THE BELL SYSTEM TECHNICAL JOURNAL
Vol. 58, No. 3, March 1979
Printed in U.S.A.

DPCM with Forced Updating and Partial Correction of Transmission Errors

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(Manuscript received September 28, 1978)

A speech signal is dpcm (differential pulse code modulation) encoded at the sampling rate f_s and also pcm encoded at a rate f_s/W. Blocks of W dpcm words and one pcm word are transmitted. The receiver compares the decoded dpcm signal at the end of every block with the decoded pcm sample. If the difference is above a threshold, it is assumed that one error exists within the block, and a search is made for the erroneous dpcm code word. Correction is accomplished by inverting the bits in this code word until the difference is below the threshold. Whether or not the error is corrected, the dpcm signal at the end of the block is forced to the value of the pcm sample, thereby preventing error propagation outside the block. The system improves segmental s/n ratio by 7 dB for $W = 64$ and bit error rates between 0.1 and 0.5 percent. Larger improvements are available with smaller block sizes. In the absence of transmission errors, there is no perceptible distortion due to the correction system.

I. INTRODUCTION

We describe a new method of protecting dpcm (differential pulse code modulation) speech signals against the effects of transmission errors. The dpcm bit stream is divided into blocks, and one pcm sample is transmitted with each block. At the receiver, the appropriate sample of the integrated dpcm signal is compared with the pcm sample. A disparity between the two samples is evidence of a transmission error within the block.

When an error is thus detected, the dpcm integrator is reset to the value of the pcm sample and an algorithm is invoked to locate the error within the block. When the algorithm is successful, the transmission error is completely corrected. Even when the algorithm is unsuccessful, the resetting of the integrator at the receiver prevents the error from propagating outside the block in which it occurs.

This approach to error protection is different in spirit from conventional channel coding aimed at protecting a digital information stream regardless of its nature. Our method is directly keyed to the dpcm character of the message. Because it introduces its own redundancy to a dpcm signal, it is more powerful than the DDC (difference detection and correction) system described by the authors.[1,2] DDC is implemented at the receiver only and infers transmission errors from anomalies within the integrated dpcm sample sequence.

Other authors have reported on the periodic transmission of pcm code words in a dpcm picture coding system.[3] The pcm samples were used to update the receiver integrator and thereby curtail visible streaks caused by dpcm transmission errors. The use of pcm samples for error detection and correction is new to this paper.

II. SYSTEM DEFINITION

2.1 The transmitter

The transmitter (Fig. 1) sends one block of data every W sampling intervals. The data consist of the W code words of a conventional dpcm encoder plus one pcm code word formed by quantizing the input sample at the end of the block. In our implementation, the pcm and dpcm samples are formed by the same quantizer. As a consequence, for intermediate and high level inputs, the pcm signal is unable to code the entire range of signal amplitudes. Therefore, at the receiver, the error control mechanism is disabled when the pcm quantizer is overloaded so that errors affecting high amplitude speech samples go uncorrected and are allowed to propagate beyond the blocks in which they occur. We accept this penalty in order to derive the convenience

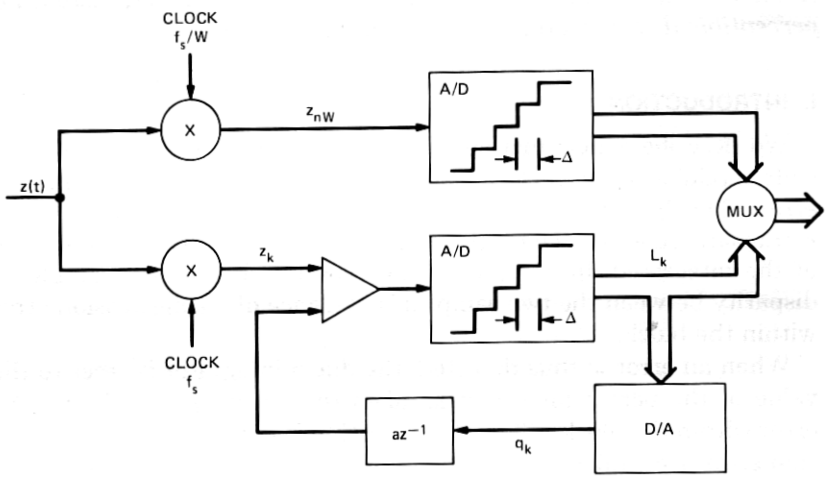

Fig. 1—Transmitter. The dpcm and pcm bit streams are multiplexed to form the transmitted sequence. The symbol rate is $f_s(W+1)/W$ Hz.

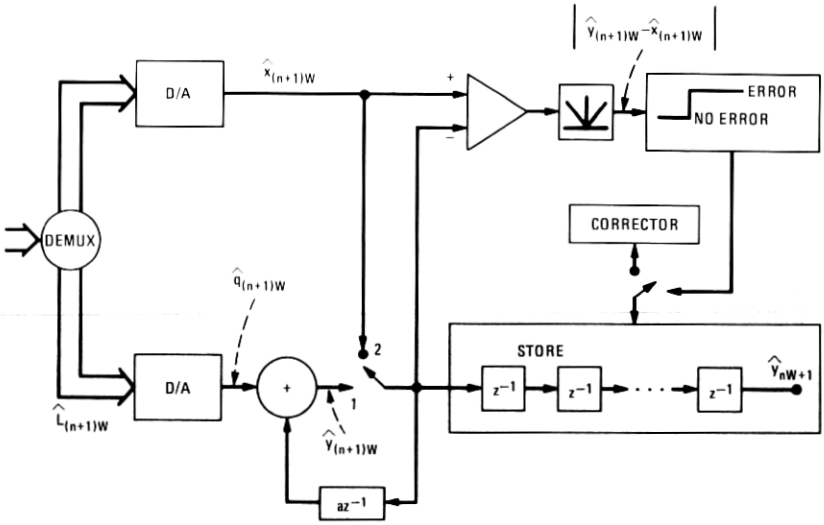

Fig. 2—Receiver. If, at the end of the block, the pcm sample and the dpcm sample differ by more than Δ, the correction logic is introduced. The switch in the integrator is in position 1 for W samples and goes to position 2 at the end of each block provided $\hat{x}$ is not an extreme value.

of a system with one quantizer, because high amplitude samples occur with relatively low probability in speech, and because we anticipate that errors in the idle channel and low amplitude regions of the signal are the most damaging ones subjectively.

2.2 The receiver

During reception of the nth block of data (Fig. 2), the integrator generates W dpcm output words $\hat{y}_{nW+j}$, from the received dpcm sequence $\{\hat{q}_k\}$ according to:

$$\hat{y}_{nW+j} = a\hat{y}_{nW+j-1} + \hat{q}_{nW+j}, \qquad j = 1, 2, \cdots, W. \tag{1}$$

These W samples are stored for possible revision by the error correction algorithm. At the end of the block, $\hat{y}_{(n+1)W}$ is reset to the pcm sample $\hat{x}_{(n+1)W}$ if $\hat{x}_{(n+1)W}$ is not the most positive or the most negative code word. When $\hat{x}_{(n+1)W}$ is at an extreme value, $\hat{y}_{(n+1)W}$ remains set at the value computed from (1).

2.2.1 Error detection

Provided $\hat{x}_{(n+1)W}$ does not indicate quantizer overload, it is compared with $\hat{y}_{(n+1)W}$ derived from (1). If these two samples differ by more than Δ, the quantizer step size, an error in the nth block is inferred and a search for the error is initiated. Otherwise the samples

$$\hat{y}_{nW+1}, \hat{y}_{nW+2}, \cdots, \hat{y}_{(n+1)W} \tag{2}$$

are sent to the system low-pass filter, $\hat{y}_{(n+1)W}$ is reset to $\hat{x}_{(n+1)W}$, and block $n + 1$ is processed.

2.2.2 Error correction

When $|\hat{x}_{(n+1)W} - \hat{y}_{(n+1)W}| > \Delta$, the system scans the samples in its buffer and finds the largest sample-to-sample difference in the block. That is, it computes

$$\delta_j = |\hat{y}_{nW+j} - \hat{y}_{nW+j-1}|, \qquad j = 1, 2, \cdots, W$$

and determines the maximum δ_j. If this maximum occurs at the rth position in the block, the corrector modifies $\hat{q}_r$ by successively inverting bits in the code word $\hat{L}_r$. With each bit inversion, $\tilde{q}_r$, a version of $\hat{q}_r$, is formed and (1) is iterated to produce a new trial value of $\hat{y}_{(n+1)W}$, which we call $\tilde{y}_{(n+1)W}$. Thus we have

$$\tilde{y}_{nW+r} = a\hat{y}_{nW+r-1} + \tilde{q}_r$$

and

$$\tilde{y}_{nW+j} = a\hat{y}_{nW+j-1} + \hat{q}_{nW+j}, \qquad j = r + 1, r + 2, \cdots, W.$$

Now, if $\tilde{y}_{(n+1)W}$ satisfies the test,

$$|\tilde{y}_{(n+1)W} - \hat{x}_{(n+1)W}| \leqslant \Delta,$$

the modified sequence

$$\hat{y}_{nW+1}, \cdots, \hat{y}_{nW+r-1}, \tilde{y}_{nW+r}, \cdots, \tilde{y}_{(n+1)W}$$

becomes the system output. Otherwise, a new value of $\tilde{q}_r$ is obtained by inverting another bit in $\hat{L}_r$.

If no single-bit inversion in $\hat{L}_r$ succeeds in bringing $\tilde{y}_{(n+1)W}$ sufficiently close to $\hat{x}_{(n+1)W}$, the correction attempt ceases and the samples in (2) are sent to the output filter.

Clearly, this correction scheme is effective only when there is one bit in error in a block of W samples and this single error leads to a large sample-to-sample difference. However, even when there is more than one error in the block, the updating of the integrator signal prevents long-term propagation of error effects.

Errors in pcm samples induce distortions that would not be present in ordinary dpcm. Our performance evaluations indicate that the effects of these distortions are substantially smaller than the benefits of error correction and integrator updating.

III. EVALUATION

The technique was evaluated by means of computer simulation of a 7-bit, 8-kHz single-integration dpcm system with prediction coefficient 0.9. To obtain the objective measurements displayed in Figs. 3 to 5, we repeatedly played a single sentence through the system: "I have two

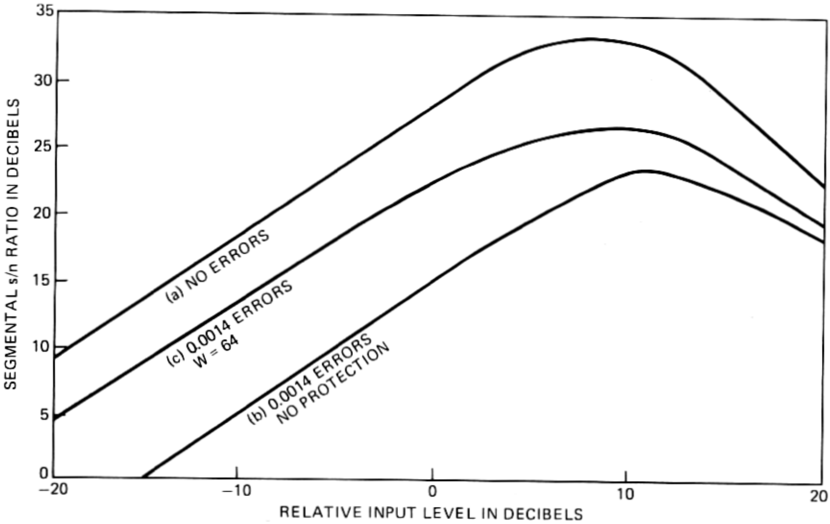

Fig. 3—Performance as a function of input level. (a) No transmission errors. Segmental s/n ratio reflects only quantizing noise. The curves with and without error protection virtually coincide. (b) Transmission error rate is 0.0014 and the system is unprotected. (c) Error rate is 0.0014 and protection block size is 64.

daughters, Lorna and Susan," spoken by a male. By listening to many samples of processed speech from a variety of talkers, we confirmed that the error protection indicated in these graphs is subjectively meaningful and not peculiar to a single utterance.

For each system configuration, we measured segmental signal-to-noise ratio,[4] defined as the decibel average of the signal-to-noise ratios in 214 speech segments, each of duration 16 ms. These are the segments (out of 224 in the 3.5-second utterance) in which the rms signal level exceeds −60 dB relative to the peak signal. Segmental s/n ratio is considered a better indicator of speech quality than ordinary s/n ratio. In a study of adaptive dpcm, there was a correlation of 0.93 between segmental s/n ratio and subjective ratings of speech quality.[5] The comparable correlation with ordinary s/n ratio was only 0.69.

Variables in our experiment were input signal level, block size W, and transmission error rate. The simulated channel introduces random errors to the serial bit stream consisting of W 7-bit dpcm code words and one 7-bit pcm code word per block.

Figure 3 shows segmental s/n ratio as a function of input level for a block size $W = 64$ and for unprotected dpcm. The top curve is for zero error rate, in which case s/n ratio is the same (to within 0.3 dB) with and without the error protection mechanism. Curve b shows the effect on dpcm of a channel with error rate 0.0014. In the granular noise region (signal level below the peak of curve a), these errors cause a degradation of about 13 dB in s/n ratio. Over most of this region, the

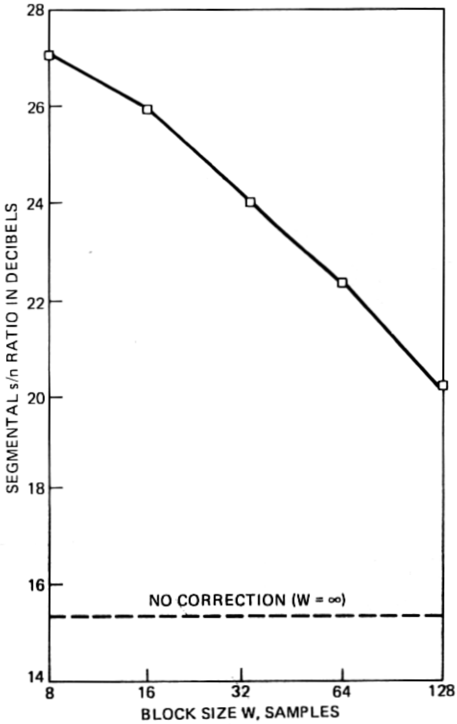

Fig. 4—The effect of block size on performance. The relative input level is 0 dB and the error rate is 0.0014. Over most of the range, s/n ratio decreases by about 2 dB per octave increase in block size.

protection system redeems about 9 dB of this loss. For reasons discussed in Section 2.1, the effectiveness of the error protection diminishes as signal level increases.

Figure 4 pertains to the 0-dB relative input level (see Fig. 3) and 0.0014 error rate. It shows the effect on segmental s/n ratio of varying the block size, W, over the range 8 to 128 samples. A small block size offers more protection but exacts a greater penalty in transmitted bit rate than a large block size. With $W = 8$ there is almost one extra bit per code word and an improvement of 11.5 dB in s/n ratio. The improvement decreases by about 2 dB per octave change in W over the range we investigated.

The effectiveness of the error protection as a function of channel quality is shown in Fig. 5 for the input level 0 dB and $W = 64$. Our error protection mechanism results in an s/n ratio improvement of about 6 to 8 dB at error rates between 0.001 and 0.01. For comparison, we also show the performance of a dpcm system protected by the DDC (difference detection and correction) scheme.[1,2] DDC requires no modification of the dpcm transmitter and no additional transmitted bits.

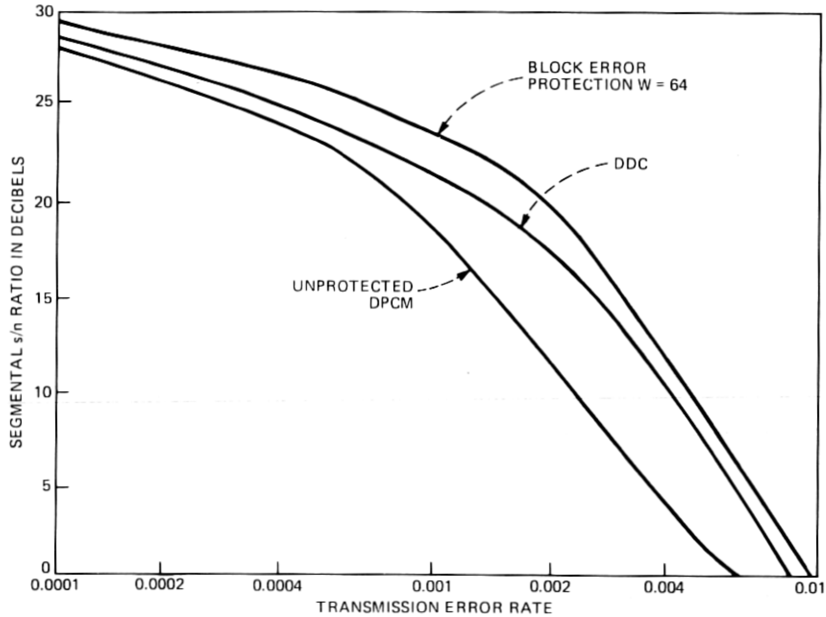

Fig. 5—Performance as a function of channel error rate, 0 dB relative input level. Here, the block protection scheme is compared with ordinary dpcm and with dpcm augmented at the receiver by DDC, a difference detection and correction system.

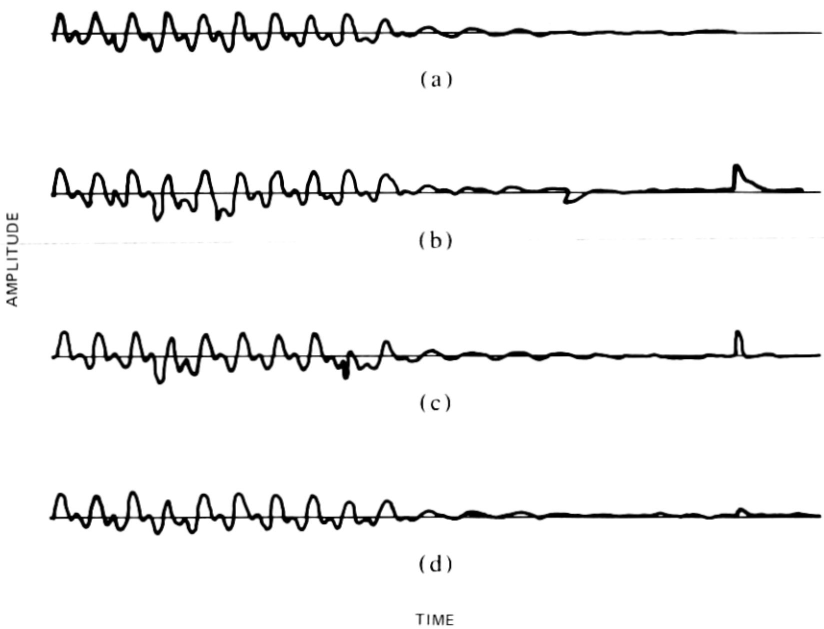

Fig. 6—Oscillograms of a segment of the test utterance. (a) Original speech, 0 dB relative input level. (b) dpcm output with 0.0042 error per bit. (c) After block protection with $W = 64$. (d) After selective smoothing of (c) by means of DDC.

Its performance, however, is about 2 dB poorer than that of the new scheme.

IV. REPROCESSING THE PARTIALLY CORRECTED SPEECH

The forced updating of the integrator signal itself causes sharp transients in the system output when a detected error cannot be corrected. Thus, the output signal often contains many spurious spikes. This phenomenon is illustrated by the speech waveforms in Fig. 6. A segment of the original speech signal is shown in Fig. 6a, and Fig. 6b shows the same segment corrupted by transmission errors (which occurred with probability 0.0042). Figure 6c shows the output of the protected dpcm system, which has suppressed most of the channel error noise but left a residual spike (click) in the signal. DDC[6] (as applied to pcm) is designed to smooth out such spikes and the effect of reprocessing the system output with DDC is seen in Fig. 6d. In general, appending DDC to the forced updating method is effective in suppressing residual impulse noice.

V. CONCLUSIONS

By periodically introducing pcm samples to a dpcm signal sequence, it is possible to reduce substantially the propagation of transmission errors in dpcm. Furthermore, at the cost of some delay, storage, and elementary signal processing in the receiver, many errors can be completely corrected. Small block sizes (frequent pcm transmissions) are more effective but introduce larger transmission rate penalties than large block sizes.

We have shown that this block protection scheme improves segmental s/n ratio of speech signals, and our experience of listening to several speech samples processed this way confirms that subjective quality is correspondingly enhanced. The method is also very appropriate to dpcm transmission of video signals in which error propagation causes very objectionable streaks in pictures.[3]

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