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Speech Compression by Using Differential Pulse Code Modulation

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***Abstract*— Compression is a process of converting an input data stream into another data stream that has a smaller size. Compression is possible only because data is normally represented in the computer in a format that is longer than necessary i.e. the input data has some amount of redundancy associated with it. The main objective of compression systems is to eliminate this redundancy. The basic goal of speech data compression is to reduce the bit rate for transmission and storage while either maintaining the original quality. Analysis of speech samples, obtained from analog to digital conversion process, reveals that a high correlation is found between consecutive speech samples. The DPCM algorithm takes advantage of this high correlation property of speech data. This algorithm does not encode the actual speech samples. This algorithm actually encodes the difference between a predicted speech sample and the actual speech sample. This speech encoding provides an efficient compression with a significant reduction in the number of bits per sample. The quality of speech signal is also significantly preserved by this speech processing algorithm.**

Data compression along with analog to digital conversion is an important factor in data Communication and data storage. The Differential Pulse Code Modulation is a technique that gives above mentioned advantages along with Excellent Speech Quality.

***Keywords*— ARM Controller, DPCM Algorithm, Audio Amplifier**

I. INTRODUCTION

In the recent years, large scale information transfer by remote computing and the development of massive storage and retrieval systems have witnessed a tremendous growth. To cope up with the growth in the size of databases, additional storage devices need to be installed and the modems and multiplexers have to be continuously upgraded in order to permit large amounts of data transfer between computers and remote terminals. This leads to an increase in the cost as well as equipment. One solution to these problems is-“COMPRESSION” where the database and the transmission sequence can be encoded efficiently.

Compression is a process of converting an input data stream into another data stream that has a smaller size. Compression is possible only because data is normally represented in the computer in a format that is longer than necessary i.e. the input data has some amount of redundancy associated with it. The main objective of compression systems is to eliminate this redundancy. When compression is used to reduce storage requirements, overall program execution time may be reduced. This is because reduction in storage will result in the reduction of disc access attempts.

With respect to transmission of data, the data rate is reduced at the source by the compressor (coder), it is then passed through the communication channel and returned to the original rate by the expander (decoder) at the receiving end. The compression algorithms help to reduce the bandwidth requirements and also provide a level of security for the data being transmitted. A tandem pair of coder and decoder is usually referred to as codec.

II. COMPRESSION OF SPEECH SIGNAL

The advantage of DPCM is the reduced amount of information that must be transmitted if we maintain the same SNR or an improved SNR if we maintain the same amount of information. To get an idea on the improvement in performance that we can get from using DPCM as compared to the performance of regular PCM, DPCM can increase the SNR for some signals by as much as 20 dB. This corresponds to an improvement in the signal power compared to the noise power by 100 times, or a reduction in the amount of information by more than 3 bits/sample.

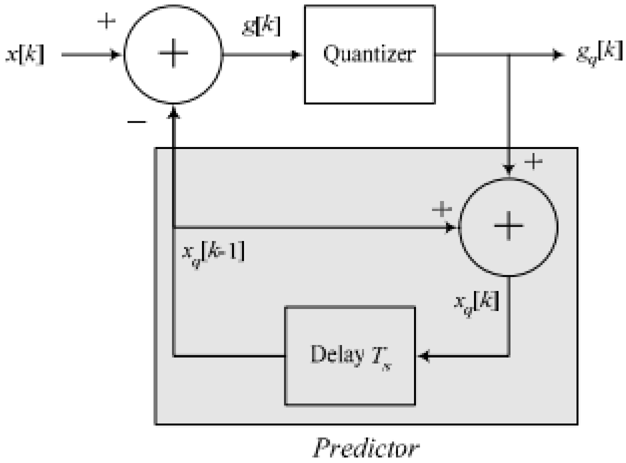

Fig.1 Differential Pulse Code Modulation

In the above system, we can easily prove that the resulting signal Xq[k] is the quantized form of X[k].

First we see that

$$g[k\,] = X\,[k] - Xq\,[k-1]$$

Now, the output of the quantizer is the quantized form of *g*[*k*] which can be represented by adding a quantization noise *q*[*k*] to the input of the quantizer. Therefore,

$$g_q[k\,] = g[k\,] + q[k]$$

Substituting for g[k] in gq[k] gives

$$g_q[k\,] = X\,[k] - Xq[k-1] + q[k]$$

From the block diagram,

$$\begin{aligned} Xq\,[k] &= g_q[k\,] + Xq\,[k-1] \\ &= X\,[k] - Xq[k-1] + q[k] + Xq\,[k-1] \\ &= X\,[k] + q[k] \end{aligned}$$

So, in fact, the function Xq[k] is the quantized form of X[k] as seen by the last line of the above equation. A word of caution here, the above derivation does not mean that if we quantized X[k] directly by the quantizer we will get Xq[k]. It just says that Xq[k] is a quantized form of X[k].

III. Working Block Diagram

Figure 2 Shows that Hardware diagram of Speech compression by using Differential Pulse Code Modulation.

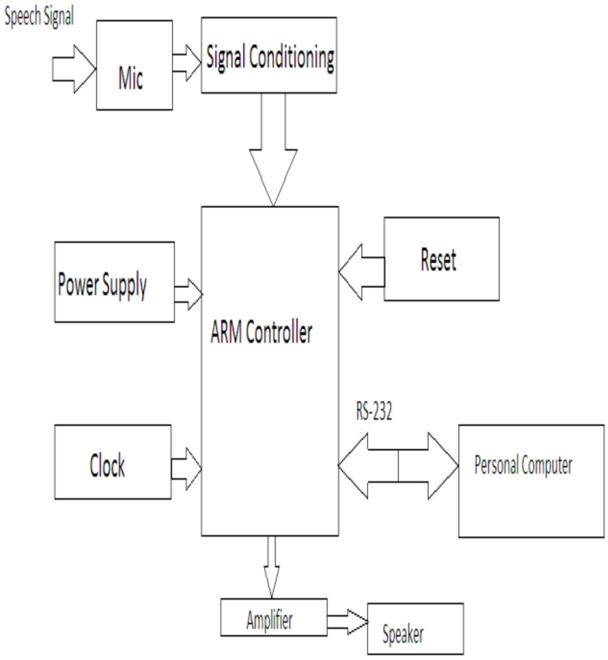

Fig.2 Speech Compression by using Differential Pulse Code Modulation

In above diagram the ARM Controller plays very important role. The ARM Controller contains inbuilt analog to digital conversion & Digital to analog Conversion, also the memory of ARM Controller is sufficient for this project. Because of this we reduces the external hardware.

This project contains the minimum hardware circuit.

1) ARM Controller
2) Mice
3) Audio amplifier
4) Speaker

Here we take the speech signal from mice. We get Output of mice is Variable amplitude analog speech. This analog signal we directly apply to the arm controller. In ARM Controller this analog speech converted to the digital speech signal(ADC). After that we apply the above differential pulse code modulation technique on this speech signal.

When compression is achieve we apply this signal to digital to analog conversion (DAC) of in built ARM Controller. This analog signal we amplify by using audio amplifier and then given to the speaker.

On speaker we listen the original signal sound & Compressed signal sound. We check the quality of original sound & Compressed sound. We compare the original sound waveform & Compressed sound waveform on personal computer.

IV. RESULTS

When we uses the algorithm of Differential Pulse Code Modulation for compression of signal. We get the quality of original & compressed signal is same on speaker.

We also check the waveform of original & compressed signal.

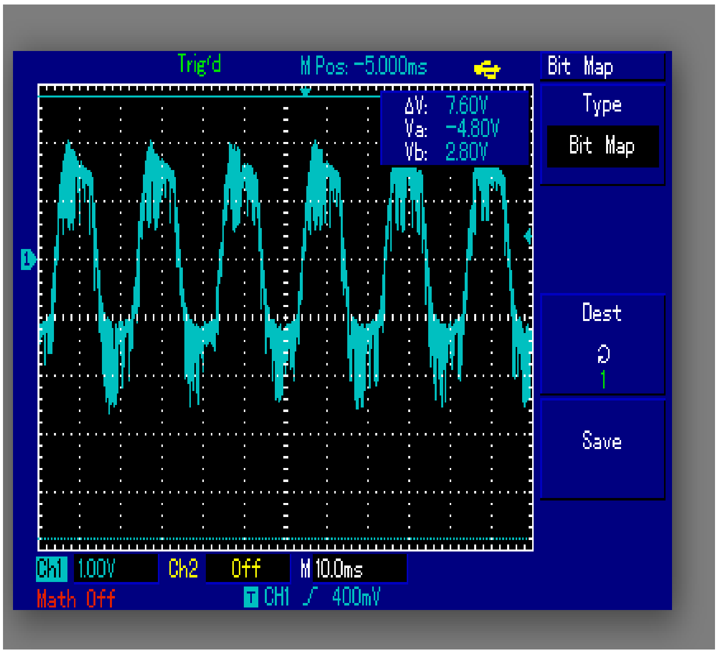

Fig.3 Original sound signal on DSO

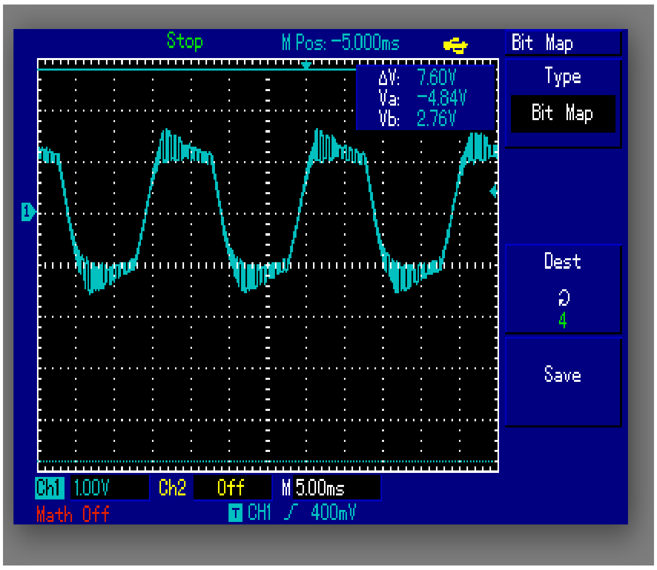

Fig.4 Pulse Code Modulated sound signal of original signal on DSO

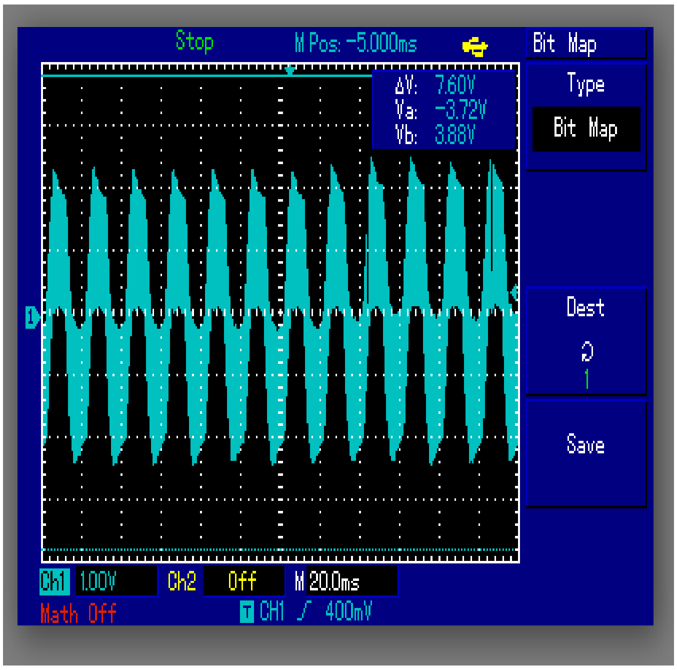

Fig.4 Differential Pulse Code Modulated sound signal of original signal on DSO

The Figure 2 shows original sound waveform on DSO. This sound signal amplitude is variable. Sound signal frequency is 20hz to 20Khz. For differential pulse code modulation (DPCM) we required the sample signal for operation. To obtain the correct sample signal we uses the Sampling theorem,

Sampling theorem,

$$Fs > 2Fm$$

Fs = Sampling Frequency
Fm =Original Frequency

Here we sample the original sound signal at 8Khz sample signal.
When we get the sample signal as sampling theorem, then we uses the Differential Pulse code modulation technique for compression.
Fig.4 shows the pulse code modulated sound signal.
Fig.5 shows the Differential Pulse Code Modulation signal means compressed signal.

V. CONCLUSIONS

By uses the algorithm of Differential Pulse Code Modulation we compressed the sound signal. The quality of signal is very good. By uses this method we reduces bits of original sound signal.
Because of this technique we save the memory size & bandwidth of signal. This technique we uses in save memory & transmission of data. This system we also uses in transferring the data between the computers, In communication system for communicating the data.
Hardware implementation of this system is very simple, complexity of circuit is reduces.
This system is very simple to design & implement.

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