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VARIABLE-RATE ADM WITH EFFICIENT RESIDUAL CODING FOR PACKET VOICE TRANSMISSION

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ABSTRACT

In this paper, we propose a new variable-rate adaptive delta modulation (ADM) coder and its modified version that may be used for packet voice transmission. Also, we investigate analytically the performances of the proposed coders which cover the transmission rate of 16 to 64 kbps. In the proposed scheme, the residual signal is coded efficiently by adaptive pulse code modulation (APCM) or by an adaptive quantizer using the previous step size information. The performances of the new variable-rate continuously variable slope delta modulation (CVSD) with APCM and its modified version are better by 3 to 7 dB in segmented signal-to-noise ratio (SNR_{SEG}) than that of the conventional CVSD coder. Also, according to our simulation results, the new variable-rate CVSD with APCM and the modified variable-rate CVSD with an adaptive quantizer yield better performance by 3 to 6 dB in SNR_{SEG} than the conventional variable-rate CVSD with a fixed quantizer.

The modified variable-rate CVSD with an adaptive quantizer can be used directly as an embedded waveform coder as well as a variable-rate coder. Also, the variable-rate CVSD with APCM may be used as an embedded coder with slight modification of the APCM quantizer.

I. INTRODUCTION

As the interest for voice/data integration is growing rapidly, several researchers have recently studied statistical voice/data multiplexing for existing voice networks [1],[2]. In statistical voice/data multiplexing, the transmission rate of a speech coder is changed according to the variation of available channel bandwidth and traffic load. The transmission rate of speech may be varied by one of the following two methods. One method is the use of a variable-rate coder in which the rate of a speech coder is controlled to have the same rate adjusted by the transmission system. The other method is the use of embedded coding in which the rate is controlled within the transmission network without the interaction between the network and the speech coder.

Up to now, several variable-rate coding structures have been proposed in waveform coding and vocoding. The variable-rate coder alleviates overloading in a packet transmission system by reducing the bit rate rather than by discarding whole packets [3],[4],[5]. In this scheme, bit allocation for given speech signal is decided according to the input speech activity and channel loading. Hence, the variable-rate waveform coder can have a significant gain in signal-to-noise ratio (SNR) or segmented SNR (SNR_{SEG}) over the corresponding fixed-rate coder. In vocoding, several multi-rate vocoders have been developed by modifying either the residual-excited linear prediction vocoder (RELP) or the voice-excited channel vocoder [6],[7]. In these systems, residual or baseband signal is transmitted at variable rates, while the speech model parameters for pitch and vocal tract are transmitted at a constant bit rate.

On the other hand, in embedded coding, network nodes adjust the bit rate automatically without the interaction between the network and the speech coder. In general, a medium-low rate hybrid coder, such as the RELP vocoder, may be considered as a two-rate embedded coder, since it operates like a low-rate linear prediction vocoder if the baseband signal is not transmitted. Among the waveform coders, the PCM coder of which code word has a hierarchy of bits with different significance may be regarded as an embedded coder. As for the DPCM system, Goodman proposed a robust embedded DPCM system for a Gauss Markov signal [8]. Also, Tierney and Malpass proposed an embedded CVSD coder for 16 to 64 kbits/s transmission rate [9].

In this study, two new algorithms of variable-rate ADM with residual coding that may be used for packet voice transmission are suggested and compared to the conventional variable-rate ADM. Also, the performance gains of the proposed variable-rate ADM's with residual coding over the conventional variable-rate ADM with fixed quantizer are investigated.

Following this introduction, algorithms of the variable-rate ADM with residual coding are described in Section II. In Section III, we investigate the performance of a variable-rate ADM with residual coding. In Section IV, the performances of new variable-rate ADM coders with residual coding are evaluated by computer simulation. Finally, conclusions are made in Section V.

II. ALGORITHMS OF VARIABLE-RATE ADM WITH RESIDUAL CODING

In this section various algorithms of variable-rate ADM coders are described briefly. Our discussion will focus only on important features of the coding algorithms. Among various ADM algorithms, only the variable-rate CVSD with an adaptive quantizer or adaptive pulse code modulation (APCM) as a residual coder is considered.

A. Variable-Rate CVSD with Adaptive Quantizer (CVSD-AQ)

The block diagram of a variable-rate CVSD-AQ is shown in Fig. 1. This scheme was proposed by Lee and Un [10]. It consists of a conventional

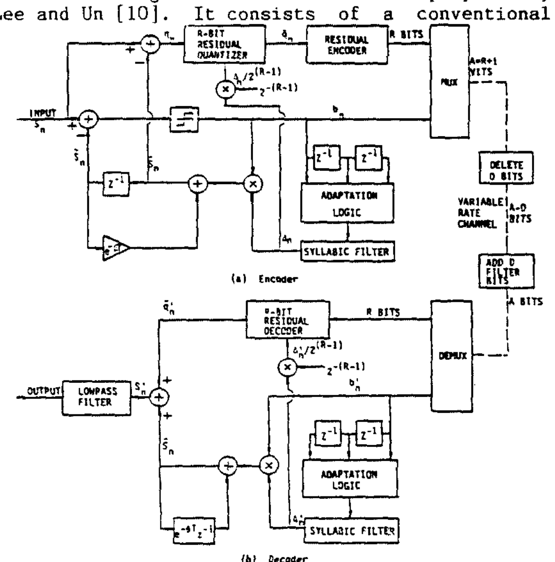

Fig. 1 A block diagram of the variable rate CVSD-AQ.

CVSD encoder and a residual encoder. The step size (Δ_n) adaptation rule of the CVSD is given as follows:

$$\Delta_n = e^{-\alpha T}\Delta_{n-1} + \Delta_{max}(1-e^{-\alpha T}), \text{ if } b_n=b_{n+1}=b_{n+2}$$

$$= e^{-\alpha T}\Delta_{n-1} \qquad , \text{ otherwise} \qquad (1)$$

where α is the inverse of the time constant of the syllabic filter, $\{b_n\}$ is the output bit sequence, and Δ_{max} is the maximum step size. The input speech s_n is predicted by a leaky integrator with a prediction time constant $\frac{1}{\beta}$. The reconstructed speech $\hat{s}_n$ is given by

$$\hat{s}_n = e^{-\beta T}\tilde{s}_n + b_n\Delta_n \qquad (2)$$

where $\tilde{s}_n$ is the predicted value of s_n.

The residual signal q_n is obtained by subtracting $\hat{s}_n$ from s_n. This signal is quantized and encoded. We note that the ADM quantizer with an ideal integrator is a one-bit mid-riser quantizer with the step size of $2\Delta_n$. Hence, the step size of the R-bit mid-riser residual quantizer is determined by

$$\Delta_q = \frac{2\Delta_n}{2^R} = \frac{\Delta_n}{2^{R-1}}. \qquad (3)$$

The residual quantizer operates at the same sampling rate as the ADM coder.

Alternatively, we propose a modified variable-rate CVSD-AQ. It is shown in Fig. 2. In this scheme, the difference signal between the original

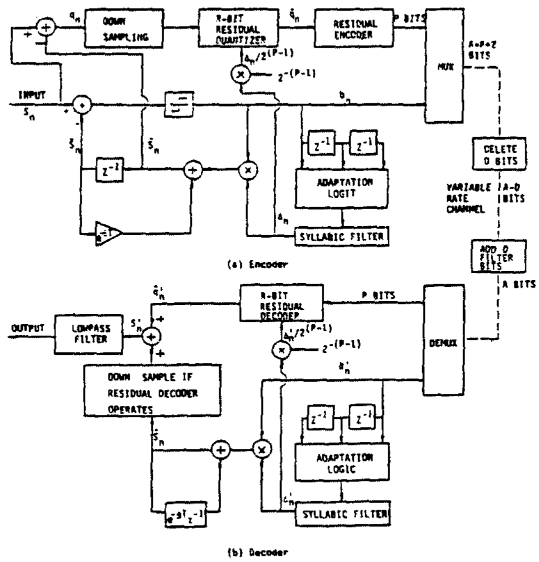

Fig. 2 A block diagram of the modified variable rate CVSD-AQ.

input speech and the reconstructed speech is down-sampled to 8 kHz and encoded by the adaptive quantizer mentioned above. At the receiver, the reconstructed value of the 16 kbits/s CVSD encoder is down-sampled if the residual encoder is used. This scheme makes it possible to improve the performance of a variable-rate CVSD-AQ significantly. The reason is that encoding bits for the difference signal can be increased by twice in the modified variable-rate CVSD-AQ in comparison to the variable-rate CVSD-AQ discussed earlier. Moreover, the modified variable-rate CVSD-AQ can be used directly as an embedded coder like the conventional variable-rate CVSD-PCM.

B. Variable-Rate CVSD with APCM Residual Coding (CVSD-APCM)

A block diagram of the variable-rate CVSD-APCM is shown in Fig. 3. This system is composed of a conventional CVSD coder and an APCM coder for residual coding. The operation of a conventional CVSD is the same as the variable-rate CVSD-AQ. The residual signal is coded by APCM [11]. In general, there exist two methods for step size adaptation of an APCM coder. One is instantaneous companding in which the quantizer basic step size is changed instantaneously at each sampling time in response to the bit pattern. The step size adaptation logic for instantaneously companding examines the quantizer output bits for the n-th sample and computes the quantizer step size, Δ_{n+1}, for the (n+1)-th sample according to the following relation:

$$\Delta_{n+1} = \Delta_n \cdot M(|H_n|) \qquad (4)$$

and

$$\Delta_{min} \leq \Delta_{n+1} \leq \Delta_{max}, \qquad (5)$$

where Δ_n is the step size used for the n-th sample and $M(|H_n|)$ is a multiplication factor whose value is decided according to the quantizer magnitude level H_n at time n. If the value of $M(|H_n|)$ is less than one, the next step size Δ_{n+1} is reduced; if it is greater than one, Δ_{n+1} is increased. In this way, the APCM coder tracks the short-time variance of the input signal adaptively.

The other step size adaptation method is syllabic companding. In this method, the basic step size Δ_n of the quantizer is adjusted at a much slower rate compared to instantaneous variations of input speech signal. When the mean value of the L consecutive quantizer magnitude level $\{|H_n|\}$ is larger than the threshold value C, the voltage Δ_{max} is applied to an RC integrator with a time constant $\frac{1}{\alpha}$s and the step size is increased toward the maximum step size Δ_{max}. For the input of lower amplitudes, no voltage is applied to the RC integrator and the step size Δ_n is decreased toward Δ_{min}. The step size adaptation rule can be expressed as the following:

$$\begin{aligned}\Delta_n &= e^{-\alpha T}\Delta_{n-1} + \Delta_{max}(1-e^{-\alpha T}), \text{ if } \frac{\sum_{\ell=1}^{L}|H_{n-\ell}|}{2^{b-1}L} > C \\ &= e^{-\alpha T}\Delta_{n-1} \qquad\qquad , \text{ otherwise} \qquad (6)\end{aligned}$$

where T is a sampling interval and Δ_n is larger than or equal to the minimum step size Δ_{min}. One can note that the step size adaptation scheme of the syllabic companding is similar to that of CVSD.

On the other hand, in the variable-rate CVSD-APCM, APCM operates at the sampling rate of 8 kHz. Also, unlike the variable-rate CVSD-AQ, the variable-rate CVSD-APCM cannot directly be used as an embedded coder. However, by modifying the step size multiplication factor of APCM with instantaneous companding, the variable-rate CVSD-APCM may be operated as an embedded coder with slight performance degradation. In this case, the step size multiplication factor must have only two distinct multipliers, one for the inner half of the quantizer range and the other for the outer half.

III. PERFORMANCE ANALYSIS OF VARIABLE-RATE ADM WITH RESIDUAL CODING

Here, we wish to analyze the performance of a variable-rate ADM with residual coding for real speech. A block diagram of a variable-rate ADM with APCM for residual coding is shown in Fig. 3. We assume that the step sizes of the ADM and the residual encoder are adjusted so that the system may be operated at the maximum SNR point. We note that, although the input to the ADM is real speech, the input of the residual coder can be assumed to be a Laplacian source.

It may be shown that SNR of ADM for real speech is in general given by [12]

$$SNR_{ADM} = k\cdot\left(\frac{f_s}{2f_m}\right)^3 \qquad (7)$$

where k is an empirical constant, f_s is sampling frequency and f_m is the bandwidth of real speech. Also, SNR of a residual encoder (such as PCM, adaptive quantizer and APCM) for a Laplacian source can be expressed approximately as [14]

$$SNR_{APCM} = 10^{\frac{CB+\alpha}{10}} \qquad (8)$$

where B is the number of bits per sample, C is an empirical performance gain, and α is an empirical constant depending on the residual encoder system.

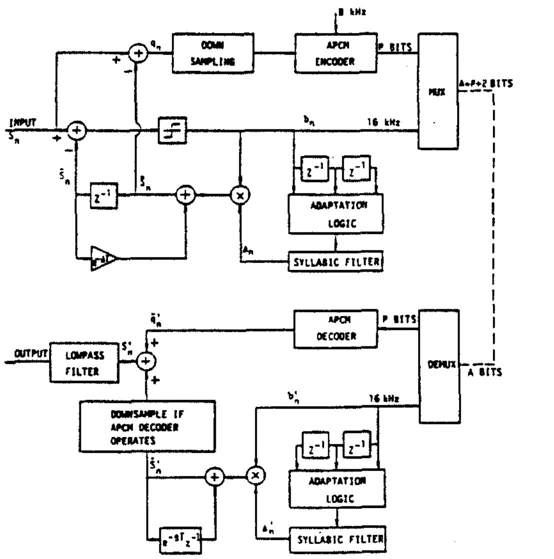

Fig. 3 A block diagram of the variable rate CVSD-APCM.

Using (7) and (8), the overall SNR of a variable-rate ADM with a residual encoder whose sampling frequency is 16 kHz is given by

$$SNR_{ADM,RE-16} = k\left(\frac{f_s}{2f_m N}\right)^3\cdot\left(10^{\frac{C(N-1)+\alpha}{10}}\right) \qquad (9)$$

where N-1 is the number of bits per residual sample when the sampling frequency of the ADM coder is 16 kHz. Also, the overall SNR of the variable-rate ADM with a residual encoder of which sampling frequency is 8 kHz is given by

$$SNR_{ADM,RE-8} = k\left(\frac{f_s}{2f_m N}\right)^3\cdot\left(10^{\frac{C\cdot 2(N-1)+\alpha}{10}}\right). \qquad (10)$$

From (7), (9) and (10), we obtain the improvement factor in decibels of the variable-rate ADM with a residual encoder over the conventional ADM system as

$$I = -30\log N + r(N-1) + \alpha \qquad (11)$$

where r is C or 2C depending on the sampling frequency of the residual encoder. Here, according to the simulation results, the values of C and α for PCM are approximately 4 and -3, respectively. Also, the values of C and α for APCM are approximately 5 and -3, respectively.

Table I shows the performance gain of the variable-rate ADM with residual coding over the conventional CVSD system at 32, 48 and 64 kbits/s. We can note from this table that little performance improvement can be obtained when the number of encoding bits for residual coding is small and the transmission rate of ADM coder is 16 kbits/s. However, it is seen that the performance of a variable-rate ADM with residual encoding is improved as the encoding rate of residual increases. In addition, it is noted that the performance of a variable-rate ADM with APCM encoding whose sampling frequency is 8 kHz is better than that of the conventional ADM at the same trans-

Table I. Performance gain of variable rate CVSD coders with residual coding over the CVSD coder.

Transmission rate	CVSD performance	Performance gain: CVSD-PCM	Performance gain: CVSD-APCM
32 kbps	24	-4	-2
48 kbps	29	-1	3
64 kbps	33	3	9

Note: The sampling rates of PCM, and APCM are both 8 kHz.

mission rate. Moreover, we can note that the performance of the variable-rate ADM with APCM coding is significantly better than that of the variable-rate ADM with PCM coding.

IV. COMPUTER SIMULATION RESULTS AND DISCUSSION

Simulations of the variable-rate CVSD-AQ, the modified variable-rate CVSD-AQ, the variable-rate CVSD-PCM and the variable-rate CVSD-APCM were done with real speech bandlimited to 3.4 kHz and sampled at 16 kHz. To measure the performances of various variable-rate CVSD schemes, we have used the SNR_{SEG} measure as a performance criterion. With the optimum parameter values for various variable-rate CVSD schemes, we obtained SNR_{SEG} as a function of the transmission rate and the input signal level.

The performances of the variable-rate CVSD-AQ and its modified version are shown in Figs. 4 and 5. It is noted from these figures that the performance of the modified variable-rate CVSD-AQ is much better than that of the variable-rate CVSD-AQ. The reason may be that in a fixed or adaptive quantizer system, a sampling rate above the Nyquist rate is not needed for reconstruction of speech signal. Also, we can note from Figs. 4 and 5 that the modified variable-rate CVSD-AQ gives much better performance than the conventional CVSD at the same transmission rate. This phenomenon may be due to the fact that in the conventional CVSD system, as the sampling rate is doubled, a gain of only about 8 to 9 dB can be achieved.

On the other hand, the performances of the variable-rate CVSD-PCM and CVSD-APCM are shown in

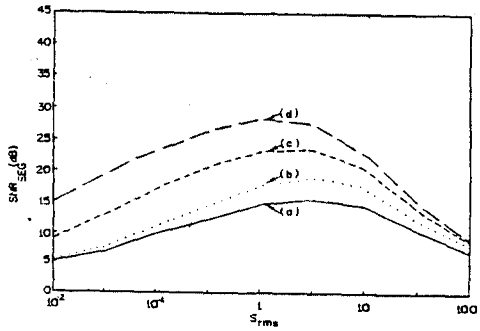

Fig. 4 SNR_{SEG} of variable rate CVSD-AQ at different transmission rates vs. input signal level.
(a) 16 kbits/s (b) 32 kbits/s
(c) 48 kbits/s (d) 64 kbits/s

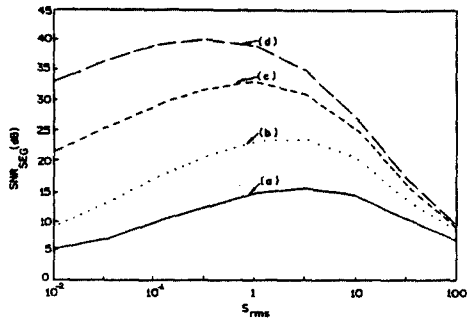

Fig. 5 SNR_{SEG} of modified variable rate CVSD-AQ at different transmission rates vs. input signal level.
(a) 16 kbits/s (b) 32 kbits/s
(c) 48 kbits/s (d) 64 kbits/s

Fig. 6 through 8. It is noted from Figs. 6 and 8 that the performance of the variable-rate CVSD-APCM is better by about 1 dB at 32 kbits/s, 6 dB at 48 kbits/s and 6 dB at 64 kbits/s in SNR_{SEG} than that of the variable rate CVSD-PCM. We can note that this gain is approximately the same as that obtained by analysis in Section III. In addition, we can note from Figs. 7 and 8 that the performance of the variable-rate CVSD-APCM is better by 3 dB at 48 kbits/s and 8 dB at 64 kbits/s in SNR_{SEG} than that of the conventional CVSD. This result is approximately in agreement with the analysis. Moreover, we can note from the same figures that there exists little difference between the performance of APCM with syllabic companding and that of APCM with instantaneous companding for residual coding.

In addition, it can be seen from Figs. 5 and 7 that the performance of the modified variable-rate CVSD-AQ is slightly better than that of the variable-rate CVSD-APCM in peak SNR_{SEG} and dynamic range. Also, we can note from Figs. 4 through 8 that peak SNR occurs at a low signal level. As the transmission rate increases, the maximum SNR becomes peakier and shifts toward the lower signal level. This phenomenon may be explained as follows. For the input signal with high amplitude, the quantization noise is dominated by slope overload distortion, which cannot be reduced effectively by the addition-

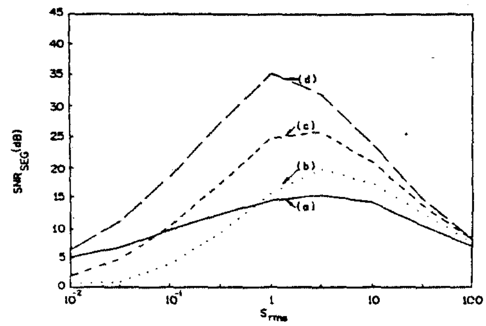

Fig. 6 SNR_{SEG} of variable rate CVSD-PCM at different transmission rates vs. input signal level.
(a) 16 kbits/s (b) 32 kbits/s
(c) 48 kbits/s (d) 64 kbits/s

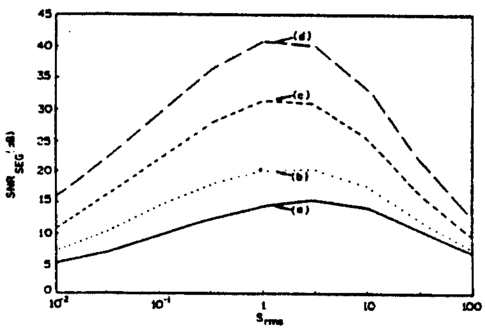

Fig. 7 SNR_{SEG} of variable rate CVSD-APCM at different transmission rates vs. input signal level (APCM has a syllabic companding logic).
(a) 16 kbits/s (b) 32 kbits/s.
(c) 48 kbits/s (d) 64 kbits/s

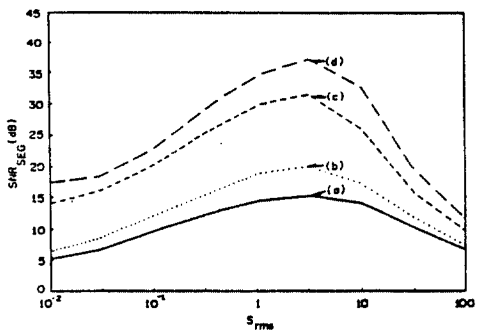

Fig. 8 SNR_{SEG} of variable rate CVSD-APCM at different transmission rates vs. input signal level (APCM has an instantaneous companding logic).
(a) 16 kbits/s (b) 32 kbits/s
(c) 48 kbits/s (d) 64 kbits/s

al residual coder that codes the quantization error signal. Hence, the performance degradation can occur in a higher range of input signal level. On the other hand, for the input signal with lower amplitude,the granular noise is dominant in quantization noise. Hence, in this case, it is possible to reduce the quantization noise using an additional residual coder.

V. CONCLUSIONS

New variable-rate ADM coders with efficient residual coding that may be used for transmission of packetized voice have been suggested. Also, the improvement factors of variable-rate ADM with residual coding over the conventional ADM have been derived analytically. According to the computer simulation results, the performance of the modified variable-rate CVSD-AQ is superior to that of CVSD-PCM by about 3 dB at 32 kbits/s, 6 dB at 48 kbits/s and 5 dB at 64 kbits/s in SNR_{SEG}. Moreover, it has been shown that the performance of the modified variable-rate CVSD-AQ is better by about 3 dB at 48 kbits/s and 7 dB at 64 kbits/s in SNR_{SEG} than that of the conventional CVSD system.

In addition, it has been shown that, the modified variable-rate CVSD-AQ can be used directly as an embedded coder as well as a variable-rate coder, and the variable-rate CVSD-APCM may be used as an embedded coder with a slight modification of the quantizer.

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