# PCM-Based Digital Audio Coding & Transmission over Serial Line

## Session 2005-2009

**Project Advisor:**
**Mr. Khalid Asghar**

## Submitted By:



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| **M. Aamir Saeed Khan** | **050620-156** |
| **Saad Hassan** | **050520-010** |

Department of Electrical & Communication Engineering
University of Management and Technology
Lahore

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By

**Muhmmad Aamir Saeed Khan**
**&**
**Saad Hassan**

# University of Management and Technology

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**Signed:**

Aamir Saeed Khan
050620-156

Saad Hassan
050520-010

**Dated:**
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## Abstract

Pulse-code modulation (PCM) is a digital representation of an analog signal where the magnitude of the signal is sampled regularly at uniform intervals, then quantized to a series of symbols in a numeric (usually binary) code. The main purpose of this project is to analyze different aspects of Pulse Code Modulation including sampling theory, quantizing and effect of non-uniform quantization. "Sampling is the first step in any pulse modulation system. In fact by sampling, a signal is represented by set of discrete samples. If the frequency of the sampling is high enough, the original signal can be recovered from the samples. From the mathematical point of view, the sampling process can be considered as the multiplication of the message signal by a train of Dirac impulses.'' The sampling rate, or number of samples per second, is several times the maximum frequency of the analog waveform in cycles per second or hertz. The instantaneous amplitude of the analog signal at each sampling is rounded off to the nearest of several specific, predetermined levels. This process is called quantization. The number of levels is always a power of 2 -- for example, 8, 16, 32, or 64. These numbers can be represented by three, four, five, or six binary digits (bits) respectively. The output of a pulse code modulator is thus a series of binary numbers, each represented by some power of 2bits.

## Project Objective:

Our Final Project consists of two part 1) is Senior Project 1 & 2) is Senior Project 2. The name of the project is PCM-Based Digital Audio Coding & Transmission over Serial Line. It has two chapters. 1) Is PCM Introduction, & 2) is PCM Communication System. 3) PCM based Digital Audio Transmission. In first part we discussed four essential parts of PCM. These are 1) Filtering, 2) Sampling 3) Quantizing & 4) Encoding. We discussed each of them individually and how it plays a key role in designing PCM. In Chapter#2 PCM Communication System, we talked about 1) Anti-Alias Filter 2) Sample And Hold 3) Analog to Digital Converter 4) Channels 5) Digital To Analog Converter 6) Reconstruction Filter. They too take part in manipulating PCM. In second part 'Senior Project' we would implement it practically.

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# Chapter 1

# PCM

# Introduction

## 1.1 Introduction:

Pulse code modulation is the heart of technology in communications in today's digital world. It's a process in which analog signals are converted to digital form. The analog signal is represented by a series of pulses and non-pulses (1 or 0 respectively). At this stage to fully understand we can refer back to the notes on signals.

## 1.2 What is PCM ?

Pulse code modulation (PCM) is a digital scheme for transmitting analog data. The signals in PCM are binary; that is, there are only two possible states, represented by logic1 (high) and logic0 (low). This is true no matter how complex the analog waveform happens to be. Using PCM, it is possible to digitize all forms of analog data, including full-motion video, voices, music, telemetry, and virtual reality (VR).

## 1.3 Why PCM ?

The stream of pulses and non-pulse streams of 1's and 0's are not easily affected by interference and noise. Even in the presence of noise, the presence or absence of a pulse can be easily determined. Since PCM is digital, a more general reason would be that digital signals are easy to process by cheap standard techniques. This makes it easier to implement complicated communication systems such as telephone networks

The practical implementation of PCM makes use of other processes. The processes are carried out in the order in which they appear below:

- **Filtering**
- **Sampling**
- **Quantizing**
- **Encoding**

**[R1]**

### 1.3.1. Filtering:

The first step to convert the signal from analog to digital is to filter out the higher frequency component of the signal. This makes things easier downstream to convert this signal. Most of the energy of spoken language is somewhere between 200 or 300 hertz and about 2700 or 2800 hertz. Roughly 3000–hertz bandwidth for standard speech and standard voice communication is established. Therefore, they do not have to have precise filters (it is very expensive). A bandwidth of 4000 hertz is made from an equipment point if view. This band-limiting filter is used to prevent aliasing (anti aliasing). This happens when the input analog voice signal is undersampled, defined by the Nyquist criterion as Fs < 2(BW). The sampling frequency is less than the highest frequency of the input analog signal. This creates an overlap between the frequency spectrum of the samples and the input analog signal. The low−pass output filter, used to reconstruct the original input signal, is not smart enough to detect this overlap. Therefore, it creates a new signal that does not originate from the source. This creation of a false signal when sampling is called aliasing. **[R2]**

The filtering stage removes frequencies above the highest signal frequency. These frequencies if not removed, may cause problems when the signal is going through the stage of sampling. Sampling of a waveform means determining instantaneous amplitudes of a signal at fixed intervals. You may have a problem in understanding what this sentence means, but if you take time to look at Figure 1, you should be able to understand **[R1]**

### 1.3.2. Sampling:

In signal processing, **sampling** is the reduction of a continuous signal to a discrete signal. A common example is the conversion of a sound wave (a continuous-time signal) to a sequence of samples (a discrete-time signal).

A **sample** refers to a value or set of values at a point in time and/or space.

A **sampler** is a subsystem or operation that extracts samples from a continuous signal. A theoretical ideal sampler produces samples equivalent to the instantaneous value of the continuous signal at the desired points. **[R3]**

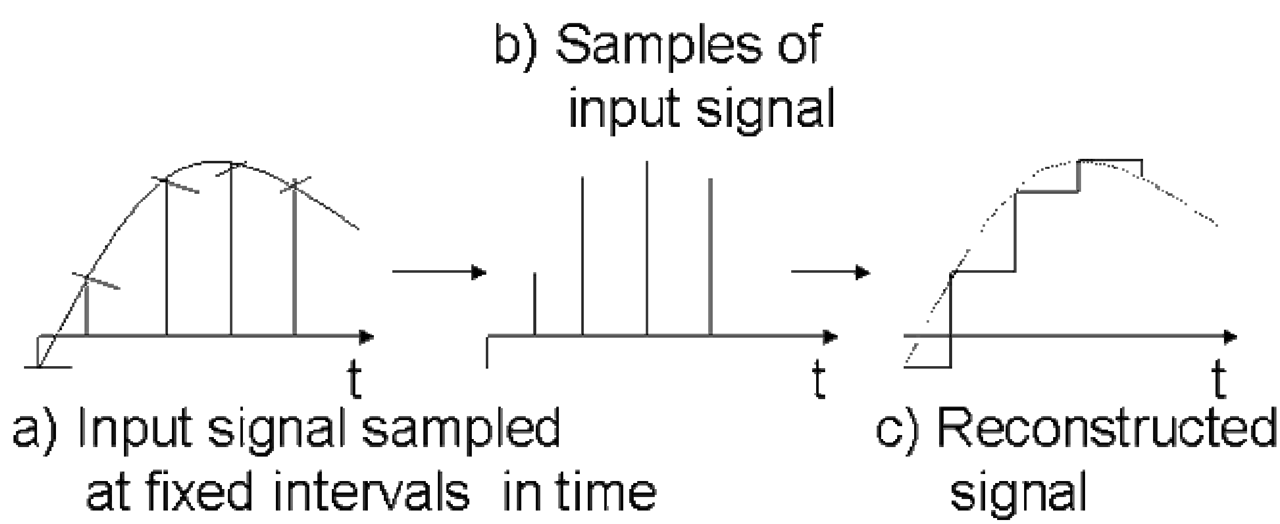


fig.1.1 a, b, c: The Action of Sampling **[R1]**

### 1.3.3. Sampling Theorem:

. **A Signal is said to be band-limited**

if  g (t), its spectrum (FT) G(ω)

.  $G(\omega) = 0$ as $|\omega| > 2\pi B$

.

**Sampling Theorem:**

The signal can be reconstructed from its samples taken uniformly at a rate

$$R > 2B.$$

That is, the minimum sampling frequency is

fs = 2B [Ts = 1/2B]

(Ts: sampling interval)

fs: Nyguist rate for g(t)

Ts: Nyguist interval for g(t)

Ts = 1/fs

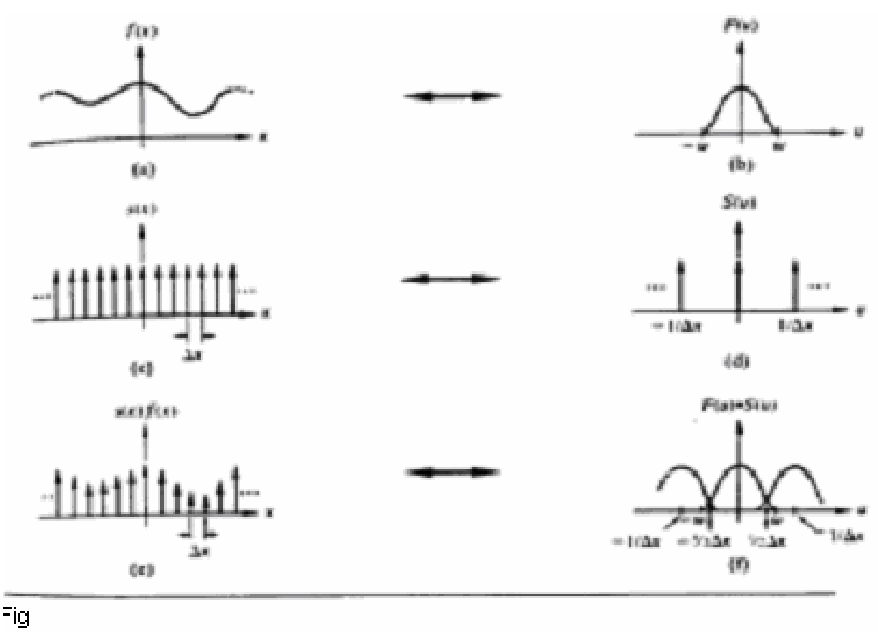

Fig 1.2 Sampling Theorems

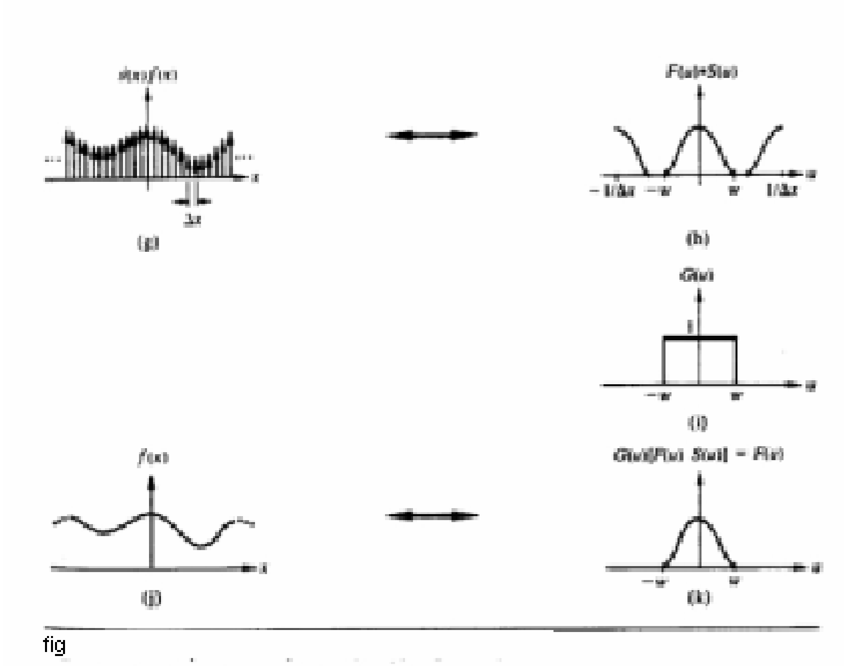


Fig 1.3 Sampling Theorems

**[R4]**

### 1.3.3.1. A−law and u−law Companding:

A-law and u-law are audio compression schemes (codec's) defined by Consultative Committee for International Telephony And Telegraphy (CCITT) G.711 which compress 16−bit linear PCM data down to eight bits of logarithmic data.

**A−law Compander:**

Limiting the linear sample values to twelve magnitude bits, the A−law compression is defined by this equation, where A is the compression parameter (A=87.7 in Europe), and x is the normalized integer to be compressed.

$$F(x) = \left[ \begin{array}{ll} \dfrac{A * |x|}{1+\ln(A)} & 0 \le |x| < \dfrac{1}{A} \\ \dfrac{\mathrm{sgn}(x) * (1+\ln(A|x|))}{1+\ln(A)} & \dfrac{1}{A} \le |x| \le 1 \end{array} \right.$$

**u−law Compander:**

Limiting the linear sample values to thirteen magnitude bits, the u−law (u−law and Mu−law are used interchangeably in this document) compression is defined by this equation, where m is the compression parameter (m =255 in the U.S. and Japan) and x is the normalized integer to be compressed.

A−law standard is primarily used by Europe and the rest of the world. u−law is used by North America and Japan.

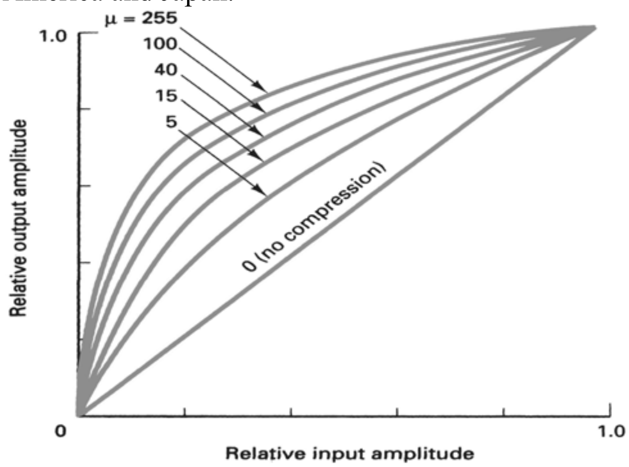


Fig 1.4 Relative Input Amplitude

### 1.3.3.2. Similarities Between A−law and u−law:

• Both are linear approximations of logarithmic input/output relationship.

Both are implemented using eight−bit code words (256 levels, one for each quantization interval).

Eight−bit code words allow for a bit rate of 64 kilobits per second (kbps). This is calculated by

multiplying the sampling rate (twice the input frequency) by the size of the code word (2 x 4 kHz x 8 bits = 64 kbps).

- ❖ Both break a dynamic range into a total of 16 segments:
- ❖ Eight positive and eight negative segments.
- ❖ Each segment is twice the length of the preceding one.
- ❖ Uniform quantization is used within each segment.
- ❖ Both use a similar approach to coding the eight−bit word:
- ❖ First (MSB) identifies polarity.
- ❖ Bits two, three, and four identify segment.
- ❖ Final four bits quantize the segment are the lower signal levels than A−law.

### 1.3.3.3. Differences Between A−law and u−law

- ❖ Different linear approximations lead to different lengths and slopes. The numerical assignment of the bit positions in the eight−bit code word to segments and the quantization levels within segments are different.
- ❖ A−law provides a greater dynamic range than u−law.
- ❖ u−law provides better signal/distortion performance for low level signals than A−law.
- ❖ A−law requires 13−bits for a uniform PCM equivalent. u−law requires 14−bits for a uniform PCM equivalent.
- ❖ An international connection needs to use A−law, u to A conversion is the responsibility of the u−lawcountry. **[R2]**

## 1.4. Quantization

If eight bits are allowed for the PCM sample, this gives a total of 256 possible values. PCM assigns these 256 possible values as 127 positive and 127 negative encoding levels, plus the zero-amplitude level. (PCM assigns two samples to the zero level.) These levels are divided up into eight bands called chords. Within each chord is sixteen steps. Figure shows the chord/step structure for a linear encoding scheme.:

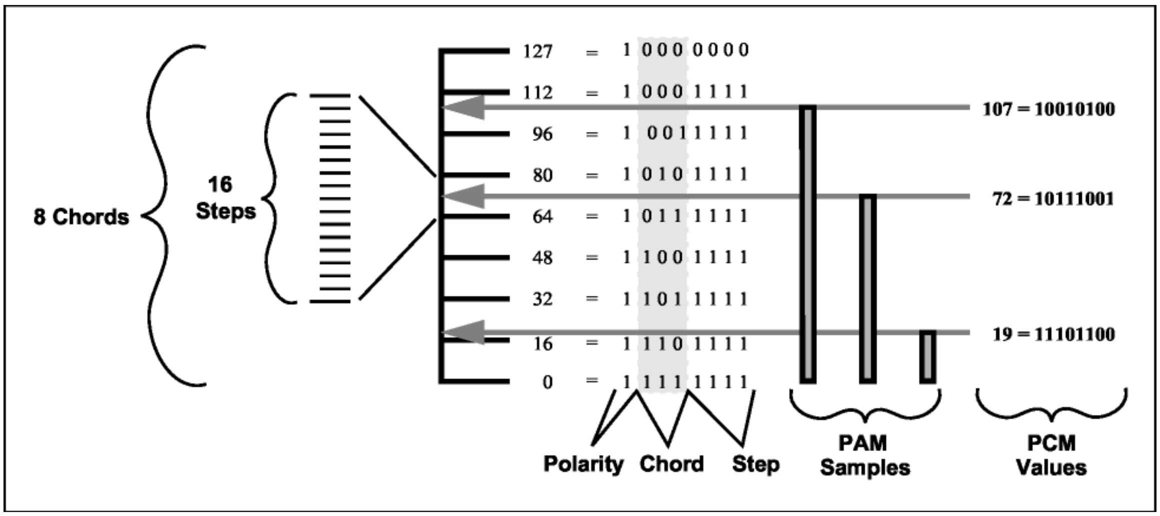


Figure 1.5 : PCM Quantization levels - Chords and Steps

Three examples of PAM samples are shown in Figure. Each PAM sample's peak falls within a specific chord and step, giving it a numerical value. This value translates into a binary code which becomes the corresponding PCM value. Figure 23 only shows the positive-value PCM values, for simplicity.

Figure 24 shows the conversion function for a linear quantization process. As a voice signal sample increases in amplitude the quantization levels increase uniformly. The 127 quantization levels are spread evenly over the voice signal's dynamic range. This gives loud voice signals the same degree of resolution (same step size) as soft voice signals. Encoding an analog signal in this manner, while conceptually simplistic, does not give optimized fidelity in the reconstruction of human voice.

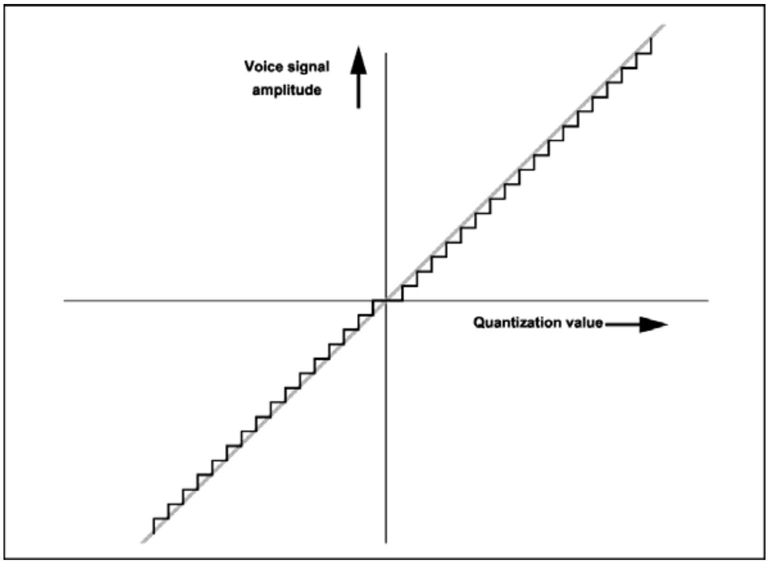


Figure 1.6 : Linear Quantization, Signal Amplitude versus Quantization Value

Quantization levels distributed according to a logarithmic, instead of linear, function gives
finer resolution, or smaller quantization steps, at lower signal amplitudes. Therefore, higher-fidelity reproduction of voice is achieved. Figure 25 shows a conversion function for a logarithmic quantization process.
A vocoder that places most of the quantization steps at lower amplitudes by using a nonlinearfunction, such as a logarithm, is said to compress voice upon encoding, then expand the PCM samples to re-create an analog voice signal. Such a vocoder is hence called a **compander** (from compress and expand).

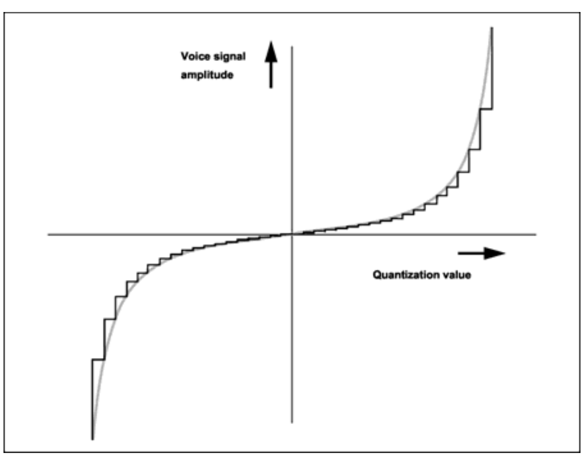


Figure 1.7: Logarithmic Quantization, Signal Amplitude versus Quantization Value **[R5]**

The output of a sampler is still continuous in amplitude. Each sample can take on any value e.g. 3.752, 0.001, etc. The number of possible values is infinite. To transmit as a digital signal we must restrict the number of possible values.

**Quantization**

is the process of "rounding off" a sample according to some rule.
E.g. suppose we must round to the nearest tenth, then:

3.752 --> 3.8     0.001 --> 0 **[R6]**

### 1.5 Encoding:

The process for encoding an analog signal is conceptually simple. The image to the right provides a simple example. A single cycle of a sine wave is depicted. If this is a one hertz signal, the period of the cycle is one second. The selected time interval is 16ths of a second, indicated by the hash marks on the horizontal axis. The range of possible amplitude has been broken into 8 ranges, indicated by the hash marks on the left vertical axis. At each interval, the amplitude is measured and related back to one of the eight intervals on the left axis. The sample is encoded using the binary version of the value of that range (indicated on the right side of the image). The resulting bit stream is depicted at the bottom of the image. In this example, the eight ranges require a 3-bit code, and 16 samples per second would result in a 48 bits per second of signal. **[R7]**

The output of the quantizer is one of *M* possible signal levels. If we want to use a binary transmission system, then we need to map each quantized sample into an ***n*** bit binary word. **[R6]**

## 1.5.1. Criteria For Signal Encoding:

- What determines how successful a receiver will be in interpreting an incoming signal?
  - Signal-to-noise ratio
  - Data rate
  - Bandwidth
- An increase in data rate increases bit error rate
- An increase in SNR decreases bit error rate
- An increase in bandwidth allows an increase in data rate **[R8]**

## 1.5.2. Basic Encoding Techniques

- Analog data to analog signal
- Amplitude modulation (AM)
- Angle modulation
- Frequency modulation (FM)
- Phase modulation (PM)

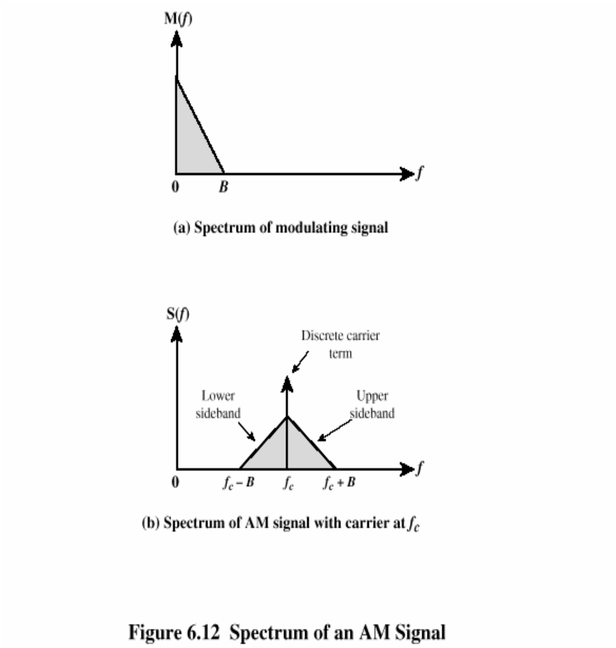


Fig 1.8 Spectrum of an AM Signal [R9]

### 1.5.3. A Basic PCM Encoder

This figure shows a simplified block diagram of a PCM encoder. A number of transducer signals are applied to the input of a multiplexer switch. Signals are sampled in any order and at any rate as defined by the user of the system. Many systems provide programmability in order to select channel sampling conditions.

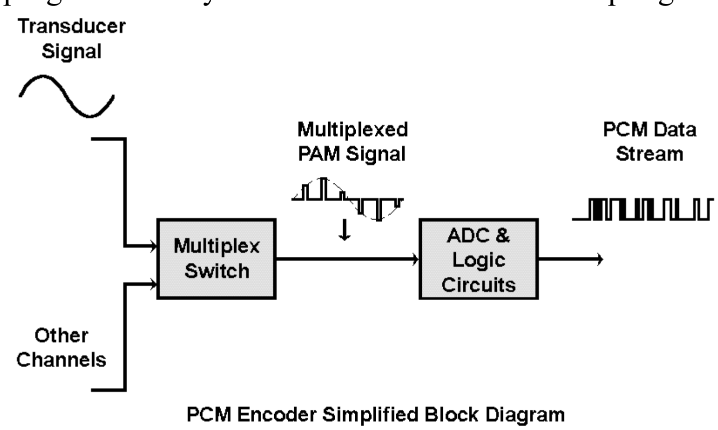


Fig 1.9 PCM Encoder Simplified Block Diagram

The output of the multiplexer switch is a PAM signal which carries time-division-multiplexed samples of each input channel. The ADC samples this PAM signal and executes an analog-to-digital conversion on each sample. The output from the ADC is serialized and formatted into a PCM wavetrain in accordance with the applicable telemetry standards. **[R10]**