

A NOVEL ERROR - CONCEALABLE DIGITAL TRANSMISSION  
SYSTEM WITH HIGHER QUALITY AND MORE BIT - REDUCTION

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**AN AUDIO ENGINEERING SOCIETY PREPRINT**

A NOVEL ERROR - CONCEALABLE DIGITAL TRANSMISSION SYSTEM  
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A number of systems have been proposed for PCM transmission and storage by waveform coding, including Near-Instantaneous Companding PCM (NIPCM) {Refs. 1, 2, 3}, Differential PCM (DPCM) {4} and Delta Modulation (DM) {4}. Unfortunately, none of these systems offer both high sound quality and high efficiency. Let's briefly examine DPCM and NIPCM. DPCM has two advantages. First because it transmits differential data, it is relatively efficient and requires fewer transmission bits than when transmitting an original signal. Second, it is able to transmit low-frequency signals with relatively high performance. The problem with DPCM is that the differential data for high frequencies is greater than that for low frequencies making it impossible to transmit all the signals within a given frequency band. NIPCM was originally developed by the British Broadcasting Corporation (BBC). It features wide frequency response and wide dynamic range, and therefore is said to be suitable for high-quality sound transmission {3}. A proposal has also been made to improve the SNR of this system at low frequencies by the application of the CCITT pre-emphasis curve {1}.

Combining the best parts of DPCM and NIPCM seems like a natural step. Its realization however has been hindered by the following problems:

(1) Due to bits that are removed during the compression of differential data, a receiver suffers from DC shift, which makes reproduction practically impossible.

(2) When a number of signals are transmitted, a signal corresponding to the removed bits may not be totally transmitted, or it may suffer from excessive distortion. We solved these problems using a method we call "accumulating of removed bits", and thereby developed a PCM transmission system that provides both high sound quality and high efficiency. We have called this system "DC-PCM" {5, 6, 7}. In its original form, there was a problem with DC-PCM which was a DC shift in the integrator of the receiver when there was an error in transmitted

data. We have solved this problem with an error concealment method based on two essential circuits:

(1) A non-perfect differentiator which converges the propagation of error by keeping coefficients of both the encoder's differentiator and the decoder's integrator less than 1.

(2) A rough refresher which prevents error propagation by grouping a predetermined number of samples as a block, transmitting rough data in block form, and then rough-refreshing it.

The refresh data is rough rather than full, because the latter involves the transmission of the entire 16 bits of linear data, which results in low transmission efficiency. In the rough refresh we employed, a shorter bit word is used than is used with linear data, yet it provides enough accuracy without any adverse effect on the reproduced signal.

In the new DC-PCM system we are proposing in this paper, the error concealment function is added to the original DC-PCM, forming a system that offers both high sound quality and high efficiency. In the following paragraphs we will discuss the operating principle and signal format of the new DC-PCM system, as well as its characteristics and performance based on measurements of data error inherent in a test transmission system we set up.

### 3. Configuration of the original DC-PCM system

The original DC-PCM system consisted of an encoder and decoder, as shown in Fig. 1. The encoder consists of a digital converter (LPF and ADC) to sample a sound signal and quantize it into linear data, a differentiator to obtain the difference between the present data and the previous one, a NIPCM encoder to compress bits, and an accumulator to accumulate bits removed in the encoder.

The encoder groups a predetermined number of samples into a block and calculates the scale value from the maximum value -- an absolute value -- of the differential data contained in each block. Based on this scale value the encoder compresses the differential data for the block and accumulates the removed bits, and combines them as transmission data.

The decoder is composed of a NIPCM decoder to expand the received transmission data, an integrator to convert differential data into linear data and an analog converter (DAC and LPF) to change digital data into sound signals.

Let's look at the operation of the accumulator that accumulates removed bits generated during the compression of data, an operation that is an important function of the new DC-PCM system described above.

#### ● Operation of the accumulator

Fig. 2 shows the operation of an accumulator in the DC-PCM system. In this system, the differential data of the original 16 bits is compressed by the action of NIPCM to obtain 8-bit compressed data and removed bits. The accumulation is achieved by adding accumulated data from the previous data to the removed bits. The added data then serves as the next accumulated data. When a carry occurs due to an overflow in the addition by the accumulator, it is added to the compressed data as part of the transmission data. Should the sign of the compressed data reverse when the data is added to the carry, 1 will be added to the scale value for the block. Because of the action of the accumulator, the transmission data thus obtained contains removed bits, though the lower bits are truncated.

The accumulator in the new DC-PCM system has the following advantages:

1) Reproduced signals do not suffer DC fluctuation or DC shift.

● 2) During transmission of high-frequency and low-frequency signals whose differential data differ widely in level, low frequencies that are completely removed can be recreated.

#### 5. New DC-PCM system

When compared with the original DC-PCM system, the new DC-PCM system in Figure 3 has a different differentiator and integrator, and an additional error concealment block including a non-perfect differentiator and integrator. Fig. 4 shows its transmission data format.

### 5.1. Transmission data format

The data in the new DC-PCM system is composed of frames in which each comprises a predetermined number of blocks. The block format (Fig. 4A) consists of a scale and a predetermined number of transmission data -- the same format as used in the original DC-PCM system. The frame format (Fig. 4B) contains the same frame as used in the original DC-PCM system, plus additional rough data. Rough data corresponds to the upper 8 bits of linear data. In the original DC-PCM system, the original data from which the compressed data is derived is obtained by perfect differentiation, but in the new system, it is obtained by non-perfect differentiation.

### 5.2. Error concealment block

The error concealment block in the encoder (Fig. 5) is comprised of a non-perfect differentiator and a rough data generator. Fig. 6 shows the same block but in the decoder; it consists of a non-perfect integrator and a rough refresher. Functions and operation of each block are as follows.

#### 5.2.1. Non-perfect differentiator and non-perfect integrator

A non-perfect differentiator obtains the difference between the present data and the previous data which is multiplied by the differentiator coefficient, which is smaller than 1. In actual multiplication by hardware, the result has a limited bit length; therefore, a least significant product (LSP) that occurs during multiplication is accumulated to produce a most significant product (MSP). In the same manner, the non-perfect integrator obtains the MSP data by first multiplying the previous non-perfect integrated data by an integrator coefficient (which is less than 1) and then by accumulating the LSP that is generated during multiplication. To the MSP data we add the present linear data in order to generate non-perfect integrated data. As the result of the non-perfect integration, a DC shift due to data errors in the transmission system is rapidly suppressed according to the integrator coefficient, preventing DC shift from accumulating over a long period of time. By making the coefficient for the differentiator the same as the one for the integrator, it is possible to obtain a reproduced signal which is a replica of the input signal.

## 5.2.2. Rough refresher

As a result of the data format shown in Fig. 4, the decoder can check the DC shift that occurs due to data error in the transmission system by comparing the non-perfect integrated data with the rough data. If the rough data is correct, and the reproduced data suffers a DC shift which is beyond the precision range of the rough data, the rough refresher is activated. It replaces the non-perfect integrated data with rough data, thus keeping the DC value within the error range of the rough data. In this way, any DC shift in the reproduced signal is quickly suppressed by non-perfect integration, and instantaneously eliminated by rough refresh.

## 6. Measuring the new DC-PCM

We measured the performance of the new DC-PCM system using a test system with the following transmission parameters.

Conversion system	16-8-16
Differentiator/integrator coefficient	$1-1/2^8$
Sampling frequency	48kHz
Block length	16 samples
Frame length	64 blocks

### 6.1. DC suppression by non-perfect integration

Fig. 7 shows the result of the DC suppression effect of the system by supplying to the decoder an impulse whose value is equivalent to the maximum value an error can have.

### 6.2. DC suppression by rough refresh

Fig. 8 shows the effect of rough refresh by using the same input as used for Fig. 7 and setting the rough data value to zero.

### 6.3. SNR characteristics

Fig. 9 shows the system's SNR vs. input level, and Fig. 10, SNR vs. frequency.

## 7. Conclusion

We have established that by using non-perfect integration and rough refresh, we can solve the problem of DC shift due to data error in a

transmission system. The new DC-PCM system is a practical digital transmission system that offers both high quality sound and high efficiency. We believe that the new DC-PCM system is suitable for high-quality music and speech transmissions and that it can be put to practical use for digital transmission and storage systems such as studio to transmitter links (STL), microwave channels, satellite broadcasting and digital audio tape recording.

#### 8. Acknowledgment

The authors would like to thank H. Suzuki, M. Kato, and H. Kanzaki the company's central research laboratory for their assistance during the development of this system.

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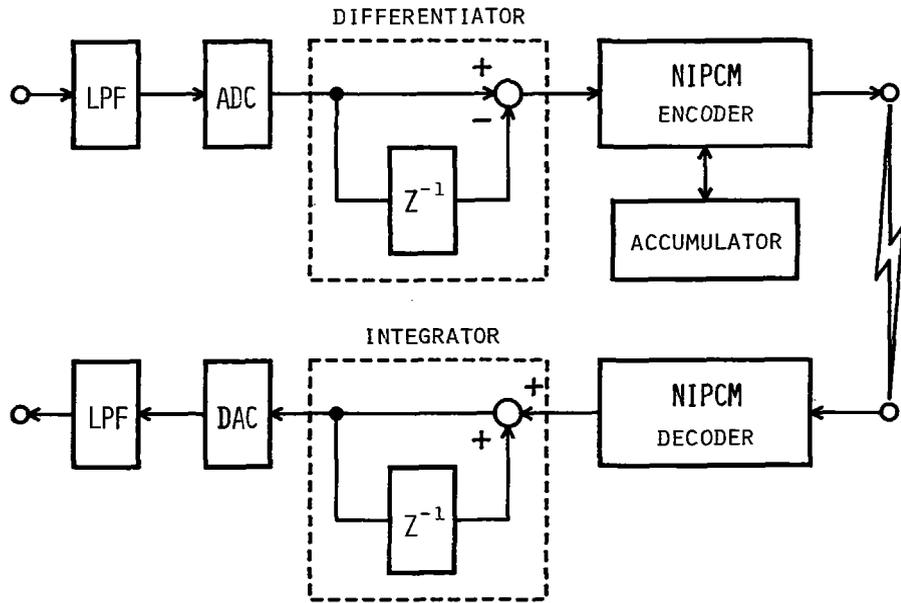


FIG. 1 BLOCK DIAGRAM OF THE ORIGINAL DC-PCM SYSTEM

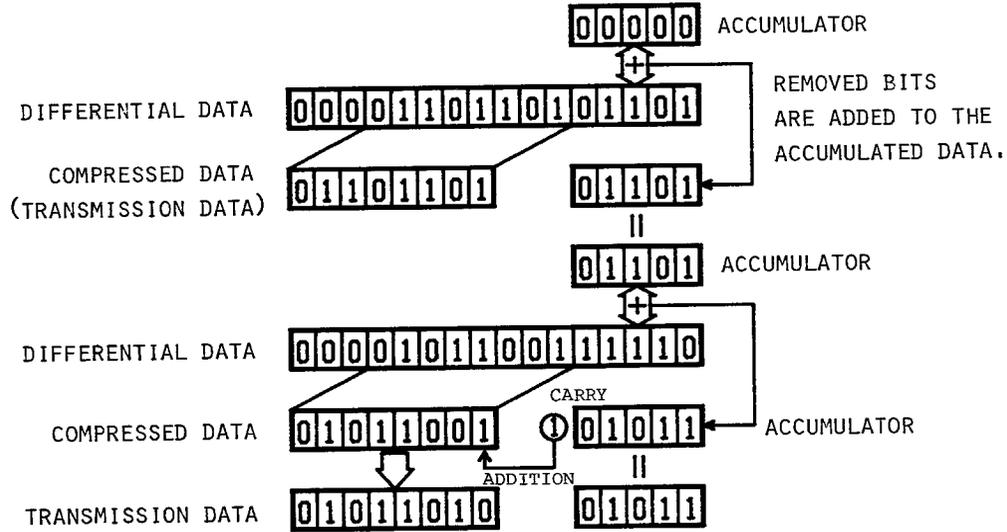


FIG. 2 OPERATION OF THE ACCUMULATOR

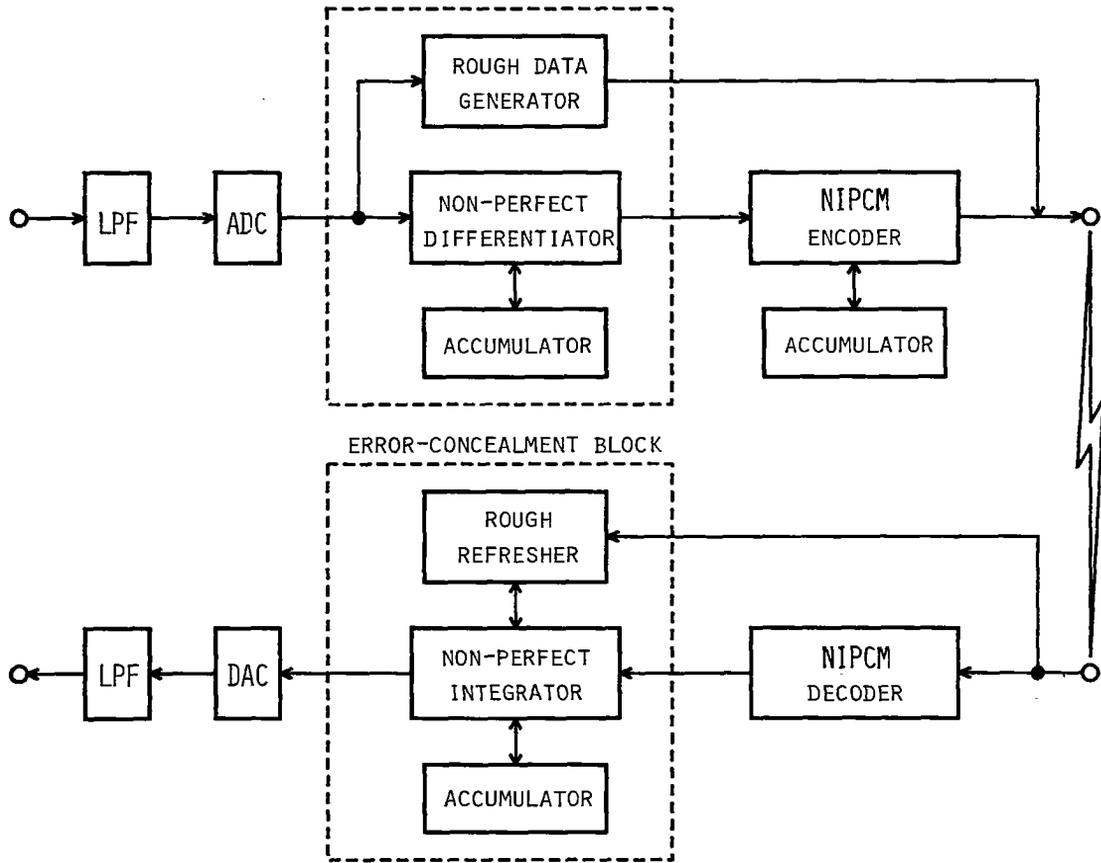
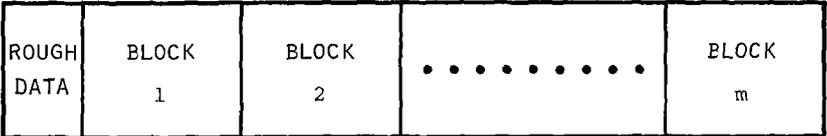


FIG. 3 BLOCK DIAGRAM OF NEW ERROR-CONCEALABLE DC-PCM SYSTEM



(A) BLOCK FORMAT



(B) FRAME FORMAT

FIG. 4 TRANSMISSION DATA FORMAT

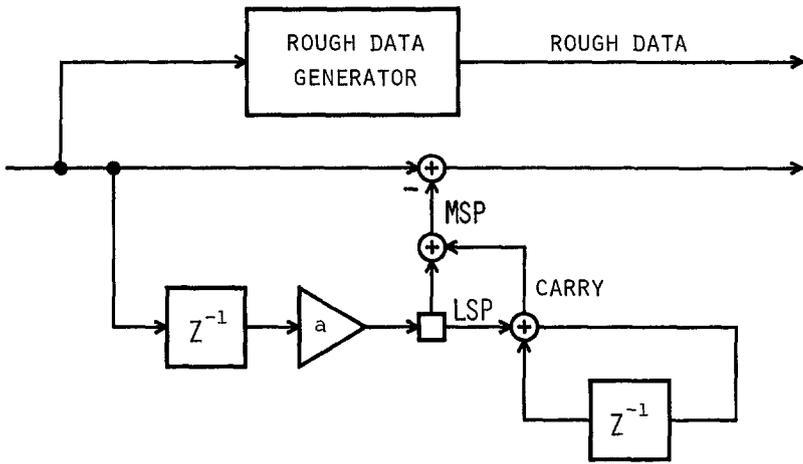


FIG. 5 BLOCK DIAGRAM OF ERROR-CONCEALMENT (ENCODER)

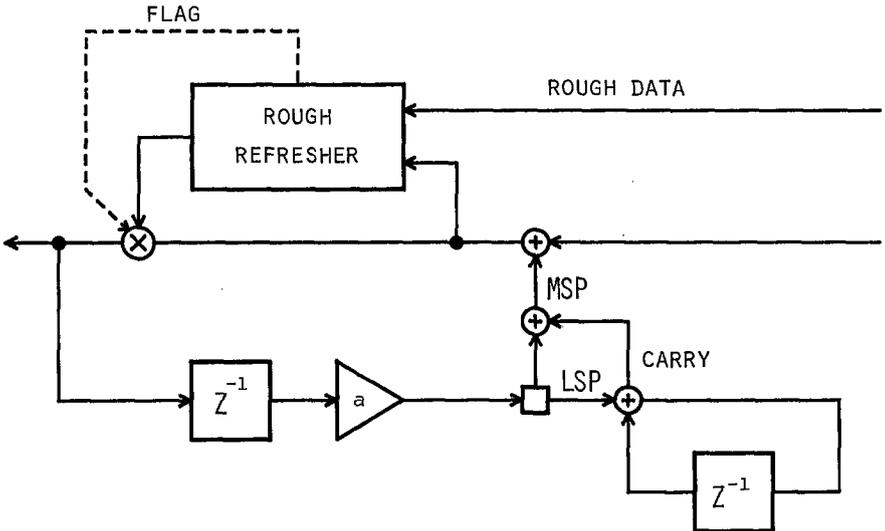
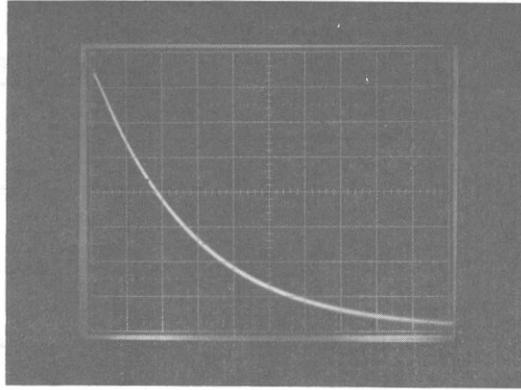
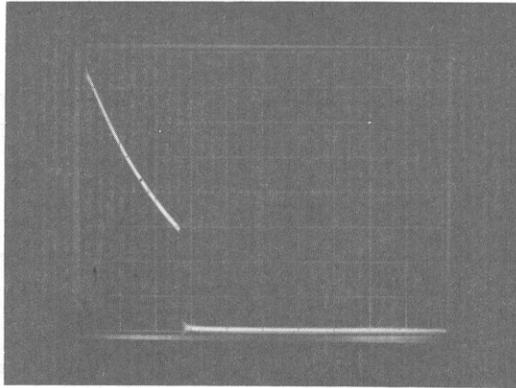


FIG. 6 BLOCK DIAGRAM OF ERROR-CONCEALMENT (DECODER)



VERTICAL: 0.2 V/Div.  
HORIZONTAL: 2 ms/Div.  
INTEGRATOR COEFFICIENT:  $1-1/2^8$

FIG. 7 IMPULSE RESPONSE (NON-PERFECT INTEGRATION)



VERTICAL: 0.2 V/Div.  
HORIZONTAL: 2 ms/Div.  
INTEGRATOR COEFFICIENT:  $1-1/2^8$

FIG. 8 IMPULSE RESPONSE (ROUGH REFRESHMENT)

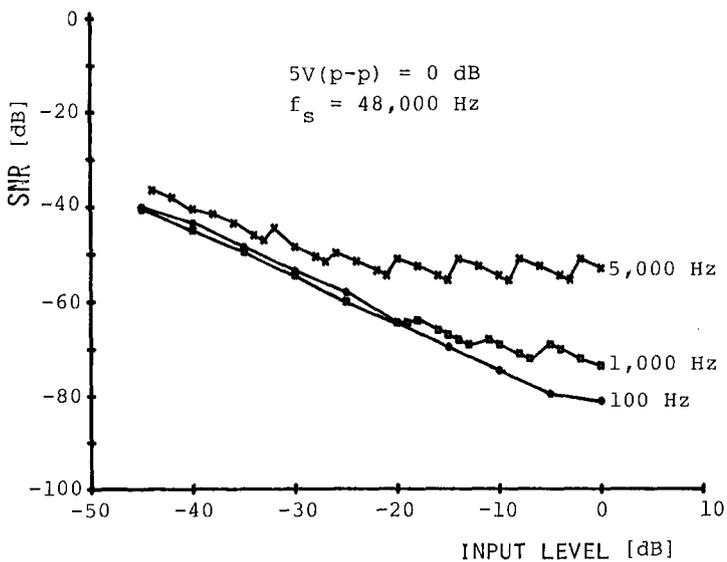


FIG. 9 SNR vs. INPUT LEVEL (MEASURED)

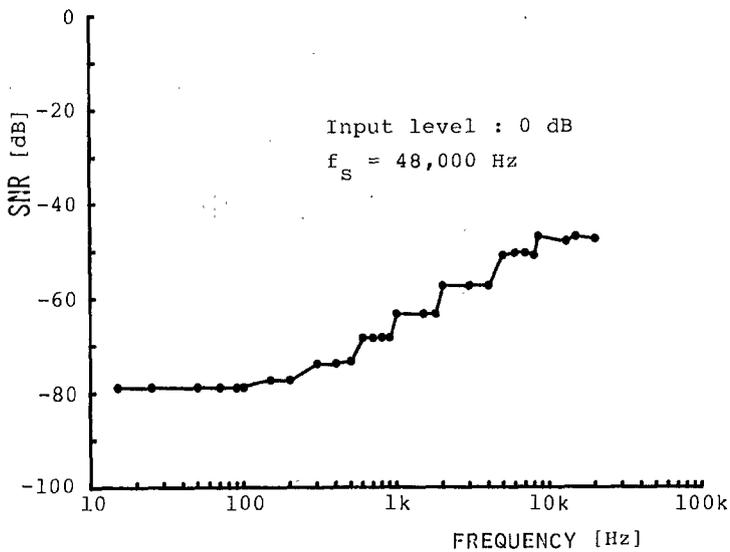


FIG. 10 SNR vs. FREQUENCY (MEASURED)