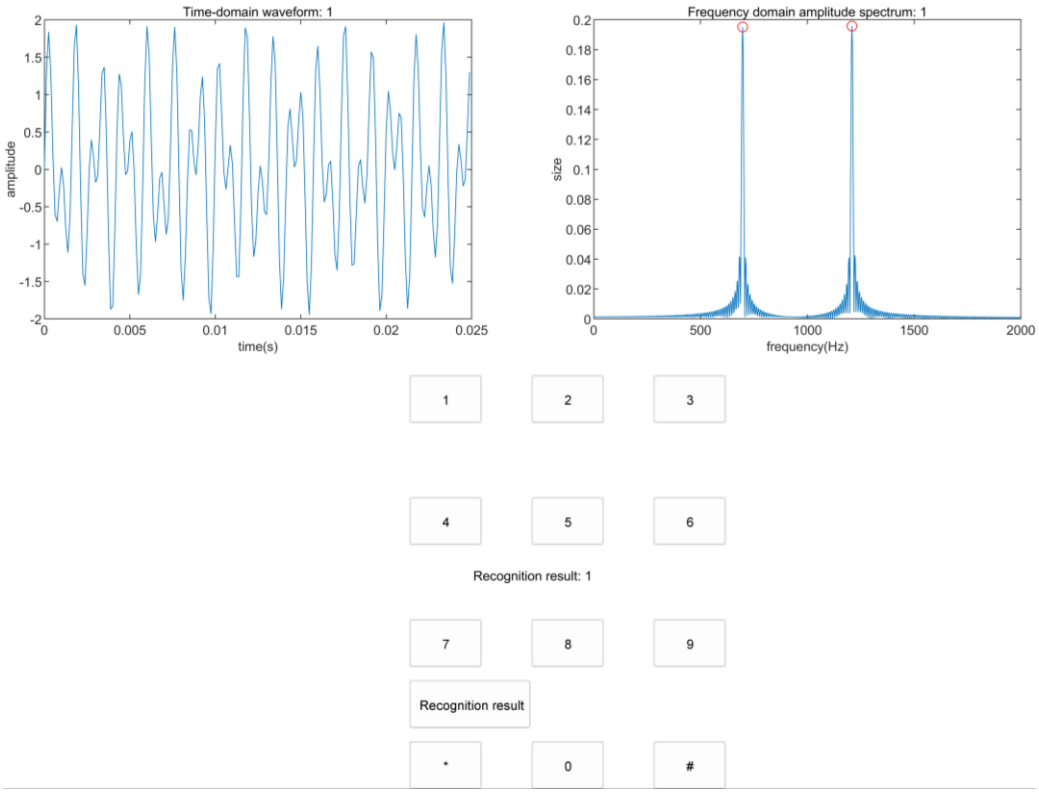

Figure 3. The time-domain waveform and frequency-domain waveform of a given signal

Save the sampling point and sampling frequency of the signal tobe recognized in the text file, and click the recognition button to select the file to identify the signal saved in the file.

5. Conclusion

Fast Fourier Transform (FFT) is a technique used in digital signal processing to efficiently calculate discrete Fourier transform (DFT) by converting signals from the time domain to the frequency domain. Compared with direct DFT calculation, FFT greatly reduces the amount of computation and reduces the time complexity to O (N log N). Combined with FFT, signal processing and analysis can be performed faster.

The DTMF signal contains two sets of audio signals, and the decoder's task is to convert them from the time domain to the frequency domain through mathematical transformations, and then derive the corresponding numerical information. The algorithm studied in this paper has a similar function to the decoder and can recognize DTMF signals. Since the chip processes digital signals, the input signals must be digitized, which is also reflected in the algorithm.

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