

# Comparative Study of 24 kb/s ADPCM Algorithms

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Received in final form July 17, 2013

**Abstract**—The paper is devoted to comparison of four algorithms of 24 kb/s ADPCM, standard ADPCM (ADPCM-1), and three new modified algorithms (ADPCM-2, ADPCM-3, ADPCM-4). The purpose of the modified algorithms is to reduce the nonlinear distortion introduced by ADPCM when high data rate signal passes through it. The performances of the four algorithms are researched using QAM signal at data rate of 9.6kb/s. The simulation results show that the performance of ADPCM-4 is better than that of ADPCM-3, and the performance of ADPCM-3 is better than that of ADPCM-2, and the performance of ADPCM-2 is better than that of ADPCM-1.

DOI: 10.3103/S0735272714060065

## 1. INTRODUCTION

Increase of demand for efficient use of digital communication channel leads to necessity of various types of highly effective speech coding methods development [1–7]. One of such coding methods is international standard 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.

The main problem of ADPCM is introduced severe nonlinear distortion to the voiceband data signal at high data rate. This problem can be solved either by modifying the algorithm of ADPCM [15–21], or by modifying the model of data transmission system [22–24].

## 2. STRUCTURE OF ADPCM

### 2.1. General Structure

In Fig. 1 it is shown the 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  $r(n)$  at the output instead of  $c(n)$ . The adaptive predictor 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

$$r(n) = s(n) + e(n), \quad (1)$$

where