Proceedings of ITC-CSCC 2000, Pusan, Korea

Signal Analysis and Performance Evaluation of the PCMA System based on the QFT

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Abstract - The system has a function of acquirement PCM signal of the preferred channel from the subhighway (SHW), connecting a universal signal transceiver unit and time switch unit, and then it classifies the type of signal such as R2MFC/ DTMF/ CCT/ VOICE, and finally discriminates the digit.

This paper describes the spectral analysis of the PCM acquisition system usng the quick Fourier transform(QFT), and discusses the algorithm of signal analysis and discrimination.

1. Introduction

The switching services are allowed to us for the sake of BISDN(Broadband Integrated Service Digital Network) and digital switching technique improvement. There are several switching signal service functions, which is used to control the communication path including R2MFC(Multi-Frequency Compelled) signal transmission and reception for the information interchange between the switching stations DTMF/MFC/CCT/VOICE [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]. In other words, there are the problems to estimate the input/output status of various signals on subhighway(SHW), the transmission line, a switching board match, and the performance to the relative switching board. Therefore, this paper is focused on analyzing the state of signal send to/receive from a signaling equipment. In order to analyze the signal status of the PCMA(PCM acquisition) system, it is used for the QFT(quick Fourier transform) algorithm.

2. Signal acquisition equipment

PCMA(PCM Acquisition) system is connected between universal signal transceiver unit(USTU) and time switch unit,(TSU). 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[11].

The PCMA system can select the desired channel to investigate and control among PCM signals along SHW, and analyze the status of the acquisited signals. The channel data on PC are converted PCM signals to decimal number by μ-law expansion. It is proposed for the QFT algorithm to analyze signal spectrum and to display the kind of signals. The decimal data reserved in PC are converted to analog signal to be enable to process digital signal processing operation during the linearization[3, 4].

The program controls an input/output using the PPI mode[5,6]. In addition, program as an IFG, constitutes 1μs clear pulse generation routine and file entity using random block method, taking one block into account to 256 bytes. Files of the binary type are converted into the decimal type by $\mu-law$ technique[7]. 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. 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.

3. Theary of signal analysis using QFT

The compressed decimal data by the PCMA are expanded by linearization[7]. Each 256 bytes data measured in 32 ms are transformed into the signal of the frequency domain by FFT. After that, it can happen the Gibbs phenomenon and the ripple. The Blackman window function and the zero-padding were used to reduce them[9, 13].

The obtained impulse response s(n) is given by

$$s(n) = s_d(n)w(n),\ 0 \le n \le N-1 \qquad (1)$$

Where $s_d(n)$ is real impulse response and w(n) represent window function. In equation (1), we use the symmetric properties of DFT to derive an efficient algorithm of s(n), and develop a basic QFT algorithm for arbitrary data lengths[14]. Compared with Goertzel' method or other direct methods, the QFT will reduce the number of floating-point operations necessary compute the DFT by a factor of two or four. The algorithm has an interesting structure related to that of the DCT and DST, and it is well suited for the DFT's of real data.

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 given by

$$S(k) = \sum_{n=0}^{N-1} s(n)e^{-j2\pi nk/N} \quad \cdots\cdots (2)$$

for $0 \le k \le N-1$. The equation (2) has an even real part and odd imaginary part. 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. We have

$$\begin{aligned} s(n) &= u(n) + jv(n) \\ &= [u_e(n) + u_o(n)] + j[v_e(n) + v_o(n)] \end{aligned} \quad \cdot (3)$$

$$e^{-j2\pi nk/N} = \cos(2\pi nk/N) - j\sin(2\pi nk/N) \quad \cdot\cdot (4)$$

In equation (3), the respective even and odd parts of the real part of x(n) are given by

$$u_e(n) = [u(n) + u(N-n)]/2 \qquad (5)$$

$$u_o(n) = [u(n) - u(N-n)]/2 \qquad (6)$$

Using a simpler notation with $\theta_{nk} = 2\pi nk/N$, the DFT of (2) becomes

$$S(k) = \sum_{n=0}^{N-1} [u(n) + jv(n)][\cos\theta_{nk} - j\sin\theta_{nk}] \quad \cdot (7)$$

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 (7) becomes

$$\begin{aligned} S(k) = 2\sum_{n=0}^{N/2-1} \{ & [u_e(n)\cos\theta_{nk} + v_o(n)\sin\theta_{nk}] \\ & + j[v_e(n)\cos\theta_{nk} - u_o(n)\sin\theta_{nk}]\} \end{aligned} \quad \cdot (8)$$

for $0 \le k \le N-1$. The evaluation of the DFT using (8) requires half as many real multiplications and half as many real additions as evaluating it using (2)-(7). 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.

The total algorithm, along with the mofied second-order Goertzel algorithm[9] and the direct calculation of the FFT, requires N^2 real multiplications and N^2+4N real additions for complex data in

Algorithm	real mults.	real adds.	trig. eval.
DFT	$4N^2$	$4N^2$	$2N^2$
Goertzel	N^2+N	$2N^2+N$	N
QFT	N^2	N^2+4N	$2N$

Table 1. Comparison of the number operations for $O(N^2)$ DFT algorithms

Of the various algorithms of Table 1, the QFT seems to be the most efficient for an arbitrary length N. After processing the equation (8) by the FFT, 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).

$$P_T = \sum_{k=-N}^{N} |S(k)|^2, N > k \quad \cdots (9)$$

$$P_{NS} \cong P_T - P_S, \quad P_S \cong \sum_{i=-k}^{k} S(f_{\max} + i\Delta f) \quad \cdot (10)$$

$$SNR = 20 \log_{10}\left(\frac{P_S}{P_{NS}}\right) \text{ [dB]} \quad \cdots\cdot (11)$$

Where P_{NS} is the noise power. Eventually discriminating the kind and ID number of signal. Spectrum of DTMF signal contains surely both high-frequency group component and low-frequency group component. Hence that of R2MFC and CCT signal have low-frequency component which contains 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 1162㎐, and comparing them with some threshold values Th_1.

step 1. Signal classification

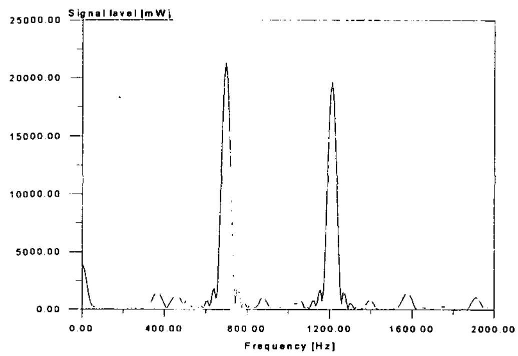
step 2. Voice signal extraction

step 3. Chack up ITU-T power twist spec

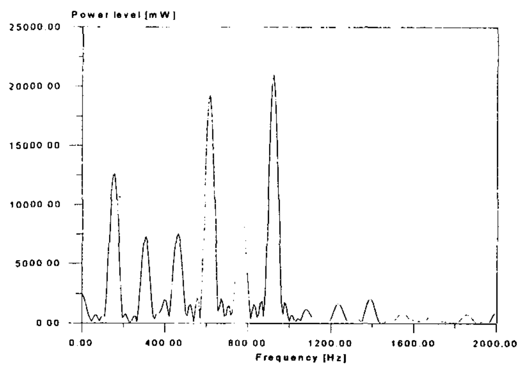
step 4. Signal ID number decision

4. Experimental results

In other to test channels of TDX signaling service equipment, the PCMA card is inserted at PC slot, saving the PCM signal and analyzing channel to the computer in the file type. Then an analyzed signal will be one of DTMF/R_2MFC/CCT/VOICE signal, it is displayed in monitor.



Spectrum of DTMF No. 1.



Spectrum of voice"아"

V. Conclusion

To rapid diagnose the cause of trouble in the signal service, provided by the fully electronic switching system such as TDX, and to solve these problems, this paper has been developed the signal status analysis system for discriminating what kinds of a R_2MFC/DTMF/CCT/VOICE are analyzed. Also, by concluding of a tolerance limit for practical operation, this system can apply to control the signal qualities in the TDX signaling service equipment, and can use this system to repair a trouble case and to maintain the signal status of the switching system.

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