VOIP : A New Compression Technique

Introduction

The VoIP technology has been a very fascinating and a convenient way for transferring the voice signals over the IP. This proves to be a very cheap way of transferring the data. The fact that now a days we are able to get high speed internet has always added to the popularity that VoIP is gaining.

The basic functionality of VoIP is converting the voice signals into packets which can be sent over the IP network. We make use of certain components like the phone adapter which basically converts the audio signals into digital signals. This then makes the signals into packets and puts them on the IP. The packets are then sent to the required destination.

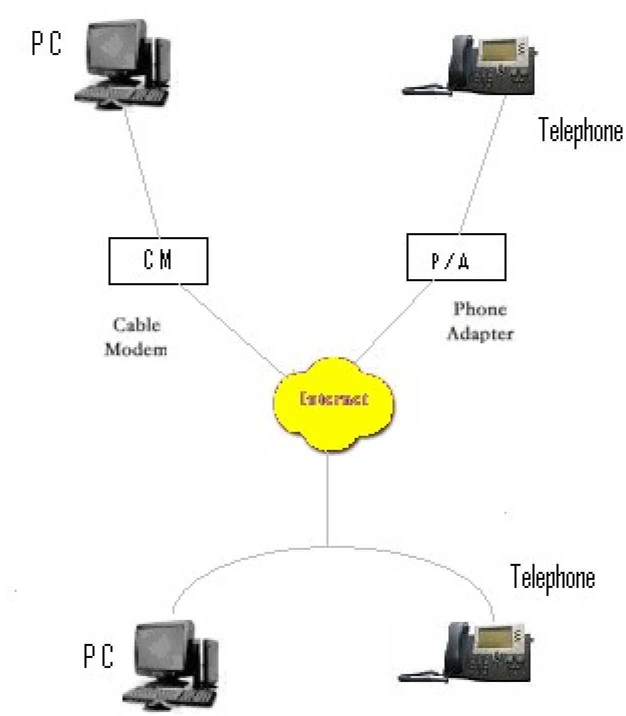

Fig 1 : V oIP Overview

There are lot many other significant concepts to be discussed about VoIP. But we will restrict the discussion only till here.

The Actual Procedure

Digitization is the process of converting the analog signals into digital signals. This paper explains an optimization technique which can be used to reduce the size of the audio content. The amount of reduction achieved would be 50% and the quality of the audio signals is not much hampered.

Lets take a sample audio signal. The sampling of this audio signal is as explained below.

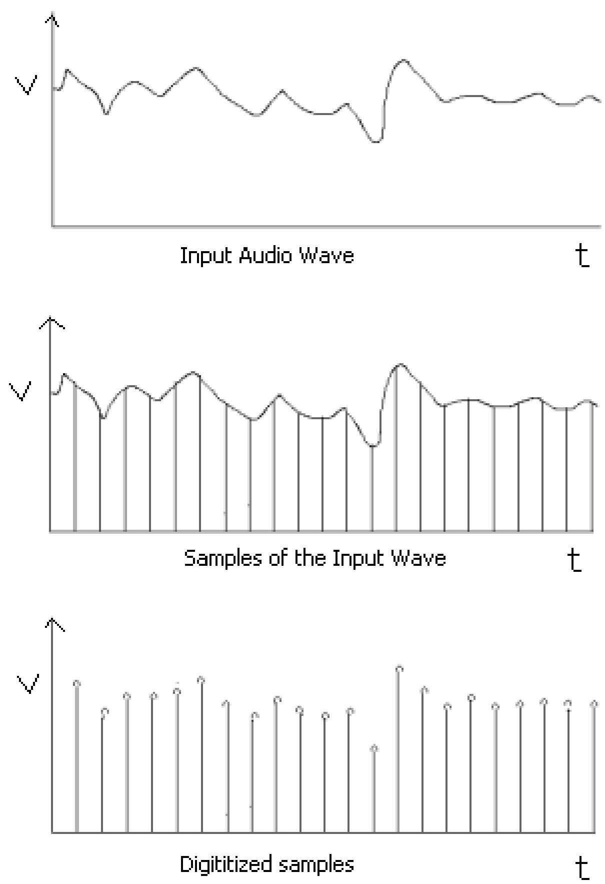

Fig 2 : Digitization

The samples finally obtained are converted into packets and are sent along the IP. Lets examine a few samples. We can convert the samples to integer sample values which have a range from 0 to 65536 (64K). In this case all the samples are in this specified range only.

This paper is intended to present a compression technique which can be used so that we can reduce the amount of data to be transferred by half the size. Let us consider a few sample values that we might come across. The values when represented in 64K format would have values like

23412, 6454, 12546, 44562, … etc

Let us take two of these sample values. 23412 and 6454. Let us represent these as V1 and V2. The following calculations explain the concept involved in the compression technique.

V1 = 23412
V2 = 6454

Let us take the square roots of both these values and represent in integer format. They would be

S1 = 153
S2 = 80

Let us combine both the values into a single value by using a formula.

C1 = S1*256 + S2
C1 = 153*256 + 80 = 39248.

According to this techniques after the samples have been digitized there is only one sample sent in the place of 2 samples. Thus the amount of data to be sent is directly reduced to half the size. Let us see how the decoding would be done at the receiver side.

R2 = C1 % 256 = 80
R1 = (C1 – R2)/256 = 153

The actual output values would be got by squaring the both these values which we had got. So the final output values would be

F1 = 23409
F2 = 6400

Thus we see that we got back both the values which we had sent from the source side.

Now, let us calculate the efficiency of this techniques. The deviation seen in the sample values would be

D1 = 3 (as compared to input value of 23412)
D2 = 54 (as compared to input value of 6454)

Ideally the deviation would be less for smaller sample values and it will increase as the sample values increase. (But in reality it all depends on the sample values).

Considering the normal speech samples these changes would have hardly any impact on the quality of the voice being transmitted. So the voice quality would remain almost the same. But we would have achieved a compression rate of 50%.

Advantages of this concept:

The implementation of this concept would be very simple and the efficiency achieved would be very high. This is an algorithm which can be ideally used for the voice signals since a slight change in the quality of the input voice is hardly noticeable.

Disadvantages of this concept:

There are practically no disadvantages apart from the fact that additional blocks for compression and decompression need to be introduced in the existing VoIP implementation flow.

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