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Data Transmission at 16.8kb/s Over 32kb/s ADPCM Channel

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***Abstract*: -** *This paper presents three models of Quadrature Amplitude Modulation (QAM) modem operating at data rate of 16.8kb/s to be transmitted over 32kb/s Adaptive Differential Pulse Code Modulation (ADPCM) channel. These modems operate at symbol rates of 2400, 2800, and 3360 baud with associated number of bits per symbol of 7, 6, and 5 respectively. The performance of ADPCM is studied considering these modems with different constellations. The simulation results show that the performance of ADPCM degrades as the symbol rate decreases or number of bits per symbol increases. Also, the performance with circular constellation is better than rectangular one.*

Index Terms: - *QAM modem, 32kb/s ADPCM.*

I. IMPORTANCE OF STUDY

With the increase in demand for efficient use of digital communication channel, various types of highly effective speech coding methods have been developed [2],[5], [8],[11],[24],[27],[28]. As one of these coding methods is international standard 32kb/s Adaptive Differential Pulse Code Modulation (ADPCM)[2]. The superior performance, economy and application flexibility of ADPCM relative to other bandwidth reduction techniques were the prime reasons for its selection.

The specification of ADPCM opens the door to a host of applications in telecommunication networks [12],[13],[14],[15],[22],[23],[25].These applications can be divided into three categories: telephone company use, end customer applications, and new service offerings.

II. STATEMENT OF PROBLEM

A recommended definition of ADPCM algorithm was published by International Telephone Telegraph Consultative Committee [CCITT, the new name is International Telecommunication Union (ITU)] as Recommendation G.721 [2].

It was recognized at study group XVIII meeting [2] that voiceband data performance at 9.6 kb/s would not be acceptable with standard 32 kb/s ADPCM because ADPCM adds severe nonlinear distortion to the voiceband data signal with speed greater than 4.8 kb/s.

Thus, the interest of many research workers has been directed towards the ADPCM codec capable of providing better performance for speech and voiceband data signal at speed greater than 4.8 kb/s.

III. METHODOLOGY OF SOLUTION

Exhaustive work had been done to accommodate high speed voiceband data signal either by modifying the algorithm of ADPCM [3],[4],[6], [7], [9], [10], [17], [18], [21], [29] or by modifying the model of data transmission[16],[18],[19],[20]. One way to modify the model of data transmission is to use different constellations of Quadrature Amplitude Modulation (QAM) signal. This idea was firstly studied by AL-Rawi in [16],[18] to improve the performance of ADPCM.

IV. OBJECTIVE

The main objective of this paper is to modify the model of data transmission at data rate of 16.8 kb/s in order to reduce the distortion of ADPCM

V. STRUCTURE OF ADPCM

The algorithm of 32 kb/s ADPCM which is described here is as in CCITT G.726 [3]. Fig.1 shows simplified block diagram of ADPCM codec. Two major components form the algorithm: an adaptive quantizer and an adaptive predictor. The relation between the encoder and the decoder is also depicted. The difference between them is that the encoder has adaptive quantizer(**Q**) and inverse adaptive quantizer($\mathbf{Q}^{-1}$) ,while, the decoder has inverse adaptive quantizer only. The decoder is simply a subset of the encoder and transmits r(n) as its output instead of c(n). The adaptive predictor, which is composed of two poles and six zeros, computes an input signal estimate ŝ(n) which is subtracted from input signal s(n) resulting in a difference signal d(n). The adaptive quantizer codes d(n) into 4-bit codeword c(n) which is sent over the transmission facility. At the receiving end, an ADPCM decoder uses c(n) to attempt to reconstruct the original signal s(n). Actually, only r(n) can be reconstructed which is related to the original input signal s(n) by

$$r(n) = s(n) + e(n) \tag{1}$$

where

$$e(n) = dq(n) - d(n) = r(n) - s(n) \tag{2}$$

is the error introduced by the quantizer, and dq(n) is the output of inverse adaptive quantizer .
A typical measure of the ADPCM performance is given by signal-to-noise ratio (SNR)

$$SNR= E[s^2(n)]/E[e^2(n)] = \sigma_s^2/ \sigma_e^2 \tag{3}$$

Where E denotes expectation, σ_s^2 is the power (or variance) of input signal, &σ_e^2 is the power (or variance) of the error signal.

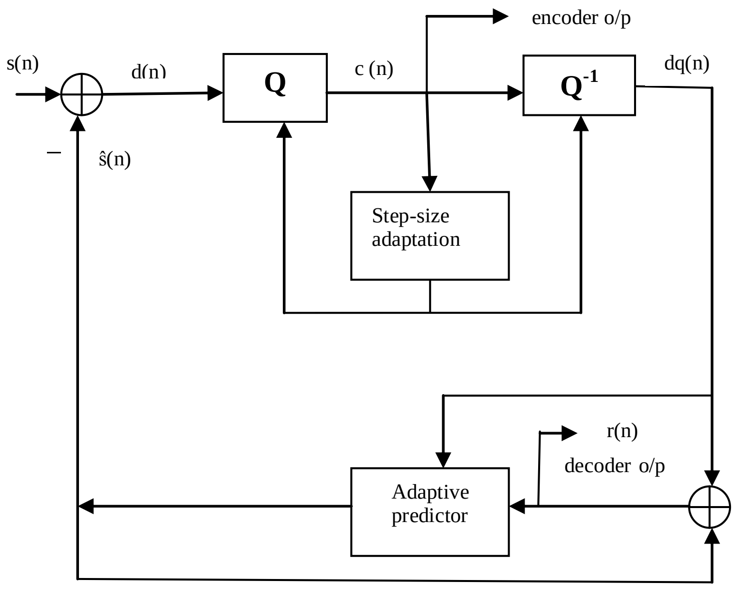


Fig.1 ADPCM Codec

VI. QAM MODEM

The first QAM modem named modem-I operates at symbol rate of 2400 baud with each symbol is represented by 7 bits (trellis coding is excluded) giving data rate of 2400x7=16.8 kb/s. The number of points in M-ary QAM constellation is equal to 2^7=128 points, while, modem V.34[1] uses the same symbol rate but with 192-point or 224-point constellation. The design of QAM constellation plays important role in reducing the effect of channel noise [26], also, in reducing the distortion of ADPCM[16]. Some of constellations which are considered here are shown in Fig.2, for 128-point, rectangular, & (6,12,19,27) circular. Because of symmetry, parts of rectangular and circular constellations are drawn.

The second QAM modem named modem-II operates at symbol rate of 2800 baud with each symbol is represented by six bits (trellis coding is excluded) giving data rate of 2800x6=16.8kb/s, with 2^6=64-point constellation, while, modem V.34 uses the same symbol rate but with 96-point or 112-point constellation. Fig.3 shows some of 64-point constellations, rectangular,& (6,12,19,27) circular.

The third QAM modem named modem-III operates at symbol rate of 3360 baud (the maximum allowable symbol rate is 3429 baud[1]) with each symbol is represented by five bits (trellis coding is excluded) giving data rate of 3360x5=16.8kb/s, with 2^5=32-point constellation. Fig.4 shows some of 32-point constellations, rectangular, (4,11,17),& (5,11,16) circular.

VII. COMPUTER SIMULATION TEST

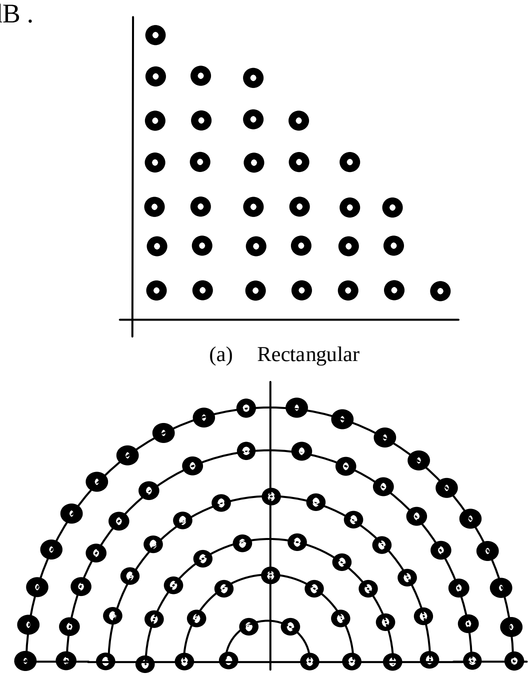
A series of computer simulation tests have been carried out on ADPCM codec using the three QAM modem signals at 16.8kb/s with constellations shown in Figs.2-4. The performance of ADPCM is measured by calculating SNR in equation 3.

Table 1 shows the results of testing ADPCM using modem-I. It seems that the performance of ADPCM with circular constellation is better than rectangular one by approximately 0.3dB.

Table 2 shows the results of testing ADPCM using modem-II. It seems that the performance of ADPCM with circular constellation is better than rectangular one by approximately 0.4dB.

Table 3 shows the results of testing ADPCM using modem-III. It seems that the performance of ADPCM with circular constellation is better than rectangular one by approximately 0.5dB.

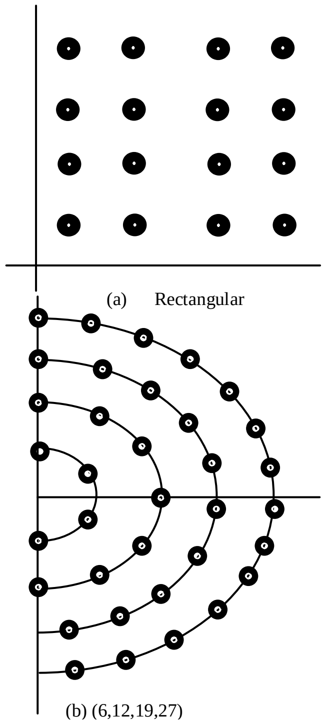
The comparison among the three modems shows that the performance of ADPCM with modem-III is better than its performance with modem-II by approximately 1.1dB and the later is better than modem-I by approximately 1.2dB .



(a) Rectangular

(b) (6,12,18,24,30,38)

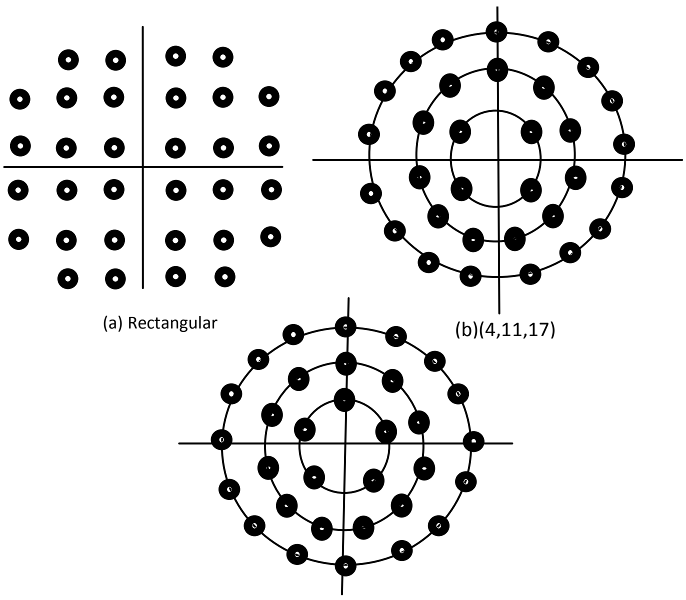
Fig.2 128-ary QAM constellations



(a) Rectangular

(b) (6,12,19,27)

Fig.3 64-ary QAM constellations



(a) Rectangular

(b)(4,11,17)

(c)(5,11,16)

Fig.4 32-ary QAM constellations

Table 1 Performance of ADPCM

	Modem-I	
	Rect	(6,12,19,27)
SNR(dB)	19.7	20

Table 2 Performance of ADPCM

	Modem-II	
	Rect	(6,12,19,27)
SNR(dB)	20.9	21.3

Table 3 Performance of ADPCM

	Modem-III		
	Rect	(4,11,17)	(5,11,16)
SNR(dB)	21.8	22.2	22.3

VIII. SUMMARY AND CONCLUSION

Three QAM modems operate at data rate of 16.8kb/s have been considered in order to reduce the nonlinear distortion of ADPCM. The simulation results show that the performance of ADPCM with modem-III is better than modem-II and the later is better than modem-I. Also, the performance with circular constellation is better than rectangular one.

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