

STATISTICAL TIME DIVISION MULTIPLEXING ARCHITECTURES AND DESIGN

A2

15 mV

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200mV

20mV



0.1 500ns

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4. Low Bit Rate Speech Coding Standards

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4.0 Abstract

The main area of speech coding standards is digital telephone, cellular mobile digital radio, and satellite fixed, and mobile system. Various digital speech coding standards which are currently in use and others are being developed for such applications. Most of these standards are International Telecommunication Union (ITU) based recommendations, Inmarsat-M, Japanese or American standards. In this chapter all such standards are provided along with their year of standardization, coding type, bit rate and Mean Opinion Scores (MOS) which is a subjective, perceptual judgment.

4.1 Introduction

The Code-Excited Linear Predictive (CELP) coding algorithm developed by Atal and Schroeder in [65] [45] made major progress in the history of digital speech coding as it offered the potential of encoding high quality speech at low bit rates. It is a widely recognised fact that CELP coding outperforms (in speech quality) all other speech coding techniques at bit rates in the 6 to 8 Kb/s range. The CELP speech quality at around 8 Kb/s is acceptable for most applications but at lower rates it needs to be improved, especially around 6 to 4 kb/s.

The CELP coder speech suffers from some inherent problems, such as excitation modeling and constraints on predictors order. To minimise the coder rate and maintain speech quality, some modifications needed to be done. Firstly, the design of the codebook is excitation contribution in such a way as to cater for the changing characteristics of the speech signal, and secondly, to correct the short comings of the Short Term (ST) and Long Term (LT) predictors.

4.2 Speech Coding Standards and Applications

The main application areas for digital speech transmission include: (a) digital telephony systems which employs high and medium bit rates; (b) cellular digital mobile radio (DMR) systems, and (c) satellite fixed, mobile and leased line systems which employ medium and low bit rate coders. Table provides a summary of various digital speech coding standards which are currently in use or advanced stages of development for some of those applications.