Using Advanced Signal Processing Methods for DTMF Detection

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***Abstract* – In this paper we explore the possibility of using time-frequency representations (namely the Short Time Fourier Transform) for the process of decoding Dual-tone Multi-Frequency (DTMF) signals. The paper shows the advantages of using time-frequency representation based detection in the case of DTMF signals covered with noise due to improper telecommunication channels.**

***Keywords:* time-frequency representations, non-stationary signals, DTMF**

I. INTRODUCTION

Dual-tone multi-frequency (DTMF) signaling is a standard in telecommunication systems [1]. DTMF detection is used to detect DTMF signals in the presence of speech and dialing tone pulses. Besides being used to set up regular calls on a telephone line, DTMF detection is suitable for computer applications such as voice mail and electronic mail, and telephone control features such as conference calling and call forwarding.

A DTMF signal consists of the sum of two sinusoids with frequencies taken from two mutually exclusive groups. These frequencies were chosen to prevent any harmonics from being incorrectly detected by the receiver as some other DTMF frequency. Each pair of tones contains one frequency of the low group (697 Hz, 770 Hz, 852 Hz, 941 Hz) and one frequency of the high group (1209 Hz, 1336 Hz, 1477Hz) and represents a unique symbol. The frequencies allocated to the push-buttons of the telephone pad are shown in Fig 1. It has to be noted that the A, B, C, and D keys are usually not present on a regular telephone keypad.

Analog DTMF detection is done using bandpass filter banks with center frequencies at the DTMF signal

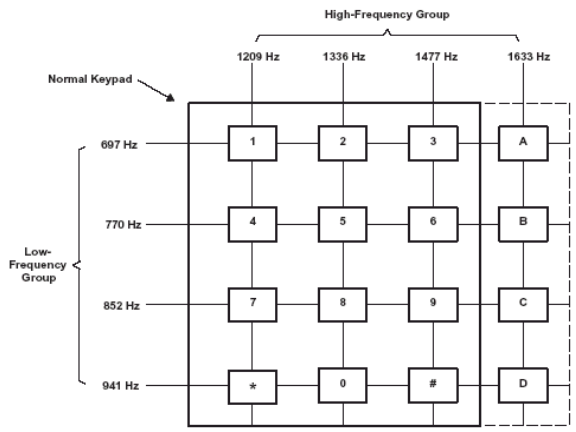

Fig 1: Phone pad with DTMF frequencies

frequencies. Analog receivers have wide tolerances to compensate for distortion caused by aging transmitters, variations in keying characteristics, and transmission line distortion. These distortions compound the problem of digital DTMF detection [2].

In digital switching systems is desirable to treat all signals uniformly, bringing all signals through A/D converters and switching them through the PCM system. Therefore the need for digital DTMF detection is justifiable to avoid the costs of hardware and D/A conversion needed to use analog detectors. With the constant advances in VLSI driving DSP costs downward, it is economically useful to replace analog detectors with their digital counterparts which are more reliable, maintenance cost effective, and spatially minimal.

Several techniques for digital DTMF detection have been used, but most designers have settled on either digital filtering or discrete Fourier transform (DFT). In digital filtering, DTMF signals are passed through digital bandpass filters centered at the signaling frequencies (shades of analog). The power at each frequency is then measured repeatedly to detect the DTMF tones. A DSP then interprets and translates them for the proper switching. The DFT also measures signal power at the signaling frequencies but has the additional need to check for signals of some minimum duration. This will help to ensure robustness toward speech and noise. The actual DFT is based on the Goertzel algorithm.

The DTMF signaling frequencies are very closely spaced. It is obvious that the bandwidth of the filter used for detection must be narrow enough to avoid any bleeding of adjacent frequencies. An even more limited bandwidth is introduced when one considers that some of today's DTMF signal generators (phones, modems, etc.) are poorly made to such an extent that the actual frequencies generated are not as shown in Fig. 1. On the other hand, the design must be efficient enough to run in real time. So we have a trade-off between the number of DFT points that must be taken to give proper bandwidth (and accurate spectrum) compared with the amount of time it takes to evaluate that number of points.

Since the DTMF signal is actually a multicomponent nonstationary signal, time-frequency methods can be applied to properly detect these signals. For non-stationary signals found in many important applications, the right tools to apply are time-frequency representations (TFRs), which measure how the frequency content of a signal changes over time. TFRs like the short-time Fourier transform (STFT), the wavelet transform, and the Wigner distribution are very useful in solving different problems that occur in fields like geophysics; data compression; image coding and analysis; communications; speech and acoustic signal processing, and medical signal processing. The time-frequency plane is a rich feature space for analyzing the signal's attributes. A TFR of a multi-component non-stationary signal consists of sets of ridges, the orientations and widths of which characterize the signal. For example, once computed, time-frequency images can be processed using edge detection and other image processing algorithms to automatically determine the ridge parameters. The STFT and continuous wavelet transform have been suggested for the first, image generation step of the feature extraction procedure. In this paper, we will explore the advantages of using TFRs in DTMF detection.

Bilinear time-frequency distributions, offer a wide range of methods designed for the analysis of non stationary signals. Nevertheless, a critical point of these methods is their readability, which means both a good concentration of the signal components and no misleading interference terms. Some efforts have been made recently in that direction, and in particular a general methodology referred to as reassignment. TFRs are two dimensional functions of time t and frequency f that indicate how the frequency content of a signal x changes over time. The simplest TFR is based on the Short-Time Frequency Transform (STFT):

$$S_x(t,f) = \int x(\tau)\omega^*(\tau - t)e^{-j2\pi f\tau}d\tau \tag{1}$$

Another useful TFR is the spectrogram, the squared magnitude of the STFT. The classical time-frequency resolution tradeoff of the spectrogram, which is controlled by the analysis window w, has prompted the development of more advanced bilinear TFRs, including the Wigner distribution:

$$W_x(t,f) = \int x\left(t + \frac{\tau}{2}\right)x^*\left(t - \frac{\tau}{2}\right)e^{-j2\pi f\tau}d\tau \tag{2}$$

This TFR can be interpreted as a short-time Fourier transform with the window matched to the signal. While the Wigner distribution is highly concentrated, due to its nonlinearity it generates cross-components and is very sensitive to noise. The spectrogram and Wigner distribution both belong to Cohen's class of TFRs. Because the high amount of interference terms generated by these nonlinear transforms we will not use them in our example.

II. ANALYZING DTMF SIGNALS USING TIME-FREQUENCY REPRESENTATIONS

The problem addressed in this paper is how, given a digital single-channel data stream, to detect the presence of valid DTMF tones. The algorithm must be capable of accurately determining:

- which of the eight DTMF frequencies are present,
- the relative signal power at each of the frequencies,
- the duration that these signals are present. These signal characteristics must then be compared to industry standard criteria to determine whether a valid DTMF tone is present

Using time-frequency representations, these problems can be solved by calculating the local maxima of the energy contents of the signals in the time-frequency plane. First we generated a DTMF signal having minimum tone duration of 220 milliseconds, and minimum tone-pause duration of 20 milliseconds. There is no additional synchronization constraint on maintaining a fixed spacing or duration of the tones. The signal contained the frequencies of all phone numbers from 1 to 9 and the final number was 0.

Then the Short Time Fourier Representation of this signal was calculated. This representation is shown in Fig. 2. The multi-component spectral content of the signal is obvious in this figure. When the useful signal is multicomponent or it is perturbed by additive noise the estimation problem is more complicated and the algorithms already reported generally don't work. This is the reason why we propose here a method based on the use of time-frequency representations. These distributions have two useful properties:

- They have a very good concentration around the curve of the instantaneous frequency of the analyzed signal, [3];

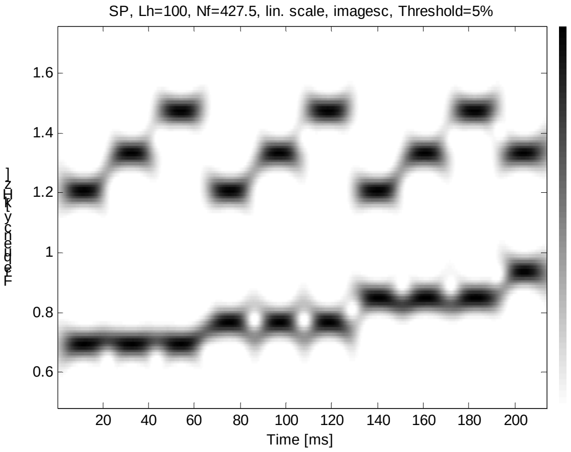

Fig. 2: STFT of a DTMF signal

- They realize a diffusion of the perturbation noise's power in the time-frequency plane.

Computing the time-frequency representation of the signal with noise, we can obtain a good estimation of the ridges of the time-frequency representation of the original signal. Projecting these ridges on the time-frequency plane we obtain a good estimation of the frequency content of the analyzed signal.

The frequency localization of the representation presented in Fig. 2 is not the best because the resolution limitations in the time-frequency plane due to the Heisenberg-Gabor uncertainty principle. One way to improve the representation is to use the reassignment method. The method of reassignment is a technique for sharpening a time-frequency representation by mapping the data to time-frequency coordinates that are nearer to the true region of support of the analyzed signal. In the case of the spectrogram or the short-time Fourier transform, the method of reassignment sharpens blurry time-frequency data by relocating the data according to local estimates of instantaneous frequency and group delay. This mapping to reassigned time-frequency coordinates is very precise for signals that are separable in time and frequency with respect to the analysis window. In Fig. 3 is presented the reassigned spectrogram of the DTMF signal. Although the results are much better, in the case of a real DTMF detector the increase of computing time needed to obtain such a representation may be impractical. More useful is the direct detection of the ridges of the simple STFT representation.

The role of the time-frequency representation in our estimation method is to spread the noise in the time-frequency plane and to locate the ridges of the time-frequency representation of the useful signal.

For frequency estimation of multicomponent signals or of signals perturbed by noise, the linear time-frequency representations are more useful due to the presence of the interference terms of bilinear time-frequency representations.

Then the time-frequency representation can be processed using mathematical morphology elements. The image obtained above is converted in binary form.

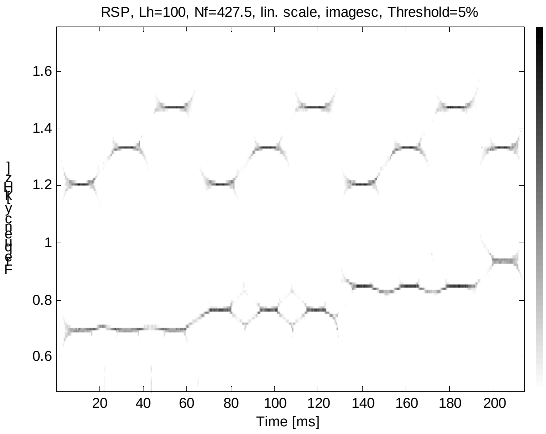

Fig 3: Reassigned spectrogram of DTMF signal

The Short Time Fourier Representation of a DTMF signal covered with gaussian noise was also calculated. This representation is shown in Fig. 4.

This realizes a thresholding of the time-frequency representation image. The effect of the use of this operation is a denoising of the image of the time-frequency representation. Applying the dilation operator on the image and calculating the skeleton of the last image, an estimation of the signal's spectrum is being obtained. The conversion in binary form realizes a denoising of the time-frequency distribution. The skeleton produces the ridges estimation. Comparing the results obtained by the ridges estimation with the known spectral content that belong to a DTMF signal, the original phone number can be obtained. The result of the mathematical morphology operations is presented in Fig. 5.

Although the method is computation-intensive compared to classic methods, it has the advantage of being able to correctly detect phone numbers even when the signal is deeply covered with noise.

Our DTMF detection algorithm exhibits excellent voice rejection performance while correctly detecting the dual-tones in signal to noise ratio levels as low as -3 dB. The results obtained with this method show that we can detect tones with SNR's as low as -10 dB.

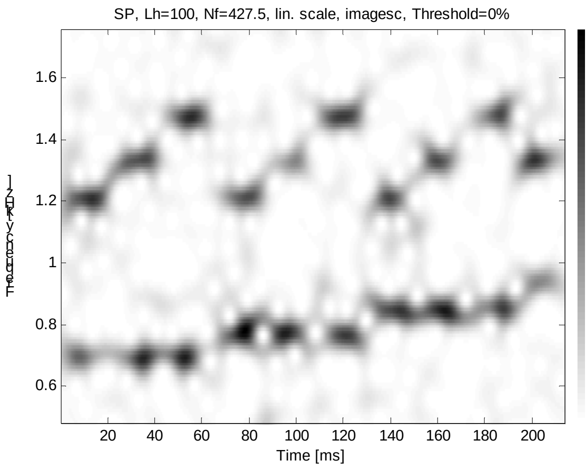

Fig. 4: STFT of a DTMF signal covered with noise

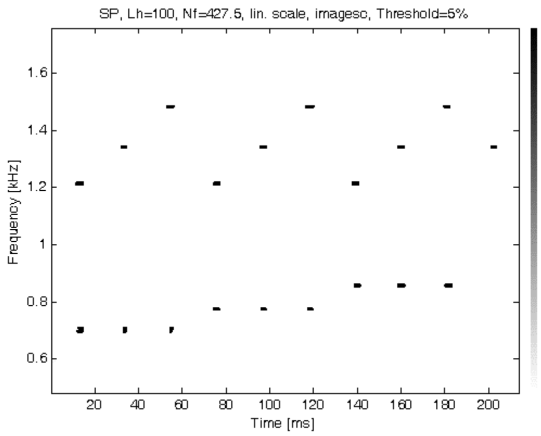

Fig 5: Result of the morphological operations applied to the time-frequency representation of the DTMF signal

III. CONCLUSIONS

The method shows great promise for analysis of non stationary DTMF signals with the goal of extracting instantaneous attributes. The DTMF signals were chosen as an example. The method may be applied to detect many types of encoded signals. Further analysis is still required and the computational speed will be a real issue given that one needs to first compute the time-frequency representation before instantaneous attributes can be extracted based on a TFR, and all should be done in real-time. Using more advanced time-frequency methods, like for example wavelet-based representations, the results should be even better. The estimation method proposed in this paper can be used in a lot of other applications. Some of them, like radar, sonar, or seismic signal processing are already recognized as applications of the time-frequency representations theory. This method can be used in measurements, too.

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