On Implementing the Digital R2 MF Receiver using PARCOR Analysis Method

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 widely 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
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 (PARC) 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 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>
<thead>
<tr>
<th>Table 1. System Requirements</th>
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<tr>
<td>Nominal Frequencies</td>
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<td>Forward Frequency Group</td>
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<td>Allowable Frequency</td>
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<td>Deviation from the Nominal value</td>
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<td>Allowable Signal Level</td>
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<td>Allowable Level Difference</td>
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<td>between Two Tones (Twist)</td>
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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

\[ H_k(z) = \frac{G}{1 + \sum_{i=1}^{K} a_i^{(K)} z^{-i}} \]

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

\[
\begin{bmatrix}
    r(0) & r(1) & \cdots & r(K) \\
r(1) & r(2) & \cdots & r(K-1) \\
\vdots & \vdots & \ddots & \vdots \\
r(K) & r(K-1) & \cdots & r
\end{bmatrix}
\begin{bmatrix}
    1 \\
a_1^{(K)} \\
\vdots \\
a_k^{(K)}
\end{bmatrix}
= \begin{bmatrix}
    G \\
0 \\
\vdots \\
0
\end{bmatrix}
\]

\[
(2)
\]

\( \{r(i)\} \), \( i=1, 2, ..., k \) 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

\[
\Lambda_k(z) = 1 + \sum_{i=1}^{K} a_i^{(K)} z^{-i}
\]

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

\[
f_i = \frac{\theta_i}{2\pi} \left( \frac{1}{T} \right) \left( \frac{1}{Hz} \right),
\]

\[
b_i = \ln(1/r_i)/\left( \pi T \right) \left( \frac{1}{Hz} \right),
\]

\[
II
\]

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 implemented by 4 sections of lattice filters in cascade in terms of PARCOR (Partial Correlation) coefficients derived from the LPC coefficients \( \{a_i^{(K)}\}, i=1, 2, ..., k \). Then, the R2 MF signal can be differentiated from the normalized residual energy

\[
en - \frac{1}{T} \left( \frac{1}{K} \right) \left( \frac{1}{K} \right) = 0
\]

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 \( (\beta \sim 10) \) than any other windows (Hamming, Hanning, rectangular, triangular, and Kaiser window). The Kaiser window is of the form
where $\beta$ 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 $I_0(\cdot)$ 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.

For a pure sinusoidal signal $u_m \sin(2\pi ft)$, 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 $2ft = \pi/2$, where its derivative is zero. The optimum sampling frequency is obtained as $f_s = 1/t=4$ KHz if we use average signaling frequency $f=\text{avg}=1$ 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 4 KHz 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 affects 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 dBm, -10 dBm),
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


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.