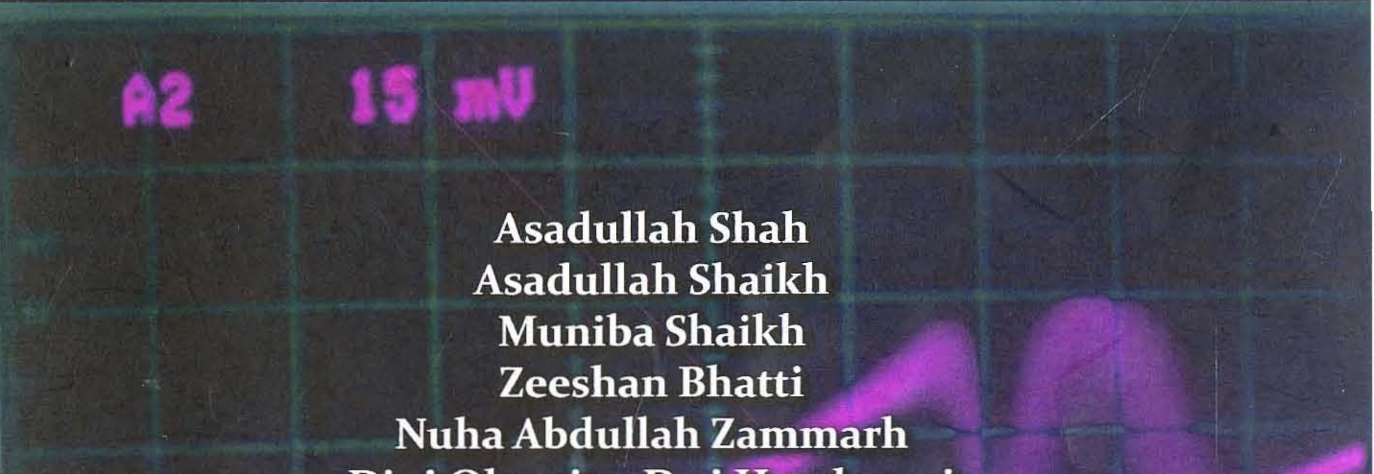
STATISTICAL TIME DIVISION MULTIPLEXING ARCHITECTURES AND DESIGN



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3. Low Bit Rate Speech Coding

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3.0 Abstract

In recent years much work has be concentrated on encoding speech. Various coding methods are developed such as Low-Delay Codebook Excited Linear Prediction (LD-CELP), Codebook Excited Linear Prediction (CELP), Pulse Residual Excited Linear Prediction (PRELP) and others. These encoders operate at very low bit rates (4.8Kbps) as well as maintained speech quality. In this chapter some of these encoding algorithms are explained.

Developments in new' speech compression techniques have provided reasonable speech quality at various bit rates lower than 64 kb/s PCM. The use of low bit rate voice digitisation can significantly increase the bandwidth efficiency of the system. For example, on a 64 kb/s link, a single PCM coded speech user can be allocated. But two users can be accommodated on the same link if adaptive differential pulse-code modulation (ADPCM) 32 kb/s [16] are used. The capacity of the channel in this case is doubled. Linear Predictive coding methods [71] enable bit rates as low as 4.8 kb/s. But this was not the end of it, researchers are still working on new hybrid compression techniques to bring- coding rates still lower and at the same time maintain the speech quality.

Low bit rate coders have the capability to reduce the bits transmitted per speech sample. In PCM 8 bits sampling at 8 kHz, with 8000 samples/s transmitted at 64 Kb/s, code each single speech sample, Whereas in CELP at 4.8 kb/s a smaller number of bits is required to encode a single sample of speech. Assuming that the data link transmits at 64 kb/s as a single channel such as a PCM-30 system, a single user on PCM channel is accommodated. But on the other hand, with low bit rates of 6.4 k b/s, the same channel can be shared among 10 users. The service ratio here is 10 times greater than the PCM user. The capacity of the channels can still be increased by employing sophisticated DSI techniques by statistical multiplexing of the active users only. Such a multiplexer system based on low bit rate digitisation models can be provided in digital switching centres (PBXs) owned by the private companies for their communications and much benefit can be gained on bandwidth.

3.1 Speech Frame Substitution Techniques