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Study on performance analysis of Pulse Code Modulation (PCM)

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***Abstract*—Pulse Code Modulation (PCM) is digital scheme for transmitting analog signal. The concept of PCM came into existence to overcome the problem of the additive nature of noise when an analog signal underwent multiple amplifications along a long distance line.**

Our paper concentrates on the performance analysis of PCM using Simulink. The methods involved in this process are sampling, quantization and encoding of analog signal to obtain the PCM output. We have used sinusoidal input and then collected the samples, quantized, finally encoded them to successfully obtained the result which satisfied us.

The main idea behind it was to solve the noise issue by transmitting the digitized version of analog signal. Being a digital signal, PCM output is encoded and hence secure. Low noise susceptibility, repeatability are its other benefits. Its area of implementation includes CD laser disks, digital audio recording, digitized video special effects, voice mails, most importantly radio control.

***Index Terms*- Companding, Encoding, Quantization, Quantization error, Sampling etc...**

I. INTRODUCTION

PCM is digital scheme for transmitting analog signal. So, to obtain PCM from an analog waveform at the transmitter end, the amplitude of signal is sampled at regular time interval. This process is known as SAMPLING. At each sample unit, the instantaneous amplitude of the analog signal is rounded off to the nearest of the several specific, predetermined levels. This process is known as QUANTIZATION. The number of levels are always a power of 2 such as 8, 16, 32 which can be represented by 3, 4, or 5 binary bits respectively. This process in called ENCODING [2,5].

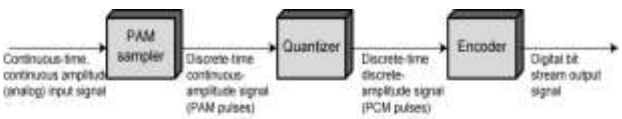

Fig 1- Block diagram of PCM

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At the receiver's end, reverse process takes place. The demodulator converts binary bits back into pulses of same quantum levels. These pulses are further processed to get the original analog waveform.

Before we proceed further let us review our concepts about signal and its various types.

SIGNAL: - In general, signal is a single-valued function of one or more independent variables. But for communication purposes signal is a time varying quantity.

CLASSIFICATION OF SIGNALS: - Signals can be classified on various basis. On the basis of nature in time domain, signals can be broadly categorized as-

a). Continuous time signals- A signal, x(t) is continuous time signal if 't' is continuous variable. An analog signal is the example of continuous time signal.

b). Discrete time signal- If x(t) is defined at discrete times then x(t) is a discrete time signal. A digital signal is the example of discrete time signal.

Generation of PCM output involves following processes:

II. SAMPLING

By the process of Sampling Discrete Signals are created by taking samples of an Analog Signal. At regular intervals like time base (for time domain) or space based (for space domain) or any other independent variables these samples are taken. Here we will prefer, intervals to be time based and these regular intervals define the Sampling Rate as the rate at which samples are taken, also called Sampling Frequency Fs.

Sampling is governed by Nyquist Sampling Theorem, which implies when a signal is sampled, the Sampling Rate (or Sampling Frequency) must be at least twice the highest frequency the signal contains to avoid Aliasing (Overlapping) [3].

Mathematically, $F_s \geq 2F_m$.
Here, Fs= Sampling Frequency
Fm= Modulating Frequency

Sampling Techniques are of three types-

a). Impulse Sampling: - Impulse sampling can be performed by multiplying analog input signal x(t) by impulse train of period 'T'.

Sampling output, y(t)=x(t)* impulse train.

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b). Natural Sampling: - Natural Sampling similar to Impulse Sampling except the fact that it has pulse train instead of impulse train of some finite width and time period 'T'.

Sampling output, y(t)=x(t)* pulse train.

c). Flat Top Sampling: - During transmission noise is introduced at the top of the transmission pulse which can easily be removed if pulse is in the form of Flat Top i.e. top of pulse are flat and have constant amplitude. Hence, it is referred as Flat Top Sampling [3].

For PCM, we will bound our study on flat top as it is noise free and along with that practical as well as simpler.

In Fig 2, the output from the circuit must be held at a constant level for the sampling duration. Vcontrol switches the MOSFET ON until the charge on C is equal to the amplitude of the sampled voltage. Vcontrol then goes LOW, the MOSFET is OFF and the charge is held by the capacitor. This is called a sample and holds circuit and is usually used as the input to an ADC [5].

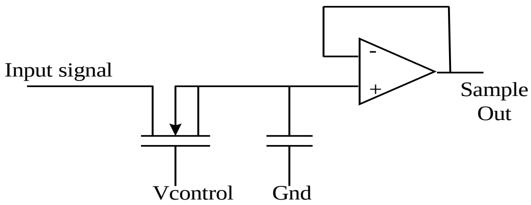

Fig 2- Sampling circuit

Theoretically, the sampled signal can be obtained by convolution of rectangular pulse p(t) with ideally sampled signal say y(t) as shown in diagram below:

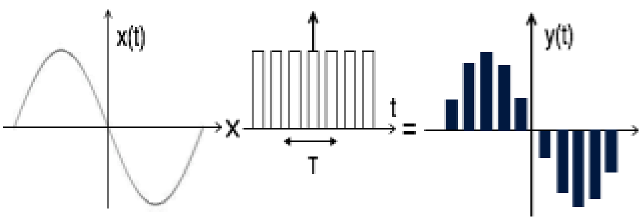

Fig 3- Flat Top Sampling

As shown in Fig 3, the output obtained after sampling an analog signal is a Pulse Amplitude modulated signal. This process is known as Pulse Amplitude Modulation (PAM). The PAM signal is further quantized and encoded to get the PCM output.

III. QUANTIZATION

Quantizing a signal means to limit the possible values that a signal can take on. For instance, a discrete signal can have any range of amplitudes at a discrete time interval [1,2]. Well in order to represent such a signal digitally, we would have to have an infinite amount of bits to do this. Since infinite bit data sample cannot be handled by any computer, we quantize the data to fit within a range so that we are comfortable with it and fits our requirements for the system.

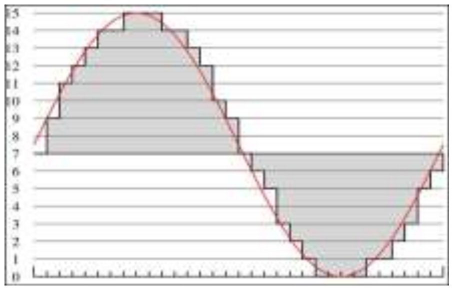

Fig 4. Quantization

While quantizing the sampled signal in uniform manner, we come across new challenge known as QUANTIZATION ERROR. It is the result of quantization in analog -to-digital convertor in the telecommunication systems and signal processing. It is a rounding error between the analog input voltage to the ADC and the output digitized value. The noise is non-linear and signal dependent.

To overcome such problems in uniform quantization, we prefer non-uniform quantization which has relatively less error. The non-uniform quantization is practically achieved through a process called COMPANDING.

Non-uniform uniform quantization may be achieved by first passing the message through a compressor, non-linear device which compresses the peak amplitude. This is followed by uniform quantizer, such that uniform zones at the output correspond to the non-uniform zones at the input. At the receiving end, the compressed signal is passed through an expander, another non-linear device used to cancel the non-linear effect of the compressor. This process is called COMPANDING [2,4]. Two international companding standards retain up to 5 bits of precision by encoding signal data into 8 bits are μ -law and A-law. μ – law is the accepted standards in U.S. and Japan, while A-law is the European accepted standards. Both are discussed in the following sections.

μ-Law Companding

The μ -law compression portion of this standard is defined mathematically by continuous equation [5]:-

$$F(x) = sgn(x)\frac{(\ln(1+\mu|x|)}{\ln(1+\mu)}, \qquad \text{where } -1 \le x \le 1$$

'μ' is the compression parameter and 'x' is the normalized integer to be compressed

.

A-Law Companding

The compression portion of this standard is defined in the given continuous equation [5]:-

$$F(x) = sgn(x)\frac{A|x|}{(1+lnA)}, \qquad where\ 0 \le |x| < \frac{1}{A}$$
$$= sgn(x)\frac{1+lnA|x|}{(1+lnA)}, \qquad where\ 1/A \le |x| \le 1$$

Where, 'A' is the compression parameter and 'x' is the normalized integer to be compressed.

In our context, we have focused on uniform quantization only.

IV. ENCODING

The process of assigning the binary codes to the quantizer output is known as Encoding. Encoding provides security, withstand capability with the channel noise and flexible operation of signal.

There are two types of encoding [2]: -

a). Linear Encoding- In linear encoding quantization levels are evenly spaced. It is simpler but overall signal distortion takes place in it.

b). Non-linear Encoding- In non-linear encoding, the quantization levels are not evenly spaced. As a result, there is reduction of overall signal distortion. It can also be done by companding and more complex as compared to linear encoding.

Now, let's elaborate linear encoding using an example: we consider that the obtained quantized values are 2, 6, 7, etc. then their encoded value will be given as 010, 110, 111 respectively as shown in Fig 5 given below.

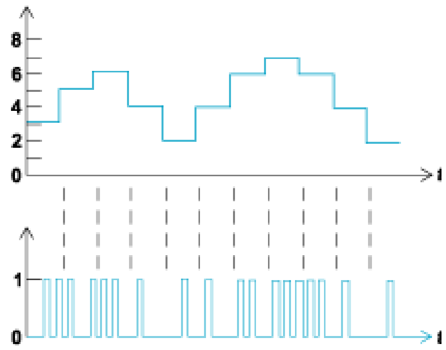

Fig 5- Quantized and Encoded output

Now these digital data are converted into digital signal using various encoding scheme. Encoding scheme is simply mapping of data bits to signal elements. NRZ-L, RZ, AMI, Manchester etc. are the examples of such encoding schemes.

Fig 6 given below depicts show Analog signal is sampled, quantized and then encode to get the PCM output. The corresponding output of PAM sampler, quantizer and encoder has also been shown in the figure given below [5].

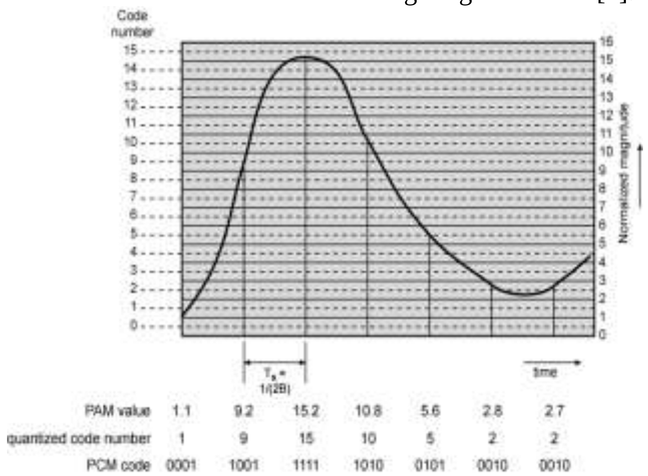

Fig 6- An example of PCM

Thus, there are three elements of PCM system. They are transmitter, transmission medium and receivers. Transmitter part includes sampling, quantization and encoding of Analog signal to PCM signal which have been discussed in the above sections. Our study is limited to transmitter part only.

In transmission medium, regenerative repeaters are present to generate the distorted PCM wave.

Finally, at receiver's end, the reverse process of the transmitter takes place. The original Analog signal is reconstructed using decoders and low pass filters.

ADVANTAGES OF PCM

1. Effect of noise is reduced.
2. PCM permits the use of pulse regeneration.
3. Multiplexing of various PCM signals is possible.
4. Due to digital nature of the signal, we can place repeaters in the transmission medium. In fact, the repeaters regenerate the received PCM signal. This can't be possible in Analog systems since amplifiers are used in it which amplifies the signal as well as noise [4].

DISADVANTAGES OF PCM

1. It is complex as it involves various processes like sampling, quantization, encoding.
2. It requires large bandwidth since all bits obtained after encoding needs to be transferred [4].

PCM transmits all the bits obtained after encoding the quantized signal which was removed in the enhanced version known as Differential Pulse Code Modulation (DPCM). DPCM takes the advantages of the fact that the sample of Analog signals are highly correlated. It utilizes the redundancy present in the signal waveform. Under this, the difference between actual sample value and predicted value based on previous sample is quantized and then encoded which results a digital value.

DPCM will be our concern for further study.

V. SIMULINK MODEL

We have carried out our study of the PCM using Simulink platform.

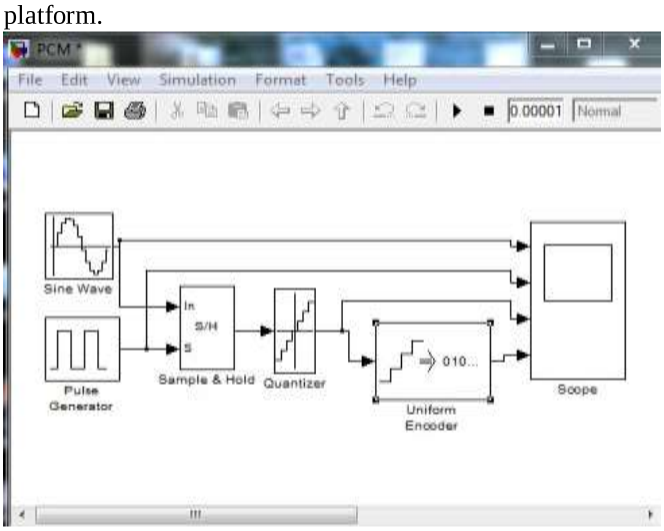

Fig 7- Simulink model of PCM

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Sine wave is taken as an input having frequency in order of Mega-hertz. The duty cycle of the pulses selected are 50% and 99.5%. Input and pulses are sent further in the sample and hold, quantizer and uniform encoder blocks respectively. We have selected zero as initial condition for sample and hold and analysis is performed with quantization interval 0.5, 1 and 2. Encoding of samples are done using 3 and 4 bits.

VI. SIMULATION RESULT

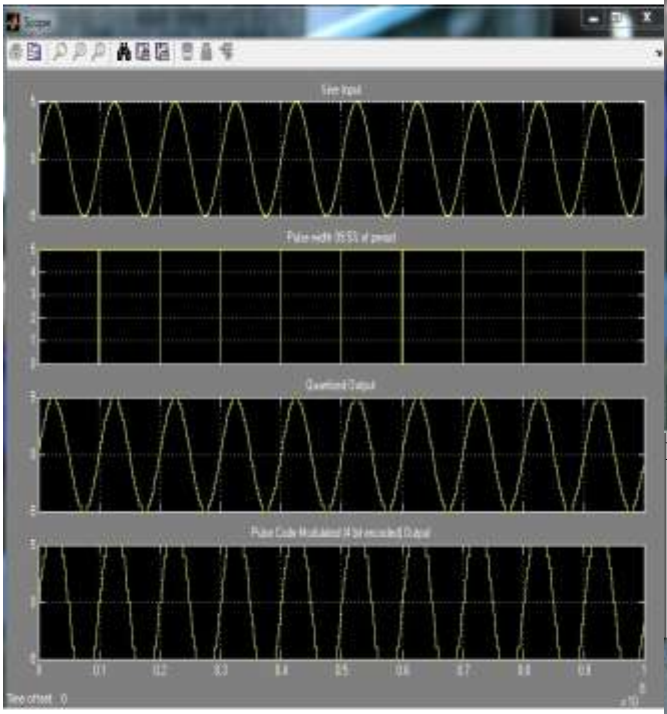

Fig 8 -PCM O/P encoded with 4 bits (99.5% pulse width, 0.5 quantization interval)

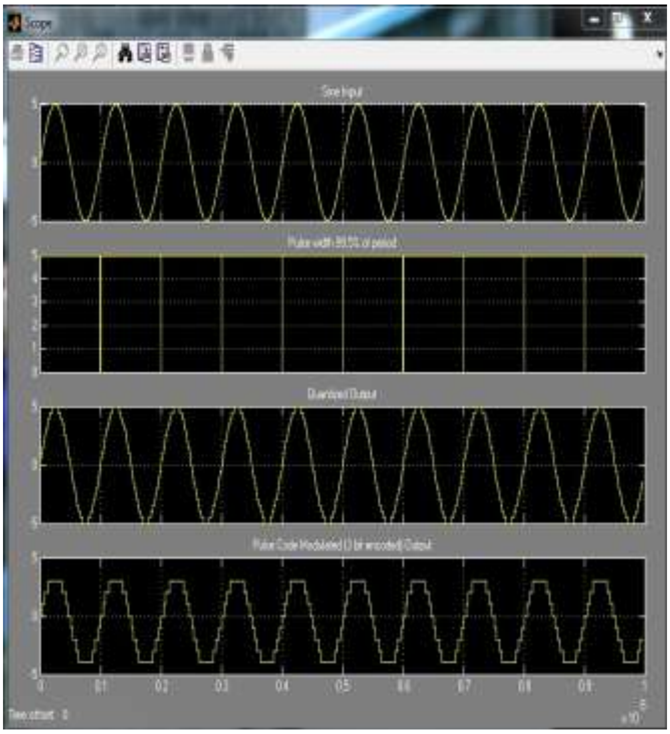

Fig 9 -PCM O/P encoded with 3 bits (99.5% pulse width, 0.5 quantization interval).

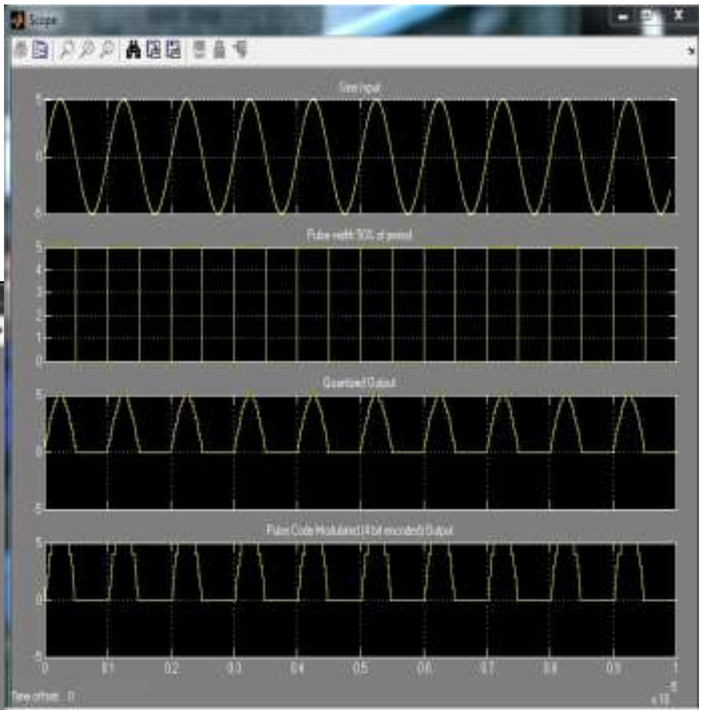

Fig 10 -PCM O/P encoded with 4 bits (50% pulse width, 0.5 quantization interval).

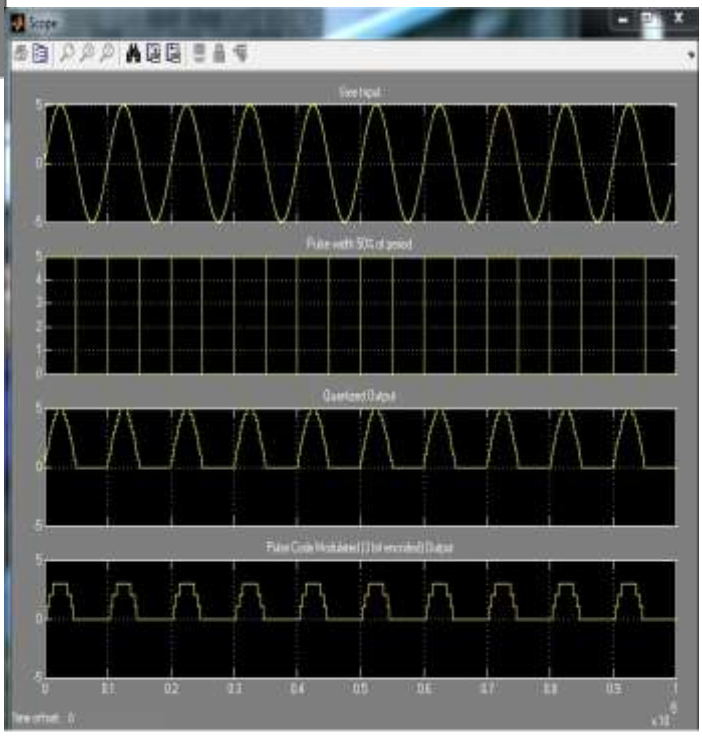

Fig 11 -PCM O/P encoded with 3 bits (50% pulse width, 0.5 quantization interval).

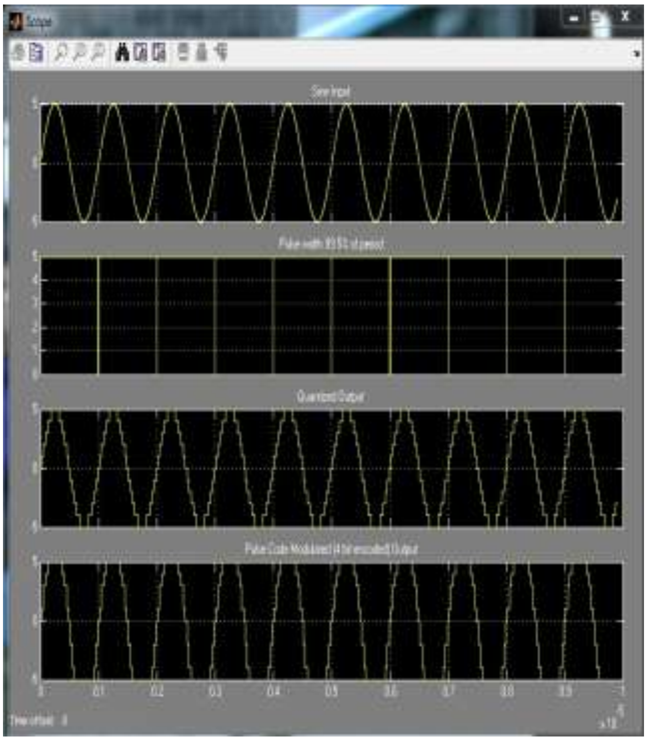

Fig 12 -PCM O/P encoded with 4 bits (99.5% pulse width, 1 quantization interval).

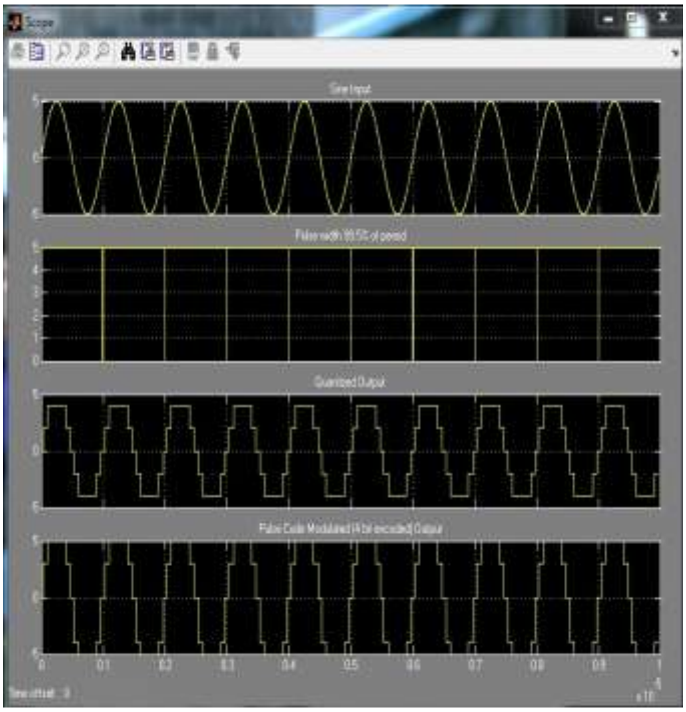

Fig 13 -PCM O/P encoded with 4 bits (99.5% pulse width, 2 quantization interval).

VII. CONCLUSION

We have successfully done performance analysis of Pulse Code Modulation and the output has been depicted in the figures above.

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