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25. An Application of the Adaptive Differential PCM (ADPCM) Method to the Seismic Wave Signal Compression.

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Abstract

We have examined the Adaptive Differential Pulse Code Modulation (ADPCM) method which has been used for a digital speech synthesis on compressing seismic wave data. Response of the ADPCM for the seismic wave data was examined not only by hardlogic which used the speech analyzer but also by software which used the FORTRAN programs. The results indicate that the ADPCM is practically useful for seismic wave data, with the compression ratio ranging from one-third without filtering to one-forth with filtering.

1. Introduction

Recently, a digital processing of seismic wave signals has been utilized to transmit the signal from a local network to a research center, and to record it onto digital magnetic tapes near the sensors. Digital transmission and recording systems present higher signal quality than that obtained by the analog systems, e.g. wide dynamic range and higher signal-to-noise ratio. However, the data length in terms of bit rate is limited by storage constraints, bandwidth in communication links and number of tele-communication channels. For the above reasons, the seismic wave data need to be compressed before being recorded onto tapes and transmitted through communication channels.

In general, the logarithmic signal compression technique has been used for compression of the seismic wave data and expansion of dynamic range. However, this method has the nature that the data loose resolution on mantissa through the logarithmic compression-expansion process. Other data compression techniques have been proposed for data resampling and interpolation in the time domain. In notable works, the Walsh inverse transform in the Walsh domain was used for the Vibroseis (WOOD, 1974), and explicit reversible data compression was also used for the Vibroseis (LEE and YARLAGADDA, 1982).

Data compression techniques have made remarkable progress in digital speech interpolation. A number of methods have been achieved to compress speech and voice-band data. For example, the PARCOR method uses the partial autocorrelation coefficient and the transform coding method uses Walsh, Hadamal and FFT conversions. The data compression ratio is contrary to the identity of original and synthetic waveforms. In general, speech analysis attaches more importance to the compression ratio than to the identity of waveforms, because speech analysis necessitates the extraction of the attributes for the voice, syllable, word and paragraph from synthetic data. In the analysis of seismic waves, the identifying the original and synthetic waveforms takes priority over the compression ratio.

This paper describes that the ADPCM (Adaptive Differential Pulse Code Modulation) bit reduction technique has a high compression ratio and a high waveform agreement after the synthesis. The ADPCM can be applied to practical uses when the seismic wave signal is transferred through communication channels and is stored in some recording medium.

2. ADPCM encoding and other methods

Various methods for speech synthesis and data compression are described in many books (e.g., AGUI and NAKAJIMA, 1980). Therefore, several methods are briefly mentioned in this paper.

In Delta Modulation(DM), one bit ($+\Delta$ or $-\Delta$) is used for the expression of residual between real and forcasted signal. This method must use high sampling rates to reproduce high-quality signals. The method contains two principal noises: granular noise and overload noise. The former occurs when the change of signal amplitude is less than delta amplitude. The latter occurs when the forcasted signal cannot follow the abrupt change of signal amplitude. The delta amplitude must be small to reduce the granular noise and a large sampling rate is required to overcome the overlaod noise.

The Adaptive Delta Modulation (ADM) method uses a variable delta amplitude according to the change of signal amplitude. A small delta amplitude prevents the growth of granular noise when the signal amplitude does not change very much. Conversely, the growth of overload noise is prevented by an increase of the delta amplitude when signal amplitude changes greatly.

The Differential Pulse Code Modulation (DPCM) method digitizes the residual value between the real signal and the forcasted signal by plural bits. In principle, it is required that neighboring data values be highly correlated as in the case of the delta modulation.

The Adaptive Differential Pulse Code Modulation (ADPCM) method uses variable amplitudes for each sampling. It defines the quantitative amplitudes (Δ_i) of differential signal amplitude between the real signal and the previous signal. Then, the relation of quantitative amplitudes Δ_i and Δ_{i-1} are expressed by

$$\Delta_i = K \cdot \Delta_{i-1}$$

where coefficient K is defined by function of the previous data value.

3. Data analysis

Hardlogic examination

Recently, the CMOS-IC Chip MSM-5218RS (manufactured by OKI Electric Industry Co., Ltd.) for ADPCM has been put on the market. It has been applied to the voice replying system with a digital speech interpolation. This LSI contains an analyzing circuit and a synthesizing circuit. The analyzing circuit encodes a 12-bit signal to 4 or 3 bits ADPCM data. The synthesizing circuit reproduces analog data through the internal 10 bit D-A converter after the synthesizing 12-bit PCM data using 4 or 3 bits ADPCM data. The LSI has following features:

1. switchable modes (analyze or synthesize),
2. low power consumption,
3. high-speed real time analyzing (at 8, 6, 4 KHz or external),
4. internal 10 bit A-D converter, and
5. low cost.

The speech analyzer, OKI speech analyser which uses the device of MSM-5218RS, was used for analysis of the seismic wave signal. The analog seismic data was put in the microphone terminal, and the out-put signals were monitored by the inkjet recorder connected to the speaker of the equipment. Fig. 1 is the result of the analysis

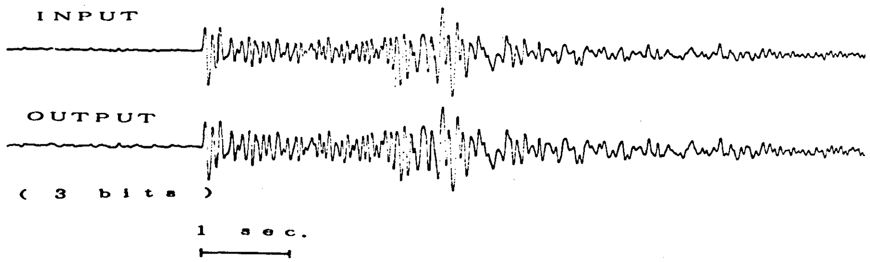

Fig. 1. The result of the analysis. The upper figure: input seismic data from magnetic tape recorder reproduced at ten times of the recording speed. The bottom figure: analog output of decompressed seismic signal from the ADPCM 3 bits encoding data sampled by 8 KHz with 2.27 KHz low pass filter (18 dB/oct.)

showing a noticeable agreement between the input and the decoded seismic waveforms. The equipment can memorize the encoding signals on Random Access Memories (RAM). In practical use, the compressed signal may be transferred to communication channels or recording mediums.

Software examination

The ADPCM encoding technique has been proved even by computer simulation in a digital speech interpolation. The FORTRAN programs of ADPCM four bits encoding and decoding (AGUI and NAKAJIMA, 1980) are shown in Appendixes 1 and 2 respectively. The FORTRAN programs are expressed by very short steps. Fig. 2 shows the seismic wave signal of numerical calculation by computer simulation using 3 bits encoding. A local event in Kanto district, Japan,

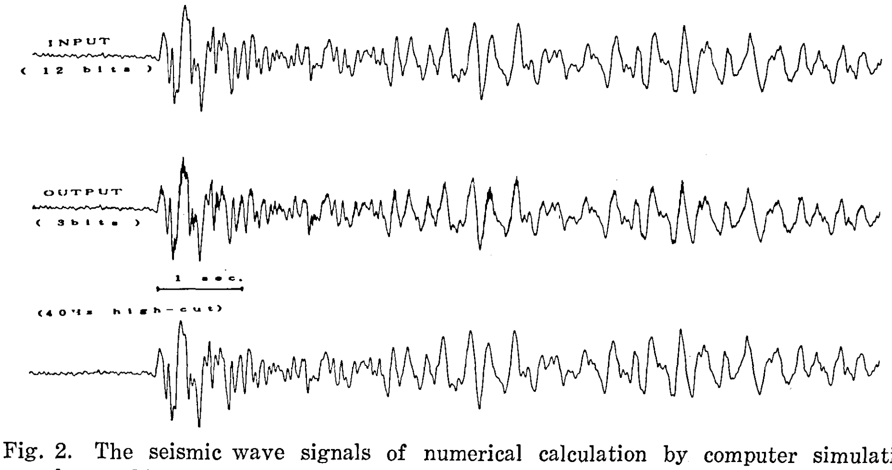

Fig. 2. The seismic wave signals of numerical calculation by computer simulation about 3 bits encoding. Top: input seismic wave data sampled by 12 bits. Middle: decoded signal. Bottom: 40 Hz high-cut filtered signal.

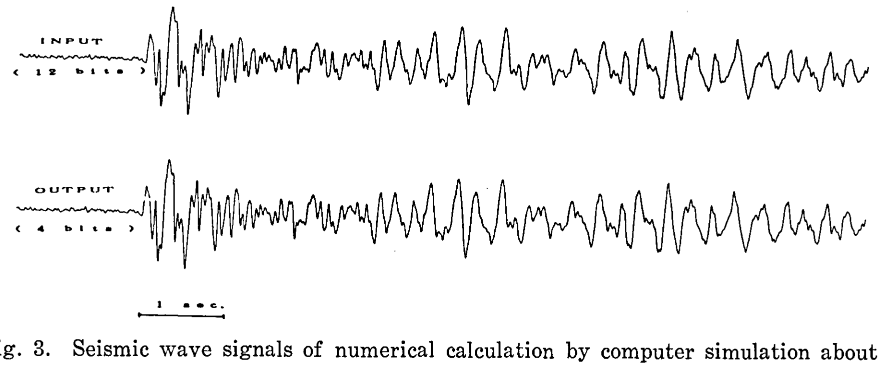

Fig. 3. Seismic wave signals of numerical calculation by computer simulation about 4 bits encoding. Top: input seismic wave signal sampled by 12 bits. Bottom: Decoded wave signal.

was used as the input sesmic wave signal that was observed at the Dodaira Micro-earthquake Observatory, University of Tokyo. The input-signal was sampled on 12 bits at 200 Hz. The decoded seismic wave signal contains the granular noise of 50 Hz near the low fluctuation of data values. The granular noise is removed by the digital Chebyshev high-cut filter (cut off frequency 40 Hz).

Fig. 3 shows a sample of 4 bit ADPCM encoding. The input wave signal is drawn on the top of the figure and the decoded wave signal without filtering is drawn on the bottom of the figure. The figure indicates fine agreement between the input and the decoded wave from. The result suggests the high possibility of practical use of "ADPCM encoding" by computer simulation for compression of seismic signals.

4. Conclusion

We confirmed the applicability of the ADPCM encoding technique to the seismic-wave data compression in the viewpoint of hardlogic and software. The seismic wave is identically reproduced through the ADPCM compression and decompression process. Even three bits compression gave the same quality data as one for the four bits when the high-cut filter was used to eliminate the granular noise. According to the present results, we designed and made digital seismic wave recording systems for temporary observation on land and ocean bottom (Fig. 4) (KASAHARA *et al.*, 1984). The system consists of analog amplifier and low-pass filter, 12-bits A/D, 4 bit ADPCM, CMOS 8 bits CPU (Hitachi 6303), D/A and 20 MB cartrige magnetic tape devices operated by a portable power supply. The system is able to record more than 800

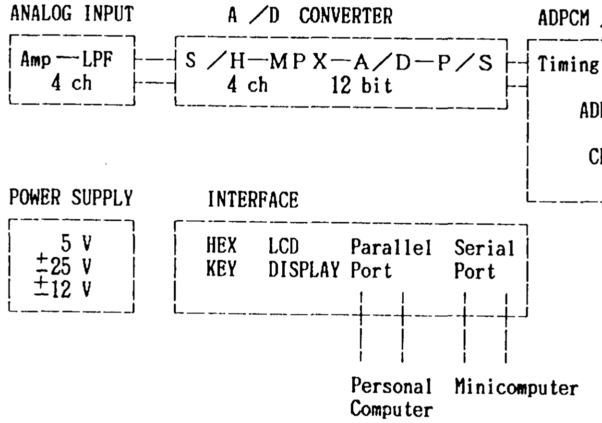

Fig. 4. Block diagram of the digital seismic wave recording system for temporary observation on land and ocean bottom.

events with 4 channels data of 90 seconds duration per one seismic event. The recorded data can be analyzed by any computer system through an interface device.

In addition to in situ data compression, the ADPCM encoding technique which uses the computer simulation makes it possible to treat seismic wave signals under low bit rate without losing the quality of the original wave signals.

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Appendix 1. FORTRAN program of ADPCM four bits encoding.

```
      SUBROUTINE CADCM(IVOCE,N,DMIN,DMAX,ICODE)
      DIMENSION IVOCE(N),ICODE(N)
      DIMENSION DD(8)
      DD(1)=0.9
      DD(2)=0.9
      DD(3)=0.9
      DD(4)=0.9
      DD(5)=1.2
      DD(6)=1.6
      DD(7)=2.0
      DD(8)=2.4
      DLT=DMIN
      ICO=FLOAT(IVOCE(1))/DLT
      ICABS=IABS(ICO)+1
      IF(ICABS-8) 20,20,10
   10 ICABS=8
   20 IF(ICO)30,40,40
   30 ICODE(1)=-ICABS
      IPRID=DLT*FLOAT(ICODE(1))
      GO TO 50
   40 ICODE(I)=ICABS
      IPRID=DLT*FLOAT(ICODE(1))
   50 DLN=DLT*DD(ICABS)
      IF (DLN-DMIN)38,38,34
   34 IF(DLN-DMAX)36,36,38
   36 DLT=DLN
   38 DO 200 K=2,N
      ICO=FLOAT((IVOCE(K)-IPRID))/DLT
      ICABS=IABS(ICO)+1
      IF(ICABS-8) 70,70,60
   60 ICABS=8
   70 IF (ICO) 80,90,90
   80 ICODE(K)=-ICABS
      IPRID=FLOAT(IPRID)+DLT*FLOAT(ICODE(K))
      GO TO 100
   90 ICODE(K)=ICABS
      IPRID=FLOAT(IPRID)+DLT*FLOAT(ICODE(K))
  100 DLN=DLT*DD(ICABS)
      IF(DLN-DMIN) 200,200,110
  110 IF(DLN-DMAX) 120,120,200
  120 DLT=DLN
  200 CONTINUE
      RETURN
      END
```

Appendix 2. FORTRAN program of ADPCM four bits decoding

```
      SUBROUTINE DADCM(IVOCE,N,DMIN,DMAX,ICODE)
      DIMENSION IVOCE(N),ICODE(N)
      DIMENSION DD(8)
      DD(1)=0.9
      DD(2)=0.9
      DD(3)=0.9
      DD(4)=0.9
      DD(5)=1.2
      DD(6)=1.6
      DD(7)=2.0
      DD(8)=2.4
      DLT=DMIN
      IVOCE(1)=FLOAT(ICODE(1))*DLT
      IDD=IABS(ICODE(1))
      DLN=DLT*DD(IDD)
      IF (DLN-DMIN)38,38,34
   34 IF (DLN-DMAX)36,36,38
   36 DLT=DLN
   38 DO 100 K=2,N
      IVOCE(K)=FLOAT(IVOCE(K-1))+DLT*FLOAT(ICODE(K))
      IDD=IABS(ICODE(K))
      DLN=DLT*DD(IDD)
      IF (DLN-DMIN)100,100,70
   70 IF (DLN-DMAX)75,75,100
   75 DLT=DLN
  100 CONTINUE
      RETURN
      END
```

25. ADPCM (適応差分 PCM) 法の地震波信号圧縮への応用

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地震波信号をディジタル化することにより波形精度が向上することは言うまでもないが，それと共にディジタルデータを伝送する回線及び収納する媒体の容量の増大化は著しく，データの圧縮が要求される．

これまで地震波信号の伝送等に用いられている対数圧縮は簡単ではあるが波形の再現性はあまり良くない．データ圧縮の技術は音声合成法の発展により急速に進歩しつつあり，種々の方法が試みられている．多くの圧縮法のなかでも，ADPCM (適応差分 PCM) 法は圧縮度は PARCOR 法，アダマール・ウォルシュ変換に比べ高くないが，地震波信号を扱う場合に最も重要な波形の再現性は高い．ADPCM 法は波形信号そのものを数値化するものではなく，前のデータ値との差 (増加又は減少分) だけを数値化したものを最適化し，効率の良い変換を行う方法である．

ADPCM 用 CMOS—IC (OKI—MSM-5218) 評価用機器を用いて ADPCM 法による地震波信号を圧縮する試みを行った．また，FORTRAN プログラムによるソフトウエアからの検討も行った．実験は堂平微小地震観測所，及び海底地震計で観測された地震波信号を 12 bit, 200 Hz でサンプリングしたデータを 3 bit (1/4 圧縮)，及び 4 bit (1/3 圧縮) に圧縮伸張を施した．その結果，4 bit については通常の地震波信号はほぼ忠実に再現され，3 bit についてもノイズ除去のためのフィルターを併用すれば地震波信号の伝送・収録に充分実用出来る事が確認出来た．