Performance of 32kb/s ADPCM for Data Transmissin at 14.4kb/s

Muhanned AL-Rawi

***Abstract*— This study introduces a brand-new modified QAM modem that transmits data at a rate of 14.4 kbps over a 32 kbps Adaptive Differential Pulse Code Modulation (ADPCM) channel. This modified QAM modem's goal is to lessen the nonlinear distortion that ADPCM introduces. The effectiveness of ADPCM is investigated using modified and regular QAM modems with various constellations. The simulation findings demonstrate that ADPCM performs better when using a customized QAM modem than when using a normal QAM modem. Additionally, a circular constellation performs better than a rectangular one.**

***Index Terms*— Data transmission, QAM modem, 32 kbps ADPCM**

INTRODUCTION

Different sorts of extremely effective speech coding techniques have been developed in response to the growing demand for efficient use of digital communication channels. Adaptive Differential Pulse Code Modulation (ADPCM) is one of these coding techniques that was standardized by the International Telecommunication Union (ITU; formerly known as the International Telephone & Telegraph Consultative Committee (CCITT)) as recommendation G726[1]. The main reasons for choosing ADPCM over alternative bandwidth reduction approaches were its superior performance, economics, and application flexibility.

ADPCM's standard makes a wide range of applications in telecommunication networks possible. Three groups can be made out of these applications: ones for telephone companies, ones for end users, and ones for brand-new services. The most recent ADPCM applications[2–5], which are application

A packet loss concealing scheme using an ADPCM codec and a packet loss compensation (PLC) circuit is presented in [2]. The method establishes a predetermined transition period between the correct signal and a replacement signal, and the difference between the replacement signal and a computed prediction signal is combined with a dequantized prediction error to produce a dequantized combined prediction error that is added to a predicted signal to produce a combined transition signal as the basis for an output signal throughout the predetermined transition period for adjusting all decoder parameters. According to one implementation, the error combiner circuit has an analysis filterbank at one input for downsampling the replacement signal received from the PLC circuit into subband signals and an adaptive dequantization unit at another input.quantization unit for the prediction error that the ADPCM decoder receives after being quantized, downsampled, and encoded. One of the two outputs of an adaptive prediction unit is connected to a subtractor, which receives the subband substitution signal from the analysis filterbank, while the other output is connected to an adder. The output of the adder has a feedback loop to the adaptive prediction unit and leads to a synthesis filterbank for recombining the combined subband substitute signals to gain an output signal. A concealment prediction error shaper is connected to the output of the adaptive dequantization unit between the subtractor and the adder. A weighted sum of the dequantized prediction error and the prediction error of the concealment prediction error shaper is created in a predetermined way.

A method for coordinating an encoder and a decoder in a wireless communication system using discontinuous transmission was developed in [3]. The method involves giving the encoder and the decoder a syncless reset command to put them in stable and compatible states. The syncless reset command can be executed by either performing a conventional reset command followed by the encoding or decoding of a predefined number of predetermined sample values, or by setting the variables of the encoder and decoder into prepared values. In a communication system that uses discontinuous transmission, one aspect of the disclosure relates to a method for coordinating an ADPCM-based encoder in the transmitting unit and an ADPCM-based decoder in the receiving unit after a period of silence. The method includes: the encoder performing a first syncless reset operation after receiving audio data; the encoder using the audio data and the decoder performing a second syncless reset operation; the encoder operating on the audio data; the transmitting unit transmits the content of the buffer of samples; the receiving unit receiving the content of the buffer of samples; and the decoder decoding the received content.

A device that separates a sampled sound signal into a high frequency signal and a low frequency signal, individually encrypts each signal, and generates error detection code specific to the high frequency ADPCM data and the low frequency ADPCM data was developed in[4]. The device transmits the error detection code in place of data applicable to some of the multiple bits used to configure the low frequency ADPCM data. The high frequency ADPCM data and the low frequency ADPCM data are received by a receiver side, which then individually processes each set of data in accordance with the error detection code value. The current invention proposes a wireless wideband voice. An error detection code is delivered after replacing some low range ADPCM data bits, and it is used to analyze voice communication, high frequency ADPCM data, and low frequency ADPCM data separately. A fixed-size packet can therefore retain wideband sound quality even in poorer connectivity settings without affecting the sound's data rate.

The shape of encoding noise during the ADPCM encoding of a digital audio input signal is controlled using a technique and equipment in [5]. Through the use of feedback, which includes noise filtering, noise is shaped. The process consists of the following steps: obtaining a parameter to indicate a high spectral dynamic range of the signal, the parameter indicating a risk of feedback instability; detecting a risk of instability by comparing the indication parameter to at least one predetermined threshold; limiting the feedback in the event that a risk of instability is detected; and gradually reactivating the feedback over a predetermined number of frames after the current frame for which the feedback is limitedA feedback encoder with a control module that uses the described control mechanism is also offered.

As a result, the method enables quick and accurate detection of potentially troublesome signals that run the risk of causing feedback instability phenomena. As a result, this occurrence is prevented because the feedback is limited as soon as these risky signals appear. Even total deletion of this feedback may constitute this restriction. Therefore, the technique acts as a preventative measure against the emergence of problematic events. After that, the feedback loop is gradually activated so as not to cause the coded signals to change in an unnaturally abrupt way.

The primary issue with ADPCM is that, even at high data rates, it adds significant nonlinear distortion to the voiceband data signal. This issue can be resolved either by changing the ADPCM algorithm[6–9] or the model of the data transmission system[10–12].

STRUCTURE OF ADPCM

The 32 kb/s ADPCM algorithm used in this document is the same as that in CCITT G.726 [1]. ADPCM codec's simplified block diagram is shown in Fig. 1. The algorithm is made up primarily of two elements: an adaptive quantizer and an adaptive predictor. Also shown is the relationship between the encoder and the decoder. The encoder has an adaptive quantizer (Q) and an inverse adaptive quantizer (Q-1) which makes a distinction between them. In contrast, the decoder merely features an inverse adaptive quantizer. The decoder merely transmits r(n) as opposed to c(n) as its output. It is a subset of the encoder. The two poles and six zeros of the adaptive predictor compute an input signal estimate, or (n), which is then subtracted from the input signal, or s(n), to produce a difference signal, or d(n). The transmission facility converts d(n) into the 4-bit codeword c(n), which is then transferred. An ADPCM decoder employs c(n) to try to reconstruct the original signal s(n) at the receiving end. Actually, only r(n), which is connected to the initial input signal s(n), may be recovered as follows

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

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.

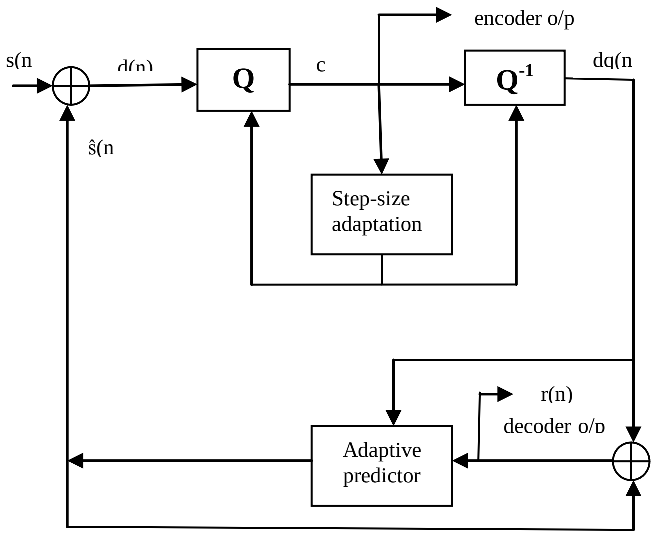

Fig.1 ADPCM Codec

STANDARD AND MODIFIED MODEM

When trellis coding is removed, the conventional QAM modem V.32 bis[13] works at a symbol rate of 2400 baud with a six-bit representation for each symbol, giving it a data rate of 2400x6=14.4 kb/s. The M-ary QAM constellation contains 64 points total, or 26 points. In order to lessen the impact of channel noise and the distortion of ADPCM, the QAM constellation design is crucial. Fig. 2 depicts some of the 64-point, rectangular, and (6,12,19,27) circular constellations that are taken into consideration in this article. Parts of rectangular and circular constellations are drawn due to symmetry.

The modified QAM modem runs at a symbol rate of 2880 baud with a five-bit representation for each symbol (trellis coding is removed), providing a data throughput of

2880x5=14.4kb/s with a 25=32-point constellation. Fig. 3 displays some 32-point constellations, including circular, rectangular, and (4,11,17), and (5,11,16).

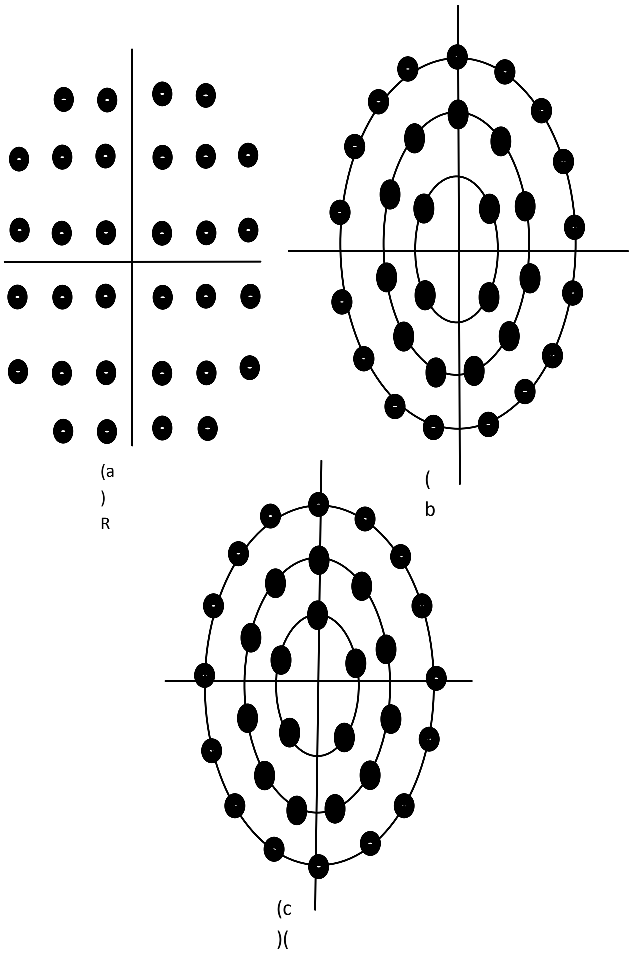

Fig.3 32-ary QAM constellations

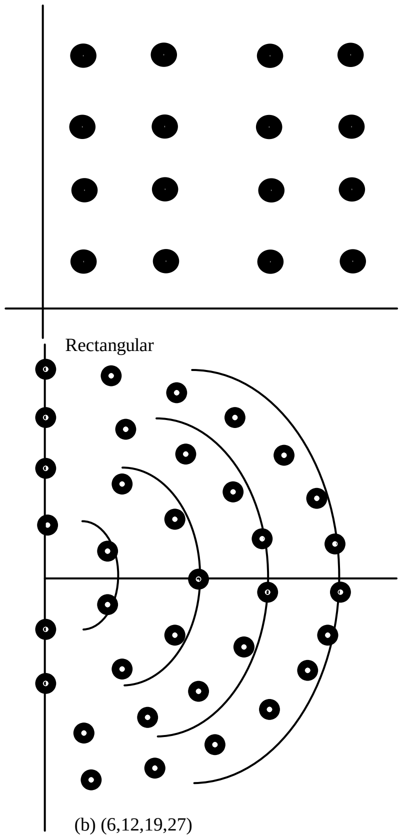

Fig.2 64-ary QAM constellations

MODEL OF DATA TRANSMISSION

The data transmission model used to assess the effectiveness of ADPCM is shown in Fig. 4. There are four pieces to this model. Each 6-bit/5-bit of binary data is mapped into one of the 64-point or 32-point QAM constellations in the first component, which is a random data generator. The second component is a QAM transmitter with a symbol output speed of 2400/2880 baud and a data rate of 2400 6 bits 2880 5 bits 14.4 kb/s. The third component is an ADPCM codec that has an input (s(n)) and an output (r(n)) for calculating the SNR specified in Eq. (3). The QAM receiver is the fourth component.

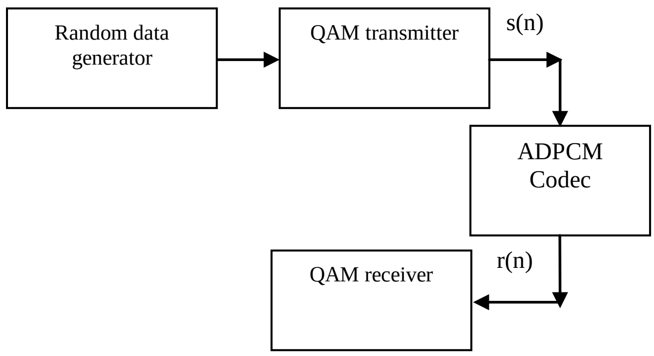

Fig.4 Model of data transmission

COMPUTER SIMULATION TEST

ADPCM codec has been put through a number of computer simulation tests utilizing both standard and customized QAM modem signals at 14.4 kbps with the constellations seen in Figs. 2 and 3. SNR in equation 3 is used to determine ADPCM performance.

Tables 1 and 2 display the outcomes of the ADPCM testing. ADPCM with a circular constellation performs somewhat better for regular modems by around 0.4 dB—than one with a rectangular constellation. ADPCM with a circular constellation performs somewhat better than one with a rectangular constellation for modified modems (by about 0.5 dB).

The performance of ADPCM with modified modem is superior to standard modem by roughly 0.9dB for rectangular constellation and 1dB for circular constellation, according to the comparison between them

Table 1 Performance of ADPCM

	Standard QAM	
	Rect	(6,12,19,27)
SNR(dB)	20.9	21.3

Table 2 Performance of ADPCM

	Modified QAM		
	Rect	(4,11,17)	(5,11,16)
SNR(dB)	21.8	22.2	22.3

SUMMARY AND CONCLUSION

In order to lessen the nonlinear distortion of ADPCM, a modified QAM modem that transmits data at a rate of 14.4kb/s has been developed. According to the simulation results, ADPCM with a modified QAM modem performs better than a conventional QAM modem. Additionally, a circular constellation performs better than a rectangular one.

REFRENCES

[1] *40, 32,24,16 kb/s ADPCM,* CCITT Recommendation G.726-1990.

[2] Z. Markus, "Method of packet loss concealment in ADPCM Codec and ADPCM decoder with PLC circuit", U.S. Patent 20160148619, 2016.

[3] A. Gur, "Non-synchronized ADPCM with discontinuous transmission", U.S. Patent 9332585, 2016.

[4] A. Shibuta, "Wireless communication device and communication terminal", U.S. Patent 9270419, 2016.

[5] S. Ragot, " Controlling a noise-shaping feedback loop in a digital audio signal encoder avoiding instability risk of the feedback", U.S. Patent 9489961, 2016.

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

[7] M. AL-Rawi, and M. AL-Rawi, "Modified 24kb/s ADPCM System", *Journal of The institution of Engineers(India): Series B(Springer),* Vol.94, Issue 3, 2013.

[8] M. AL-Rawi, and M. AL-Rawi, "Comparative study of 24kb/s ADPCM algorithms", *Radioelectronics and Communications Systems(Springer)*, Vol.57, Issue 6, 2014.

[9] M. AL-Rawi, and M. AL-Rawi, "Modified algorithm of 16kb/s ADPCM ", *Pacific Science Review (Elsevier)*, Vol.17, Issue 3,2015.

[10] M. AL-Rawi, "Detection techniques for mitigating the nonlinear distortion of ADPCM link ", *Australian Journal of Multi-disciplinary Engineering(Taylor and Francis),* Vol.14, Issue 1,2018.

[11] M. AL-Rawi, and M. AL-Rawi, "Analysis and design of ADPCM system", *Journal of Advanced Engineering*, Vol.10, No.3, 2015.

[12] M. AL-Rawi, and M. AL-Rawi, "Performance study of multistage of 40kb/s ADPCM systems", *International Review of Applied Sciences and Engineering(Akademiai Kiado)* , Vol.6, No.2, 2015.

[13] *A duplex modem operating at data signaling rates of up to 14400bit/s for use on the general switched telephone network and on leaswd point-to-point 2-wire telephone-type circuits,* CCITT Recommendation V.32 bis-1991.

Muhanned AL-Rawi - Bandung Institute of Technology (email: muhrawi@yahoo.com)