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Speech Compression by using Adaptive Differential Pulse Code Modulation (ADPCM) technique With Microcontroller

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Abstract

Compression is a process of reducing an input data (Speech Signal) bit stream into a small bit size with high quality. Analog signal is a continuous signal takes more space to store the data in memory devices with original size (Bit). All sensor data (Analog Data) stored in computer with original size (Bit), but because of compression technique we store the same data in reduced format we same quality. In compression the unwanted data is eliminate. The main purpose of speech compression is to reduce the data bits for transmission of original data from one place to other & store this data that maintaining the quality as same as original signal. In this compression technique the analog to digital conversion (ADC) process played important role, because of analog to digital conversion analog to digital conversion (ADC) we get quantized sample signal. In that sample signal high correlation property is present between the sampled speech signal. The Adaptive Delta Pulse Code Modulation (ADPCM) techniques use the high correlation property of sampled data for compression of signal. This algorithm cannot compress the sampled data as it. It takes the difference between the predicted sample signal and actual sample signal then encode this difference signal which is explained in details below. The Adaptive Delta Pulse Code Modulation (ADPCM) methods have very efficient methods for the compression of signal by reduction of number of bits per sample from original signal with maintaining the quality of signal.

There are so many data compression technique available, but some technique algorithm operation not gives actual quality of signal after compression. That type of technique is called as lossy type algorithm. Because this type lossy algorithm the human ear cannot detect the word. Human voice frequency ranges from 300 Hz to 3400 Hz. Adaptive Delta Pulse Code Modulation (ADPCM) is a well known encoding scheme used for speech processing. This project focuses on the simplification in the technique so that the hardware complexity can be reduced for the portable speech compression & decompression devices.

In this project we used ARM controller which is heart of this project that contains 10-bit channel analog to digital conversion (ADC) pin. Means we get the sample upto 1024. this sample we have going uses for the Adaptive Delta Pulse Code Modulation (ADPCM) algorithm. Also in ARM controller Digital to Analog coversion (DAC) pin has available to check the compressed signal with original signal. Because of ARM controller we reduce the circuitry. Also, we use the Digital signal oscilloscope (DSO) & personal computer to check the behavioural of signal.

Because of compression we save the memory & transmission time with same quality. Also when we want this data we check from stored from memory. In so many government offices, private colleges, laboratory & industry requires to store the original data in computer or in memory devices as it is, so many memory devices are required hence wastage of money is take place. Because of compression by using adaptive differential pulse code modulation technique with microcontroller we achieve the compression of signal with same quality.

Keywords: *ARM Controller, ADPCM Algorithm, Quantizer, Audio Amplifier, PC, DSO*

INTRODUCTION

In the recent years in so many areas the information (Original data) transfer by remote computing to other computer & the required more storage devices to store the original data & recover that data is increased. So, for that purpose we required so many additional storage devices to store the original data & modems & multiplexers required to transfer the data between computers and remote centres. Because of that the cost of equipment is increased. This type of major issues face by so many companies, colleges & government office. So, avoid this problem we want the compression technique to reduce the bit rate of original data.

Compression is a process of converting an input data stream into another data stream that has a size is a small. Compression is achieved only because data size is normally represented in the computer in a format that is longer than necessary i.e. the input data has contains amount of redundancy. The objective of compression is to eliminate this redundancy.

The compression algorithms help to reduce the storage requirement, bandwidth requirements and also provide a level of security for the transmitted data. A combination of coder and decoder is called as codec.

COMPRESSION OF SPEECH SIGNAL

ADPCM algorithm has used to reduce amount of analog data that must be transmitted. In ADPCM Signal to noise ratio (SNR) is played very important role, hence if we maintain the Signal to noise ratio (SNR), then we get the same amount information a with same quality. Because of this signal power is improved as compared to the noise power by 100 times, or amount of information is reduced by more than 3 bits/sample.

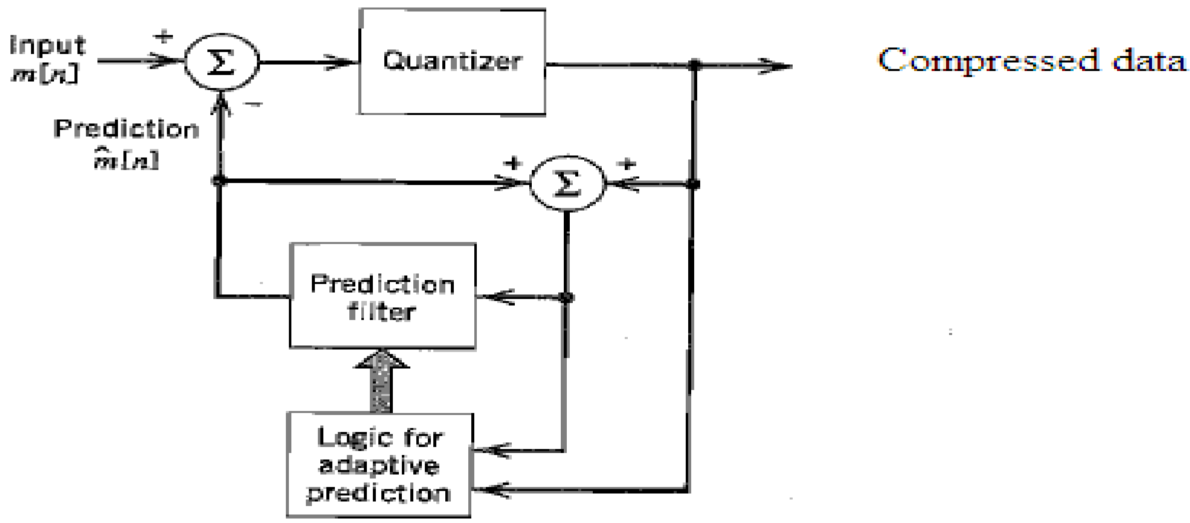

***Fig.1** Adaptive Differential Pulse Code Modulation*

In the above system, we can easily prove that the resulting signal $m\varnothing_q$ [n] is the quantized form of m [n] First we see that

$g[k] = m[n] - m\varnothing_q[n-1]$

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,

$gq[k] = g[k] + q[k]$

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

$gq[k] = m[n] - m\varnothing_q[n-1] + q[k]$

From the block diagram,

$Xq[k] = gq[k] + Xq[k-1]$

$= \quad m[n] - m\varnothing_q[n-1] + q[k] + m\varnothing_q[n-1]$
$= \quad m[n] + q[k]$
So, in fact, the function Xq[k] is the quantized form of X[k] as shown in above equation.
From above we does not mean that if we quantized X[k] directly by the quantizer then we will get Xq[k]. It just says that Xq[k] is a quantized form of X[k].
Figure 1 shows the basic operational block diagram of Adaptive differential pulse code modulation (ADPCM). i.e. contains the basic operation of

1) Quantizer
2) Prediction filter

Quantizer
An analog is a continuous signal like as speech, sensor data with continuous range of amplitude and hence we get samples accordance with continuous amplitude range.
This means that the original continuous signal may be approximated by a signal constructed of discrete amplitudes selected on a minimum error basis from an available set. The existence of a finite number of discrete amplitude levels is a basic condition of pulse-code modulation.

The speech signal is not continues signal, its amplitude is changes with time. So that here we uses the non-uniform quantizer.

Non-uniform quantizer having step size are not fixed, step size changes with accordance with speech signal.

Quantizer values are dependent on how many bits are present for one sample.
In this project we use the analog to digital conversion (ADC) of ARM controller.
ARM controller LPC2148 contains the 10-bit ADC internally.
Number of level of quantizer =2^N
N= Bits per sample =10
Number of level of quantizer =2^10
Number of level of quantizer =1024
Means quntizer level is 1024.
Here, for simplicity I divided this level in 8 groups. From every group I considered the middle value. If the value of sample is in between the group value, then automatically this value is nothing but middle value of this group. Like that I organized all this group.

Table 1: *Level are divided*
1024 level are divided in following group:

Group	**Group Range**	**Approximate value**
1	0-127	64
2	128-255	192
3	256-383	320
4	384-511	448
5	512-639	576
6	640-767	704
7	768-895	832
8	896-1023	960

Linear Prediction: FIR (Finite Impulse Response) Filter
Consider a finite –duration FIR discrete time configured in figure 2.

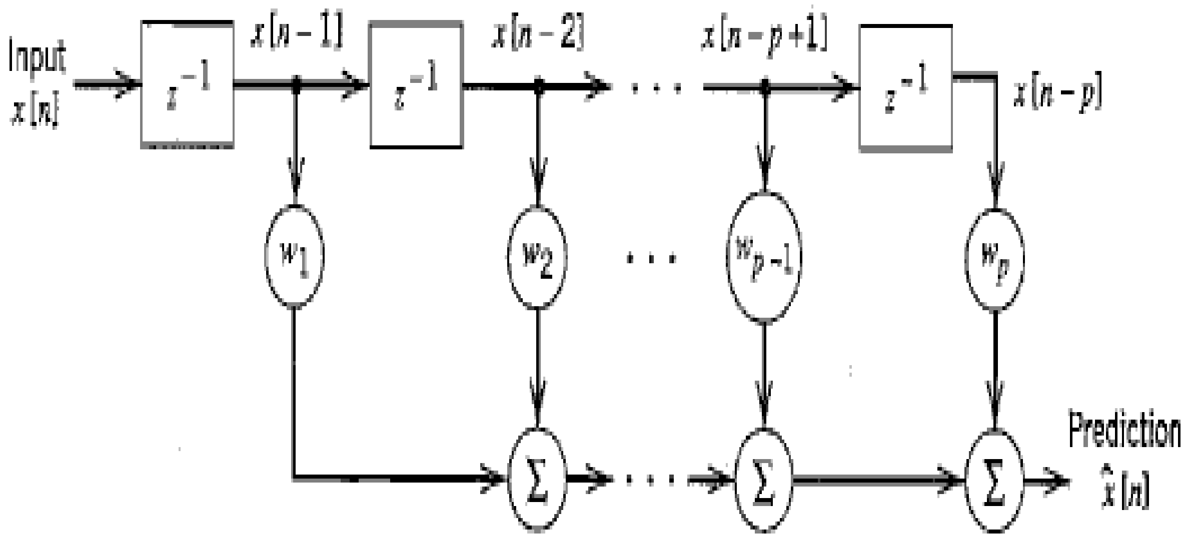

Fig.2 *Block diagram of FIR Filter of order p*

Which involves the use of three functional blocks,

1) Set of 'P' unit delay elements, each of which represented be Z^-1.
2) Set of multipliers involving the filter coefficients w1,w2,…..,wp.
3) Set of adders used to sum the scaled versions of the delayed inputs x(n-1),x(n-2),……......,
   x(n-P) to produce the output x(n).
   The filter output x(n) or the linear prediction of input signal, is thus defined by convolution sum
   $m\varnothing_q[n]=\hat{x}(n)= \sum_{k=0}^{10} Wk * x(n-K)$

Where, P is the number of unit delay element

The actual sample at time (t) is m(n). The prediction error, denoted by e(n), is defined as the difference between m(n) and prediction $m\varnothing_q$ [n], as shown by

$g(k) = m(n) - m\varnothing_q [n]$

Here, I consider the 11 sample signal per signal. The filter coefficients are find by using low pass filter of finite impulse response.

Table 2: *Filter coefficients of FIR*

Sr. No.s	Filter coefficients	Value
1	W0	0.06366
2	W1	0
3	W2	-0.106
4	W3	0
5	W4	0.3183
6	W5	0.5
7	W6	0.3183
8	W7	0
9	W8	-0.106
10	W9	0
11	W10	0.06366

***Table 3:** Filter coefficients and delayed signal*

Sr. No.s	Filter coefficients *Delayed signal	value
1	W0*x(n)	0.06366*x(n)
2	W1*x(n-1)	0* x(n-1)
3	W2*x(n-2)	-0.106*x(n-2)
4	W3*x(n-3)	0*x(n-3)
5	W4*x(n-4)	0.3183*x(n-4)
6	W5*x(n-5)	0.5*x(n-5)
7	W6*x(n-6)	0.3183*x(n-6)
8	W7*x(n-7)	0*x(n-7)
9	W8*x(n-8)	-0.106*x(n-8)
10	W9*x(n-9)	0*x(n-9)
11	W10*x(n-10)	0.06366*x(n-10)

$m\O q[n]=\hat{x}(n)=W0*x(n)+W1*x(n-1)+W2*x(n-2)+W3*x(n-3)+W4*x(n-4)+W5*x(n-5)+W6*x(n-6)+W7*x(n-7)+W8*x(n-8)+W9*x(n-9)+W10*x(n-10)$

WORKING BLOCK DIAGRAM

Figure 3 Shows that Hardware diagram of Speech compression by using Adaptive Differential Pulse Code Modulation.

This project contains the minimum hardware circuit.

ARM Controller

ARM controller plays very important role. In this controller in built 10- bit analog to digital conversion(ADC) & Digital to Analog conversion. Internal memory of ARM controller is sufficient for this project.

1) Mice
2) Audio amplifier
3) Speaker

The human voice frequency signal having ranges 300 Hz to 3 KHz which has taken from mice. This signal having amplitude is variable. This signal is applied to ADC pin of ARM controller. Internal ADC makes the 1024 sample of signal because ADC has 10-bit. That sample we apply in ADPCM algorithm technique.

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.

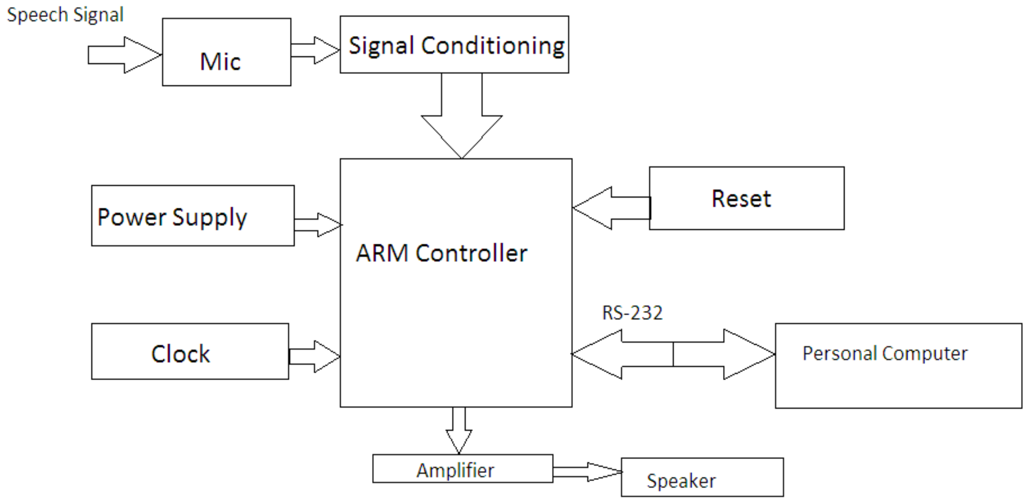

***Fig.3** Speech Compression by using Differential Pulse Code Modulation*

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RESULTS

After completion of operation, we check the compressed sound signal & original sound signal on same speaker. The quality of signal remains same. Both the signal checking on DSO also.

The Figure 4 shows original sound waveform on DSO. This sound signal amplitude is variable. Sound signal frequency is 20hz to 20Khz. For differential pulse code modulation (ADPCM) 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.5 shows the pulse code modulated sound signal.

Fig.6 shows the Differential Pulse Code Modulation signal means compressed signal.

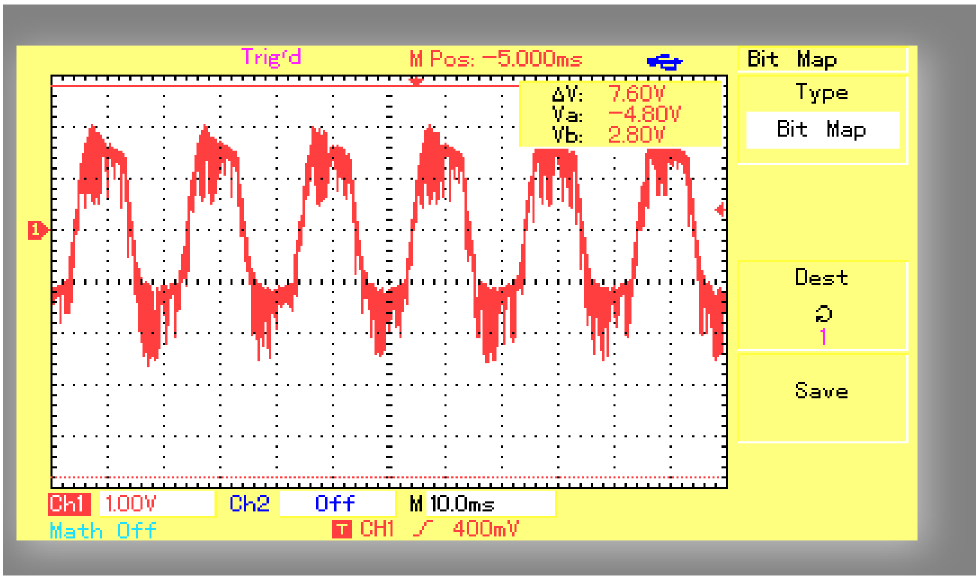

Fig.4 *Original sound signal on DSO*

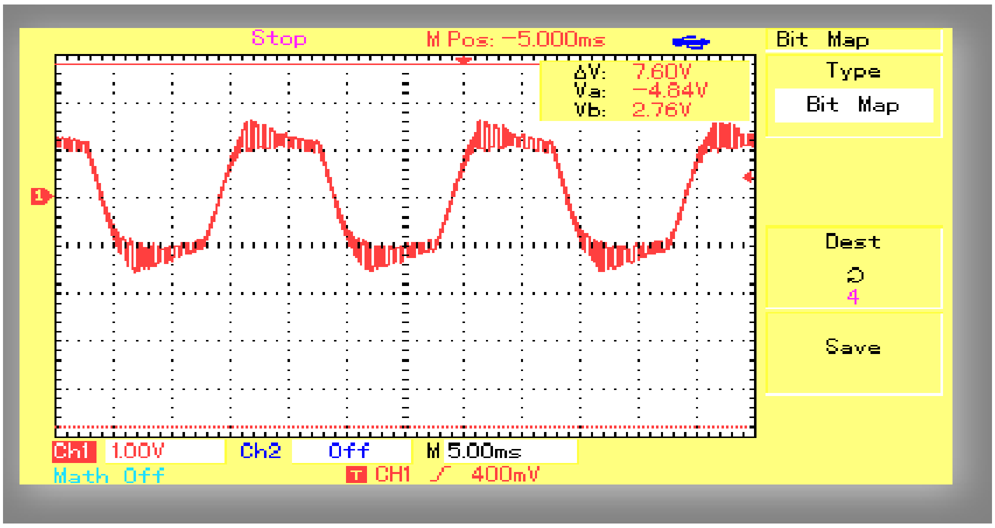

Fig.5 *Pulse Code Modulated sound signal of original signal on DSO*

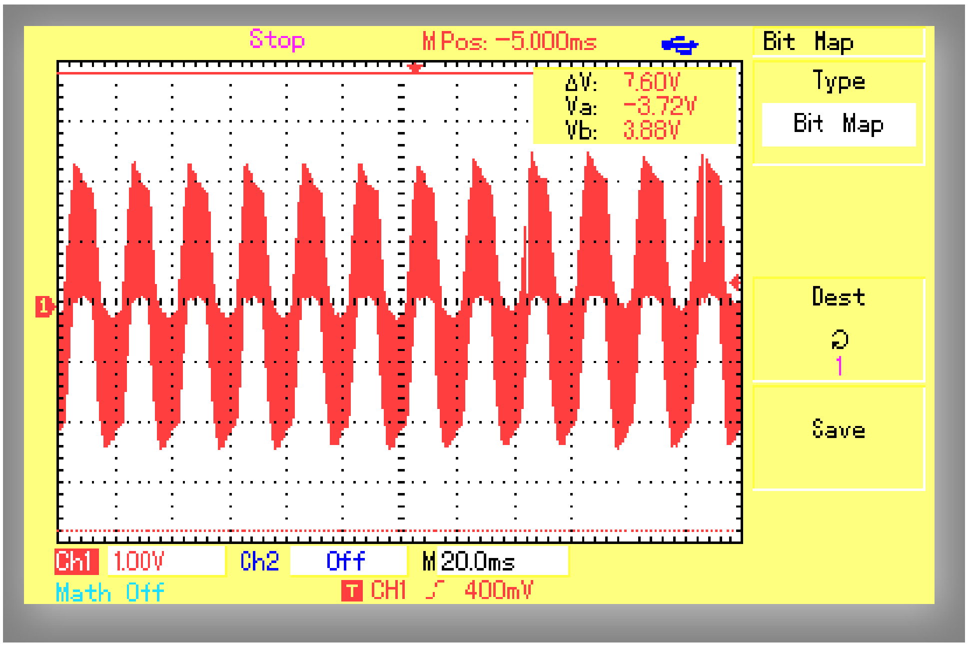

Fig.6 *Differential Pulse Code Modulated sound signal of original signal on DSO*

FLOWCHART

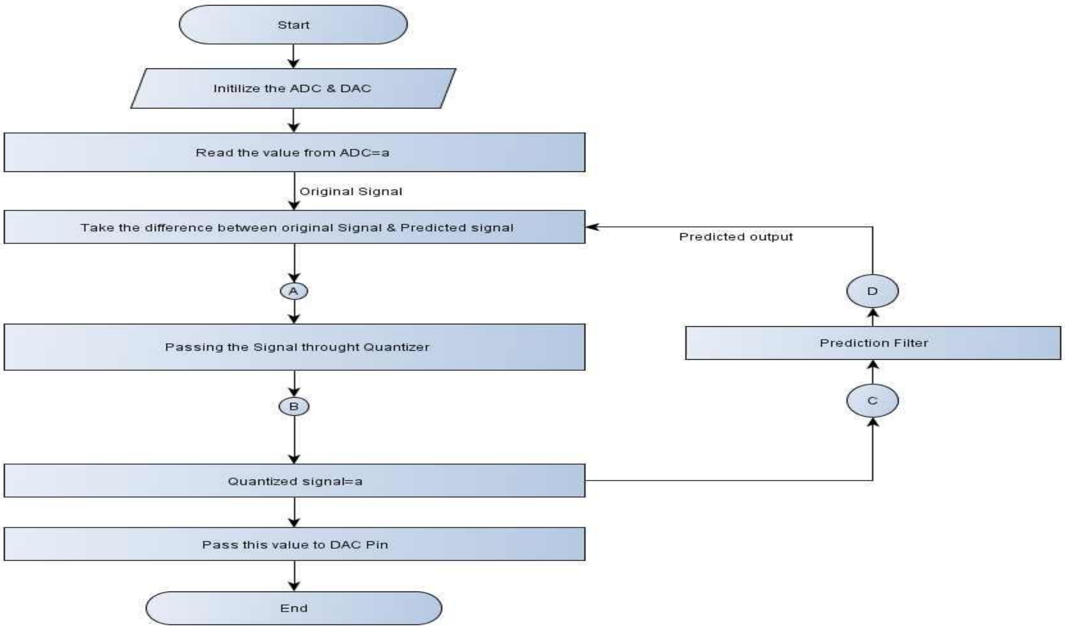

FIG.7 *FLOW CHART OF PROJECT*

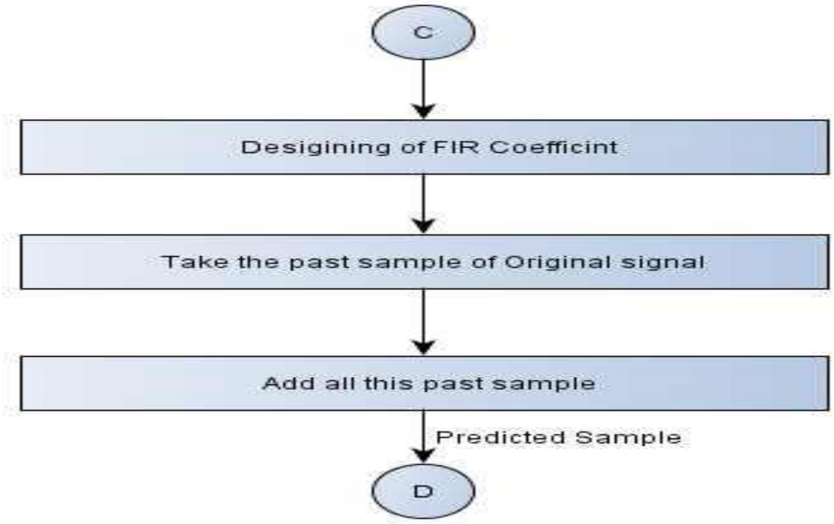

FIG.8 *FLOW CHART OF PROJECT*

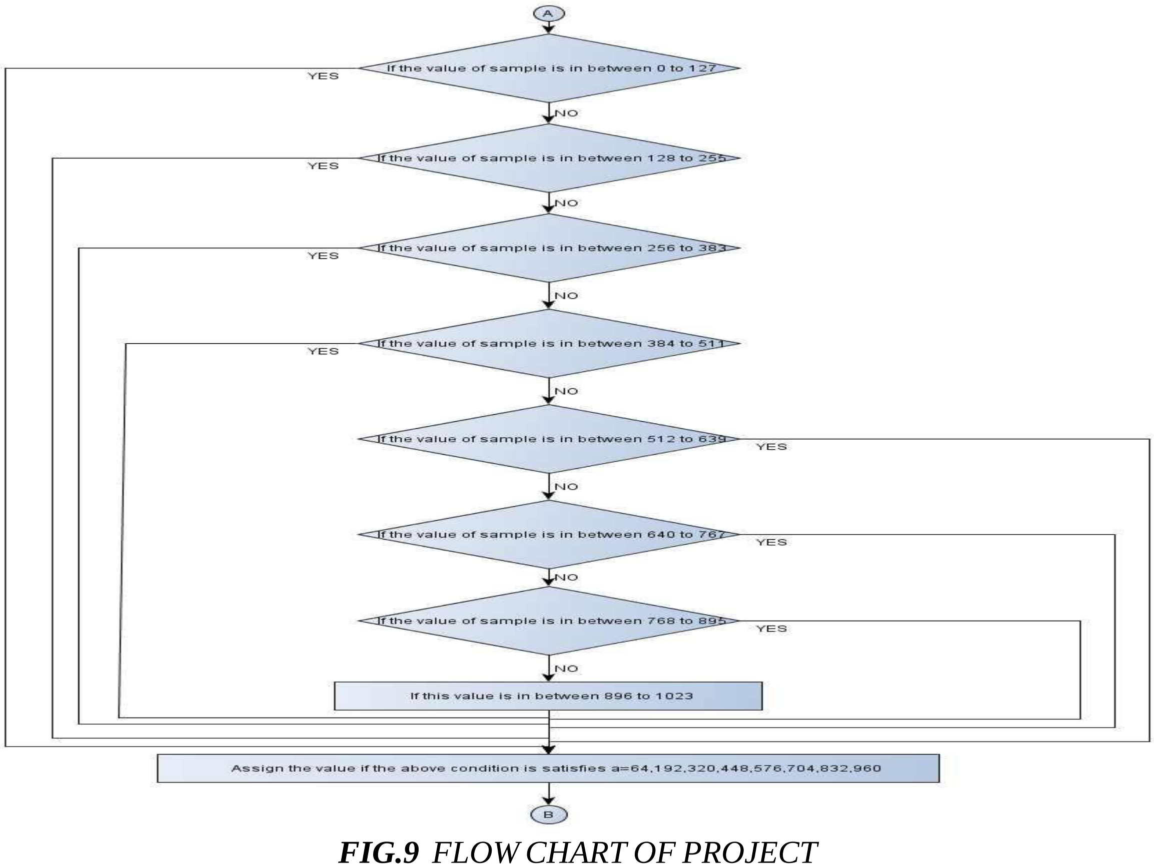

FIG.9 *FLOW CHART OF PROJECT*

The Figure 7, 8 & 9 shows flow chart of project. In which Figure 7 shows main project flowchart, figure 7 shows flowchart of between point C & D and figure 8 shows flowchart of between point A & B.

CONCLUSION

In this work we studied about Speech compression by Differential pulse code modulation (DPCM) & Adaptive Differential pulse code modulation (ADPCM) , How to ADPCM takes the advantage of highly correlated signal. Also we studied about the how the bit per sample is reduced. We also study about the prediction filter effect by using the finite impulse response on ADPCM. Also studied about how to implement quantizer for the pulse code modulation. Because of this system we save the memory of data. Also bandwidth is reduced for transmission of data.

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