Departament d'Arquitectura de Computadors

Facultat d'informàtica de Barcelona

Capítol 2. Elements tecnològics d'Internet

Source Book: Data and Computer Communications. 10th Ed. W. Stallings

1 Tecnologies de Xarxes de Computadors

Aquest capítol farem un anàlisi dels fonaments bàsics en la transmissió de dades. Analitzarem els diferents medis de transmissió en l'actualitat fen una atenció especial en la fibra òptica i satèl·lit. La possibilitat de utilitzar un medi per a la transmissió de dades comporta una sèrie de requisits tècnics que limiten el seu ús. Caldrà analitzar-ho i fonamentar les bases en les tècniques de comunicacions de dades. Per transmetre dades es poden utilitzar dues tècniques, la codificació i la modulació. Estudiarem les seves possibilitats i requeriments en relació als medis de transmissió. Per transmetre físicament els bits caldrà organitzar-los per mantenir en tot moment la sincronització en base a la multiplexació, sobre tot quan es transmeten canals. La definició de canals i paquets ens permetrà utilitzar les xarxes en formats diferents. Això té repercussió sobre els sistemes de transmissió físics i de nivell d'enllaç.

Un aspecte fonamental de tot plegat per al funcionament dels sistemes de transmissió de dades serà la sincronització en qualsevol nivell OSI. Això vol dir saber quan comença i quan acaba una unitat de dades. Molts dels conceptes anteriors aniran en aquest sentit i ho tindrem present constantment.

Finalment estudiarem el protocol HDLC com una base i model de la majoria de protocols en funcionament a nivell de capa d'enllaç.

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2.1 Medis de transmissió

Consultar Capítols 3.1, 3.2, 3.3, 3.4 i 4 Stallings.

Anàlisi dels medis de transmissió més utilitzats per interconnectar sistemes de transmissió de dades. Amb fils: parell de coure, cable coaxial i fibra òptica. Sense fils: atmòsfera ràdio i satèl·lit.

Primer s'analitza el concepte de transmissió frequencial i els seus requeriments.

Veure capítol 3 i 4 de Stallings.

Velocitats en transmissió de dades

- Transmissió

  Vt = bits/seg

- Propagació

  Vp = Km/seg

- Llargària d'un bit = Vp / Vt

La velocitat de transmissió és el ritme en que el sistema posa els bits a la línia.

La velocitat de propagació és la velocitat a la que el senyal que representa el bit circula per la línia.

Així doncs podem calcular quan de llarg és un bit o bé quans bit hi caben en una distància de línia determinada dividint aquestes velocitats.

La divisió dona bits/km o Km/bit.

Frequency, Spectrum and Bandwidth

Time Domain Concepts

- **analog signal**
  - *signal intensity varies smoothly with no breaks*
- **digital signal**
  - *signal intensity maintains a constant level and then abruptly changes to another level*
- **periodic signal**
  - *signal pattern repeats over time*
- **aperiodic signal**
  - *pattern not repeated over time*

We are concerned with electromagnetic signals used as a means to transmit data. The signal is a function of time, but it can also be expressed as a function of frequency; that is, the signal consists of components of different frequencies. It turns out that the **frequency domain** view of a signal is more important to an understanding of data transmission than a **time domain** view. Both views are introduced here. Viewed as a function of time, an electromagnetic signal can be either analog or digital. An **analog signal** is one in which the signal intensity varies in a smooth, or **continuous**, fashion over time. In other words, there are no breaks or discontinuities in the signal. A **digital signal** is one in which the signal intensity maintains a constant level for some period of time and then abruptly changes to another constant level, in a **discrete** fashion. Figure 3.1 shows an example of each kind of signal. The analog signal might represent speech, and the digital signal might represent binary 1s and 0s. The simplest sort of signal is a **periodic signal**, in which the same signal pattern repeats over time. Stallings DDC9e Figure 3.2 shows an example of a periodic continuous signal (sine wave) and a periodic discrete signal (square wave). Mathematically, a signal $s(t)$ is defined to be periodic if and only if $s(t + T) = s(t) \quad -\infty < t < +\infty$ where the constant T is the period of the signal (T is the smallest value that satisfies the equation). Otherwise, a signal is **aperiodic**. This is an idealized definition. In fact, the transition from one voltage level to another will not be instantaneous, but there will be a small transition period. Nevertheless, an actual digital signal approximates closely the ideal model of constant voltage levels with instantaneous transitions.

Analog and Digital Signals

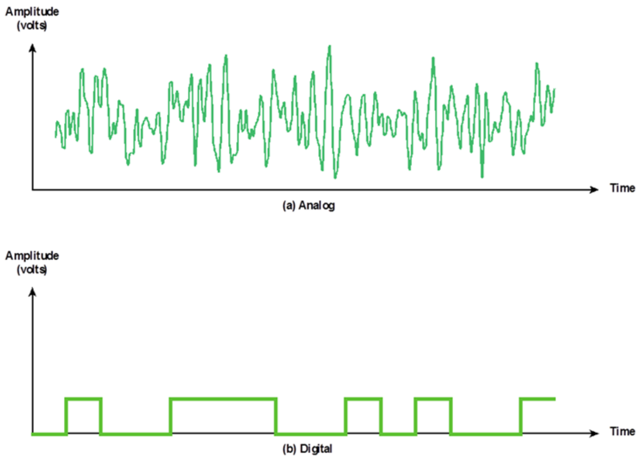

Figure 3.1 Analog and Digital Waveforms

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Viewed as a function of time, an electromagnetic signal can be either analog or digital. An analog signal is one in which the signal intensity varies in a smooth, or continuous , fashion over time. In other words, there are no breaks or discontinuities in the signal. A digital signal is one in which the signal intensity maintains a constant level for some period of time and then abruptly changes to another constant level, in a discrete fashion. Figure 3.1 shows an example of each kind of signal. The analog signal might represent speech, and the digital signal might represent binary 1s and 0s.

Periodic Signals

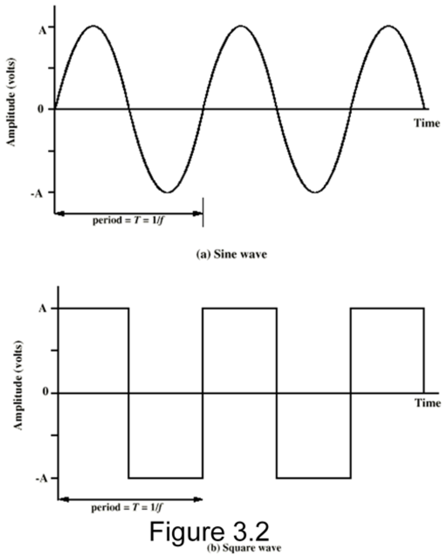

Figure 3.2

The simplest sort of signal is a periodic signal , in which the same signal pattern repeats over time. Figure 3.2 shows an example of a periodic continuous signal (sine wave) and a periodic discrete signal (square wave).

Sine Wave

(periodic continuous signal)

- **peak amplitude (A)**
  - maximum strength of signal
  - typically measured in volts
- **frequency (f)**
  - rate at which the signal repeats
  - Hertz (Hz) or cycles per second
  - period (T) is the amount of time for one repetition
  - T = 1/f
- **phase (ϕ)**
  - relative position in time within a single period of signal

The sine wave is the fundamental periodic signal. A general sine wave can be represented by three parameters: peak amplitude (*A*), frequency (*f*), and phase (f). The **peak amplitude** is the maximum value or strength of the signal over time; typically, this value is measured in volts. The **frequency** is the rate [in cycles per second, or hertz (Hz)] at which the signal repeats. An equivalent parameter is the **period** (*T*) of a signal, which is the amount of time it takes for one repetition; therefore, $T = 1/f$. **Phase** is a measure of the relative position in time within a single period of a signal, as is illustrated subsequently. More formally, for a periodic signal $f(t)$, phase is the fractional part t/T of the period T through which t has advanced relative to an arbitrary origin. The origin is usually taken as the last previous passage through zero from the negative to the positive direction.

Varying Sine Waves
$s(t) = A \sin(2\pi ft + \Phi)$

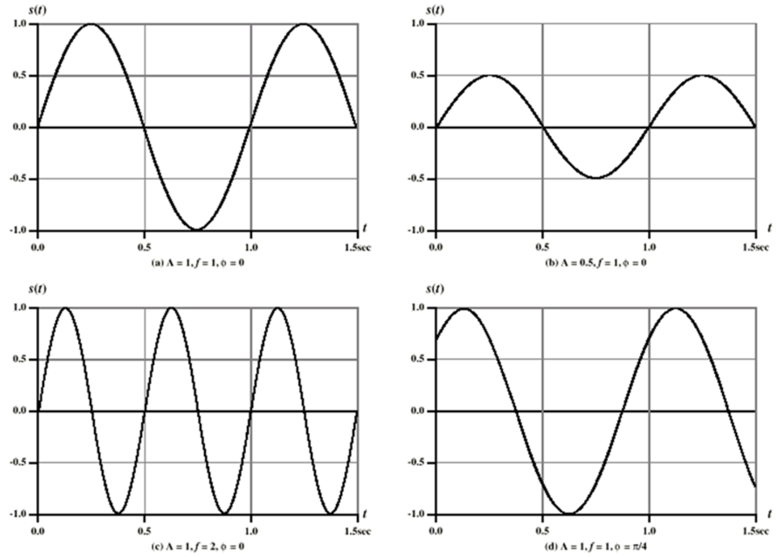

Figure 3.3

The general sine wave can be written $s(t) = A \sin(2\pi ft + f)$. A function with the form of the preceding equation is known as a **sinusoid**. Figure 3.3 shows the effect of varying each of the three parameters. In part (a) of the figure, the frequency is 1 Hz; thus the period is $T = 1$ second. Part (b) has the same frequency and phase but a peak amplitude of 0.5. In part (c) we have $f = 2$, which is equivalent to $T = 0.5$. Finally, part (d) shows the effect of a phase shift of $\pi/4$ radians, which is 45 degrees (2π radians = 360˚ = 1 period). In Figure 3.3, the horizontal axis is time; the graphs display the value of a signal at a given point in space as a function of time. These same graphs, with a change of scale, can apply with horizontal axes in space. In this case, the graphs display the value of a signal at a given point in time as a function of distance. For example, for a sinusoidal transmission (e.g., an electromagnetic radio wave some distance from a radio antenna, or sound some distance from a loudspeaker), at a particular instant of time, the intensity of the signal varies in a sinusoidal way as a function of distance from the source.

Frequency Domain Concepts

- signals are made up of many frequencies
- components are sine waves
- Fourier analysis can show that any signal is made up of components at various frequencies, in which each component is a sinusoid
- can plot frequency domain functions

In practice, an electromagnetic signal will be made up of many frequencies. For example, the signal $s(t) = [(4/\pi) \times (\sin(2\pi ft) + (1/3)\sin(2\pi(3f)t)]$ is shown in Figure 3.4c. The components of this signal are just sine waves of frequencies f and $3f$; parts (a) and (b) of the figure show these individual components. There are two interesting points that can be made about this figure: The second frequency is an integer multiple of the first frequency. When all of the frequency components of a signal are integer multiples of one frequency, the latter frequency is referred to as the **fundamental frequency**. Each multiple of the fundamental frequency is referred to as a **harmonic frequency** of the signal. The period of the total signal is equal to the period of the fundamental frequency. The period of the component $\sin(2\pi ft)$ is $T = 1/f$, and the period of $s(t)$ is also T, as can be seen from Figure 3.4c. It can be shown, using a discipline known as Fourier analysis, that any signal is made up of components at various frequencies, in which each component is a sinusoid. By adding together enough sinusoidal signals, each with the appropriate amplitude, frequency, and phase, any electromagnetic signal can be constructed. Put another way, any electromagnetic signal can be shown to consist of a collection of periodic analog signals (sine waves) at different amplitudes, frequencies, and phases. The importance of being able to look at a signal from the frequency perspective (frequency domain) rather than a time perspective (time domain) should become clear as the discussion proceeds. For the interested reader, the subject of Fourier analysis is introduced in Appendix A.

Addition of Frequency Components (T=1/f)

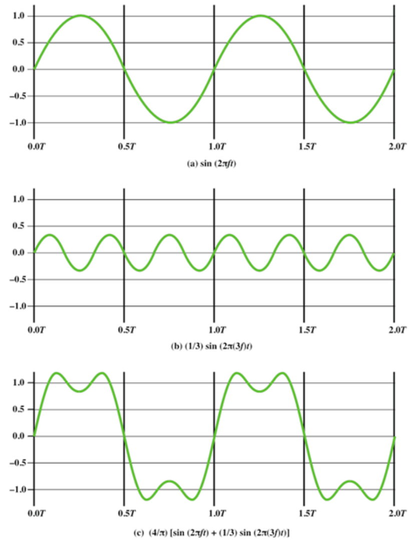

Figure 3.4

Figure 3.4 Addition of Frequency Components ($T = 1/f$)

In Figure 3.4c, the components of this signal (c) are just sine waves of frequencies *f* and 3f, as shown in parts (a) and (b). (c)= (a) + (b)

Frequency Domain Representations

- frequency domain function
- frequency domain function of single square pulse

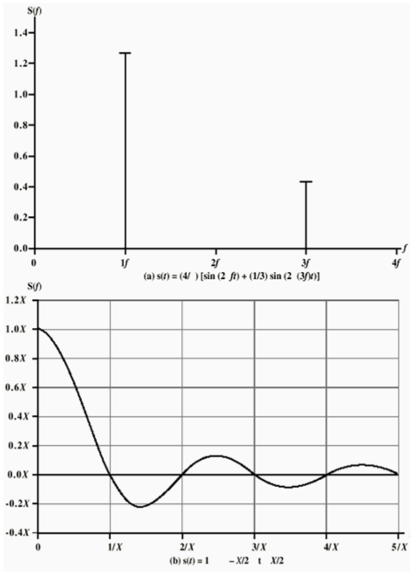

Figure 3.5

So we can say that for each signal, there is a time domain function $s(t)$ that specifies the amplitude of the signal at each instant in time. Similarly, there is a frequency domain function $S(f)$ that specifies the peak amplitude of the constituent frequencies of the signal. Figure 3.5a shows the frequency domain function for the signal of Figure 3.4c. Note that, in this case, $S(f)$ is discrete. Figure 3.5b shows the frequency domain function for a single square pulse that has the value 1 between $–X/2$ and $X/2$, and is 0 elsewhere. Note that in this case $S(f)$ is continuous and that it has nonzero values indefinitely, although the magnitude of the frequency components rapidly shrinks for larger f. These characteristics are common for real signals.

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Spectrum & Bandwidth

spectrum

- range of frequencies contained in signal

absolute bandwidth

- width of spectrum

effective bandwidth

- often just bandwidth
- narrow band of frequencies containing most energy

dc component

- component of zero frequency

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The **spectrum** of a signal is the range of frequencies that it contains. For the signal of Figure 3.4c, the spectrum extends from f to $3f$. The **absolute bandwidth** of a signal is the width of the spectrum. In the case of Figure 3.4c, the bandwidth is $3f - f = 2f$. Many signals, such as that of Figure 3.5b, have an infinite bandwidth. However, most of the energy in the signal is contained in a relatively narrow band of frequencies. This band is referred to as the **effective bandwidth**, or just **bandwidth**. One final term to define is **dc component**. If a signal includes a component of zero frequency, that component is a direct current (dc) or constant component.

Signal with dc Component

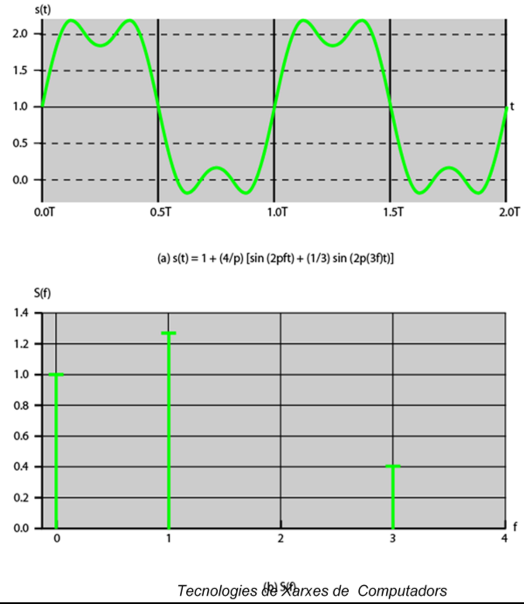
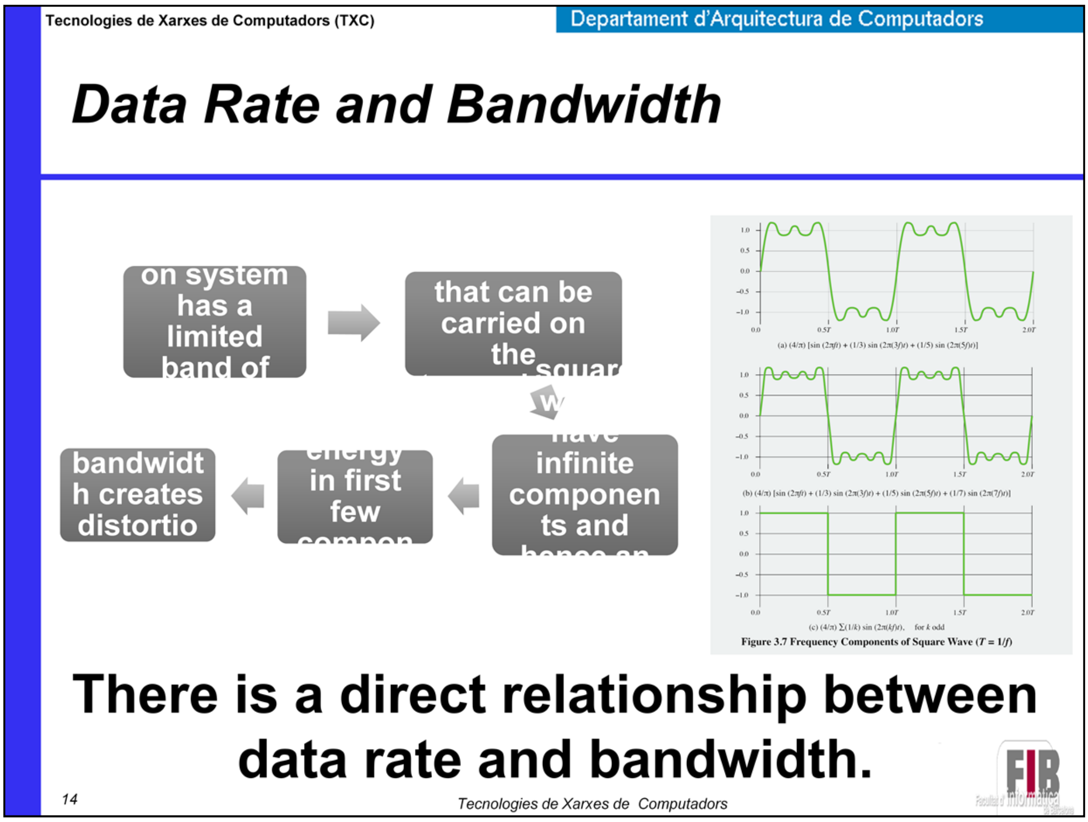


Figure 3.6

For example, Figure 3.6 shows the result of adding a dc component to the signal of Figure 3.4c. With no dc component, a signal has an average amplitude of zero, as seen in the time domain. With a dc component, it has a frequency term at $f = 0$ and a nonzero average amplitude.



We have said that effective bandwidth is the band within which most of the signal energy is concentrated. The meaning of the term *most* in this context is somewhat arbitrary. The important issue here is that, although a given waveform may contain frequencies over a very broad range, as a practical matter any transmission system (transmitter plus medium plus receiver) will be able to accommodate only a limited band of frequencies. This, in turn, limits the data rate that can be carried on the transmission medium. To try to explain these relationships, consider the square wave of Figure 3.2b. Suppose that we let a positive pulse represent binary 0 and a negative pulse represent binary 1. Then the waveform represents the binary stream 0101…. The duration of each pulse is $1/(2f)$; thus the data rate is $2f$ bits per second (bps). What are the frequency components of this signal? To answer this question, consider again Figure 3.4. By adding together sine waves at frequencies f and $3f$, we get a waveform that begins to resemble the original square wave. Let us continue this process by adding a sine wave of frequency $5f$, as shown in Figure 3.7a, and then adding a sine wave of frequency $7f$, as shown in Stallings Figure 3.7b. As we add additional odd multiples of f, suitably scaled, the resulting waveform approaches that of a square wave more and more closely. Thus, this waveform has an infinite number of frequency components and hence an infinite bandwidth.

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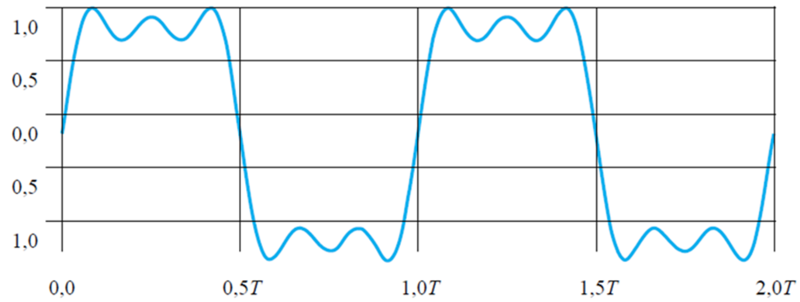
Data Rate and Bw



(a) $(4/\pi)\,[\sin(2\pi ft) + (1/3)\sin(2\pi(3f)t) + (1/5)\sin(2\pi(5f)t)]$

Si f = 1 Mhz ; Bw = 4 Mhz (5f-1f)
T = 1/f = 1 microseg ; temps bit = 0,5 microseg
2 bits/hertz ; Vt. = 2x 1Mhz = 2 Mbps
Si f = 2 Mhz ; Bw = 8 Mhz ; T = 0,5 microseg
2 bits/hertz Vt = 4 Mbps

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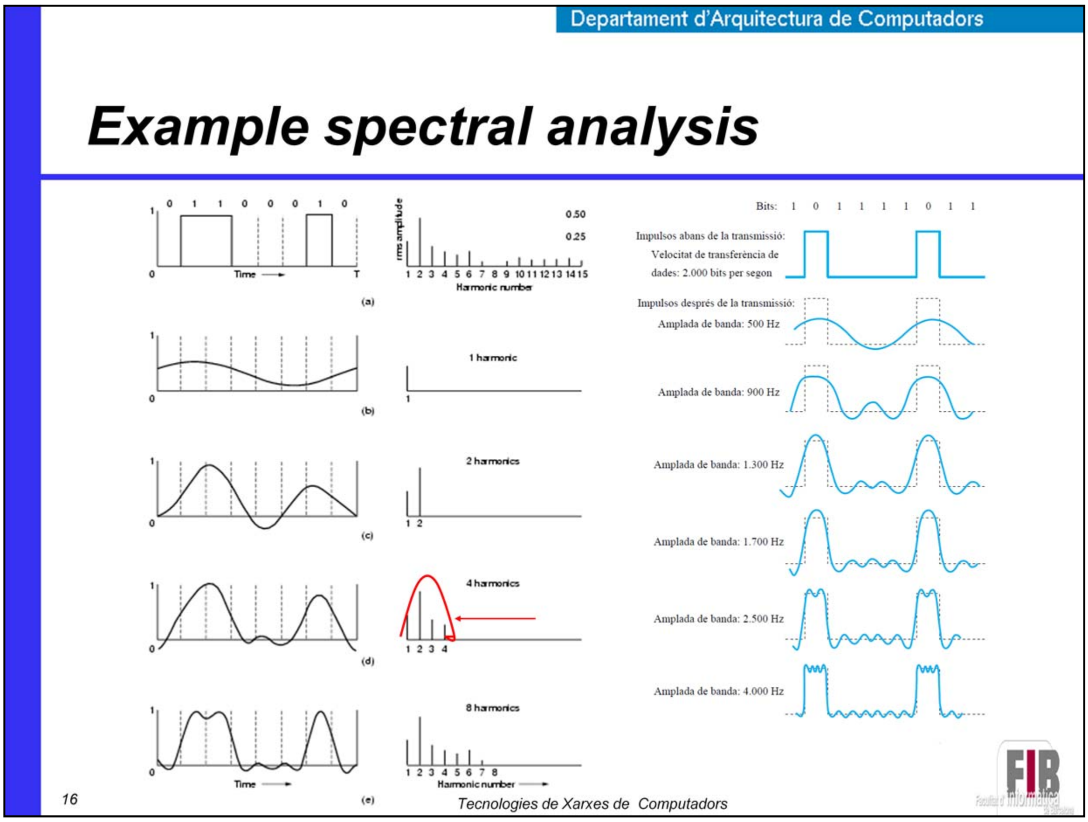
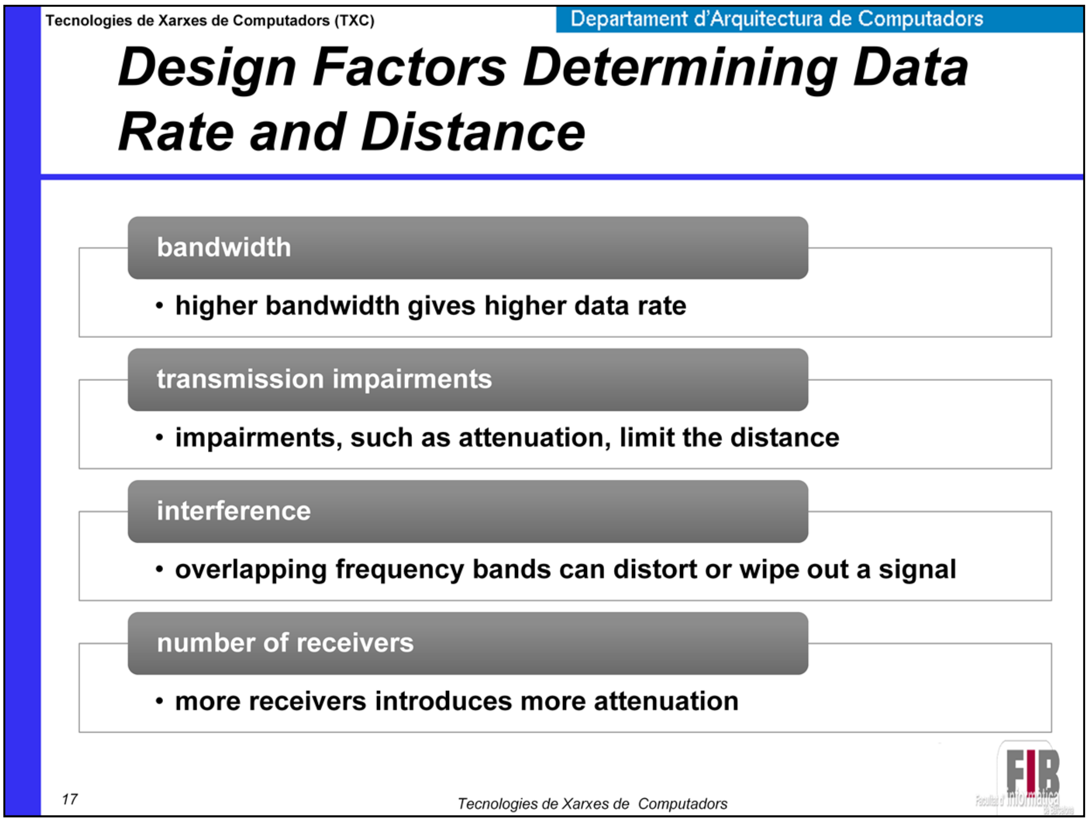
Contra més ampla de banda disponible, més velocitat de transmissió



Si transmetem un senyal quadrat com l'indicat a la primera figura, segons fourier, tindríem una distribució de freqüències com les indicades, tenint en compte que com que és un senyal periòdic la distribució de freqüències no inclou totes les freqüències sinó que només aquelles que són múltiples de la freq. Fonamental. Recordem que la freq. fonamental és l'invers del període.

A la figura dos se suposa que per el canal només passa la freq. Fonamental. Veiem que el senyal que arriba a la destinació només inclou aquesta freq. I per tant és molt diferent del senyal original i no es pot recuperar amb garanties. O sigui no es pot saber si el que arriba és un 1 o un 0 dins de cada símbol.

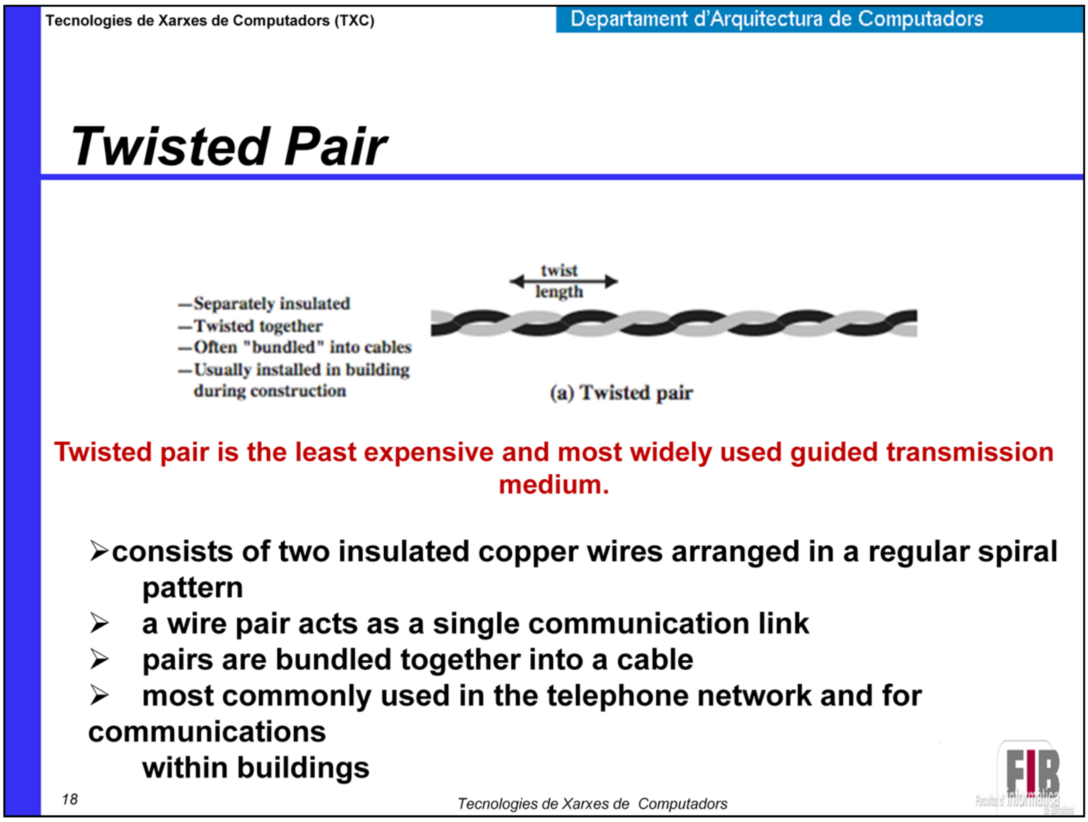
Així anem deixar passar més freq. en funció de la disponibilitat del canal i correspon als dibuixos posteriors. Contra més freq. pasen, cada cop el senyal que arriba se sembla més al original. Això determina quin és el canal mínim per a que el senyal que arriba permeti distingir clarament el 1 del 0.



Data rate and distance are the key considerations in data transmission system design; with emphasis placed on achieving the highest data rates over the longest distances. A number of design factors relating to the transmission medium and the signal determine the data rate and distance: **Bandwidth:** All other factors remaining constant, the greater the bandwidth of a signal, the higher the data rate that can be achieved.

Transmission impairments: Impairments, such as attenuation, limit the distance. For guided media, twisted pair generally suffers more impairment than coaxial cable, which in turn suffers more than optical fiber.

Interference: Interference from competing signals in overlapping frequency bands can distort or cancels out a signal. Interference is of particular concern for unguided media, but is also a problem with guided media. For guided media, interference can be caused by emanations coupling from nearby cables (alien crosstalk) or adjacent conductors under the same cable sheath (internal crosstalk). **Number of receivers:** A guided medium can be used to construct a point-to-point link or a shared link with multiple attachments. In the latter case, each attachment introduces some attenuation and distortion on the line, limiting distance and/or data rate.



The least expensive and most widely used guided transmission medium is twisted pair.

Unshielded vs. Shielded Twisted Pair

Unshielded Twisted Pair (UTP)

- ordinary telephone wire
- cheapest
- easiest to install
- suffers from external electromagnetic interference

Shielded Twisted Pair (STP)

- has metal braid or sheathing that reduces interference
- provides better performance at higher data rates
- more expensive
- harder to handle (thick, heavy)

Twisted pair comes in two varieties: unshielded and shielded. As the name implies, **unshielded twisted pair** (UTP) consists of one or more twisted-pair cables, typically enclosed within an overall thermoplastic jacket, which provides no electromagnetic shielding. The most common form of UTP is ordinary voice-grade telephone wire, which is pre-wired in residential and office buildings. For data transmission purposes, UTP may vary from voice-grade to very high-speed cable for local area networks (LANs). For high-speed LANs, UTP typically has four pairs of wires inside the jacket, with each pair twisted with a different number of twists per centimeter to help eliminate interference between adjacent pairs. The tighter the twisting, the higher the supported transmission rate and the greater the cost per meter. Unshielded twisted pair is subject to external electromagnetic interference, including interference from nearby twisted pair and from noise generated in the environment. In an environment with a number of sources of potential interference (e.g., electric motors, wireless devices, and RF transmitters), **shielded twisted pair** (STP) may be a preferred solution.

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UTP Categories

	Category 3 Class C	Category 5 Class D	Category 5E	Category 6 Class E	Category 7 Class F
Bandwidth	16 MHz	100 MHz	100 MHz	200 MHz	600 MHz
Cable Type	UTP	UTP/FTP	UTP/FTP	UTP/FTP	SSTP
Link Cost (Cat 5 =1)	0.7	1	1.2	1.5	2.2

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In response to the need to support higher speeds, EIA-568-A was issued in 1995. The new standard reflects advances in cable and connector design and test methods. It covers 150-ohm shielded twisted pair and 100-ohm unshielded twisted pair. EIA-568-A recognizes three categories of UTP cabling:

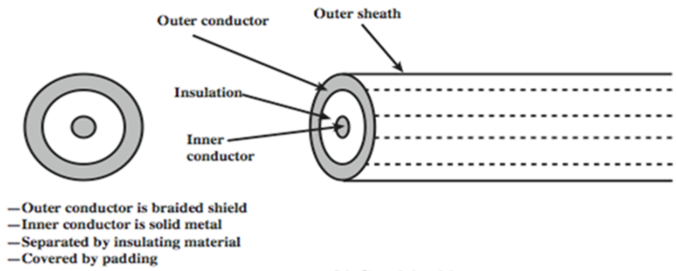
•**Category 3:** UTP cables and associated connecting hardware whose transmission characteristics are specified up to 16 MHz

•**Category 4:** UTP cables and associated connecting hardware whose transmission characteristics are specified up to 20 MHz

•**Category 5:** UTP cables and associated connecting hardware whose transmission characteristics are specified up to 100 MHz

Of these, it is Category 3 and Category 5 cable that have received the most attention for LAN applications. Category 3 corresponds to the voice-grade cable found in abundance in most office buildings. Over limited distances, and with proper design, data rates of up to 16 Mbps should be achievable with Category 3. Category 5 is a data-grade cable that is becoming standard for preinstallation in new office buildings. Over limited distances, and with proper design, data rates of up to 100 Mbps are achievable with Category 5. Since the publication of EIA-568-A, there has been ongoing work on the development of standards for premises cabling, driven by two issues. First, the Gigabit Ethernet specification requires the definition of parameters that are not specified completely in any published cabling standard. Second, there is a desire to specify cabling performance to higher levels, namely Enhanced Category 5 (Cat 5E), Category 6, and Category 7.

Coaxial Cable



(b) Coaxial cable

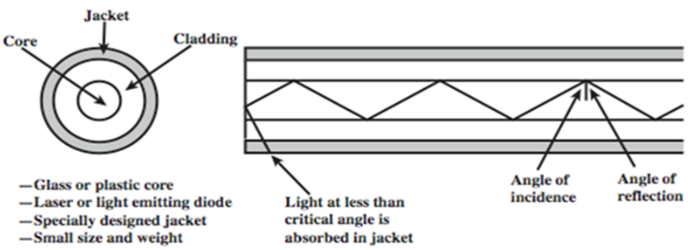
Coaxial cable can be used over longer distances and support more stations on a shared line than twisted pair.

- **consists of a hollow outer cylindrical conductor that surrounds a single inner wire conductor**
- **is a versatile transmission medium used in a wide variety of applications**
- **used for TV distribution, long distance telephone transmission and LANs**

Coaxial cable is widely used as a means of distributing TV signals to individual homes—cable TV. From its modest beginnings as Community Antenna Television (CATV), designed to provide service to remote areas, cable TV reaches almost as many homes and offices as the telephone. A cable TV system can carry dozens or even hundreds of TV channels at ranges up to a few tens of kilometers. Coaxial cable has traditionally been an important part of the long-distance telephone network. Today, it faces increasing competition from optical fiber, terrestrial microwave, and satellite. Using frequency division multiplexing (FDM, see Chapter 8), a coaxial cable can carry over 10,000 voice channels simultaneously. Coaxial cable is also commonly used for short-range connections between devices. Using digital signaling, coaxial cable can be used to provide high-speed I/O channels on computer systems.

Optical Fiber

Figure 4.2



(c) Optical fiber

Optical fiber is a thin flexible medium capable of guiding an optical ray.

- **various glasses and plastics can be used to make optical fibers**
- **has a cylindrical shape with three sections – core, cladding, jacket**
- **widely used in long distance telecommunications**
- **performance, price and advantages have made it popular to use**

An optical fiber is a thin (2 to 125 µm), flexible medium capable of guiding an optical ray. Various glasses and plastics can be used to make optical fibers. The lowest losses have been obtained using fibers of ultrapure fused silica. Ultrapure fiber is difficult to manufacture; higher-loss multicomponent glass fibers are more economical and still provide good performance. Plastic fiber is even less costly and can be used for short-haul links, for which moderately high losses are acceptable. An optical fiber cable has a cylindrical shape and consists of three concentric sections: the core, the cladding, and the jacket (Figure 4.2c). The **core** is the innermost section and consists of one or more very thin strands, or fibers, made of glass or plastic; the core has a diameter in the range of 8 to 50 µm. Each fiber is surrounded by its own **cladding**, a glass or plastic coating that has optical properties different from those of the core and a diameter of 125 µm. The interface between the core and cladding acts as a reflector to confine light that would otherwise escape the core. The outermost layer, surrounding one or a bundle of cladded fibers, is the **jacket**. The jacket is composed of plastic and other material layered to protect against moisture, abrasion, crushing, and other environmental dangers.

Optical Fiber - Benefits

- greater capacity
  - data rates of hundreds of Gbps
- smaller size and lighter weight
  - considerably thinner than coaxial or twisted pair cable
  - reduces structural support requirements
- lower attenuation
- electromagnetic isolation
  - not vulnerable to interference, impulse noise, or crosstalk
  - high degree of security from eavesdropping
- greater repeater spacing
  - lower cost and fewer sources of error

Five basic categories of application have become important for optical fiber: 1.Long-haul fiber transmission is becoming increasingly common in the telephone network. Long-haul routes average about 1500 km in length and offer high capacity (typically 20,000 to 60,000 voice channels). These systems compete economically with microwave and have so underpriced coaxial cable in many developed countries that coaxial cable is rapidly being phased out of the telephone network in such countries. Undersea optical fiber cables have also enjoyed increasing use. 2. Metropolitan trunking circuits have an average length of 12 km and may have as many as 100,000 voice channels in a trunk group. Most facilities are installed in underground conduits and are repeaterless, joining telephone exchanges in a metropolitan or city area. Included in this category are routes that link long-haul microwave facilities that terminate at a city perimeter to the main telephone exchange building downtown. 3. Rural exchange trunks have circuit lengths ranging from 40 to 160 km and link towns and villages. Most of these systems have fewer than 5000 voice channels. The technology used in these applications competes with microwave facilities. 4. Subscriber loop circuits are fibers that run directly from the central exchange to a subscriber. These facilities are beginning to displace twisted pair and coaxial cable links as the telephone networks evolve into full-service networks capable of handling not only voice and data, but also image and video. The initial penetration of optical fiber in this application has been for the business subscriber, but fiber transmission into the home is now a significant presence in many areas. 5. A final important application of optical fiber is for local area networks.

Optical Fiber Transmission Modes

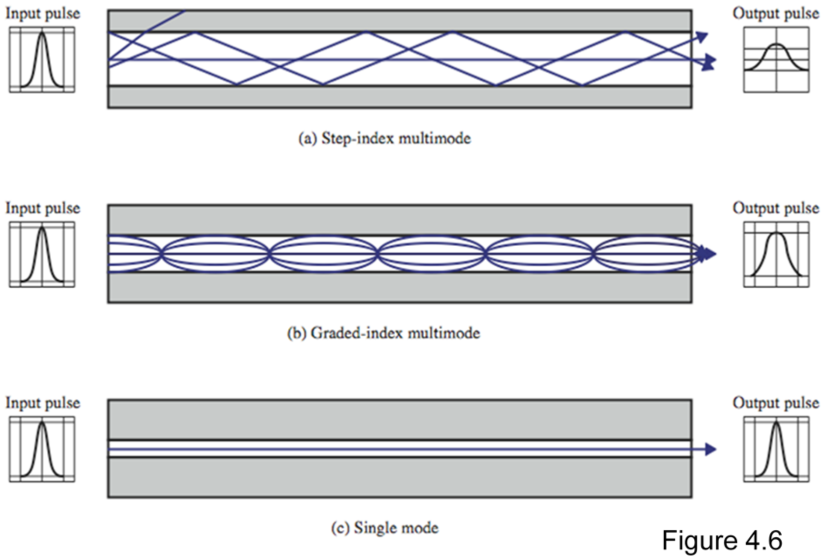
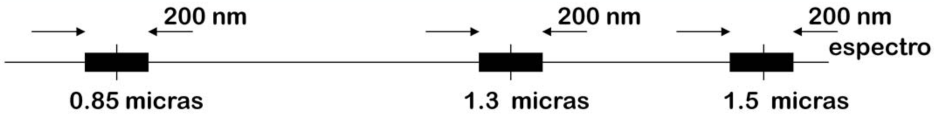


Figure 4.6

Figure 4.6 shows the principle of optical fiber transmission. Light from a source enters the cylindrical glass or plastic core. Rays at shallow angles are reflected and propagated along the fiber; other rays are absorbed by the surrounding material. This form of propagation is called **step-indexmultimode**, referring to the variety of angles that reflect. With multimode transmission, multiple propagation paths exist, each with a different path length and hence time to traverse the fiber. This causes signal elements (light pulses) to spread out in time, which limits the rate at which data can be accurately received. Put another way, the need to leave spacing between the pulses limits data rate. This type of fiber is best suited for transmission over very short distances. When the fiber core radius is reduced, fewer angles will reflect. By reducing the radius of the core to the order of a wavelength, only a single angle or mode can pass: the axial ray. This **single-mode** propagation provides superior performance for the following reason. Because there is a single transmission path with single-mode transmission, the distortion found in multimode cannot occur. Single-mode is typically used for long-distance applications, including telephone and cable television. Finally, by varying the index of refraction of the core, a third type of transmission, known as **graded-index multimode**, is possible.Two different types of light source are used in fiber optic systems: the light-emitting diode (LED) and the injection laser diode (ILD). Both are semiconductor devices that emit a beam of light when a voltage is applied. The LED is less costly, operates over a greater temperature range, and has a longer operational life. The ILD, which operates on the laser principle, is more efficient and can sustain greater data rates.

Departament d'Arquitectura de Computadors

Frequency Utilization for Fiber Applications



Wavelength (in vacuum) range (nm)	Frequency Range (THz)	Band Label	Fiber Type	Application
820 to 900	366 to 333		Multimode	LAN
1280 to 1350	234 to 222	S	Single mode	Various
1528 to 1561	196 to 192	C	Single mode	WDM
1561 to 1620	192 to 185	L	Single mode	WDM

Table 4.3

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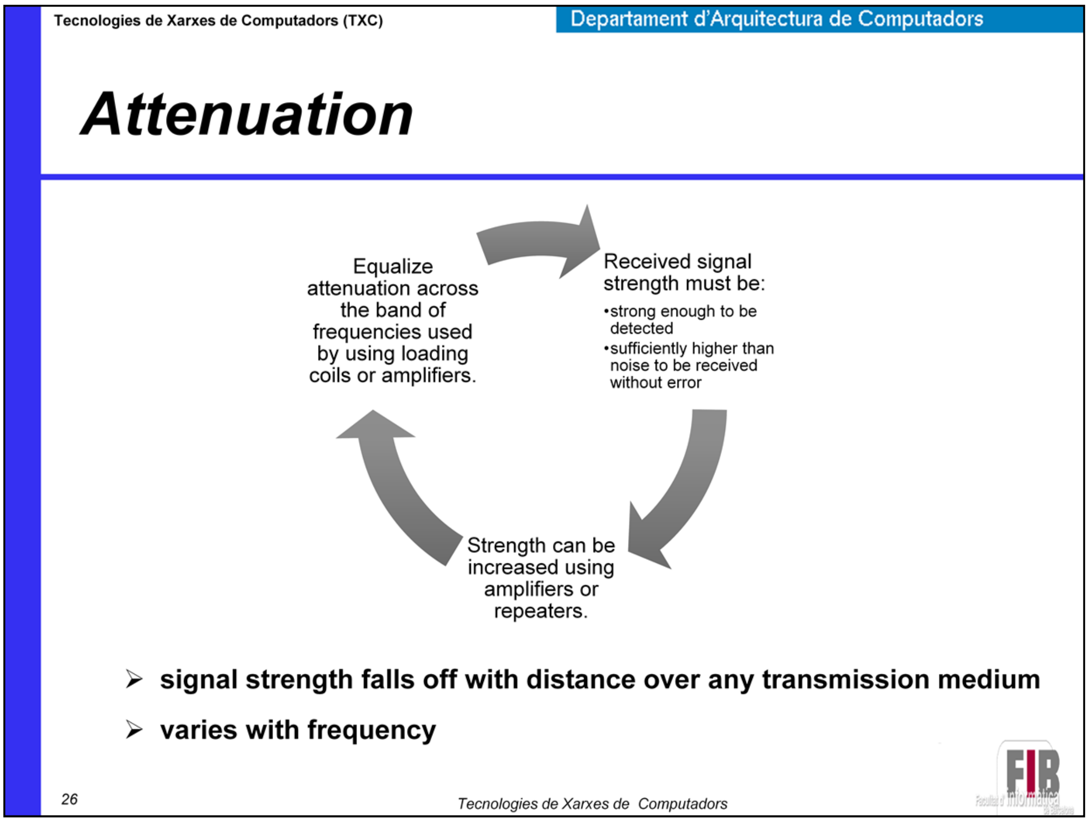
In optical fiber, based on the attenuation characteristics of the medium and on properties of light sources and receivers, four transmission windows are appropriate, as shown in Table 4.3. Note the tremendous bandwidths available. For the four windows, the respective bandwidths are 33 THz, 12 THz, 4 THz, and 7 THz. This is several orders of magnitude greater than the bandwidth available in the radio-frequency spectrum. The four transmission windows are in the infrared portion of the frequency spectrum, below the visible-light portion, which is 400 to 700 nm. The loss is lower at higher wavelengths, allowing greater data rates over longer distances. Many local applications today use 850-nm LED light sources. Although this combination is relatively inexpensive, it is generally limited to data rates under 100 Mbps and distances of a few kilometers. To achieve higher data rates and longer distances, a 1300-nm LED or laser source is needed. The highest data rates and longest distances require 1500-nm laser sources. nb. WDM = wavelength division multiplexing (see Chapter 8).



The strength of a signal falls off with distance over any transmission medium. For guided media (e.g., twisted-pair wire, optical fiber), this reduction in strength, or attenuation, is generally exponential and thus is typically expressed as a constant number of decibels per unit distance. For unguided media (wireless transmission), attenuation is a more complex function of distance and the makeup of the atmosphere. Attenuation introduces three considerations for the transmission engineer. **1.**A received signal must have sufficient strength so that the electronic circuitry in the receiver can detect and interpret the signal. **2.**The signal must maintain a level sufficiently higher than noise to be received without error. **3.**Attenuation is greater at higher frequencies, and this causes distortion.

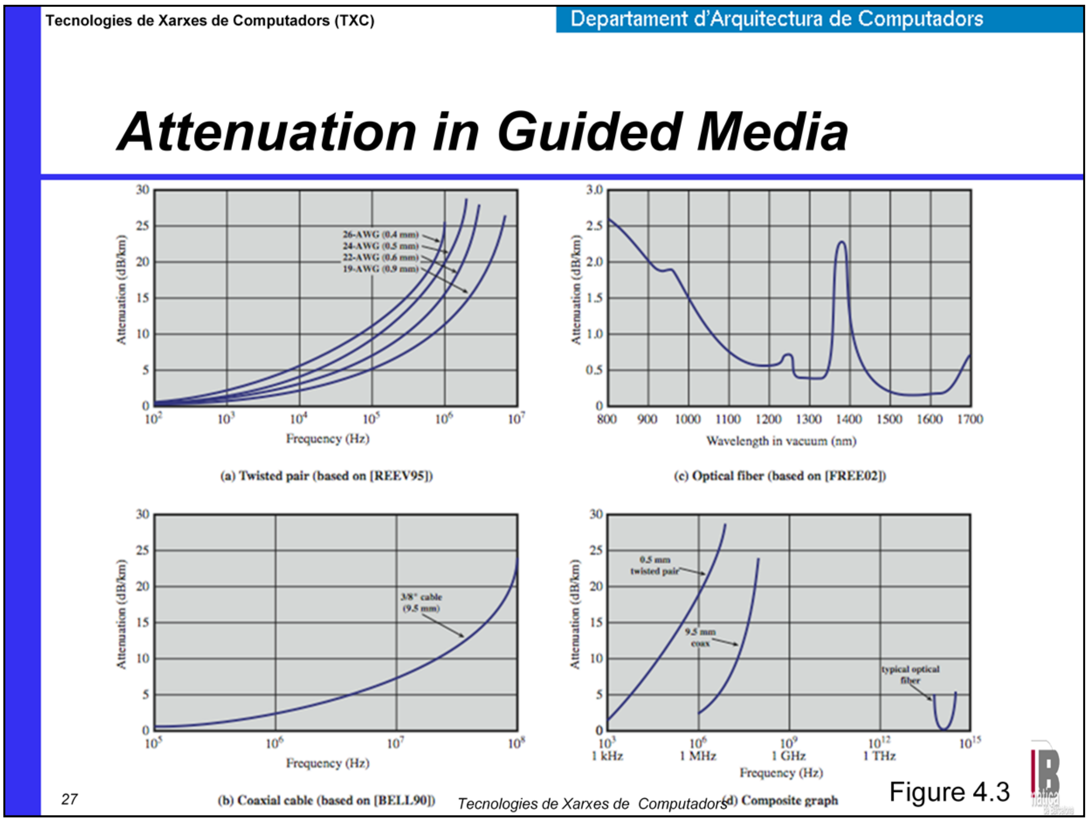


Figure 4.3 shows attenuation versus wavelength for the various types of wired media we have discussed.

Fig 4.3a shows that attenuation for twisted pair is a very strong function of frequency.

Fig 4.3b shows, coaxial cable has frequency characteristics that are superior to those of twisted pair and can hence be used effectively at higher frequencies and data rates.

Fig 4.3c shows the attenuation vs. wavelength for a typical optical fiber. The unusual shape of the curve is due to the combination of a variety of factors that contribute to attenuation. The two most important of these are absorption and scattering, which is the change in direction of light rays after they strike small particles or impurities in the medium.

Attenuation Distortion

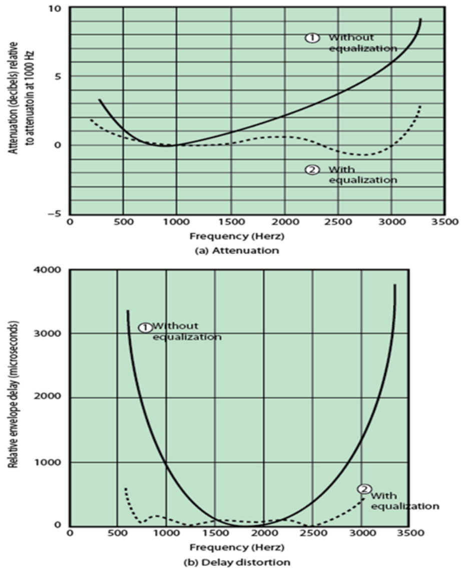


Figure 3.14

An example is provided in Figure 3.14a, which shows attenuation as a function of frequency for a typical leased line. In the figure, attenuation is measured relative to the attenuation at 1000 Hz. Positive values on the *y*-axis represent attenuation greater than that at 1000 Hz. A 1000-Hz tone of a given power level is applied to the input, and the power, *P*1000, is measured at the output.

The solid line in Figure 3.14a shows attenuation without equalization. As can be seen, frequency components at the upper end of the voice band are attenuated much more than those at lower frequencies. It should be clear that this will result in a distortion of the received speech signal. The dashed line shows the effect of equalization. The flattened response curve improves the quality of voice signals. It also allows higher data rates to be used for digital data that are passed through a modem.

Attenuation distortion can present less of a problem with digital signals. As we have seen, the strength of a digital signal falls off rapidly with frequency (Figure 3.5b); most of the content is concentrated near the fundamental frequency or bit rate of the signal.

Delay Distortion

- occurs because propagation velocity of a signal through a guided medium varies with frequency
- various frequency components arrive at different times resulting in phase shifts between the frequencies
- particularly critical for digital data since parts of one bit spill over into others causing intersymbol interference

Delay distortion is a phenomenon that occurs in transmission cables (such as twisted pair, coaxial cable, and optical fiber); it does not occur when signals are transmitted through the air by means of antennas. Delay distortion is caused by the fact that the velocity of propagation of a signal through a cable is different for different frequencies. For a signal with a given bandwidth, the velocity tends to be highest near the center frequency of the signal and to fall off toward the two edges of the band. Thus, various components of a signal will arrive at the receiver at different times.

This effect is referred to as delay distortion because the received signal is distorted due to varying delays experienced at its constituent frequencies. Delay distortion is particularly critical for digital data. Consider that a sequence of bits is being transmitted, using either analog or digital signals. Because of delay distortion, some of the signal components of one bit position will spill over into other bit positions, causing **intersymbol interference**, which is a major limitation to maximum bit rate over a transmission channel. Equalizing techniques can also be used for delay distortion. Again using a leased telephone line as an example, Figure 3.14b shows the effect of equalization on delay as a function of frequency.

Satellite Point-to-Point Link

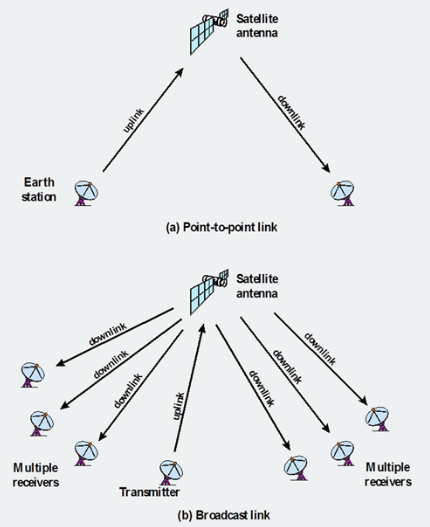


Figure 4.9 Satellite Communication Configurations

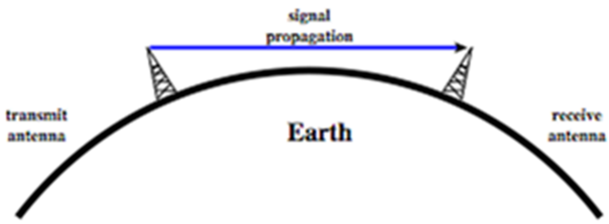
Figure 4.9 depicts in a general way two common configurations for satellite communication. In the first, the satellite is being used to provide a point-to-point link between two distant ground-based antennas.

In the second, the satellite provides communications between one ground-based transmitter and a number of ground-based receivers.

For a communication satellite to function effectively, it is generally required that it remain stationary with respect to its position over the earth. Otherwise, it would not be within the line of sight of its earth stations at all times. To remain stationary, the satellite must have a period of rotation equal to the earth's period of rotation. This match occurs at a height of 35,863 km at the equator.

Two satellites using the same frequency band, if close enough together, interfere with each other. To avoid this, current standards require a 4° spacing (angular displacement as measured from the earth) in the 4/6-GHz band and a 3° spacing at 12/14 GHz. Thus the number of possible satellites is quite limited.

Wireless Propagation Line of Sight

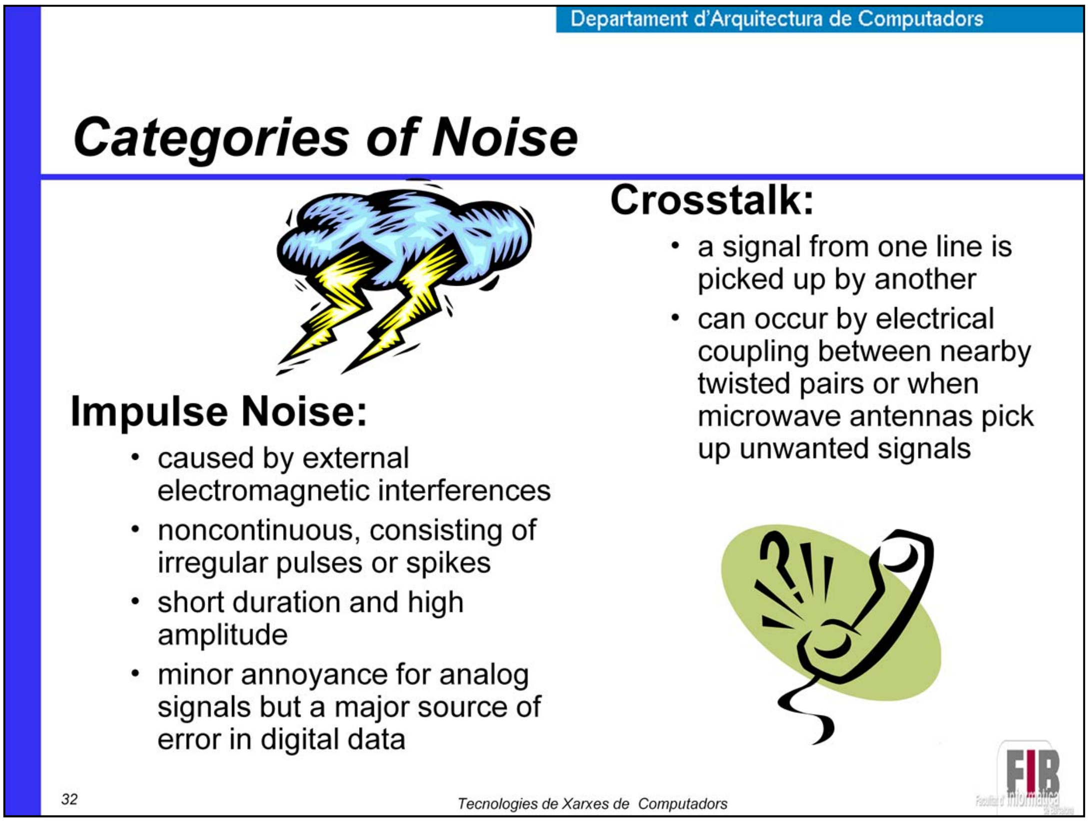


(c) Line-of-sight (LOS) propagation (above 30 MHz)

- ground and sky wave propagation modes do not operate above 30 MHz - - communication must be by line of sight

Figure 4.11

Above 30 MHz, neither ground wave nor sky wave propagation modes operate, and communication must be by line of sight(Figure 4.11c). For satellite communication, a signal above 30 MHz is not reflected by the ionosphere and therefore a signal can be transmitted between an earth station and a satellite overhead that is not beyond the horizon. For ground-based communication, the transmitting and receiving antennas must be within an *effective* line of sight of each other. The term *effective* is used because microwaves are bent or refracted by the atmosphere. The amount and even the direction of the bend depend on conditions, but generally microwaves are bent with the curvature of the earth and will therefore propagate farther than the optical line of sight.



Crosstalk has been experienced by anyone who, while using the telephone, has been able to hear another conversation; it is an unwanted coupling between signal paths. It can occur by electrical coupling between nearby twisted pairs or, rarely, coax cable lines carrying multiple signals. Crosstalk can also occur when microwave antennas pick up unwanted signals; although highly directional antennas are used, microwave energy does spread during propagation. Typically, crosstalk is of the same order of magnitude as, or less than, thermal noise.

All of the types of noise discussed so far have reasonably predictable and relatively constant magnitudes. Thus it is possible to engineer a transmission system to cope with them. **Impulse noise**, however, is noncontinuous, consisting of irregular pulses or noise spikes of short duration and of relatively high amplitude. It is generated from a variety of causes, including external electromagnetic disturbances, such as lightning, and faults and flaws in the communications system. Impulse noise is generally only a minor annoyance for analog data. For example, voice transmission may be corrupted by short clicks and crackles with no loss of intelligibility. However, impulse noise is the primary source of error in digital data communication. For example, a sharp spike of energy of 0.01 s duration would not destroy any voice data but would wash out about 560 bits of digital data being transmitted at 56 kbps

Nyquist Bandwidth

- Difference between symbol/s and b/s

In the case of a channel that is noise free:

- if rate of signal transmission is *2B* then can carry signal with frequencies no greater than *B*
  - given bandwidth B, highest signal rate is 2B
- for binary signals, *2B* bps needs bandwidth *B* Hz
- can increase rate by using *M* signal levels
- Nyquist Formula: $C\ (b/s) = 2B\ (symbol/s)\ \log_2 M$
- data rate can be increased by increasing signals
  - however this increases burden on receiver
  - noise & other impairments limit the value of *M*

To begin, let us consider the case of a channel that is noise free. In this environment, the limitation on data rate is simply the bandwidth of the signal. A formulation of this limitation, due to Nyquist, states that if the rate of signal transmission is 2*B*, then a signal with frequencies no greater than *B* is sufficient to carry the signal rate. The converse is also true: Given a bandwidth of *B*, the highest signal rate that can be carried is 2*B*. This limitation is due to the effect of intersymbol interference, such as is produced by delay distortion. The result is useful in the development of digital-to-analog encoding schemes and is, in essence, based on the same derivation as that of the sampling theorem, described in Appendix G. Note that in the preceding paragraph, we referred to signal rate. If the signals to be transmitted are binary (two voltage levels), then the data rate that can be supported by *B* Hz is 2*B* bps. However, as we shall see in Chapter 5, signals with more than two levels can be used; that is, each signal element can represent more than one bit. For example, if four possible voltage levels are used as signals, then each signal element can represent two bits. With multilevel signaling, the Nyquist formulation becomes $C = 2B \log2 M$ where *M* is the number of discrete signal or voltage levels.

So, for a given bandwidth, the data rate can be increased by increasing the number of different signal elements. However, this places an increased burden on the receiver: Instead of distinguishing one of two possible signal elements during each signal time, it must distinguish one of *M* possible signal elements. Noise and other impairments on the transmission line will limit the practical value of *M*.

Shannon Capacity Formula

- considering the relation of data rate, noise and error rate:
  - faster data rate shortens each bit so bursts of noise corrupts more bits
  - given noise level, higher rates mean higher errors
- Shannon developed formula relating these to signal to noise ratio (in decibels)
- SNR_{db}=10 $\log_{10}$ (signal/noise)
- capacity $C = B \log_2(1+SNR)$
  - theoretical maximum capacity
  - get much lower rates in practice

Nyquist's formula indicates that, all other things being equal, doubling the bandwidth doubles the data rate. Now consider the relationship among data rate, noise, and error rate. The presence of noise can corrupt one or more bits. If the data rate is increased, then the bits become "shorter" so that more bits are affected by a given pattern of noise. If the data rate is increased, then more bits will occur during the interval of a noise spike, and hence more errors will occur. A high SNR will mean a high-quality signal and a low number of required intermediate repeaters.

The signal-to-noise ratio is important in the transmission of digital data because it sets the upper bound on the achievable data rate. Shannon's result is that the maximum channel capacity, in bits per second, obeys the equation $C = B \log2 (1 + SNR)$ where C is the capacity of the channel in bits per second and B is the bandwidth of the channel in hertz. The Shannon formula represents the theoretical maximum that can be achieved. In practice, however, only much lower rates are achieved. One reason for this is that the formula assumes white noise (thermal noise). Impulse noise is not accounted for, nor are attenuation distortion or delay distortion. Even in an ideal white noise environment, present technology still cannot achieve Shannon capacity due to encoding issues, such as coding length and complexity.

Capítol 5.1 Stallings.

La codificació de senyals permet transmetre senyals digitals en línia ocupant tot l'ampla de banda del medi de transmissió, si això no és un problema.

Proporciona avantatges en relació a la relació senyal soroll, al sincronisme i al control d'errors.

Signal Encoding Techniques

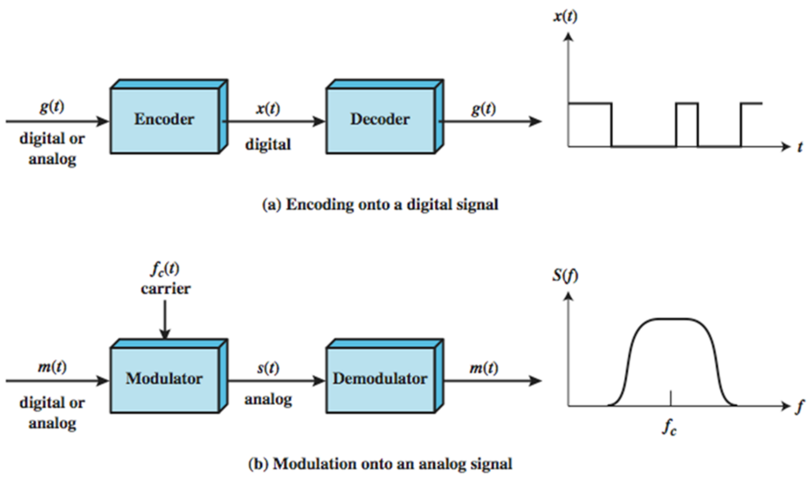


Figure 5.1 Encoding and Modulation Techniques

Have already noted in Ch 3 that both analog and digital information can be encoded as either analog or digital signals:

♦**Digital data, digital signals:** simplest form of digital encoding of digital data

♦**Digital data, analog signal:** A modem converts digital data to an analog signal so that it can be transmitted over an analog

♦**Analog data, digital signals:** Analog data, such as voice and video, are often digitized to be able to use digital transmission facilities

♦**Analog data, analog signals:** Analog data are modulated by a carrier frequency to produce an analog signal in a different frequency band, which can be utilized on an analog transmission system

Fig 5.1 emphasizes the process involved in this. For **digital signaling**, a data source $g(t)$, which may be either digital or analog, is encoded into a digital signal $x(t)$. The basis for **analog signaling** is a continuous constant-frequency f_c signal known as the **carrier signal**. Data may be transmitted using a carrier signal by modulation, which is the process of encoding source data onto the carrier signal. All modulation techniques involve operation on one or more of the three fundamental frequency domain parameters: amplitude, frequency, and phase. The input signal $m(t)$ may be analog or digital and is called the modulating signal, and the result of modulating the carrier signal is called the modulated signal $s(t)$.

Comparison of Encoding Schemes

- signal spectrum
- clocking
- error detection
- signal interference and noise immunity
- cost and complexity

Before describing the various encoding techniques, consider the following ways of evaluating or comparing them:

•Signal Spectrum - Lack of high frequencies reduces required bandwidth, lack of dc component allows ac coupling via transformer, providing isolation, should concentrate power in the middle of the bandwidth

•Clocking - need for synchronizing transmitter and receiver either with an external clock or with a sync mechanism based on signal

•Error detection - useful if can be built in to signal encoding

•Signal interference and noise immunity - some codes are better than others

•Cost and complexity - Higher signal rate (& thus data rate) lead to higher costs, some codes require signal rate greater than data rate

Encoding Schemes

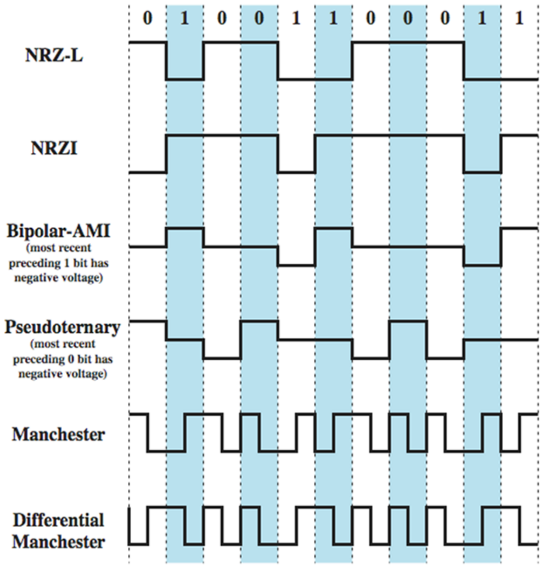
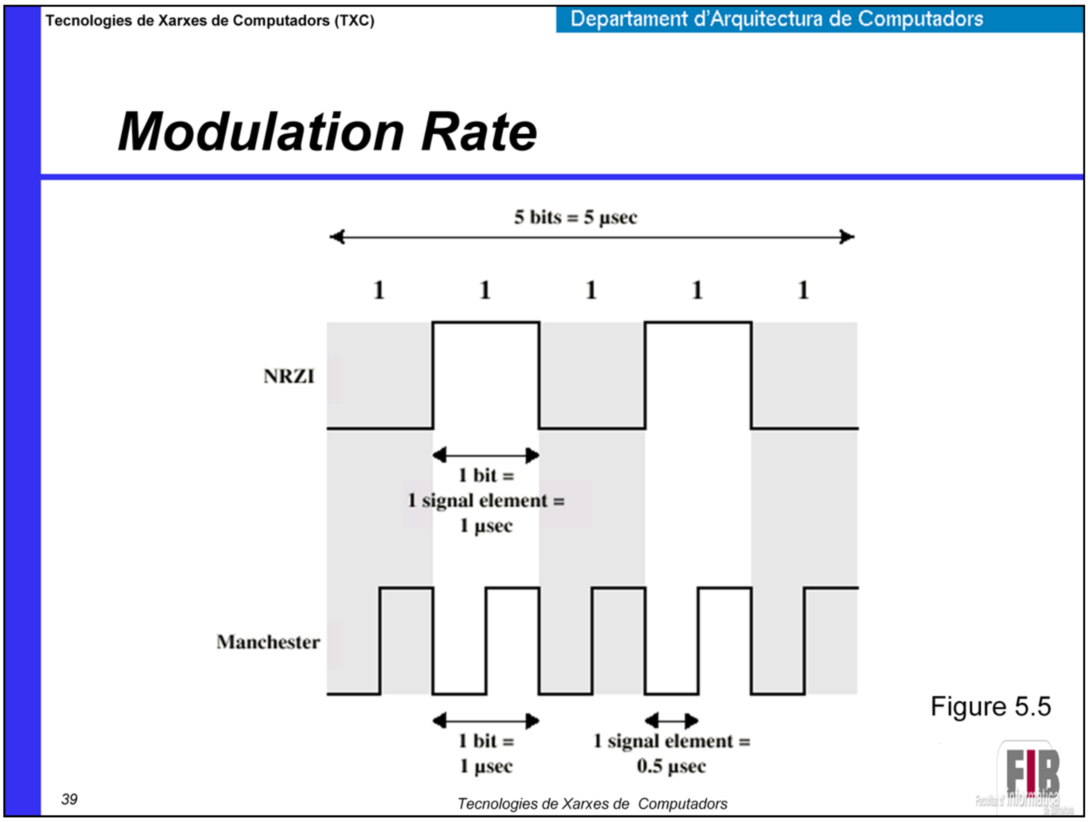
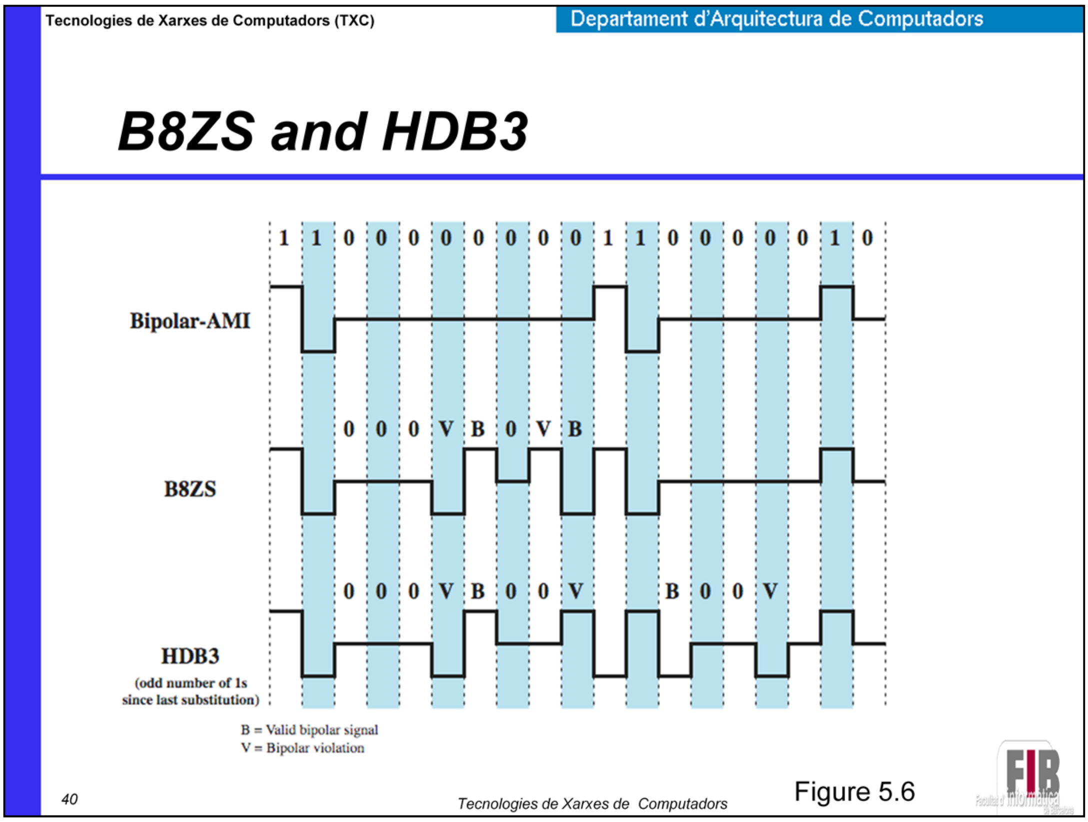


Figure 5.2

We now turn to a discussion of various techniques, which are defined in Table 5.2 and depicted in Figure 5.2 as shown above.



When signal-encoding techniques are used, a distinction needs to be made between data rate (expressed in bits per second) and modulation rate (expressed in baud). The data rate, or bit rate, is $1/T_b$, where T_b = bit duration. The modulation rate is the rate at which signal elements are generated. Consider, for example, Manchester encoding. The minimum size signal element is a pulse of one-half the duration of a bit interval. For a string of all binary zeroes or all binary ones, a continuous stream of such pulses is generated. Hence the maximum modulation rate for Manchester is $2/T_b$. This situation is illustrated in Figure 5.5, which shows the transmission of a stream of binary 1s at a data rate of 1 Mbps using NRZI and Manchester. One way of characterizing the modulation rate is to determine the average number of transitions that occur per bit time. In general, this will depend on the exact sequence of bits being transmitted. Table 5.3 compares transition rates for various techniques.



Two techniques are commonly used in long-distance transmission services; these are illustrated in Figure 5.6.

A coding scheme that is commonly used in North America, based on a bipolar-AMI, is known as **bipolar with 8-zeros substitution (B8ZS).** To overcome the drawback of the AMI code that a long string of zeros may result in loss of synchronization, the encoding is amended with the following rules:

•If an octet of all zeros occurs and the last voltage pulse preceding this octet was positive, then the eight zeros of the octet are encoded as 000+–0–+.

•If an octet of all zeros occurs and the last voltage pulse preceding this octet was negative, then the eight zeros of the octet are encoded as 000–+0+–.

This technique forces two code violations (signal patterns not allowed in AMI) of the AMI code, an event unlikely to be caused by noise or other transmission impairment. The receiver recognizes the pattern and interprets the octet as consisting of all zeros. A coding scheme that is commonly used in Europe and Japan is known as the **high-density bipolar-3 zeros (HDB3)** code. It is also based on the use of AMI encoding. In this case, the scheme replaces strings of four zeros with sequences containing one or two pulses. In each case, the fourth zero is replaced with a code violation. In addition, a rule is needed to ensure that successive violations are of alternate polarity so that no dc component is introduced. Thus, if the last violation was positive, this violation must be negative and vice versa.

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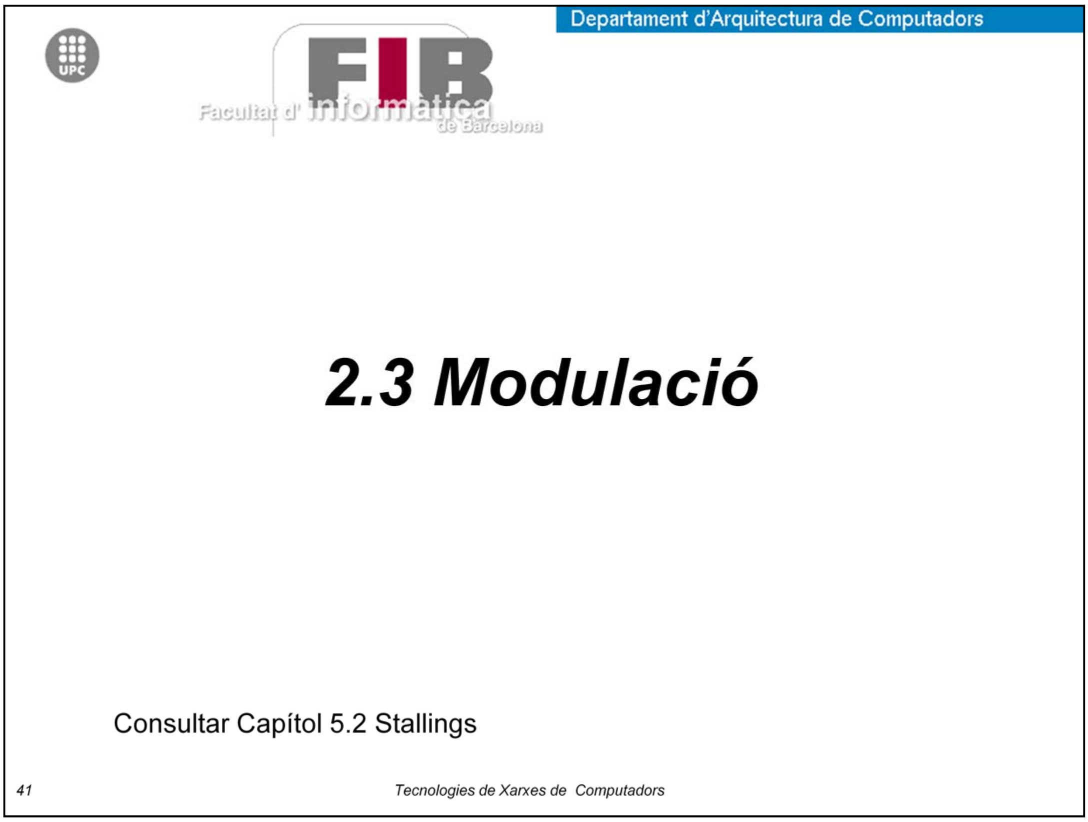


Capítol 5.2 Stallings.

La modulació permet adaptar el senyal que s'està transmetent a l'ampla de banda del medi que s'està utilitzant. És útil quan el canal utilitzabe està limitat en freqüències.

Modulation Techniques

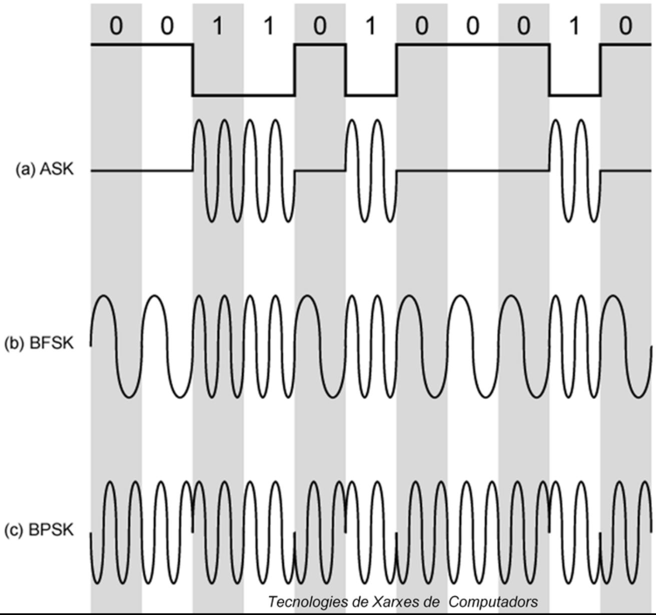


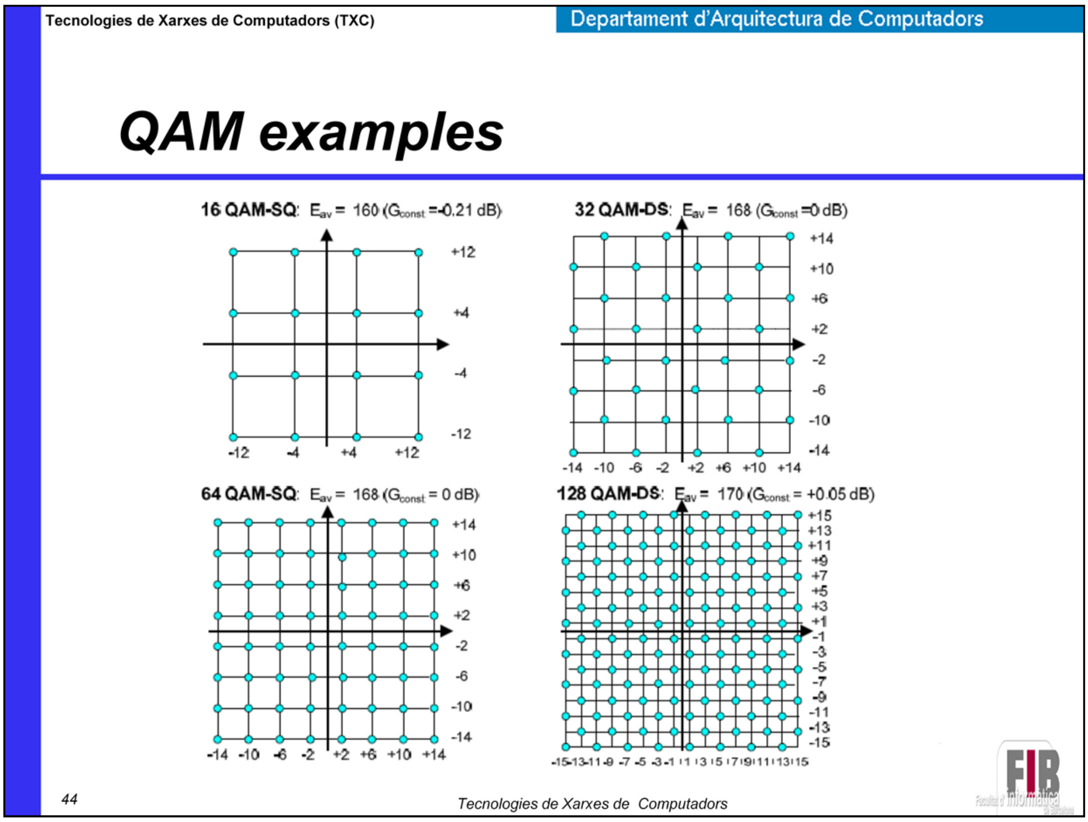
Figure 5.7

Have stated that modulation involves operation on one or more of the three characteristics of a carrier signal: amplitude, frequency, and phase. Accordingly, there are three basic encoding or modulation techniques for transforming digital data into analog signals, as illustrated in Figure 5.7 (above): amplitude shift keying (ASK), frequency shift keying (FSK), and phase shift keying (PSK). In all these cases, the resulting signal occupies a bandwidth centered on the carrier frequency.

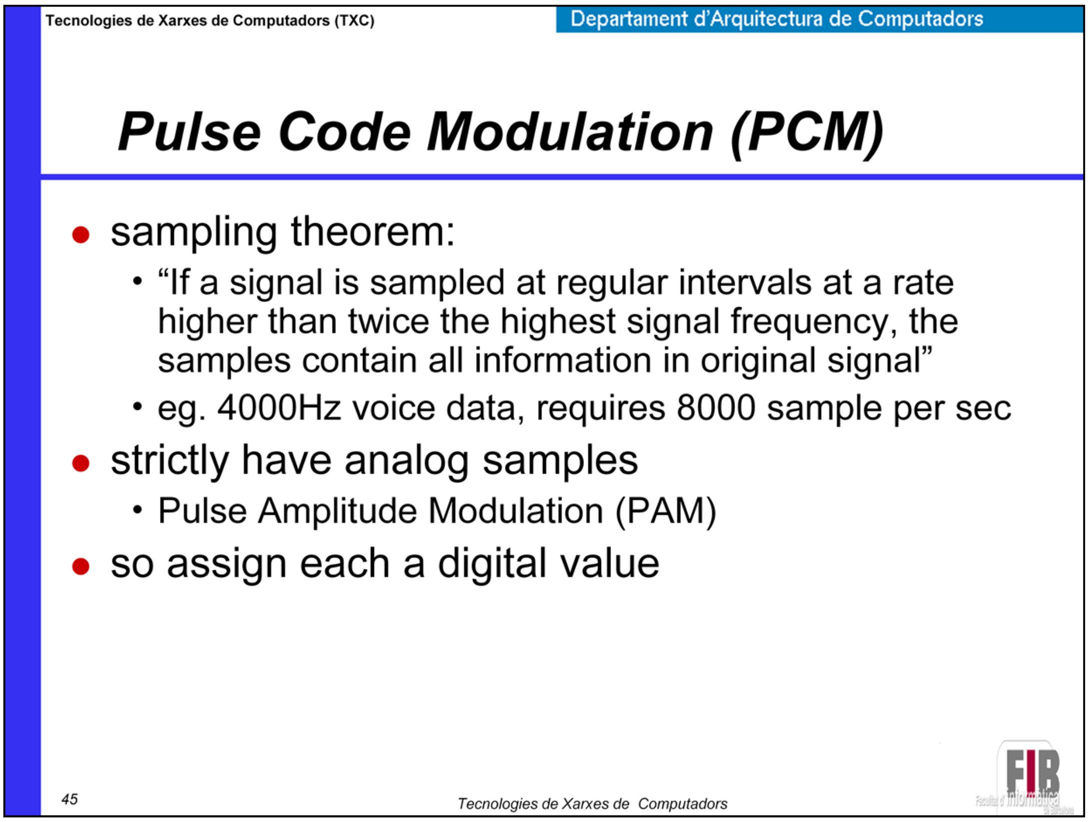
Quadrature Amplitude Modulation

- QAM used on asymmetric digital subscriber line (ADSL) and some wireless
- combination of ASK and PSK
- send two different signals simultaneously on same carrier frequency
  - use two copies of carrier, one shifted 90°
  - each carrier is ASK modulated
  - two independent signals over same medium
  - demodulate and combine for original binary output

Quadrature amplitude modulation (QAM) is a popular analog signaling technique that is used in the asymmetric digital subscriber line (ADSL), described in Chapter 8, and in some wireless standards. This modulation technique is a combination of ASK and PSK. QAM can also be considered a logical extension of QPSK. QAM takes advantage of the fact that it is possible to send two different signals simultaneously on the same carrier frequency, by using two copies of the carrier frequency, one shifted by 90° with respect to the other. For QAM, each carrier is ASK modulated. The two independent signals are simultaneously transmitted over the same medium. At the receiver, the two signals are demodulated and the results combined to produce the original binary input.

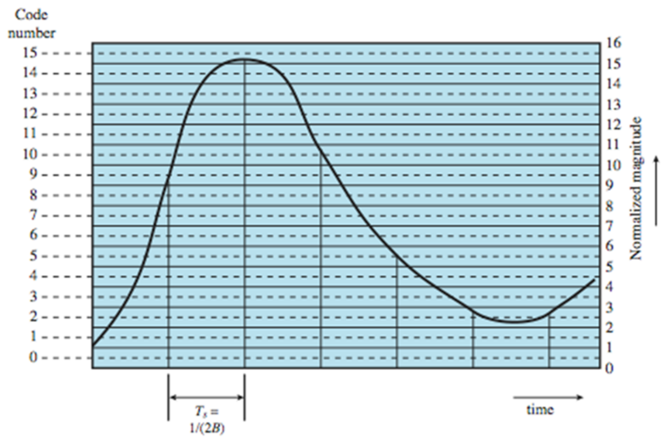


Cada punt reprenta un símbol diferent amb una amplada i una fase diferent amb la mateixa freq. per tots ells. En funció del nombre de punts disponibles podem codificar cada un amb un nombre de bits. 16 QAM vol dir 16 símbols diferents i cada símbol es pot codificar amb 4 bits. Mantenint el nombre de símbols per segon (depén de l'ampla de banda disponible) podem tenir més velocitat de transmissió en bits/seg. Un símbol n bits. El límit ve donat per la relació senyal/soroll que pot fer confondre un símbol amb un altra en el receptor.



The simplest technique for transforming analog data into digital signals is pulse code modulation (PCM), which involves sampling the analog data periodically and quantizing the samples. Pulse code modulation (PCM) is based on the sampling theorem (quoted above). Hence if voice data is limited to frequencies below 4000 Hz (a conservative procedure for intelligibility), 8000 samples per second would be sufficient to characterize the voice signal completely. Note, however, that these are analog samples, called **pulse amplitude modulation (PAM)** samples. To convert to digital, each of these analog samples must be assigned a binary code.

PCM Example



PAM value	1.1	9.2	15.2	10.8	5.6	2.8	2.7
quantized code number	1	9	15	10	5	2	2
PCM code	0001	1001	1111	1010	0101	0010	0010

Figure 5.17

Figure 5.17 shows an example in which the original signal is assumed to be bandlimited with a bandwidth of B. PAM samples are taken at a rate of $2B$, or once every $T_s = 1/2B$ seconds. Each PAM sample is approximated by being *quantized* into one of 16 different levels. Each sample can then be represented by 4 bits. But because the quantized values are only approximations, it is impossible to recover the original signal exactly. By using an 8-bit sample, which allows 256 quantizing levels, the quality of the recovered voice signal is comparable with that achieved via analog transmission. Note that this implies that a data rate of 8000 samples per second × 8 bits per sample = 64 kbps is needed for a single voice signal.

Departament d'Arquitectura de Computadors

Facultat d' informàtica de Barcelona

2.4 Multiplexació

Consultar Capítol 8.1 i 8.2 Stallings

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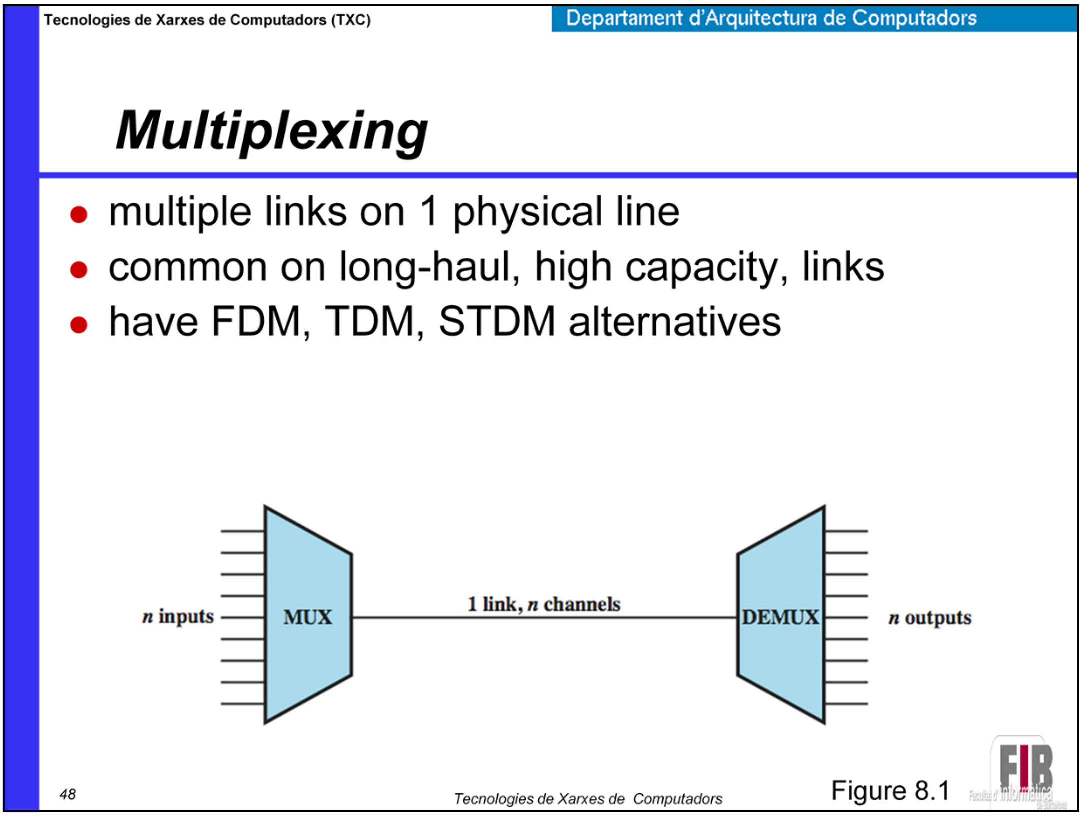
La multiplexació permet compartir medis de transmissió per diferents comunicacions de dades (usuaris).

Capítol 8.1 i 8.2 Stallings.

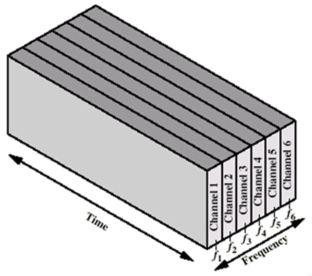


To make efficient use of high-speed telecommunications lines, some form of multiplexing is used. Multiplexing allows several transmission sources to share a larger transmission capacity. A common application of multiplexing is in long-haul communications. Trunks on long-haul networks are high-capacity fiber, coaxial, or microwave links. These links can carry large numbers of voice and data transmissions simultaneously using multiplexing. Common forms of multiplexing are frequency division multiplexing (FDM), time division multiplexing (TDM), and statistical TDM (STDM).

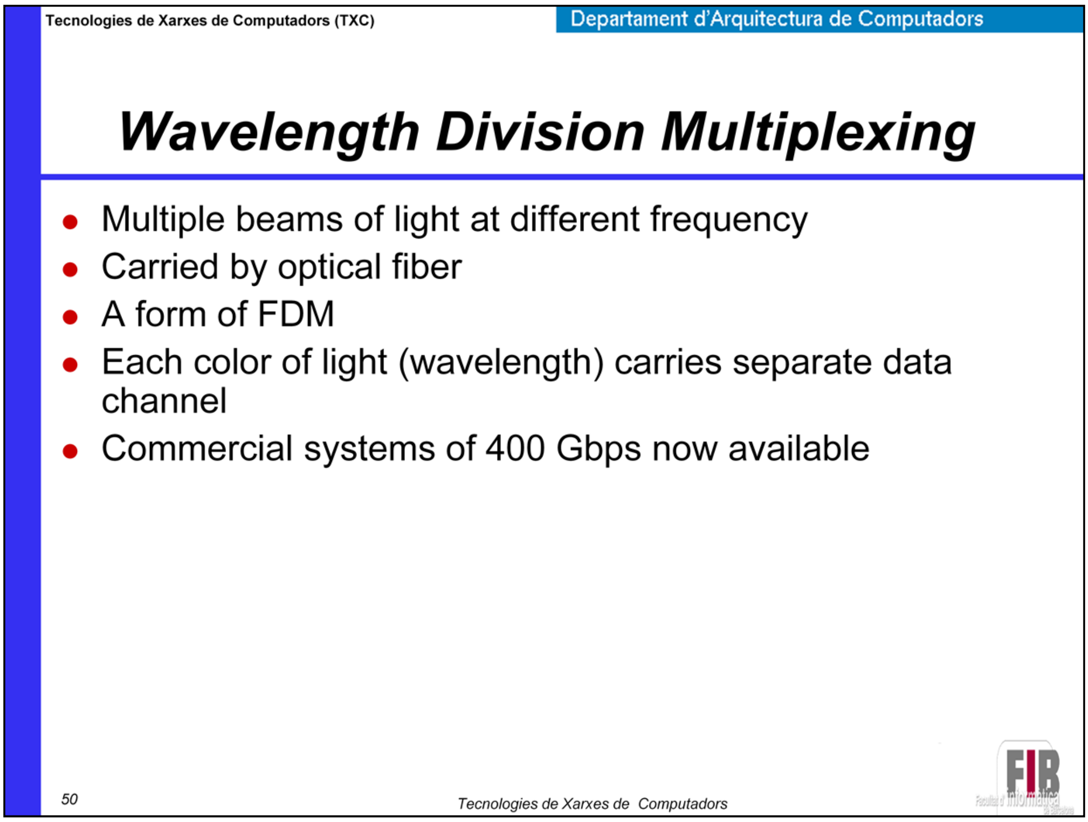
Figure 8.1 depicts the multiplexing function in its simplest form. There are *n* inputs to a multiplexer. The multiplexer is connected by a single data link to a demultiplexer. The link is able to carry *n* separate channels of data. The multiplexer combines (multiplexes) data from the *n* input lines and transmits over a higher-capacity data link. The demultiplexer accepts the multiplexed data stream, separates (demultiplexes) the data according to channel, and delivers data to the appropriate output lines.

Frequency Division Multiplexing

- FDM
- Useful bandwidth of medium exceeds required bandwidth of channel
- Each signal is modulated to a different carrier frequency
- Carrier frequencies separated so signals do not overlap (guard bands)
- e.g. broadcast radio
- Channel allocated even if no data



Cada usuari té un grup de freq. diferents durant tot el temps.

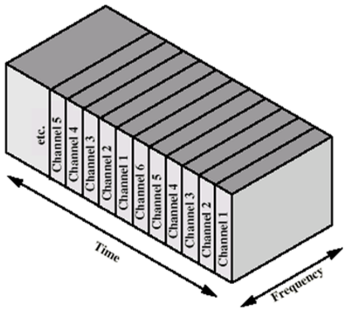


Multiplexació per divisió de la longitud d'onda. És el mateix que la multiplexació per divisió de freq. Però en longitud d'onda. Cada usuari té una longitud d'onda diferent tot el temps. Exemple la llum blanca porta tots els colors. Cada color és una longitud d'onda diferent.

Tecnologies de Xarxes de Computadors (TXC) Departament d'Arquitectura de Computadors

Time Division Multiplexing

- Data rate of medium exceeds data rate of digital signal to be transmitted
- Multiple digital signals interleaved in time
- May be at bit level of blocks
- Time slots preassigned to sources and fixed
- Time slots allocated even if no data
- Time slots do not have to be evenly distributed amongst sources



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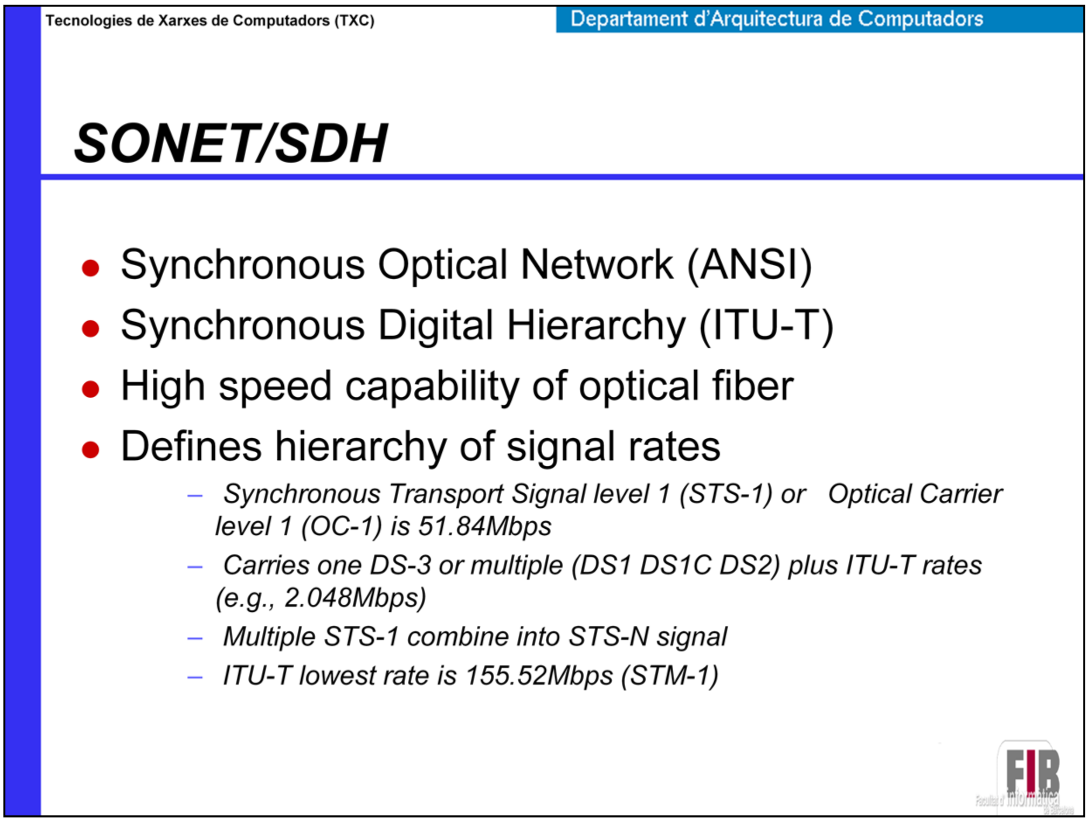
Cada usuari té totes les freq. durant una part del temps. Es diuen ranures temporals o slots. O sigui cada usuari transmet a tota la velocitat que permet l'ampla de banda però no durant tot el temps.



SONET (Synchronous Optical Network) is an optical transmission interface originally proposed by BellCore and standardized by ANSI. A compatible version, referred to as Synchronous Digital Hierarchy (SDH), has been published by ITU-T in Recommendation G.707. SONET is intended to provide a specification for taking advantage of the high-speed digital transmission capability of optical fiber.

SONET/SDH Signal Hierarchy

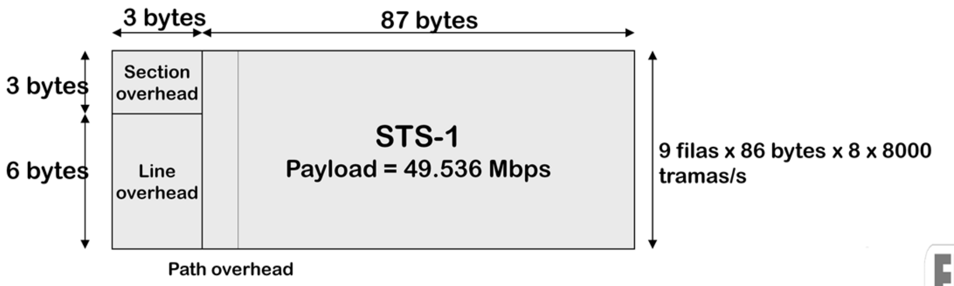
SONET Designation	ITU-T Designation	Data Rate	Payload Rate (Mbps)
STS-1/OC-1		51.84 Mbps	50.112 Mbps
STS-3/OC-3	STM-1	155.52 Mbps	150.336 Mbps
STS-12/OC-12	STM-4	622.08 Mbps	601.344 Mbps
STS-48/OC-48	STM-16	2.48832 Gbps	2.405376 Gbps
STS-192/OC-192	STM-64	9.95328 Gbps	9.621504 Gbps
STS-768	STM-256	39.81312 Gbps	38.486016 Gbps
STS-3072		159.25248 Gbps	153.944064 Gbps

Table 8.4

The SONET specification defines a hierarchy of standardized digital data rates (Table 8.4). The lowest level, referred to as STS-1 (Synchronous Transport Signal level 1) or OC-1 (Optical Carrier level 1), is 51.84 Mbps. This rate can be used to carry a single DS-3 signal or a group of lower-rate signals, such as DS1, DS1C, DS2, plus ITU-T rates (e.g., 2.048 Mbps). An OC-N rate is the optical equivalent of an STS-N electrical signal. End user devices transmit and receive electrical signals; these must be converted to and from optical signals for transmission over optical fiber. Multiple STS-1 signals can be combined to form an STS-N signal. The signal is created by interleaving bytes from *N* STS-1 signals that are mutually synchronized. For the ITU-T Synchronous Digital Hierarchy, the lowest rate is 155.52 Mbps, which is designated STM-1. This corresponds to SONET STS-3.

SONET

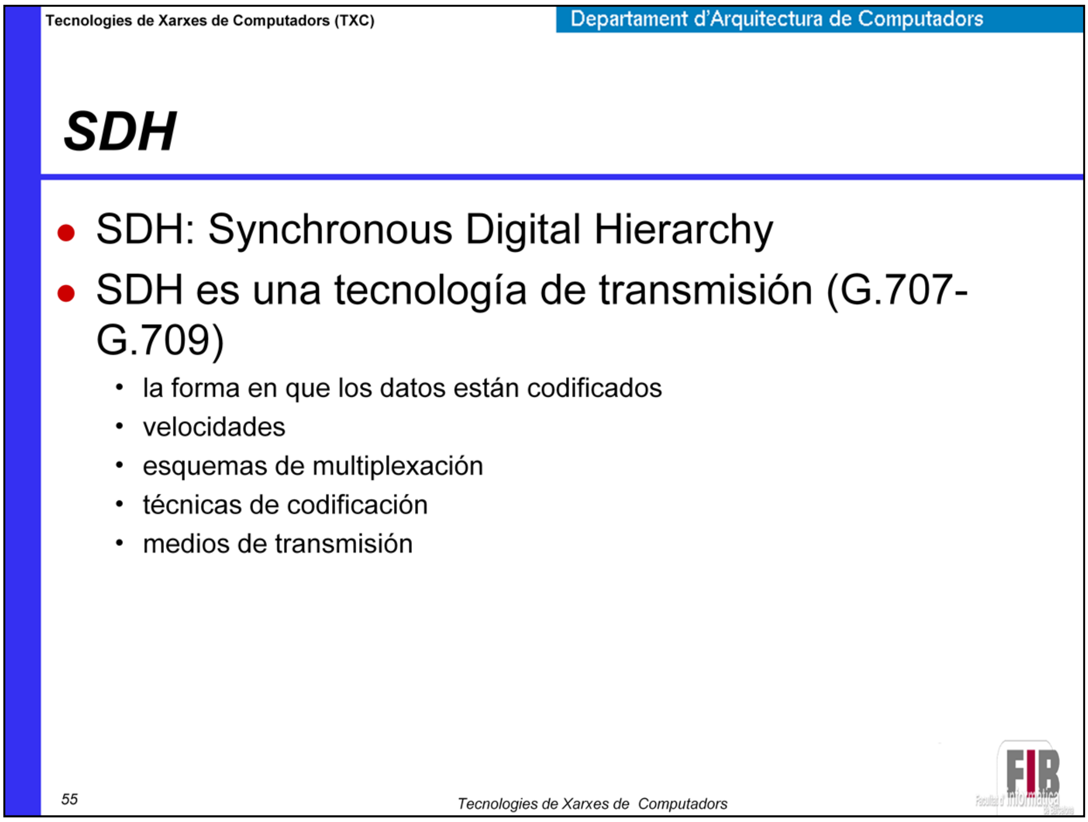
- SONET transmite datos en tramas:
  - La trama OC-1 es un conjunto bidimensional de 90 columnas por 9 filas de octetos (bytes).
    - *Las primeras 3 columnas (27 bytes) son el overhead de transporte*
    - *La velocidad es 8000 tramas por segundo (cada 125 microsegundos):*
    - *90 x 9 x 8 x 8000=90 x 9 x 64 kbps = 51.84 Mbps*
  - La trama de OC-n son n tramas de OC-1



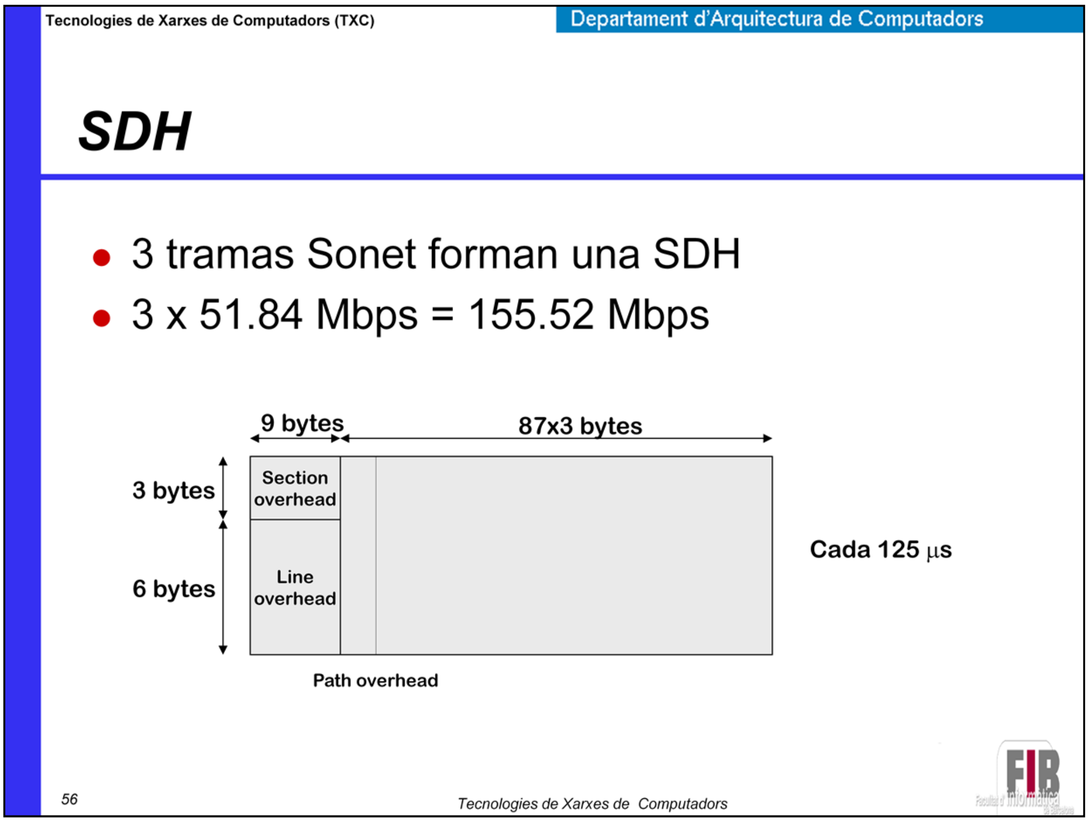
Normativa americana per organitzar les dades que s'envien a nivell físic en format de canals de 64 kbps.

Matrius de 90x9 octets. El Payload és de 86x9 octets.

El bits s'envien en fila índia però van organitzats d'aquesta forma. Cada matriu es repeteix cada 125 microsegons.

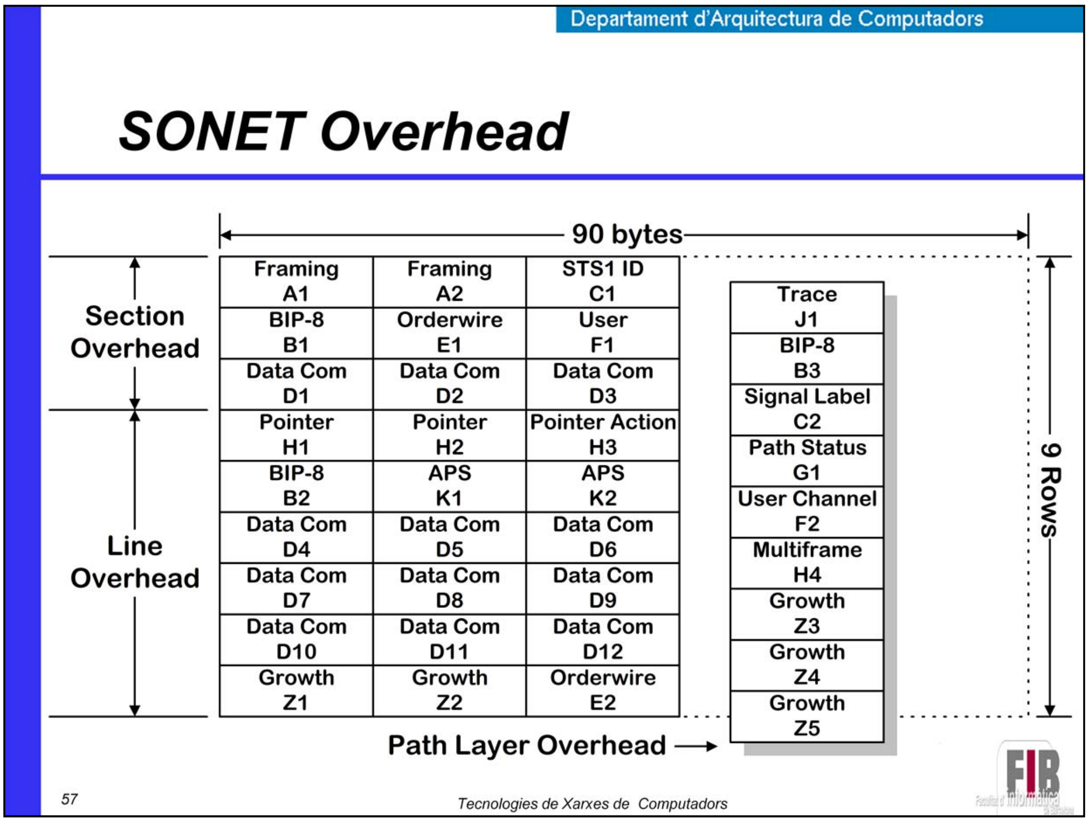


SDH és una organització de bits en forma matricial que correspon a 3 vegades Sonet. O sigui 270 columnes x 9 files. Es una normativa europea i mundial. Sonet és americana.



El concepte és el mateix que SONET. SDH Normativa mundial.

A SDH el payload és 260x9 octets. La capçalera és 10x9 octets (inclou capçaleres: Section, Line i Path).



La capçalera a Sonet porta canals de 64 Kbks (8 bits / 125 microsegons) que serveixen per diferents funcions. Els dos primers són de sincronisme. Hi ha altres de punters.

A SDH el nombre de columnes de la capçalera és de 9+1

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2.5 Commutació

Consultar Capítol 9 Stallings.

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Capítol 9 Stallings.

Switched Network

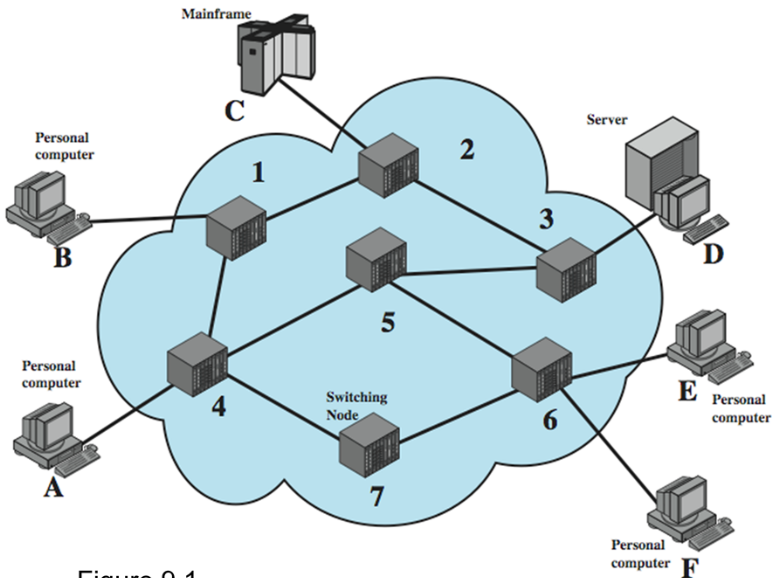


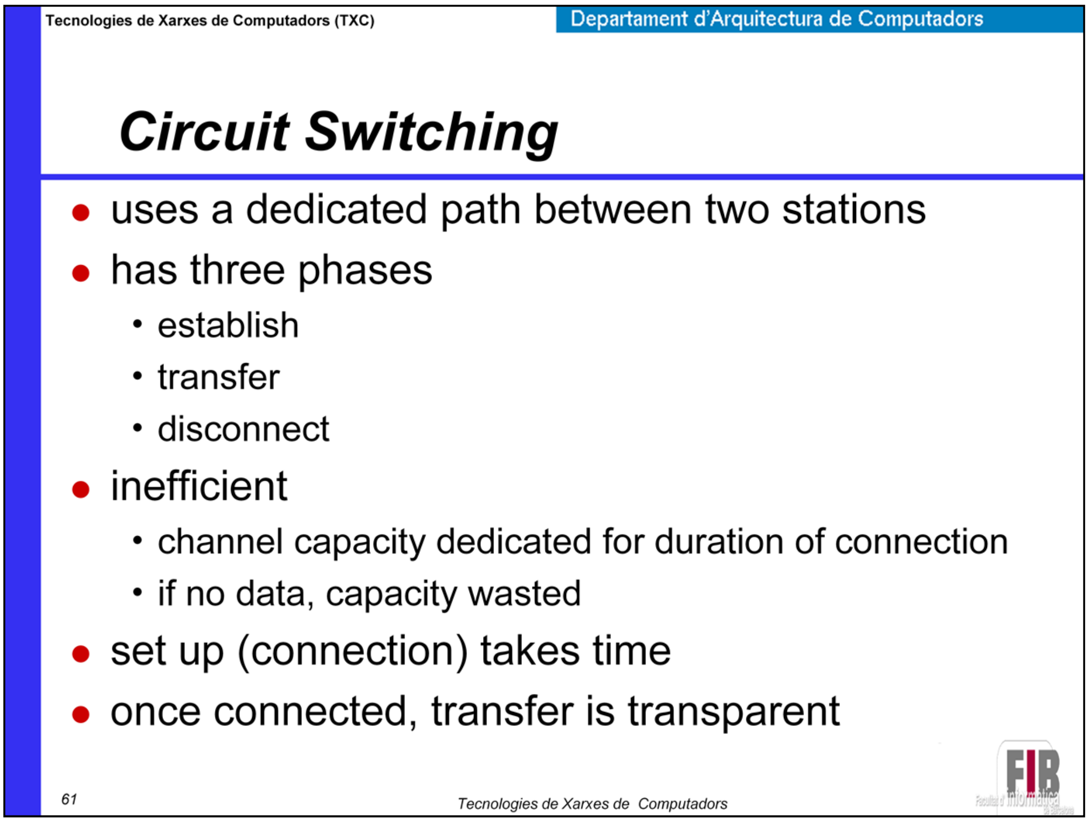
Figure 9.1

For transmission of data beyond a local area, communication is typically achieved by transmitting data from source to destination through a network of intermediate switching nodes; this switched network design is typically used to implement LANs as well. The switching nodes are not concerned with the content of the data; rather, their purpose is to provide a switching facility that will move the data from node to node until they reach their destination. Figure 9.1 illustrates a simple network. The devices attached to the network may be referred to as *stations*. The stations may be computers, terminals, telephones, or other communicating devices. We refer to the switching devices whose purpose is to provide communication as *nodes*. Nodes are connected to one another in some topology by transmission links. Node-station links are generally dedicated point-to-point links. Node-node links are usually multiplexed, using either frequency division multiplexing (FDM) or time division multiplexing (TDM). In a *switched communication network*, data entering the network from a station are routed to the destination by being switched from node to node. For example, in Figure 9.1, data from station A intended for station F are sent to node 4. They may then be routed via nodes 5 and 6 or nodes 7 and 6 to the destination.

Nodes

- a collection of nodes and connections is a communications network
- nodes may connect to other nodes only, or to stations and other nodes
- network is usually partially connected
  - some redundant connections are desirable
- have two different switching technologies
  - circuit switching
  - packet switching

Each station attaches to a node, and the collection of nodes is referred to as a *communications network*. Some nodes connect only to other nodes (eg., 5 and 7 on previous slide). Their sole task is the internal (to the network) switching of data. Other nodes have one or more stations attached as well; in addition to their switching functions, such nodes accept data from and deliver data to the attached stations. Usually, the network is not fully connected; that is, there is not a direct link between every possible pair of nodes. However, it is always desirable to have more than one possible path through the network for each pair of stations. This enhances the reliability of the network. Two different technologies are used in wide area switched networks: circuit switching and packet switching. These two technologies differ in the way the nodes switch information from one link to another on the way from source to destination.



Communication via circuit switching implies that there is a dedicated communication path between two stations. That path is a connected sequence of links between network nodes. On each physical link, a logical channel is dedicated to the connection. Communication via circuit switching involves three phases:

1. **Circuit establishment -** Before any signals can be transmitted, an end-to-end (station-to-station) circuit must be established.
2. **Data transfer -** Data can now be transmitted through the network between these two stations. The transmission may be analog or digital, depending on the nature of the network. As the carriers evolve to fully integrated digital networks, the use of digital (binary) transmission for both voice and data is becoming the dominant method. Generally, the connection is full duplex.
3. **Circuit disconnect -** After some period of data transfer, the connection is terminated, usually by the action of one of the two stations. Signals must be propagated to the intermediate nodes to deallocate the dedicated resources.

Circuit switching can be rather inefficient. Channel capacity is dedicated for the duration of a connection, even if no data are being transferred. For a voice connection, utilization may be rather high, but it still does not approach 100%. For a client/server or terminal-to-computer connection, the capacity may be idle during most of the time of the connection. In terms of performance, there is a delay prior to signal transfer for call establishment. However, once the circuit is established, the network is effectively transparent to the users.

Public Circuit Switched Network

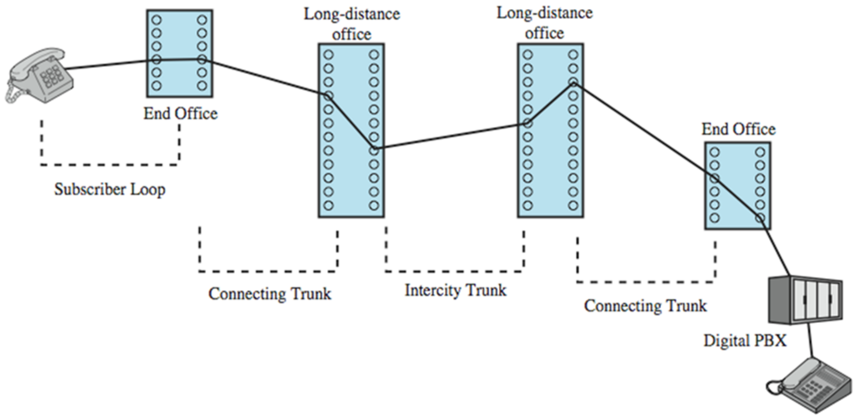


Figure 9.2

Circuit switching was developed to handle voice traffic but is now also used for data traffic. The best-known example of a circuit-switching network is the public telephone network (see Figure 9.2 above). This is actually a collection of national networks interconnected to form the international service. A public telecommunications network can be described using four generic architectural components:

• **Subscribers:** The devices that attach to the network, typically telephones, but the percentage of data traffic increases year by year.

• **Subscriber line:** The link between the subscriber and the network, also referred to as the *subscriber loop* or *local loop*, mostly using twisted-pair wire.

• **Exchanges:** The switching centers in the network. A switching center that directly supports subscribers is known as an end office.

• **Trunks:** The branches between exchanges. Trunks carry multiple voice-frequency circuits using either FDM or synchronous TDM

Packet Switching

- circuit switching was designed for voice
- packet switching was designed for data
- transmitted in small packets
- packets contains user data and control info
  - user data may be part of a larger message
  - control info includes routing (addressing) info
- packets are received, stored briefly (buffered) and past on to the next node

The long-haul circuit-switching telecommunications network was originally designed to handle voice traffic, and resources within the network are dedicated to a particular call. With increasing use of data communications its limitations become more apparent.

In contrast a packet-switching network is designed for data use. Data are transmitted in short packets. A typical upper bound on packet length is 1000 octets (bytes). If a source has a longer message to send, the message is broken up into a series of packets. Each packet contains a portion (or all for a short message) of the user's data plus some control information. The control information, at a minimum, includes the information that the network requires to be able to route the packet through the network and deliver it to the intended destination. At each node in route, the packet is received, stored briefly, and passed on to the next node.

Packet Switching

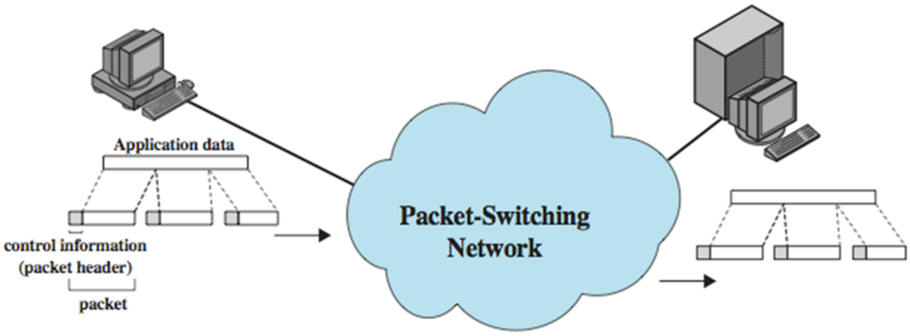


Figure 9.10

Figure 9.10 illustrates how a longer message is broken up into a series of packets, which are transferred over the packet-switched network.

Advantages

- line efficiency
  - single link shared by many packets over time
  - packets queued and transmitted as fast as possible
- data rate conversion
  - stations connects to local node at own speed
  - nodes buffer data if required to equalize rates
- packets accepted even when network is busy
- priorities can be used

A packet-switching network has a number of advantages over circuit switching:

•Line efficiency is greater, because a single node-to-node link can be dynamically shared by many packets over time. The packets are queued up and transmitted as rapidly as possible over the link. By contrast, with circuit switching, time on a node-to-node link is preallocated using synchronous time division multiplexing. Much of the time, such a link may be idle because a portion of its time is dedicated to a connection that is idle.

•A packet-switching network can perform data-rate conversion. Two stations of different data rates can exchange packets because each connects to its node at its proper data rate.

•When traffic becomes heavy on a circuit-switching network, some calls are blocked; that is, the network refuses to accept additional connection requests until the load on the network decreases. On a packet-switching network, packets are still accepted, but delivery delay increases.

•Priorities can be used. If a node has a number of packets queued for transmission, it can transmit the higher-priority packets first. These packets will therefore experience less delay than lower-priority packets.

Datagram Diagram

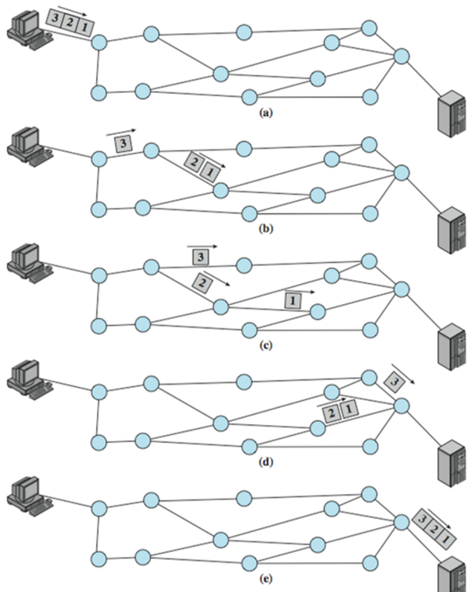


Figure 10.9 Packet Switching: Datagram Approach

Figure 9.11

In the **datagram** approach, each packet is treated independently, with no reference to packets that have gone before. This approach is illustrated in Figure 9.11, which shows a time sequence of snapshots of the progress of three packets through the network. Each node chooses the next node on a packet's path, taking into account information received from neighboring nodes on traffic, line failures, and so on. So the packets, each with the same destination address, do not all follow the same route, and they may arrive out of sequence at the exit point. In this example, the exit node restores the packets to their original order before delivering them to the destination. In some datagram networks, it is up to the destination rather than the exit node to do the reordering. Also, it is possible for a packet to be destroyed in the network. For example, if a packet-switching node crashes momentarily, all of its queued packets may be lost. Again, it is up to either the exit node or the destination to detect the loss of a packet and decide how to recover it. In this technique, each packet, treated independently, is referred to as a datagram.

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Virtual Circuit Diagram

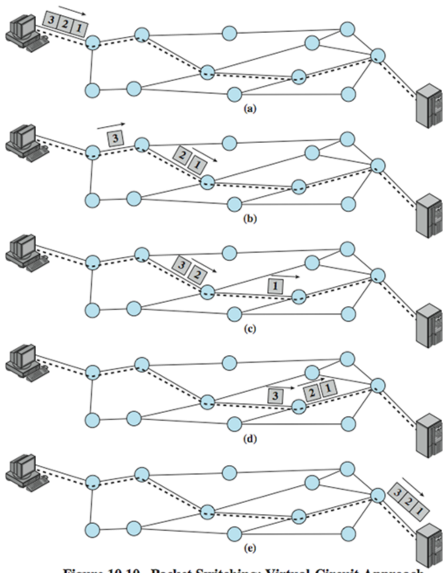

Figure 10.10 Packet Switching: Virtual-Circuit Approach

Figure 9.12

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In the **virtual circuit** approach, a preplanned route is established before any packets are sent. Once the route is established, all the packets between a pair of communicating parties follow this same route through the network. This is illustrated in Figure 9.12. Because the route is fixed for the duration of the logical connection, it is somewhat similar to a circuit in a circuit-switching network and is referred to as a virtual circuit. Each packet contains a virtual circuit identifier as well as data. Each node on the preestablished route knows where to direct such packets; no routing decisions are required. At any time, each station can have more than one virtual circuit to any other station and can have virtual circuits to more than one station.

So the main characteristic of the virtual circuit technique is that a route between stations is set up prior to data transfer. Note that this does not mean that this is a dedicated path, as in circuit switching. A transmitted packet is buffered at each node, and queued for output over a line, while other packets on other virtual circuits may share the use of the line. The difference from the datagram approach is that, with virtual circuits, the node need not make a routing decision for each packet. It is made only once for all packets using that virtual circuit.

Virtual Circuits v Datagram

- virtual circuits
  - network can provide sequencing and error control
  - packets are forwarded more quickly
  - less reliable
- datagram
  - no call setup phase
  - more flexible
  - more reliable

If two stations wish to exchange data over an extended period of time, there are certain advantages to virtual circuits. First, the network may provide services related to the virtual circuit, including sequencing and error control. Sequencing refers to the fact that, because all packets follow the same route, they arrive in the original order. Error control is a service that assures not only that packets arrive in proper sequence, but also that all packets arrive correctly. Another advantage is that packets should transit the network more rapidly with a virtual circuit; it is not necessary to make a routing decision for each packet at each node.

One advantage of the datagram approach is that the call setup phase is avoided. Thus, if a station wishes to send only one or a few packets, datagram delivery will be quicker. Another advantage of the datagram service is that, because it is more primitive, it is more flexible. For example, if congestion develops in one part of the network, incoming datagrams can be routed away from the congestion. With the use of virtual circuits, packets follow a predefined route, and thus it is more difficult for the network to adapt to congestion. A third advantage is that datagram delivery is inherently more reliable. With the use of virtual circuits, if a node fails, all virtual circuits that pass through that node are lost. With datagram delivery, if a node fails, subsequent packets may find an alternate route that bypasses that node. A datagram-style of operation is common in internetworks.

Packet Size

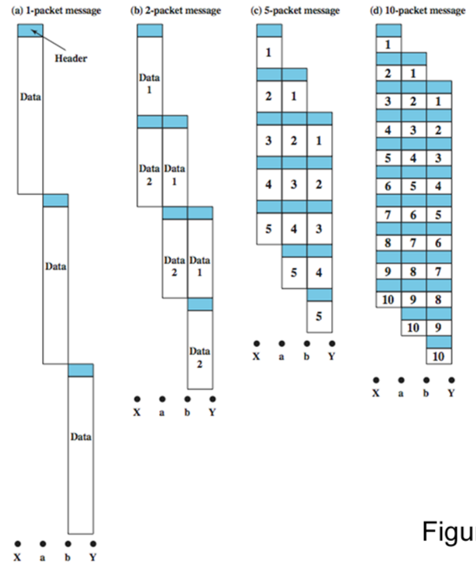

Figure 9.13

There is a significant relationship between packet size and transmission time, as shown in Figure 9.13, which assumes a virtual circuit exists from station X through nodes a and b to station Y. The message to be sent comprises 40 octets, with 3 octets of control information at the beginning of each packet in the header. If the entire message is sent as a single packet of 43 octets (3 octets of header plus 40 octets of data), then the packet is first transmitted from station X to node a (Figure 10.11a). When the entire packet is received, it can then be transmitted from a to b. When the entire packet is received at node b, it is then transferred to station Y. Ignoring switching time, total transmission time is 129 octet-times (43 octets × 3 packet transmissions).

If we break the message into two packets with 20 octets of message and 3 octets of header each. In this case, node a can begin transmitting the first packet as soon as it has arrived from X. Because of this overlap in transmission, the total transmission time drops to 92 octet-times. By breaking the message into five packets, each intermediate node can begin transmission even sooner, with a total of 77 octet-times for transmission. This process of using more and smaller packets eventually results in increased, rather than reduced, delay as illustrated in Figure 10.11d, since each packet contains a fixed amount of header, and more packets Also, processing and queuing delays at each node are greater when more packets are handled for a single message. However, we shall see in the next chapter that an extremely small packet size (53 octets) can result in an efficient network design.

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Event Timing

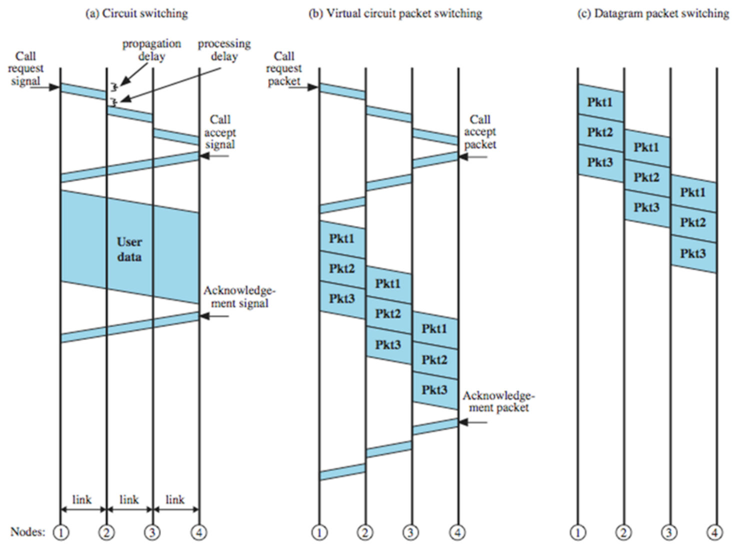

Figure 9.15

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A simple comparison of circuit switching and the two forms of packet switching is provided in Figure 9.15. The figure depicts the transmission of a message across four nodes, from node 1 to 4.

For circuit switching, there is a delay before the message is sent. First, a Call Request signal is sent. If the destination station is not busy, a Call Accepted signal returns. Note a processing delay is incurred at each node during the call request. On return, this processing is not needed because the connection is already set up. After the connection is set up, the message is sent as a single block, with no noticeable delay at the switching nodes.

Virtual circuit packet switching appears similar to circuit switching. A virtual circuit is requested using a Call Request packet, which incurs a delay at each node, and is accepted with a Call Accept packet. In contrast to the circuit-switching case, the call acceptance also experiences node delays, since each packet is queued at each node and must wait its turn for transmission. Once the virtual circuit is established, the message is transmitted in packets. Packet switching involves some delay at each node in the path. Worse, this delay is variable and will increase with increased load.

Datagram packet switching does not require a call setup. Thus, for short messages, it will be faster than virtual circuit and perhaps circuit switching. However, because each individual datagram is routed independently, the processing at each node may be longer than for virtual circuit packets. For long messages virtual circuits may be superior.

comparing the techniques

Circuit Switching	Datagram Packet Switching	Virtual Circuit Packet Switching
Dedicated transmission path	No dedicated path	No dedicated path
Continuous transmission of data	Transmission of packets	Transmission of packets
Fast enough for interactive	Fast enough for interactive	Fast enough for interactive
Messages are not stored	Packets may be stored until delivered	Packets stored until delivered
The path is established for entire conversation	Route established for each packet	Route established for entire conversation
Call setup delay; negligible transmission delay	Packet transmission delay	Call setup delay; packet transmission delay
Busy signal if called party busy	Sender may be notified if packet not delivered	Sender notified of connection denial
Overload may block call setup; no delay for established calls	Overload increases packet delay	Overload may block call setup; increases packet delay
Electromechanical or computerized switching nodes	Small switching nodes	Small switching nodes
User responsible for message loss protection	Network may be responsible for individual packets	Network may be responsible for packet sequences
Usually no speed or code conversion	Speed and code conversion	Speed and code conversion
Fixed bandwidth	Dynamic use of bandwidth	Dynamic use of bandwidth
No overhead bits after call setup	Overhead bits in each packet	Overhead bits in each packet

Table 9.1

Besides performance, there are a number of other characteristics that may be considered in comparing the techniques we have been discussing. Table 9.1 summarizes the most important of these. Most of these characteristics have already been discussed. A few additional comments follow. As was mentioned, circuit switching is essentially a transparent service. Once a connection is established, a constant data rate is provided to the connected stations. This is not the case with packet switching, which typically introduces variable delay, so that data arrive in a choppy manner. Indeed, with datagram packet switching,

data may arrive in a different order than they were transmitted. An additional consequence of transparency is that there is no overhead required to accommodate circuit switching. Once a connection is established, the analog or digital data are passed through, as is, from source to destination. For packet switching, analog data must be converted to digital before transmission; in addition, each packet includes overhead bits, such as the destination address.

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Facultat d' informàtica de Barcelona

2.6 Protocols control d'enllaç

Consultar Capítol 7 Stallings

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Consultar capítol 7 del llibre Stallings.

Data Link Control Protocols

- when sending data, to achieve control, a layer of logic is added above the Physical layer
  - data link control or a data link control protocol
- to manage exchange of data over a link:
  - frame synchronization
  - flow control
  - error control
  - addressing
  - control and data
  - link management

In this chapter, we shift our emphasis to that of *sending data over a data communications link*. To achieve the necessary control, a layer of logic is added above the physical layer, referred to as **data link control** or a **data link control protocol**. Some of the requirements and objectives for effective data communication between two directly connected transmitting-receiving stations:

•**Frame synchronization:** Data are sent in blocks called frames. The beginning and end of each frame must be recognizable.

•**Flow control:** The sending station must not send frames at a rate faster than the receiving station can absorb them.

•**Error control:** Bit errors introduced by the transmission system should be corrected.

•**Addressing:** On a shared link, such as a local area network (LAN), the identity of the two stations involved in a transmission must be specified.

•**Control and data on same link:** the receiver must be able to distinguish control information from the data being transmitted.

•**Link management:** Procedures for the management of initiation, maintenance, and termination of a sustained data exchange over a link.

Flow Control

- ensure sending entity does not overwhelm receiving entity
  - prevent buffer overflow
- influenced by:
  - transmission time
    - *time taken to emit all bits into medium*
  - propagation time
    - *time for a bit to traverse the link*
- assumption is all frames are successfully received with no frames lost or arriving with errors

Flow control is a technique for assuring that a transmitting entity does not overwhelm a receiving entity with data. The receiving entity typically allocates a data buffer of some maximum length for a transfer. When data are received, the receiver must do a certain amount of processing before passing the data to the higher-level software. In the absence of flow control, the receiver's buffer may fill up and overflow while it is processing old data.

Will assume data sent in a sequence of frames, with each frame containing a portion of the data and some control information. The time it takes for a station to emit all of the bits of a frame onto the medium is the transmission time; this is proportional to the length of the frame. The propagation time is the time it takes for a bit to traverse the link between source and destination. For this section, we assume that all frames that are transmitted are successfully received; no frames are lost, they arrive in order sent, and none arrive with errors. However, each transmitted frame suffers an arbitrary and variable amount of delay before reception.

Model of Frame Transmission

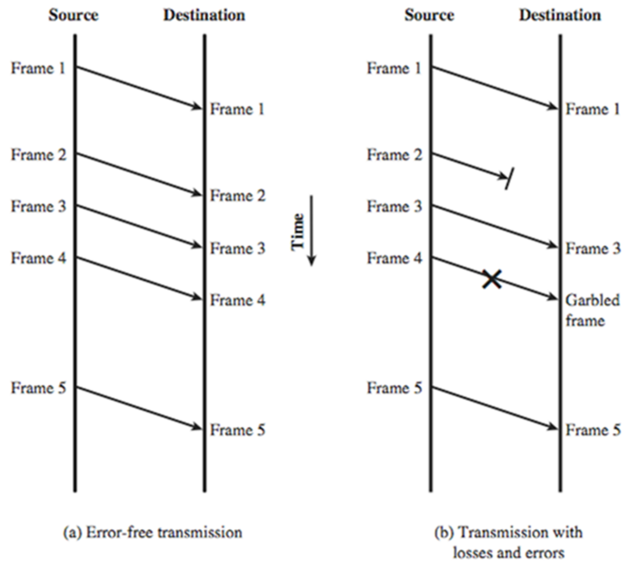
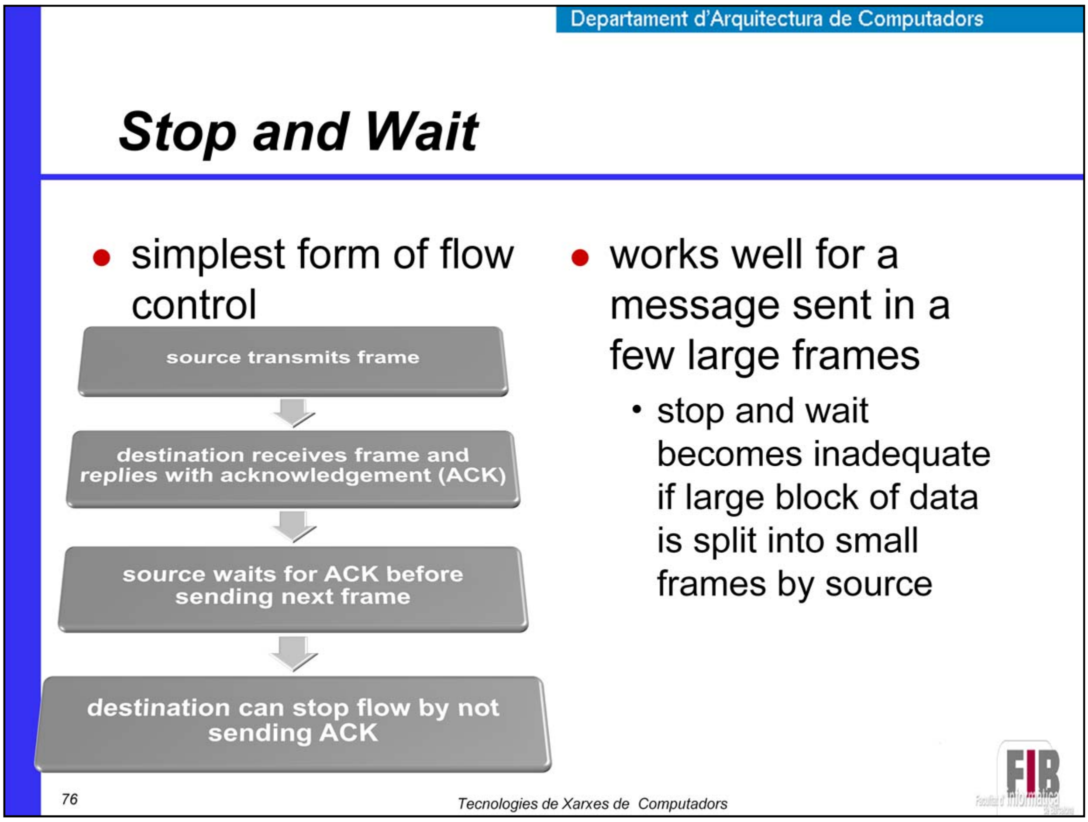


Figure 7.1

To start will examine mechanisms for flow control in the absence of errors. The model we will use is depicted in Figure 7.1a, which is a vertical-time sequence diagram. It has the advantages of showing time dependencies and illustrating the correct send-receive relationship. Each arrow represents a single frame transiting a data link between two stations. The data are sent in a sequence of frames, with each frame containing a portion of the data and some control information.



The simplest form of flow control, known as stop-and-wait flow control, works as follows. A source entity transmits a frame. After the destination entity receives the frame, it indicates its willingness to accept another frame by sending back an acknowledgment to the frame just received. The source must wait until it receives the acknowledgment before sending the next frame. The destination can thus stop the flow of data simply by withholding acknowledgment. This procedure works fine and, indeed, can hardly be improved upon when a message is sent in a few large frames. However, it is often the case that a source will break up a large block of data into smaller blocks and transmit the data in many frames (because of limited buffer size, errors detected sooner with less to resend, to prevent media hogging). With the use of multiple frames for a single message, the stop-and-wait procedure may be inadequate, mainly since only one frame at a time can be in transit.

Stop and Wait Link Utilization

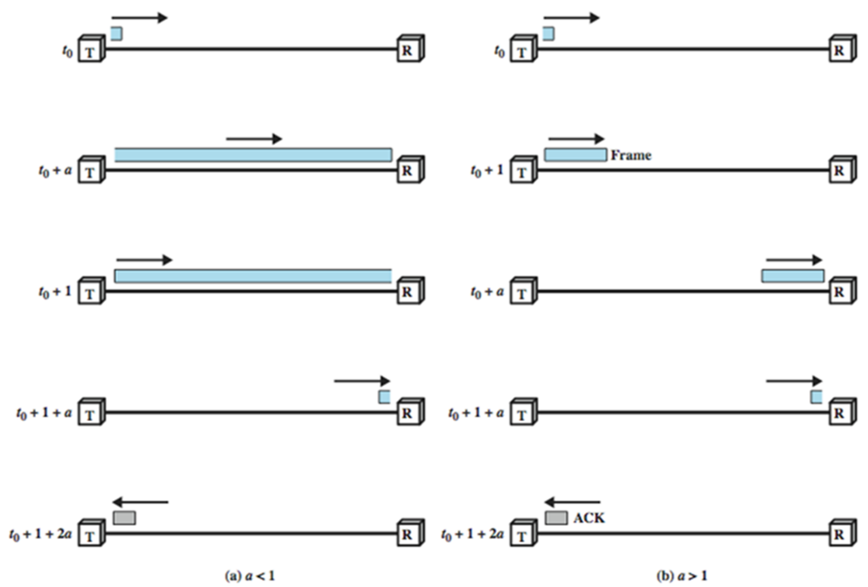


Figure 7.2

With the use of multiple frames for a single message, the stop-and-wait procedure may be inadequate, mainly since only one frame at a time can be in transit.

To show this, start by defining the **bit length B of a link** as the number of bits present on the link at an instance in time when a stream of bits fully occupies the link. In situations where the bit length of the link is greater than the frame length, serious inefficiencies result, as shown in Figure 7.2. In the figure, the transmission time (the time it takes for a station to transmit a frame) is normalized to one, and the propagation delay (the time it takes for a bit to travel from sender to receiver) is expressed as the variable a = B / L (where L is the number of bits in the frame).

When a is less than 1, the propagation time is less than the transmission time. In this case, the frame is sufficiently long that the first bits of the frame have arrived at the destination before the source has completed the transmission of the frame. When a is greater than 1, the propagation time is greater than the transmission time. In this case, the sender completes transmission of the entire frame before the leading bits of that frame arrive at the receiver. Both parts of Figure 7.2 (a and b) consist of a sequence of snapshots, the first four show the process of transmitting a data frame, and the last shows the return of a small acknowledgment frame. Note that for a> 1, the line is always underutilized and even for a< 1, the line is inefficiently utilized. In essence, for very high data rates, for very long distances between sender and receiver, stop-and-wait flow control provides inefficient line utilization.

Sliding Windows Flow Control

- allows multiple numbered frames to be in transit
  - receiver has buffer W long
  - transmitter sends up to W frames without ACK
  - ACK includes number of next frame expected
  - sequence number is bounded by size of field (k)
    - *frames are numbered modulo 2^k*
    - *giving max window size of up to $2^k – 1$*
  - receiver can ACK frames without permitting further transmission (Receive Not Ready)
  - must send a normal acknowledge to resume
- if have full-duplex link, can piggyback ACKs

The essence of the problem described so far is that only one frame at a time can be in transit. Efficiency can be greatly improved by allowing multiple frames to be in transit at the same time. Consider two stations, A and B, connected via a full-duplex link. Station B allocates buffer space for *W* frames. Thus, B can accept *W* frames, and A is allowed to send *W* frames without waiting for any acknowledgments. To keep track of which frames have been acknowledged, each is labeled with a a *k*-bit sequence number. This gives a range of sequence numbers of 0 through $2^k – 1$, and frames are numbered modulo 2^k, with a maximum window size of $2^k – 1$. The window size need not be the maximum possible size for a given sequence number length k. B acknowledges a frame by sending an acknowledgment that includes the sequence number of the next frame expected. This scheme can also be used to acknowledge multiple frames, and is referred to as **sliding-window flow control**. Most data link control protocols also allow a station to cut off the flow of frames from the other side by sending a Receive Not Ready (RNR) message, which acknowledges former frames but forbids transfer of future frames. At some subsequent point, the station must send a normal acknowledgment to reopen the window. If two stations exchange data, each needs to maintain two windows, one for transmit and one for receive, and each side needs to send the data and acknowledgments to the other. To provide efficient support for this requirement, a feature known as **piggybacking** is typically provided. Each **data frame** includes a field that holds the sequence number of that frame plus a field that holds the sequence number used for acknowledgment.

Sliding Window Diagram

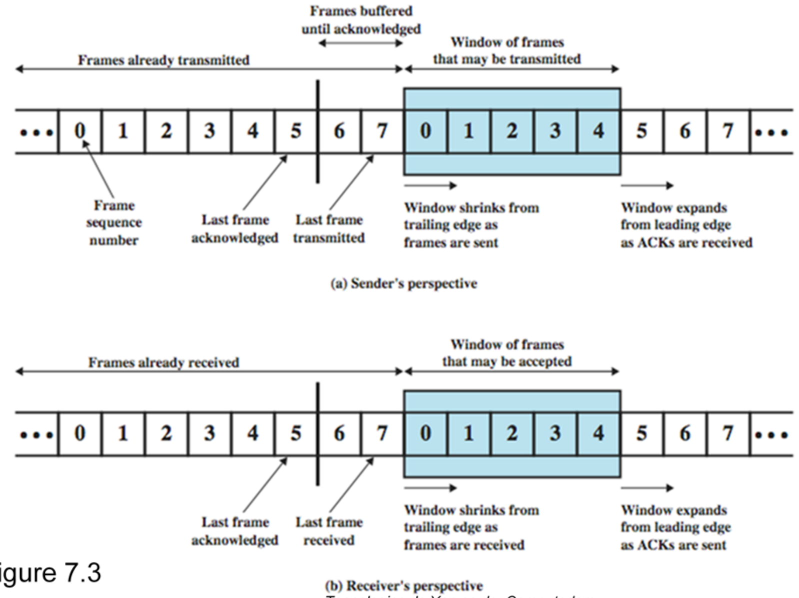


Figure 7.3

Figure 7.3 depicts the sliding-window process. It assumes the use of a 3-bit sequence number, so that frames are numbered sequentially from 0 through 7, and then the same numbers are reused for subsequent frames. The shaded rectangle indicates the frames that may be sent; in this figure, the sender may transmit five frames, beginning with frame 0. Each time a frame is sent, the shaded window shrinks; each time an acknowledgment is received, the shaded window grows. Frames between the vertical bar and the shaded window have been sent but not yet acknowledged. As we shall see, the sender must buffer these frames in case they need to be retransmitted. Sliding-window flow control is potentially much more efficient than stop-and-wait flow control. The reason is that, with sliding-window flow control, the transmission link is treated as a pipeline that may be filled with frames in transit. In contrast, with stop-and-wait flow control, only one frame may be in the pipe at a time.

Sliding Window Example

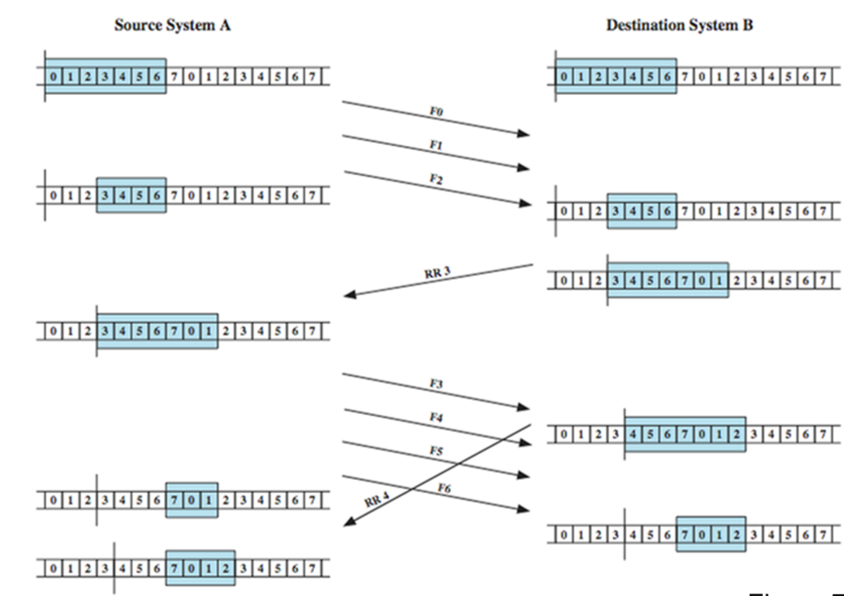
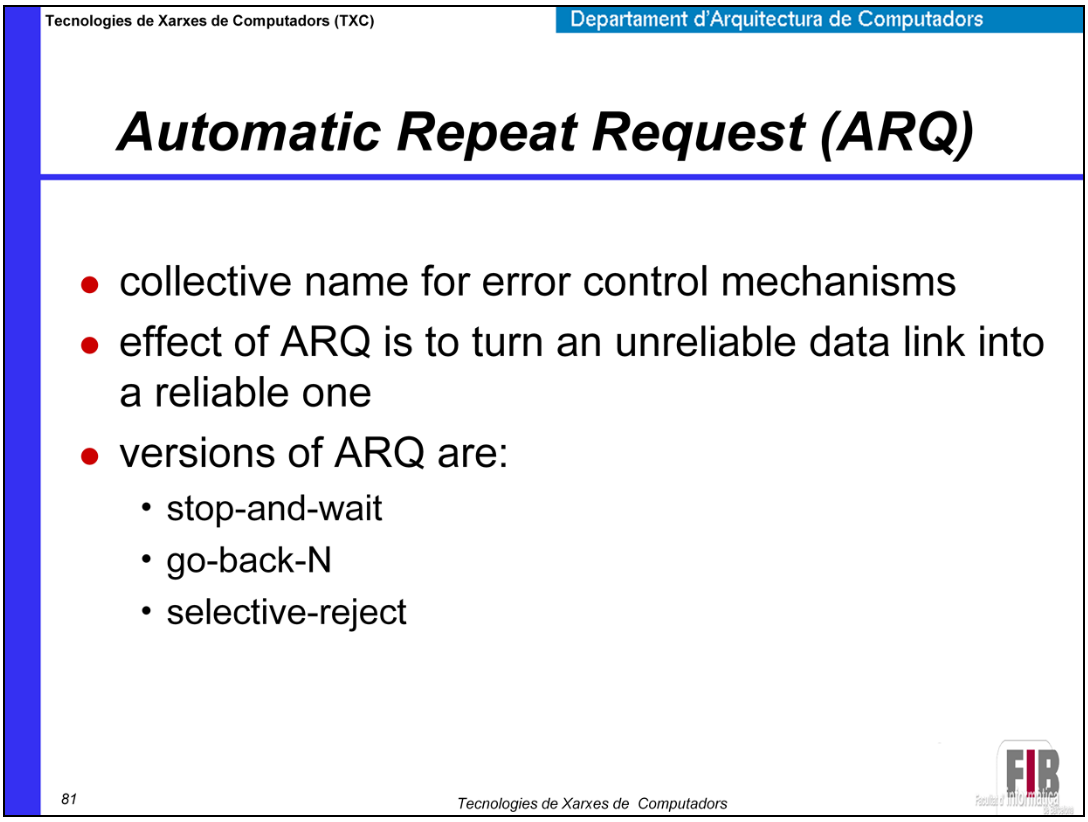


Figure 7.4

An example is shown in Figure 7.4. The example assumes a 3-bit sequence number field and a maximum window size of seven frames. Initially, A and B have windows indicating that A may transmit seven frames, beginning with frame 0 (F0). After transmitting three frames (F0, F1, F2) without acknowledgment, A has shrunk its window to four frames and maintains a copy of the three transmitted frames. The window indicates that A may transmit four frames, beginning with frame number 3. B then transmits an RR (receive ready) 3, which means "I have received all frames up through frame number 2 and am ready to receive frame number 3; in fact, I am prepared to receive seven frames, beginning with frame number 3." With this acknowledgment, A is back up to permission to transmit seven frames, still beginning with frame 3; also A may discard the buffered frames that have now been acknowledged. A proceeds to transmit frames 3, 4, 5, and 6. B returns RR 4, which acknowledges F3, and allows transmission of F4 through the next instance of F2. By the time this RR reaches A, it has already transmitted F4, F5, and F6, and therefore A may only open its window to permit sending four frames beginning with F7.

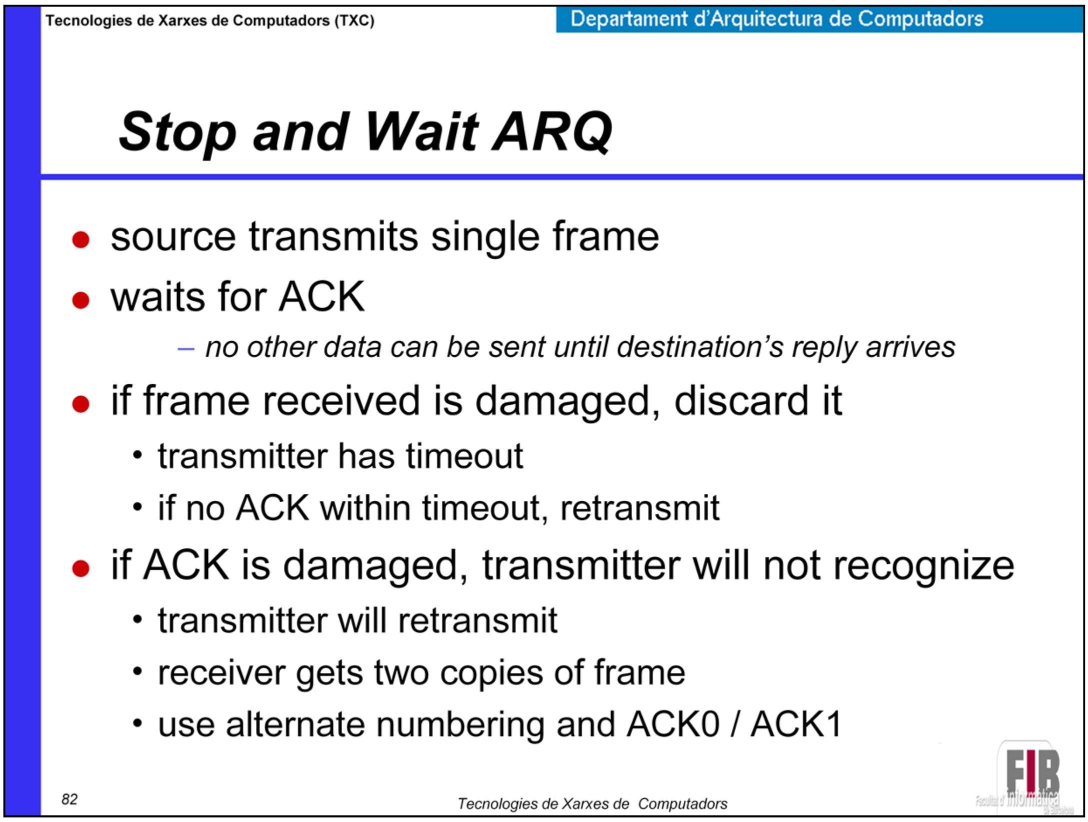


Collectively, these mechanisms are all referred to as **automatic repeat request** (ARQ); the effect of ARQ is to turn an unreliable data link into a reliable one. Three versions of ARQ have been standardized:

•Stop-and-wait ARQ

•Go-back-N ARQ

•Selective-reject ARQ



Stop-and-wait ARQ is based on the stop-and-wait flow control technique outlined previously. The source station transmits a single frame and then must await an acknowledgment (ACK). No other data frames can be sent until the destination station's reply arrives at the source station. Two sorts of errors could occur. First, the frame that arrives at the destination could be damaged. The receiver detects this by using the error-detection technique referred to earlier and simply discards the frame. To account for this possibility, the source station is equipped with a timer. After a frame is transmitted, the source station waits for an acknowledgment. If no acknowledgment is received by the time that the timer expires, then the same frame is sent again. Note that this method requires that the transmitter maintain a copy of a transmitted frame until an acknowledgment is received for that frame. The second sort of error is a damaged acknowledgment, which is not recognizable by A, which will therefore time out and resend the same frame. This duplicate frame arrives and is accepted by B. B has therefore accepted two copies of the same frame as if they were separate. To avoid this problem, frames are alternately labeled with 0 or 1, and positive acknowledgments are of the form ACK0 and ACK1. In keeping with the sliding-window convention, an ACK0 acknowledges receipt of a frame numbered 1 and indicates that the receiver is ready for a frame numbered 0.

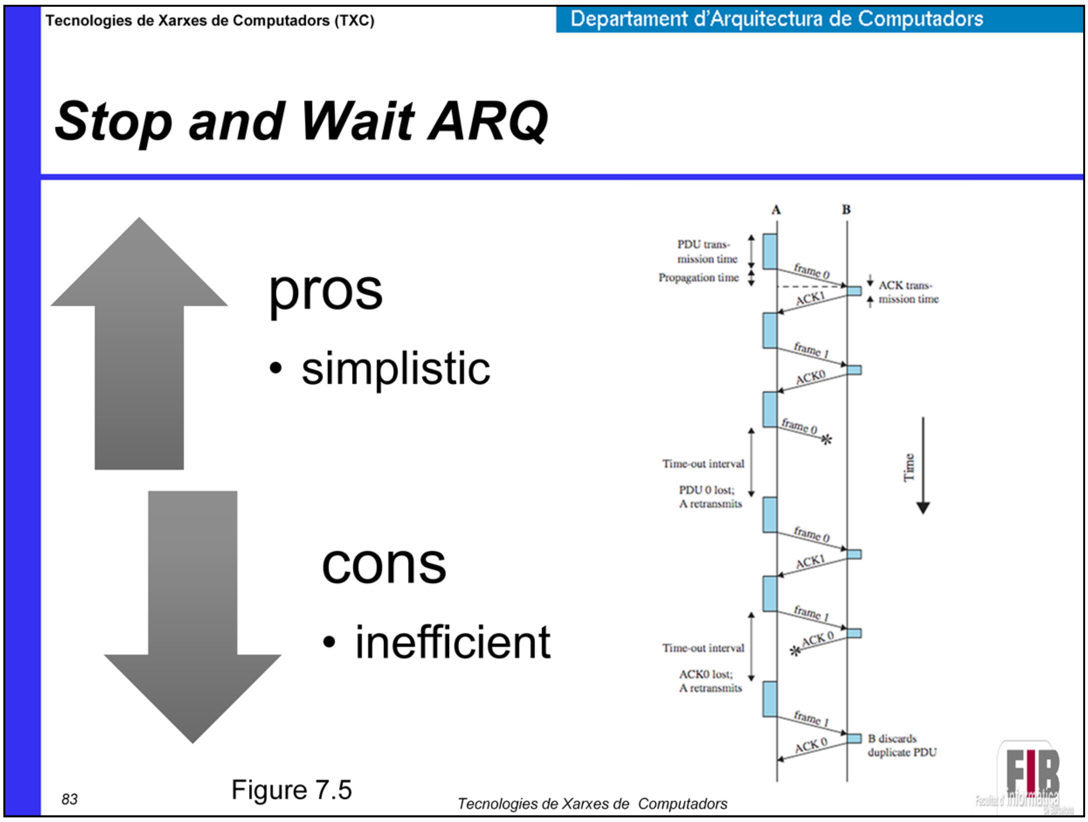


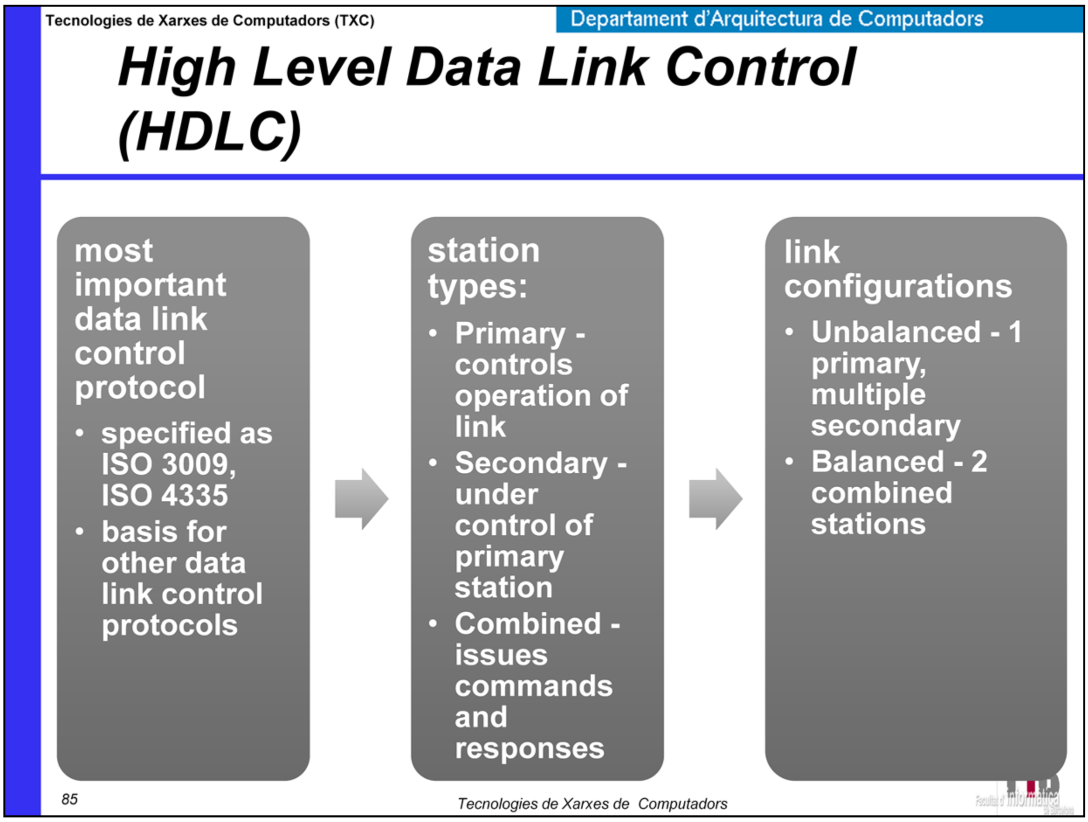
Figure 7.5 gives an example of the use of stop-and-wait ARQ, showing the transmission of a sequence of frames from source A to destination B. The figure shows the two types of errors just described. The third frame transmitted by A is lost or damaged and therefore B does not return an ACK. A times out and retransmits the frame. Later, A transmits a frame labeled 1 but the ACK0 for that frame is lost. A times out and retransmits the same frame. When B receives two frames in a row with the same label, it discards the second frame but sends back an ACK0 to each.

The principal advantage of stop-and-wait ARQ is its simplicity. Its principal disadvantage is that stop-and-wait is an inefficient mechanism.

Go-Back-N ARQ

- most commonly used error control
- based on sliding-window
- use window size to control number of outstanding frames
- if no error, ACK as usual
- if error, reply with rejection
  - destination will discard that frame and all future frames until frame in error is received correctly
  - transmitter must go back and retransmit that frame and all subsequent frames

The sliding-window flow control technique can be adapted to provide more efficient line use, sometimes referred to as *continuous ARQ*. The form of error control based on sliding-window flow control that is most commonly used is called go-back-N ARQ. In this method, a station may send a series of frames sequentially numbered modulo some maximum value. The number of unacknowledged frames outstanding is determined by window size, using the sliding-window flow control technique. While no errors occur, the destination will acknowledge incoming frames as usual (RR = receive ready, or piggybacked acknowledgment). If the destination station detects an error in a frame, it may send a negative acknowledgment (REJ = reject) for that frame, as explained in the following rules. The destination station will discard that frame and all future incoming frames until the frame in error is correctly received. Thus, the source station, when it receives a REJ, must retransmit the frame in error plus all succeeding frames that were transmitted in the interim.



The most important data link control protocol is HDLC (ISO 3009, ISO 4335). Not only is HDLC widely used, but it is the basis for many other important data link control protocols, which use the same or similar formats and the same mechanisms as employed in HDLC.

To satisfy a variety of applications, HDLC defines: three station types:

• **Primary station:** Responsible for controlling the operation of the link. Frames issued by the primary are called commands.

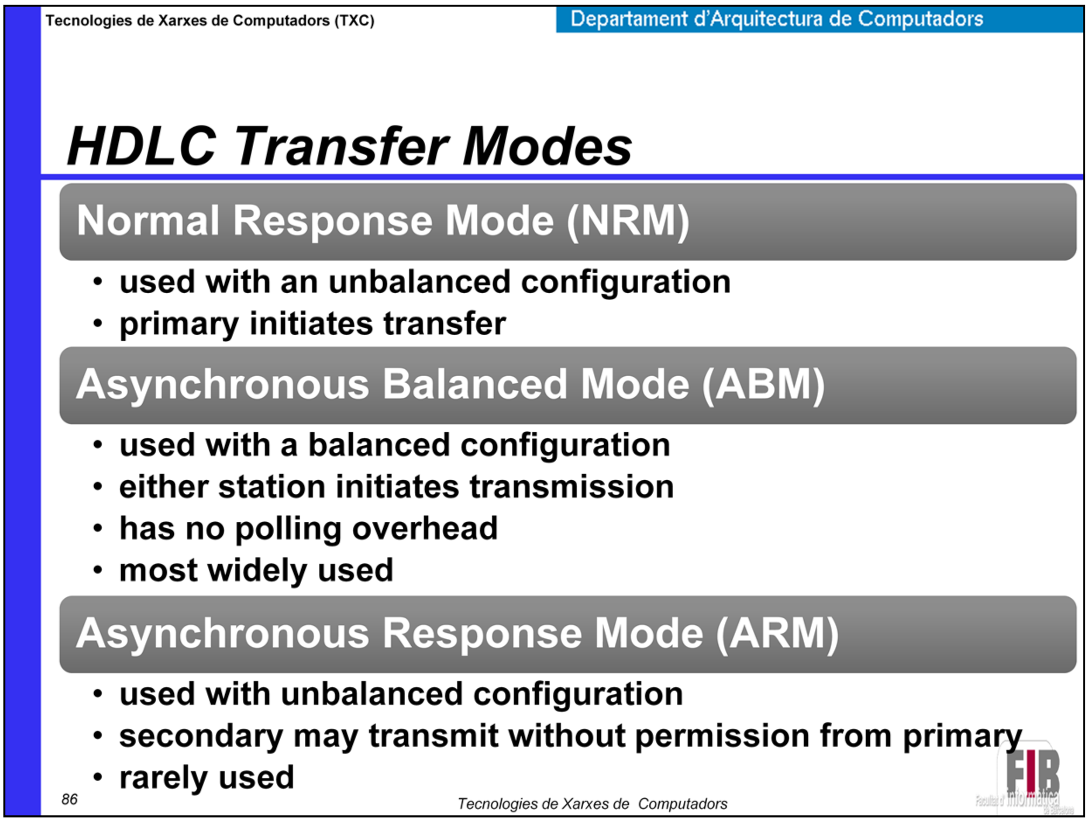
• **Secondary station:** Operates under the control of the primary station. Frames issued by a secondary are called responses. The primary maintains a separate logical link with each secondary station on the line.

• **Combined station:** Combines the features of primary and secondary. A combined station may issue both commands and responses.

It also defines two link configurations:

• **Unbalanced configuration:** Consists of one primary and one or more secondary stations and supports both full-duplex and half-duplex transmission.

• **Balanced configuration:** Consists of two combined stations and supports both full-duplex and half-duplex transmission.

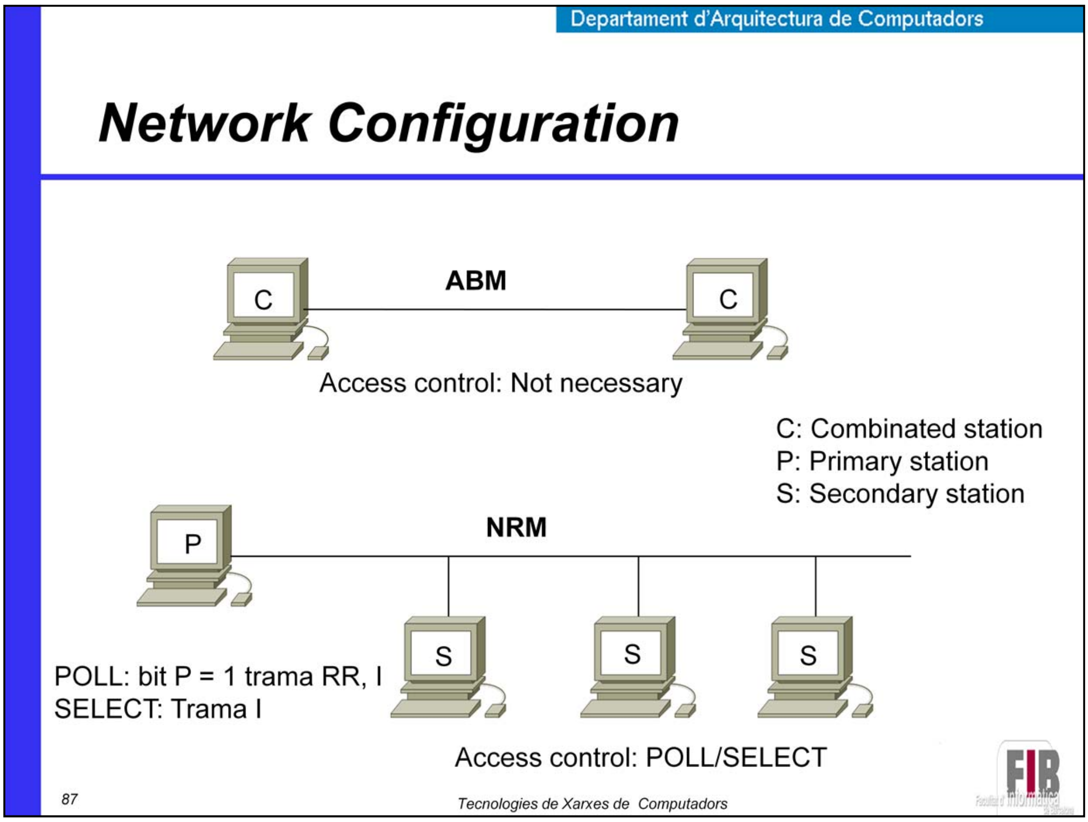


HDLC defines three data transfer modes:

•**Normal response mode (NRM):** Used with an unbalanced configuration. The primary may initiate data transfer to a secondary, but a secondary may only transmit data in response to a command from the primary. NRM is used on multi-drop lines, in which a number of terminals are connected to a host computer.

•**Asynchronous balanced mode (ABM):** Used with a balanced configuration. Either combined station may initiate transmission without receiving permission from the other combined station. ABM is the most widely used of the three modes; it makes more efficient use of a full-duplex point-to-point link because there is no polling overhead.

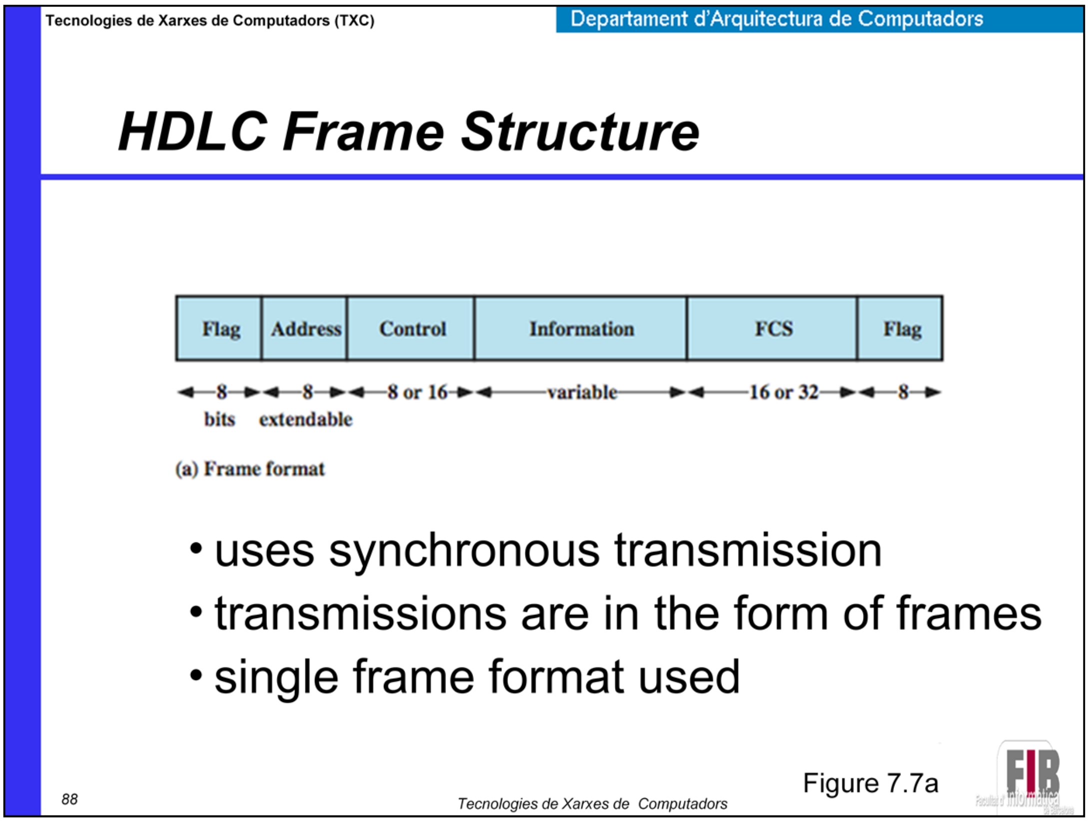
•**Asynchronous response mode (ARM):** Used with an unbalanced configuration. The secondary may initiate transmission without explicit permission of the primary. The primary still retains responsibility for the line, including initialization, error recovery, and logical disconnection. ARM is rarely used; it is applicable to some special situations in which a secondary may need to initiate transmission.



Topologia de xarxa (la xarxa física pot ser qualsevol configuració):

ABM només dues estacions les dues combinades amb la mateixa capacitat de control de l'enllaç. No hi ha POLL. És el cas més habitual en xarxes WAN a nivell local.

NRM té una estació primària i una o més secundàries. El control de l'enllaç el té la primària. Per accedir al medi les secundàries han de rebre un POLL de la primària (bit P a 1 en trames RR o I). La primària envia dades a les secundàries amb un SELECT (trames I). Les secundàries no es poden connectar entre si.



HDLC uses synchronous transmission. All transmissions are in the form of frames, and a single frame format suffices for all types of data and control exchanges.

Figure 7.7a depicts the structure of the HDLC frame. The flag, address, and control fields that precede the information field are known as a **header**. The FCS and flag fields following the data field are referred to as a **trailer**.

Flag Fields and Bit Stuffing

- delimit frame at both ends with 01111110
- receiver hunts for flag sequence to synchronize
- bit stuffing used to avoid confusion with data containing flag sequence 01111110
  - 0 inserted after every sequence of five 1s
  - if receiver detects five 1s it checks next bit
  - if next bit is 0, it is deleted (was stuffed bit)
  - if next bit is 1 and seventh bit is 0, accepted as flag
  - if sixth and seventh bits 1, sender is indicating abort

Original Pattern:

111111111111011111101111110

After bit-stuffing

1111101111101101111101011111010

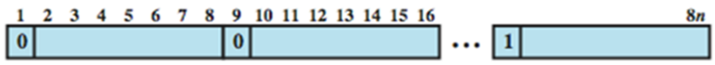
Figure 7.8

Flag fields delimit the frame at both ends with the unique pattern 01111110. A single flag may be used as the closing flag for one frame and the opening flag for the next. On both sides of the user-network interface, receivers are continuously hunting for the flag sequence to synchronize on the start of a frame. While receiving a frame, a station continues to hunt for that sequence to determine the end of the frame.

Because the protocol allows the presence of arbitrary bit patterns, there is no assurance that the pattern 01111110 will not appear somewhere inside the frame, thus destroying synchronization. To avoid this problem, a procedure known as *bit stuffing* is used. For all bits between the starting and ending flags, the transmitter inserts an extra 0 bit after each occurrence of five 1s in the frame. After detecting a starting flag, the receiver monitors the bit stream. When a pattern of five 1s appears, the sixth bit is examined. If this bit is 0, it is deleted. If the sixth bit is a 1 and the seventh bit is a 0, the combination is accepted as a flag. If the sixth and seventh bits are both 1, the sender is indicating an abort condition. With the use of bit stuffing, arbitrary bit patterns can be inserted into the data field of the frame. This property is known as **data transparency**. Figure 7.8 shows an example of bit stuffing. Note that in the first two cases, the extra 0 is not strictly necessary for avoiding a flag pattern but is necessary for the operation of the algorithm.

Address Field

- identifies secondary station that transmitted or will receive frame
- usually 8 bits long
- may be extended to multiples of 7 bits
  - leftmost bit indicates if is the last octet (1) or not (0)
- address 11111111 allows primary to broadcast

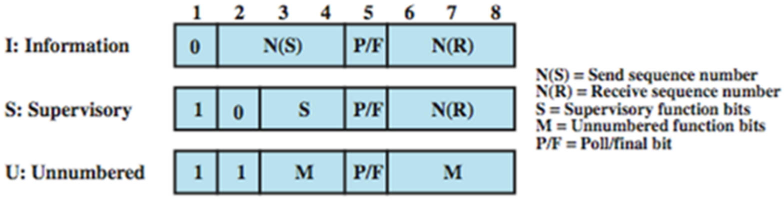


(b) Extended Address Field

Figure 7.7b

The address field identifies the secondary station that transmitted or is to receive the frame. This field is not needed for point-to-point links but is always included for the sake of uniformity. The address field is usually 8 bits long but, by prior agreement, an extended format may be used in which the actual address length is a multiple of 7 bits. The leftmost bit of each octet is 1 or 0 according as it is or is not the last octet of the address field. The remaining 7 bits of each octet form part of the address. The single-octet address of 11111111 is interpreted as the all-stations address in both basic and extended formats. It is used to allow the primary to broadcast a frame for reception by all secondaries.

Control Field



(c) 8-bit control field format

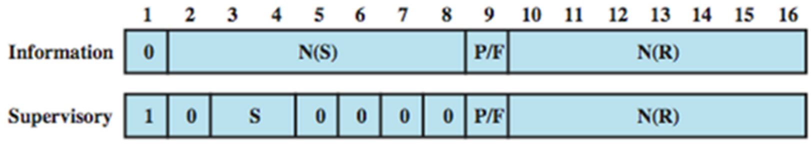
Figure 7.7c

- different frame types
  - Information - data transmitted to user (next layer up)
    - *flow and error control piggybacked on information frames*
  - Supervisory - ARQ when piggyback is not used
  - Unnumbered - supplementary link control functions
- first 1-2 bits of control field identify frame type

HDLC defines three types of frames, each with a different control field format. **Information frames** (I-frames) carry the data to be transmitted for the user (the logic above HDLC that is using HDLC). Additionally, flow and error control data, using the ARQ mechanism, are piggybacked on an information frame. **Supervisory frames** (S-frames) provide the ARQ mechanism when piggybacking is not used. **Unnumbered frames** (U-frames) provide supplemental link control functions. The first one or two bits of the control field serves to identify the frame type. The remaining bit positions are organized into subfields as indicated in Figures 7.7c.

Control Field

- use of Poll/Final (P/F) bit depends on context
- in command frame P bit set to 1 to solicit (poll) response from peer
- in response frame F bit set to 1 to indicate response to soliciting command
- sequence number usually 3 bits
  - can extend to 8 bits as shown below



(d) 16-bit control field format

Figure 7.7d

All of the control field formats contain the poll/final (P/F) bit. Its use depends on context. Typically, in command frames, it is referred to as the P bit and is set to 1 to solicit (poll) a response frame from the peer HDLC entity. In response frames, it is referred to as the F bit and is set to 1 to indicate the response frame transmitted as a result of a soliciting command.

Note that the basic control field for S- and I-frames uses 3-bit sequence numbers, as shown on previous slide. With the appropriate set-mode command, an extended control field can be used for S- and I-frames that employs 7-bit sequence numbers. U-frames always contain an 8-bit control field.

Information and Frame Check Sequence (FCS) Fields

Information Field

- in I-frames and some U-frames
- must contain integral number of octets
- variable length

Frame Check Sequence Field (FCS)

- used for error detection
- either 16 bit CRC or 32 bit CRC

The information field is present only in I-frames and some U-frames. The field can contain any sequence of bits but must consist of an integral number of octets. The length of the information field is variable up to some system-defined maximum.

The frame check sequence (FCS) is an error-detecting code calculated from the remaining bits of the frame, exclusive of flags. The normal code is the 16-bit CRC-CCITT defined in Section 6.5. An optional 32-bit FCS, using CRC-32, may be employed if the frame length or the line reliability dictates this choice.

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HDLC Operation

- consists of exchange of I-frames, S-frames and U-frames
- involves three phases

Initialization	Data Transfer	Disconnect
• either side may request by issuing one of the six set-mode commands	• with flow and error control • using both I and S-frames (RR, RNR, REJ, SREJ)	• when fault noted or at request of higher-layer user • sends a disconnect (DISC) frame

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HDLC operation consists of the exchange of I-frames, S-frames, and U-frames between two stations. The various commands and responses defined for these frame types are listed in Stallings DCC9e Table 7.1. In describing HDLC operation, we will discuss these three types of frames. The operation of HDLC involves three phases: First, one side or another initializes the data link so that frames may be exchanged in an orderly fashion. During this phase, the options that are to be used are agreed upon, such as which of the three modes (NRM, ABM, ARM) is requested, and whether 3- or 7-bit sequence numbers are to be used. After initialization, the two sides exchange user data and the control information to exercise flow and error control. Both sides may begin to send user data in I-frames, starting with sequence number 0. The N(S) and N(R) fields of the I-frame are sequence numbers that support flow control and error control. S-frames are also used for flow control and error control. The receive ready (RR) frame acknowledges the last I-frame received. Receive not ready (RNR) acknowledges an I-frame, as with RR, but also asks the peer entity to suspend transmission of I-frames. When again ready, it sends an RR. REJ initiates the go-back-N ARQ. It indicates that the last I-frame received has been rejected and that retransmission of all I-frames beginning with number N(R) is required. Selective reject (SREJ) is used to request retransmission of just a single frame. Finally, one of the two sides signals the termination of the operation, either on its own initiative if there is some sort of fault, or at the request of its higher-layer user. HDLC issues a disconnect by sending a disconnect (DISC) frame.

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Commands and Responses (1)

Name	Command/ Response	Description
Information (I)	C/R	Exchange user data
Supervisory (S)		
Receive ready (RR)	C/R	Positive acknowledgment; ready to receive I-frame
Receive not ready (RNR)	C/R	Positive acknowledgment; not ready to receive
Reject (REJ)	C/R	Negative acknowledgment; go back N
Selective reject (SREJ)	C/R	Negative acknowledgment; selective reject

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Llista de trames I i S. La trama SREJ no es fa servir usualment ja que correspon a la retransmissió selectiva que no s'utilitza per la seva complexitat.

Tecnologies de Xarxes de Computadors (TXC) Departament d'Arquitectura de Computadors

Commands and Responses (2)

Table 7.1

Unnumbered (U)		
Set normal response/extended mode (SNRM/SNRME)	C	Set mode; extended = 7-bit sequence numbers
Set asynchronous response/extended mode (SARM/SARME)	C	Set mode; extended = 7-bit sequence numbers
Set asynchronous balanced/extended mode (SABM, SABME)	C	Set mode; extended = 7-bit sequence numbers
Set initialization mode (SIM)	C	Initialize link control functions in addressed station
Disconnect (DISC)	C	Terminate logical link connection
Unnumbered Acknowledgment (UA)	R	Acknowledge acceptance of one of the set-mode commands
Disconnected mode (DM)	R	Responder is in disconnected mode
Request disconnect (RD)	R	Request for DISC command
Request initialization mode (RIM)	R	Initialization needed; request for SIM command
Unnumbered information (UI)	C/R	Used to exchange control information
Unnumbered poll (UP)	C	Used to solicit control information
Reset (RSET)	C	Used for recovery; resets N(R), N(S)
Exchange identification (XID)	C/R	Used to request/report status
Test (TEST)	C/R	Exchange identical information fields for testing
Frame reject (FRMR)	R	Report receipt of unacceptable frame

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Llista de trames U. Cal considerar SABM, SNRM, SABME, SNRME, UA, UI i FRMR.

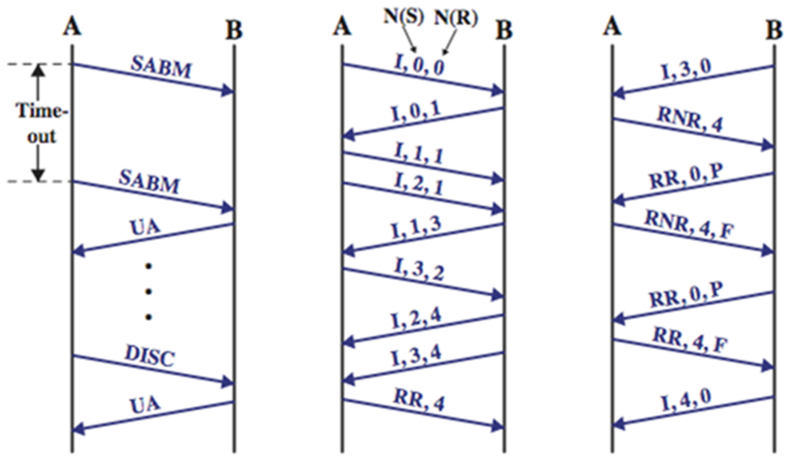
SABM i SNRM per començar en ABM o NRM. Amb E és la versió “extended” per a numeració 128 i 7 bits.

UA és la confirmació positiva de trama U.

UI es fa servir quan es transmet poca informació i es fa sense numerar.

FRMR serveix per sortir de situacions inexplicables. Per exemple quan es rep una confirmació d'una trama que no s'ha enviat.

HDLC Operation Example

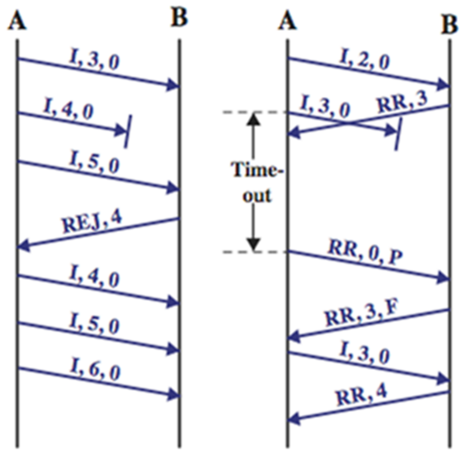


(a) Link setup and disconnect (b) Two-way data exchange (c) Busy condition

Figure 7.9

Examples of HDLC operation are shown here in Figure 7.9. Figure 7.9a shows the frames involved in link setup and disconnect. One side issues an SABM command and starts a timer. Upon receiving the SABM, the recipient returns a UA response and sets local variables and counters to initial values. The initiating entity receives the UA response, sets its variables and counters, and stops the timer. The logical connection is now active, and both sides may begin transmitting frames. Should the timer expire without a response to an SABM, the originator will repeat the SABM, as illustrated. The same figure (Figure 7.9a) shows the disconnect procedure. One side issues a DISC command, and the other responds with a UA response. Figure 7.9b illustrates the full-duplex exchange of I-frames. When an entity sends a number of I-frames in a row with no incoming data, then the receive sequence number is simply repeated (e.g., I,1,1; I,2.1 in the A-to-B direction). When an entity receives a number of I-frames in a row with no outgoing frames, then the receive sequence number in the next outgoing frame must reflect the cumulative activity (e.g., I,1,3 in the B-to-A direction). Figure 7.9c shows an operation involving a busy condition, where the entity's receive buffer fills up and it must halt the incoming flow of I-frames, using an RNR command. In this example, A issues an RNR, which requires B to halt transmission of I-frames. The station receiving the RNR will usually poll the busy station at some periodic interval by sending an RR with the P bit set. This requires the other side to respond with either an RR or an RNR. When the busy condition has cleared, A returns an RR, and I-frame transmission from B can resume.

HDLC Operation Example



(d) Reject recovery (e) Timeout recovery

Stallings Figure 7.9d shows an example of error recovery using the REJ command. In this example, A transmits I-frames numbered 3, 4, and 5. Number 4 suffers an error and is lost. When B receives I-frame number 5, it discards this frame because it is out of order and sends an REJ with an N(R) of 4. This causes A to initiate retransmission of I-frames previously sent, beginning with frame 4. A may continue to send additional frames after the retransmitted frames. An example of error recovery using a timeout is shown in Figure 7.9e. In this example, A transmits I-frame number 3 as the last in a sequence of I-frames. The frame suffers an error. B detects the error and discards it. However, B cannot send an REJ, because there is no way to know if this was an I-frame. If an error is detected in a frame, all of the bits of that frame are suspect, and the receiver has no way to act upon it. A, however, would have started a timer as the frame was transmitted. This timer has a duration long enough to span the expected response time. When the timer expires, A initiates recovery action. This is usually done by polling the other side with an RR command with the P bit set, to determine the status of the other side. Because the poll demands a response, the entity will receive a frame containing an N(R) field and be able to proceed. In this case, the response indicates that frame 3 was lost, which A retransmits.