- Next the segments from TCP or user datagrams from UDP are accepted and multiplexed by IP at the network layer.
- IP can also multiplex the packets from some other protocols such as ICMP or IGMP etc.
- The frames at the data link layer level can carry the payload coming from the network layer protocols such as IP or ARP etc.

Review Questions

Q. 1 State the names of two network models.
Q. 2 Define the word protocol.
Q. 3 What is protocol layering ?
Q. 4 Explain the concept of logical connections.
Q. 5 Draw the layers of TCP/IP suite.
Q. 6 Explain the layered architecture of TCP/IP suite.
Q. 7 Explain in detail the physical layer in TCP/IP suite.
Q. 8 Explain in detail the data link layer in TCP/IP suite.
Q. 9 Explain in detail the network layer in TCP/IP suite.
Q. 10 Explain in detail the transport layer in TCP/IP suite.
Q. 11 Explain in detail the application layer in TCP/IP suite.
Q. 12 Name any three network layer protocols.
Q. 13 Write a short note on : IP.
Q. 14 State various functions of network layer.
Q. 15 State the two most important transport layer protocols.
Q. 16 State various duties of transport layer.
Q. 17 State any four application layer protocols.
Q. 18 Explain the concept of encapsulation in TCP/IP.
Q. 19 Explain the concept of decapsulation in TCP/IP.
Q. 20 Write a note on following in TCP/IP suite :
1. Addressing.
2. Multiplexing and demultiplexing.

□□□

CHAPTER 3
Unit I

Data and Signals

Syllabus :

Data and signals, Analog and digital data, Analog and digital signals, Sine wave phase, Wavelength, Time and frequency domains, Composite signals, Bandwidth, Digital signal, Bit rate, Bit length, Transmission of digital signals, Transmission impairments, Attenuation, Distortion, Noise, Data rate limits, Performance, Bandwidth, Throughput, Latency (Delay).

3.1 Analog and Digital Signals :

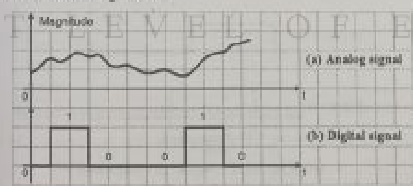
- Signals can be of two types :
  1. Analog signals. 2. Digital signals.

1. Analog signal :

It is the signal in which the signal magnitude varies in a smooth fashion without any break with respect to time, as shown in Fig. 3.1.1(a).

2. Digital signal :

It is the signal in which the signal magnitudes has a constant level for some period of time, then it changes suddenly to another constant level as shown in Fig. 3.1.1(b). The examples of digital signal are binary signal, hexadecimal signal etc.



(L-24) **Fig. 3.1.1 : Types of signals**

3.1.1 Analog and Digital Data :

- Data are the entities which convey meaning, or information such as temperature, pressure etc. Signals are electric or electromagnetic representation of data. Thus signal is the representation of data.
- Data can be of two types :
  1. Analog data 2. Digital data.

1. Analog data :

Analog data is the type of data that varies continuously (smoothly) with respect to time. Voice and video are the best examples of analog data. The other examples are temperature, pressure etc.

2. Digital data :

Digital data is the type of data that can take on discrete values i.e. it is discrete in nature. The examples of digital data are text and integers.

3.1.2 Sources of Digital Signal :

- The digital signals can be obtained directly from the computers. All the data used by the computers is digital.
- We can also use an A to D converter (Analog to digital converter) so as to convert analog signals into digital signals.

3.1.3 Advantages of Digital Signals :

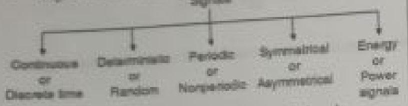
1. Digital signals can be processed and transmitted more efficiently and reliably than analog signals.
2. It is possible to store the digital data.
3. Play back or further processing of the digital data is possible.
4. The effect of "noise" (unwanted voltage fluctuations) is less. So digital data does not get corrupt.
5. It is possible to separate signal and noise and use repeaters between the transmitter and receiver.
6. Use of microprocessor and digital systems is possible.

3.1.4 Comparison of Digital and Analog Signals :

Sr. No.	Parameter	Analog signals	Digital signals
1.	Number of values	Infinite	Finite (2, 8, 16 etc.)
2.	Nature	Continuous	Discrete
3.	Sources	Signal generators, transducers etc.	Computers, A to D converters
4.	Examples	Sinewave, triangular wave	Binary signal

3.1.5 Classification of Signals :

- Fig. 3.1.2 shows the classification of signals.



(G-1250) **Fig. 3.1.2 : Classification of signals**

- Out of these we will concentrate only on periodic or non periodic signals.

3.1.6 Periodic and Non-periodic Signals :

Periodic signal :

- A signal which repeats itself after a fixed time period is called as a periodic signal. The periodicity of a signal can be defined mathematically as follows :

$$x(t) = x(t + T_o) \text{ : Condition of periodicity} \quad ...(3.1.1)$$

Where T_o is called as the period of signal x (t), in other words, signal x (t) repeats itself after a period of T_o sec.

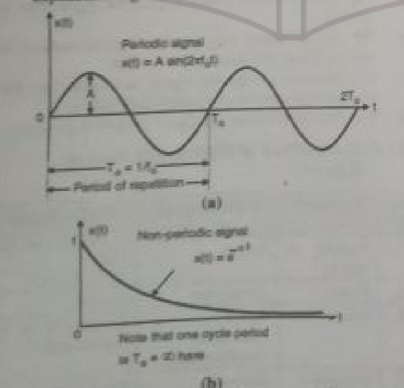
- Examples of periodic signals are sine wave, cosine wave, square wave etc. Fig. 3.1.3(a) shows a sine wave which is periodic because it repeats itself after a period T_o.

Non-periodic signal :

- A signal which does not repeat itself after a fixed time period or does not repeat at all is called as a non-periodic or aperiodic signal.
- The non-periodic signals do not satisfy the condition of periodicity stated in Equation (3.1.1).

∴ For a non-periodic signal $x(t) \neq x(t + T_o)$ ...(3.1.2)

- Sometimes it is said that an aperiodic signal has a period $T_o = \infty$. Fig. 3.1.3(b) shows a decaying exponential signal.

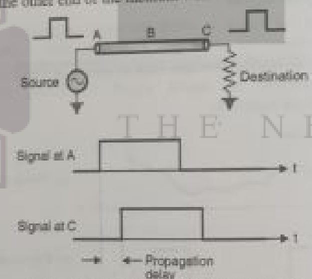


(G-1251) **Fig. 3.1.3 : Periodic and non-periodic signals**

- This exponential signal is non-periodic but it is deterministic because we can mathematically express it as $x(t) = e^{-at}$.

3.1.7 Signal Propagation :

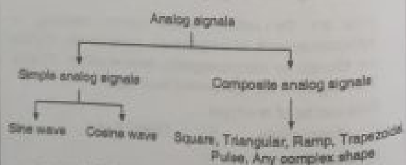
- Refer Fig. 3.1.4 which shows a signal source, a communication channel and the destination or signal receiver.
- The signal containing the data information is in the electric form. It is applied at point A of the conducting medium.
- The electrons in the conducting medium will transfer the charge to the adjacent electrons and the signal at point A gets transferred to B and then to C which is the receiving point.
- The shape of the signal at the receiver (point C) is almost same as that at the source (point A), but the signal reaches point C after a finite delay called **propagation delay.**
- The signal producing source in Fig. 3.1.4 can be a person talking on the phone or a computer producing a data signal or a video camera producing a video signal etc.
- Thus if we apply a signal at one end of the conducting medium then eventually this signal gets propagated to the other end of the medium with some time delay.



(G-1252) **Fig. 3.1.4 : Signal propagation**

3.2 Analog Signals :

We can classify the analog signals as follows :



(L-770) **Fig. 3.2.1 : Classification of analog signals**

3.2.1 Simple Analog Signal :

- It is the analog signal which cannot be decomposed into simpler signals. So this is the most basic analog signal which can be used as basic building block to build other composite signals.
- Examples of simple analog signal are sine and cosine waves.

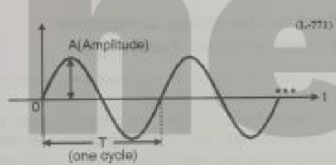
3.2.2 Composite Analog Signal :

- A composed analog signal is made of multiple sine or cosine waves of different amplitudes added to / subtracted from each other.
- Examples of composite analog signals are square wave, triangular wave, ramp, trapezoidal signal, pulse etc.

3.2.3 Sinewaves :

- It is the most basic type of periodic analog signal. That means it is the simple analog signal.
- Table 3.2.1 shows the graphical representation along with some of its characteristics.

Table 3.2.1 : Sinewave and its characteristics

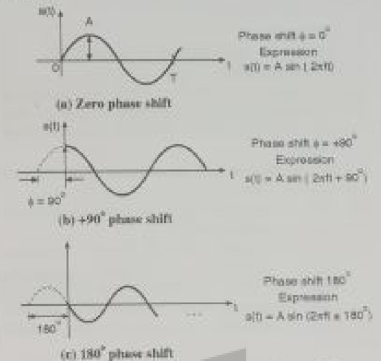


Mathematical expression	: $s(t) = A \sin(2\pi ft + \phi)$
Peak amplitude	: A
Frequency	: f Hz = 1 / T
Period	: T = (1/f) sec.
Phase	: ϕ radians

Important points about frequency and time period :

1. Frequency and time period (T) are reciprocals of each other.
2. Frequency can be defined as rate of change of signal magnitude with respect to time. Change of signal magnitude in a shorter time span corresponds to high frequency.
3. Change of signal magnitude over a long time span corresponds to low frequency.
4. The signal which does not change at all has zero frequency. Example of such a signal is dc signal.
5. The signal that changes instantaneously has an infinite frequency. Example is delta signal or an impulse.

Phase :



(L-772) **Fig. 3.2.2 : Concept of phase shift**

- The letter ϕ in the mathematical expression for a sinewave denotes phase of the sinewave.
- It describes the position of the waveform with respect to time zero (i.e. t = 0). This is explained in Fig. 3.2.2.

Wavelength :

- Wavelength is another important characteristics of a signal travelling through a transmission medium.
- Wavelength relates the frequency of a sinewave with the speed of propagation as follows :

$$\text{Wavelength } \lambda = \frac{\text{Propagation speed}}{\text{Frequency}}$$

$$= \text{Propagation speed} \times \text{Period}$$

$$\therefore \lambda = \frac{C}{f} = C \times T \quad ...(3.2.1)$$

- Wavelength of a signal is dependent on the medium. In data communication the wavelength is used to describe the transmission of light in an optical fiber.
- Wavelength is the distance travelled by a signal or an electromagnetic wave during the time period of one cycle of the signal.
- In vaccum the speed of light (c) is 3×10^8 m/s but it does not remain same in a cable. Hence the wavelength of the same signal will be different in air and on a cable.

3.3 Time and Frequency Domain :

It is possible to display a signal in either time domain or in frequency domain.

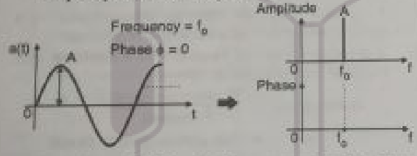
3.3.1 Time Domain Display of Analog Signals :

- It is the graph of signal magnitude with respect to time. That means on the x-axis we plot the time t and on the y-axis the magnitude is plotted. (Refer Fig. 3.3.1(a)).

- A time domain display tell us about the following :
  1. Shape of the signal
  2. Its frequency
  3. Type of the signal (periodic or nonperiodic).
  4. One cycle period.
- But we cannot know anything about what frequency components are present and in what proportion they have been mixed in order to obtain the particular shape of the signal.
- All this information can be obtained from the line spectrum of a signal.
- The phase and frequency cannot be explicitly measured from the time domain plot.
- In case of a composite waveform, which is composed of multiple sinewaves of different amplitudes, frequencies and phases, the time domain plot does not reveal any information about the relationship between amplitude, frequencies and phases of the various sinewaves.

3.3.2 Frequency Domain Display of a Signal :

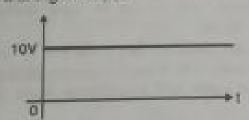
- It is the graph of signal amplitude (peak or sometimes rms), plotted on Y-axis with respect to frequency plotted on the x-axis.
- It is also possible to plot phase ϕ with respect to frequency as shown in Fig. 3.3.1.



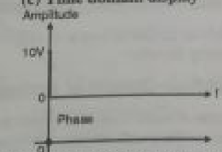
(a) Time domain display (b) Frequency domain display

(L-773) **Fig. 3.3.1**

- Now refer Fig. 3.3.1(c) which shows a dc signal with amplitude 10 V. The frequency and phase both will be zero. The frequency domain display of this signal is shown in Fig. 3.3.1(d).



(c) Time domain display



(d) Frequency domain display

(L-774) **Fig. 3.3.1 : DC signal**

Conclusions :

1. In the frequency domain we can show two characteristics of the signal using one vertical line. First is the amplitude and second is the frequency of the signal.
2. We can also show the phase angle.
3. The analog signal is best represented in the frequency domain.

3.4 Composite Signals :

- Sine and cosine waves are the basic and purest signals but they are not useful in data communication.
- To use the sinewave for data communication, we have to change one or more of its characteristics, such as amplitude in accordance with the data to be transmitted.
- But then the sinewave does not remain the simple sinewave, instead it becomes a **composite signal** which is made up of many simple sinewaves.

  Thus when we change amplitude, frequency or phase of a single frequency signal, it gets transformed into a **composite signal** made of many frequencies.

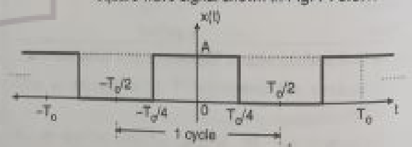
3.5 Tool for Analyzing Composite Signals (Fourier Analysis) :

- Fourier series is one of the well known tools used for the analysis of composite signals.
- According to fourier analysis, any composite signal can be represented as sum of simple sinewaves with different frequencies, amplitudes on phases as follows :

  $s(t) = a_0 + a_1 \sin(2\pi f_1 t + \phi_1) + a_2 \sin(2\pi f_2 t + \phi_2) + \dots$

  where $a_0, a_1, a_2 \dots$ are called as the Fourier coefficients.

Ex. 3.5.1 : Obtain the quadrature fourier series for the square wave signal shown in Fig. P. 3.5.1.



(G-819) **Fig. P. 3.5.1**

Soln. :

- Let us obtain the quadrature fourier series for the given square wave. The quadrature fourier series is given by the following expression :

$$x(t) = a_0 + \sum_{n=1}^{\infty} a_n \cos(n\omega_0 t) + \sum_{n=1}^{\infty} b_n \sin(n\omega_0 t) \quad \dots(1)$$

- Substituting $\omega_0 = 2\pi f_0 = \frac{2\pi}{T_0}$ into Equation (1) we get,

$$\therefore \quad x(t) = a_0 + \sum_{n=1}^{\infty} a_n \cos\left[\frac{2\pi nt}{T_0}\right] + \sum_{n=1}^{\infty} b_n \sin\left[\frac{2\pi nt}{T_0}\right] \quad \dots(2)$$

- To find the fourier coefficients, we must consider one complete cycle of x (t) for integration.
- Here we will consider one cycle from $t = -\frac{T_0}{2}$ to $t = \frac{T_0}{2}$. Let us obtain the fourier coefficients now.

1. **To obtain the value of a_0 :**

$$a_0 = \frac{1}{T_0} \int_{-T_0/2}^{T_0/2} x(t)\, dt = \frac{1}{T_0} \int_{-T_0/4}^{T_0/4} x(t)\, dt$$

.....As x (t) exists from $-T_0/4$ to $T_0/4$ only

$$= \frac{1}{T_0} \int_{-T_0/4}^{T_0/4} A\, dt \quad \text{.....As } x(t) = A \text{ from } -T_0/4 \text{ to } T_0/4$$

$$\therefore \quad a_0 = \frac{A}{T_0} [t]_{-T_0/4}^{T_0/4} = \frac{A}{T_0} \times \frac{T_0}{2} = \frac{A}{2} \quad \dots(3)$$

2. **To obtain the value of a_n :** a_n is given by,

$$a_n = \frac{2}{T_0} \int_{-T_0/2}^{T_0/2} x(t) \cdot \cos\left[\frac{2\pi nt}{T_0}\right] dt$$

$$= \frac{2}{T_0} \int_{-T_0/4}^{T_0/4} A \cos\left[\frac{2\pi nt}{T_0}\right] dt$$

$$= \frac{2A}{T_0} \cdot \frac{1}{\frac{2\pi n}{T_0}} = \left[\sin\left(\frac{2\pi nt}{T_0}\right)\right]_{-T_0/4}^{T_0/4}$$

$$= \frac{A}{\pi n}\left[\sin\left(\frac{2\pi n\, T_0/4}{T_0}\right) - \sin\left(\frac{-2\pi n\, T_0/4}{T_0}\right)\right]$$

$$= \frac{A}{\pi n}\left[\sin\left(\frac{n\pi}{2}\right) + \sin\left(\frac{n\pi}{2}\right)\right] = \frac{2A}{\sin} \sin\left[\frac{\pi nt}{T_0}\right]$$

$$\therefore \quad a_n = \frac{2A}{n\pi} \sin\left(\frac{n\pi}{2}\right) \quad \dots(4)$$

3. **To obtain the value of b_n :** b_n is given by,

$$b_n = \frac{2}{T_0} \int_{-T_0/2}^{T_0/2} x(t) \sin\left[\frac{2\pi nt}{T_0}\right] dt$$

$$= \frac{2}{T_0} \int_{-T_0/4}^{T_0/4} A \sin\left[\frac{2\pi nt}{T_0}\right] dt$$

$$= \frac{-2A}{T_0} \cdot \frac{1}{\frac{2\pi n}{T_0}} = \left[\cos\left(\frac{2\pi nt}{T_0}\right)\right]_{-T_0/4}^{T_0/4}$$

$$= \frac{-A}{\pi n}\left[\cos\left(\frac{2\pi n\, T_0/4}{T_0}\right) - \cos\left(\frac{2\pi n\, T_0/4}{T_0}\right)\right]$$

$$\therefore \quad b_n = \frac{-A}{\pi n}[\cos(n\pi/2) - \cos(n\pi/2)]$$

$$\therefore \quad b_n = 0 \quad \dots(5)$$

4. Substitute the values of a_0, a_n and b_n in the Equation (2) we get the quadrature fourier series for x (y) as :

$$x(t) = \frac{A}{2} \sum_{n=1}^{\infty} \frac{2A}{n\pi} \sin\left(\frac{n\pi}{2}\right) \cos n\omega_0 t \quad \dots(6)$$

Explanation : Equation (6) can be expanded by opening the summation sign as follows :

$$x(t) = \frac{A}{2} + \frac{2A}{n\pi} \sin\left(\frac{\pi}{2}\right) \cos \omega_0 t + \frac{2A}{2\pi} \sin \pi \cos 2\omega_0 t + \frac{2A}{3\pi} \sin \frac{3\pi}{2} \cos 3\omega_0 t + \dots$$

$$= \frac{A}{2} + \frac{2A}{n\pi} \cos(\omega_0 t) - \frac{2A}{3\pi} \cos(3\omega_0 t) + \frac{2A}{5\pi} \cos(5\omega_0 t) + \dots \quad \dots(7)$$

- Substitute $\frac{2A}{\pi} = A_1$ to get,

$$x(t) = A/2 + A_1 \cos \omega_0 t - A_1/3 \cos(3\omega_0 t) + A_1/5 \cos(5\omega_0 t) + \dots \quad \dots(8)$$

DC value
Fundamental
Third harmonic
Fifth harmonic

(G-828)

Conclusions :

- Equation (8) shows that a square wave consists of a dc component, fundamental and only the odd harmonic components such as third, fifth, seventh ... etc.
- The even harmonic components are automatically cancelled.
- The amplitude of harmonic components decreases with increase in their order.

3.5.1 Fundamental Component :

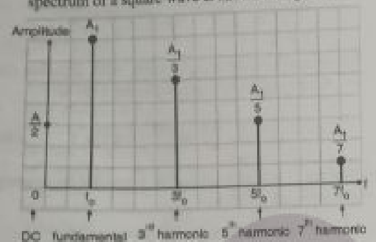
The composite signal x(t) has a frequency of f_0. Now refer to Equation (8) in Ex. 3.5.1. The second term $A_1 \cos \omega_0 t$ has the same frequency as that of composite signal. It is called as fundamental frequency or fundamental component.

3.5.2 Harmonics :

- The other sinewaves (cosine terms in Equation (8)) having frequencies which are integral multiples of fundamental frequency are called as harmonics.
- The sinewave having a frequency $3f_0$ is called as third harmonic, then the sinewave with frequency $5f_0$ is fifth harmonic and so on.

3.5.3 Frequency Spectrum :

- The frequency domain description of a signal is called as frequency spectrum.
- A square wave is made of fundamental and odd harmonics (third, fifth, seventh etc). The frequency spectrum of a square wave is shown in Fig. 3.5.1.



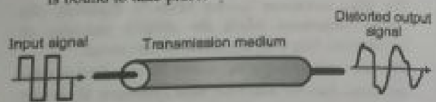
(G-821) **Fig. 3.5.1 : Frequency spectrum of a square wave**

3.6 Composite Signal and Transmission Medium :

- The data is generally in the form of pulses and pulse is a composite signal which contains many frequencies.
- Note that the peculiar shape of a pulse is due to the sum of specific frequencies at specific amplitudes and phases.
- If there is any change in the amplitudes or phases of these frequency components, then the shape of the pulse will not remain the same.

3.6.1 Medium :

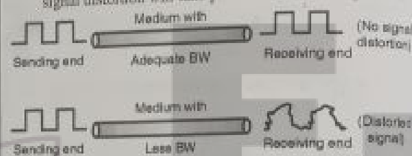
- The signal always travels over some medium from sender to destination.
- The medium can be a coaxial cable or optical fiber etc. A medium does not pass all frequencies equally due to its inadequate frequency spectrum.
- It may pass some frequencies and weaken or block the other frequencies.
- Hence when a composite signal is passed over such a transmission medium, at the receiving end we get a wave, having a different shape as shown in Fig. 3.6.1.
- To avoid the signal distortion, the medium should pass all the frequencies present at the input without any change.
- But no medium is perfect and so some signal distortion is bound to take place.



(G-822) **Fig. 3.6.1 : Signal distortion on a transmission medium**

3.7 Channel Bandwidth :

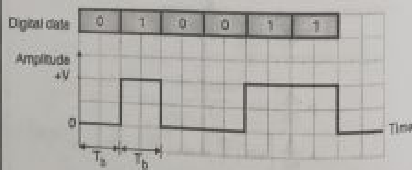
- The range of frequencies that contain the information is called as the bandwidth. But the term channel bandwidth is used to describe the range of frequencies required to transmit the desired information.
- For example the amplitude modulation (AM) system needs a channel bandwidth of 10 kHz to transmit a signal of 5 kHz bandwidth.
- But the single sideband system (SSB) needs only 5 kHz channel bandwidth to transmit the same signal.
- All the efforts should be made to reduce the required channel bandwidth so that we can fit in more number of channels in the same available EM spectrum.
- Bandwidth of a medium (also called as channel bandwidth) is defined as the maximum frequency it can allow to pass through it without attenuating it and without distorting the shape of the signal.
- If the medium has less bandwidth than required, then signal distortion will take place as shown in Fig. 3.7.1.



(L-30) **Fig. 3.7.1 : Importance of channel BW**

3.8 Digital Signals :

- The input data which is either analog or digital can also be represented by a digital signal.
- A digital signal is a discrete time signal having finite number of amplitudes. For example see the digital signal shown in Fig. 3.8.1.
- A 0 is represented by zero volts and a 1 by some positive voltage.



(L-26) **Fig. 3.8.1 : Digital signal**

3.8.1 Bit Interval (T_b) :

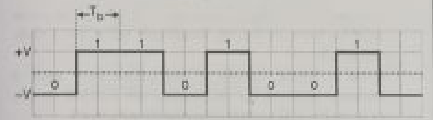
- The bit interval is the time corresponding to one single bit (0 or 1).
- As shown in Fig. 3.8.2, time corresponding to a 0 or a 1 is T_b hence it is the bit interval or bit length.

3.8.2 Bit Rate (Data Rate) :

- Bit rate is defined as the number of bits transmitted or sent in one second. It is expressed in bits per second (bps).
- Relation between bit rate and bit interval is as follows :

$$\text{Bit rate} = \frac{1}{\text{Bit interval}}$$

- Bit rate is also called as signalling rate and is defined as the number of bits which can be transmitted in a second.
- If the bit duration is "T_b" then bit rate will be $1 / T_b$. Look at Fig. 3.8.2, you will see that the bit duration is necessarily equal to the pulse duration.
- In Fig. 3.8.2 the first pulse is of two bit duration.

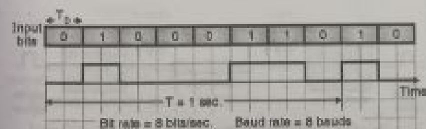


(L-27) **Fig. 3.8.2 : A bit stream**

- Bit rate is also called as signalling rate and it should be as high as possible.
- However with increase in bit rate the bandwidth of transmission medium (channel bandwidth) must be increased, in order to ensure that the signal is received without any distortion.

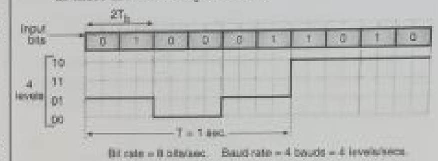
3.8.3 Bauds (or Baud Rate) :

- Baud is the unit of signalling speed or modulation rate or the rate of symbol transmission.
- It indicates the rate at which a signal level changes over a given period of time.
- When binary bits are transmitted as an electrical signal with two levels "0" and "1" the bit rate and the modulation rate i.e. baud rate are same. This is as shown in Fig. 3.8.3(a).
- Note that for a two level signal (binary signal) the bit rate and bauds are equal.
- Now consider Fig. 3.8.3(b) where four different levels are used to represent the data.



(L-28) **Fig. 3.8.3(a) : Baud rate for two level modulation**

- Each level is being represented by a combination of two bits i.e. 00 or 01 etc.
- The bit rate is therefore not equal to the baud rate.
- The bit rate is 8 bits/sec. but baud rate is only 4 bauds as there are 4-levels per second.



(L-29) **Fig. 3.8.3(b) : Baud rate for a four level modulation**

Ex. 3.8.1 : For a binary PCM system, the number of bits per transmitted word is 8 and the sampling frequency f_s = 8 kHz. Calculate the bit rate and baud rate.

Soln. :

Given : $N = 8$, $f_s = 8$ kHz

Bit rate = $N \times f_s = 8 \times 8$ kHz

= 64 k bits/sec.

Baud rate = Bit rate = 64 kHz

(as transmission is binary)

Ex. 3.8.2 : For the same data in the previous example, calculate the bit rate and baud rate if a QPSK system is used.

Soln. :

In the QPSK system, two successive bits are clubbed together to form one message. Hence one symbol corresponds to 2 bit duration.

$\therefore$ Baud rate = $\frac{1}{2} \times$ bit rate = 32 k bits/sec.

Bit rate does not change.

Ex. 3.8.3 : A system sends a signal that can assume 8 different voltage levels. It sends 400 of these signals per second. What are the baud and bit rates ?

Soln. :

1. As the signal assumes 8 different voltage levels we need 3 bit digital signal to have 8 different combinations. Hence the number of bits per voltage level is 3. Let each voltage level represent one symbol.

   $\therefore$ Number of bits / symbol = 3

2. The system sends 400 signal / sec. Hence the number of symbols transmitted per second is also 400.

   $\therefore$ Symbol rate = Number of symbols / sec.

   = 400 symbols / sec.

3. The baud rate is defined as the number of symbols per second.

   $\therefore$ Baud rate = Symbol rate.

   $\therefore$ Baud rate = 400 symbols / sec. ...Ans.

4. We are using 3 bit to represent each symbol.

   So bit rate = $3 \times$ symbol rate = 3×400

   = 1200 bits / sec. ...Ans.

Ex. 3.8.4 : A system sends a signal that can assume 4 different voltage levels. It sends 200 of such signals per second. What is the baud rate ?

Soln. :

This example is similar to Ex. 3.8.3. A system has 4 different voltage levels and it sends 200 of such signals/sec. Hence number of voltage levels transmitted will be 200/sec. Therefore the symbol rate is 200 symbols / sec.

$\therefore$ Baud rate = 200 symbols / sec. ...Ans.

Ex. 3.8.5 : A system sends a signal that can assume two different voltage level. It sends 100 of signal per second, what is baud rate ?

Soln. :

1. As the signal assumes 2 different voltage levels we need, 1 bit digital signal to have 2 different combinations. Hence the number of bits per voltage is 1. Let each voltage level represent one symbol.

   $\therefore$ Number of bits / symbol = 1

2. The system sends 100 signals / sec. Hence the number of symbols transmitted per second is also 100.

   $\therefore$ Baud rate = 100 symbols / sec.

3.8.4 Bit Length :

- The bit length for a digital signal is similar to the term wavelength for an analog signal.
- Bit length of a digital signal is defined as the distance corresponding to one bit on the transmission medium. It is measured in meters or cm.

$\therefore$ Bit length = Propagation speed $\times$ Bit duration $= \frac{\text{meters}}{\text{sec}} \times \text{sec}$

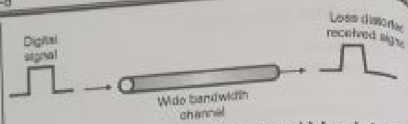
3.9 Transmission of Digital Signals :

The digital signals can be transmitted from one point to the other using one of the following two approaches :

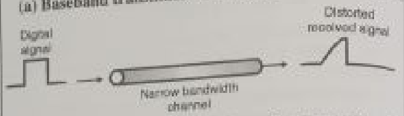
1. Baseband transmission
2. Bandpass transmission (with modulation)

3.9.1 Baseband Transmission :

- A baseband digital signal is the original signal without any modulation. In the modulation process the baseband digital signal is converted into analog signal.
- Baseband transmission is the transmission of baseband digital signal.
- A baseband signal occupies bandwidth from 0 to f_1 Hz.
- Hence baseband transmission requires the use of low pass channel. This low pass channel can have a narrow bandwidth or wide bandwidth.
- If we send the digital signals over the low pass channel with a small bandwidth (telephone cable) then same frequency components in the digital signal get blocked and the shape of the received signal will be badly distorted as shown in Fig. 3.9.1(b).



(a) Baseband transmission through a wideband channel



(b) Baseband transmission through a narrowband channel

(G-1301) Fig. 3.9.1

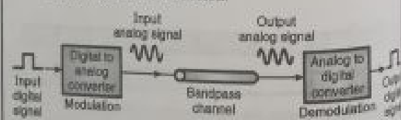
- A practically available medium such as coaxial cable does not have an infinite bandwidth but it has a wide bandwidth.
- When a digital signals is transmitted over such a medium, some of the frequencies are blocked by the medium but still enough frequencies are passed.
- So the digital signal at the receiving end will have a different shape with a small distortion as shown in Fig. 3.9.1(a).

Conclusion :

Baseband transmission of digital signals over a low pass channel without waveform distortion is possible if and only if the channel has a wide or infinite bandwidth.

3.9.2 Broadband Transmission (With Modulation) :

- A baseband signal is passed through a D to A converter to obtain an equivalent analog signal. This is modulation and the modulated analog signal is called a broadband signal.
- Transmission of a broadband signal is known as broadband transmission of digital signal. The spectrum of a broadband signal extends from f_1 to f_2 so it is a bandpass spectrum.
- We have to use a bandpass channel to carry this transmission. Fig. 3.9.2 shows the modulation process and broadband transmission.



(G-1302) Fig. 3.9.2 : Broadband transmission of digital signal

- Note that the output digital signal is distortion free.
- We can send the digital signals over a bandlimited medium. The best example of this is the data transfer over telephone cables in internet.

- The most important question here is that what should be minimum bandwidth of the medium (B Hz) if we want to transmit a signal of n bps.
- The answer to this question will be given when we study the Nyquist theorem and Shannon capacity.

3.9.3 Relation between Required Bandwidth and Bit Rate :

- In computer communication we have to send as many bits as possible per second for fast data transfer.
- That means the bit rate should be as high as possible. But increase in bit rate has an undesired side effect.
- The signal bandwidth and the required bandwidth of the medium (channel bandwidth) increase with increase in bit rate. If we double the bit rate then the required channel bandwidth needs to be doubled.
- Thus bit rate and bandwidth are proportional to each other.
- The general relation between required bandwidth (B) and bit rate (n) is as follows

$$B \geq \frac{n}{2} \text{ or } n \leq 2B.$$

- Thus over a medium having a bandwidth of 4 kHz we can send a digital signal with a bit rate upto 8 kbps.
- In practice the maximum bit rate can be more than 30 kbps using the traditional MODEMS.

3.10 Some Important Definitions :

Channel Capacity (C) :

- The channel capacity is defined as the maximum data rate at which the digital data can be transmitted over the channel reliably.
- The various other concepts related to channel capacity are as follows :

1. Data rate 2. Bandwidth
3. Noise 4. Error rate

Data Rate :

- It is defined as the number of bits transmitted by the transmitter per second. It indicates how fast a signal can be transmitted reliably over the given medium.
- This capability depends on the following factors :
  1. The amount of energy put into transmitting each signal.
  2. Distance to be travelled.
  3. Noise.
  4. Channel bandwidth

Channel bandwidth :

- The bandwidth of the communication medium should be large enough to transmit the digital signal reliably.
- An inadequate bandwidth will distort the signal and introduce errors into the received signal.

Noise : This is the average level of noise over the communication path.

Error rate : It is defined as the rate at which errors occur in the received (or detected) signal.

3.11 Transmission Impairments :

- In any communication system, the received signal is never identical to the transmitted one due to some transmission impairments.
- The quality of analog signals will deteriorate due to the transmission impairment whereas errors get introduced if the signal is digital.
- The most important reasons for the impairments are as follows :
  1. Attenuation and attenuation distortion
  2. Delay distortion
  3. Noise.

(G-86) Fig. 3.11.1 : Impairment types

3.11.1 Attenuation :

- The strength of a signal (signal energy) decrease with increase in distance travelled over a medium. This is known as attenuation.
- When any signal travels over a medium or channel, it loses some of its energy in the form of heat in the resistance offered to the signal by the medium.
- Attenuation is expressed in decibels as,

$$\text{Attenuation (dB)} = 10 \log_{10} \frac{P_{out}}{P_{in}} \quad ...(3.11.1)$$

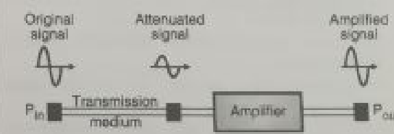
Where P_{in} = Power at the sending end

P_{out} = Power at the receiving end.

- Attenuation decides the signal strength and hence the signal to noise ratio and the quality of received signal.

Remedy to reduce attenuation :

We can introduce amplifiers to compensate for any loss of signal as shown in Fig. 3.11.2. Note that amplification and attenuation are exactly opposite to each other.



(G-87) Fig. 3.11.2 : Concept of attenuation

Ex. 3.11.1 : Calculate the attenuation in dB if input power is three times higher than the received power.

Soln. :

$P_{in} = 3 P_{out}$

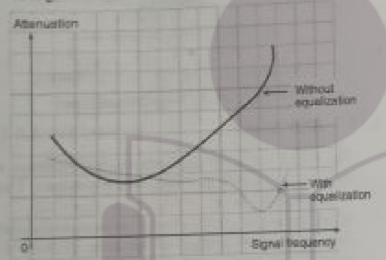
$\therefore \frac{P_{out}}{P_{in}} = \frac{1}{3}$

$\therefore$ Attenuation in dB $= 10 \log_{10} \frac{P_{out}}{P_{in}} = 10 \log_{10} (1/3)$

$= -4.8$ dB ...Ans.

Effect of frequency on attenuation :

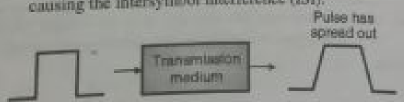
- Attenuation increases with increase in signal frequency.
- The SNR can be improved by using amplifiers or repeaters to boost the signal strength.
- The effect of frequency on attenuation is important if the signal is analog. This effect can be reduced by using equalizers. Another way to do this is to use an amplifier that amplifies the higher frequencies more than lower frequencies.
- The effect of equalization on attenuation is illustrated in Fig. 3.11.3.



(G-88) Fig. 3.11.3 : Effect of equalizer on attenuation

3.11.2 Delay Distortion :

- This problem is particularly present in the wired media.
- This distortion is caused due to a property which states that the velocity of propagation of a signal through a medium varies with frequency.
- The velocity is maximum near the center frequency and reduces as the signal frequency deviates away from the center frequency on both sides of frequency spectrum.
- Hence various frequency components of a signal arrive at the receiver at different instants of time resulting in phase shifts between different frequencies.
- This distorts the signal and the distortion is called as delay distortion.
- Delay distortion is particularly important for digital data.
- Due to delay distortion a digital pulse transmitted over a medium tends to spread out as shown in Fig. 3.11.4 causing the intersymbol interference (ISI).

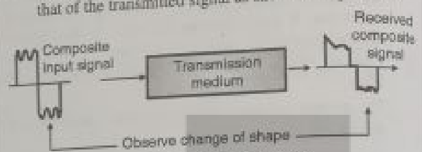


(G-89) Fig. 3.11.4 : Delay distortion

- The Intersymbol interference (ISI) put a major limitation on the maximum bit rate over a transmission channel.
- Delay distortion can be reduced by using equalizers.

3.11.3 Harmonic Distortion :

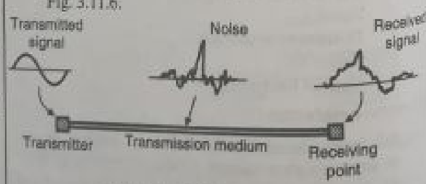
- Another meaning of distortion is change in shape of the signal. The shape of the received signal can be substantially different than the transmitted signal.
- If the medium is not perfect, then all the frequency components present at the input will not be equally attenuated and will not be proportionally delayed.
- Hence the shape of the received signal is different from that of the transmitted signal as shown in Fig. 3.11.5.



(G-90) Fig. 3.11.5 : Distortion

3.11.4 Noise :

- When the data travels over a transmission medium, noise gets added to it.
- Noise is a major limiting factor in communication system performance.
- Noise can be categorized into four types as follows :
  1. Thermal noise 2. Intermodulation noise
  3. Crosstalk 4. Impulse noise
- Thermal noise is due to the random motion of electrons in a wire. But the noise induced into a wire is generated by the sources such as electric motors and various appliances.
- Crosstalk is the interference caused by one wire to the other one. Here one wire radiates its signal by acting as a sending antenna and the other one acts as a receiving antenna to induce voltage into it.
- Impulse noise is in the form of a high energy spike. It is basically a pulse of short duration which comes from power lines, lightning etc.
- The effect of noise on a signal has been illustrated in Fig. 3.11.6.



(L-779) Fig. 3.11.6 : Effect of noise

Thermal Noise :

- It is due to thermal agitation of electrons.

- Thermal noise is present in all electronic devices and the transmission media.
- Thermal noise is proportional to temperature and it is uniformly distributed across the frequency spectrum. So it is called as white noise.
- We cannot eliminate the thermal noise completely. So it puts a limit on the performance of the communication system.
- The thermal noise power is given by,

$$N = kTB \text{ Watts} \quad ...(3.11.2)$$

where k = Boltzman's constant

T = Absolute temperature °K

B = Bandwidth

- It can be expressed in dBW as :

$$N = 10 \log k + 10 \log T + 10 \log B$$

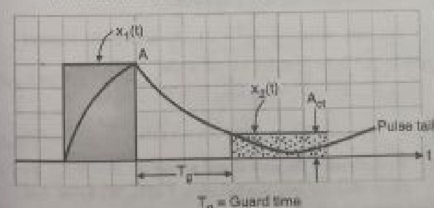
- Thermal noise can be reduced by reducing temperature or bandwidth.

Intermodulation Noise :

- If signals at different frequencies are transmitted simultaneously on a common transmission medium (e.g. FDM) then it results in intermodulation noise.
- The intermodulation produces frequency components at sum and difference frequencies of the frequencies travelling on the medium.
- Suppose signals of frequencies f_1 and f_2 are travelling simultaneously over the same medium, then due to intermodulation produces frequency components $(f_1 + f_2)$ and $(f_1 - f_2)$, which are called intermodulation noise.
- This is similar to a mixer. So whenever there is any non-linearity in the transmitter or channel, the intermodulation noise is produced.

Crosstalk :

- Crosstalk basically means interference between the adjacent telephone channels.
- It is the unwanted coupling of information from one channel to the other adjacent channels. The guard time (T_g) is the time spacing introduced between the adjacent telephone channels represented by $x_1(t)$ and $x_2(t)$ in Fig. 3.11.7.
- A signal without baseband filtering also has crosstalk if the pulse tail or postcursor overlaps into the next time slot of the frame as shown in Fig. 3.11.7.

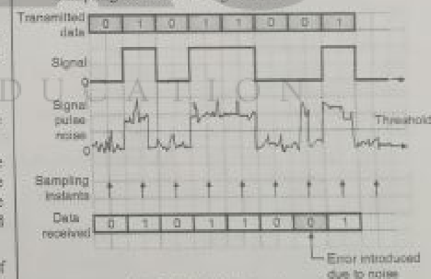


(G-91) Fig. 3.11.7 : Crosstalk

- Pulse overlap shown in Fig. 3.11.7 can be controlled by using "guard time" (T_g) between the pulses. The guard time (T_g) is analogous to the guard bands between the channels in the FDM systems.
- In practice crosstalk can be experienced while using a telephone. We can hear another conversation due to crosstalk.
- It can occur due to electrical coupling between nearby twisted pairs or sometimes even in the coaxial cables.
- Typically the magnitude of crosstalk is almost same as that of the thermal noise.

Impulse Noise :

- All the types of noise discussed so far are reasonably predictable and their magnitude is relatively constant.
- So it is a bit easier for the system engineers to deal with them.
- But impulse noise is completely different from the other types. It is non continuous, by nature and it is made of noise spikes of short duration of high amplitude. These noise pulses occure irregularly and they are extremely unpredictable.
- Impulse noise gets generated due to many reasons such as external electromagnetic disturbances, lightning etc.
- Impulse noise does not affect the quality of analog signal to a great extent but it affect the digital data badly because it introduces errors in the digital data as shown in Fig. 3.11.8.
- It is possible to recover the original data by means of sampling as shown in Fig. 3.11.8.



(G-92) Fig. 3.11.8 : Effect of impulse noise

3.11.5 Signal to Noise Ratio :

- The signal to noise ratio (SNR) is defined as :

$$SNR = \frac{\text{Average Signal Power}}{\text{Average Noise Power}}$$

- SNR can be used to find the theoretical bit rate limit of a given communication medium. SNR is the ratio of the desired portion (signal) and the undesired portion (noise) in the transmitted or received waveform. Its value should be as high as possible.
- SNR is a ratio of two powers. So it is often defined in decibels (dB).

$$[SNR]_{dB} = 10 \log_{10} SNR$$

Note :

1. A high SNR indicates that the signal is less degraded due to noise.
2. A low SNR indicates that the signal is heavily degraded due to the noise.

3.12 Data Rate Limits :

- In data communication a large data is required to be transferred from one place to the other.
- It is necessary to transfer it as quickly as possible. In other words the data rate in bits per second over a channel should be as high as possible.
- The data rate is decided by the following factors :
  1. The maximum bandwidth.
  2. The signal level.
  3. The noise presented by the channel.
- Two theorems were developed to calculate the data rate and we can use them on the basis of the type of channel as follows :
  1. A noiseless channel : Nyquist theorem
  2. A noisy channel : Shannon's theorem

3.12.1 Noiseless Channel : Nyquist Bit Rate :

- As we know a transmission channel is a medium over which the electrical signals from a transmitter travel to the receiver. Two important characteristics of a transmission channel are :
  1. Signal to Noise ratio (SNR) and
  2. Channel bandwidth.
- These two characteristics will ultimately decide the maximum capacity of a channel to carry information.
- Nyquist and Shannon worked on finding the maximum channel capacity of a bandlimited channel.
- Nyquist's theorem states that if the bandwidth of a transmission channel is "B" which carries a signal having "L" number of levels, then the maximum data rate "R" on this channel is given by.

$$R = 2 B \log_2 L \qquad ...(3.12.1)$$

- As maximum data rate for reliable transmission is defined as channel capacity C, the above expression gets modified as :

$$C = 2 B \log_2 L \qquad ...(3.12.2)$$

- This expression indicates that the data rate can be increased by increasing the number of different signal elements (L).

3.12.2 Noisy Channel : Shannon's Channel Capacity :

- A noiseless channel is not possible in the real world, so Shannon introduced a theorem called Shannon's capacity theorem to determine the highest possible data rate on the noisy channel.
- We have seen the Nyquist bandwidth earlier in this chapter.
- Shannon extended Nyquist's work. He included the effect of noise present on the transmission channel.
- According to Shannon's theorem, if (S/N) is the signal to noise ratio then the maximum data rate is given by

$$C = R_{max} = B \log_2 \left[1 + \frac{S}{N}\right] \text{ bits/sec} \qquad ...(3.12.3)$$

- Shannon's theorem puts a limit on the maximum number of levels for a given (S/N) ratio and bandwidth. This expression shows that the maximum data rate for a communication channel is dependent on the channel bandwidth B and signal to noise ratio (S/N).
- It is important to note that the Shannon's formula does not indicate the signal level. It says no matter how much is the signal level, it is not possible to achieve a data rate (R) which is greater than the capacity of the channel (C).

Importance of channel bandwidth :

- Bandwidth of the communication channel should be higher than the bandwidth of the signal that is to be transmitted over it.
- This is essential in order to preserve the shape of the signal being transmitted.
- If the channel bandwidth is less than the signal bandwidth then the signal shape will be distorted when it travels over this channel.

3.12.3 Solved Examples :

Ex. 3.12.1 : A channel has a bandwidth of 5 kHz and a signal to noise power ratio 63. Determine the bandwidth needed if the S/N power ratio is reduced to 31. What will be the signal power required if the channel bandwidth is reduced to 3 kHz ?

Soln. :

1. **To determine the channel capacity :**

It is given that B = 5 kHz and $\frac{S}{N} = 63$. Hence using the Shannon Hartley theorem the channel capacity is given by,

$$C = B \log_2 \left[1 + \frac{S}{N}\right] = 5 \times 10^3 \log_2 [1 + 63]$$

$$\therefore \quad C = 30 \times 10^3 \text{ bits/sec.} \qquad ...(1)$$

2. **To determine the new bandwidth :**

The new value of $\frac{S}{N} = 31$. Assuming the channel capacity "C" to be constant we can write,

$$30 \times 10^3 = B \log_2 [1 + 31]$$

$$\therefore \quad B = \frac{30 \times 10^3}{5} = 6 \text{ kHz} \qquad ...(2)$$

3. **To determine the new signal power :**

Given that the new bandwidth is 3 kHz. We know that noise power $N = N_0 B$.

Let the noise power corresponding to a bandwidth of 6 kHz be $N_1 = 6 N_0$ and the noise power corresponding to the new bandwidth of 3 kHz be $N_2 = 3 N_0$.

$$\therefore \quad \frac{N_1}{N_2} = \frac{6 N_0}{3 N_0} = 2 \qquad ...(3)$$

The old signal to noise ratio $= \frac{S_1}{N_1} = 31$

$$\therefore \quad S_1 = 31 N_1 \qquad ...(4)$$

The new signal to noise ratio $= \frac{S_2}{N_2}$. We do not know its value, hence let us find it out.

$$30 \times 10^3 = 3 \times 10^3 \log_2 \left(1 + \frac{S_2}{N_2}\right)$$

$$\therefore \quad \frac{S_2}{N_2} = 1023 \qquad ...(5)$$

$$\therefore \quad S_2 = 1023 N_2$$

But from Equation (3), $N_2 = \frac{N_1}{2}$, substituting we get,

$$\therefore \quad S_2 = 1023 \frac{N_1}{2} \qquad ...(6)$$

Dividing Equation (6) by Equation (4) we get,

$$\therefore \quad \frac{S_2}{S_1} = \frac{1023 N_1}{2 \times 31 N_1} = 16.5$$

$$\therefore \quad S_2 = 16.5 S_1 \qquad ...Ans.$$

Thus if the bandwidth is reduced by 50% then the signal power must be increased 16.5 times i.e. 1650% to get the same capacity.

Ex. 3.12.2 : Calculate the maximum bit rate for a channel having bandwidth 3100 Hz and S/N ratio 20 dB.

Soln. :

Given : B = 3100 Hz

$$\frac{S}{N} = 20 \text{ dB.}$$

But 20 dB = 10 log (S/N)

$\therefore$ S/N = 100

The maximum bit rate is given by,

$$R_{max} = B \log_2 \left[1 + \frac{S}{N}\right]$$

$$= 3100 \log_2 [1 + 100]$$

$$= \frac{3100 \log_{10} 101}{\log_{10} 2}$$

$$= 20{,}640 \text{ bits/sec.} \qquad ...Ans.$$

Ex. 3.12.3 : Calculate the maximum bit rate for a channel having bandwidth 3100 Hz and S/N ratio 10 dB.

Soln. :

Given : B = 3100 Hz

$$\left(\frac{S}{N}\right)_{dB} = 10$$

$$\therefore \quad 10 = 10 \log_{10} \left(\frac{S}{N}\right)$$

$$\therefore \quad \frac{S}{N} = 10$$

$$\therefore \text{ Maximum bit rate} = R_{max} = B \log_2 \left[1 + \frac{S}{N}\right]$$

$$= 3100 \log_2 (1 + 10)$$

$$= \frac{3100 \log_{10} 11}{\log_{10} 2}$$

$$= 10{,}724 \text{ bits/s} \qquad ...Ans.$$

Ex. 3.12.4 : Calculate the maximum bit rate for a channel having bandwidth 1600 Hz if :
(a) S/N ratio is 0 dB (b) S/N ratio is 20 dB.

Soln. :

Given : B = 1600 Hz

(a) R_{max} for S/N = 0 dB :

$$\left(\frac{S}{N}\right)_{dB} = 10 \log_{10} \left(\frac{S}{N}\right)$$

$$\therefore \quad \frac{S}{N} = 1$$

$$\therefore \quad R_{max} = B \log_2 (1 + S/N) = 1600 \log_2 (1 + 1)$$

$$= 1600 \text{ bits/sec} \qquad ...Ans.$$

(b) R_{max} for S/N = 20 dB :

$$\left(\frac{S}{N}\right)_{dB} = 10 \log_{10} \left(\frac{S}{N}\right)$$

$$\therefore \quad 20 = 10 \log_{10} (S/N)$$

$$\therefore \quad \frac{S}{N} = 100$$

$$\therefore \quad R_{max} = B \log_2 \left(1 + \frac{S}{N}\right)$$

$$= 1600 \log_2 (101) = \frac{1600 \log_{10} (101)}{\log_{10} (2)}$$

$$R_{max} = 10{,}654 \text{ bits/sec} \qquad ...Ans.$$

Using both the limits :

In practice we have to use both the methods to calculate the required bandwidth and signal level. Consider the following example for the same.

Ex. 3.12.5 : The bandwidth of a channel is 2 MHz and its signal to noise ratio is 63. Calculate the appropriate bit rate and signal level.

Soln. :

Step 1 : Calculate the upper limit using Shannon's theorem :

$$C = B \log_2 \left(1 + \frac{S}{N}\right)$$

$$= 2 \times 10^6 \log_2 (1 + 63)$$

$$= 12 \text{ M bits / sec.}$$

This is the upper limit. For ensuring a better performance we select a somewhat lower value say 8 Mbps.

Step 2 : Calculate number of signal levels using Nyquist theorem :

$$C = 2\,B \log_2 L$$

$$\therefore \quad 8 \times 10^6 = 2 \times 2 \times 10^6 \log_2 L$$

$$\therefore \quad 2 = \log_2 L$$

$$\therefore \quad L = 4 \qquad \text{...Ans.}$$

Ex. 3.12.6 : Calculate the maximum bit rate of channel having bandwidth 1200 Hz if :

1. S/N ratio is 0 dB 2. S/N ratio is 20 dB

Soln. :

Given : B = 1200 Hz.

1. R_{max} for S/N = 0 dB

$$\left(\frac{S}{N}\right)_{dB} = 10 \log_{10}\left(\frac{S}{N}\right)$$

$$\frac{S}{N} = 1$$

$$\therefore \quad R_{max} = B \log_2\left(1 + \frac{S}{N}\right) = 1200 \log_2 (1 + 1)$$

$$= 1200 \text{ bits/sec} \qquad \text{...Ans.}$$

2. R_{max} for S/N = 20 dB

$$\left(\frac{S}{N}\right)_{dB} = 10 \log_{10}\left(\frac{S}{N}\right)$$

$$20 = 10 \log_{10}\left(\frac{S}{N}\right)$$

$$\therefore \quad \frac{S}{N} = 100$$

$$\therefore \quad R_{max} = B \log_2\left(1 + \frac{S}{N}\right)$$

$$= 1200 \log_2 (101) = 1200 \frac{\log_{10} (101)}{\log_{10} (2)}$$

$$= 7990 \text{ bits/sec} \qquad \text{...Ans.}$$

Maximum bit rate = 1200 bits/sec for 0 dB

Maximum bit rate = 7990 bits/sec for 20 dB

Ex. 3.12.7 : Find the number of coding or symbol levels if C = 31000 bits/s and B is 3100 Hz.

Soln. : C is the channel capacity while B is the bandwidth.

According to Shannon's theorem,

$$C = B \log_2\left(1 + \frac{S}{N}\right)$$

where S/N is the signal to noise ratio.

$$\therefore \quad 31000 = 3100 \log_2\left(1 + \frac{S}{N}\right)$$

$$\therefore \quad \log_2\left(1 + \frac{S}{N}\right) = 10$$

$$\therefore \quad \frac{S}{N} = 1023 \text{ or } 30\text{dB}$$

$$\left(\frac{S}{N}\right) \text{dB} = 1.8 + 6 \text{ NdB}$$

$$\therefore \quad 30 = 1.8 + 6N$$

where N = Number of bits per word.

$$\therefore \quad N = 4.72 \approx 5$$

Number of symbol levels $Q = 2^N = 2^5 = 32$ ...Ans.

Ex. 3.12.8 : Calculate the channel capacity for a noisy channel having bandwidth = 5 kHz and SNR = 0 using appropriate formula.

Soln. :

Given : B = 5 kHz, $\frac{S}{N} = 0$

Find : Channel capacity.

1. **To determine channel capacity :**

$$C = B \log_2\left[1 + \frac{S}{N}\right] = 5 \times 10^3 \log_2 [1 + 0]$$

$$C = 0 \text{ bits/sec.} \qquad \text{...Ans.}$$

Ex. 3.12.9 : An analog signal has a bit rate of 8000 bps and a baud rate of 1000 baud. How many data elements are carried by each signal element ? How many signal elements do we need ?

Soln. :

Given : Bit rate (n) = 8000 bps, Baud rate = 1000 baud.

To find : 1. Data elements carried by each signal element (R)

2. Total signal elements (L)

Step 1 : Calculate R :

Bit rate = Numbers of data elements per signal × Baud rate

$$\therefore \text{ Number of data elements per signal (R)} = \frac{\text{Bit rate (n)}}{\text{baud rate}}$$

$$\therefore \quad R = \frac{8000}{1000} = 8 \text{ bits/ baud} \qquad \text{...Ans.}$$

Step 2 : Calculate L :

Total signal elements $(L) = 2^R = 2^8 = 256$ ...Ans.

Ex. 3.12.10 : State and explain the Nyquist theorem and Shannon capacity and solve the following example :

Calculate the maximum bit rate for noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels.

Soln. : For Nyquist theorem and Shannon capacity refer sections 3.12.1 and 3.12.2.

Given : B = 3000 Hz, L = 2

To find : Maximum bit rate

$$\text{Maximum bit rate (R)} = 2B \log_2 L$$

$$= 2 \times 3000 \log_2 (2)$$

$$= 2 \times 3000 \times \left(\frac{\log_{10} (2)}{\log_{10} 2}\right)$$

$$\therefore \quad R = 6000 \text{ Bits/sec.} \qquad \text{...Ans.}$$

3.13 Performance :

- The performance of a data communication network can be measured with the help of the following parameters :
  1. Bandwidth 3. Jitter
  2. Delay 4. Throughput
- We will discuss them one by one in the following sections.

3.14 Bandwidth :

- Bandwidth is a very important characteristics of a network, which can be used for measuring the network performance.
- Bandwidth can have two different values :
  1. BW in hertz and 2. BW in bits per second.

BW in Hz :

It is the range of frequencies present in a composite signal. It can also be defined as a range of frequencies that a channel can pass through without much attenuation.

BW in bits per second :

We can also define bandwidth as the number of bits per second (bps) that a channel or network can transmit. For example the BW of Fast Ethernet is 100 Mbps i.e. that network can transmit 100 Mbps.

Relationship :

- There is a clear relationship between the bandwidth in Hz and BW in bps. With increase in BW in Hz, there is an increase in bps bandwidth.
- The relation between them depends on whether baseband transmission is being used or transmission with modulation is being used.

3.14.1 Signal and Channel Bandwidths :

- We can define two different bandwidths :
  1. Signal bandwidth 2. Channel bandwidth
- Signal bandwidth B_s is defined as the range of frequencies contained in the signal.
- Whereas the bandwidth of a channel B_c is the range of frequencies that is passed by a channel.
- If the bandwidth of the input signal is larger than the channel bandwidth, then the output of the channel will not contain all the frequencies of the input signal.
- Fig. 3.14.1 shows a typical digital signal at the input and the output is of different shape than input, if the channel BW is less than signal BW.

(G-698) Fig. 3.14.1 : Effect of $B_c < B_s$

- If the signaling rate of input signal is increased, then the channel bandwidth has to be increased so as to pass the signal without any change in shape of the signal.
- The maximum rate at which pulses can be transmitted through the channel is given by,

$$r_{max} = 2\,W \text{ pulses/sec.}$$

where, $W = 1/2\tau$ and τ = Smallest pulse width of input signal.

- The bandwidth is also important in deciding the channel capacity C of a transmission system.
- **The channel capacity** C of a transmission system is the maximum rate at which bits can be transferred reliably.
- The relation between C and channel bandwidth B_c is given by,

$$C = B_c \log_2\left(1 + \frac{S}{N}\right) \text{ bits/sec.} \qquad \text{...(3.14.1)}$$

- The channel capacity should be as high as possible and to increase C. We have to increase the channel bandwidth B_c.
- For a telephone channel B_c = 3.4 kHz. If SNR = 10,000 then the channel capacity is given by,

$$C = 3400 \log_2 (1 + 10000) = 45,200 \text{ bits/sec.}$$

- That means we can transmit at a maximum rate of 45.2 kbits per second on the telephone channel.

3.15 Throughput (T) :

- The throughput is a parameter that is used to know the speed of data transmission over a network.
- The throughput of a system is defined as the actual rate at which the information is sent over the channel. It is measured in bits/second or frames/second.
- Throughput is a measure of performance of any network.
- The definitions of bandwidth and throughput appear to be the same but they are not. They are different.
- If B is the bandwidth and T is the throughput of a network then T is always less than and at the most equal to B.
- Thus bandwidth is the theoretical measurement while throughput is the actual measurement of how fast data can be sent.

Ex. 3.15.1 : A network has a bandwidth of 20 Mbps. It can pass 15,000 frames per minute and each frame contains 10,000 bits. Calculate the throughput. Comment on the result.

Soln. :

Given : $B = 20 \times 10^6$ bps, Number of frames = 15000 per min, Number of bits per frame = 10,000.

Throughput T = Number of frames per sec. × Number of bits per frame

$$= \frac{15000}{60} \times 10{,}000 = 3.5 \text{ Mbps.} \qquad \text{... Ans.}$$

Comment :

The value of B = 20 Mbps shows the potential of the network and T = 3.5 Mbps shows the actual capability.

3.16 Latency (Delay) :

- The latency or delay is defined as the time required for an entire message to reach its destination from the instant at which the first bit was sent out from the source.
- Latency is the sum of four delay components viz :
  1. Processing delay 2. Queueing delay
  3. Transmission delay 4. Propagation delay.
- The sum of all these delays amounts to the total delay or latency.

1. **Processing delay :**
   - A packet consists of a header and a data field as shown in Fig. 3.16.1. The header contains the destination address.
   - The time required to examine the packet header and deciding the direction in which the packet is to be sent is a part of the processing delay.
   - The processing delay may also include some other factors such as the time required to check the errors.
   - The processing delay is of the order of few microseconds or less.

Header	Data

(G-696(a)) **Fig. 3.16.1 : Format of a packet**

2. **Queueing delay :**
   - At the queue, the packets experience a queueing delay, when they wait to get transmit on the links.
   - The queueing delay depends on the number of packets arrived earlier in the queue.
   - With no waiting packets in the queue, the queueing delay will be equal to zero.
   - Queueing delays can be of the order of microseconds or milliseconds in practice.
3. **Transmission delay :**
   - The packets are transmitted on the first come first served basis. So a particular packet can get transmitted only after all the earlier packets are transmitted.
   - Transmission delay is also called as store and forward delay. It is the time required to push (transmit) all the packet bits into the link.
   - Typically, the transmission delay is of the order of microseconds or milliseconds in practice.

$$\text{Transmission delay} = \frac{\text{Message size}}{\text{Bandwidth}}$$

4. **Propagation delay :**
   - The time required for the packet bits to reach from the beginning of the link to the desired router is called as propagation delay.
   - The signals travel at a speed which is slightly less than that of light.
   - So propagation delay is the ratio of the distance to be travelled by the signal to the speed of propagation.
   - Practically the propagation delays are of the order of few milliseconds.

$$\text{Propagation delay} = \frac{\text{Distance}}{\text{Propagation speed}}$$

3.17 Jitter :

Definition :

- The delay introduced by the data communication networks is not constant. It varies packet to packet. The jitter measures the variability in packet delays and it is measured in terms of the difference of the minimum delay and maximum value of delay.
- Jitter is defined as the variation in delay for the packets belonging to the same flow.
- The real time audio and video cannot tolerate jitter on the other hand the jitter does not matter if the packets are carrying any data information contained in a file.
- For the audio and video transmission if the packets take 20 msec to 30 msec (delay) to reach the destination, it does not matter, provided that the delay remains constant.
- The quality of sound or video will be hampered if the delays associated with different packets have different values.

Jitter control :

- When a packet arrives at a router, the router will check to see whether the packet is behind or ahead and by what time.
- This information is stored in the packet and updated at every hop.
- If the packet is ahead of the schedule (early) then the router will hold it for a slightly longer time and if the packet is behind the schedule (late), then the router will try to send it out as quickly as possible.
- This will help in keeping the average delay per packet constant and will avoid time jitter.

3.17.1 Difference between Delay and Jitter :

End to end delay :

- End to end delay is the time required for the signal to travel from transmitter to receiver.
- This delay is due to the time required for buffering, queueing, switch and routing.
- This time delay remains the same for all the types of packets in the same flow.

Jitter :

Jitter is defined as the variation in delay for the packets belonging to the same flow. This is the difference between end to end delay and jitter.

Review Questions

Q. 1 What are the different methods of representing the data ?

Q. 2 Explain the importance of frequency spectrum in analysis of a signal.

Q. 3 Define the term signal bandwidth with the help of frequency spectrum.

Q. 4 Define analog and digital signals and compare them.

Q. 5 Define signal bandwidth and channel bandwidth.

Q. 6 Define : Bit interval, Bit rate, Baud rate.

Q. 7 What is Shannon's channel capacity ?

Q. 8 State and explain the term Nyquist's bandwidth.

Q. 9 What is the significance of channel capacity ?

Q. 10 Explain the effect of channel bandwidth.

Q. 11 What is meant by impulse noise ?

Q. 12 Write a short note on crosstalk and guard time.

Q. 13 What is delay distortion and what is its affect ?

Q. 14 State and explain various transmission impairments.

Q. 15 Define bandwidth in Hz and in bps.

Q. 16 What is throughput ? How is it different from bandwidth ?

Q. 17 Define latency and its four components.

Q. 18 Explain the concept of bandwidth delay product.

❑❑❑