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DTMF and CLIP decoding in a noisy area using adaptive approach

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Abstract

Multi-frequency signals are greatly used in telecommunication fields. Signaling and speech are such an example of multi-frequency signals exchanging through the telecommunication networks. Extracting the frequencies embedded in these signals is very useful for a lot of operations: like filtering, decoding, compressing….We propose in this paper adaptive technique to process in real time multi-frequency signals and extracting the frequencies that they contain.

Keywords: DTMF, CLIP, Noise, Adaptive, Real Time

1. Introduction

Detecting frequencies of a multi-frequency signal is very used in telecommunication field. In fact, the information about a telephonic call as called number, calling number, establishment of a communication, compressing … is defined by decoding the frequencies superposed in the signal. DTMF (dual-tone multi-frequency) signaling, CLIP (Calling Identification Presentation) signal and voice are examples of multifrequency signals that the telephone network is designed to carry.
DTMF signalling [3] is typically generated by a telephon set or possibly by a PBX (Private branch telephone exchange). DTMF digits may be consumed by entities such as gateways or application servers in the IP network, or by entities such as telephone switches or IVRs (Interactive Voice Response) in the circuit switched network. In fact, DTMF signaling converts decimal digits and the symbols '*' and '#' into sounds that share enough essential characteristics with voice to easily traverse circuits designed for voice.
In the DTMF signaling, there are, as illustrated by figure 1, four frequencies associated with the four rows, and three frequencies associated with the three columns. Each key specifies two frequencies. The DTMF signal for a key is the sum of two sinusoidal waves, one at each frequency. For example, when we push the key ‘8’ of keypad’s phone a sound with two tones, one at 852hz and the other at 1336hz will be generated.
The CLIP signal is transmitted during the ringing phase of a telephone call and contains information related to the caller as phone number, date and time of the call. The information that the PSTN (Public Switched Telephone Network) is transmitting to the terminal equipment is in a series of 8-bit words each bounded by a start bit (0) and a stop bit (1). Frequency Shift Keying modulation is used to transmit bits over a phone line. It assigns to “0” the frequency 2125hz while “1” is represented by the tone 1275hz.
Recently, a great number of algorithms for identifying the frequencies of a multi-frequency signal have been introduced into signal processing field. These algorithms compute first dsp (power spectrum density) of signals to extract the frequency with high energy. GOERTZEL, LEVINSON, and BURG[14,9] algorithms are such examples. These ones are widely used to compute the signal dsp and overcome mismatch in sampling that appears in DFT method [9]. But they are not very efficient in noisy area [11,13]. Also these algorithms can process only stationary signal [13]. While in reality, signals are non stationary due to the impact of the transmission medium noise.
In this paper we introduce adaptive techniques to decode both DTMF and CLIP signal. We propose the RLS algorithm to process in real time these multi-frequency signals and extract the frequencies that they contain. In fact, even if RLS method is complex and not very fast as LMS algorithm [24], it is

stable and does not require preliminary the choice scrupulously of the adaptive parameter which introduces the divergence of the LMS method particularly in presence of noises to decode those signals, we model them with a 4th adaptive all-pole filter. Then we deploy its coefficients that we compute using RLS algorithm to extract signal frequencies. The RLS method is presented below.

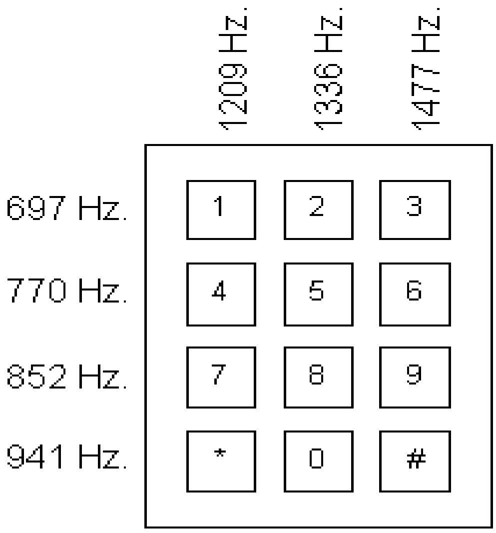

Figure 1: Allocation of frequencies for a push-button keypad

2. RLS Algorithm

The RLS (Recursive Least Squares) filtering [22,24] permits to determine the best predictor coefficients ($1, a_1, \ldots. a_p$) that minimizes the prediction error e(n) defined by equation (2).

$$\tilde{x}(n) = -\sum_{k=0}^{p} a_k x(n-k) \tag{1}$$

$$e(n) = x(n) - \tilde{x}(n) \tag{2}$$

x(n) is the desired signal at time n, while $\tilde{x}(n)$ is the N-length data vector used in prediction of x(n). The RLS problem requires finding the set of predictor coefficients $A_N(n)=[a1,a2,\ldots\ldots a_N]^T$ such that the cumulative squared error measure

$$E_m = \sum_{i=1}^{n} \lambda^{n-i} e(i)^2 \tag{3}$$

is minimized. The parameter λ, where $0 < \lambda < 1$, is a data weighting factor that may be used to weight recent data more heavily in the RLS computations. A value of λ in the range $0.95 < \lambda < 0.9995$ has proved to be effective in tracking local nonstationaries.

The best coefficient vector is obtained by setting the gradient of equation (3) to zero.

$$\frac{\partial E_m}{\partial a_k} = 0 \tag{4}$$

Using equation (2), it is easy to show that solution of (4) for $A_N(n)$ gives equation.

$$\left[\sum_{k=0}^{n} \lambda^{n-k} . x_N(k) . x_N(k)^T\right] . A^T = \sum_{k=0}^{n} \lambda^{n-k} . x_N(k) . x(k) \tag{5}$$

This equation may be resolved using RLS algorithm .

In other hand, spectrum S(Z) of the signal x(n) is given by equation (6):

$$S(Z) = |X(Z^{-1})|^2 = \frac{E_m}{|1 + a_1 Z^{-1} + a_2 Z^{-2} + \dots + a_p Z^{-p}|} = \frac{E_m}{|A(Z^{-1})|^2} \quad (6)$$

If we replace Z by $e^{2j\Pi f}$ in the equation (6) , $S(e^{2j\Pi f})$ will have peaks at the frequencies composites of the signal. Those frequencies may also be computed using the polynomials bellow.

$$Q(Z) = A_p(Z) + Z^{-(p+1)} A_p(Z^{-1}) \quad (7)$$

$$R(Z) = A_p(Z) - Z^{-(p+1)} A_p(Z^{-1}) \quad (8)$$

The roots of these polynomials called LSP (Line Spectrum Pair) are the frequencies superposed in the signal [6]. We find in the reference [10] some algorithms for computing the roots of polynomials Q(Z) and R(Z).

3. Results and discussions

In this section, we decode the real CLIP signal depicted in figure 2 using the DSP ADSP210 [25]. The CLIP signal is sent between the first and second ring. In this case the ring itself signals the burst of data.

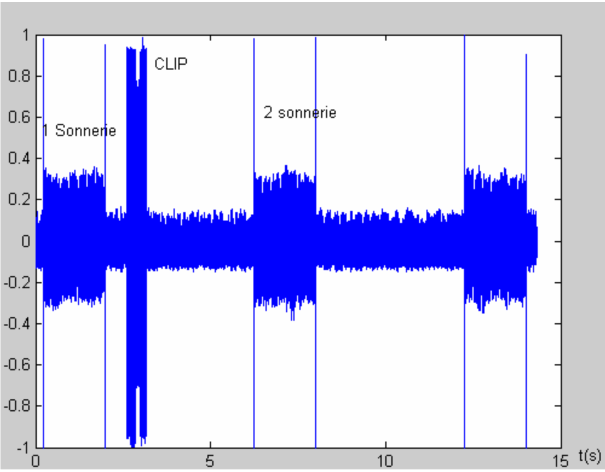

Figure2: Apparition of a Real CLIP signal

We take the received 24 KHz line value in accumulator. In fact, while the transmission speed is fixed, according to the V24 UIT recommendation, to 1200 bps the duration of a bit is 0.83 ms. By sampling the clip signal with 24 KHz, a bloc of 19 samples is available. The current bloc and the previous one affected with a scale factor are added. The process of the result permits to detect simultaneously the two bits send (figure 3) using a 4^{th} adaptive filter. Data weighting factor used is 0.995. It was determined experimentally.

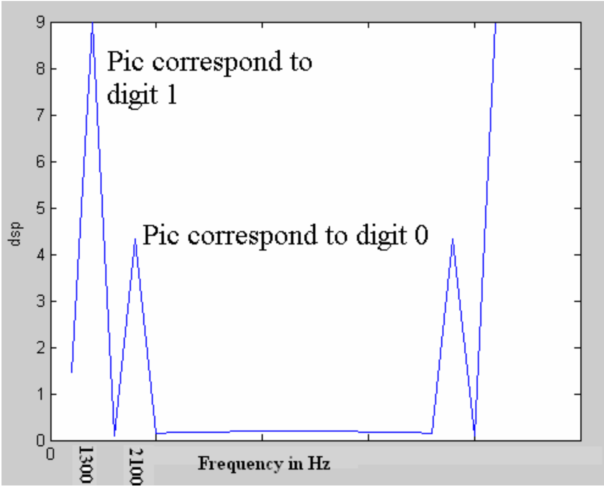

Figure 3: Decoding of the sequence 10

The same filter is also used to decode the DTMF signal 852+1336 Hz generated when we push the key 8 of the telephonic keypad. The signal generated is given as follow:

$$s(n) = a_1 \sin\left(\frac{2\pi n f_1}{Fe}\right) + a_2 \sin\left(\frac{2\pi n f_2}{Fe}\right) + \sigma^2 rand$$

Fe is sampling frequency. σ2 rand(1) is a MATLAB's function that simulates noise.
In Fact, when we push a key of keypad's phone, a DTMF signal with two frequencies is transmitted [1]. Frequencies are allocated to the various digits and symbols of a push-button keypads as shown in Figure 1.
Filter coefficients are computed using this method converge very fast to their optimal values (figure 4). Even if we decrease the ration S/B, the coefficients convergences are not influenced (figure 5). This shows the capability of RLS to decode multi-frequency signals in a noisy area.
Furthermore, roots of the polynomials Q(Z) and R(Z) established, coincide accurately with frequencies of th code 8 (table 1). The decoding time of a DTMF code is shown by table 2.

Table 1: Frequencies determined using RLS

Samples Number / S/R(db}	100		50		25	
	f1	f2	f1	f2	f1	f2
39.46	851.5 (0.06%)	1335.7 (0.02%)	850.5 (0.18%)	1336 (0.00%)	847.2 (0.56%)	1334.2 (0.13%)
20	850 (0.23%)	1336 (0.0%)	850.3 (0.2%)	1334.3 (0.13%)	849.7 (0.27%)	1338.3 (-0.17%)
10.34	850.5 (0.18%)	1337 (0.07%)	848.3 (0.43%)	1339.7 (-0.28%)	847.7 (0.5%)	1335.1 (0.07%)

Table 2: time required for the detection of a DTMF code

Number of samples	100	50	40	30
Decoding time (ms)	0.85	0.43	0.37	0.27

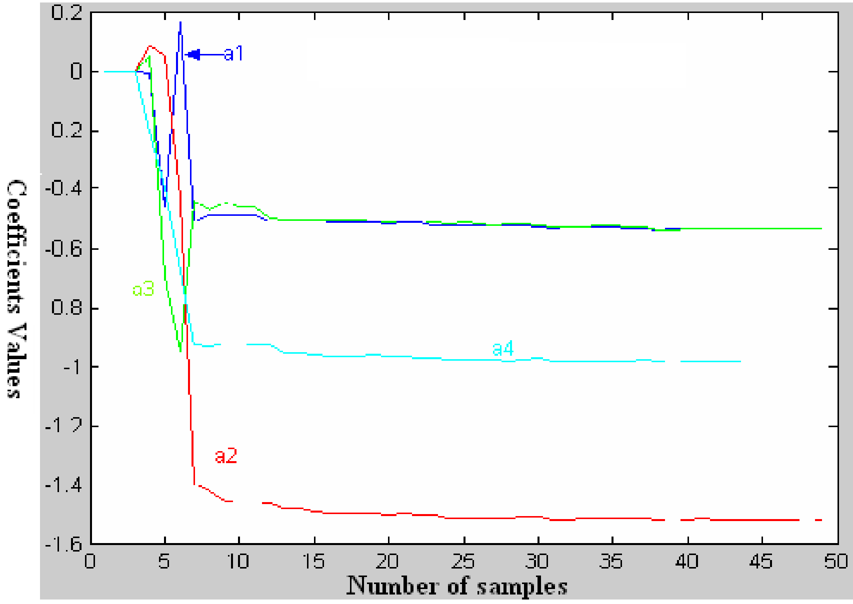

Figure 4: Convergence of adaptive filter coefficients to their optimal value

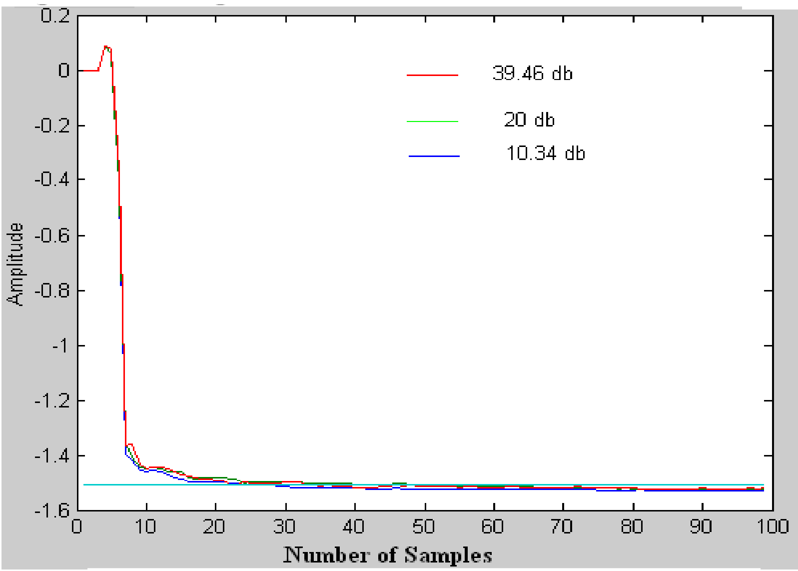

Figure 5: Convergence of coefficients a2

4. Conclusion

Adaptive filtering is an important basis for signal processing. We have shown in this paper that adaptive filter may be useful to decode multi-frequency signal in noisy area. We have illustrated their performance. We have also used RLS algorithm to decode simultaneously both DTMF and CLIP signal and we have noted that the RLS algorithm exhibits extremely fast convergence and permits to predict perfectly frequencies superposed in a multi-frequency signal in a real time.
The next step in our work will be the study of DTMF and CLIP events through H323 and SIP signaling features over IP networks.

5. Acknowledgements

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