University of Southern California

Viterbi School of Engineering

EE450

Computer Networks

Physical Layer

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Spring 2013

Physical Layer Main Responsibilities

- **Definition of Hardware Specifications**: The details of operation of cables, connectors, wireless radio transceivers, network interface cards and other hardware devices are generally a function of the physical layer (although also partially data link layer)
  - Specifications of Physical connectors / cards
  - Specifications of cables connecting that hardware
  - Signaling standards like voltage levels, voltage changes & its timings
  - Bit stream is transmitted & received as 0's / 1's & can be sent as electrical impulse (over Copper media,) a light signals (over Fiber optics media,) and a radio signal (over air / Wireless Media

Main Responsibilities (Cont.)

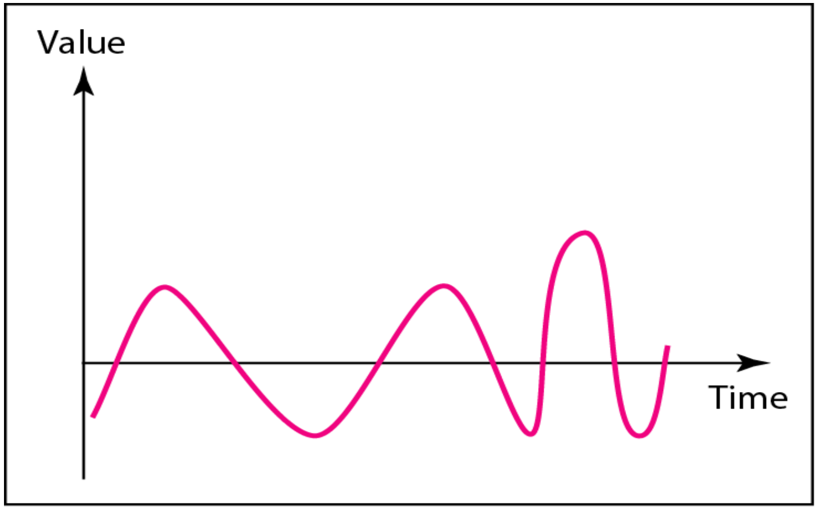
- **Encoding and Signaling**: The physical layer is responsible for various encoding and signaling functions that transform the data from bits that reside within a computer or other device into signals that can be sent over the network
  - It deals with how a Layer 2 Frame data will be converted to bits (0s and 1s) & how 0s and 1s will be converted back into Layer 2 frames. It can be about conversion of Data into Voltage / Light / Radio Signal to represent bits, and also bits in form of Voltage / Light / Radio Signal to represent data

Main Responsibilities (Cont.)

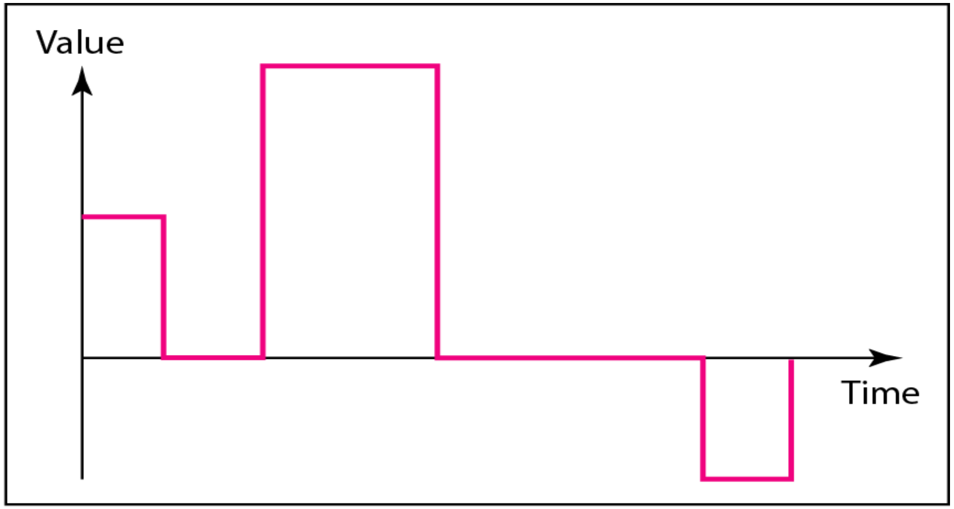
- **Data Transmission and Reception (including synchronization)** : After encoding the data appropriately, the physical layer actually transmits the data, and of course, receives it. Note that this applies equally to wired and wireless networks, even if there is no tangible cable in a wireless network
  - Actual transmission of data over the medium & defines standards for data transmission rates and maximum transmission distance
  - Synchronization of the sender and receiver physical layers
- **Topology and Physical Network Design:** The physical layer is also considered the domain of many hardware-related network design issues for LANs, WANs, etc.

Basics: Data and Signal

- Data can be analog or digital. Analog data are continuous and take continuous values. Digital data have discrete states and take discrete values
- Signals can be analog or digital. They are a way of representing the data. The representation could be electrical, electromagnetic (i.e., transmission via radio), or optical (i.e., transmission via optical fiber)



a. Analog signal

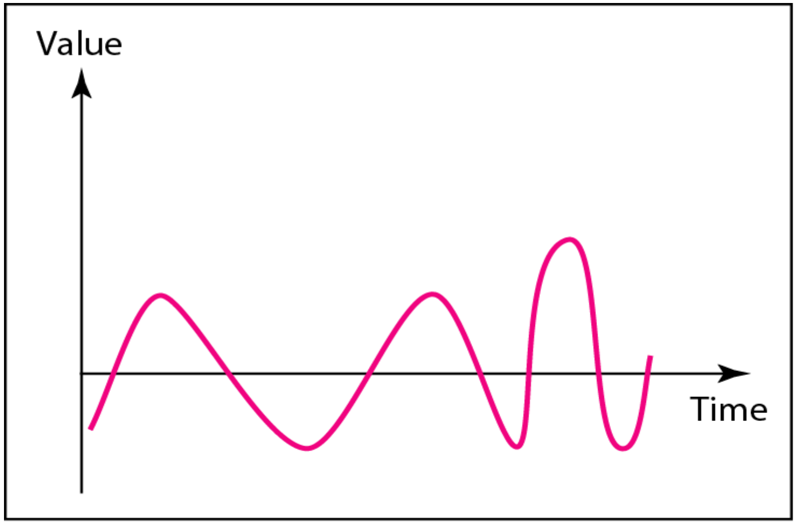


b. Digital signal

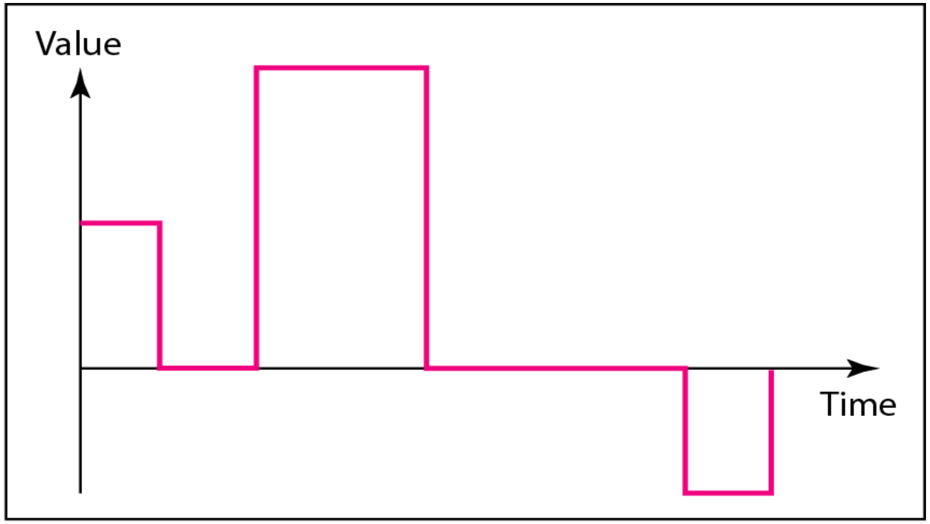
Basics: Data and Signal (Cont.)

- Analog signals can have an infinite number of values in a range, for example sound or video (like real numbers)

- Digital signals can have only a limited number of discrete values like integer numbers, however they typically have two levels, e.g., 0 and 1 or -1 and +1



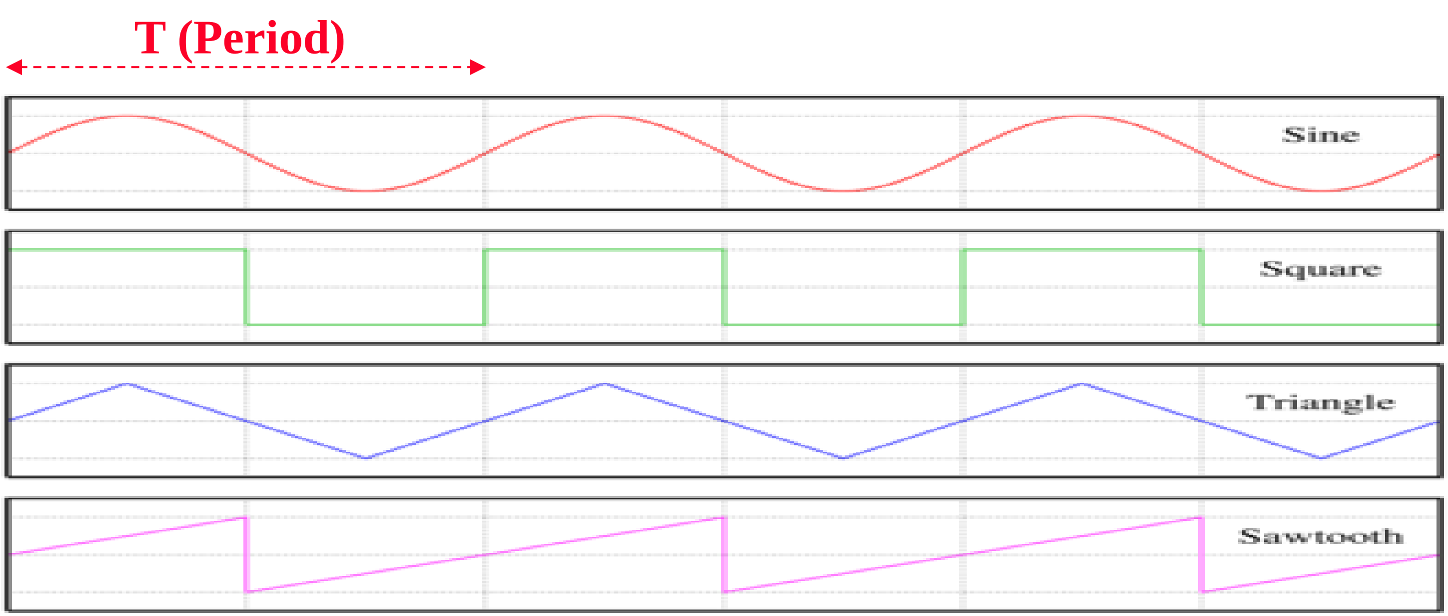
a. Analog signal



b. Digital signal

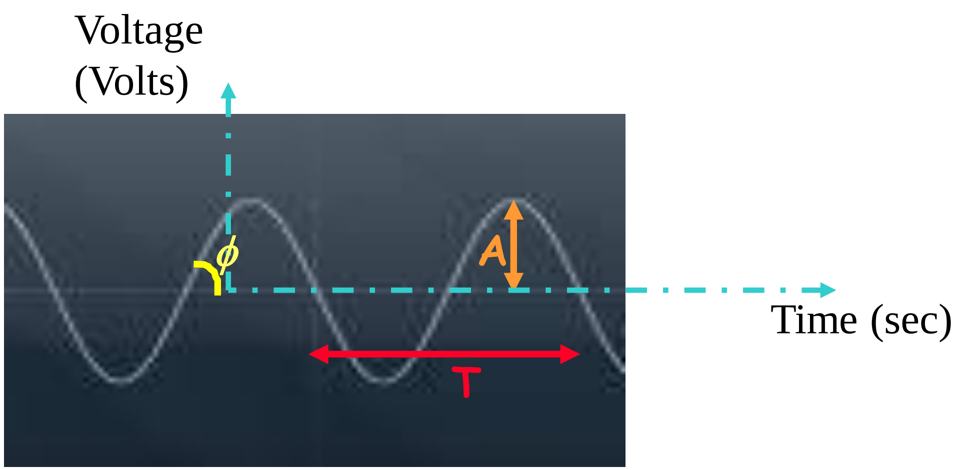
Periodic Signals

- A signal that repeats itself is periodic, e.g., periodic sine or cosine signal waveforms
- A periodic signal can be recognized by three parameters, its frequency, its peak amplitude and its phase



Signal Key Parameters (frequency, Amplitude and Phase

- **Frequency** (or inverse of period, i.e., 1/T,) denoted by f, is the signal's rate of change and is measured in **Hertz** or **cycles per second**
- **Peak amplitude**, A, is the maximum strength of a signal and is measured in volts or amps
- **Phase**, ϕ, describes the position of the waveform relative to time 0
- **Example:** Sine wave representation: $A\sin(2\pi f t + \phi)$



Periodic Signals

- **A periodic signal with period T can be represented as sum of sinusoids using Fourier Series:**

$$x(t) = a_0 + a_1\text{Sine}(2\pi f_0 t + \phi_1) \quad + \quad a_2\text{Sine}(2\pi 2 f_0 t + \phi_2) \quad + \ldots$$
$$+ \quad a_k\text{Sine}(2\pi k f_0 t + \phi_k) \quad + \ldots$$

"DC" long-term average

fundamental frequency $f_0 = 1/T$ first harmonic

k^{th} harmonic

- Amplitude Spectrum $|a_0|, |a_1|, |a_2|, \ldots$
- $|a_k|$ determines amount of power in kth harmonic

Fourier Series

- In general Fourier series of f(t) is defined as:

$$f(t) = 0.5c + \Sigma \{a_n \text{Cosine}(2\pi nt/T) + b_n \text{Sine}(2\pi nt/T)\} \quad 1 \le n \le \infty$$

- Where f=1/T is the fundamental frequency, a_n and b_n are the sine and cosine amplitudes of the n^{th} harmonics and c is a constant
- It can be shown:

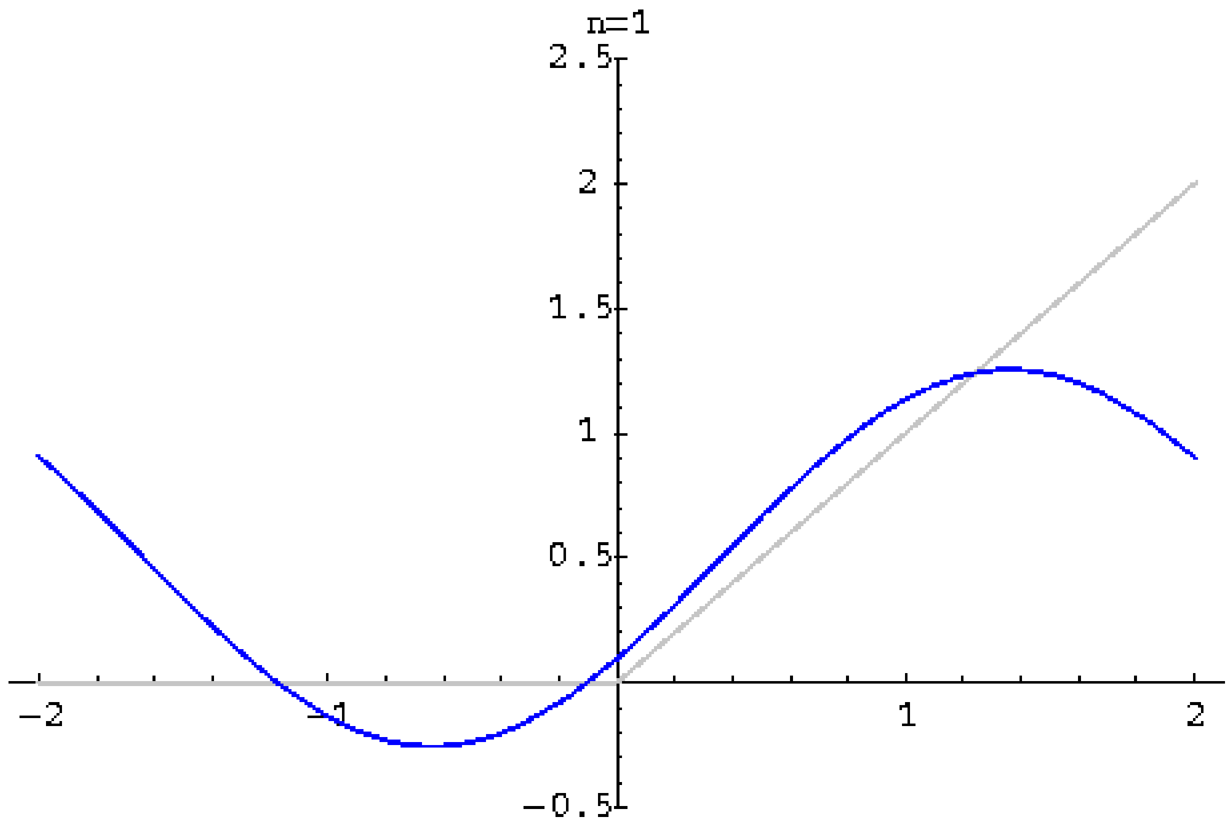
$$c = \frac{2}{T}\int_0^T f(t)\,dt$$

$$a_n = \frac{2}{T}\int_0^T f(t)\text{Cosine}(2\pi n/T)dt$$

$$b_n = \frac{2}{T}\int_0^T f(t)\text{Sine}(2\pi n/T)dt$$

[Optional] Higher Harmonics – Example

- Example: The Fourier series for a function that is defined as 0 if the input is between -2 and 0, and is the inputs if the input is between 0 and 2



- http://www.sosmath.com/fourier/fourier1/fourier1.html

[Optional] Higher Harmonics Example (Cont.)

- **Example: The Fourier series for a function that is defined as 0 if the input is between -2 and 0, and is the input if the input is between 0 and 2**

$$0.5c = \frac{1}{4}\int_0^2 x\,dx = \frac{1}{2},$$

$$a_n = \frac{1}{2}\int_0^2 x\cos\left(n\frac{\pi x}{2}\right)dx = \frac{1}{2}\left(\frac{2}{n\pi}\right)^2(\cos(n\pi)-1) = \frac{1}{2}\left(\frac{2}{n\pi}\right)^2((-1)^n-1),$$

$$b_n = \frac{1}{2}\int_0^2 x\sin\left(n\frac{\pi x}{2}\right)dx = -\frac{2\cos(n\pi)}{n\pi} = \frac{2}{n\pi}(-1)^{n+1}$$

- http://www.sosmath.com/fourier/fourier1/fourier1.html

Periodic Signals (Cont.)

- Change in a short span of time means high frequency
- Change over a long span of time means low frequency
- If a signal does not change at all, its frequency is 0
- If a signal changes instantaneously, its frequency is infinite

- Information
  - The interesting fact about periodic signals is that if you know the waveform for one cycle (for T seconds) you will be able to predict the signal behavior at any time
    - In other words if you know its frequency, its peak amplitude, and its phase you can predict it

Units of Period and Frequency

Unit	*Equivalent*	*Unit*	*Equivalent*
Seconds (s)	1 s	Hertz (Hz)	1 Hz
Milliseconds (ms)	10^{-3} s	Kilohertz (kHz)	10^{3} Hz
Microseconds (μs)	10^{-6} s	Megahertz (MHz)	10^{6} Hz
Nanoseconds (ns)	10^{-9} s	Gigahertz (GHz)	10^{9} Hz
Picoseconds (ps)	10^{-12} s	Terahertz (THz)	10^{12} Hz

- **Example 1: Express a period of 100ms in microseconds**

$$100\text{ ms} = 100 \times 10^{-3}\text{ s} = 100 \times 10^{-3} \times 10^{6}\ \mu\text{s} = 10^{2} \times 10^{-3} \times 10^{6}\ \mu\text{s} = 10^{5}\ \mu\text{s}$$

- **Example 2: The voltage of a battery is a constant; this constant value can be considered a sine wave with frequency zero and peak amplitude of 1.5V**
- **Example 3: The power we use at home has a frequency of 60Hz. The period of this sine wave can be determined as follows:**

$$T = \frac{1}{f} = \frac{1}{60} = 0.0166\text{ s} = 0.0166 \times 10^{3}\text{ ms} = 16.6\text{ ms}$$

Periodic Signals – Examples (Cont.)

- Example 4: The period of a signal is 100ms. What is its frequency in kilohertz?

$$100 \text{ ms} = 100 \times 10^{-3} \text{ s} = 10^{-1} \text{ s}$$

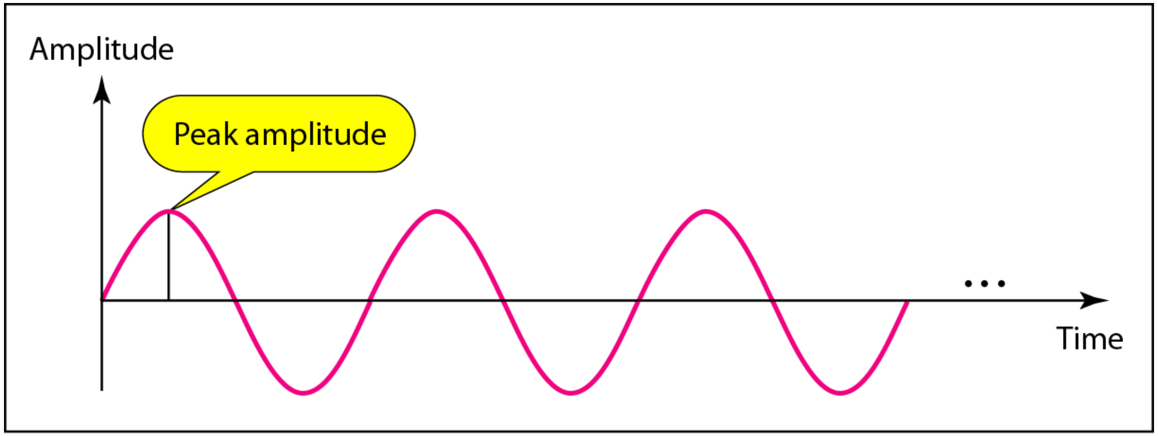
$$f = \frac{1}{T} = \frac{1}{10^{-1}} \text{ Hz} = 10 \text{ Hz} = 10 \times 10^{-3} \text{ kHz} = 10^{-2} \text{ kHz}$$

- Example 5: The power in your house can be represented by a sine wave with a peak amplitude of 155 to 170V. However, it is common knowledge that the voltage of the power in U.S. homes is 110 to 120V. This discrepancy is due to the fact that these are root mean square (rms) values. The signal is squared and then the average amplitude is calculated
- The peak amplitude is equal to $(2)^{\frac{1}{2}}$ × rms value
- Example 6: A sine wave is offset 1/6 cycle with respect to time 0. What is its phase in degrees and radians?
  - We know that 1 complete cycle is 360° (degree) or 2π Radians (or rad.) 1/6 cycle is hence:

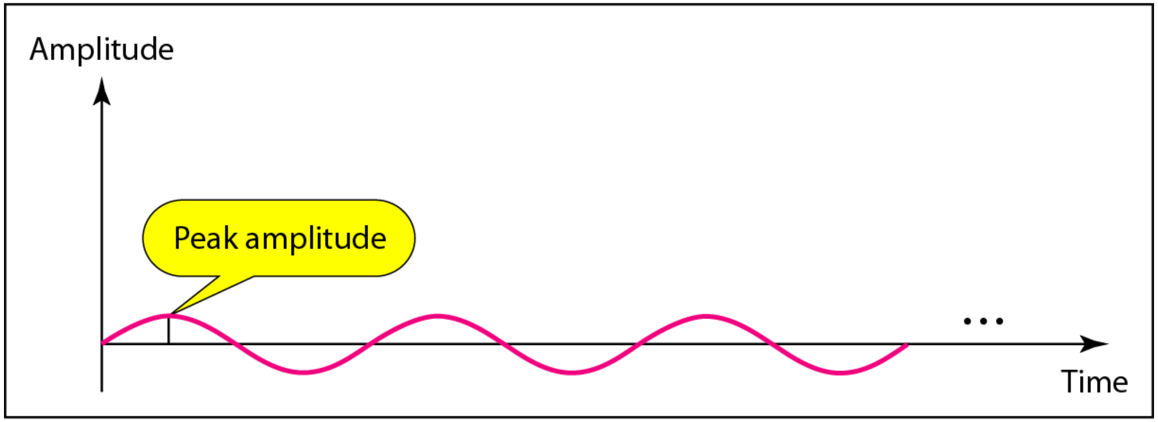
$$\frac{1}{6} \times 360 = 60° = 60 \times \frac{2\pi}{360} \text{ rad} = \frac{\pi}{3} \text{ rad} = 1.046 \text{ rad}$$

Periodic Signals – Different Amplitudes

- **Two signals with the same frequency and phase, but different peak amplitudes**



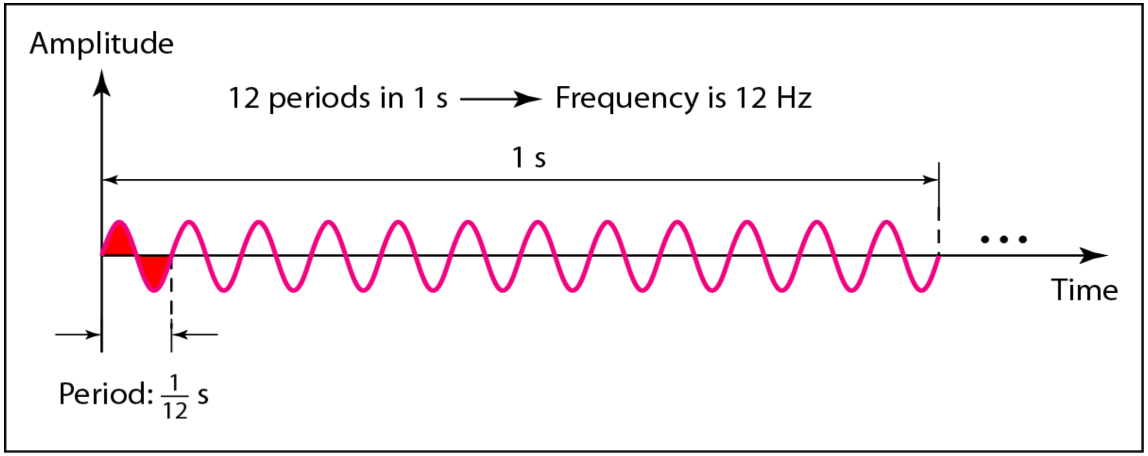
a. A signal with high peak amplitude



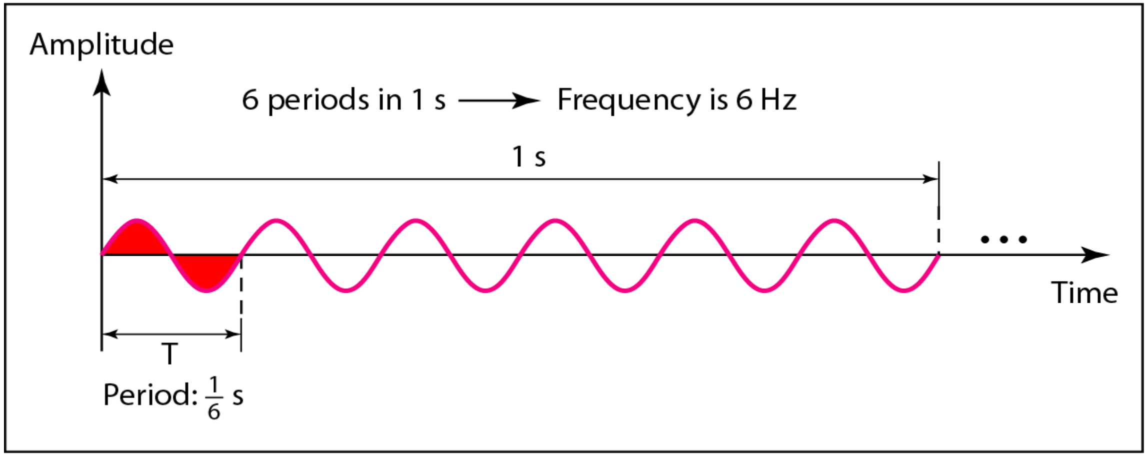
b. A signal with low peak amplitude

Periodic Signals – Different Frequencies

- **Two signals with the same phase and peak amplitude, but different frequencies**



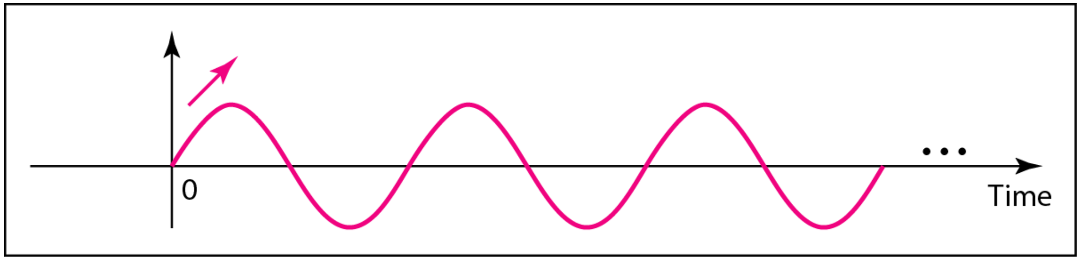
a. A signal with a frequency of 12 Hz



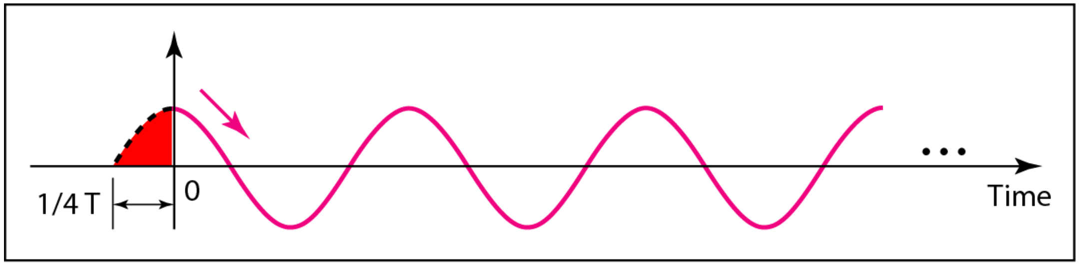
b. A signal with a frequency of 6 Hz

Periodic Signals – Different Phases

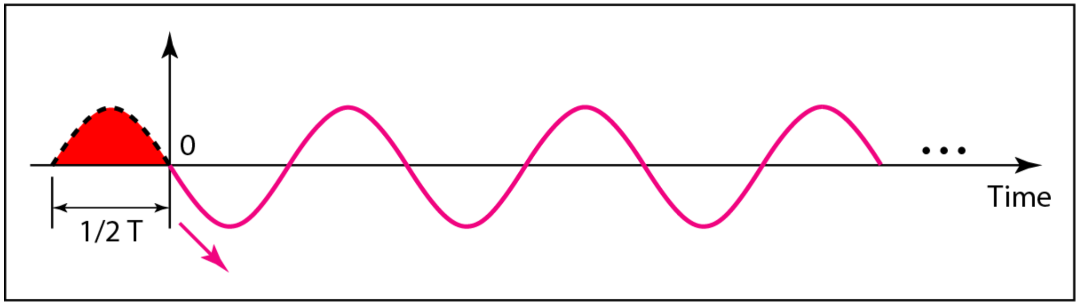
- **Three sine waves with the same frequency and peak amplitude, but different phases**



a. 0 degrees



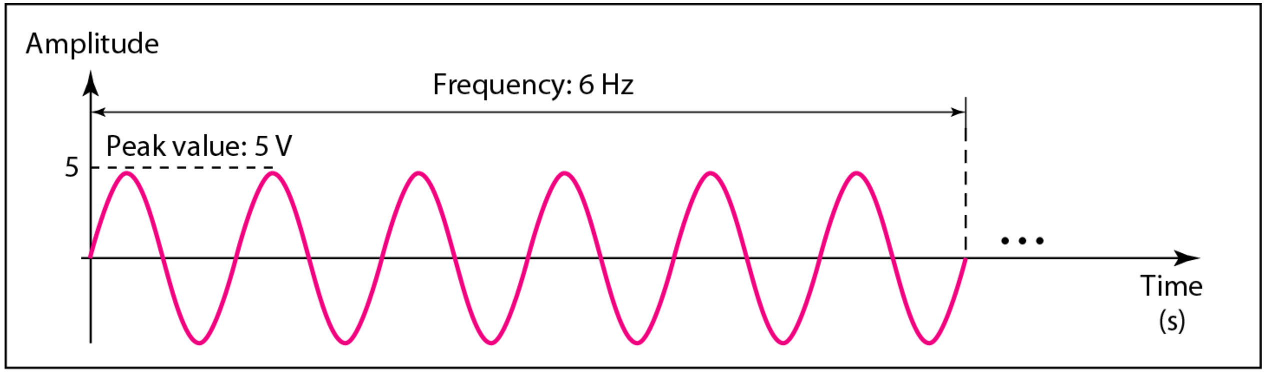
b. 90 degrees



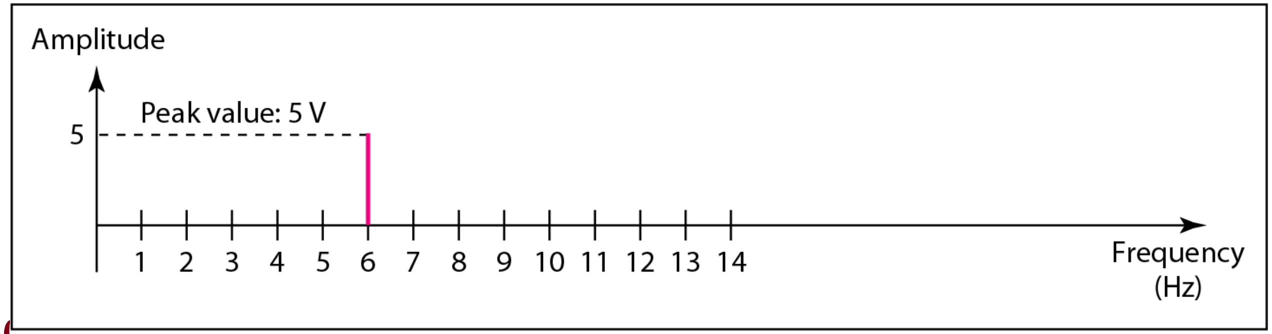
c. 180 degrees

Time Domain and Frequency Domain

- A complete (periodic) sine wave in the time domain can be represented by one single spike in the frequency domain
- Spectrum of a signal (or its frequency domain) is the magnitude of amplitudes as a function of frequency



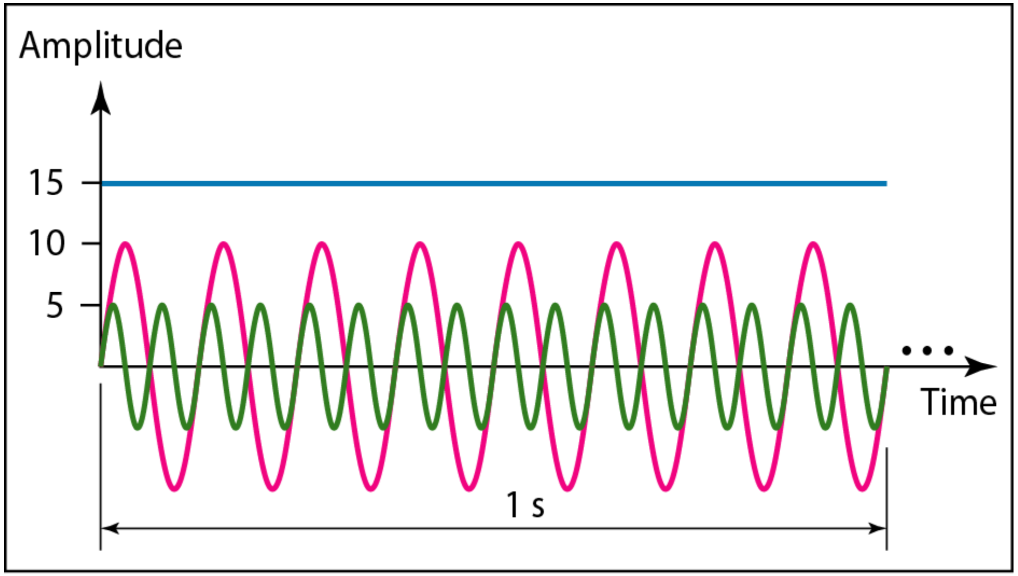
a. A sine wave in the time domain (peak value: 5 V, frequency: 6 Hz)



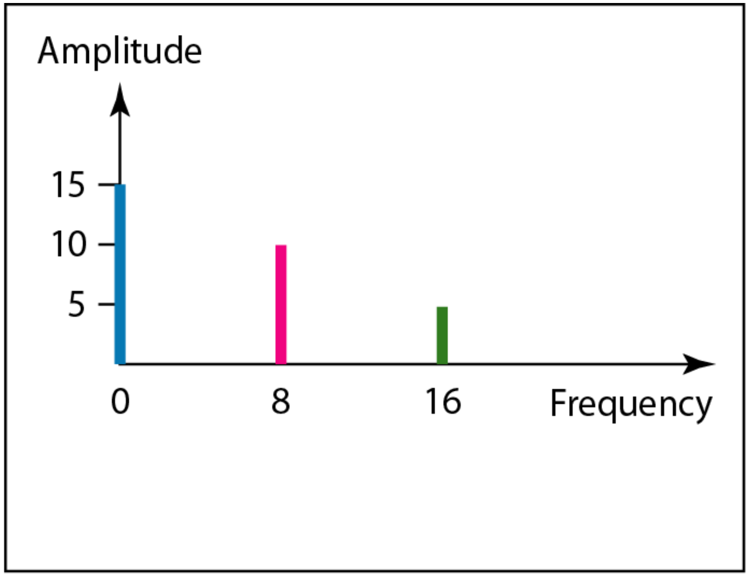
b. The same sine wave in the frequency domain (peak value: 5 V, frequency: 6 Hz)

Frequency Domain

- The frequency domain is more compact and useful when we are dealing with more than one sine wave.
- For example, the following shows three sine waves, each with different amplitude and frequency. All can be represented by three spikes in the frequency domain



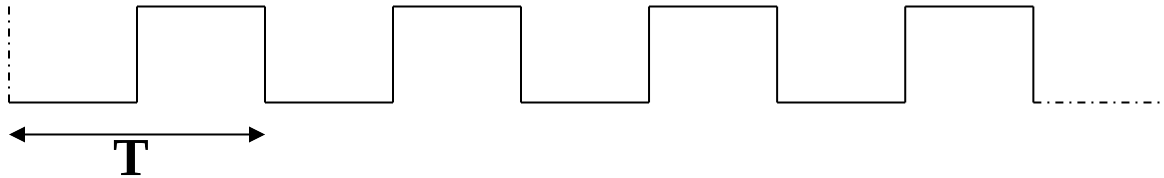
a. Time-domain representation of three sine waves with frequencies 0, 8, and 16



b. Frequency-domain representation of the same three signals

Frequency Domain (Cont.)

- Example: What are the frequencies in a square wave (pulse train) periodic signal?



  - Pulse train has a fundamental frequency of $f_0 = 1/T$. It also has the harmonic frequencies of $f_n = nf_0$ (with peak amplitude of $1/n$ of the peak amplitude of f_0 waveform and $n = 1, 3, 5, \ldots$
  - Again note the low amount of information in this periodic signal (even if this contains a lot of sine waves, but still it is predictable)
- 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

Baseband, Passband and Broadband

- **Baseband signals** run from 0 to maximum frequency
- **Passband signals** are shifted to occupy a higher range of frequencies, e.g., wireless transmissions use passband signals
- A **baseband transmission** devotes the entire capacity of the medium to one communication channel, e.g., most of the LANs are baseband
- Baseband to passband conversion: (0,B) -> (S,S+B)
- A **broadband transmission** enables two or more communication channels to share the bandwidth of the communications medium, such as TV

Composite Signal

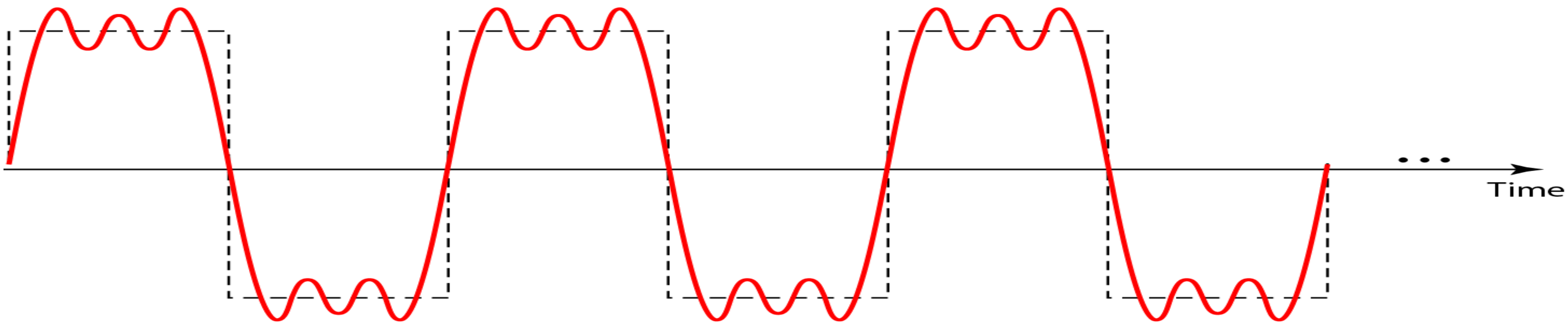
- According to Fourier analysis, any composite signal is a combination of simple sine waves with different frequencies, amplitudes, and phases

- If the composite signal is periodic, the decomposition gives a series of signals with discrete frequencies

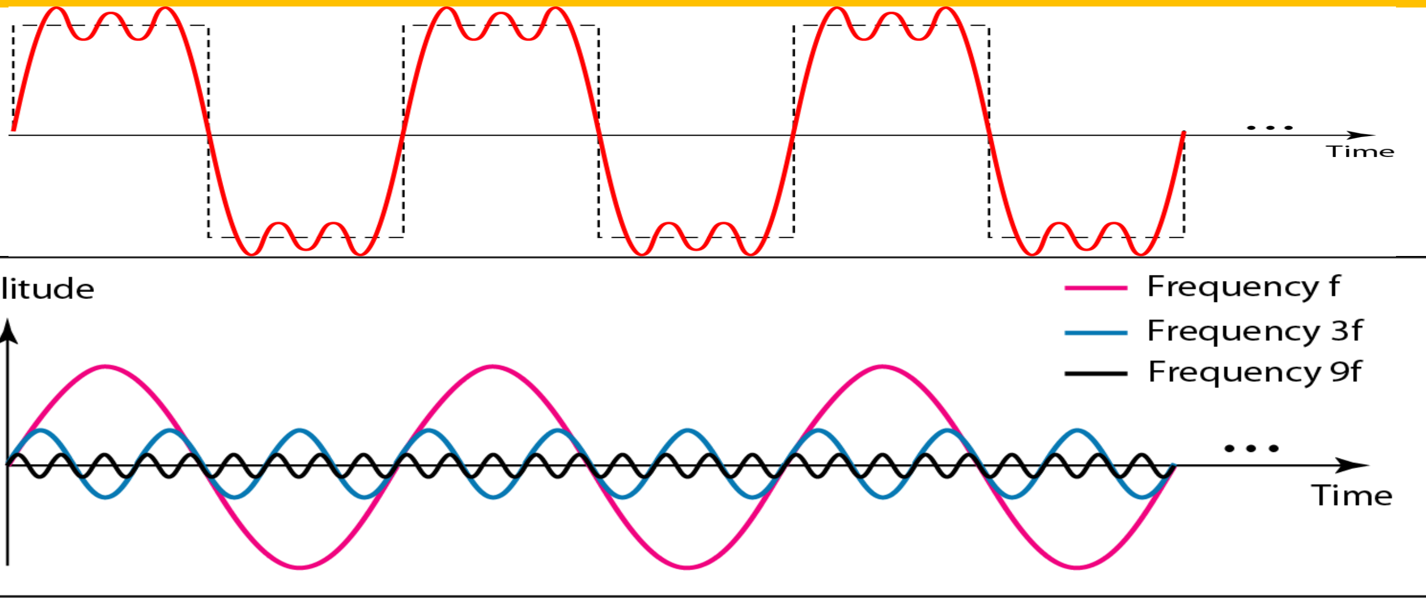
- If the composite signal is non-periodic, the decomposition gives a combination of sine waves with continuous frequencies

Composite Signal (Cont.)

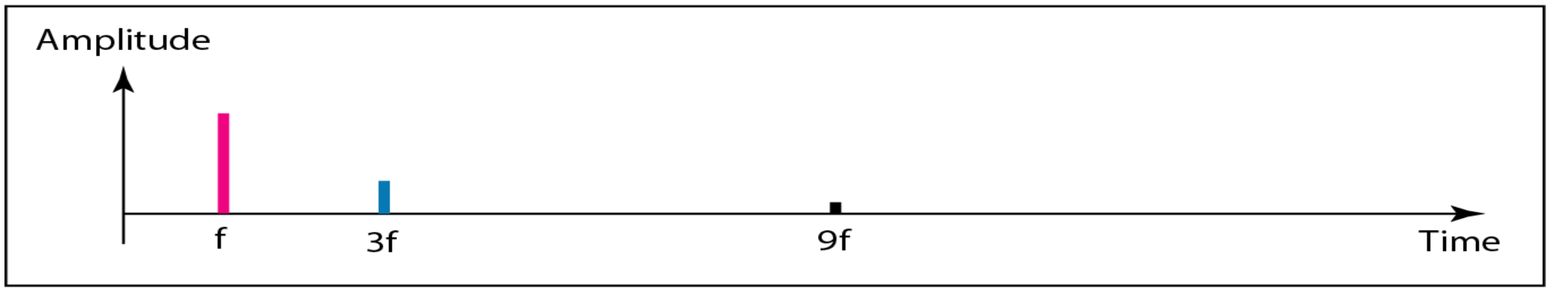
- Example:
  - The following periodic composite signal with frequency f consists of three waveforms with different frequencies
  - Signal decomposition analysis using a spectrum analyzer or oscilloscope



Composite Signal (Cont.)



a. Time-domain decomposition of a composite signal

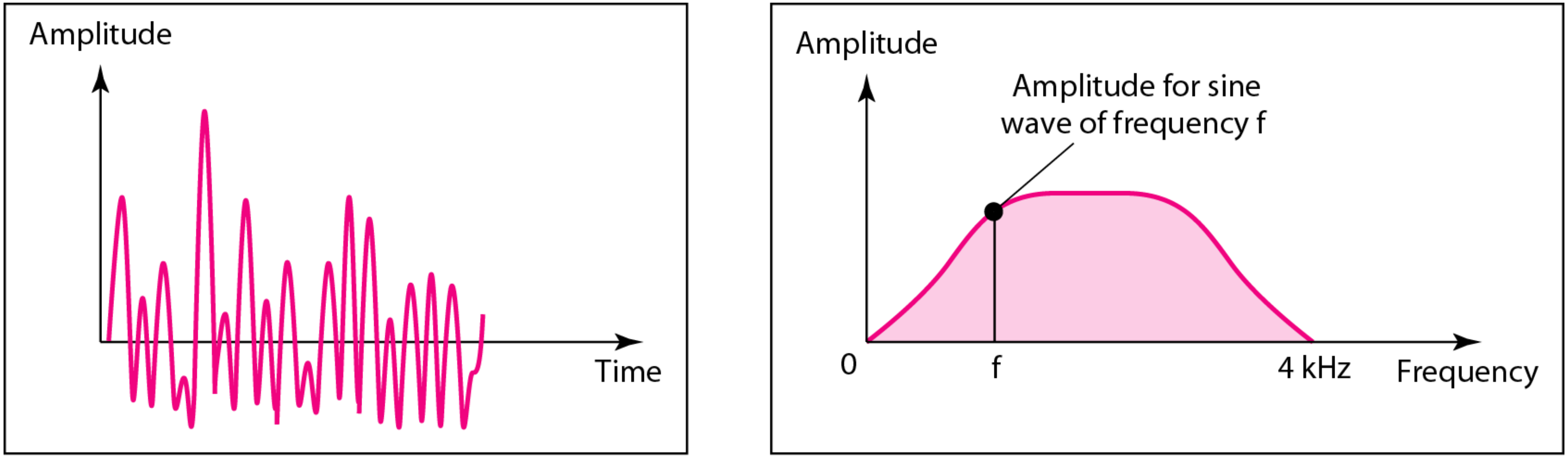


b. Frequency-domain decomposition of the composite signal

- **Note that adding the first two harmonic frequencies to original sine wave has made it look more similar to a square wave. Still there is a difference; what does that difference represent?**

Composite Signal (Cont.)

- The following figure shows a nonperiodic composite signal. It can be the signal created by a microphone or a telephone set when a word or two is pronounced. In this case, the composite signal cannot be periodic, because that would imply repeating the same word or words with exactly the same tone

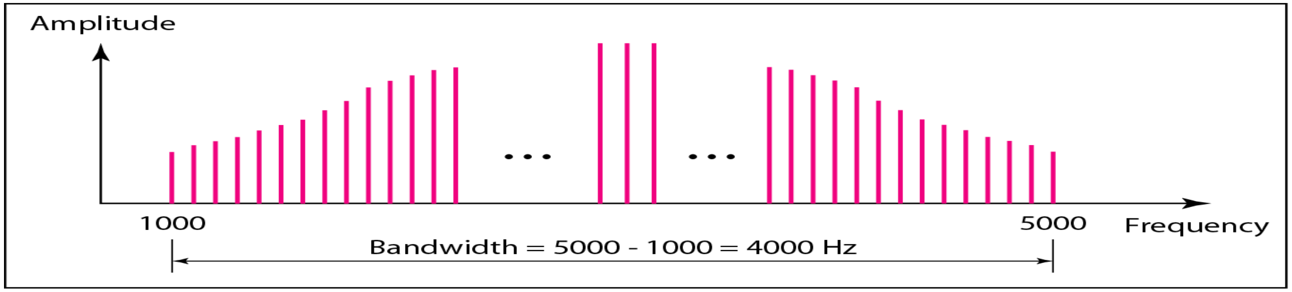


a. Time domain

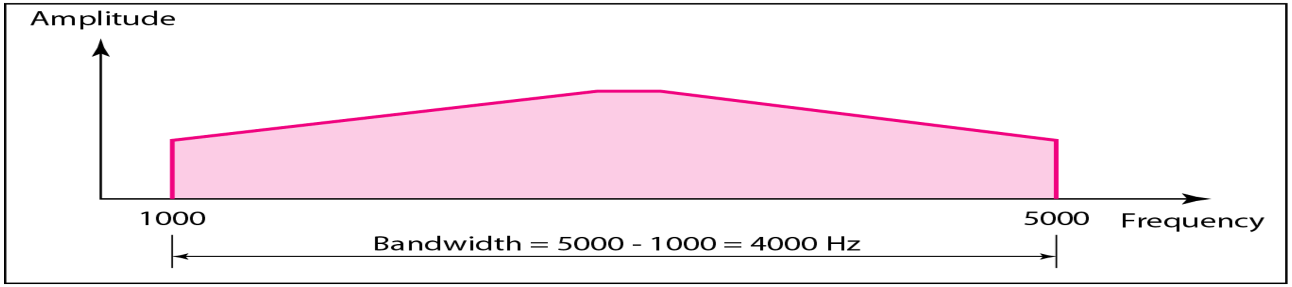
b. Frequency domain

Bandwidth of the Composite Signal

- Bandwidth of a signal is defined as its range of frequencies
- Bandwidth implies how wide the (frequency) spectrum is
- Example: The bandwidth of a periodic and nonperiodic composite signals:

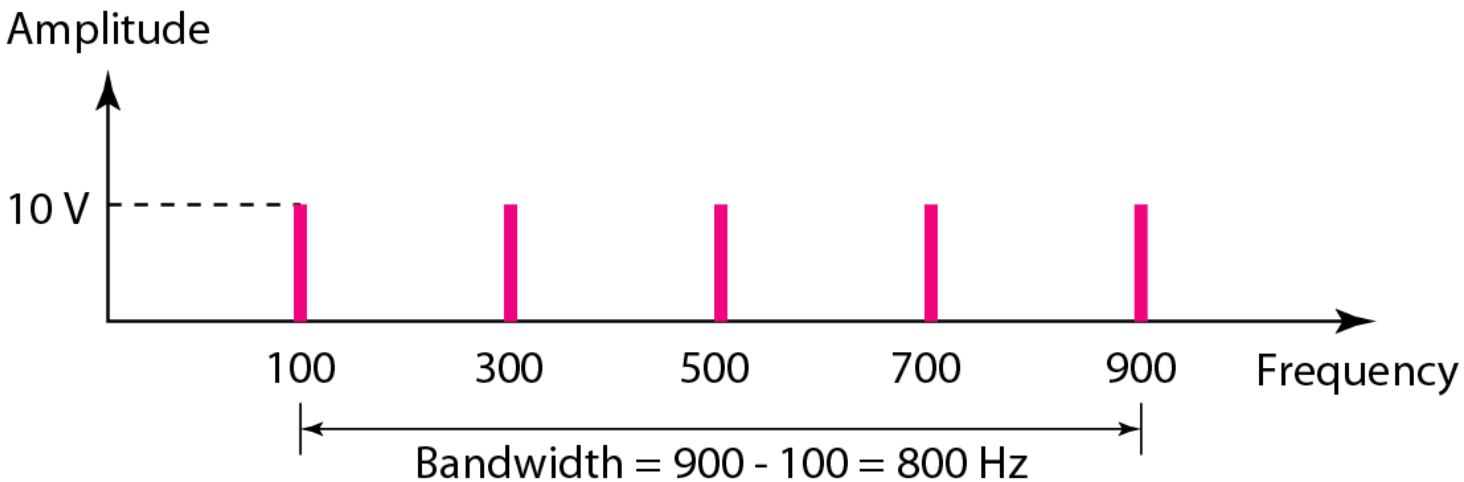


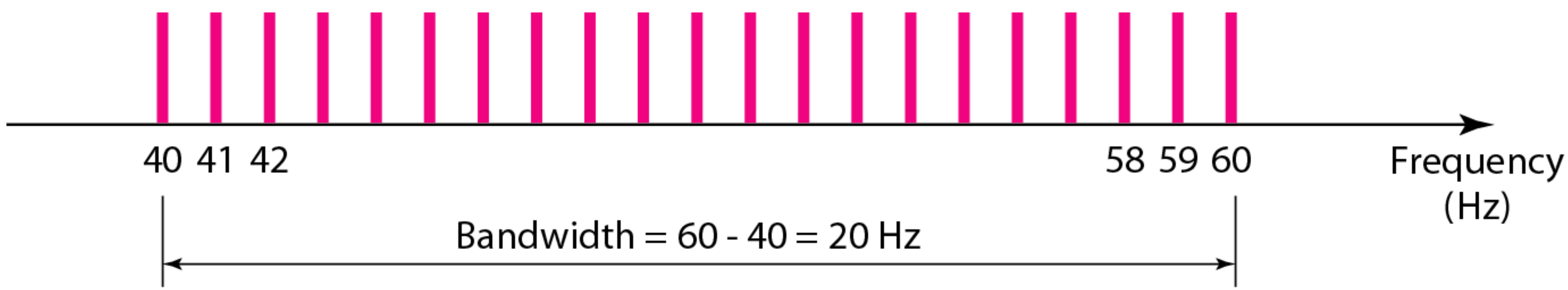
a. Bandwidth of a periodic signal



b. Bandwidth of a nonperiodic signal

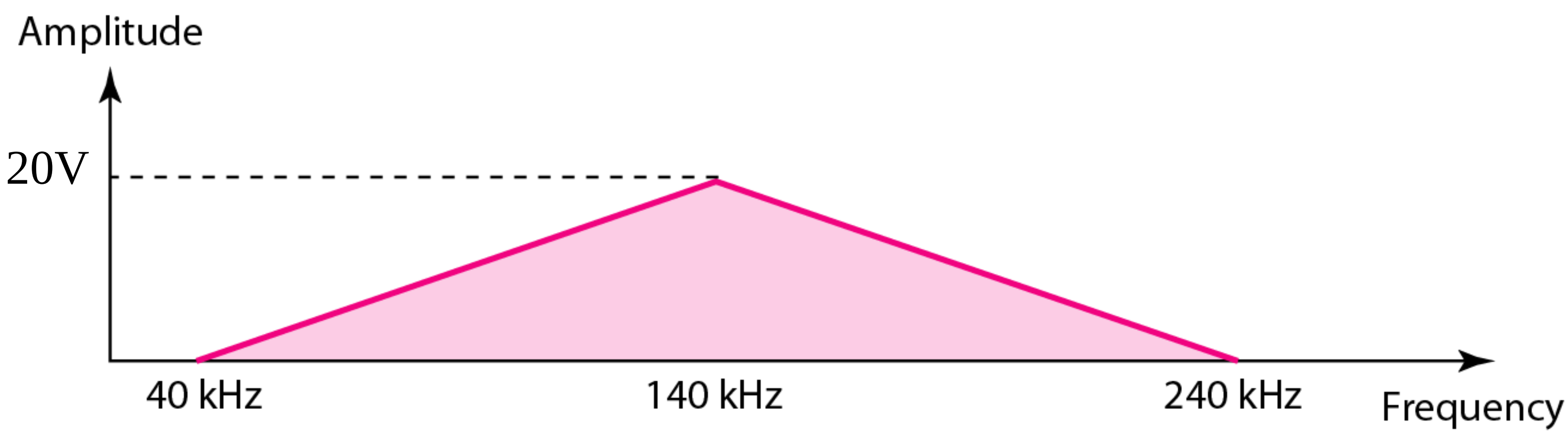
Composite Signal Bandwidth - Example





Composite Signal Bandwidth - Example

- A nonperiodic composite signal has a bandwidth of 200kHz, with a middle frequency of 140kHz and peak amplitude of 20V. The two extreme frequencies have an amplitude of 0. Draw the frequency domain of the signal
  - The lowest frequency must be at 40kHz and the highest at 240kHz



Composite Signal Bandwidth - Examples

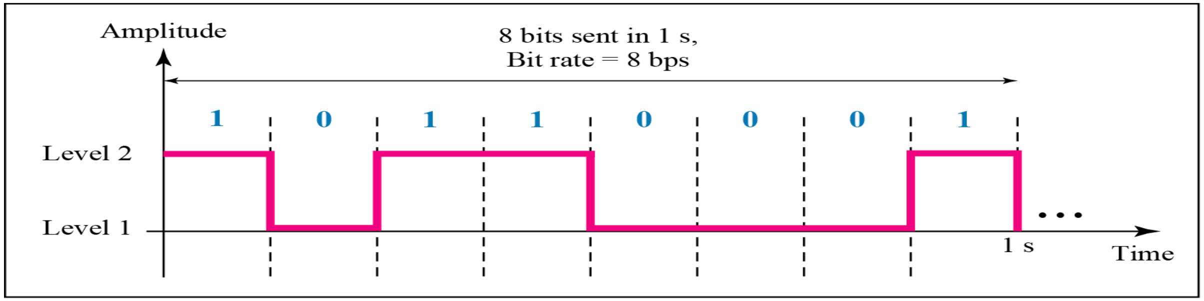
- Example I: The signal propagated by an AM radio station is a nonperiodic composite signal
  - The total bandwidth dedicated to AM radio ranges from 530 to 1700kHz
- Example II: The signal propagated by an FM radio station is a nonperiodic composite signal.
  - The total bandwidth dedicated to FM radio ranges from 88 to 108MHz
- Example III: The telephone system for which the bandwidth is 3.1KHz (each channel ranges from 300Hz to 3400Hz. Bandwidth may be rounded up and considered as 4KHz

Bandwidth Examples (Cont.)

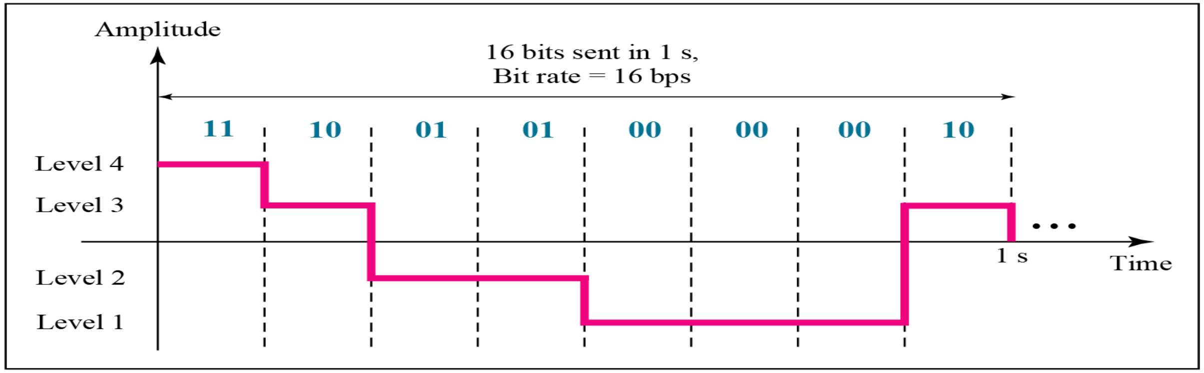
- Example IV: The signal received by an old-fashioned analog black-and-white TV is a nonperiodic composite signal. A TV screen is made up of pixels. If we assume a resolution of 525 × 700, we have 367,500 pixels per screen. If we scan the screen 30 times per second, this is 367,500 × 30 = 11,025,000 pixels per second. The worst-case scenario is alternating black and white pixels. We can send 2 pixels per cycle. Therefore, we need 11,025,000/2 = 5,512,500 cycles per second, or Hz. The bandwidth needed is hence 5.5125 MHz
- Bandwidth of video signal is considered as about 4MHz to 6MHz, therefore compared to telephone channel a TV channel has much higher bandwidth to be able to carry video (which has a lot more information that voice)

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



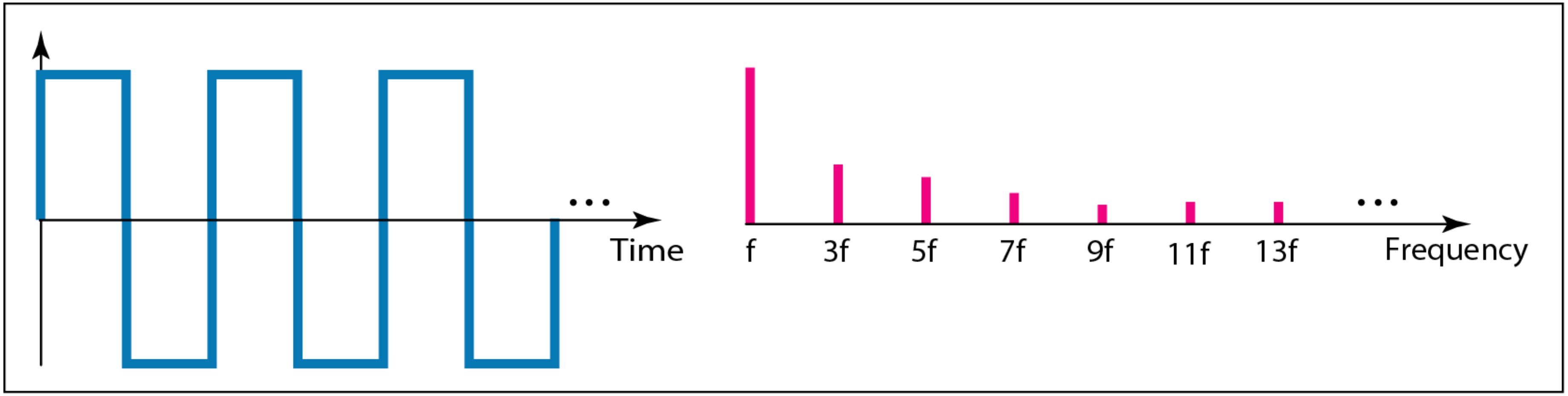
a. A digital signal with two levels



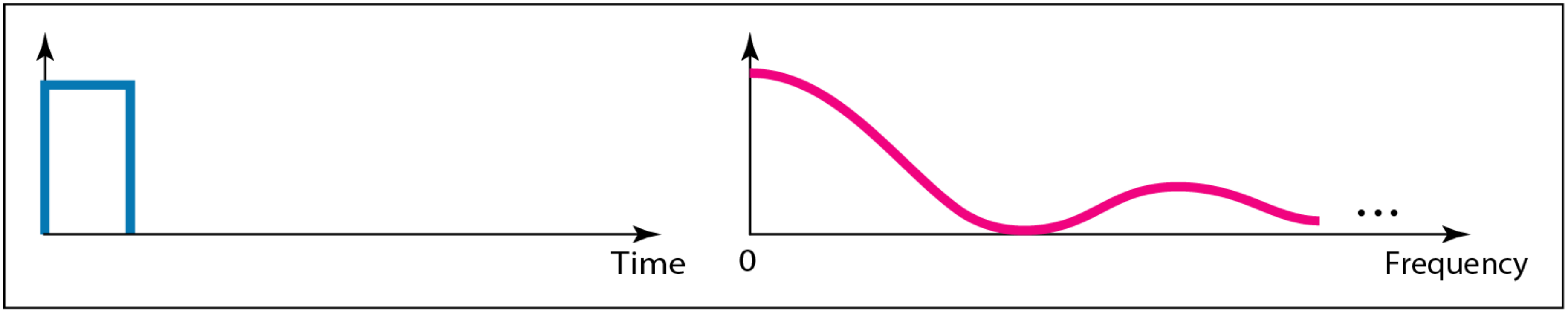
b. A digital signal with four levels

Digital Signal Bandwidth and Domain

- The time and frequency domains of periodic and nonperiodic digital signals



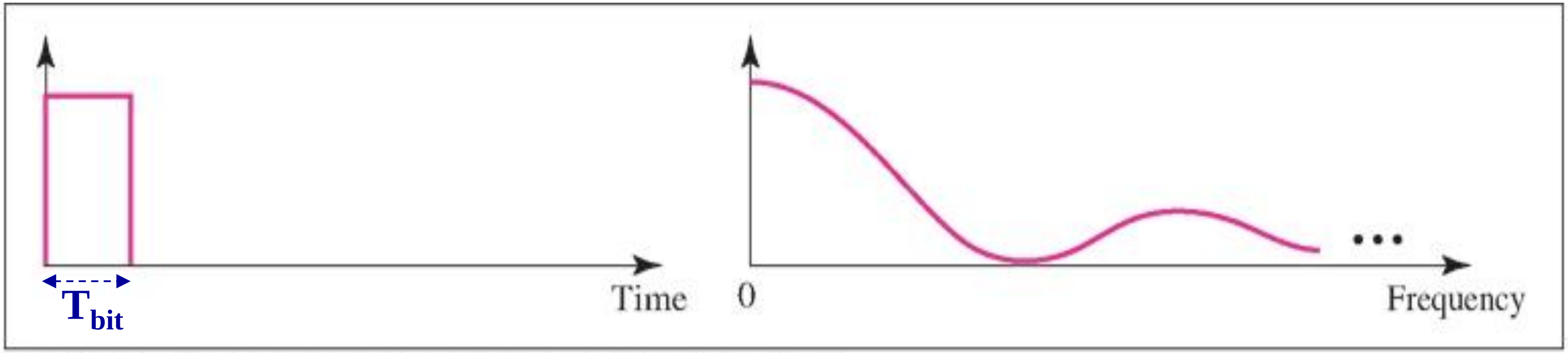
a. Time and frequency domains of periodic digital signal



b. Time and frequency domains of nonperiodic digital signal

Digital Signal Bandwidth

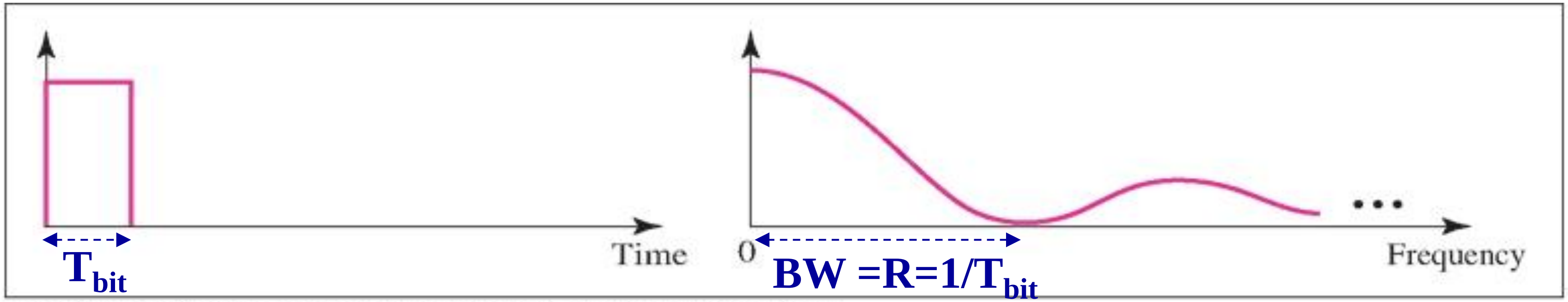
- Remember that the bandwidth of a composite signal is the difference between the highest and the lowest frequencies contained in that signal
- The following digital signal is a single nonperiodic pulse (composite signal). Bandwidth as highest frequency minus lowest frequency is infinity. However the contribution of higher frequencies is low. Bandwidth for this case is considered as the bit rate, R



b. Time and frequency domains of nonperiodic digital signal

Digital Signal Bandwidth (Cont.)

- Bit rate, R (and hence the bandwidth) is inversely proportional to the bit duration T_{bit}. Therefore the narrower the pulse, the shorter the bit duration and then the higher the bit rate (and the bandwidth)
- Also bit rate is the frequency of the first lobe in the frequency domain, so assuming bandwidth as the bit rate is like ignoring the following lobes (i.e., higher frequencies.) In fact this is reasonable because their contribution in the digital pulse signal is negligible



b. Time and frequency domains of nonperiodic digital signal

Signal to Noise Ratio, Bit Error Rate

- While signal (digital or analog) that is carrying information is being transmitted, the data experiences degradation, so what is received is different from what was sent

- The signal will degrade due to noise such as thermal or crosstalk noise, interference, amplitude attenuation or delay distortion (due to system's low bandwidth)

- This happens to both digital and analog signals

SNR, BER (Cont.)

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

$$SNR = \frac{10{,}000\ \mu W}{1\ \mu W} = 10{,}000$$

$$SNR_{dB} = 10 \log_{10} 10{,}000 = 10 \log_{10} 10^4 = 40$$

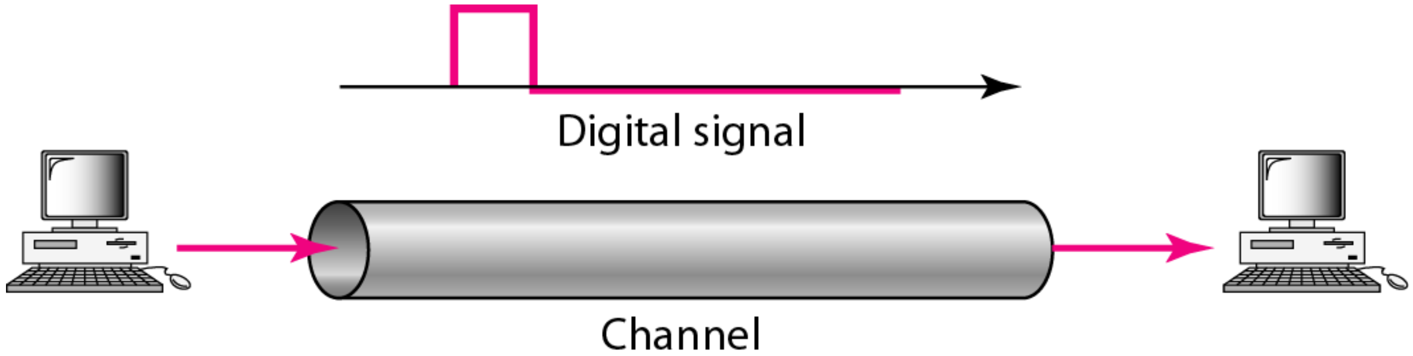
- Example II: What does a BER of 0.005 mean?
  - It means out of each 1000 bits, 5 of them are going to be erroneous in average
- Example III: Calculate SNR and SNR_{dB} for a noiseless channel

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

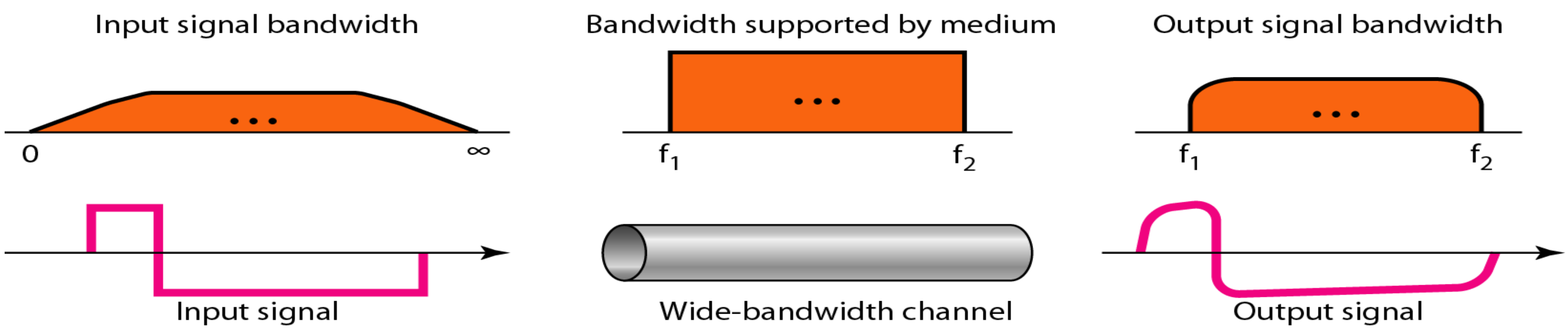
$$SNR_{dB} = 10 \log_{10} \infty = \infty$$

Channel Capacity

- The channel capacity needs to be equal to or greater than the bandwidth of the signal to avoid any signal degradation



- Example:



Shannon's Theorem

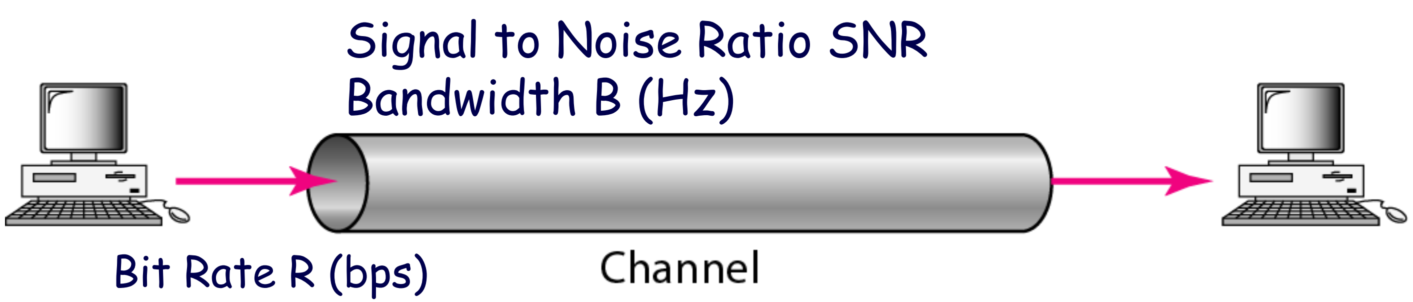
- Assume the system (medium or channel) bandwidth is B (in Hz) then the channel capacity is

$$C = B \log_2(1 + SNR)$$

- To increase the channel capacity, the channel bandwidth or SNR should be increased (i.e., increased signal power or reduced noise power)
- **Shannon's theorem** states that the maximum transmission bit rate, R should be the channel capacity to achieve reliable transmission, i.e., low BER

$$R \leq C = B\log_2 (1+SNR)$$

- This is the foundation of digital communication and network



Shannon's Theorem – Examples

- Example I: A telephone line normally has a bandwidth of 3KHz. For a SNR of 3162 find the channel capacity:

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

  - If R is selected higher than C, e.g., equal to 700kbps, based on Shannon Theorem the transmission will not be reliable

- Example II: $\text{SNR}_{\text{dB}} = 36$ and the channel bandwidth is 2MHz. The theoretical channel capacity can be calculated as:

$$\text{SNR}_{\text{dB}} = 10 \log_{10} \text{SNR} \rightarrow \text{SNR} = 10^{\text{SNR}_{\text{dB}}/10} \rightarrow \text{SNR} = 10^{3.6} = 3981$$
$$C = B \log_2 (1 + \text{SNR}) = 2 \times 10^6 \times \log_2 3982 = 24 \text{ Mbps}$$

Shannon's Theorem – Examples (Cont.)

- Example III: Consider an extremely noisy channel in which the value of the signal-to-noise ratio is almost zero. In other words, the noise is so strong that the signal is faint. For this channel the capacity C is calculated as:

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

- Example IV: We have a channel with a 1MHz bandwidth. The SNR for this channel is 63. What is the appropriate bit rate?

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

Analog Data Transmission

- **A/A**: Data signal can be analog and represented by analog signal. E.g., sound signal is analog and a telephone system would convert analog sound into analog signal

- **D/A**: Data signal can be digital and represented by analog signals. E.g., if you use dialup, DSL, or cable connection to your computer, the data sent out from your PC is digital. A Modem would convert the digital signal to analog to make the data suitable for the medium to transport

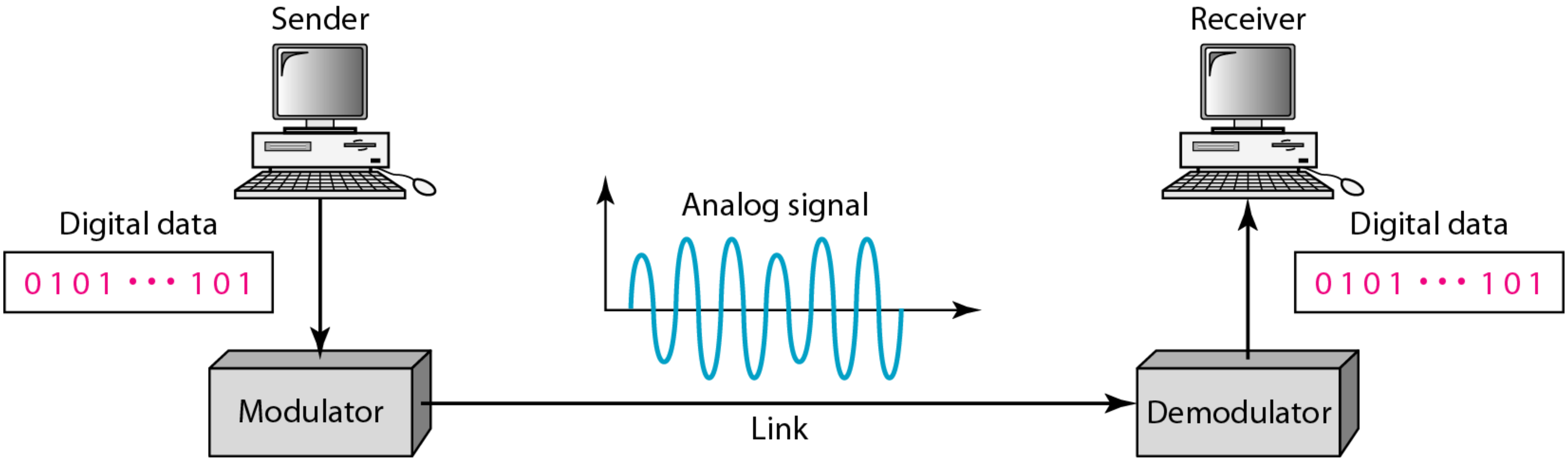
Digital Data Transmission

- **D/A**: Data signal can be analog and represented by digital signal. E.g., the telephone signal travels from telephone system to the CO (Central Office) and there it will be converted into digital signal to make the signal suitable for the telephone network (note that every part of the telephone network is digital and the only analog part is from telephone systems (customers) to COs. Also cell phones convert the analog voice signal to digital

- **D/D**: The data can be digital and represented by another digital signal, meaning that the code would be different for them. For example PC data is based on 0 and 1, but during transmission the digital levels of -1 and +1 may be used

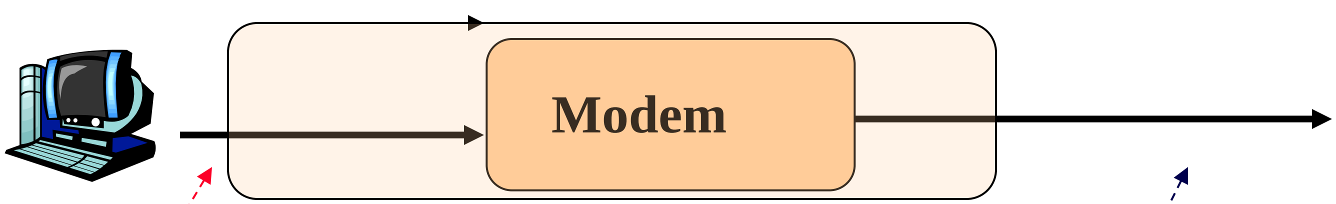
Modulation

- **Modulation** is the process of varying one or more of the key parameters (amplitude, frequency, phase) of a carrier signal in relation to the information signal



Baud Rate

- **Bit rate**, **R**, is the number of <u>bits</u> per second
- **Baud rate**, R_{baud}, is the number of signal elements per second. It is also called the signal rate, pulse rate, modulation rate, and synonymous to pulses per second or symbols per second
- In the analog transmission of digital data, the baud rate is less than or equal to the bit rate, $R_{baud} \leq$ Bandwidth=R therefore baud rate and bit rate are different

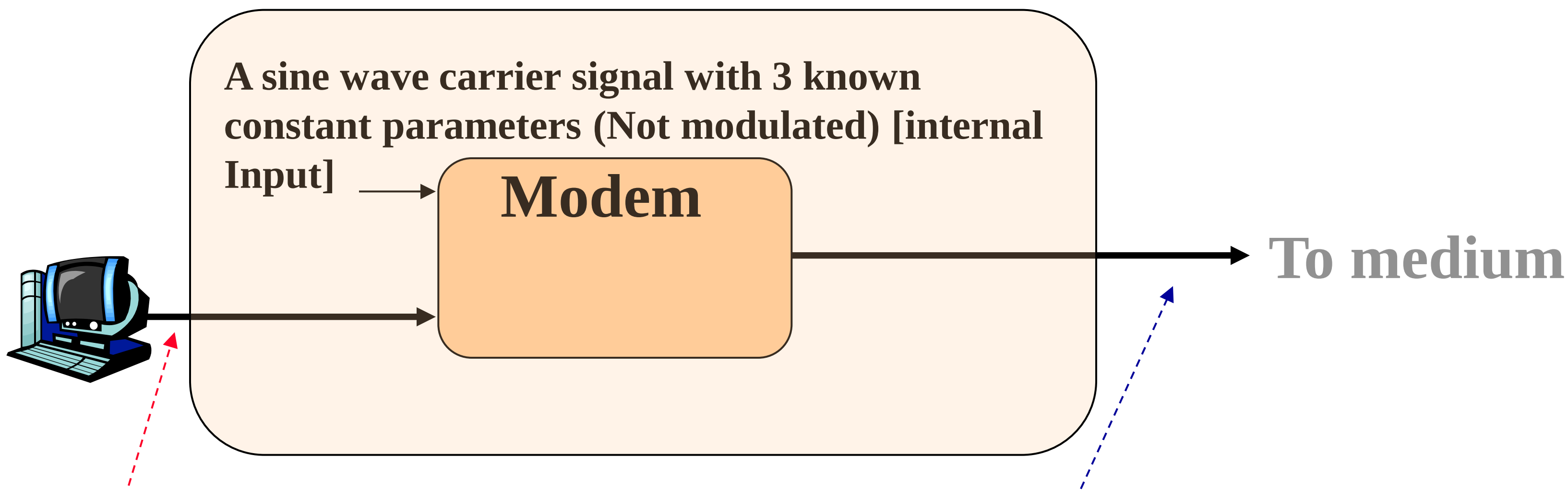


Information signal (digital) with rate R (bps)

Modulated Carrier (analog) which carries the digital information with Baud rate R_{baud}

Analog Transmission – D/A or Modulation

- **Binary modulation**: for every bit of information that enters the modem, one modulated signal comes out: $R_{baud} = R$
- **Multi-level modulation**: for every k bits of information that enters the modem one modulated signal comes out: $R_{baud} = R/k$

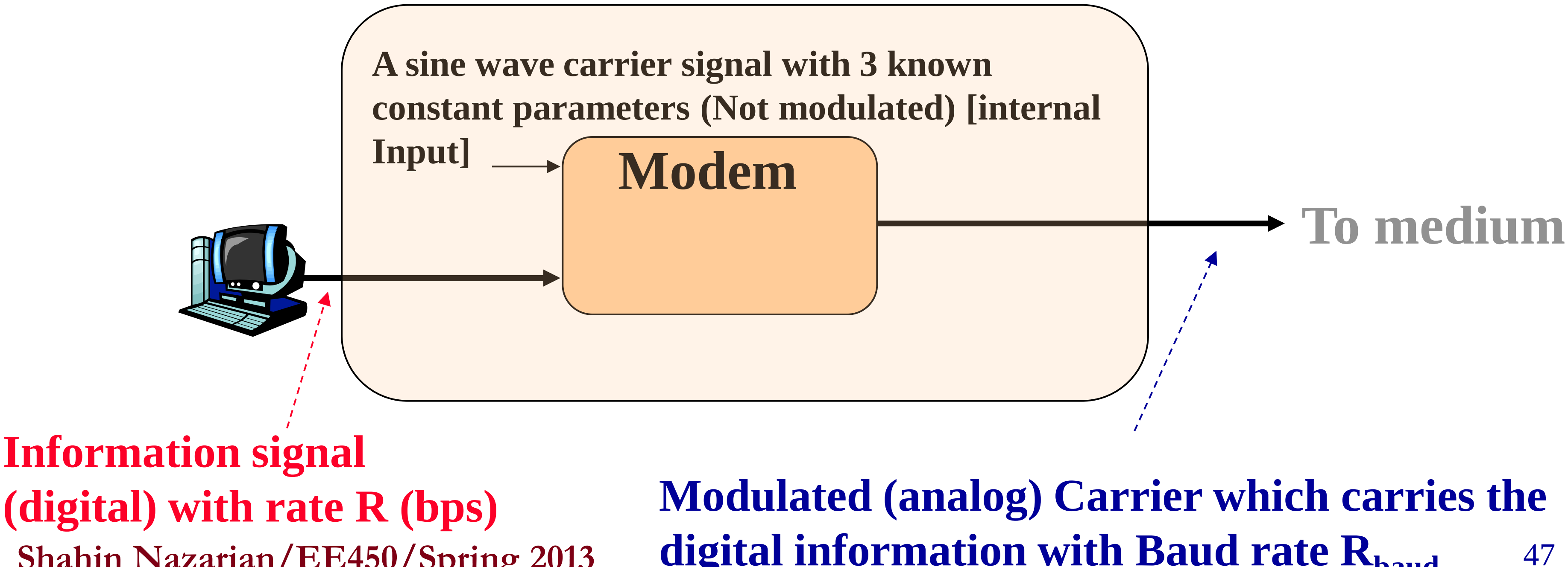


Information signal (digital) with rate R (bps)

Modulated (analog) Carrier which carries the digital information with Baud rate R_{baud}

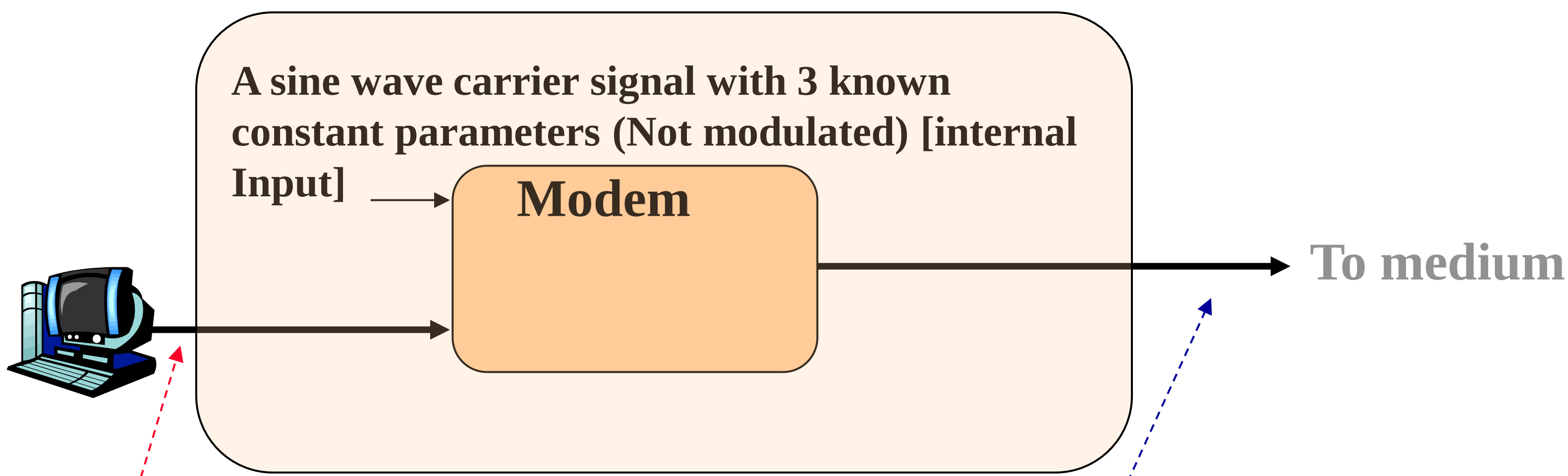
D/A or Modulation – Carrier Signal

- For all modem types, such the ones for ADSL, cable, dialup, satellite and wireless 802.11 the internal input of the modem (i.e., the sine wave carrier signal) is generated by an oscillator inside the modem



Multi-level Modulation

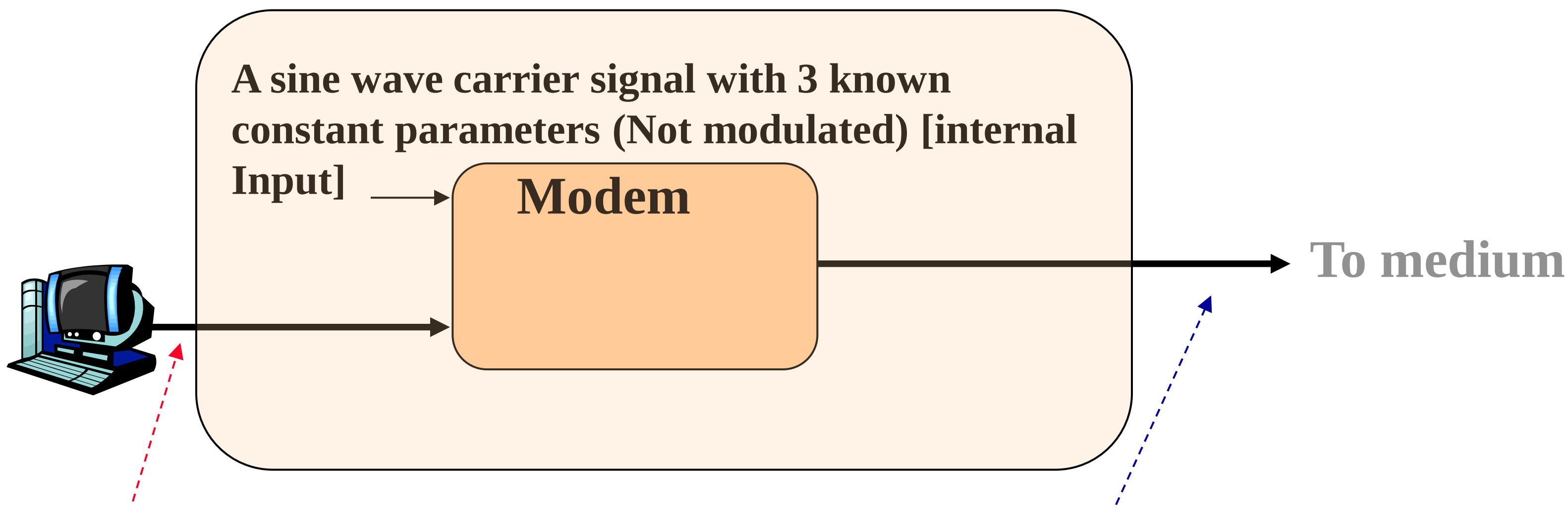
- Multi-level modems must have a buffer to keep the bits
- Example I: for k=2, there are 4 different signals for 00, 01, 10, and 11. The 1st bit goes into the modem and goes to a buffer; the 2nd bit goes in and then signal is generated
- Example II: For K=3, 8 numbers are 000 up to 111 and for each we need a signal, so a total of 8 signals is required and $R = 3\ R_{baud}$



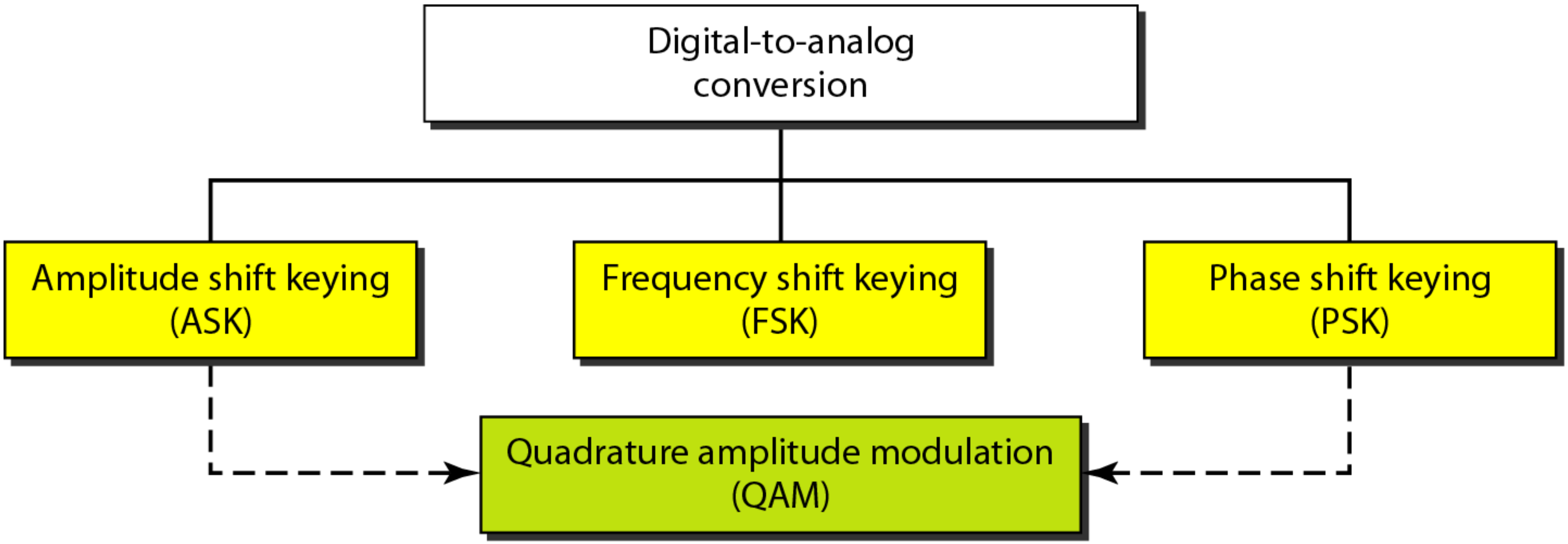
Bit Rate and Baud Rate Relation

- The user's goal is to increase the bit rate, this means either k, or R_{baud} or both should increase. Increasing R_{baud} means increasing the bandwidth

- How about increasing k?



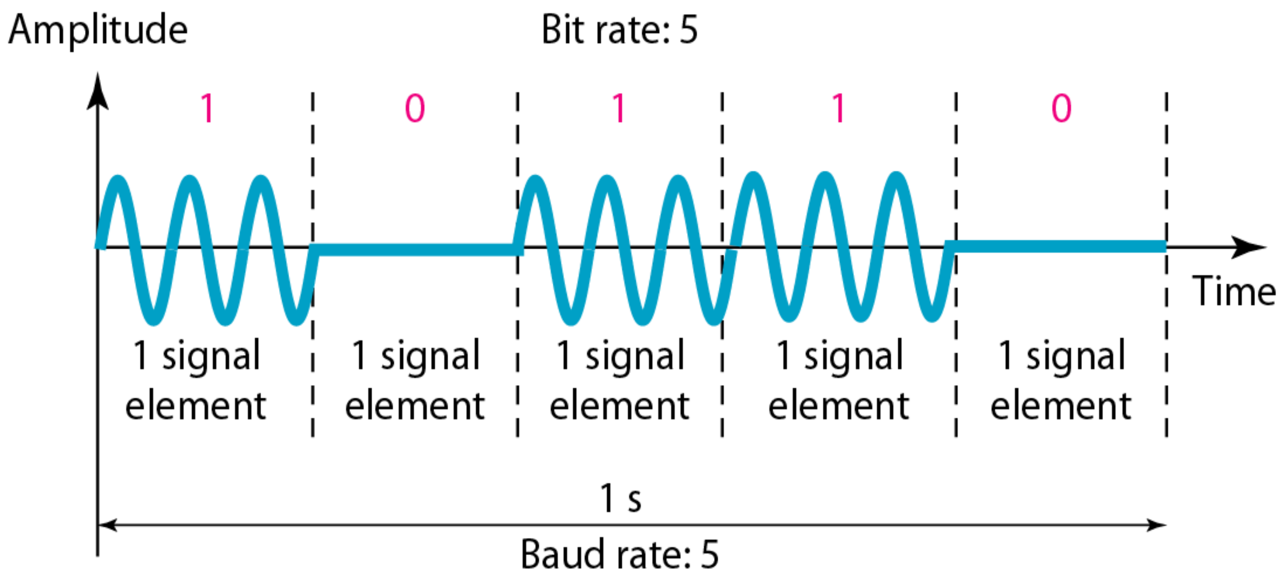
D/A – Modulation Techniques



Amplitude Modulation

- **Amplitude modulation** aka **Amplitude Shift Keying (ASK)**

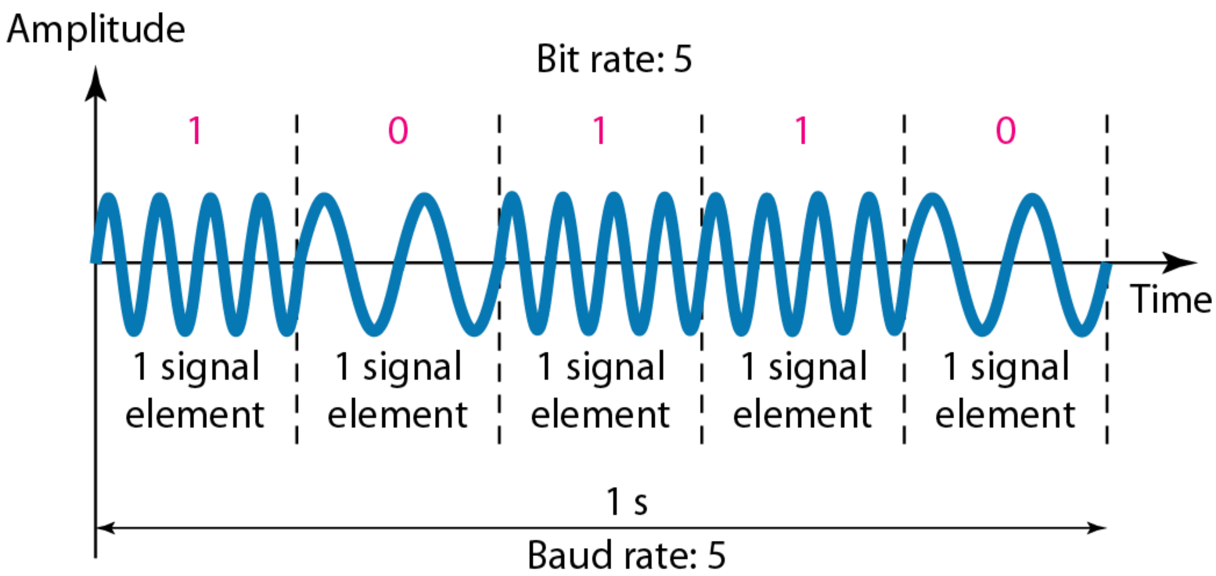
- **Binary ASK:**



Frequency Modulation

- Frequency modulation aka Frequency Shift Keying (FSK)

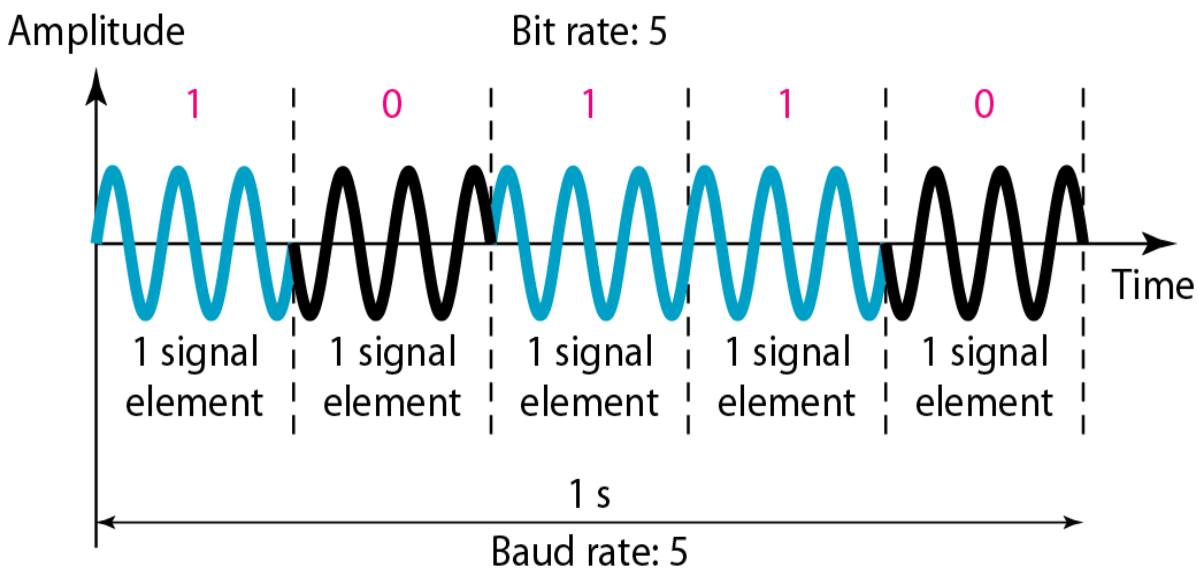
- Binary FSK:



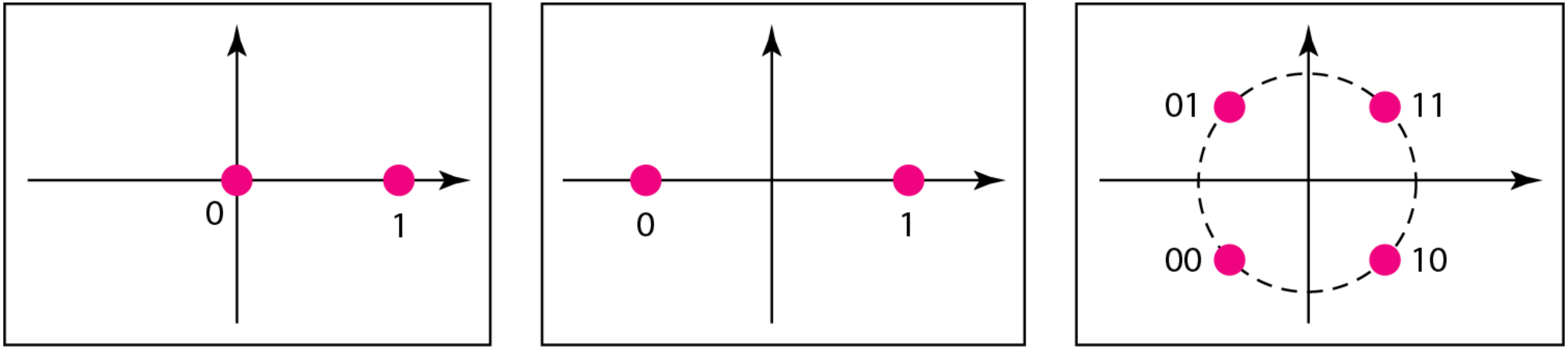
Phase Modulation

- **Phase modulation** aka **Phase Shift Keying (PSK)**

- **Binary PSK:**



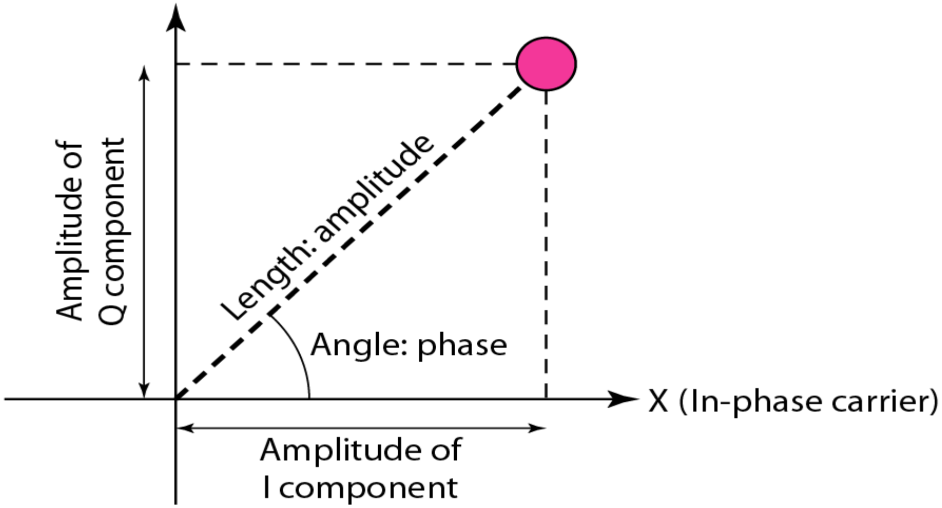
Constellation Diagrams



a. ASK

b. BPSK

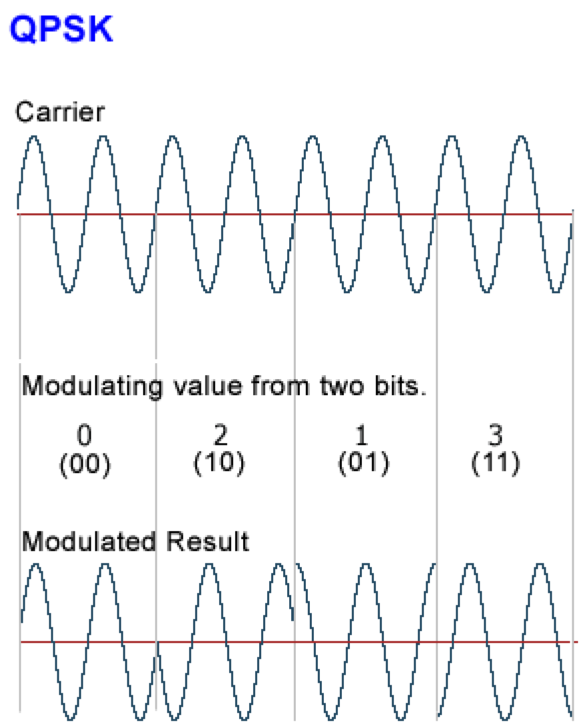
c. QPSK

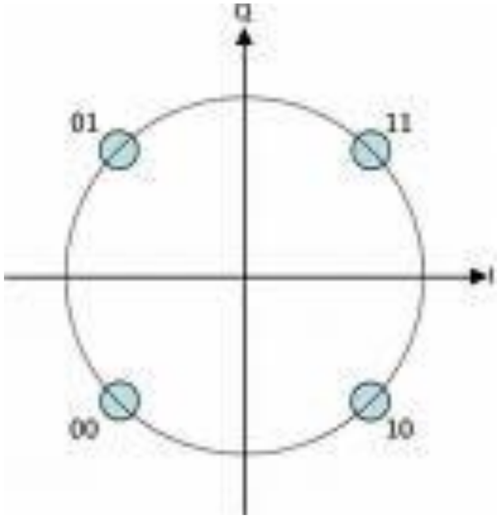


QPSK (Quadrature Phase Shift keying)

- QPSK is used in TDMA (Time Division Multiple Access) cell phones
- Four signals are required to represent 00, 01, 10, 11, in QPSK ($2^K=4$ i.e., K=2) and 4 different phases are therefore required

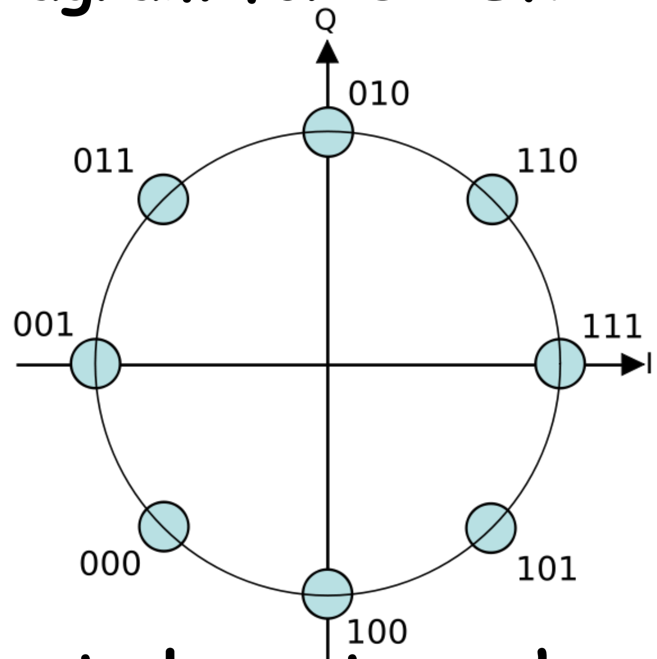
K=2 $\Rightarrow$ 2^2 signals





2^k-PSK

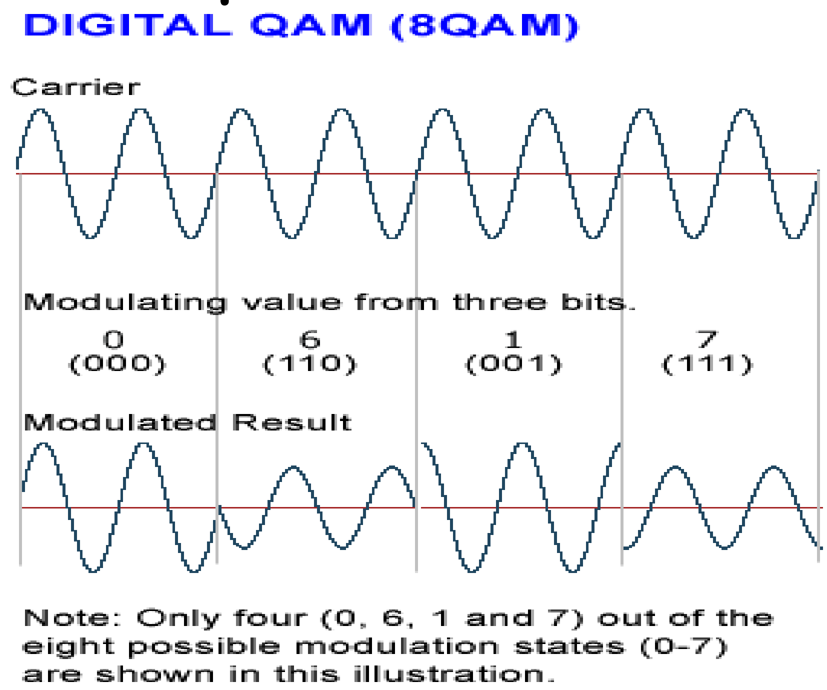
- Note: 2^k-PSK implies k bits
- Example: Show the constellation diagram for 8-PSK
- k=3



- As k increases in 2^k-PSK, phases get closer to each other and the receiver finds it more challenging to differentiate among 2^k phases while demodulation. Then the receiver is more likely to make mistakes and due to noise the bit error rate increases

Quadrature Amplitude Modulation (QAM)

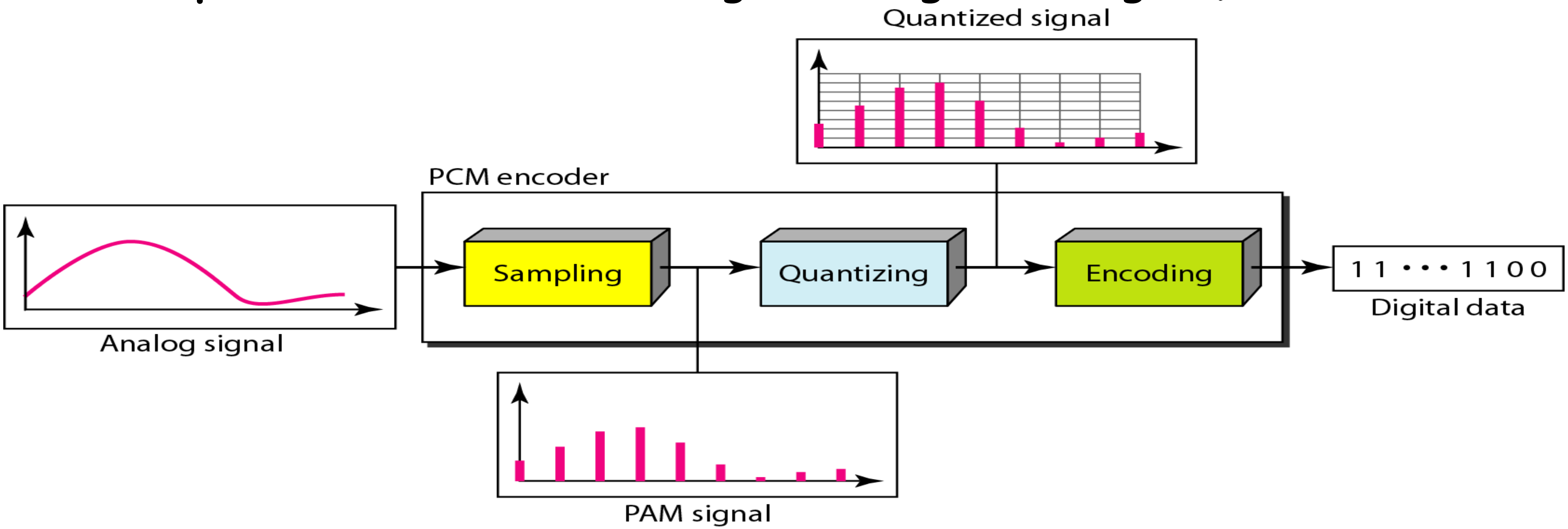
- **QAM** is a combination of ASK and PSK
- For 8-QAM, k=3: There are two amplitudes and four phases, so overall 8 different signals can exist, and therefore each can represent a unique 3-bit information signal



- Example: what is the value of k for 128-QAM?
$K=\log_2 128=7$

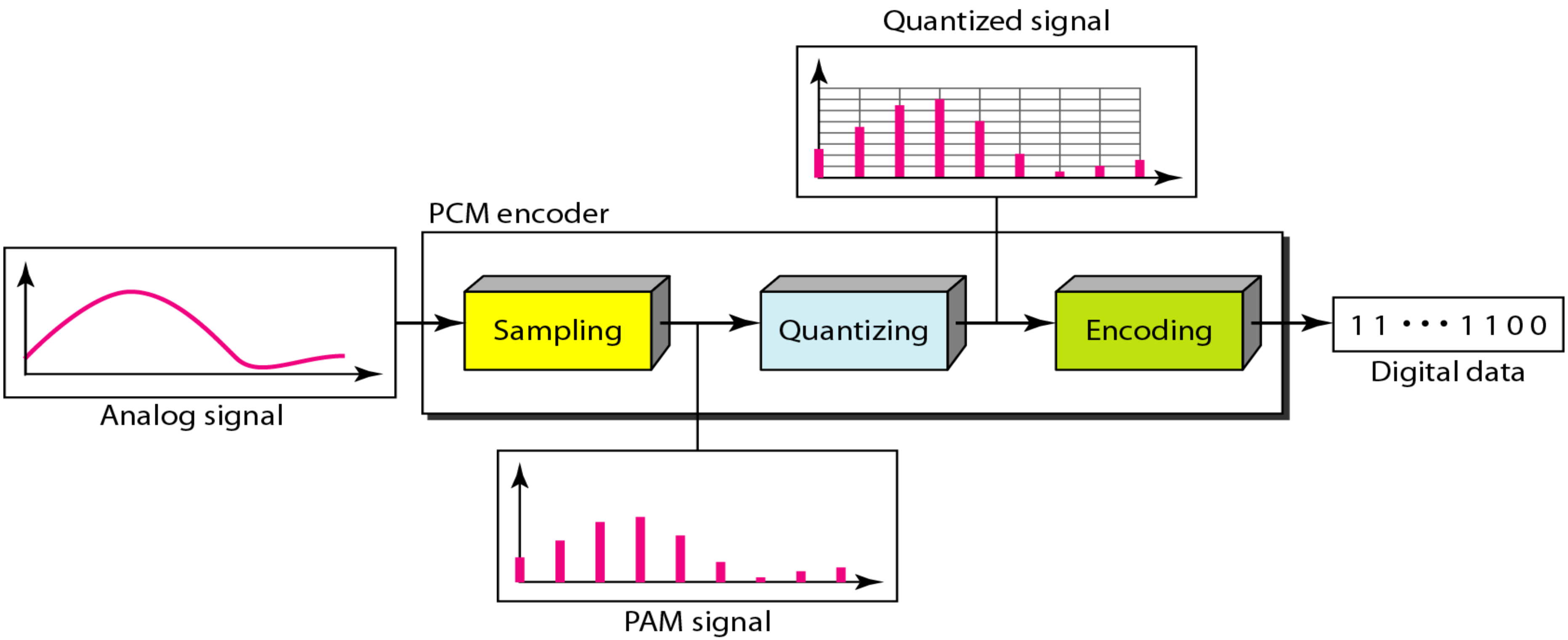
Digital Transmission – A/D

- In digital transmission, in case of analog to digital conversion, 3 steps are required: **sampling**, **quantizing**, and **encoding**
- For example, the telephone signal travels from telephone system to the CO (Central Office) and there it will be converted into digital signal to make the signal suitable for the telephone network (note that every part of the telephone network is digital and the only analog part is the telephone line from customer to CO. Also cell phones convert the analog voice signal to digital)



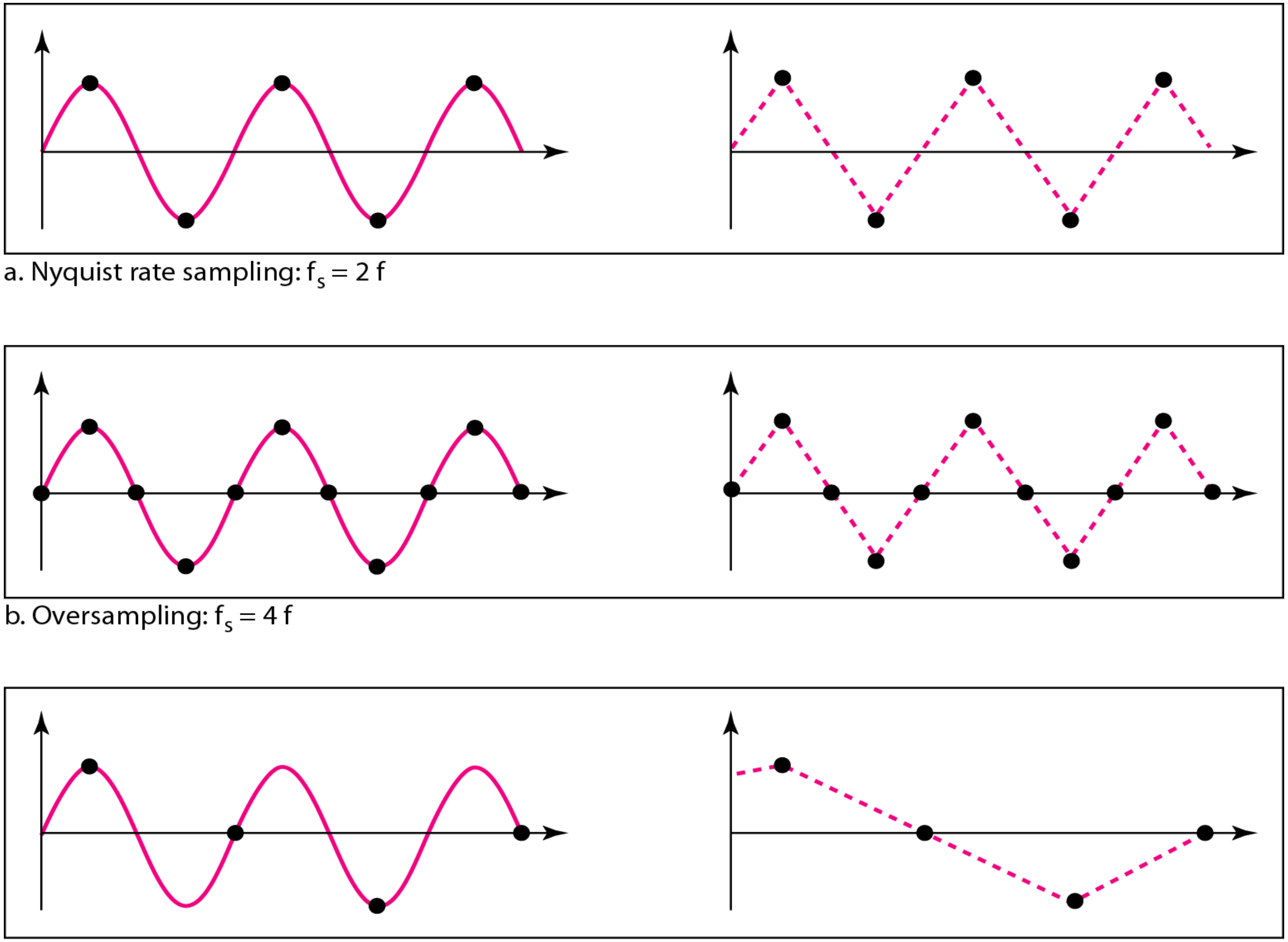
Digital Transmission – A/D (Cont.)

- **PCM (Pulse Code Modulation)** – The A/D technique that involves sampling the magnitude of analog signal and quantizing to numeric (binary) code
- **PAM (Pulse Amplitude Modulation)**



A/D – Sampling and Recovery at Receiver

- Recovery of a sampled sine wave for different sampling rates



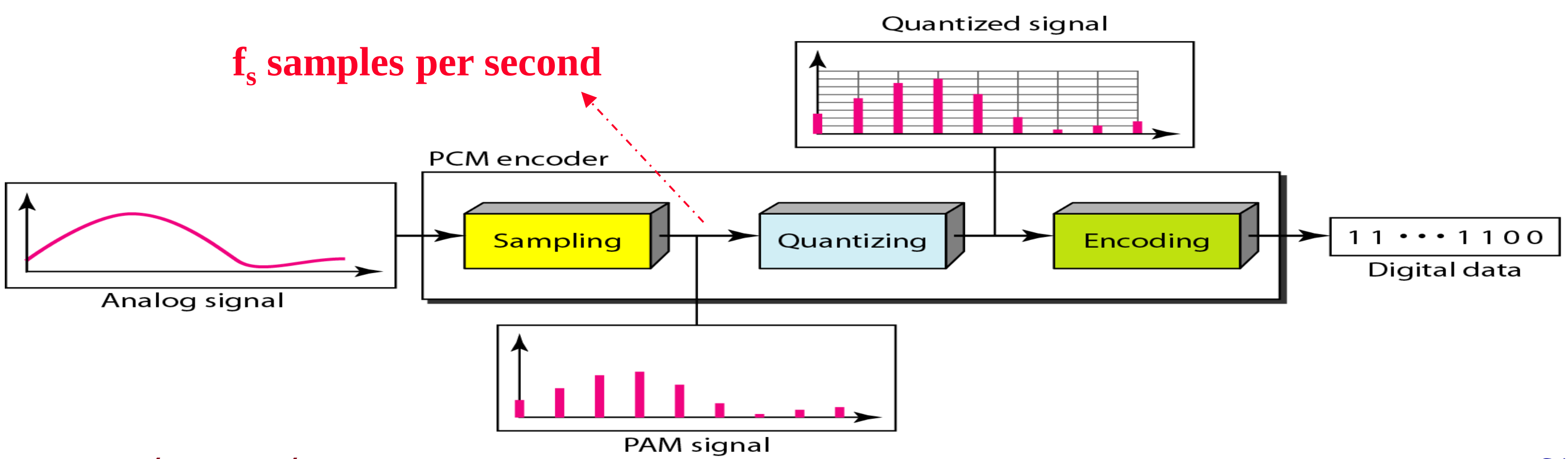
a. Nyquist rate sampling: $f_s = 2\,f$

b. Oversampling: $f_s = 4\,f$

c. Undersampling: $f_s = f$

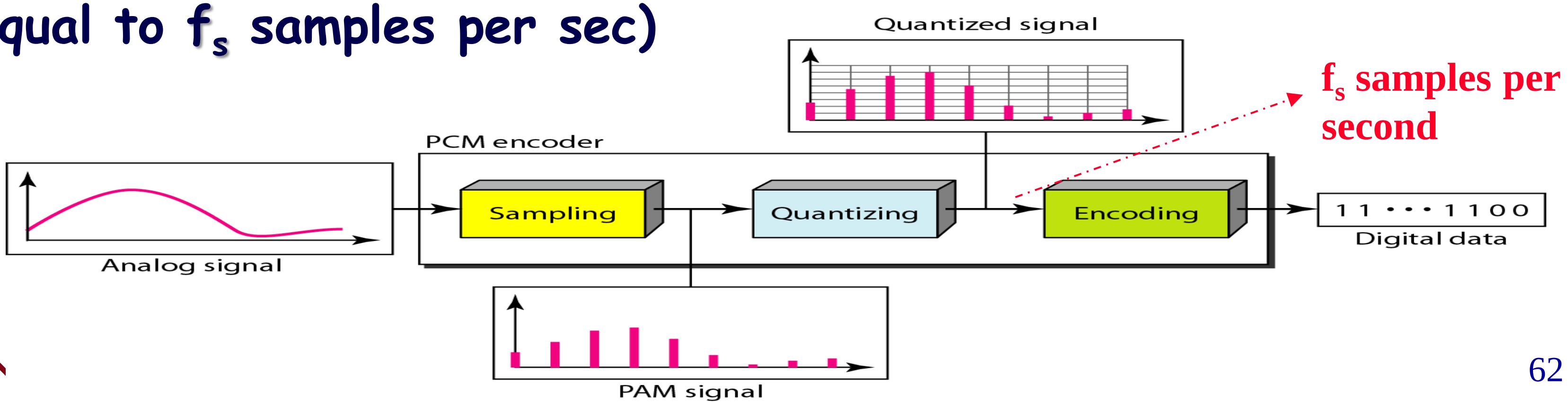
A/D – Sampling Using Nyquist Theorem

- According to the **Nyquist Theorem**, the sampling rate must be at least 2 times the highest frequency contained in the signal
- Example: Telephone companies digitize voice by assuming a maximum frequency of 4000 Hz. The sampling rate therefore is $f_s = 2 \times 4000 = 8000$ samples per second
  - Note that sampling rates higher than 8000 can also be used, but based on Nyquist theorem, 8000 samples/sec are enough
- Example: The sampling rate for video signals with max frequency of 6MHz is 12 Msamples per second



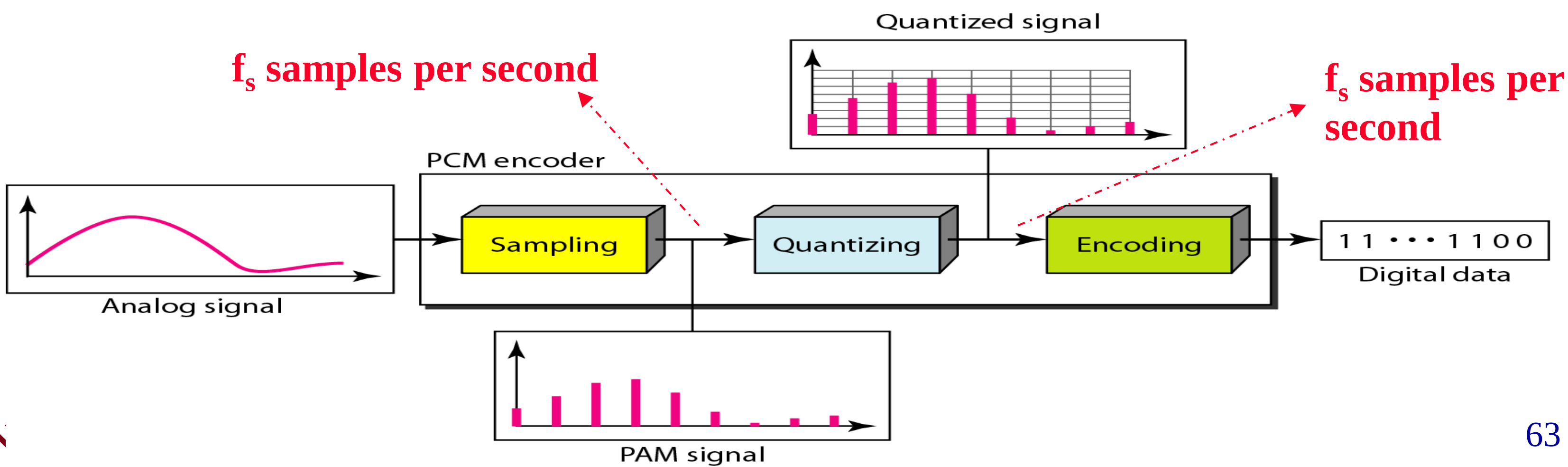
A/D – Quantization

- The sampled values are still analog with infinite number of levels, therefore the quantization step is required to have finite number of levels (we denote it by M_Q)
- Unlike sampling error that can be recovered, quantization error won't be recovered. For example For 8 quantization levels, i.e., M_Q = 8, and if the sampled values 0.912V and 0.645V are quantized into 7 and 5, respectively; later on, the receiver won't be able to recover 0.912V and 0.645V; for example, 7 and 5 could mean 0.9 and 0.6 to the receiver
- Example: In PSTN, quantization is done with M_Q=256 levels ranging from -127 to 128
- The sampling rate after quantization is the same as that before (is equal to f_s samples per sec)



A/D – Quantization and Encoding

- Therefore quantization step will not generate new samples, so the rate is going to stay at f_s however it is just going to shape the samples into M_Q specific levels
- Note: The M_Q levels are not yet digital. Encoding step will convert those quantized values into digital
- An encoder is a device that will map every one of the levels to a unique binary code (number of bits required is $Log_2 MQ$)
- Question: In PSTN quantization is done with M_Q=256 levels ranging from -127 to 128. How many bits are required to encode them into binary number? $Log_2 M_Q$ = 8 bits



A/D – Bit Rate of the Encoder Output

- Example: Bit rate of the telephone line: Remember that the maximum bandwidth of 4KHz is considered for a telephone channel; based on Nyquist Theorem we need 8000 samples/sec. Each sample is quantized into a level shown by 8 bits, so for 8000 samples per second we will have 64000bits per second. wow…! we now know why the telephone call bit rate is 64Kbps!

$$\text{Bit Rate} = f_s \times \text{Log}_2 M_Q$$

- Remember that this is the bit rate of the digital signal that was resulted from sampling, quantizing and encoding your voice call in the CO. This digital signal with this bit rate will reach the CO connected to the call receiver and converted back to analog electrical signal and sent to the call receiver. When callee hears the voice, she/he can recognize the caller's voice because based on Nyquist Thm there were enough samples for a BW of 4KHz (i.e., 8Ksamples/sec) and M_Q was chosen big enough (i.e., 8bits/sample) so the voice quality did not degrade much

A/D – Encoding (Cont.)

- Let's assume we have considered f_s big enough (e.g., 8000 for the telephone line) but we quantize the sampled signal into only two levels, i.e., $M_Q = 2$, we then would not need an encoder, because we can use 0 and 1 to represent those two levels
- However this will result in high errors at the receiver side. We can look at this noise as the quantization noise and based on Shannon's Theorem $C = B\log_2(1+SNR)$ this would reduce the capacity of the system. In other words for a bit rate of 64Kpbs, we need to make sure the channel capacity is enough. We can consider the quantization error in SNR and ensure the error is low enough such that C stays big enough for 64kbps

A/D – Compressing

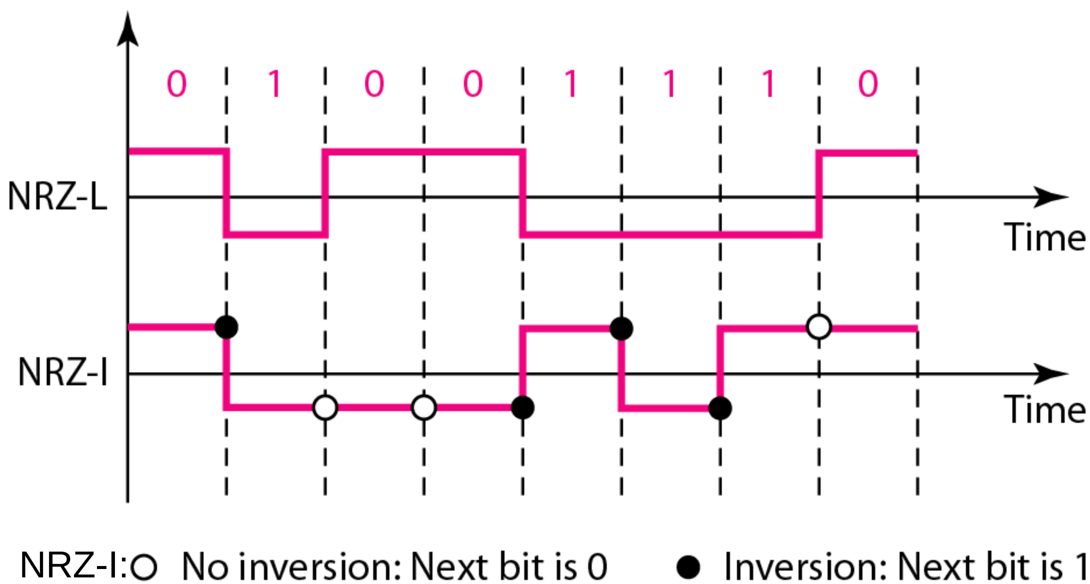
- If one considers a lot of quantization levels to avoid high quantization errors, that adds into the complexity of the system, so there is a tradeoff
- Example: Bit rate of the cell phone: Since uses radio signal its A/D includes a **compressor** (after sampler, quantizer, and encoder) to compress the encoded digital signal to 8kbps. This way the bandwidth required is 1/8 of the bandwidth required on a wired telephone line in PSTN. Of course the quality is lower than that of the wired telephone line
- Optional: http://searchunifiedcommunications.techtarget.com/tip/How-voice-compression-saves-bandwidth

Digital Transmission – D/D

- It may be the case that the digital data that is created by a system is not suitable for transmission (channel or medium) therefore it needs to be converted into another digital signal, e.g., a PC generates binary data consisting of 0s and 1s, however for Ethernet, -1 and +1 are suitable
- The process of converting a digital signal into another digital signal is called **Line Coding**
- There are many line coding techniques, such as **NRZ-L** (Non-Return-to-Zero,) **NRZ-I** (Non-Return-to-Zero-Inverted,) **Manchester**, etc.

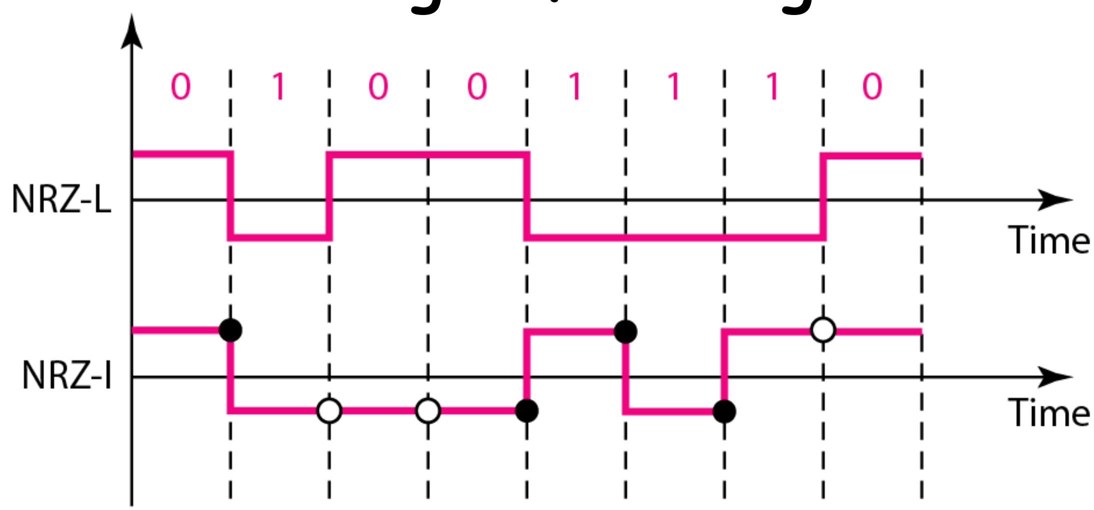
NRZ

- In NRZ-L the level of the voltage determines the value of the bit, e.g., a positive voltage for bit 0, a negative voltage for bit 1. the bit to voltage mapping should be known by the receiver too, to be able to decode
- Note: During bit duration there is no transition



NRZ (Cont.)

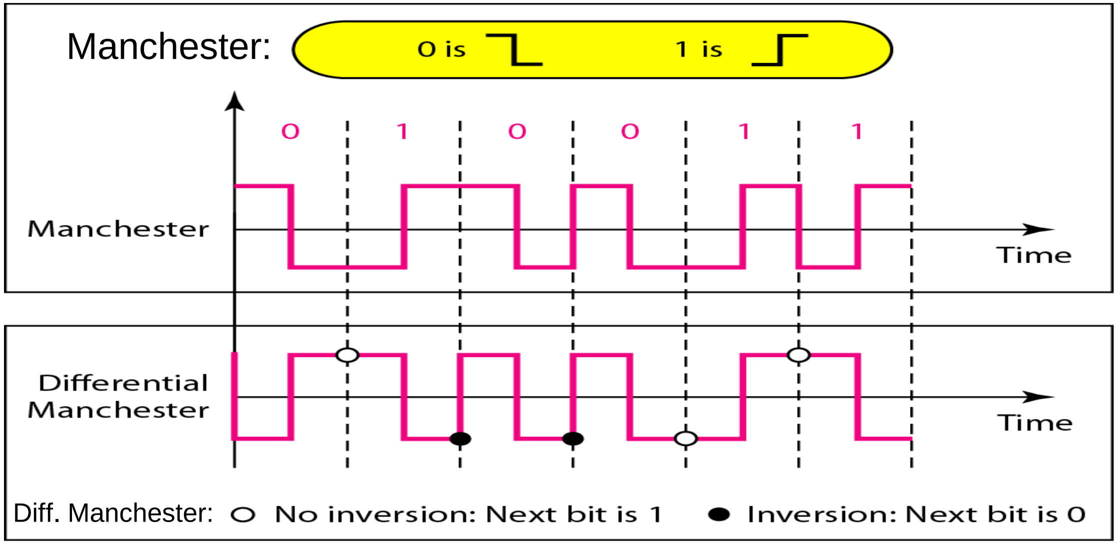
- In NRZ-I the inversion or the lack of inversion determines the value of the bit
- NRZ weakness: Receiver side would have a synchronization problem for long strings of 0s or 1s in the coded signal (e.g., in NRZ-L when the original message has long strings of 0s or 1s)
- NRZ also suffers from nonzero DC component problem. DC component is the average of the signal



 NRZ-I: ○ No inversion: Next bit is 0 ● Inversion: Next bit is 1

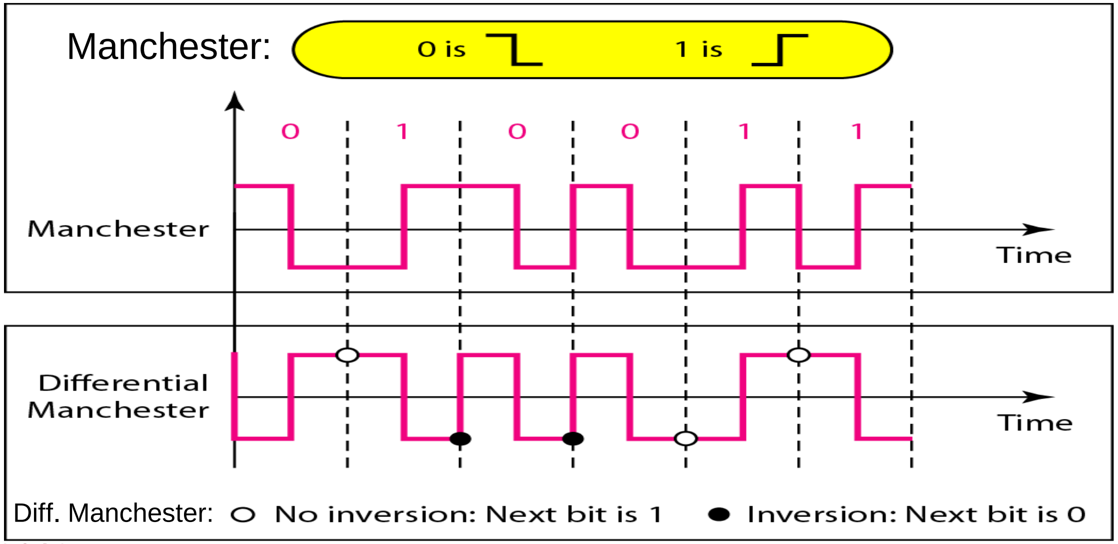
Manchester/Differential Manchester

- In **Manchester** and **Differential Manchester** encoding, the DC component is zero
- Also the transition at the middle of the bit is used for synchronization, which guarantees transitions even for long strings of 0s or 1s, so the receiver finds it more unfussy to synchronize itself with the sender



Manchester/Differential Manchester

- In Manchester every 0 is represented by half positive half negative and every 1 by half negative half positive
- In differential Manchester, a transition happens at the beginning of the bit period for bit 0, no transition for bit 1. Also there is always a transition in the middle of the bit period which is used for clocking
- The disadvantage of Manchester/Differential Manchester is that bandwidth of Manchester code is two times that of the NRZ



Just a Reminder: An example on Clock Synchronization

- In a digital transmission, the receiver clock is 0.1 percent faster than the sender clock. How many extra bits per second does the receiver receive if the data rate is 1kbps?
  - At 1kbps, the receiver receives 1001bps instead of 1000bps

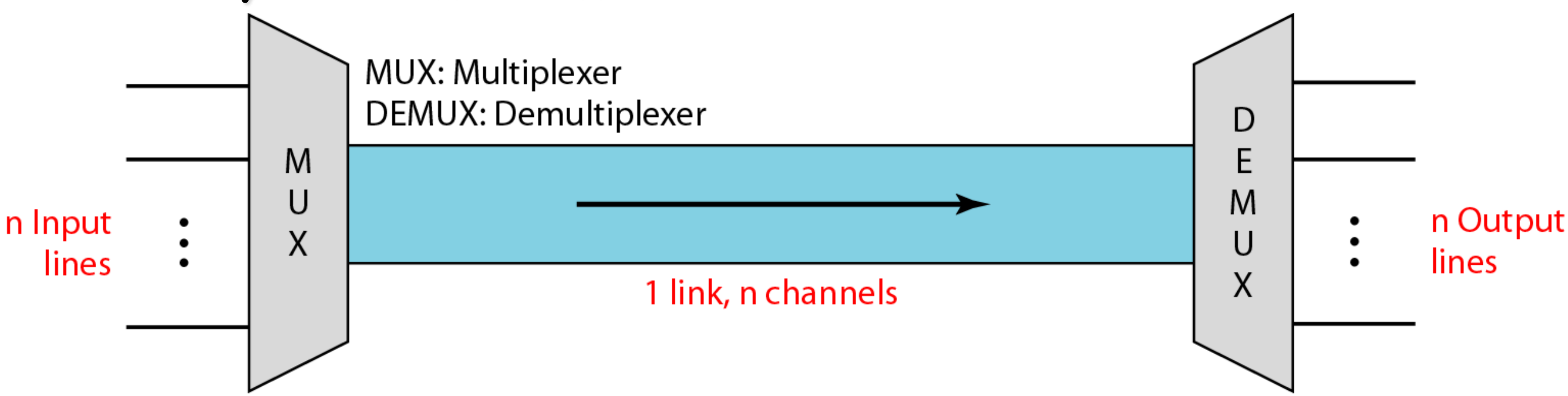
1000 bits sent	1001 bits received	1 extra bps

- How many extra if the data rate is 1 Mbps?
  - At 1 Mbps, the receiver receives 1,001,000bps instead of 1,000,000bps

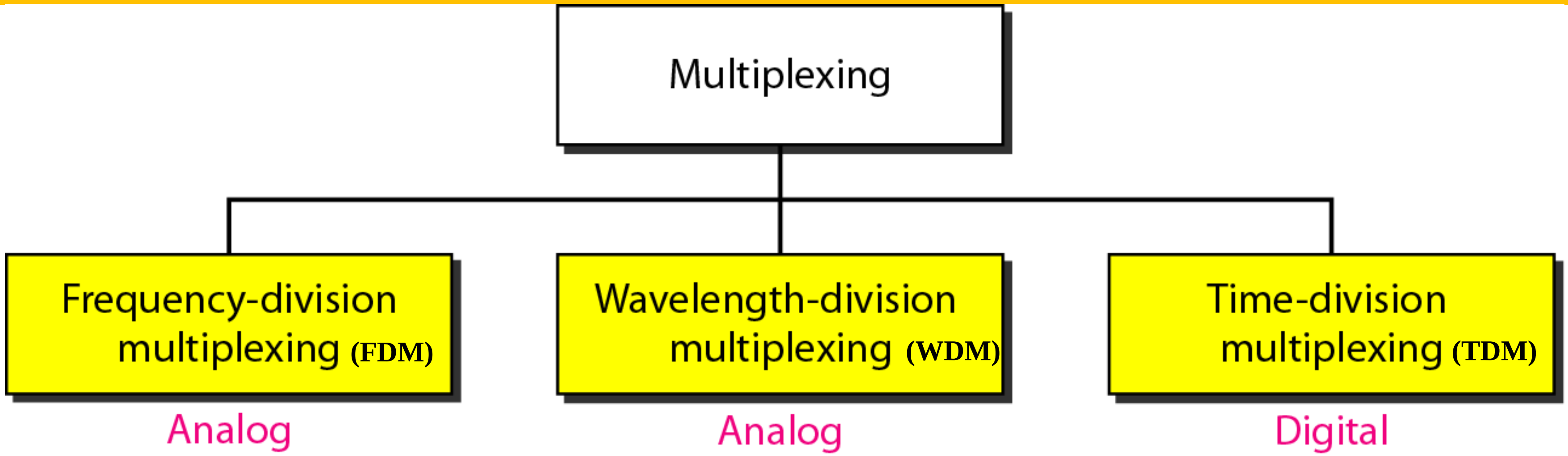
1,000,000 bits sent	1,001,000 bits received	1000 extra bps

Multiplexing (Bandwidth Utilization)

- **Multiplexing** (aka **resource sharing**) is the set of techniques that allows the simultaneous transmission of multiple signals across a single data link. Therefore multiplexing results in efficiency
- Bandwidth utilization is the wise use of available bandwidth to achieve specific goals
- Whenever the bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices, the link can be shared



Multiplexing Classification



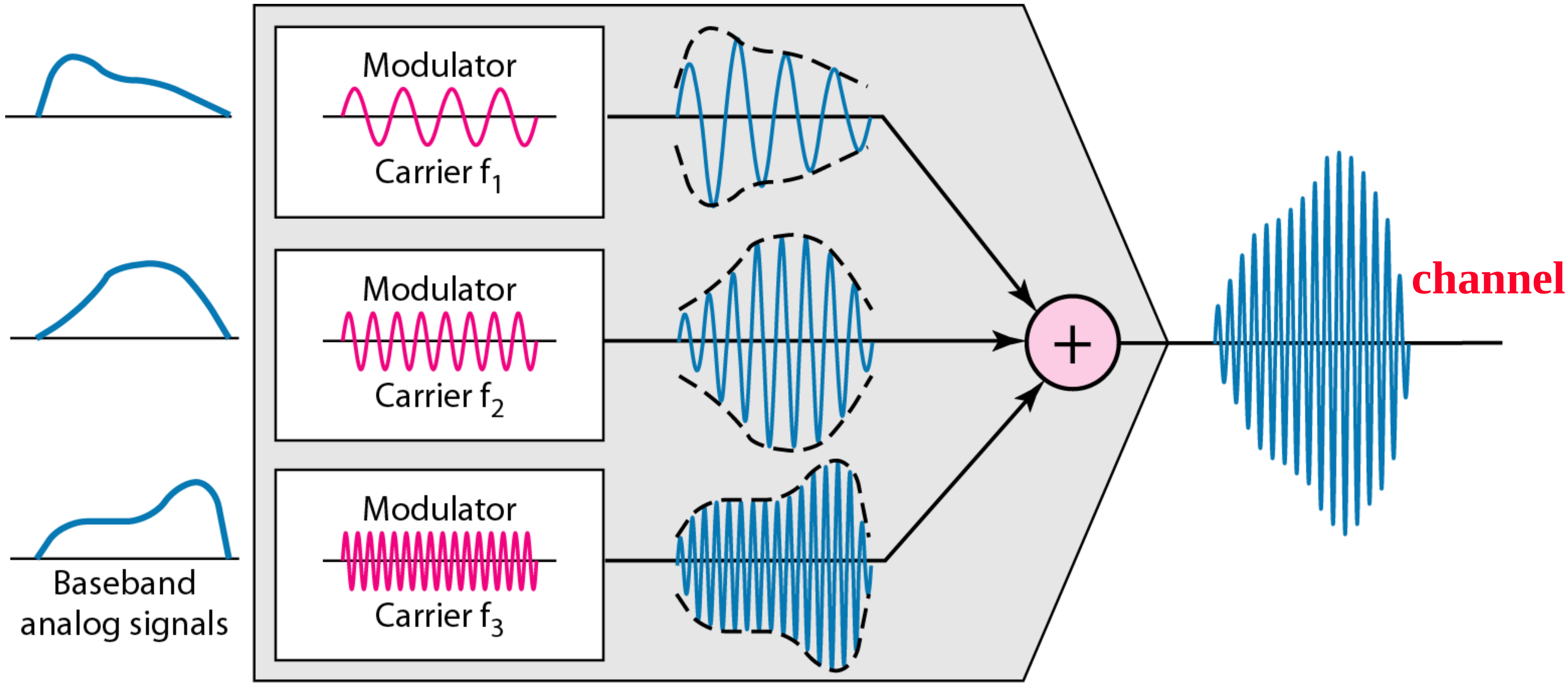
Notes:

- TDM is further classified into **Synchronous TDM** which is used in PSTN or **Statistical TDM** aka (**Asynchronous TDM**) which is used in the Internet
- In **WDM**, wavelength (denoted by λ) is equal to c/f where c is the speed of light and f is the frequency

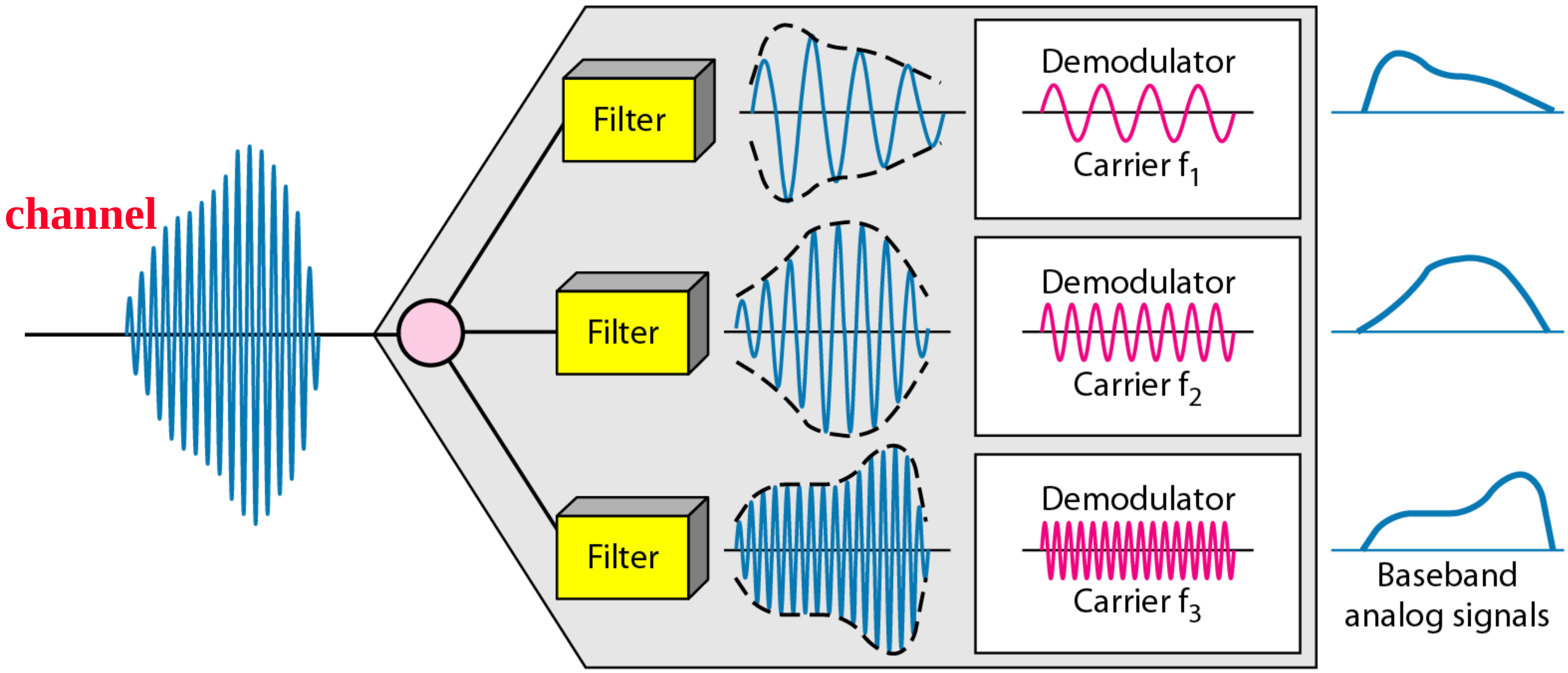
FDM (Frequency Division Multiplexing)

- Each signal is modulated with a different carrier frequency
- Each carrier frequency represents the channel
- FDM is used in Radio both FM and AM, Cable TV, ADSL, cable access and cell phones

FDM Multiplexing Process

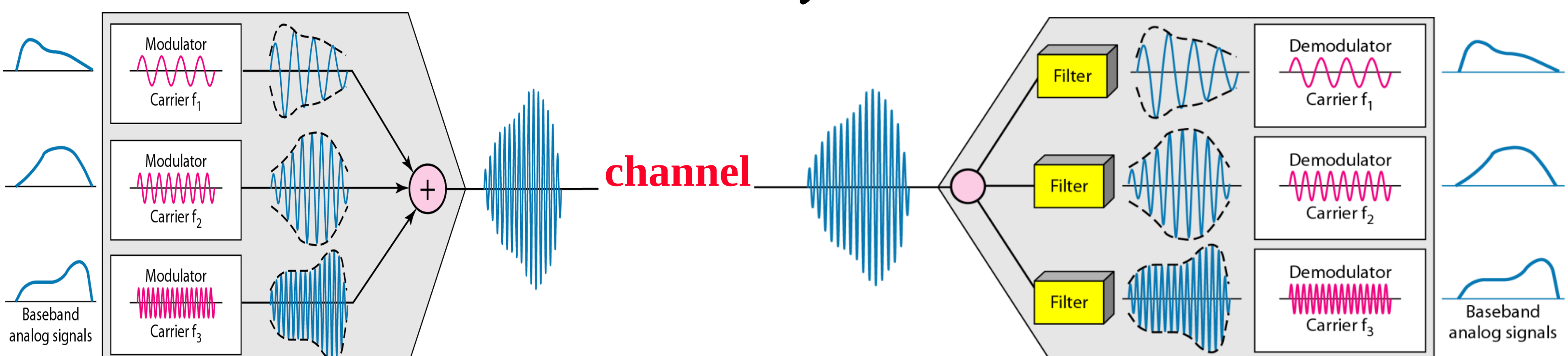


FDM Demultiplexing Process



FDM (Cont.)

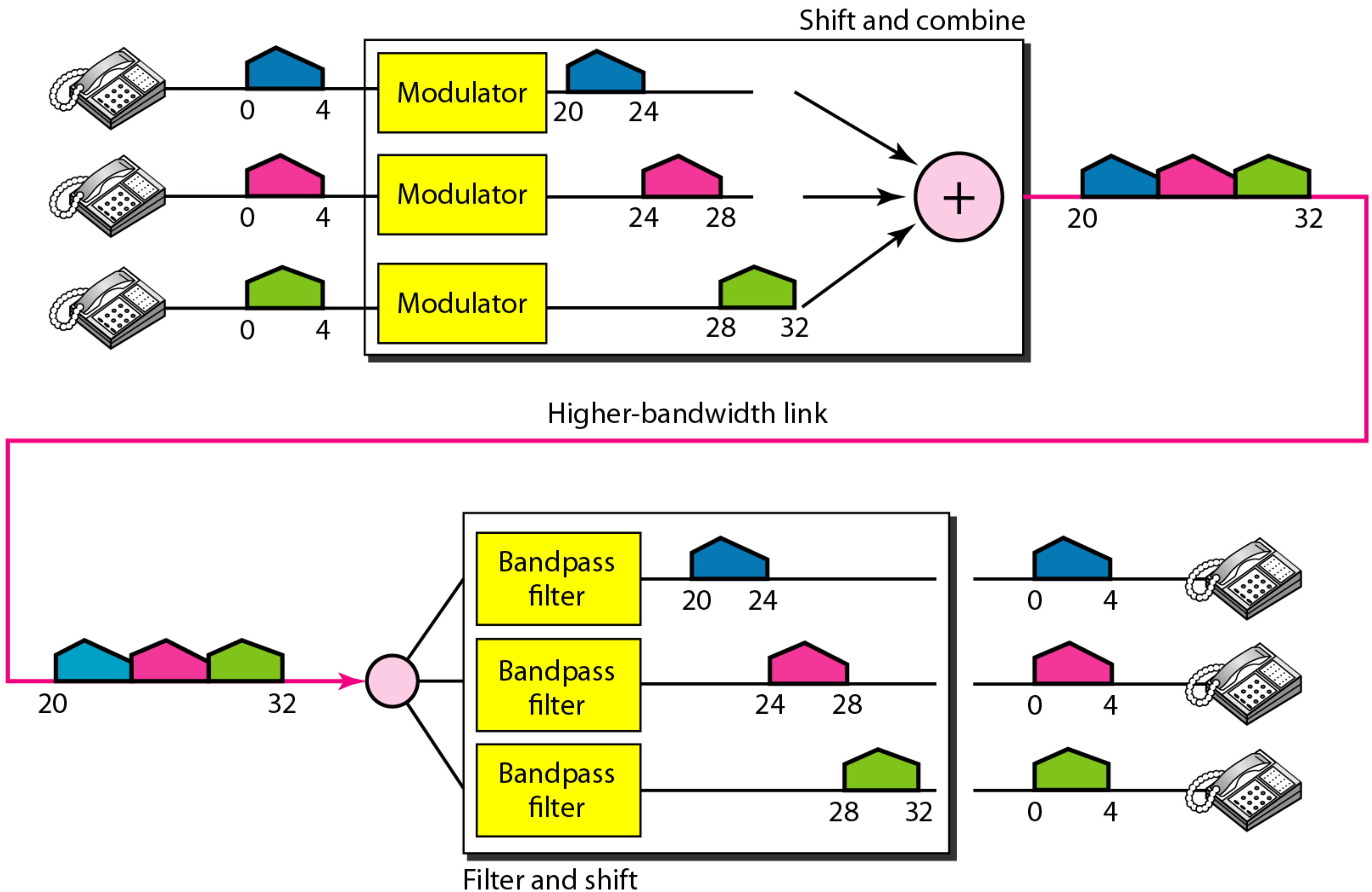
- Note: the adder is not a real adder, e.g., in Radio, the signals from different channels are added in the air. If we had not modulated the channels, they would add in the same frequency range and later on in the receiver we could not separate the channels
- But now that we have modulated them with different carriers, at the receiver we can use filters to tune to each channel (in reality we have one tunable filter instead of multiple filters, e.g., Radio tuner or the remote controller of the TV)



FDM – Example I

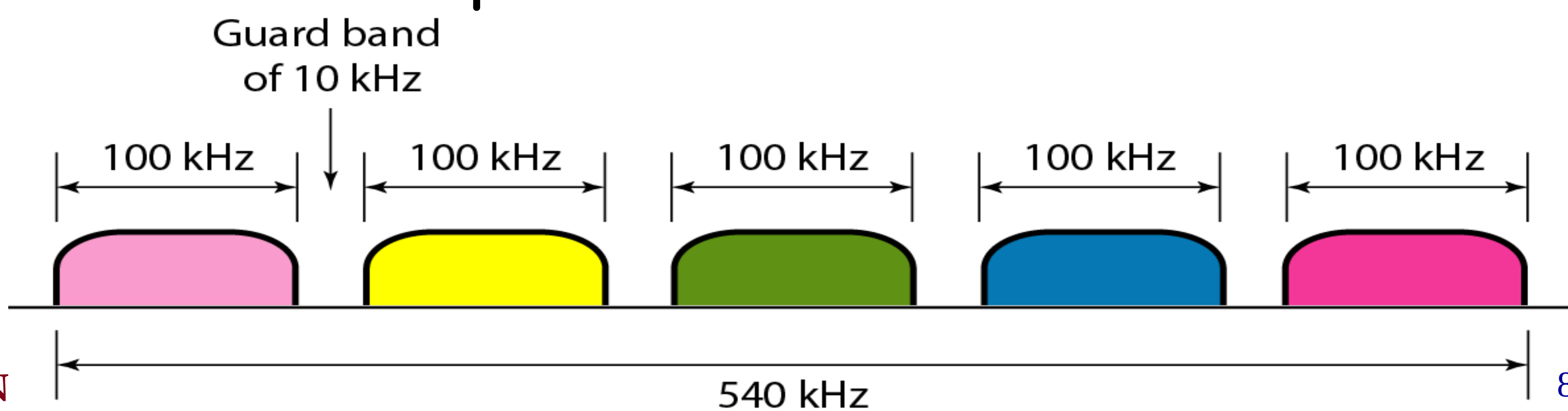
- Assume that a voice channel occupies a bandwidth of 4kHz. We need to combine three voice channels into a link with a bandwidth of 12kHz, from 20 to 32kHz. Show the configuration, using the frequency domain. Assume there are no guard bands
  - We shift (modulate) each of the three voice channels to 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 (It is illustrated on the following page

FDM – Example I (Cont.)



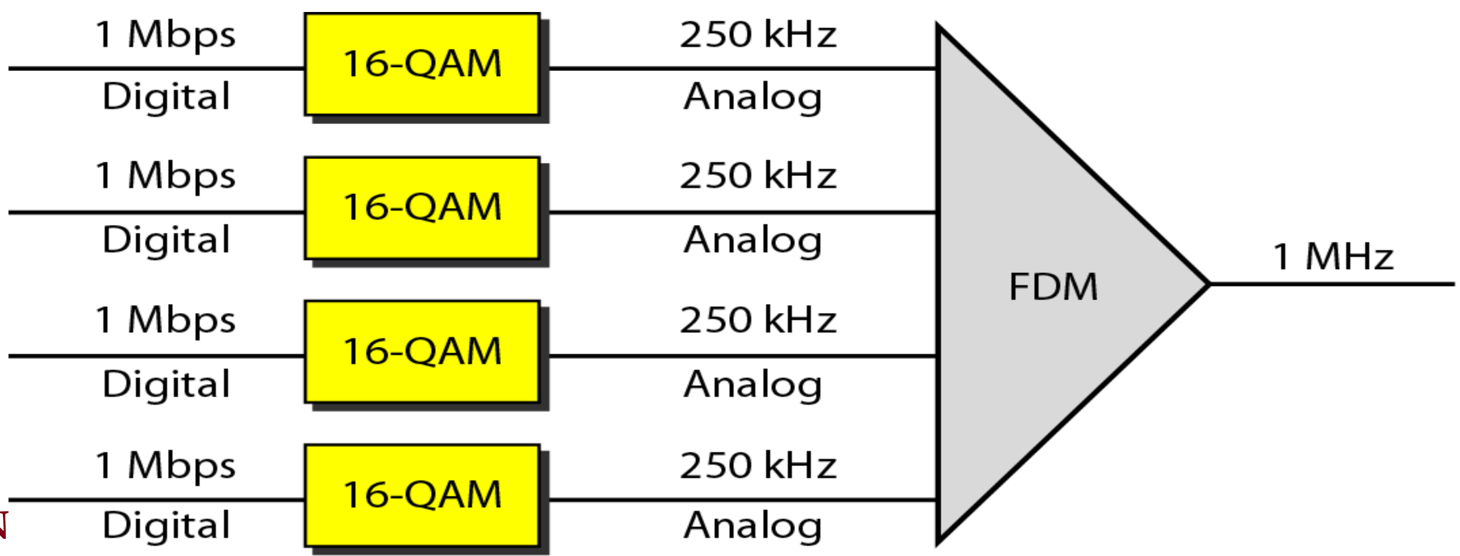
FDM – Example II

- Usually a guard band is considered between the neighboring channels in the frequency domain
- Example: Five channels, each with a 100kHz bandwidth, are to be multiplexed together. What is the minimum bandwidth of the link if there is a need for a guard band of 10kHz between the channels to prevent interference?
  - For five channels, we need at least four guard bands, i.e., the required bandwidth is at least $5 \times 100 + 4 \times 10 = 540$ kHz
- Note: In reality and even with the guard bands, when we tune a radio, we may hear another radio in the background, the reason is that in real life the frequency domain shape is not square and still there is an overlap



FDM – Example III

- Four data channels (digital), each transmitting at 1Mbps, use a satellite channel of 1MHz. Design an appropriate configuration, using QAM for D/A modulation and then resource sharing with a 4 channel FDM
  - The satellite channel is analog. For 4 channel FDM, each channel has a 0.25MHz bandwidth. Each digital channel is 1 Mbps so we need to modulate each digital channel to get a 250KBd=250KHz



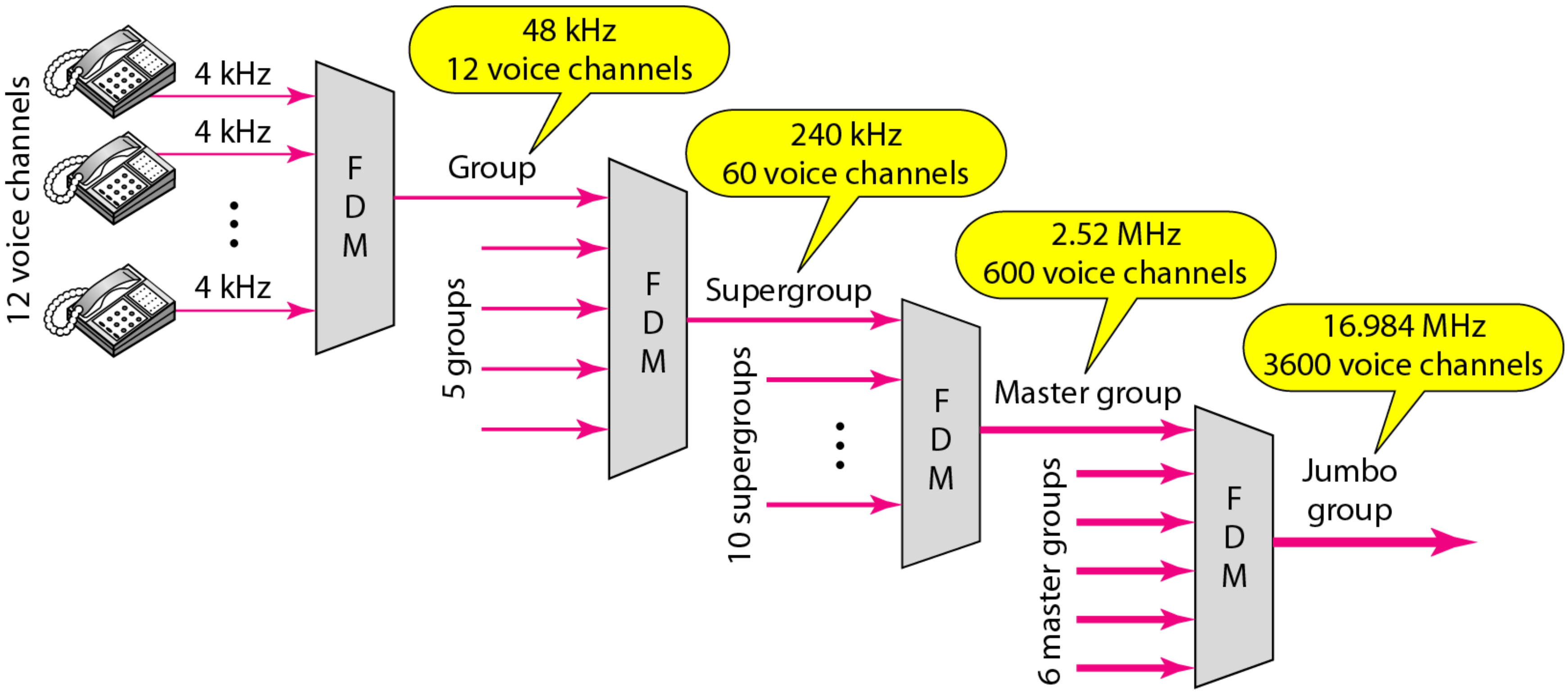
FDM – Example IV – AM Radio

- We talked about nonperiodic composite signals before. An example of a nonperiodic composite signal is the signal propagated by an AM radio station. The total bandwidth dedicated to AM radio ranges from 530KHz to 1700KHz (about 1.17MHz total bandwidth) and they are assigned by FCC (Federal Communication Commission) to AM broadcasters. Looking at the range of 530 to 1700KHz, there are about 100 channels that are broadcasting (transmitting) simultaneously, i.e., they overlap in time
- Let's focus on a radio station, e.g., 670AM. 670KHz is the carrier frequency and the channel's bandwidth is 10KHz. How are the stations distinguished at the receiver side (the Radio)? Since they do not overlap in frequency domain, a filter can be used to pass the certain bandwidth corresponding to that radio channel, and then the signal will be demodulated to receive the baseband signal. It is then converted into sound signal

FDM – Example V – FM Radio

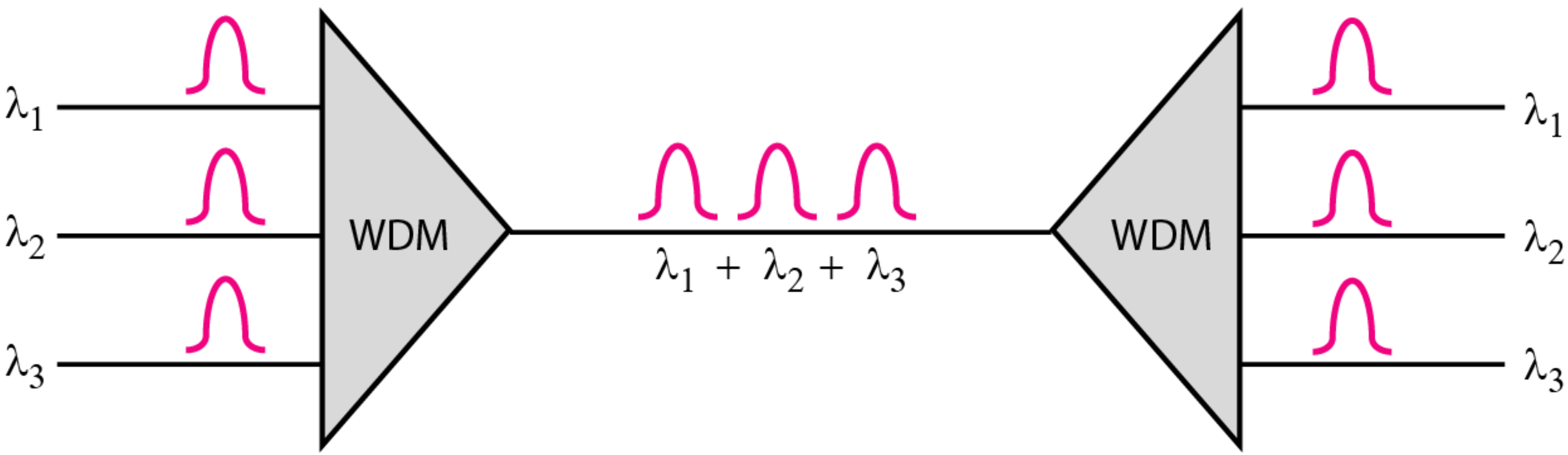
- Another example of a nonperiodic composite signal is the signal propagated by an FM radio station. The total bandwidth dedicated to FM radio ranges from 88 to 108 MHz (about 20MHz total bandwidth.) Therefore there are about 100 channels that are broadcasting (transmitting) simultaneously (i.e., they overlap in time)
- Note: AM and FM radio channel are region based, so in each region a maximum of 100 AM broadcasters and 100 FM broadcasters exist. The same channel can be reused by another broadcaster as far as its transmitter is far enough from other broadcaster(s) with the same carrier frequency
- The signal will lose its strength (amplitude degrades,) when goes through an RF medium

FDM – Analog Hierarchy



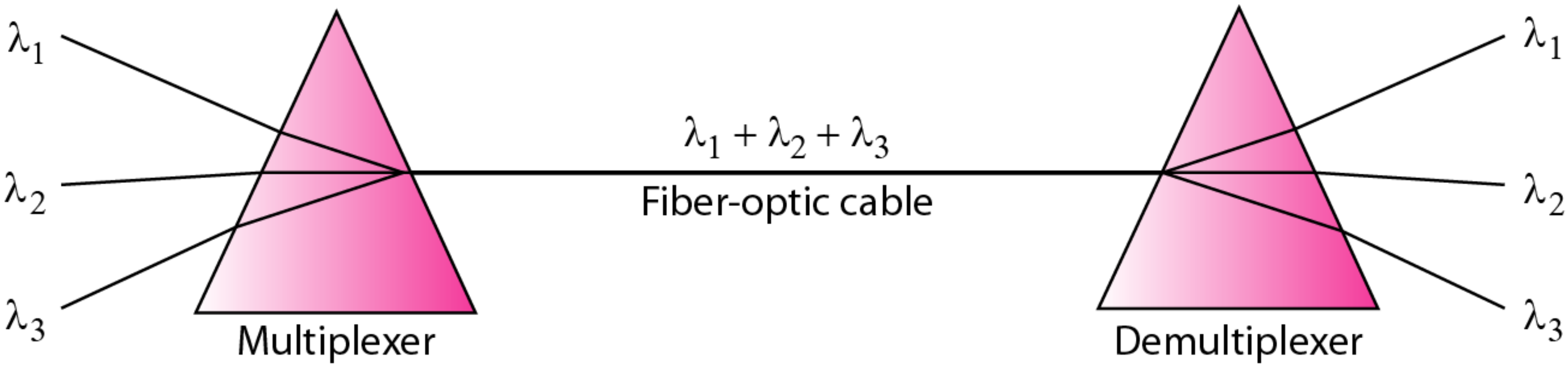
WDM (Wavelength Division Multiplexing)

- WDM is an analog multiplexing technique to combine optical signals
- WDM is the optical version of FDM
- Wavelength λ is equal to **c/f** where **c** is the speed of light and **f** is the frequency
- Examples: 16 wavelengths at OC-48. 32 wavelengths at OC-192



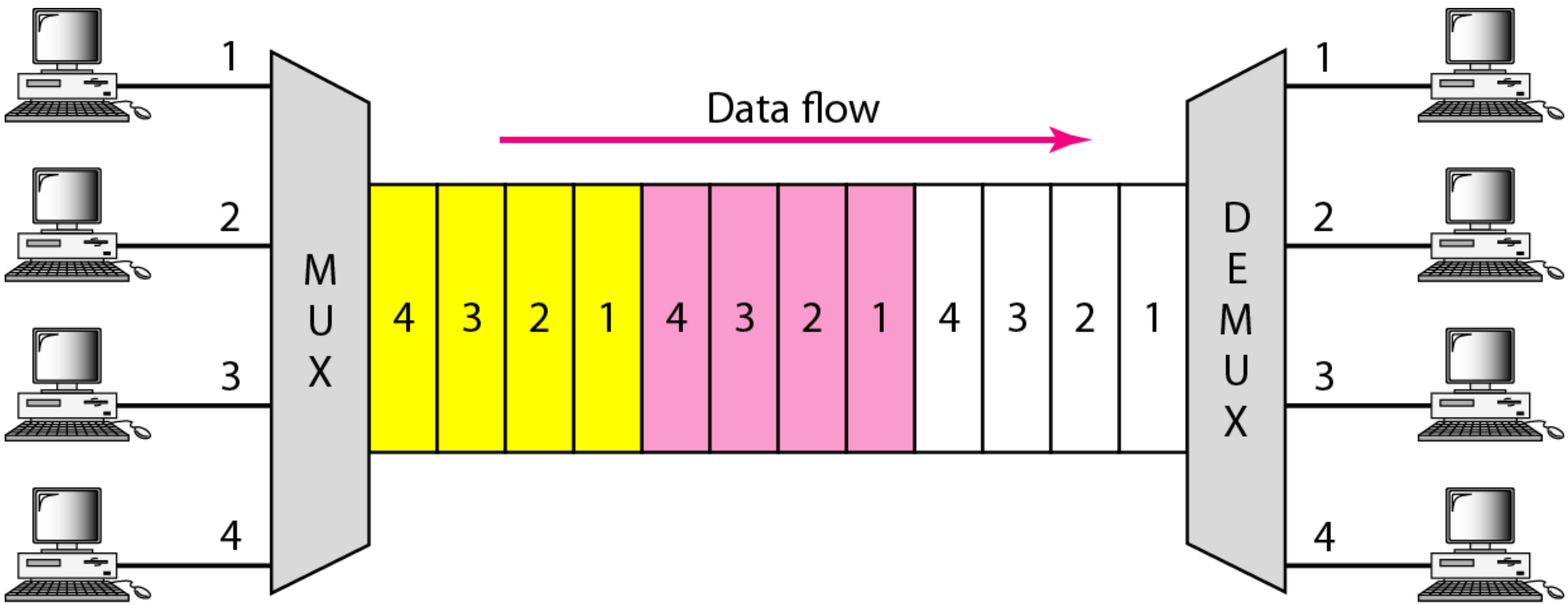
WDM (Cont.)

- **Multiplexing and demultiplexing are used**



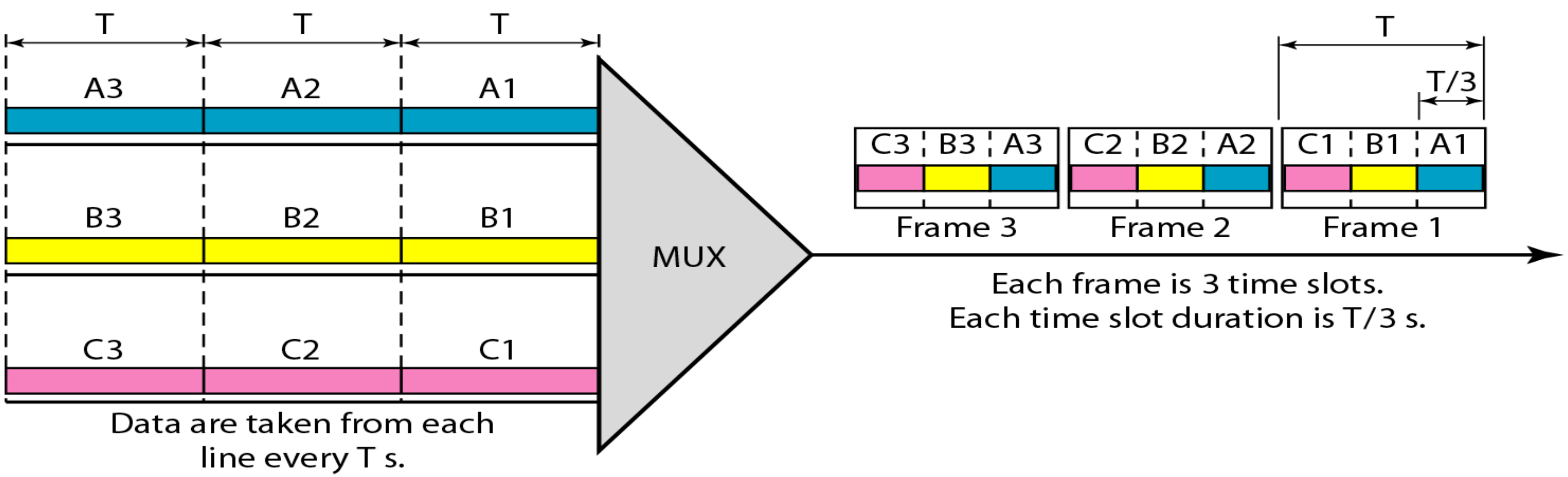
TDM

- **TDM** is a digital **multiplexing technique** for combining several low-rate channels into one high-rate one
- The main channel is shared in the time domain, i.e., time is divided into non-interleaving time slots (remember that in FDM the main channel was shared in the frequency domain)
- TDM is digital so if the information is analog, it needs to be converted to digital before TDM
- Note that channels overlap in frequency but interleaved in time



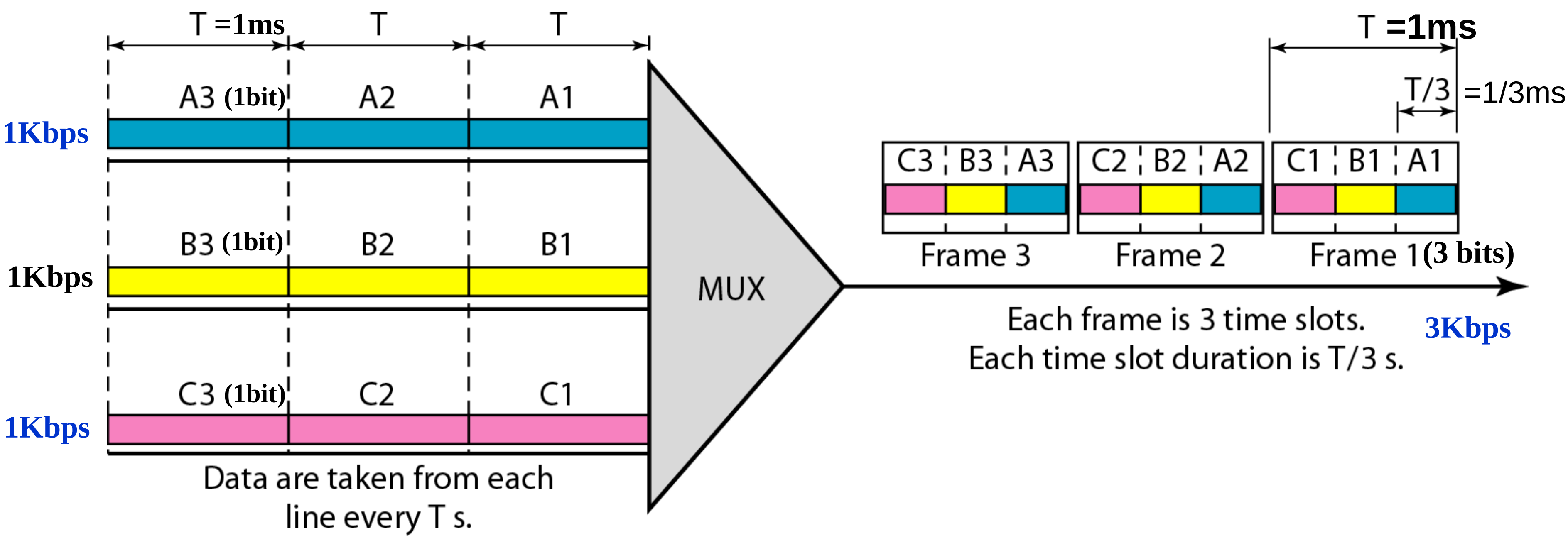
TDM (Cont.)

- Each input has a buffer (in reality inside the Mux) and writes its generated data into its buffer (slowly). Then Mux will visit and read the data from the buffer faster than the input writing to it
- E.g., calls are occurring simultaneously, but each is digitized and buffered and then read into a time slot with a fast rate, but during our call using a cell phone we don't feel that our voice channel is read fast and is time interleaved with other calls



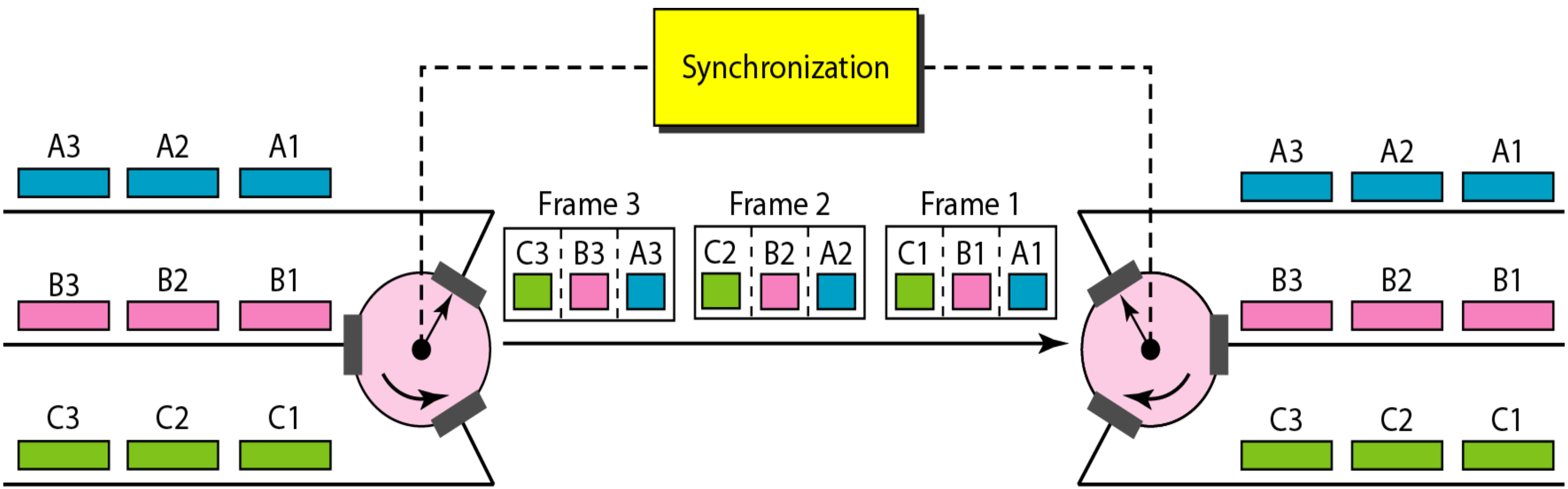
TDM (Cont.)

- Multiplexing may happen at bit level or block of bits
- Example: The data rate of each input is 1Kbps. 1 bit is multiplexed at a time (i.e., multiplexing unit is one bit.) Using the TDM multiplexer it picks up one bit from A, one from B, and one from C



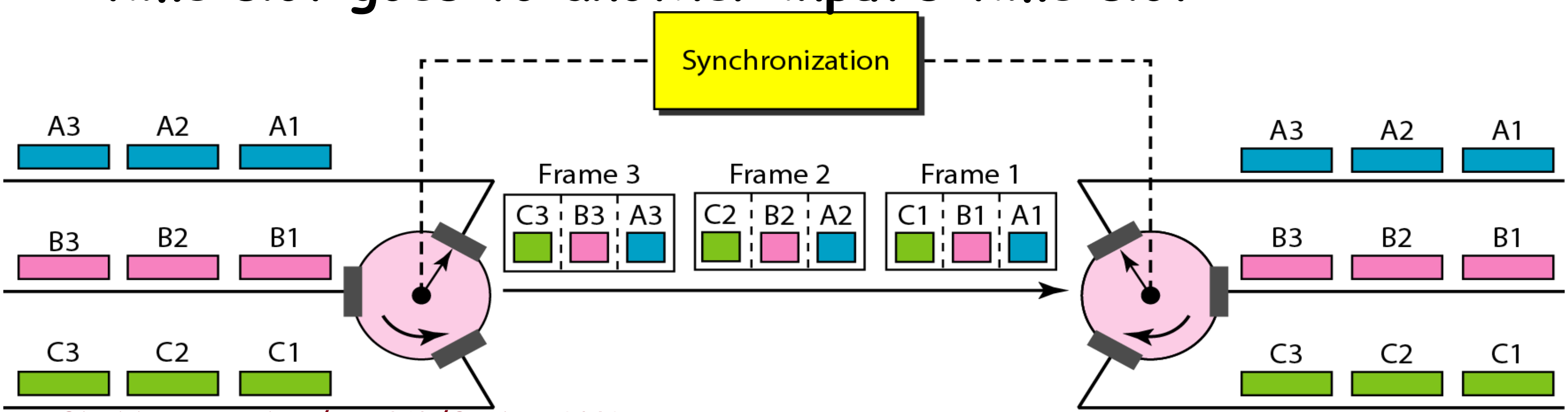
TDM Multiplexer is Like a Rotating Switch

- TDM multiplexer is like a rotating switch that in first time slot is at A, so the unit data from A goes into main channel
- When the time slot of A expires, the switch rotates to B and puts the unit data from B into the main channel during the next time slot
- Note that channel A has the entire bandwidth during the time slot that is given to A. The cases for B and C are similar



TDM Multiplexer – a Rotating Switch (Cont.)

- While the inputs A, B, and C are generating data simultaneously, the transmission of data is interleaved in time, however the transmission bit rate, R_{Mux} (bit rate of the channel) is faster [in fact : $R_{Mux} \geq \Sigma_i R_i$ where R_i is the rate of input i]
- Whenever we use a Mux we need to use a Demux on the other side. The Mux and Demux need to be synchronized, otherwise the content of some input's time slot goes to another input's time slot

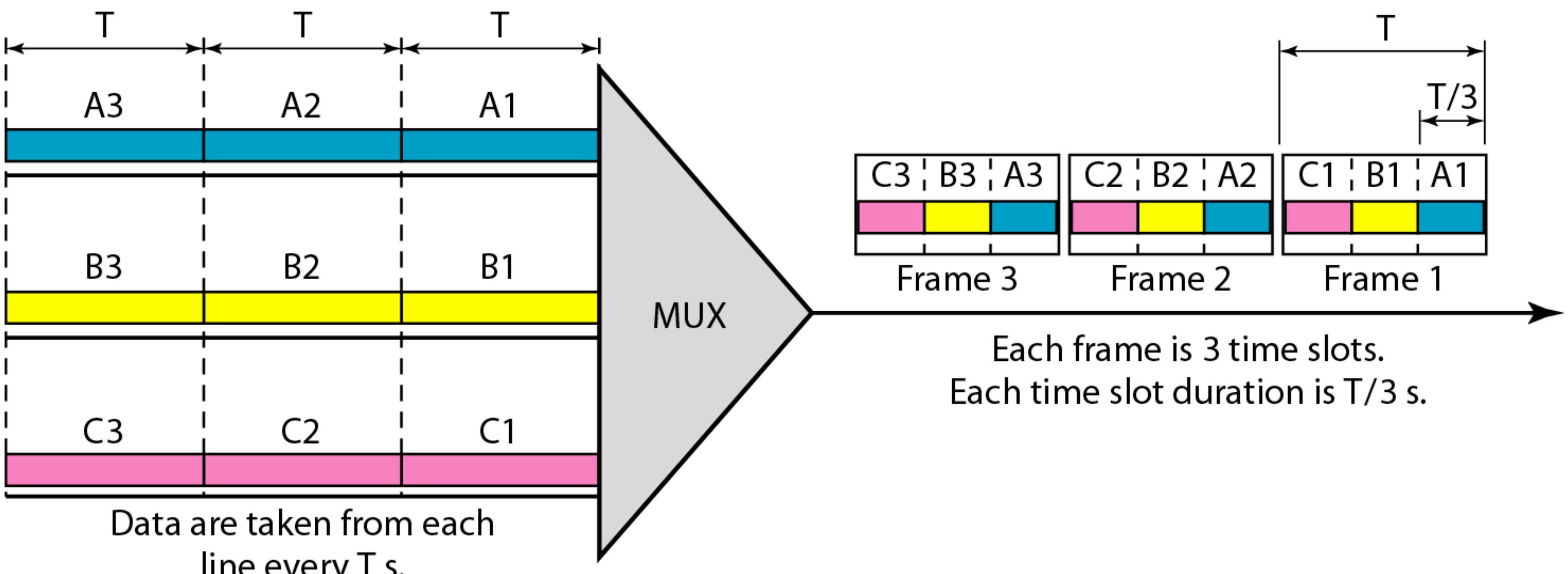


TDM Example – Grading Your Exam Papers!

- Suppose I plan to grade your exam (which consists of multiple choice questions) real time. Each paper has one question. As soon as you are done with a question you hand in its paper for grading. Assume my grading speed is 60 times as fast as your answering speed. Therefore while you are answering Q#2 I am able to grade all your Q#1 papers. This means the time I spent on each paper was $1/60^{th}$ of the time each of you spent to answer Q#1. I am allocating $1/60^{th}$ of my grading time to each paper, but I dedicate all my grading power to your paper, the time I am grading it (resembles time interleave)
- If I wanted to have a grading system that works based on the idea of FDM, I would've asked 59 other faculty members to be join me (a trunk of 60 faculties ☺) and each of us would grade one Q#1 paper, with our rate of grading the same as your rate of answering. The trunk of faculty allocates $1/60^{th}$ of its grading power (bandwidth) to each paper, but that power is dedicated to the paper the whole time (resembles frequency interleaved, time overlapped)
- In both cases the grading would be finished at the same time. For the first case time was divided (like TDM) and for 2^{nd} case faculty was divided (like FDM that frequency or bandwidth is divided)

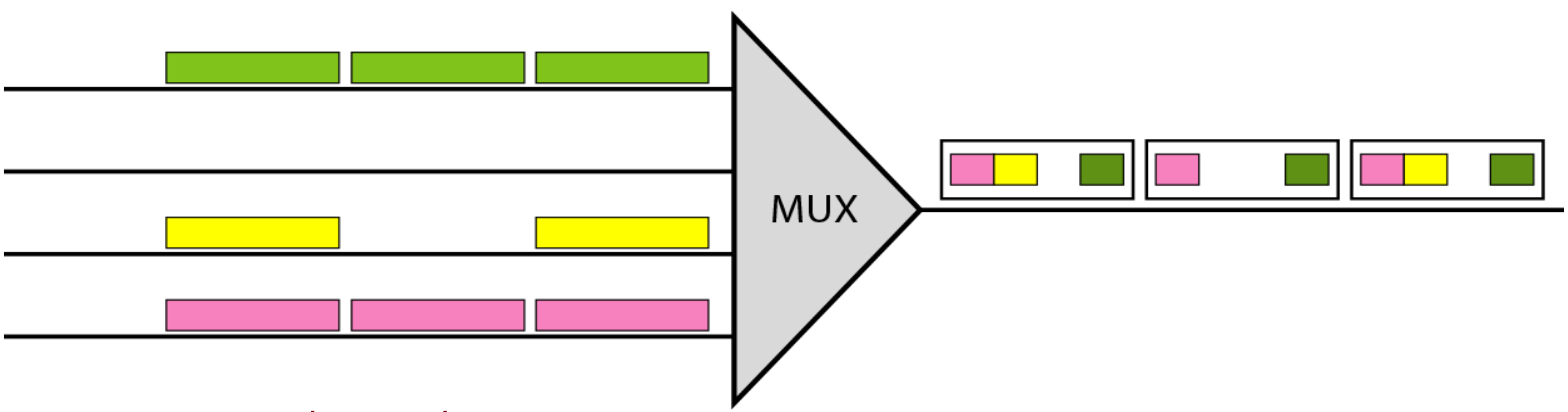
Synchronous TDM

- In **synchronous TDM** for n inputs the data rate of the link is *n* times faster, and the unit duration is *n* times shorter
- For example, in PSTN the trunk is shared in the time domain
- Each input has a buffer and writes its generated data into its buffer (slowly). Then Mux will visit and read the data from the buffer faster than the input is writing to



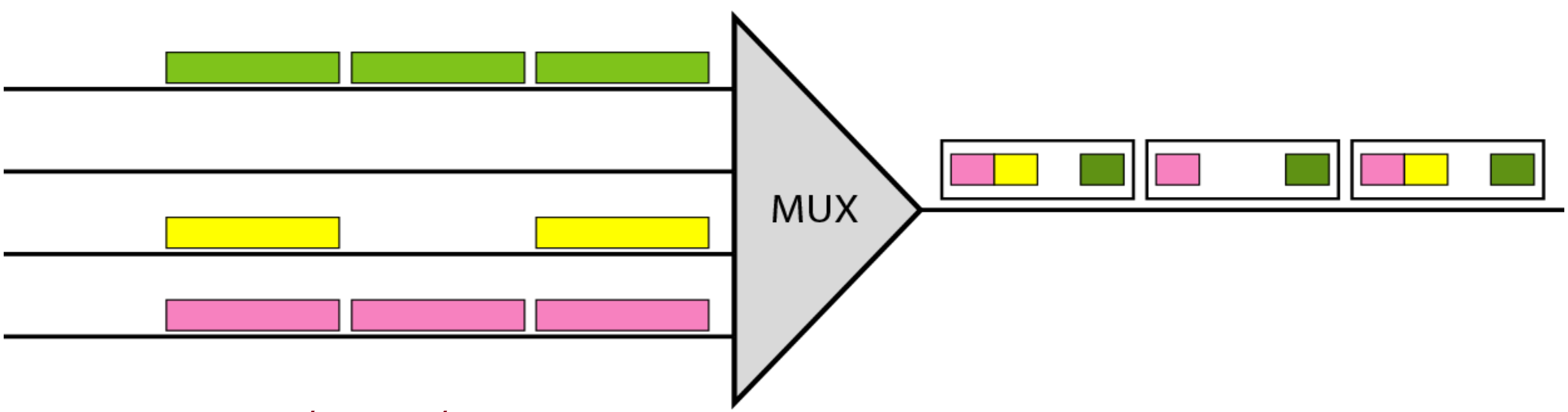
Synchronous TDM (Cont.)

- In synchronous TDM, the slot is assigned to the input and fixed even if that input is idle (does not have any data in its buffer;) this means the time slot for that input is empty and the entire bandwidth of the link is wasted in that time slot
- Synchronous TDM is suitable when the traffic is **streamy** so not many idle slots are encountered. It is not good for **bursty** traffic

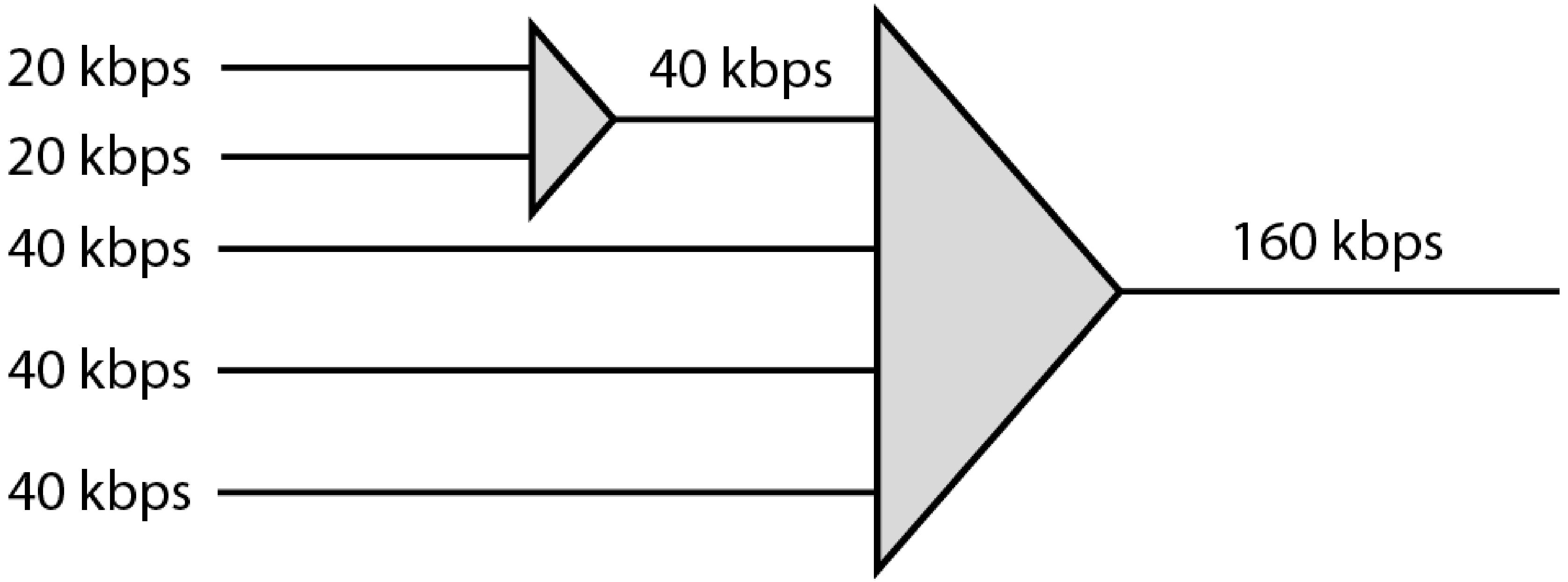


Synchronous TDM – Example

- The TDM that is used in the optical trunks of PSTN has bit rates in the order of Gbps, therefore many 64kbps calls can be TDMed
- The slots may not be equal, however they are pre-assigned and won't be changed during the call. If all the time slots on a trunk are assigned, then if one attempts to make a call, she/he would hear the message that all lines are busy at the time

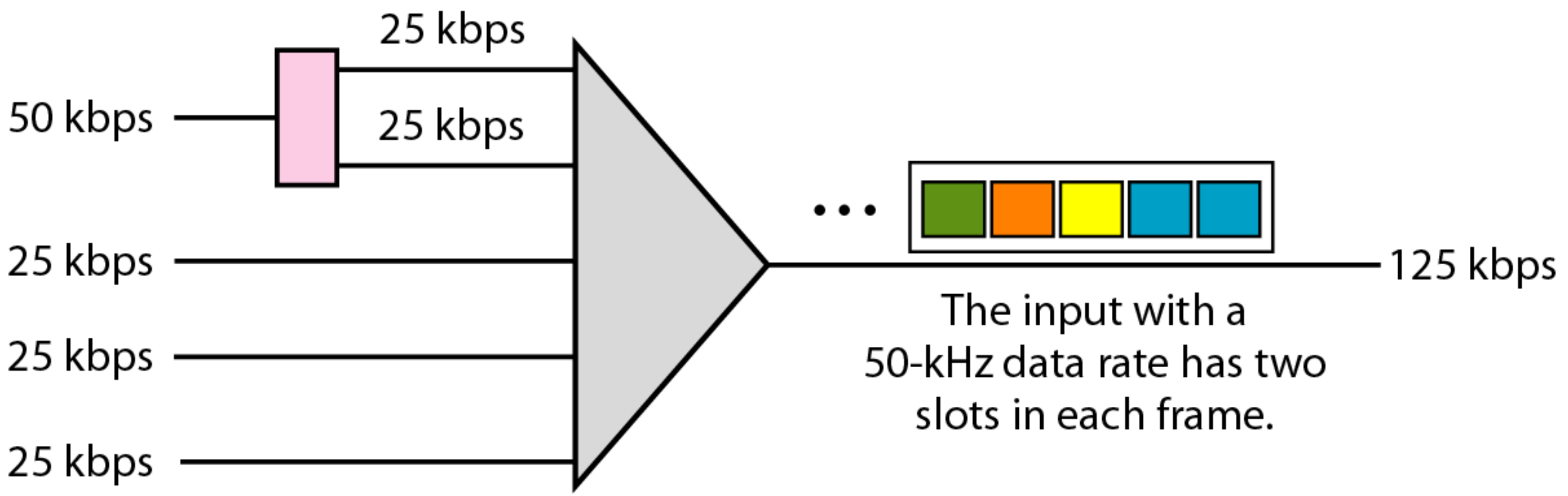


Multilevel Multiplexing



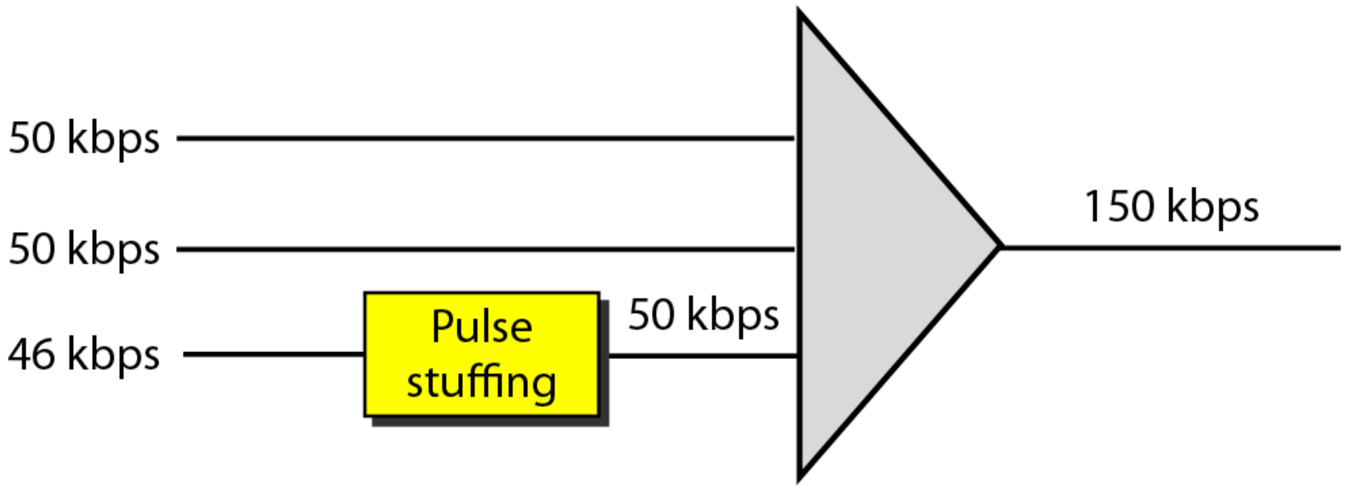
Multiple Slot Multiplexing

- In this example one of the inputs generates data twice as fast as the others (i.e., 50Kbps vs 25Kbps) we can assign two time slots to that input for every other time slot we assign to any other input [Note: the pink box is for demonstration purposes and need not exist. We simply assign two time slots to the first input and one to any other input]



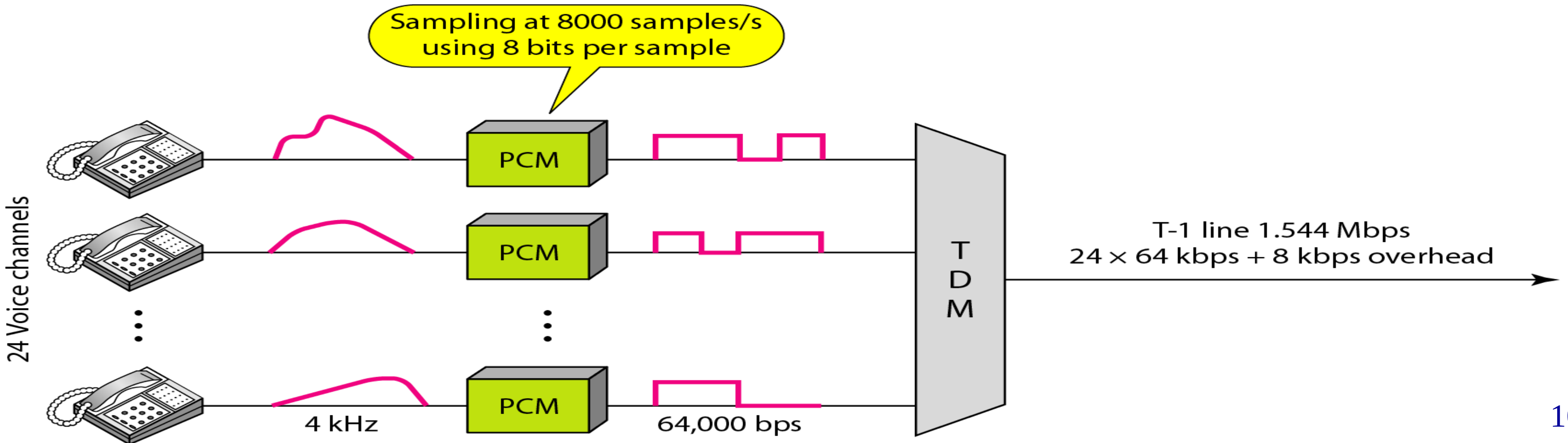
Pulse Stuffing

- Time slots do not need to be evenly distributed among the inputs
- Here the bit rate of one of the inputs is 46Kbps. Redundant bits to the data generated by that input can be added to round up the bit rate number to 50Kbps. This is called **pulse stuffing**



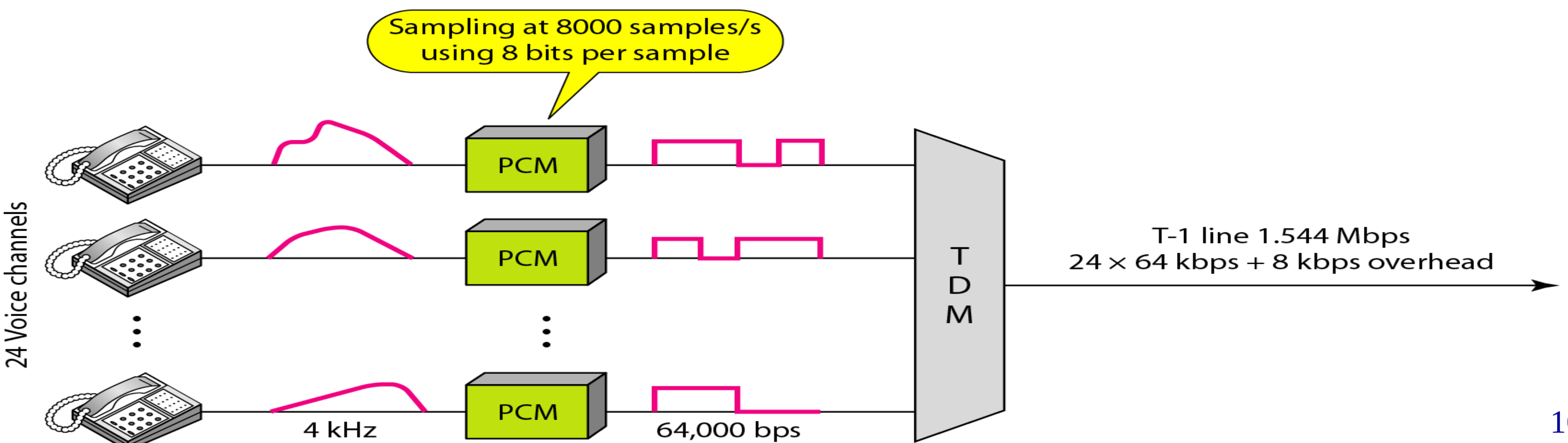
T1

- **T1** is a Full duplex (FDX) dedicated digital service at a rate of 1.544Mbps. It is supported over a 2-pair wire, each pair in one direction [Note: telephone line has only one pair used for both directions]
- It multiplexes 24 channels. Each channel bandwidth is 4KHz, so based on Nyquist Thm, there are 8000 samples/sec. Each sample is quantized into M_Q=256 levels and then encoded into 8 bits. The time duration for each sample (8bits) is 1/8000= 125μsec



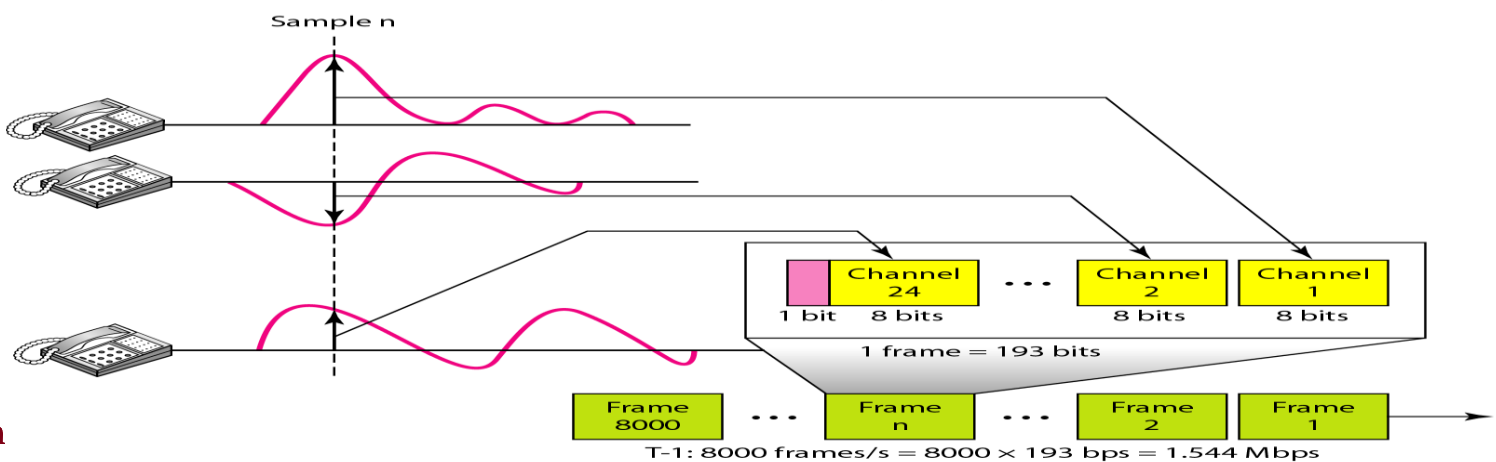
T1 (Cont.)

- For TDM, each time slot includes 8 bits from each channel, so the frame is 24 ×8 = 192bits plus 1 bit so each frame overall has 193 bits. Time duration of T1 frames is 125μsec, therefore the bit rate for T1 is 1.544Mbps
- T1 is for business (but only 24 simultaneous calls) and not residential service because it is very expensive compared to DSL and cable access services. The rate of 1.544Mbps is not that great either!



T1 Frame Structure

- Remember that each voice sample is represented with (encoded into) 8 bits. For a 4KHz voice channel we need 8000 samples (each sample time duration = 1/8000= 125μsec), therefore the bit rate at the output of the encoder is 64Kbps
- The Mux in T1 visits channel 1 picks one sample then goes to channel 2 picks one sample…and finally picks up a sample from channel 24, then goes back to channel 1
- Mux needs to visit each channel every 125μsec (the time duration of each channel sample,) otherwise the generated sample will get lost. That's the reason the time duration for frame should be considered as 125μsec

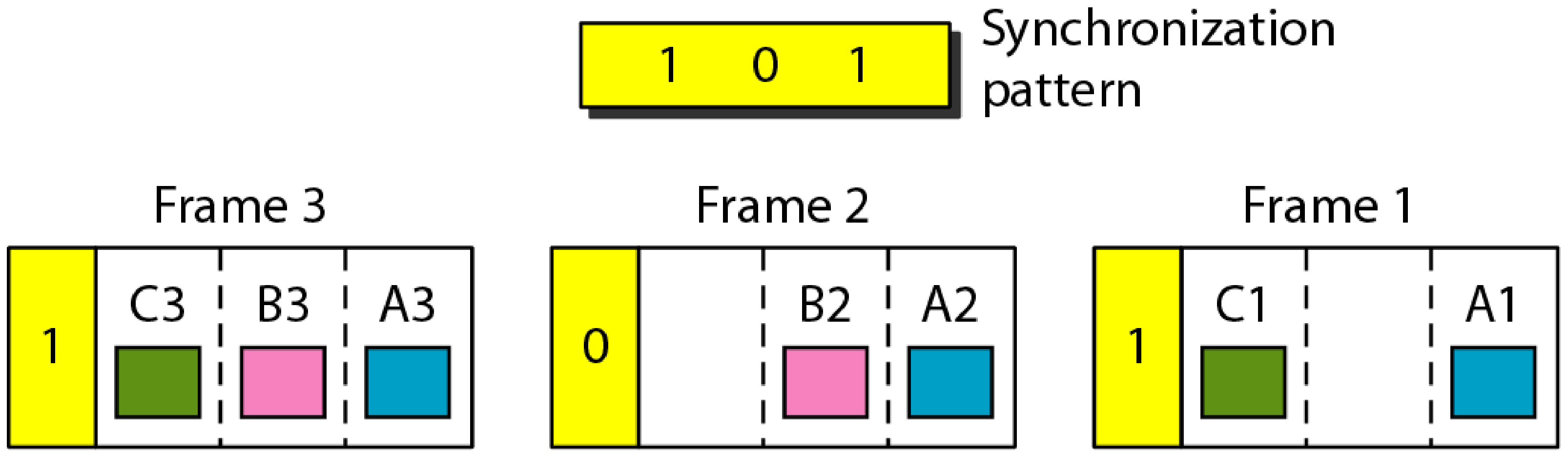


T1 Frame Structure (Cont.)

- Question: for 24 channel each with a bit rate of 64Kbps the total bit rate should be 24×64Kbps=1.536Mbps. Explain the difference between 1.536Mbps and 1.544Mbps calculated previously
- Each frame has 192 bits from channels plus one overhead bit and there are 8000 frames per second so the total rate is 8000 $\times$193bits = 1.544Mbps however the users overall receive only 1.536Mbps as the bit rate
- The efficiency of T1 (the rate of information to the total rate) is hence 1.536Mbps/1.544Mbps=99.48%

TDM in T1

- Framing bit is to allow Mux to synchronize with the incoming stream

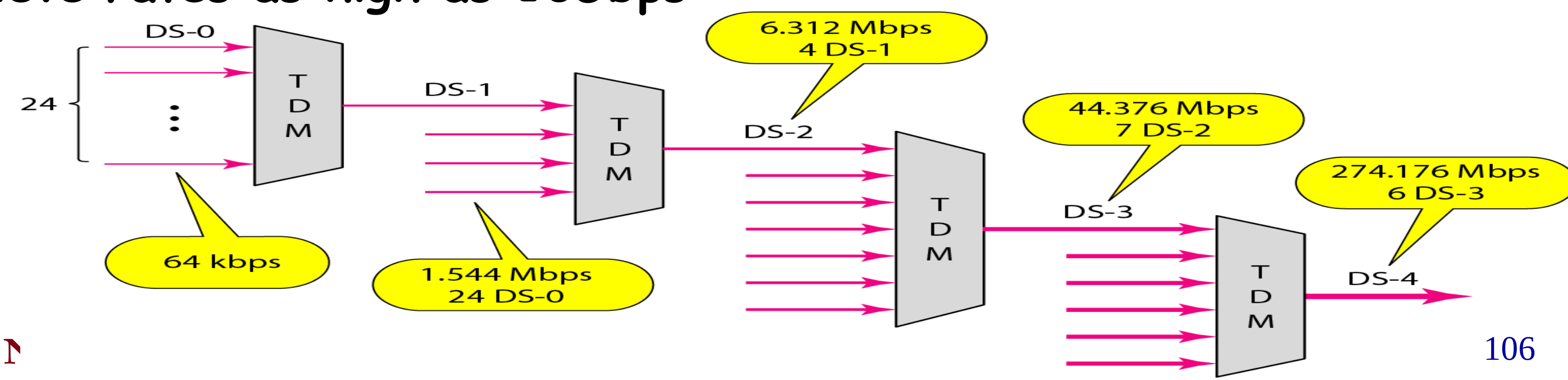


TDM in T1

- During time slot 1 the rate of the output of the Mux is 1.544Mbps. That is the case for any other time slot. For each time slot the rate of writing to its buffer is 64Kbps but Mux would read the bits with a rate of 1.544Mbps, so it's as if the rate of input 1 during time slot 1 was 1.544Mbps
- The rate of input 1 in any other time slot than time slot 1 is zero, therefore if you make an average of the rate of input 1 (time slot 1: 1.544Mbps, 23 other slots: 0bps) it is going to be 64Kbps
- channel, but the outcome is overall the same as that in T1
- Question: Describe an alternative solution to T1 lines
- Question: A startup company has 56 employees. How many T1 lines does it need to lease for its employees?

TDM – Digital Hierarchy

- Shows the hierarchy of TDM in the telephone network. 24 **DS-0** (digital signal level 0, or input channel) will be muxed into one **Ds-1** T1 line
- For **DS-2** (4 T1 or 96 channels) 4×1.544=6.176Mbps differs from (is less than) 6.312Mbps due to overhead
- **T3** (aka **DS-3**) which suits bigger businesses and is more expensive, has 7 DS-2 (alternatively 28 T1) or 672 channels. Unlike T1 which is on 2-pairs of copper wires, T3 is on either cable or radio (microwave or satellite)
- T3 (44.376Mbps) is still not fast enough for big companies
- The hierarchy was upgraded using SONET (optical hierarchy) to achieve rates as high as 10Gbps

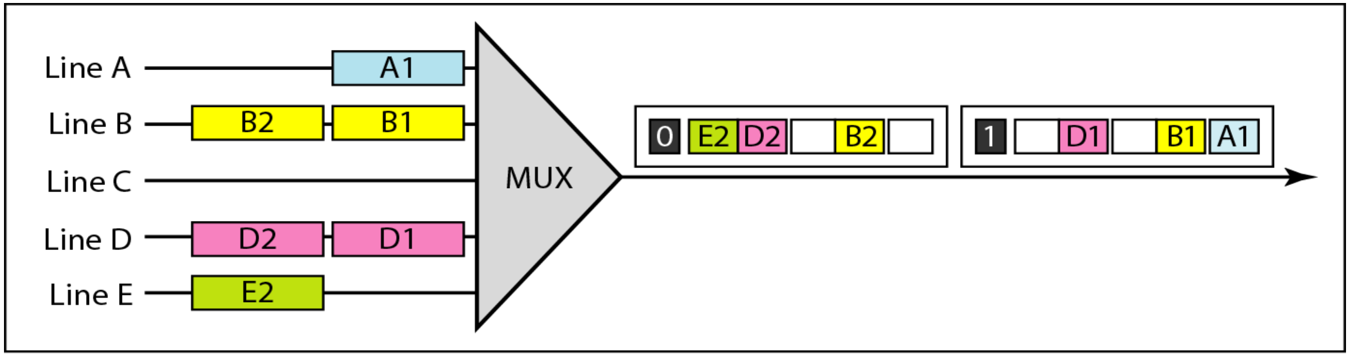


DS and T Line Rates

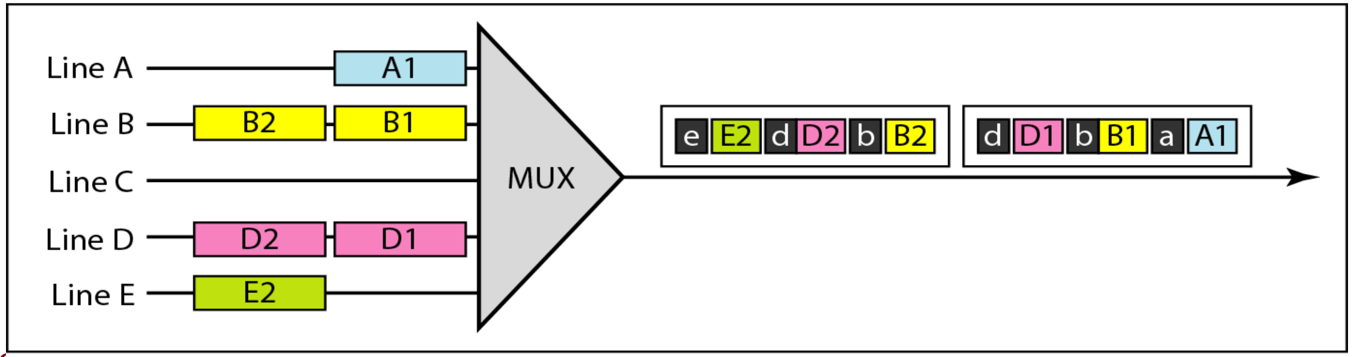
Service	*Line*	*Rate (Mbps)*	*Voice Channels*
DS-1	T-1	1.544	24
DS-2	T-2	6.312	96
DS-3	T-3	44.736	672
DS-4	T-4	274.176	4032

Statistical TDM

- In **statistical TDM** if an input does not have anything to transmit, i.e., its buffer is empty, then the Mux need not transmit an empty time slot. It assigns the slots dynamically and based on demand, so mux will look at the buffer and assigns a bigger time slot (TS) to the user with higher demand



a. Synchronous TDM



b. Statistical TDM

Statistical TDM (Cont.)

- Unlike synchronous TDM, time slots are not pre-allocated, therefore the network (routers or switches) should have a way of knowing which input the TS (time slot) belongs to
- Each TS is preceded by a header (an address) to identify whose TS it is. Note that this not like a framing bit. In fact it is the source and destination IP addresses
- Note: header is an overhead. So part of the channel capacity is wasted to transmit a header for each frame, however, this overall is better than wasting a lot of pre-allocated empty TS'es as occurs in the synchronous TDM for the case of idle inputs
- Question: How is the synchronization realized in statistical TDM? There is no synchronization in statistical TDM!

Statistical TDM (Cont.)

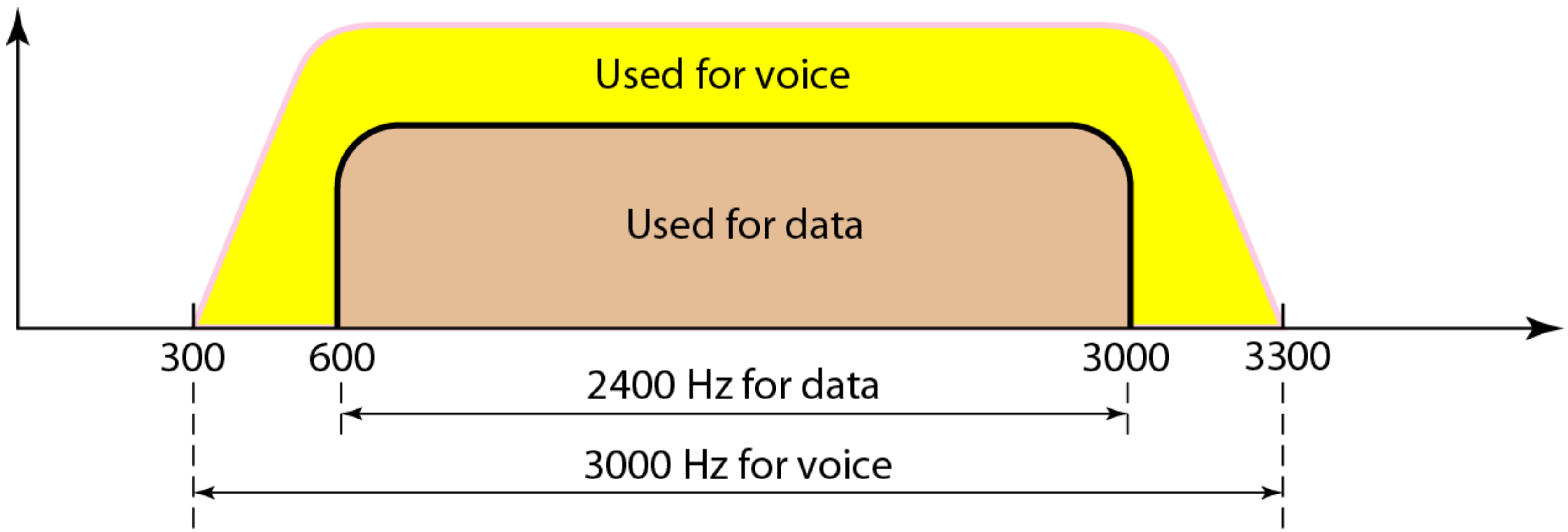
- Compare synchronous TDM and statistical TDM by assuming the same Mux (output channel) rate for them. Which one can support more users in average?
  - Statistical TDM, because it can take more users assuming some of them would be idle
  - For example, for an identical rate of 1Mbps and user rate of 50Kbps, the synchronous TDM can only support 20 users, whereas the statistical can support close to 80 users if it knows in average more than 75% of them are idle
  - Note that in the above discussion we ignored the overhead of the addresses in the header. But even 60 user support for the statistical TDM above is a lot better support than only 20 users in synchronous TDM

Statistical TDM (Cont.)

- Question: What if in the previous example all 80 users become active at the same time?
  - Then 60 go into backlog. This is the disadvantage of statistical TDM, meaning that it goes into backlog when all users are active

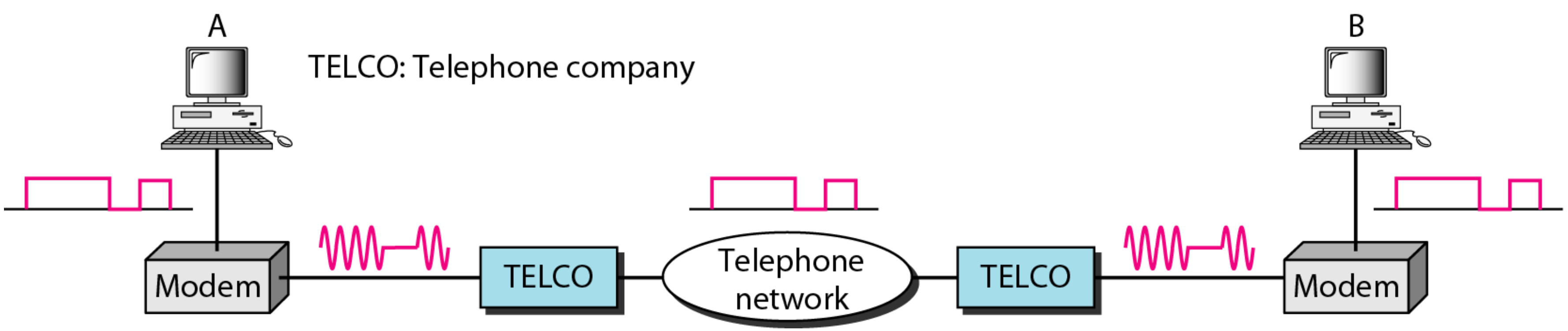
Access Technologies – Dialup

- In dial-up service the bandwidth of the voice and data overlap, therefore it is not possible to later separate them, i.e., we cannot make a phone call and data call at the same call
- For voice a great deal of interference and distortion may be accepted
- Data signal is different from voice and needs higher degree of accuracy, therefore the edges of the frequency range are not used for data



Dialup – Modulation and Demodulation

- By access technologies we refer to the part from the residential or office to the CO (central office) of the Internet service provider (ISP)
- Many of the most popular modems are based on V-series for modem communication standards and interfaces by ITU-T (International Telecommunications Union-Telecommunication Standardization Sector)
- Dialup Service:



DSL (Digital Subscriber Line) Technologies

- After traditional modems reached their peak data rate, telephone companies developed another technology, DSL, or xDSL to provide higher-speed access to the Internet. DSL technology is one of the most promising ones for supporting high-speed communication over the existing local loops (DSL is not digital, but it is to carry digital information

- Some DSL Types:
  - ADSL (Asymmetric DSL)
  - ADSL Lite (Splitterless ADSL)
  - HDSL (High Data rate DSL)
  - SDSL (Symmetric DSL)
    VDSL (Very high Data rate DSL)
  - IDSL (ISDN [Integrated Services Digital Network] DSL)

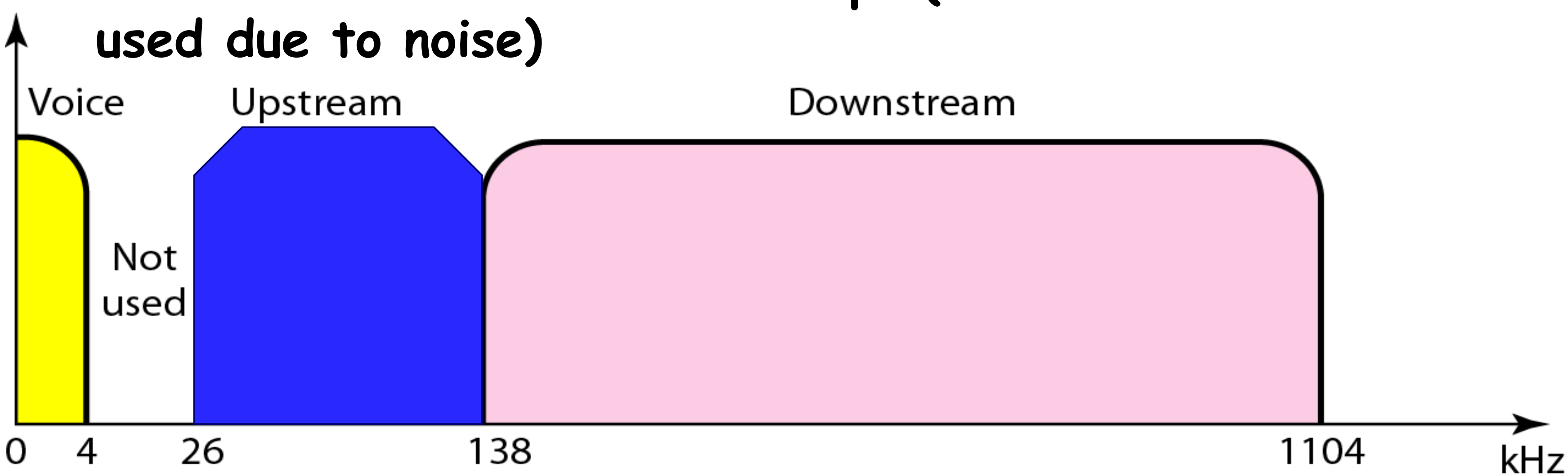
Summary of DSL Technologies

- **Local loop** is a twisted-pair cable that connects the subscriber telephone to the nearest end office or local CO (central office)
- The existing local loops can handle theoretical bandwidth of 1.1MHz (which is a lot higher than 4KHz)
- Factors such as distance btn residence and switching office, size of the cable, the signaling used, etc. affect the bandwidth
- ADSL uses an adaptive technology that tests the condition and bandwidth availability of the line before settling the data rate

Technology	*Downstream Rate*	*Upstream Rate*	*Distance (ft)*	*Twisted Pairs*	*Line Code*
ADSL	1.5–6.1 Mbps	16–640 kbps	12,000	1	DMT
ADSL Lite	1.5 Mbps	500 kbps	18,000	1	DMT
HDSL	1.5–2.0 Mbps	1.5–2.0 Mbps	12,000	2	2B1Q
SDSL	768 kbps	768 kbps	12,000	1	2B1Q
VDSL	25–55 Mbps	3.2 Mbps	3000–10,000	1	DMT

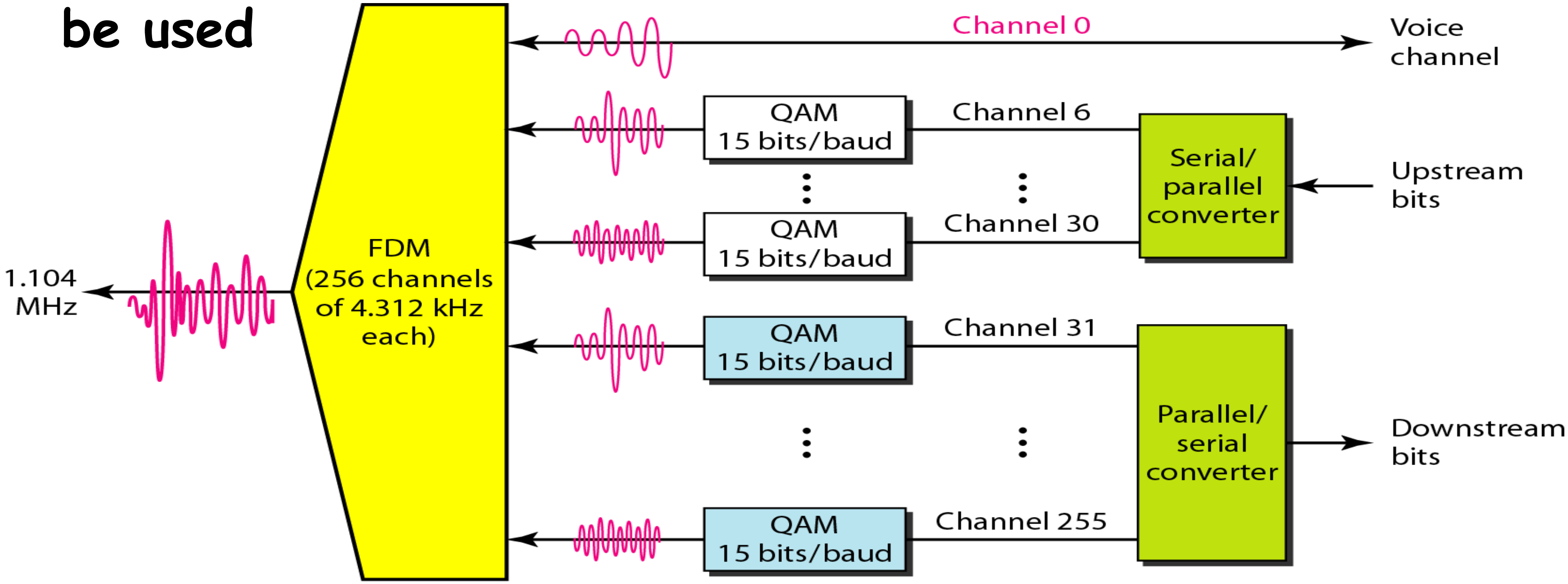
ADSL

- ADSL is an asymmetric communication technology designed for residential users and is not suitable for businesses
- Using FDM, non-overlapping bandwidths are considered in DSL
- Downstream rate is theoretically up to 13.4Mbps, and upstream of about 1.44Mbps (depending on how far the user is from CO , noise condition, etc. Distance is the main limitation of DSL)
- In reality the upstream rate is less than 500Kbps and downstream rate is below 8Mbps (cause some channels are not used due to noise)



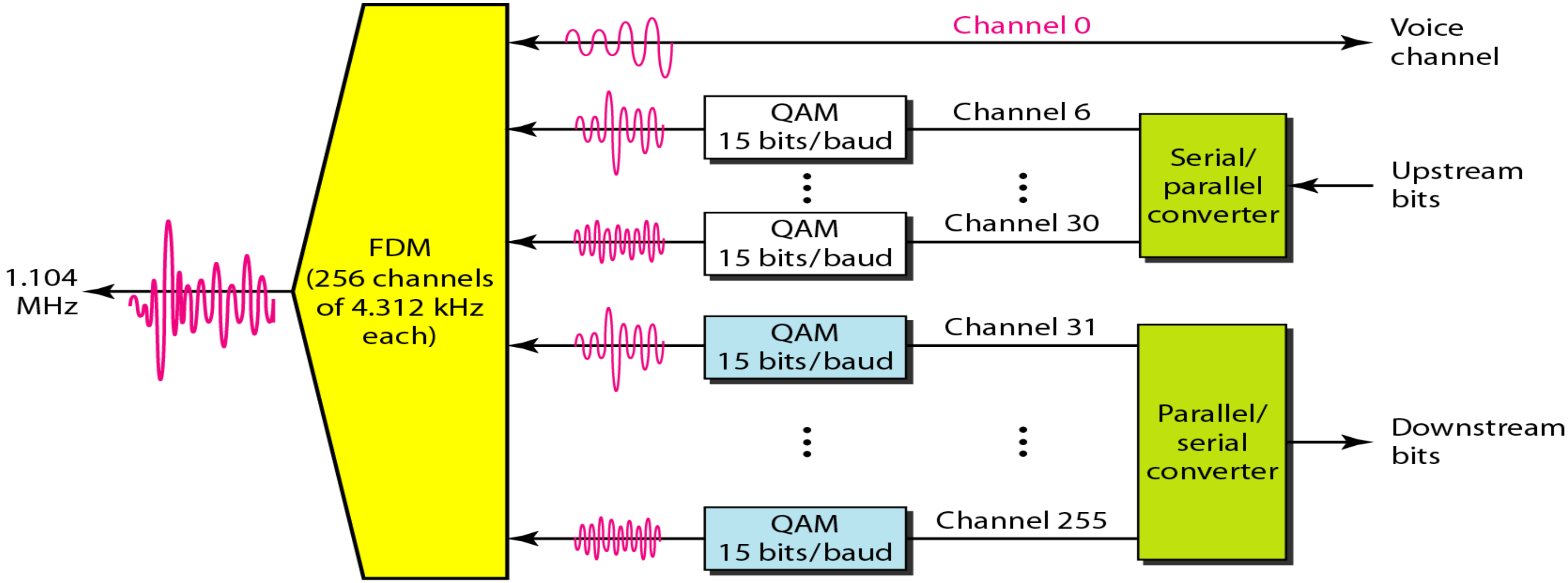
ADSL Modulation Technique

- ADSL modulation technique is a standard called **DMT** (**Discrete Multitone Technique**) which combines QAM and FDM
- Channels 6 to 30 (one for control, 24 for data) are each 4KHz (out of 4.312 available,) therefore maximum upstream rate is 24×4000×15=1.44Mbps, however the normal rate is below 500Kbps, as in large noise levels, some channels won't be used



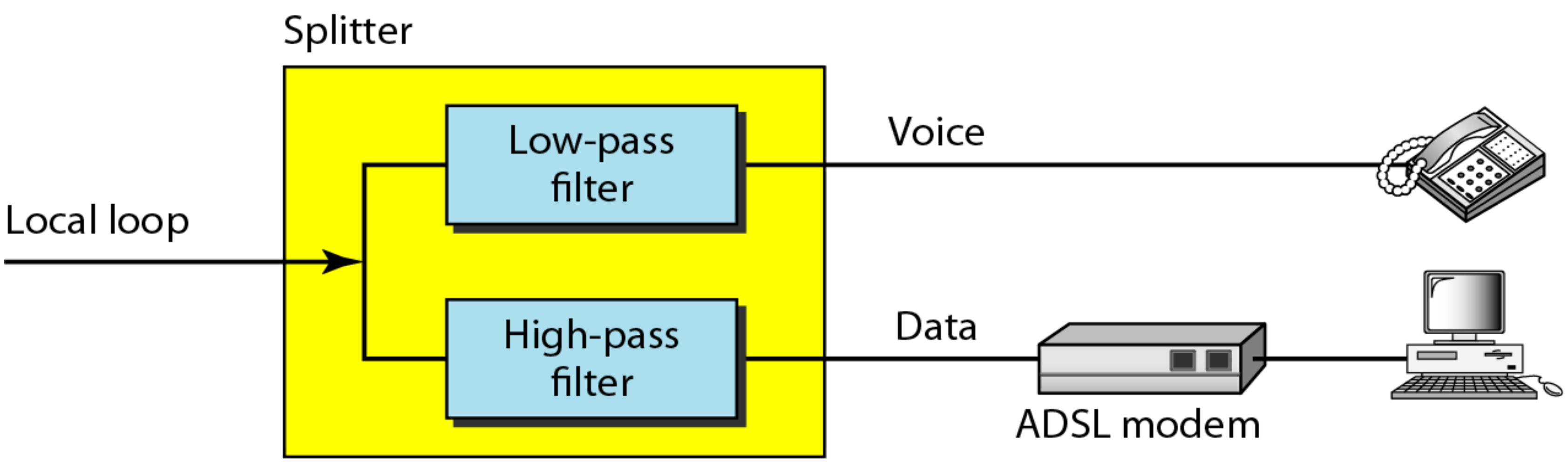
DMT (Cont.)

- Channel 0 is reserved for voice. Channels 1 to 5 are not used (to have gap btn voice and data)
- Channels 31 to 255 (one for control and 224 for downstream data transfer) can result in maximum downstream rate of 224×4000×15=13.4Mbps, however the normal rate is below 8Mbps as some of the channels are deleted (unused) due to noise



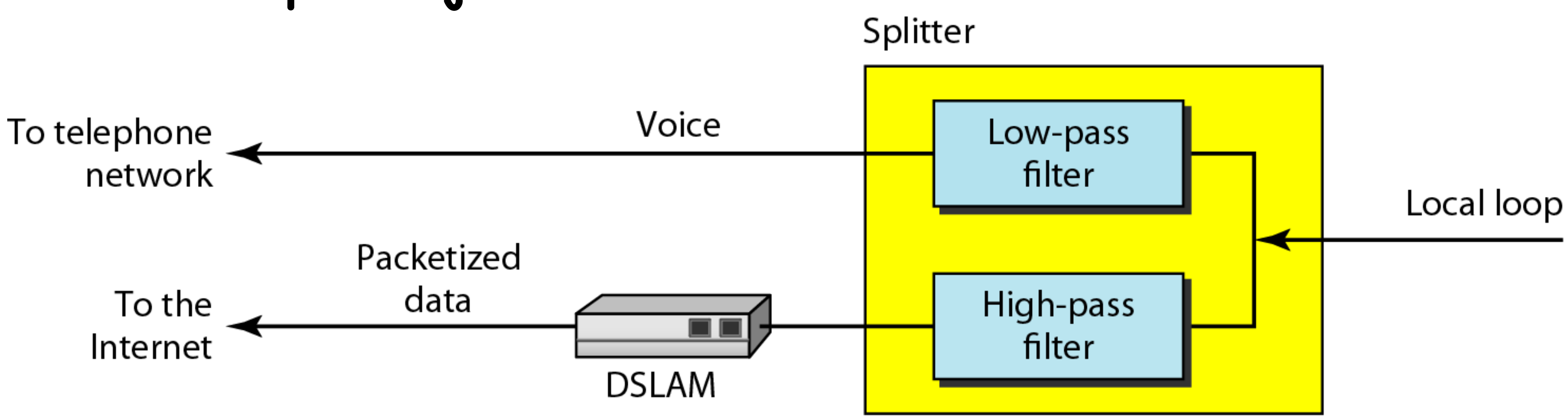
Customer Site: ADSL Modem

- **The local loop connects to a splitter (installed at the customer's premises) which separates voice and data. Using DMT, the ADSL modem at the customer site, modulates and demodulates to create downstream and upstream channels, respectively**



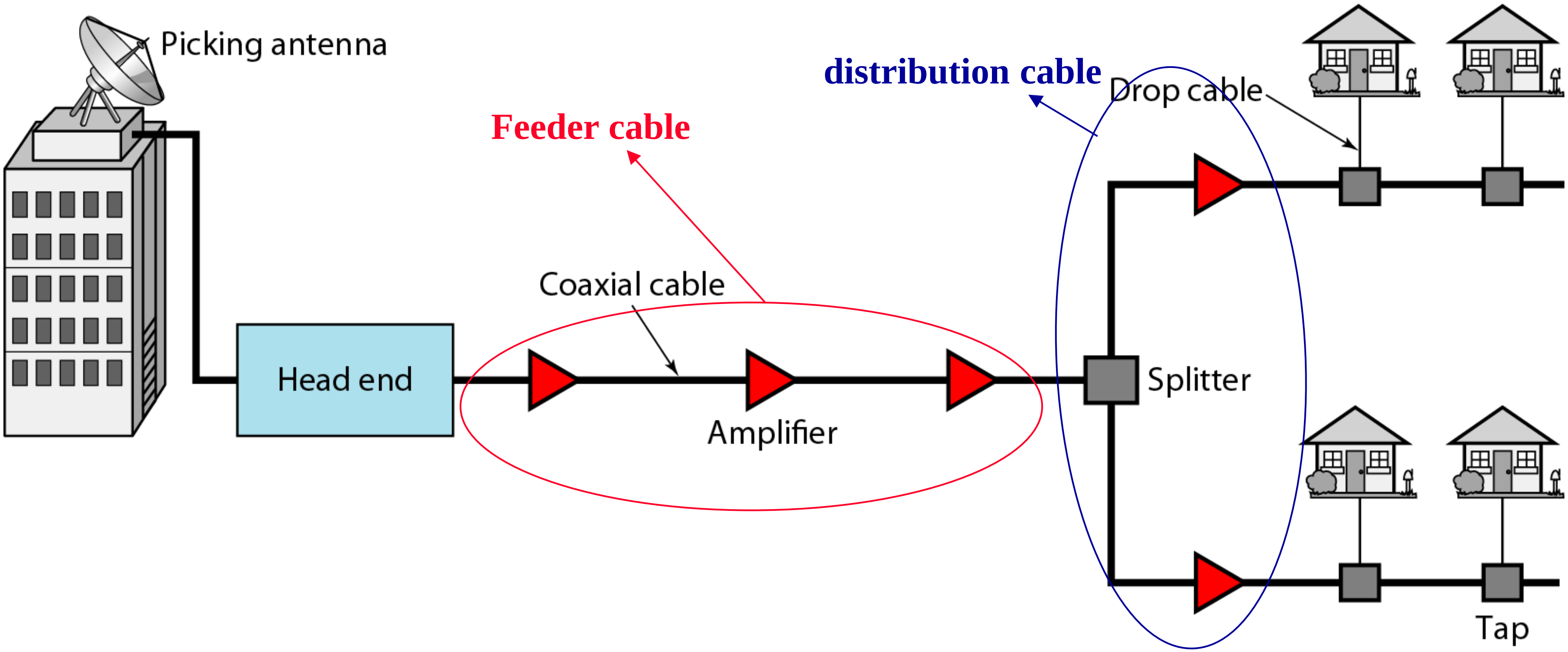
Telephone Company Site: DSLAM

- At the telephone company site, instead of an ADSL modem a **DSLAM** (**DSL Access Multiplexer**) is used that functions similarly to the ADSL modem, but additionally packetizes the data to be sent to the Internet (ISP). It is a statistical Mux (in fact Demux) that is used as a demodulator
- Note: In CO there is one DSLAM for each customer
- ADSL is expensive mostly because of the cost of the splitter installation. In **ADSL Lite** the splitting is done at the telephone company, so ADSL Lite modem is directly plugged into a telephone jack



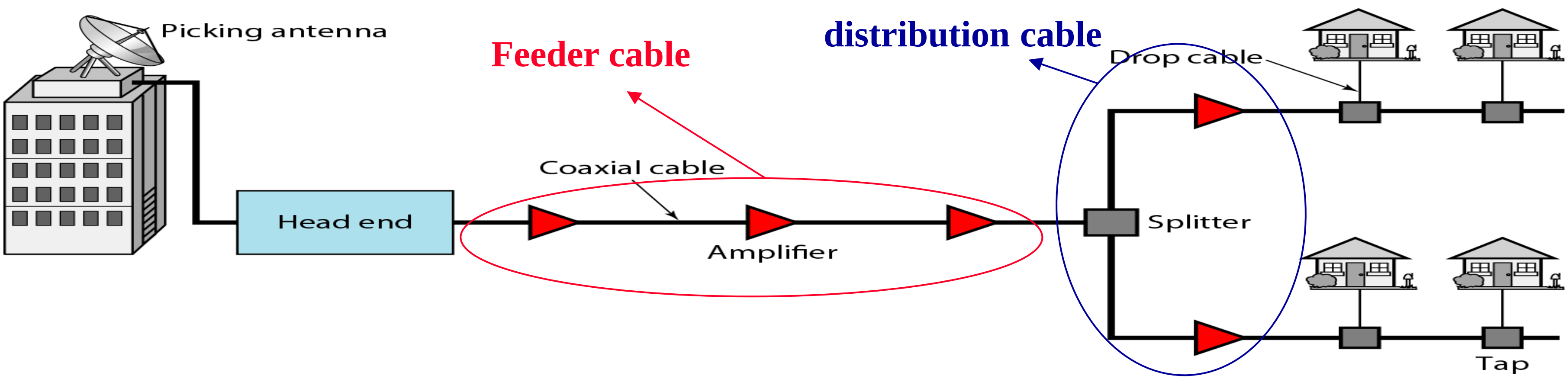
Traditional Cable TV Network

- Traditional cable TV was called **CATV** (Community Antenna TV)
- The cable TV office (aka the **head end**) receives video signal from broadcasting stations and feeds them into coaxial cables



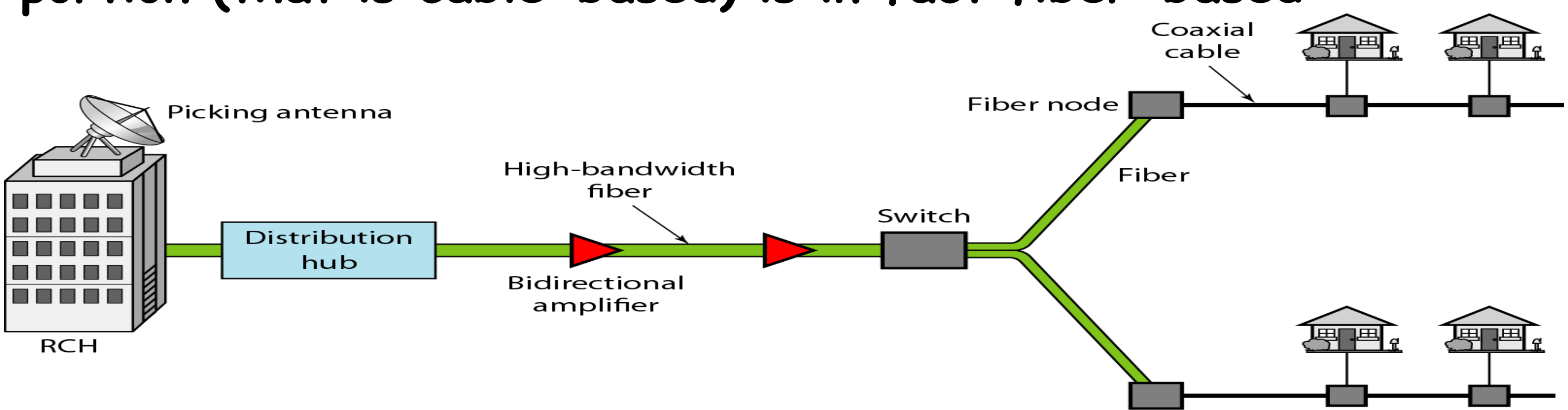
CATV (Cont.)

- The signals would become weaker with distance, therefore up to 35 amplifiers were installed btn the head end and the subscriber premises
- At other end, splitters split the cable. Taps and drop cables made the connection to the subscriber
- CATV used coaxial cable end to end
- Due to attenuation of the signals and the use of a large number of amplifiers, communication in CATV was unidirectional (one-way)
- Therefore, traditional Cable TV could not be used for data



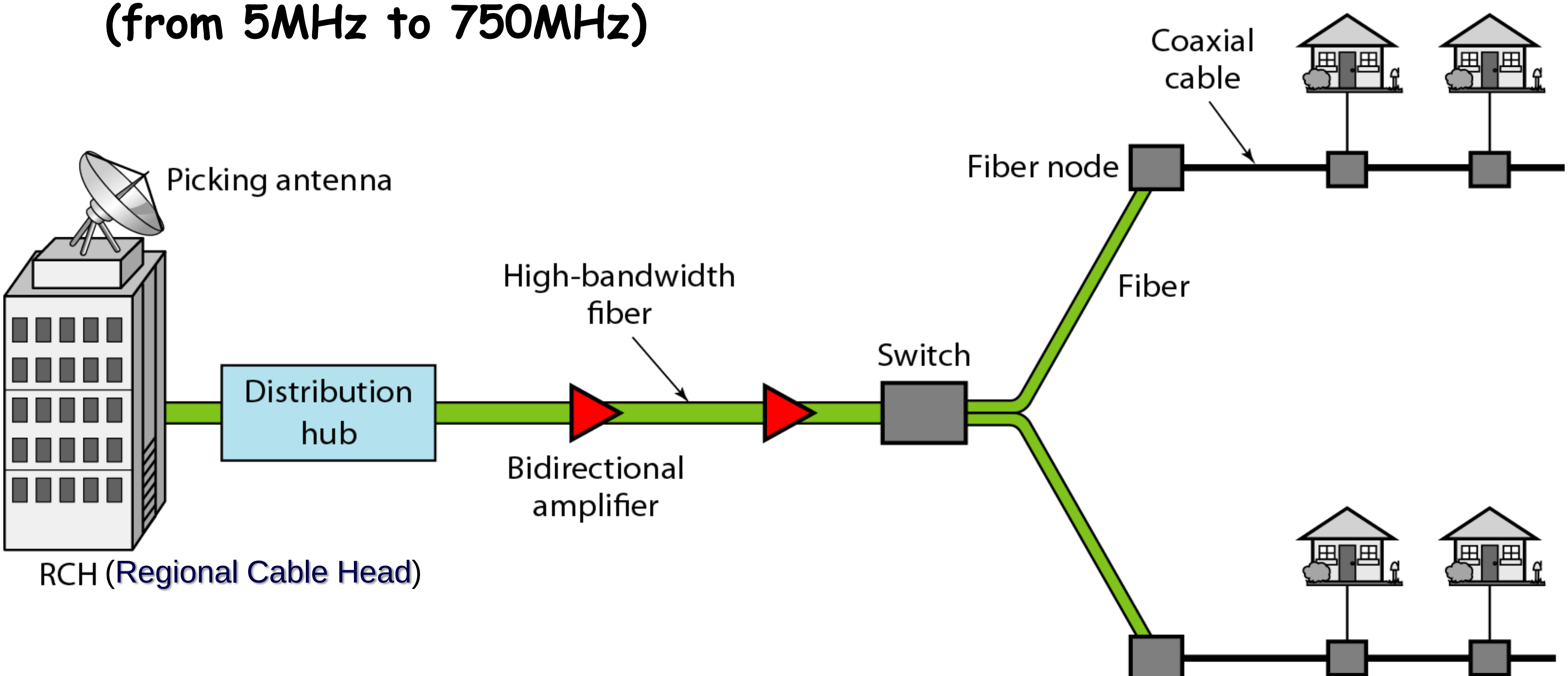
Hybrid Fiber-Coaxial (HFC) network

- The 2nd generation of cable networks uses a combination of fiber-optic and coaxial cable
- The cable companies wanted to use dialup for upstream, but later on they made more modifications to make HFC suitable for bidirectional service (using two way amplifiers, splitters, combiners, etc
- The feeder cable was replaced by a fiber cable (which needs less amplifiers) that connects to to an optical switch to be distributed to shorter fibers
- The fiber node changes the optical signal back to electrical signal and sends it over the distribution cable, therefore all, but the last portion (that is cable-based) is in fact fiber-based



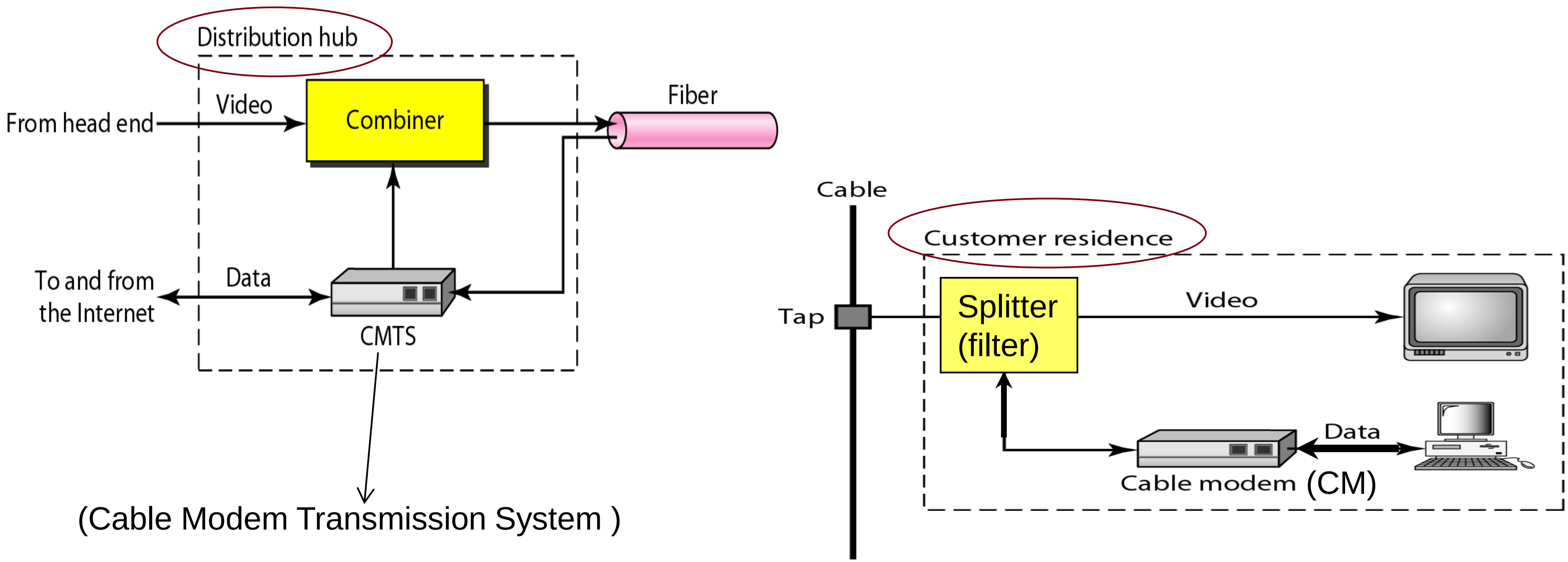
HFC network (Cont.)

- The RCH (regional cable head) serves up to 400,000 subscribers. RCHs feed distribution hubs, each of which serves up to 40,000 subscribers. The modulation and distribution of signals are done here. The signals are then fed to fiber nodes through fiber optic cables
- The coaxial cable has an approximate bandwidth of 750MHz, (from 5MHz to 750MHz)



Two Way Cable (HFC) Access

- To use cable network (HFC) for data transmission, two devices are needed:
  - **CM (Cable Modem)** – installed on the subscriber premises. It is similar to an ADSL modem
  - **CMTS (Cable Model Transmission System)** – installed inside the distribution hub by the cable company. Receives data from Internet and passes them to subscribers, vice versa

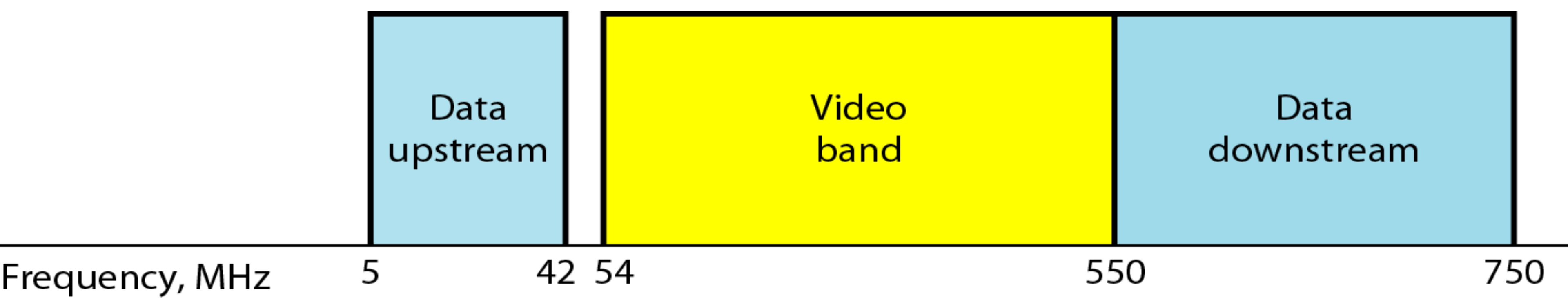


Division of Coaxial Cable band by CATV

- FDM is used over a bandwidth of 750MHz (compare with ADSL bandwidth of 1.104MHz.) This is the main reason ADSL may lose competition against cable service, cause TV channels with bandwidths of up to 6MHz cannot fit the ADSL, this makes ADSL strictly a data service, whereas cable is the whole package (data, voice (audio, and music,) TV broadcasting (analog and digital, Video on Demand (VoD: digital)
- Satellite companies such as DishNetwork and DirectTV are the competitors of the cable companies (there are a lot of small cable companies in the US)
- VoD is digital, so video signal gets sampled, quantized, encoded and compressed and sent to the receiver side
- There are up to about 80 channels (each 6MHz) in the range of 54 to 550MHZ for video

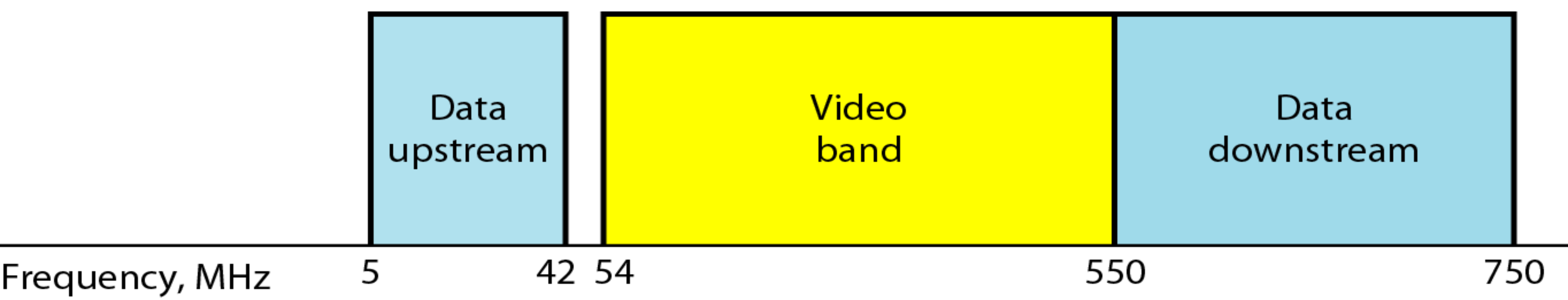
CATV Band – Data Bands

- 200MHz bandwidth (from 550 to 750MHz) is the downlink for data and also VoD, so there are about 33 channels (each 6MHz)
- Downstream uses 64-QAM (k=6, one bit for error correction and 5 bits of data per baud or 1Hz), therefore the theoretical downstream data rate is 30Mbps (5bits/Hz×6MHz), however cable model is typically connected through a 10Base-T cable which limits the data to 10Mbps



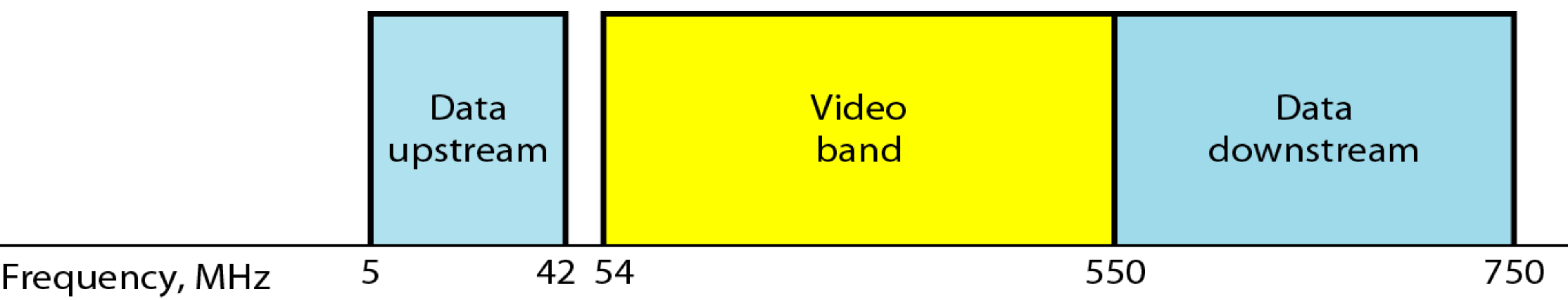
CATV Band – Data Bands

- 37MHz bandwidth (from 5 to 42MHz) is dedicated to uplink, from the resident to the distribution hub
- Upstream uses QPSK (k=2, 2 bits per baud or 1 Hz), therefore the theoretical upstream data rate is 12Mbps (2bits/Hz×6MHz)
- Both upstream and downstream bands should be shared by subscribers



CATV Band – Data Bands (Cont.)

- The customers on the residential share the cable, down stream would not have much problem other than the fact that for a shared network, the speed degrades if there are many customers trying to use the service
- Note that all subscribers sharing a channel receive the corresponding downstream data, but only the one with its address matching keeps the data



CATV Band – Data Bands (Cont.)

- However for the upstream direction there is a collision problem: if two or more customers want to send data at the same time, there will be a collision (note for 42-5MHz bandwidth the maximum number of parallel channels can be 20 with upstream bandwidths as high as 2MHz or only 6 with 6MHz bandwidth,) so for more users, we need to have a procedure to avoid collision
- The solution to this problem is called MAC access control (it is mainly used in Ethernet, and some other technologies such as Token Ring, ALOHA, Wireless 802.11.) This will be one of the topics of Data Link layer