Method for Dual-Tone Multiple Frequency Detection Using Constrained Adaptive Second-Order Notch Filters

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Abstract - **An original method for dual-tone multiple frequency (DTMF) detection is proposed. It is based on constrained adaptive second-order notch filters. The method is computationally efficient and reliable and meets the International Telecommunications Union (ITU) Recommendations. The talk-off tests and tests in the presence of noise show the robustness of the detection method. This method can replace the more complex realizations based on Goertzel algorithm, or those using classical filter design methods.**

Keywords – **Adaptive filters, Adaptive signal processing, Multiple frequency detection**

I. INTRODUCTION

Dual-tone multiple frequency (DTMF) signaling is a standard in telecommunications systems [1]. A DTMF codec incorporates an encoder that translates digit information into dual-tone signals, as well as a decoder that detects incoming DTMF tone signals. A DTMF signal consists of two superimposed sinusoidal waveforms with frequencies chosen from a set of eight standardized frequencies. These frequencies should be generated and detected according to the ITU Recommendations Q.23 and Q.24.

The task to detect DTMF tones in an incoming signal and to convert them into actual digits is certainly more complex than the encoding process. As the incoming signal is a sum of two frequencies, which uniquely determine the transmitted digit, we need a technique for extracting spectral information from the input signal.

First option is to design a set of narrow-band filters for the eight allowed frequencies and other eight for the second harmonics in order to be able to discriminate DTMF tones from possible speech and music. Here the classical method for digital filter design can be used, or alternatively the Goertzel algorithm could be applied, the latter proved to be more efficient [2]. Yet both methods need the computational power for calculating the parameters of 16 digital filters.

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Here we propose a new method for the purpose of DTMF detection. Bearing in mind that at every particular time the incoming signal is composed of two frequencies the task is to identify these frequencies. We design our DTMF detector on the basis of adaptive digital second-order notch filters.

II. METHOD

A realization based on a second order lattice circuit has been used - Fig.1 [3]. Using this circuit it becomes possible to implement a second order notch/bandpass section - Fig. 2. This realization is very efficient for the present application because it is possible to control independently the notch frequency and the bandwidth.

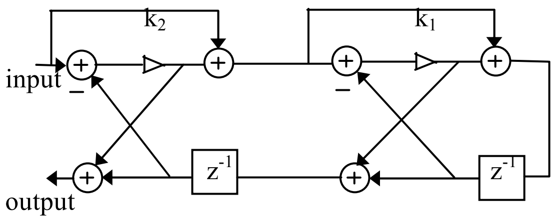

Fig. 1. Second order lattice circuit realizing all-pass function A(z)

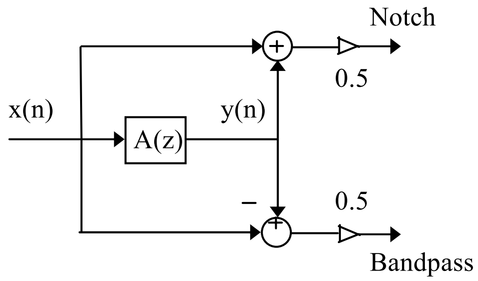

Fig. 2. Second order notch/bandpass section

Thus if the all-pass function A(z) is

$$A(z) = \frac{k_2 + k_1(1+k_2)z^{-1} + z^{-2}}{1 + k_1(1+k_2)z^{-1} + k_2 z^{-2}} \tag{1}$$

then k_1 controls the notch frequency ω_0 while k_2 is related to the bandwidth (BW) via

$$k_1 = -\cos \omega_0 \tag{2}$$

$$k_2 = \frac{1-\tan(BW/2)}{1+\tan(BW/2)}. \qquad (3)$$

Using the basic structure shown in Fig. 2, the final arrangement of our system for extracting spectral information is shown in Fig. 3.

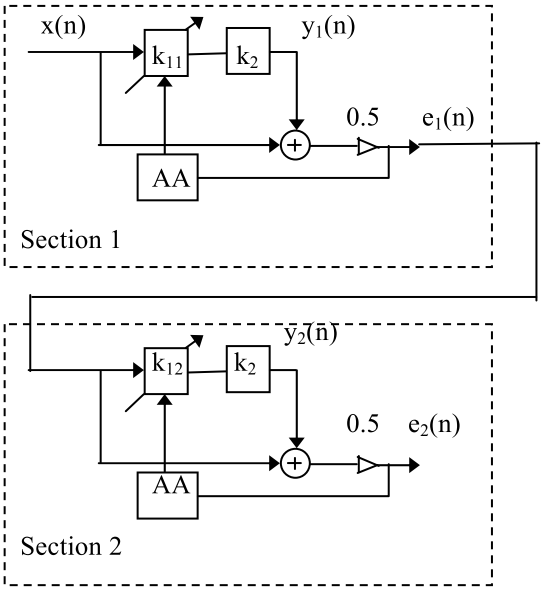

Fig. 3. Adaptive system for extracting spectral information

The system works in the following manner: each section identifies one of the two frequencies using an appropriate adaptive algorithm (AA). As shown in Fig. 3 we propose to update only the coefficients k_{11} and k_{12}, while k_2 is a priori determined from equation (3). Applying this constraint we fix BW and make the distance from the pole to the unity-circle constant. Thus we can reduce considerably the number of computations and can guarantee the stability of the adaptive structure. Here we introduce the normalized least mean square (NLMS) algorithm for adjusting the filter coefficients as shown below:

$$e_i(n) = 0.5[e_{i-1}(n) + y_i(n)] \qquad (4)$$

for $i = 1,2$ and $e_0(n) = x(n)$

$$k_{1i}(n+1) = k_{1i}(n) - \mu \frac{e_i(n)\, y'_i(n)}{[y'_2(n)]^2} \qquad (5)$$

$$y'_i = \frac{d\, y'_i(n)}{d\, k_{1i}(n)}$$

for $i=1,2$

where $e_i(n)$ is the error signal, μ is the step size and $y'_i(n)$ is the derivative of $y_i(n)$ with respect to the coefficient subject to adaptation.

The application of a normalized adaptive algorithm makes the chose of the step size at some extent independent of the amplitude of the input signal x(n). For our experiments we use a step $\mu=0.01$.

TABLE I
COEFFICIENT-TO-DIGIT MAPPING FOR DTMF DETECTOR BASED ON ADAPTIVE DIGITAL SECOND-ORDER NOTCH FILTERS ACCORDING TO ITU BANDWIDTH SPECIFICATIONS

Digit	Section 1	Section 2
0	$-0.4660<k_{11}<-0.5297$	$-0.7215<k_{12}<-0.7563$
1	$-0.5547<k_{11}<-0.6087$	$-0.8437<k_{12}<-0.8637$
2	$-0.4660<k_{11}<-0.5297$	$-0.8437<k_{12}<-0.8637$
3	$-0.3618<k_{11}<-0.4362$	$-0.8437<k_{12}<-0.8637$
4	$-0.5547<k_{11}<-0.6087$	$-0.8104<k_{12}<-0.8345$
5	$-0.4660<k_{11}<-0.5297$	$-0.8104<k_{12}<-0.8345$
6	$-0.3618<k_{11}<-0.4362$	$-0.8104<k_{12}<-0.8345$
7	$-0.5547<k_{11}<-0.6087$	$-0.7696<k_{12}<-0.7986$
8	$-0.4660<k_{11}<-0.5297$	$-0.7696<k_{12}<-0.7986$
9	$-0.3618<k_{11}<-0.4362$	$-0.7696<k_{12}<-0.7986$

Next task is to map the values of k_{11} and k_{12} to the corresponding digits in order to identify them (see Table I). Here we round the values of coefficients with a precision of four digits after the decimal point. That turns out to be quite sufficient for the proper work of our decoder and also suggests robustness in implementations with finite precision.

Using the adaptive system for extracting spectral information (ASESI) and Table I for the mapping of the coefficients to the relevant digit we design our system for DTMF detection as depicted in Fig.4.

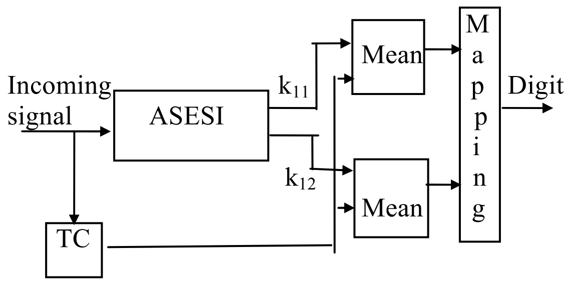

Fig. 4. A system for DTMF detection using adaptive notch filters

The system works in the following manner: ASESI collects continuously spectral information as at its output we have the coefficients k_{11} and k_{12} which are related directly to the angular frequency ω_0. We employ a threshold control (TC) unit. Its role is twofold. First, to detect the presence of a DTMF signal and second, to control the units “Mean” which take the mean value of the coefficients over a predetermined number of observations. Finally the unit “Mapping” identify the transmitted digit according to Table I.

III. EXAMPLE

An example is given in order to clarify the basic operations of our system. The incoming signal is a mixture of DTMF

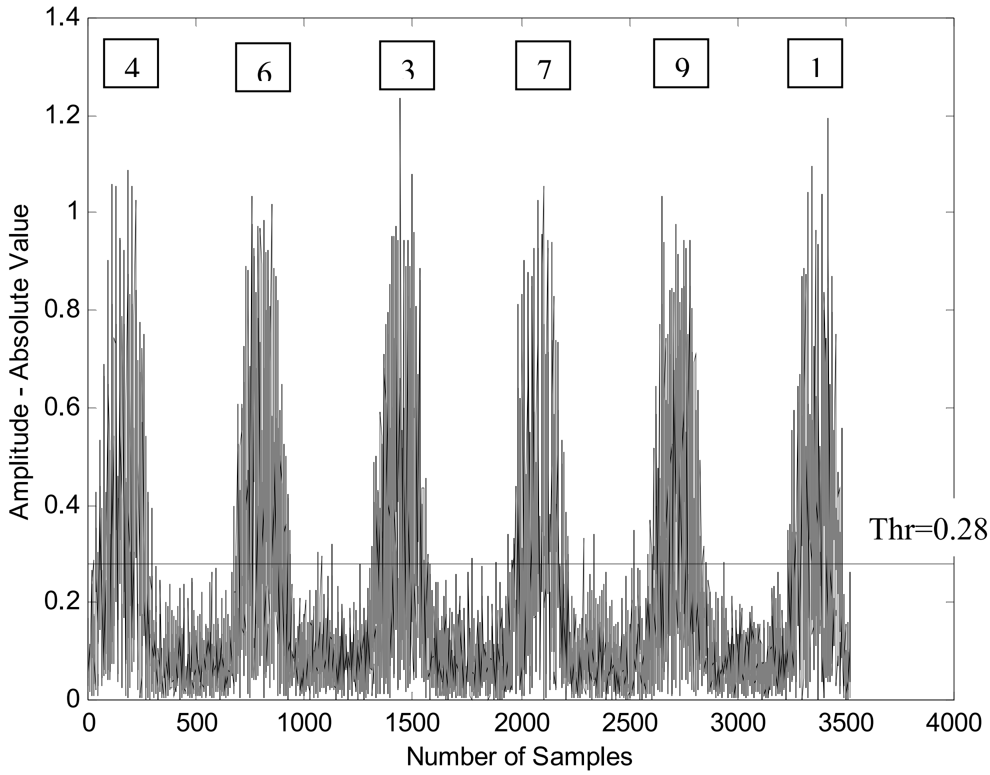

Fig. 5. A mixture of DTMF signal and White Noise, a threshold is used for end-points detection of the different digits

signal and White Noise (-20 dBV). DTMF signal corresponds to the number 463 791 and complies with the ITU Recommendations (signal duration min 40 ms and pause duration min 40 ms). Here the threshold is set to 0.28 to detect the presence of a DTMF signal. We use a "sliding" window spanning 100 samples and take the mean value of the signal defined by the window. This mean value is compared with the threshold. We use the absolute value of the signal (Fig. 5.). Once "beginning" and "end" points of DTMF signal are determined, TC sets the time interval during which the "Mean" units operate. In this case using 8 kHz sampling frequency and 40 ms signal duration we have 320 samples for each transmitted digit. We let ASESI to adapt for the first 270 samples and take the mean value of the coefficient over the rest 50 samples when ASESI is reached steady state. Taking the mean value of the coefficients provides robustness in noise environment.

IV. EXPERIMENTS

Here we test how our method addresses the basic issues related to DTMF detection namely:

- capability of correctly identifying different digits;
- robustness in a noise environment;
- speech rejection.

The capability of identifying different digits is related closely to the tracking of the two sinusoidal waveforms presented in the incoming DTMF signal. This is illustrated with an example shown in Fig. 6. The DTMF signal corresponds to a six-digit telephone number (532 184) composed according to the Recommendations of ITU. Apparently the adaptive system depicted in Fig. 3 is able to track the changing frequencies of the sinusoidal waveforms. Next step is to map the values of the coefficients to the relevant digit (see Table I). We tested our method in noise environment of –24 dBV AWGN (MITEL Specification) [4]. The system was able to detect correctly all 1000 tone bursts. Finally we set an experiment where our detector was exposed to speech samples (previously recorded speech in .wav format). We tested our detector on 1000 .wav files (about 35 minutes of speech) and it did not respond a single time.

V. CONCLUSIONS

We propose a computationally efficient and reliable method for DTMF detection in telecommunications applications according to ITU Recommendations. The method is implemented and tested in software but a realization on the basis of a signal processor is possible too. The presented method can replace the more complex realizations based on Goertzel algorithm, or those using classical filter design methods.

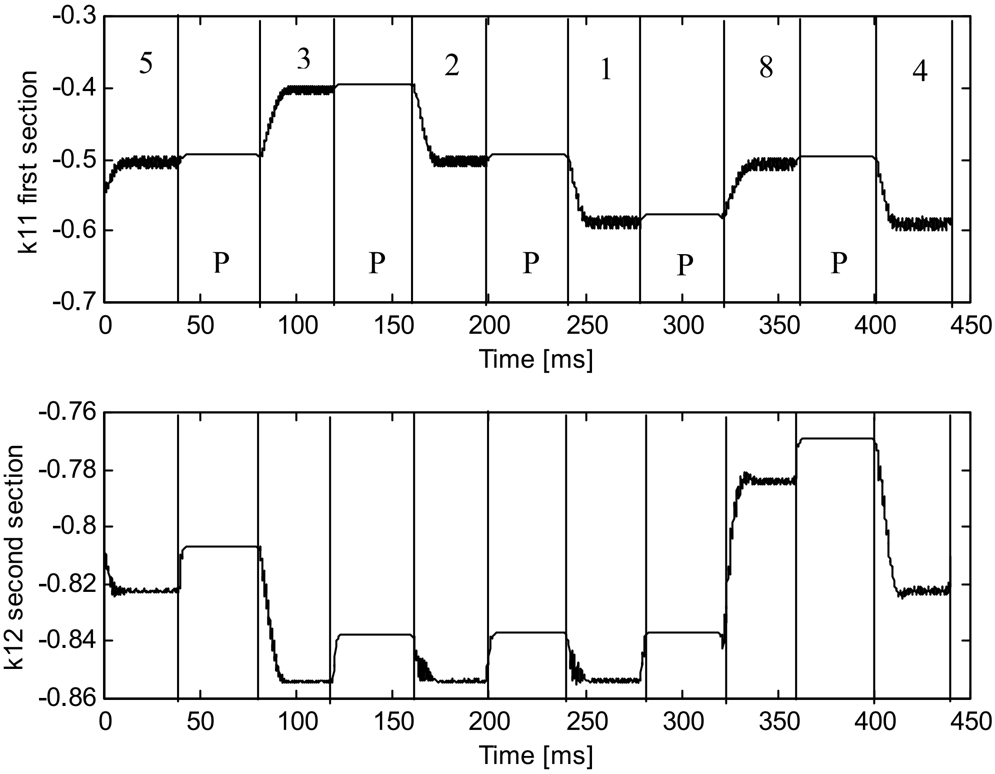

Fig. 6. Trajectories of coefficients k_{11} and k_{12} (the DTMF signal corresponds to a six-digit telephone number (532 184, P - pause) composed according to the Recommendations of ITU)

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