論文 87-24-2-3

PARCOR 분석 방법에 의한 디지탈 DTMF 수신기 구현에 관한 연구

(On Implementing the Digital DTMF Receiver Using PARCOR Analysis Method)

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要 約

디지탈 DTMF 수신기를 구현하는 방법으로는 IIR 디지탈 필터, 계수기(counter)방법, DFT방법 및 FFT방법 등이 제안되어 왔다.[2]

DTMF 신호를 검출하는데, 음성신호 처리분야에서 널리 이용되고 있는 PARCOR 분석 방법을 적용하였다. 이 방법은 디지탈적으로 구현하기가 용이하며, 지금까지 제안된 어느 방법보다도 음성의 digit simulation에 강하다. 또한 원래 8KHz로 표본화된 DTMF 신호를 검출할 때는 4KHz로 재표본화하므로 다중화 효율을 2배 높일 수 있다.

Abstract

The following methods are proposed for implementing digital dual tone multi-frequency (DTMF) receiver: using infinite impulse response (IIR) digital filters, period-counting algorithm, discrete Fourier transform (DFT), and fastFourier transform (FFT) [2].

The PARCOR (Partial Correlation) analysis method which has been widly used in the speech signal processing area is applied to the dual tone multi-frequency (DTMF) signal detection. This method is easy to implement digitally and stronger to digit simulation of speech than any other methods proposed up to date. Since sampling rate of 4 KHz is used in the DTMF receiver for the detection of input DTMF signal originally sampled at 8 KHz, it effects two times higher multiplexing efficiency.

I. Introduction

Digital devices superiority over analog devices has been recognized with advances in large scale integration (LSI) and digital signal processing technologies. Digitalization has the effect of making a device small and of simple manufacture and maintenance. Especially for dual tone multi-frequency (DTMF) receiver which must satisfy severe specifications, it is attractive that aging degradations become negligible when it is digitalized.

In the following, this paper describes results of studies on all-digital DTMF receiver for push-button service using partial correlation (PARCOR) analysis method, which has been successfully applied to speech signal processing area.

Pushbutton service is a voice (or in-band)

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接受日字 : 1986年 8月 5日

frequency signaling system in which any one of 16 digits may be transmitted by simultaneously sending two tones. The frequency of one of the tones may be either 697,770,852, or 941 Hz (called the low frequency group) and the frequency of the other tone 1209, 1336, 1447, or 1633 Hz (called the high frequency group). The receiver must tolerate frequency shifts in the transmitter, operate over a wide dynamic range in the presence of noise, and be insensitive to speech (digit simulation).

II. Digital DTMF Receiver Algorithm Using PARCOR Analysis

1. PARCOR Analysis [5]-[6]

If input signal is a pure sinusoidal signal

$$f(nT) = a_m \sin(2\pi f nT), \tag{1}$$

its autocorrelation function is

$$v(nT) = a_m/2 \ \cos(2\pi f nT). \tag{2}$$

From this result we can obtain PARCOR (Partial Correlation) coefficients

$$k_1 = \frac{v(T)}{v(0)} = \cos(2\pi fT), \ |k_1| \leq 1, \tag{3}$$

$$k_2 = \frac{v(0)v(2T) - v(T)^2}{v(0)^2 - v(T)^2} = \frac{\cos(4\pi fT) - \cos^2(2\pi fT)}{1 - \cos^2(2\pi fT)} = -1 \tag{4}$$

And the normalized residual energy can be written as

$$en = (1 - k_1{}^2)(1 - k_2{}^2) = 0. \tag{5}$$

That is, a pure sinusoidal signal can be differentiated from the speech signal using $k_2 \simeq -1$ or en$\simeq$0 and its frequency is identified using the distribution of k [3].

Now we describe the method how we can obtain the PARCOR coefficients k_1 and k_2 for each group signaling frequency from the input DTMF signal. Since PARCOR analysis is done block by block like finite impulse response (FIR) digital filtering, not sample by sample like infinite impulse response (IIR) digital filtering, it requires a buffer to save a block of data (frame).

At first, the input DTMF signal must be expanded to the linear code used in the DTMF receiver and saved in the buffer. According to μ-255 encoding law, μ-255 pulse coded modulation (PCM) corresponds to 14 bits linear code. Then the data in the buffer are weighted by 64-point Hamming window, and its power spectrum is obtained by fast Fourier transform (FFT) method. Low group signaling frequency band is selected around between 11 (about 698 Hz) and 18 (about 1143 Hz) and high group signaling frequency band around between 19 (about 1206 Hz) and 29 (about 1841 Hz). Therefore dial tone band is rejected as a result since the frequencies of dial tone are 350 Hz and 440 Hz in our country. And their autocorrelation coefficients v(0), v(T) and v(2T) are obtained by discrete cosine transform (DCT) That is, the inverse FFT of power spectrum is by definition autocorrelation coefficients

$$v(nT) = 1/N \sum_{k=0}^{N-1} |X(k)|^2 e^{j2\pi kn/N}, \quad 0 \leq n \leq (N-1), \tag{6}$$

and if we use the symmetry property of power spectrum

$$|X(k)|^2 = |X(N-k)|^2, \tag{7}$$

we can obtain the DCT formula

$$v(nT) = 1/N \ \{ |X(0)|^2 + 2 \sum_{k=1}^{N/2} |X(k)|^2 \cos(2\pi kn/N) + (-1)^n |X(N/2)|^2 \}, \quad 0 \leq n \leq (N-1), \tag{8}$$

where $|X(k)|^2$ is the power spectrum in each selected signaling frequency band. And finally, we can get PARCOR coefficients by using equations (3) and (4).

Since it is required that the DTMF receiver should respond in between 24 ms to 40 ms as shown in Table 1, the required frame length must be below 20 ms (80 samples for 4 KHz sampling rate). At most cases, we select one of exact powers of two as the number of data in a frame when we use FFT algorithm, we choose 64 as the frame length since it is the maximum power of two below 80. The flow chart of implementation is shown in Fig. 1.

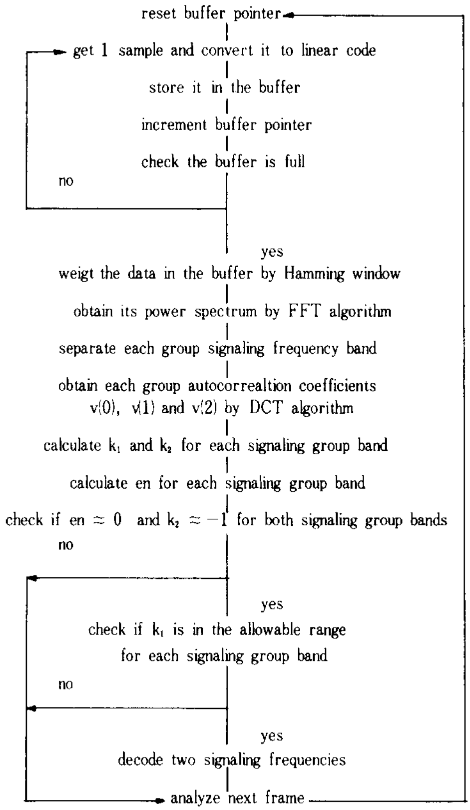

Fig. 1. Flow Chart of DTMF receiver implementation.

2. Sampling Frequency

PARCOR coefficients are obtained from the autocorrelation coefficients as we saw in section 2.1. It means that the autocorrelation coefficients must be high sensitive to the signaling frequencies for the PARCOR coefficients to be so.

For a pure sinusoidal signal $a_m \sin(2\pi ft)$, the normalized autocorrelation coefficient is $\Phi(t)=\cos(2\pi ft)$ and thus we can see that it is most sensitive to the frequency f satisfying $2\pi ft=\pi/2$ where its derivative is zero. Since the signaling frequencies are around 1 KHz and below 2 KHz, sampling frequency is $f_0=1/t=4$ KHz if we use average frequency $f=f_{avg}=1$ KHz.

Since input DTMF signal was originally sampled at the frequency of 8 KHz and it is down-sampled to 4KHz sampling rate for detection in the DTMF receiver, the 2 to 4 KHz band is aliased to the 2 KHz to 0 Hz band by the 2 to 1 reduction in sampling rate. It effects two times higher multiplexing efficiency and prevents speech signals from being detected to be DTMF signals (digit simulation) more preferably due to the aliasing effect.

3. Computation of Power Spectra

The main use of the fast Fourier transform (FFT) is to compute power spectra. Frequently the local library package requires an exact power of two for the number of data points, and usually you do not have that many. And even if you do have exactly the right number, remember that the Fourier expansion implies that the function is periodic; if there is a significant difference between the starting values and the ending values, there will be a discontinuity in the function and you will see the effect that discontinuity in the spectrum.

Conventional wisdom these days suggests first removing the mean of the data and then using the set of weights

$$w(k) = 1/2 \{1+\cos(\pi k/N1)\}, 0 \leq k \leq (N1-1) \quad (9)$$

on the data, where N1 is about 10% of the data that you have. This operation produces a taper at the starting end. A similar (but reversed) sequence of weights at the other end produces a second taper. Then you can pad out the rest with zeros. Thus the flat part of the window is more than about 80% of the data, and there are no sharp corners in the data going into the FFT. Whether you pad at both ends or only one end of the data is a matter of phase shift only.

If you have exactly the right amount of data and merely face a discontinuity around ends, a taper of about 10% of the data to smooth the transition is suitable [1].

In this paper, Hamming window is used to taper out the transitions around ends, that is, for N even,

$$w(k) = 0.54+0.46\cos\{2\pi k/(N-1)\},$$
$$-(N-1)/2 \leq k \leq (N-1)/2, \quad (10)$$

and for N odd,

$$w(k) = 0.54+0.46\cos\{2\pi(2k+1)/2(N-1)\},$$
$$-(N/2) \leq k \leq (N/2-1). \quad (11)$$

III. DTMF Receiver Requirements

For use in PABX or central office applications, there are requirements that a DTMF receiver must meet. These are shown in Tables 1 and 2 and have been influenced greatly by CCITT recommendations [4].

While most of the criteria in Tables 1 and 2 can easily be met individually, it becomes more difficult to meet them collectively [2].

Table 1. System requirements.

Nominal Frequencies Low Frequency Group High Frequency Group	 697, 770, 852, 941 Hz 1209, 1336, 1477, 1633 Hz
Allowable Frequency Deviation from the Nominal value	−2.0%~2.0%
Unallowable Frequency Deviation from the Nominal value	≤ −2.8% ≥ 2.8
Allowable Signal Level	−3dBm~−24dBm
Unallowable Signal Level	≤ −29dBm
Allowable Level Difference between Two Tones (Twist)	± 15 dB
Responce Time	24~40ms

Table 2. Timing requirements

DTMF receiver must

1) detect proper DTMF signal if their duration is 40ms or longer.
2) not detect two pairs whose duration is 20 ms or less.
3) detect as two separate DTMF signals any separation of valid tone pairs which is 35ms or greater.
4) not detect as two distinct DTMF signals if the separation of valid tone pairs is 5 ms or less.

IV. Simulation and Results

We investigated by computer simulation that the designed receiver met the requirements of CCITT recommendations, by displaying the results of each stage on the IBM PC/XT graphic display for varying input parameters: low group signaling frequency, high group signaling frequency, signal level, dial tone level, gaussian noise level, frequency deviation, twist, etc.

In Table 3 are shown results of k_1 parameter distribution obtained by computer simulation, which are the average values over the 32 frames for each signaling frequency deviated by ±2.4% from its nominal frequency without gaussian noise.

Table 3. Distribution of k_1 parameters.

Signaling Frequency	Allowable Range of k_1
697Hz	0.4340472~0.4803848
770Hz	0.3254744~0.3800102
852Hz	0.1986982~0.2613614
941Hz	0.05732099~0.1274292
1209Hz	−0.3650256~−0.2789645
1336Hz	−0.5458455~−0.4590502
1477Hz	−0.7196066~−0.6383145
1633Hz	−0.8690504~−0.8017178

Fig. 2 to 7 show the output results for the following input parameters: low group signaling frequency = 697 Hz, high group signaling frequency = 1209 Hz, low group signal level = −10 dBm, high group signal level = −10 dBm, dial tone level = −15 dBm, gaussian noise level = −30 dBm (SNR =23 dB), frequency deviation = 0%, simulation interval = 0 ~ 500 ms, data interval = 100 ~400 ms, and so on. The amplitudes are normalized to their maximum value in Fig. 2.

Fig. 2. Input DTMF signal (horizontal=16ms, vertical=0.1v)

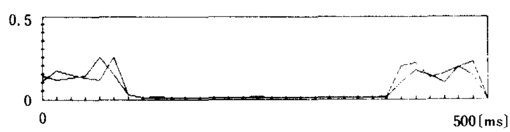

Fig. 3. Normalized residual energy (horizontal=16ms, vertical=0.05)

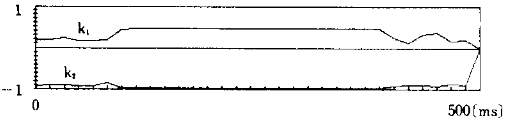

Fig. 4. k_1 and k_2 for low group frequency (horizontal=16ms, vertical=0. 1).

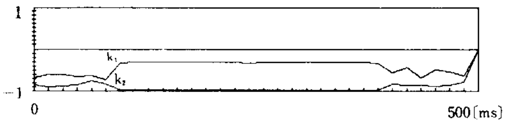

Fig. 5. k_1 and k_2 for high group Frequency (horizontal=16ms, vertical=0. 1)

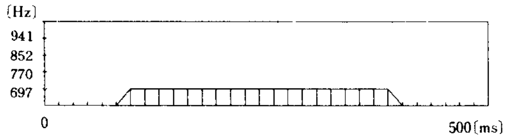

Fig. 6. Detected result for low group frequency (horizontal=16ms)

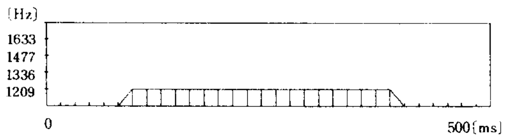

Fig. 7'. Detected result for high group frequency (horizontal=16ms)

V. Conclusions

In this paper, we have studied an all-digital DTMF receiver using PARCOR analysis method. According to simulation results, the designed receiver meets all the requirements of CCITT recommendations and it is expected that these results can be used without any modifications in implementing with available general-purpose microprocessors or digital signal processors (DSP) large scale integration (LSI).

This method is easy to implement digitally and stronger to digit simulation of speech than any other methods proposed up to now. And it effects two times higher multiplexing efficiency.

References

[1] R.W. Hamming, "Digital filters," *Prentice-Hall, Inc.*, Englewood Cliffs, New Jersey, 1977.

[2] P.B. Ha and S.G. Ann, "On implementing the digital DTMF receiver using DSP LSI," *JASK*, vol. 5, no. 2, June 1986.

[3] A. Ichikawa and K. Nakada, "A New digital multi-frequency signaling receiver," *Trans. of IECE*, vol. J62-B, no. 9, 1979.

[4] *CCITT Study Group XI*, "Report on the meeting held in tokyo from 3 to 12 July 1967," COM. XI-no, 121-E, p. 47, July 1967.

[5] L.R. Rabiner and R.W. Schafter, "Digital processing of speech singal," *Prentice-Hall, Inc.*, Englewood Cliffs, New Jersey, 1978.

[6] J.D. Markel and A.H. Gray, Jr., "Linear prediction of speech," *Spring-Verlag*, Berlin, Heidelberg, New York, 1976.