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Effects of Channel Errors on the Signal-to-Noise Performance of Speech-Encoding Systems

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The signal-to-noise ratios of different speech-encoding schemes have been measured in the case where the channel contains errors. Those types and probabilities of errors have been considered that are of interest for mobile telephone applications. Most encoding schemes use an adaptive three-bit quantizer with an explicit transmission of the step-size information. A scheme with an adaptive prediction algorithm has also been studied. It has been assumed in all cases that the side information about the quantizer step size and the predictor coefficients is transmitted in an error-protected format.

Measurements were made by simulating the coding schemes and the noisy channel on a digital computer. The results include upper bounds of the improvements that can be reached with error protection of the most significant bits.

I. INTRODUCTION

The suitability of digital coding and transmitting speech signals for mobile telephone systems is a question of current interest. In UHF systems, Rayleigh fading causes the carrier-to-interference ratio and the carrier-to-noise ratio to be low in frequent intervals. This leads to high bit-error probabilities in the transmission of the coded signal; the errors occur in bursts.

This paper compares the effects of channel transmission errors on the objective signal-to-noise ratio (s/n) for different encoding schemes. Most of these schemes use an adaptive quantizer with time-varying step sizes or, equivalently, a time-varying gain control of an amplifier in front of a quantizer with fixed step sizes. The side information about the step size (or about the amplifier gain) is derived from stored samples of the input signal and has to be transmitted together with the message block of coded samples (adaptive quantization with forward estimation = AQF). These AQF schemes have an excellent idle channel performance, even in the presence of channel errors, if the side information can be transmitted in an error-protected format. Previous studies[1–3] always assumed a nonadaptive μ-logarithmic quantizer with its specific problems of allowable peak clipping and changing performance caused by different mean levels of the speech signal. The purpose of the present study is to show the s/n performance of some adaptive speech-encoding schemes suitable for mobile telephone applications. Our measurements were made by simulating the coding schemes and the noisy channels on a digital computer. The measurements include upper bounds for the s/n that can be reached by using error-protection schemes to reduce the effective channel error probability. We have studied independent, as well as clustered, channel error patterns. No attempt was made to study practical error-detection or error-correction schemes. However, the results allow us to predict the overall performance that can be reached with nonideal error-correction schemes. The s/n values of the coding systems with independently distributed channel errors were measured because it is possible to produce a nearly equivalent error pattern by scrambling the bit stream at the output of the transmitter.[6] Additionally, independent errors will be the primary impairment if the interference is low.

The organization of this paper is as follows. Section II discusses the dependence of the total s/n on the bit-error rate P, assuming gaussian-distributed quantizer input data. It has been shown[4] that the gaussian probability density function is a good approximation for signals occurring in speech AQF schemes. Both the natural binary code (NBC) and the folded (symmetrical) binary code (FBC) are considered. It is shown that the FBC code has a better s/n performance if channel transmission errors cannot be ignored. Section III discusses those speech-encoding systems used in this study, and Section IV considers the types of errors on the channel and gives the main results obtained by simulating the encoding schemes and the noisy channels on a digital computer. Comparison is made on the basis of the objective overall signal-to-noise ratio. Tape-recorded examples were used to compare the subjective and perceptual effects of signal-quantization and channel errors. Some conclusions are given in Section V.

A detailed comparison of various speech-encoding schemes on the assumption of an error-free transmission is described in another paper.[5] A companion paper by Jayant[6] gives numerous examples of the performance of practical error-protecting schemes in the presence of channel errors.

II. DERIVATION OF THE SIGNAL-TO-NOISE RATIO FOR GAUSSIAN SIGNALS

We calculate the overall signal-to-noise ratio of a three-bit PCM-quantization scheme on the assumption that a speech signal can be

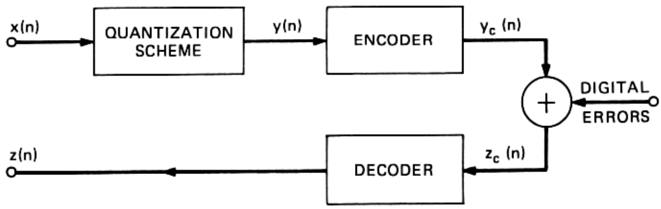

Fig. 1—Digital transmission system.

represented by a gaussian probability density function (PDF) at the quantizer input. This assumption is approximately valid if adaptive quantization with forward estimation is applied to the speech samples and if the message blocks are not too long.[4] Figure 1 shows a block diagram of the system under consideration. The quantizer with $M = 8$ steps (and $m = \log_2 M$ bits per code word) maps each input sample $x(n)$ into one of a set of eight rational numbers $y(n)\epsilon\{v_k\}_{k=1,2,\cdots,M}$. The representation level v_i is chosen if $u_{i+1} \geqq x(n) > u_i$, as illustrated in Fig. 2. The index i of the input symbol v_i of the transmission system is transmitted to the receiver in a binary format [binary code word $y_c(n)$]; the received code word $z_c(n)$ is interpreted as one of the eight output symbols $z(n)\epsilon\{w_k\}_{k=1,2,\cdots,M}$. We obtain a change $\delta_{ij} = |v_i - w_j|$ in amplitude if the transmitted quantizer index i is changed to j because of channel errors (Fig. 3). The total mean-squared error is

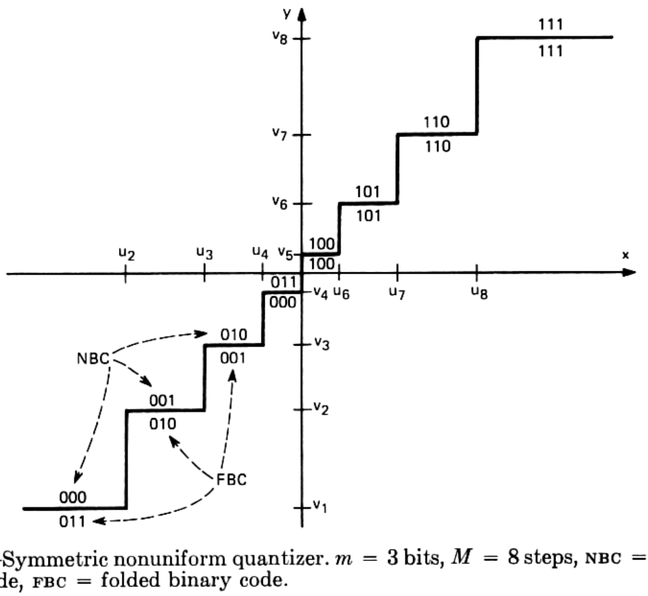

Fig. 2—Symmetric nonuniform quantizer. $m = 3$ bits, $M = 8$ steps, NBC = natural binary code, FBC = folded binary code.

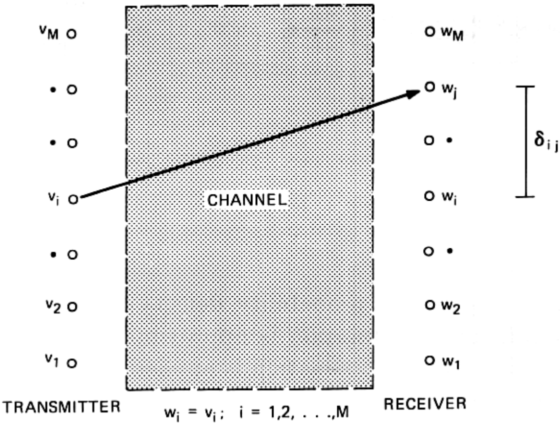

Fig. 3—Channel transmission errors.

given by

$$\begin{aligned}\epsilon_t^2 &= E[x(n) - z(n)]^2 \\ &= \epsilon_q^2 + \epsilon_c^2,\end{aligned} \tag{1}$$

where the quantization error ϵ_q^2 and the channel error ϵ_c^2 are given by

$$\epsilon_q^2 = E[x(n) - y(n)]^2 \tag{2}$$

$$\epsilon_c^2 = E[y(n) - z(n)]^2. \tag{3}$$

Equation (1) is only true on the assumption of a vanishing correlation between quantization error and channel error;[7] this is the case if the quantizer structure is that of Max[8] (these quantizers lead to a maximum s/n performance for a given PDF).

The mean-squared error caused by digital line errors is

$$\epsilon_c^2 = \sum_{i=1}^{M} \sum_{j=1}^{M} P(v_i, w_j) \cdot \delta_{ij}^2. \tag{4}$$

With

$$P(v_i, w_j) = P(v_i) \cdot P(w_j/v_i), \tag{5}$$

we have

$$\begin{aligned}\epsilon_c^2 &= \sum_{i=1}^{M} \sum_{j=1}^{M} P(v_i) \cdot P(w_j/v_i) \cdot \delta_{ij}^2 \\ &= \text{trace}\,\{\mathbf{P}_v \cdot \mathbf{P}_c \cdot \boldsymbol{\delta}^2\},\end{aligned} \tag{6}$$

where

$P(v_i, w_j)$ is the joint probability of an input symbol v_i at the transmitter and an output symbol w_j at the receiver,

$\delta_{ij} = \lvert v_i - w_j \rvert$	is the amplitude of the error occurring if the input symbol v_i has been chosen at the transmitter and if the output symbol w_j has been interpreted at the receiver,
$P(v_i)$	is the probability of input symbol v_i occurring,
$P(w_j/v_i)$	is the conditional probability that the output symbol w_j will be received if the input symbol v_i is sent,
$\mathbf{P}_v$	is a diagonal matrix with elements $P(v_i)$,
$\mathbf{P}_c$	is the channel transition matrix with elements $P(w_j/v_i)$, and
$\boldsymbol{\delta}^2$	is the matrix of squared error amplitudes with elements δ_{ij}^2.

The conditional probabilities $P(w_j/v_i)$ can be calculated easily if the bit errors on the channel are distributed independently. The values depend on the bit-error probability P and on the code. The probability $P(w_j/v_i)$ that w_j will be received if v_i is sent is just the probability that digital errors will occur in the D places where they differ and that no errors will occur in the $m - D$ remaining places,

$$P(w_j/v_i) = P^D(1 - P)^{m-D}, \tag{7}$$

where P is the bit-error probability on the channel, D is the Hamming distance between the code words representing the symbols v_i and w_j, and m is the number of bits per code word. The Hamming distance D depends on the code; two codes have been considered (see Fig. 2): (*i*) the natural binary code (NBC), (*ii*) the folded binary code (FBC). For this code, the most significant bit gives polarity information; the remaining bits represent the signal magnitude in natural binary code. For example, the transition from input symbol v_1 to output symbol $w_8 = v_8$ causes an error δ_{18} in amplitude. The Hamming distances are $D = 3$ and $D = 1$ for the NBC code and FBC code, respectively (see Fig. 2). Using (7) we get $P(w_8/v_1) = P^3$ with the NBC code and $P(w_8/v_1) = P(1 - P)^2$ with the FBC code.

The mean-squared error caused by digital line errors can also be calculated if (ideal) error-protection schemes are applied:

Scheme EP1: The most significant bit of each code word is perfectly error-protected.

Scheme EP2: The two most significant bits are perfectly error-protected.

The channel transition matrix $\mathbf{P}_c$ has to be modified in these cases because some elements of the matrix are zero then. These necessary modifications are not described in this paper. Using (1) and (6), we obtain the signal-to-noise ratio

$$\text{s/n} = 10 \cdot \log_{10} \frac{\sigma_x^2}{\epsilon_t^2} = 10 \cdot \log_{10} \frac{\sigma_x^2}{\epsilon_q^2 + \text{trace}\,\{\mathbf{P}_v \cdot \mathbf{P}_c \cdot \boldsymbol{\delta}^2\}}, \tag{8}$$

where

$$\sigma_x^2 = E[x^2(n)] \tag{9}$$

is the mean-squared power of the input signal. The normalized quantization noise variance ϵ_q^2/σ_x^2 has a value of 0.03451 if the structure of the three-bit quantizer is that of Max.[8] Using this value, the s/n performance has been calculated as a function of the bit-error rate P (Fig. 4). The two lower curves show that the FBC code outperforms the NBC code. The folded binary code has therefore been used in all simulations. The upper curves demonstrate the advantage of an (ideal) error protection of the most significant bit (EP1) and of the two most significant bits (EP2). It may be relevant to mention that this error protection leads to a reduction of the effective bit-error rate. To confirm the theoretical results, we have measured some s/n values with the simulation program that has been used for the study of the speech-

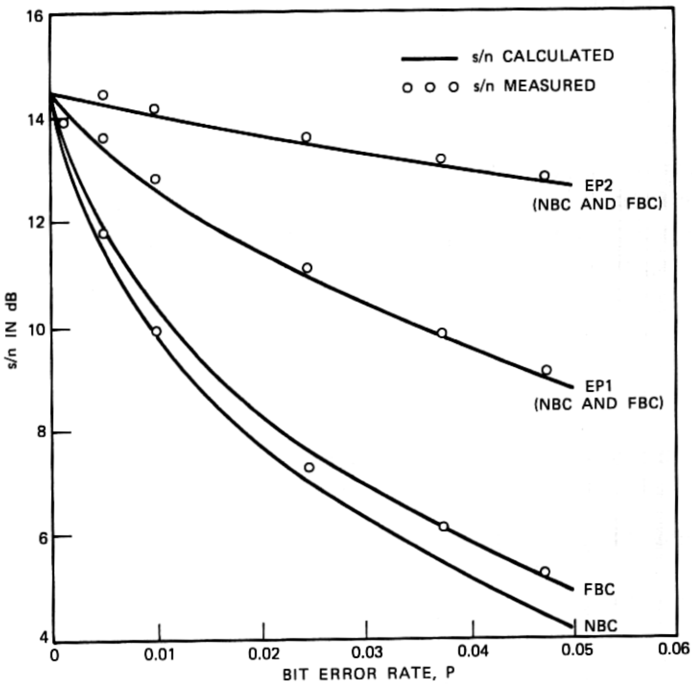

Fig. 4—Signal-to-noise ratio performance of a gaussian data transmission system. NBC = natural binary code, FBC = folded binary code, EP1 = error protection of most significant bit, EP2 = error protection of two most significant bits.

Table I — Channel coefficients of optimum three-bit Gauss quantizers. NBC = natural binary code, FBC = folded binary code.

	α_1	α_2	α_3
Optimum nonuniform quantizer			
NBC	6.9124	−2.7373	−0.3133
FBC	5.7445	−0.4015	−0.3133
Optimum uniform quantizer			
NBC	7.2113	−3.3611	0.
FBC	5.5672	−0.0128	0.

encoding systems. The results obtained with gaussian input data and the FBC code compare favorably with those determined using (8) (see Fig. 4).

The channel error variance ϵ_c^2 as given in (6) can be transformed into

$$\epsilon_c^2 = \sum_{j=1}^{m} \alpha_j \cdot P^j. \tag{10}$$

The coefficients α_j contain the total information about the effects of channel errors on the performance of an encoding-decoding scheme. Thus, different schemes can be compared easily. Table I lists the α_j coefficients of optimum uniform and nonuniform gaussian three-bit quantizers. Note that there are only very small differences between nonuniform and uniform quantizers. Note also that the FBC code should be chosen; it has a nearly 1-dB advantage over the NBC code if the bit-error rate is very high (see also Fig. 4).

In the discussions so far, a quantizer has been assumed that is optimum (in the sense of a maximum s/n) if the channel is error-free. A higher overall s/n performance can be reached, however, if the quantizer is reoptimized for a given channel transition matrix $\mathbf{P}_c$.[9] Figure 5 shows an example with a three-bit quantizer optimum for a bit-error rate of 0.025 and the NBC code. In fact, we get a better s/n performance if the bit-error rates P are high, but the shortcoming is the decrease in s/n for low P values; this reduction is approximately 2.3 dB in the error-free case ($P = 0$). The reoptimization of the quantizer can therefore only be of interest if the channel noise statistics are time-invariant, or if the s/n decrease for small error rates can be accepted. This will be the case if a quantizer with a higher number of step sizes is chosen.

The principal aim of this section was to show the calculation of the s/n of a (nonadaptive) PCM scheme if channel errors are present. These calculations can be extended to DPCM systems and to burst error

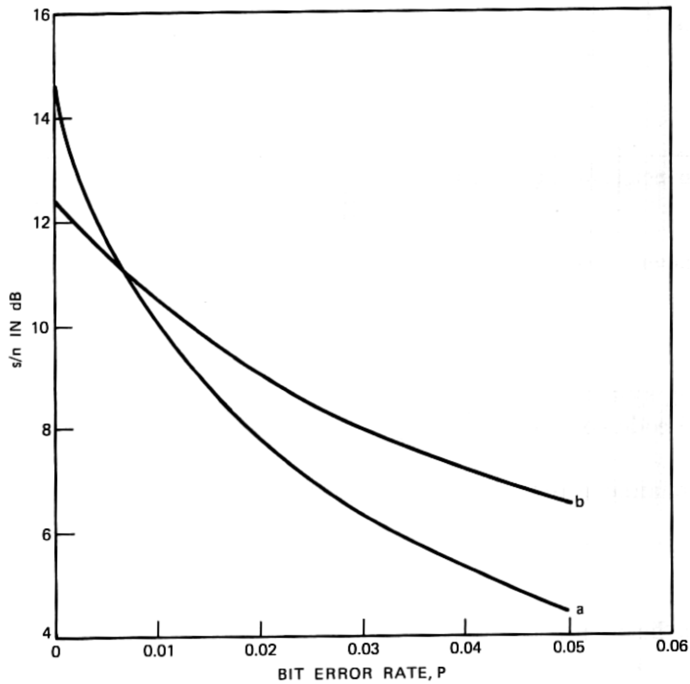

Fig. 5—Signal-to-noise ratio performance of two nonuniform gaussian three-bit quantizers. (a) Quantizer is optimized for an error-free transmission. (b) Quantizer is optimized for independent channel errors of rate $P = 0.025$. NBC code has been used in both cases.

channels. But it seems very difficult to find theoretical solutions for coding schemes that apply adaptive quantization and adaptive prediction strategies. Instead of looking for such solutions, we simulated different encoding schemes and channels on a digital computer and measured the overall s/n. We shall find some similarities to the results we obtained in this section. Particularly, we shall use the channel coefficients α_j to compare the performance of the coding schemes.

III. DESCRIPTION OF THE CODING SYSTEMS

To get a good quality of the coded speech with low bit rates, we have used PCM and differential PCM schemes that employ adaptive quantizers. Both nonadaptive prediction (DPCM) and adaptive prediction (ADPCM) have been applied. The advantage of adaptive quantization is that the quantizer is always adjusted to the highly variable variance of the speech signals. Thus, a better s/n performance is achieved.[4,10] DPCM and ADPCM schemes provide an additional s/n gain over PCM; this is especially true if the predictor responds to changes in the short-term spectrum of speech (ADPCM).[10,11] A nonadaptive logarithmic companded PCM has been included in our study because it very often serves as a standard reference in coder comparisons. A great number of speech-encoding schemes have been compared in a companion paper[5] on the basis of s/n as performance measure using the same speech signal employed in this paper. The effect of channel errors on s/n performance has been studied using the following encoding schemes:

Scheme 1: *PCM, nonadaptive* (*Fig. 6a*). The quantizer has a μ100 characteristic,[12] and the loading is four times the standard deviation of the speech signal to be quantized.

Scheme 2: *PCM-AQF* (*Fig. 6b*). Thirty-two samples of the input signal are buffered, and the maximum value of this block determines the gain of the amplifier in front of the quantizer (adaptive quantization

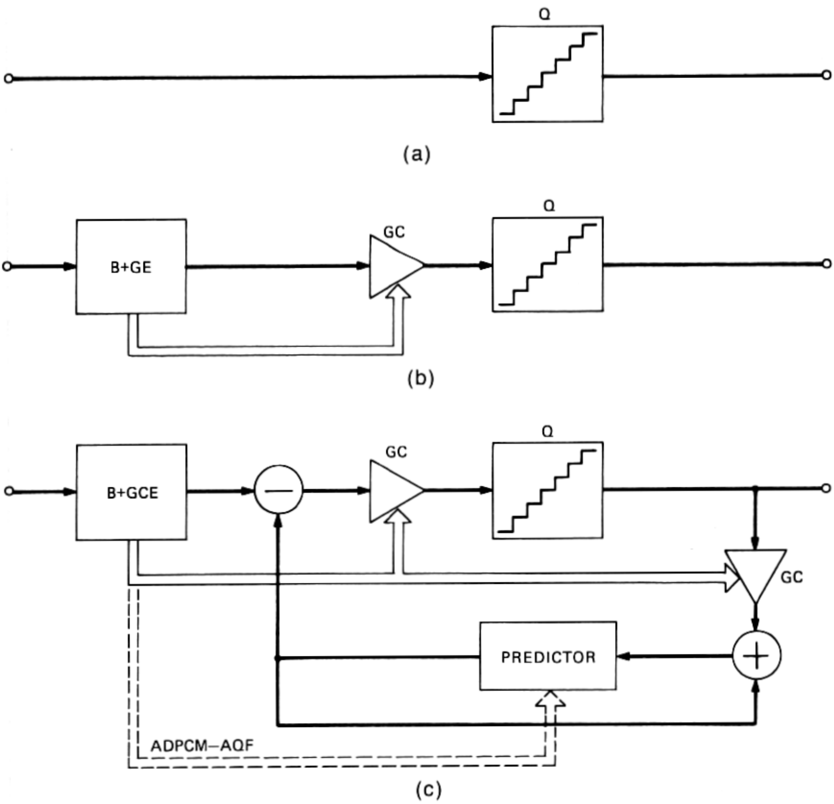

Fig. 6—Speech-encoding schemes. (a) Nonadaptive PCM. (b) PCM with adaptive quantization (PCM-AQF). (c) DPCM and adaptive DPCM (ADPCM) with adaptive quantization (DPCM-AQF and ADPCM-AQF). Q = quantizer, GC = gain control, B + GCE = buffer and gain and coefficients estimation.

with forward estimation = AQF). The characteristic of the quantizer is optimum for signals with a gaussian probability density.

Scheme 3: DPCM1-AQF (Fig. 6c). A predictor with one time-invariant coefficient is being used in connection with adaptive (forward estimation) quantization (AQF). Thirty-two samples of the input signal are buffered, and the maximum difference between neighbored samples determines the gain of the amplifier in front of the quantizer. The characteristic of the quantizer is optimum for signals with a gaussian probability density. The predictor coefficient that is optimum for the speech signal being used is $h_1 = 0.85$. Lower values lead to a better performance of the DPCM scheme in the case of high error probabilities; this will be shown in Section V.

Scheme 4: ADPCM1-AQF (Fig. 6c). In this adaptive prediction scheme, 32 samples of the input signal are buffered, the normalized short-term correlation coefficient between neighbored samples of this block is calculated, and the predictor coefficient is set to this correlation coefficient. The gain of the amplifier in front of the quantizer is determined by calculating an estimation value of the standard deviation of the difference signal; the amplifier gain is set to the inverse of this estimation value. The characteristic of the quantizer is optimum for signals with a gaussian probability density.

Scheme 5: ADPCM4-AQF (Fig. 6c). In this adaptive prediction scheme, four optimum predictor coefficients are calculated for each segment of 32 samples from the first values of the short-term autocorrelation function; see the description of Scheme 4 for further details.

The folded binary code (FBC) was used in all cases. It should be mentioned that the combination of controlled amplifier and fixed quantizer in the adaptive quantization schemes is equivalent to a quantizer with a step-size adaptation. Some adaptive quantization schemes that use the transmitted code words for the control of the amplifier gain (adaptive backward estimation) have also been studied. The simulations have shown that these schemes cannot be used for channels with high bit-error probabilities; the overall s/n turned out to be less than 0 dB in most cases.

IV. ERROR PERFORMANCE: RESULTS AND DISCUSSION

4.1 Simulation system and types of errors

The dependence of the overall signal-to-noise ratios of five speech coding schemes (see Section III) on the average bit-error probability P has been determined for different types of noisy channels. The s/n is given by

$$\text{s/n}(P) = 10 \log_{10} \frac{\sigma_x^2}{\epsilon_t^2}, \tag{11}$$

where ϵ_t^2 and σ_x^2 are defined in (1) and (9), respectively. The s/n values have been measured for bit-error rates of 0, 0.001, 0.0125, 0.025, and 0.05. The measurements were made by simulating the coding schemes and the noisy channels on a digital computer. Channels with independent, as well as correlated, error patterns have been studied. The statistically independent errors have been generated by using a pseudorandom noise generator program. Tape recordings with error patterns of actual fading signals have been used in the channel simulation of burst errors. The error patterns are typical for UHF mobile radio transmission. The statistics of these errors are described in Ref. 6. In all simulations, it has been assumed that it is possible to transmit the information about the gain of the amplifier (AQF scheme) and/or about the predictor coefficients (adaptive prediction) without any error. Increased signal-to-noise ratios can be reached for a given P value using error-detection and error-correction schemes. In studying these error-protected cases, it has been assumed that all errors are corrected. Practical schemes will not always be able to correct all errors. Therefore, the s/n values given in this paper for the error-protected case represent an upper bound on the performances of error-protecting techniques. Two types of error correction have been studied:

EP1: Protection of the most significant bit; this bit is the sign bit.
EP2: Protection of the two most significant bits. Only changes to neighbored output symbols are possible in this case (if the quantizer has eight step sizes).

A 2.3-second utterance ("the boy was mute about his task"; female voice; bandwidth 200 to 3200 Hz; sampling rate 8 kHz) has been used in all simulations.

4.2 Results

The s/n performances of the coding schemes that have been described in Section III have been measured using three-bit quantizers and the folded binary code. Figures 7 to 11 show the measured dependence of the s/n on the average bit-error rate P in the case of burst errors. The lower, middle, and upper curves refer to the unprotected transmission and to the EP1 and EP2 error-protection schemes; note that the effective bit-error rate is reduced then. We show this using Fig. 9 as an example. The s/n value for $P = 0$ is due to the quantization noise only. The lower curve shows a considerable loss in s/n for high bit-error rates P. An increase in s/n can be obtained by protecting the most significant bit (EP1; middle curve) or the two most significant bits (EP2; upper curve). For example, if $P = 0.025$, the s/n value without error protection is 10.4 dB. A value of 14 dB is obtained with

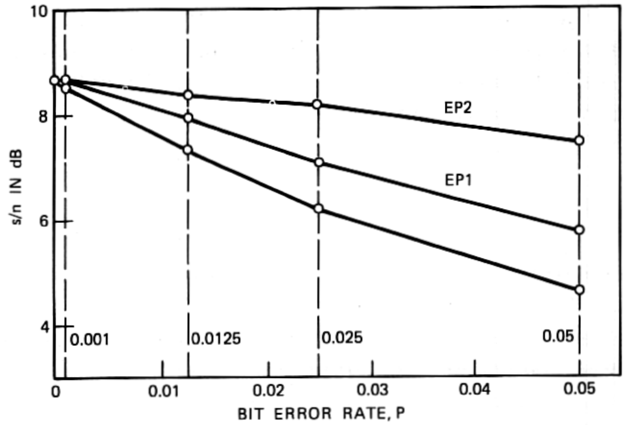

Fig. 7—Signal-to-noise ratio performance of a three-bit log PCM scheme in the presence of correlated errors.

the protection of the most significant bit (EP1). The effective bit-error rate is reduced to $\frac{2}{3} \times 0.025 = 0.0167$ in this case because $\frac{1}{3}$ of the errors are assumed to be corrected at the receiver. The 14-dB value of the EP1 curve is 2 dB higher than the s/n value we get for $P = 0.0167$

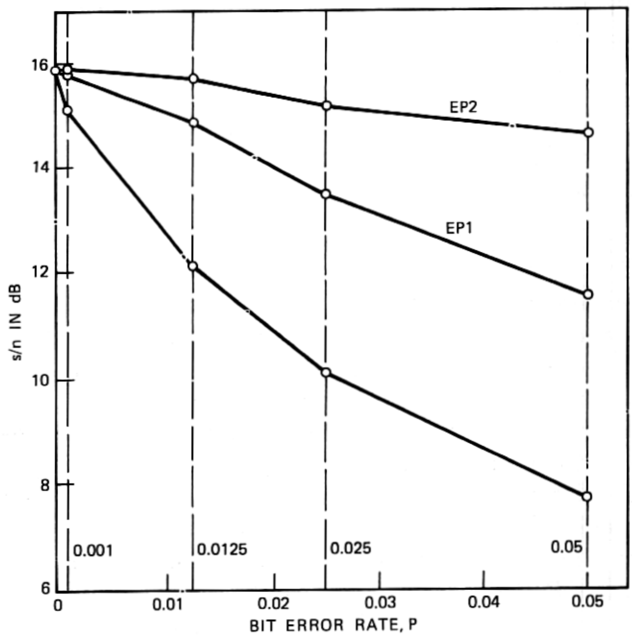

Fig. 8—Signal-to-noise ratio performance of a three-bit PCM-AQF scheme in the presence of correlated errors.

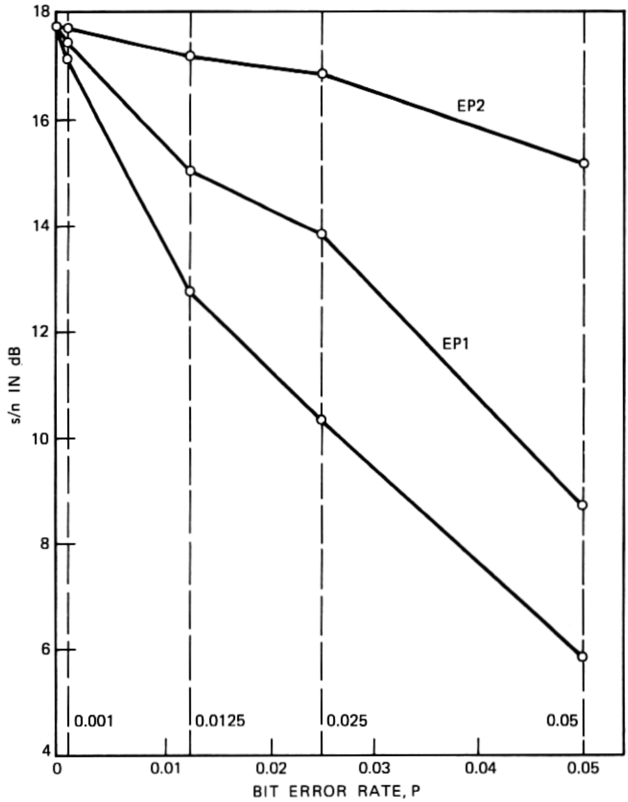

Fig. 9—Signal-to-noise ratio performance of a three-bit DPCM1-AQF scheme in the presence of correlated errors. The predictor coefficient has a value $h_1 = 0.6$.

on the lower curve (no error protection). We expect this result because no errors occur on the most significant bit for the effective 0.0167 bit-error rate of the EP1 curve. An s/n value of 16.9 dB is obtained with the protection of the two most significant bits (EP2). The s/n value for the effective bit-error rate of $\frac{1}{3} \times 0.025 = 0.083$ is 14.3 dB if the errors occur on all bits of the code words (lower curve); therefore, a 2.6-dB increase in s/n is due to the fact that only the least significant bits are affected.

Error protection, however, is only possible by inserting redundancy into the code words. Let us assume that it is possible to obtain an error protection of the two most significant bits by using three redundant bits for each three-bit code word. The total bit rate is now 6 bits per sample. Let us further assume that doubling the transmission rate causes doubling the bit-error rate (this is true for phase-modulation systems). Again using Fig. 9, we find an s/n value of 15.2 dB for

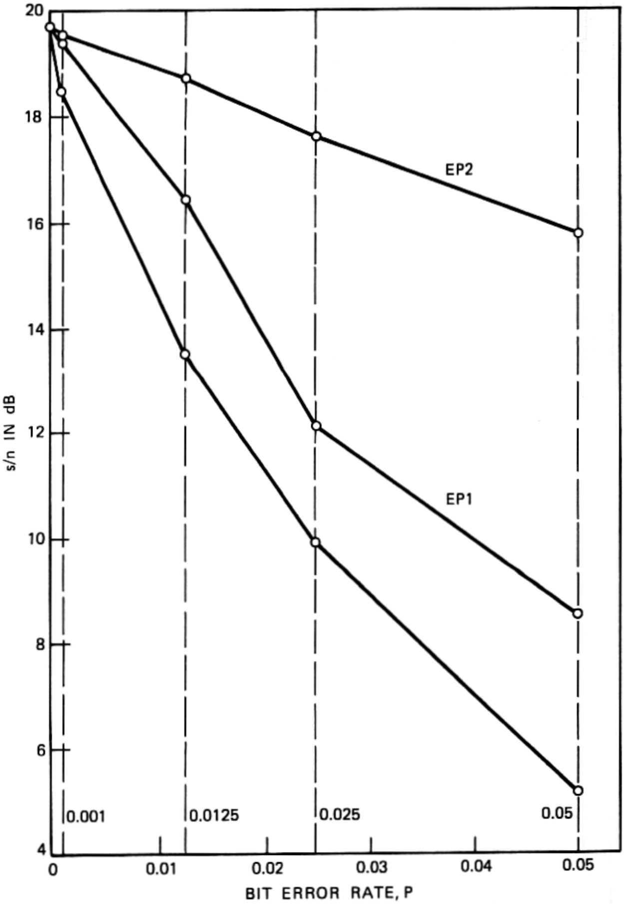

Fig. 10—Signal-to-noise ratio performance of a three-bit ADPCM1-AQF scheme in the presence of correlated errors.

$P = 0.05$ (EP2 curve). On the other hand, the s/n value without error protection is 10.4 dB for $P = 0.025$. Therefore, an improvement of nearly 5 dB over the transmission without error protection has been obtained. A similar discussion using the EP1 values shows that we get only a small s/n advantage then: an error protection of the sign bit is not sufficient for improving the overall performance.

To better compare the performances of the coding schemes in the presence of errors, we have plotted the s/n values of these schemes with P as a parameter (Fig. 12). The s/n values for $P = 0$ are due to the quantization errors only; the increase in s/n as compared to log-PCM starts with 7 dB (PCM-AQF) and goes up to 14 dB (ADPCM4-AQF). At

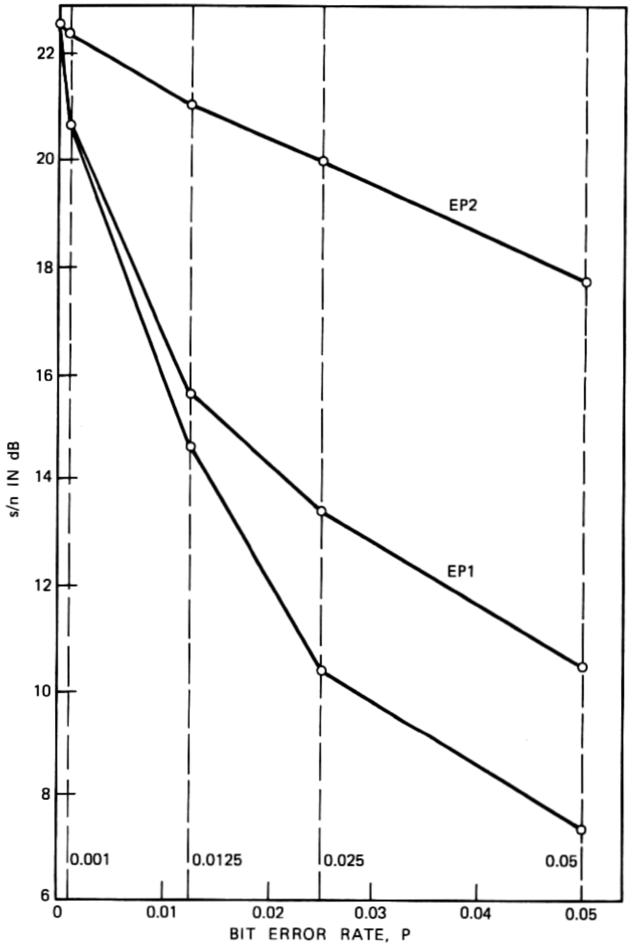

Fig. 11—Signal-to-noise ratio performance of a three-bit ADPCM4-AQF scheme in the presence of correlated errors.

higher bit-error rates, the different coding schemes have approximately identical performance because the output noise due to channel errors predominates over the quantization noise due to the quantizer.

Recall from (1) that the total error variance can be expressed as the sum of the quantization error variance ϵ_q^2 and the channel error variance ϵ_c^2 if the mutual error is neglected. The term ϵ_q^2 can be determined from the s/n for $P = 0$; hence, we can separate the values ϵ_c^2 for the four bit-error rates P that have been used in the simulations; these are the values $P = 0.001$, 0.0125, 0.025, and 0.05. The channel error can be

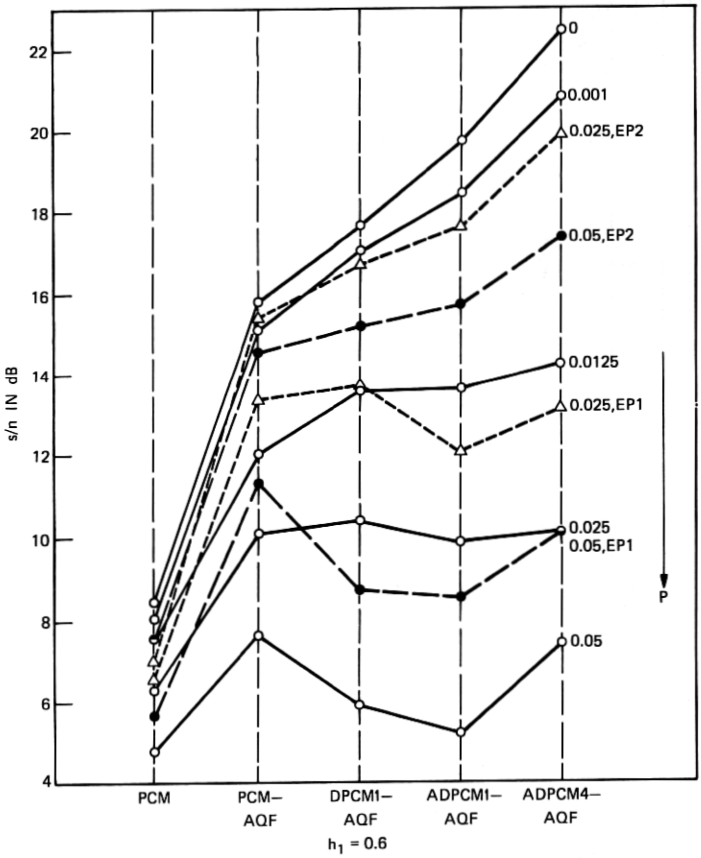

Fig. 12—A comparison of the s/n performance of three-bit encoding schemes at different bit-error rates P (correlated errors).

expressed approximately as

$$\epsilon_c^2 = \alpha_1 P + \alpha_2 P^2 \tag{12}$$

because the third term of (10) can be neglected.

The coefficients α_1 and α_2 have been determined using the measured data by searching for the minimum of the mean-squared differences between measured and calculated s/n values. These coefficients α_1 and α_2 describe the effect of channel errors on the performance of the coding schemes. The total channel error variance is mainly determined by the channel error coefficient α_1. Figure 13 shows that the α_1 values of those encoding schemes that use the same gaussian quantizer (all AQF schemes) are not very different. From this, we conclude that transmission errors are no more serious for DPCM and ADPCM schemes than for PCM; this has already been mentioned for DPCM in Ref. 2. The channel error performance is better for burst errors than for independently distributed errors. This has partly to do with the fact that some bursts appear in low-level parts of the speech sample. On the other hand, we cannot neglect the α_2 term in the case of burst errors; α_2 is the coefficient of the P^2 term in (12); this term mainly represents the channel error contribution caused by two bit errors in a code word.

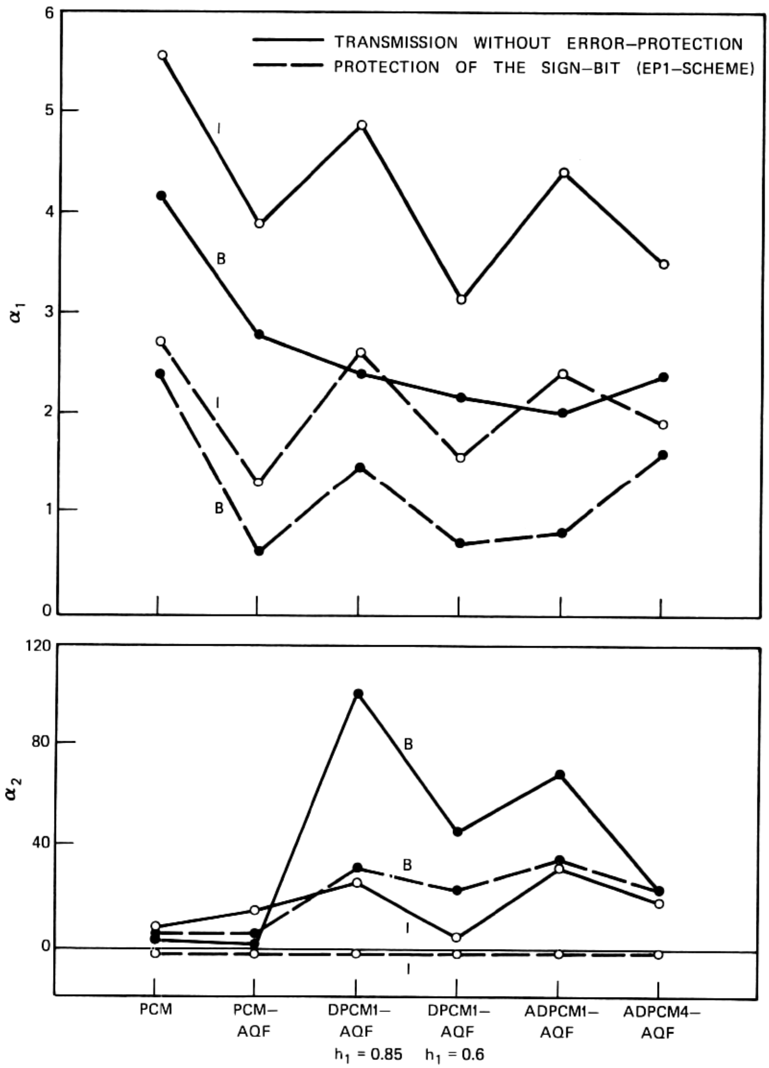

Fig. 13—Channel coefficients α_1 and α_2. I = independent errors, B = burst errors.

This contribution is higher in the case of burst errors; it causes a stronger decrease in signal-to-noise ratio as can be seen in Fig. 14 showing the s/n performance of a DPCM1-AQF scheme both for independent and correlated errors. We find the same tendency if we protect the most significant bit (EP1; see Fig. 13) or the two most significant bits. We have used the average α_1 values to calculate the s/n increase if we apply (perfect) error protection: the increases are approximately 3 and 11 dB for the EP1 scheme and EP2 scheme, respectively. Note, from Fig. 13, that the α_2 term can be neglected in the case of PCM schemes; we therefore have a slower decrease in s/n at high bit-error rates. This can also be seen from Fig. 15, where we compare the s/n performance of a PCM-AQF scheme with DPCM1-AQF schemes that have different values of the predictor coefficient. Lowering this value, we obtain a higher channel error resistance, but PCM-AQF outperforms the

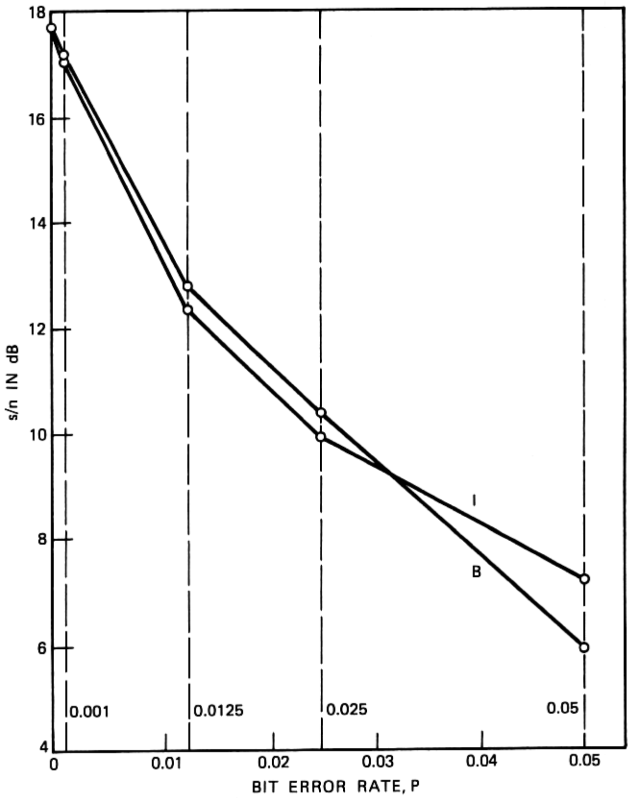

Fig. 14—Signal-to-noise ratio performance of a three-bit DPCM1-AQF scheme in the presence of independent errors (I) and burst errors (B). The value of the predictor coefficient is $h_1 = 0.6$.

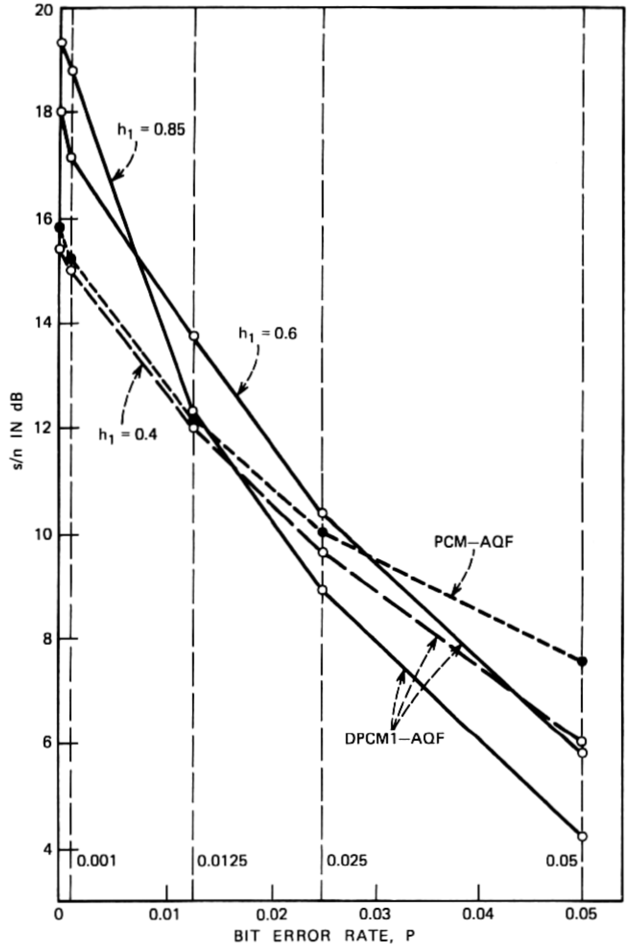

Fig. 15—Comparison of the s/n performance of DPCM1-AQF schemes with different values h_1 of the predictor coefficient with a PCM-AQF scheme.

DPCM1-AQF scheme if the predictor coefficient is too low (note the different slopes of the curves at high bit-error rates).

Our simulations involved not only three-bit quantization but also quantization schemes with a greater number of step sizes. Figure 16 illustrates a typical example: the signal-to-noise ratios of the four-bit quantization schemes are nearly 6 dB higher than the signal-to-noise ratios of the corresponding three-bit quantization schemes if the channel is error-free. But this increase is lost in the presence of high channel error rates: all three-bit and four-bit systems have a similar s/n performance for high bit-error rates. Higher s/n values can only

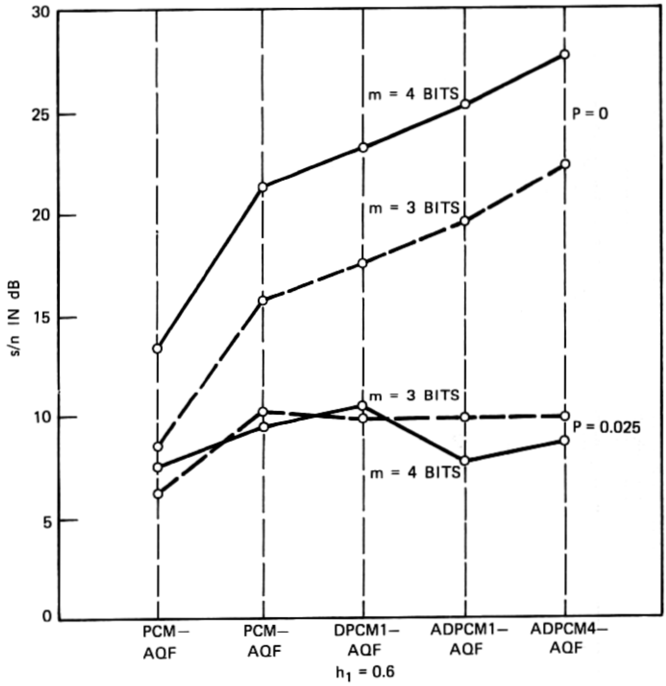

Fig. 16—Comparison of three- and four-bit encoding schemes.

be reached by error protection, that is, by reducing the effective bit-error rate.

V. CONCLUSIONS

In this paper, we have determined the s/n performance of various speech-encoding schemes in the presence of high bit-error rates (up to 5 percent); both independent and correlated error patterns have been used. Some conclusions can be drawn from the quantitative results that have been obtained and from informal subjective listening tests.

(*i*) It is possible, without pitch tracking, to quantize speech signals with three bits per sample such that the decoded signal is nearly indistinguishable from the original signal (adaptive prediction schemes in connection with adaptive quantization). A simple scheme with a fixed predictor (one coefficient) and an adaptive quantization can be chosen for a bit rate of four bits per sample.

(*ii*) Adaptive quantization lowers the idle channel noise and increases the s/n (the three-bit quantizer with a logarithmic characteristic has a very poor performance).

(*iii*) Only the adaptive quantization schemes (AQF schemes) with an explicit error-protected transmission of the step-size information can be used in the case of high channel error probabilities.

(*iv*) The folded binary code (FBC) outperforms the natural binary code (NBC) for large bit-error rates.

(*v*) Burst errors cause a stronger decrease in signal-to-noise ratio for large bit-error rates than independent errors.

(*vi*) Transmission errors are no more serious for DPCM and ADPCM schemes than for PCM.

(*vii*) All coding schemes show approximately the same s/n performance for high bit-error rates, because the contribution of the noisy channel to the total error is much higher than the contribution of the quantizer. Therefore, a better s/n performance can only be reached by using error-protection schemes, not by increasing the number of quantizer steps.

(*viii*) A high-quality decoded signal can be obtained with a protection of the two most significant bits. An improvement in decoded signal quality can be realized even if a doubling of the bit-error rate (caused by the higher transmission rate) has to be tolerated.

(*ix*) Significant-bit-packed codes that provide only protection of the sign bit (EP1 scheme) are not very efficient.

(*x*) Adaptive quantization schemes suppress the idle channel noise; therefore, channel errors produce decoded noise only with very small amplitudes in silent intervals. This fact makes the decoded speech perceptually more pleasing.

(*xi*) Nonadaptive and adaptive prediction schemes have a better perceptual quality than PCM schemes when bit errors occur on the channel. The power density spectrum of the error sequence is shaped in the DPCM or ADPCM feedback loop such that the main contribution of the error is in the low-frequency range. This error spectrum is perceptually less objectionable.

It is important to realize that the numerical results of this paper are based on a single speech record of one speaker. However, we expect the broad conclusions of this paper, as summarized above, to be true of a wide range of input speech material.

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