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Analog Data Signal Compression by using Differential Pulse Code Modulation with Microcontroller

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Abstract— Compression is a process of converting an input data stream into another data stream that has a small bit size. Compression is take place because of data is normally represented in the computer in a format that is longer than necessary i.e. the input information has some amount of repetition of data associated with it. The main aim of compression systems is to eliminate this repeated of data. The idea of speech data compression is to reduce the bit rate for transmission and storage while either maintaining the quality original information. In analog to digital conversion process, we get the sample signal. a high correlation is found between consecutive speech samples. The DPCM algorithm takes advantage of this high correlation property of speech sample data. The DPCM algorithm does not encode the original speech samples. This algorithm takes the difference between a predicted speech sample and the actual speech sample & then encode that difference. This methods do the efficient compression with a significant reduction in the number of bits per sample. Speech signal quality also remains same.

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, Quantizer, 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 face this problem we need to growth in the size of databases, additional storage devices and the modems and multiplexers have to be continuously upgraded in order to permit large amounts of data transfer between computers and remote centres. Because of this 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 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. When compression is used to reduce storage requirements, overall program execution time is also reduced. This is because reduction in storage will result in the reduction access attempts.

With respect to transferring of data, the data (Bit) 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 transmitted data. A combination of coder and decoder is called as codec.

II. COMPRESSION OF SPEECH SIGNAL

The benefit 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 pulse code modulation (PCM), DPCM can increase the signal to noise ratio (SNR) for some signals by 20 dB. 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.

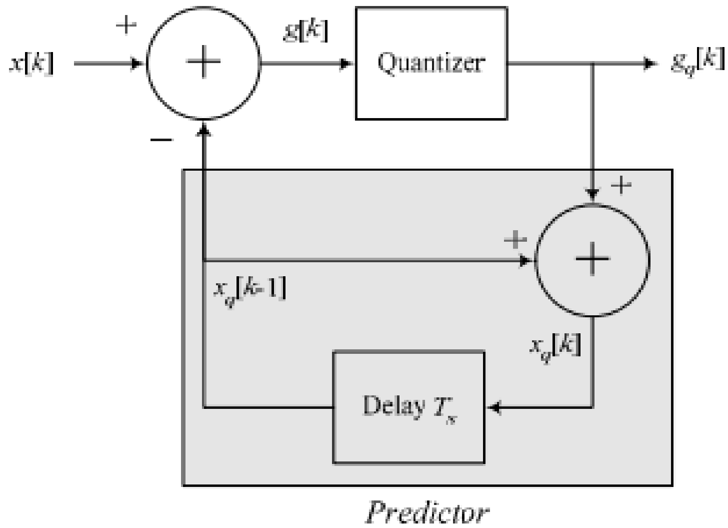

Fig.1 Differential Pulse Code Modulation

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,

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

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

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

From the block diagram,

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

So, in fact, the function Xq[k] is the quantized form of X[k] as 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].

III. WORKING BLOCK DIAGRAM

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

A. ARM Controller
B. Mice
C. Audio amplifier
D. 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.

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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.

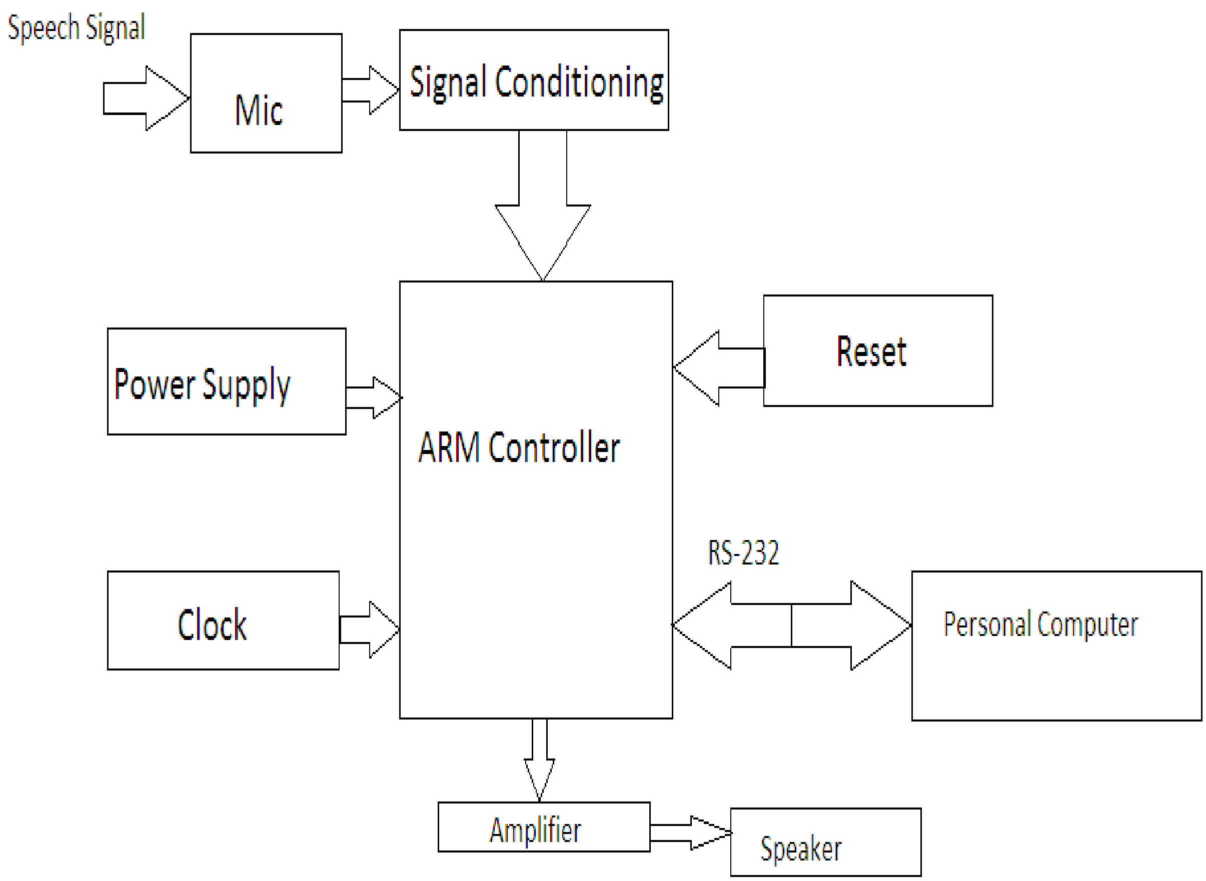

Fig.2 Speech Compression by using Differential Pulse Code Modulation

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.

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.

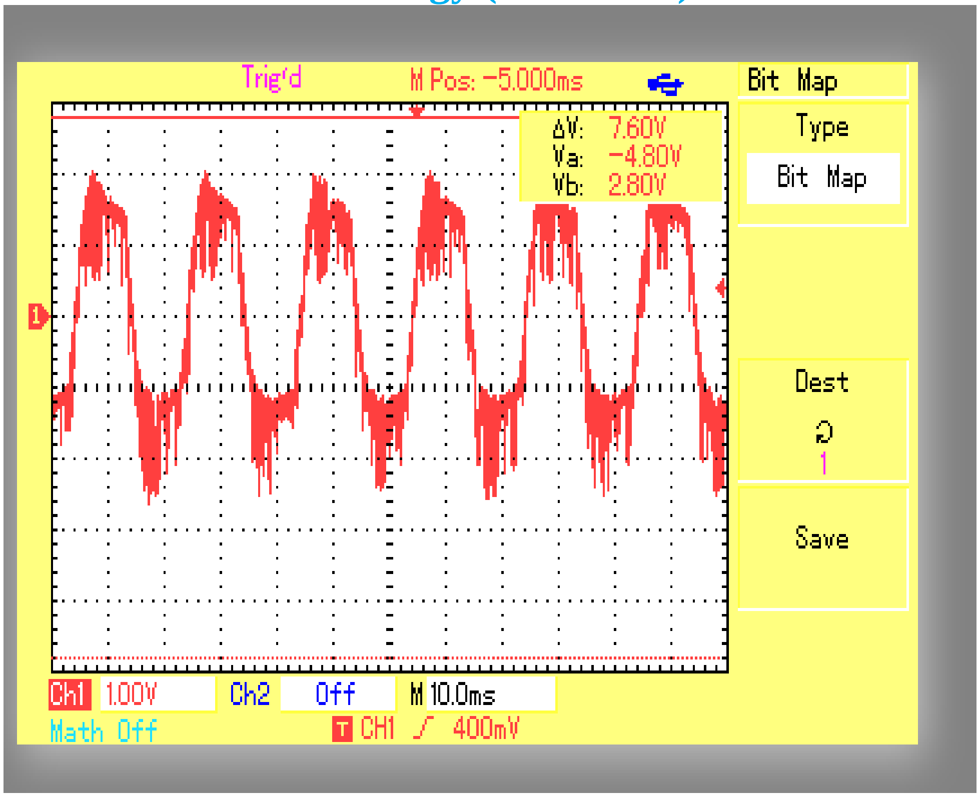

Fig.3 Original sound signal on DSO

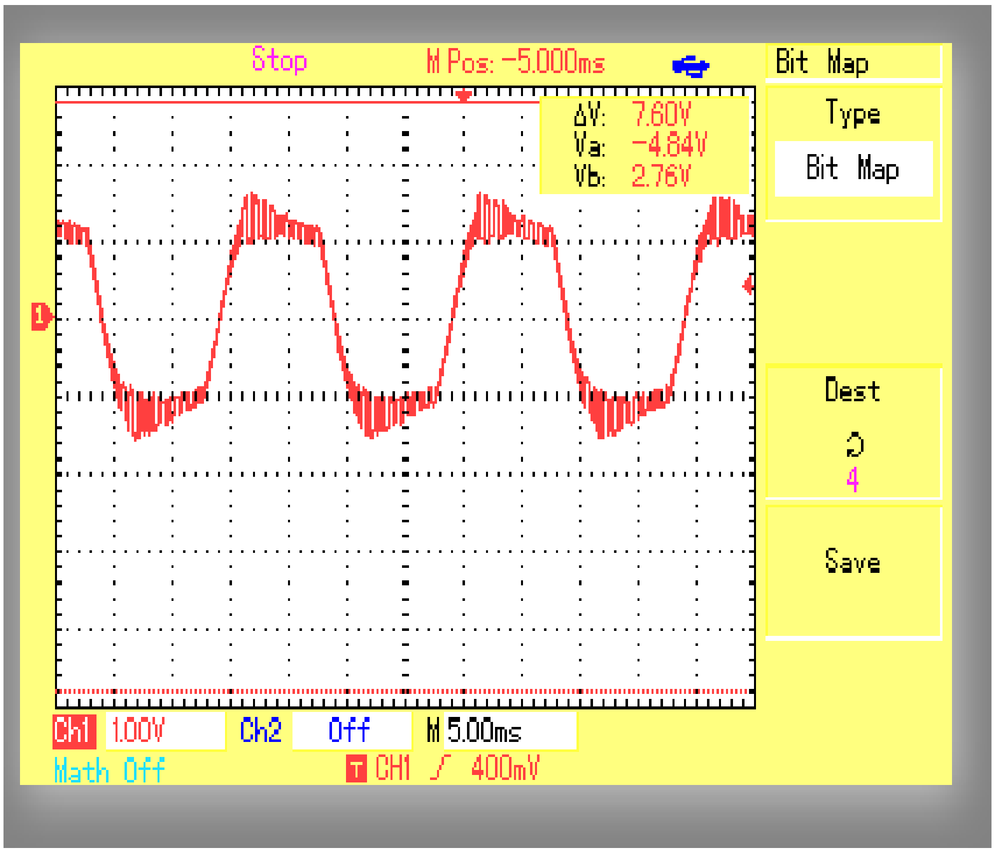

Fig.4 Pulse Code Modulated sound signal of original signal on DSO

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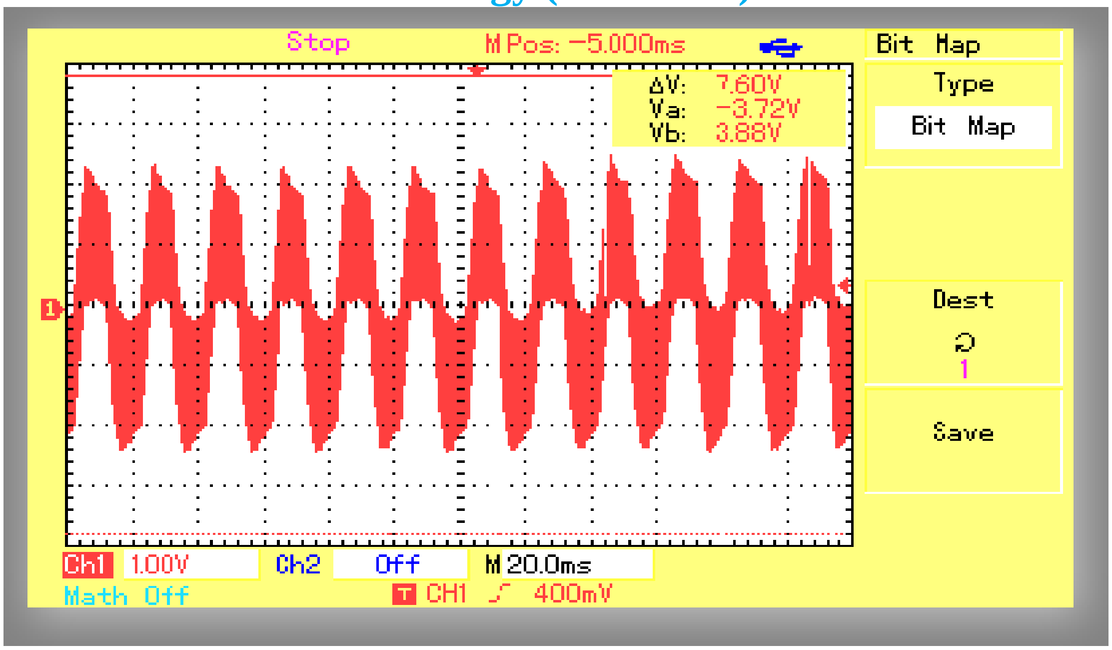

Fig.5 Differential Pulse Code Modulated sound signal of original signal on DSO

V. FLOWCHART

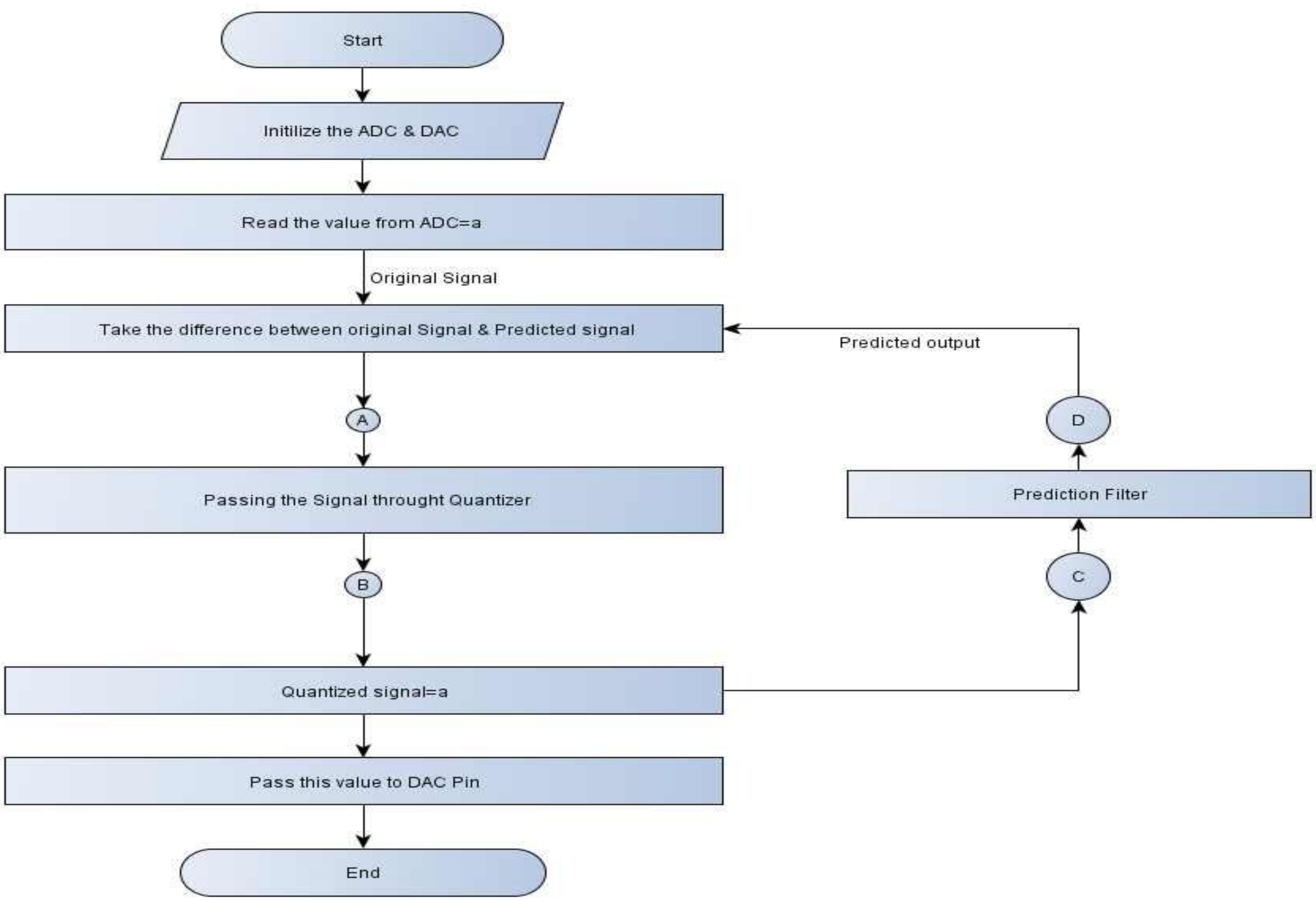

Fig.6 FLOW CHART OF PROJECT

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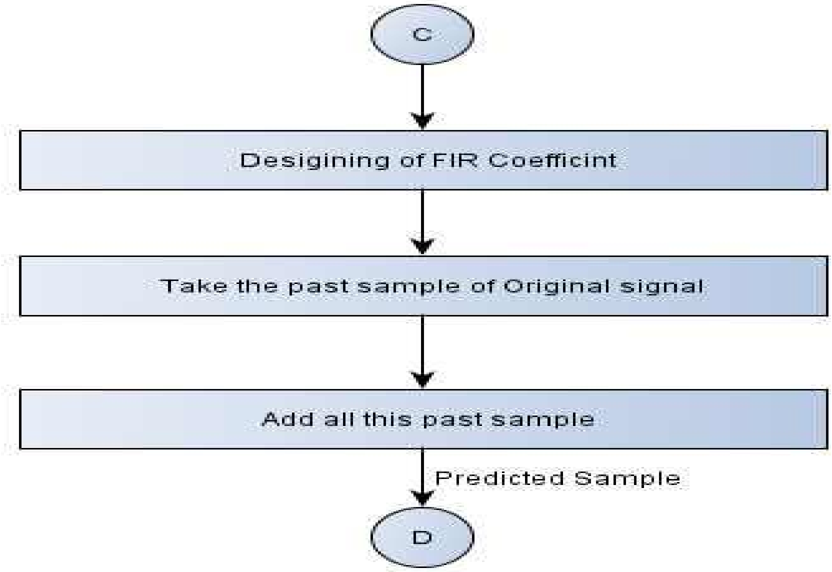

Fig.7 FLOW CHART OF PROJECT

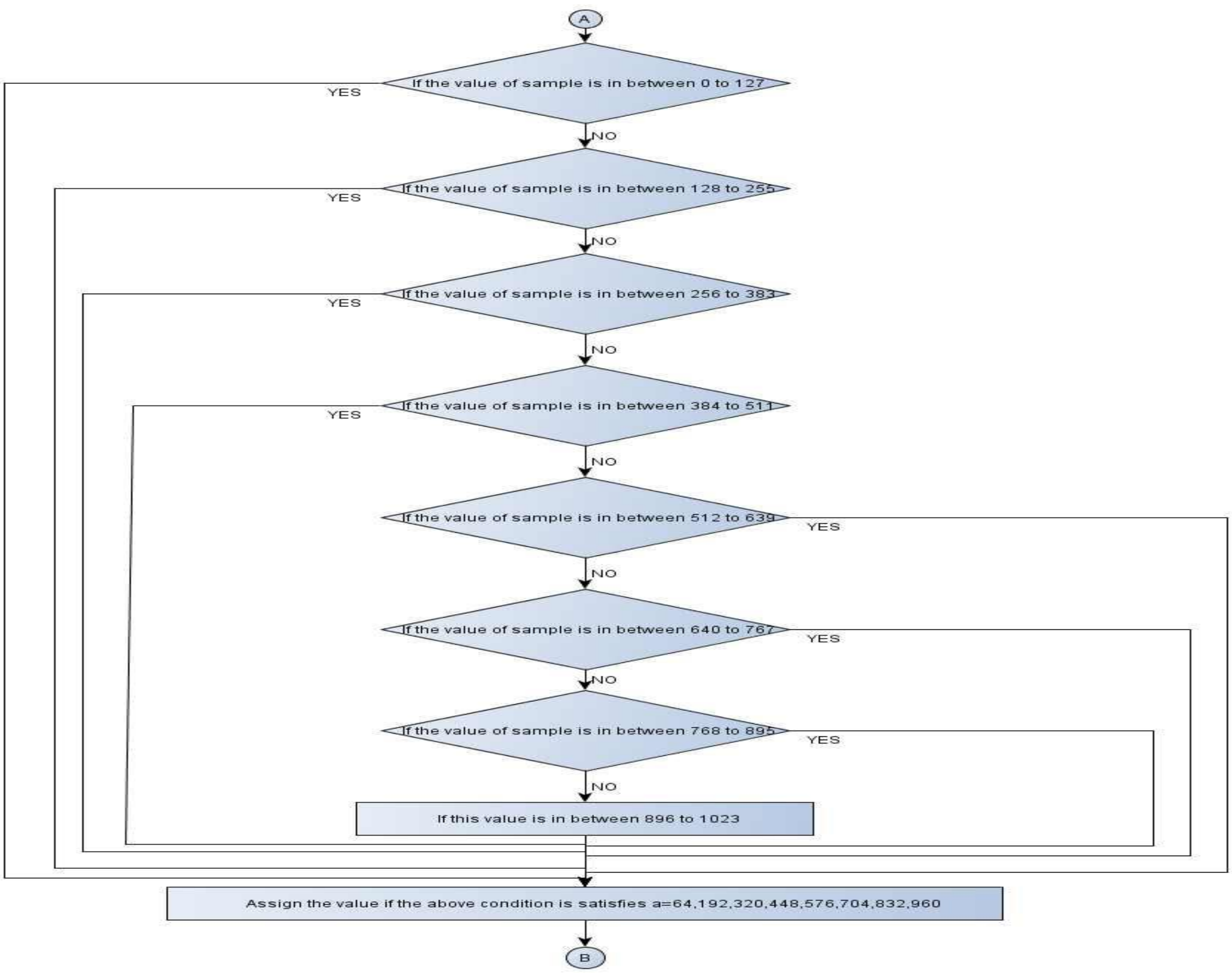

Fig.8 FLOW CHART OF PROJECT

The Figure 6, 7 & 8 shows flow chart of project. In which Figure 6 shows main project flowchart, figure 7 shows flowchart of between point C & D and figure 8 shows flowchart of between point A & B.

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