Performance Evaluation and Design of DTMF Receiver with a Subset of 2^M Data Point

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Abstract: In this paper, we have analyzed the power spectra and evaluate the performance of DTMF receiver by using the quick Fourier transform(QFT) algorithm. The economical signals detection of dual-tone multifrequency (DTMF) receiver is an important factor when developing cost-effective telecommunication equipment. In experimental results, it shows that reducing memory waste and can process the real-time.

Keyword: quick Fourier transform(QFT), multifrequency, reducing memory, real time.

I. Introduction

The economical detection of DTMF(dual-tone multiple frequency) signals is critical importance in developing cost-effective telecommunications equipment. While many single-chip DTMF detectors currently exist, a multiple channel implementation is more appropriate in environments that have a concentration of many lines. In addition, a digital signal processor(DSP) implementation is often more desirable in applications such as switches where a single hardware resource may be shared among many channels and be used to perform many different signal processing functions at different times[1,2].

The DTMF receiver has independently each channel, and it informs the detected signal to the processor. The implementation of a DTMF receiver involves the detection of each of the signaling tones, validation of a correct tone pair, and timing to determine that a digit is present for the correct amount of time and with the correct spacing between tones. In addition, depending on the algorithm used to detect frequencies, it is sometimes necessary to perform additional tests to improve the performance of the decoder in the presence of speech. Current DSP technology allows several DTMF receivers to be implemented on a single device[3,4].

In order to detect the signal, there are several methods using IIR filter, FIR filter and PARCOR method. Recently, the method to detect the signal

use Goertzel algorithm and modified Goertzel(MG) algorithm. This method has an advantage that the calculation is easy and simply.

Also, when the input is real, the modified DFT(MDFT) pair have proposed, and its advantages have been examined[5].

In order to analyze the power spectra and decide the detected signal, the QFT method proposed is compared with other algorithms. Especially, it shows highly efficient in both real-time and memory waste.

Ⅱ. The QFT algorithm

The data received in DTMF receiver are transformed into the signal of the frequency domain by QFT. After that, it can happen the Gibbs phenomenon and the ripple. The Blackman window function and the zero-padding were used to reduce them. The definition of DFT is

$$S(k) = \sum_{n=0}^{N-1} s(n) e^{-j\frac{2\pi nk}{N}}, 0 \le k \le N-1 \quad (1)$$

When the input is real, the modified DFT(MDFT) pair is defined as

$$S_{k,i} = \sum_{n=0}^{N-1} x_{i-n} \cos(\frac{2\pi nk}{N}) \quad (2)$$

$$x_i = \frac{X_{0,i}}{N} + \frac{2}{N} \sum_{k=1}^{N/2-1} X_{k,i} \quad (3)$$

where N is the number of samples and assumed to be even[5].

The QFT algorithm can be easily modified to compute the DFT with only a subset either of input or output points. By using the respective even and odd symmetries of the cosine function and the sine function, the kernel of the DFT or the basis functions of the expansion is used. So, the quick Fourier transform(QFT) will reduce the number of floating-point operations necessary compute the DFT by a factor of two or four.

In equation (1), the complex data s(n) can be decomposed into its real and imaginary parts and those parts further decomposed into their even symmetric and odd symmetric parts. The sum over an integral number of periods of an odd function is zero, and the sum of an even function over half of the period is one half the sum over the whole period. Then the equation (2) becomes

$$S(k) = 2 \sum_{n=0}^{N/2-1} \{[u_e(n)\cos\theta_{nk} + v_o(n)\sin\theta_{nk}]$$

$$+ j[v_e(n)\cos\theta_{nk} - u_o(n)\sin\theta_{nk}]\} \quad (3)$$

for $0 \le k \le N-1$.

The evaluation of the DFT using (3) requires half as many real multiplications and half as many real additions as evaluating it using (1)-(3). This saving is independent of whether the length is composite or not. We should add the data points first then multiply the sum by the sine or cosine which requires one rather than two multiplications. Next, we take advantages of the symmetries of the sine and cosine as functions of the frequency index k, Using these symmetries on (3) gives

$$S(N-k) = 2 \sum_{n=0}^{N/2-1} [u_e(n)\cos\theta_{nk} + v_o(n)\sin\theta_{nk}]$$

$$+j[v_e(n)\cos\theta_{nk} - u_o(n)\sin\theta_{nk}] \quad (4)$$

After processing the equation (4) by the QFT, a spectrum of frequency can be divided into low frequency and high frequency bands. It must be analyze the power spectrum of the split signal for each bands and decide the limits of the computed signal level. That is, we can define SNR as a total power and signal power P_S and signal power P_T which is a sum of the powers from each, $f_{\max}$ f_{next} whose power is $P_{\max}$ to $\pm i\Delta f$ (Δf : frequency sampling interval). Eventually discriminating the kind and ID number of signal. By using equations (1)-(4), we can evaluate the performance of DTMF receiver. The experimental terms include the magnitude tests, the twist tests, frequency offset tests, and the tone-to-total energy tests, so on[6].

Ⅲ. Experimental results

Comparison of the number operations for O(N2) DFT algorithms. For the various algorithms, the QFT seems to be the most efficient for an arbitrary length N. The total algorithm, along with the modified second-order Goertzel algorithm and the direct calculation of the QFT, requires N^2 real multiplications and N^2+4N real additions for complex data[7].

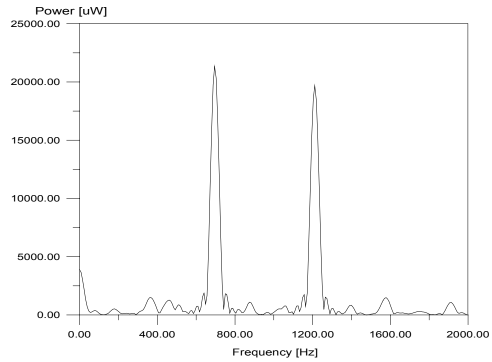
Since the complex operations only occur at the last stage, the QFT algorithm is well suited for DFT`s of real data. So, we can obtain the number of operations required for QFT on real. Then O_M and O_A stand for the number of real multiplies and adds, respectively.

$$O_M = \frac{N}{2}\log_2(N) - \frac{11}{8}N + 1 \tag{5}$$

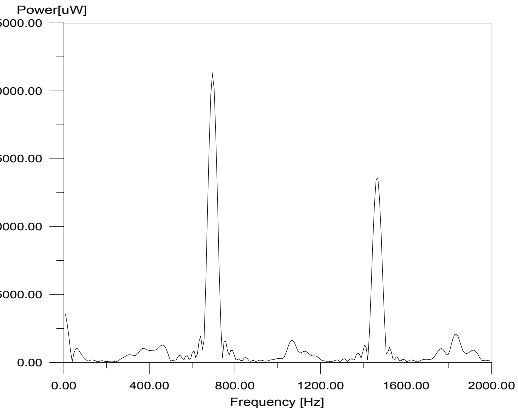
$$O_A = \frac{7}{4}N\log_2(N) - 3N + 2 \tag{6}$$

In this experiments, we use the length-2^8 data. Therefore, the number of operation is $O_M = 673$ and $O_A = 2818$.

Fig. 1 show power spectra of (a) DTMF No. 1 and (b) DTMF No.3, and the spectrum (a) are combined signal of 697 Hz and 1447 Hz, (b) 697 Hz and 1447 Hz..



(a) DTMF No. 1

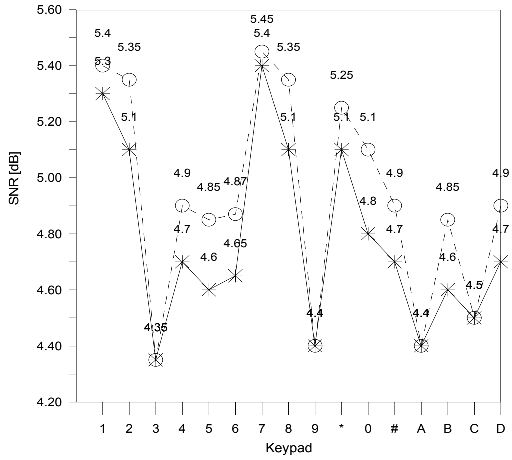


(b) DTMF No. 3

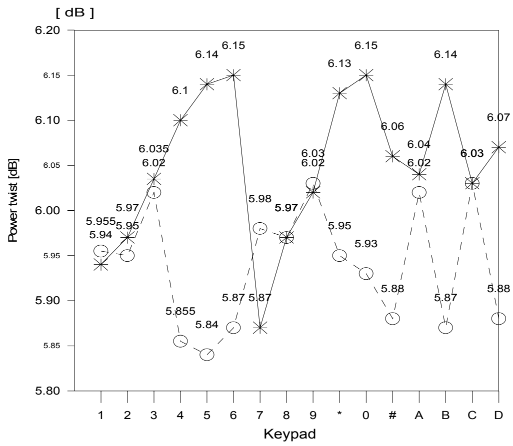
Fig. 1 Power levels of DTMF No. 1 and 3.

In other to decide a signal classification, a system will set a power level that satisfies simultaneously the signal specification of the ITU-T and the signal level of the practical operation. Then we must find threshold values.

Figure 2 show (a) a typical SNR and (b) power twist of the DTMF signals of the keypad. In the specification of ITU-T, the maximum and the minimum SNR level of DTMF signal are 0 dB and -35 dB, respectively. Also, a tolerance limit of the power twist is 6 dB. But a tolerance limit such a case can't apply the practical decision.



(a) SNR



(b) Power twist

Fig. 2 SNR and power twist of DTMF signals

The results of computer simulation like fig. 2 produce that the SNR are 5 and 0.5 dB, and the power twist is 6±0.5 dB. So, considering the allowable threshold, we can decide that the tolerance of the SNR and the power twist are 5.5 dB and 6.5 dB, respectively.

IV. Conclusion

In this paper, DTMF detection algorithm, highly efficient in both real-time and memory, were described. For the QFT based algorithm can be implemented on several DSP chip, and was found to have better digit simulation performance.

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