Unit 3
History of data communication

Data Communications History
• 1838: Samuel Morse & Alfred Veil Invent Morse CodeTelegraph System
• 1876: Alexander Graham Bell invented Telephone
• 1910:Howard Krum developed Start/Stop Synchronisation
History of Computing
• 1930: Development of ASCII Transmission Code
• 1945: Allied Governments develop the First Large Computer
• 1950: IBM releases its first computer IBM 710
• 1960: IBM releases the First Commercial Computer IBM 360
Main Contributors of Data Comm.
• Transmission Technology
• Packet Switching Technology
• Internet – 1967: ARPANET by Advanced Research Project Agency (ARPA) of U.S.
– 1975: TCP/IP protocol
• LAN Technology
– DIX-Ethernet & IEEE 802 Networks
• WAN
– 1976: ISO releases HDLC & CCITT releases X.25 (PSPDN)

Data communication codes.
ASCII code
ASCII is defined in ANSI X3.4
o Corresponding CCITT recommendation is IA5 (International Alphabet No.5)
o – ISO specification is ISO 646
Total 128 codes
o – 96 codes are graphic symbols (in Col. 2~7).
o 94 codes are printable
o And 2 codes viz. SPACE & DEL characters arenon printable
o 32 codes control symbols (Col. 0 & 1)
All are non printable
• EBCDIC code
o It is an 8-bit code with 256symbols
o No parity bit for error checking
o The graphic symbols are almostsame as ASCII
o Several differences in Controlcharacters as compared to ASCII
• BAUDOT code
o It is a 5-bit code also known asITA2 (International TelegraphAlphabet No.
2).
o 32 codes are possible. With thehelp of Letter shift & Figure shift key same code is used torepresent two symbols.
o Maximum symbols in this code are58
o Used in Telegraphy/Telex

Concepts and Terminology

- **Transmission Terminology**

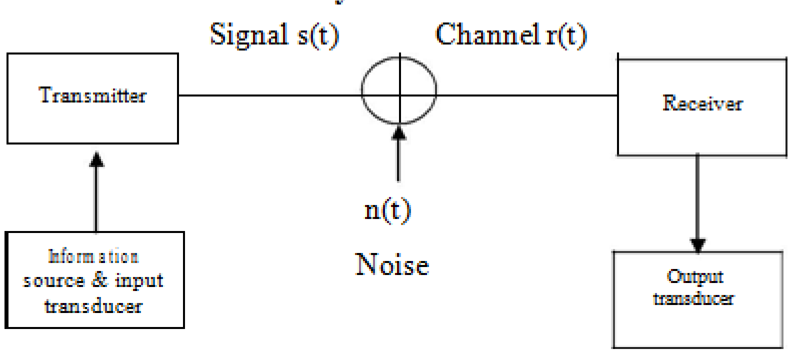
Transmission from transmitter to receiver goes over some transmission medium using electromagnetic waves.

- **Guided Media:** waves are guided along a physical path; twisted pair, optical fibre, coaxial cable.
- **Unguided Media:** waves are not guided; air waves radio waves.
- **Direct Link:** signal goes from transmitter to receiver without intermediate devices, other than amplifiers and repeaters.
- **Point-to Point Link:** guided media with direct link between two devices.
- **Multipoint Guided Configuration:** more than two devices can share the same medium.

Data Communications

Data communications is the transfer of information that is in digital form, before it enters the communication system.

- **Basic Elements of a Communication System**



Information: generated by the source may be in the form of voice, a picture or a plain text. An essential feature of any source that generates information is that its output is described in probabilistic terms; that is, the output is not deterministic.

A transducer is usually required to convert the output of a source in an electrical signal that is suitable for transmission.

- **Transmitter:** a transmitter converts the electrical signal into a form that is suitable for transmission through the physical channel or transmission medium. In general, a transmitter performs the matching of the message signal to the channel by a process called modulation.

  The choice of the type of modulation is based on several factors, such as:
  - the amount of bandwidth allocated,
  - the type of noise and interference that the signal encounters in transmission over the channel,
  - and the electronic devices that are available for signal amplification prior to

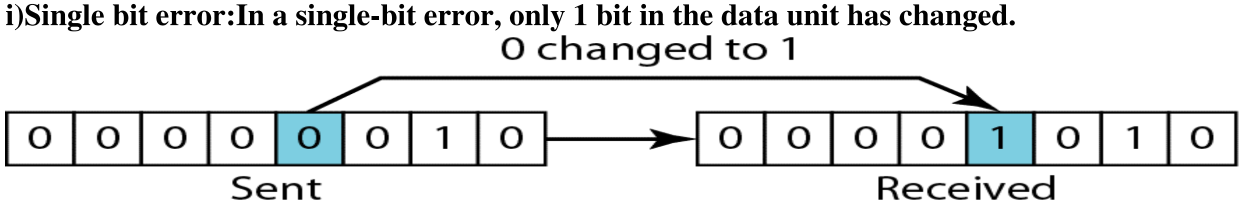
transmission.

- **Channel:** the communication channel is the physical medium that connects the transmitter to the receiver. The physical channel may be a pair of wires that carry the electrical signals, or an optical fibre that carries the information on a modulated light beam or free space at which the information-bearing signal are electromagnetic waves.
- **Receiver:** the function of a receiver is to recover the message signal contained in the received signal. The main operations performed by a receiver are demodulation, filtering and decoding.

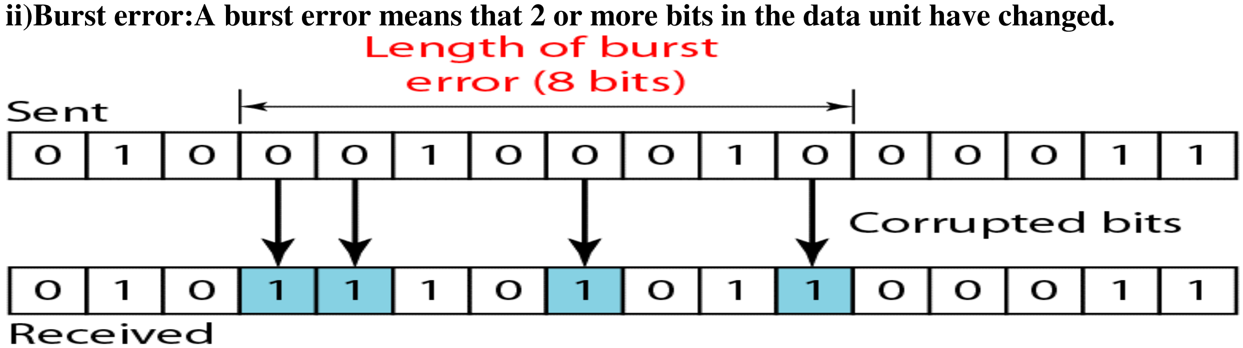
Error Detection and Correction

No communication channel or storage device is completely error-free
As the number of bits per area or the transmission rate increases, more errors occur.
Impossible to detect or correct 100% of the errors

Types of error:
i)Single bit error:In a single-bit error, only 1 bit in the data unit has changed.



ii)Burst error:A burst error means that 2 or more bits in the data unit have changed.



Error detection code:
*Parity coding:
A common type of error-detecting code is called a **parity check**. For example, consider the message 1101. We add a 0 or 1 to the end of this message so that the resulting message has an even number of 1's. We would thus encode 1101 as 11011. If the original message were 1001, we would encode that as 10010, since the original message already had an even number of 1's. Now consider receiving the message 10101. Since the number of 1's in this message is odd, we know that an error has been made in transmission. However, we do not know how many errors happened in transmission or which digit(s) were e_ected. Thus a parity check scheme detects errors, but does not locate them for correction.

***Cyclic Redundancy Check (CRC)**

- Mostly used in data communication

- Tells us whether an error has occurred, but does not correct the error.
- This is a type of "systematic error detection"
  - The error-checking bits are appended to the information byte

Calculating and Using CRCs

1. Let the information byte F = 1001011
2. The sender and receiver agree on an arbitrary binary pattern P. Let P = 1011.
3. Shift F to the left by 1 less than the number of bits in P. Now, F = 1001011000.
4. Let F be the dividend and P be the divisor. Perform "modulo 2 division".
5. After performing the division, we ignore the quotient. We got 100 for the remainder, which becomes the actual CRC checksum.
   Division

```
                        ____1010        ← Quotient
   Ex:     1011 | 1001011
                          1011
                          001001
                            1011
                            00101       ← Remainder
```

6. Add the remainder to F, giving the message M:
   1001011 + 100 = 1001011100 = M
7. M is decoded and checked by the message receiver using the reverse process.

```
              ____1010100
   1011 | 1001011100
                1011
                001001
                  1001
                  0010
                  001011
                    1011
                    0000        ← Remainder
```

Error correction code:
Hamming Codes

1. **One of the most effective codes for error-recovery**
2. **Used in situations where random errors are likely to occur**
3. **Error detection and correction increases in proportion to the number of parity bits (error-checking bits) added to the end of the information bits**
   code word = information bits + parity bits
   Hamming distance: the number of bit positions in which two code words differ.

```
            10001001
            10110001
              * * *
```

Minimum Hamming distance or D(min) : determines its error detecting and correcting capability.

3.1 Pulse Modulation

The continuous time signal $x(t)$ to be transmitted is sampled at frequency f_s sufficiently above the highest frequen cy present in $x(t)$. The amplitude of the modulating signal $x(t)$ modulates some parameter of the pulse train. These parameters are amplitude, duration (width) and position. Fig. 3.1.1 shows different types of analog pulse modulation techniques with message waveform $x(t)$.

For PAM the modulated pulse parameter is amplitude, for PDM it is width and for PPM it is relative position. These parameters vary in direct proportion to amplitude of $x(t)$ at the sampling instant. As shown in waveforms of Fig. 3.1.1.

$$f_s = \frac{1}{T_s} = \text{Sampling frequency}$$

$$A_0 = \text{Amplitude of the pulse}$$

and

$$\tau_0 = \text{Width of the pulse}$$

Since the waveforms are unipolar, they have some dc value. Also the shape of the pulse should be preserved (rising and falling edges, amplitude, duration etc.). Thus the transmission bandwidth needed for these pulse transmission is quite high compared to the message signal bandwidth. Therefore normally single channel PAM, PPM or PDM are seldom used. Always time division multiplexing (TDM) is used to utilize the channel bandwidth.

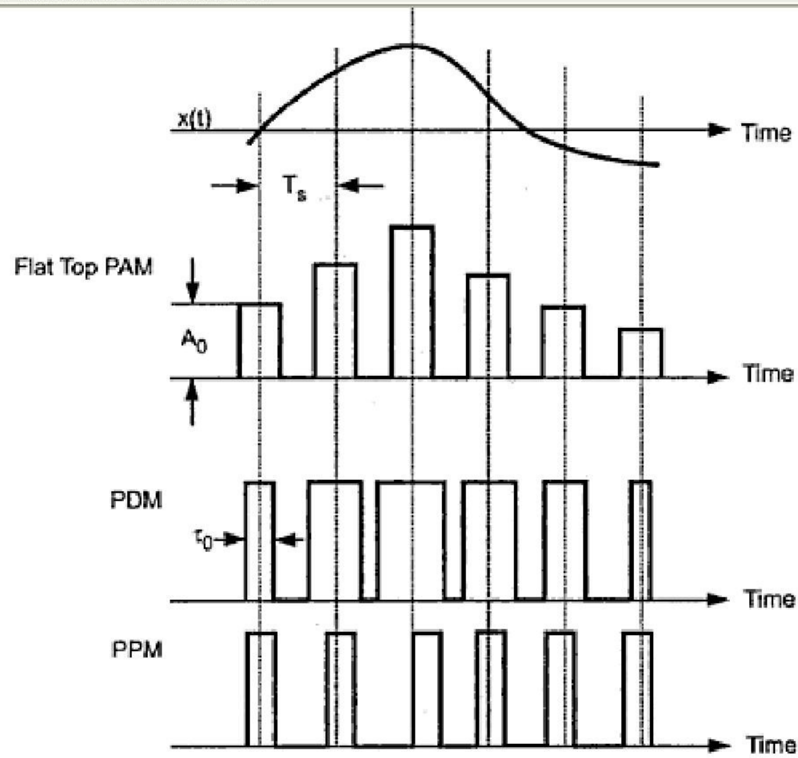


Fig. 3.1.1 Different types of analog pulse modulation techniques

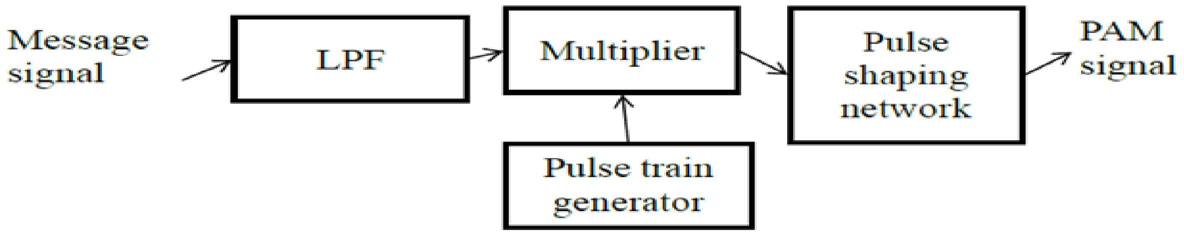
Pulse Amplitude Modulation (PAM)

Amplitude of pulse is proportional to the amplitude of the modulating signal.The width and position of the pulse remains unchanged.

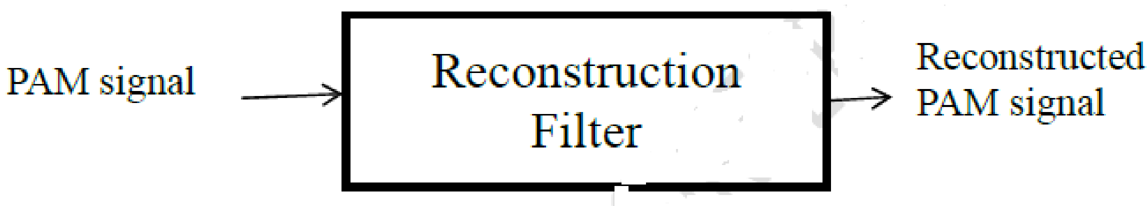
PAM Modulator

- Message signal is transmitted to LPF
- LPF performs bandlimiting

• Band limited signal is then sampled at the multiplier.
• Multiplier samples with the help of pulse train generator
• Pulse train generator produces the pulse train
• The multiplication of message signal and pulse train produces PAM signal



PAM demodulator



Pulse Width Modulation

Width of pulse is proportional to the amplitude of the modulating signal.The amplitude and position of the pulse remains unchanged.

PWM Modulator

• It is basically a monostablemultivibrator with message signal applied at the control voltage input.
• Externally applied modulating signal changes the control voltage and hence the threshold voltage level
• The time period required to charge the capacitor upto threshold level changes giving pulse modulated signal

PPM Modulator

• Sawtooth generator generates sawtooth signal of frequency which is applied to inverting input of comparator
• Modulating signal is applied to the non-inverting input of comparator
• When the value of message signal is higher than value of sawtooth ,then the output is high
• When the value of message signal is lower than value of sawtooth ,then the output is high

3.2 Pulse Code Modulation (PCM)

3.2.1 PCM Generator

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The pulse code modulator technique samples the input signal $x(t)$ at frequency $f_s \geq 2W$. This sampled 'Variable amplitude' pulse is then digitized by the analog to digital converter. The parallel bits obtained are converted to a serial bit stream. Fig. 3.2.1 shows the PCM generator.

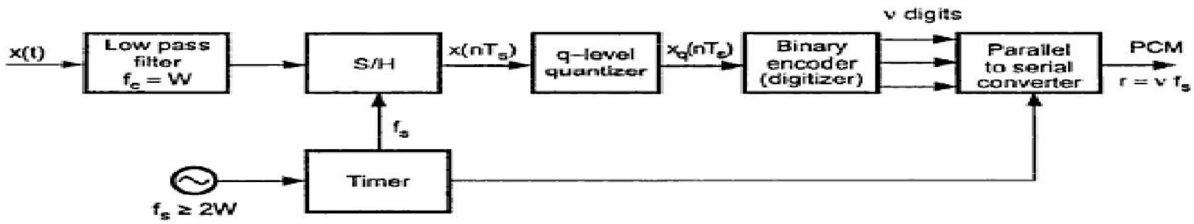


Fig. 3.2.1 PCM generator

In the PCM generator of above figure, the signal $x(t)$ is first passed through the low-pass filter of cutoff frequency 'W' Hz. This low-pass filter blocks all the frequency components above 'W' Hz. Thus $x(t)$ is bandlimited to 'W' Hz. The sample and hold circuit then samples this signal at the rate of f_s. Sampling frequency f_s is selected sufficiently above Nyquist rate to avoid aliasing i.e.,

$$f_s \geq 2W$$

In Fig. 3.2.1 output of sample and hold is called $x(nT_s)$. This $x(nT_s)$ is discrete in time and continuous in amplitude. A q-level quantizer compares input $x(nT_s)$ with its fixed digital levels. It then assigns any one of the digital level to $x(nT_s)$ which results in minimum distortion or error. This error is called *quantization error*. Thus output of quantizer is a digital level called $x_q(nT_s)$.

Quantization error is given as,

$$\varepsilon = x_q(nT_s) - x(nT_s) \qquad \text{... (3.2.1)}$$

3.2.2 Transmission Bandwidth in PCM

Let the quantizer use 'v' number of binary digits to represent each level. Then the number of levels that can be represented by 'v' digits will be,

$$q = 2^v \qquad \text{... (3.2.2)}$$

Here 'q' represents total number of digital levels of q-level quantizer.

For example if $v = 3$ bits, then total number of levels will be,

$$q = 2^3 = 8 \text{ levels}$$

Each sample is converted to 'v' binary bits. i.e. Number of bits per sample = v

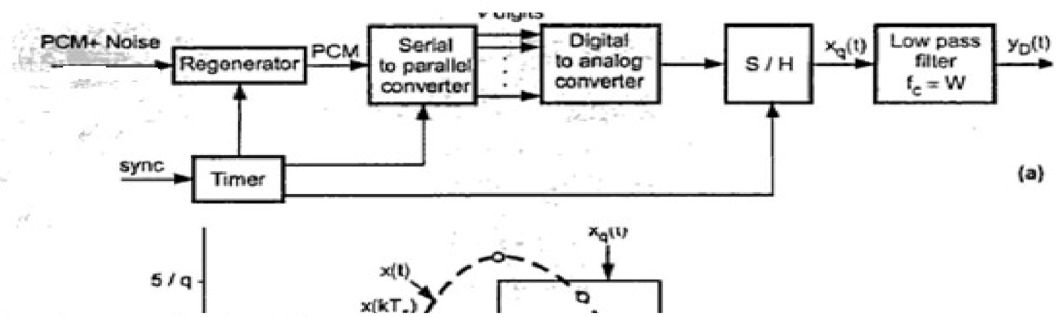
We know that, Number of samples per second = f_s

∴ Number of bits per second is given by,

(Number of bits per second) = (Number of bits per samples) × (Number of samples per second)

= v bits per sample × f_s samples per second

Fig. 3.2.2 (a) shows the block diagram of PCM receiver and Fig. 3.2.2 (b) shows the reconstructed signal. The regenerator at the start of PCM receiver reshapes the pulses and removes the noise. This signal is then converted to parallel digital words for each sample.



1.8.6.3 Companding in PCM

Normally we don't know how the signal level will vary in advance. Therefore the nonuniform quantization (variable step size 'δ') becomes difficult to implement. Therefore the signal is amplified at low signal levels and attenuated at high signal levels. After this process, uniform quantization is used. This is equivalent to more step size at low signal levels and small step size at high signal levels. At the receiver a reverse process is done. That is signal is attenuated at low signal levels and amplified at high signal levels to get original signal. Thus the compression of signal at transmitter and expansion at receiver is called combinely as *companding*. Fig. 1.8.9 shows compression and expansion curves.

As can be seen from Fig. 1.8.9, at the receiver, the signal is expanded exactly opposite to compression curve at transmitter to get original signal. A dotted line in the Fig. 1.8.9 shows uniform quantization. The compression and expansion is obtained by passing the signal through the amplifier having nonlinear transfer characteristic as

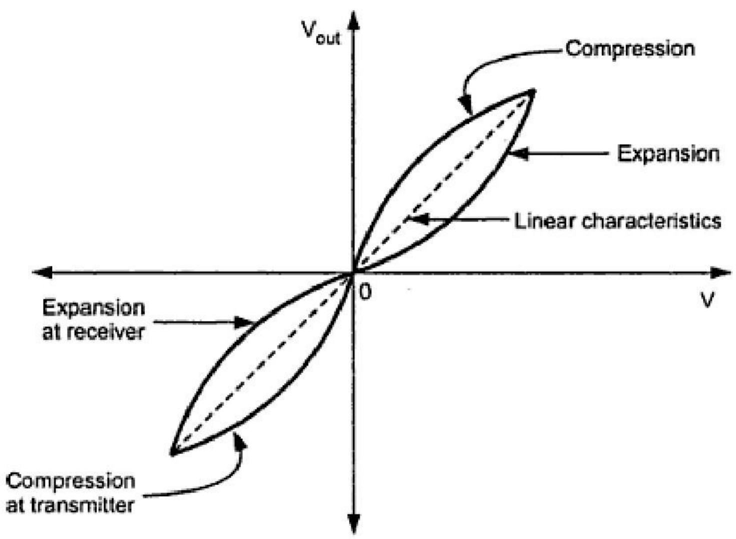


Fig. 1.8.9 Companding curves for PCM

1.8.6.4 μ - Law Companding for Speech Signals

Normally for speech and music signals a μ - law compression is used. This compression is defined by the following equation,

$$Z(x) = (\text{Sgn } x) \frac{ln(1+\mu|x|)}{ln(1+\mu)} \quad |x| \leq 1 \qquad \ldots (1.8.52)$$