PARCOR분석방법에 의한 R 2 MF 수신기 구현에 관한 연구

On Implementing the Digital R 2 MF Receiver using PARCOR Analysis Method

* 하 판 봉(Ha, P. B.)
** 안 수 길(Ann, S. G.)

요 약

디지탈 R 2 MF 수신가를 구현하는 방법으로는 디지탈 필터, 계수기(counter) 방법, DFT 방법 즉 Goertzel 알고리즘[8], 및 FFT방법 등이 제안되어 왔다.

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

ABSTRACT

The following methods have been proposed for implementing R2 multi-frequency (MF) receiver digitally: using digital filters, period-counting algorithm, discrete Fourier transform (DFT) or Goertzel algorithm[8], and fast Fourier transform (FFT).

The PARCOR (Partial Correlation) analysis method which has been widly used and successfully applied in the speech signal processing area is applied to the R2 multi-frequency (MF) signal detection. This method is easy to implement digitally and stronger to digit simulation of speech than any other

* 서울대학교 전자공학과 통신연구실
** 서울대학교 전자공학과 교수

methods proposed up to date. Since sampling rate of 4 KHz is used in this R2 MF receiver for the detection of input R2 MF 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 R2 multi-frequency (MF) 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 R2 MF receiver using partial correlation (PARCOR) analysis method, which has been successfully applied in the speech signal processing area.

Dual tone multi-frequency (DTMF) signaling is a voice frequency signaling system used mainly between the customer or coin telephone set and the local office. The signaling format is known as a 2-out-of-8 code in which any one of sixteen signaling 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 group) and the frequency of the other tone may be either 1209, 1336, 1477 or 1633 Hz (called the high group)[2]-[4].

On the contrary, R2 multi-frequency (MF) signaling is generally used to establish inter-office connections. The signaling format is known as a 2-out-of-6 code in which any one of fifteen signaling digits may be transmitted by simultaneously sending two of six tones. The frequencies of the tones are 540, 660, 780, 900, 1020 and 1140 Hz for forward group or 1380, 1500, 1620, 1740, 1860 and 1980 Hz for backward group[5]-[8].

II. R2 RECEIVER REQUIREMENTS

For use in PABX or central office applications, there are requirements that a R2 MF receiver must meet. These are shown in Table 1 and have been influenced greatly by CCITT recommendations[5].

While most of the criteria in Table 1 can easily be met individually, it becomes more difficult to meet them collectively.

Table 1. System Requirements

Nominal Frequencies Forward Frequency Group Backward Frequency Group	 1380, 1500, 1620, 1740, 1860, 1980 540, 660, 780, 900, 1020, 1140 Hz
Allowable Frequency Deviation from the Nominal value	-10 ~ 10 Hz
Allowable Signal Level	-5 ~ -35 dBm
Allowable Level Difference between Two Tones (Twist)	± 5 dB (adjacent freq) ± 7 dB (non-adjacent freq)
Recognition Time	13 ~ 27 ms
Operate and Release Time	70 ms
Signal Interrupt Time	7 ms

III. DIGITAL R2 MF RECEIVER ALGORITHM USING PARCOR ANALYSIS

1. PARCOR Analysis[6], [7]

If input signal is described as a k-th order all-pole model, its system function can be written as

$$H_K(Z) = \frac{G}{1+\sum_{i=1}^{k} \alpha_i(K)\ Z^{-i}}, \quad (Z=e^{j2\pi fT}) \tag{1}$$

where $\{\alpha_i^{(K)}\}$, i=1, 2, ..., k are LPC (Linear Predictive Coding) coefficients obtained from matrix equation

$$\begin{bmatrix} r(0) & r(1) & \cdots r(K) \\ r(1) & r(2) & \cdots r(K-1) \\ \vdots & \vdots & \vdots \quad \vdots \\ r(K) & r(K-1) & \cdots r \end{bmatrix} \begin{bmatrix} 1 \\ \alpha_1^{(K)} \\ \vdots \\ \alpha_K^{(K)} \end{bmatrix} = \begin{bmatrix} G \\ 0 \\ \vdots \\ 0 \end{bmatrix} \tag{2}$$

$\{r(i)\}$, i=1, 2, ..., k are autocorrelation coefficients of the input signal. Many fast recursive algorithms to solve this form of matrix equations have been proposed[9]. The poles of $H_K(z)$, $\{z_i = r_i \exp(\theta_i)\}$, i=1, 2, ..., k are obtained by solving the following k-th order LPC polynomial

$$A_K(z) = 1 + \sum_{i=1}^{K} \alpha_i^{(K)} z^{-i} \tag{3}$$

The real-time root factoring algorithm of the LPC polynomial was also proposed[10]. Then, since the poles occur in complex conjugate pairs, the frequencies $\{f_i\}$ and bandwidths $\{b_i\}$ i=1, 2, ..., k/2 of the poles $\{z_i\}$ can be obtained by the following relations

$$f_i = \theta_i / (2\pi T) \ [Hz], \tag{4}$$

$$b_i = \ln(1/r_i)/(\pi T) \ [Hz], \tag{5}$$

Π

where T is sampling period. In general, for the input signal consisting of two pure sinusoidal waves like R2 MF signal, the system function can be described as a fourth order model (k=4). The system function can be implenented by 4 sections of lattice filters in cascade in terms of PARCOR (Partial Correlation) coefficients derived from the LPC coefficients $\{\alpha_i^{(K)}\}$, i=1, 2, ..., k. Then, the R2 MF signal can be differentiated from the normalized residual energy

$$e_n = \prod_{i=1}^{K} (1 - K_i^2) \simeq 0 \tag{6}$$

Now we describe the detection algorithm. Since LPC or 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 R2 MF signal taken alternately must be expanded to the linear code used in the R2 MF receiver and saved in the buffer. According to μ-255 encoding law, μ-255 pulse coded modulation (PCM) corresponds to 14-bit linear code. Then the data in the buffer are weighted by 64-point (16 ms) window. Kaiser window are used in this simulation since less fluctuating frequency outputs are obtained with the Kaiser window ($\beta = 10$) than any other windows (Hamming, Hanning, rectangular, triangular, and Kaiser window). The Kaiser window is of the form

$$W_K(n) = \frac{I_0\left(\beta\sqrt{1-[2n/(N-1)]^2}\right)}{I_0(\beta)},$$

$$|n| \leq (N-1)/2 \qquad (7)$$

where β is a constant that specifies a frequency response tradeoff between the peak height of the side lobe ripples and the width or energy of the main lobe and Io(x) is the modified zeroth-order Bessel function. Then LPC analysis is done on the windowed data. The flow chart of implementation is shown in Fig. 1.

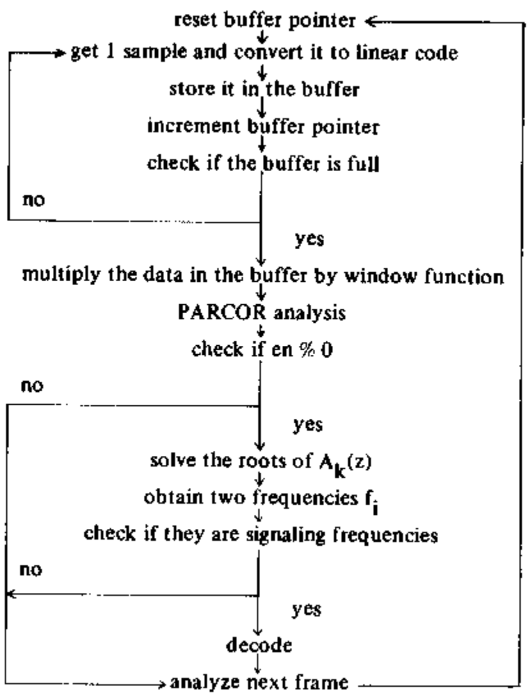

Fig. 1. Flow Chart of R2 MF Receiver

2. Sampling Frequency

Now, we find out the sampling rate of R2 MF signal at which the R2 MF receiver operates optimally. LPC or PARCOR coefficients are obtained from the autocorrelation coefficients as we saw in section 3.1. It means that the autocorrelation coefficients must be high-sensitive to the signaling frequencies for the R2 MF receiver to be so.

For a pure sinusoidal signal $a_m \sin(2\pi ft)$, the normalized autocorrelation function is $\Phi(t)=\cos(2\pi ft)$. So, we can see that it is most sensitive to the frequency f satisfying 2 ft= $\pi/2$, where its derivative is zero. The optimum sampling frequency is obtained as $f_s = 1/t=4$ KHz if we use average signaling frequency f=f avg =1 KHz since the signaling frequencies are around 1 KHz and below 2 KHz,.

Since R2 MF signal was originally sampled at the rate of 8 KHz and it is down-sampled to 4KHz sampling rate for detection in the R2 MF 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 R2 MF signals (digit simulation) more preferably due to the aliasing effect.

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: two signaling frequencies, signal levels, gaussian noise level, frequency deviations, twist, etc. And we were able to see that the designed receiver operated properly for over signal-to-noise ratio (SNR) of about 21 dB.

Fig. 2 to 4 show the output results for the following input parameters: signaling frequencies (1020 Hz, 1140 Hz), signal levels (-10 dBm, -10

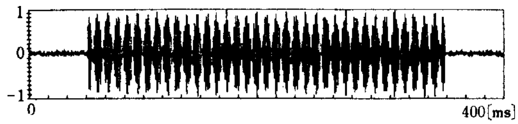

Fig. 2. Input Signal (1020 Hz + 1140 Hz)
(horizontal = 16 ms, vertical = 0.1 v)

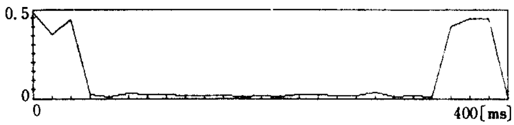

Fig. 3. Normalized Residual Energy
(horizontal = 16 ms, vertical = 0.05)

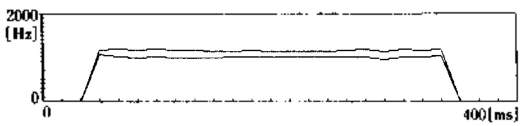

Fig. 4. Detected Result
(horizontal = 16 ms, vertical = 100 Hz)

dBm), gaussian noise level = -30 dBm (SNR = 23 dB), frequency deviations (0 %, 0 %), simulation interval = 0 ~ 400 ms, data interval = 50 ~ 350 ms, and so on. The amplitudes are normalized to their maximum value in Fig. 2.

V. CONCLUSIONS

In this paper, we have studied an all-digital R2 MF receiver using PARCOR analysis method. According to simulation results, the designed receiver met all the requirements of CCITT recommendations and operated properly for the signal-to-noise ratio (SNR) of over 21 dB or so. It is expected that these results can be used without any modifications in implementing with floating-point digital signal processors (DSP) large scale integration (LSI) available on the markets.

This method is easy to implement digitally and stronger to digit simulation of speech than any other methods proposed up to now, and effects two times higher multiplexing efficiency due to 2 to 1 down-sampling.

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] P.B. Ha and S.G. Ann, "On Implementing the Digital DTMF Receiver Using PARCOR Analysis Method", KIEE, to be published, 1986.

[4] A. Ichikawa and K. Nakada, "A New Digital Multi-Frequency Signaling Receiver", Trans. of IECE, Vol. J62-B, No. 9, 1979.

[5] CCITT Recommendations, Orange Books Vol. VI, ITU, Geneva, 350-368.

[6] L.R. Rabiner and R.W. Schafer, Digital Processing of Speech Signal, Prentice-Hall, Inc., Englewood Cliffs, New Jersey, 1978.

[7] J.D. Markel and A.H. Gray, Jr., Linear Prediction of Speech, Spring-Verlag, Berlin, Heidelberg, New York, 1976.

[8] Ivan Koval and George Gara, "Digital MF Receiver Using Discrete Fourier Transform", IEEE Trans. on Communications, Vol. COM-21, No. 12, Dec. 1973.

[9] George Carayannis, Nicholas Kalouptsidis, and Dimitris G. Manolakis, "Fast Recursive Algorithms for a Class of Linear Equations", IEEE Trans. on Acoustics, Speech, and Signal Processing, Vol. ASSP-30, No. 2, April 1982.

[10] Periagaram K. Rajasekaran and George R. Doddington, "Real-Time Factoring of the Linear Prediction Polynomial of Speech Signals", Proceedings of the International Conference on Digital Signal Processing, Florence, Italy, Sep. 1984.