# Data Transmission at 12kb/s over 32kb/s ADPCM

Muhanned AL-Rawi, *Member, IACSIT* and Muaayed AL-Rawi

***Abstract*—This paper presents new modified QAM modem that operates at data rate of 12kb/s to be transmitted over 32kb/s Adaptive Differential Pulse Code Modulation (ADPCM). The purpose of this modified QAM modem is to reduce the nonlinear distortion introduced by ADPCM. The performance of ADPCM is studied considering the standard QAM modem and modified QAM modem with different constellations. The simulation results show that the performance of ADPCM with modified QAM modem is better than its performance with standard QAM modem. 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[1]-[7]. As one of these coding methods is international standard 32kb/s Adaptive Differential Pulse Code Modulation (ADPCM) [1]. 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 [8]-[14].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 [1].

It was recognized at study group XVIII meeting[1] 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.

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The authors are the Department of Electrical Engineering, University of Ibb, Yemen (e-mail: muhrawi@yahoo.com, muaayed@yahoo.com ).

## III. Methodology of Solution

Exhaustive work had been done to accommodate high speed voiceband data signal either by modifying the algorithm of ADPCM [15] - [24] or by modifying the model of data transmission [25]-[27]. 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 [25],[22] 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 12 kb/s by modifying the standard QAM modem 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 [15]. 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($Q^{-1}$) ,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 $\hat{s}(n)$ 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, $\sigma_s^2$ is the power (or variance) of input signal, &$\sigma_e^2$ is the power (or variance) of the error signal.

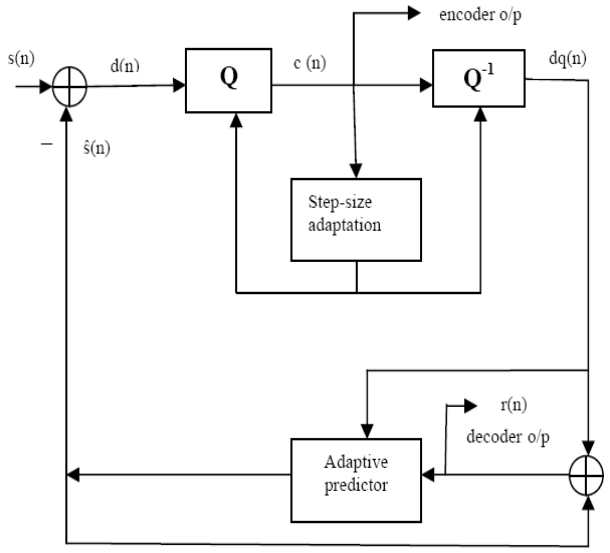



Fig.1. ADPCM codec

## VI. Standard and Modified Modem

The standard QAM modem V.32 bis[28] operates at symbol rate of 2400 baud with each symbol is represented by five bits (trellis coding is excluded) giving data rate of 2400x5=12 kb/s. The number of points in M-ary QAM constellation is equal to $2^5$=32 points. The design of QAM constellation plays important role in reducing the effect of channel noise[29], also, in reducing the distortion of ADPCM[25]. Some of constellations which are considered here are shown in Fig.2, for 32-point, rectangular, (4, 11,17), &(5,11,16) circular

The modified QAM modem operates at symbol rate of 3000 baud with each symbol is represented by four bits (trellis coding is excluded)) giving data rate of 3000x4=12kb/s, with $2^4$=16-point constellation, while, modem V.34[30] uses the same symbol rate but with 24-point or 28-point constellation. Fig.3 shows some of 16-point constellations, rectangular, (5,11), (4,12), & (8,8) circular.

## VII. Computer Simulation Test

A series of computer simulation tests have been carried out on ADPCM codec using standard and modified QAM modem signals at 12kb/s with constellations shown in Figs.2&3. The performance of ADPCM is measured by calculating SNR in equation 3.

Tables I ,II show the results of testing ADPCM. For standard modem, the performance of ADPCM with circular constellation is better than rectangular one by approximately 0.5dB. For modified modem, the performance of ADPCM with circular constellation is better than rectangular one by approximately 1dB.

The comparison between standard an modified modems shows that the performance of ADPCM with modified modem is better than standard modem by approximately 1dB for rectangular constellation and 1.5dB for circular constellation.

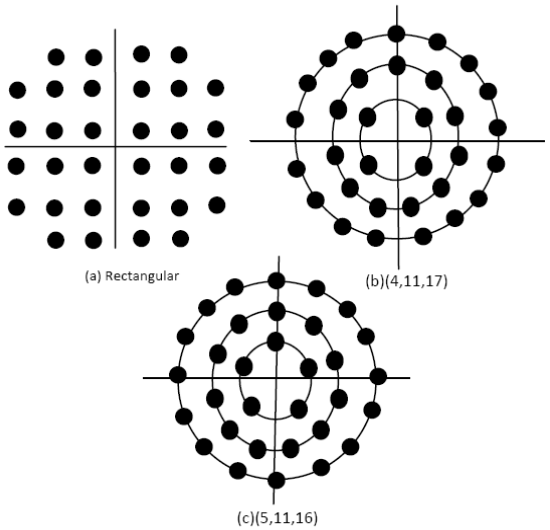



Fig. 2. 32-ary QAM constellations

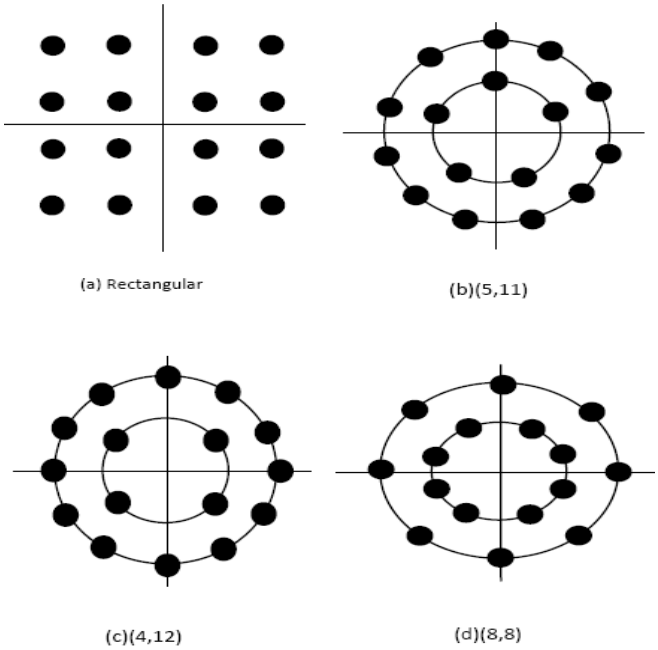



Fig. 3. 16-ary QAM constellations

TABLE I: Performance of ADPCM

| | Standard QAM | | |
|---|---|---|---|
| | Rect | (4,11,17) | (5,11,16) |
| SNR(dB) | 21.8 | 22.2 | 22.3 |

TABLE II: Performance of ADPCM

| | Modified QAM | | | |
|---|---|---|---|---|
| | Rect | (5,11) | (4,12) | (8,8) |
| SNR(dB) | 22.9 | 23.9 | 23.6 | 23.7 |

## VIII. Summary and Conclusion

Modified QAM modem operates at data rate of 12kb/s has been developed in order to reduce the nonlinear distortion of ADPCM. The simulation results show that the performance of ADPCM with modified QAM modem is better than standard QAM modem. Also, the performance with circular

constellation is better than rectangular one.

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**Muhanned AL-Rawi** was born in Iraq, 1969. He received B.Sc. degree in electrical engineering from AL-Mustansiryia University, Iraq, in 1990, M.Sc. degree in communication and electronics engineering from Baghdad University, Iraq, in 1993, and Ph.D. degree in telecommunication engineering from Bandung Institute of Technology, Indonesia, in 1999.

He is currently with the department of electrical engineering, University of Ibb, Yemen. His research interests include digital communications, and digital signal processing.



**Muaayed AL-Rawi** was born in Iraq, 1971. He received B.Sc. degree in electrical and nuclear engineering from Baghdad University, Iraq, in 1992, and M.Sc. degree in communication and electronics engineering from Jordan University of Science and Technology, Jordan, in 1999.

He had worked as nuclear and electrical engineer for several years at Iraqi Atomic Energy Organization, Iraq. Currently he is with department of electrical engineering, AL-Mustansiryia University, Iraq. His research interests include biomedical engineering, digital communications, and digital signal processing.