Ch 3 Summary

- Data must be transformed to electromagnetic signals to be transmitted .
- Data can be analog or digital .
- Analog data are continuous and take continuous values .
- Digital data have discrete states and take discrete values .
- Signals can be analog or digital .
- Analog signals can have an infinite number of values in a range; digital signals can have only a limited number of values .
- In data communications, we commonly use periodic analog signals and nonperiodic digital signals .
- Frequency and period are the inverse of each other .
- Frequency is the rate of change with respect to time .
- Phase describes the position of the waveform relative to time 0 .
- A complete sine wave in the time domain can be represented by one single spike in the frequency domain .
- A single-frequency sine wave is not useful in data communications; we need to send a composite signal, a signal made of many simple sine waves .
- According to Fourier analysis, any composite signal is a combination of simple sine waves with different frequencies, amplitudes, and phases .

- The bandwidth of a composite signal is the difference between the highest and the lowest frequencies contained in that signal .
- A digital signal is a composite analog signal with an infinite bandwidth .
- Baseband transmission of a digital signal that preserves the shape of the digital signal is possible only if we have a low-pass channel with an infinite or very wide bandwidth .
- If the available channel is a bandpass channel, we cannot send a digital signal directly to the channel; we need to convert the digital signal to an analog signal before transmission .
- For a noiseless channel, the Nyquist bit rate formula defines the theoretical maximum bit rate .
- For a noisy channel, we need to use the Shannon capacity to find the maximum bit rate .
- Attenuation, distortion, and noise can impair a signal .
- Attenuation is the loss of a signal's energy due to the resistance of the medium .
- Distortion is the alteration of a signal due to the differing propagation speeds of each of the frequencies that make up a signal .
- Noise is the external energy that corrupts a signal .
- The bandwidthdelay product defines the number of bits that can fill the link .

3.8.2 Questions

Q3-1. What is the relationship between period and frequency?

Q3-1. The period of a signal is the inverse of its frequency and vice versa: $T = 1/f$ and $f = 1/T$.

Q3-2. What does the amplitude of a signal measure? What does the frequency of a signal measure? What does the phase of a signal measure?

The amplitude of a signal measures the value of the signal at any point.

The frequency of a signal measures the number of periods in one second.

The phase of a signal measures describe the position of the waveform relative to time zero.

Q3-3. How can a composite signal be decomposed into its individual frequencies?

Q3-3. *Fourier series* gives the frequency domain of a periodic signal; *Fourier analysis* gives the frequency domain of a nonperiodic signal.

Q3-4. Name three types of transmission impairment.

The three types of transmission impairment are:

1. Attenuation
2. Distortion
3. Noise

Q3-5. Distinguish between baseband transmission and broadband transmission.

Q3-5. *Baseband transmission* means sending a digital or an analog signal without modulation using a low-pass channel. *Broadband transmission* means to modulate signal using a band-pass channel.

Q3-6. Distinguish between a low-pass channel and a band-pass channel.

The difference between a low-pass channel and a band-pass channel is:

A low-pass channel contains a channel with a bandwidth that starts from zero. This is the case if we have a dedicated medium with a bandwidth constituting only one channel.

A band pass channel contains a channel with a bandwidth that does not start from zero. This type of channel is more available than a low-pass channel.

Q3-7. What does the Nyquist theorem have to do with communications?

Q3-7. The *Nyquist theorem* defines the maximum bit rate of a noiseless channel.

Q3-8. What does the Shannon capacity have to do with communications?

The capacity is to determine the theoretical highest data rate for a noisy channel. It is called the channel capacity theory, where double the bandwidth equals to double the highest data rate. This is of course theoretically and does not take into account thermal noise, impulse noise, attenuation distortion or delay distortion.

Capacity=bandwidth $\times(1+\mathrm{SNR})$

In this formula, bandwidth is the bandwidth of the channel, SNR is the signal-to-noise ration and capacity is the capacity of the channel in bits per second.

Q3-9. Why do optical signals used in fiber optic cables have a very short wave length?

Q3-9. A fiber-optic cable uses light (very high frequency). Since *f* is very high, the wavelength, which is $\lambda = c / f$, is very low.

Q3-10. Can we say whether a signal is periodic or nonperiodic by just looking at its frequency domain plot? How?

Yes, we can say. If a signal is periodic, the decomposition gives a series of signals with **discrete** frequencies.

If a signal is non-periodic, the decomposition gives a combination of sine waves with **continuous** frequencies.

Q3-11. Is the frequency domain plot of a voice signal discrete or continuous?

Q3-11. The frequency domain of a voice signal is normally *continuous* because voice is a nonperiodic signal.

Q3-12. Is the frequency domain plot of an alarm system discrete or continuous?

The frequency domain plot of an alarm system is **discrete** because alarm system is a periodic signal. A periodic signal completes a pattern within a measurable time frame, called a period, and repeats that pattern over subsequent identical periods.

Q3-13. We send a voice signal from a microphone to a recorder. Is this baseband or broadband transmission?

Q3-13. This is *baseband transmission* because no modulation is involved.

Q3-14. We send a digital signal from one station on a LAN to another station. Is this baseband or broadband transmission?

A digital signal transmitted from one station on a LAN to another station does not require any change or modulation.

• Baseband transmission is used to transfer digital signals without converting the signals.

• In broadband transmission the digital signal is converted to analog signal, modulation of the signal is required to transmit the signal.

• Therefore as no modulation of the signal is required the signal transmission is a baseband transmission.

Q3-15. We modulate several voice signals and send them through the air. Is this baseband or broadband transmission?

Q3-15. This is *broadband transmission* because it involves modulation.

3.8.3 Problems

P3-1. Given the frequencies listed below, calculate the corresponding periods.

a. 24 Hz **b.** 8 MHz **c.** 140 KHz

P3-1.

a.	$T = 1/f$	$= 1 / (24\ \text{Hz})$	$= 0.0417\ \text{s}$	$= 41.7\ \text{ms}$
b.	$T = 1/f$	$= 1 / (8\ \text{MHz})$	$= 0.000000125\ \text{s}$	$= 0.125\ \text{ms}$
c.	$T = 1/f$	$= 1 / (140\ \text{kHz})$	$= 7.14 \times 10^{-6}\ \text{s}$	$= 7.14\ \text{ms}$

P3-2. Given the following periods, calculate the corresponding frequencies.

a. 5 s **b.** 12 μs **c.** 220 ns

b)

Given Period (T) =12 μs =12 $\times 10^6$Hz [Since, μs =10^{-6}Hz]

Frequency (f) = $\frac{1}{\text{Period(T)}}$

= $\frac{1}{12\times10^6}$

=83333Hz

=83.333 $\times 10^3$Hz

= 83.333 KHz [1KHz=10^3Hz]

Therefore, Frequency (f) =83.333KHZ

a)

Given Period (T) = 5s

Frequency (f) = $\frac{1}{\text{Period(T)}}$

= $\frac{1}{5}$

=0.2Hz

Therefore, Frequency (f) =0.2Hz

c)

Given Period (T) =220ns=220 $\times 10^9$Hz [Since, ns=10^9Hz]

Frequency (f) = $\frac{1}{\text{Period(T)}}$

= $\frac{1}{220\times10^9}$

=4550000

=4.55 $\times 10^6$Hz

= 4.55MHz [1MHz=10^6Hz]

Therefore, Frequency (f) =4.55MHZ

P3-3. What is the phase shift for the following?

a. A sine wave with the maximum amplitude at time zero

b. A sine wave with maximum amplitude after 1/4 cycle

c. A sine wave with zero amplitude after 3/4 cycle and increasing

P3-3.

a. 90 degrees ($\pi/2$ radians)

b. 0 degrees (0 radians)

c. 90 degrees ($\pi/2$ radians) (Note that it is the same wave as in part *a*.)

P3-4. What is the bandwidth of a signal that can be decomposed into five sine waves with frequencies at 0, 20, 50, 100, and 200 Hz? All peak amplitudes are the same. Draw the bandwidth.

Given frequencies are:

0, 20, 50, 100 and 200Hz

Let f_h be the highest frequency, f_l be the lowest frequency and B is the bandwidth. Then

From given data,

f_h=200 and f_l=0

Bandwidth B= $f_h - f_l$

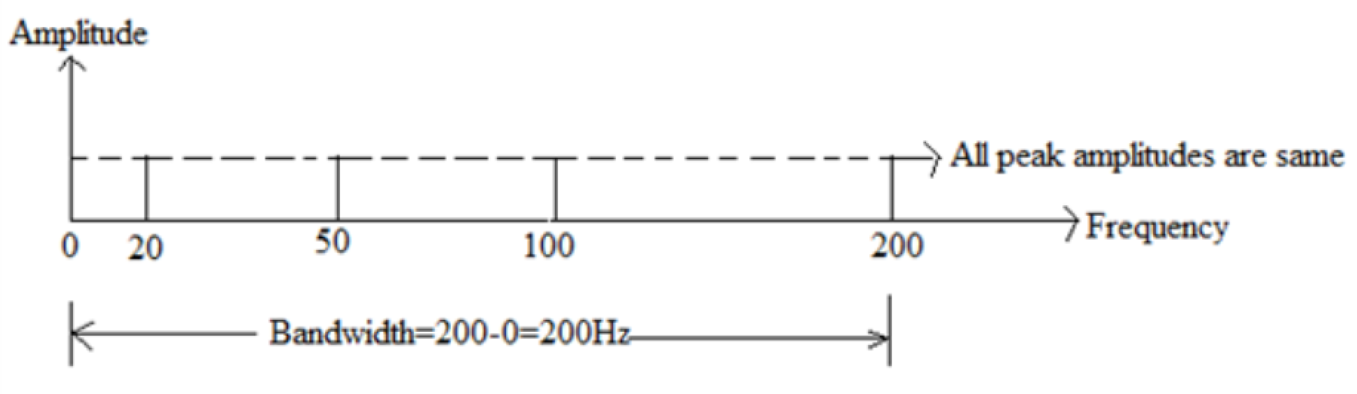
=200-0

=200Hz

Therefore, the Bandwidth=200Hz

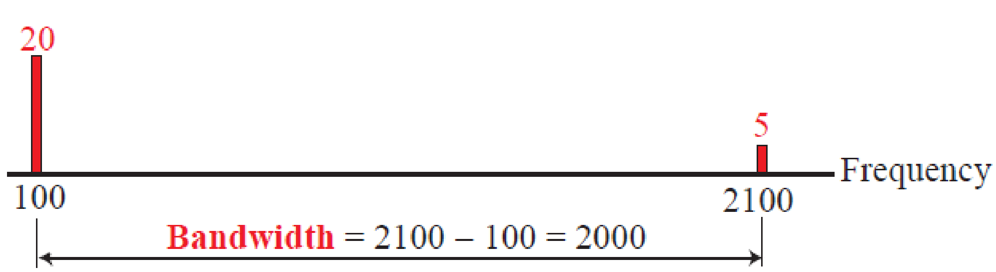
The bandwidth has only five spikes, at 0, 20, 50,100 and 200Hz. All peak amplitudes are the same.

The diagram of Bandwidth is:



P3-5. A periodic composite signal with a bandwidth of 2000 Hz is composed of two sine waves. The first one has a frequency of 100 Hz with a maximum amplitude of 20 V; the second one has a maximum amplitude of 5 V. Draw the bandwidth.

P3-5. We know the bandwidth is 2000. The highest frequency must be 100 + 2000 = 2100 Hz. See below:



P3-6. Which signal has a wider bandwidth, a sine wave with a frequency of 100 Hz or a sine wave with a frequency of 200 Hz?

A sine wave with a frequency of 200 Hz signal has the wider bandwidth because the general bandwidth is 2fm.

Bandwidth (B) = 2fm

$=2 \times 200$

= 400Hz

P3-7. What is the bit rate for each of the following signals?

a. A signal in which 1 bit lasts 0.001 s

b. A signal in which 1 bit lasts 2 ms

c. A signal in which 10 bits last 20 μs

P3-7.

a. bit rate = 1/ (bit duration) = 1 / (0.001 s) = 1000 bps = 1 Kbps

b. bit rate = 1/ (bit duration) = 1 / (2 ms) = 500 bps

c. bit rate = 1/ (bit duration) = 1 / (20 μs/10) = 1 / (2 μs) = 500 Kbps

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P3-8. A device is sending out data at the rate of 1000 bps.

a. How long does it take to send out 10 bits?

b. How long does it take to send out a single character (8 bits)?

c. How long does it take to send a file of 100,000 characters?

Given data:

Data rate=1000 bps

a)

The data rate takes to send out 10 bits then

$$= \frac{10}{1000}$$

=0.01s

Therefore, 0.01s time taken to send out 10 bits.

b)

The data rate takes to send out a single character (8 bits) then

$$= \frac{8}{1000}$$

$=8 \times 10^{-3}$s

=8 ms ($1ms=10^{-3}$)

Therefore,8ms time taken to send out a single character

c)

The data rate takes to send a file of 100,000 characters then

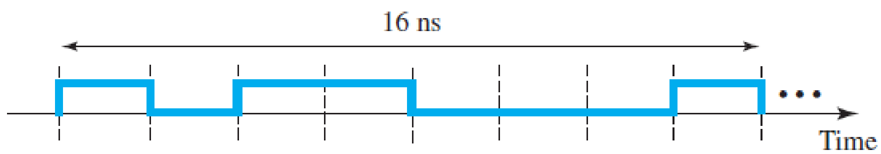
$= \frac{100,000 \times 8}{1000}$ (1 character = 8 bits)

=800s

Therefore,800s time taken to send a file of 100,000 characters

P3-9. What is the bit rate for the signal in Figure 3.35?

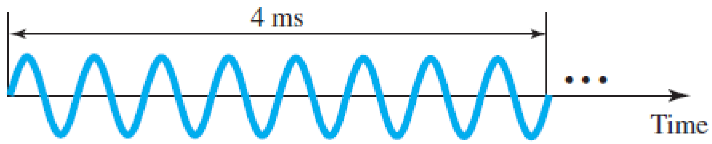
Figure 3.35 *Problem P3-9*



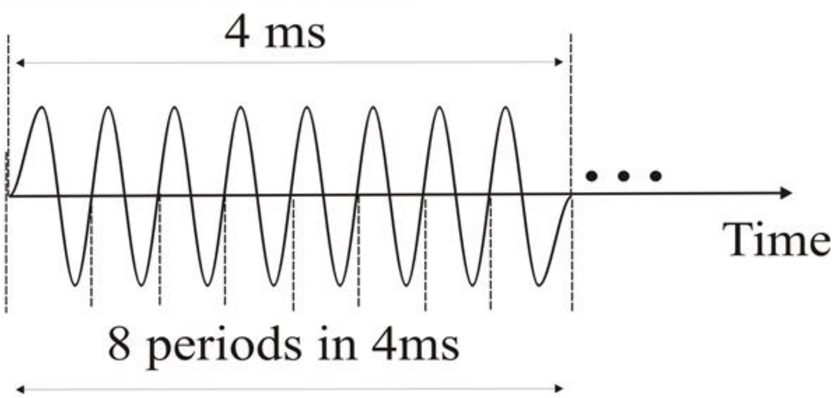
P3-9. There are 8 bits in 16 ns. Bit rate is $8 / (16 \times 10^{-9}) = 0.5 \times 10^{9} = 500$ Mbps

P3-10. What is the frequency of the signal in Figure 3.36?

Figure 3.36 *Problem P3-10*



The given signal has 8 periods (cycles) in 4ms.



Time (T) =Period= $\frac{4\text{ms}}{8}$ (Since, Period refers to the amount of time, in seconds, a signal needs to complete 1 cycle. Then here 8cycls in 4ms)

= $\frac{4\times10^{-3}\text{s}}{8}$ (1Milliseconds (ms) = 10^{-3}s)

= $\frac{1\times10^{-3}}{2}$

Now, Frequency (f) = $\frac{1}{T}$ Hz (Frequency and period are the inverse of each other)

= $\frac{1}{\frac{10^{-3}}{2}}$

= $\frac{2}{10^{-3}}$ Hz

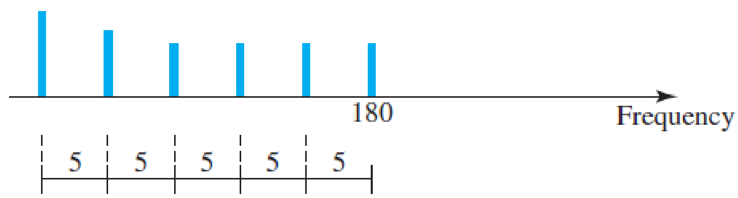
= 2×10^{3}Hz

= 2 KHz (1KHZ=10^{3}Hz)

Therefore, Frequency (f) = 2 KHz

P3-11. What is the bandwidth of the composite signal shown in Figure 3.37?

Figure 3.37 *Problem P3-11*

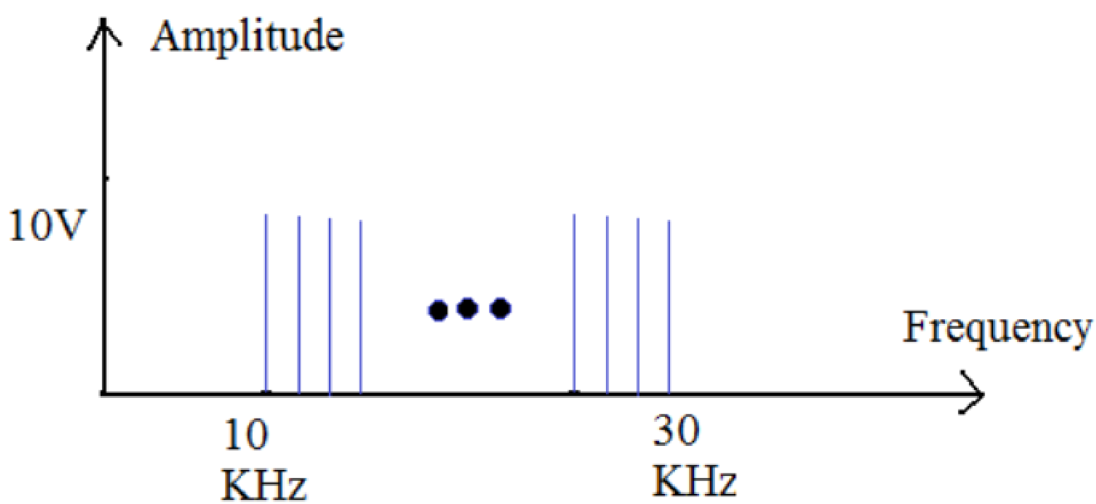


P3-11. The bandwidth is $5 \times 5 = 25$ Hz.

P3-12. A periodic composite signal contains frequencies from 10 to 30 KHz, each with an amplitude of 10 V. Draw the frequency spectrum.

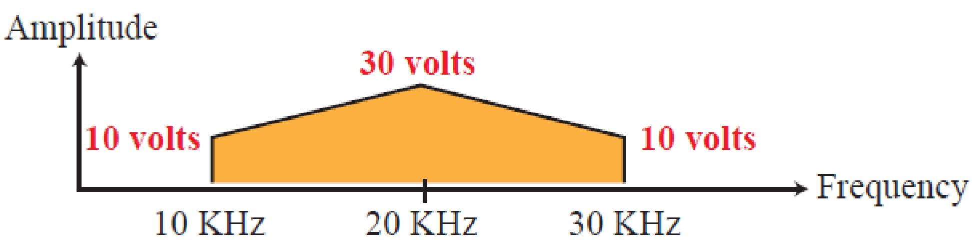
A periodic composite signal contains a series of signals with discrete frequencies. The discrete frequencies have integer values from 10, 11, 12, 13, … to 30KHz.

Then, the frequency spectrum is as follows:



P3-13. A nonperiodic composite signal contains frequencies from 10 to 30 KHz. The peak amplitude is 10 V for the lowest and the highest signals and is 30 V for the 20-KHz signal. Assuming that the amplitudes change gradually from the minimum to the maximum, draw the frequency spectrum.

P3-13. The signal is nonperiodic, so the frequency domain is made of a continuous spectrum of frequencies as shown below:



P3-14. A TV channel has a bandwidth of 6 MHz. If we send a digital signal using one channel, what are the data rates if we use one harmonic, three harmonics, and five harmonics?

Given data:

TV channel Band width (B) = 6 MHz

Using the first harmonic:

The minimum bandwidth, a rough approximation is

Band width (B) = $\frac{\text{(data rate (or) bit rate)}}{2}$

Data rate=2 $\times$B

=2 $\times$6

Therefore, data rate=12 Mbps

Using the first and three harmonics:

A better result can be achieved by using the first and the third harmonics with the required band width (B) =3 $\times \frac{\text{data rate}}{2}$

Data rate= $\frac{2\times B}{3}$

= $\frac{2\times 6}{3}$

Therefore, data rate =4Mbps

Using the first, third and fifth harmonics:

Still a better result can be achieved by using the first, third and fifth harmonics with

Band width (B) =5 $\times \frac{\text{data rate}}{2}$

Data rate= $\frac{2\times B}{5}$

= $\frac{2\times 6}{5}$

Therefore, the data rate=2.4 Mbps

P3-15. A signal travels from point A to point B. At point A, the signal power is 100 W. At point B, the power is 90 W. What is the attenuation in decibels?

P3-15. We can calculate the attenuation as shown below:

$$dB = 10 \log_{10} (90 / 100) = -0.46 \text{ dB}$$

P3-16. The attenuation of a signal is −10 dB. What is the final signal power if it was originally 5 W?

Given data:

The attenuation (dB) = -10dB

The signal power P1= 5W

The attenuation (db) = $10\log_{10}(\frac{P2}{P1})$

$-10=10\log_{10}(\frac{P2}{5})$

$\log_{10}(\frac{P2}{5}) = \frac{-10}{10}$

$\log_{10}(\frac{P2}{5}) = -1$

$\frac{P2}{5}=10^{-1}$ (Since, $\log_{10}(x)=a \Rightarrow x=10^{a}$)

$P2=5 \times 10^{-1}$

P2 =0.5W

Therefore, the final signal power= 0.5W

P3-17. A signal has passed through three cascaded amplifiers, each with a 4 dB gain. What is the total gain? How much is the signal amplified?

P3-17. The total gain is 3 × 4 = 12 dB. To find how much the signal is amplified, we can use the following formula:

$12 = 10 \log (P_2/P_1) \quad \rightarrow \quad \log (P_2/P_1) = 1.2 \quad \rightarrow \quad P_2/P_1 = 10^{1.2} = 15.85$

The signal is amplified almost 16 times.

P3-18. If the bandwidth of the channel is 5 Kbps, how long does it take to send a frame of 100,000 bits out of this device?

The bandwidth=5 Kbps

=5000 bps (Since, 1Kbps=1000bps)

It takes time to send a frame of 100,000 bits out of this device (T) = $\frac{100,000}{5000}$

=20s

Therefore, 20s time take to send a frame of 100,000 bits out of the bandwidth of the channel is 5 Kbps device.

P3-19. The light of the sun takes approximately eight minutes to reach the earth. What is the distance between the sun and the earth?

P3-19. $480 \text{ s} \times 300,000 \text{ km/s} = 144,000,000 \text{ km}$

P3-20. A signal has a wavelength of 1 μm in air. How far can the front of the wave travel during 1000 periods?

Given data:

Wave length= $1\mu m$

Time =1000 periods

The speed of sound in air is not important because already given wavelength $=1\mu m$.

Thus, in 1 period of the signal travels $1\mu m$.

So, 1000period of the signal travels $1\mu m \times 1000$ =1000 μm =1mm

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P3-21. A line has a signal-to-noise ratio of 1000 and a bandwidth of 4000 KHz. What is the maximum data rate supported by this line?

P3-21. We use the Shannon capacity $C = B \log_2 (1 + SNR)$

$$C = 4{,}000 \log_2 (1 + 1{,}000) \approx 40 \text{ Kbps}$$

P3-22. We measure the performance of a telephone line (4 KHz of bandwidth). When the signal is 10 V, the noise is 5 mV. What is the maximum data rate supported by this telephone line?

Signal-to-noise ratio (SNR) = $\frac{\text{average signal power}}{\text{average noise power}}$

$= \frac{10}{0.005}$

=2000V

The maximum data rate = $B\log_2(1+SNR)$

$= 4000\log_2(1+2000)$

$= 4000\left[\frac{\log_{10}^{2001}}{\log_{10}^{2}}\right] \qquad (\text{since}, \log_b^a = \frac{\log_x^a}{\log_x^b})$

$= 4000\left[\frac{3.3010}{0.3010}\right]$

$= 4000(10.9667)$

≈ 44867 bps

Therefore, the maximum data rate approximately 44867 bps supported.

Given data:

The signal power =10V

The noise power=5mV =>5 $\times 10^{-3}$V=0.005V

Band width B= 4 KHz =>4 $\times 10^{3}$ Hz=4000Hz

P3-23. A file contains 2 million bytes. How long does it take to download this file using a 56-Kbps channel? 1-Mbps channel?

P3-23. The file contains 2,000,000 × 8 = 16,000,000 bits.

a. With a 56-Kbps channel, it takes 16,000,000/56,000 = 289 s ≈ 5 minutes.

b. With a 1-Mbps channel, it takes 16,000,000/1,000,000 = 16 s.

P3-24. A computer monitor has a resolution of 1200 by 1000 pixels. If each pixel uses 1024 colors, how many bits are needed to send the complete contents of a screen?

Given data:

The resolution of computer monitor =1200 $\times$ 1000 pixels

The number of colors used by each pixel = 1024.so, this can be rewrite like $\log_2^{1024}$

The simplification is:

$\log_2^{1024} = \log_2^{(2^{10})}$

$= 10\log_2^2 \quad \left[\text{since}, \log_a^{x^y} = y\log_a^x\right]$

$\log_2^{1024} = 10 \quad \left[\log_2^2 = 1\right]$

Therefore, the total number of bits = 1200 $\times$ 1000 $\times$ 10

= 1, 20, 00, 000 bits

P3-25. A signal with 200 milliwatts power passes through 10 devices, each with an average noise of 2 microwatts. What is the SNR? What is the SNRdB?

P3-25. We have

$$\text{SNR} = (200\ \text{mW}) / (10 \times 2 \times \mu\text{W}) = 10{,}000$$
$$\text{SNR}_{\text{dB}} = 10 \log_{10} \text{SNR} = 10 \log_{10} 10000 = 40$$

P3-26. If the peak voltage value of a signal is 20 times the peak voltage value of the noise, what is the SNR? What is the SNR_{dB}?

Assume that noise voltage = Vs,

Signal voltage =20Vs because if the peak voltage value of a signal is 20 times the peak voltage value of noise.

Signal-to-noise ratio (SNR) = $\frac{\text{signal power}}{\text{noise power}}$

Here, power is proportional to square of voltage. This means that

Signal-to-noise ratio (SNR) $=\frac{\left[(\text{signal voltage})^2\right]}{\left[(\text{noise voltage})^2\right]}$

$=\left[\frac{(\text{signal voltage})}{(\text{noise voltage})}\right]^2$

$=\left[\frac{(20\times \text{Vs})}{(\text{Vs})}\right]^2$

$=(20)^2$

= 400

There fore, SNR =400 microwatts

Signal-to-noise ratio in decibels $(SNR_{dB})=10\log_{10}^{SNR}$

= $10\log_{10}^{400}$

=26.02 dB

There fore, $SNR_{dB}=26.02\,\text{dB}$

P3-27. What is the theoretical capacity of a channel in each of the following cases?

a. Bandwidth: 20 KHz $SNR_{dB} = 40$

b. Bandwidth: 200 KHz $SNR_{dB} = 4$

c. Bandwidth: 1 MHz $SNR_{dB} = 20$

P3-27. We can approximately calculate the capacity as

a. $C = B \times (SNR_{dB} /3) = 20 \text{ KHz} \times (40 /3) = 267 \text{ Kbps}$

b. $C = B \times (SNR_{dB} /3) = 200 \text{ KHz} \times (4 /3) = 267 \text{ Kbps}$

c. $C = B \times (SNR_{dB} /3) = 1 \text{ MHz} \times (20 /3) = 6.67 \text{ Mbps}$

P3-28. We need to upgrade a channel to a higher bandwidth. Answer the following questions:

a. How is the rate improved if we double the bandwidth?

b. How is the rate improved if we double the SNR?

To upgrade a channel to a higher bandwidth, consider the channel capacity (or) data rate.

The channel Capacity(C) = $B \times \log_2(1+\text{SNR})$

Where B= bandwidth

SNR=signal-to-noise ratio

a.

If the bandwidth is doubled, then the channel capacity (or) data rate is also doubled, because the channel capacity and bandwidth are directly proportional to each other. If one parameter is increased, another one is automatically increased.

Therefore the channel capacity or rate is as shown below for the doubled bandwidth:

Channel capacity(C) = $2B \times \log_2(1+\text{SNR})$

a.

If the bandwidth is doubled, then the channel capacity (or) data rate is also doubled, because the channel capacity and bandwidth are directly proportional to each other. If one parameter is increased, another one is automatically increased.

Therefore the channel capacity or rate is as shown below for the doubled bandwidth:

Channel capacity(C) = $2B \times \log_2(1+\text{SNR})$

b.

If we double the SNR then the channel capacity or rate increases slightly.

$$
\begin{aligned}
C_1 &= B \times \log_2(1+2\times\text{SNR}) \\
&\approx B \times \log_2(2\text{SNR}+1) \\
&\approx B\log_2 2 + B\log_2(\text{SNR}+1) \\
&\approx B + B\log_2(\text{SNR}+1) \text{ where } \log_2 2 \text{ is } 1 \\
&\approx B + C \text{ where } B\log_2(\text{SNR}+1) \text{ is } C \\
C_1 &\approx B + C
\end{aligned}
$$

Therefore, the rate increases slightly for the doubled SNR.

P3-29. We have a channel with 4 KHz bandwidth. If we want to send data at 100 Kbps, what is the minimum SNR_{dB}? What is the SNR?

P3-29. We can use the approximate formula

$$C = B \times (SNR_{dB} / 3) \text{ or } SNR_{dB} = (3 \times C) / B$$

We can say that the minimum of SNR_{dB} is

$$SNR_{dB} = 3 \times 100 \text{ Kbps} / 4 \text{ KHz} = 75$$

This means that the minimum

$$SNR = 10 \; SNR_{dB}/10 = 107.5 \approx 31{,}622{,}776$$

P3-30. What is the transmission time of a packet sent by a station if the length of the packet is 1 million bytes and the bandwidth of the channel is 200 Kbps?

Given data:

The Packet length= 1 million bytes

$=10^6$ bytes (Since, 1 million bytes = 10^6bytes)

$=10^6 \times 8$ bits (Since, 1 byte= 8 bits)

=8000000 bits

Band width=200 Kbps

$=200 \times 10^3$bps

The transmission time of a packet sent by station = $\frac{\text{Packet length}}{\text{Bandwidth}}$

$= \frac{8000000}{200 \times 10^3}$

=40s

Therefore, the transmission time=40s

P3-31. What is the length of a bit in a channel with a propagation speed of 2×10^8 m/s if the channel bandwidth is

a. 1 Mbps? **b.** 10 Mbps? **c.** 100 Mbps?

P3-31. The bit duration is the inverse of the bandwidth. We have

(bit length) = (propagation speed) ´ (bit duration)

a. Bit length = $(2 \times 10^8$ m$) \times [(1 / (1$ Mbps$)] = 200$ m. This means a bit occupies 200 meters on a transmission medium.

b. Bit length = $(2 \times 10^8$ m$) \times [(1 / (10$ Mbps$)] = 20$ m. This means a bit occupies 20 meters on a transmission medium.

c. Bit length = $(2 \times 10^8$ m$) \times [(1 / (100$ Mbps$)] = 2$ m. This means a bit occupies 2 meters on a transmission medium.

P3-32. How many bits can fit on a link with a 2 ms delay if the bandwidth of the link is

a. 1 Mbps? **b.** 10 Mbps? **c.** 100 Mbps?

The product of bandwidth and delay is the number of bits.

Thus, the number of bits=bandwidth $\times$ delay

a)

Delay=2ms

$=2 \times 10^{-3}$s

Bandwidth=1 Mbps

$=1 \times 10^{6}$bps

Therefore, the number of bits=$1 \times 10^{6} \times 2 \times 10^{-3}$

$=2 \times 10^{6-3}$

=2000 bits

b)

Delay=2ms

$=2 \times 10^{-3}$s

Bandwidth=10 Mbps

$=10 \times 10^{6}$bps

Therefore, the number of bits=$10 \times 10^{6} \times 2 \times 10^{-3}$

$=20 \times 10^{6-3}$

=20,000 bits

c)

Delay=2ms

$=2 \times 10^{-3}$s

Bandwidth=100 Mbps

$=100 \times 10^{6}$bps

Therefore, the number of bits=$100 \times 10^{6} \times 2 \times 10^{-3}$

$=200 \times 10^{6-3}$

=2, 00, 000 bits.

P3-33. What is the total delay (latency) for a frame of size 5 million bits that is being sent on a link with 10 routers each having a queuing time of 2 μs and a processing time of 1 μs. The length of the link is 2000 Km. The speed of light inside the link is 2×10^8 m/s. The link has a bandwidth of 5 Mbps. Which component of the total delay is dominant? Which one is negligible?

P3-33. We have Latency = $\text{Delay}_{pr} + \text{Delay}_{qu} + \text{Delay}_{tr} + \text{Delay}_{pg}$

$\text{Delay}_{pr} = 10 \times 1\ \mu s = 10\ \mu s$ **// Processing delay**

$\text{Delay}_{qu} = 10 \times 2\ \mu s = 20\ \mu s$ **// Queuing delay**

$\text{Delay}_{tr} = 5{,}000{,}000\ /\ (5\ \text{Mbps}) = 1\ s$ **// Transmission delay**

$\text{Delay}_{pg} = (2000\ \text{Km})\ /\ (2 \times 10^8) = 0.01\ s$ **// Propagation delay**

This means

Latency = $10\ \mu s + 20\ \mu s + 1 s + 0.01\ s \approx 1.01\ s$

The transmission time is dominant here because the packet size is huge.

Ch 4 Summary

- Digital-to-digital conversion involves three techniques: line coding, block coding, and scrambling.
- Line coding is the process of converting digital data to a digital signal.
- We can roughly divide line coding schemes into five broad categories: unipolar, polar, bipolar, multilevel, and multitransition.
- Block coding provides redundancy to ensure synchronization and inherent error detection.
- Block coding is normally referred to as mB/nB coding; it replaces each m-bit group with an n-bit group.
- Scrambling provides synchronization without increasing the number of bits.
- Two common scrambling techniques are B8ZS and HDB3.
- The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (PCM).
- The first step in PCM is sampling.
- The analog signal is sampled every T_s second, where T_s is the sample interval or period.
- The inverse of the sampling interval is called the *sampling rate* or *sampling frequency* and denoted by f_s, where $f_s = 1/T_s$.
- There are three sampling methods—ideal, natural, and flat-top.
- According to the *Nyquist theorem*, to reproduce the original analog signal, one necessary condition is that the *sampling rate* be at least twice the highest frequency in the original signal.
- Other sampling techniques have been developed to reduce the complexity of PCM.

- The simplest is delta modulation.

- PCM finds the value of the signal amplitude for each sample; DM finds the change from the previous sample.

- While there is only one way to send parallel data, there are three subclasses of serial transmission: asynchronous, synchronous, and isochronous.

- In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte.

- In synchronous transmission, we send bits one after another without start or stop bits or gaps.

- It is the responsibility of the receiver to group the bits.

- The isochronous mode provides synchronization for the entire stream of bits.

- In other words, it guarantees that the data arrive at a fixed rate.

4.5.2 Questions

Q4-1. List three techniques of digital-to-digital conversion.

Q4-1. The three different techniques described in this chapter are *line coding*, *block coding*, and *scrambling*.

Q4-2. Distinguish between a signal element and a data element.

The difference between data element and signal element is:

In data communications, our goal is to send data elements. A "data element" is the smallest entity that can represent a piece of information: this is the bit.

In digital data communications, a "signal element" carries data elements. A signal element is the shortest unit (time wise) of a digital signal.

In other words, data elements are what we need to send; signal elements are what we can send. Data elements are being carried; signal elements are the carriers.

Q4-3. Distinguish between data rate and signal rate.

Q4-3. The *data rate* defines the number of data elements (bits) sent in 1s. The unit is bits per second (bps). The *signal rate* is the number of signal elements sent in 1s. The unit is the baud.

Q4-4. Define baseline wandering and its effect on digital transmission.

In decoding a digital signal, the receiver calculates a running average of the received signal power. This average is called the “baseline”. The incoming signal power is evaluated against this baseline to determine the value of the data element. A long string 0s or 1s can cause a drift in the baseline (baseline wandering) and make it difficult for the receiver to decode correctly. A good line coding scheme needs to prevent baseline wandering.

Q4-5. Define a DC component and its effect on digital transmission.

Q4-5. When the voltage level in a digital signal is constant for a while, the spectrum creates very low frequencies, called *DC components*, that present problems for a system that cannot pass low frequencies.

Q4-6. Define the characteristics of a self-synchronizing signal.

A self-synchronizing digital signal includes timing information in the data being transmitted. This can be achieved if there are transitions in the signal that alert the receiver to the beginning, middle, or end of the pulse. If the receiver’s clock is out of synchronization, these points can reset the clock.

Q4-7. List five line coding schemes discussed in this book.

Q4-7. In this chapter, we introduced *unipolar*, *polar*, *bipolar*, *multilevel*, and *multi-transition* coding.

Q4-8. Define block coding and give its purpose.

Block coding:

• In block coding, message is divided into blocks of size k bits each known as data words.

• For each block, r redundant bits are added.

• After adding redundant bits block length is $n = k + r$

• The n bits block is known as code words.

Purpose of block coding:

• Block coding provides redundancy to ensure synchronization.

• Error detecting.

• In general, block coding changes a block of m bits into a block of n bits, where n is larger than m.

Q4-9. Define scrambling and give its purpose.

Q4-9. *Scrambling*, as discussed in this chapter, is a technique that substitutes long zero-level pulses with a combination of other levels without increasing the number of bits.

Q4-11. What are the differences between parallel and serial transmission?

Q4-11. In *parallel transmission* we send data several bits at a time. In serial transmission we send data one bit at a time.

Q4-12. List three different techniques in serial transmission and explain the differences.

The three different techniques in serial transmission are

1. Asynchronous

2. Synchronous

3 .Isochronous

In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte. In asynchronous transmission, the bytes inside each frame are also independent.

In synchronous transmission, we send bits one after another without start or stop bits or gaps. It is the responsibility of the receiver to group the bits. In synchronous transmission, the bytes inside each frame are synchronized.

The isochronous mode provides synchronized for the entire stream of bits must. In other words, it guarantees that the data arrive at a fixed rate.

4.5.3 Problems

P4-1. Calculate the value of the signal rate for each case in Figure 4.2 if the data rate is 1 Mbps and $c = 1/2$.

P4-1. We use the formula $s = c \times N \times (1/r)$ for each case. We let $c = 1/2$.

a. $r = 1 \rightarrow s = (1/2) \times (1\text{ Mbps}) \times 1/1 =$ 500 kbaud

b. $r = 1/2 \rightarrow s = (1/2) \times (1\text{ Mbps}) \times 1/(1/2) =$ 1 Mbaud

c. $r = 2 \rightarrow s = (1/2) \times (1\text{ Mbps}) \times 1/2 =$ 250 Kbaud

d. $r = 4/3 \rightarrow s = (1/2) \times (1\text{ Mbps}) \times 1/(4/3) =$ 375 Kbaud

---

P4-2. In a digital transmission, the sender clock is 0.2 percent faster than the receiver clock. How many extra bits per second does the sender send if the data rate is 1 Mbps?

If the data rate is 1 Mbps, then the sender clock is **0.2 percent faster** than the receiver clock

The extra bits per second does the sender send if the data rate 1Mbps

$$= \frac{0.2}{100} \times 1\text{Mbps}$$

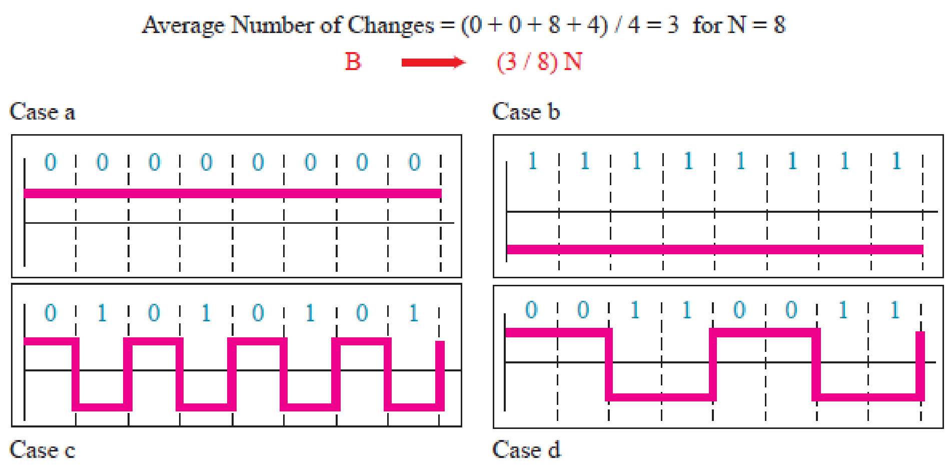
$$= \frac{0.2}{100} \times 1000{,}000\text{ bps}$$ (Since 1Mbps=1,000,000 bps)

= 2000bits

P4-3. Draw the graph of the NRZ-L scheme using each of the following data streams, assuming that the last signal level has been positive. From the graphs, guess the bandwidth for this scheme using the average number of changes in the signal level. Compare your guess with the corresponding entry in Table 4.1.

a. 00000000 **b.** 11111111 **c.** 01010101 **d.** 00110011

P4-3. See the following figure. Bandwidth is proportional to (3/8)N which is within the range in Table 4.1 (B = 0 to N) for the NRZ-L scheme.

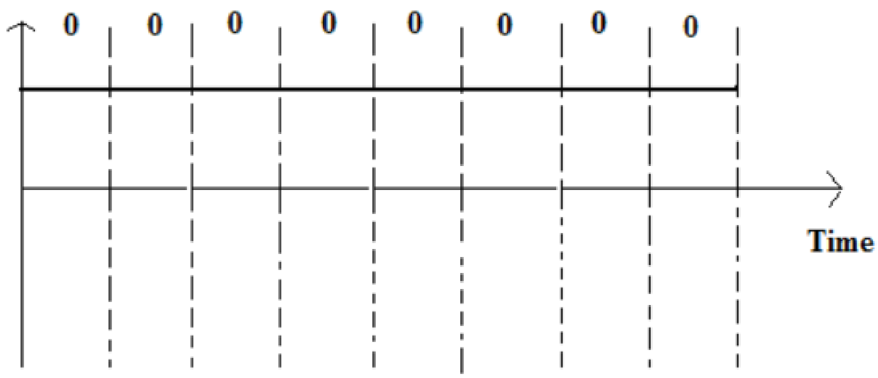


P4-4. Repeat Problem P4-3 for the NRZ-I scheme.

a)

Given data stream 00000000

NRZ-I:

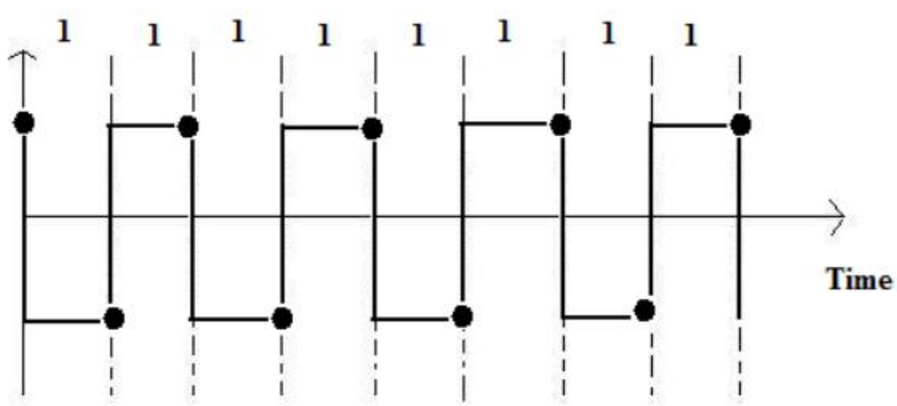


If there is a long sequences of 0s in NRZ-I, then the number of changes =0

b)

Given data stream 11111111

NRZ-I:



○ :-No inversion: next bit is 0

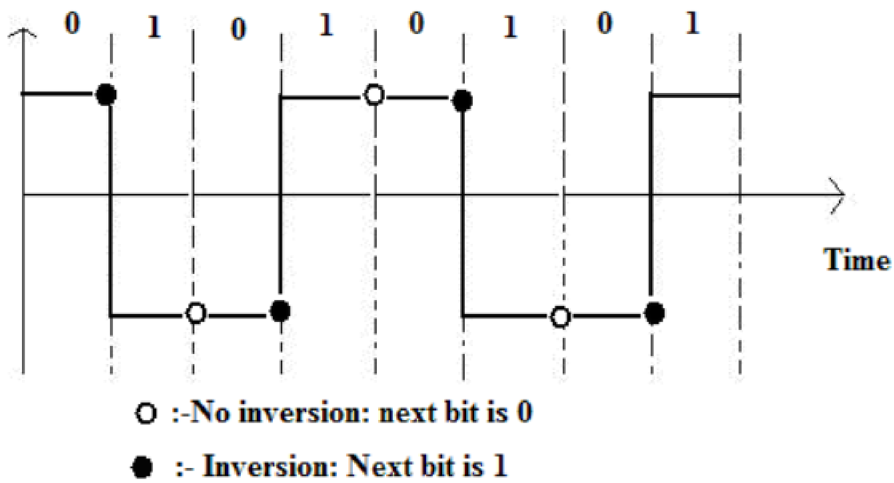
● :- Inversion: Next bit is 1

The number of changes=9

c)

Given data stream 01010101

NRZ-I:

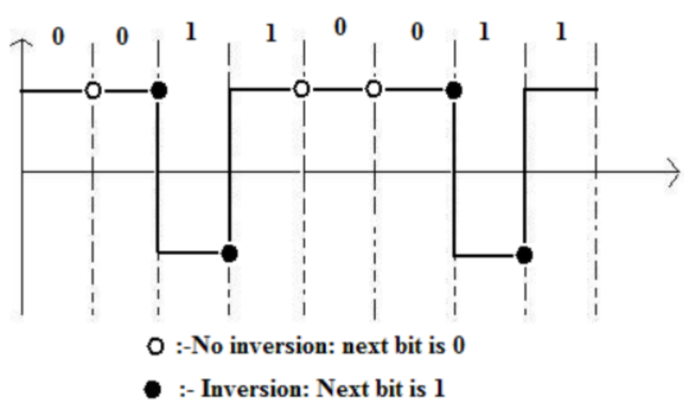


The number of changes =4

d)

Given data stream 00110011

NRZ-I:



The number of changes=4

Now, consider a, b, c, d then the average number of changes = $\frac{(0+9+4+4)}{4}$

=4.25

The Data rate (N) defines the number of data elements. So, here given 8 data stream. Thus, N=8.

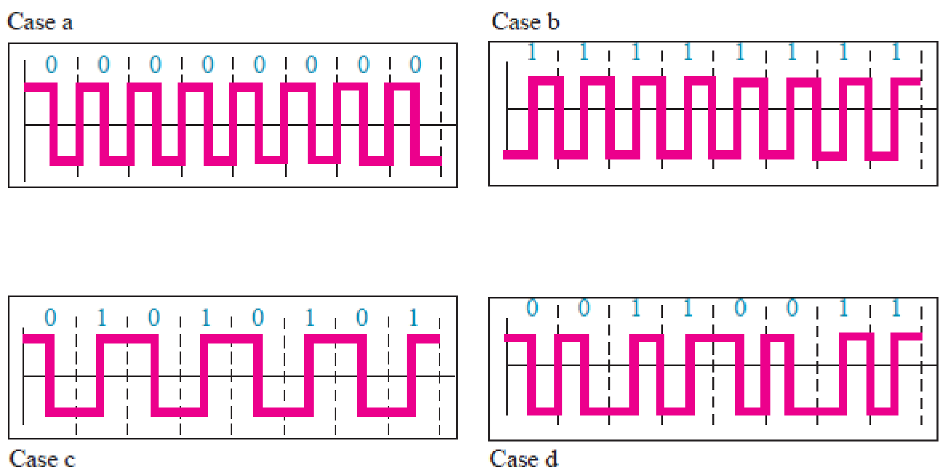
Therefore, the Band width B is proportional to $\left(\frac{4.25}{8}\right)N$ which is range B=0 to N.

P4-5. Repeat Problem P4-3 for the Manchester scheme.

P4-5. See the following figure. Bandwidth is proportional to (12.5 / 8)N which is within the range in Table 4.1 (B = N to B = 2N) for the Manchester scheme.

Average Number of Changes = (15 + 15+ 8 + 12) / 4 = 12.5 for N = 8

B ⟶ (12.5 / 8) N



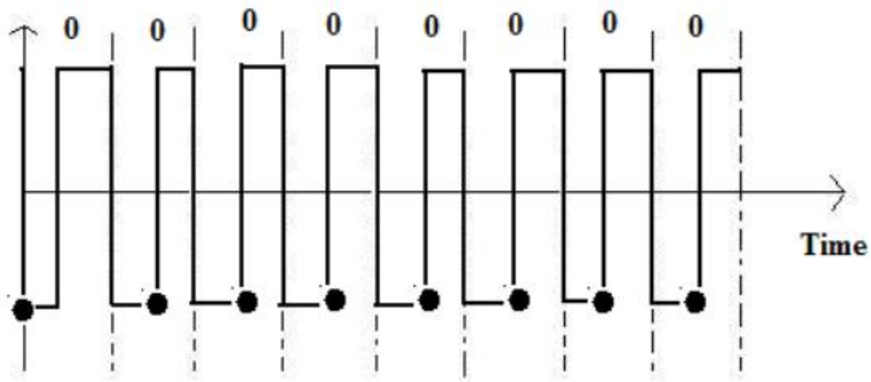
P4-6. Repeat Problem P4-3 for the differential Manchester scheme.

In Differential Manchester scheme, there is always a transition at the middle of the bit, but the bit values are determined at the beginning of the bit. If the next bit is 0, there is a transition; if the next bit is 1, there is none.

a)

Given data stream 00000000

Differential :



● :-No inversion: next bit is 0

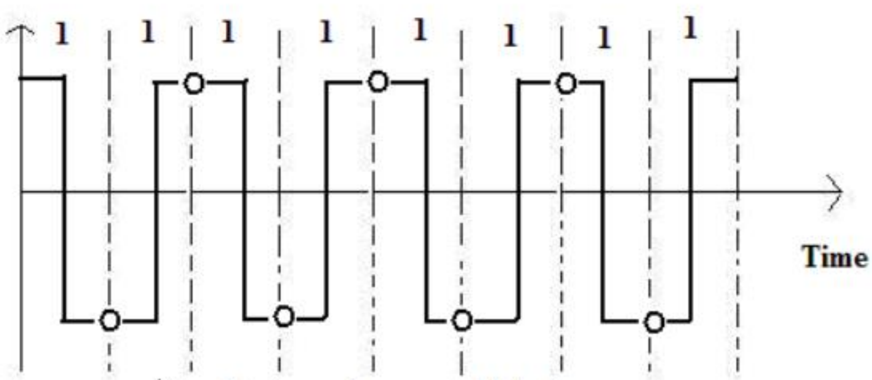
○ :- Inversion: Next bit is 1

The number of changes=16

b)

Given data stream 11111111

Differential :



● :-No inversion: next bit is 0

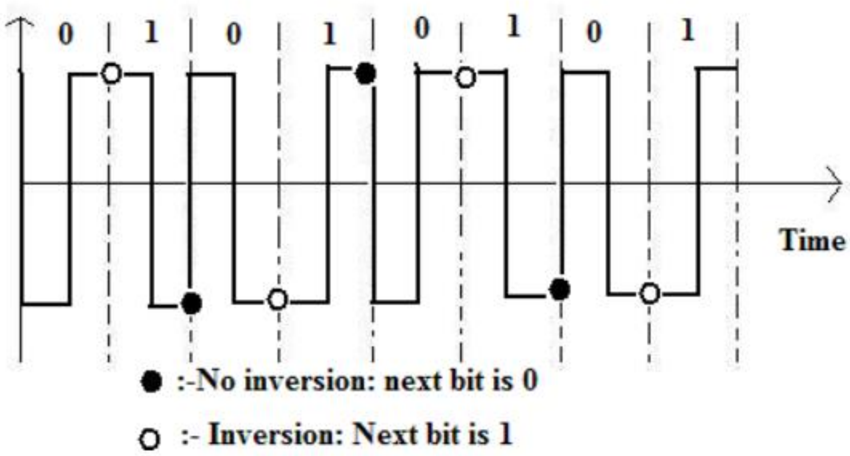
○ :- Inversion: Next bit is 1

The number of changes=8

c)

Given data stream 0101010101

Differential

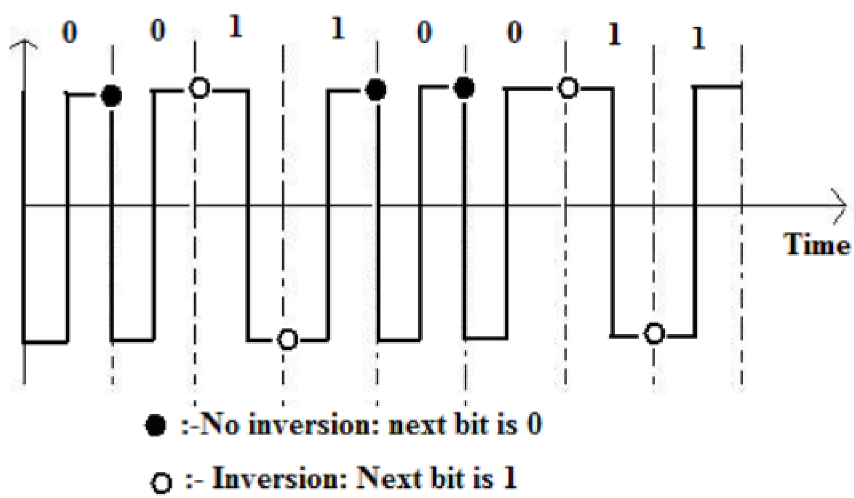


The number of changes=12

d)

Given data stream 00110011

Differential



The number of changes=12

Now, consider a, b, c, d then the average number of changes= $\frac{(16+8+12+12)}{4}$

=12

The Data rate (N) defines the number of data elements. So, here given 8 data stream. Thus, N=8.

Therefore, the Band width B is proportional to $\left(\frac{12}{8}\right)N$ which is range B=N to 2N.

P4-7. Repeat Problem P4-3 for the 2B1Q scheme, but use the following data streams.

a. 0000000000000000

b. 1111111111111111

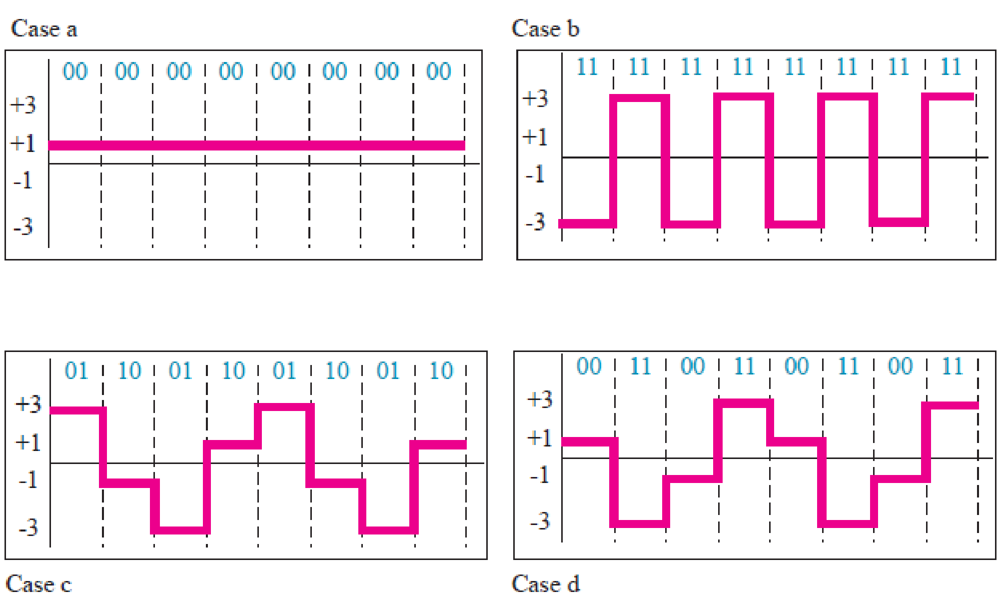
c. 0101010101010101

d. 0011001100110011

P4-7. See the following figure. B is proportional to (5.25 / 16)N which is inside range in Table 4.1 (B = 0 to N/2) for 2B/1Q.

Average Number of Changes = (0 + 7 + 7 + 7) / 4 = 5.25 for N = 16

B ⟶ (5.25 / 8) N



P4-8. Repeat Problem P4-3 for the MLT-3 scheme, but use the following data streams.

a. 00000000 **b.** 11111111 **c.** 01010101 **d.** 00011000

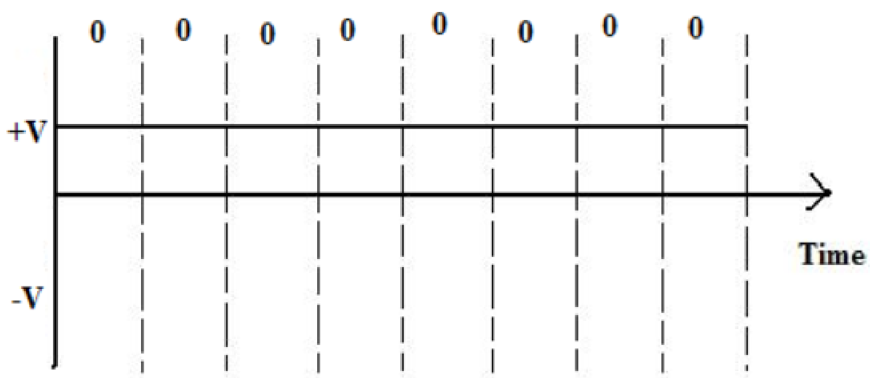
In MLT-3 (Multilane Transmission, three levels):

1) If the next bit is 0, there are no transitions.

2) If the next bit is 1 and the current level is not 0, the next level is 0

3) If the next bit is 1 and the current level is 0, the next level is the opposite of the last nonzero level.

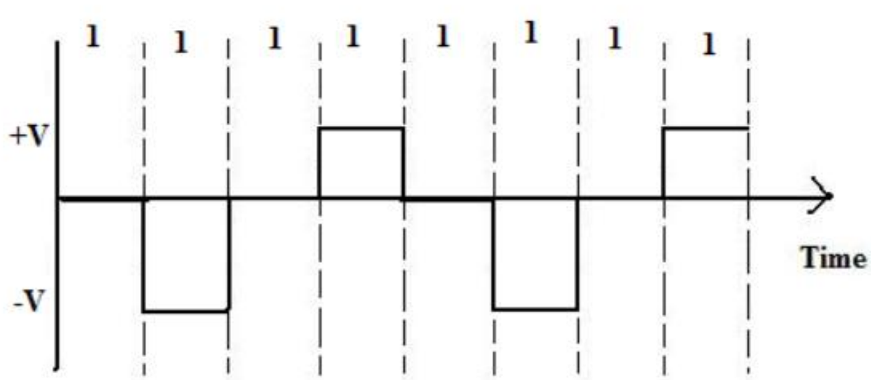
a)Given data stream 00000000



The number of changes=0

b)

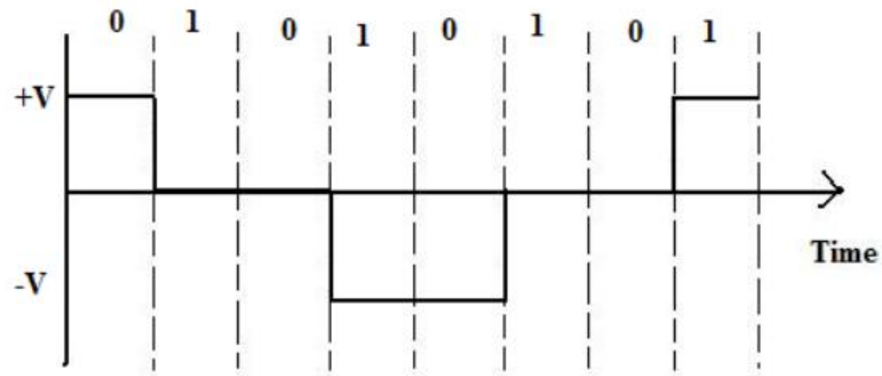
Given data stream 11111111



The number of changes=7

c)

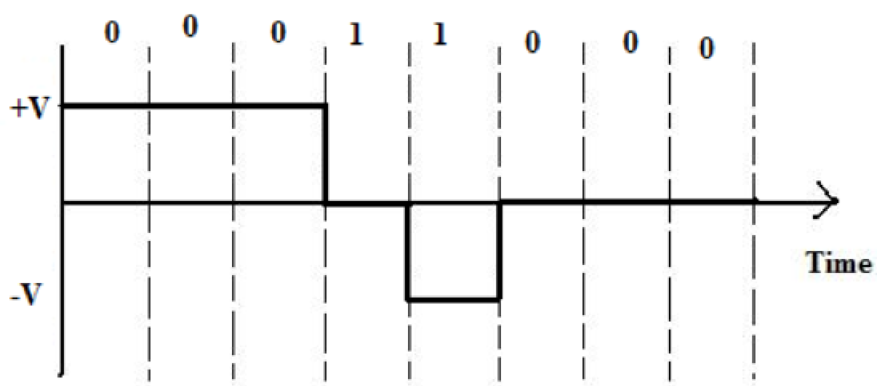
Given data stream 01010101



The number of changes=4

d)

Given data stream 00011000



The number of changes =3

Now, consider a, b, c, d then the average number of changes= $\frac{(0+7+4+3)}{4}$

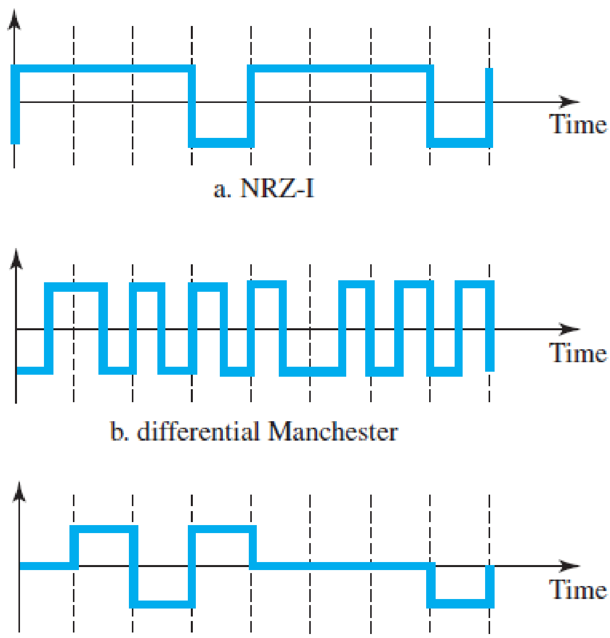
=3.5

The Data rate (N) defines the number of data elements. So, here given 8 data stream. Thus, N=8.

Therefore, the Band width B is proportional to $\left(\frac{3.5}{8}\right)\mathbf{N}$ which is range B=0 to N/2 for MLT-3..

P4-9. Find the 8-bit data stream for each case depicted in Figure 4.36.

Figure 4.36 *Problem P4-9*



P4-9. The data stream can be found as

a. NRZ-I: 10011001.

b. Differential Manchester: 11000100.

c. AMI: 01110001.

P4-10. An NRZ-I signal has a data rate of 100 Kbps. Using Figure 4.6, calculate the value of the normalized energy (P) for frequencies at 0 Hz, 50 KHz, and 100 KHz.

Given data:

Data rate N=100 Kbps

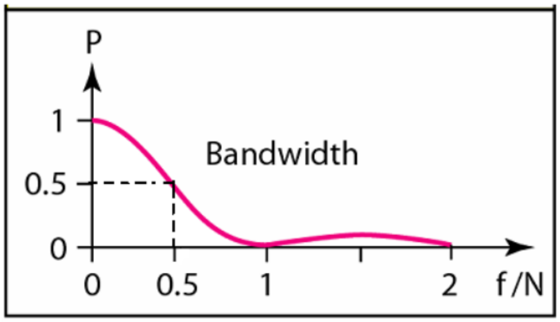
Now first calculate $\frac{f}{N}$ value then after find energy P value by using the given figure.

f=frequency

N=data rate

P=energy per Hz

The given figure is:



Case 1: f= 0Hz then $\frac{f}{N} = \frac{0}{100} = 0$

$\frac{f}{N} = 0$ .So, P=1

Case 2: f=50 KHz then $\frac{f}{N} = \frac{50}{100} = 0.5$

$\frac{f}{N} = 0.5$ So, P=0.5

Case 3: f=100 KHz then $\frac{f}{N} = \frac{100}{100} = 1$

$\frac{f}{N} = 1$. So, P=0

P4-11. A Manchester signal has a data rate of 100 Kbps. Using Figure 4.8, calculate the value of the normalized energy (P) for frequencies at 0 Hz, 50 KHz, 100 KHz.

P4-11. The data rate is 100 Kbps. For each case, we first need to calculate the value of f / N. We then use Figure 4.8 in the text to find P (energy per Hz). All calculations are approximations.

a. $f/N = 0/100 = 0 \rightarrow P = 0.0$

b. $f/N = 50/100 = 1/2 \rightarrow P = 0.3$

c. $f/N = 100/100 = 1 \rightarrow P = 0.4$

d. $f/N = 150/100 = 1.5 \rightarrow P = 0.0$

P4-12. The input stream to a 4B/5B block encoder is

0100 0000 0000 0000 0000 0001

Answer the following questions:

a. What is the output stream?

b. What is the length of the longest consecutive sequence of 0s in the input?

c. What is the length of the longest consecutive sequence of 0s in the output?

P4-13. How many invalid (unused) code sequences can we have in 5B/6B encoding? How many in 3B/4B encoding?

P4-13. In 5B/6B, we have $2^5 = 32$ data sequences and $2^6 = 64$ code sequences. The number of unused code sequences is $64 - 32 = 32$. In 3B/4B, we have $2^3 = 8$ data sequences and $2^4 = 16$ code sequences. The number of unused code sequences is $16 - 8 = 8$.

P4-14. What is the result of scrambling the sequence 11100000000000 using each of the following scrambling techniques? Assume that the last non-zero signal level has been positive.

a. B8ZS

b. HDB3 (The number of nonzero pulses is odd after the last substitution.)

In 5B/6B (5 Binary/ 6 Binary), A sequence of 5 bits can have only 32 (2^5=32) different combinations while a sequence of 6 bits can have 64(2^6=64) different combinations.

Therefore, the number of unused code sequence is 64-32=32.

In 3B/4B (3 Binary/ 4 Binary), A sequence of 3 bits can have only 8 (2^3=8) different combinations while a sequence of 4 bits can have 16(2^4=16) different combinations.

Therefore, the number of unused code sequence is 16-8=8.

P4-15. What is the Nyquist sampling rate for each of the following signals?

a. A low-pass signal with bandwidth of 200 KHz?

b. A band-pass signal with bandwidth of 200 KHz if the lowest frequency is 100 KHz?

a. In a low-pass signal, the minimum frequency 0. Therefore, we have

$$f_{max} = 0 + 200 = 200 \text{ KHz.} \quad \rightarrow \quad f_s = 2 \times 200{,}000 = 400{,}000 \text{ samples/s}$$

b. In a bandpass signal, the maximum frequency is equal to the minimum frequency plus the bandwidth. Therefore, we have

$$f_{max} = 100 + 200 = 300 \text{ KHz.} \quad \rightarrow \quad f_s = 2 \times 300{,}000 = 600{,}000 \text{ samples /s}$$

P4-16. We have sampled a low-pass signal with a bandwidth of 200 KHz using 1024 levels of quantization.

a. Calculate the bit rate of the digitized signal.

b. Calculate the SNR_{dB} for this signal.

c. Calculate the PCM bandwidth of this signal.

P4-17. What is the maximum data rate of a channel with a bandwidth of 200 KHz if we use four levels of digital signaling.

P4-17. The maximum data rate can be calculated as

$$N_{max} = 2 \times B \times n_b = 2 \times 200 \text{ KHz} \times \log_2 4 = 800 \text{ kbps}$$

P4-18. An analog signal has a bandwidth of 20 KHz. If we sample this signal and send it through a 30 Kbps channel, what is the SNR_{dB}?

Given data:

Band width(B) =20 KHz

Sample signal channel=30Kbps=300000bps

The bandwidth of a low-pass signal is between 0 and f, where f is the maximum frequency in the signal. Here minimum frequency 0 and maximum frequency 20 KHz.

Thus, sample this signal at 2 times the highest frequency.

So, the sample rate $f_s = 2 \times 20$

f_s **=40,000** samples per second.

The number of bits per level= $\frac{30000}{40000}$

n_b **=0.75** bits/sample

The formula:

Quantization Error (noise)

$SNR_{dB} = 6.02\, n_b + 1.76\, dB$

Here,n_b = number of bits per level=0.75 bits/sample

Therefore,

$SNR_{dB} = 6.02\, n_b + 1.76\, dB$

$= 6.02 \times 0.75 + 1.76$

$= 6.275$ dB

P4-19. We have a baseband channel with a 1-MHz bandwidth. What is the data rate for this channel if we use each of the following line coding schemes?

a. NRZ-L **b.** Manchester **c.** MLT-3 **d.** 2B1Q

P4-19. We can calculate the data rate for each scheme:

a. NRZ → $N = 2 \times B = 2 \times 1$ MHz = 2 Mbps
b. Manchester → $N = 1 \times B = 1 \times 1$ MHz = 1 Mbps
c. MLT-3 → $N = 3 \times B = 3 \times 1$ MHz = 3 Mbps
d. 2B1Q → $N = 4 \times B = 4 \times 1$ MHz = 4 Mbps

P4-20. We want to transmit 1000 characters with each character encoded as 8 bits.

a. Find the number of transmitted bits for synchronous transmission.

b. Find the number of transmitted bits for asynchronous transmission.

c. Find the redundancy percent in each case.

Given data:

The number of Transmit characters = 1000

The number of each character encoded bits =8

a)

Synchronous transmission:

In synchronous transmission send bits one after another without start or stop bits. It is the responsibility of the receiver to group the bits.

Therefore, the number of transmitted bits =1000 $\times$ 8

= 8000bits.

b)

Asynchronous transmission:

In asynchronous transmission, send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte. There may be a gap between each byte. So each character encoded as 8bits +2 bits (Take Start bit and Stop bit =2bits).

Therefore, the number of transmitted bits =1000 $\times$ 10

=10000bits.

c)

Case a:

The transmitted bits =8000 bits. These bits are correctly required bits. So, the redundancy percentage is 0%

Case b:

The transmitted bits=10000 bits. So, send 2000 extra bits (10000-8000=2000) for 8000 required bits.

So, the redundancy percentage $= \frac{2000}{8000} \times 100$

=25%

Ch 8 Summary

- A switched network consists of a series of interlinked nodes, called *switches*.
- Traditionally, three methods of switching have been important: circuit switching, packet switching, and message switching.
- We can divide today's networks into three broad categories: circuit-switched networks, packet-switched networks, and message-switched networks.
- Packet-switched networks can also be divided into two subcategories: virtual-circuit networks and datagram networks.
- A circuit-switched network is made of a set of switches connected by physical links, in which each link is divided into *n* channels.
- Circuit switching takes place at the physical layer.
- In circuit switching, the resources need to be reserved during the setup phase; the resources remain dedicated for the entire duration of the datatransfer phase until the teardown phase.
- In packet switching, there is no resource allocation for a packet.
- This means that there is no reserved bandwidth on the links, and there is no scheduled processing time for each packet.
- Resources are allocated on demand.
- In a datagram network, each packet is treated independently of all others.
- Packets in this approach are referred to as datagrams.
- There are no setup or teardown phases.

- A virtual-circuit network is a cross between a circuit-switched network and a datagram network.

- It has some characteristics of both.

- Circuit switching uses either of two technologies: the space-division switch or the timedivision switch.

- A switch in a packet-switched network has a different structure from a switch used in a circuit-switched network.

- We can say that a packet switch has four types of components: input ports, output ports, a routing processor, and switching fabric

8.6.2 Questions

Q8-1. Describe the need for switching and define a switch.

Q8-1. *Switching* provides a practical solution to the problem of connecting multiple devices in a network. It is more practical than using a bus topology; it is more efficient than using a star topology and a central hub. Switches are devices capable of creating temporary connections between two or more devices linked to the switch.

Q8-2. List the three traditional switching methods. Which are the most common today?

The three traditional switching methods are:

1) Circuit switching

2) Packet switching

3) Message switching

Circuit switching and packet switching are most commonly used today.

Q8-3. What are the two approaches to packet switching?

Q8-3. There are two approaches to packet switching: *datagram approach* and *virtual-circuit approach*.

Q8-4. Compare and contrast a circuit-switched network and a packet-switched network.

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A circuit-switched network is made of a set of switches connected by physical links, in which each link is divided into n channels.

In circuit switching, data are not packetized, the resources need to be reserved during the setup phase; the resources remain dedicated for the entire duration of data transfer phase until the teardown phase.

In packet switching, data are packetized, there is no resource allocation for a packet. This means that there is no reserved bandwidth on the links, and there is no scheduled processing time for each packet. Resources are allocated on demand.

Q8-5. What is the role of the address field in a packet traveling through a datagram network?

Q8-5. The address field defines the *end-to-end* (source to destination) addressing.

Q8-6. What is the role of the address field in a packet traveling through a virtual-circuit network?

The role of the address filed in a packet traveling through a virtual-circuit network is involved two types of addressing. There are global addressing and local (virtual-circuit identifier) addressing.

Global addressing:

A source to a destination needs to have a global address- an address that can be unique in the scope of the network or internationally if the network is part of an international network.

Local addressing:

The identifier that is actually used for data transfer is called the virtual-circuit identifier (VCI). A VCI, unlike a global address, is a small number that has only switch scope; it is used by a frame between two switches.

Q8-7. Compare space-division and time-division switches.

Q8-7. In a *space-division* switch, the path from one device to another is spatially separate from other paths. The inputs and the outputs are connected using a grid of electronic microswitches. In a *time-division* switch, the inputs are divided in time using TDM. A control unit sends the input to the correct output device.

Q8-8. What is TSI and what is its role in time-division switching?

Time-slot interchange (TSI) is most popular technology in a "time-division switching" uses time-division multiplexing (TDM) inside a switch. A TSI consists RAM (Random Access Memory) with several memory locations. The RAM fills up with incoming data from time slots in the order received. Slots are then sent out in an order based on the decisions of a control unit.

Q8-9. Compare and contrast the two major categories of circuit switches.

Q8-9. The two categories of circuit switches are *space-division* and *time-division*. In a space-division switch, the paths are separated from one another spatially. In a time-division switch, TDM technology is used to separate paths from one another.

Q8-10. List four major components of a packet switch and their functions.

The four major components of a packet switch are:

1. Input ports
2. Output ports
3. Routing processor
4. Switching fabric

The functions of these components are:

An 'input port' performs the physical and data link functions of the packet switch.

The 'output port' performs the same functions as the input port, but in the reverse order.

The 'routing processor' performs the functions of table lookup in the network layer.

The 'switching fabric' performs to move the packet from the input queue to the output queue.

8.6.3 Problems

P8-1. A path in a digital circuit-switched network has a data rate of 1 Mbps. The exchange of 1000 bits is required for the setup and teardown phases. The distance between two parties is 5000 km. Answer the following questions if the propagation speed is 2×10^8 m:

a. What is the total delay if 1000 bits of data are exchanged during the data-transfer phase?

b. What is the total delay if 100,000 bits of data are exchanged during the data-transfer phase?

c. What is the total delay if 1,000,000 bits of data are exchanged during the data-transfer phase?

d. Find the delay per 1000 bits of data for each of the above cases and compare them. What can you infer?

P8-1. We assume that the setup phase is a two-way communication and the teardown phase is a one-way communication. These two phases are common for all three cases. The delay for these two phases can be calculated as three propagation delays and three transmission delays or

$$3\,[(5000\text{ km})/\,(2 \times 10^8\text{ m/s})] + 3\,[(1000\text{ bits}/1\text{ Mbps})] = 75\text{ ms} + 3\text{ ms} = 78\text{ ms}$$

We assume that the data transfer is in one direction; the total delay is then

delay for setup and teardown + propagation delay + transmission delay

a. 78 + 25 + 1 = 104 ms

b. 78 + 25 + 100 = 203 ms

c. 78 + 25 + 1000 = 1103 ms

d. In case a, we have 104 ms. In case b we have 203/100 = 2.03 ms. In case c, we have 1103/1000 = 1.101 ms. The ratio for case c is the smallest because we use one setup and teardown phase to send more data.

P8-2. Five equal-size datagrams belonging to the same message leave for the destination one after another. However, they travel through different paths as shown in Table 8.1.

Table 8.1 *P8-2*

Datagram	*Path Length*	*Visited Switches*
1	3200 km	1, 3, 5
2	11,700 km	1, 2, 5
3	12,200 km	1, 2, 3, 5
4	10,200 km	1, 4, 5
5	10,700 km	1, 4, 3, 5

We assume that the delay for each switch (including waiting and processing) is 3, 10, 20, 7, and 20 ms respectively. Assuming that the propagation speed is 2×10^8 m, find the order the datagrams arrive at the destination and the delay for each. Ignore any other delays in transmission.

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Given data:

Five equal size data grams belonging to the same message leave for the destination one after another.

Assume the transmission time is negligible in this case. This means all data grams start at time 0.

Assume that **the delay for each** switch (including waiting and processing) **is 3, 10, 20, 7 and 20.**

Generally, calculate data grams arrival time by using formula t= $\frac{d}{s}$

Here, d=distance of path length and Propagation speed s=2 $\times 10^8$m/s.

But here, have to calculate the datagram arrive at the destination and the delay for each.

So, in below all data gram cases, the delay for each visited switches are corresponds to above delay for each switch values added to arrival time.

Datagram 1:

Given data:

Distance of path length (d) =3200 Km

Propagation speed (s) =2 $\times$ 108m/s

Visited switches = 1, 3, 5=>3, 20, 20

The datagram arrive at the destination and the delay for each $= \frac{3200\text{km}}{2\times10^8\text{m/s}} + (3+20+20)$

$=59 \times 10^{-3}$ s

$=59$ ms ($1\text{ms}=10^{-3}$s)

Datagram 2:

Given data:

Distance of path length (d) =11,700 Km

Propagation speed(s) =2 ×108m/s

Visited switches = 1, 2, 5=>3, 10, 20

The datagram arrive at the destination and the delay for each $= \frac{1700\text{km}}{2\times10^8\text{m/s}} + (3+10+20)$

$=91.5 \times 10^{-3}$s

$=91.5$ ms ($1\text{ms}=10^{-3}$s)

Datagram 3:

Given data:

Distance of path length (d) =12,200 Km

Propagation speed(s) =2 ×108m/s

Visited switches = 1, 2, 3, 5=>3, 10, 20, 20

The datagram arrive at the destination and the delay for each $=\frac{12200\text{km}}{2\times10^{8}\text{m/s}}+(3+10+20+20)$

=114 $\times 10^{-3}$ s

=114 ms (1ms=10^{-3} s)

Datagram 4:

Given data:

Distance of path length (d) =10,200 Km

Propagation speed(s) =2 $\times$108m/s

Visited switches = 1, 4, 5=>3, 7, 20

The data-grams arrive at the destination and the delay for each $=\frac{10200\text{km}}{2\times10^{8}\text{m/s}}+(3+7+20)$

=81 $\times 10^{-3}$ s

=81 ms (1ms=10^{-3} s)

Datagram 5:

Given data:

Distance of path length (d) =10,700 Km

Propagation speed (s) =2 $\times$108m/s

Visited switches = 1, 4, 3, 5=>3, 7, 20, 20

The data-grams arrive at the destination and the delay for each $=\frac{10700\text{km}}{2\times10^{8}\text{m/s}}+(3+7+20+20)$

=103.5 $\times$10-3s

=103.5 ms (1ms=10-3s)

Therefore, the order of the datagrams arrive at the destination is

Datagram 3-> Datagram 5-> Datagram 2-> Datagram 4-> Datagram

P8-3. Transmission of information in any network involves end-to-end addressing and sometimes local addressing (such as VCI). Table 8.2 shows the types of networks and the addressing mechanism used in each of them.

Table 8.2 *P8-3*

Network	*Setup*	*Data Transfer*	*Teardown*
Circuit-switched	End-to-end		End-to-end
Datagram		End-to-end	
Virtual-circuit	End-to-end	Local	End-to-end

Answer the following questions:

a. Why does a circuit-switched network need end-to-end addressing during the setup and teardown phases? Why are no addresses needed during the data transfer phase for this type of network?

b. Why does a datagram network need only end-to-end addressing during the data transfer phase, but no addressing during the setup and teardown phases?

c. Why does a virtual-circuit network need addresses during all three phases?

P8-3.

a. In a *circuit-switched* network, end-to-end addressing is needed during the setup and teardown phase to create a connection for the whole data transfer phase. After the connection is made, the data flow travels through the already-reserved resources. The switches remain connected for the entire duration of the data transfer; there is no need for further addressing.

b. In a *datagram network*, each packet is independent. The routing of a packet is done for each individual packet. Each packet, therefore, needs to carry an end-to-end address. There is no setup and teardown phases in a datagram network (connectionless transmission). The entries in the routing table are somehow permanent and made by other processes such as routing protocols.

c. In a *virtual-circuit* network, there is a need for end-to-end addressing during the setup and teardown phases to make the corresponding entry in the switching table. The entry is made for each request for connection. During the data transfer phase, each packet needs to carry a virtual-circuit identifier to show which virtual-circuit that particular packet follows.

P8-4. We mentioned that two types of networks, datagram and virtual-circuit, need a routing or switching table to find the output port from which the information belonging to a destination should be sent out, but a circuit-switched network has no need for such a table. Give the reason for this difference.

A datagram and virtual circuit network handles packetized data. In this type of network, each packet has a routing table which is based on the destination address. The routing tables are dynamic and are updated periodically. The destination addresses and the corresponding forwarding output ports are recorded in the tables of data-gram network and the virtual circuit identifier in the case of a virtual circuit network.

So, data-gram and virtual circuit need a routing or switching table to find the output port from which the information belonging to a destination should be sent out.

But, a circuit switched network, data are not packetized and no routing information is carried with the data in which each entry is created when the setup phase is completed and deleted when the teardown phase is over. So, a circuit switched network has no need for such a table.

P8-5. An entry in the switching table of a virtual-circuit network is normally created during the setup phase and deleted during the teardown phase. In other words, the entries in this type of network reflect the current connections, the activity in the network. In contrast, the entries in a routing table of a datagram network do not depend on the current connections; they show the configuration of the network and how any packet should be routed to a final destination. The entries may remain the same even if there is no activity in the network. The routing tables, however, are updated if there are changes in the network. Can you explain the reason for these two different characteristics? Can we say that

a virtual-circuit is a *connection-oriented* network and a datagram network is a *connectionless* network because of the above characteristics?

P8-5. In *circuit-switched* and *virtual-circuit* networks, we are dealing with connections. A connection needs to be made before the data transfer can take place. In the case of a circuit-switched network, a physical connection is established during the setup phase and the is broken during the teardown phase. In the case of a virtual-circuit network, a virtual connection is made during setup and is broken during the teardown phase; the connection is virtual, because it is an entry in the table. These two types of networks are considered *connection-oriented*. In the case of a *datagram* network no connection is made. Any time a switch in this type of network receives a packet, it consults its table for routing information. This type of network is considered a *connectionless* network.

P8-6. The minimum number of columns in a datagram network is two; the minimu number of columns in a virtual-circuit network is four. Can you explain tl reason? Is the difference related to the type of addresses carried in the packe of each network?

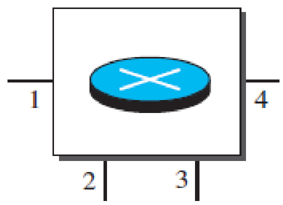
In a datagram network, each switch (or packet switch) has a routing table which is based on the destination address. The minimum number of columns in a datagram networks is two; there are **destination address** and the corresponding forwarding **output port**s are recorded in the tables. As in a datagram network, data are packetized and each packet carries an address in the header.

The routing table in a virtual-circuit network is based on the virtual circuit identifier (VCI), which is local jurisdiction (if defines what should be the next switch and the channel on which the packet is being carried). So, the two different input or output ports may use the same virtual circuit number. Therefore, the minimum number of columns is required four in a virtual-circuit network. There are input port, input virtual circuit number, output port and output virtual circuit number.

P8-7. Figure 8.27 shows a switch (router) in a datagram network.

Figure 8.27 *Problem P8-7*

Destination address	Output port
1233	3
1456	2
3255	1
4470	4
7176	2
8766	3
9144	2



Find the output port for packets with the following destination addresses:

a. Packet 1: 7176

b. Packet 2: 1233

c. Packet 3: 8766

d. Packet 4: 9144

P8-7.

a. Packet 1: 2

b. Packet 2: 3

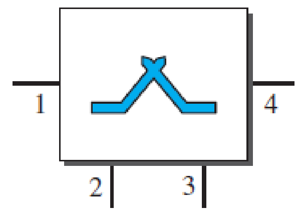
c. Packet 3: 3

d. Packet 4: 2

P8-8. Figure 8.28 shows a switch in a virtual-circuit network.

Figure 8.28 *Problem P8-8*

Incoming		Outgoing	
Port	VCI	Port	VCI
1	14	3	22
2	71	4	41
2	92	1	45
3	58	2	43
3	78	2	70
4	56	3	11



Find the output port and the output VCI for packets with the following input port and input VCI addresses:

a. Packet 1: 3, 78

b. Packet 2: 2, 92

c. Packet 3: 4, 56

d. Packet 4: 2, 71

Given data:

A switch in a virtual circuit network:

Incoming		Outgoing	
Port	VCI	Port	VCI
1	14	3	22
2	71	4	41
2	92	1	45
3	58	2	43
3	78	2	70
4	56	3	11

The output port and the output VCI for packets with the following input port and input VCI address follows from above table.

	Incoming		Outgoing	
	Port	VCI	Port	VCI
Packet1	3	78	2	70
Packet2	2	92	1	45
Packet3	4	56	3	11
Packet4	2	71	4	41

P8-9. Answer the following questions:

a. Can a routing table in a datagram network have two entries with the same destination address? Explain.

b. Can a switching table in a virtual-circuit network have two entries with the same input port number? With the same output port number? With the same incoming VCIs? With the same outgoing VCIs? With the same incoming values (port, VCI)? With the same outgoing values (port, VCI)?

P8-9.

a. In a *datagram* network, the destination addresses are unique. They cannot be duplicated in the routing table.

b. In a *virtual-circuit* network, the VCIs are local. A VCI is unique only in relationship to a port. In other words, the (port, VCI) combination is unique. This means that we can have two entries with the same input or output ports. We can have two entries with the same VCIs. However, we cannot have two entries with the same (port, VCI) pair.

P8-10. It is obvious that a router or a switch needs to search to find information in the corresponding table. The searching in a routing table for a datagram network is based on the destination address; the searching in a switching table in a virtual-circuit network is based on the combination of incoming port and incoming VCI. Explain the reason and define how these tables must be ordered (sorted) based on these values.

Since the datagram network is a connectionless network, it simply needs the destination address for routing. It will not have any ports and connections. So the searching in a routing table for a datagram network is based on the destination address.

Whereas, a virtual circuit network is connection oriented and virtual connections are established for routing. So in order to route packets it need to receive a packet from one port with a VCI (Virtual-Circuit-Index) send through another port with some other VCI. Hence the searching in a switching table in a virtual circuit network is based on the combination of incoming port and incoming VCI.

As the searching in a routing table for a datagram network is done based on the destination address, these tables must be sorted based on the destination address so that the searching becomes easy.

And as the searching in a switching table in a virtual circuit network is based on the incoming port and incoming VCI, these tables must be sorted based on these values. Since each port number may have many virtual circuits, it will be easy to sort the table first based on the port numbers and then sort each port according to their input VCIs.

P8-11. Consider an $n \times k$ crossbar switch with n inputs and k outputs.

a. Can we say that the switch acts as a multiplexer if $n > k$?

b. Can we say that the switch acts as a demultiplexer if $n < k$?

P8-11.

a. If $n > k$, an $n \times k$ crossbar is like a *multiplexer* that combines n inputs into k outputs. However, we need to know that a regular multiplexer discussed in Chapter 6 is $n \times 1$.

b. If $n < k$, an $n \times k$ crossbar is like a *demultiplexer* that divides n inputs into k outputs. However, we need to know that a regular demultiplexer discussed in Chapter 6 is $1 \times n$.

P8-12. We need a three-stage space-division switch with $N = 100$. We use 10 crossbars at the first and third stages and 4 crossbars at the middle stage.

a. Draw the configuration diagram.

b. Calculate the total number of crosspoints.

c. Find the possible number of simultaneous connections.

d. Find the possible number of simultaneous connections if we use a single crossbar (100×100).

e. Find the blocking factor, the ratio of the number of connections in part c and in part d.

a)

Given data:

N=100

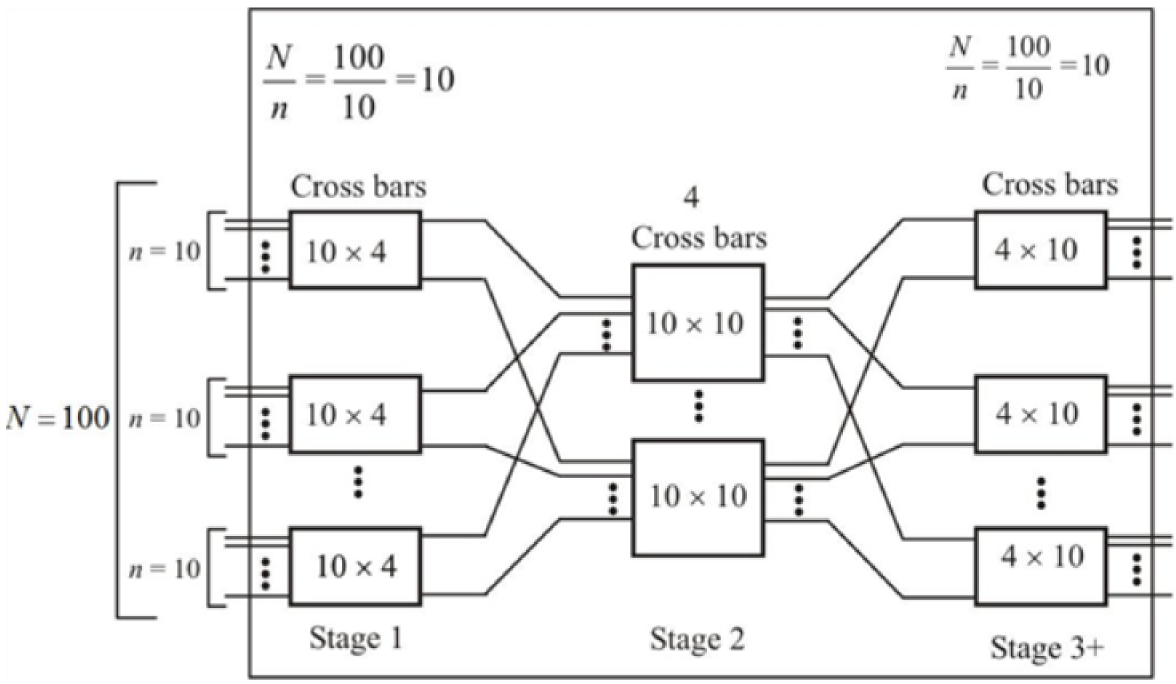
n=10

k=4

In the first stage we have N/n=100/10=10 crossbars, each of size is 10 $\times$4.

In the second stage, we have 4 cross bars, each of size is 10 $\times$10.

In the third stage, we have 10 cross bars, each of size is 4 $\times$10.



b)

In a three-state switch, the total number of cross-points = $2kN+k\left(\frac{N}{n}\right)^2$

= $2\times4\times100+4\left(\frac{100}{10}\right)^2$

= $800+4(10)^2$

=1200

c)

In first stage, each crossbar have 4 simultaneous connections are possible.

So, the possible number of simultaneous connections=10 $\times$4

=40

d)

If we use one single cross bar (100 $\times$ 100) then all input lines can have a connection at the same time. This means the possible number of simultaneous connections is 100.

e)

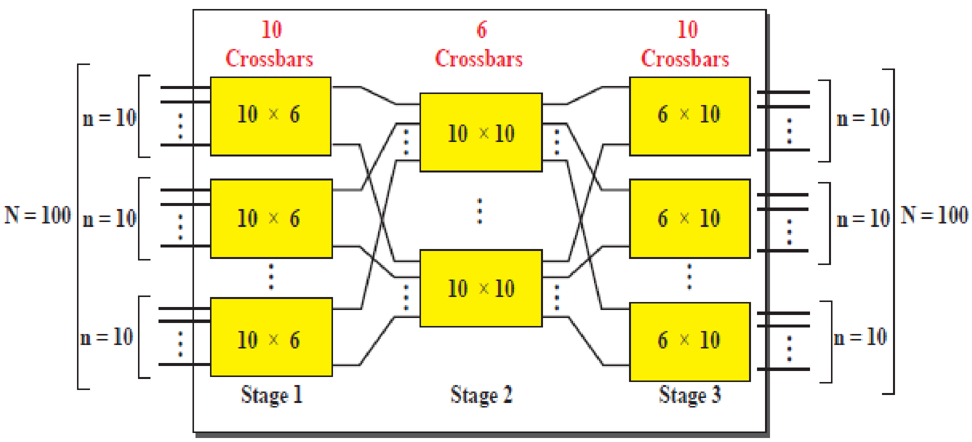
The ratio of the number of connections in above (c) and (d) = 40:100

Therefore, the blocking factor= $\frac{40}{100}$ or 40%

P8-13. Repeat Problem 8-12 if we use 6 crossbars at the middle stage.

P8-13.

a. See the following figure.



b. The total number of crosspoints are

Number of crosspoints = $10\,(10 \times 6) + 6\,(10 \times 10) + 10\,(6 \times 10) = 1800$

c. Only six simultaneous connections are possible for each crossbar at the first stage. This means that the total number of simultaneous connections is 60.

d. If we use one crossbar (100×100), all input lines can have a connection at the same time, which means 100 simultaneous connections.

e. The blocking factor is 60/100 or 60 percent.

P8-14. Redesign the configuration of Problem 8-12 using the Clos criteria.

a)

Given data:

N=100

k=4

According to Clos criterion:

$$n=\left(\frac{N}{2}\right)^{\frac{1}{2}}$$

$$=\left(\frac{100}{2}\right)^{\frac{1}{2}}$$

=7.07

n $\approx$ 8 (approximately)

k>2n-1

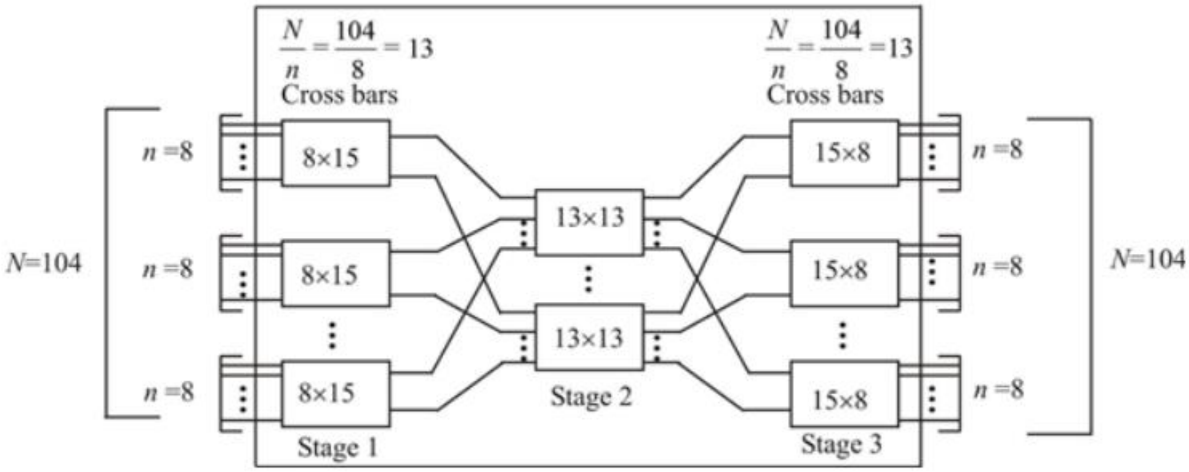
k=2(8)-1

=15

In the first stage we have N/n= $\frac{100}{8} \approx 13$ crossbars, each of size 8 $\times$ 15 (n $\times$ k).

In the second stage, we have 4 cross bars, each of size 13 × 13 (N/n × N/n).

In the third stage, we have 10 cross bars, each of size 15 × 8 (k × n).



b)

In a Clos criterion, the total number of cross-points is $\geq 4N[(2N)^{1/2} - 1]$

$= 4 \times 100\left[(2 \times 100)^{1/2} - 1\right]$

≈ 5257

c)

In first stage, each crossbar have 8 simultaneous connections are possible.

So, the possible number of simultaneous connections=10 × 8

=80

d)

If we use one single cross bar (100 × 100) then all input lines can have a connection at the same time. This means the possible number of simultaneous connections is 100.

e)

The ratio of the number of connections in above (c) and (d) = 80:100

Therefore, the blocking factor= $\frac{80}{100}$ or 80%

P8-15. We need to have a space-division switch with 1000 inputs and outputs. What is the total number of crosspoints in each of the following cases?

a. Using a single crossbar.

b. Using a multi-stage switch based on the Clos criteria.

P8-15.

a. Total crosspoints = $N^2 = 1000^2 = 1{,}000{,}000$

b. Total crosspoints $\geq 4N[(2N)^{1/2} - 1] \geq 174{,}886$. With less than 200,000 crosspoints we can design a three-stage switch. We can use $n = (N/2)^{1/2} = 23$ and choose $k = 45$. The total number of crosspoints is 178,200.