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Performance Evaluation of Spectral Analysis System for TDX-families Signaling Service Equipment

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ABSTRACT

It has developed a PCM signal acquisition(PCMA) system which can analyze status of signals in order to establish rapid diagnosing in TDX-families signaling service equipment. We develop the quick Fourier transform(QFT) for length 2^M data to analyze the power spectral of the PCMA system. This algorithm can reduces the number of floating-point operations necessary to compute the DFT by a factor of two or four over direct methods or Goertzel's method for prime lengths. In the experimental results, the system classifies the type of signals and finally discriminates the digit.

Key words : PCMA, QFT

I. Introduction

The switching services are allowed to us for the sake of 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 to interchange the information between the switching systems, DTMF(Dual Tone Multi-Frequency) signal reception to identify the PB(Push Button) subscribers, some AT(Audible Tone) transmission and reception for the connection status of the call from switching to phone, and CCT transmission and reception to confirm the status of the path between switching systems using CCS(Common Channel Signaling)[1].

On establishing the TDX-switching systems in the field, the practical operators which to through continuous operating process require delicate controls between two switching systems. To do easily 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 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.

In order to analyze the signal status of the PCMA(PCM acquisition) system, we develop an algorithm called the quick Fourier transform(QFT) that reduces the number of floating-point operations necessary to compute the DFT by a factor of two or four over direct methods or Goertzel's method for prime lengths[4]. The classical Cooley-Turkey FFT and prime factor FFT exploit the periodic properties of the cosine and sine functions to remove redundancies[5]. So, we use the symmetric properties of these functions to drive an efficient algorithm.

II. Signal acquisition system

The PCMA system is connected between universal signal transceiver unit(USTU) and time switch unit(TSU) in part of signaling service

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unit[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[6]. The configuration of PCMA system is shown in Fig. 1.

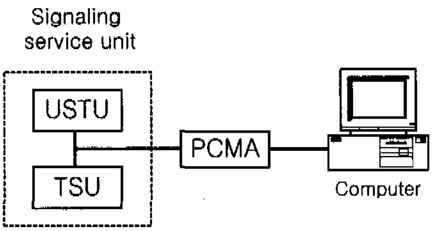


Fig. 1 Configuration of PCM acquisition system.

In Fig. 1, the PCMA hardware connects the signaling service unit and the PC. Then the PC operates the software converting the signal[7,8].

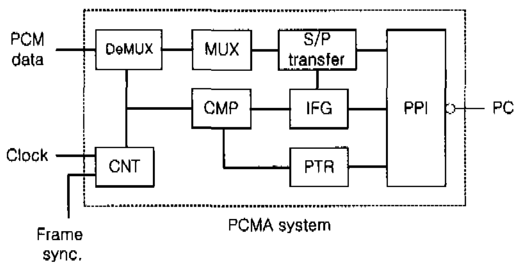


Fig. 2 Block diagram of PCMA system

Fig. 2 shows the hardware block diagram of the PCMA system[12, 13, 14]. The PCM data line and the frame synchronization line are connected the SHW in the TDX-signaling service unit. Then the system can select the desired channel to investigate and control among PCM signals along SHW, and analyze the status of the acquisited signals.

As shown in the fig. 2, the DeMUX (demultiplexer) is the data divider to share the PCM signal in each channel, and the MUX (multiplexer) plays a selector role in selecting the required channel. The CNT(counter) count a channel slot by synchronization between frame signal and DeMUX signal. The S/P(serial to parallel converter), the serial data from the MUX converts into the parallel data, is a shift register which conveys into I/O port. The CMP(comparator) compares the status of the CNT and the PTR(Pointer). In addition, the IFG(input flag F.F) is the input flag informing the status of the S/P to the computer.

The I/F(interface) used the general PPI chip for the interface circuit connecting with the SHW and the computer. Now, it is necessary to the PCM data(DX) on SHW and the clock signal and the frame synchronization signal, and the Quad Differential Line Receivers can do it.

By using these system, the PCM data reserved in PC are converted to analog signal to be enable to process digital signal processing operation during the linearization [9]. Then, in order to analyze signal spectrum and to display the kind of signals use the QFT algorithm proposed.

Ⅲ. Signal analysis algorithm

The compressed decimal data by the PCMA are expanded by linearization[9]. Each 256 bytes data measured in 32 ms 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[15].

The QFT algorithm can be easily modified to compute the DFT with only a subset either input or output points[7]. 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

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

for $0 \le k \le N-1$. The equation (1) 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

$$\begin{aligned} s(n) &= u(n) + jv(n) \\ &= [u_e(n) + u_o(n)] + j[v_e(n) + v_o(n)] \end{aligned} \qquad (2)$$

In equation (2), the respective even and odd parts of the real part of x(n) are given by

$$u_e(n) = [u(n) + u(N-n)]/2 \quad (3)$$

$$u_o(n) = [u(n) - u(N-n)]/2 \quad (4)$$

Using a simpler notation with $\theta_{nk} = 2\pi nk/N$, the DFT of (1) becomes

$$S(k, N, s) = \sum_{n=0}^{N-1} [u(n) + jv(n)][\cos\theta_{nk} - j\sin\theta_{nk}] \quad (5)$$

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 (5) becomes

$$S(k, N, s) = 2\sum_{n=0}^{N/2-1} \{[u_e(n)\cos\theta_{nk} + v_o(n)\sin\theta_{nk}] + j[v_e(n)\cos\theta_{nk} - u_o(n)\sin\theta_{nk}]\} \quad (6)$$

for $0 \le k \le N-1$. The evaluation of the DFT using equation (6) requires half as many real multiplications and half as many real additions as evaluating it using above equations (2)-(6). 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 equation (6) gives

$$S(k, N-k, s) = 2\sum_{n=0}^{N/2-1} \{[u_e(n)\cos\theta_{nk} - v_o(n)\sin\theta_{nk}] + j[v_e(n)\cos\theta_{nk} + u_o(n)\sin\theta_{nk}]\} \quad (7)$$

for $0 \le k \le N/2-1$. Equations (6) and (7) together reduce the number of operations by a factor of two because they calculate two output values at a time. The total algorithm, along with the modified 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. Of the various algorithms of Table 1, the basic QFT of the equation (6) and (7) seems to be the most efficient for an arbitrary length N.

Table 1. Comparison of operations for $O(N^2)$

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	$2N$

However the length of the data sequence is even, it is possible to repeat some of the operations used for the basic QFT. In deed, we use the lengths 2^M data. So, a recursive formation can be developed much as can be done for the Cooley-Turkey FFT. The consequence is the appearance of the discrete cosine and discrete sine transforms in the formation.

For $N = 2^M$, the equation (1) can decompose DCT of a length-(N+1) into DST of a length-(N-1) about the sequence $s(n)$:

$$DCT(k, N+1, s) = \sum_{n=0}^{N} s(n)\cos\left(\frac{\pi nk}{N}\right), \quad 0 \le k \le N \quad (8)$$

$$DST(k, N-1, s) = \sum_{n=1}^{N-1} s(n)\sin\left(\frac{\pi nk}{N}\right), \quad 1 \le k \le N-1 \quad (9)$$

Comparing to the definitions by Wang[8, 9], one sees that we are using the types DCT-I and DST-I, respectively. We will use the following symmetric relations of sine and cosine functions.

$$\cos\left(\frac{2\pi(N-n)k}{N}\right) = \cos\left(\frac{2\pi nk}{N}\right) \quad (10)$$

$$\sin\left(\frac{2\pi(N-n)k}{N}\right) = -\sin\left(\frac{2\pi nk}{N}\right) \quad (11)$$

By defining a length-(N/2+1) sequence $s_e(n)$, which is two times the even part of $s(n)$, as

$$s_e(n) = s(n) + s(N-n), \quad 1 \le n \le N/2-1 \quad (12)$$

with $s_e(0) = s(0)$ and $s_e(N/2) = s(N/2)$, and defining a length-(N/2-1) sequence $s_o(n)$, which is two times the odd part of $s(n)$, as

$$s_o(n) = s(n) - s(N-n) \quad 1 \le n \le N/2-1 \tag{13}$$

So, we can decompose a length-N DFT into a length-(N/2+1) DCT and a length-(N/2-1) DST for the first half as

$$\begin{aligned} S(k,N,s) &= DCT(k, N/2+1, s_e) \\ &\quad - j\,DST(k, N/2-1, s_o) \end{aligned} \tag{14}$$

where $S(0,N,s) = DCT(0, N/2+1, s_e)$, $0 \le k \le N/2-1$ and the second half as

$$\begin{aligned} S(N-k,N,s) &= DCT(k, N/2+1, s_e) \\ &\quad + j\,DST(k, N/2-1, s_o) \end{aligned} \tag{15}$$

where $S(N/2,N,s) = DCT(N/2, N/2+1, s_e)$, $0 \le k \le N/2-1$.

After processing the equation (14) and (15), 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).

$$P_T = \sum_{k=-N}^{N} |S(k)|^2,\ N > k \tag{16}$$

$$P_{NS} \cong P_T - P_S,\ P_S \cong \sum_{i=-k}^{k} S(f_{\max} + i\Delta f) \tag{17}$$

$$SNR = 20 \log_{10}\left(\frac{P_S}{P_{NS}}\right) \text{[dB]} \tag{18}$$

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 $P_{\max,\text{low}}$ and $P_{\max,\text{high}}$ with respect to low and high frequency region whose criterion is 1162 Hz, and comparing them with some threshold values Th_1.

Step 1, Signal classification

$$\text{① } \textit{DTMF signal or voice signal} \quad P_{\max.low} > Th_1,\ P_{\max.high} > Th_1 \tag{19}$$

$$\text{② } \textit{backward } R_2\textit{MFC signal or voice signal} \quad P_{\max.low} > Th_1,\ P_{\max.high} < Th_1 \tag{20}$$

$$\text{③ } \textit{forward } R_2\textit{MFC signal or CCT signal} \quad P_{\max.low} < Th_1,\ P_{\max.high} > Th_1 \tag{21}$$

Although step 1 has been done clearly, additive work is needed to decide the result signal be a correct one; ITU-T spec. tolerance check, especially distinction with voice signal.

Step 2, Voice signal extraction

The step 2 can analyze the frequency sampling interval of power for special service signal of two frequency components to distinguish voice and service signal. Voice signal has more frequency components than service signal, so SNR of the real service signal is relatively high. After previous processing, when SNR is smaller than threshold values Th_2, it is the voice signal.

$$S/N \ge Th_2 : \text{service signal} \tag{22}$$

$$S/N < Th_2 : \text{voice signal} \tag{23}$$

Now, it isn't voice signal, maybe it can be non-voice signal, but it should check ITU-T power twist spec.

Step 3, Check up ITU-T power twist spec.

$$\text{DTMF signal} : P_{twist} < 6\,\text{dB} \tag{24}$$

$$\text{MFC signal} : P_{twist} < 7\,\text{dB} \tag{25}$$

If the signal satisfies these specification, then it can really judge this as service signal. For the case of CCT signal, since power twist is a criterion for forward MFC signal, they don't satisfy the spec. in Step 2 and 3.

Classified its kinds, finally, service signal is tested for its own ID number. It is possible to decide the signal by looking for two dominant frequency(one for CCT) to be within the acceptable variance interval from specified frequency. That is,

Step 4, Signal ID number decision

Decided ID numbers of DTMF signal are as followings. $No. = W_L + W_H$. That is, let it be weights of the low frequency group f_L for 1, 4, 7 and 17, and let it be weights of the high frequency group f_H for 0, 1, 2 and 9. By assigning the index(x) and weight(y) in the each frequency, the identity number is resulted in the sum of the index and weight value[11]. So, by using these methods, No. = 10, 13, 16 and 17 present 'A', 'B', 'C' and '*'.

Ⅳ. Experimental results

A. The measured signals

Fig. 3 shows the software flow to process the program. The software program in PC 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

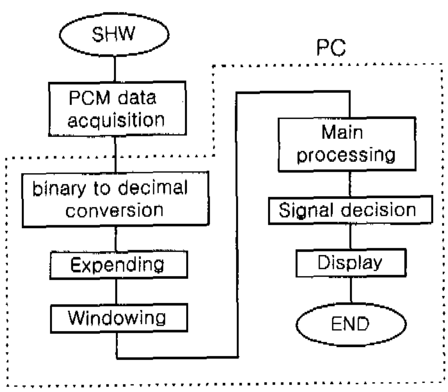


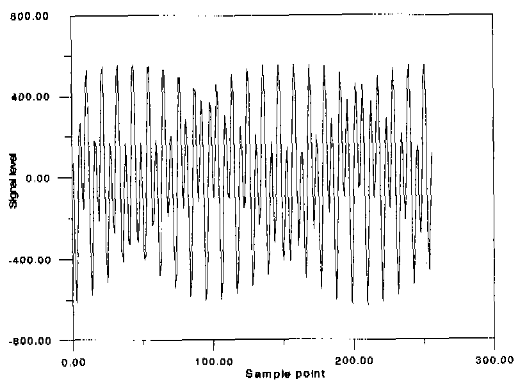
Fig. 3 Flow of PCM signal acquisition

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.

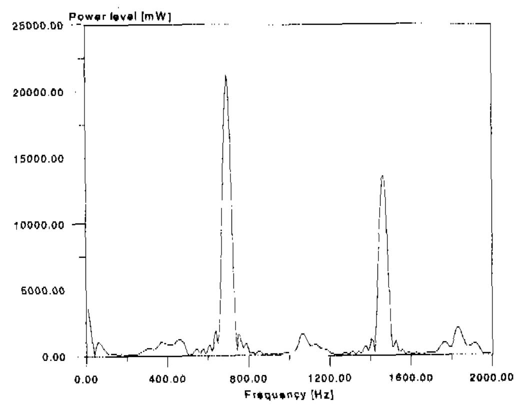
$$O_M = \frac{N}{2}\log_2(N) - \frac{11}{8}N + 1 \tag{26}$$

$$O_A = \frac{7}{4}N\log_2(N) - 3N + 2 \tag{27}$$

In this experiments, we use the length-2^8 data. Therefore, the number of operations are O_M = 673 and O_A = 1378. Fig. 4 show the signal waveform and the power spectrum of the DTMF No.3. In fig. 4, the spectra is a combined signal of 697 Hz and 1447 Hz.



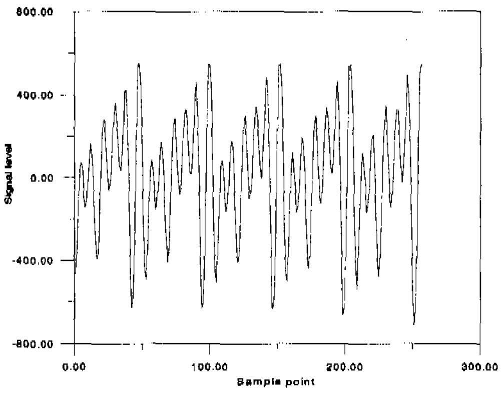
(a) Signal waveform of No. 3



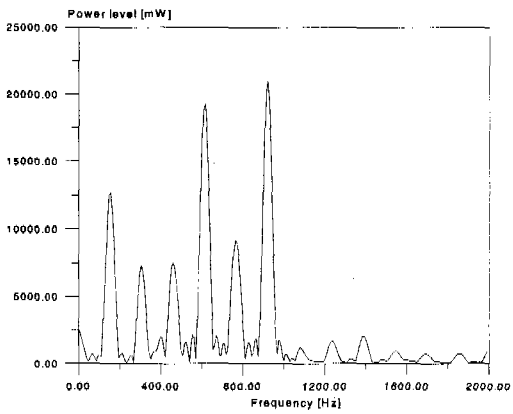
(b) Power spectrum of No. 3

Fig. 4 Signal and power level of DTMF No. 3.

The 256 bytes data with the sampling rate of 8 kHz by the zero-padding have the data length of 1024.



(a) voice "아"



(b) Power spectrum of "아"

Fig. 5 Signal and power level of "아"

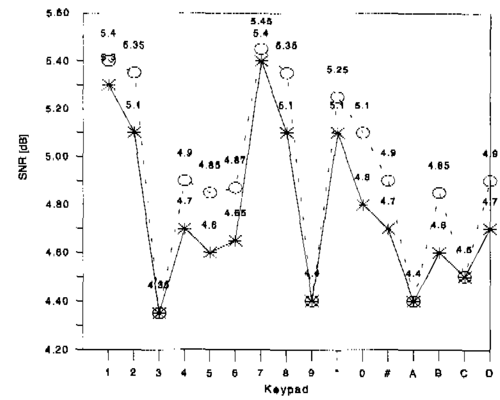
Fig. 5 present the signal waveform and the power spectrum of the Korean language voice "아". The spectrum of the voice "아" is dispersed in the middle of the 340 Hz.

B. Signal classification and decision

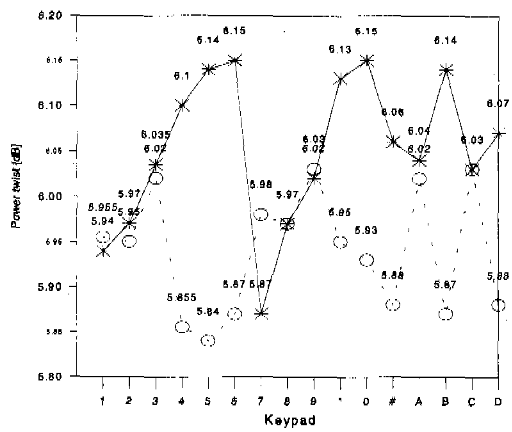
In other to decide an experimental propriety, a system will set a power level that satisfies simultaneously the signal specification of the ITU-T and the signal level of the practical operation.

An acquired PCM signal by PCMA system is one of DTMF/MFC/CCT/Voice signals. Then we must find threshold values Th_1 and Th_2 to have classfy and decide signals through steps 1 ~ 4 . Considering the transmission noise, we could decide the allowable power level Th_1 to the 9000. If the measured signal level is less then Th_1, we consider that the signal doesn't exist. Also, we could obtain the threshold value Th_2 to the 3.5 dB.

Fig. 6 show a typical SNR and power twist of the DTMF. Where the dotted line is the experimental results of transmitting line with the noise of - 6 dB and 0 dB, an oblique line is the experimental results of receiving line with the noise of -32 dB and -26 dB.



(a) SNR

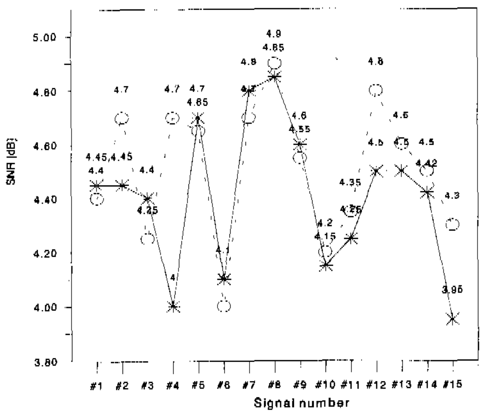


(b) Power twist

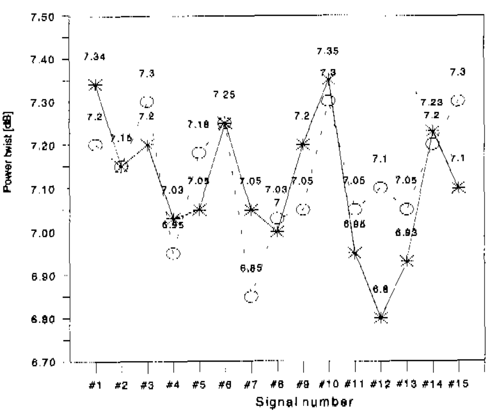
Fig. 6 SNR and power twist of DTMF signals

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. The results of computer simulation like fig. 6 produce that the SNR is 5±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.

Fig. 7 presents a typical SNR and power twist of forward R2MFC signal.



(a) SNR



(b) Power twist

Fig. 7 The SNR and power twist of the forward MFC signal

Where the dotted line is the experimental results of transmitting line with the noise of - 6 dB and 0 dB, an oblique line is the experimental results of receiving line with the noise of -32 dB and -26 dB. In the specification of ITU-T, the maximum and the minimum SNR level of R2MFC signal are 0 dB and - 32 dB, respectively.

Also, a tolerance limit of the power twist is 7 dB for MFC signal. But a tolerance limit such a case can't apply the practical decision. The results of computer simulation in fig. 6 produces that the SNR is 4±0.5 and the power twist is 7±0.5 dB. So, considering the allowable threshold, we can decide that the tolerance of the SNR and the power twist are 4.5 dB and 7.5 dB, respectively

The results for analyzing an arbitrary channel is the DTMF No. 0 signal, the SNR is 2.97 dB and its frequency component consists of the 945 Hz(-25.83 dBm) and 1328 Hz(-23.61 dBm). In the other experiments of the DTMF No. 9 signal, the SNR is 2.61 dB and its frequency component consists of the 843 Hz(-26.49 dBm) and 1469 Hz(-23.44 dBm).

As a result of Korean voice "아", the SNR is - 8.83 dB and its frequency component consists of the combination of 906 Hz(-28.44 dBm) and 773 Hz(- 34.15 dBm). Besides, the tested experiments of various signals give us to the proper results.

Finally, in the experimental results of MFC signal, we can obtain the forward frequency that f_1 signal is 1156 Hz and f_2 signal is 1742 Hz. Then the signal level are -26.44 dBm and 0.45 dBm, respectively. And the SNR is 4.04 dB. In case of CCT signal, we could obtain the decision frequency of 1781 Hz and the signal level of 0.4 dBm. Besides, the tested experiments of various signals give us to the proper results.

V. Conclusion

To rapid diagnose the cause of trouble in the signal service, and to solve these problems, this paper has used the PCMA system for discriminating what kinds of an analyzed R_2MFC/DTMF/CCT/VOICE. In order to analyze the power spectra and conclude a tolerance limit for practical operation, We develop the quick Fourier transform(QFT) for length 2^M data. This algorithm can reduces the number of floating-point operations necessary to compute the DFT by a factor of two or four over direct methods or Goertzel's method for prime lengths. Also, 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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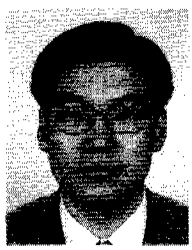

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