Analog Signal Compression & Decompression by using ADSP Processor

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Abstract

The analog speech signal is digitized by sampling. For maintaining the voice quality, each sample has to be represented by 13 or 16 bits. The compression is nothing but to reduce the original data bits from higher bits to lower bits with good quality of signal as compared to original signal and decompression is reconstructed original signal from the compressed signal.

Adaptive differential pulse code modulation is very useful & efficient for compression & decompression designed by Bell labs in 1970 for the reduction of bits of Analog signal. The ADPCM uses the difference techniques of next samples of original signal & predicted signal of last sample.

The analog speech signals are amplified by the pre-amplifier and fed to the CODEC for analog to digital conversion. The CODEC transmits the digitized signal to the ADSP 2105/2115 processor, which then compresses the speech data using the ADPCM techniques and store in RAM. When the processor is interrupted, it reads the compressed data from RAM expands the data and send the data to CODEC. CODEC are used for the conversion of digital signal data to analog signal data which amplified by amplifier and output applied for the speaker. The ADPCM algorithms are going to use in ADSP processor, the compressed signal is stored in RAM, that signal is applied to the speaker & going to check the quality of signal with original signal. That stored compressed signal is then applied to the decompression techniques of ADPCM in ADSP processor.

The ADSP-2100 processor is a digital signal processing (DSP) with high speed processing applications built in single chip. That contains the computation units, address data generators & on chip program & data memory, serials ports that operate on 25 Mhz with 40ns instruction cycle time.

This project played very important role because this project is going to use in different application, like data transfer from one place to another place by using wire in computer application. In communication field this compression techniques is very useful to stores the data with less memory.

Keywords: *ADSP 2105/2115 processor**,** ADPCM Algorithm, Quantizer, Audio Amplifier, PC, DSO*

INTRODUCTION

The Human voice signal in its inherent form of acoustic signal. For the purpose of transmission we required to converts this signal to in Electrical signal with the help of transducers.

1. It is a one-dimensional signal, with time as its independent variable.
2. It is random in nature.
3. It is non-stationary, i.e. the frequency spectrum is not constant in time.
4. The human voice signal is audible to human hear is 20Hz to 20 KHz audible frequency range. Means, this frequency range we going to compress the signal

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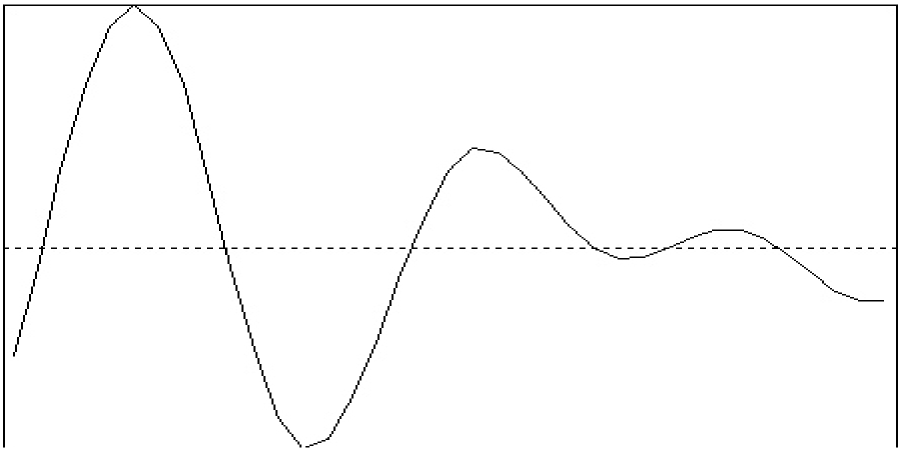

***Fig 1.1**: Sound signal voltage v/s Time*

Digital representation of speech
Digital signal which is nothing but the sample of original signal with same frequency, that samples we going to quantized at discrete levels.
Thus, parameters of digital speech are:
1. Sampling rate
2. Bits per second
3. Number of channels.

Goal
Data compression along with analog to digital conversion is an important factor in data Communication and data storage. The Adaptive Differential Pulse Code Modulation is a technique that gives above mentioned advantages along with Excellent Speech Quality, 4:1 or 8:1 compression ratios.

Compression – An Overview
Today the communication techniques increases rapidly and because of that storage of data as it is in memory become costly, means this become one problem to stores the original message as it. So on that problems one solution is that compression the data with same quality.

Why Compression?
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.

The bits rate reduces by using compression which is nothing but coder and data decompress by using decompression is called as decoder. The combination of coder & decoder which is nothing but "CODEC".

Compression Terminology
Compression ratio: - The compression ratio is defined as:-
Compression ratio = size of the output stream/size of the input stream
Compression factor: - It is the inverse of compression ratio.
Values greater than 1 indicate compression and less than 1 indicates expansion.
Signal:
There are so many analog signal available which taken the more size of memory, for the purpose we want the compression.
1) Sine wave signal
Sine wave is a continuous analog signal.

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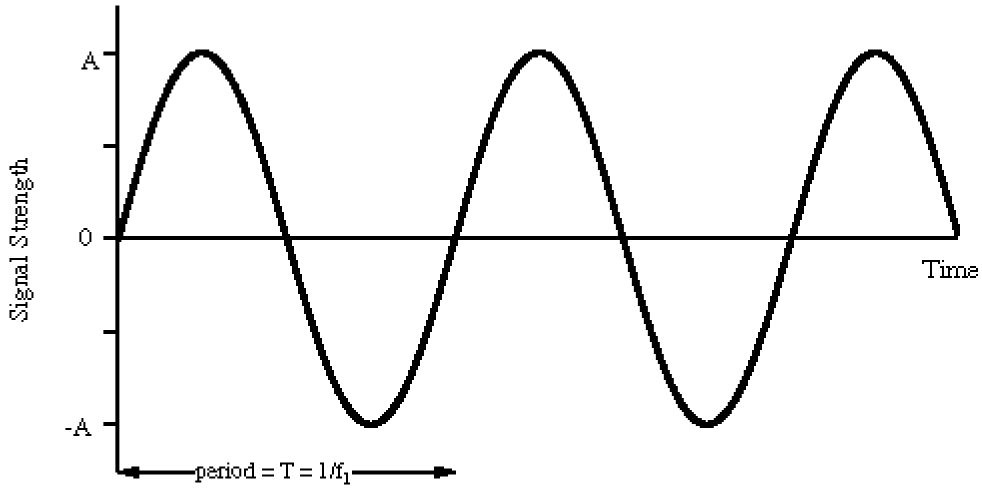

(a) Sine Wave

2) All sensor signals
a) Temperature Sensor
b) Pressure Sensor
c) Humidity sensor
d) Speech signal
e) Video signal

OPERATION OF ADPCM

The ADPCM algorithms take advantage of samples in which that takes the difference between the next sample signal & last predicted samples, that difference signal are encode by the ADPCM. Means this is very efficient techniques of compression which reduces the number of bits per sample.

Compression

The input signal s_i to the speech encoder is supposed to be the 16-bit speech sample. Fig.1.2 shows the block diagram for ADPCM compression algorithm. For the next iteration of the encoder, the predicted sample s_p and the quantized step size index are saved in a structure. For this iterative process, the quantization step size index and the predicted sample s_p are initially set to zero. The input s_i to the speech encoder is supposed to be a 16-bit 2's complement speech sample, while the value returned by the speech encoder is an 8-bit number which contains the 4-bit sign magnitude ADPCM code.

The difference ***d*** is produced by subtracting the predicted sample s_p from the input signal s_i. The 4-bit ADPCM value ***t*** is obtained by performing adaptive quantization on the difference obtained in the previous step. Using that ADPCM value the encoder and decoder update their internal variables.

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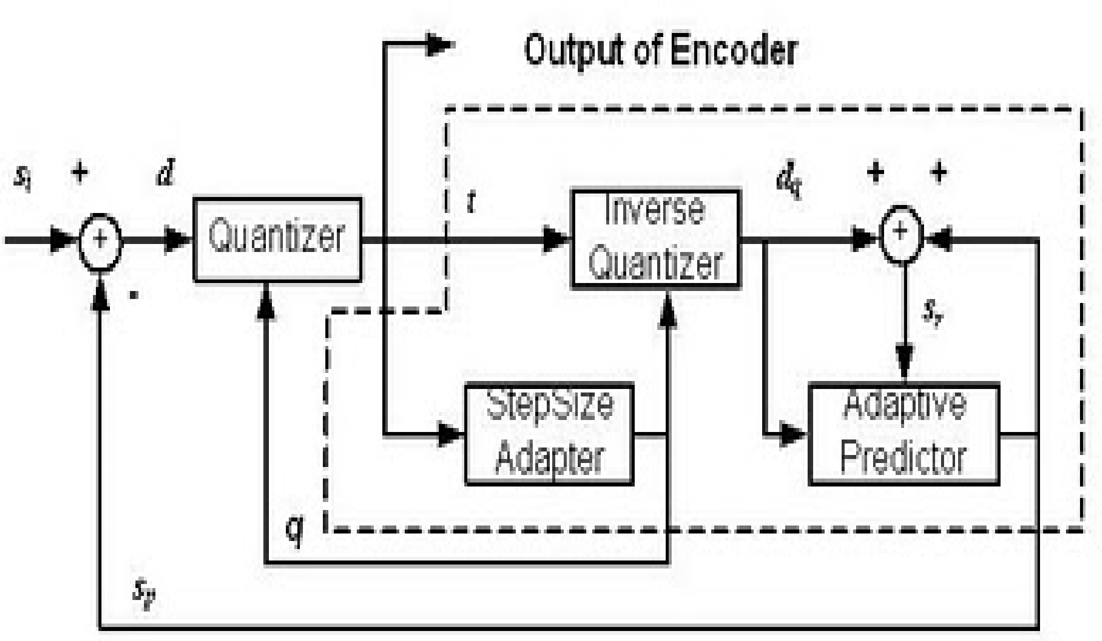

***Fig 1.2:** ADPCM Encoder block diagram*

Algorithm 1: ADPCM Encoder Algorithm

1. ADPCM Encoder takes a 16-bit signed number (Speech sample,-32768 to +32767) returns an 8-bit number containing the 4-bit ADPCM Code(0-15).
2. This stores the previous values of prediction values (**s_p**) and the quantizer step size index.
3. Take the difference(**d**) between the next sample (**s_i**) and the predicted sample (**s_p**).
4. Then quantizer of ADPCM Quantize the difference(**d**) signal.
5. Fixed predictor is designed by using FIR filter computes new predicted sample(**s_r**) and adding the old predicted sample signal (**s_p**) to the predicted difference(**d_q**).
6. Check for overflow of the new predicted sample (**s_r**). **s_r**, which is a signed 16-bit sample, must be in the range of -32768 to +32767.
7. Find the new quantizer step size(**q**) by adding the previous index and a look up table using the ADPCM code (**t**).
8. Check for overflow of the new quantizer step size index.
9. Save the new predicted sample (**s_r**) and quantizer step size index for next iteration. Return the ADPCM code (**t**).

Decompression

The block diagram of the ADPCM decompression is shown in Figure 1.3. The decoder is same as the one used by the encoder routine. For updating the inverse quantizer it uses the ADPCM value, which in turn produces the difference d_q. The output sample s_r is produced by adding the difference d_q to the predicted sample s_p. For next iteration of the decoder the output sample s_r is saved into the predicted sample.

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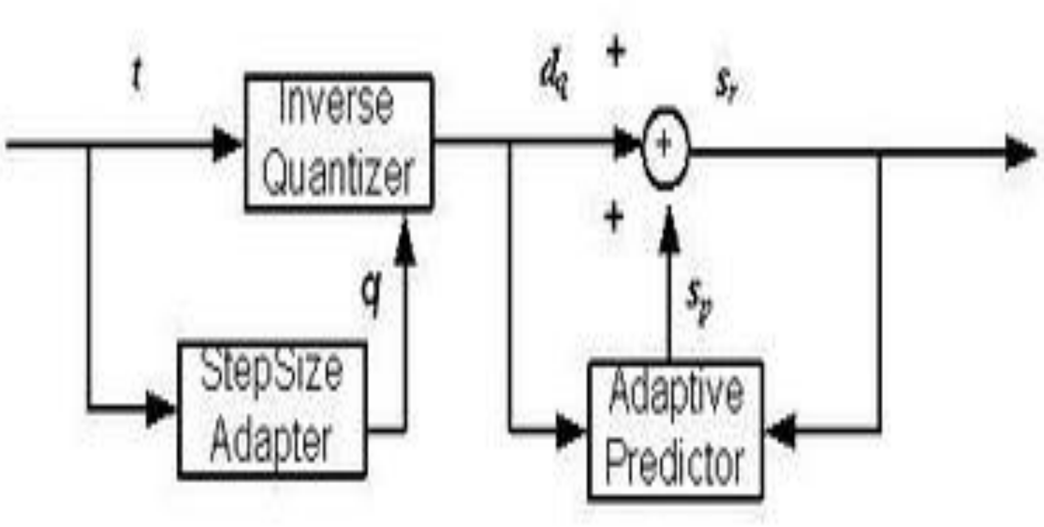

FIG 1.3: *ADPCM DECODER BLOCK DIAGRAM*

Algorithm 2: ADPCM Decoder Algorithm

1. ADPCMDencoder takes an 8-bit signed number containing the 4-bit ADPCM Code (0-15) and returns a 16-bit signed number (Speech Sample,-32768 to +32767).
2. Restore the previous values of the predicted sample (**s_p**) and the quantizer step size index.
3. Find the quantizer step size (**q**) from a lookup table using the quantizer step size index..
4. Inverse quantize the ADPCM code (**t**) into
   a predicted difference (**d_q**) using the quantizer step size (**q**).
5. Fixed predictor finds out the new predicted sample (**s_r**) by the addition of the old predicted sample (**s_p**) to the predicted difference (**d_q**).
6. Check the new quantizer step size.
7. Save the new predicted sample (**s_r**) and quantizer step size index for next iteration.
8. Return the new sample (**s_r**).

WORKING BLOCK DIAGRAM

The figure 1.4 shows the block diagram of compression & decompression contains the ADSP processor, CODEC, Memory block (RAM & ROM), amplifier, mice and speakers. Also this system is interfaced by using serial port with personal computer.

The TTL logic levels of serial port are converted to RS232 level using level converter, so that the system can directly communicate with the standard serial port (com1/com2) of personal computer.

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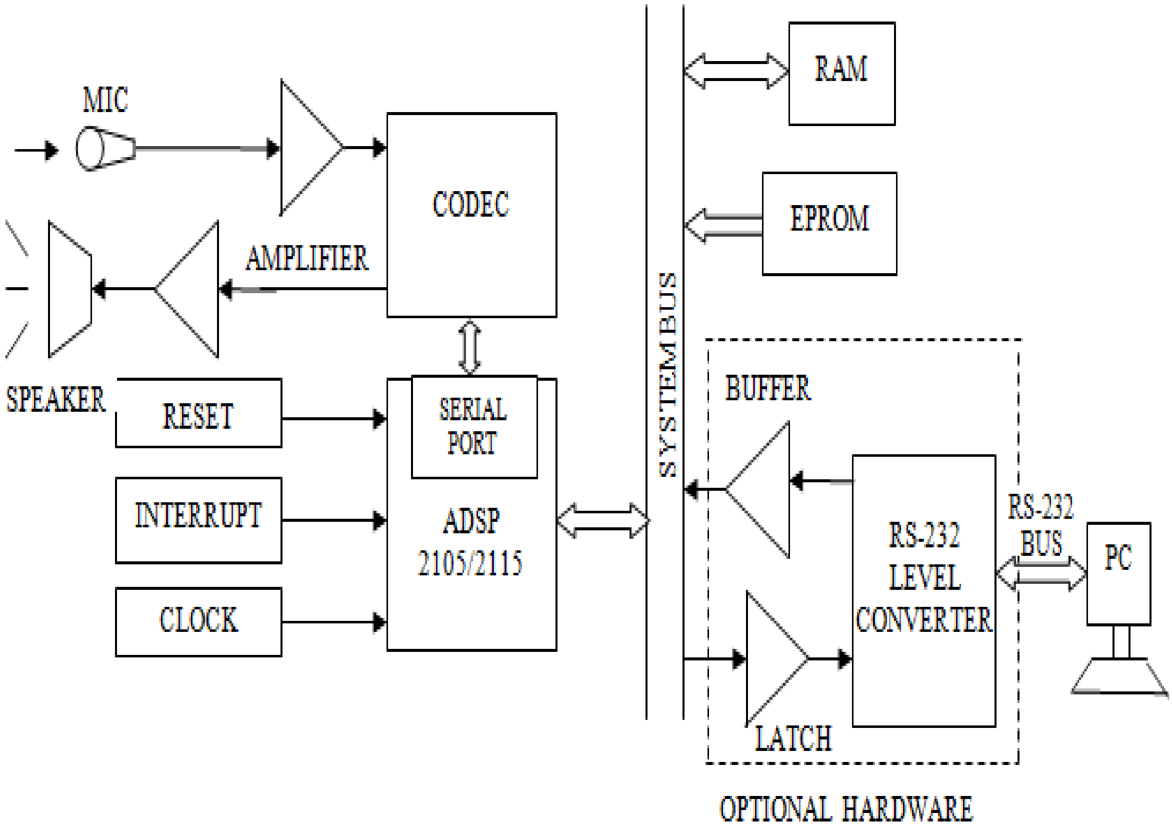

FIG 1.4: *BLOCK DIAGRAM OF COMPRESSION & DECOMPRESSION*

The system amplified the analog speech signal and fed to CODEC for the analog to digital conversion. The CODEC transmits the digitized signal to the ADSP 2105/2115 processor, which then compresses the speech data using the ADPCM techniques and store in RAM. When the processor is interrupted, it reads the compressed data from RAM expands the data and send the data to CODEC. The CODEC converts the digital data to analog signal, which is amplified and output through the speaker.

CONCLUSION

This system works on compression & decompression of any analog signal. Because of that we going to saves the memory block, also because of compression we save the transmission time between the two computers, hence the bandwidth of signal is also reduces.

ACKNOWLEDGMENT

We would like to thanks to my guide Parents, Friend & all faculty members for their valuable suggestions and consistent encouragement.

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