DETECTION OF INFORMATION CARRIER SIGNALS USING BAND-PASS FILTERS

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Abstract

Band-pass filters have an important role in the spectral analysis of information carrier signals. A carrier signal which is usually like a plain sine oscillation is varied in its amplitude, frequency or phase when conveying information. In this paper we tried to detect the carrier frequencies of dual tone multiple frequency (DTMF) signals generated by regular push-button telephones and corrupted with various levels of noise. The detection is done using designed finite impulse response (FIR) band-pass filters that can pass only one particular frequency. The filter design has been realized using the Matlab system and DSP toolbox.

1 Introduction

The purpose of DTMF detection is to detect periodical signals in the presence of noise. Estimating the frequencies of noisy signals is an important problem in signal processing. The presented paper describes a solution of one of the individual student's projects studied in the optional lessons "Multimedia in Signal Processing". Three telephone signals with various levels of noise are given and the goal of the project is the recognition of the phone numbers represented by each signal. The Discrete Fourier Transform (DFT) and the finite impulse response (FIR) filters were used for analyzing and filtering the original signals.

2 DTMF Specifications

Dual Tone Multiple Frequency (DTMF) signals are commonly used in telephone systems. They consist of two components with different frequencies (see Tab. 1). When a key of a phone touch pad is pressed the tones with the frequencies from the given column and row are summed and the corresponding tone is generated. Each digit lasts for 50 msec and is followed by a silence interval duration of 50 msec. The DTMF signals are usually sampled at 8 kHz.

	High frequency			
Low frequency	1209 Hz	1336 Hz	1447 Hz	1633 Hz
697 Hz	1	2	3	A
770 Hz	4	5	6	B
852 Hz	7	8	9	C
941 Hz	$\star$	0	#	D

Table 1: A telephone keypad and the DTMF frequencies for each column and row.

3 Spectral Analysis of DTMF Signals

One of the possibilities to analyze DTMF signals is to compute its Discrete Fourier Transform (DFT) [2, 3, 4]. In its simplest form, the DFT is given by

$$X(k) \;=\; \sum_{n=0}^{N-1} x(n)\, e^{-jk2\pi/N} \tag{1}$$

where $x(n)$ is the input signal of length N samples and $X(k)$ is its Fourier transform. Thanks to the DFT properties, it would be possible to determine two characteristic frequencies from the set of 8 frequencies given by Tab. 1 in the case of an ideal signal segment with absence of noise. The signal segment selection and corresponding spectrum estimation can be realized by means of Short Time Fourier Transform (STFT). The various shapes of sliding window can be used [4] in this case. An example of application of DFT to the selected signal is presented in Fig. 1.

4 DTMF Signal Detection

Three telephone signals with various levels of noise were given in this project with the goal of signal analysis and decoding the phone numbers. It is necessary to perform the following steps to achieve the given signal decoding:

- Computation of the spectrum of the original signal by means of the STFT and noise removing using the corresponding FIR band-pass filter. Using the MATLAB function `fir1` a filter of length 64 was designed with the cutoff frequencies 650 and 1700 Hz. We used the function `freqz` to find the frequency response of the filter (Fig. 1). The following figures (Figs. 2, 3 and 4) show how the noisy signals were filtered using the FIR filter. The signals were corrupted with different levels of noise. As shown in the figures every signal contains 14 digits.

- Generation of the DTMF signal using the frequencies shown in Tab. 1. The input to this procedure is a scalar value for the time duration of each button signal (the implicit value is 50msec) and a vector of numbers to be dialed. Results of these two steps serves as the inputs to the third part of the algorithm

- Detection of dialed number in the analyzed signal by means of calculation a matrix of correlation coefficients for an array of the given signal spectrum and the spectra of the generated DTMF frequencies. A matched filter concept is applied for each signal segment spectrum to determine the frequency at which the incoming signal has maximum energy. This is performed by using the MATLAB function `corrcoef` which calculates a matrix of correlation coefficients for an array of the telephone signal spectrum and the spectra of the generated frequencies. A maximum value corresponds to the index of the digit to be detected.

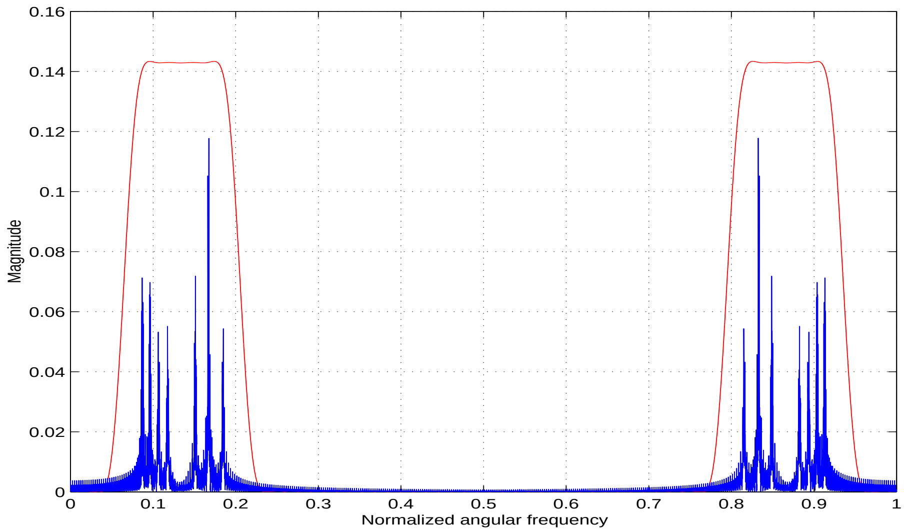

Figure 1: Spectrum of a given signal and frequency response of a FIR band-pass filter

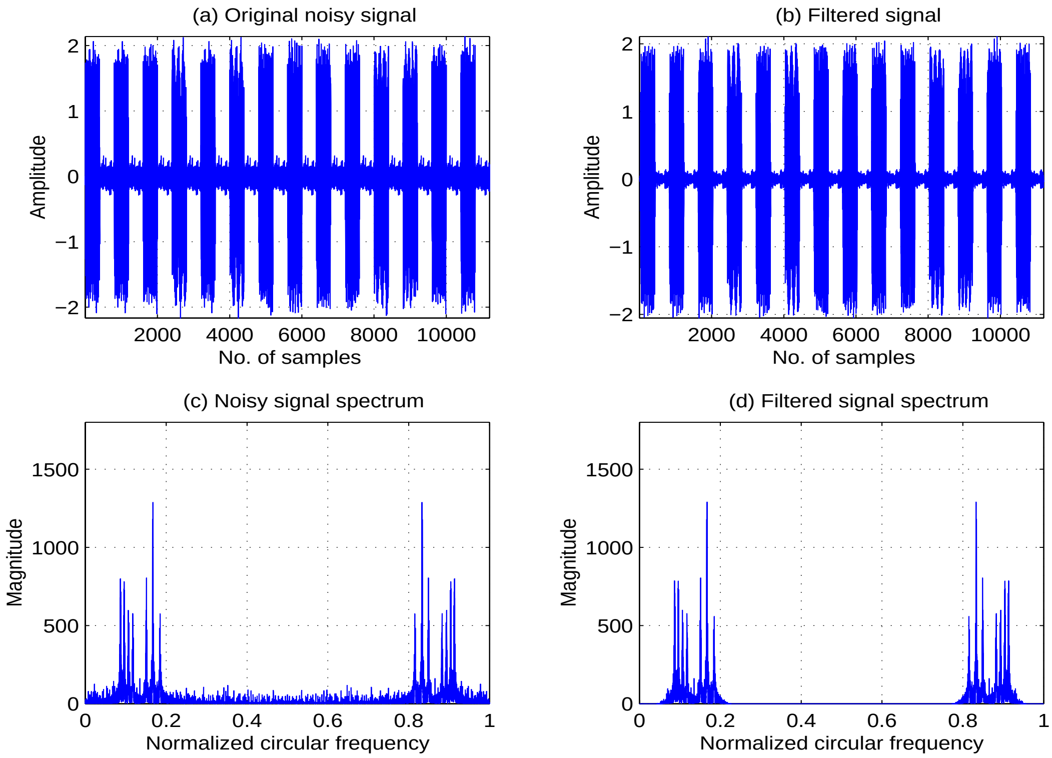

Figure 2: Results of the application of FIR filter - telephone signal 1.

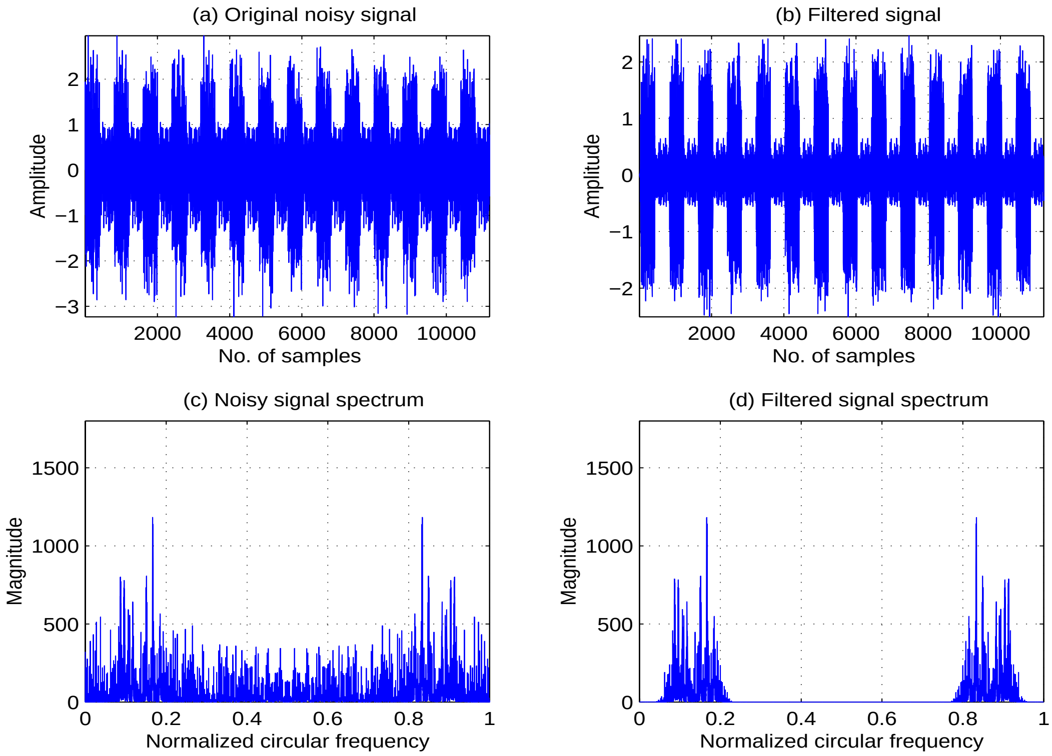

Figure 3: Results of the application of FIR filter - telephone signal 2.

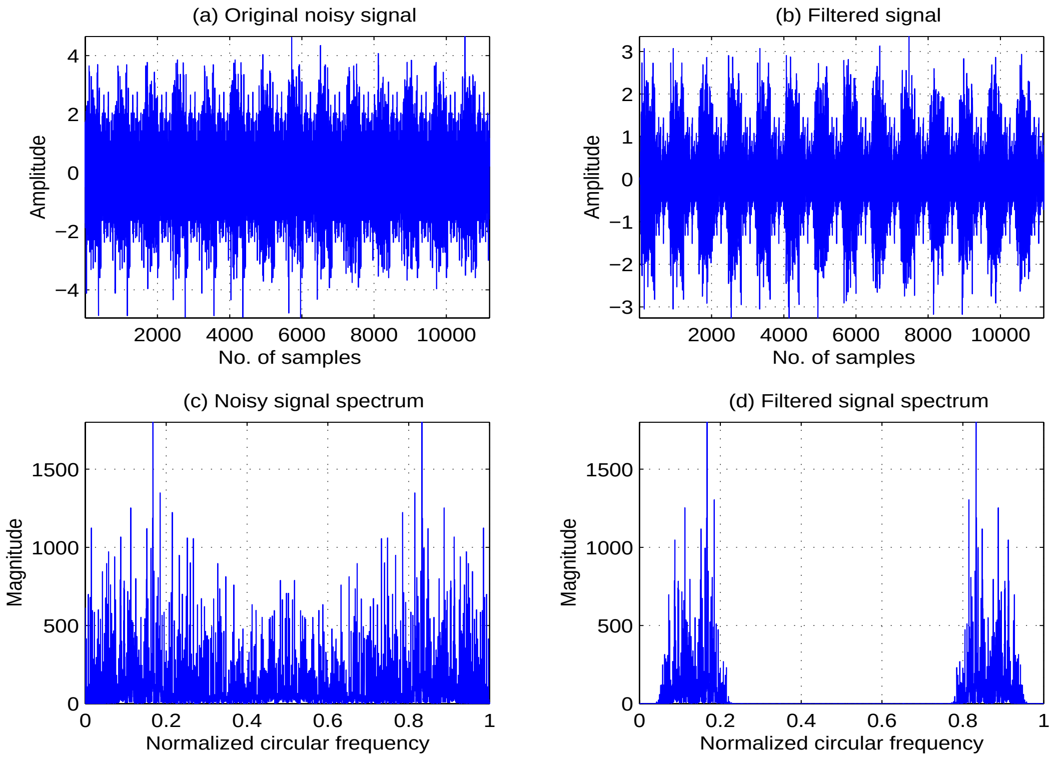

Figure 4: Results of the application of FIR filter - telephone signal 3.

5 Conclusion

The presented paper deals with one of the possibilities of analysis and decoding of DTMF signal. Its main contribution is the package of programs in the Matlab environment involving the DTMF generation, signal analysis including noise rejection and phone number detection. The function of the DTMF generator is formation of a selected number. The DTMF detector is based on spectral analysis by means of DFT and correlation analysis. The functions of the algorithms were verified on the signals with various noise levels. Apart from the quality of input signal, the number detection was successful in the given cases.

Acknowledgments

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