UTI Report Vol. 12, No. 1, pp. 77~83, 1981 　　연구논문집 제12권 제 1 호 pp. 77~83, 1981

A Study on ADM/ADPCM Code Converter

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(Received November 5, 1980)

〈Abstract〉

In order to communicate between two systems which have diffent systems, there must be code converters as interfaces. In this paper, ADM/ADPCM code converter, which is used to convert codes of ADM system to these of ADPCM system, is investigated. All signal processings are done in digital domain by using the decimation filter and the FFT algorithm.

ADM/ADPCM 부호 변환기에 관한 연구

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〈요 약〉

서로 다른 부호 체계를 갖고 있는 시스템 간에 신호를 보내기 위해서는 자국(自局)의 신호를 상대국(相對局)의 부호체계에 맞는 신호로 바꾸어 보내야만 한다. 이때 부호를 변환해 주기 위한 부호 변화기가 필수적으로 들어 가야 한다. 본 논문에서는 ADM을 사용하는 시스템에서 ACPCM을 사용하는 시스템으로 신호를 보내고자 할 때 필요한 ADM/ADPCM부호 변환기에 관하여 연구하였다. 모든 신호 처리는 Decimation 여파기와 FFT 알고리즘을 이용하여 디지탈 영역에서 하였다.

I. Introduction

Since the early 1960's, pulse code modulation (PCM) has been mainly used for waveform coding. And for effective bandwidth compression, adaptive delta modulation (ADM), adaptive differential pulse code modulation (ADPCM)and linear predictive coding (LPC) have been developed. As the result, several methods of coding are used, nowadays, at the several nodes in the communication networks. If the same coding systems are used between the transmitters and the receivers, there are no difficult problems. But if the different coding systems are used between them, there must be code converters as the interfaces. In this paper, we consider the code converter, which is needed when coded signals are transmitted from one node, using ADM, to the theother node, using ADPCM.

When coded signals are transmitted or received between the two systems, using different coding methods, there can be several code conversion methods. First, after decoding the coded signals of the transmitter, we convert these decoded signals to baseband analog signals. Then, we encode these analog signals by the coding methods of the receiver. (Fig. 1. a) In this case, since there must be some processes, which are D/A coversion, sampling, and quantization noise is increasing, and then signal-to-quantization noise ratio (SQNR) is decreasing. To get rid of this shortcoming, we do not convert to analog signals, we do code conversion in the digital domain. In that case, digital filters must be constructed and we must know how to convert the sampling rate between the two nodes. (Fig.1.b) [1]

No one have studied, until now, ADM to ADPCM code converter. [2] Because of the problems of the bandwidth and error rates of PCM (Table 1), in the future, ADM and ADPCM are mainly and widely used. [3] It is the motive of the investigtion about ADM to ADPCM code converter.

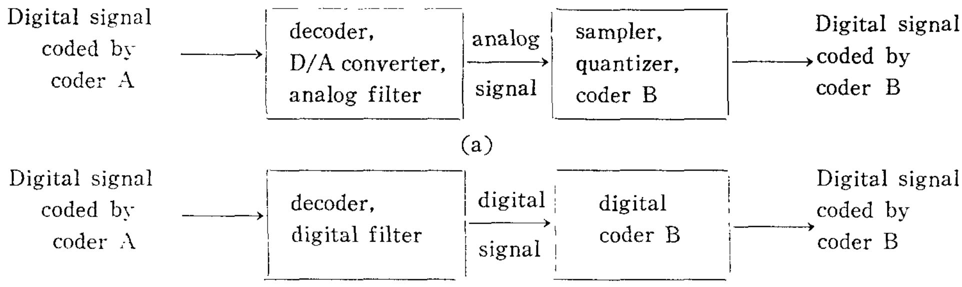

Fig.1. Code conversion methods.
(a) analog code conversion (b) digital code conversion

Table.1. Comparison of PCM with ADM and ADPCM.

	Log PCM	ADPCM	ADM
Transmission rate	48—64Kbps	24—48Kbps	16—48Kbps
Quantization level	2^6—2^8	2^3—2^6	2
Framing	yes	yes	no

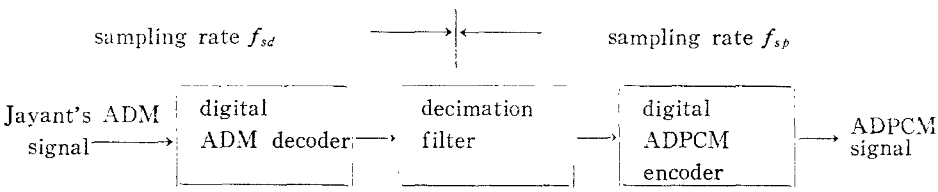

Fig.2. Block diagram of overall system.

Ⅱ. Overall Systems [3]

Among several methods of ADM, we consider the Jayant's ADM, in which the companding, suggested by Jatant, is used. We now convert the codes of the above to those of the ADPCM. We, first, define the sampling rate f_{sd} in ADM and f_{sp} in ADPCM. The number of the quantizer levels is L-level. And one sample is encoded to s bits. Therefore, although the bit rate of ADM is f_{sd}, that of ADPCM is s times of f_{sp}.

The block diagram of overall systems is shown in Fig.2. Decimation digital filter is an FIR filter with Kaiser window, and FFT algorithm is used for decreasing the computation time.

Ⅲ. Detail Description of Subsystems

1. Jayant's ADM coder/decoder [3]

The block diagram of Jayant's ADM is shown in Fig.3. In Fig.3. the logic [means the following;

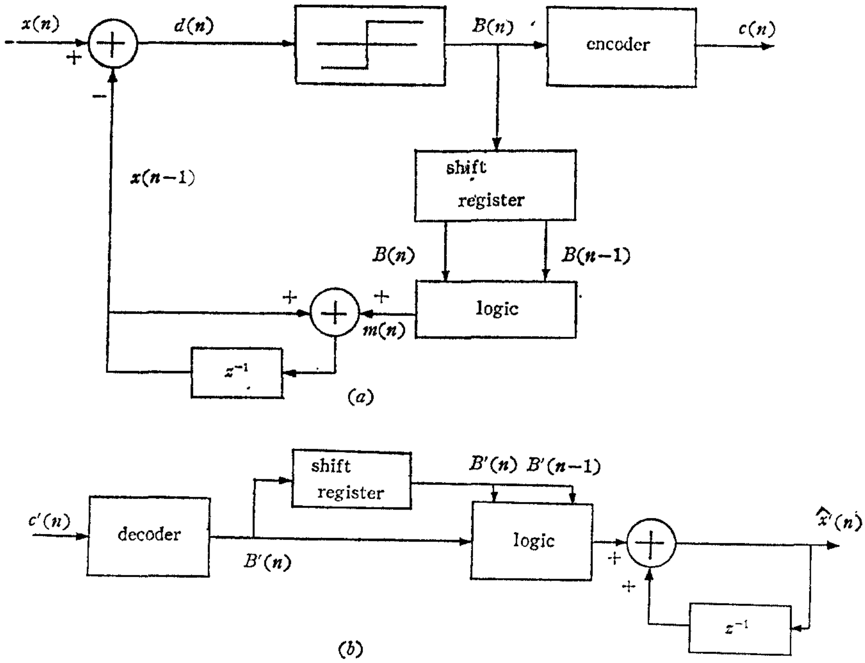

Fig. 3. Jayant's ADM system.
(a) encoder (b) decoder

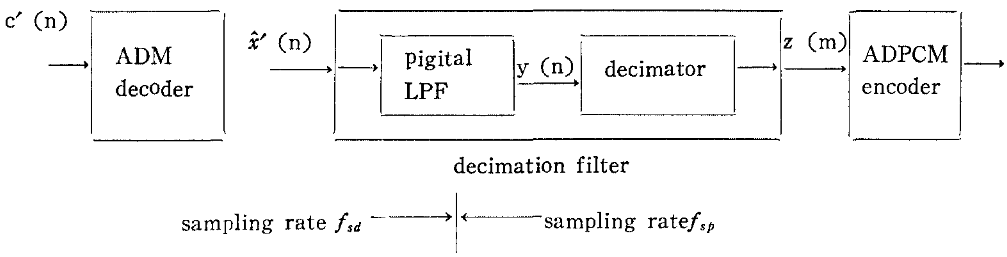

Fig. 4. Decimation filter.

$$m(n)=P_0 m(n-1), \quad \text{if } B(n)=B(n-1),$$
$$m(n)=-Q_0 m(n-1), \quad \text{if } B(n)\neq B(n-1). \qquad (1)$$

2. Digital Filter [1, 4]

In this case, the sampling rate f_{sd} of ADM is greater than that of ADPCM (f_{sp}). To match the two different systems, there must be an interface. (Fig. 4) According to the old methods of code conversion, after $\hat{x}'(n)$ is converted to baseband analog signal by D/A conversion, it is passed through analog low pass filter(LPF), then resampled by f_{sp} and encoded. But, in this case, much quantization noise is generated. Therefore, to minimize the quantization noise, all code conversions are processed in the digital domain.

If $\hat{x}'(n)$ is directly decimated, aliasing would occur in the process of decimation. However, $\hat{x}'(n)$ must be bandlimited. (In general, the cutoff frequency of the speech systems is 3.4KHz.) The signal $\hat{x}'(n)$ is first passed through a digital low pass filter which undisturbed the desired information and attenuates the signal components having frequencies beyond 3.4KHz to prevent aliasing. The output of the low pass filter $y(n)$ is passed through the decimator. The use of an infinite impulse response (IIR) filter, in this case, has an obvious shortcoming, that is, no computational saving. On the other had, if an FIR filter is used in this case, we can do computations at the rate of f_{sp}.

The relation of decimation digital filter between the input and output sequences is represented by

$$z(m)=\sum_{i=0}^{N-1}\hat{x}'(bm-i)h(i), \tag{2}$$

where $h(i)$ is the impulse response of the FIR filter with a tap size of N, and 'b' is the ratio of f_{sd} to f_{sp}, i.e., f_{sd}/f_{sp}.

The characteristics of digital low pass filter are as following [4];

1) The magnitude of passband ripple and stopband ripple are equal, and the stopband attenuation is greater than or equal to 'a'(dB).

2) The cutoff frequency is f_c.

3) The transition bandwidth is less than Δf. with these above specifications, we get the following equations,

$$N=\frac{a-7.95}{14.36\Delta f} \tag{3}$$

$$\alpha=\begin{cases}0.1102(a-8.7), & 50>a21\\ 0.5842(a-21)^{0.4}+0.007886(a-21), & 21<a\leq 50\\ & a\leq 21.\end{cases} \tag{4}$$

Using the tap size of N and parameter α, which are determined by the above two equations, we get the following Kaiser window function,

$$w(n)=\begin{cases}\dfrac{I_0[\alpha\sqrt{1-2n/(N-1))^2}]}{I_0(\alpha)}, & 0\leq n\leq N-1\\ 0, & \text{otherwise}\end{cases} \tag{5}$$

where $I_0(x)$ is the modified Bassel function of the first kind and zeroth order. Since an ideal low pass filter has the impulse response,

$$h_d(n)=\frac{\sin(w_c(n-k))}{\pi(n-k)}, \qquad k=\frac{N-1}{2} \tag{6}$$

where $w_c(=2\pi f_c)$ is the cutoff frequency. There, we can determine by a windowing method the implulse response $h(n)$ of an FIR low pass filter as

$$h(n)=h_d(n)\cdot w(n), \quad 0\leq n\leq N-1. \tag{7}$$

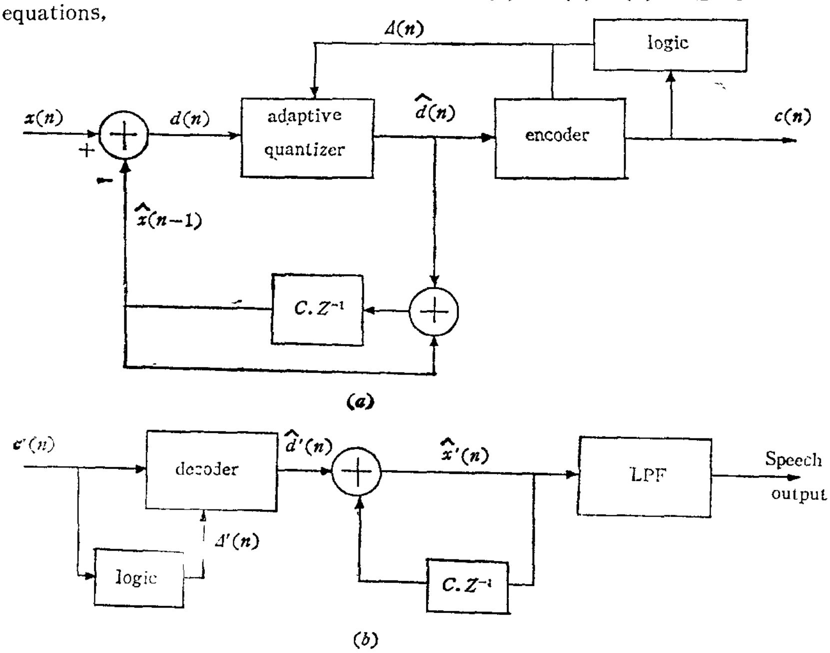

Fig. 5. ADPCM system.
(a) coder (b) decoder

3. ADPCM coder/decoder [3]

The block diagram of ADPCM is shown in Fig. 5. Adaptive predictor is not used in this system, but adaptive quantizer is used. In this system, the step size of the quantizer is determined by using the output sequences of the encoder. It is well known that the optimum value of the predictor gain 'c' is 0.85. The adaptive quantizer used in this system is shown in Fig. 6. 1-word memory is used so that the step size of the quantizer is matched to the input signal variance. For example, let the output of the s-bit quantizer be as following;

$$Y(n)=H(n)\frac{\Delta(n)}{2},$$
$$\pm H(n)=1,3,5\cdots,2^s-1 \quad (\Delta(n)>0,\ s\geq 2). \quad (8)$$

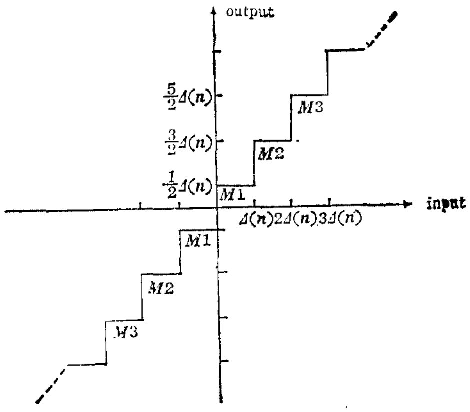

Fig. 6. ADPCM adaptive quantizer

Then we get

$$\Delta(n)=\Delta(n-1)M(\,|H(n-1)|\,) \quad (9)$$

where $M(\,|H(n-1)|\,)$ is the fixed multiplier function. The values of $M(.)$ which maximize the SQNR are shown in Table 2.

Table. 2. ADPCM quantizer multiplication factors.

	M1	M2	M3	M4	M5	M6	M7	M8
16 levels	0.9	0.9	0.9	0.9	1.2	1.6	2.0	2.4

IV. Computer Simulation

Since most energy of voice is within 3.4KHz, the signal (speech) bandwidth of the overall systems is 3.4KHz. The sampling rate of ADM (f_{sd}) is 32KHz, and the output bit rate is 32Kbps. Since the optimum values of P_0 and Q_0 in Eq. (1) is rewritten as following;

$$m(n)=1.5m(n-1), \quad \text{if } B(n)=B(n-1),$$
$$m(n)=-0.66m(n-1), \quad \text{if } B(n)\neq B(n-1). \quad (1)'$$

The sampling rate of ADPCM(f_{sp}) is 8 KHz, which is greater than the Nyquist sampling rate, 6.8KHz, and the bandwidth of the guard band is 1.2KHz to prevent aliasing. One sample is encoded to 4 bits through encoder, that is s is equal to 4. Also, the number of the adaptive quantizer levels(L) is 16.

Therefore, the decimation filter must convert ADM, of which sampling rate is 32KHz, to ADPCM, 8KHz. In this case, 'b' is equal to 4, and substituting this value from Eq. (2) yields

$$z(m)=\sum_{i=0}^{N-1}\hat{x}'(4m-i)\ h(i). \quad (2)'$$

Eq. (2)'is the most important equation in this paper.

In practical system, it is very important that we determine the number of taps of FIR filter. As the number of taps increases, the system cost also increases, and the computation time becomes long. The characteristics of the filter are shown below,

stopband attenuation: $a\geq 30$dB
cutoff frequency : $f_c=3.4$KHz
transition bandwidth: $\Delta f=1/8$(when the sampling rate is 8KHz, $\Delta f=1KHz$).

From Eq. (3), Eq. (4), and the above specification, we can get the values of N and α, that is, 12 and 2.11 respectively. By using the above values, we can rewrite Eq. (5) as the following form;

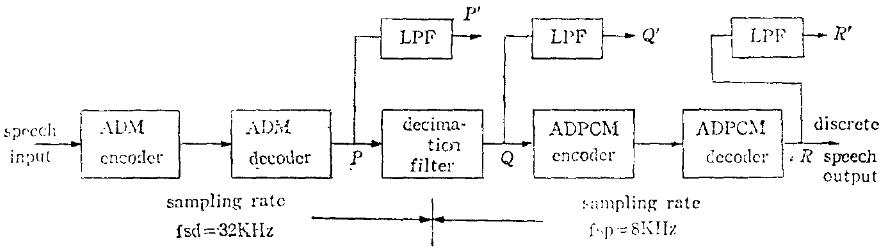

Fig. 7. Block diagram in computer simulation.

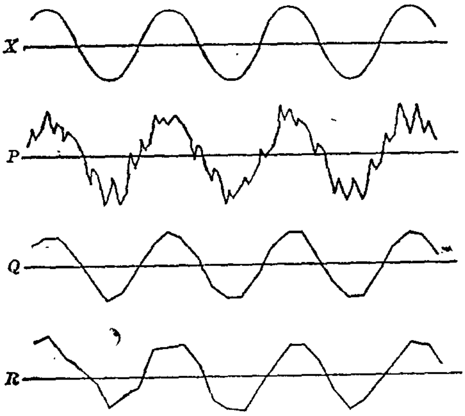

Fig. 8. (X) Input speech signal(1 KHz sine wave)
(P) ADM decoder output before low pass filtering
(Q) Decimation filter output before LP filtering
(R) Discrete speech output before LP filtering

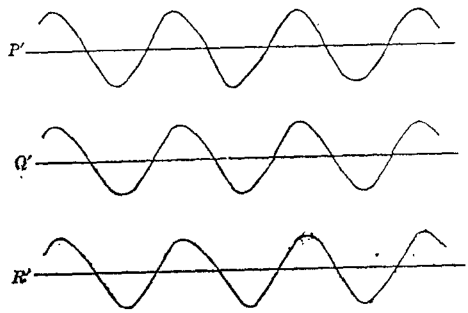

Fig. 9. (P′) Low pass filtered ADM decoder output
(Q′) Low pass filtered decimation filter output
(R′) Low pass filtered speech output, i.e., output speech signal

$$w(n)=\begin{cases}\dfrac{I_0(2.11\sqrt{1-(1-2n/11)^2}}{I_0(2.11)}, & 0\leq n\leq 11\\ 0, & \text{otherwise.}\end{cases} \quad (5)'$$

The block diagram of the overall systems used in computer simulation is as shown in Fig. 7. The waveforms at P, Q, R and P', Q', R' are shown in Fig. 8 and Fig. 9. 1KHz sine wave is used as the speech input of the system.

Ⅴ. Performance

The low pass filtered signal (i.e., P', Q',

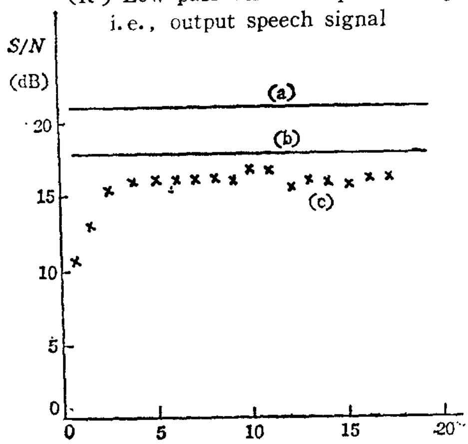

Fig. 10. (a) SQNR of LP filtered ADM decode output signal (at P′ in Fig. 7)
(b) SQNR of LP filtered decimation filter output signal(at Q′ in Fig. 7)
(c) SQNR of output speech signal (at R′ in Fig. 7)

R') is transformed to frequency domain through FFT (N=128), then SQNR is computed in each case. As the amplitude of the input signal increa from 1 to 50, SQNR at P' is 20.84 dB, SQNR at Q' is 17.06dB. But SQNR at R', that is, SQNR at the output, increases according to the increase of the amplitude of the input signal. But, it is apparent that SQNR at Q' is always greater than *SQNR* at R'. Fig. 10 shows SQNR in each case.

Ⅶ. Conclusion

It has been studied that the code of Jayant's ADM system of which the sampling rate is 32KHz is transformed to the code of ADPCM system of which the sampling rate is 8KHz and of which the quantization levls are 16 levels. The key point of this study is the processing the digital signal in the digital domain by using the decimation filter instead of conversion the digital signal to the analog signal.

Not only the ADM/ADPCM code converter but also the ADPCM/ADM code converter which is not considered in this paper will be needed as the interface between the ADM system and the ADPCM system. In this case the interpolation filter, instead of the decimation filter, will be used in the ADPCM/ADM code converter.

Reference

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