Encoding and decoding of DTMF signal based on DSP

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***Abstract*—Dual-tone multi-frequency (DTMF) Signals are used in touch-tone telephones as well as many other areas. The design method of the traditional DTMF signal encoding and decoding circuit is more complex and the number of chip is more much that easy to cause the system unstable. The encoding and decoding of DTMF signal based on digital signal processor (DSP) not only can improve the real-time performance of DTMF signal processing, but also streamline the entire system module. In paper using TMS320vf5402 as the hardware platform and CCS as the software platform and programming in asm language, the encoding and decoding of DTMF signal are realized. The Goertzel optimization algorithm is used to reduce the computational complexity in the decoding part. The experimental results show that the encoding and decoding of DTMF signal based on DSP makes the signal processing system more simple and flexible.**

keywords—DTMF signal; DSP; encoding; decoding; Goertzel algorithm

I. DTMF INTRODUCTION

Dual-tone-multi-frequency (DTMF, also known as touch-tone) is the basis for voice communications control. Modern telephony uses DTMF to dial numbers, configure telephone exchanges (switchboards), and so on. Occasionally, simple floating codes are transmitted using DTMF - usually via a CB transceiver (27MHz). It is used to transfer information between radio transceivers, in voice mail applications, etc. DTMF signaling has also found applications requiring interactive control like in voice main, e-mail, telephone banking, and ATM machines[1-3].

DTMF is a tone composed of two sine waves of given frequencies. Frequency table as follow:

Table I Frequency table

		High Frequency Group			
		1209Hz	1336Hz	1447Hz	1633Hz
Low Frequency Group	697Hz	1	2	3	A
	770Hz	4	5	6	B
	852Hz	7	8	9	C
	941Hz	*	0	#	D

The table I resembles a matrix keyboard. The X and Y coordinates of each code give the two frequencies that the code is composed of. For example, Pressing "6" generates a 770-Hz tone form the low frequency group and a 1447-Hz tone from the high frequency group. Notice that the keypad has four keys ("A" through "D") that are not normally seen on most phones. They are used to configure phone exchanges or to perform other special functions.

II. ENCODING DTMF USING DSP

2.1 The encoding principle of DTMF signal

If the transmission function $H(z)$ of a system has no zero point and Only a pair of conjugate poles point on the unit circle , the unit impulse response of the system is the constant oscillating . That is, the sinusoidal signal is generated.

So, we have the following expressions:

$$H(z)=\frac{b_0}{1+a_1z^{-1}+a_2z^{-2}} \qquad (2\text{-}1)$$

where: $b_0 = A\sin w_0$, $a_1 = -2\cos w_0$, $a_2 = 1$

the conjugate poles point: $P_{1,2} = e^{\pm jw}$ (2-2)

The unit impulse response in Time domain:

$$h(n) = A\sin((n+1)w_0)\bullet u(n) \qquad (2\text{-}3)$$

So the second-order difference equation can be obtained as follows:

$$y(n) = -a_1y(n-1) - a_2y(n-2) + b_0\delta(n) \qquad (2\text{-}4)$$

simplied: $y(n) = 2\cos w_0 \bullet y(n-1) - y(n-2)$ (2-5) where : $y(-1)=0$ $y(-2)=-A\sin w_0$

$w_0 = 2\pi f_0 / f_s$

f_0 ----the input signal frequency

f_s ----the sample signal frequency

w_0 ----the normalized angular frequency

A ---- the Amplitude of sinusoidal signal

According to the Nyquist Theorem, the sampling rate must be more than twice the maximum frequency component of the signal being measured. Because of the telephone company's official digitalizing rate is 8k sample/sec, 8k sample/se is more than adequate for generating any valid pair of using the dsp-TMS3205402 the highest frequency involved is 1663 kHz. So we choose 8000Hz as the sampling frequency. T=50ms is taken as the standard duration of the DTMF, so each signal contains information of 800 points and the interval of 800 points.

Table. II shows the corresponding coefficient for the oscillator.

Table II The coefficients of the oscillator

f/Hz	a1	y(-1)	y(-2)/A
697	0.85382	0	-0.52047
770	0.82263	0	-0.56857
852	0.78433	0	-0.62033
941	0.73911	0	-0.67358
1209	0.58206	0	-0.81314
1336	0.49820	0	-0.86706
1477	0.39932	0	-0.91680
1633	0.28424	0	-0.95874

2.2 The simulation results of DTMF signal

Fig.1 and Fig. 2 show the simulation waveforms of the DTMF signal "1" in CCS and in Cool Edit Pro of profesional audio software. Fig.4 and Fig.5 show the simulation results of the DTMF signal "7820336".through compare the simulation results,it shows that the method of the DTMF encodiing is right.

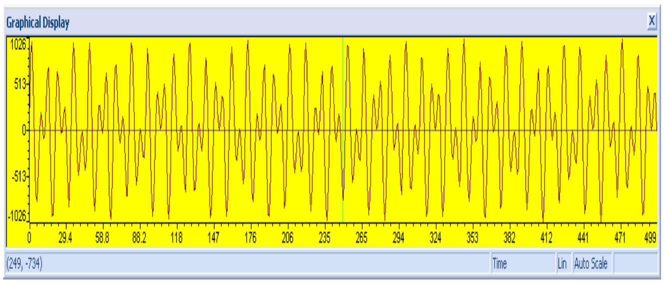

Fig.1 "1" DTMF Waveform

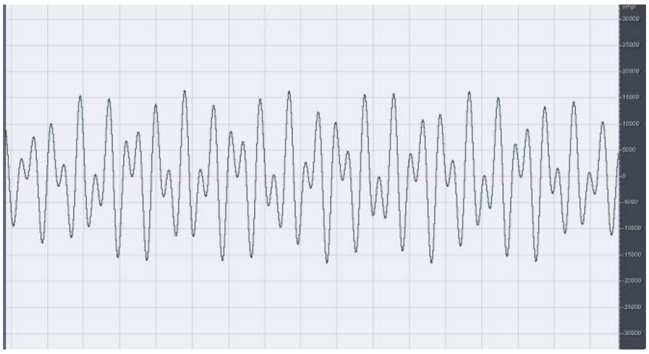

Fig. 2 "1" DTMF Waveform in Cool Edit Pro

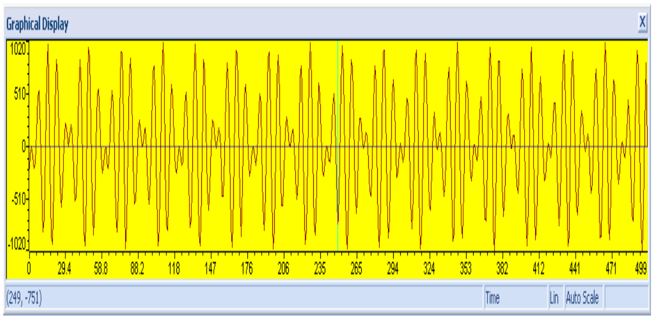

Fig .3 "*" DTMF Waveform

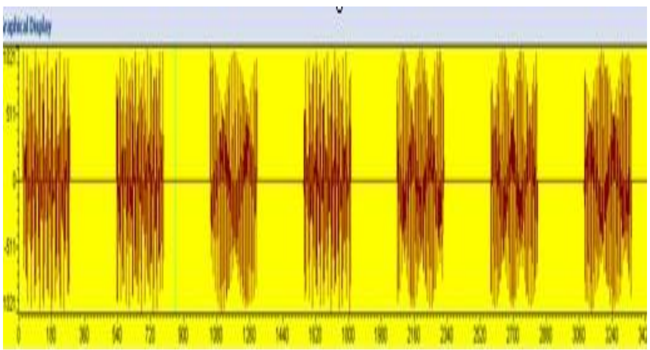

Fig. 4 "7820336" DTMF Waveform

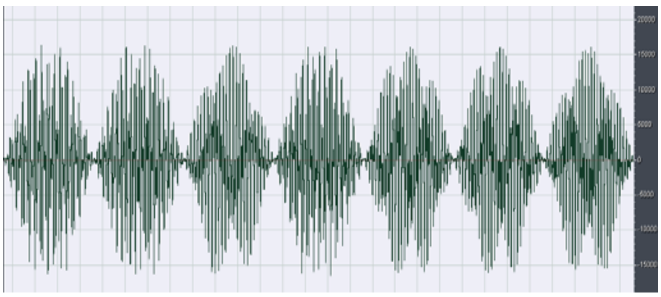

Fig .5 "7820336" DTMF Waveform in Cool Edit Pro

III. DECODING OF DTMF USING DSP

3.1 The decoding principle of DTMF

The method of DTMF Decoding is that finds the effective line and column frequency in the input signal. DFT and FFT can be used to calculate the frequency spectrum of digital signal, but Goertzel algorithm is faster than FFT when realizing DTMF decoding[4-5].

Goertzel algorithm is an IIR filter with two pole points. Fig. 6 shows its theorem diagram.

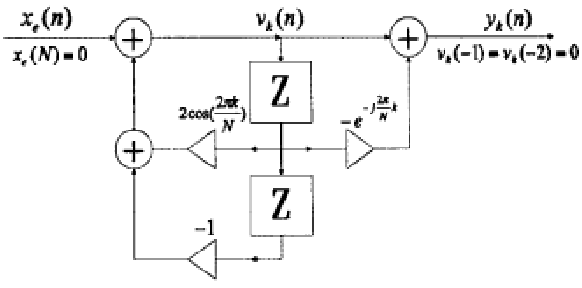

Fig. 6 Goertzel Algorithm principle block diagram

According to the above diagram, we can obtain the following equation:

$$V_k(n) = x(n) + 2\cos(\frac{2\pi k}{N})V_k(n-1) - V_k(n-2), 0 \le n \le N$$

$$X(k) = y_k(N) = V_k(N) - e^{j\frac{2\pi k}{N}} V_k(N-1) \qquad (2\text{-}6)$$

Because the input signal is real sequence when DTMF detecting ,the phase of the 8 line/colomn frequency does not need be detected and only need detect the square of the Amplitude.The internal variable VK (n) in the Goertzel algorithm is real sequence yet if the input signal is real sequence. So we can obtain the following equation:

$$|X(k)|^2 = |y_k(N)|^2 = V_k^2(n) + V_k^2(n-1) - 2\cos(\frac{2\pi k}{N})V_k(N-1)V_k(N-2)$$

(2-7)

where, N - the number of sampling points, N=205 by ITU.

The coefficients of the fundamental wave and second-order harmonic at N=205 are shown in table III:

Table III coefficient of the fundamental and its two-order harmonic at N=205

1 st Harmonics (N=205) fs=8kps			2 nd Harmonics (N=205) fs=8kps		
k	frequency (k/N)fs /Hz	coefficient cos(2pik/N)	k	frequency (k/N)fs/Hz	coefficient cos(2pik/N)
18	702	0.85162	35	1393	0.45886
20	780	0.81793	39	1552	0.34445
22	858	0.78115	43	1711	0.22470
24	936	0.74142	47	1871	0.10141
31	1210	0.58157	61	2428	-0.32974
34	1326	0.50442	67	2667	-0.50000
38	1483	0.39505	74	2945	-0.67606
42	1639	0.27972	82	3264	-0.83740

3.2 The simulation results of DTMF decoding

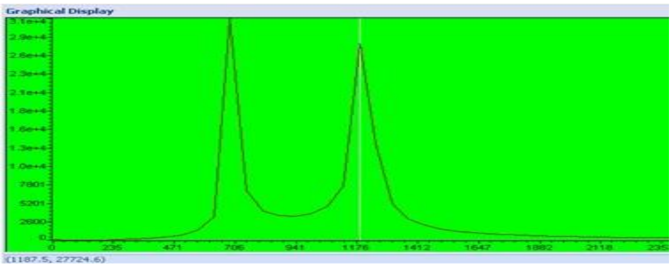

Fig. 7 "1" DTMF spectrum

Form Fig 7, we can see that the low frequency of the detected data "1" is almost 6697Hz and the high frequency is almost 1209Hz.

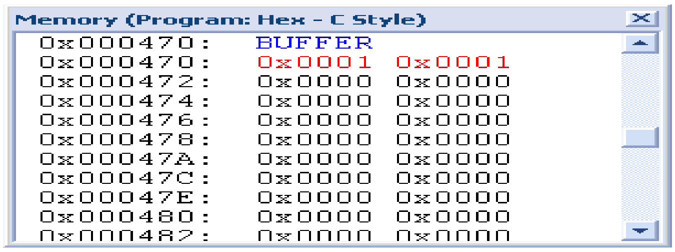

Fig.8 ROM buffer

Fig.8 shows that the two times final values of detected dial key "1" are stored in ROM. An effective detection can be obtained through getting two times consistency of the detected data

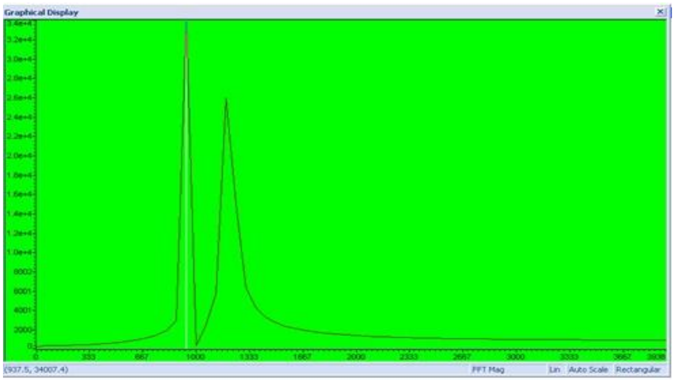

Fig .9 "*" DTMF spectrum

IV. CONCLUSION

In this paper, we clarify the concept and composition of DTMF from the perspective of complex system, analyze the Principles of the DTMF signal coding and decoding. The coding and decoding of the DTMF signal is achieved by the chip TMS320LF5402 replacing dedicated hardware chip. The whole process of the DTMF signal generation and decoding is observed with CCS development environment, which makes the coding and decoding of the DTMF signal accurate and intuitive. The test result shows the advantages of the design, namely, effective performance, low cost and flexible application, thus having broad application prospects.

REFERENCES

[1] ITUBlue Book. Recommendation Q. 24:Multi frequency Push button Signal Reception[M]. Geneva ,Switzerland:ITU ,1989.

[2] M. D. Felder ,J. C. Mason ,B. L. Evans. Efficient dual-tone multifrequency detection using the nonuniform discrete Fourier transform[J].IEEE Signal Processing Lett. :1998 ,5(7) :160 ～ 163.

[3] Ozaktas H M，Arikan O，Kutay M A，et al． Digital computation of the fractional Fourier transform［J］．IEEE Transactions on Signal processing，2006，44(9) : 2141-2150．

[4] R. Beck ,A. G. Dempster ,I. Kale. Finite-precision Goertzel filters used for signal tone detection [J]. IEEE Transactions on Circuits and Systems -II:Analog andDigital Signal Processing ,2001 ,48(6) :691～700.

[5] Wu Qingguo. On DTMF tone generation and detection using DSP implemention [J]. Computer Applications and Software. 2012,29(8):262-264.