Physical Layer

CS44 Data Communications and Networking

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Analog signal

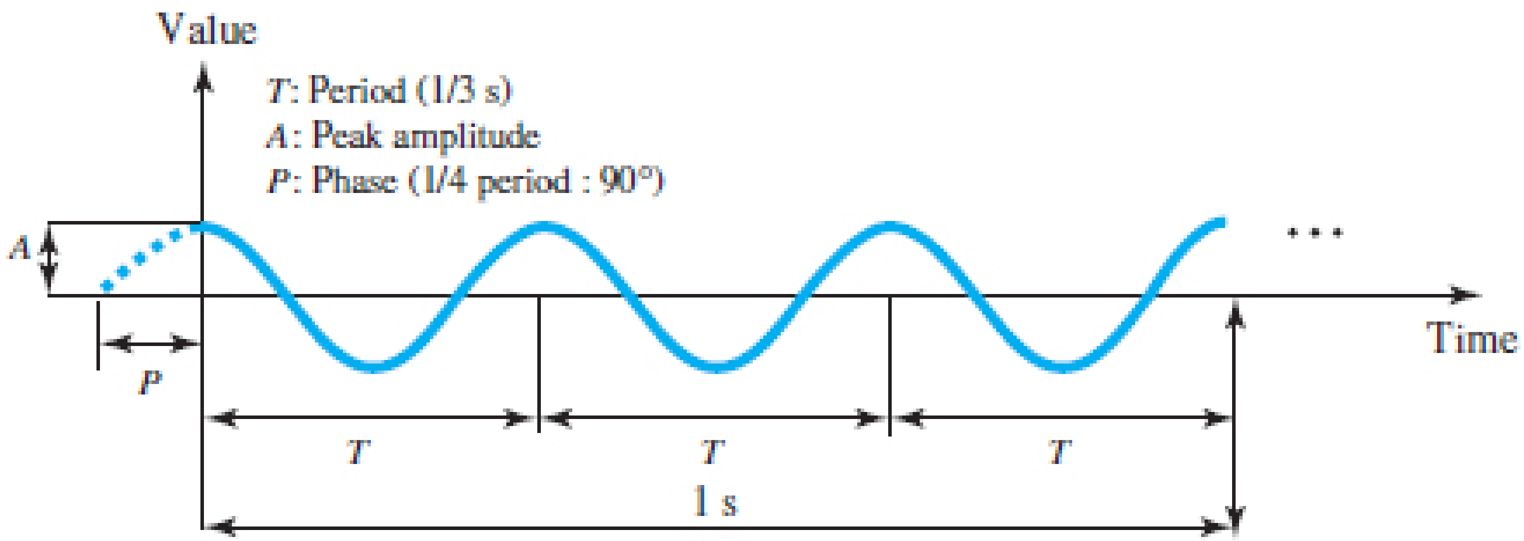
- An analog signal can take one of two forms
  - *periodic* -commonly use
    - simple - a **sine wave, cannot be** decomposed into simpler signals
    - Composite - **made up of many simple sine**
      - The range of frequencies contained in a composite signal is its **bandwidth.**
      - The bandwidth is the difference between the lowest and highest frequencies in the signal
      - For example, if a composite signal contains frequencies between 1000 and 5000, its bandwidth is 5000 – 1000, or 4000.
  - *aperiodic (nonperiodic)*

Sine wave

- The most fundamental form of a periodic analog signal.
- Represented by three parameters:
  - *Period (T)* ***- the amount of time, in seconds, a signal needs to*** complete one cycle
    - The ***frequency (f), measured in hertz (Hz), refers to the*** number of periods in 1 s.
    - Period and frequency are inverses of each other, in other words ($f = 1/T$).
  - *peak amplitude* - ***absolute value of its highest intensity.***
    - For electrical signals, peak amplitude is normally measured in volts.
  - *Phase* - position of the waveform relative to time 0
    - If we think of the wave as something that can be shifted backward or forward along the time axis, phase describes the amount of that shift
    - It indicates the status of the first cycle
    - Phase is measured in degrees or radians (360° is 2π rad)

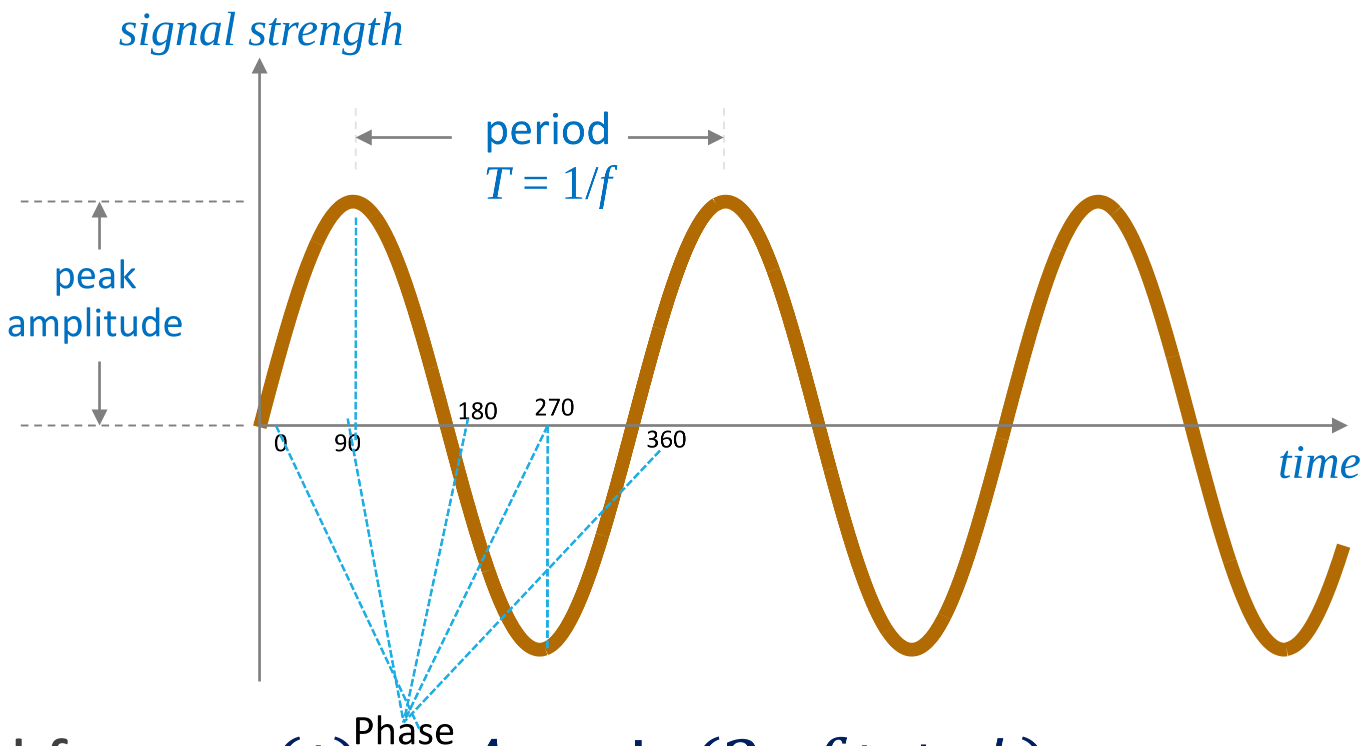
Sine wave

- The voltage of a battery is constant (for example, 1.5 V). However, this can be considered to be periodic with a frequency of 0 (and a period of infinity).
- The electrical voltage in our homes in the United States is periodic with a peak value between 110 to 120 V. Its frequency is 60 Hz.



Sine Waves: Revisited

- Simplest form of periodic signal



- General form: $y(t) = A \times \sin(2\pi f t + \phi)$

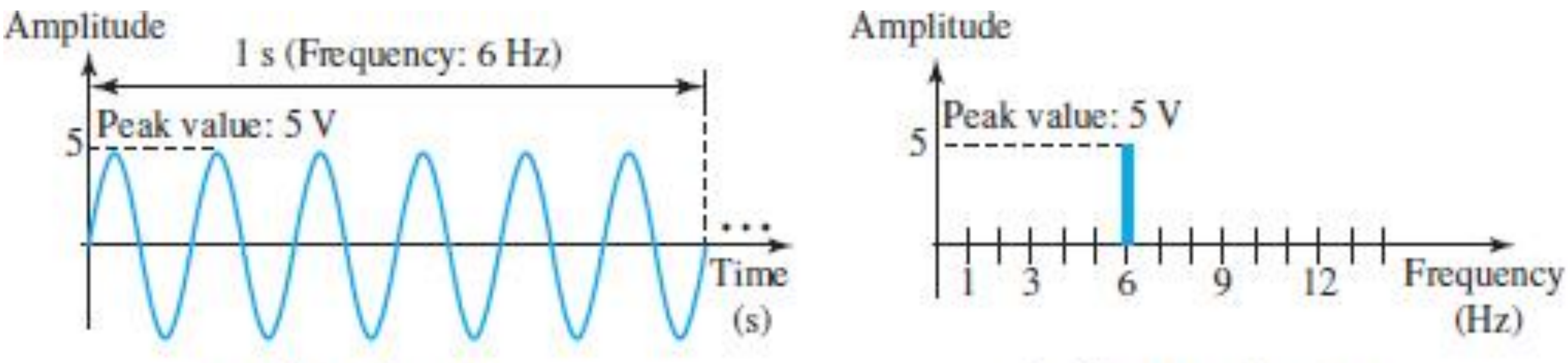
phase / phase shift

Sine Wave - Wavelength

- **Characteristic of a signal traveling through a** transmission medium
- The distance a simple signal can travel in one period
- The wavelength binds the period or the frequency of a simple sine wave to the propagation speed in the medium.\
- While the frequency of a signal is independent of the medium, the wavelength depends on both the frequency and the medium
- The wavelength can be calculated if one is given the propagation speed of the medium and the frequency of the signal
- If we represent wavelength by λ, propagation speed by *c, and frequency by f, we get* $\lambda = c / f = c \times T$

Sine Wave - *Time and Frequency Domains*

- Sine wave shown by using what is called a **timedomain plot, which shows changes in signal amplitude with respect to time.**
- To show the relationship between amplitude and frequency, we can use a **frequency-domain plot**
- In the frequency domain, a sine wave is represented by one spike
- The position of the spike shows the frequency; its height shows the peak amplitude.

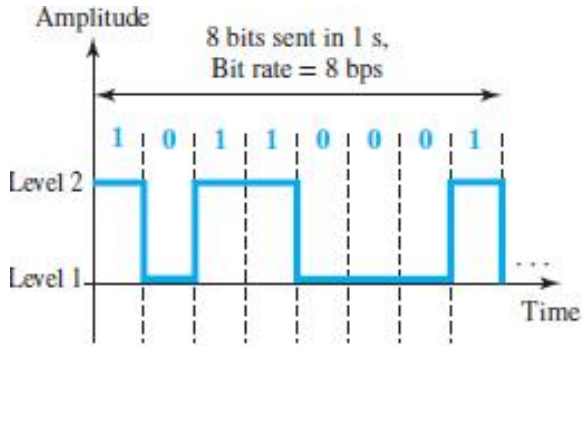


a. Time domain

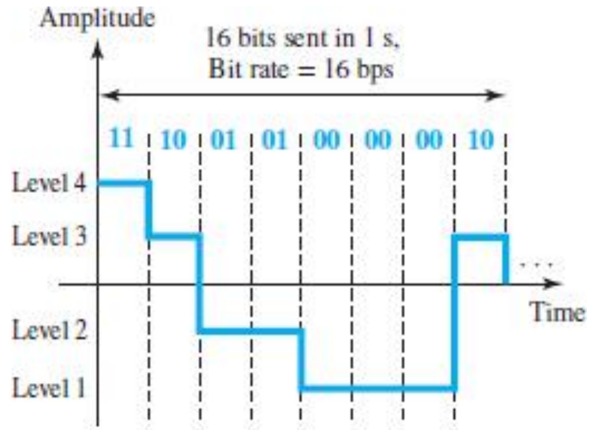
b. Frequency domain

Digital signal

- Information can also be represented by a digital signal
  - For example, a value 1 can be encoded as a positive voltage and a value 0 as zero voltage.
- A digital signal can have more than two levels- send more than 1 bit for each level
- We send 1 bit per level in Figure a and 2 bits per level in Figure b.
- In general, if a signal has *L levels, each level needs* $log_2 L$ *bits.*



a. A digital signal with two levels



b. A digital signal with four levels

Digital signal – Bit rate

- Digital signals are nonperiodic, and thus period and frequency are not appropriate characteristics of digital signals
- *bit rate (instead* of *frequency)—is used to describe digital signals.* - ***the number*** of bits sent in 1 s, expressed in bits per second (bps).
- The bit rate can be represented as kbps (kilo bits per second, where kilo means one thousand) or Mbps (mega bits per second, where mega means one million)
- Assume we need to download text documents at the rate of 100 pages per minute. What is the required bit rate of the channel?
  - A page is an average of 24 lines with 80 characters in each line.
  - If we assume that one character requires 8 bits, the bit rate is **100 × 24 × 80 × 8 = 1,536,000 bps = 1.536 Mbps**

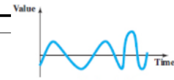
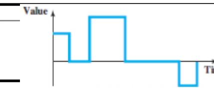
Digital signal – *Bit Length*

- We discussed the concept of the wavelength for an analog signal: the distance one cycle occupies on the transmission medium.
- We can define something similar for a digital signal: the bit length.
- The **bit length is the distance 1 bit** occupies on the transmission medium
  - **Bit length = 1 / (bit rate)**
- The length of the bit in Example of to download text documents is **1/1,536,000 = 0.000000651 s = 0.651 μs**

Transmission of Digital Signals

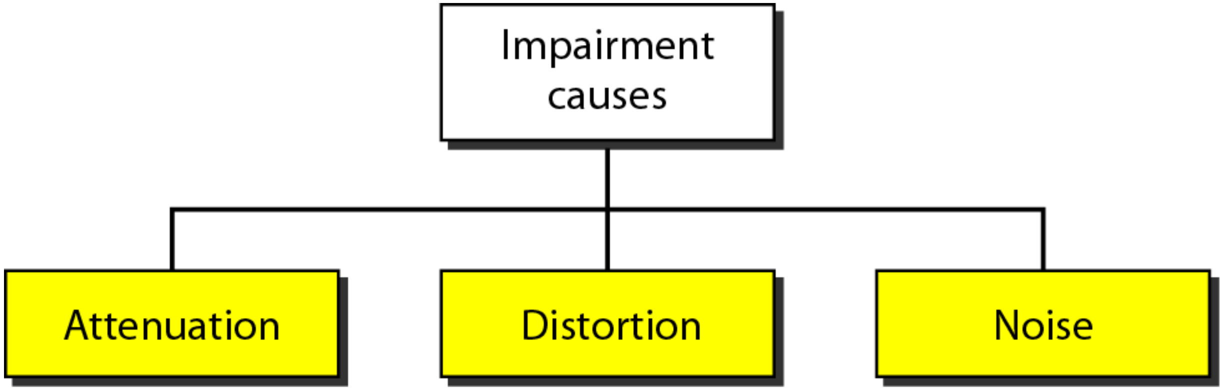
- A digital signal is a composite analog signal with frequencies between zero and infinity
- how can we send a digital signal from point A to point B?
- We can transmit a digital signal by using one of two different approaches:
  - ***Baseband transmission means*** sending a digital signal over a channel without changing it to an analog signal
  - Broadband transmission or modulation means changing the digital signal to an analog signal for transmission

Analog Vs Digital

Analog Signal 	Digital Signal 
a continuous wave that changes over a time period.	a discrete wave that carries information in binary form.
represented by a sine wave.	represented by square waves.
described by the amplitude, period or frequency, and phase.	described by bit rate and bit intervals.
has no fixed range.	has a finite range i.e. between 0 and 1.
more prone to distortion.	less prone to distortion.
transmit data in the form of a wave.	carries data in the binary form i.e. 0 and 1.
The human voice is the best example of an analog signal.	Signals used for transmission in a computer are the digital signal.
analog transmission is the only choice if we have a bandpass channel.	while digital transmission is very desirable, a low-pass channel is needed.
Converting a low-pass analog signal to a bandpass analog signal is traditionally called analog-to-analog conversion.	Converting digital data to a bandpass analog signal is traditionally called digital-to-analog conversion.

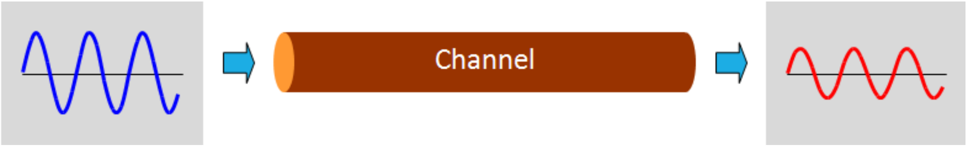
Transmission Impairments

- Signals travel through transmission media→ not perfect
- Imperfection → causes signal impairment
  - Signal at the beginning of the medium is not the same as the signal at the end of the medium
  - What is sent is not what is received
- Three causes of impairment
  - Attenuation
  - Distortion
  - Noise

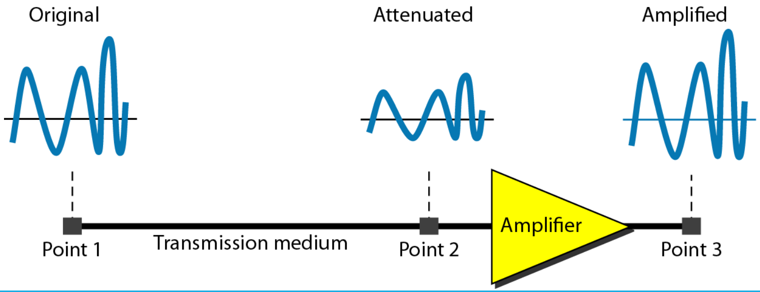


Attenuation

- **Attenuation means a loss of energy**
- **When a signal (simple or composite) travels** through a medium, it loses some of its energy in overcoming the resistance of the medium



  - Why wire carrying electric signals gets warm after a while?
    - Some of the electrical energy in the signal is converted to heat
- To compensate for this loss, amplifiers are used to amplify the signal
- The higher the frequency, the higher the attenuation



Attenuation…

- How to show that a signal has lost or gained strength?
  - engineers use the unit of the decibel
- The **decibel (dB) measures the relative strengths of two signals or one signal at two different** points
  - Positive dB → signal is amplified (gains strength)
  - Negative dB → signal is attenuated (loses strength)
- Decibel → defined in terms of voltage or power
  - because power is proportional to the square of the voltage

Attenuation - Relative Signal Strength

- Measured in *Decibel (dB)*

dB in terms of	Formula	variables
Power	dB = 10 $\log_{10}$ *(P2/P1)*	*P1 and P2 → powers of a signal at points 1 and 2, respectively*
voltage	dB = 20 $\log_{10}$ *(V2/V1)*	*V1 and V2 → voltage of a signal at points 1 and 2, respectively*

Attenuation Example 1

- Suppose a signal travels through a transmission medium and its power is reduced to one-half
  - Means that P2 is (1/2)P1
- In this case, the attenuation (loss of power) can be calculated as

$$10 \log_{10} \frac{P_2}{P_1} = 10 \log_{10} \frac{0.5 P_1}{P_1} = 10 \log_{10} 0.5 = 10(-0.3) = -3 \text{ dB}$$

- A loss of 3 dB (–3 dB) is equivalent to losing one-half the power

Attenuation Example 2

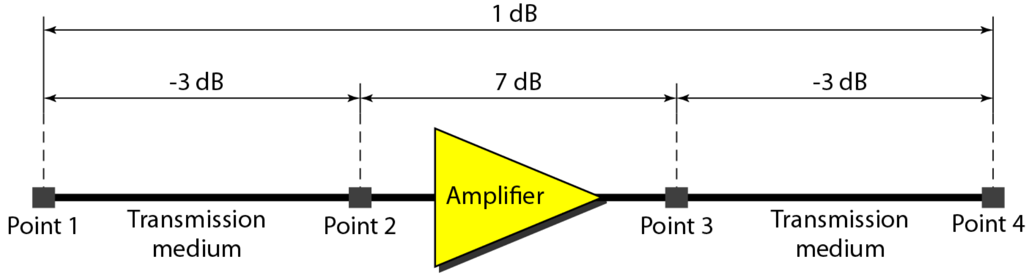
- A signal travels through an amplifier, and its power is increased 10 times
  - Means that *P2 = 10P1*
- *In this case, the amplification (gain of power) can be calculated as*

$$\boxed{10 \log_{10} \frac{P_2}{P_1} = 10 \log_{10} \frac{10P_1}{P_1}}$$

$$\boxed{= 10 \log_{10} 10 = 10(1) = 10 \text{ dB}}$$

Attenuation Example 3

- One reason that engineers use the decibel to measure the changes in the strength of a signal → decibel numbers can be added (or subtracted) when we are measuring several points (cascading) instead of just two
- A signal travels from point 1 to point 4



  - The signal is attenuated by the time it reaches point 2
  - Between points 2 and 3, the signal is amplified
  - Again, between points 3 and 4, the signal is attenuated
- In this case, the decibel value can be calculated as

$$\boxed{\mathrm{dB} = -3 + 7 - 3 = +1}$$

The signal has gained in power.

Attenuation Example 4

- Sometimes the decibel is used to measure signal power in milliwatts
  - referred as dB_m
  - calculated as $dB_m = 10 \log_{10} P_m$
    - where P_m is the power in milliwatts
- Calculate the power of a signal with $dB_m = -30$
- We can calculate the power in the signal as

$$\boxed{\begin{array}{c} \mathrm{dB_m} = 10 \log_{10} P_m = -30 \\ \log_{10} P_m = -3 \qquad P_m = 10^{-3}\ \mathrm{mW} \end{array}}$$

Attenuation Example 5

- The loss in a cable is usually defined in decibels per kilometer (dB/km)
- If the signal at the beginning of a cable with −0.3 dB/km has a power of 2 mW, what is the power of the signal at 5 km?
- Solution: The loss in the cable in decibels is 5 × (−0.3) = −1.5 dB. We can calculate the power as

$$\text{dB} = 10 \log_{10} \frac{P_2}{P_1} = -1.5$$

$$\frac{P_2}{P_1} = 10^{-0.15} = 0.71$$

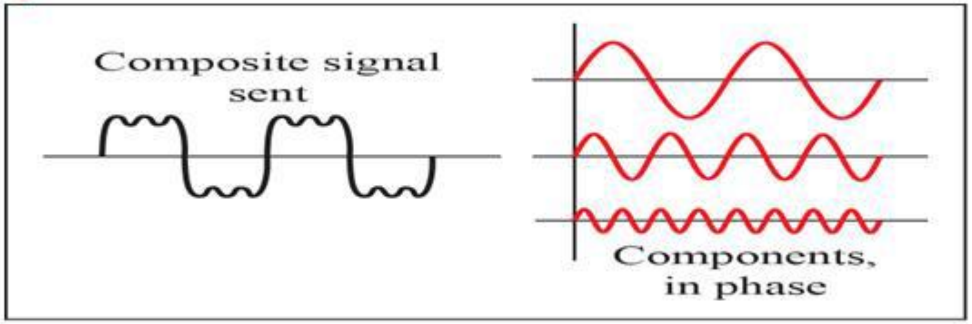
$$P_2 = 0.71 P_1 = 0.7 \times 2 = 1.4 \text{ mW}$$

Distortion

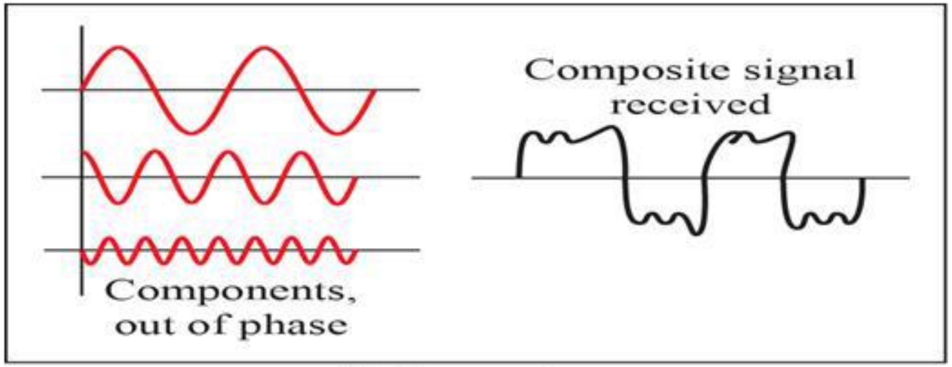
- Distortion means that the signal changes its form or shape
  - occur in a composite signal made of different frequencies.
- Each signal component has its own propagation speed through a medium
  - its own delay in arriving at the final destination
- Propagation velocity varies with frequency

Distortion …

- If the delay is not exactly the same as the period duration, it may create a difference in phase
  - Signal components at the receiver have phases different from what they had at the sender
  - The shape of the composite signal is therefore not the same



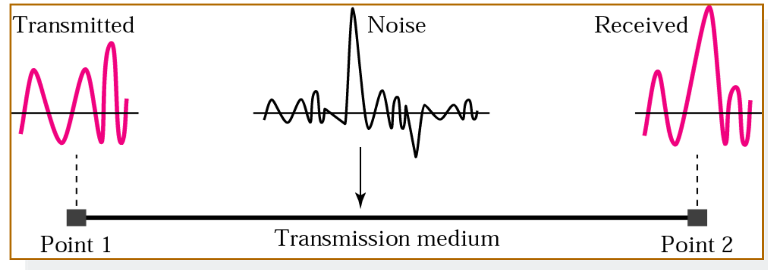
At the sender

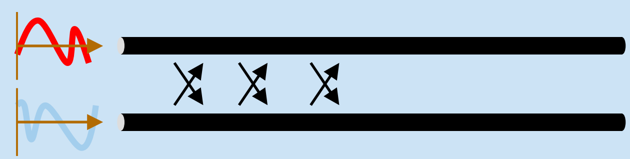
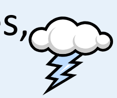


At the receiver

Noise

- Noise ⇨ Undesirable signals added between the transmitter and the receiver
  - may corrupt the signal



Types of noise	Description
thermal noise	• random motion of electrons in a wire ◦ creates an extra signal not originally sent by the transmitter
induced noise	• comes from sources such as motors and appliances ◦ Devices act as a sending antenna ◦ Transmission medium acts as the receiving antenna
crosstalk	• effect of one wire on the other ◦ One wire acts as a sending antenna ◦ Other wire as the receiving antenna 
impulse noise	a spike (a signal with high energy in a very short time) that comes from power lines, lightning, and so on 

Noise : Signal-to-Noise Ratio

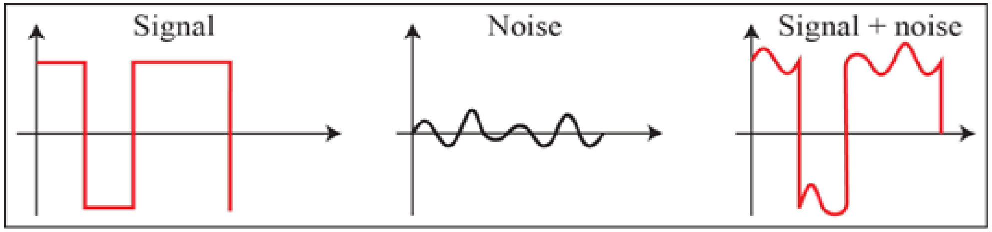
- SNR is the ratio of what is wanted (signal) to what is not wanted (noise).
  - A measurement of signal reception's quality
    - often described in decibel units, SNR_{dB}

$$SNR = \frac{\text{average signal power}}{\text{average noise power}}$$

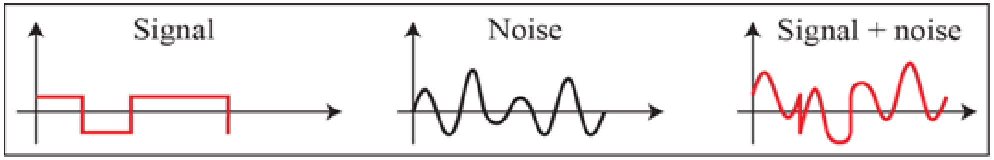
$$SNR_{dB} = 10 \log_{10} SNR$$

- high SNR means the signal is less corrupted by noise.
- Low SNR means the signal is more corrupted by noise

Noise : Signal-to-Noise Ratio



a. High SNR



b. Low SNR

- The power of a signal is 10 mW and the power of the noise is 1 µW; what are the values of SNR and SNR_{dB} ?

$$SNR = \frac{10000\mu w}{1\mu w} = 10000\mu w$$

$$SNR_{dB} = 10\log_{10} 10000 = 10\log_{10} 10^4 = 40$$

Noise : Signal-to-Noise Ratio

- The values of SNR and SNR_{dB} for a noiseless channel are

$$SNR = \frac{\text{signal power}}{0} = \infty$$

$$SNR_{dB} = 10 \log_{10} \infty = \infty$$

- We can never achieve this ratio in real life; it is an ideal

Data Rate Limits

- Data rate: How fast we can send data in bits per second, over a channel?
- Data rate depends on three factors:
  1. **The bandwidth available**
  2. **The level of the signals we use**
  3. **The quality of the channel (the level of noise)**
- Two theoretical formulas were developed to calculate the data rate:
  - Nyquist for a noiseless channel
  - Shannon for a noisy channel.

Noiseless Channel: Nyquist Bit Rate

- Nyquist bit rate formula defines the theoretical maximum bit rate

$$\boxed{\text{Bit Rate} = 2 \times \text{Bandwidth} \times \log_2 L}$$

  - Bandwidth is the bandwidth of the channel
  - *L is the number of signal* levels used to represent data
  - BitRate is the bit rate in bits per second.

Harry Nyquist (1889-1976)

- theoretically , given a specific bandwidth, any bit rate can be calculated by increasing the number of signal levels
- Practically there is a limit
  - Increase in the number of signal levels, a burden on the receiver is imposed
    - If the number of levels in a signal is just 2, the receiver can easily distinguish between a 0 and a 1
    - If the level of a signal is 64, the receiver must be very sophisticated to distinguish between 64 different levels.
      - increasing the levels of a signal reduces the reliability of the system.

Noiseless Channel: Nyquist Bit Rate

- Does the Nyquist theorem bit rate agree with the intuitive bit rate described in baseband transmission?
  - They match when we have only two levels.
  - In baseband transmission, the bit rate is 2 times the bandwidth if we use only the first harmonic in the worst case
  - However, the Nyquist formula is more general and can be applied to baseband transmission and modulation
  - It can be applied when we have two or more levels of signals.

Noiseless Channel: Nyquist Bit Rate

- Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels.
  - The maximum bit rate can be calculated as

$$\boxed{\text{BitRate} = 2 \times 3000 \times \log_2 2 = 6000 \text{ bps}}$$

- Consider the same noiseless channel transmitting a signal with four signal levels (for each level, we send 2 bits).
  - The maximum bit rate can be calculated as

$$\boxed{\text{BitRate} = 2 \times 3000 \times \log_2 4 = 12{,}000 \text{ bps}}$$

Noiseless Channel: Nyquist Bit Rate

- We need to send 265 kbps over a noiseless channel with a bandwidth of 20 kHz. How many signal levels do we need?
  - use the Nyquist formula as shown:

$$265{,}000 = 2 \times 20{,}000 \times \log_2 L$$

$$\log_2 L = 6.625 \qquad L = 2^{6.625} = 98.7 \text{ levels}$$

  - Since this result (L) is not a power of 2, we need to either increase the number of levels or reduce the bit rate
    - If we have 128 levels, the bit rate is 280 kbps
    - If we have 64 levels, the bit rate is 240 kbps

Noisy Channel: *Shannon's Capacity*

- In reality, the channel is always noisy.
- Claude Shannon introduced a formula
  - determine the theoretical highest data rate for a noisy channel: *Shannon's Capacity*

$$\boxed{\text{Capacity} = \text{Bandwidth} \times \log_2(1+\text{SNR})}$$

  - bandwidth is the bandwidth of the channel (Hz)
  - SNR is the signal-to-noise ratio
  - capacity is the capacity of the channel in bits per second

Claude Elwood Shannon (1916-2001)

- No indication of the signal level - means that no matter how many levels we have, we cannot achieve a data rate higher than the capacity of the channel
- It is a characteristic of the channel, not the method of transmission

Example 1 – *Shannon's Capacity*

- Consider an extremely noisy channel in which the value of the signal-to-noise ratio is almost zero.
  - the noise is so strong that the signal is faint.
- For this channel the capacity *C* is calculated as

$$\boxed{C = B \log_2 (1 + \text{SNR}) = B \log_2 (1 + 0) = B \log_2 1 = B \times 0 = 0}$$

- This means that the capacity of this channel is zero regardless of the bandwidth
  - We cannot receive any data through this channel

Example 2 – *Shannon's Capacity*

- We can calculate the theoretical highest bit rate of a regular telephone line. A telephone line normally has a bandwidth of 3000. The signal-to-noise ratio is usually 3162. For this channel the capacity is calculated as

$$\boxed{\begin{aligned} C &= B \log_2 (1 + \text{SNR}) = 3000 \log_2 (1 + 3162) = 3000 \log_2 3163 \\ &= 3000 \times 11.62 = 34{,}860 \text{ bps} \end{aligned}}$$

- This means that the highest bit rate for a telephone line is 34.860 kbps
- If we want to send data faster than this, we can either increase the bandwidth of the line or improve the signal-to-noise ratio.

Example 3 – *Shannon's Capacity*

- The signal-to-noise ratio is often given in decibels. Assume that $SNR_{dB} = 36$ and the channel bandwidth is 2 MHz. The theoretical channel capacity can be calculated as

$$SNR_{dB} = 10 \log_{10} SNR \rightarrow SNR = 10^{SNR_{dB}/10} \rightarrow SNR = 10^{3.6} = 3981$$

$$C = B \log_2 (1 + SNR) = 2 \times 10^6 \times \log_2 3982 = 24 \text{ Mbps}$$

Example 4 – *Shannon's Capacity*

- For practical purposes, when the SNR is very high, we can assume that SNR + 1 is almost the same as SNR. In these cases, the theoretical channel capacity can be simplified to

$$\boxed{C = B \times \frac{\mathrm{SNR_{dB}}}{3}}$$

- For example, we can calculate the theoretical capacity of the previous example as

$$\boxed{C = 2\ \mathrm{MHz} \times \frac{36}{3} = 24\ \mathrm{Mbps}}$$

Using Both Limits

- In practice, both methods used to find the limits and signal levels
- Example 1: We have a channel with a 1-MHz bandwidth. The SNR for this channel is 63. What are the appropriate bit rate and signal level?
  - Use the Shannon formula to find the upper limit.

$$\boxed{C = B \log_2 (1 + \text{SNR}) = 10^6 \log_2 (1 + 63) = 10^6 \log_2 64 = 6 \text{ Mbps}}$$

  - The Shannon formula gives us 6 Mbps, the upper limit
    - For better performance we choose something lower, 4 Mbps, for example.
  - Use the Nyquist formula to find the number of signal levels

$$\boxed{4 \text{ Mbps} = 2 \times 1 \text{ MHz} \times \log_2 L \quad \longrightarrow \quad L = 4}$$

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

Network Performance

- important issue in networking → performance of the network
- Performance parameters:
  - Bandwidth
    - Analog – Hertz
    - Digital – Bits per second (bps)
  - Throughput
    - Actual data rate
  - Latency (delay)
    - Time it takes for an entire message to completely arrive at the destination

Bandwidth

- used in two different contexts with two different measuring values:
- bandwidth in hertz
  - range of frequencies contained in a composite signal or the range of frequencies a channel can pass
    - Ex. bandwidth of a subscriber telephone line is 4 kHz
- bandwidth in bits per second
  - number of bits per second that a channel, a link, or even a network can transmit
    - Ex. bandwidth of a Fast Ethernet network (or the links in this network) is a maximum of 100 Mbps - means that this network can send 100 Mbps.
- relationship between the bandwidth in hertz and bandwidth in bits per second
  - increase in bandwidth in hertz means an increase in bandwidth in bits per second
    - depends on whether we have baseband transmission or transmission with modulation

Bandwidth – examples

- The bandwidth of a subscriber line is 4 kHz for voice or data.
- The bandwidth of this line for data transmission can be up to 56,000 bps using a sophisticated modem to change the digital signal to analog.
- If the telephone company improves the quality of the line and increases the bandwidth to 8 kHz, we can send 112,000 bps.

Throughput

- Measure of how fast we can actually send data through a network
- Bandwidth in bits per second and throughput are different
  - A link may have a bandwidth of *B bps, but we can only send T bps* through this link with *T always less than B.*
  - *Bandwidth is a potential* measurement of a link; the throughput is an actual measurement of how fast we can send data.
  - For example, we may have a link with a bandwidth of 1 Mbps, but the devices connected to the end of the link may handle only 200 kbps
    - cannot send more than 200 kbps through this link.
  - Imagine a highway designed to transmit 1000 cars per minute from one point to another.
  - Due to congestion on the road, 100 cars per minute are transmitted
    - The bandwidth is 1000 cars per minute; the throughput is 100 cars per minute.

Throughput - Example

- A network with bandwidth of 10 Mbps can pass only an average of 12,000 frames per minute with each frame carrying an average of 10,000 bits. What is the throughput of this network?

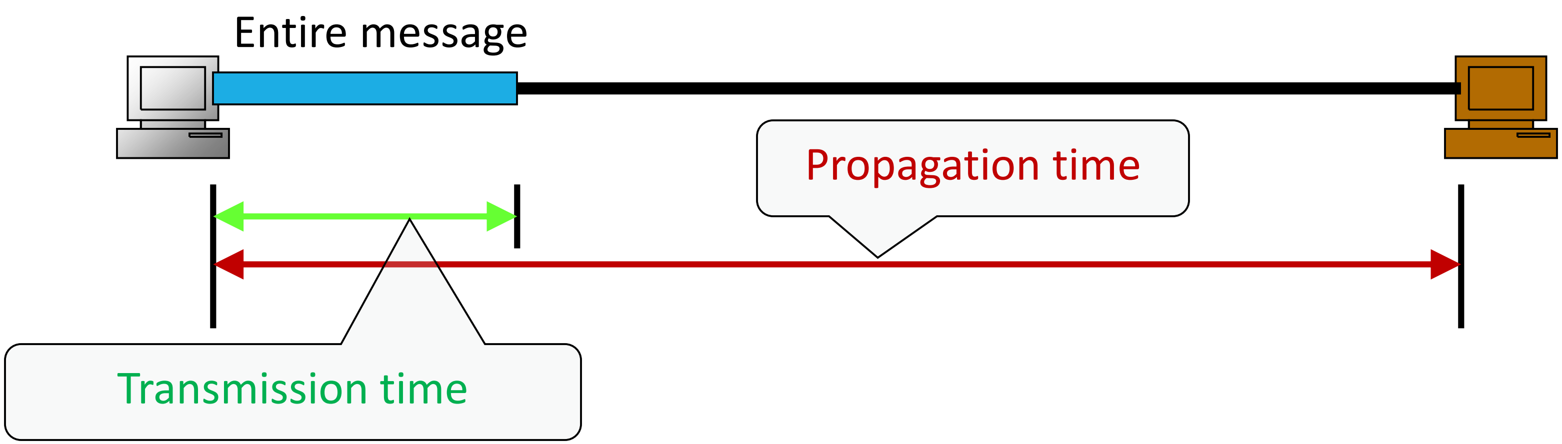
$$\boxed{\text{Throughput} = \frac{12{,}000 \times 10{,}000}{60} = 2 \text{ Mbps}}$$

- The throughput is almost one-fifth of the bandwidth in this case.

Latency

- **Defines how long it takes for an entire message to completely** arrive at the destination from the time the first bit is sent out from the source
- Composed of
  - Propagation time
  - Transmission time
  - Queuing time
  - Processing time

Latency = propagation time
+ transmission time
+ queuing time
+ processing delay



Latency - *Propagation* time

- **time required for a bit to travel from the source to the** destination.
- **Propagation time = Distance / (Propagation Speed)**
- The propagation speed of electromagnetic signals depends on the medium and on the frequency of the signal
  - For example, in a vacuum, light is propagated with a speed of 3 × 108 m/s. It is lower in air; it is much lower in cable.

Example Latency - *Propagation* time

- What is the propagation time if the distance between the two points is 12,000 km? Assume the propagation speed to be 2.4 × 108 m/s in cable.

$$\text{Propagation time} = \frac{12{,}000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

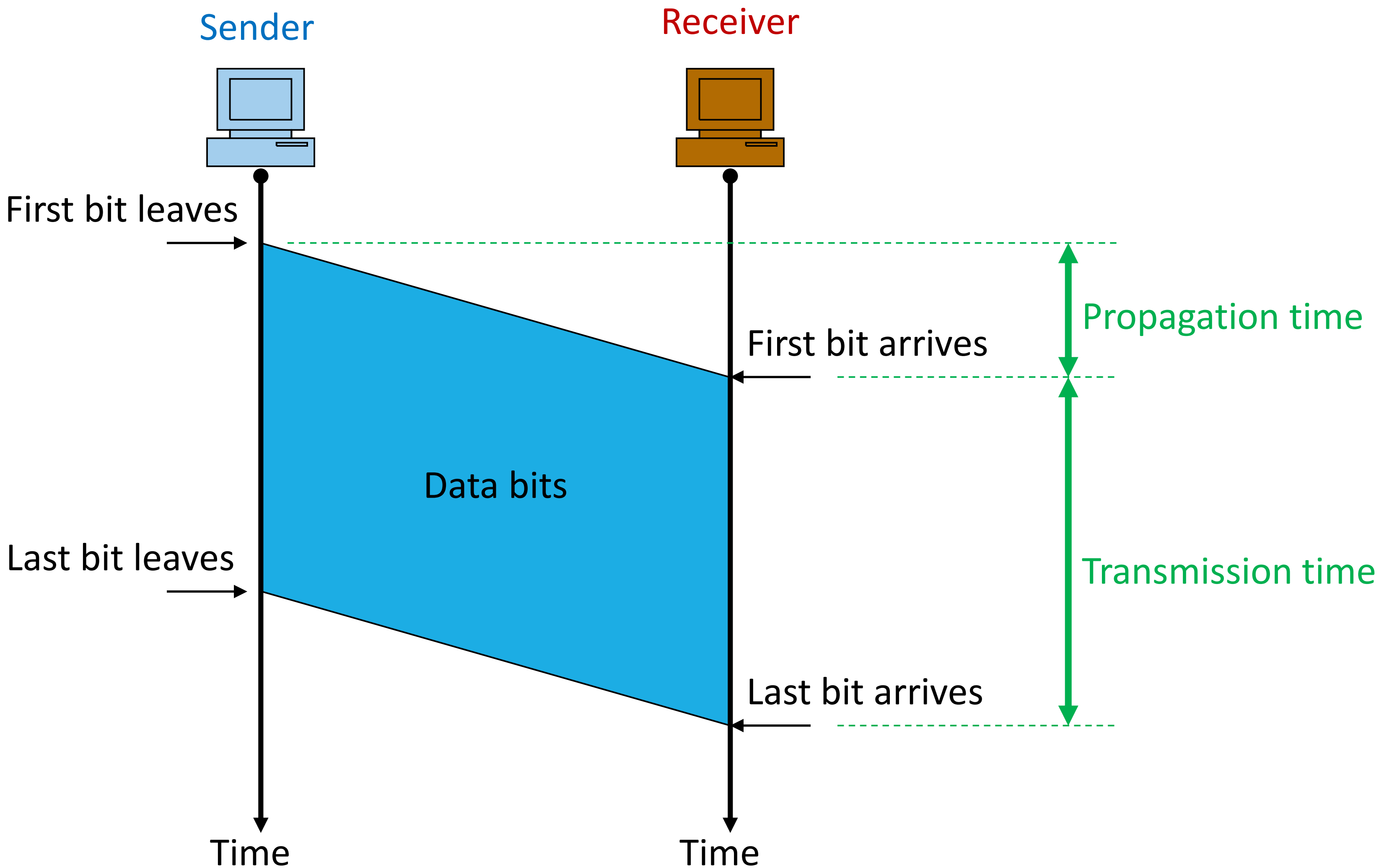
- The example shows that a bit can go over the Atlantic Ocean in only 50 ms if there is a direct cable between the source and the destination.

Latency - Transmission time

- In data communications , message is sent not just 1 bit
  - The first and last bit may take a time equal to the propagation time to reach its destination
    - The first bit leaves earlier and arrives earlier; the last bit leaves later and arrives later.
- **Transmission time →** time between the first bit leaving the sender and the last bit arriving at the receiver
- The **transmission time of a message** depends on the size of the message and the bandwidth of the channel

Transmission time = (Message size) / Bandwidth

Latency - Transmission time



Example 1 - Transmission time

- What are the propagation time and the transmission time for a 2.5-kbyte message (an e-mail) if the bandwidth of the network is 1 Gbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4×10^8 m/s.

$$\text{Propagation time} = \frac{12{,}000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

$$\text{Transmission time} = \frac{2500 \times 8}{10^9} = 0.020 \text{ ms}$$

Note : Message is short and the bandwidth is high so the dominant factor is the propagation time, not the transmission time. The transmission time can be ignored.

Example 2 - Transmission time

- What are the propagation time and the transmission time for a 5-Mbyte message (an image) if the bandwidth of the network is 1 Mbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4×10^8 m/s.

$$\text{Propagation time} = \frac{12{,}000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

$$\text{Transmission time} = \frac{5{,}000{,}000 \times 8}{10^6} = 40 \text{ s}$$

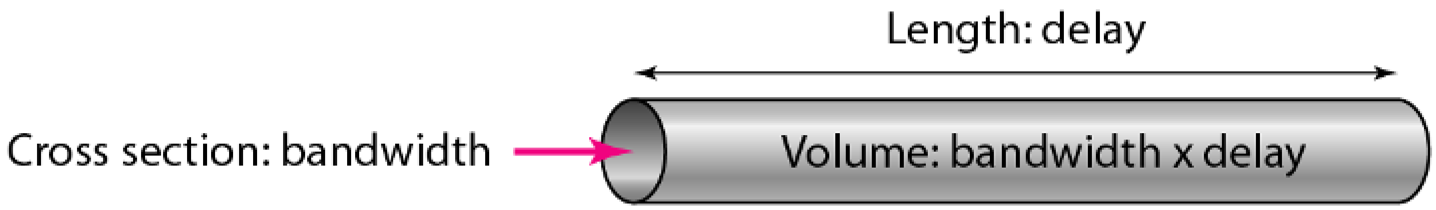
Note: Message is very long and the bandwidth is not very high so the dominant factor is the transmission time, not the propagation time. The propagation time can be ignored.

Latency – Queuing time

- time needed for each intermediate or end device to hold the message before it can be processed.
  - not a fixed factor - changes with the load imposed on the network.
    - When there is heavy traffic on the network, the queuing time increases
  - An intermediate device, such as a router, queues the arrived messages and processes them one by one
  - If there are many messages, each message will have to wait

Bandwidth-Delay Product

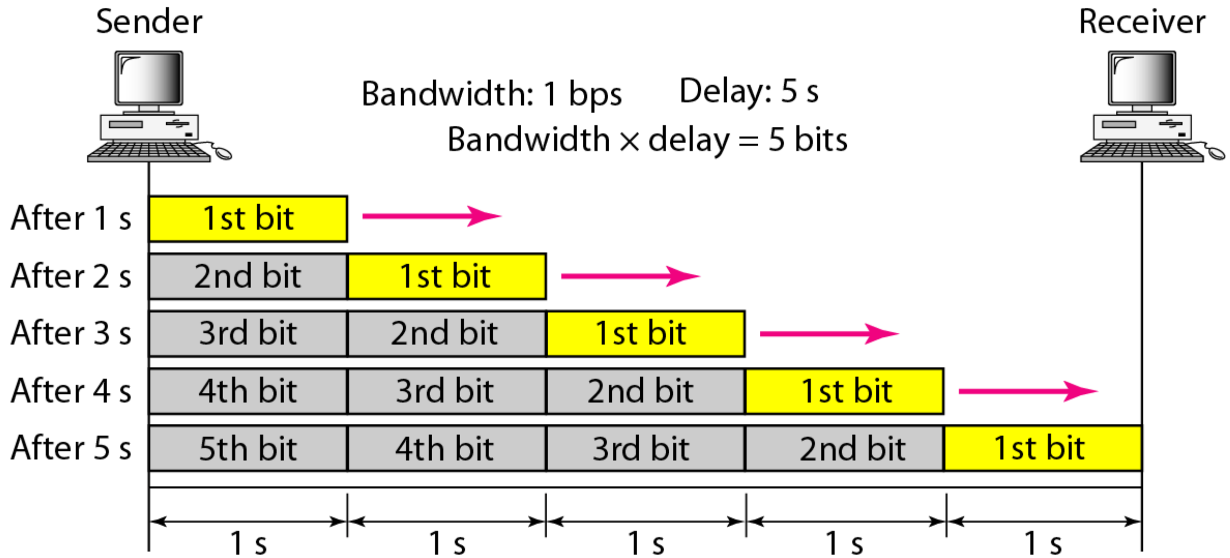
- The product of bandwidth and delay is the number of bits that can fill the link



- two hypothetical cases as examples
  - Filling the links with bits
  - Filling the pipe with bits
- Two cases show that product of bandwidth and delay is the number of bits that can fill the link

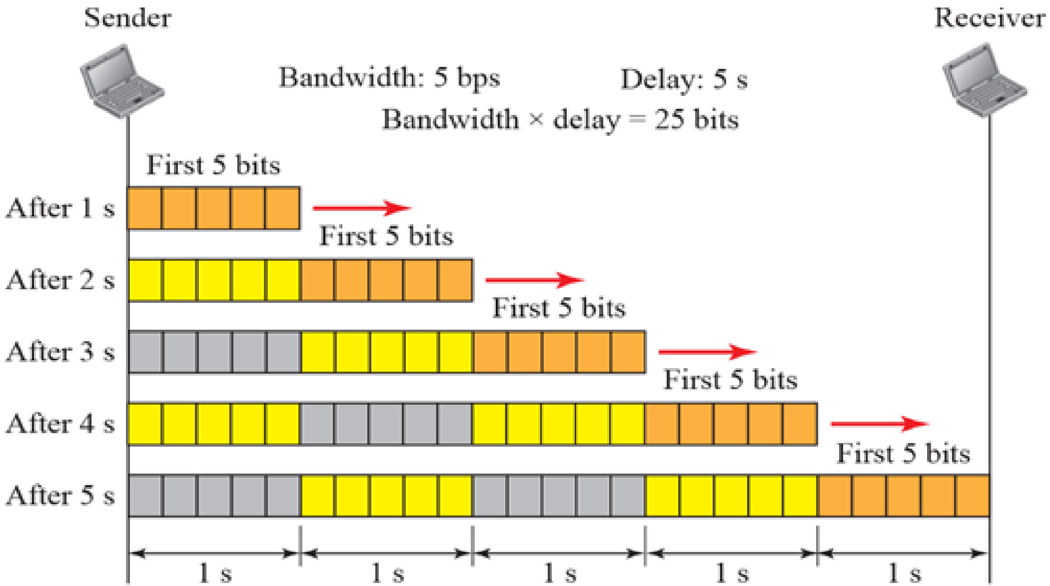
Bandwidth-Delay Product

- Filling the links with bits
  - assuming that we have a link with a bandwidth of 1 bps (unrealistic).
  - We also assume that the delay of the link is 5 s (also unrealistic).
  - We want to see what the bandwidth-delay product means in this case?
  - This product 1 × 5 is the maximum number of bits that can fill the link.
  - There can be no more than 5 bits at any time on the link.



Bandwidth-Delay Product

- Filling the pipe with bits
  - Assuming we have a bandwidth = 5 bps, and delay of the link = 5 s
  - There can be maximum 5 × 5 = 25 bits on the line.
  - The reason is that, at each second, there are 5 bits on the line
  - The duration of each bit is: 1/5 = 0.20 s.



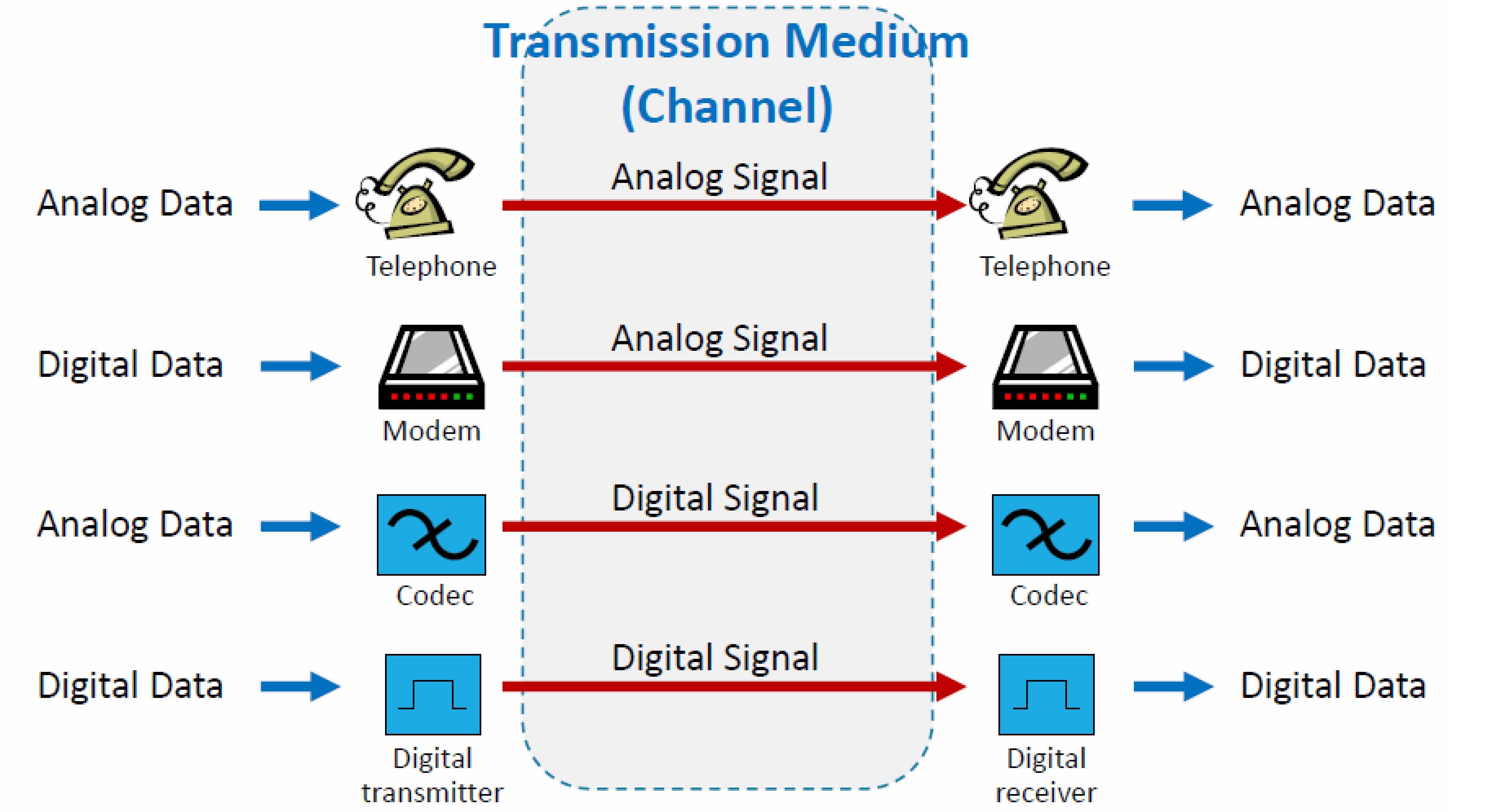
Bandwidth-Delay Product

- The number of bits that can fill the link is important if we need to send data in bursts and wait for the acknowledgment of each burst before sending the next one
- To use the maximum capability of the link, we need to make the size of our burst 2 times the product of bandwidth and delay
  - sender should send a burst of data of (2 × bandwidth × delay) bits
  - sender then waits for receiver acknowledgment for part of the burst before sending another burst
  - amount 2 × bandwidth × delay is the number of bits that can be in transition at any time

Jitter

- Related to delay is **jitter**
- **Jitter** is a problem if different packets of data encounter different delays and the application using the data at the receiver site is time-sensitive (audio and video data, for example)
- If the delay for the first packet is 20 ms, for the second is 45 ms, and for the third is 40 ms, then the real-time application that uses the packets endures jitter.

Analog and Digital



Conversion Techniques

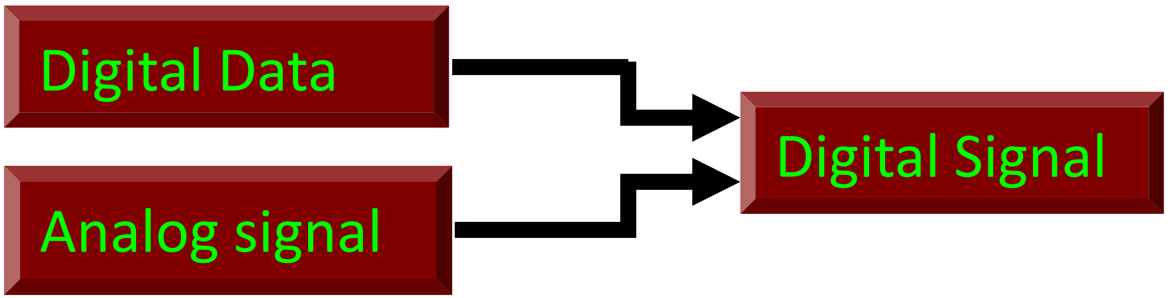
- Digital-to-digital Conversion
  - Line Coding
    - Line Coding Schemes
  - Block Coding
  - Scrambling
- Analog-to-digital Conversion
  - Pulse Code Modulation (PCM)
  - Delta Modulation (DM)
- Digital-to-analog Conversion
  - Aspects of Digital-to-Analog Conversion
  - Amplitude Shift Keying
  - Frequency Shift Keying
  - Phase Shift Keying
  - Quadrature Amplitude Modulation
- Analog-to-analog Conversion
  - Amplitude Modulation (AM)
  - Frequency Modulation (FM)
  - Phase Modulation (PM)

Digital Transmission

- Computer network →send information from one point to another
- Data can be either digital or analog
  - Analog: Continuous value data (sound, light, temperature)
  - Digital: Discrete value (text, integers, Symbols)
- Signals that represent data can also be digital or analog
  - Analog: Continuously varying electromagnetic wave
  - Digital: Series of voltage pulses (square wave)

Digital Transmission …

- Information (analog data or digital data)→ converted to either a digital signal or an analog signal for transmission
- How to represent digital data by using digital signals

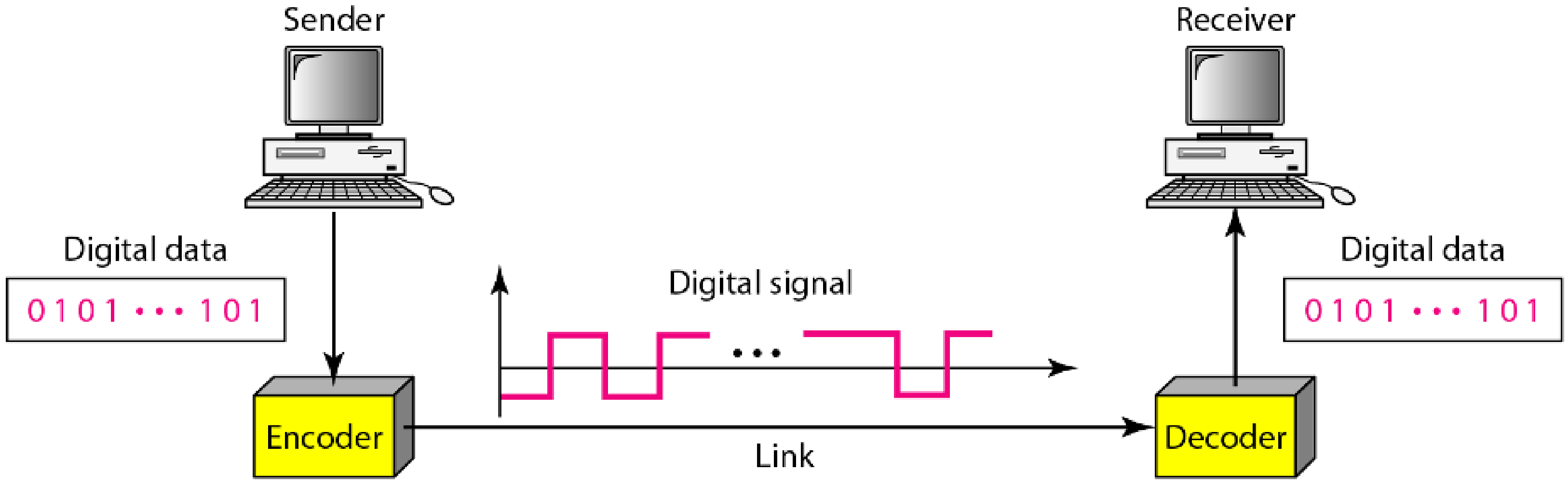


Digital Transmission…

- schemes and techniques that used to transmit data digitally
  - **Digital-to-digital conversion techniques → *represent digital data by using digital signals***
    - Line coding (always needed)
    - block coding (may or may not be needed)
    - Scrambling (may or may not be needed)
  - **Analog-to-digital conversion techniques → *represent analog data by using digital signals***
    - Pulse code modulation
    - Delta modulation
- Transmission modes:
  - parallel or serial

Line Coding

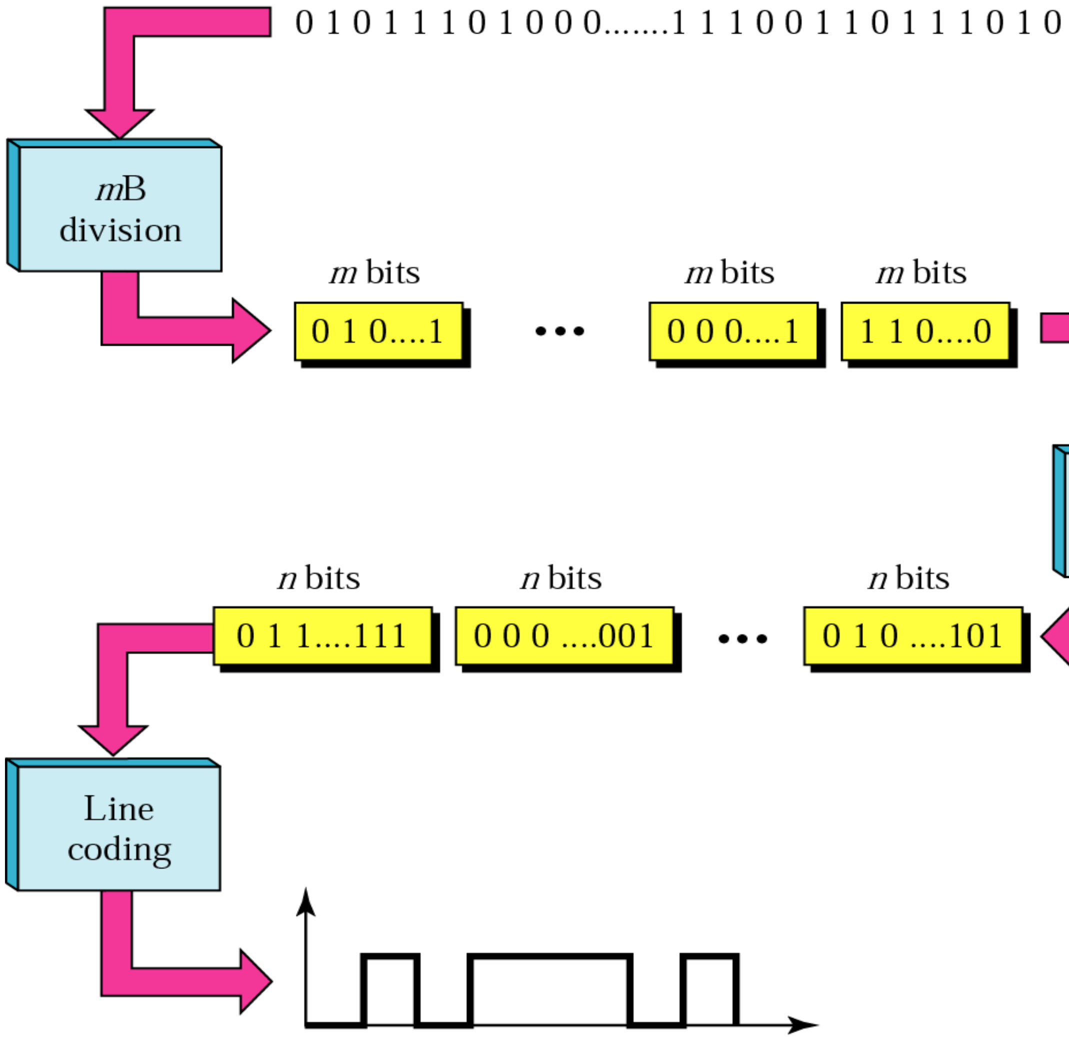
- Data→ text, numbers, graphical images, audio, or video
  - stored in computer memory as sequences of bits
- Line coding converts a sequence of bits to a digital signal
  - At the sender, digital data are encoded into a digital signal
  - At the receiver, the digital data are recreated by decoding the digital signal



Block Coding

- How to ensure synchronization and provide some kind of inherent error detecting?
  - Solution – need redundancy
    - Block coding → give this redundancy and improve the performance of line coding
- Block coding changes a block of *m bits into a block* of *n bits, where n > m*
  - *referred to as an mB/nB encoding*) → distinguishes block encoding from multilevel encoding
    - multilevel encoding → written without a slash eg. 8B6T
    - it replaces each *m*-bit group with an *n*-bit group
    - involves three steps:
      - **Division**: sequence of bits is divided into groups of *m bits*
      - **Substitution**: substitute an *m-bit group with an n-bit* group
      - **Combination**: *n-bit groups are combined to form a stream* (n > m)

Block Coding

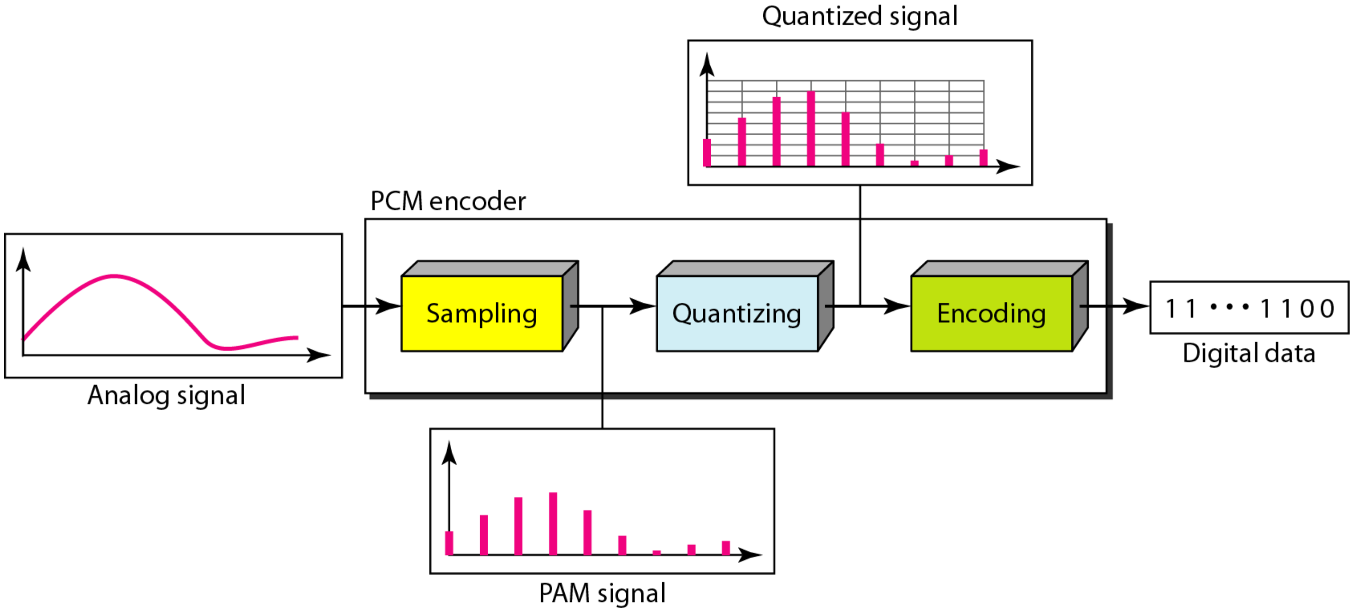


Analog to Digital Conversion

- an analog signal → created by a microphone or camera
- The amplitude of analog signal can take any value over a continuous range i.e. it can take on an infinite values.
- Digital signal amplitude can take on finite values.
- Digital signal is superior to an analog signal
- Analog signal can be converted into digital by sampling and quantizing-- digitization
  - two techniques
    - pulse code modulation
    - delta modulation
- Digital data are converted to digital signal using line coding, block coding or scrambling techniques

Pulse Code Modulation (PCM)

- Change an analog signal to digital data (digitization)
- A PCM encoder has three processes
  - Sampling - analog signal is sampled
  - Quantization- sampled signal is quantized
  - encoding - quantized values are encoded as streams of bits



PCM - Sampling

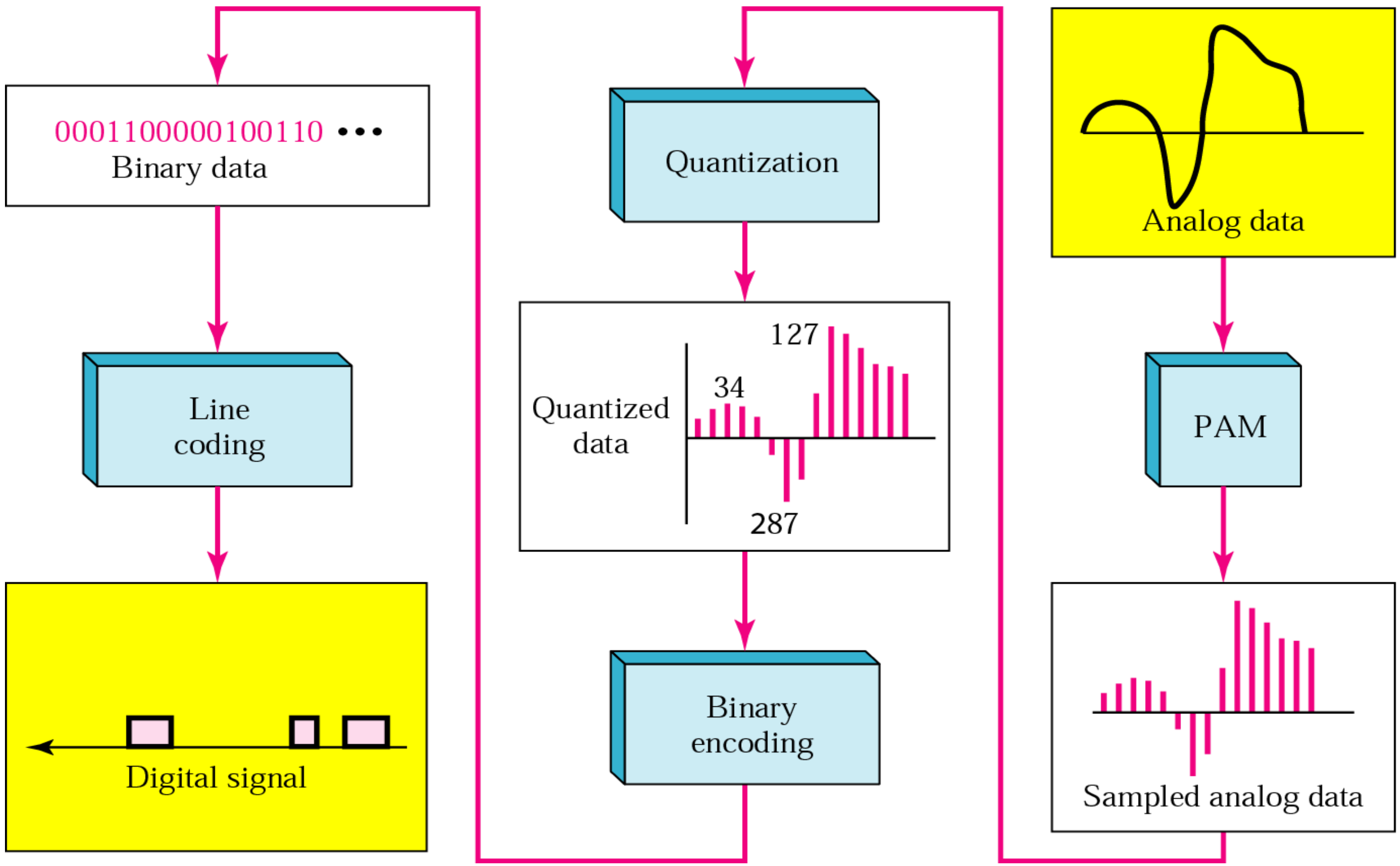
- The analog signal is sampled every T_s *s, where* T_s *is* the sample interval or period
  - **Called as pulse amplitude modulation (PAM)**
  - Result →an analog signal with nonintegral values
- The inverse of the sampling interval is called the *sampling rate or sampling frequency and denoted by* f_s
  - $f_s = 1/T_s$

PCM: Encoding 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:

$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$
$$\text{Bit rate} = 8000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

PCM: The Whole Process



Minimum Required Bandwidth

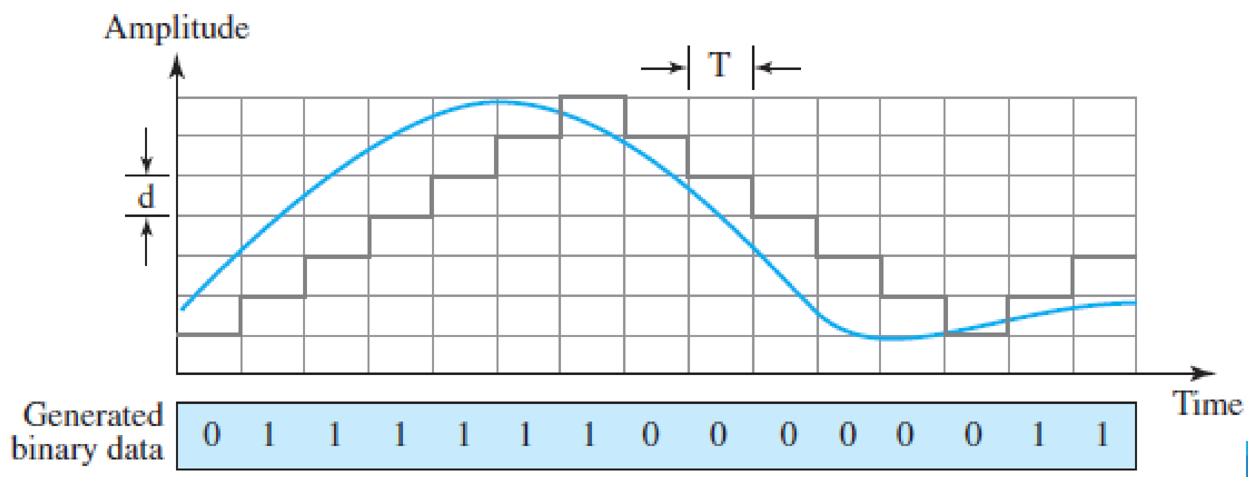
- It can be proved that the minimum bandwidth of the digital signal is $\boldsymbol{B_{min} = n_b \times B_{analog}}$

Delta Modulation (DM)

- PCM is a very complex technique
- Delta modulation reduce the complexity of PCM
- PCM finds the value of the signal amplitude for each sample whereas DM finds the change from the previous sample → bit
- DM → no code words; bits are sent one after another
    - Modulator is used at the sender site to create a stream of bits from an analog signal
    - Demodulator is used at receiver site to creates the analog signal from the received digital data
- DM is not perfect → Quantization error introduced in the process
    - The quantization error of DM is much less than that for PCM

DM...

- The process records the small positive or negative changes, called delta δ
  - If the delta is positive, the process records a 1
  - If it is negative, the process records a 0
- A better performance achieved if the value of δ is not fixed
  - Adaptive delta modulation - value of δ changes according to the amplitude of the analog signal.

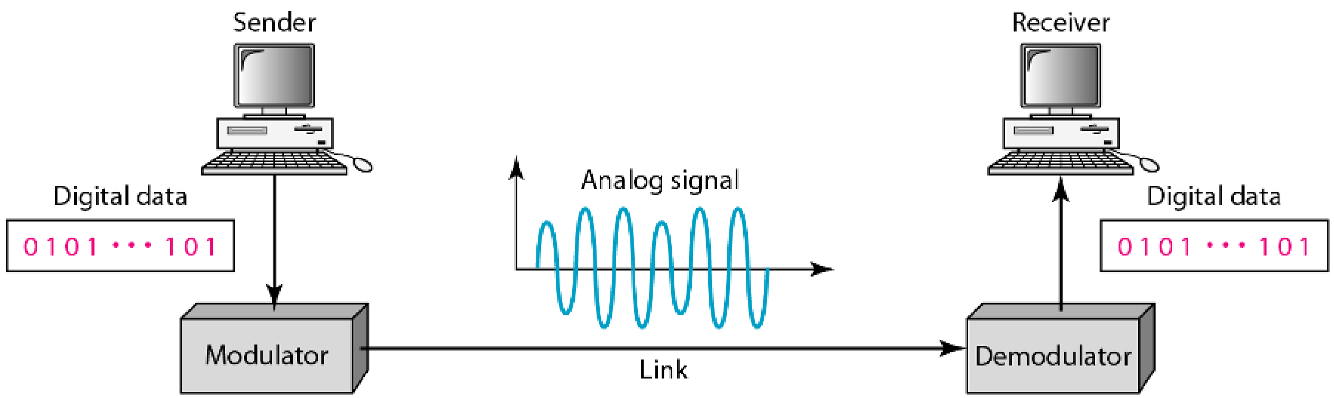


Analog-to-analog conversion

- Change an analog signal to a new analog signal with a smaller bandwidth
  - used when only a band-pass channel is available
- Three methods:
  - Amplitude modulation (AM) : amplitude of a carrier is changed based on the changes in the original analog signal
  - Frequency modulation (FM): the phase of a carrier is changed based on the changes in the original analog signal
  - Phase modulation (PM): the phase of a carrier signal is changed to show the changes in the original signal

Digital-to-Analog Conversion

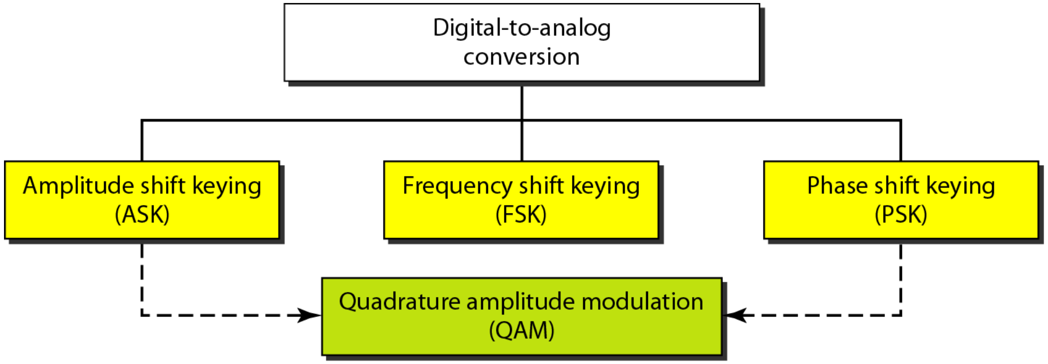
- Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data when a band-pass channel is available
- Required to send digital data over a band-pass channel
  - Also known as ***modulation***



(band-pass channel)

Digital-to-analog conversion

- Four methods:
  - Amplitude shift keying (ASK): amplitude of a carrier is changed using the digital data
  - Frequency shift keying (FSK): frequency of a carrier is changed using the digital data.
  - Phase shift keying (PSK): phase of a carrier signal is changed to represent digital data.
  - Quadrature amplitude modulation (QAM): both amplitude and phase of a carrier signal are changed to represent digital data.

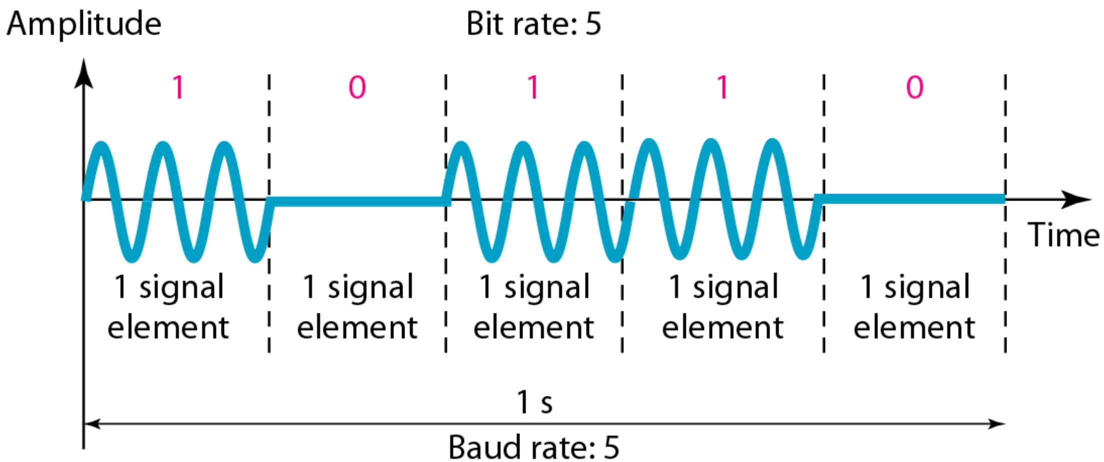


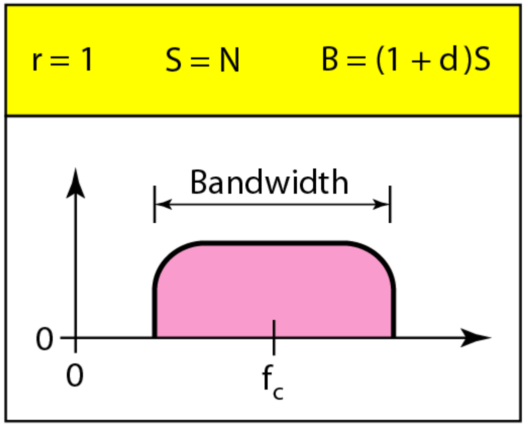
Amplitude shift keying

- Amplitude of the carrier signal is varied to create signal elements
- Both frequency and phase remain constant while the amplitude changes
- Binary ASK (BASK)
  - uses only two amplitude levels
- Multilevel ASK (implemented with QAM)
  - more than two levels

Binary Amplitude Shift Keying

- Several levels (kinds) of signal elements with a different amplitude can exists
- ASK → normally implemented using only two levels
  - *Binary amplitude shift keying or on-off keying (OOK)*
  - Peak amplitude of
    - One signal level is 0
    - Other signal level is the same as the amplitude of the carrier frequency





middle of the bandwidth - *fc* : Carrier frequency

$0 < d < 1$

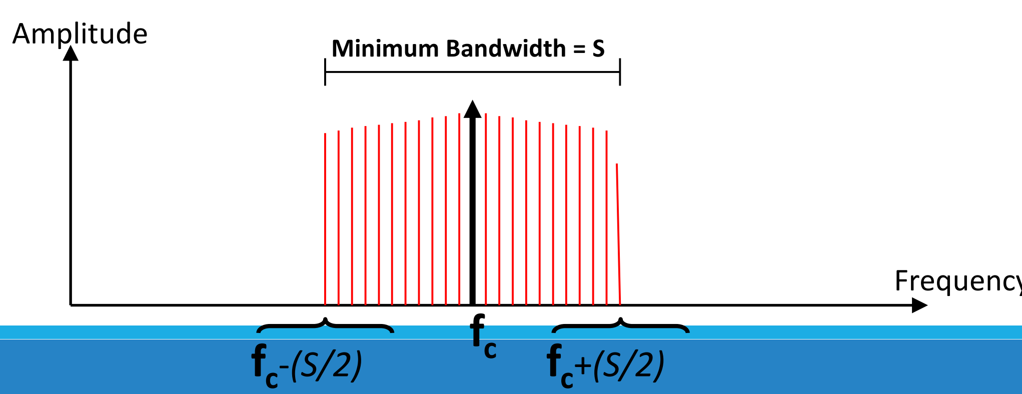
Bandwidth for ASK

- carrier signal is only one simple sine wave -> has a continuous set of frequencies
- modulation produces a nonperiodic composite signal
- bandwidth is proportional to the signal rate (baud rate)

  $B = (1 + d) \times S$
  - S is the signal rate
  - *B is the bandwidth*
  - *d depends on the modulation and filtering* process
  - The value of *d is between 0 and 1*

    $B_{min}=S$

    $B_{max}=2S$

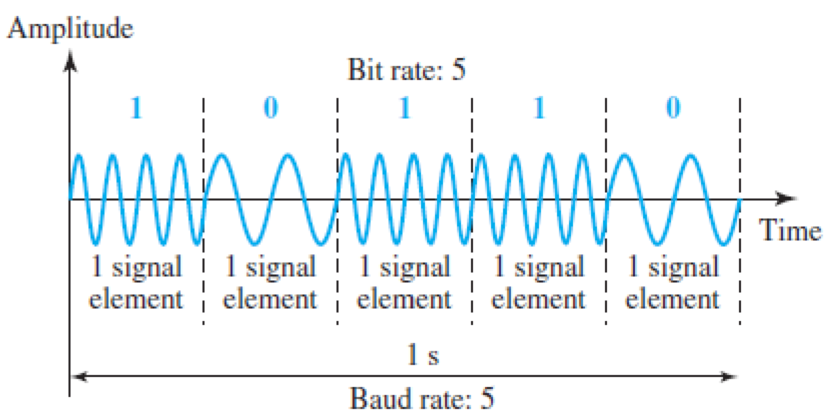


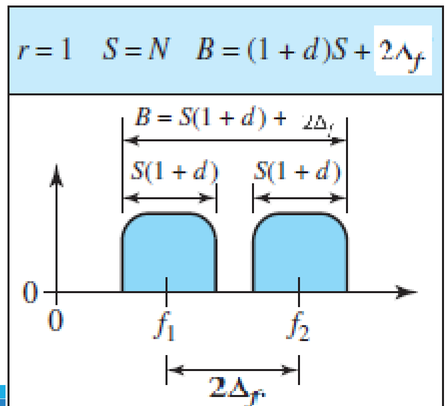
Frequency Shift Keying

- frequency of the carrier signal is varied to represent data
  - The frequency of the modulated signal is constant for the duration of one signal element
  - but changes for the next signal element if the data element changes
- Both peak amplitude and phase remain constant for all signal elements
- Binary FSK (or BFSK)
  - consider two carrier frequencies *$f1$ and $f2$*
- Multilevel FSK (MFSK)
  - more than two frequencies

Binary Frequency Shift Keying

- Consider two carrier frequencies *f1 and f2*
  - *first carrier frequency for* the data element 0
  - Second *carrier frequency for t*he data element 1
  - carrier frequencies are very high and the difference between them is very small
- *Both f1 and f2 are Δ_f apart from the midpoint between the two bands*
- *The difference* between the two frequencies is $2\Delta_f$





Bandwidth for *BFSK*

- carrier signal is only one simple sine wave -> has a continuous set of frequencies
- modulation produces a nonperiodic composite signal
- Consider FSK is two ASK signals, each with its own carrier frequency (*f1 or f2)*
- *difference between the two frequencies is* $2\Delta_f$
  - minimum value of $2\Delta_f$ should be at least *S for* the proper operation of modulation and demodulation.

$$B = (1 + d) \times S + \text{frequency shift} = (1 + d) \times S + 2\Delta_f$$

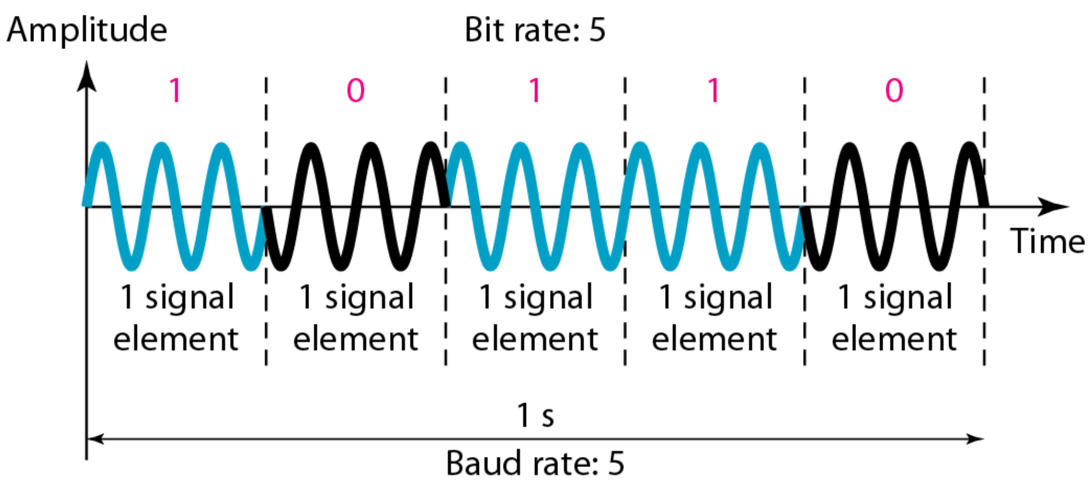
  - S is the signal rate
  - *B is the bandwidth*
  - *d depends on the modulation and filtering* process
  - The value of *d is between 0 and 1*

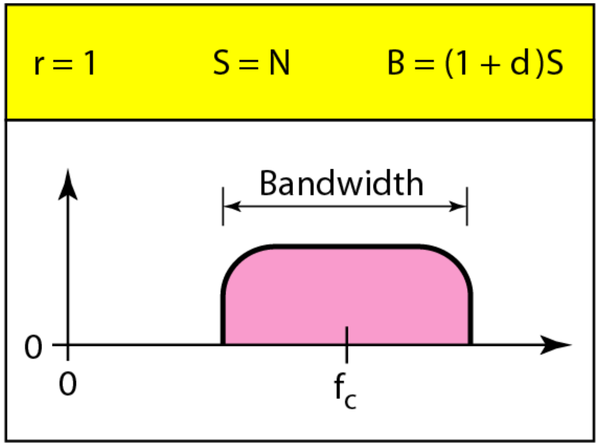
Phase Shift Keying

- Phase of the carrier is varied to represent two or more different signal elements
- Both peak amplitude and frequency remain constant as the phase changes
- Today, PSK is more common than ASK or FSK
- Binary Phase Shift Keying
  - only two signal elements
- Quadrature PSK (QPSK)
  - uses two separate BPSK modulations

Binary Phase Shift Keying

- Or Binary PSK
- only two signal elements
  - one with a phase of 0°
  - other with a phase of 180°





Binary PSK

- The bandwidth is the same as that for binary ASK, but less than that for BFSK
- No bandwidth is wasted for separating two carrier signals

PSK	ASK
PSK is less susceptible to noise than ASK as noise can change the amplitude easier than it can change the phase	ASK is more susceptible to noise than PSK
the criterion for bit detection is the phase of the signal	the criterion for bit detection is the amplitude of the signal
superior to FSK because we do not need two carrier signals	need two carrier signals
needs more sophisticated hardware to be able to distinguish between phases	No sophisticated hardware

Comparison ASK, FSK, PSK

Parameters	ASK	FSK	PSK
Variable characteristics	Amplitude	Frequency	Phase
Bandwidth	B =(1+d)S d is due to modulation & filtering ,lies between 0 & 1.	B=(1+d)×S+2Δf	B=(1+d)×S
Noise immunity	low	High	High
Complexity	Simple	Moderately complex	Very complex
Error probability	High	Low	Low
Performance in presence of noise	Poor	Better than ASK	Better than FSK
Bit rate	Suitable upto 100 bits/sec	Suitable upto about 1200 bits/sec	Suitable for high bit rates

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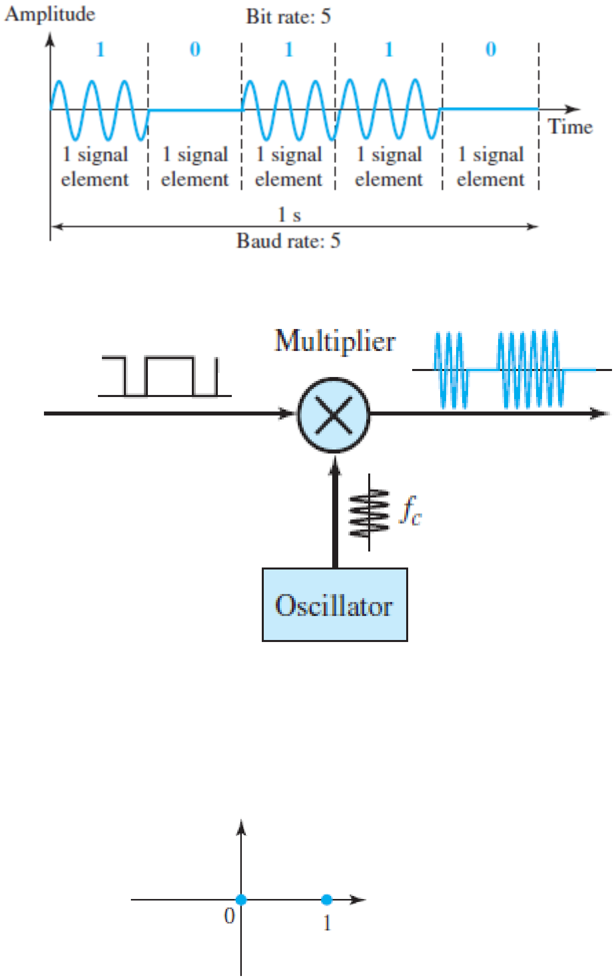
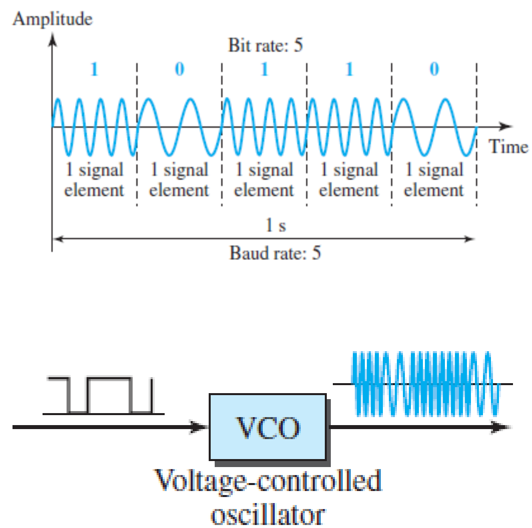
Comparison ASK, FSK, PSK

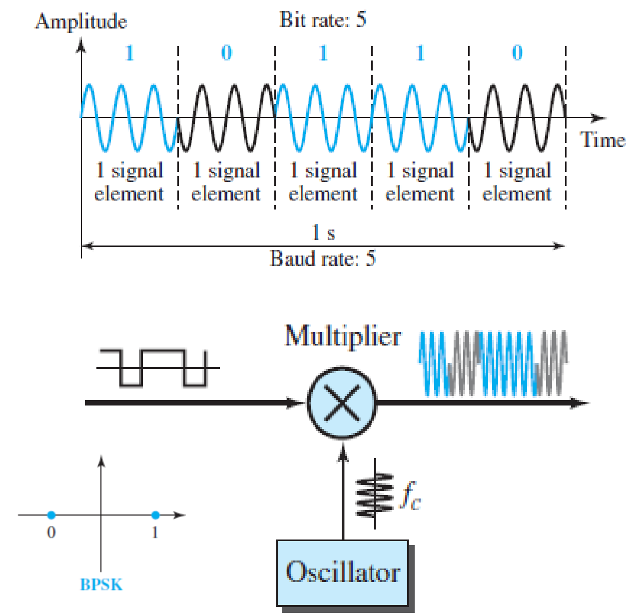
BASK



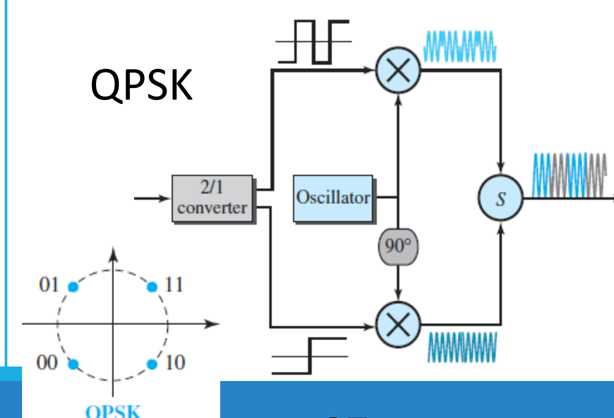
BFSK



BPSK

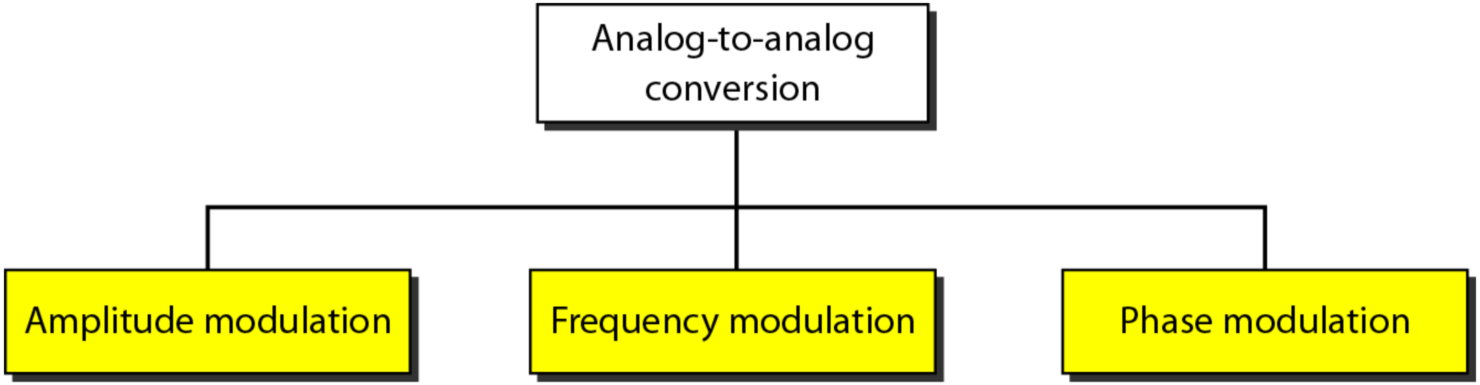


QPSK



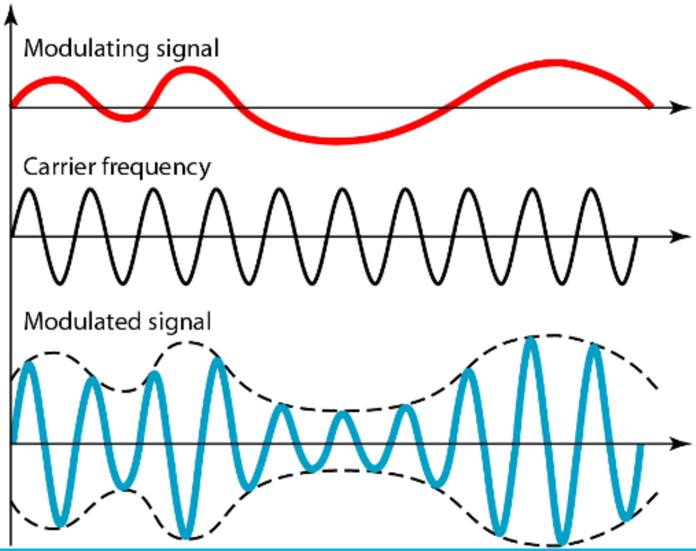
Analog-to-Analog Conversion

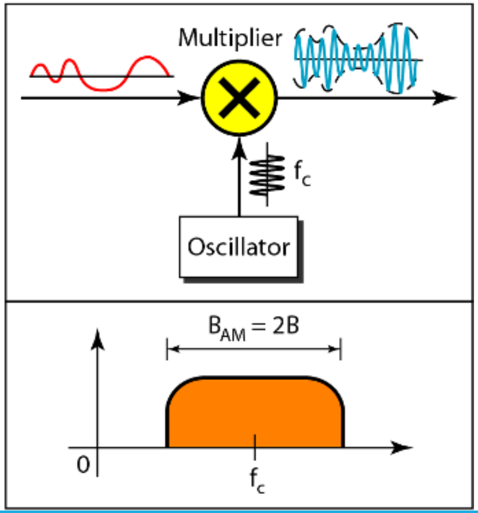
- Process of transmitting analog information by an analog signal
- Although the signal is already analog, modulation is needed if a band-pass channel is available to us eg. Radio
  - The government assigns a narrow bandwidth to each radio station
  - each radio station – produces low-pass signal
    - all stations low analog pass signal in the same range
  - Listening to different stations, the low-pass signals need to be shifted, each to a different range → bans pass signal



Amplitude Modulation (AM)

- Amplitude of the carrier signal is modulated to follow the changing amplitudes of the modulating signal
  - Only the amplitude changes to follow variations in the information
  - The frequency and phase of the carrier remain the same
- The modulating signal is the envelope of the carrier
- The amplitude of the carrier signal changed according to the amplitude of the modulating signal using a simple multiplier



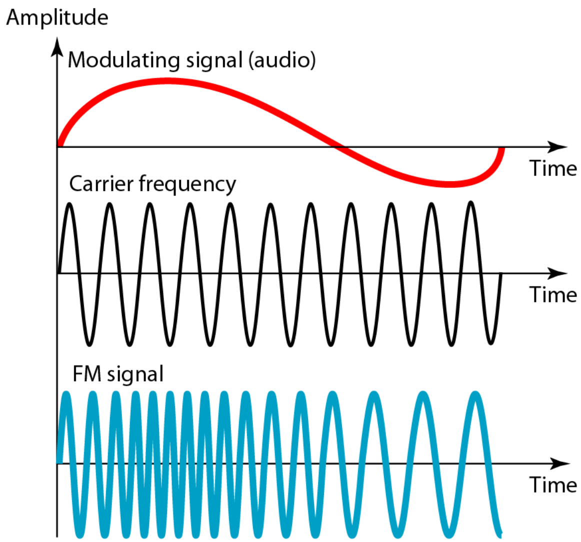


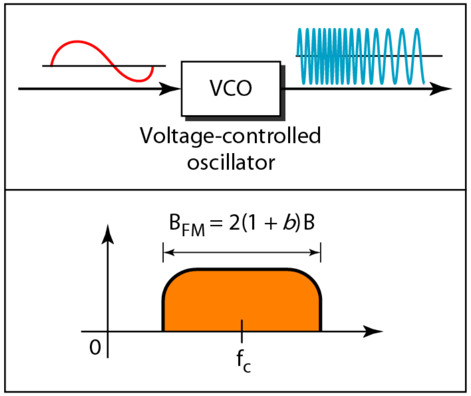
AM Bandwidth

- The modulation creates a bandwidth that is twice the bandwidth of the modulating signal and covers a range centered on the carrier frequency
  - $B_{AM}=2B$
- As the signal components above and below the carrier frequency carry exactly the same information, some implementations discard one-half of the signals and cut the bandwidth in half.

Frequency Modulation (FM)

- Frequency of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal
  - Peak amplitude and phase of the carrier signal remain constant,
  - As the amplitude of the information signal changes, the frequency of the carrier changes correspondingly
- Implemented by using a voltage-controlled oscillator as with FSK
  - The frequency of the oscillator changes according to the input voltage which is the amplitude of the modulating signal





FM Bandwidth

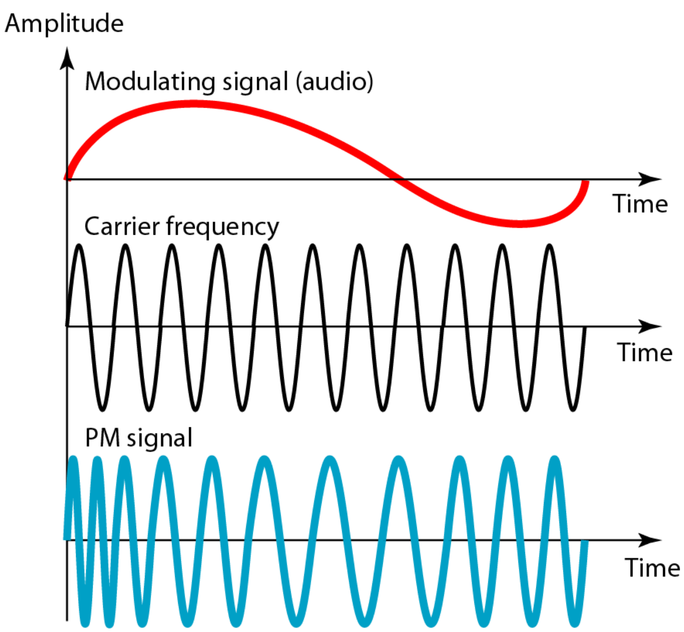
- The actual bandwidth is difficult to determine exactly
- Can be shown empirically that it is
  - several times that of the analog signal or
  - $2(1 + \beta)B$ *where* β *is a factor that depends on modulation technique* with a common value of 4.

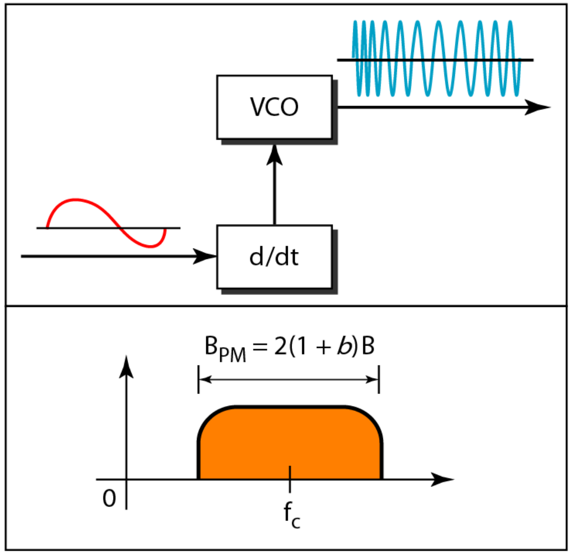
Phase Modulation (PM)

- phase of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal
  - peak amplitude and frequency of the carrier signal remain constant
  - As the amplitude of the information signal changes, the phase of the carrier changes correspondingly
- It can be proved mathematically that PM is the same as FM with one difference
  - FM: the instantaneous change in the carrier frequency is proportional to the amplitude of the modulating signal
  - PM: the instantaneous change in the carrier frequency is proportional to the derivative of the amplitude of the modulating signal

Phase Modulation (PM)

- PM is normally implemented by using a voltage-controlled oscillator along with a derivative
- Frequency of the oscillator changes according to the derivative of the input voltage, which is the amplitude of the modulating signal.





PM Bandwidth

- The actual bandwidth is difficult to determine exactly, but it can be shown empirically that it is several times that of the analog signal

$$B_{PM} = 2(1 + \beta)B$$

- Although the formula shows the same bandwidth for FM and PM, the value of β is lower in the case of PM (around 1 for narrowband and 3 for wideband)

Bandwidth: Revisited

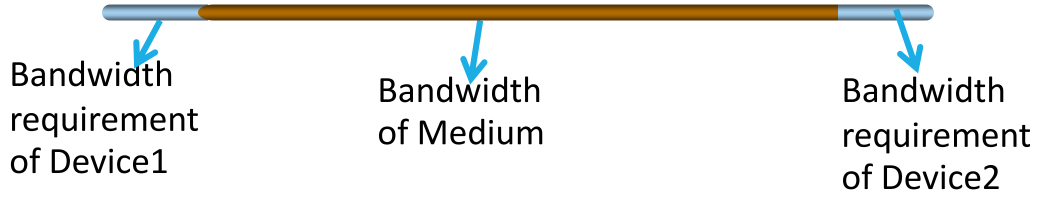
- Bandwidth = highest frequency – lowest frequency
  - Used to measures maximum data transfer rate of a network or Internet connection in a given amount of time
  - Describe network speeds
    - does not measure **how fast bits of data move** from one location to another (data rate)
    - measures **how much data can flow** through a specific connection at one time
    - The wider the bandwidth, the faster data can be sent
- Analog devices bandwidth → measured in hertz (cycles per second)
- Digital devices→ bandwidth is measured in bits per second

Bandwidth Utilization

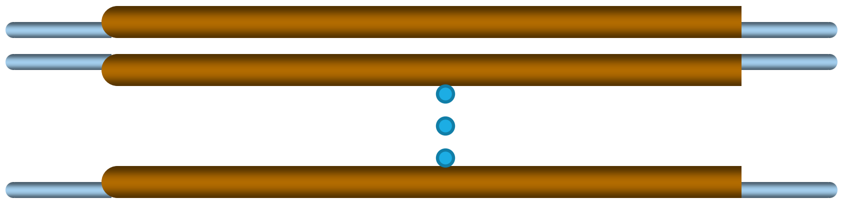
- Reality→ limited bandwidth
  - Requires wise use → main challenge of electronic communication
  - Bandwidth Utilization → depends on application
- Two goal of bandwidth utilization
  - Efficiency
    - Achieved by combining several low-bandwidth channels into one with larger bandwidth
      - Technique is called **multiplexing**
  - Privacy and anti-jamming
    - Achieved by expanding the bandwidth of a channel to insert redundancy
      - Technique is called **spectrum spreading**

Transmission Medium

- Devices connected to transmission medium as



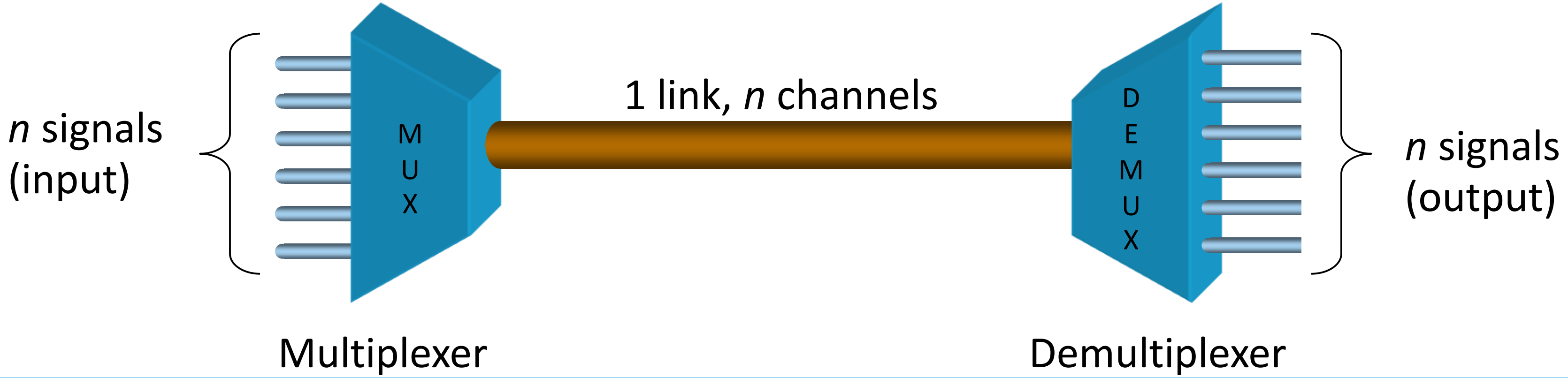
- Increased usage of data and telecommunications results into increased traffic
- How to accommodate this increase?
  - Add individual links each time a new channel is needed
- Today's technology includes high bandwidth media such as optical fiber and terrestrial and satellite microwaves



  - Each medium has a bandwidth far in excess of that needed for the average transmission signal- costly and also become difficult to manage

Shared Link

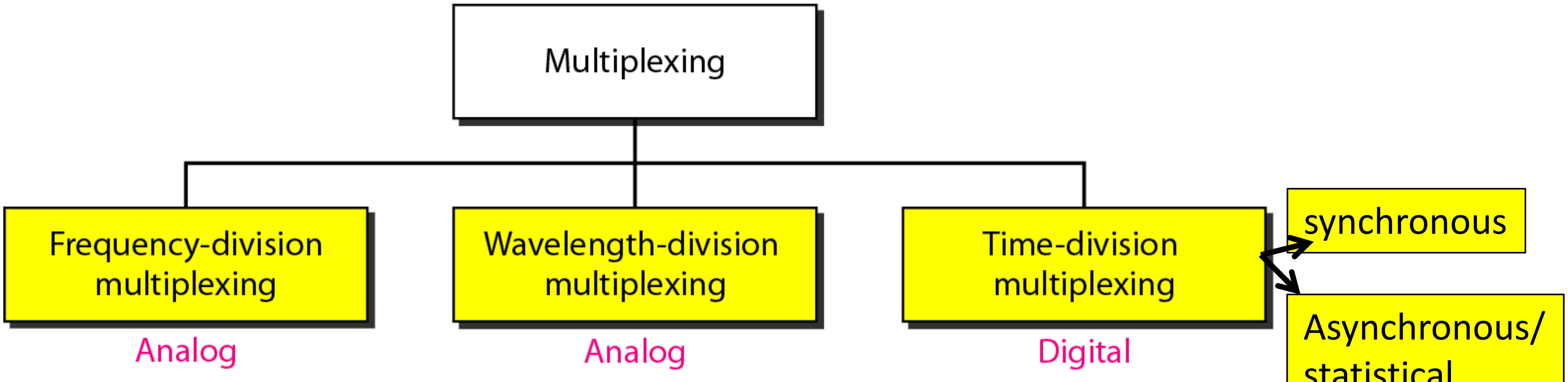
- Bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices connected to it, the bandwidth is wasted
  - Solution: the higher bandwidth link can be shared to carry multiple signals from different devices connected to it
    - Maximizes bandwidth utilization → gives efficient system
    - reduce the line cost and also it would be easier to keep track of one line than several lines
- Shared Link: uses multiplexing technique



Multiplexing

- Process of combining multiple signals (analog or digital), commonly from slow devices, onto one very fast communications link
  - achieved by a device called a Multiplexer (MUX) and Demultiplexer (DEMUX)
    - MUX: combines the lines at sender on the left to direct their transmission streams into a single stream (many-to-one)
    - DEMUX: separates the stream at receiver back into its component transmissions (one-to-many) and directs them to their corresponding lines
- allow the simultaneous transmission of multiple signals across a single data link
  - Link → physical path
  - Channel →portion of a link that carries a transmission between a given pair of lines
    - One link can have many (*n) channels*

Multiplexing …



Analog Multiplexing technique to combine analog signals

Analog Multiplexing technique to combine optical signals

Digital multiplexing technique to combine several low rate channels into high rate one

- Carrier division multiple access (CDMA) is considered as a fourth multiplexing category

Frequency Division Multiplexing (FDM)

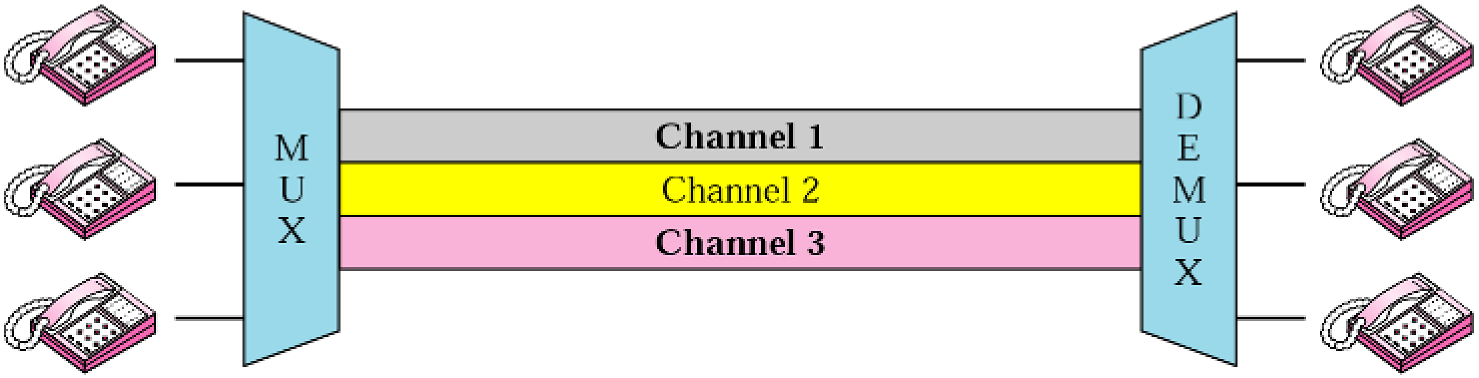
An analog multiplexing technique to combine signals

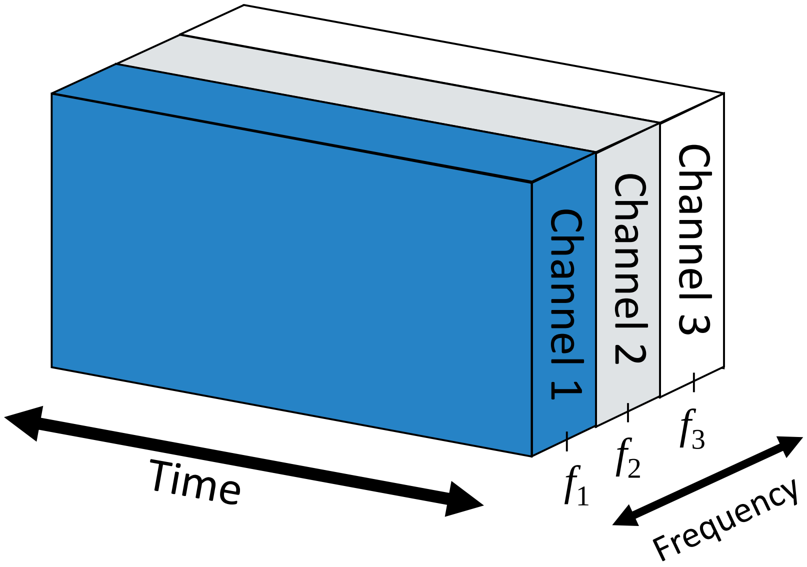
- Can be applied when the bandwidth of a link in (hertz) is greater than the combined bandwidths of the signal to be transmitted
  - Medium BW > Channel BW
- Transmitting all of the signals along the same high speed link simultaneously

FDM ...

- Each signal is modulated to a different carrier frequency
  - Carrier frequencies are separated by sufficient bandwidth to accommodate the modulated signal
    - These bandwidth ranges are the channels through which the various signals travel
      - Channels can be separated by strips of unused bandwidth -guard bands -to prevent signals from overlapping
    - In addition, carrier frequencies must not interfere with the original data frequencies
- These modulated signals are then combined into a single composite signal that can be transported by the link
- E.g., broadcast radio
- Channel allocated even if no data

Conceptual View of FDM



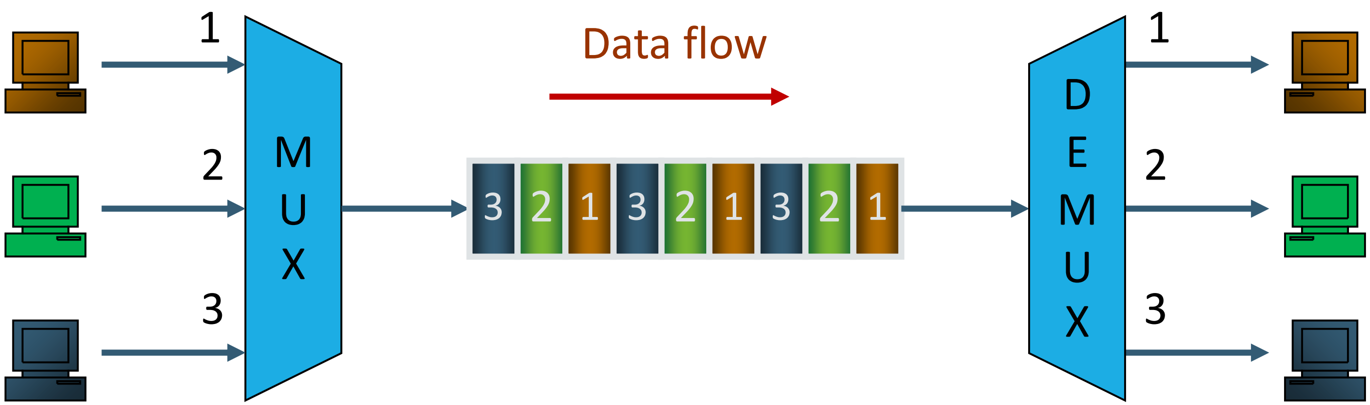


Time Division Multiplexing (TDM)

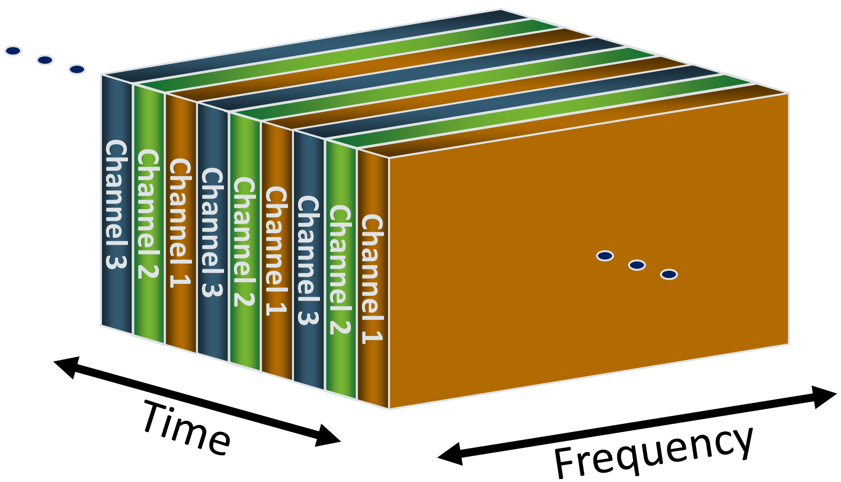
A digital multiplexing technique to combine data

- **digital process that allows several connections** to share the high bandwidth of a link
- Time is shared
  - Each connection occupies a portion of time in the link
  - Same link of FDM is shown sectioned by time rather than by frequency
- Digital data from different sources are combined into one timeshared link
- Analog data can be sampled, changed to digital data, and then multiplexed by using TDM

Conceptual View of TDM



- Portions of signals 1, 2, and 3 occupy the link sequentially
- Data in a message from source 1 always go to one specific destination 1, 2, 3, or 4
- The delivery is fixed and unvarying, unlike switching



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Summary

- Signals get impaired by attenuation, distortion, and noise
- 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 bandwidth-delay product defines the number of bits that can fill the link.

Summary

- Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in the digital data
- Digital-to-analog conversion can be accomplished in several ways: amplitude shift keying (ASK), frequency shift keying (FSK), and phase shift keying (PSK)
- Quadrature amplitude modulation (QAM) combines ASK and PSK
- A constellation diagram shows us the amplitude and phase of a signal element, particularly when we are using two carriers (one in-phase and one quadrature)
- Analog-to-analog conversion is the representation of analog information by an analog signal
- Conversion is needed if the medium is bandpass in nature or if only a bandpass bandwidth is available to us
- Analog-to-analog conversion can be accomplished in three ways: amplitude modulation (AM), frequency modulation (FM), and phase modulation (PM).