



Chapter 02

The Physical Layer

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Key Points

物理层	功能与作用	熟练掌握
	接口特性	熟练掌握
通信基础	基本概念	熟练掌握
	奈奎斯特定理, 香农定理	熟练掌握
	编码与调制	熟练掌握
传输介质	种类, 特点	掌握
物理层设备	中继器	掌握
	集线器	掌握
多路复用		熟练掌握

Chapter 2: Roadmap

2.1 Physical layer Introduction

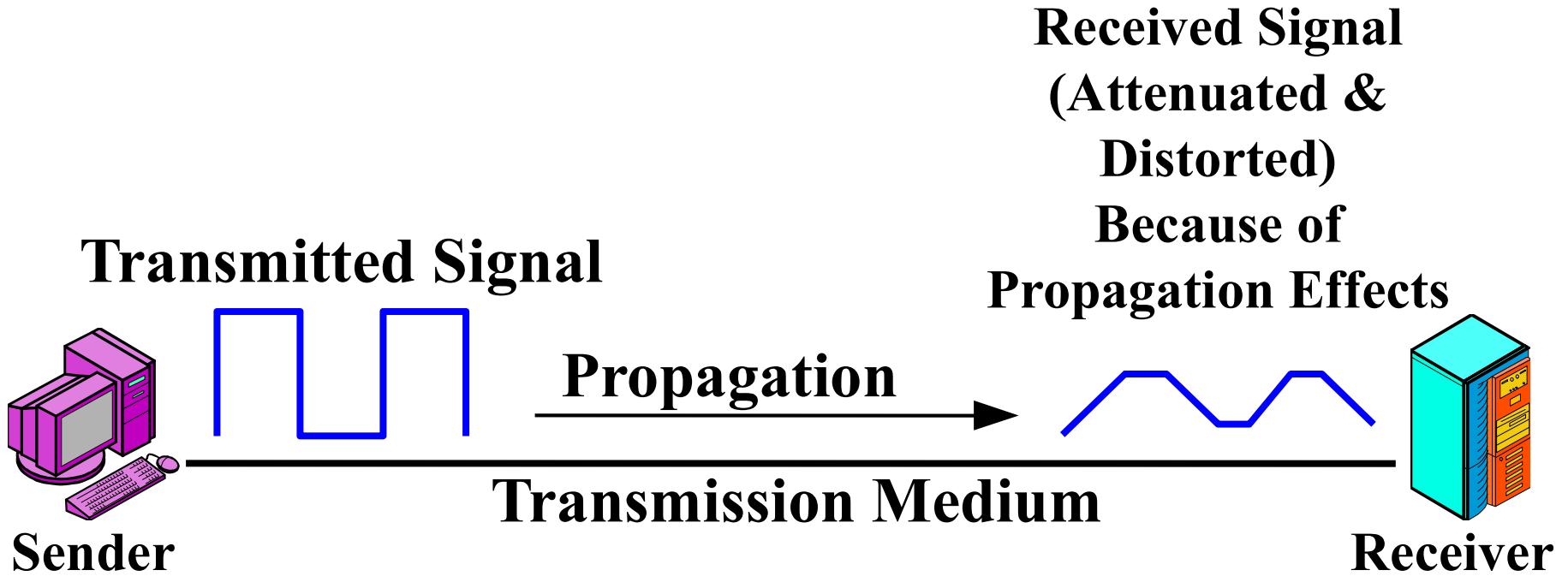
2.2 Basis for Data communication

2.3 Transmission Media

2.4 Modulation and Data Encoding

2.5 Multiplexing

Physical Layer



Physical Layer

- The lowest layer in any network architecture model
- Concerned with the transparent transmission of “**raw**” bits across a communications medium
- Deals with the **physical characteristics** (**mechanical, electrical, functional, procedural**) of data transmission and communication.

Physical Layer

- **Responsible for:**
 - **providing basic signaling (control, data)**
 - **signal modulation**
 - **encoding/decoding**
 - **activate/deactivate physical medium (PM)**
 - **bit-timing (clocking)**
 - **mapping between different formats**

Chapter 2: Roadmap

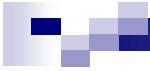
2.1 Physical layer Introduction

2.2 Basis for Data communication

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The Basis for Data Communication

- **What is communication?**

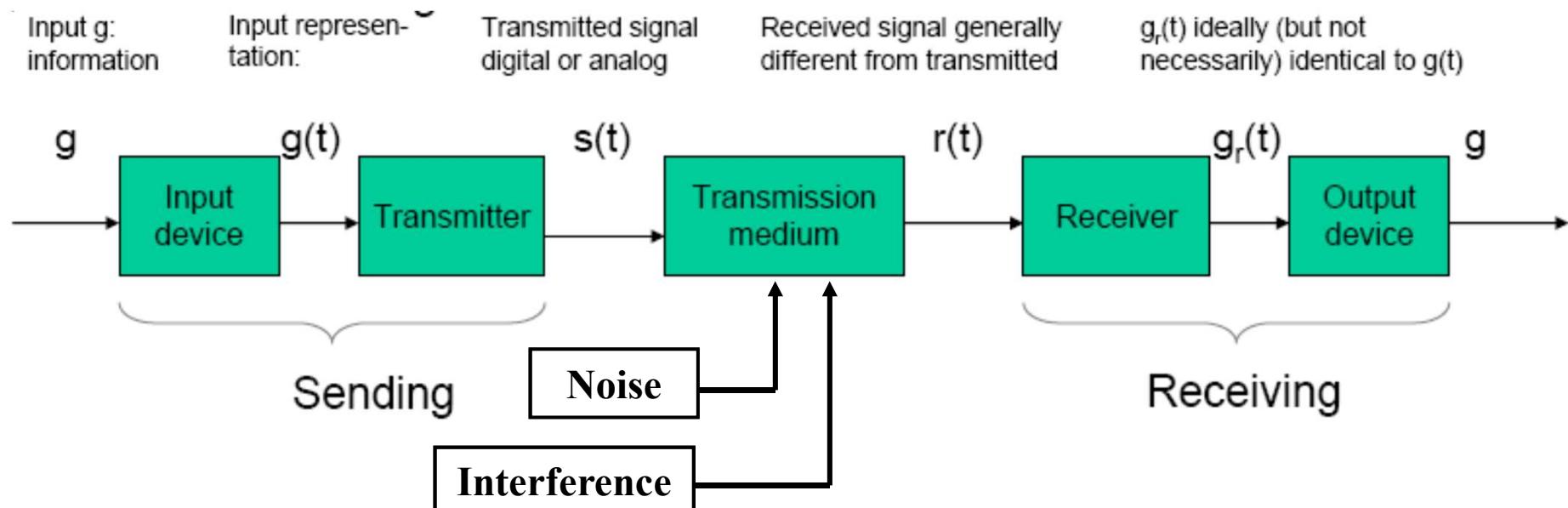
The transmission of information from one point to another through a succession of certain processes.

The Basis for Data Communication

■ Important transformations:

□ Source coding

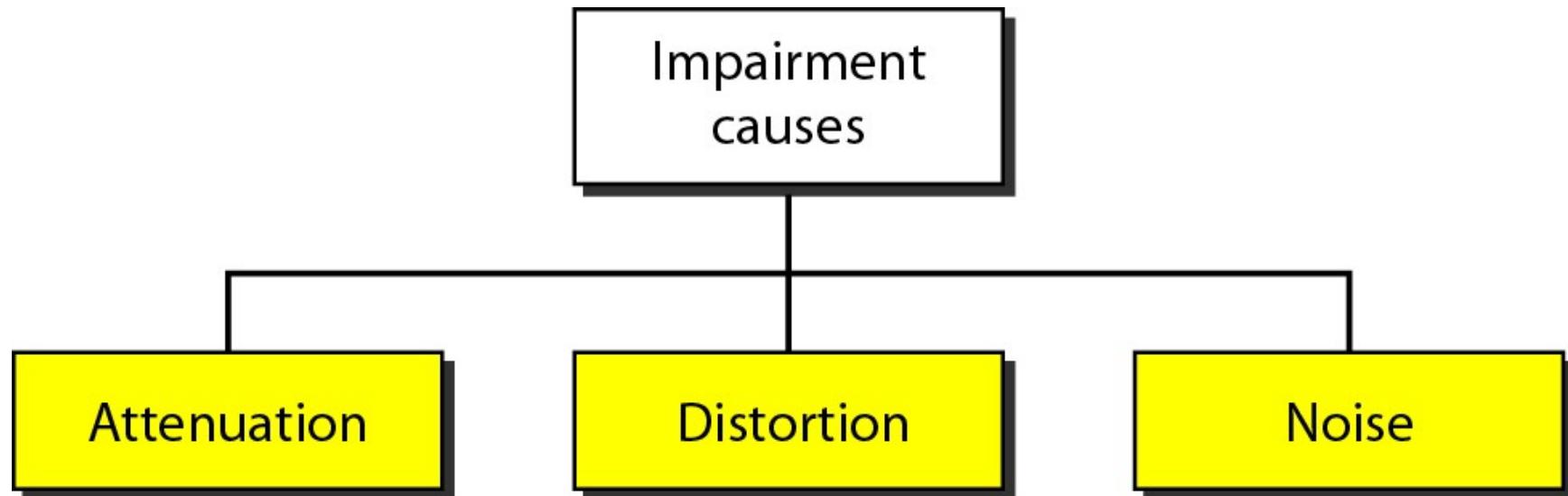
□ Channel coding



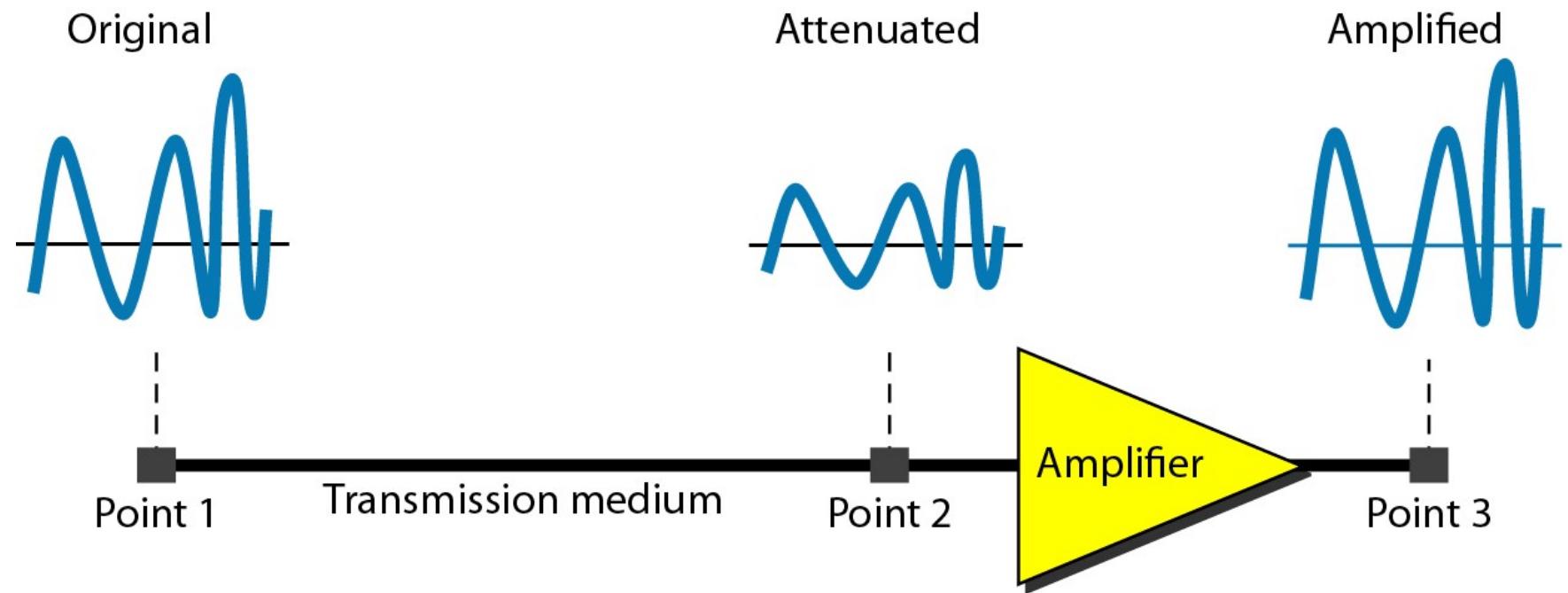
The basic model of a data comm. system

Transmission Impairment

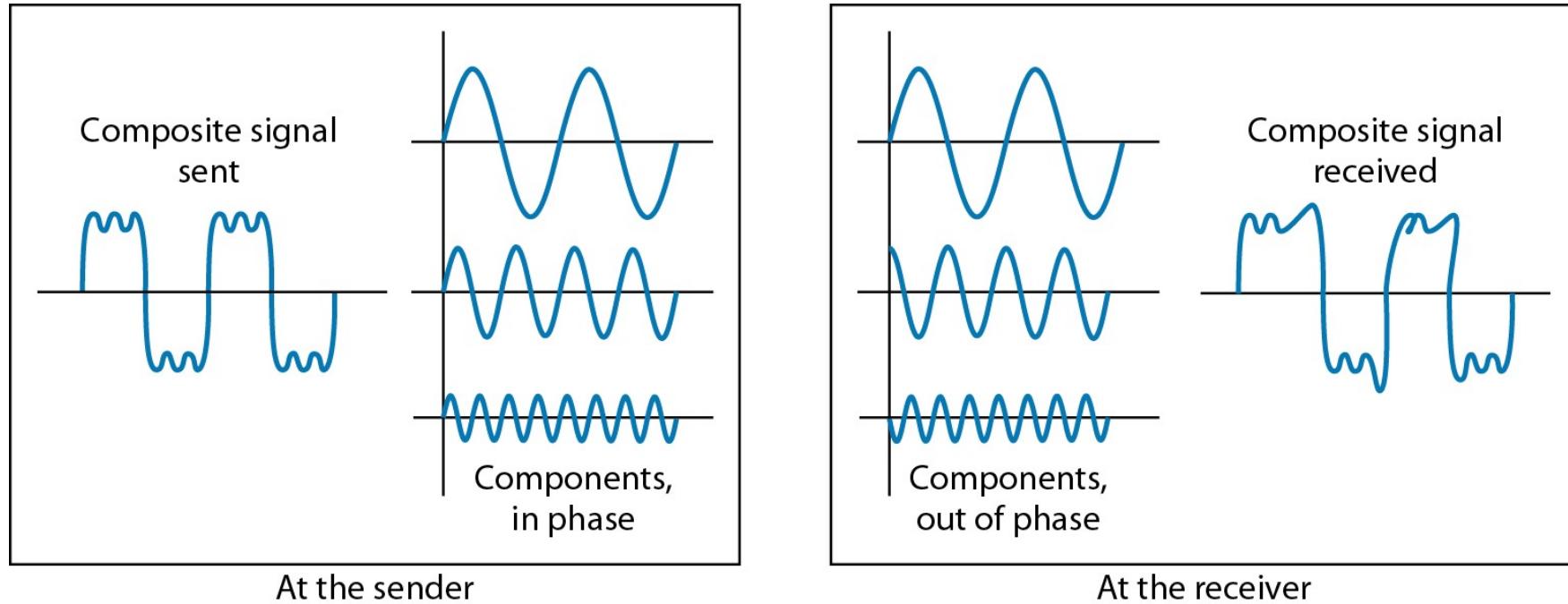
- Signals travel through transmission media, which are not perfect.
- Three causes of impairment are **attenuation, distortion, and noise.**



Attenuation: loss of energy

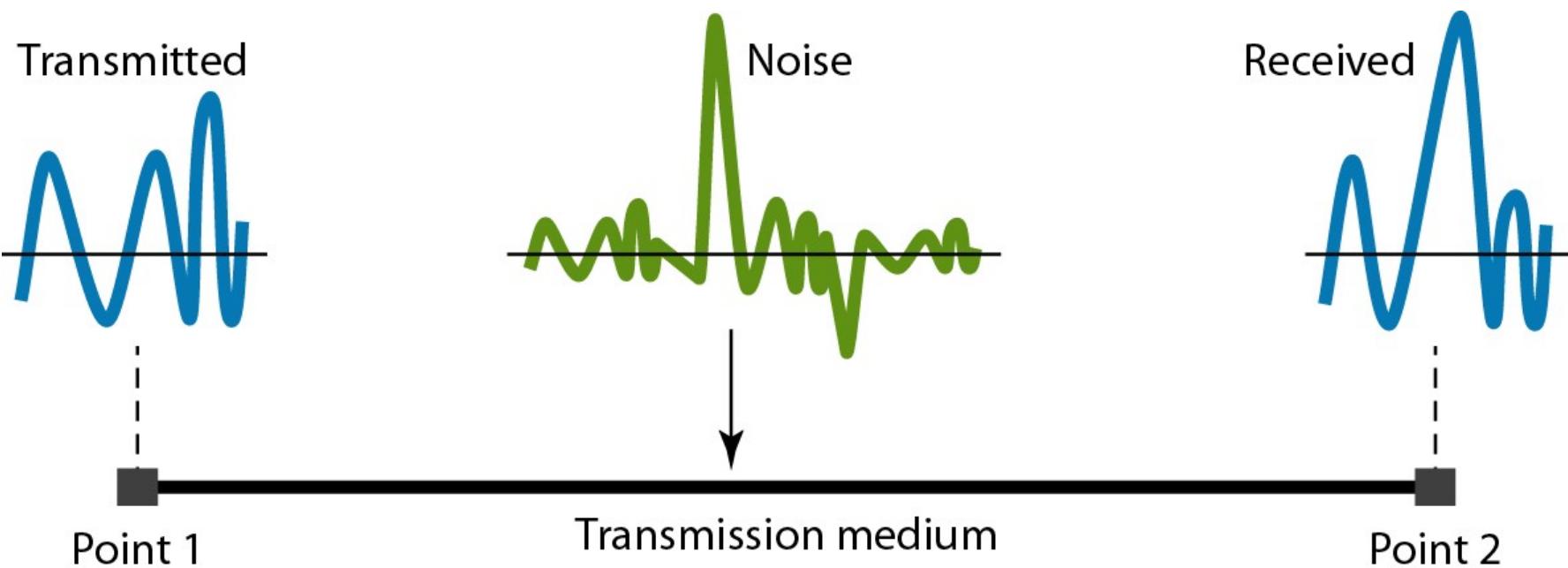


Distortion



This is because each frequency signal has its own propagation speed through a medium

Noise

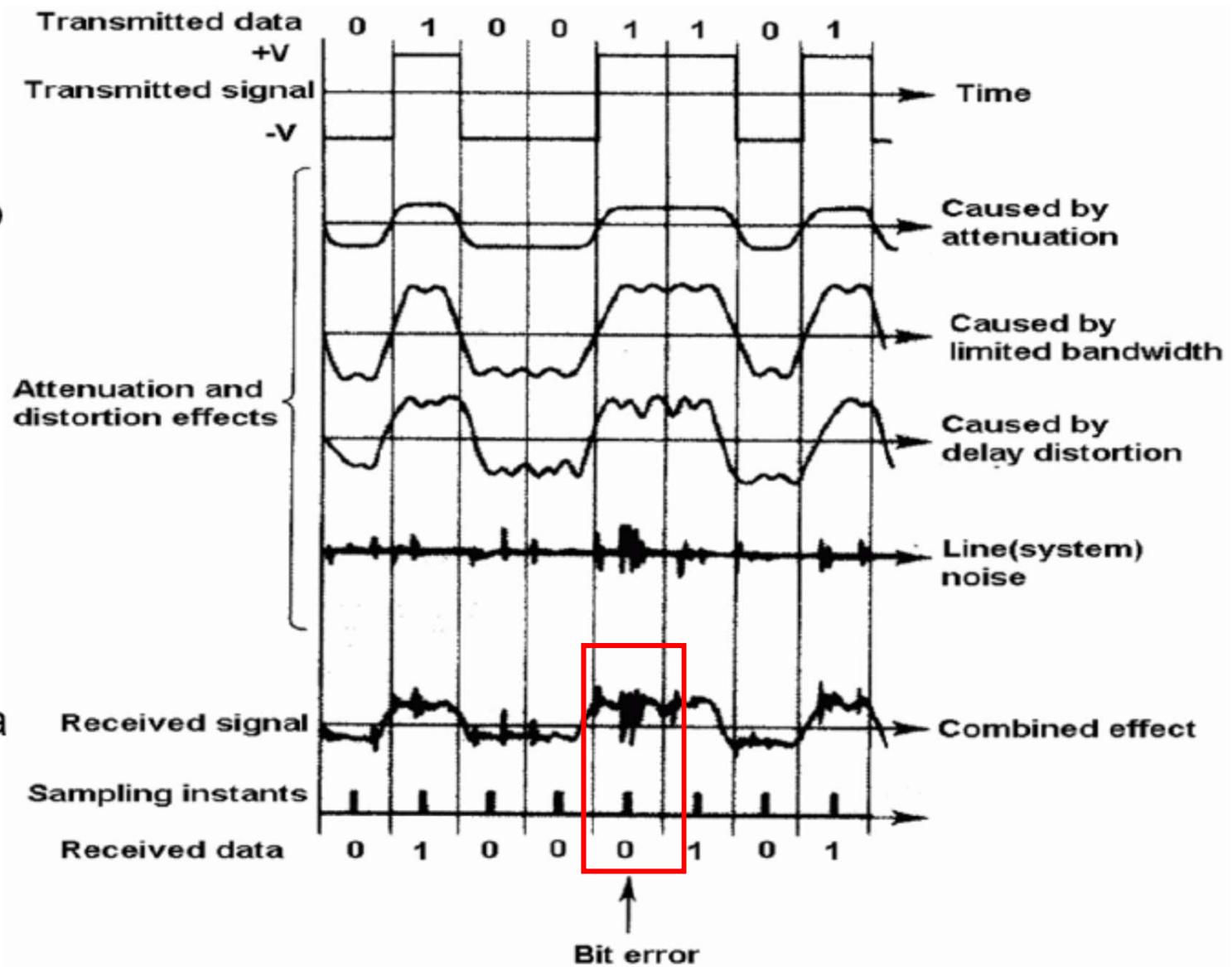


Transmission of Complex Signals

Analog signals subject to distortion by noise

Digital signals reconstructed in spite of noise

Regeneration possible for digital signals



信号的分类

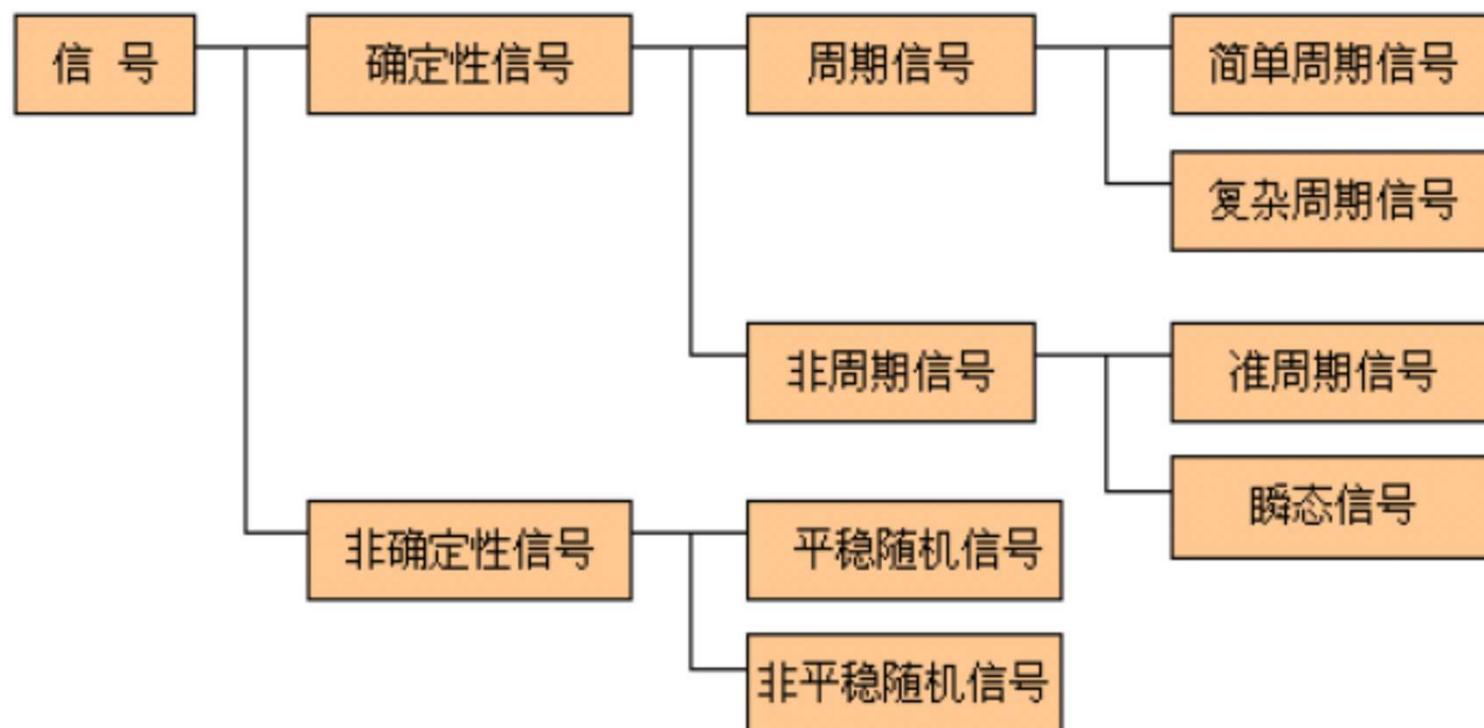
- **信号**: 定义为一个或多个独立变量的函数, 该函数含有物理系统的信息或表示物理系统状态或行为。
- **信息**: 表示对一个物理系统状态或特性的描述。(抽象性)
- **信号≠信息**, 信息需转化为传输媒质能够接受的信号形式方能传输

信号分析与处理 → 提取信息

信号的分类

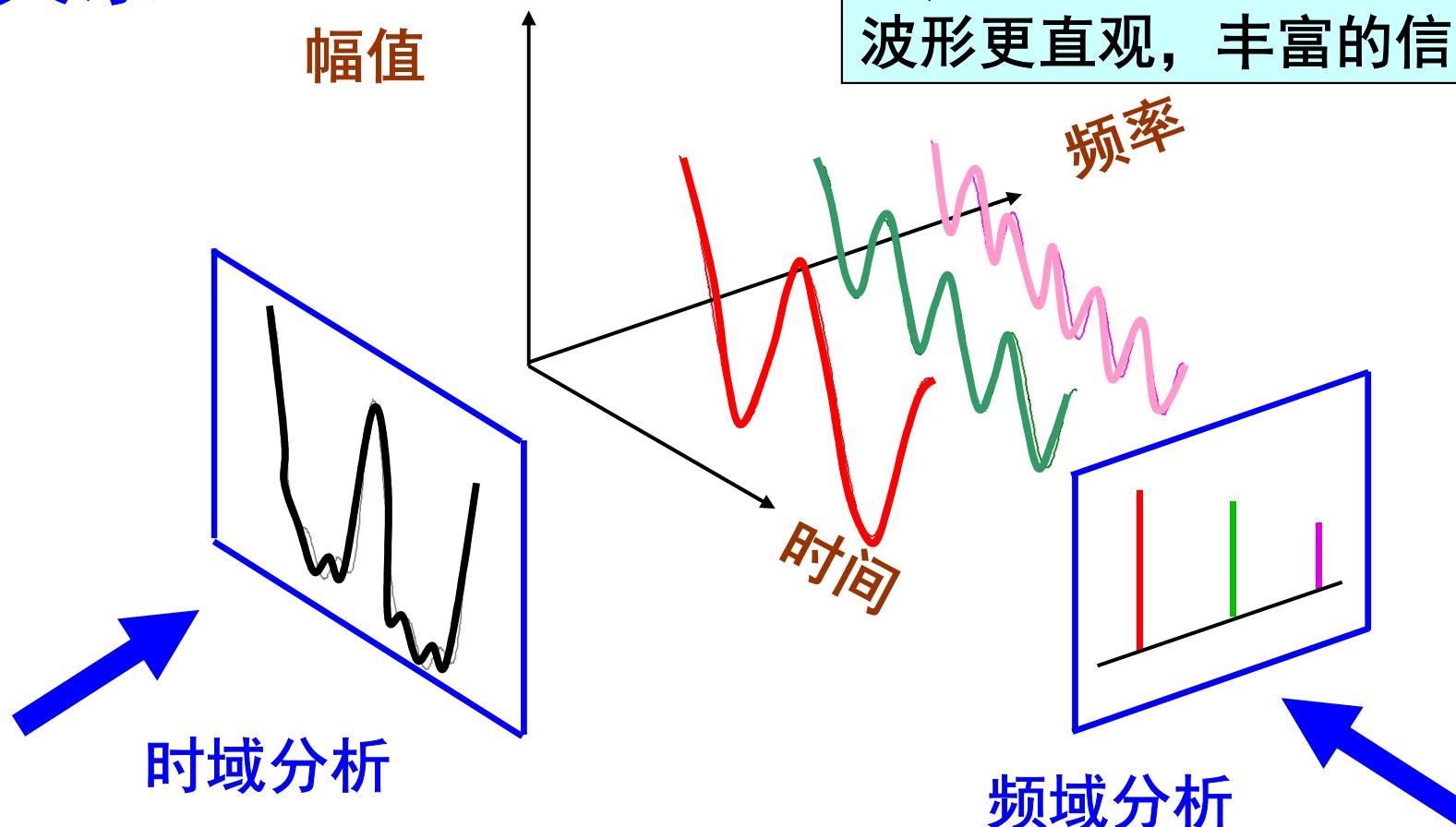
■ 确定性信号与随机(非确定性)信号：

□可以用明确数学关系式描述的信号称为**确定性信号**。
不能用数学关系式描述的信号称为**随机信号**，所描述物理现象是一种随机过程。



信号的频域分析

时域分析与频域分析的关系



Fourier Analysis



傅里叶, J.-B.-J.

法国数学家。**1768年3月21日**生于奥塞尔，**1830年5月16日**卒于巴黎。**1795年**曾在巴黎综合工科学校任讲师。**1798年**随拿破仑远征埃及，当过埃及学院的秘书。**1801年**回法国，又任伊泽尔地区的行政长官。**1817年**傅里叶被选为科学院院士，并于**1822年**成为科学院的终身秘书。**1827年**又当选为法兰西科学院院士。

Fourier Analysis

- A periodic function with period T (and frequency $f = 1/T$), $g(t)$ can be written as:

$$g(t) = \frac{1}{2}c + \sum_{n=1}^{\infty} a_n \sin(2\pi nft) + \sum_{n=1}^{\infty} b_n \cos(2\pi nft)$$

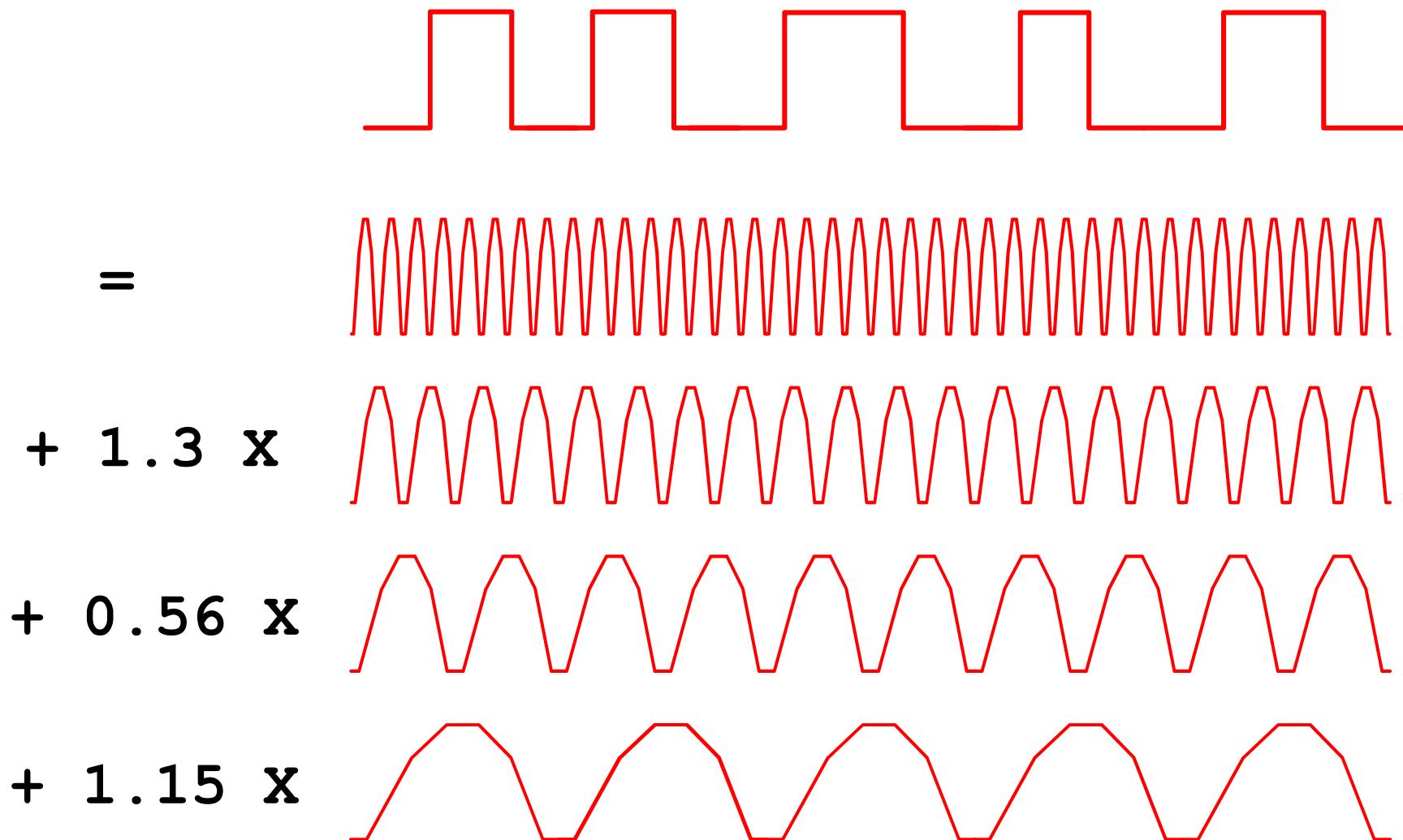
$$c = \frac{2}{T} \int_0^T g(t) dt$$

$$a_n = \frac{2}{T} \int_0^T g(t) \sin(2\pi nft) dt$$

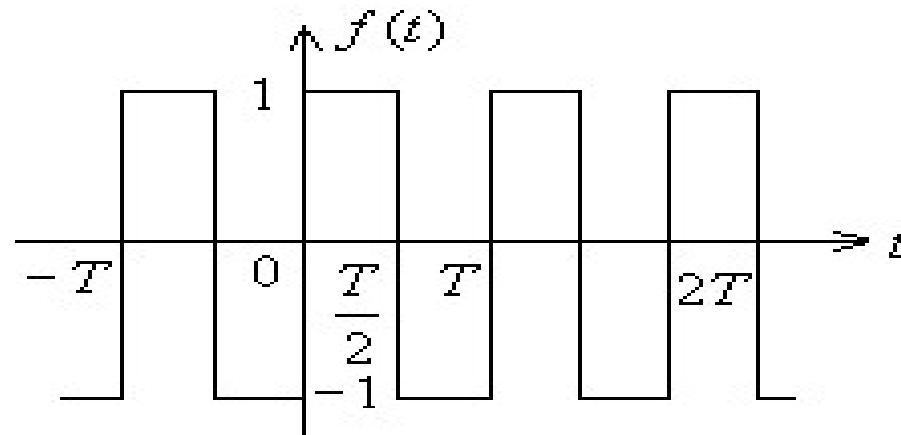
$f = 1/T$ is the fundamental frequency

$$b_n = \frac{2}{T} \int_0^T g(t) \cos(2\pi nft) dt$$

Signal = Sum of Waves



例：将下图中的方波信号展开为傅里叶级数。



$$\text{解: } f(t) = \frac{a_0}{2} + \sum_{n=1}^{\infty} a_n \cos(n\Omega t) + \sum_{n=1}^{\infty} b_n \sin(n\Omega t)$$

$$a_n = \frac{2}{T} \int_{-\frac{T}{2}}^{\frac{T}{2}} f(t) \cos(n\Omega t) dt$$

$$= \frac{2}{T} \int_{-\frac{T}{2}}^0 (-1) \cos(n\Omega t) dt + \frac{2}{T} \int_0^{\frac{T}{2}} \cos(n\Omega t) dt$$

$$= \frac{1}{n\Omega} \cdot \frac{2}{T} \left[-\sin(n\Omega t) \right]_{-\frac{T}{2}}^0 + \frac{2}{T} \cdot \frac{1}{n\Omega} \sin(n\Omega t) \Big|_0^{\frac{T}{2}}$$

$$= \frac{1}{n\Omega} \cdot \frac{2}{T} \left[-\sin(n\Omega t) \right]_{-\frac{T}{2}}^0 + \frac{2}{T} \cdot \frac{1}{n\Omega} \sin(n\Omega t) \Big|_0^{\frac{T}{2}}$$

$$\Omega = \frac{2\pi}{T} \Rightarrow a_n = 0 \quad n = 0, 1, 2, 3, \dots$$

$$\begin{aligned}
b_n &= \frac{2}{T} \int_{-\frac{T}{2}}^{\frac{T}{2}} f(t) \sin(n\Omega t) dt \\
&= \frac{2}{T} \int_{-\frac{T}{2}}^0 (-1) \sin(n\Omega t) dt + \frac{2}{T} \int_0^{\frac{T}{2}} \sin(n\Omega t) dt \\
&= \left. \frac{2}{T} \cdot \frac{1}{n\Omega} \cos(n\Omega t) \right|_{-\frac{T}{2}}^0 + \left. \frac{2}{T} \cdot \frac{1}{n\Omega} [-\cos(n\Omega t)] \right|_0^{\frac{T}{2}} \\
&= \frac{1}{n\pi} [1 - \cos(n\pi)] + \frac{1}{n\pi} [-\cos(n\pi)] + \frac{1}{n\pi} \\
&= \frac{2}{n\pi} [1 - \cos(n\pi)] = \begin{cases} 0 & , n = 2, 4, 6, 8, \dots \\ \frac{4}{n\pi} & , n = 1, 3, 5, 7, \dots \end{cases}
\end{aligned}$$

$$a_n = 0 \quad n = 0, 1, 2, 3, \dots$$

$$b_n = \begin{cases} 0 & , \quad n = 2, 4, 6, 8, \dots \\ \frac{4}{n\pi} & , \quad n = 1, 3, 5, 7, \dots \end{cases}$$

$$f(t) = \frac{4}{\pi} [\sin(\Omega t) + \frac{1}{3}\sin(3\Omega t) + \frac{1}{5}\sin(5\Omega t) + \dots + \frac{1}{n}\sin(n\Omega t) + \dots]$$
$$n = 1, 3, 5, \dots$$

它仅含有一、三、五、七.... 等奇次谐波分量。

如下页图所示，是用有限项傅里叶级数来逼近的情况：

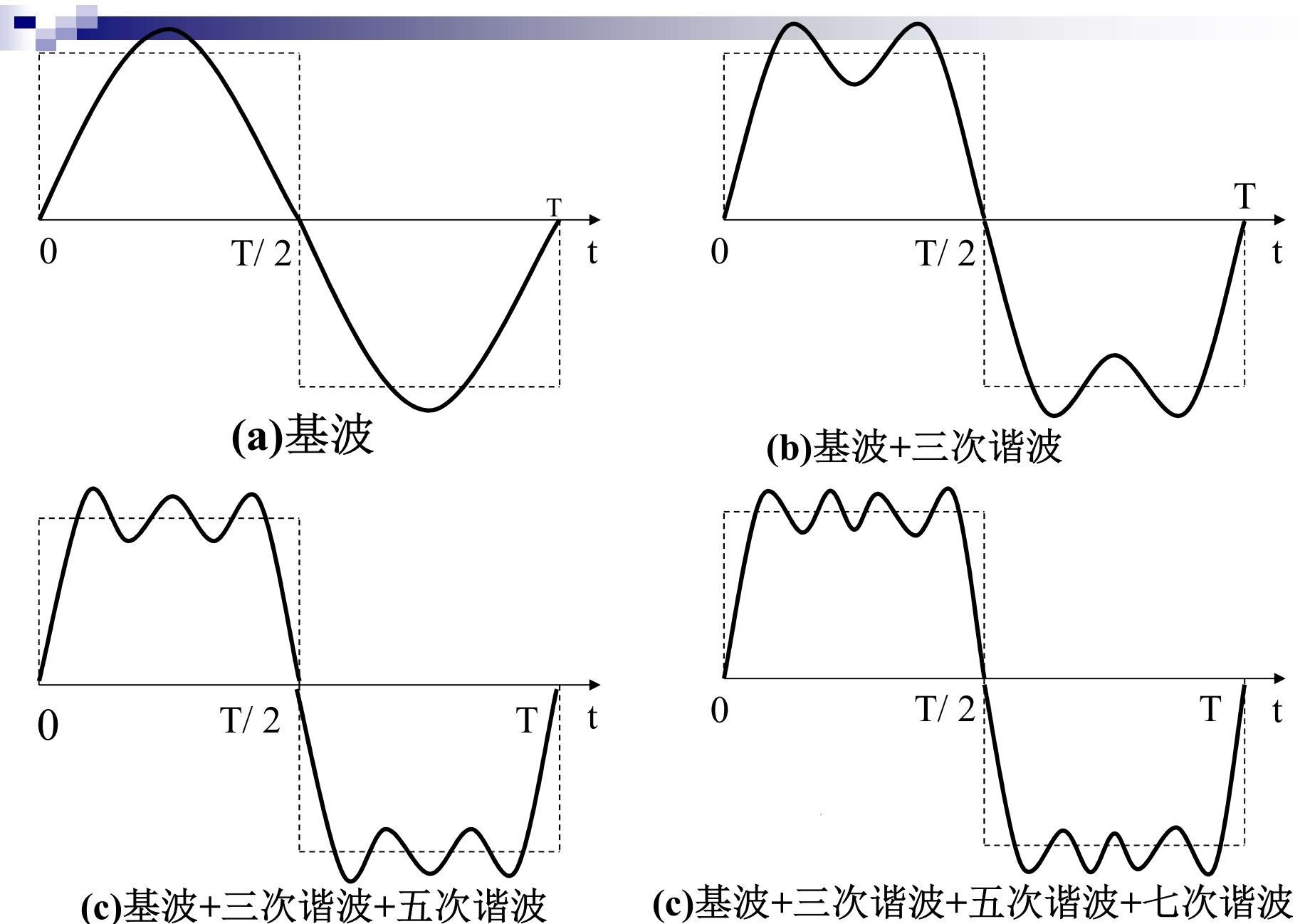
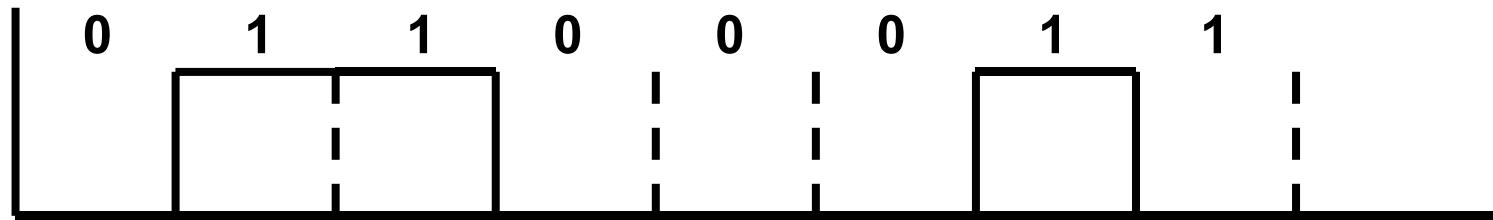


图 方波的组成

Bandwidth-Limited Signals

- Example: 01100010, 8 bit for ASCII “b”



$$a_n = 1/\pi n [\cos(\pi n/4) - \cos(3\pi n/4) + \cos(6\pi n/4) - \cos(7\pi n/4)]$$

$$b_n = 1/\pi n [\sin(3\pi n/4) - \sin(\pi n/4) + \sin(7\pi n/4) - \sin(6\pi n/4)]$$

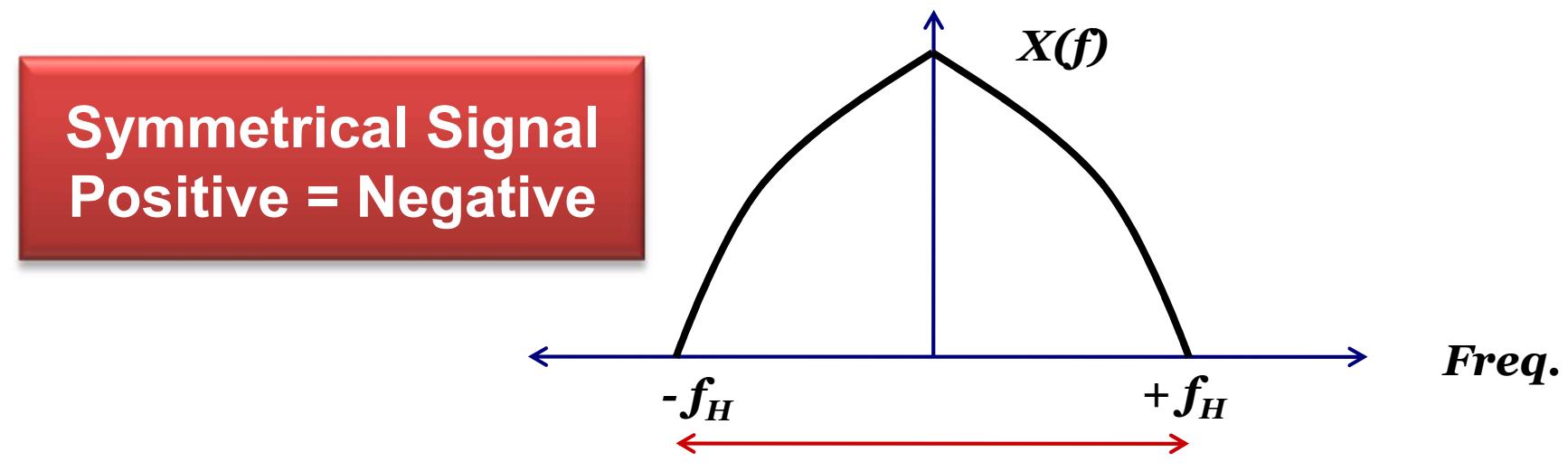
$$c_n = 3/4$$

The root mean square amplitudes is $\sqrt{a_n^2 + b_n^2}$

与对应频率处的传输能量成正比

Bandwidth-Limited Signals

- A band limited signal is a signal who has a finite spectrum.
- Most of signal energy in the spectrum is contained in a finite range of frequencies.
- After that range, the signal power is almost zero or negligible value.



Bandwidth-Limited Signals

■ What does this all mean?

- The quality of transmission is frequency dependent, not all parts of the digital signal get through the wire as you would expect.
- Digital signal transmission is subject to attenuation, distortion, etc. This is partly caused by disallowing high-frequency components to pass through.

Bandwidth

■ What is the Bandwidth?

- the range of frequency (the difference between the highest and the lowest frequencies contained in that signal.)

$$B = f_h - f_l = 900 - 100 = 800 \text{ Hz}$$

- the number of bits of a transmission medium.

Baud-rate and Bit-rate

■ **baud rate:**

The number changes in a signal per second is called the **baud rate, $B=1/T$.**

■ **bit rate:**

The number of bits transmitted per second is called the **bit rate, bits per second, b/s or bps.**

Relationship between baud-rate and bit-rate

- $S = B * \log_2 v = 1/T * \log_2 v$

S is bit rate,

B is baud rate,

T is frequency period,

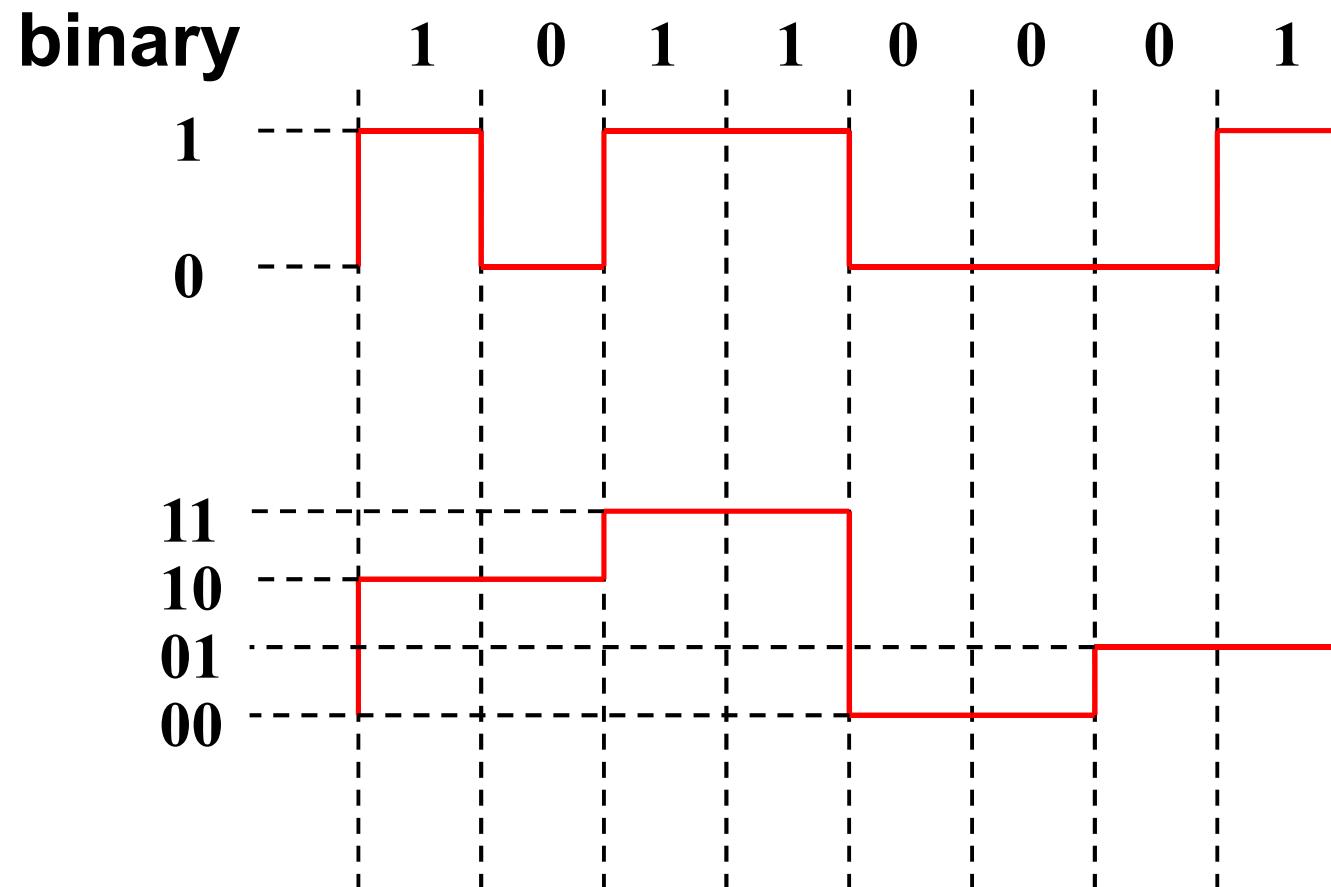
v is number of signal values (被称为码元数) .

- **Ex:**

$B=9600, v=2, s=9600 \text{ bps};$

$B=2400, v=16, s=9600 \text{ bps}$

Relationship between baud-rate and bit-rate



Baud rate=?

Bit rate=?

The maximum data rate of a channel

- 有失真，但可识别



- 失真大，无法识别



The maximum data rate of a channel

■ Nyquist criterion

for noiseless lowpass channels, if the cut-off frequency is H Hz, the filtered signal can be reconstructed by making $2H$ samples. No more, no less. Consequence:

Max. baud rate = $2H$ (baud)

Max. bit rate = $2H \log_2 V$ (bps)

(where H is the bandwidth

V is the number of signal values)

This is for noiseless channels, the maximum data rate of a 3-kHz noiseless channel is no more than 6000 bps, for binary transmission.

The maximum data rate of a channel

■ Shannon criterion

a **noisy channel** with a signal-to-noise ratio **S/N**, has a limit with respect to the **bit rate**:

$$\text{Max. bit rate} = H \log_2 (1+S/N) \text{ (bps)}$$

■ In practice, we use dB(decibel) to represent signal-to-noise ratio instead of S/N:

$$10\log_{10}(S/N) = ? \text{ dB}$$

$$\text{Ex. } 10\log_{10}(10) = 10 \text{ dB}$$

The maximum data rate of a channel

■ **Shannon criterion Example:**

A telephone line with $H = 3000$ and $10\log_{10}(S/N) = 30\text{dB}$, then
max. trans. Rate = 30kbps
no matter how you do your encoding
(excluding compression).

The maximum data rate of a channel

Note

The Shannon capacity gives us the upper limit; the Nyquist formula tells us how many signal levels we need.

Chapter 2: Roadmap

2.1 Physical layer Introduction

2.2 Data communication

2.3 Transmission Media

2.4 Modulation and Data Encoding

2.5 Multiplexing

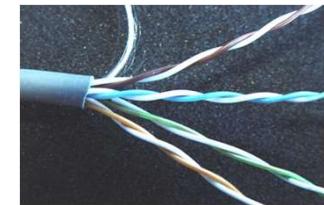
Copper Wires

- **Twisted pair**
 - **Shielded Twisted Pair (STP)**
 - **Unshielded Twisted Pair (UTP)**

Categories of Unshielded Twisted Pair

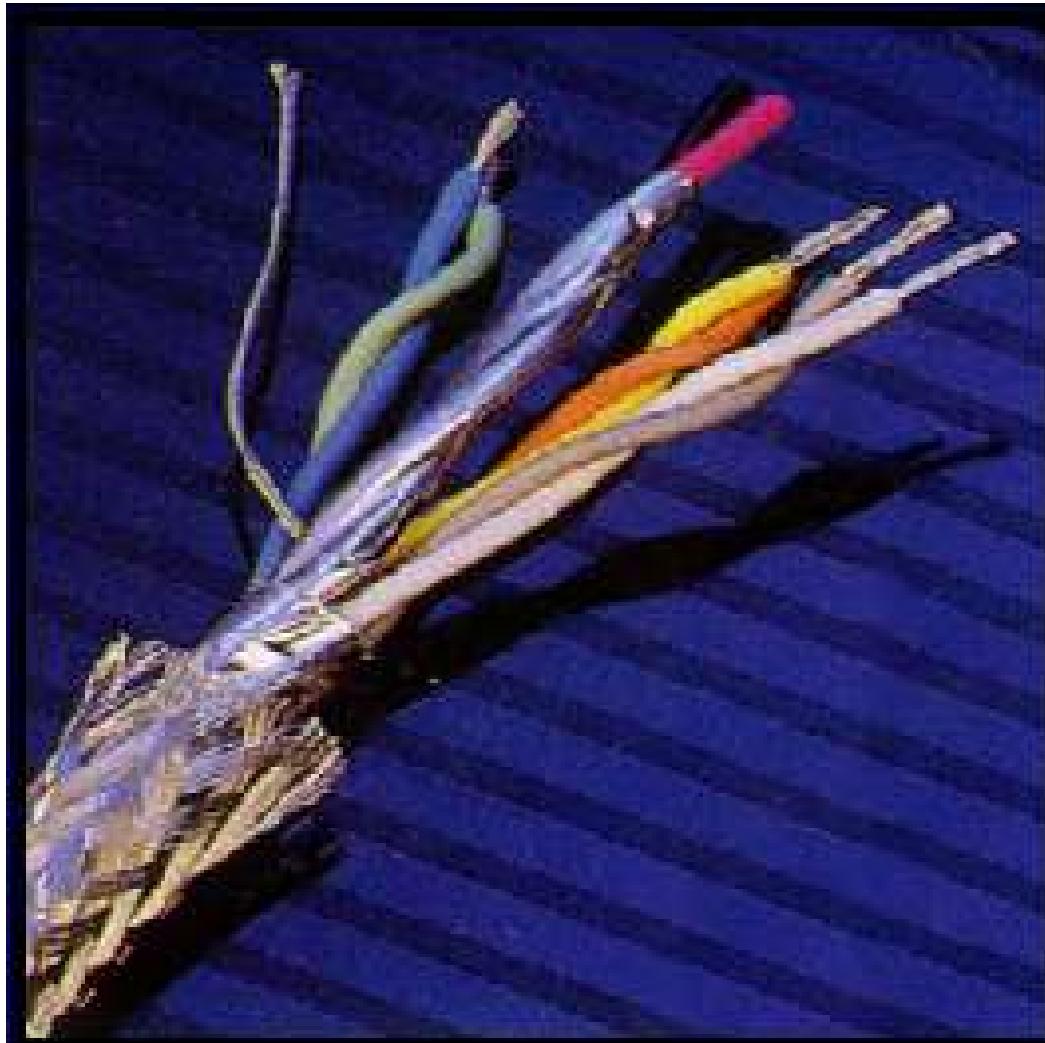
The EIA/TIA (Electronic Industry Association/Telecommunication Industry Association) has established standards of UTP.

Type	Use
Category 1	Voice Only (Telephone Wire)
Category 2	Data to 4 Mbps (LocalTalk)
Category 3	Data to 10 Mbps (Ethernet)
Category 4	Data to 20 Mbps (16 Mbps Token Ring)
Category 5	Data to 100 Mbps (Fast Ethernet)
Category 5e	Data to 100 Mbps (Fast Ethernet)
Category 6	Data to 250 MHz (Gigabit Ethernet?)
Category 7	Data to 600 MHz



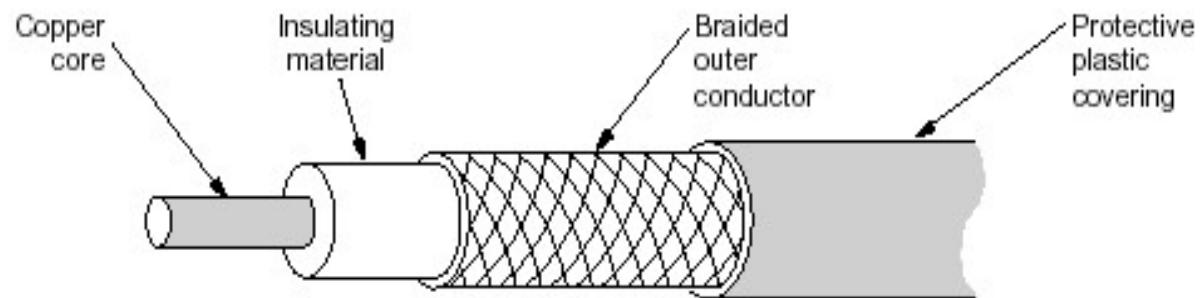


STP



Copper Wires

■ Coaxial Cable



Baseband Coax

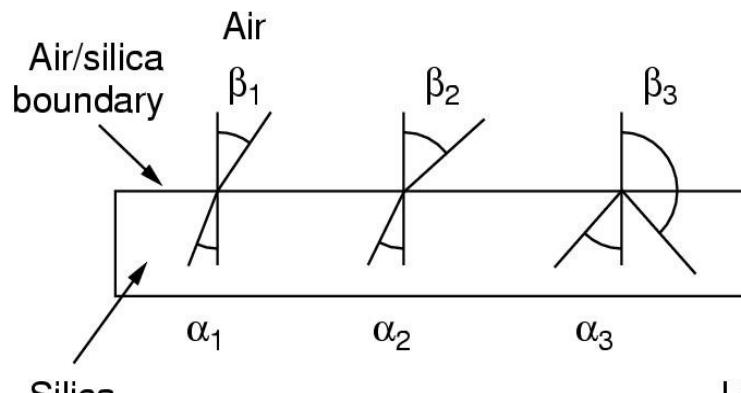
- 50-ohm cable for digital transmission
- 10Base-2, BNC, Thin-LAN, 185m/ per segment
- 10Base-5, AUI, Thick-LAN, 500m/ per segment
- At most 5 segments, up to 945m/2500m.

■ Broadband Coax

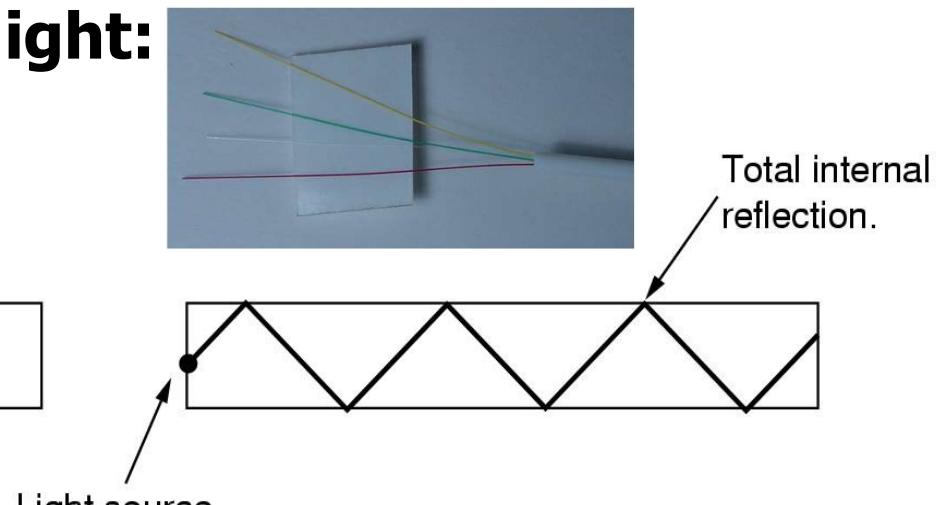
- 75-ohm cable for analog transmission, like cable TV.

Fiber Optics

- **Principle:** optical signals that are passed through optical fiber. Principal working is based on the refraction property of light:



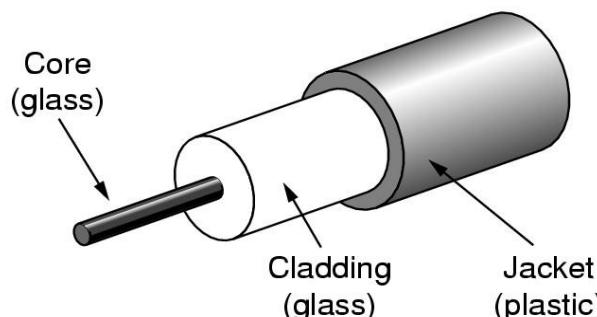
(a)



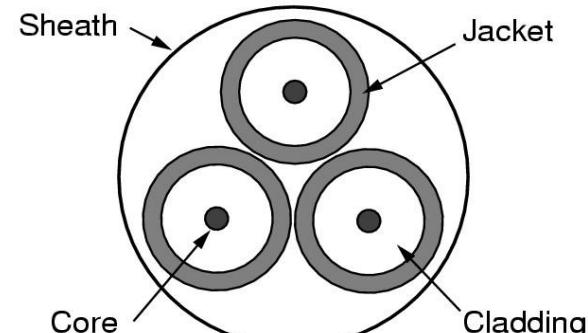
(b)

- (a) Three examples of a light ray from inside a silica fiber impinging on the air/silica boundary at different angles.
- (b) Light trapped by total internal reflection.

As it turns out, attenuation is extremely well in optical fiber. This means that they can be used for long distances. In addition, the bandwidth is enormous.



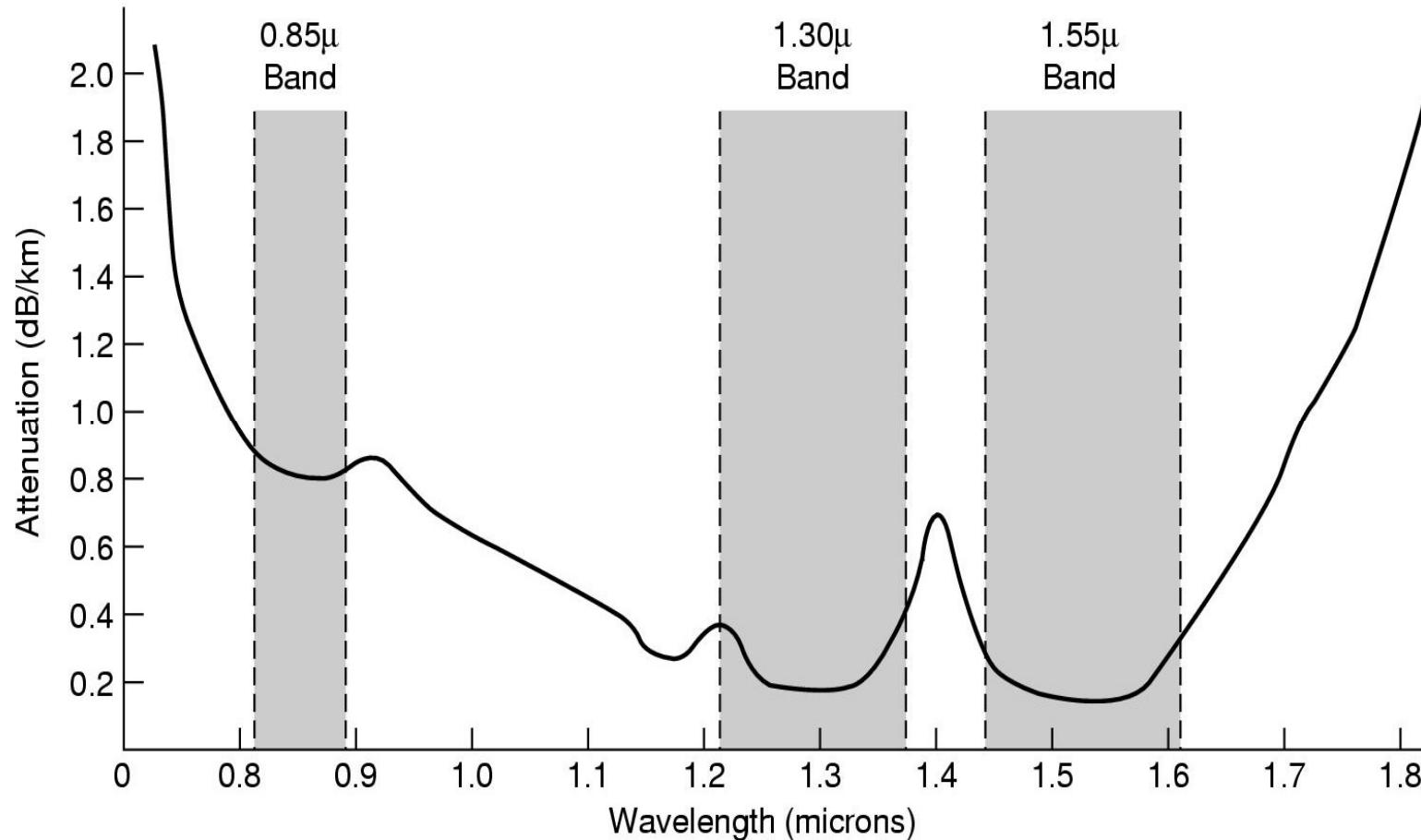
(a)



(b)

Item	LED	Semiconductor Laser
Data rate	Low	High
Mode	Multimode	Multimode or single mode
Distance	Short	Long
Lifetime	Long life	Short life
Temperature sensitivity	Minor	Substantial
Cost	Low cost	Expensive

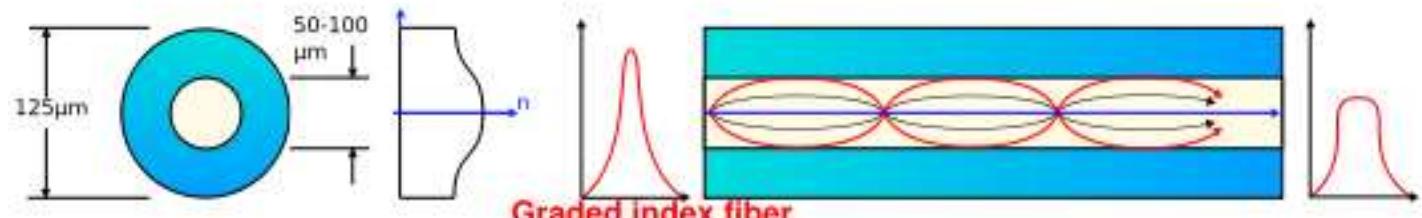
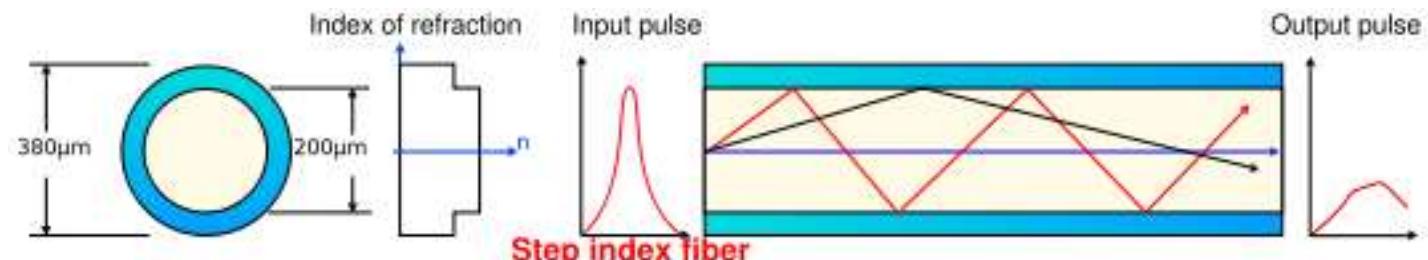
■ Attenuation in decibels =
 $10\log_{10}(\text{transmitted power}/\text{received power})$



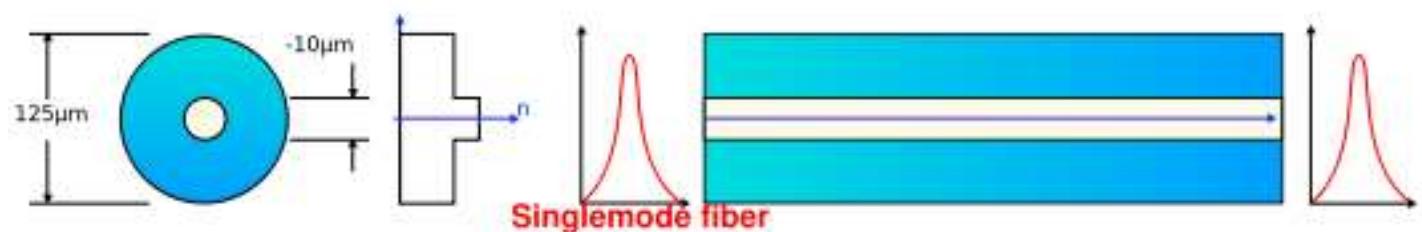
Attenuation of light through fiber in the infrared region.

Fiber Types

Multimode



Single mode



Wireless Transmission

■ The electromagnetic spectrum

- Wireless Transmission is really great for all of us who can't sit still, or feel they have to be on-line all the time (watch it – you may miss something).
- It's also convenient when wiring is needed where it can't be done, or isn't really worth the trouble (jungles, islands, mountains).

The electromagnetic spectrum

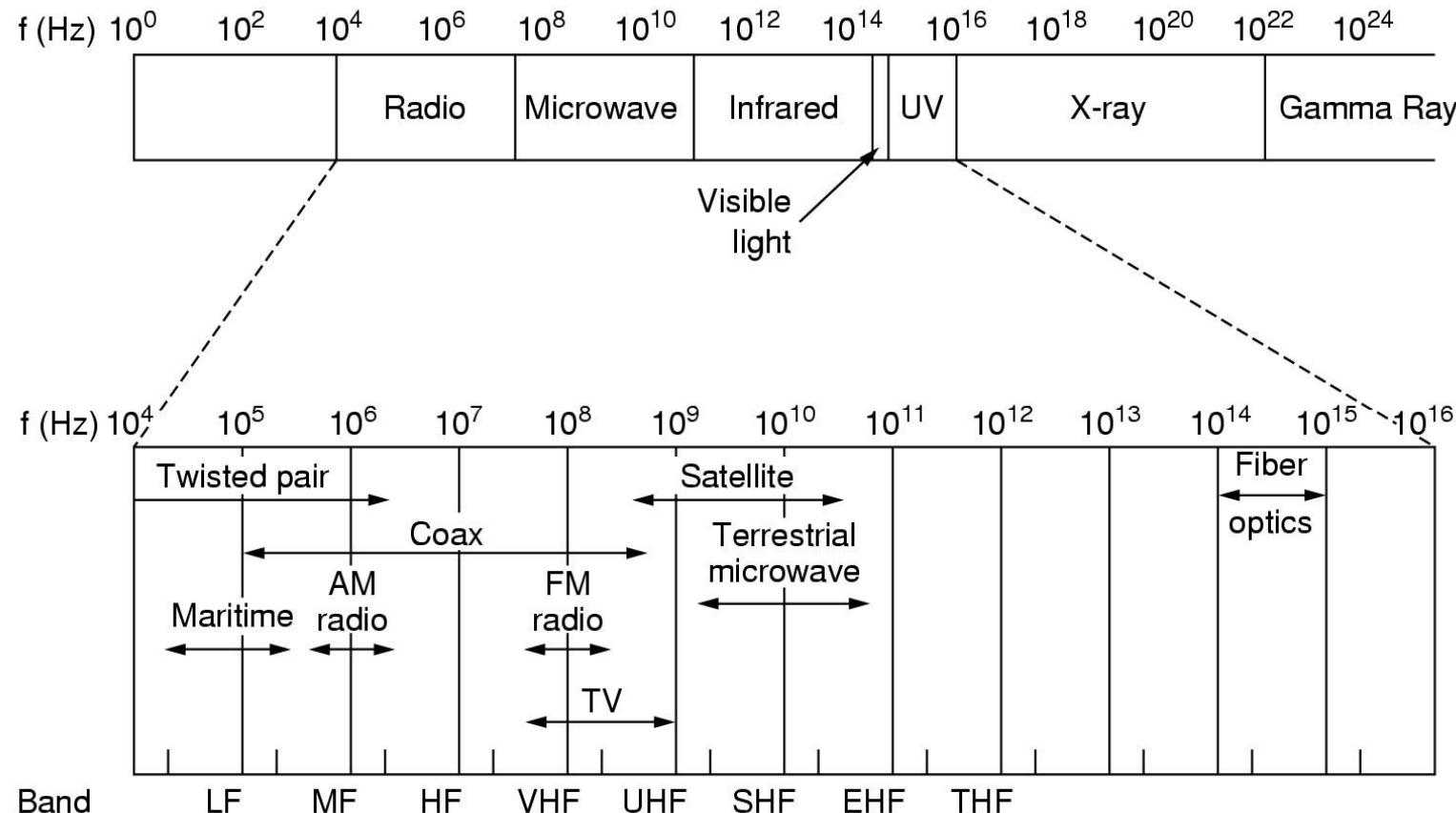
- Wireless transmissions travel at the speed of light (c), uses a frequency (f) which has a wavelength (λ). The relation is that:

$$c = \lambda * f$$

c is a constant, approximately 3×10^8 m/sec, that is theoretical in vacuum.

In practice, c is about 2×10^8 m/sec either in fiber or copper. That is **200 m/ μ sec.**
(IMPORTANT!)

The larger the wavelength is, the longer the distance it can travel without attenuation. Also, the dispersion of higher frequencies is much lower.



The electromagnetic spectrum and its uses for communication

Microwave Transmission

- Waves travel in straight lines and can be narrowly focused above 100MHz, concentrating energy into a small beam using dish-like antenna.
- Repeaters are needed periodically, for 100m high towers, repeaters can be spaced 80km apart.
- **Multipath-fading** is weather and frequency dependent.
- **Frequency range: 2.4GHz-2.484GHz, no FCC (federal communication commission) licensing needed.**

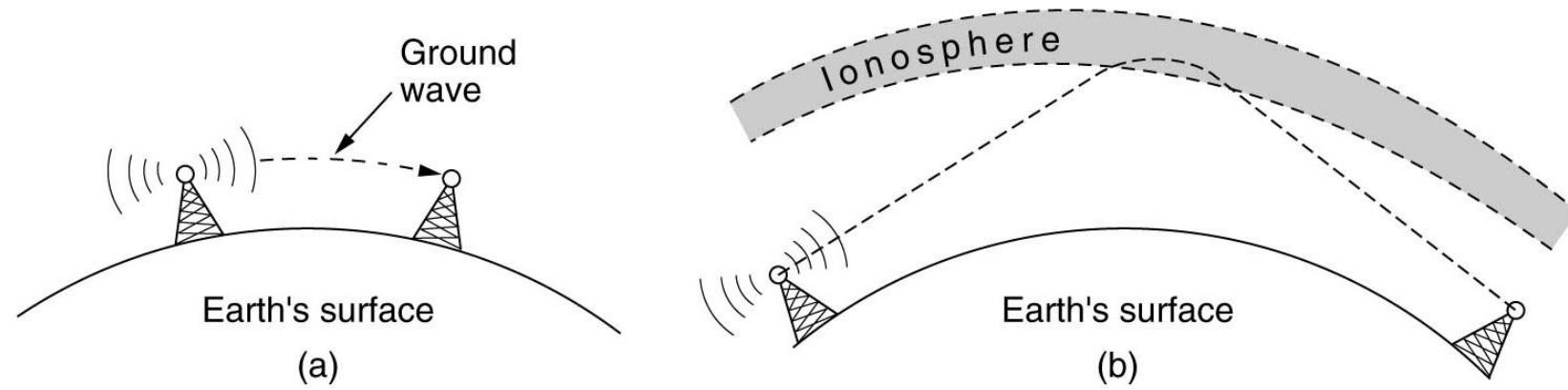
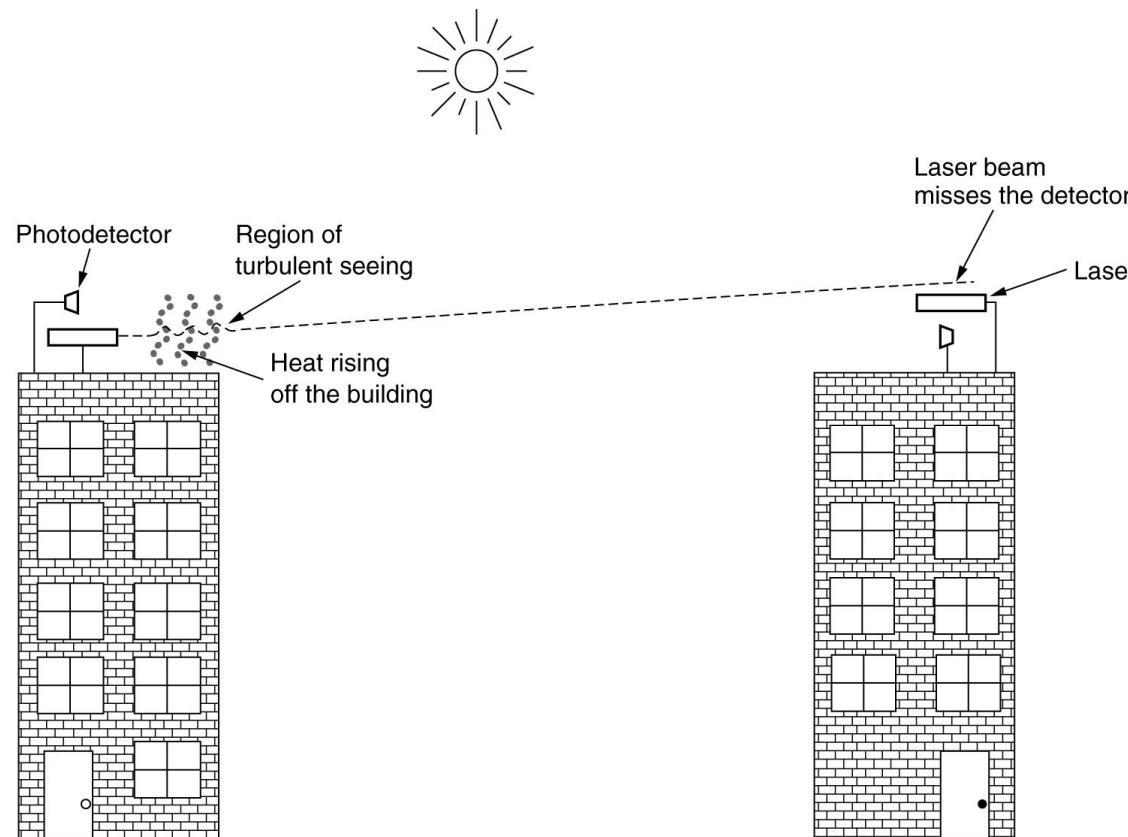


Fig. 2-12. (a) In the VLF, VF, and MF bands, radio waves follow the curvature of the earth. (b) In the HF they bounce off the ionosphere.

- **Microwave transmission is also popular and is good for long distances, as long as it's directed.**
- **Problem is the density in the spectrum, requiring higher frequency ranges (which are hard for unguided transmissions)**

Infrared and Millimeter Waves

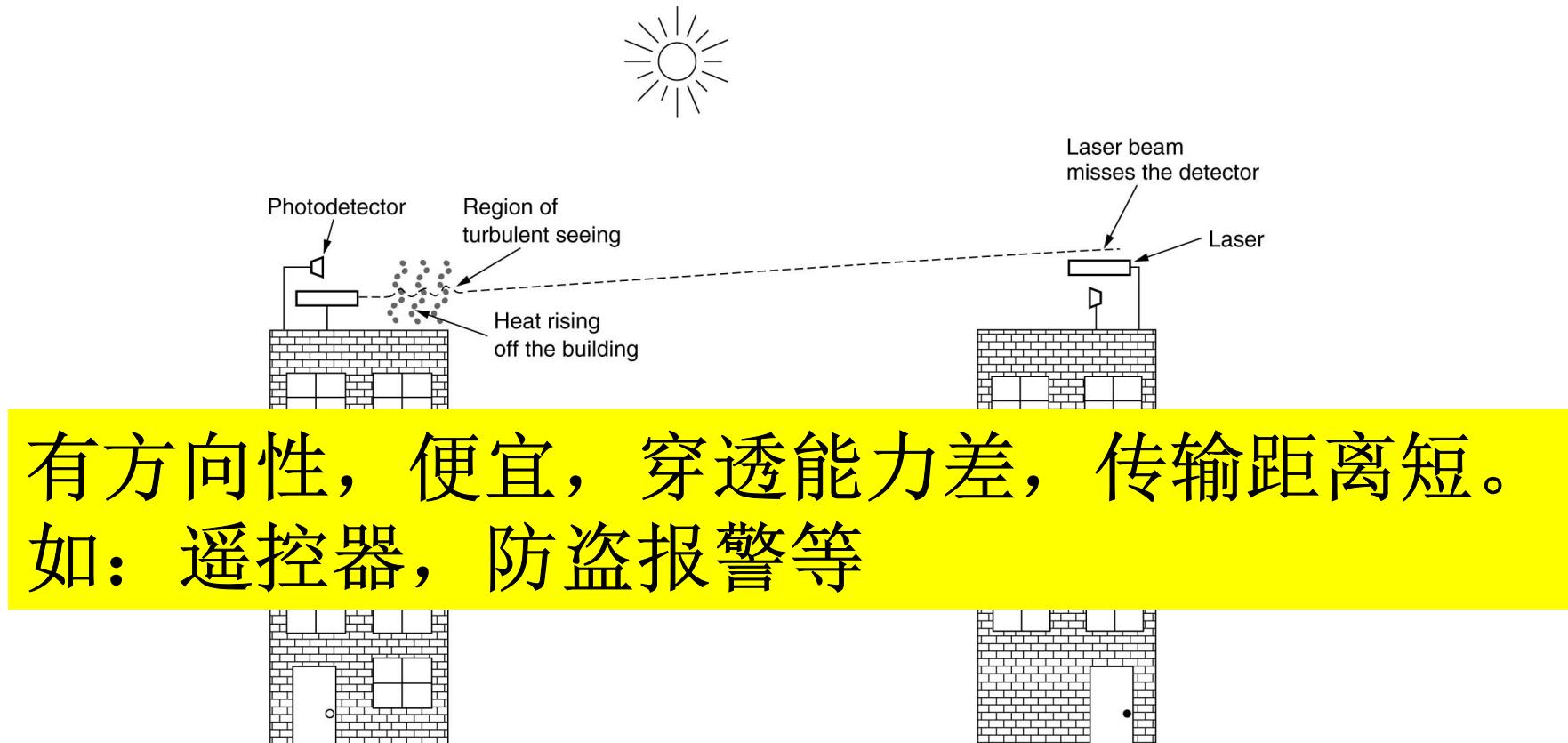
Light Wave Transmission



Convection currents can interfere with laser communication systems. A bidirectional system, with two lasers, is pictured here.

Infrared and Millimeter Waves

Light Wave Transmission

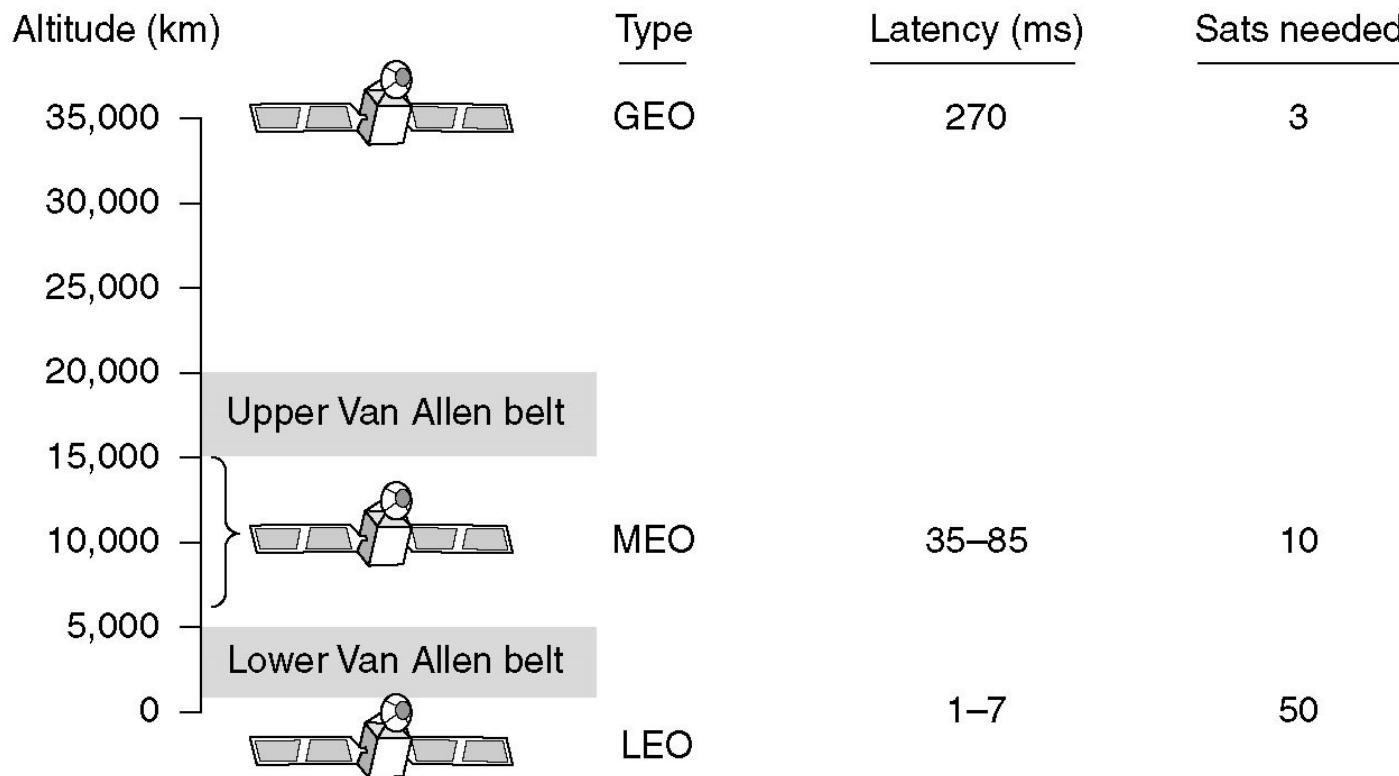


Convection currents can interfere with laser communication systems. A bidirectional system, with two lasers, is pictured here.

Communication Satellites

- **Geostationary Satellites(同步卫星)**
- **Medium-Earth Orbit Satellites**
- **Low-Earth Orbit Satellites**
- **Satellites versus Fiber**

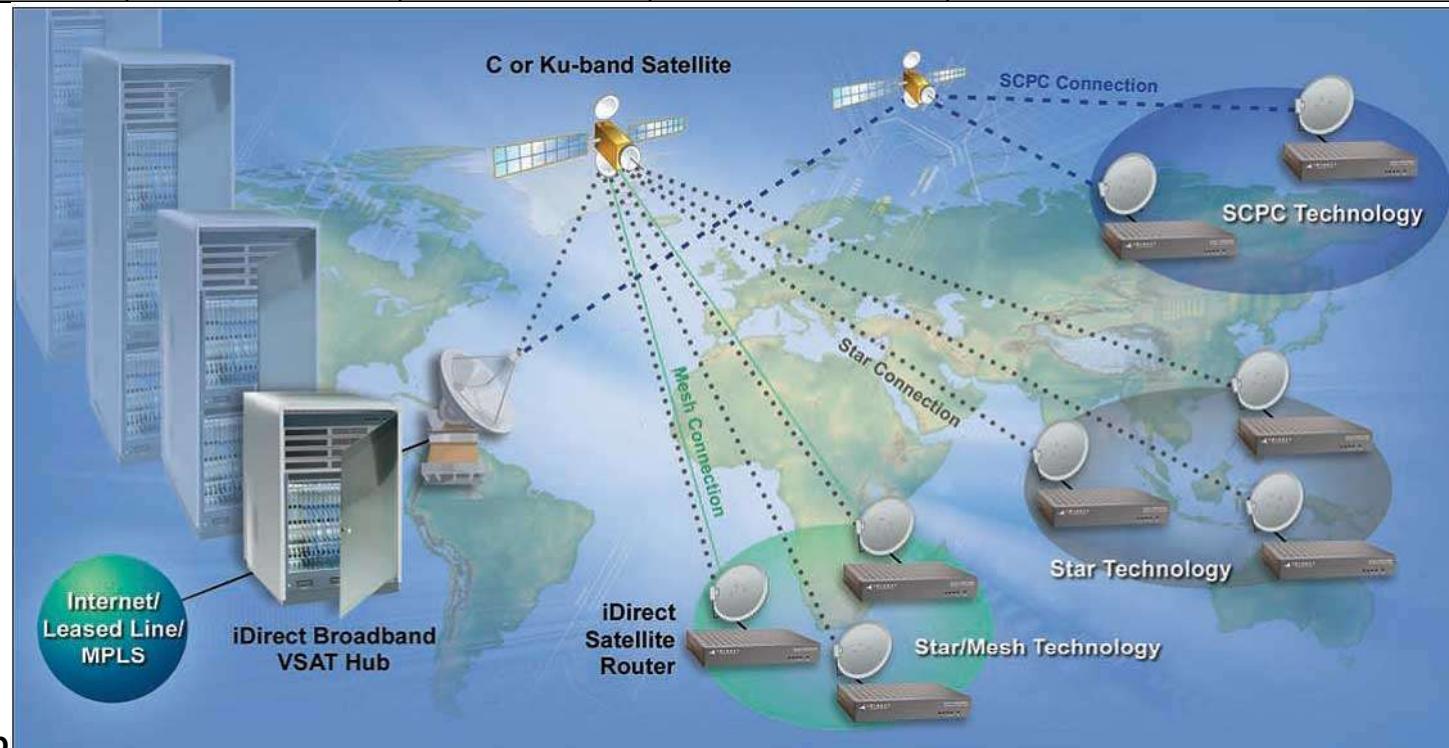
Communication Satellites



- **Communication satellites and some of their properties, including altitude above the earth, round-trip delay time and number of satellites needed for global coverage.**

The principal satellite bands

Band	Downlink	Uplink	Bandwidth	Problems
L	1.5 GHz	1.6 GHz	15 MHz	Low bandwidth; crowded
S	1.9 GHz	2.2 GHz	70 MHz	Low bandwidth; crowded
C	4.0 GHz	6.0 GHz	500 MHz	Terrestrial interference
Ku	11 GHz	14 GHz	500 MHz	Rain
Ka	20 GHz	30 GHz	3500 MHz	Rain, equipment cost

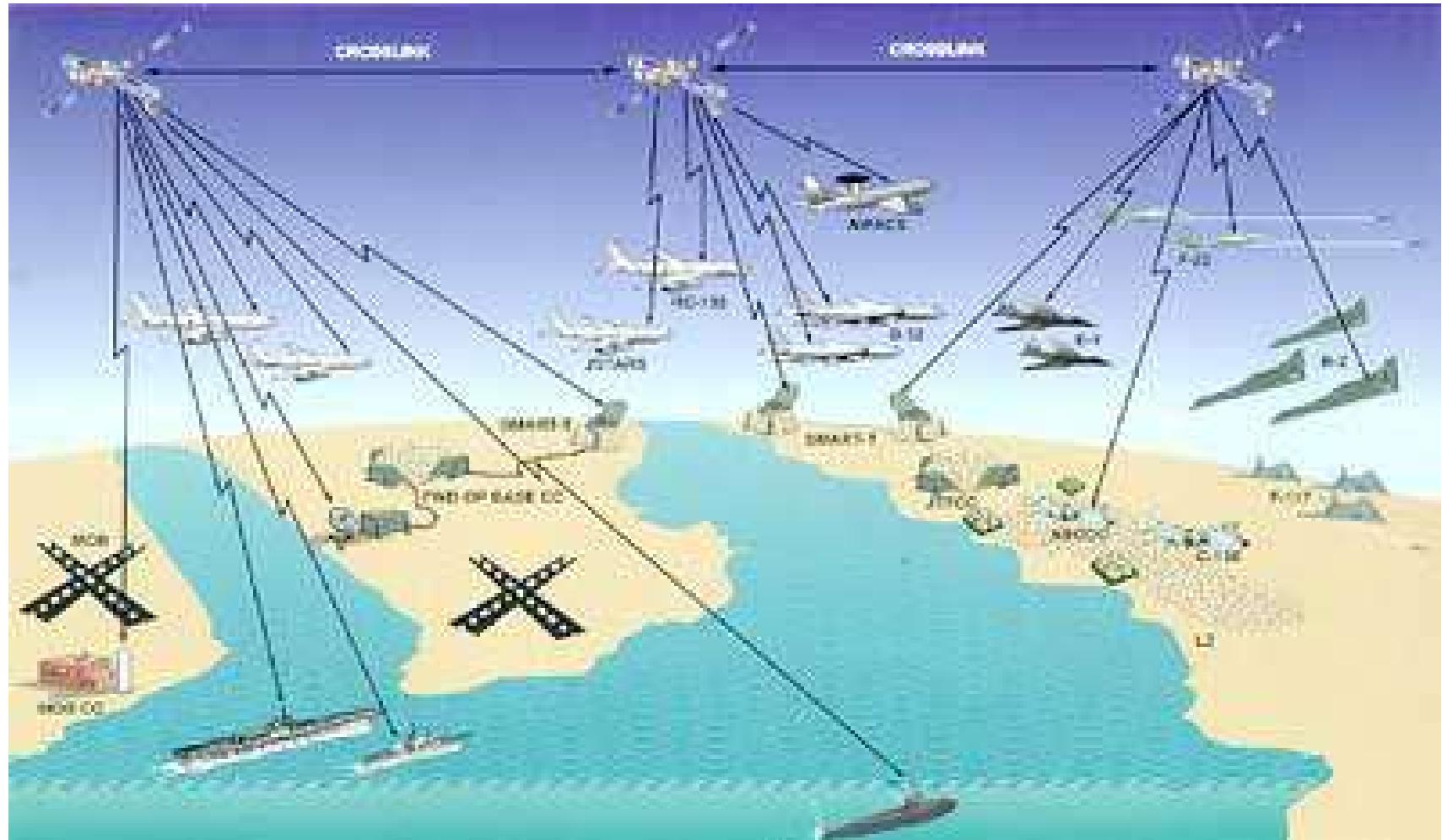


The principal satellite bands



VSATs using a hub
VSAT: very small aperture terminals

The principal satellite bands



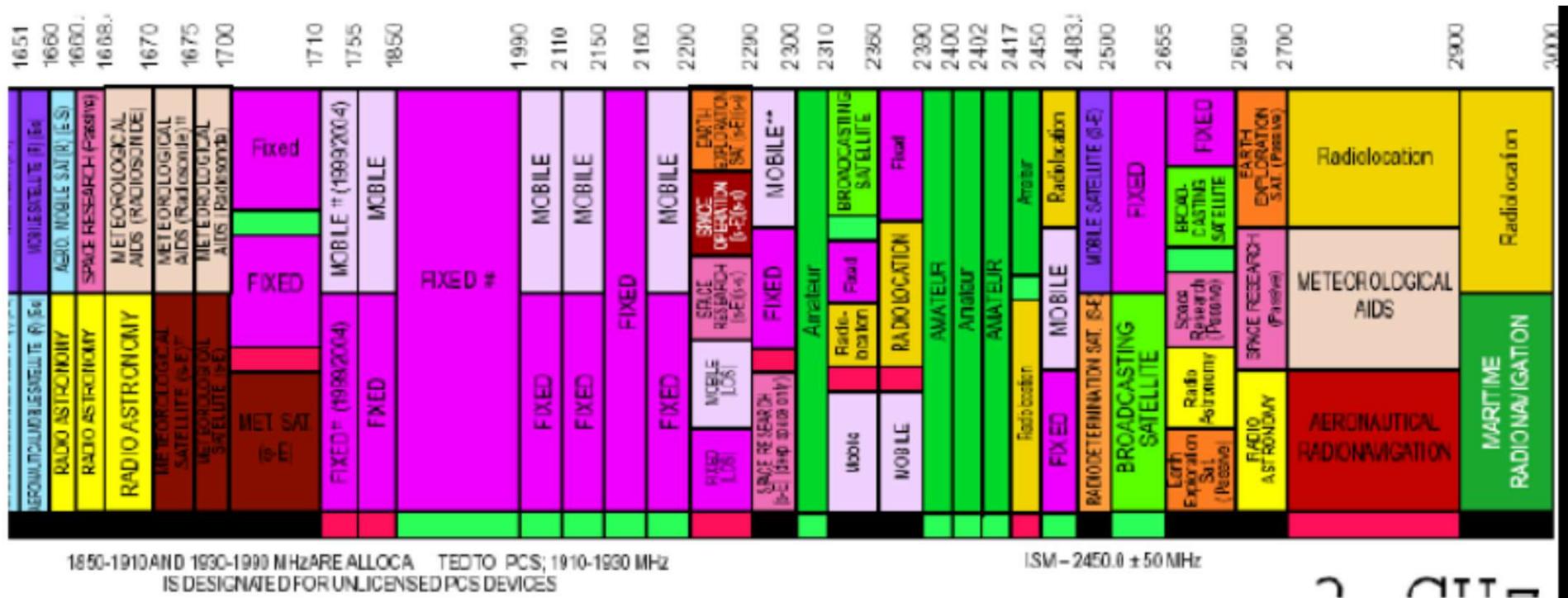
美国“军事星系统”工作示意图

Where (not) to use Satellites

- **Bandwidth:** Fiber wins, but not everyone has access to all the available bandwidth.
Satellites may make it easier to transfer data anyway
- **Mobility and remote locations:** Satellites win, although it isn't clear whether simple cellular techniques may do just fine
- **Broadcasting:** Satellites win easily: broadcasting essentially comes for free
- **Fast and reliable:** Give credits to fiber: satellites are pretty bad due to inherent high latency (230 ms round-trip for geostationary satellites), and too much Mother Nature (rain!)

Main issue: Interfering with others....

Licencing of the frequencies



Competition in (few!) bands dedicated to **unlicenced** sharing
Recent research: secondary usage of **licenced** frequencies,
temporarily not used by the "owner"

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Data and Signals

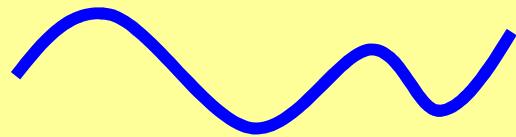
Note

To be transmitted, data must be transformed to electromagnetic signals.

Analog and Binary Data

Data can be analog or digital

Analog Data



Smoothly changing among an infinite number of states (loudness levels, etc.)

Binary Data

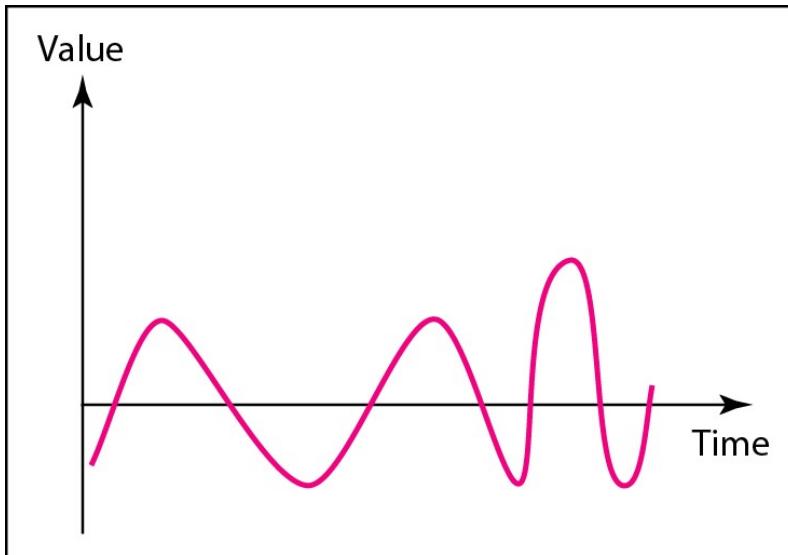
1101011000011100101

Two states:
One state represents 1
The other state represents 0

Analog and Binary Signals

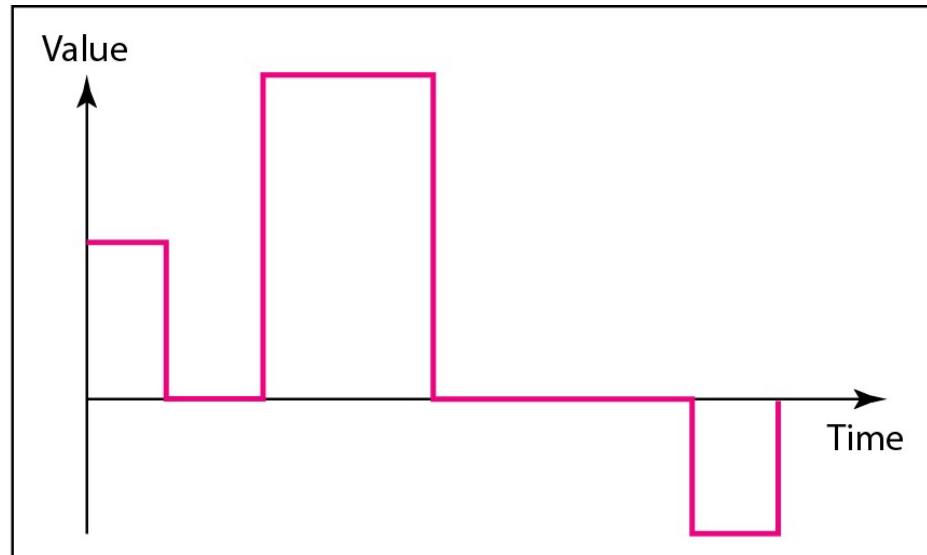
Signals can be analog or digital.

Analog signals can have an **infinite number of values** in a range



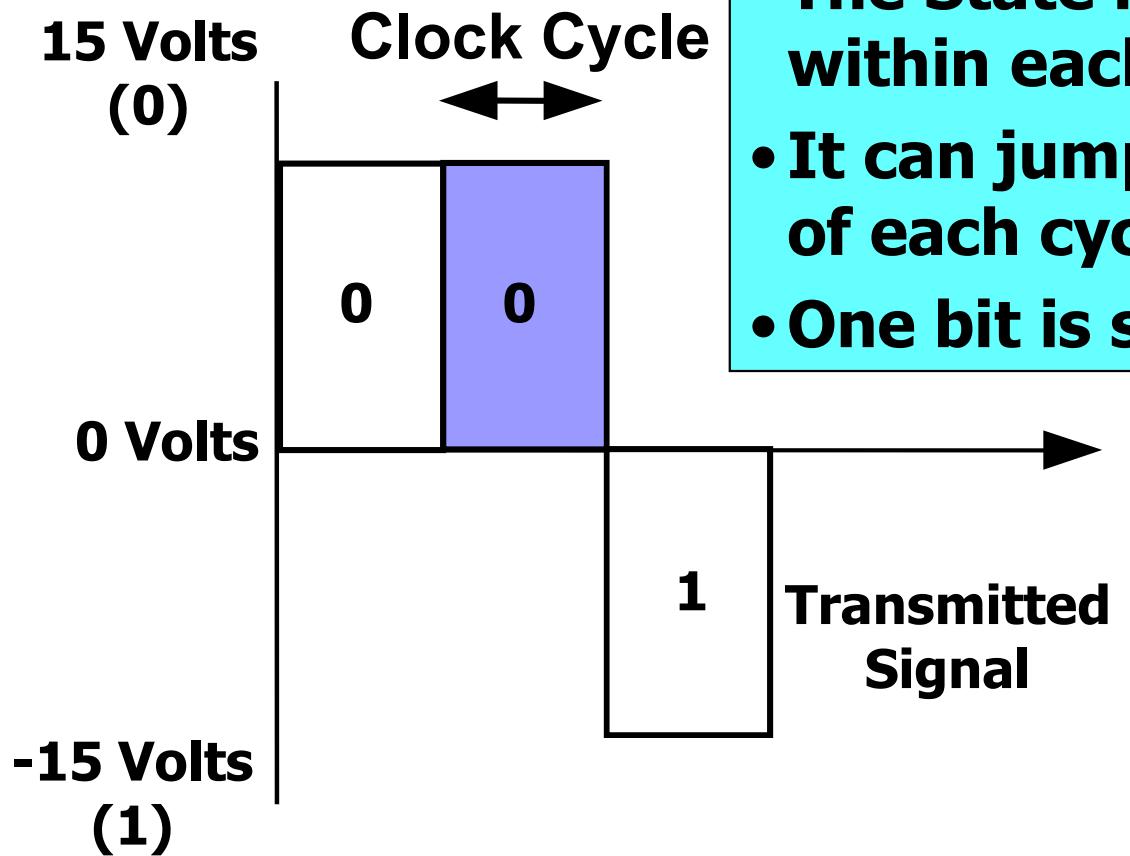
a. Analog signal

Digital signals can have only a **limited number of values**.



b. Digital signal

Binary Data and Binary Signal



- Time is divided into clock cycles
- The State is held constant within each clock cycle.
- It can jump abruptly at the end of each cycle.
- One bit is sent per clock cycle.

Analog and Binary Signals

Note

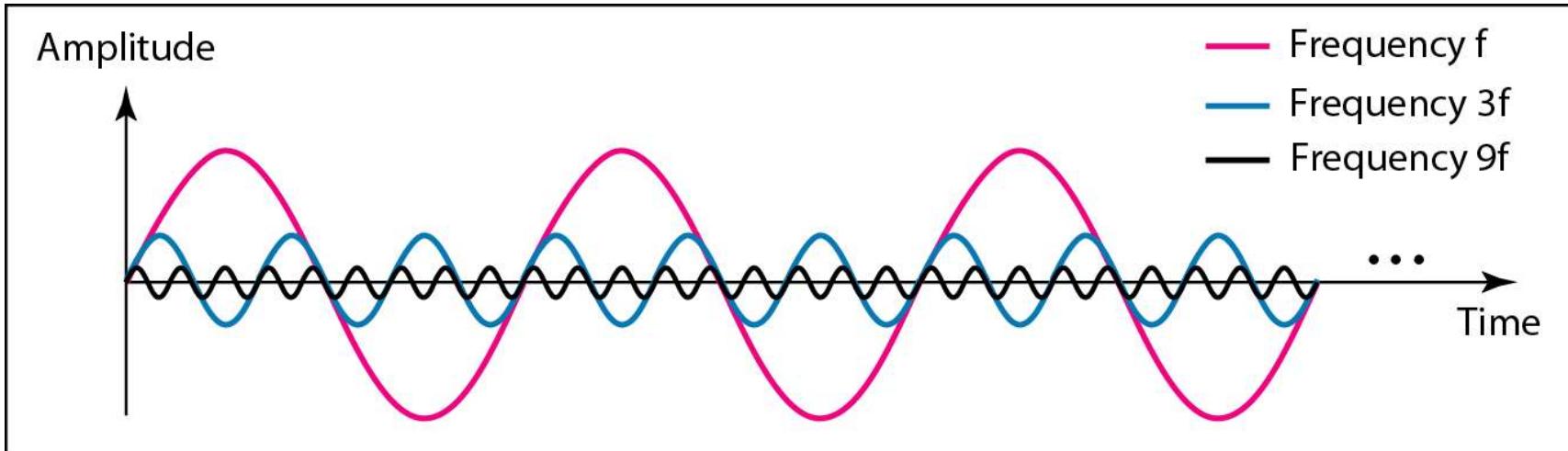
In data communications, we commonly use periodic analog signals and nonperiodic digital signals.

Periodic analog signal is used as data carrier (such as AM/FM radio)

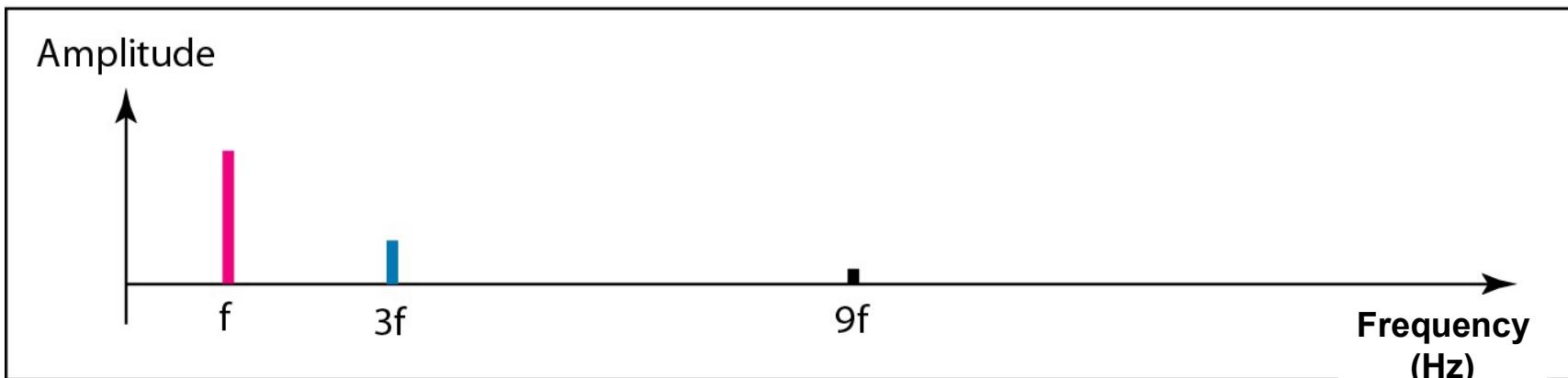
Periodic analog signals

- Periodic analog signals can be classified as **simple or composite**.
- A **simple periodic analog signal**, a **sine wave**, cannot be decomposed into simpler signals.
- A **composite periodic analog signal** is composed of multiple sine waves.

Periodic analog signals

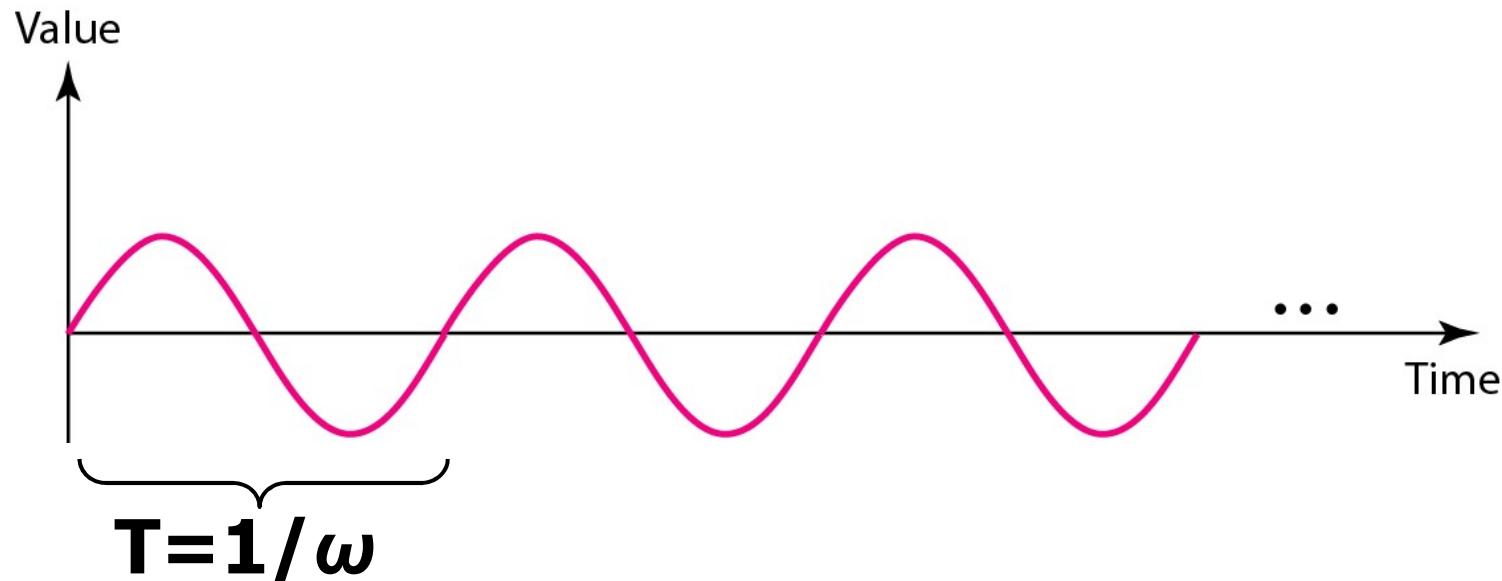


a. Time-domain decomposition of a composite signal



b. Frequency-domain decomposition of the composite signal

A sine wave



$$A \sin(\omega t + \varphi)$$

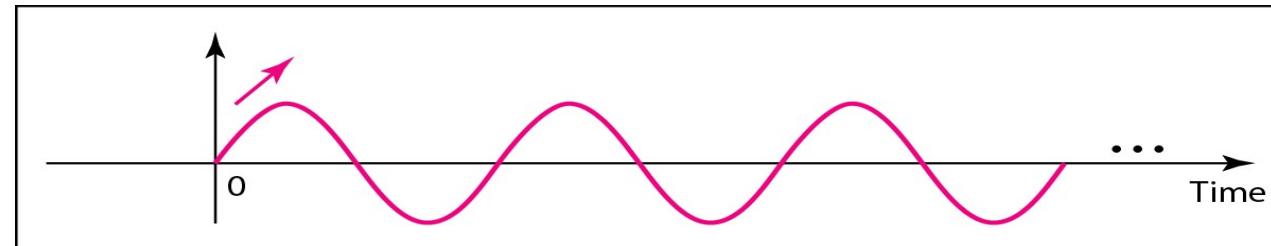
A : amplitude = maximum strength of signal (V)

ω : frequency = rate of change of signal or
cycles per second (Hz)

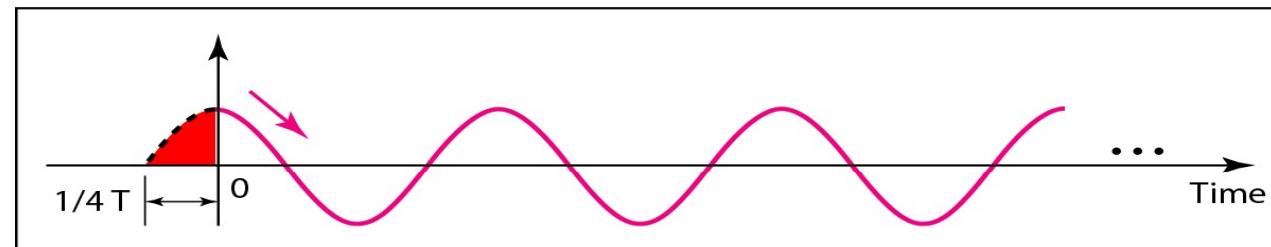
period = time for one repetition (T)

φ : phase = relative position in time

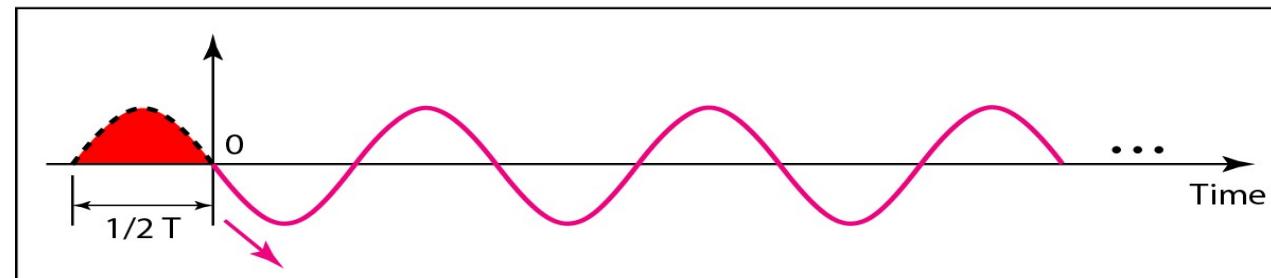
3 sine waves with different phases



a. 0 degrees



b. 90 degrees



c. 180 degrees

Phase unit: degree (360°) or radians (2π rad)

Periodic analog signals

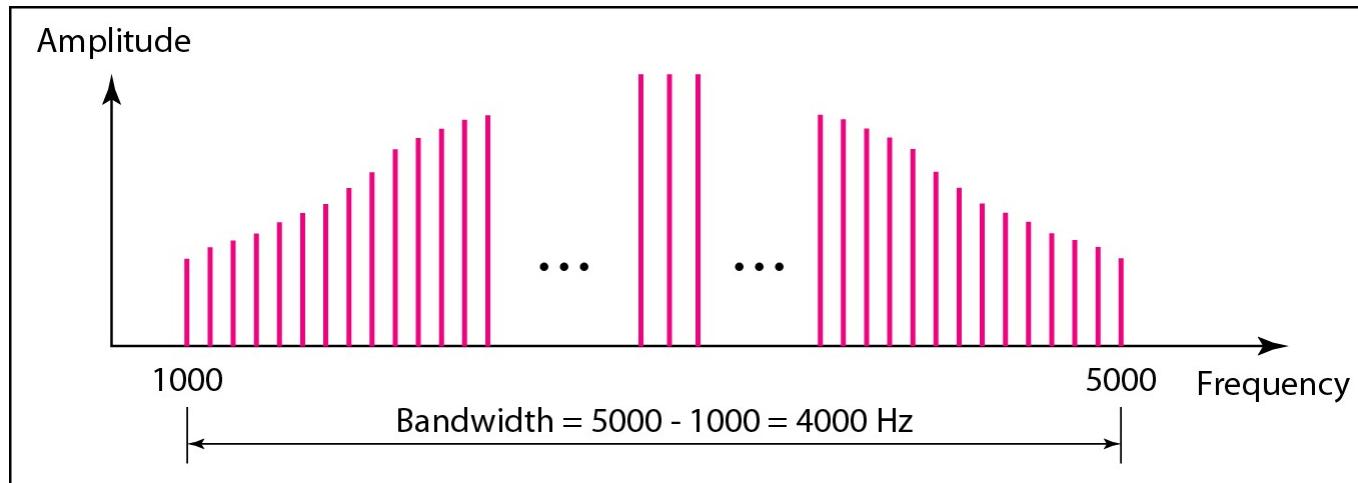
Note

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.

Note

According to Fourier analysis, any composite signal is a combination of simple sine waves with different frequencies, amplitudes, and phases.

Bandwidth



a. Bandwidth of a periodic signal

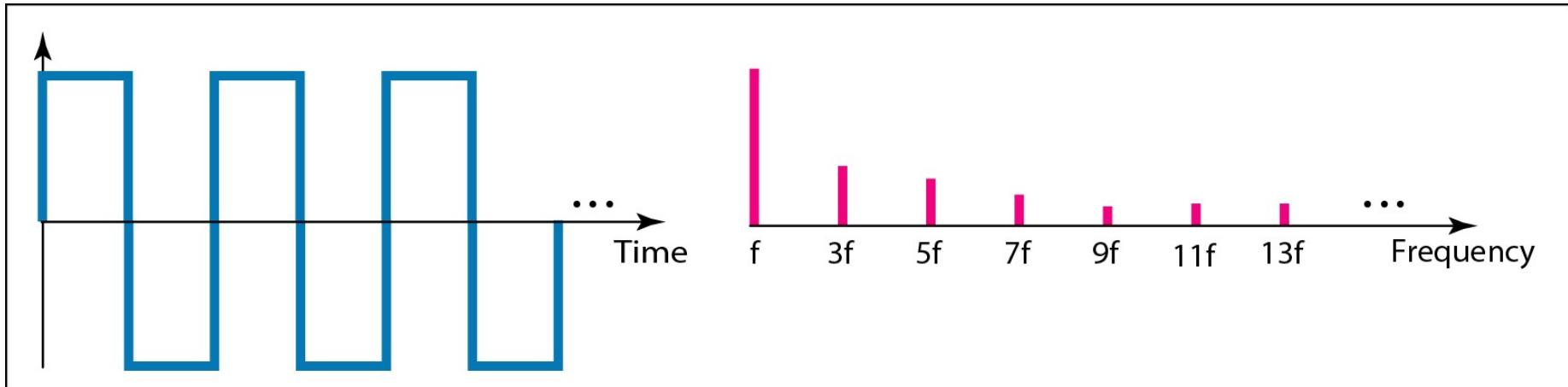
Note

The **bandwidth** of a composite signal is the difference between the highest and the lowest frequencies contained in that signal.

Digital Signals

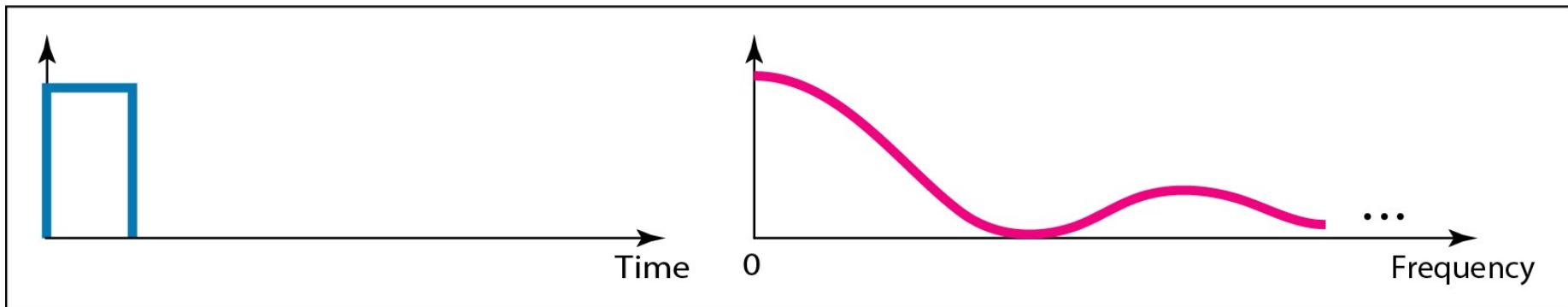
- In addition to being represented by an analog signal, information can also be represented by a **digital signal**.
- For example, a 1 can be encoded as a positive voltage and a 0 as zero voltage.
- **A digital signal can have more than two levels. In this case, we can send more than 1 bit for each level.**

Digital Signals



a. Time and frequency domains of periodic digital signal

Periodic digital signal



b. Time and frequency domains of nonperiodic digital signal

Nonperiodic digital signal

Example

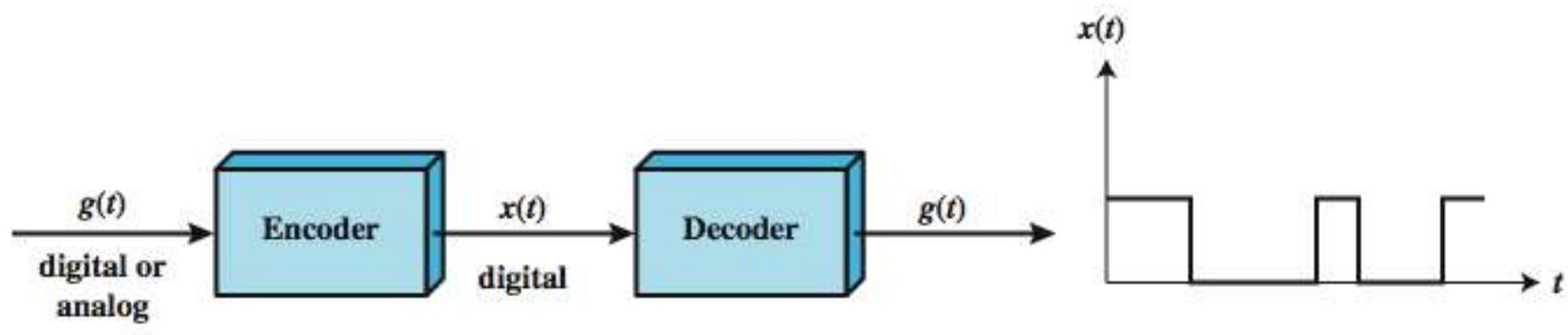
What is the bit rate for high-definition TV (HDTV)? HDTV uses digital signals to broadcast high quality video signals. The HDTV screen is normally a ratio of 16 : 9. There are 1920 by 1080 pixels per screen, and the screen is renewed 30 times per second. 24 bits represents one color pixel.

Solution:

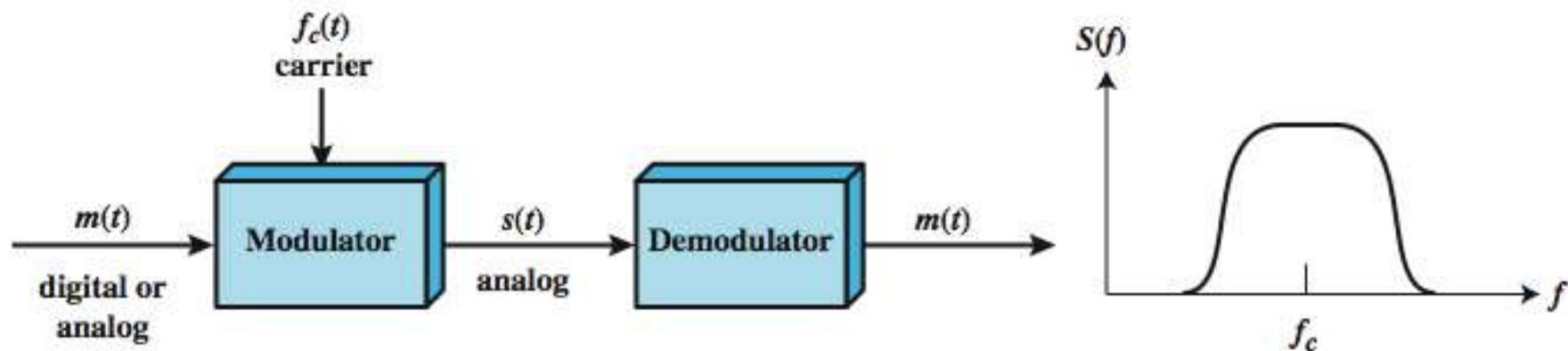
$$1920 * 1080 * 24 * 30 \text{ bit/sec} = 1.5 \text{ Gbps}$$

The TV stations reduce this rate to 20 to 40 Mbps through compression.

Encoding and Modulation Techniques



(a) Encoding onto a digital signal

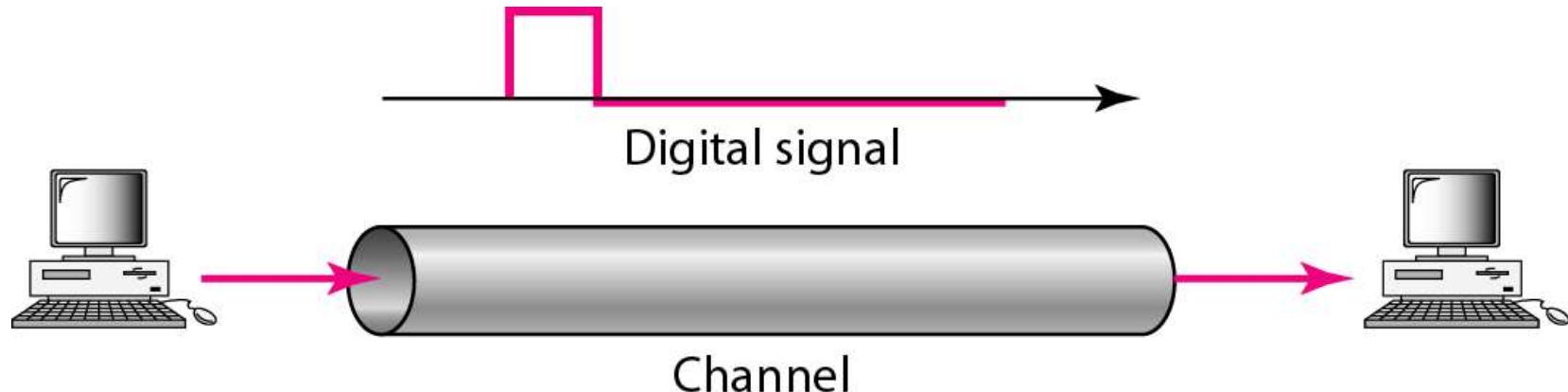


(b) Modulation onto an analog signal

Encoding and Modulation Techniques

- There are two types of Digital Data Transmission:
 - 1) **Base-Band data transmission**
 - Uses low frequency carrier signal to transmit the data
 - 2) **Band-Pass data transmission**
 - Uses high frequency carrier signal to transmit the data

Baseband Transmission



Baseband Transmission: sending digital signal without changing digital to an analog signal.

Baseband communication requires a *low-pass channel*

Example: 10Base5, 100BaseT, 1000BaseT

Baseband Transmission

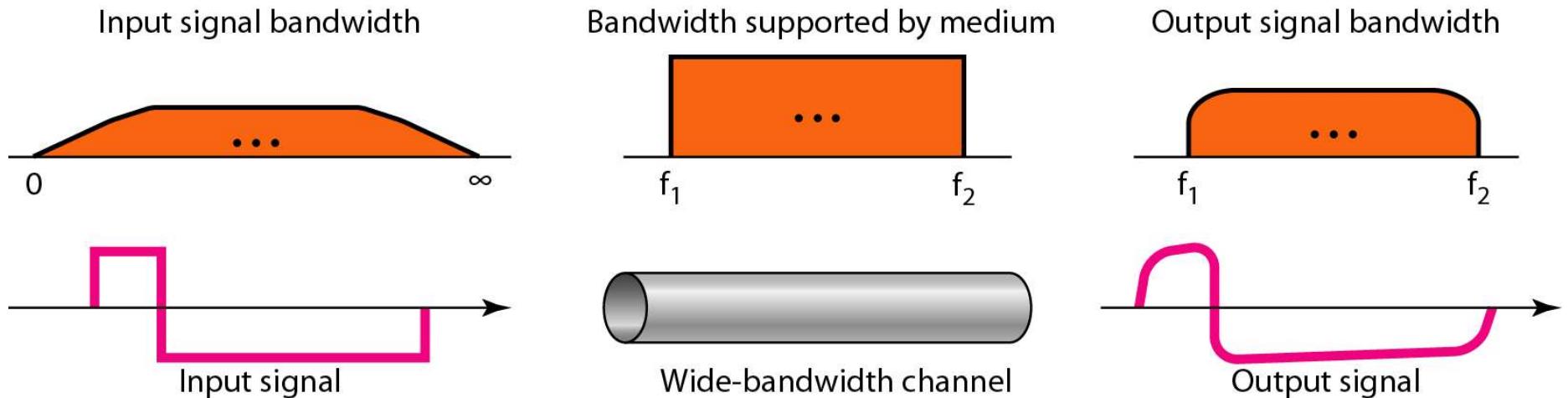
- **Baseband channel (or lowpass channel)**

a communication channel that can transfer frequencies that are very near zero.

- **Baseband bandwidth**

equal to the highest frequency of a signal or system, or an upper bound on such frequencies

Baseband transmission using a dedicated medium



Note

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.

Baseband Transmission

Note

In baseband transmission, the required bandwidth is proportional to the bit rate; if we need to send bits faster, we need more bandwidth.

Baseband Transmission

Table *Bandwidth requirements for different bit rate*

Bit Rate	Harmonics 1	Harmonics 1,3	Harmonics 1,3,5
1 kbps	B=500 Hz	B=1.5 KHz	B=2.5 KHz
10 kbps	B=5k Hz	B=15 kHz	B=25 kHz
100 kbps	B=50 kHz	B=150 kHz	B=250 kHz

Baseband Transmission

What is the required bandwidth of a low-pass channel if we need to send 1 Mbps by using baseband transmission?

Solution

The answer depends on the accuracy desired.

- a. The minimum bandwidth, is $B = \text{bit rate} / 2$, or 500 kHz.***
- b. A better solution is to use the first and the third harmonics with $B = 3 \times 500 \text{ kHz} = 1.5 \text{ MHz}$.***
- c. Still a better solution is to use the first, third, and fifth harmonics with $B = 5 \times 500 \text{ kHz} = 2.5 \text{ MHz}$.***

Baseband Transmission

We have a low-pass channel with bandwidth 100 kHz. What is the maximum bit rate of this channel?

Solution

The maximum bit rate can be achieved if we use the first harmonic. The bit rate is 2 times the available bandwidth, or 200 kbps.

Digital Data, Digital Signal Encoding

- **Digital Data:**

- **Sequence of bits to transmit**

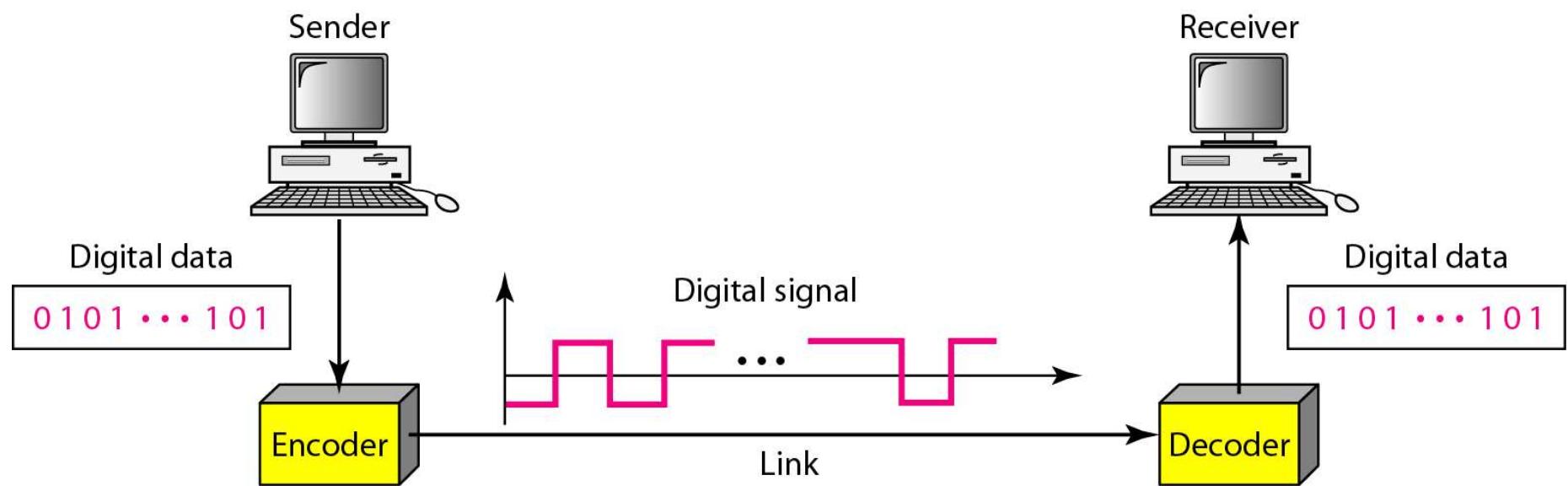
- **Physical signal:**

- **Physical value e.g. voltage, changing in discrete time epochs.**

- **Encoding (channel coding):**

- **Mapping of bit sequences to signals. This can be done in many ways...**

Digital Data, Digital Signal Encoding



Digital Data, Digital Signal Encoding

■ Interpreting Signals

□ Need to know

- Timing of bits - when they start and end
- Signal levels

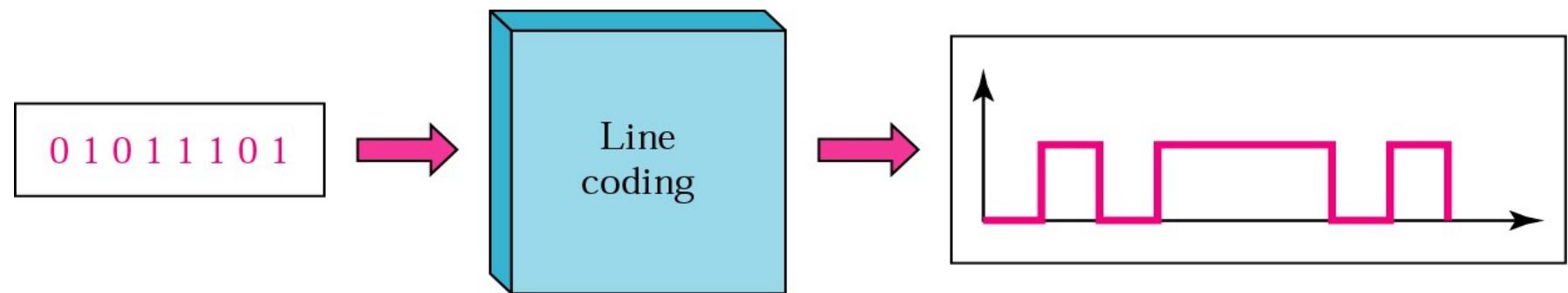
□ Factors affecting successful interpreting of signals

- Signal to noise ratio
- Data rate
- Bandwidth

■ Issues

- Bit timing
- Recovery from signal
- Noise immunity

Digital Data, Digital Signal Encoding



Line coding is the process of converting binary data, a sequence of bits, to a digital signal.

Digital Data, Digital Signal Encoding

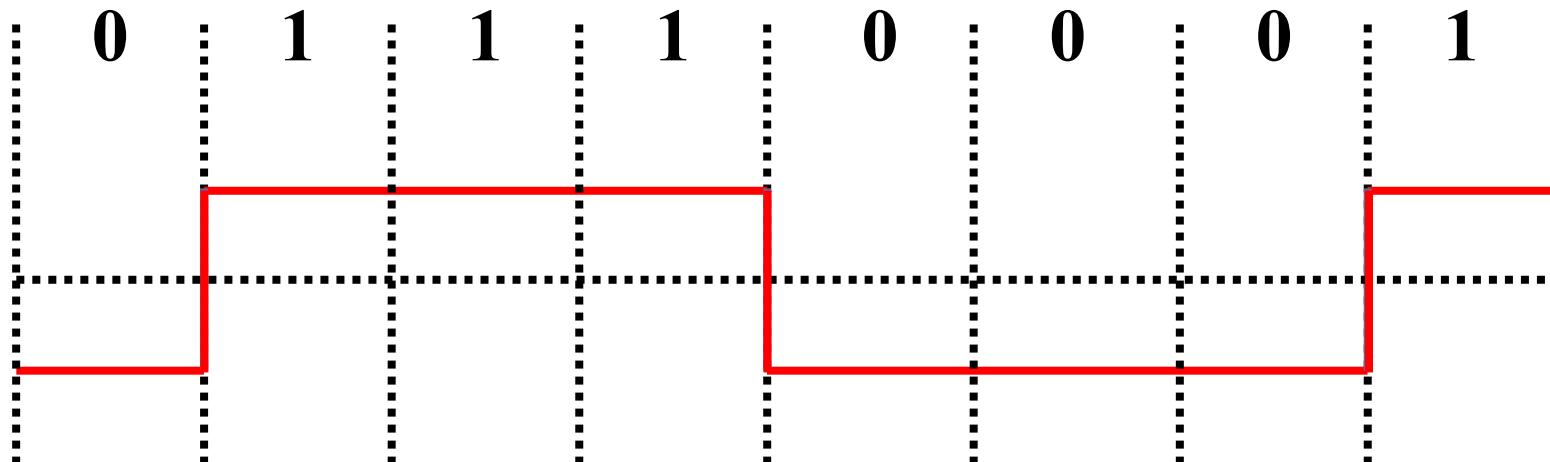
■ Encoding Schemes

- **Nonreturn to Zero-Level (NRZ-L)**
- **Nonreturn to Zero Inverted (NRZI)**
- **Manchester**
- **Differential Manchester**
-

Digital Data, Digital Signal Encoding

■ NRZ-L (Nonreturn-to-Zero-Level)

- 0 = low level
- 1 = high level
- Or exactly the other way round... ☺

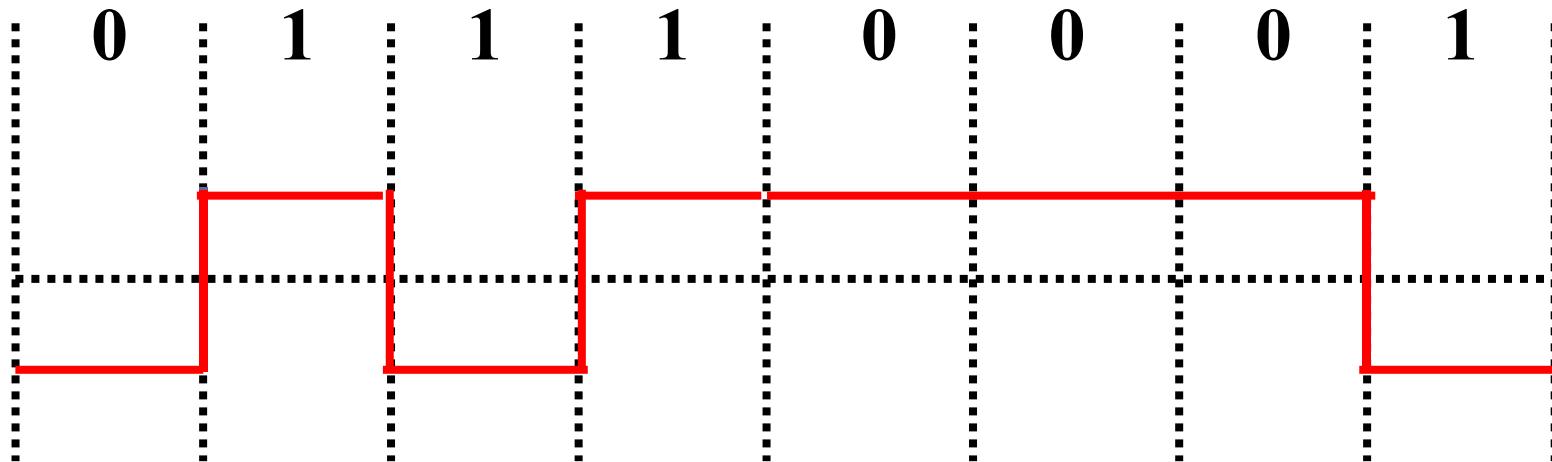


问题：接收方如何知道有几个“1”和“0”？

Digital Data, Digital Signal Encoding

■ NRZI (Nonreturn-to-Zero-Inverted) - *polarity not important!!!*

- 0 = no transition at beginning of interval
- 1 = transition at beginning of interval



问题：接收方如何知道有几个“0”？

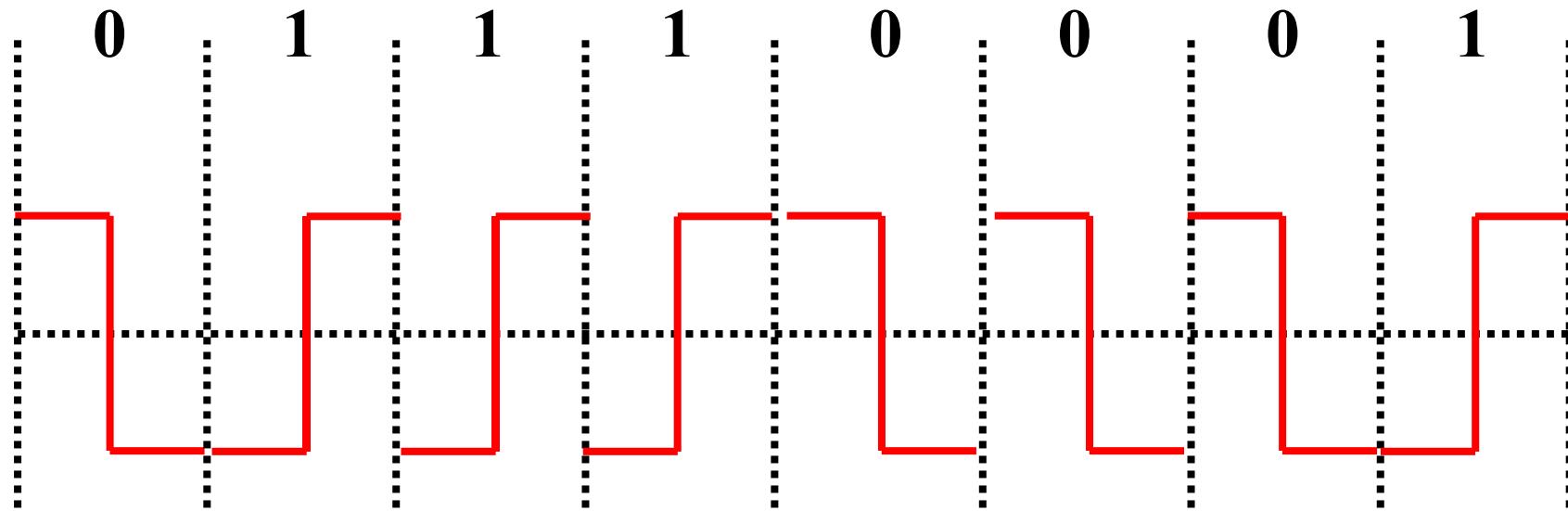
Digital Data, Digital Signal Encoding

- **Manchester:** *Solves the problem of long constant values...*

- 0 = transition **high → low** in middle of interval
- 1 = transition **low → high** in middle of interval

Note: Bit rate lower than baud rate!!

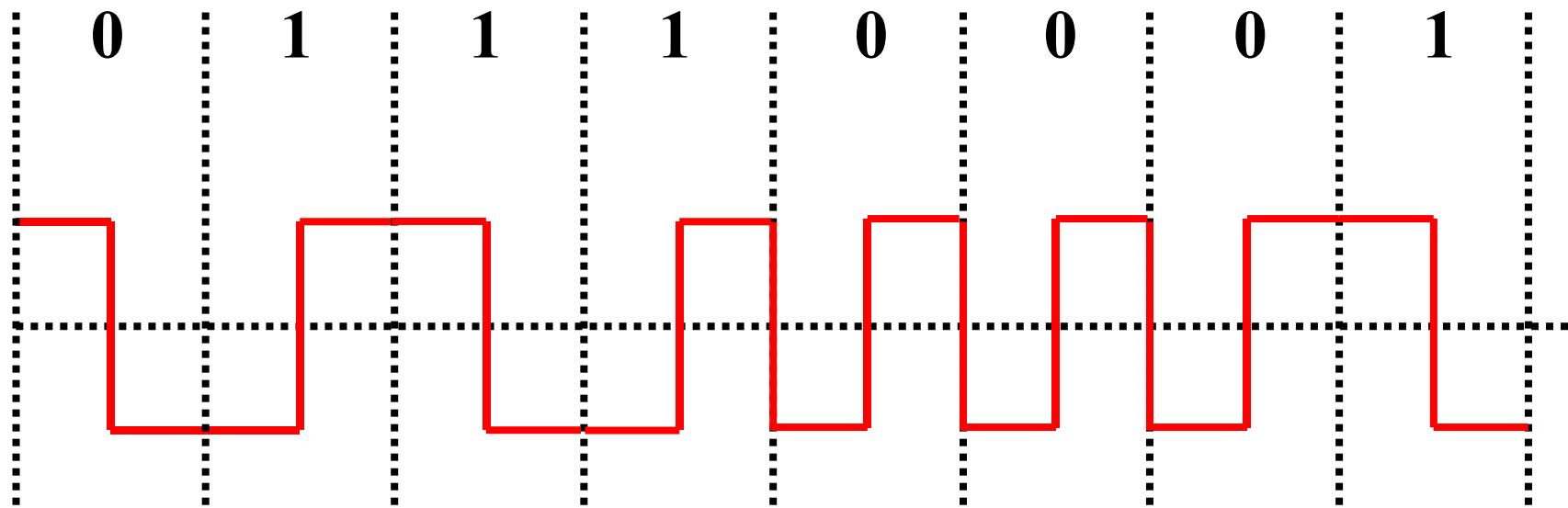
The direction of the mid-bit transition represents the digital data.



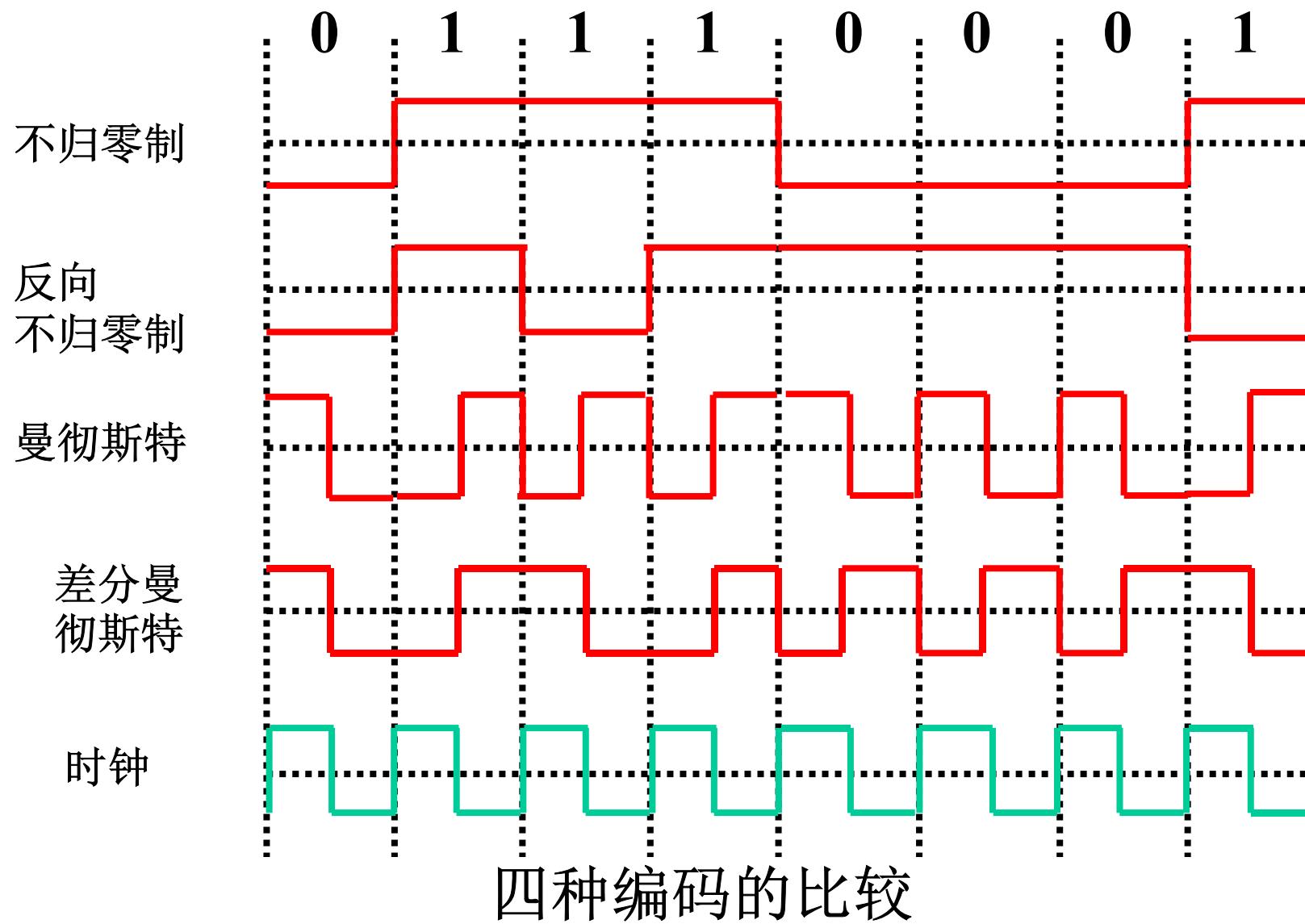
Digital Data, Digital Signal Encoding

- **Differential Manchester:** *as above, plus unimportant polarity!*

- Always a transition in middle of interval
 - mid-bit transition is **ONLY** for clocking.
 - **0 = transition at beginning of interval**
 - **1 = no transition at beginning of interval**



Digital Data, Digital Signal Encoding



NRZ Pros and Cons

■ Pros

- Easy to engineer
- Make good use of bandwidth

■ Cons

- dc component
- Lack of synchronization capability

■ Used for magnetic recording

■ Not often used for signal transmission

Manchester Pros and Cons

■ Pros

- **Synchronization on mid bit transition (self clocking)**

- **No dc component**

- **Error detection**

■ Con

- **At least one transition per bit time and possibly two**

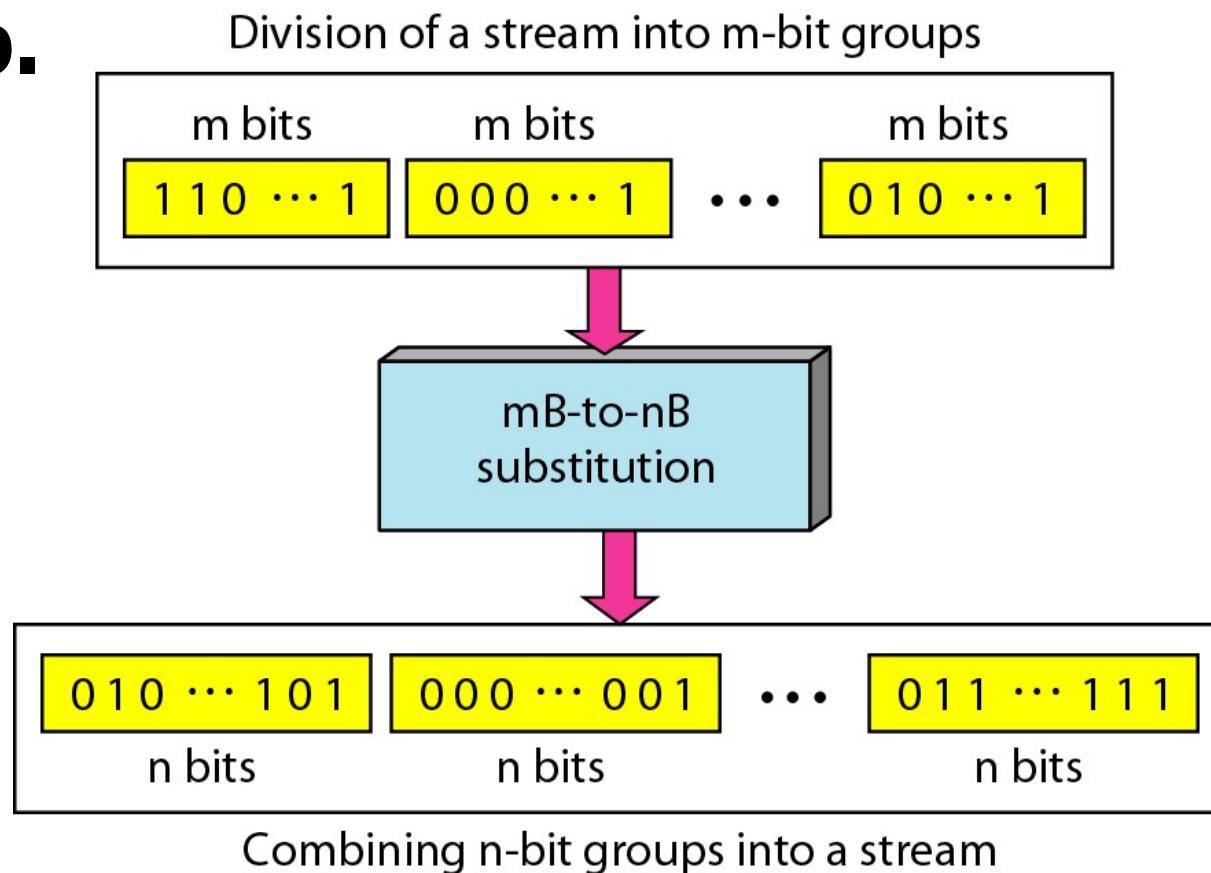
- **Maximum modulation rate is twice NRZ**

- **Requires more bandwidth**

- **Bandwidth inefficient: 50%**

Block coding

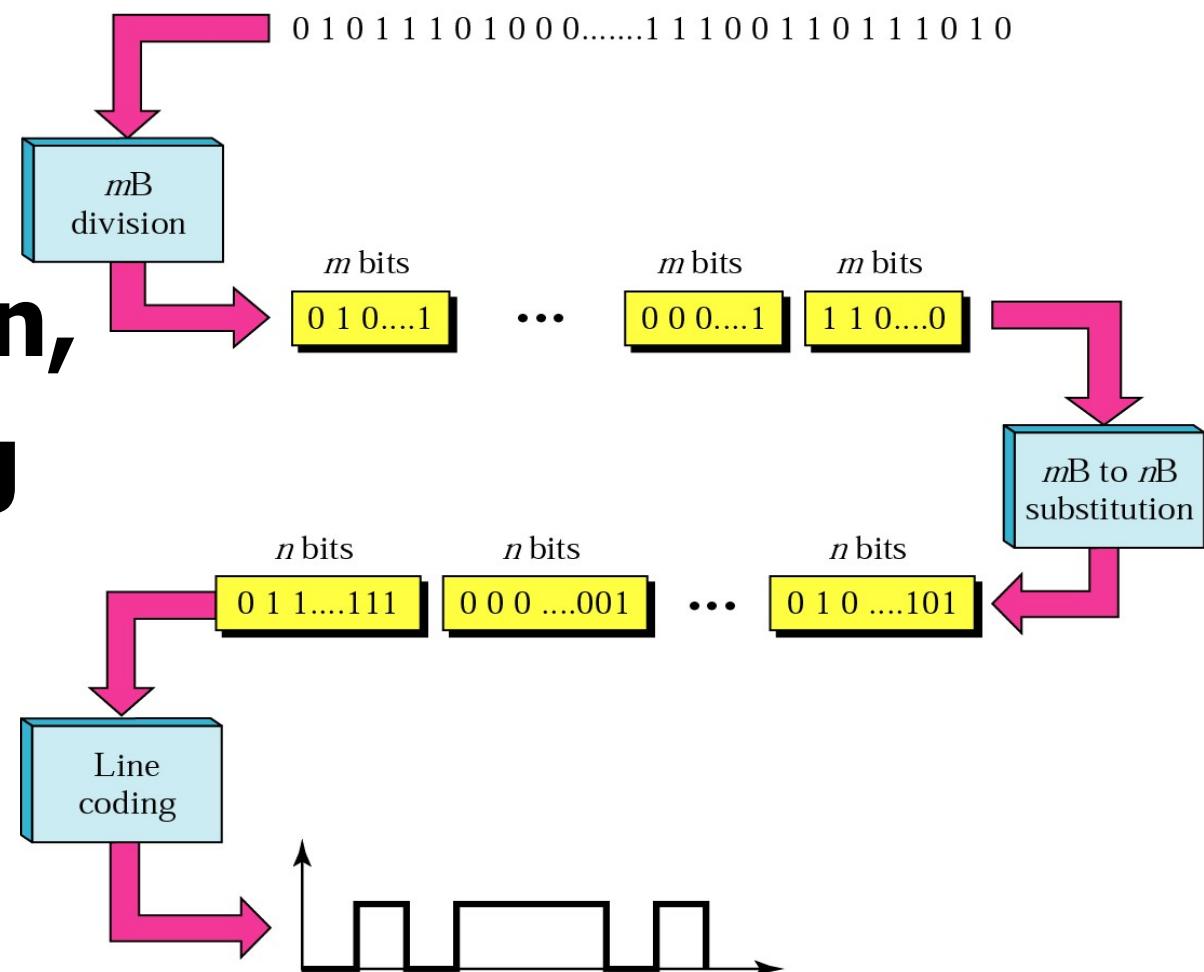
- Block coding is normally referred to as mB/nB coding;
- it replaces each **m -bit group with an n -bit group.**



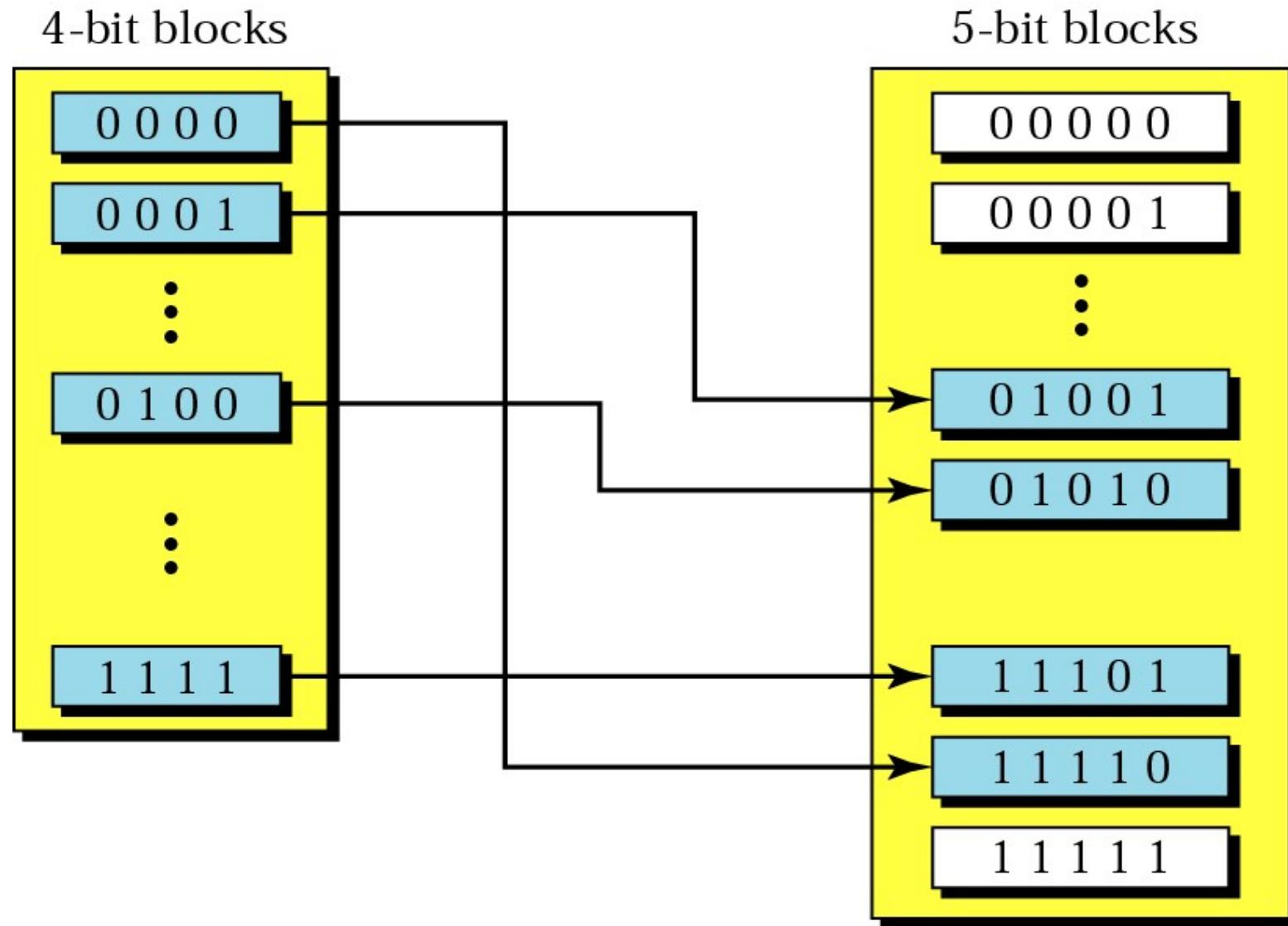
Block coding

■ Stages of operation:

- Division,
- Substitution,
- Line Coding



4B/5B Block Coding



4B/5B Mapping Codes (1/2)

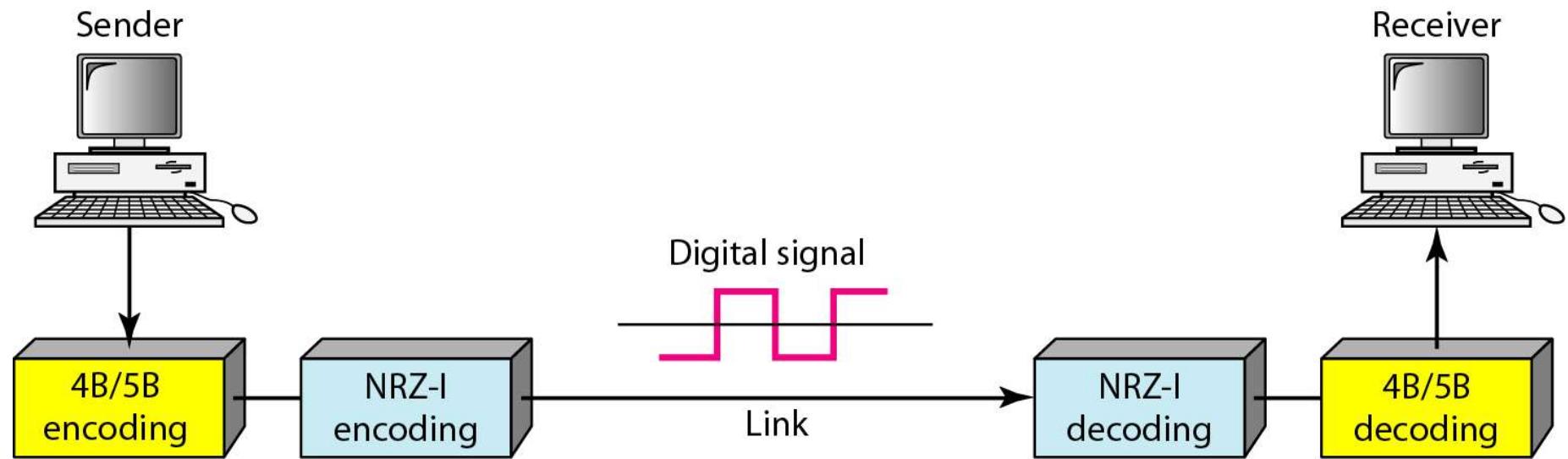
Data	Code	Data	Code
0000	11110	1000	10010
0001	01001	1001	10011
0010	10100	1010	10110
0011	10101	1011	10111
0100	01010	1100	11010
0101	01011	1101	11011
0110	01110	1110	11100
0111	01111	1111	11101

- The selection of the 5-bit code is such that each code contains no more than one leading 0 and no more than two trailing 0s.

4B/5B Mapping Codes (2/2)

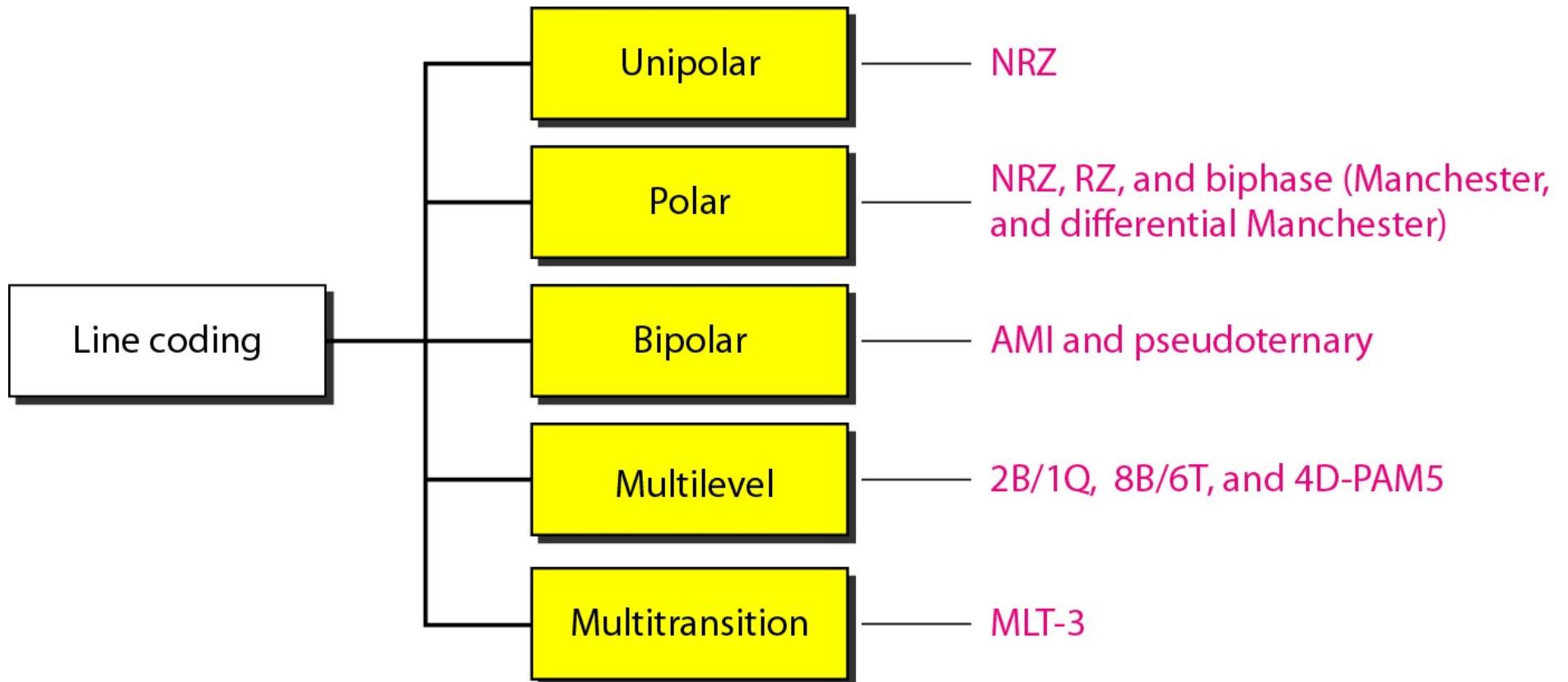
Data	Code
Q (Quiet)	00000
I (Idle)	11111
H (Halt)	00100
J (start delimiter)	11000
K (start delimiter)	10001
T (end delimiter)	01101
S (Set)	11001
R (Reset)	00111

4B/5B Block Coding



Using block coding 4B/5B with NRZ-I line coding scheme

Line coding schemes



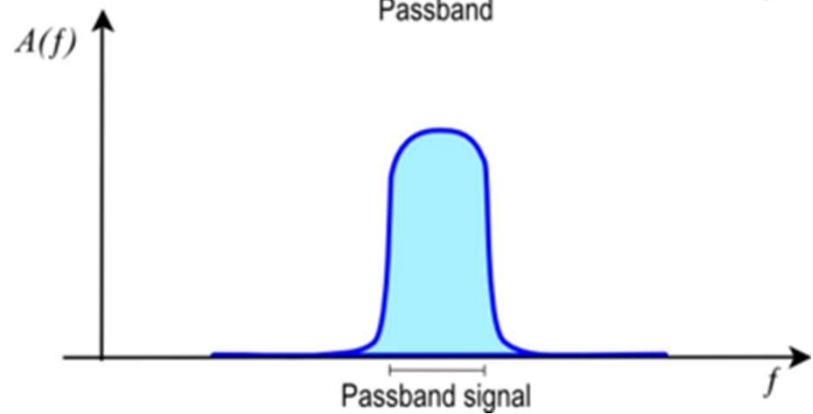
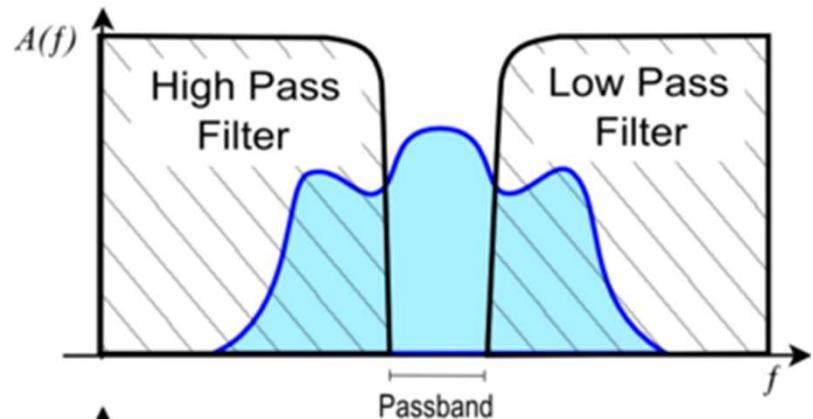
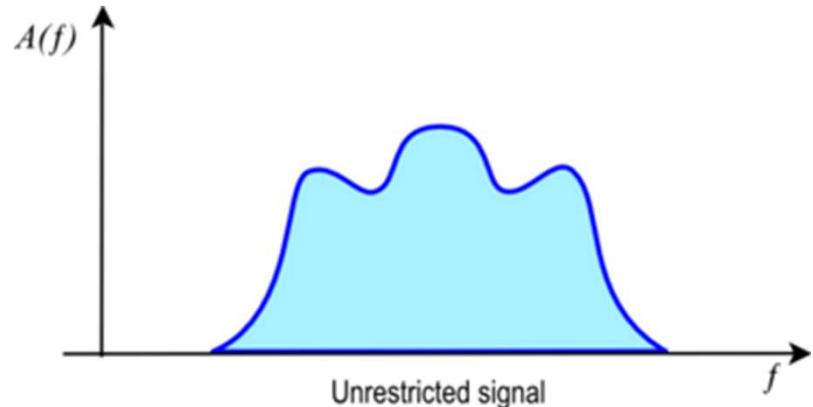
Pass-band Transmission

- Passband data transmission concerns the transmission of the data over the real passband channel.**

Pass-band

- a **passband** (通带) is the portion of the frequency spectrum that is transmitted by some filtering device.

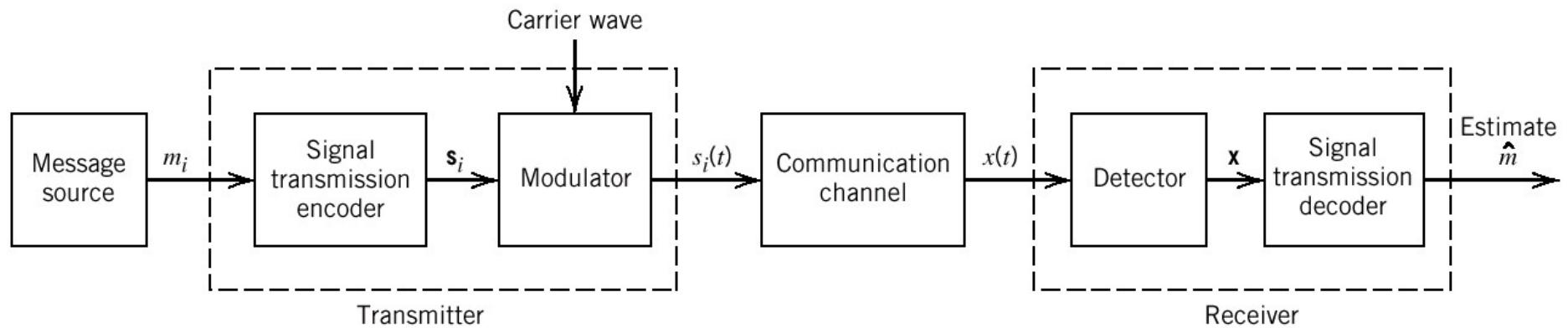
Only specific frequency bands are used.



Pass-band Transmission

- Passband **different from** the “natural spectrum of the signal”.
- The “useful” signal has to be “shifted” where desired...
- Called “**pass-band transmission**”
- We do this, using **modulation** ...

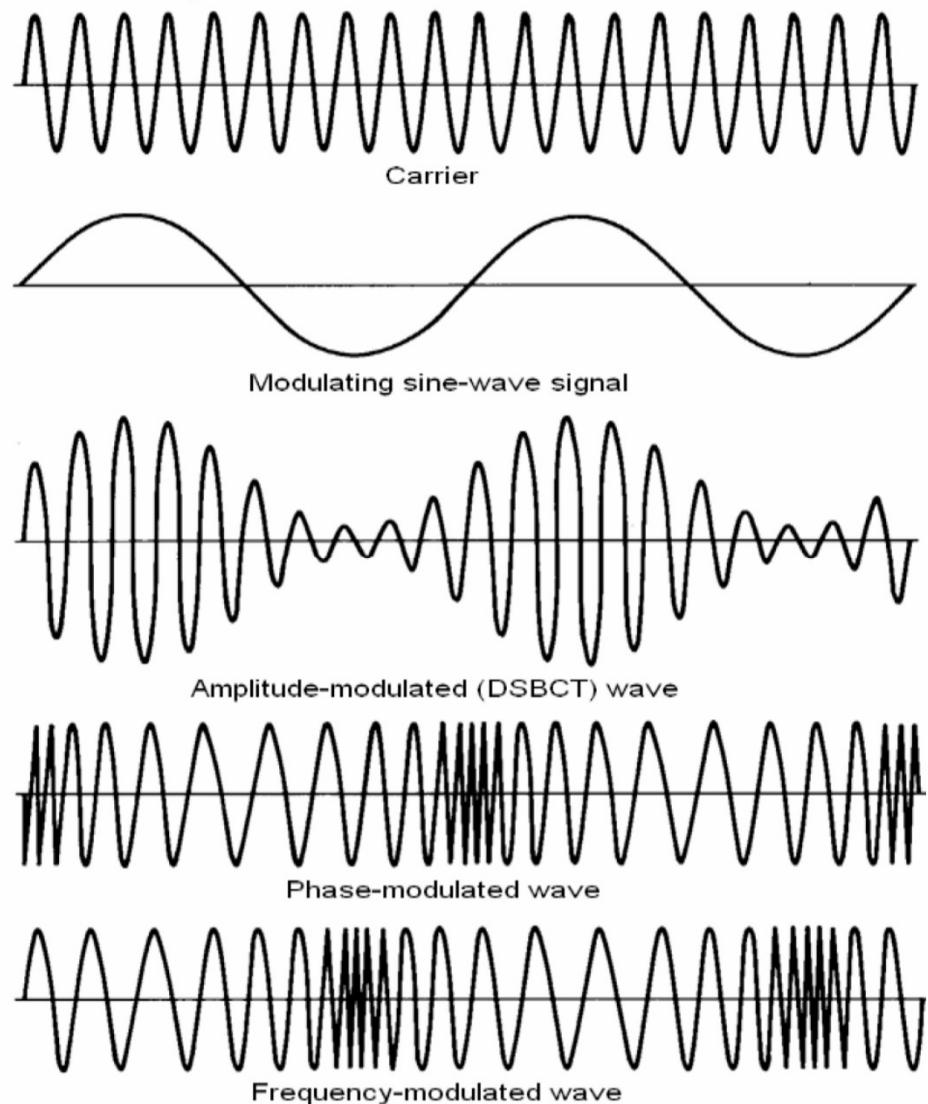
Pass-band Transmission



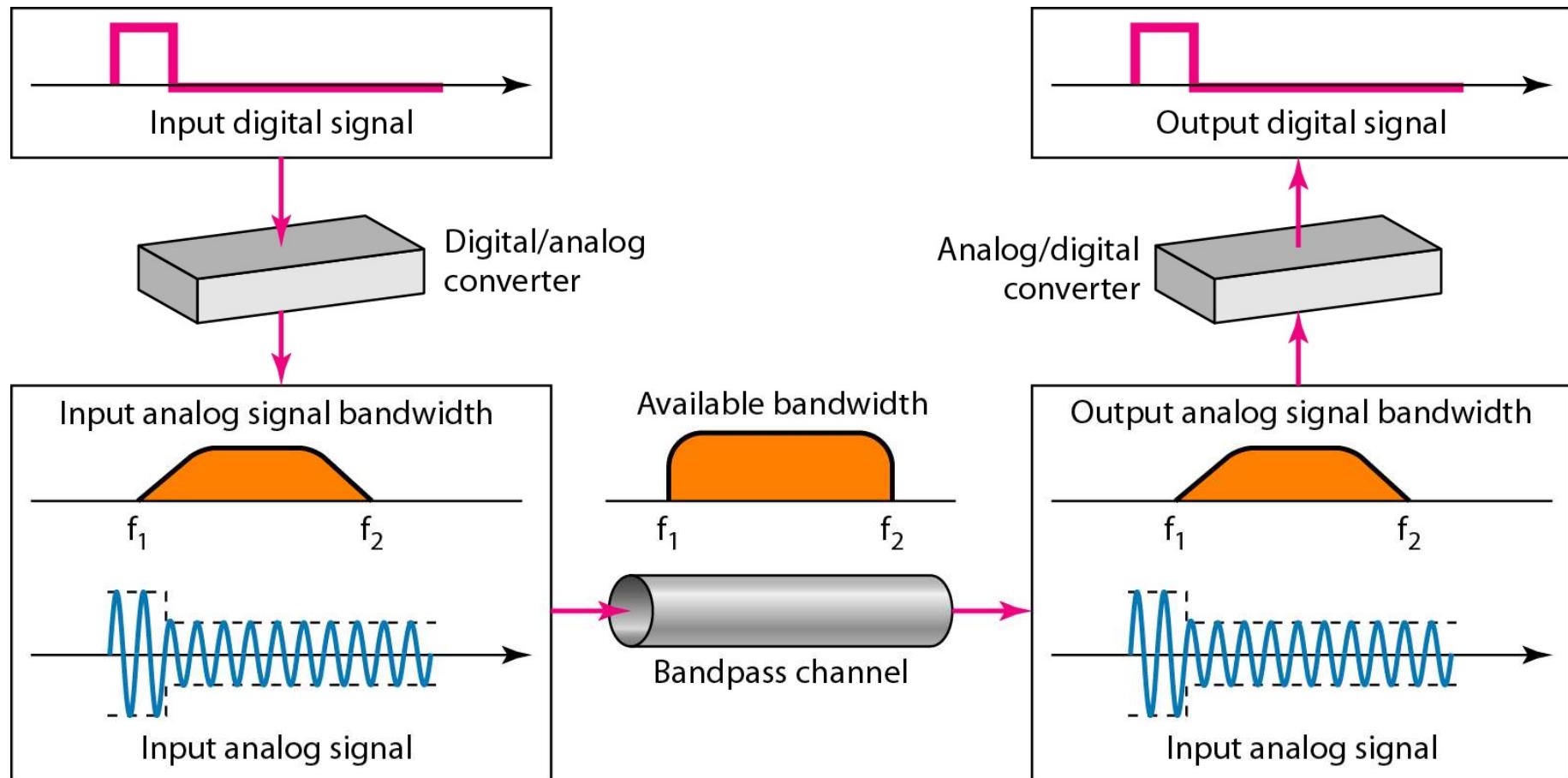
Functional model of pass-band data transmission system.

Pass-band Transmission (Analog Data and Signals!)

- We shift the frequency spectrum elsewhere...
- The different kinds of modulation can be combined



Pass-band Transmission (Digital Data and Analog Signals!)



Modulation of a digital signal for transmission on a bandpass channel

Pass-band Transmission (Digital Data and Analog Signals!)

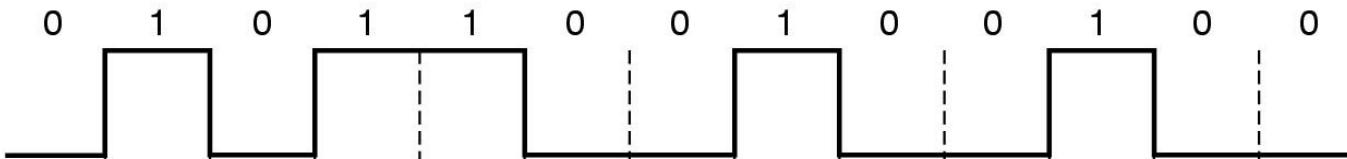
- **Digital data is encoded by modulating one of the three characteristics of the carrier:**
 - **amplitude,**
 - **frequency, or**
 - **Phase, or**
 - **some combination of these.**

Fundamental digital modulation methods

- **ASK (amplitude-shift keying):** a finite number of amplitudes are used.
- **FSK (frequency-shift keying):** a finite number of frequencies are used.
- **PSK (phase-shift keying):** a finite number of phases are used.
- **QAM (quadrature amplitude modulation):** a finite number of at least two phases and at least two amplitudes are used.



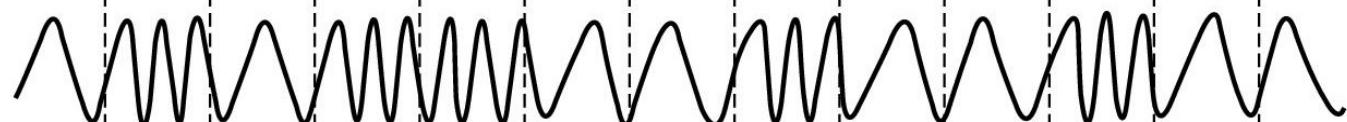
(a)



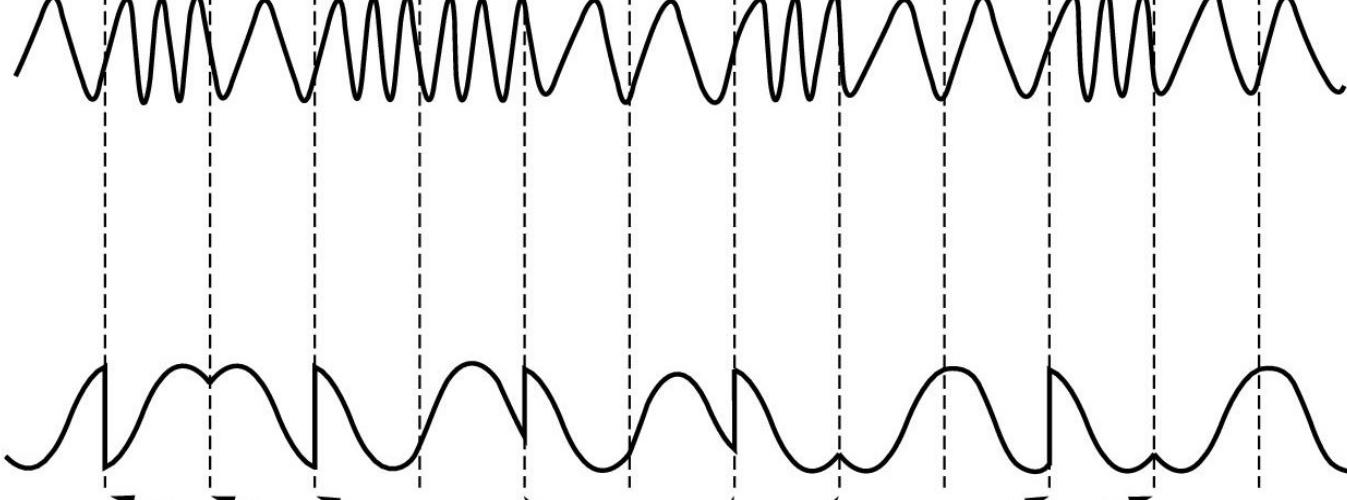
ASK (b)



FSK (c)



PSK (d)



Increasing Transmission Rates

- **Observation:** An important issue is to use low-baud modems for high transmission rates, by increasing the number of signal values—according to Nyquist, $c=2H\log_2 V$.
- **The key issue is how to present more bit information for each sample.**

Multilevel Signaling (M_{ary} Modulation)

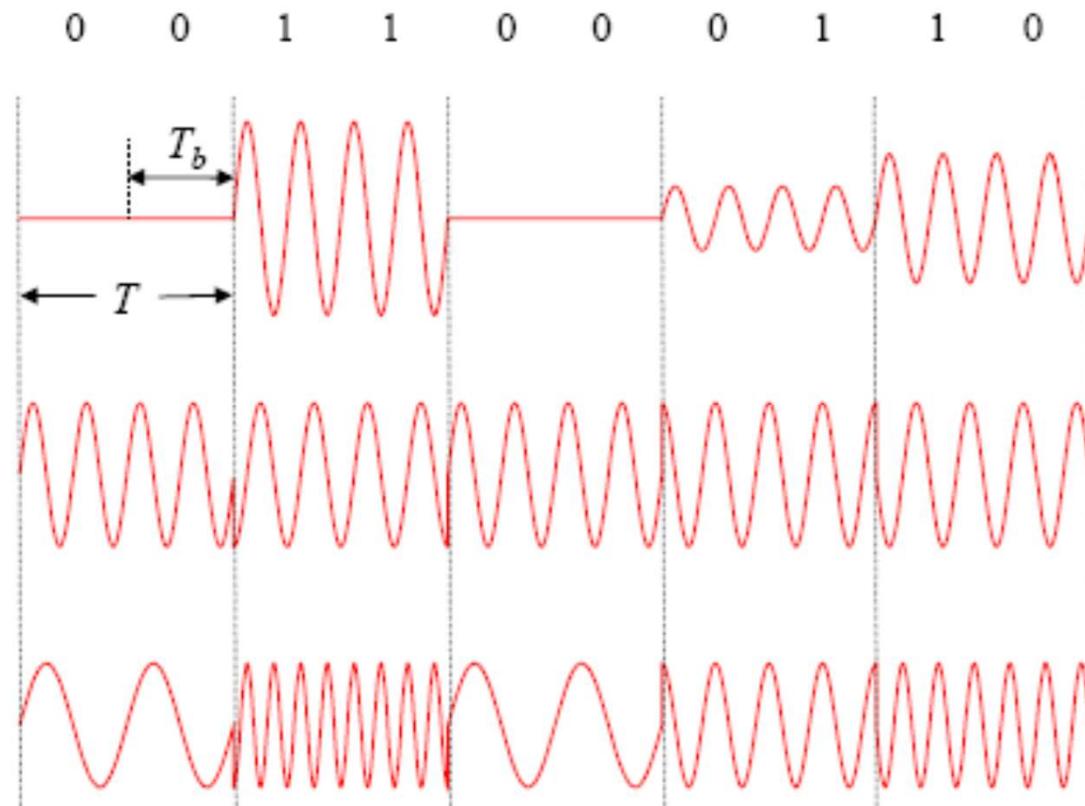
- With multilevel signaling, digital inputs with more than two modulation levels are allowed on the transmitter input.
- The data is transmitted in the form of symbols, each symbol is represented by k bits
 - → We will have $M=2^k$ different symbols

M_{ary} Modulation

- Multilevel Signaling (M_{ary} Modulation)
 - M_{ary} Amplitude Modulation
 - Changing the Amplitude using different levels
 - M_{ary} Phase Shift Keying (M_{ary} PSK)
 - Changing the phase using different levels
 - M_{ary} Frequency Shift Keying (M_{ary} FSK)
 - Changing the frequency using different levels

M_{ary} Modulation

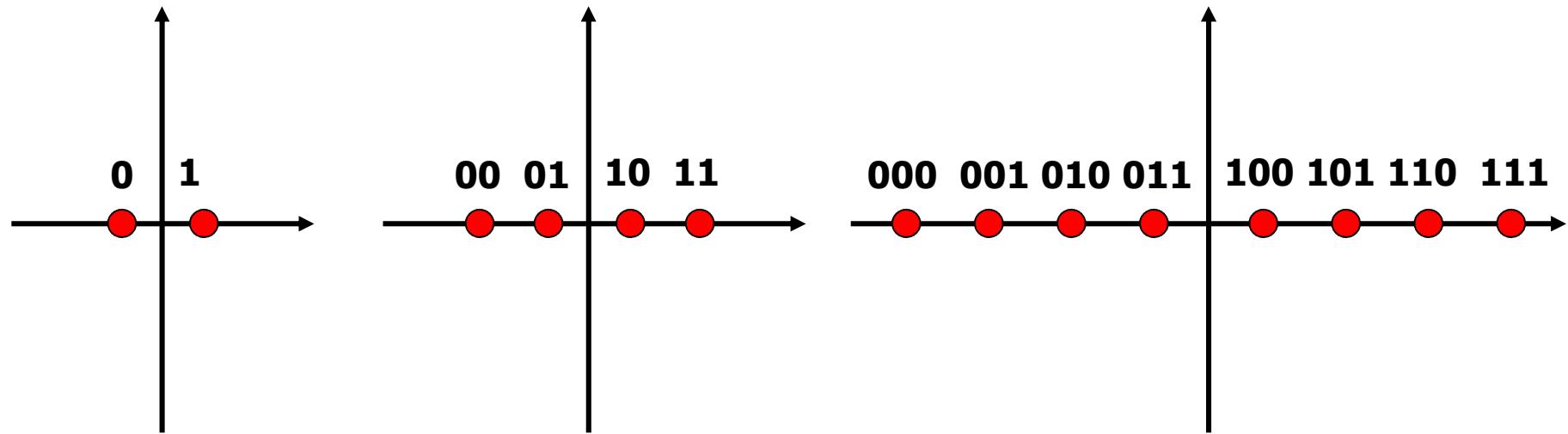
4-ary Amplitude-shift keying (ASK)



4-ary Phase-shift keying (PSK)

4-ary Frequency-shift keying (FSK)

Multiple Amplitudes (PAM)



2 “levels”

1 bits / pulse

B bits per second

4 “levels”

2 bits / pulse

$2 \times B$ bits per second

8 “levels”

3 bits / pulse

$3 \times B$ bits per second

Example

- Show how to transmit the message

m=100110001101010111

**Using 8_{ary} Pulse Amplitude Modulation.
Find the corresponding amplitudes of
the transmitted signal and calculate the
difference between the symbols. Given
that the maximum amplitude is 4 Volts**

Example - Solution

- Since we will be using 8_{ary} modulation then the signal must be divided into symbols each of 3 bits
 - Because $2^3 = 8$
- Therefore

$m = 100 \quad 110 \quad 001 \quad 101 \quad 010 \quad 111$

$S_4 \quad S_6 \quad S_1 \quad S_5 \quad S_2 \quad S_7$

Example – Solution (Cont.)

■ Amplitude calculations

$$v_i = \frac{2A}{M-1} i - A$$

$$v_4 = \frac{2(4)}{8-1}(4) - 4 = 0.5714 \text{ volts}$$

$$v_6 = \frac{2(4)}{8-1}(6) - 4 = 2.8571 \text{ volts}$$

$$v_1 = \frac{2(4)}{8-1}(1) - 4 = -2.8571 \text{ volts}$$

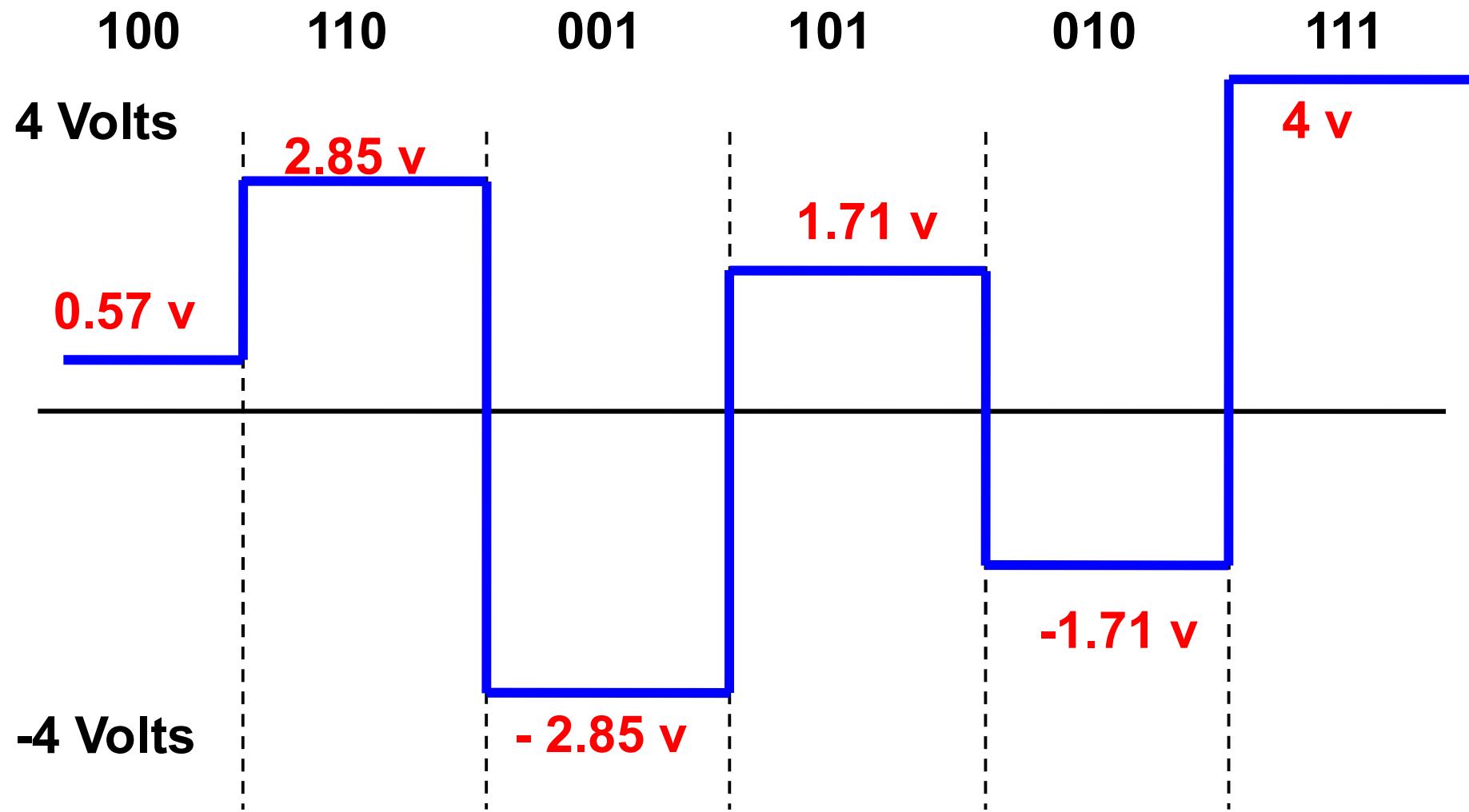
Example – Solution (Cont.)

$$v_5 = \frac{2(4)}{8-1}(5) - 4 = 1.7142 \text{ volts}$$

$$v_2 = \frac{2(4)}{8-1}(2) - 4 = -1.7142 \text{ volts}$$

$$v_7 = \frac{2(4)}{8-1}(7) - 4 = 4 \text{ volts}$$

Example – Solution (Cont.)

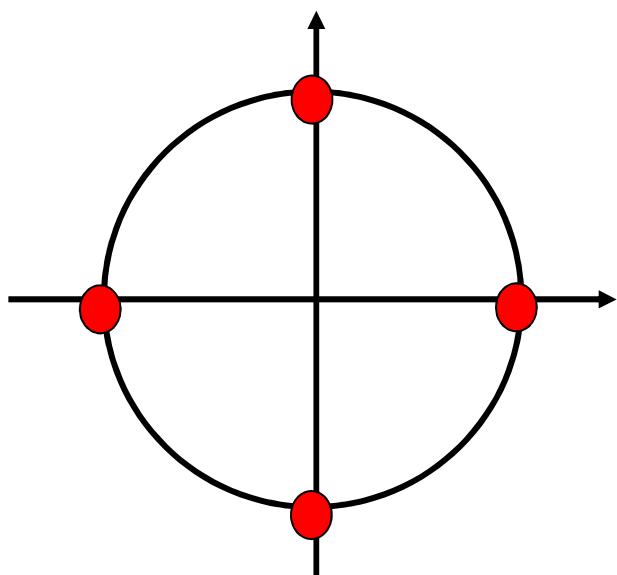


Example – Solution (Cont.)

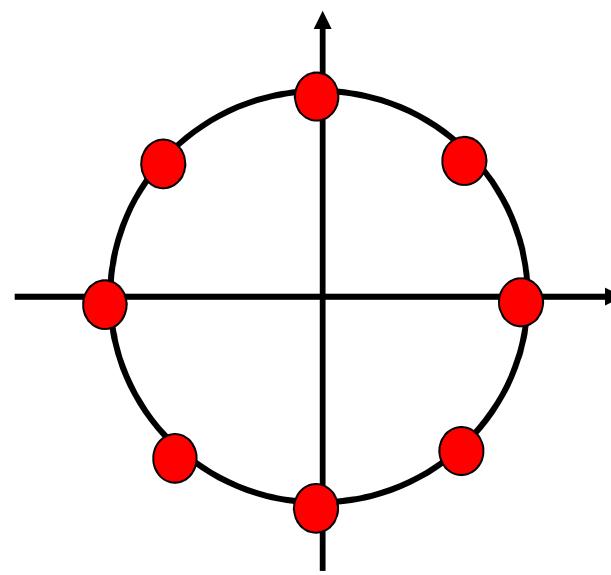
- Difference between each symbol and another can be calculated as follows:

$$\delta = \frac{2A}{M-1} = \frac{2(4)}{8-1} = 1.1428 \text{ volts}$$

Multiple Phases (MPSK)



4 “phase”
2 bits / pulse
 $2 \times B$ bits per second



8 “phases”
3 bits / pulse
 $3 \times B$ bits per second

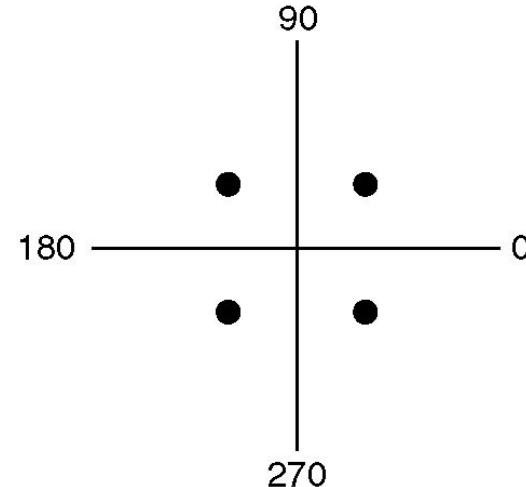
Modems

- An acronym for modulator-demodulator
- Uses a constant-frequency signal known as a carrier signal
- Converts a series of binary voltage pulses into an analog signal by modulating the carrier signal
- The receiving modem translates the analog signal back into digital data

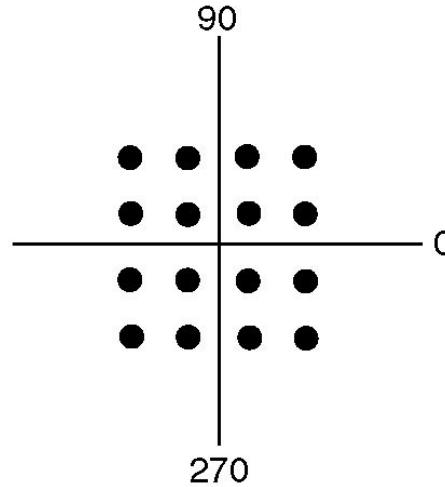
Modems

- All advanced modems use a *combination of modulation techniques* to transmit multiple bits per baud.
- Multiple amplitude and multiple phase shifts are combined to transmit several bits per symbol.
- Modems actually use Quadrature Amplitude Modulation (QAM).
- These concepts are explained using **constellation points** where a point determines a specific amplitude and phase.

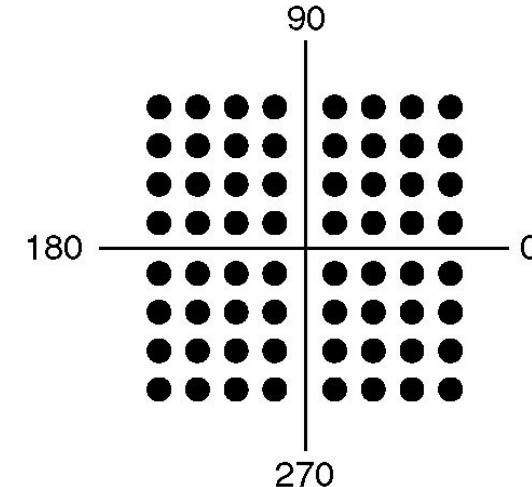
Constellation Diagrams



(a)



(b)

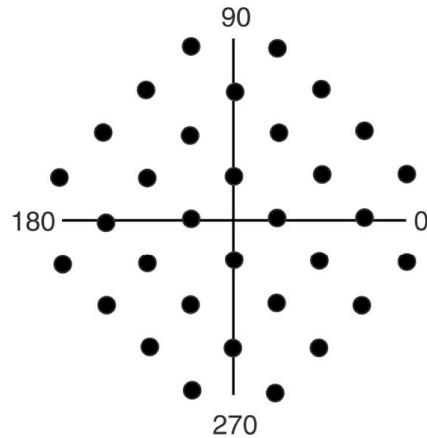


(c)

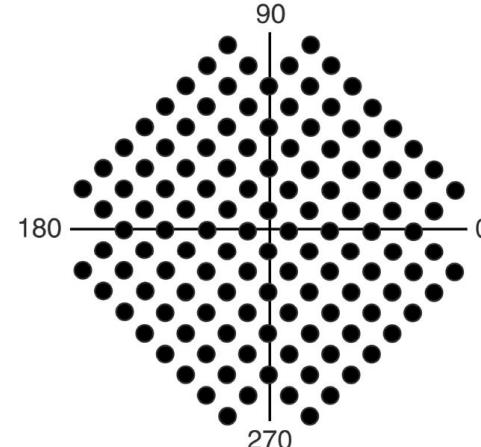
(a) QPSK.

(b) QAM-16.

(c) QAM-64.



(b)



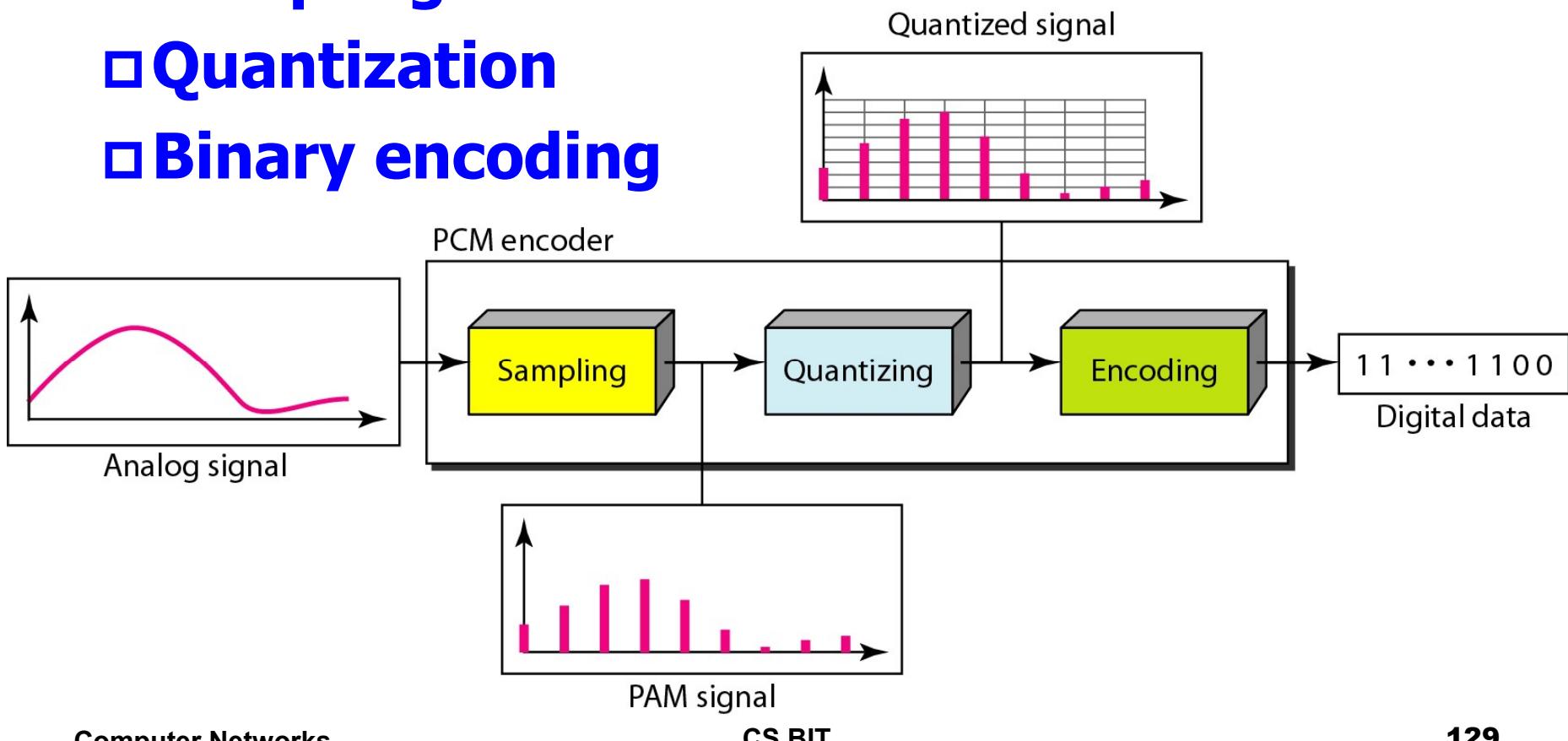
(c)

Analog-to-Digital Conversion

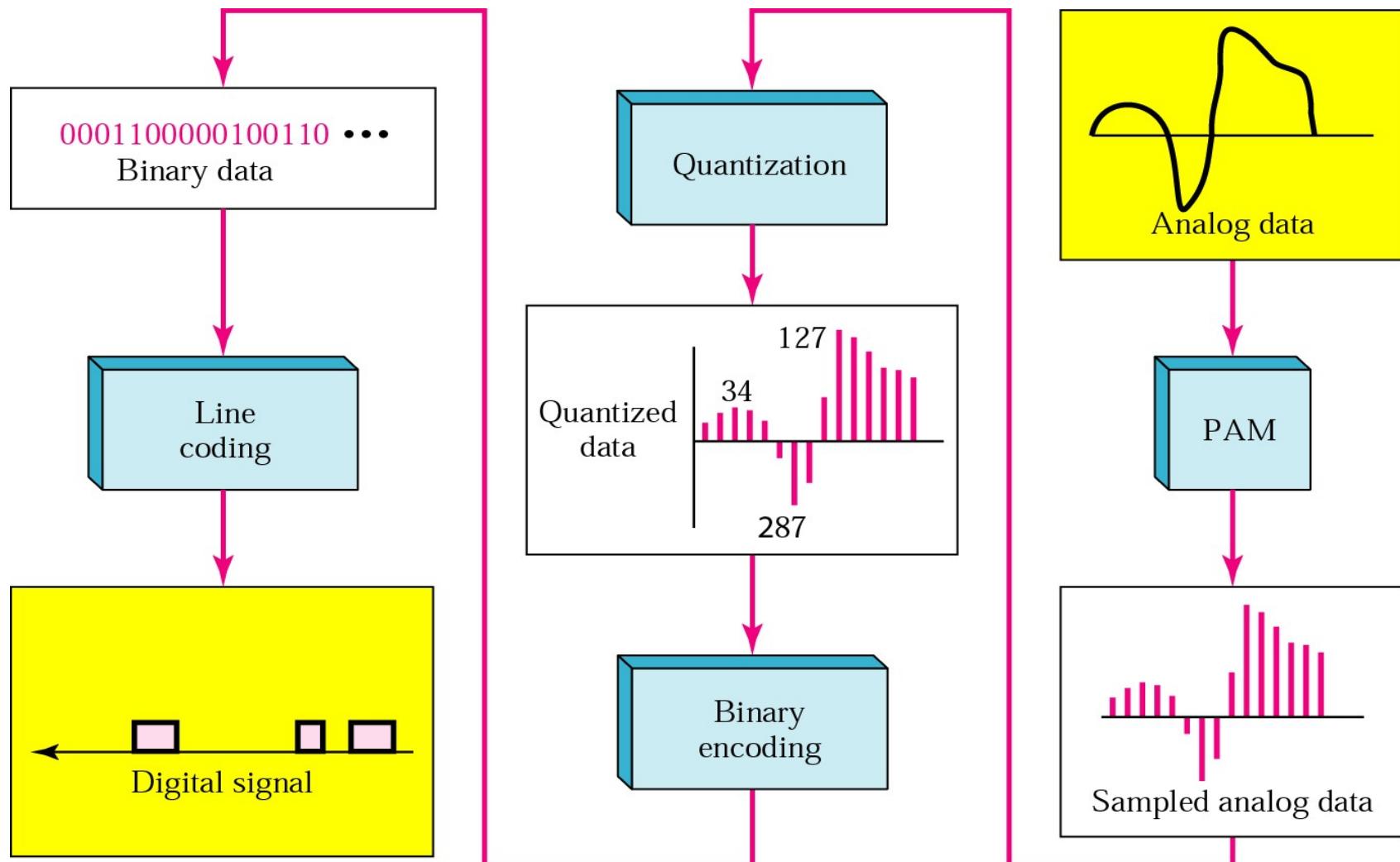
- A digital signal is superior to an analog signal because it is more robust to noise and can easily be recovered, corrected and amplified.
- For this reason, the tendency today is to change an analog signal to digital data. In this section we describe two techniques:
 - Pulse Code Modulation (PCM) and
 - Delta Modulation (DM)

PCM

- PCM consists of three steps to digitize an analog signal:
 - Sampling
 - Quantization
 - Binary encoding



PCM



From analog signal to PCM digital code

Sampling

- Analog signal is sampled every T_s secs.
- T_s is referred to as the sampling interval.
- $f_s = 1/T_s$ is called the sampling rate or sampling frequency.

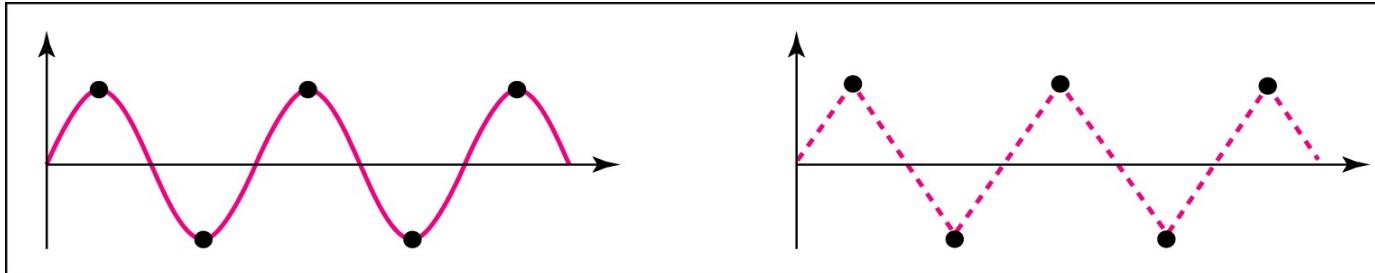
Note

According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal.

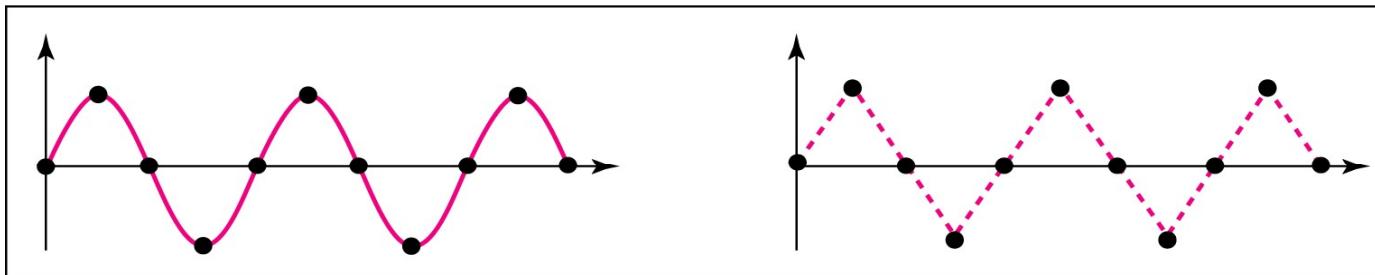
Example

- For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates:
 - $f_s = 4f$ (2 times the Nyquist rate),
 - $f_s = 2f$ (Nyquist rate), and
 - $f_s = f$ (one-half the Nyquist rate)
 - Figure shows the sampling and the subsequent recovery of the signal.

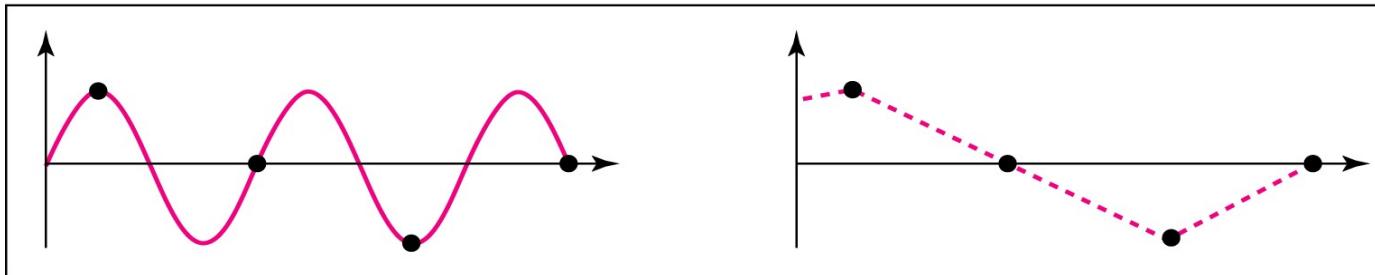
Example



a. Nyquist rate sampling: $f_s = 2 f$



b. Oversampling: $f_s = 4 f$



c. Undersampling: $f_s = f$

Recovery of a sampled sine wave for different sampling rates

Quantization

- Sampling results in a series of pulses of varying amplitude values ranging between two limits: a min and a max.
- The amplitude values are infinite between the two limits.
- We need to map the *infinite* amplitude values onto a finite set of known values.
- This is achieved by dividing the distance between min and max into **L zones**, each of height Δ .

$$\Delta = (\max - \min)/L$$

Quantization Levels

- The midpoint of each zone is assigned a value from 0 to $L-1$ (resulting in L values)
- Each sample falling in a zone is then approximated to the value of the midpoint.

Quantization Zones

- Assume we have a voltage signal with amplitudes $V_{\min} = -20V$ and $V_{\max} = +20V$.
- We want to use $L=8$ quantization levels.
- Zone width $\Delta = (20 - -20)/8 = 5$
- The 8 zones are: -20 to -15, -15 to -10, -10 to -5, -5 to 0, 0 to +5, +5 to +10, +10 to +15, +15 to +20
- The midpoints are: -17.5, -12.5, -7.5, -2.5, 2.5, 7.5, 12.5, 17.5

Assigning Codes to Zones

- Each zone is then assigned a binary code.
- The number of bits required to encode the zones, or the number of bits per sample as it is commonly referred to, is obtained as follows:

$$n_b = \log_2 L$$

- Given our example, $n_b = 3$
- The 8 zone (or level) codes are therefore:
000, 001, 010, 011, 100, 101, 110, and 111
- Assigning codes to zones:
 - 000 will refer to zone -20 to -15
 - 001 to zone -15 to -10, etc.

Bit rate and bandwidth requirements of PCM

- The bit rate of a PCM signal can be calculated from the number of bits per sample x the sampling rate

$$\text{Bit rate} = n_b \times f_s$$

- The bandwidth required to transmit this signal depends on the type of line encoding used.

Example

We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?

Solution

The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$

$$\text{Bit rate} = 8000 \times 8 = 64,000 \text{ bps} = 64 \text{ kbps}$$

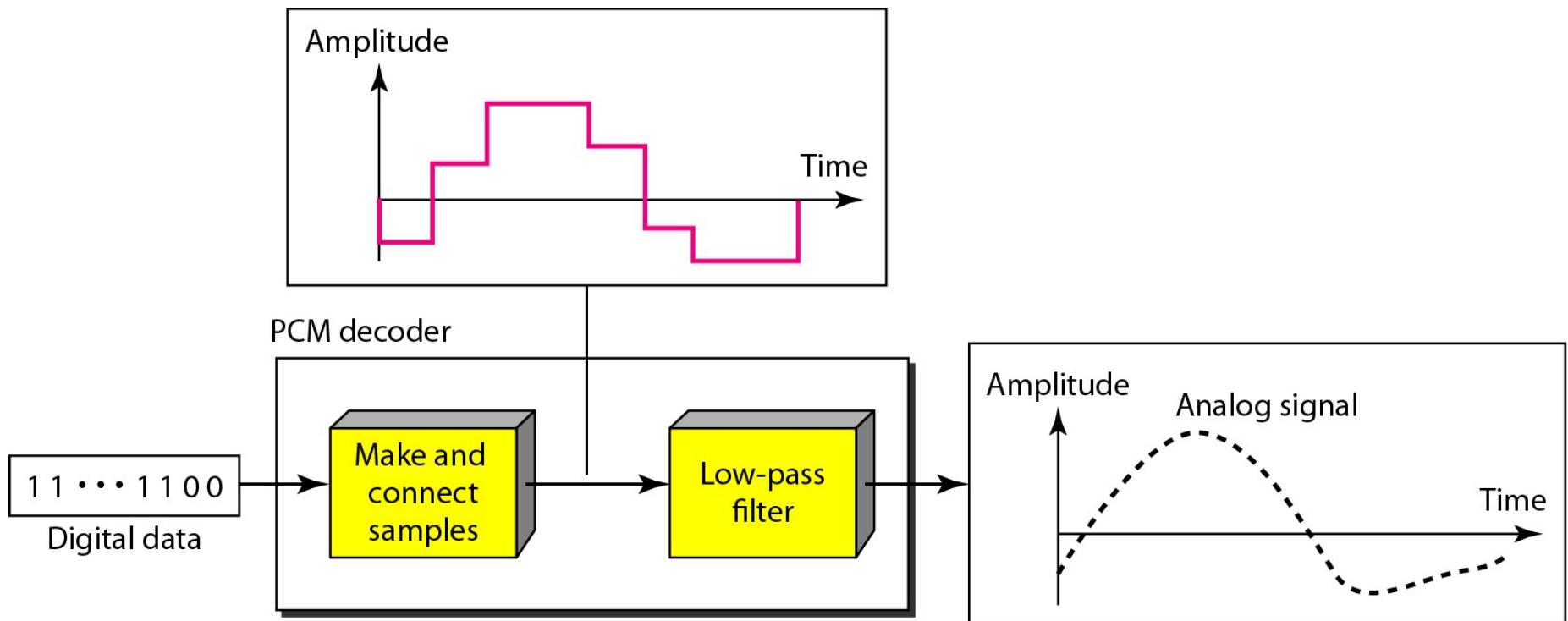
Example

- A digitized signal will always need more bandwidth than the original analog signal.
- We have a low-pass analog signal of 4 kHz.
 - If we send the analog signal, we need a channel with a minimum bandwidth of 4 kHz.
 - If we digitize the signal and send 8 bits per sample, we need a channel with a minimum bandwidth of
 $8 \times 4 \text{ kHz} = 32 \text{ kHz}$.

PCM Decoder

- To recover an analog signal from a digitized signal we follow the following steps:
 - We use a hold circuit that holds the amplitude value of a pulse till the next pulse arrives.
 - We pass this signal through a low pass filter with a cutoff frequency that is equal to the highest frequency in the pre-sampled signal.
- The higher the value of L, the less distorted a signal is recovered.

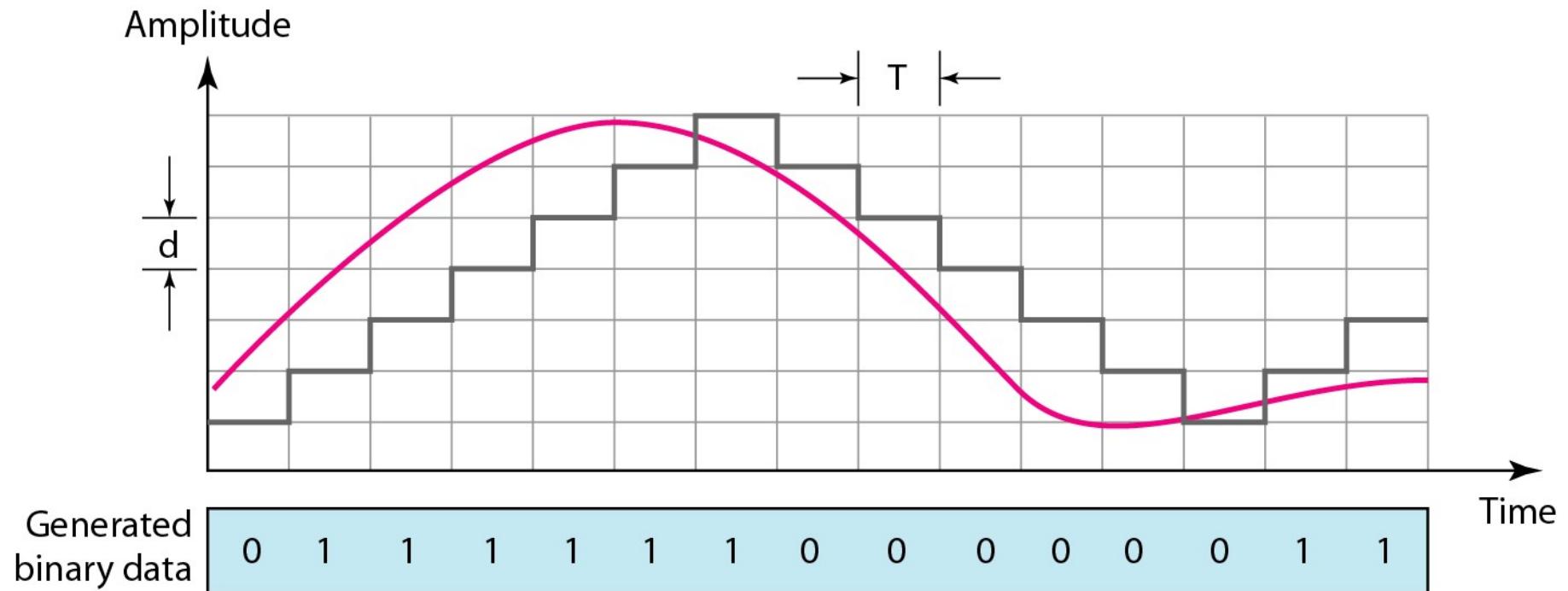
PCM Decoder



Delta Modulation

- This scheme sends only the difference between pulses, if the pulse at time t_{n+1} is higher in amplitude value than the pulse at time t_n , then a single bit, say a “1”, is used to indicate the positive value.
- If the pulse is lower in value, resulting in a negative value, a “0” is used.

Delta Modulation



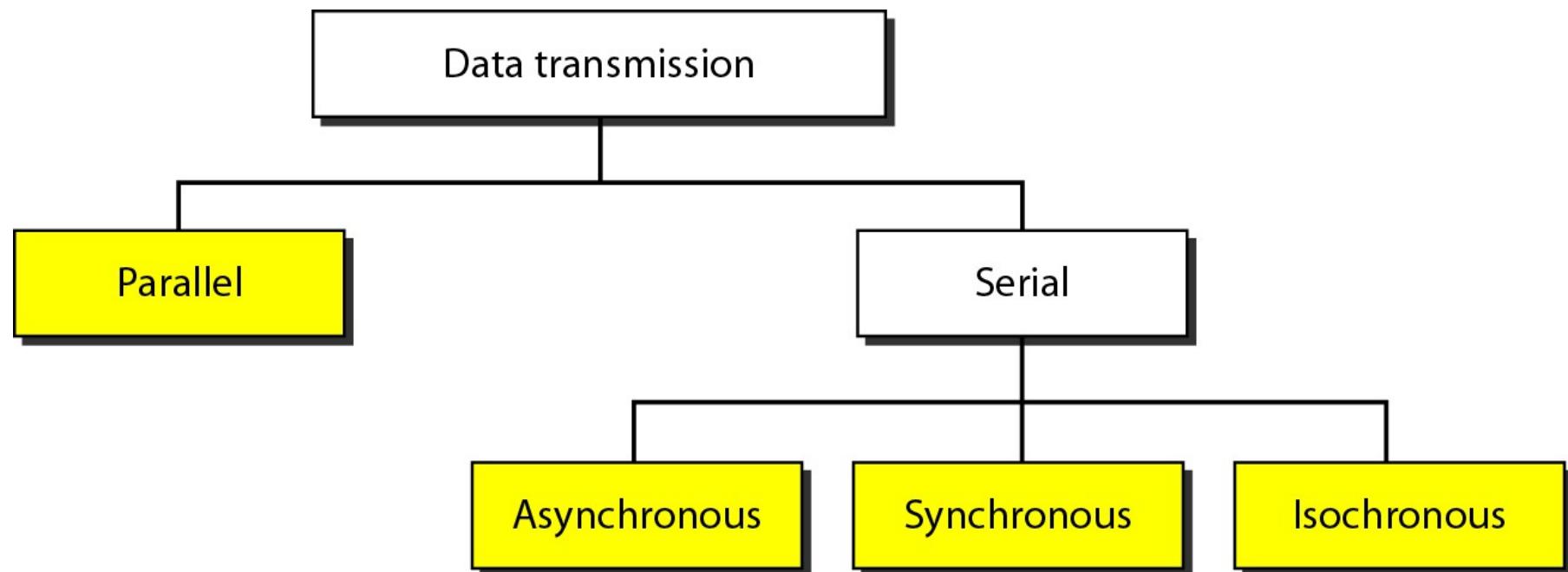
The process of delta modulation

Delta PCM (DPCM)

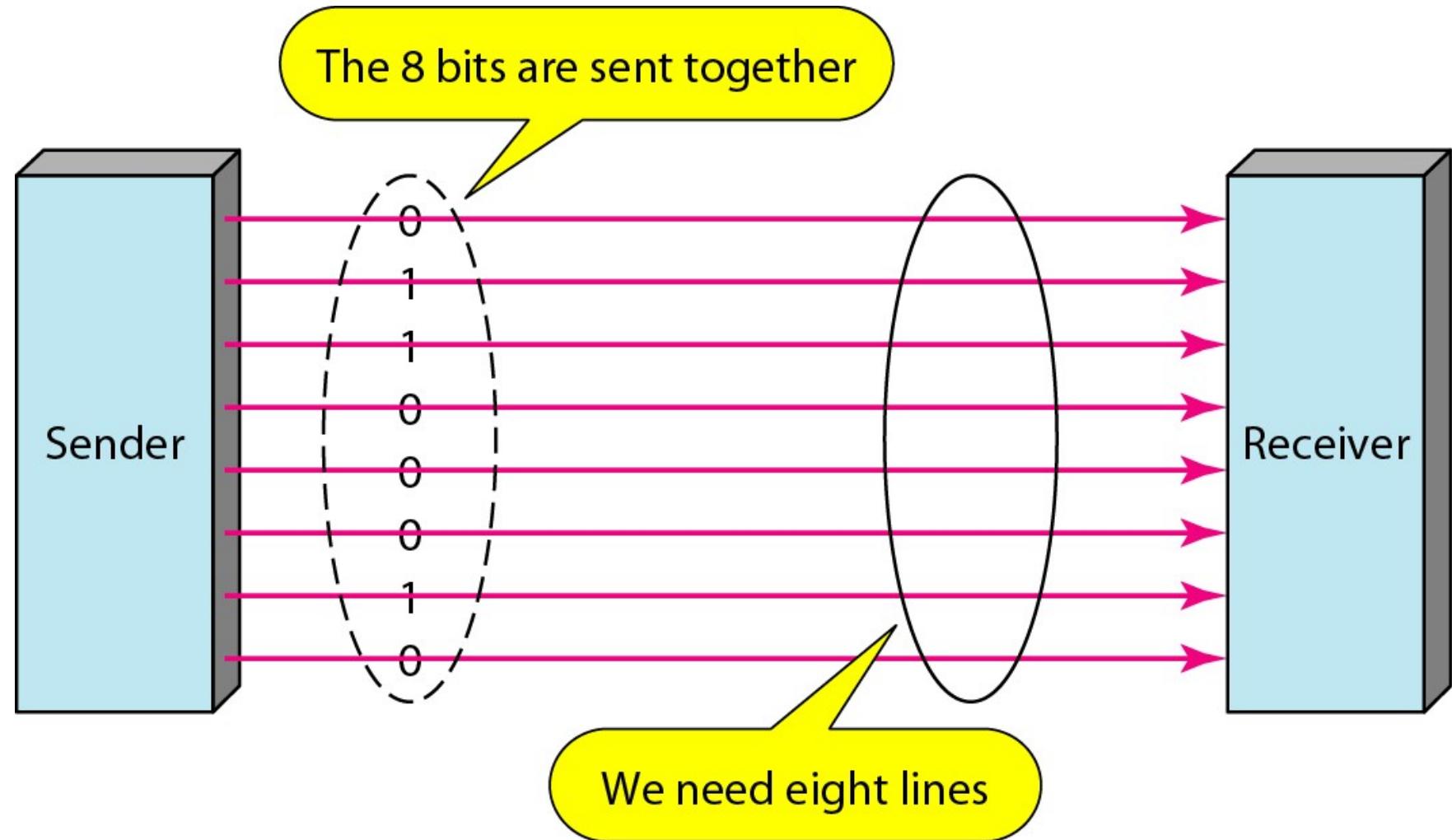
- Instead of using one bit to indicate positive and negative differences, we can use more bits -> quantization of the difference.
- Each bit code is used to represent the value of the difference.
- The more bits the more levels -> the higher the accuracy.

Transmission Modes

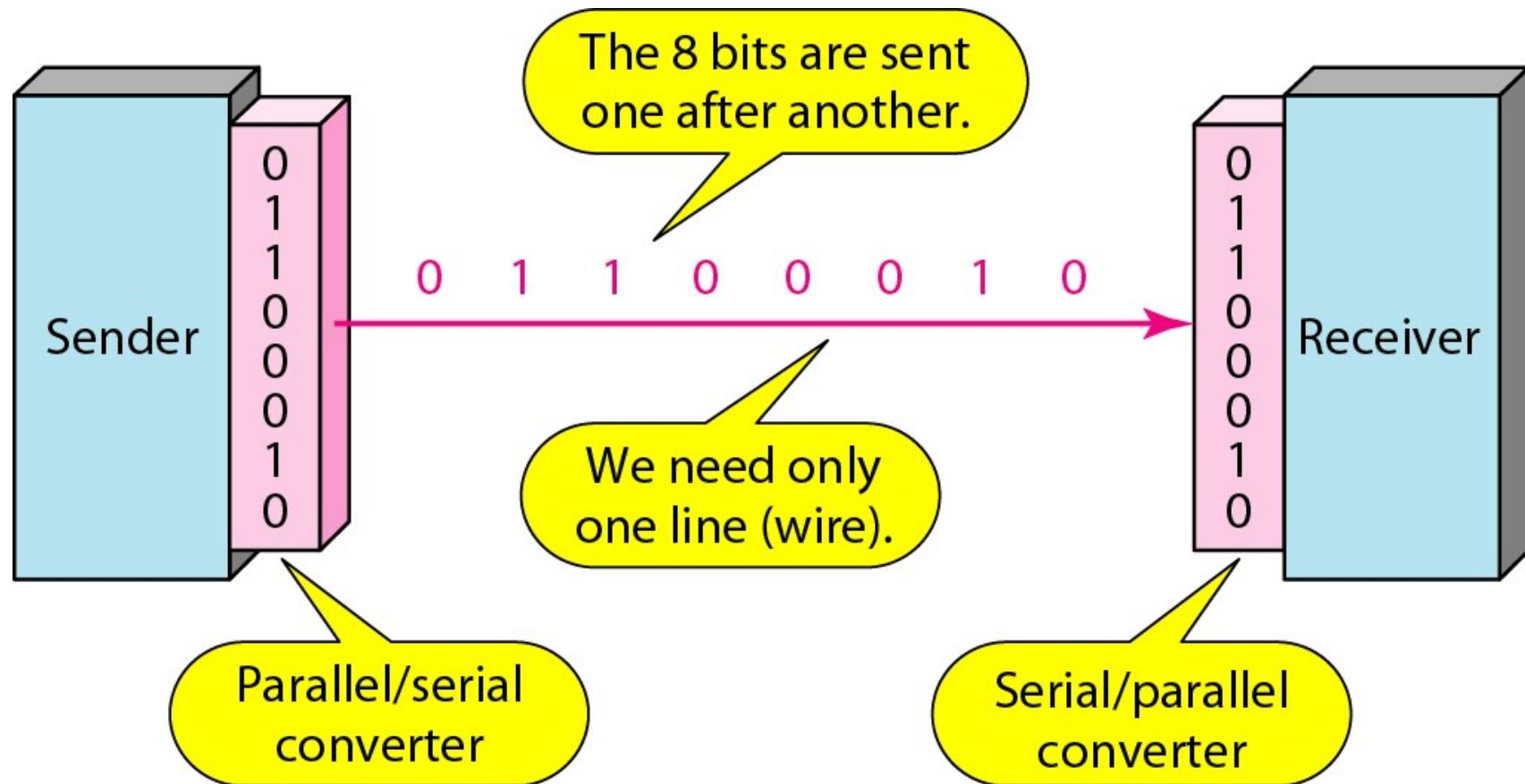
- The transmission of binary data across a link can be accomplished in either parallel or serial mode.



Parallel transmission



Serial transmission



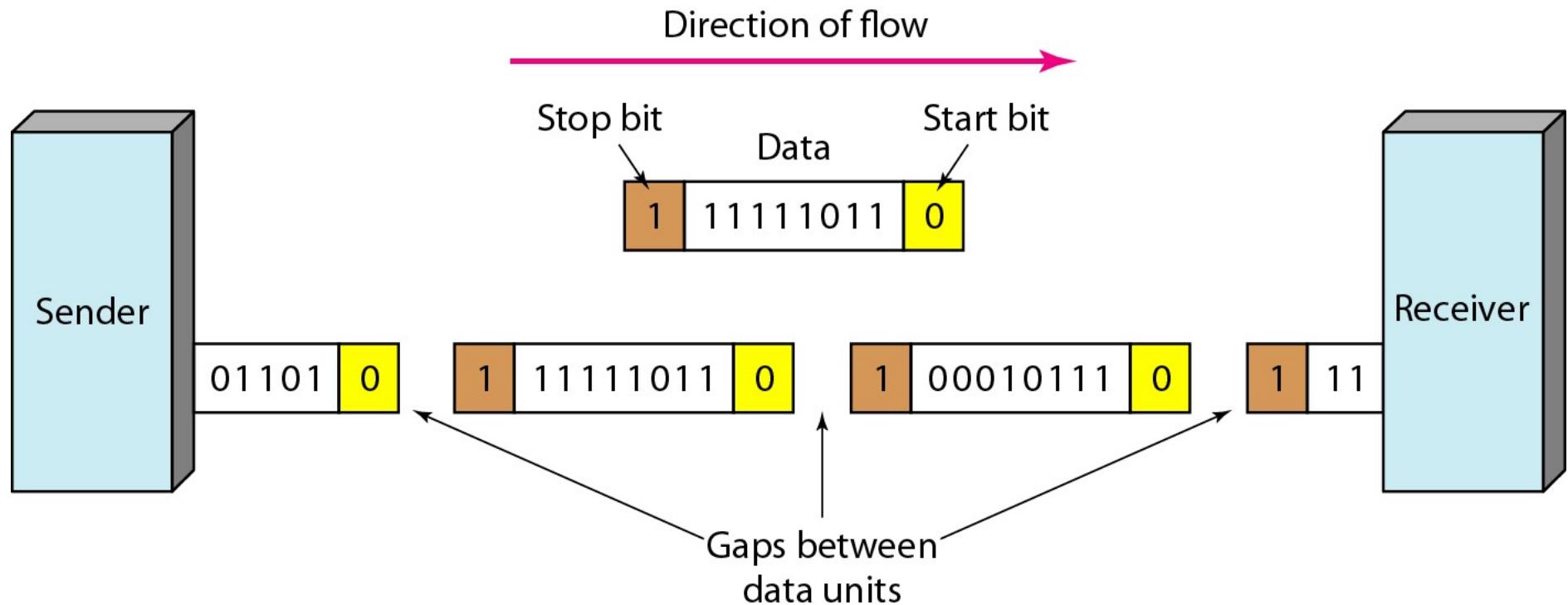
Asynchronous transmission

- 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. There may be a gap between each byte.

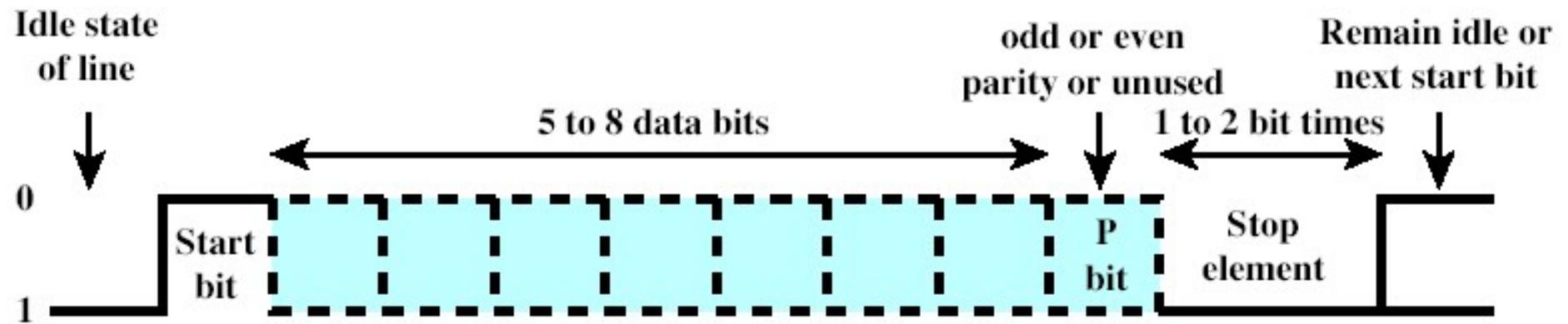
Note

Asynchronous here means “asynchronous at the byte level,” but the bits are still synchronized; their durations are the same.

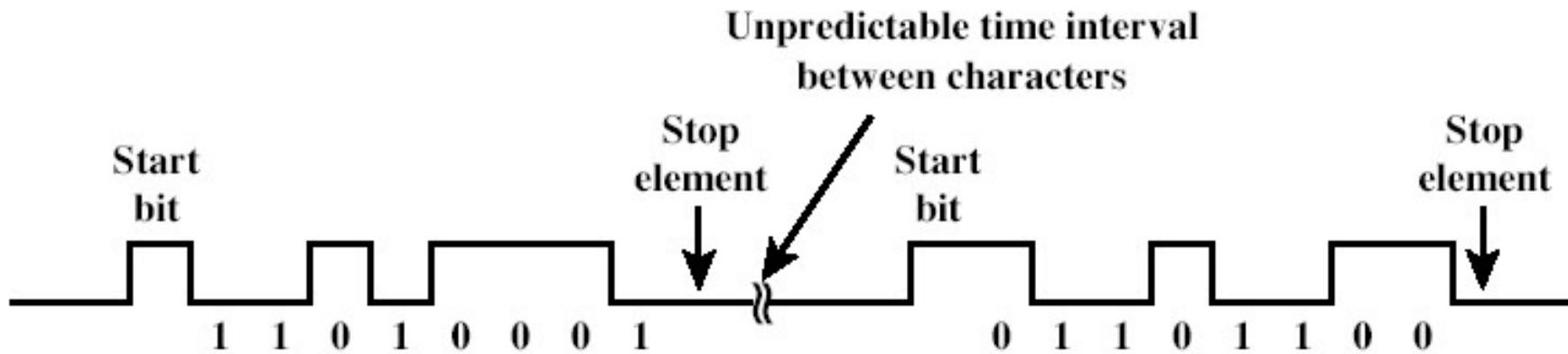
Asynchronous transmission



Asynchronous transmission



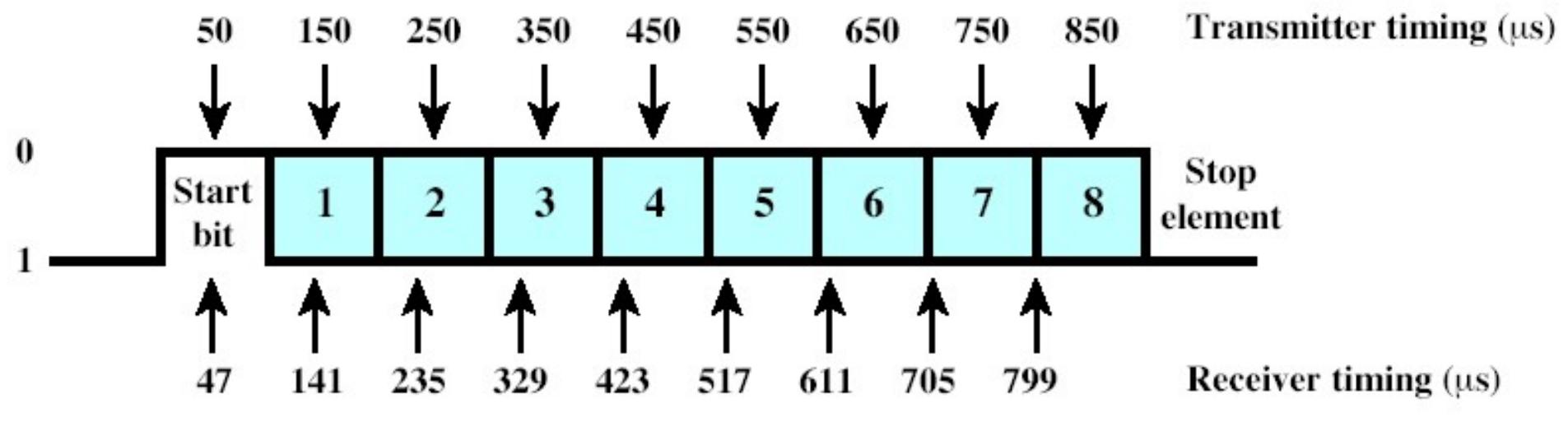
(a) Character format



(b) 8-bit asynchronous character stream

Effect of timing error in asynchronous transmission

□ Example: The figure below shows the effects of a timing error of sufficient magnitude to cause error in reception. In this example, we assume a data rate of 10Kbps; therefore each bit is 100 μ s duration. Assume that the receiver is fast by 6%, or 6 μ s per bit time. Thus, the receiver samples the incoming character every 94 μ s. As we can see, the last sample is erroneous.



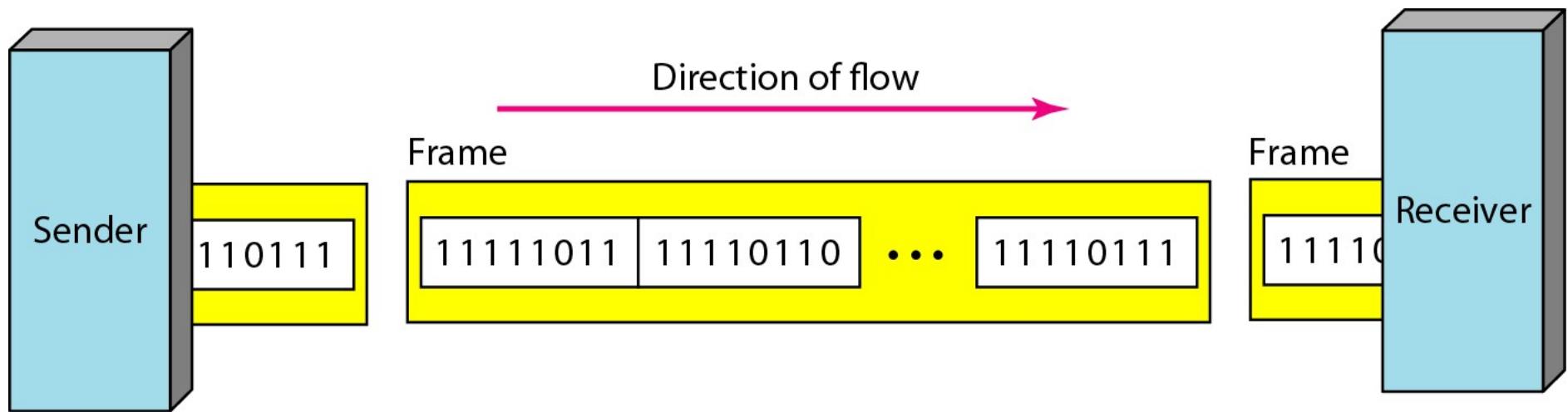
Asynchronous Transmission- Behavior

- ❑ **simple**
- ❑ **cheap**
- ❑ **overhead of 2 or 3 bits per char
(~20%)**
- ❑ **good for data with large gaps
(keyboard)**

Synchronous transmission

- 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 bits are usually sent as bytes and many bytes are grouped in a frame.
- A frame is identified with a start and an end byte.

Synchronous transmission



Isochronous

- In isochronous transmission we cannot have uneven gaps between frames.
- Transmission of bits is fixed with equal gaps.

Simplex vs. Half Duplex vs. Full Duplex

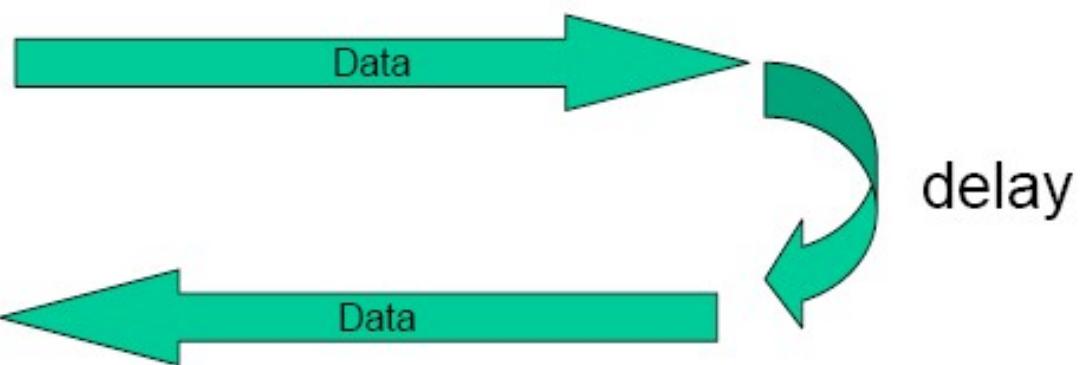
Simplex data flow



One direction, e.g. Television

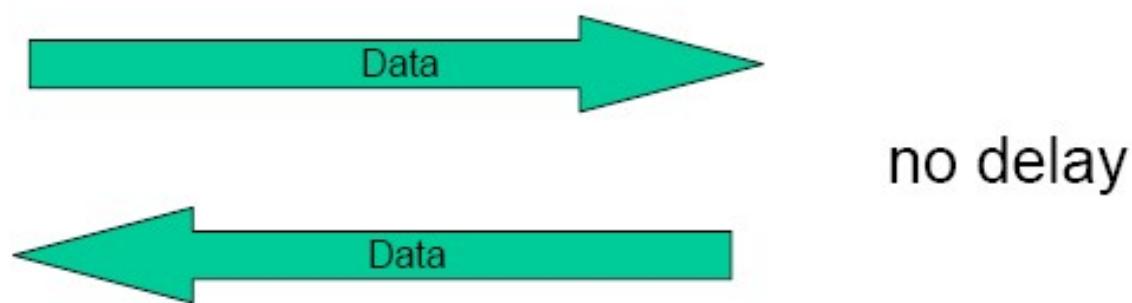
Half-Duplex data flow

Either direction, but only one way at a time, e.g. police radio



Full-Duplex data flow

Both directions at the same time e.g. telephone



no delay

Chapter 2: Roadmap

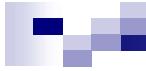
2.1 Physical layer Introduction

2.2 Data communication

2.3 Transmission Media

2.4 Modulation and Data Encoding

2.5 Multiplxing

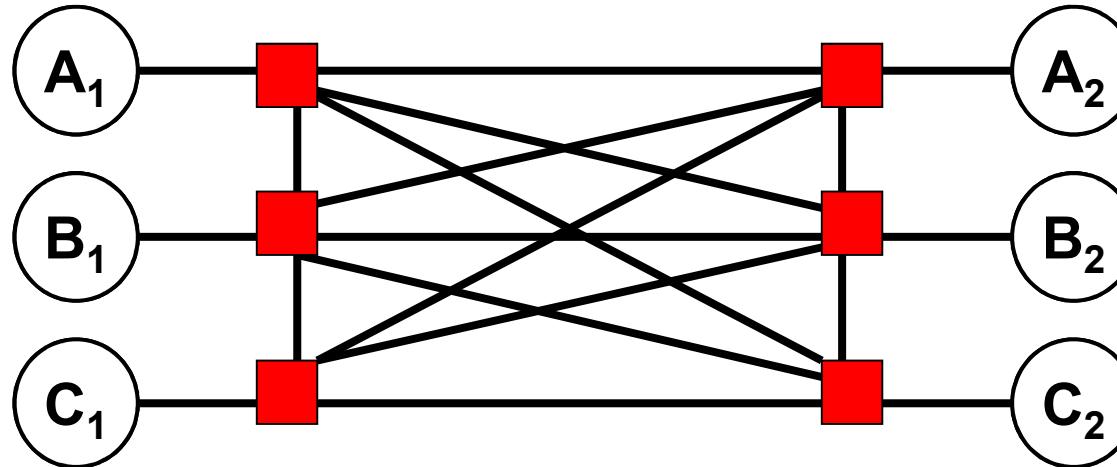


Note

Bandwidth utilization is the wise use of available bandwidth to achieve specific goals.

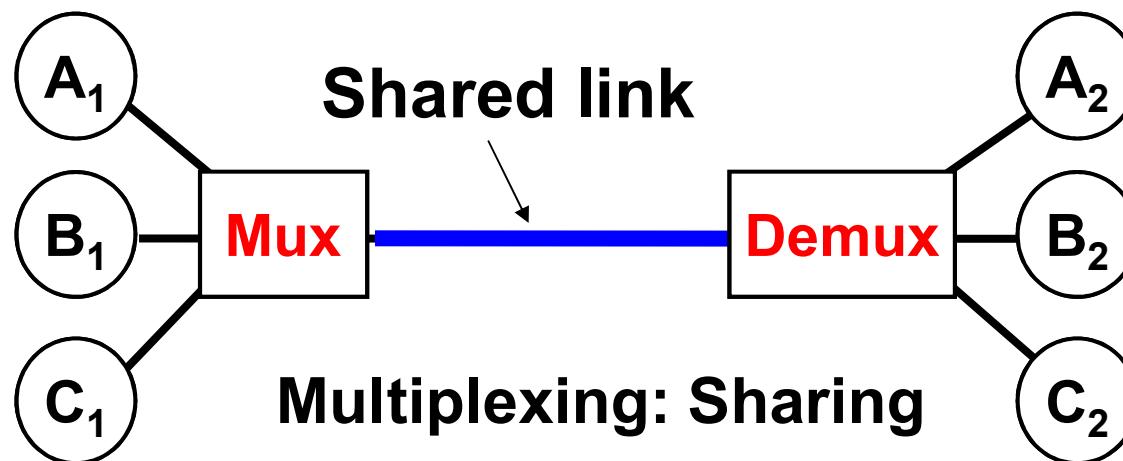
Efficiency can be achieved by multiplexing; privacy and anti-jamming can be achieved by spreading.

Multiplexing



link
link
link

Dedicated, N² problem

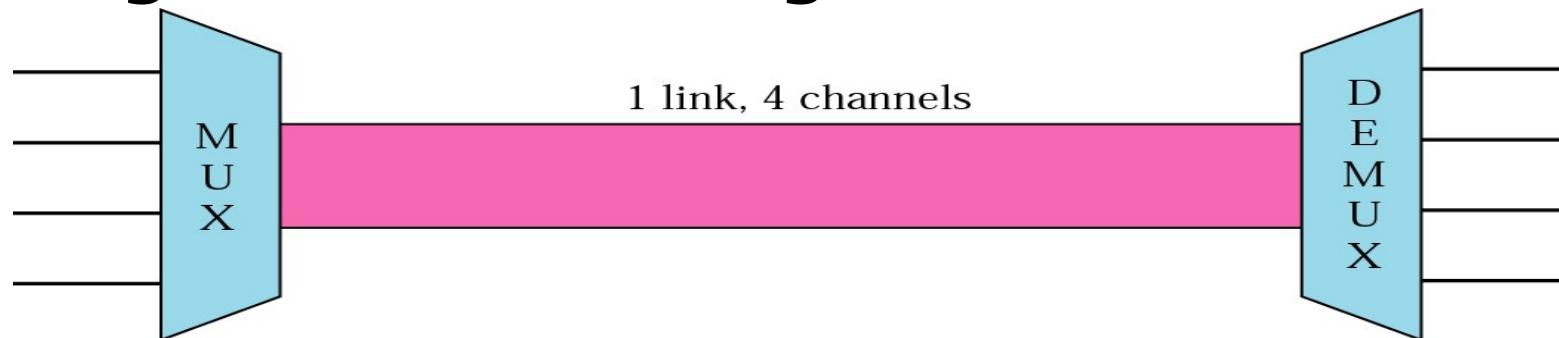


Multiplexing: Sharing

Multiplexing

■ Multiplexing

□ A set of techniques that allows the simultaneous transmission of multiple signals across a single data link.



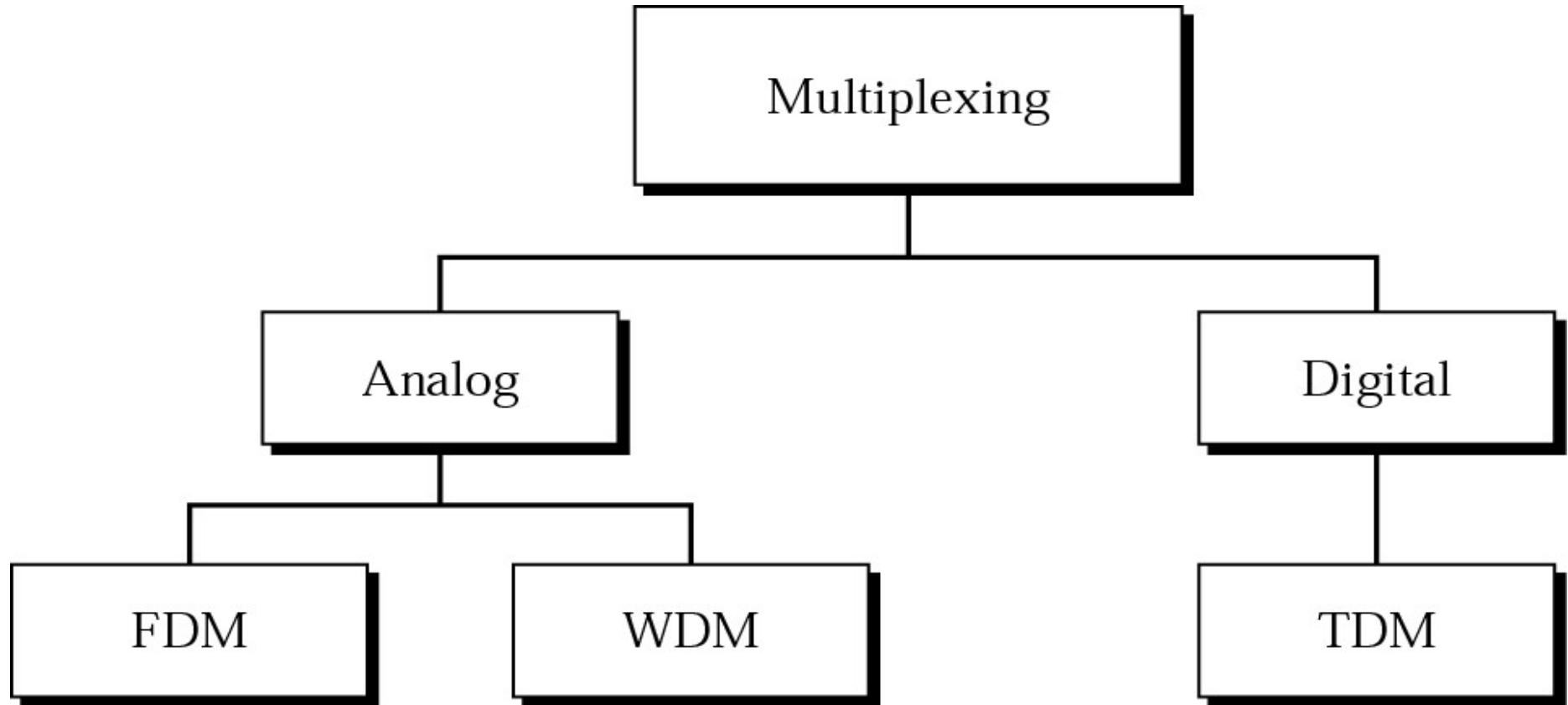
Multiplexer (MUX) :

Combines multiple streams into a single stream (many to one).

Demultiplexer (DEMUX) :

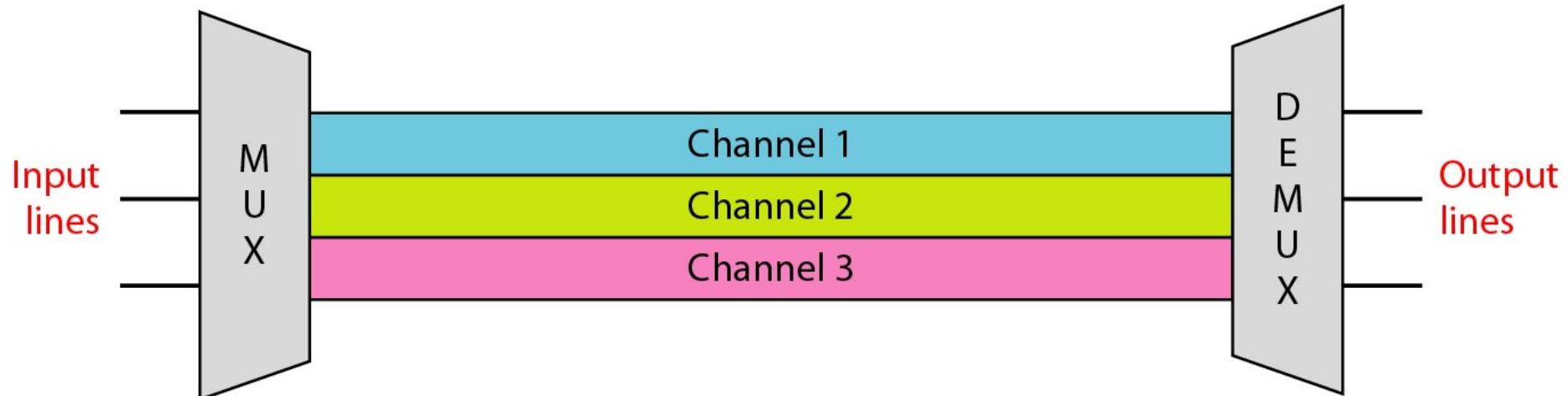
Separates the stream back into its component transmission (one to many) and directs them to their correct lines.

Category of Multiplexing



FDM : Frequency-division multiplexing

Assigns different analog frequencies to each connected device

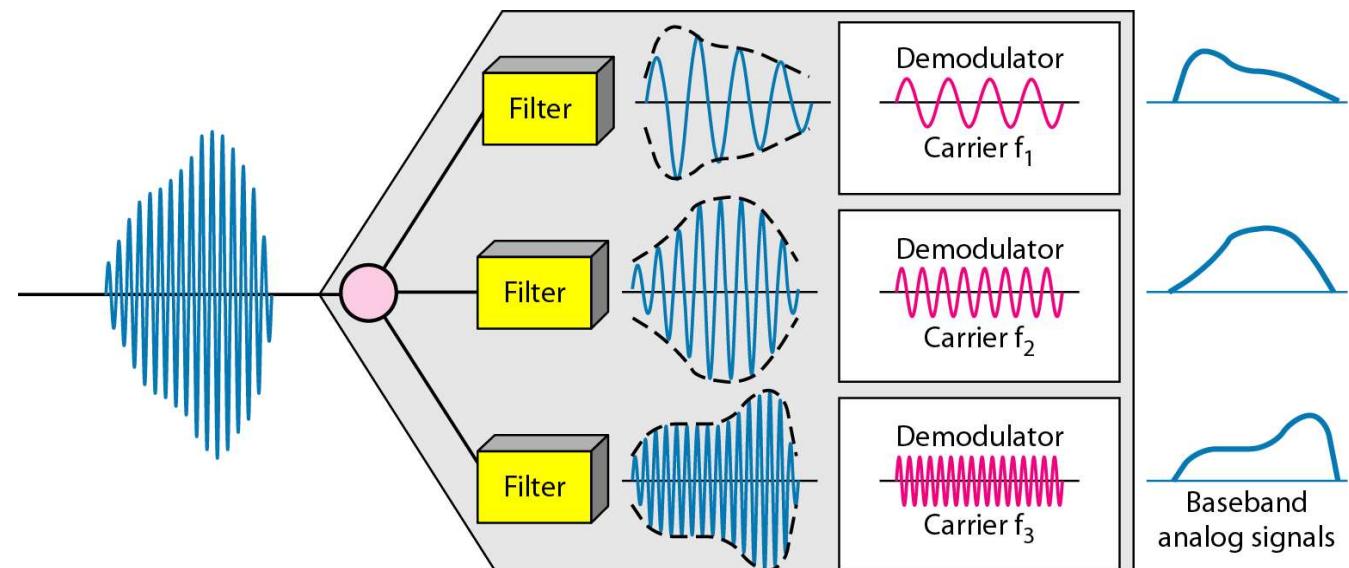
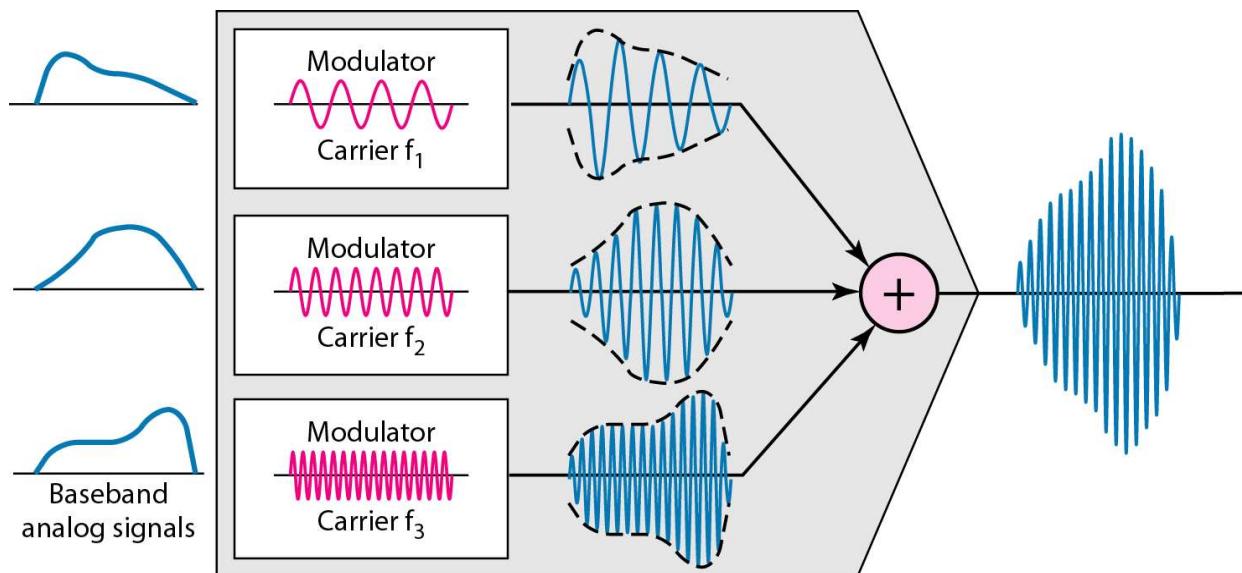


Channels must be separated by strips of unused bandwidth - *guard bandwidth*

Note

FDM is an analog multiplexing technique that combines analog signals.

FDM process



Example 1

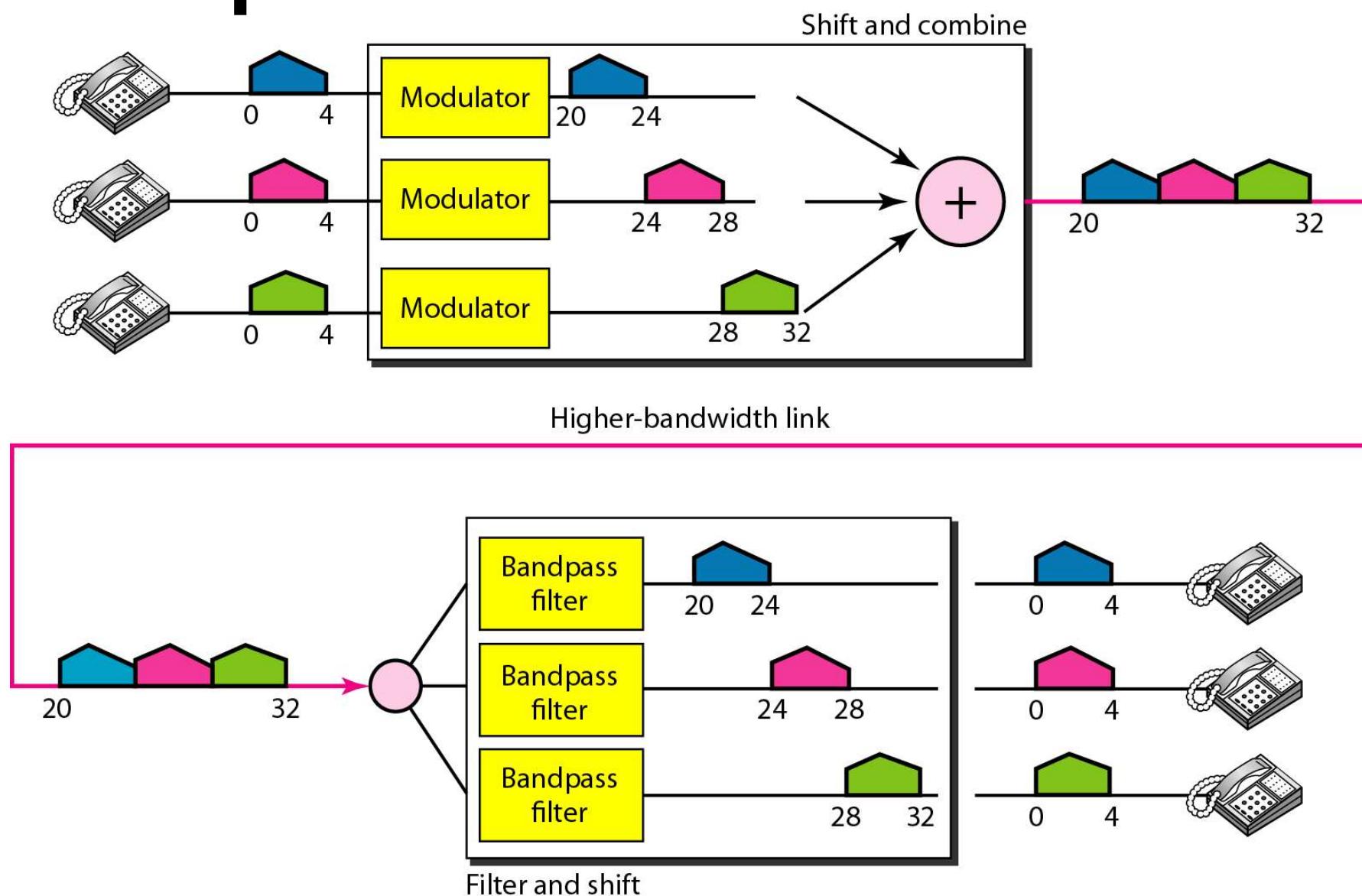
Assume that a voice channel occupies a bandwidth of 4 kHz. We need to combine three voice channels into a link with a bandwidth of 12 kHz, from 20 to 32 kHz. Show the configuration, using the frequency domain. Assume there are no guard bands.

Example 1

Solution

We shift (modulate) each of the three voice channels to a different bandwidth. We use the 20- to 24-kHz bandwidth for the first channel, the 24- to 28-kHz bandwidth for the second channel, and the 28- to 32-kHz bandwidth for the third one. Then we combine them as shown in Figure.

Example 1



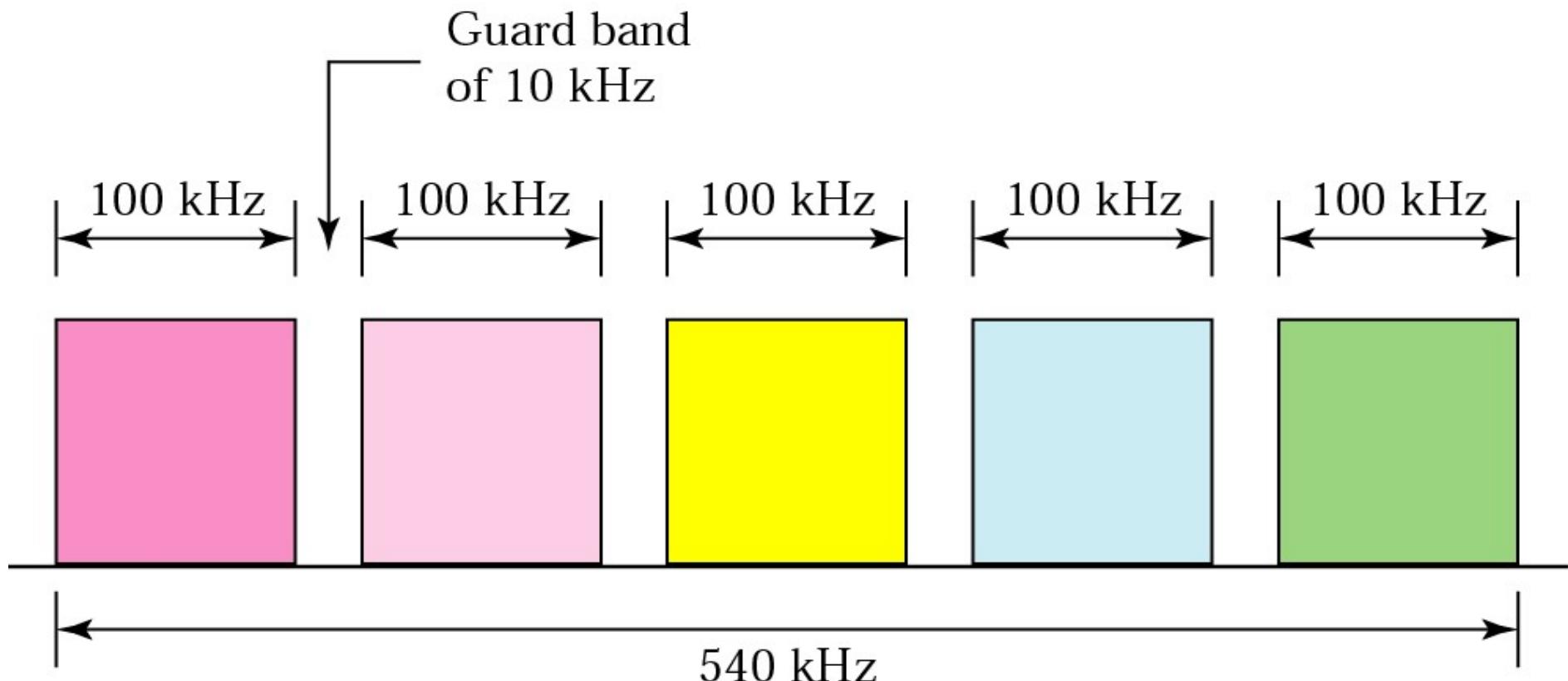
Example 2

Five channels, each with a 100-kHz bandwidth, are to be multiplexed together. What is the minimum bandwidth of the link if there is a need for a guard band of 10 kHz between the channels to prevent interference?

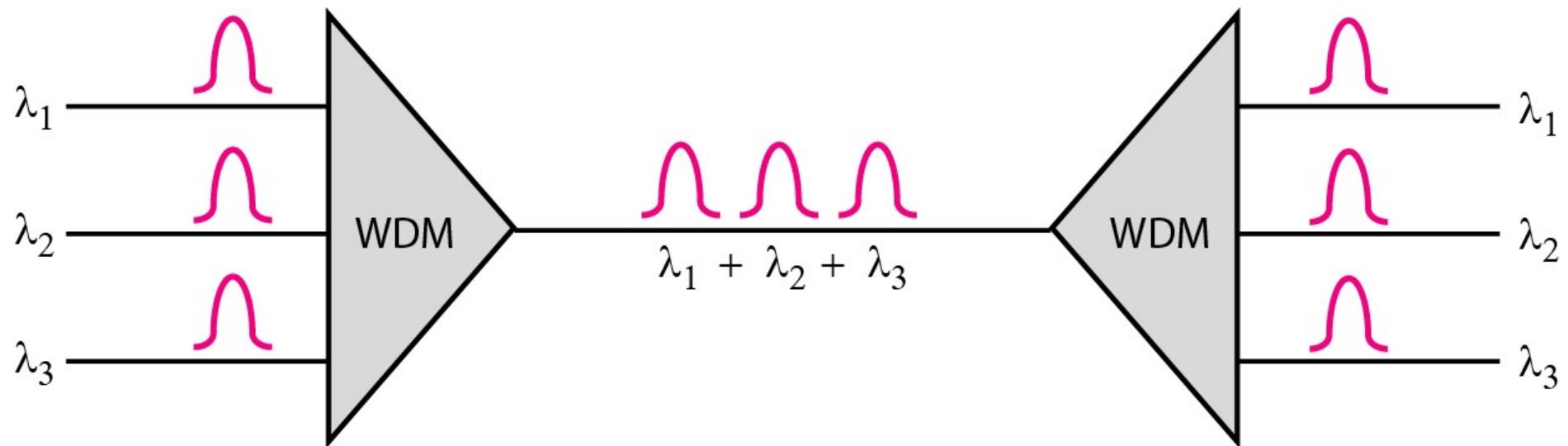
Solution

For five channels, we need at least four guard bands. This means that the required bandwidth is at least $5 \times 100 + 4 \times 10 = 540 \text{ kHz}$

Example 2



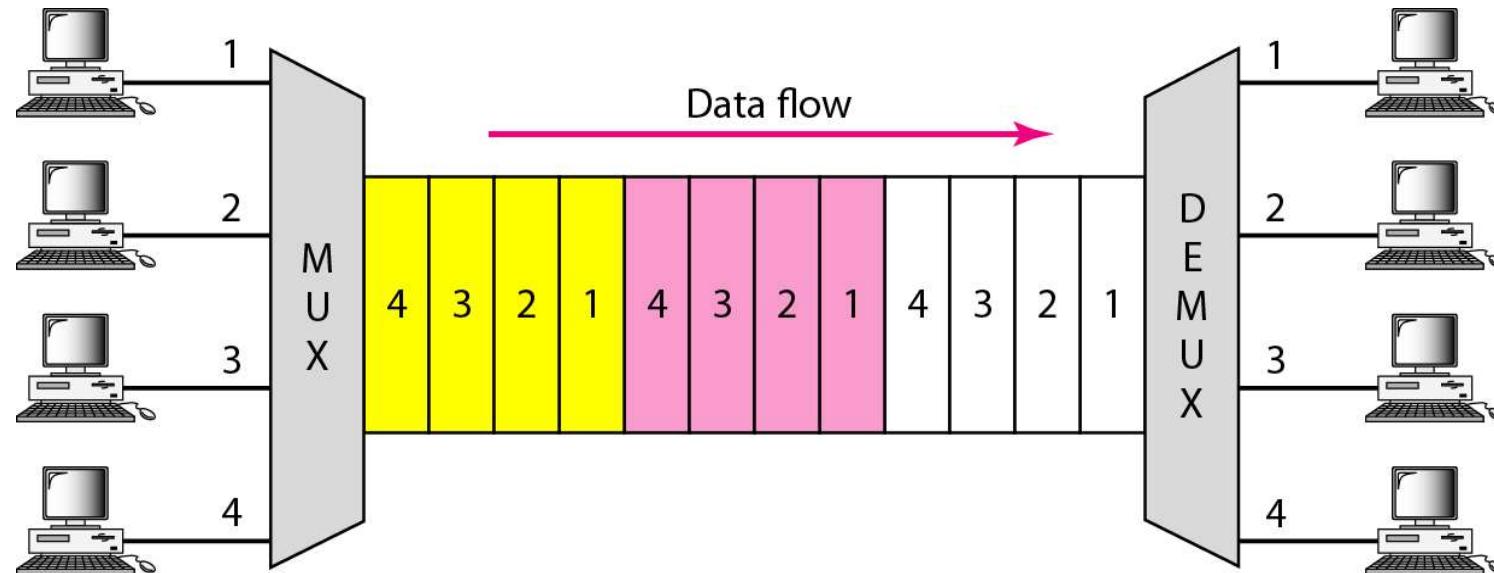
Wavelength-division multiplexing



Note

WDM is an analog multiplexing technique to combine optical signals.

TDM : Time Division Multiplexing



Sharing signal is accomplished by dividing available transmission time on a medium among users.

The whole bandwidth is used all the time, but alternatively by different channels!

TDM : Time Division Multiplexing

Note

TDM is a digital multiplexing technique for combining several low-rate channels into one high-rate one.

- Time division multiplexing comes in **two basic forms:**
 - **Synchronous time division multiplexing**
 - **Statistical, or asynchronous time division multiplexing**

TDM : Time Division Multiplexing

■ **Time Slots**

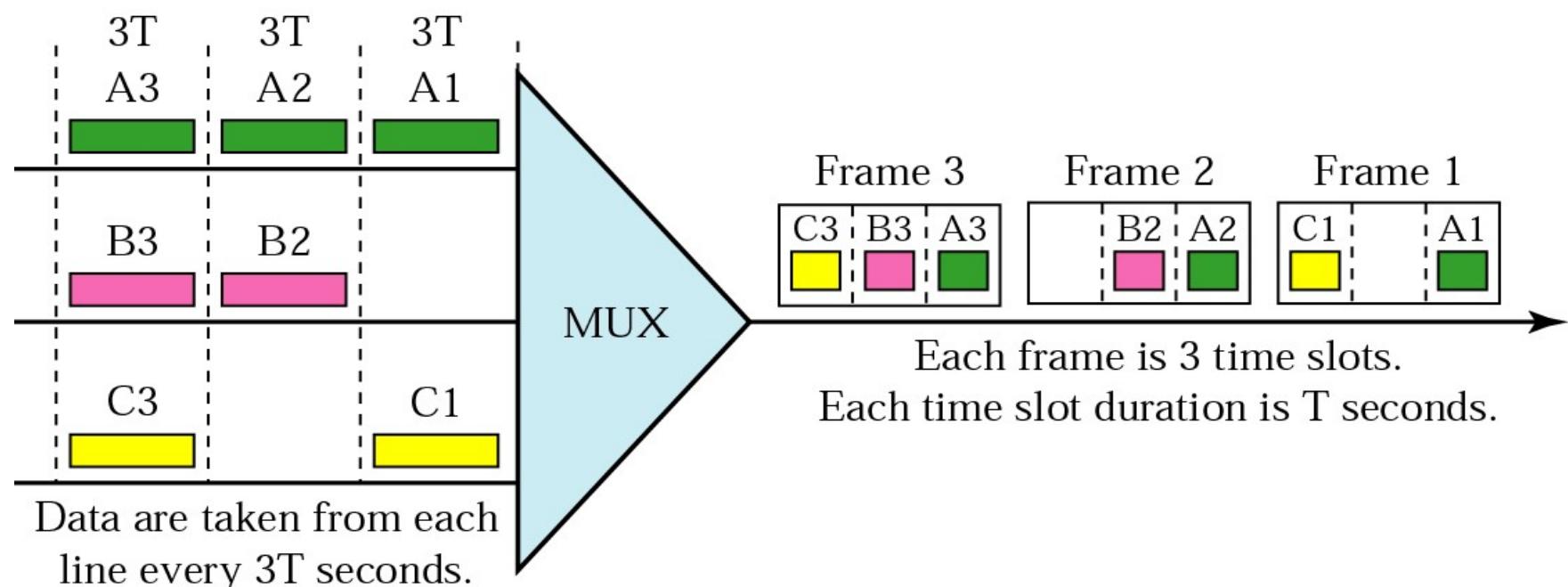
- **Each terminal/host given a “slice” of time (time slot)**

■ **Frames**

- **In TDM, a frame consists of one complete cycle of time slots, with one slot dedicated to each sending device.**

Synchronous time division multiplexing

the multiplexer allocates exactly the same time slot to each device at all times, whether or not a device has anything to transmit.



Synchronous time division multiplexing

Note

In synchronous TDM, Data rate of medium exceeds data rate of digital signal to be transmitted.

Note

In synchronous TDM, the data rate of the link is n times faster, and the unit duration is n times shorter.

Example 4

Four 1-Kbps connections are multiplexed together. A unit is 1 bit. Find (1) the duration of 1 bit before multiplexing, (2) the transmission rate of the link, (3) the duration of a time slot, and (4) the duration of a frame?

Solution

1. *The duration of 1 bit is 1/1 Kbps, or 0.001 s (1 ms).*
2. *The rate of the link is 4 Kbps.*
3. *The duration of each time slot 1/4 ms or 250 ms.*
4. *The duration of a frame 1 ms.*

Solution in detail

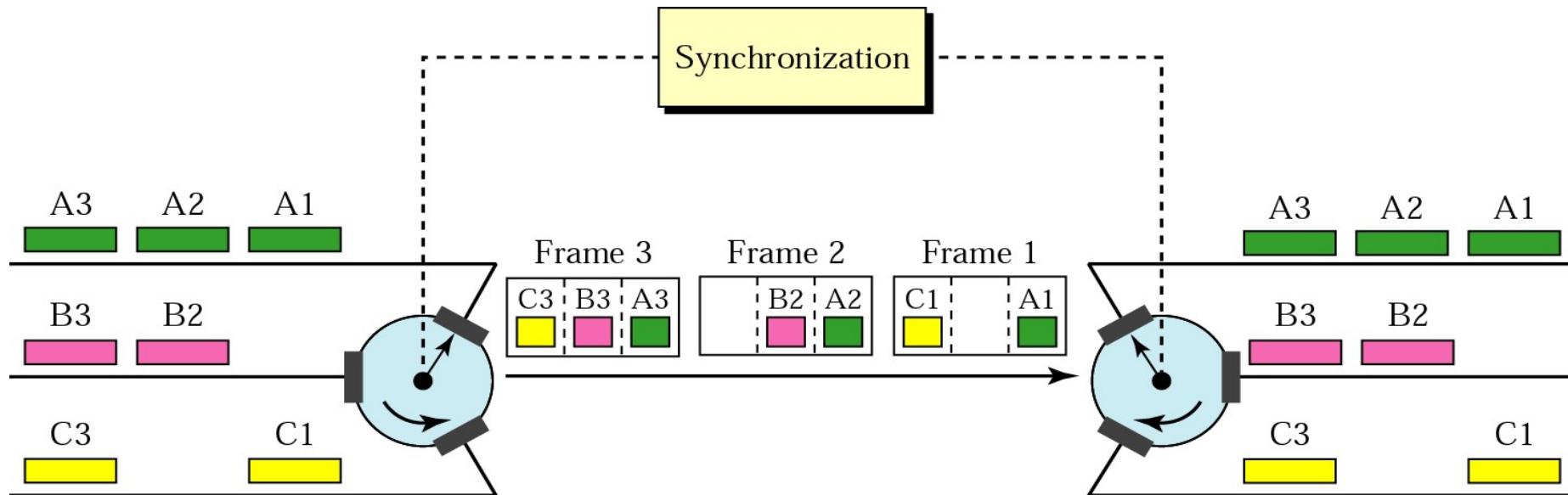
$$DataRate_{link} = 4 \times 1 kbps = 4 kbps = 4000 bps$$

$$\begin{aligned} BitDuration_{link} &= \frac{1bit}{4000bps} = \frac{1bit}{4000bit / second} \\ &= 0.25ms / bit = 250\mu s / bit \end{aligned}$$

$$\begin{aligned} TimeSlotDuration_{link} &= BitDuration \times UnitSize \\ &= 250\mu s / bit \times 1bit / TimeSlot = 250\mu s / TimeSlot \end{aligned}$$

$$\begin{aligned} FrameDuration &= TimeSlotDuration \times ChannelNumber \\ &= 250\mu s / TimeSlot * 4TimeSlot / Frame \\ &= 1ms / Frame \end{aligned}$$

Interleaving of data segments



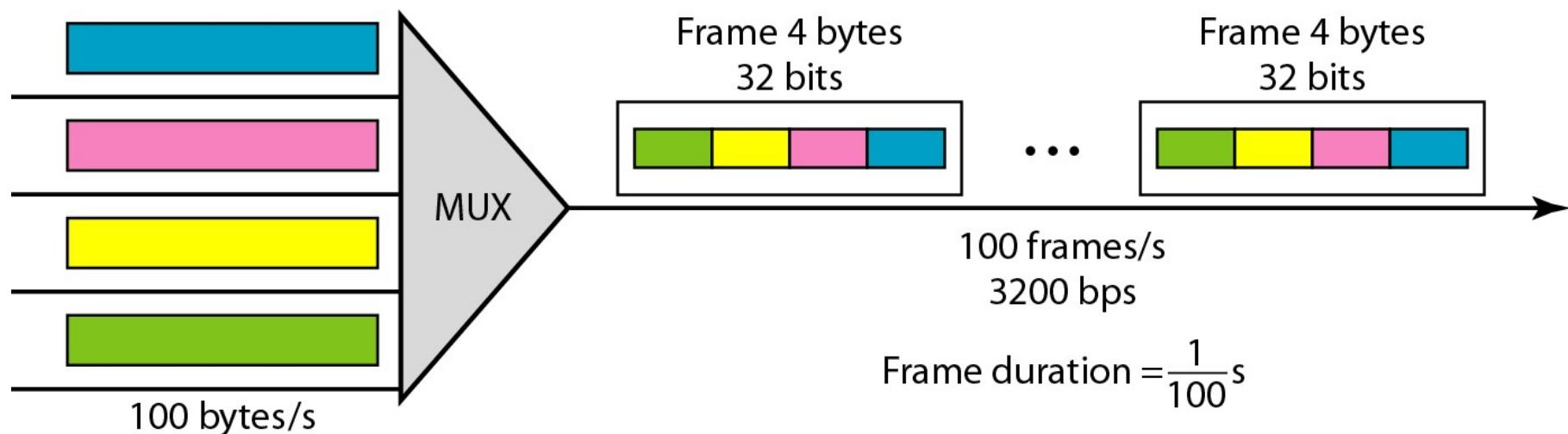
Example 5

Four channels are multiplexed using TDM. If each channel sends 100 bytes /s and we multiplex 1 byte per channel, show the frame traveling on the link, the size of the frame, the duration of a frame, the frame rate, and the bit rate for the link.

Example 5

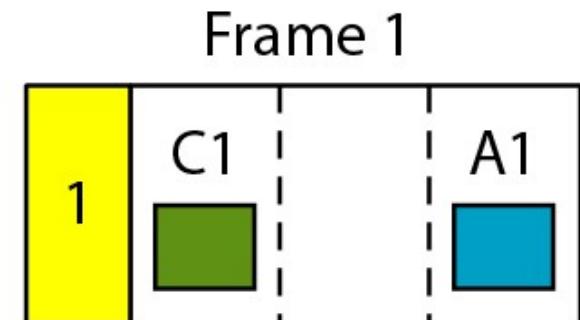
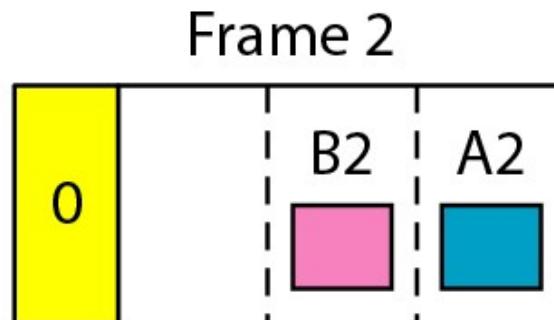
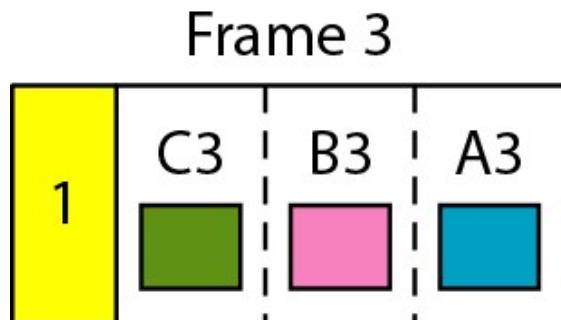
Solution

Each frame carries 1 byte from each channel; the size of each frame, therefore, is 4 bytes, or 32 bits. Because each channel is sending 100 bytes/s and a frame carries 1 byte from each channel, the frame rate must be 100 frames per second. The bit rate is 100×32 , or 3200 bps.



SYNCHRONIZING

- One or more *Framing bit (s)* is (are) added to each frame for synchronization between the multiplexer and demultiplexer
- If 1 framing bit per frame, framing bits are alternating between 0 and 1



Example 6

We have four sources, each creating 250 characters per second. If the interleaved unit is a character and 1 synchronizing bit is added to each frame, find (1) the data rate of each source, (2) the duration of each character in each source, (3) the frame rate, (4) the duration of each frame, (5) the number of bits in each frame, and (6) the data rate of the link.

Example 6

Solution

1. The data rate of each source is $250 \times 8 = 2000$ bps
= 2 Kbps.
2. The duration of a character is $1/250$ s, or 4 ms.
3. The link needs to send 250 frames per second to keep the transmission rate of each source.
4. The duration of each frame is $1/250$ s, or 4 ms.
5. Each frame is $4 \times 8 + 1 = 33$ bits.
6. The data rate of the link is 250×33 , or 8250 bps.

Example 6

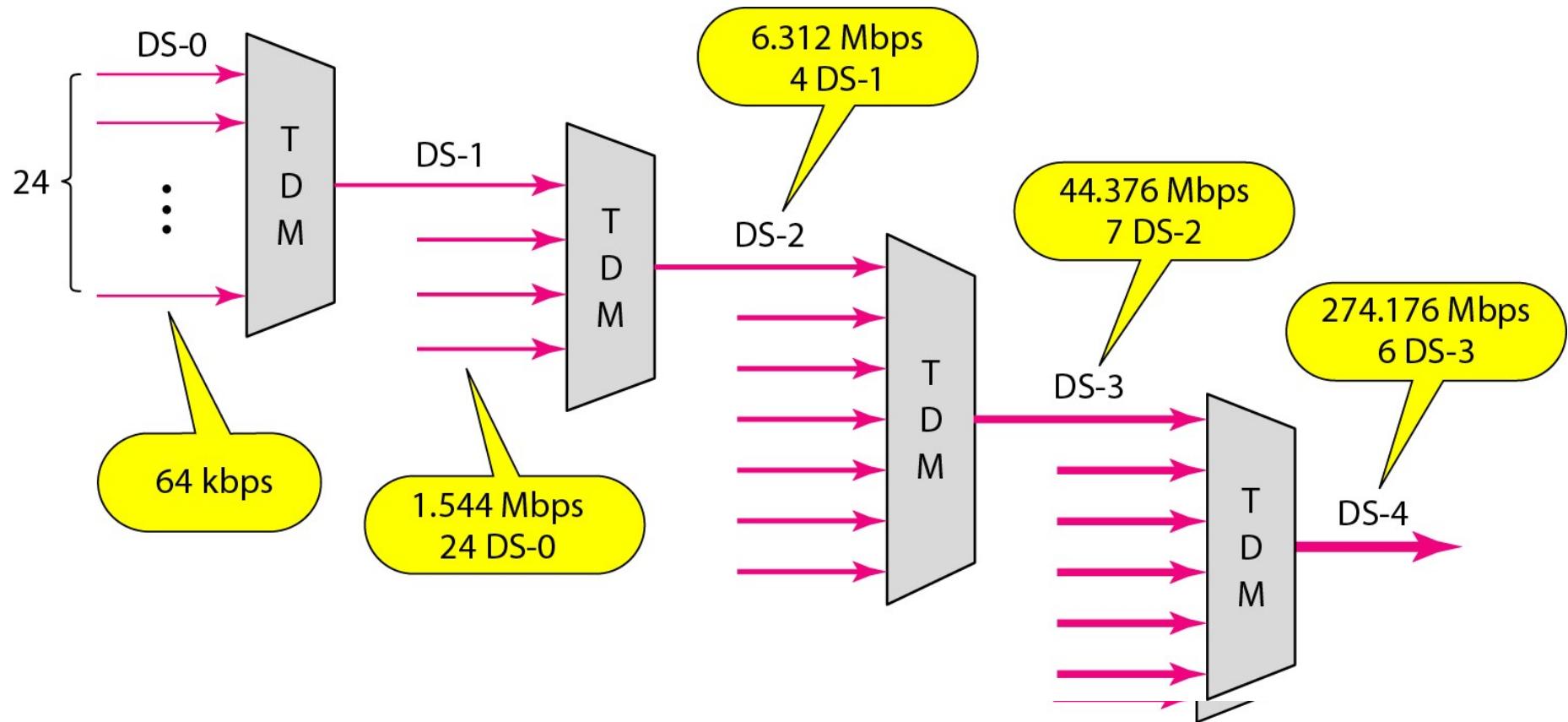
Solution in detail

$$\begin{aligned} \text{FrameSize} &= \text{ChannelNumber} \times \text{UnitSize} + \text{Fractional Bits} \\ &= 4 \text{ timeslot / frame} \times 1 \text{ character / timeslot} + 1 \text{ bit / frame} \\ &= 33 \text{ bits / frame} \end{aligned}$$

$$\text{FrameRate} = 250 \text{ frame / second}$$

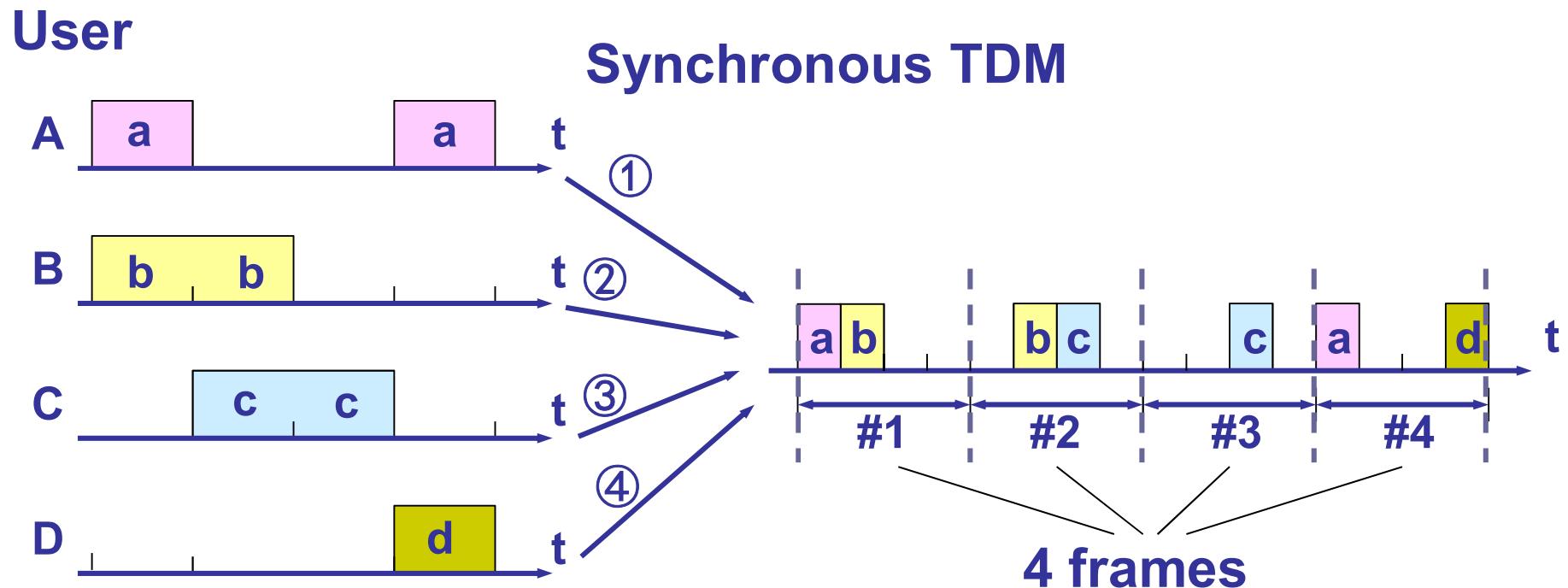
$$\begin{aligned} \text{DataRate} &= \text{FrameRate} \times \text{FrameSize} \\ &= 250 \text{ frame / second} \times 33 \text{ bits / frame} \\ &= 8250 \text{ bit / second} \end{aligned}$$

TDM hierarchy



Asynchronous TDM

- Synchronous TDM does not guarantee that the full capacity of a link is used.



Asynchronous TDM

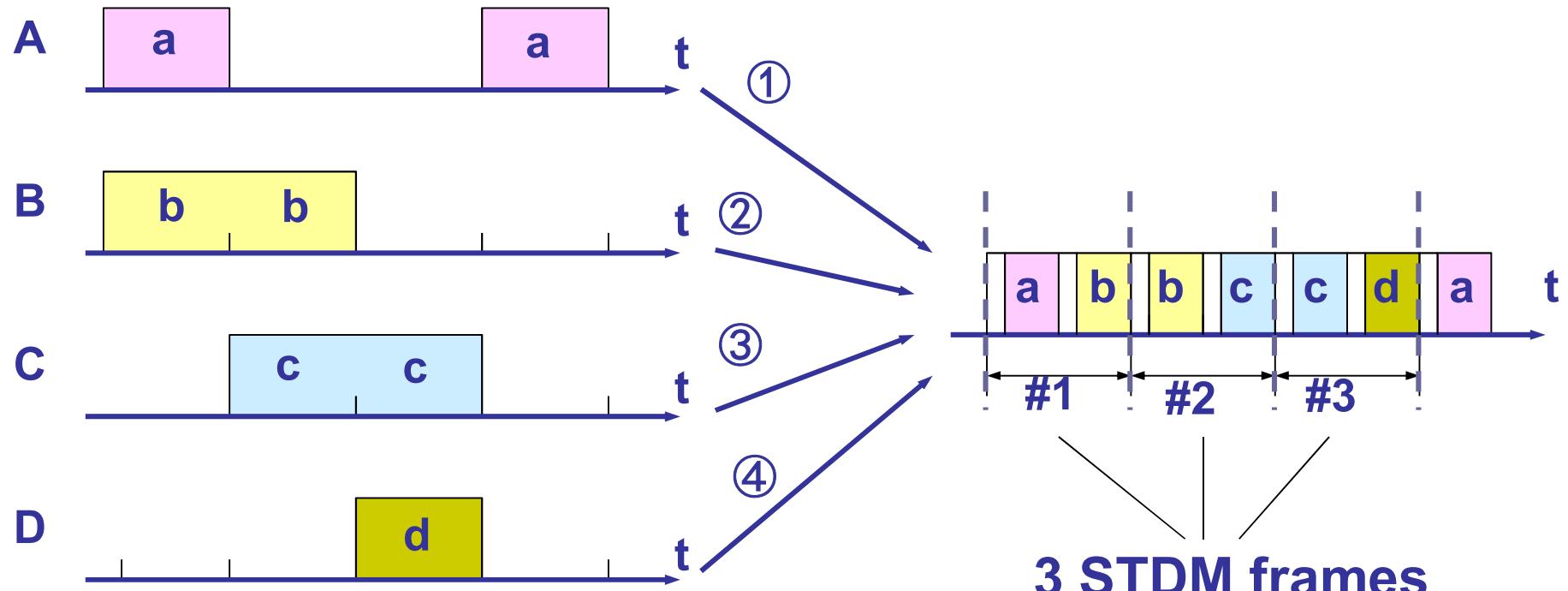
- Asynchronous TDM, or **statistical TDM (STDM)**, is designed to avoid this type of waste.
- **Statistical TDM allocates time slots dynamically based on demand.**



STDM

Users

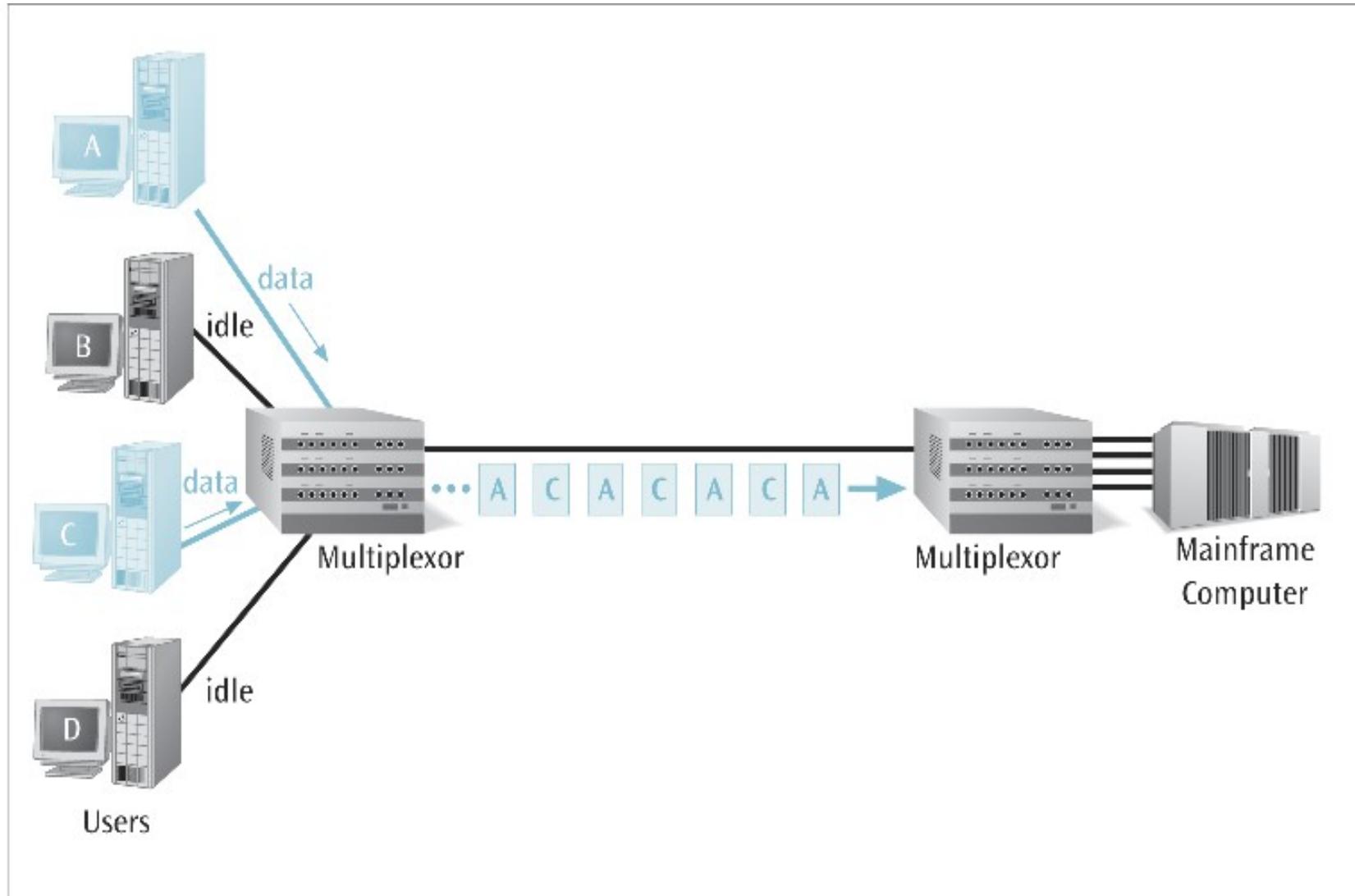
STDM



Asynchronous TDM

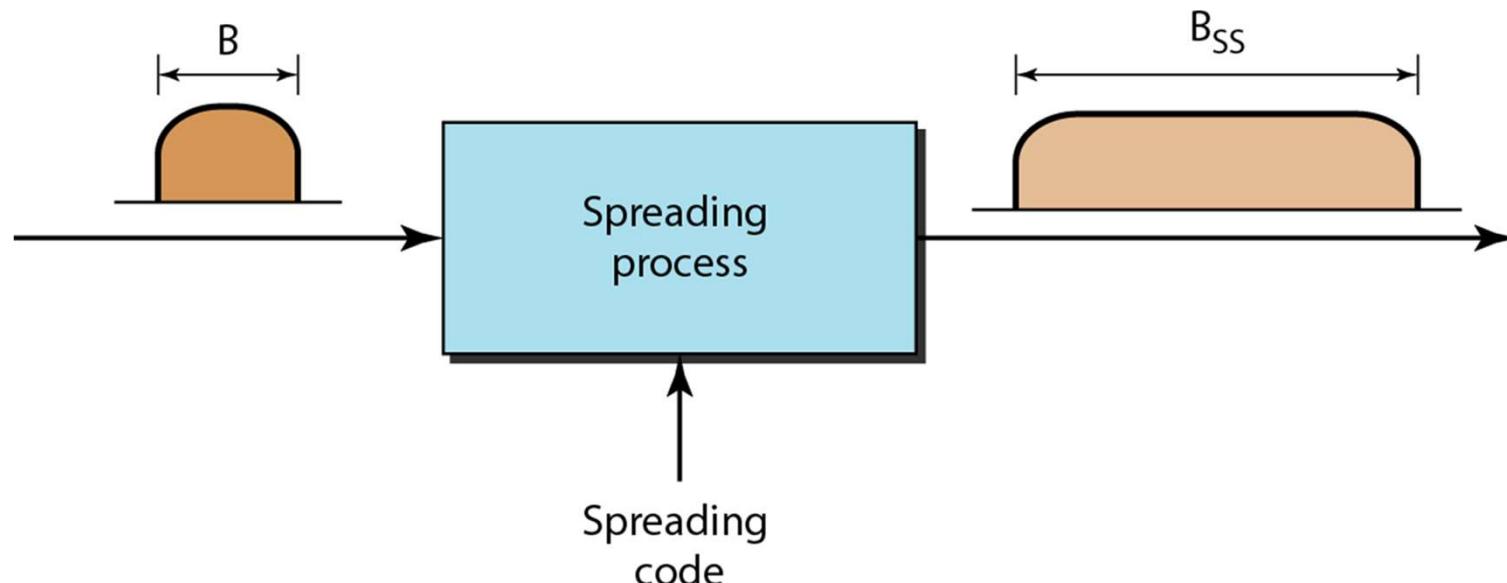
- **Statistical multiplexor transmits only the data from active workstations**
 - If a workstation is not active, no space is wasted on the multiplexed stream
- **A statistical multiplexor**
 - Accepts incoming data streams
 - Creates a frame containing only the data to be transmitted

Asynchronous TDM



Spread Spectrum

- Methods by which a signal generated with a particular bandwidth is deliberately spread in the frequency domain, resulting in a signal with a wider bandwidth.



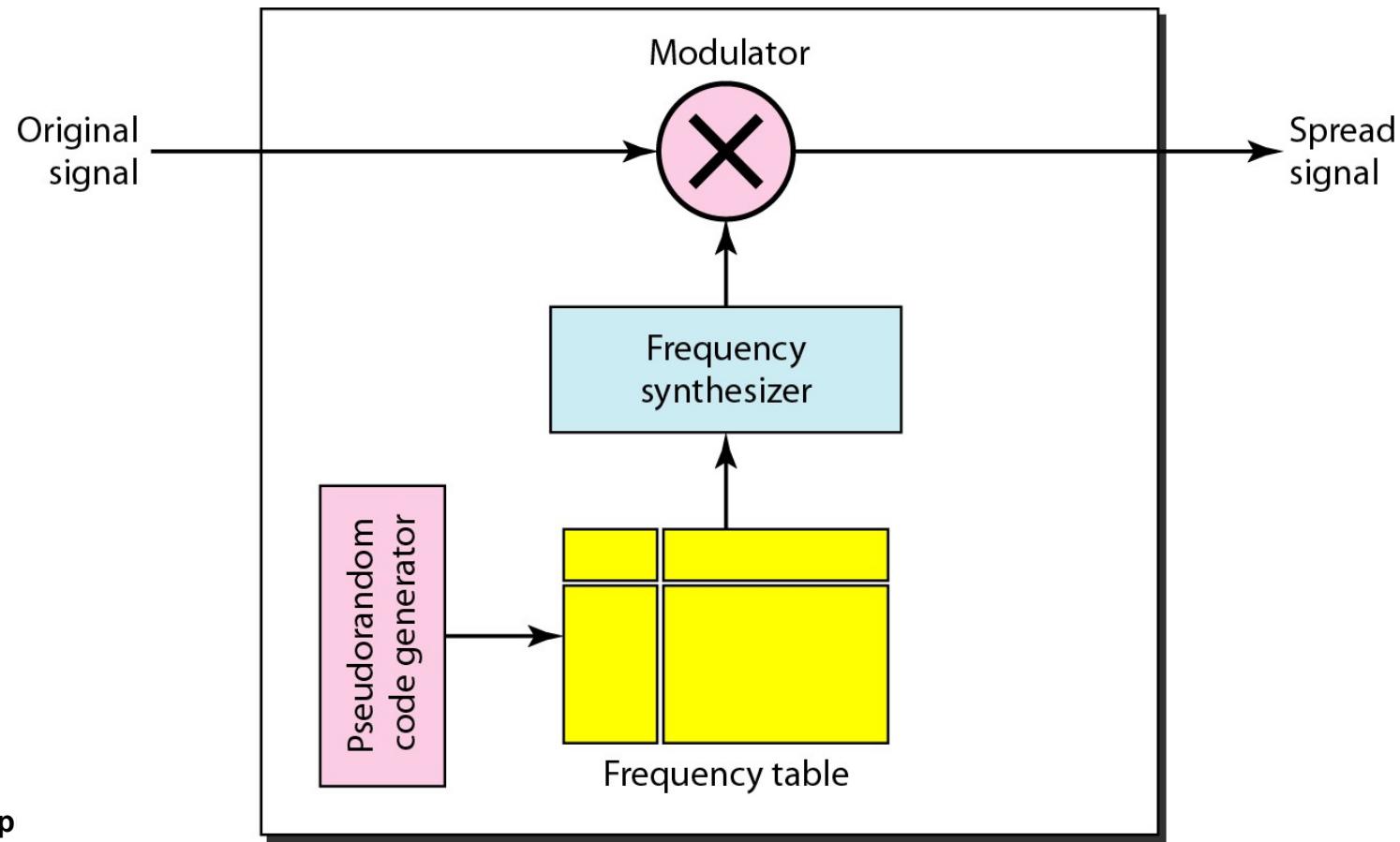
Spread Spectrum

■ Spread Spectrum Techniques

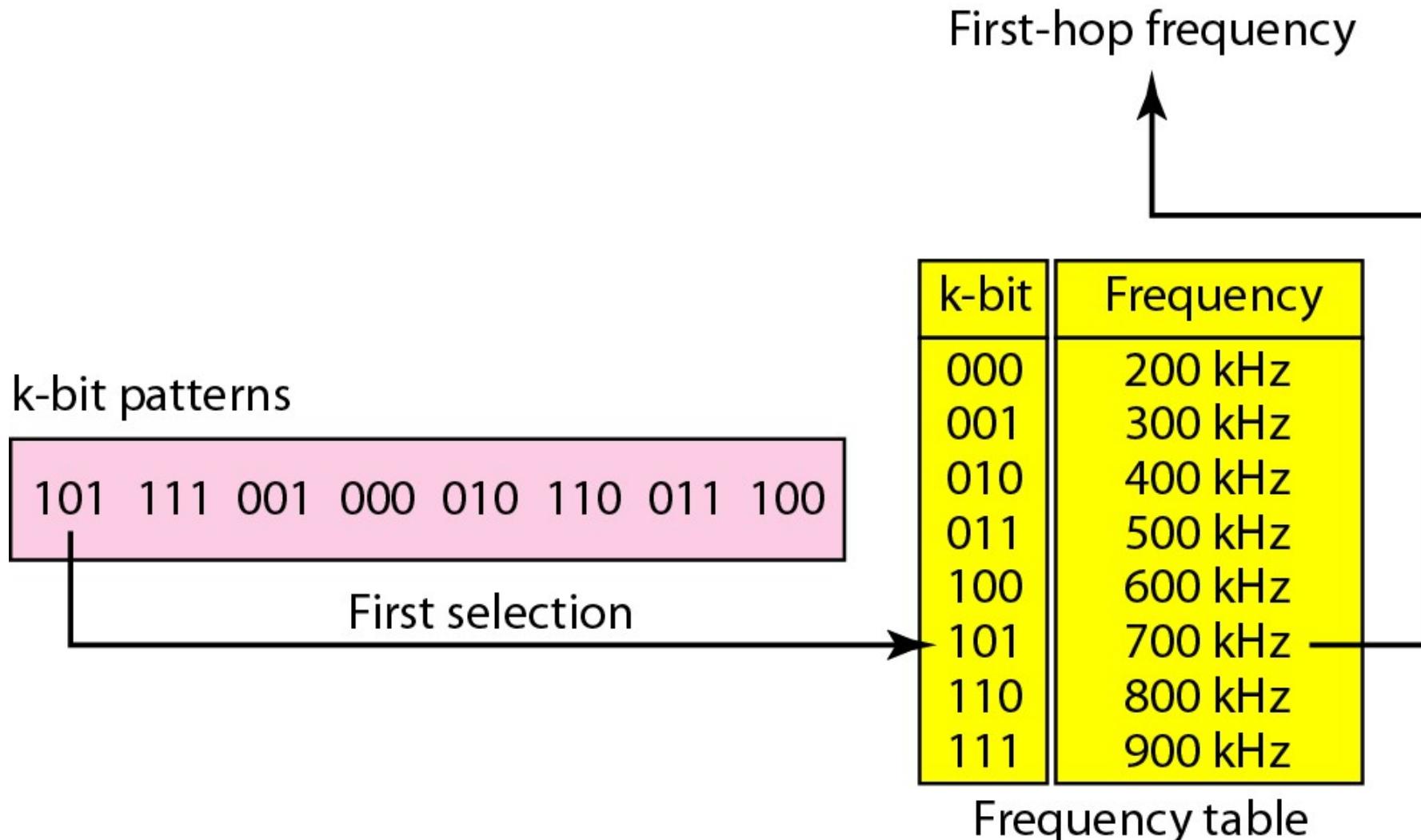
- Frequency-hopping spread spectrum (FHSS)
- direct-sequence spread spectrum (DSSS)
- time-hopping spread spectrum (THSS)
- chirp spread spectrum (CSS)
- The combinations of these techniques

FHSS

transmitting radio signals by rapidly switching a carrier among many frequency channels (**HOPS**), using a pseudorandom sequence known to both transmitter and receiver.



Frequency selection in FHSS

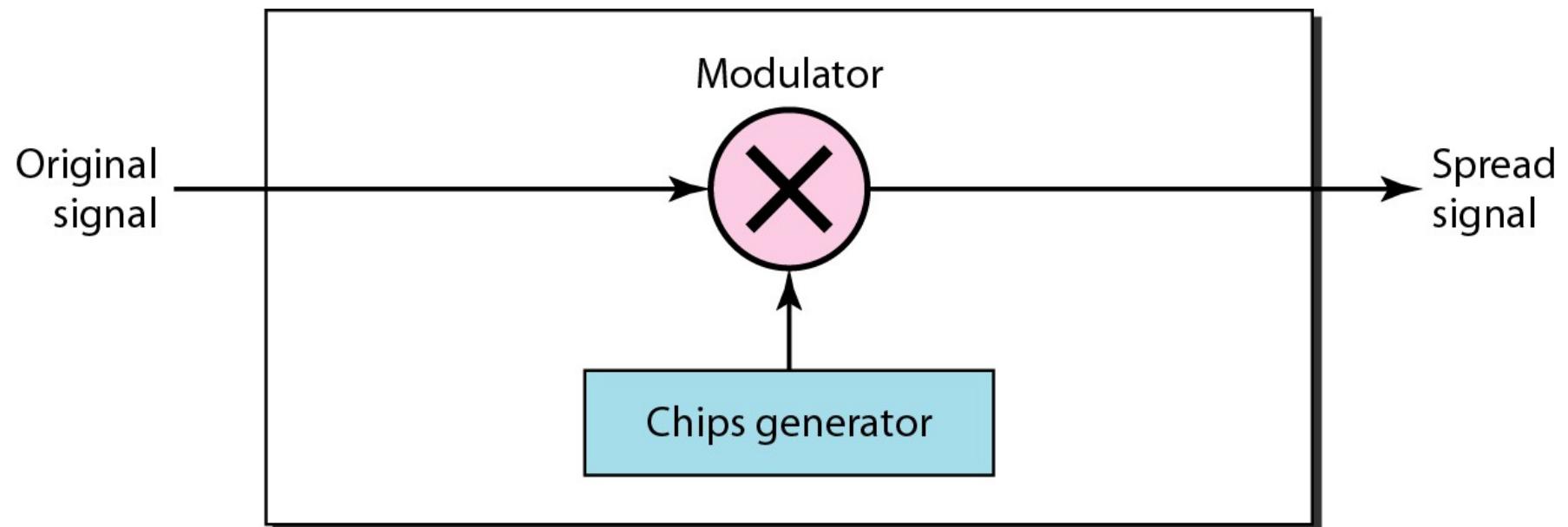


FHSS

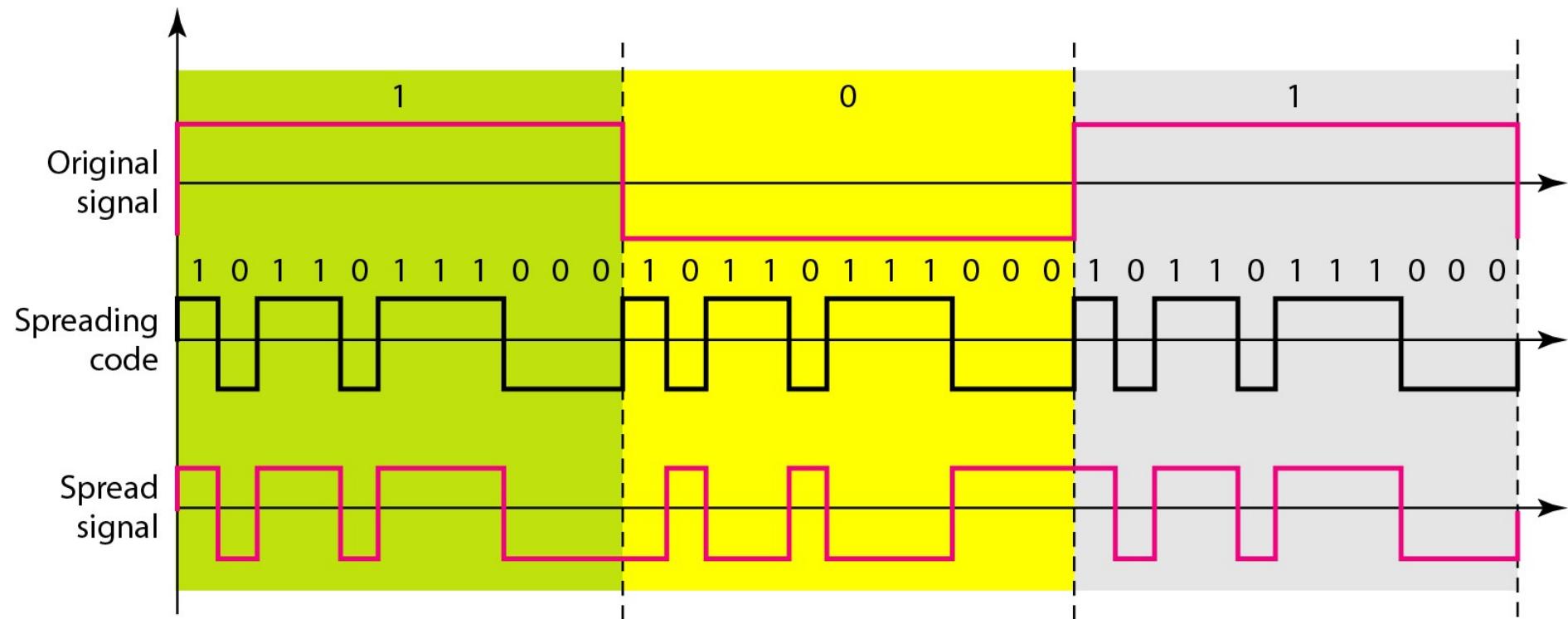
- The original 802.11 FHSS standard supports 1 and 2 Mbps data rate.
- FHSS uses the 2.402 – 2.480 GHz frequency range in the ISM band.
 - It splits the band into 79 non-overlapping channels with each channel 1 MHz wide.
 - FHSS hops between channels at a minimum rate of 2.5 times per second. Each hop must cover at least 6 MHz.
 - The hopping channels for the US and Europe are shown below.

Set	Hopping Pattern
1	{0,3,6,9,12,15,18,21,24,27,30,33,36,39,42,45,48,51,54,57,60,63,66,69,72,75}
2	{1,4,7,10,13,16,19,22,25,28,31,34,37,40,43,46,49,52,55,58,61,64,67,70,73,76}
3	{2,5,8,11,14,17,20,23,26,29,32,35,38,41,44,47,50,53,56,59,62,65,68,71,72,77}

DSSS spreads the signal by combining information *bits* (data signal) with a higher data rate bit sequence - *pseudorandom number (PN)* - called a **Chipping Code**



DSSS example



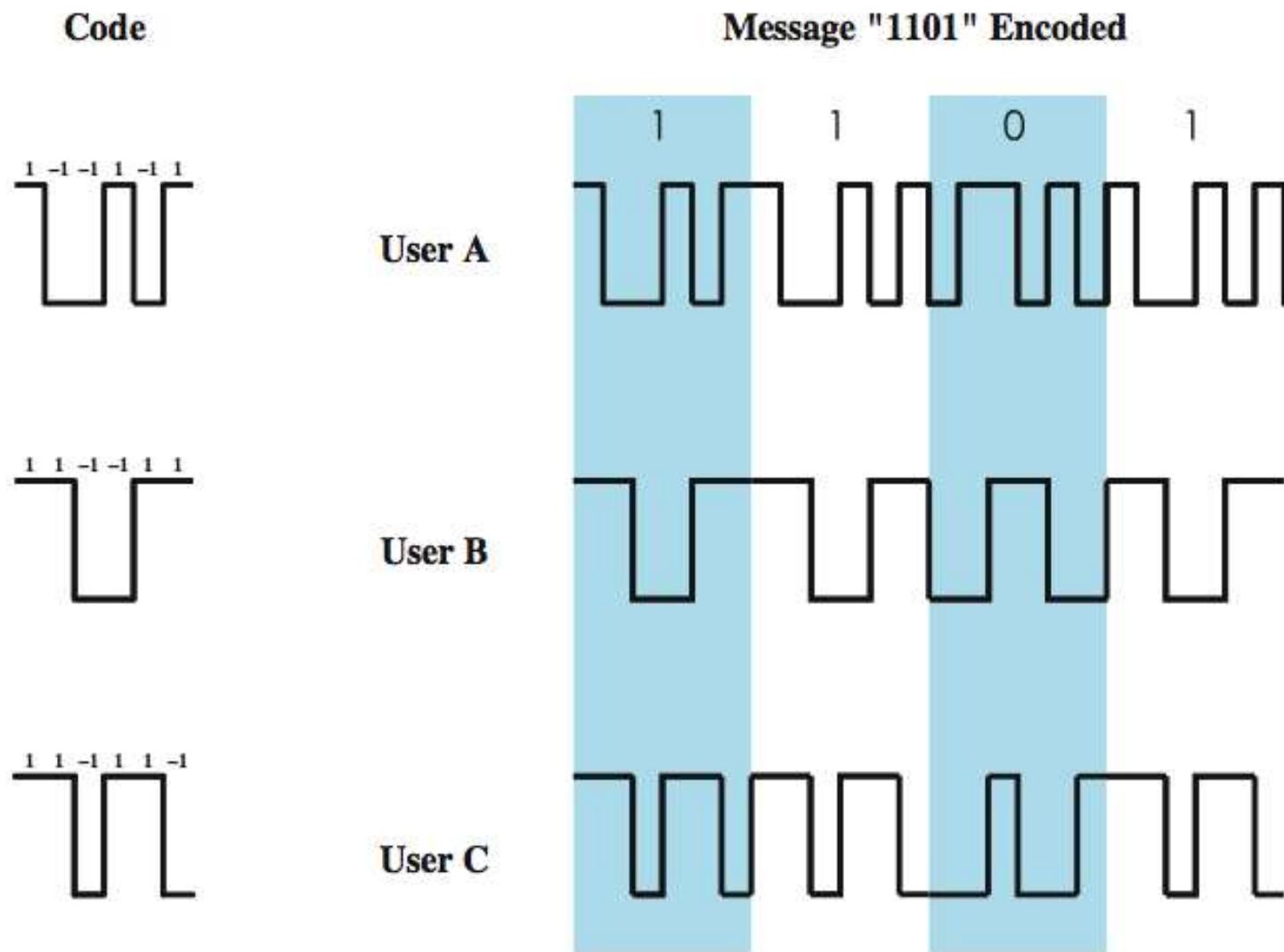
Code Division Multiple Access (CDMA)

■ Basic Principles of CDMA

- D = rate of data signal (bit data rate)
- Break each bit into k *chips*
 - Chips are a user-specific fixed pattern (user code, chipping code)
- Chip data rate of new channel = kD

■ With CDMA, the receiver can sort out transmission from the desired sender, even when there may be other users broadcasting in the same cell.

CDMA Example



码片序列(chip sequence)

- 每个站被指派一个唯一的 m bit 码片序列。
 - 如发送比特 1，则发送自己的 m bit 码片序列。
 - 如发送比特 0，则发送该码片序列的二进制反码。
- 例如，S 站的 8 bit 码片序列是 00011011。
 - 发送比特 1 时，就发送序列 00011011，
 - 发送比特 0 时，就发送序列 11100100。
- S 站的码片序列：(-1 -1 -1 +1 +1 -1 +1 +1)

每个站分配的码片序列不仅必须各不相同，并且还必须互相正交(orthogonal)。
在实用的系统中是使用伪随机码序列。

码片序列的正交关系

- 令向量 S 表示站 S 的码片向量，令 T 表示其他任何站的码片向量。
- 两个不同站的码片序列正交，就是向量 S 和 T 的规格化**内积**(inner product)都是 0：

$$S \bullet T \equiv \frac{1}{m} \sum_{i=1}^m S_i T_i = 0$$

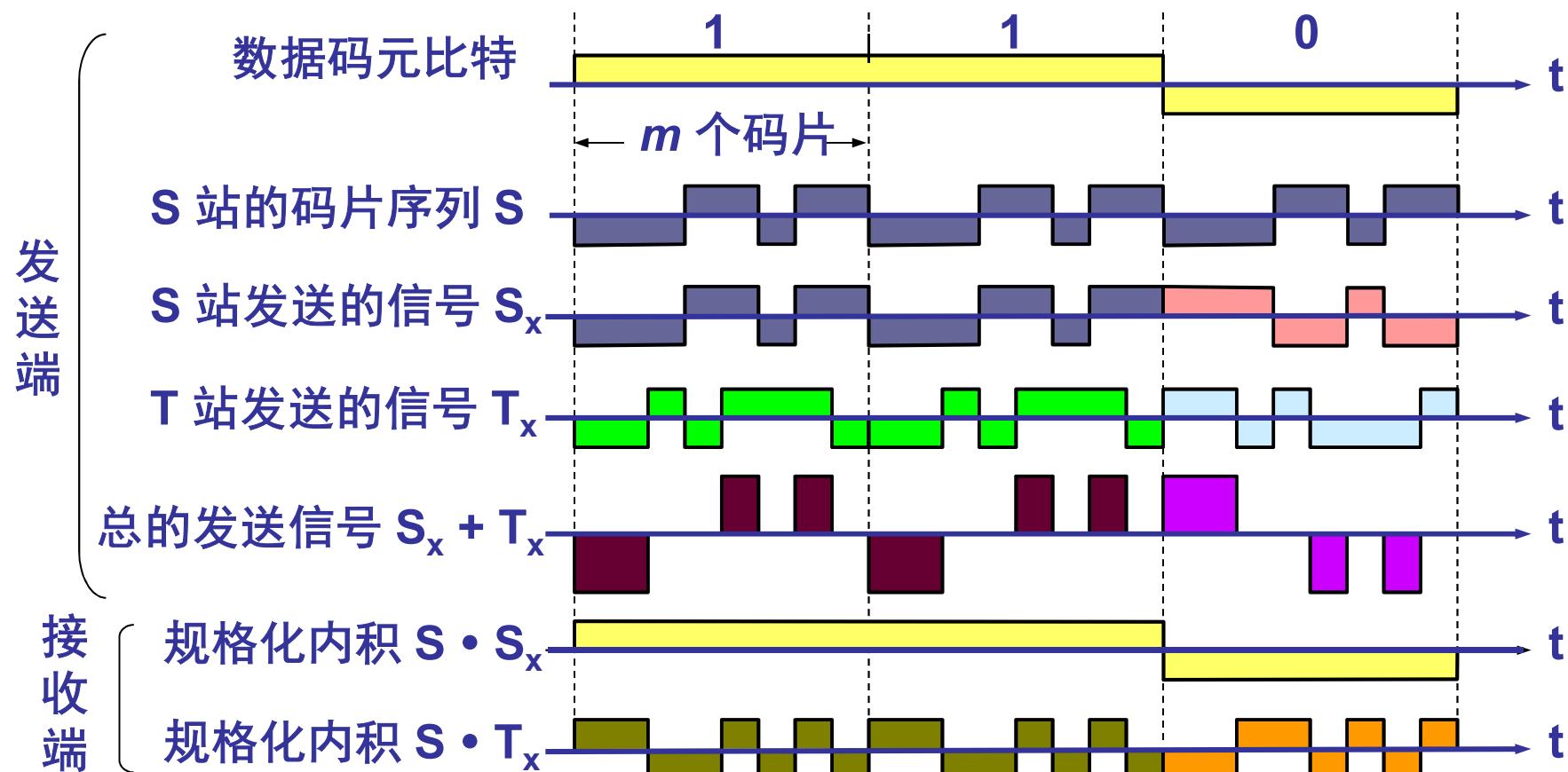
正交关系的另一个重要特性

- 任何一个码片向量和该码片向量自己的规格化内积都是1。

$$\mathbf{S} \bullet \mathbf{S} = \frac{1}{m} \sum_{i=1}^m S_i S_i = \frac{1}{m} \sum_{i=1}^m S_i^2 = \frac{1}{m} \sum_{i=1}^m (\pm 1)^2 = 1$$

- 一个码片向量和该码片反码的向量的规格化内积值是 -1。

CDMA 的工作原理



CDMA – Code Division Multiple Access

A: 0 0 0 1 1 0 1 1
B: 0 0 1 0 1 1 1 0
C: 0 1 0 1 1 1 0 0
D: 0 1 0 0 0 0 1 0

(a)

A: (-1 -1 -1 +1 +1 -1 +1 +1)
B: (-1 -1 +1 -1 +1 +1 +1 -1)
C: (-1 +1 -1 +1 +1 +1 -1 -1)
D: (-1 +1 -1 -1 -1 -1 +1 -1)

(b)

Six examples:

-- 1 - **C**
- 1 1 - **B** + **C**
1 0 -- **A** + **B**
1 0 1 - **A** + **B** + **C**
1 1 1 1 **A** + **B** + **C** + **D**
1 1 0 1 **A** + **B** + **C** + **D**

(c)

$S_1 = (-1 +1 -1 +1 +1 +1 -1 -1)$
 $S_2 = (-2 \ 0 \ 0 \ 0 +2 +2 \ 0 -2)$
 $S_3 = (\ 0 \ 0 -2 +2 \ 0 -2 \ 0 +2)$
 $S_4 = (-1 +1 -3 +3 +1 -1 -1 +1)$
 $S_5 = (-4 \ 0 -2 \ 0 +2 \ 0 +2 -2)$
 $S_6 = (-2 -2 \ 0 -2 \ 0 -2 +4 \ 0)$

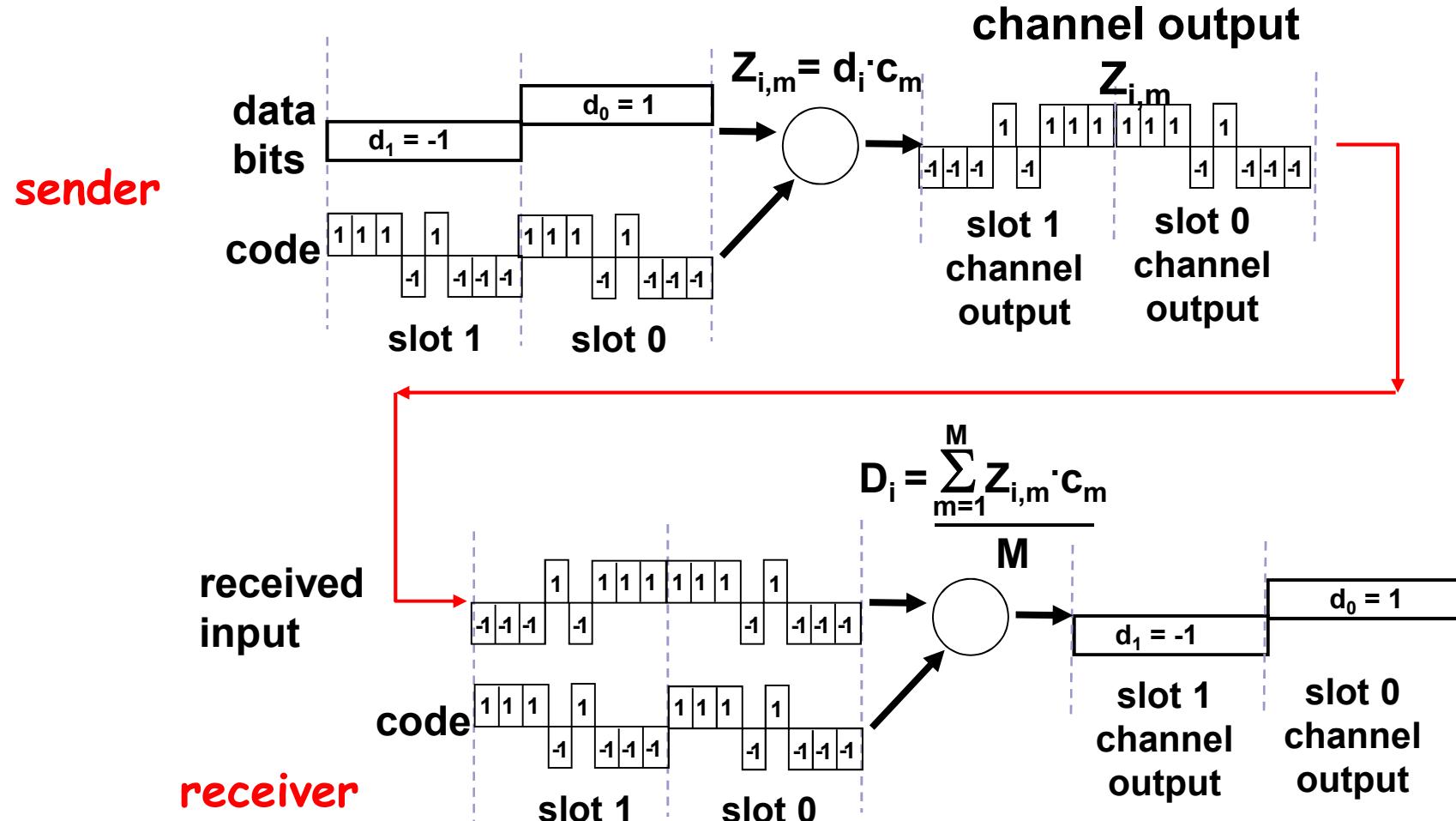
$S_1 \bullet C = (1 +1 +1 +1 +1 +1 +1 +1)/8 = 1$
 $S_2 \bullet C = (2 +0 +0 +0 +2 +2 +0 +2)/8 = 1$
 $S_3 \bullet C = (0 +0 +2 +2 +0 -2 +0 -2)/8 = 0$
 $S_4 \bullet C = (1 +1 +3 +3 +1 -1 +1 -1)/8 = 1$
 $S_5 \bullet C = (4 +0 +2 +0 +2 +0 -2 +2)/8 = 1$
 $S_6 \bullet C = (2 -2 +0 -2 +0 -2 -4 +0)/8 = -1$

(d)

$$\text{正交: } S \bullet T = \frac{1}{m} \sum_{i=1}^m S_i T_i = 0$$
$$S \bullet S = 1; \quad S \bullet \bar{S} = -1$$

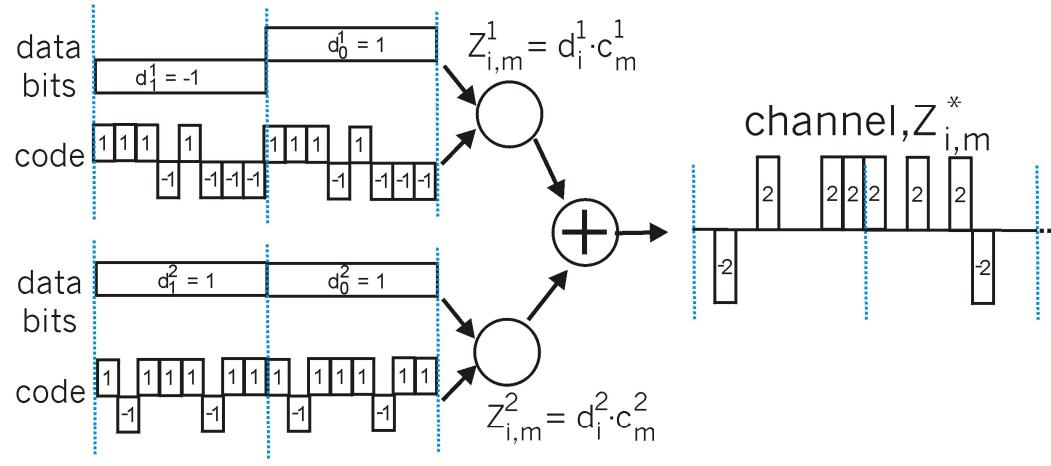
- (a) Binary chip sequences for four stations**
- (b) Bipolar chip sequences**
- (c) Six examples of transmissions**
- (d) Recovery of station C's signal**

CDMA – Code Division Multiple Access

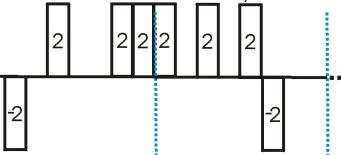


CDMA – Code Division Multiple Access

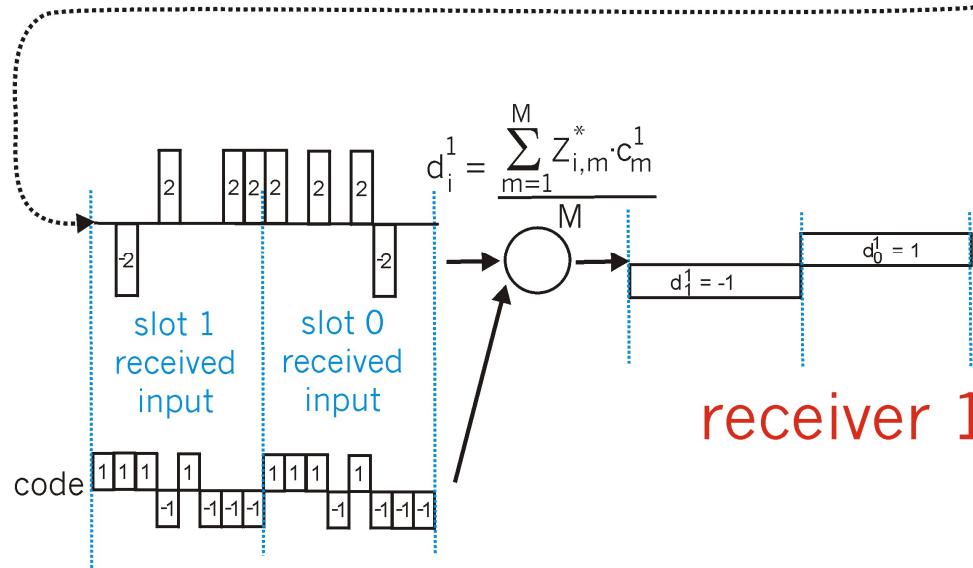
senders



channel, $Z_{i,m}^*$



receiver 1



CS BIT

RS-232-C: An Example Protocol of Physical Layer

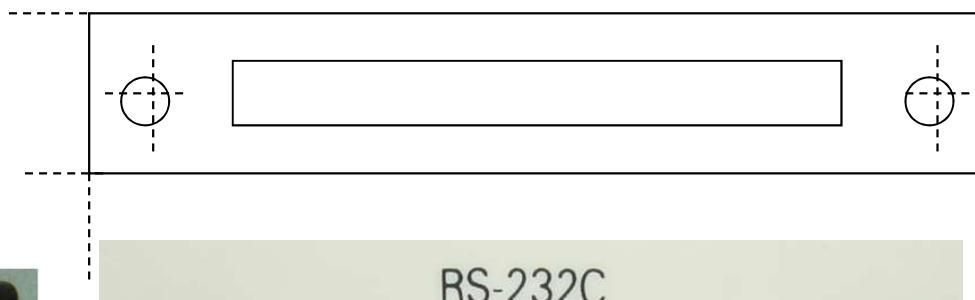
- 规范了计算机与调制解调器间的一个串行物理接口标准。
- 电子工业协会**EIA (Electronic Industries Association)**1969年制定的标准。



RS-232-C: An Example Protocol of Physical Layer

■ Mechanical

- 25芯或9芯D型连接器， DTE侧为插针， DCE侧为插孔。





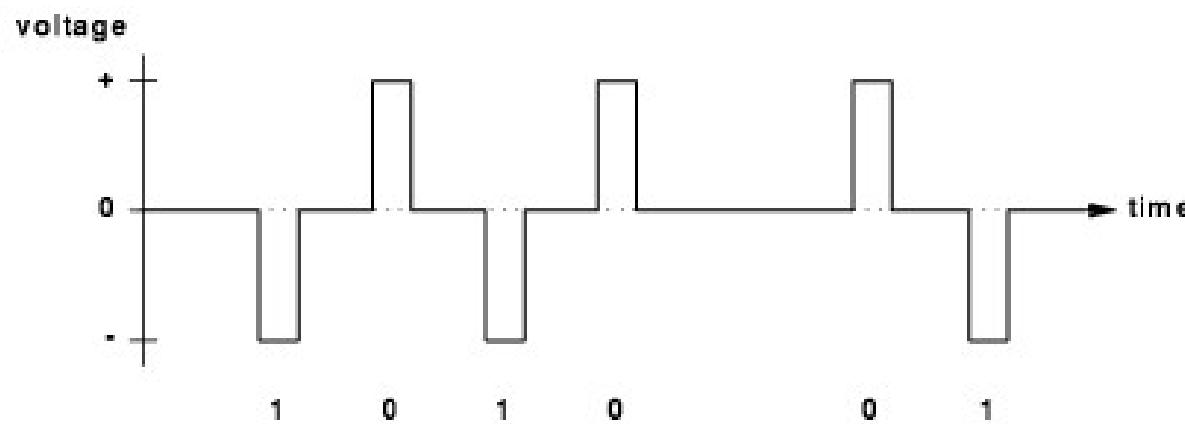
RS-232-C: An Example Protocol of Physical Layer

■ Electrical: Negative logic

“0”: +3v ~ +15v, we use +12v

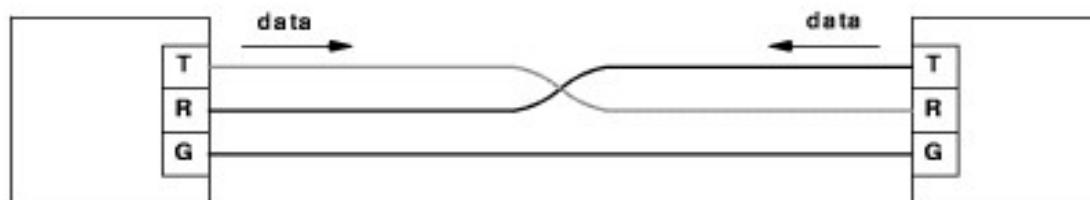
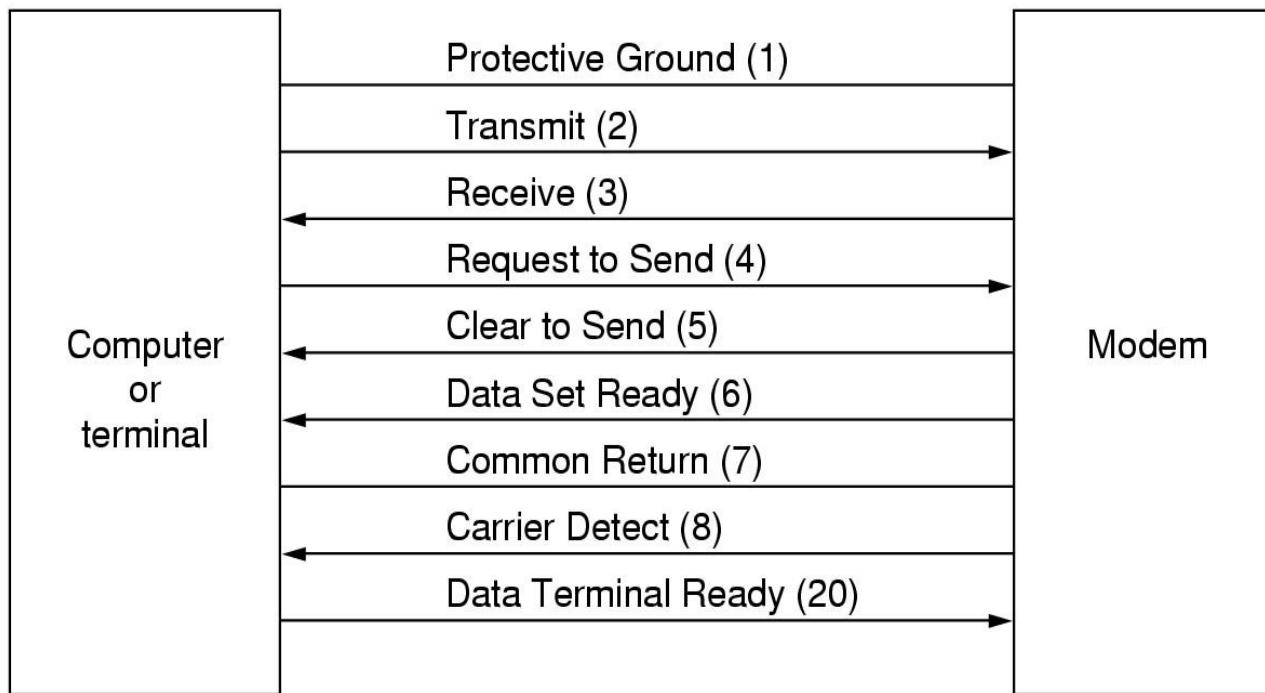
“1”: -3v ~ -15v, we use -12v

-3v ~ +3v: not defined, illegitimate



RS-232-C: An Example Protocol of Physical Layer

■ Functional



Physical-layer Device

■ NIC (网卡)

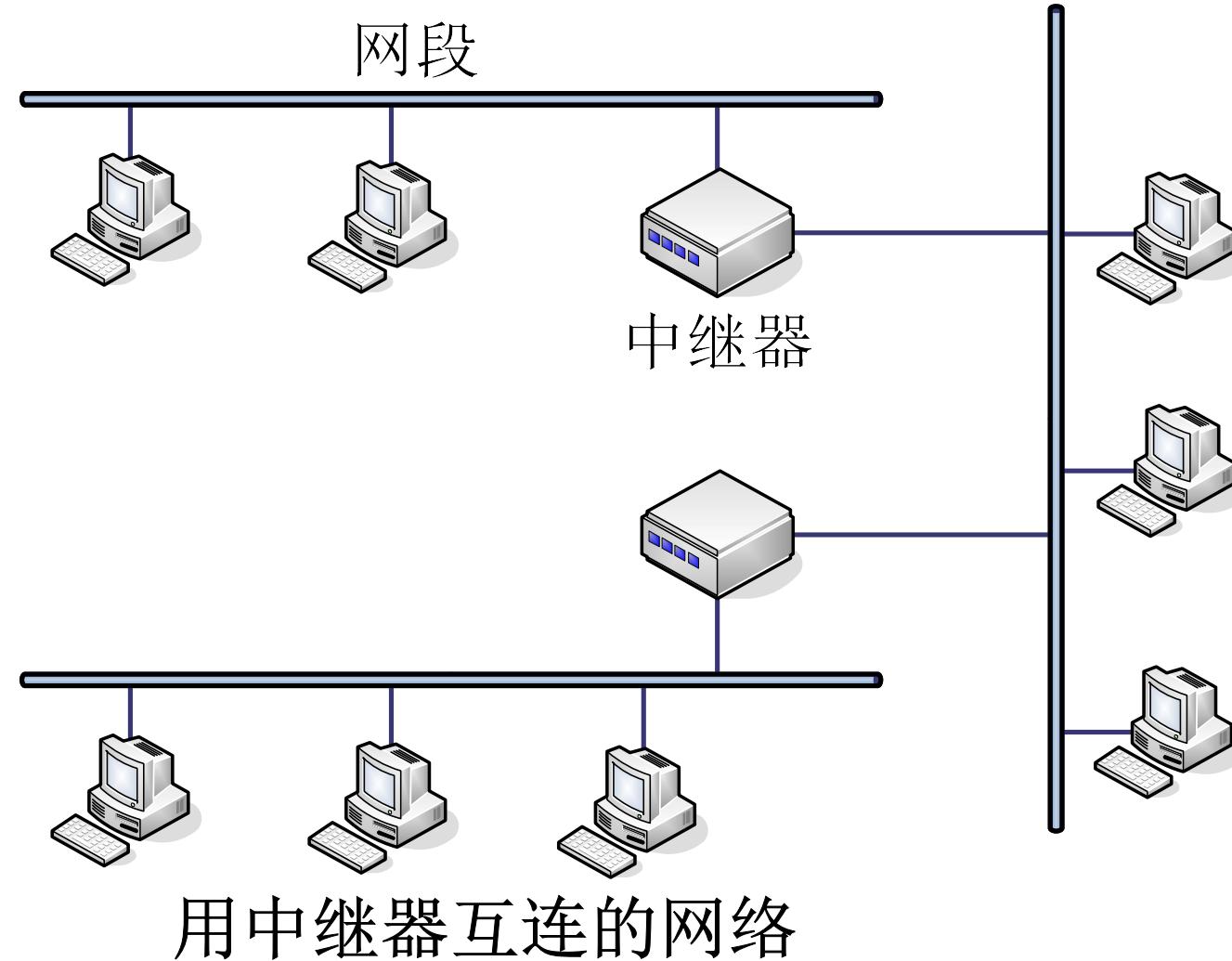
- 典型实现物理层和数据链路层功能

■ Repeater (中继器)

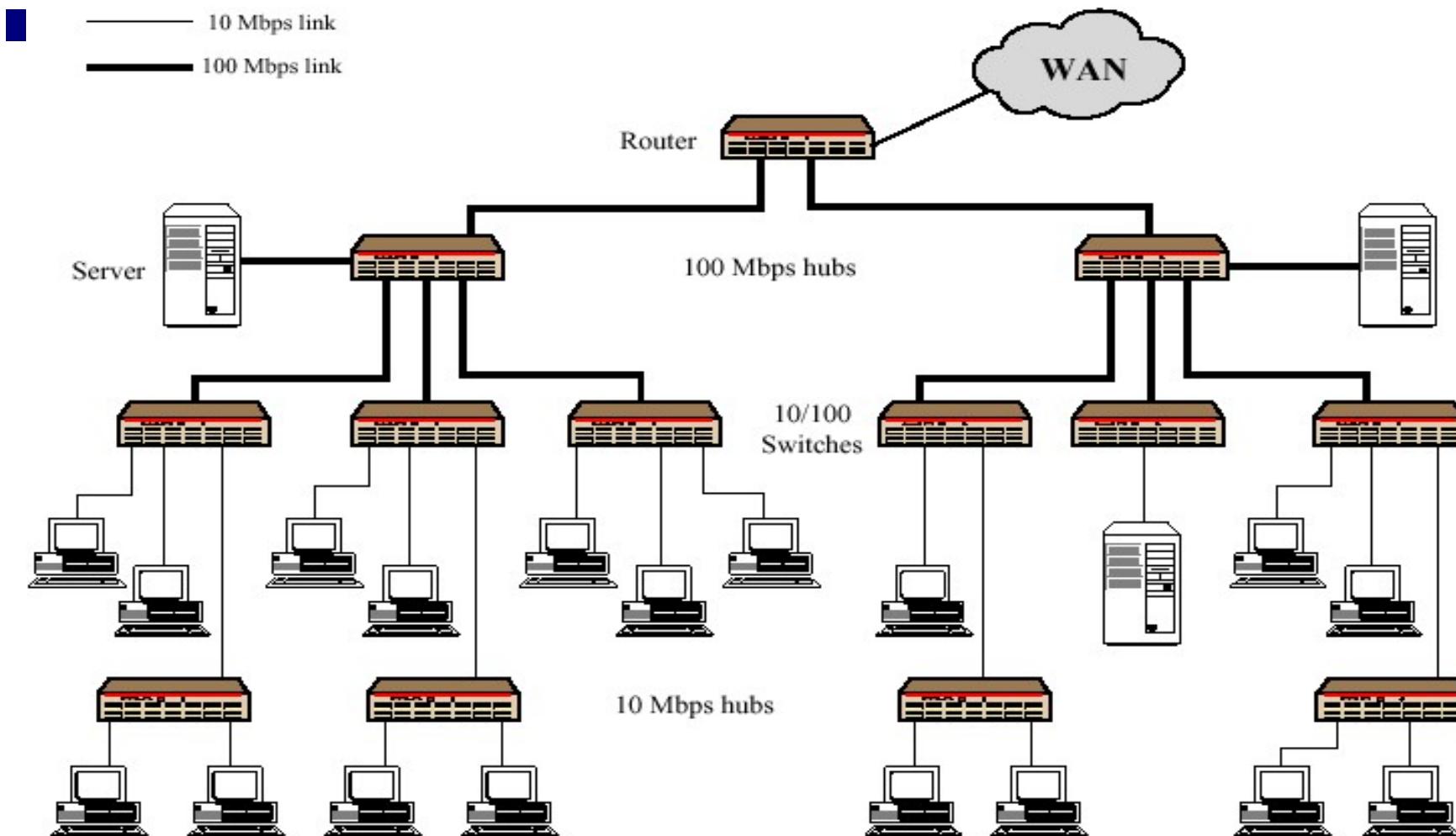
- 实现信号的再生、放大和转发
- 主要用于延迟电缆长度，扩展网段距离
- 可以互连不同的传输介质
- 不能隔离网段（即广播会通过中继器）
- 互连的网络在逻辑上是一个网络（即一个物理网络系统）。(中继器对站点透明)

■ Hub (集线器) 多端口中继器

Physical-layer Device



Physical-layer Device



用集线器构建的网络