Comparison between Two Algorithms of 32kb/s ADPCM using QAM Signal at 16.8kb/s

Muhanned Al-Rawi[1], Muaayed Al-Rawi[2]
[1]University of Ibb, Yemen
[2]University of AL-Mustansiryia, Iraq
muhrawi@yahoo.com

***Abstract*— This paper studies the comparison between two algorithms of 32kb/s ADPCM systems. The first algorithm uses 4-bit quantizer with sampling rate of 8000 sample/sec, and the second algorithm uses 5-bit quantizer with sampling rate of 6400 sample/sec. The comparison is done using QAM signal at data rate of 16.8kb/s. Two models of QAM signals are used, the first model operates at symbol rate of 2400 baud with each symbol is represented by 7-bit, while, the second model operates at symbol rate of 2800 baud with each symbol is represented by 6-bit. The contribution of this paper is that sending the second model of QAM signal over ADPCM with 5-bit quantizer. Simulation results show that the performance of ADPCM with 5-bit quantizer is better than 4-bit quantizer for both models of QAM signals. Also, the performance of ADPCM using second model is better than the first model. Furthermore, the performance with circular constellation is better than rectangular one.**

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

I. INTRODUCTION

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] and one of the 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.

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 [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.

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 [22],[25] to improve the performance of ADPCM.

II. STRUCTURE OF ADPCM

A. General Structure

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($\mathbf{Q^{-1}}$) ,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 $\hat{s}(n)$ which is subtracted from input signal s(n) resulting in a difference signal d(n). The adaptive quantizer codes d(n) into 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 and σ_e^2 is the power (or variance) of the error signal.

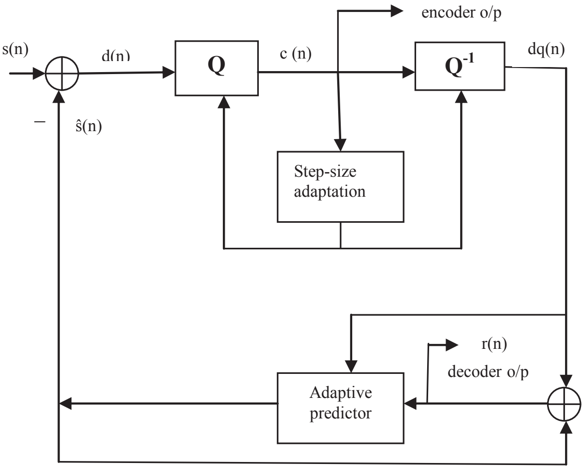


Figure 1: ADPCM Codec

B. 4-bit quantizer

The first algorithm of ADPCM (ADPCM-1) uses 4-bit quantizer with sampling rate of 8000 sample/sec. The characteristics of 4-bit quantizer are found in [18].

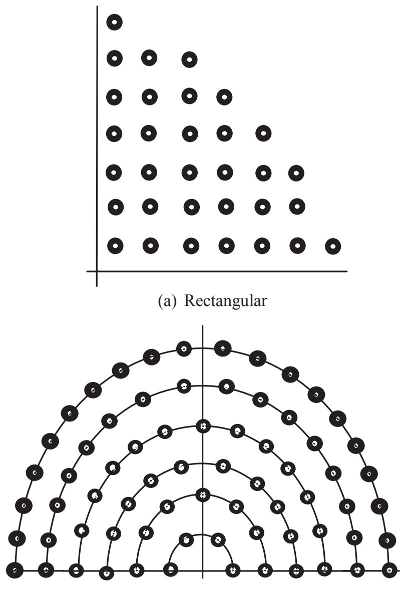
C. 5-bit Quantizer

The second algorithm of ADPCM (ADPCM-2) uses 5-bit quantizer with sampling rate of 6400 sample/sec. Its characteristics are also found in [18]. This algorithm can only be used with QAM signal having symbol rate of ≤2800 baud and carrier frequency of ≤1800 Hz. The reason is that this algorithm operates at 6400 sample/sec, and due to sampling theorem, the highest frequency should not exceed 6400/2=3200Hz which is equal to (2800/2+1800).

III. Model of QAM modem

The first model of 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 [28] 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 [29], also, in reducing the distortion of ADPCM[25]. Some of the constellations which are considered here are shown in Fig. 2, for 128-point rectangular, and (6,12,19,27) circular. Due to symmetry, only parts of rectangular and circular constellations are drawn.

The second model of 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 rectangular constellation, and (6,12,19,27) circular.



(a) Rectangular

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

Figure 2: 128-ary QAM constellations

IV. Computer simulation test

A series of computer simulation tests have been carried out on ADPCM-1 and ADPCM-2 using the two models of QAM modem signals at 16.8kb/s with constellations shown in Figs. 2-3. The performance of ADPCM is measured by calculating SNR in equation 3.

Table 1 shows the results of testing ADPCM-1 and ADPCM-2 using modem-I. It seems that the performance of ADPCM-2 is better than ADPCM-1 by approximately 1.5dB. Also, the performances of both algorithms with circular constellation are better than those with rectangular ones by approximately 0.3dB.

Table 2 shows the results of testing ADPCM-1 and ADPCM-2 using modem-II. It seems that the performance of ADPCM-2 is better than ADPCM-1 by approximately 1.7dB. Also, the performances of both algorithms with circular constellation are better than those with rectangular ones by approximately 0.4dB.

The comparison between the two modems shows that the performances of ADPCM-1 and ADPCM-2 with modem-II are better than with modem-I by approximately 1.2dB for ADPCM-1, and 1.4dB for ADPCM-2.

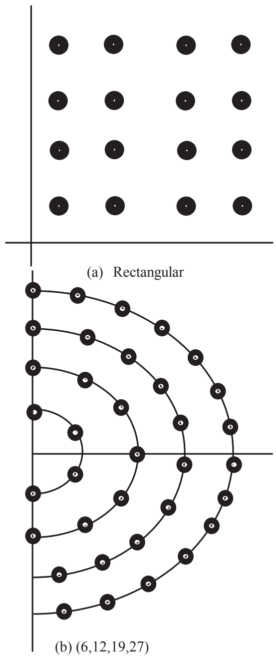


Figure 3: 64-ary QAM constellations

Table 1
Performance of ADPCM

ADPCM	Modem-I	
	Rect	Circular
ADPCM-1 SNR(dB)	19.7	20
ADPCM-2 SNR(dB)	21.2	21.6

Table 2
Performance of ADPCM

ADPCM	Modem-II	
	Rect	Circular
ADPCM-1 SNR(dB)	20.9	21.3
ADPCM-2 SNR(dB)	22.6	23.1

V. Summary and Conclusion

Two algorithms of 32kb/s ADPCM have been studied and compared using two modems operate at data rate of 16.8kb/s. The simulation results show that the performance of ADPCM-2 is better than ADPCM-1 for both modems. Also, the performances of both algorithms with modem-II are better than those with modem-I.

VI. References

[1] *32kb/s ADPCM*, CCITT Recommendation G.721-1984.

[2] E. Jahangiri, and S. Ghaemmghami, "Very low rate scalable speech coding through classified embedded matrix quantization", *EURASIP Journal on Advances in Signal Processing*, Vol.2010, 2010.

[3] *Coding of speech at 8kb/s using conjugate structure algebraic code excited linear prediction(CSACELP)*, ITU-T Recommendation G.729-1996.

[4] J. Patel, et al., " A comparison of LBG & ADPCM speech compression techniques", in *Proc. of IEEE International Joint Conference on Computer, Information, Systems Sciences, and Engineering*, 2008.

[5] S. Dusan, et al., "Speech compression by polynomial approximation", *IEEE Transaction on Audio, Speech, & Language Processing*, Vol.15, No.2, PP.387-395, 2007.

[6] W. Chu, *Speech coding algorithms: foundation and evolution of standardized coders*, John Wiley & Sons, New York, USA, 2003.

[7] W. Chu, " A scalable MELP coder based on embedded quantization of the line spectral frequencies", *in Proc. of International Symposium on Intelligent Signal Processing & Communication Systems*, Hong Kong, 2005.

[8] K. Sanjeev, "Stabilization and glitch minimization for CCITT Recommendation G.726 speech codec during packet loss scenarios by regressor control and internal state updates of the decoding process", U.S. Patent 20090125302, May 14, 2009.

[9] K. Satish, M. Mundra, and C. Daniel, "Interoperability of ADPCM encoded voice communications", U.S. Patent 20060171372, August 3, 2006.

[10] L. Chu, "Adaptive differential pulse code modulation/demodulation system and method", U.S. Patent 201001183092, July 22, 2010.

[11] L. Yen-Shih, "ADPCM encoding and decoding method and system with improved step size adaptation thereof ", U.S. Patent 20080109219, May 8, 2008.

[12] R. Mark , C. Yaakov and F. Eli , "Method and apparatus for smooth convergence during audio discontinuous transmission", U.S. Patent 20100260273, October 14, 2010.

[13] R. Mark, and F. Eli, "Adaptive error protection for wireless communications", U.S. Patent 20100138724, June 3, 2010.

[14] T. Lijun, "Rom addressing method for an ADPCM decoder implementation", U.S. Patent 20070153919, July 5, 2007.

[15] *40, 32,24,16 kb/s ADPCM,* CCITT Recommendation G.726-1990.

[16] *5-,4-,3-,2- bit/sample embedded ADPCM*, CCITT Recommendation G.727-1990.

[17] H. Mustafa, et al., "Overview and performance of CCITT/ANSI embedded ADPCM algorithm", *IEEE Transaction on Communication,* Vol.41, PP.391-399, 1993.

[18] *Comparison of ADPCM algorithm*s, ITU-T Recommendation G.726 Appendix III-1994.

[19] *Digital circuit multiplication equipment using G.726 ADPCM & digital speech interpolation,* ITU-T Recommendation G.763-1998.

[20] *ADPCM DCME configuration map report*, ITU-T Recommendation G.776.3-2000.

[21] M. AL-Rawi, et al., "Simulation results of newly designed ADPCM for data transmission at 9.6kb/s", in *Proc. of Ninth International Conference on Microelectronics*, Bandung, Indonesia, 1997.

[22] M. AL-Rawi, "Newly designed 9.6kb/s data transmission system over various algorithms of ADPCM", Ph.D. Dissertation, Dept. Elect. Eng., Bandung Institute of Technology, Indonesia, 1998.

[23] N. Bevenuto, & W.R. Daumer, "Two approaches for waveform coding of 9.6kb/s voiceband data signals through 32kb/s ADPCM", *IEEE Transaction on Communication*, Vol.3, PP.382-385, 1988.

[24] Z. Shampiro, "Adaptive differential pulse code modulation system", U.S. Patent 4860313, 1989.

[25] M. AL-Rawi, et al., "Computer simulation for newly designed 9.6kb/s data transmission system over standard ADPCM", in *Proc. of Ninth International Conference on Microelectronics*, Bandung, Indonesia, 1997.

[26] M. AL-Rawi, and M. AL-Rawi, "Modified QAM modem for data transmission at 9.6kb/s over ADPCM link", *International Proceeding of Computer Science and Information Technology* [Online] , Vol.19, 2011. Available: http://www.ipcsit.com/

[27] M. AL-Rawi, and M. AL-Rawi, "Performance of asynchronous tandem standard 32kb/s ADPCM systems using QAM signal", *Proceedings of SPIE*, Vol.8350, 2012. http://dx.doi.org/10.1117/12.920582.

[28] *A modem operating at data signaling rates of up to 33600bit/s for use on the general switched telephone network and on leased point-to point 2-wire telephone-type circuits,* CCITT Recommendation V.34-1998.

[29] T. Melvil, et al, "Digital amplitude-phase keying with M-ary alphabets", *IEEE Transaction on Communication,* Vol.22, No.2, PP.168-180, 1974.