Efficient Signal Analysis of TDX-families PCM Signal Acquisition System with the Modified DFT

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Abstract - In this paper, we have developed a PCM signal analysis system which can analyze status of signals sent from/received to the TDX-families PCM signaling service equipment. We propose the modified DFT to analyze the status of an acquired PCM signal, discuss the algorithm and the discrimination of the analyzed signal.

1. Introduction

In TDX-families signaling service equipment, there are several signal switching service functions which is used to control the communication path including DTMF/R2MFC/CCT/AT [1].

Practical operators, which to through continuous operating process, require delicate controls. Doing these, this paper has invented a system that can get PCM signal from TDX signal service equipment, and then analyze it. The system provides more accurate and quick install for test to the signaling service equipment in the field that want to check an access switching subsystem-subscriber or -trunk(ASS-S, T) or problem occurred[2, 3]. It is the problems for us to estimate the input/output status of various signals on the transmission line, a switching board match, and the performance to the relative switching board.

There are many instances when signal processing involves the measurement of spectra. The classical Cooley-Turkey FFT and prime factor FFT exploit the periodic properties of the cosine and sine functions to remove redundancies. The set of algorithms known as the fast Fourier transform(FFT) consists of a variety of tricks for reducing the computation time required to compute a discrete Fourier transform(DFT). Since the DFT is the central computation in most spectrum analysis problems, it follows that the FFT implementation of the DFT, which, in some practical cases can improve performance. A quick Fourier transform(QFT) for arbitrary data lengths emphasize prime lengths where traditional FFT's do not work[4]. Compared with Goertzel's method or other direct methods, the QFT will reduce the number of floating -point operations necessary to compute the DFT by a factor of two or four.

In this paper, we will show that the DFT algorithm can be easily modified to compute signal spectra. Then, by using the spectrum, it classifies the type of signal such as R_2MFC/DTMF/CCT/VOICE, and finally decide the digit.

2. Development of signal acquisition equipment

2.1 The PCM signal acquisition system

The configuration of PCM signal acquisition system is shown in fig. 1. This equipment is connected between universal signal transceiver unit(USTU) and time switch unit(TSU)[3]. In the switching signal, one frame has 32 channels, slot length of each channel is 8 bits, and clock speed is 2.048 MHz. The circuit board is inserted at PC slot to get PCM signal and to analyze the channel preferred[5].

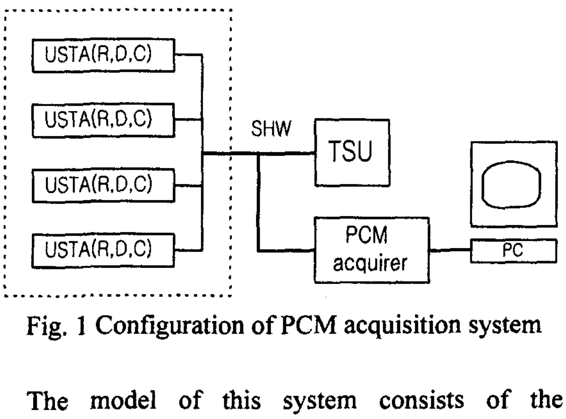


Fig. 1 Configuration of PCM acquisition system

The model of this system consists of the PCM acquisition hardware and the software part for analyzing the signal. The software program works as follows; converting the acquired PCM signal into decimal number, expanding and windowing the decimal number, classifying and deciding the signal, finally, displaying the identity number in monitor.

The PCMA can select the desired channel for investigating and analyze the status of the acquisition signals.

2.2 The signal analysis algorithm

All sample data are saved at buffer in terms of the file entity using random block method, which takes one block into account to 256 bytes in the period of 32 ms.

If we are entered the file number by using the random block method, files of the binary type are converted into the decimal type by μ-law technique[5]. If the signal isn't in channel, the computer repeats the same operations until the signal is detected. If the computer detects the signals, then it reads the signal and memorize to the buffer.

The compressed decimal data by the PCMA are expanded by linearization. After that, it can happen the Gibb's phenomenon and the ripple. The Blackman window function $w(n)$ and the zero-padding were used to reduce them[6]. To obtain a finite-duration casual impulse response is to simply truncate the signal.

When the input is real, the modified DFT(MDFT) is defined as[7]

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

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

Where N is the number of samples and assumed to be even in the following. The subscripts n, i and k are used as a time, another time and frequency index, respectively. It is important that these require only real-value operations. These equations show that the k_{th} modified DFT(MDFT) corresponds to the transversal filter and the modified inverse DFT(MIDFT) is achieved by summing all MDFT outputs

After processing the equation (1) by the MDFT, 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 P_T and signal power P_S which is a sum of the powers from each f_{max}, f_{next} whose power is P_{max}, P_{next} to $\pm i\Delta f$ (Δf : frequency sampling interval).

$$P_T = \sum_{k=-N}^{N} \left|S_{k,i}\right|^2, \quad k \le N \quad (3)$$

$$P_s \cong \sum_{i=-k}^{k} S_{k,i}(f_{\max} + i\Delta f) \quad (4)$$

$$P_N \cong P_T - P_s \quad (5)$$

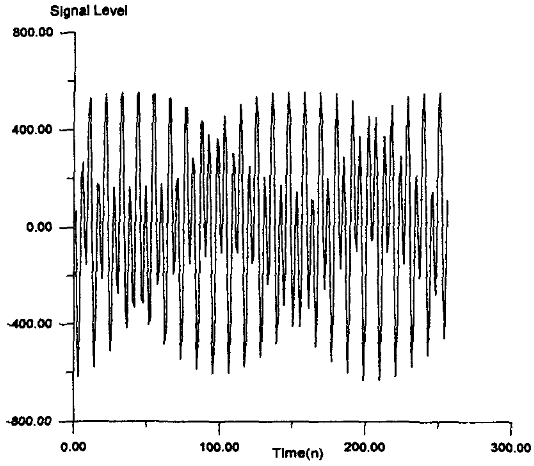
$$S/N = 20\log_{10}\left(\frac{P_S}{P_{NS}}\right) \text{[dB]} \quad (6)$$

Where P_N is the noise power. Eventually, we decide the kind and ID number of signal. Spectra of DTMF signal contain surely both high-frequency group component and low-frequency group component. Hence that of R2MFC and CCT signal have low-frequency component that contain backward MFC signal group component in the low-frequency band and forward MFC and CCT signal group component in the high-frequency band.

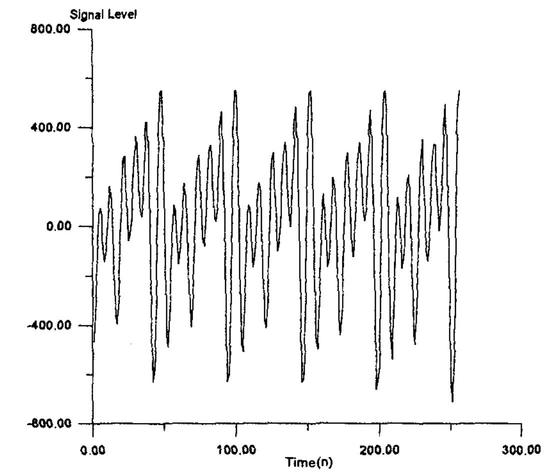
So, it can be classified the signal by calculating the maximum power $P_{max.low}$ and $P_{max.high}$ with respect to low and high frequency region whose criterion is 1162Hz, and comparing them with some threshold values Th_1 [8].

3. Experimental results

An analyzed signal will be one of DTMF/R_2MFC/CCT/VOICE signal



(a) The DTMF No. 3



(b) The korea language voice '아'

Fig. 2 Signals waveform of (a) the DTMF No. 3 and (b) the korea voice '아'

. Figure 2 show signals waveform of (a) the DTMF No.3 and (b) the korea language voice '아'.

Fig. 3 presents spectra of each signal after processing the MDFT of the Fig. 2. The 256 bytes data with the sampling rate of 8 kHz by the zero-padding have the data length of 1024. Then sample values have the zero-crossing point in the 8000/1024 = 7.8125 Hz.

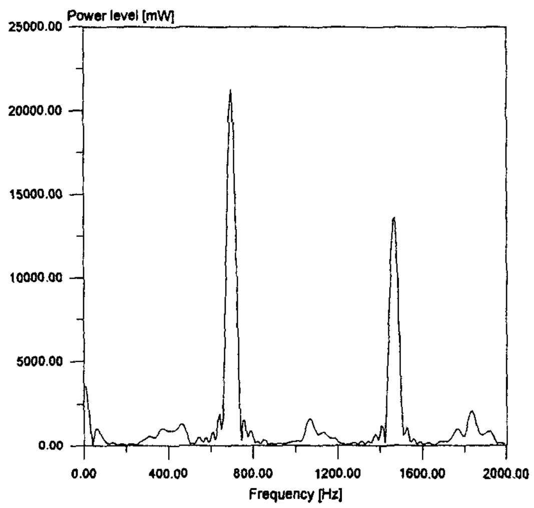


Fig. 3 The power spectrum of the DTMF No.3

In fig. 3, the signal presents a combined signal of 697 Hz and 1447 Hz. Fig. 4 presents the power spectrum of the korea language '아'.

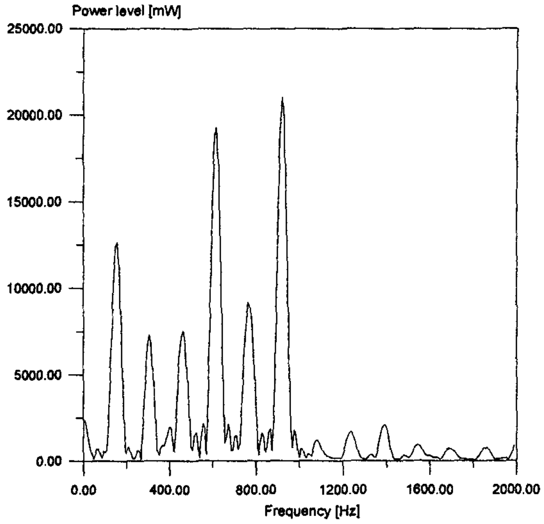


Fig. 4 The power spectrum of the voice '아'

Where the horizontal axis is the frequency domain and the vertical axis is the power level(μW).

4. Conclusion

In order to analyze the signal of acquired PCM signal, it proposes the MDFT algorithm. After that processing, we discriminate what kinds of a R_2MFC/DTMF/CCT/VOICE. Also, by concluding of a tolerance limit for practical operation, this system can use this system to repair a trouble case and to maintain the signal status of the switching system.

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