Module 3

Quantization and Coding

Lesson 10

Quantization and Preprocessing

After reading this lesson, you will learn about

- ***Need for preprocessing before quantization;***

Introduction

In this module we shall discuss about a few aspects of analog to digital (A/D) conversion as relevant for the purpose of coding, multiplexing and transmission. The basic principles of analog to digital conversion will not be discussed. Subsequently we will discuss about several lossy coding and compression techniques such as the pulse code modulation (PCM), differential pulse code modulation (DPCM), and delta modulation (DM). The example of telephone grade speech signal having a narrow bandwidth of 3.1 kHz (from 300 Hz to 3.4 kHz) will be used extensively in this module.

Need for Preprocessing Before Digital Conversion

It is easy to appreciate that the electrical equivalent of human voice is summarily a random signal, **Fig.3.10.1**. It is also well known that the bandwidth of an audible signal (voice, music etc.) is less than 20 KHz (typical frequency range is between 20 Hz and 20KHz). Interestingly, the typical bandwidth of about 20 KHz is not considered for designing a telephone communication system. Most of the voice signal energy is limited within 3.4 KHz. While a small amount of energy beyond 3.4 KHz adds to the quality of voice, the two important features of a) message intelligibility and b) speaker recognition are retained when a voice signal is band limited to 3.4 KHz. This band limited voice signal is commonly referred as 'telephone grade speech signal'.

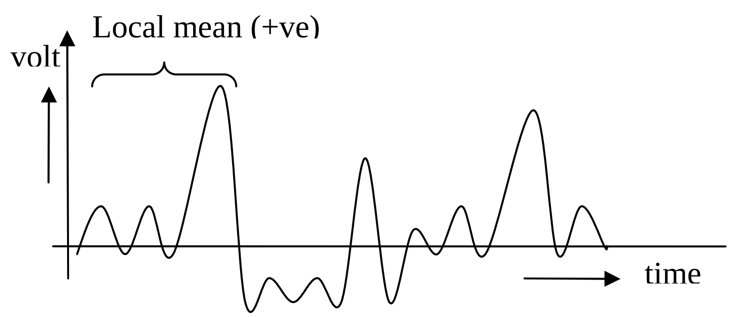

Fig. 3.10.1 *Sketch of random speech signal vs. time*

A very popular ITU-T (International Telecommunication Union) standard specifies the bandwidth of telephone grade speech signal between 300 Hz and 3.4 kHz. The lower cut off frequency of 300 Hz has been chosen instead of 20 Hz for multiple practical reasons. The power line frequency (50Hz) is avoided. Further the physical size and cost of signal processing elements such as transformer and capacitors are also suitable for the chosen lower cut-off frequency of 300 Hz. A very important purpose of squeezing the bandwidth is to allow a large number of speakers to communicate simultaneously through a telephone network while sharing costly trunk lines using the principle of multiplexing. A standard rate of sampling for telephone grade speech signal of one speaker is 8-Kilo samples/ sec (Ksps).

Usually, an A/D converter quantizes an input signal properly if the signal is within a specified range. As speech is also a random signal, there is always a possibility that the amplitude of speech signal at the input of a quantizer goes beyond this range. If no protection is taken for this problem and if the probability of such event is not negligible, even a good quantizer will lead to unacceptably distorted version of the signal. A possible remedy of this problem is to (a) study and assess the variance of random speech signal amplitude and (b) to adjust the signal variance within a reasonable limit. This is often ensured by using a variance estimation unit and a variable gain amplifier, **Fig. 3.10.2**. Another preprocessing which may be necessary is of DC adjustment of the input speech signal. To explain this point, let us assume that the input range of a quantizer is '±V' volts. This quantizer expects an input signal whose DC (i.e average) value is zero. However if the input signal has an unwanted dc bias ($\overline{x}$ in **Fig 3.10.2**) this also should be removed. For precise quantization, a mean estimation unit can be used to estimate a local mean and subtract it from the input signal. If both the processes of mean removal and variance normalization are to be adopted, the mean of the signal should be adjusted first.

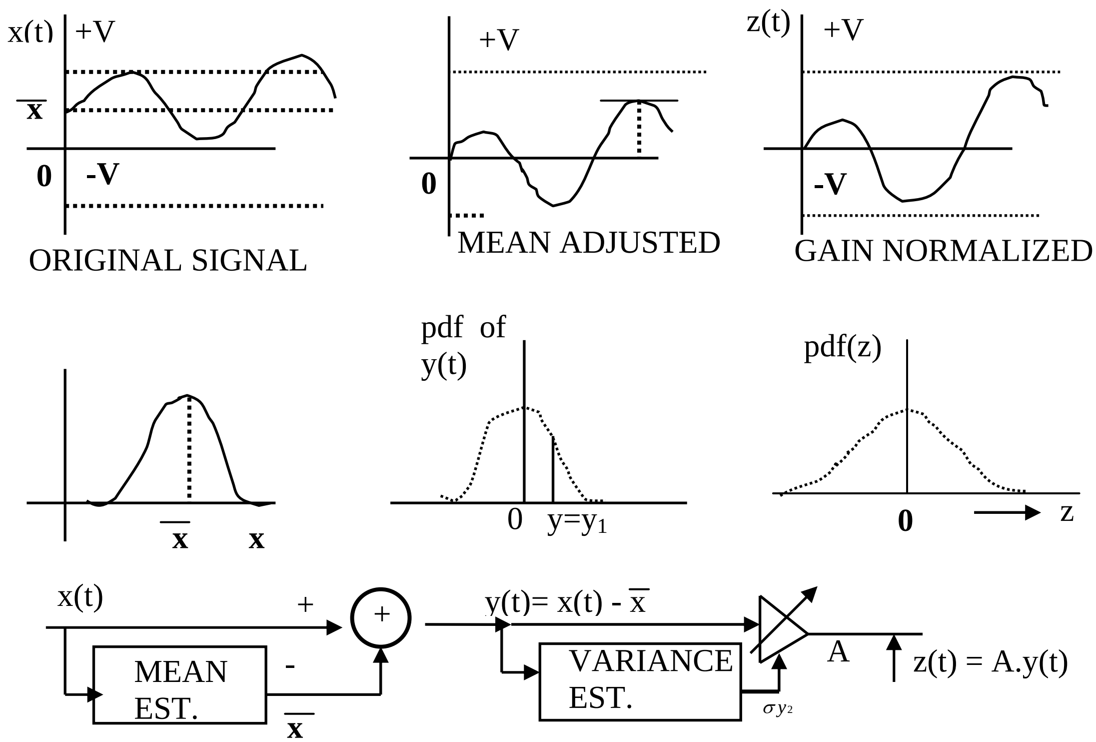

Fig. 3.10.2 *Scheme for mean removal and variance normalization*

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