UTKAL GOURAV MADHUSUDAN INSTITUTE OF TECHNOLOGY, RAYAGDA

Computer Network and Data Communication

Lecture Notes designed by

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1/1/2025

This document presents the lecture notes which is needed to the students of Diploma, CSE/IT/ETC branch. This note is in accordance with SCTE&VT syllabus.

Computer Network and Data Communication

Network and Protocol

Data Communication

- Exchange of data and information between two devices within communication system, via some transmission media.
- Effectiveness of communication depends on four fundamental characteristics, such as,
  - Delivery
  - Accuracy
  - Timeliness
  - Jitter: It is the variation in the packet arrival time.

Components of Data communication

- Sender
- Receiver
- Message
- Transmission media
- Protocol

Data Flow

- Simplex
- Half Duplex
- Full Duplex

Networks

Physical Topology

- Bus Topology
- Ring Topology
- Mesh Topology
  - Number of links= n(n-1)/2
- Star Topology
- Hybrid Topology

Categories of Network

- LAN
- MAN
- WAN
- Internet

Protocols and Standards

Prootocols

- Protocol is a set of rules that govern data communications. A protocol defines what is communicated, how it is communicated, and when it is communicated. The key elements of a protocol are syntax, semantics and timing.
- Syntax:
- Semantics:

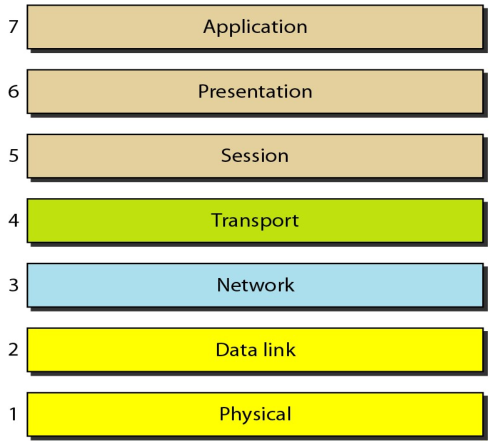
- Timing:
- Examples: http, ftp, SMTP etc.

Standards

- Standards provide guidelines to manufacturer, vendors, Govt. agencies and other service providers to ensure the kind of interconnectivity necessary in today's marketplace in the international communications.
- Two types:
  - De facto : not been approved by an organized body but have been adopted as standards through widespread use.
    - fdf
  - De jure : legislated by an officially recognized body.
- Standard Organizations
  - ISO(International Standard Organization):
    - Created in 1947
    - Multinational body (82 nation)
    - Dedicated to worldwide agreement oninternational standards in a variety of fields of network
    - Credited for building OSI model
  - ITU-T
  - ANSI
  - IEEE
  - EIA etc

OSI Model

- ISO is the organization, OSI is the mode.
- Consists of seven layers
- **Please Do Not Try Same Problem Again**
-



[Seven layers of OSI model]

Physical Layer:

- **Data encoding determines**:
    - Which **signal pattern** represents a binary 0 and a binary 1.
    - How the **receiving station recognizes** when an encoded bit starts.
    - How the **receiving station delimits** a **frame**.
- The **physical components** (such as wiring, connectors, and pin-outs) **determine**:
    - Whether an **external transceiver** is used to **connect** to the **medium**.
    - How **many pins** the connectors have and what **role each pin** performs.

Data Link Layer:

- **Frame**:

    - A frame is a **digital data transmission unit** that includes frame **synchronization**, i.e. a **sequence** of **bits** or symbols making it possible for the **receiver** to **detect** the **beginning** and **end** of the packet in the stream of symbols or bits.

    - A frame is usually **transmitted serial bit** by **bit** and contains a **header** field and a **trailer**.

    - Frame synchronization is the process of synchronizing display pixel scanning to a synchronization source. When **several systems** are **connected**, a sync signal is fed from a master system to the other systems in the network, and the displays are synchronized with each other.

- **Header** : The flag and **address fields** constitute the header.

- **Trailer** : The frame **check sequence** and second **flag fields** constitute the trailer.

- **Functions**

    - **Establishment** and **termination** of **logical links** (virtual-circuit connection) between two computers **identified** by their **unique network adapter addresses**.

- **Control** of **frame flow** by instructing the transmitting computer not to transmit frame **buffers**.
- **Sequential transmission** and **reception** of **frames**.
- **Providing** and **listening** for **frame acknowledgment**, and **detecting** and **recovering** from **errors** that **occur** in the **physical layer** by **retransmitting non-acknowledged frames** and handling **duplicate frame** receipts.
- Management of **media access** to determine when the computer is permitted to use the **physical medium**.
- Delimiting of frames to **create** and **recognize frame boundaries**.
- **Error-checking** of frames to **confirm** the **integrity** of the received frame.
- **Inspection** of the **destination address** of **each received** frame and **determination** of whether the frame should be **directed** to the **layer above**.

Network Layer

- **Packet** :
  - A packet is one unit of **binary data capable** of being **routed** through a computer network. To **improve communication performance** and **reliability**, each message sent between **two network devices** is often **subdivided** into **packets** by the **underlying hardware** and **software**.
  - **Packet formats** generally include a **header**, the **body** containing the message data, and sometimes a **footer**. The packet **header** lists the **destination** of the **packet** (in IP packets, the destination IP address) and often indicates the **length** of the **message data**. The packet **footer** contains data that signifies the **end of** the **packet**, such as a special sequence of bits known as a magic number. Both the packet header and footer may contain error-checking information.
- Functions
  - **Transfer** of **frames** to a **router** if the network address of the destination does not indicate the network to which the computer is attached.
  - **Control** of **subnet traffic** to allow an intermediate system to **instruct** a **sending station not to transmit** its frames when the **router's buffer fills up**. If the router is busy, the network layer can instruct the sending station to use an alternate router.
  - **Fragmentation** of **frames** by a **router** when the **size** of a **link** to a **downstream router's maximum transmission unit** (MTU) is **smaller** than the **frame size**. The frame fragments are reassembled by the destination station.
  - **Resolution** of the **logical computer address** (on the network layer) with the **physical network adapter address** (on the data-link layer), if necessary.

Transport Layer

- Transport layer is responsible for process-to-process delivery.
- **Segment** :
  - This **division** of the **Data Stream** into **Smaller Pieces** is called segmentation. Segmenting messages has two primary benefits.
  - First, by sending smaller individual pieces from source to destination, **many different conversations** can be **interleaved** on the network. The process used to interleave the pieces of separate conversations together on the network is called **multiplexing**.
  - Second, segmentation can **increase** the **reliability** of **network communications**. The separate pieces of each message **need not travel** the **same pathway** across the network from source to destination. If a **particular path** becomes **congested** with data traffic or **fails**, individual pieces of the message can still be directed to the destination using **alternate pathways**. If part of the message **fails** to make it to the **destination**, **only** the **missing parts** need to be **retransmitted**.
- Functions
  - Accepting messages from the layer above and, if necessary, **splitting** them into **segments**.
  - Providing reliable, **end-to-end message delivery** with **acknowledgments**.
  - Instructing the transmitting computer **not** to **transmit** when **no reception buffers** are **available**.
  - **Multiplexing** several **process-to-process** message streams or sessions onto one **logical link** and tracking which messages belong to which sessions.

Session Layer

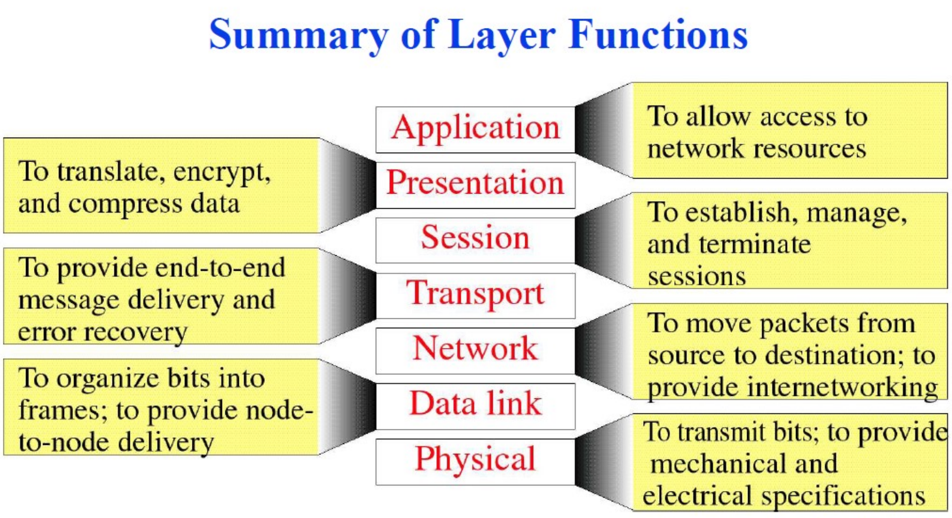
- **Permits application processes** to **register unique** process **addresses**, such as NetBIOS names. The **session layer uses** these **stored addresses** to help resolve the **addresses** of **network adapters** from **process addresses**.
- **Establishing**, **monitoring**, and **terminating** a **virtual-circuit** session between **two processes identified** by their **unique process addresses**. A virtual-circuit session is a direct link that exists between the sender and receiver.
- Delimiting messages to **add header information** that indicates where a **message starts** and **ends**. The receiving session layer can then **refrain** from indicating the presence of **any message data** to the **overlying application until** the **entire message** is **received**.
- **Performing Message Synchronization :** Coordination of the data transfer between the sending session layer and the receiving session layer. Synchronization prevents the receiving session layer from being overrun with data. This transfer is **coordinated** with **acknowledgement messages** (ACKs). ACKs are sent **back** and **forth** between **both ends** of the transfer and **notify** of the **state** of the **receiving buffer** to **accept additional data**.
- Performing other support functions that **allow processes** to communicate over the network, such as user **authentication** and **resource-access security**.

Presentation Layer

- Synchronization : While transferring data, a provision must be made to provide the **check points** at **various places** so that in case the connection is **broken** and **reestablished**, the transition runs **not lost** even if the user has not committed. This activity is called Synchronization.
- **Character-code translation**, such as from **ASCII** to **EBCDIC**.
- **Data conversion**, such as bit order reversal, CR to CR/LF, and integer to floating point.
- **Data compression**, which reduces the number of bits that need to be transmitted.
- **Data encryption** and decryption, which secures data for transmission across a potentially insecure network. One use of encryption is for transmission of a password to a receiving computer.

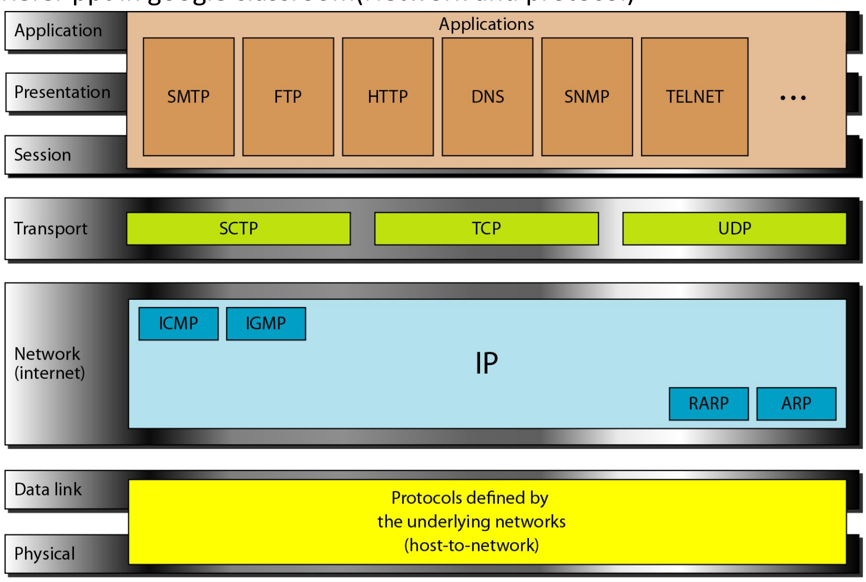
Application Layer

- **Resource sharing** and **Device Redirection**
- **Remote File Access**
- Remote **printer** access
- **Inter-Process Communication** support
- **Remote Procedure Call** support
- **Network Management**
- **Directory Services**
- **Electronic Messaging**, including e-mail messaging
- **Simulation** of **Virtual Terminals**



TCP/IP protocol suite

- Assignment
- Refer ppt in google classroom(Network and protocol)



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- Protocols:
  - Network Layer
    - IP
    - ARP
    - RARP
    - ICMP
    - IGMP
  - Transport Layer
    - UDP: Connectionless and unreliable transport protocol.
    - TCP: Connection oriented and reliable transport protocol.
    - SCTP
  - Application Layer
    - SMTP
    - FTP
    - HTTP

        - DNS
        - SNMP
        - TELNET

Addressing

- Assignment
- Physical address
    - Changes from hop to hop.
- Logical address
    - Physical addresses will change from hop to hop, but logical address usually remain same.
- Port address
    - IANA (Internet Assigned Number Authority) has divided port numbers into three ranges such as,
        - Well-known ports: 0 to 1023, assigned and controlled by IANA.
        - Registered ports: 1024 to 49151, not assigned or controlled by IANA.
        - Dynamic ports: 49152 to 65535, are neither controlled nor registered.
    - Physical addresses will change from hop to hop, but logical address and port address usually remain same.
- Specific address

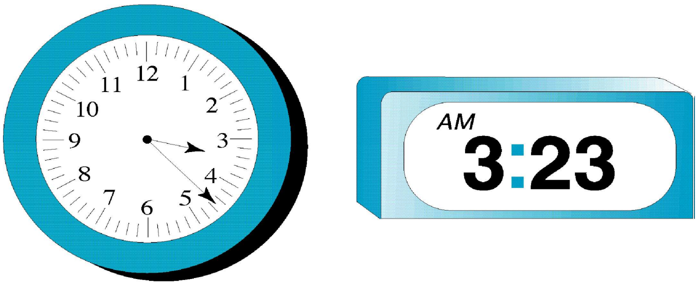
Data Transmission and media

Data transmission Concepts and Terminology

- Data and Signal: In order to send the data, it must be converted into signal.
- Signal may be electrical, electromagnetic or optical.

Analog and Digital Data/Signal

- Analog data/signal are continuous and take continuous values, whereas digital data/signal have discrete states and take discrete values.



- a. Analog b. Digital

Difference between the signals

Analog Signals	Digital signals
Continuous infinite no. of values over period of time	Limited no of defined values
Ex : microphone converts voice data into voice signal	Data in memory
Uses band pass channel	Uses low pass channel

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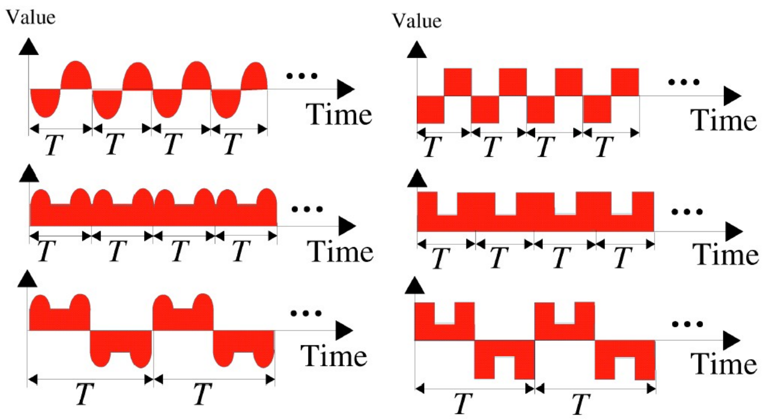
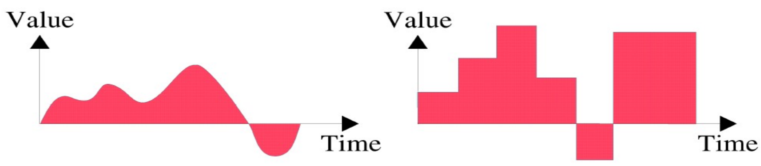
Periodic and nonperiodic signal

- A periodic signal completes a pattern within a measurable time period called periodand repeats the pattern in subsequent period.



- a. Analog b. Digital
- Non periodic or Aperiodic changes without exhibiting a pattern or acycle that repeats over time.



a. Analog b. Digital

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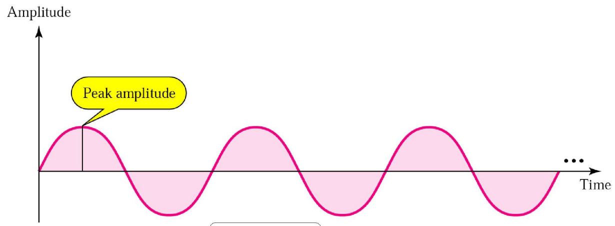
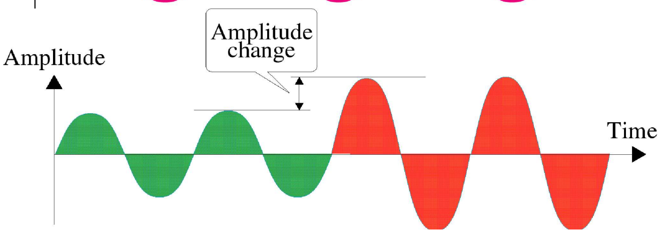
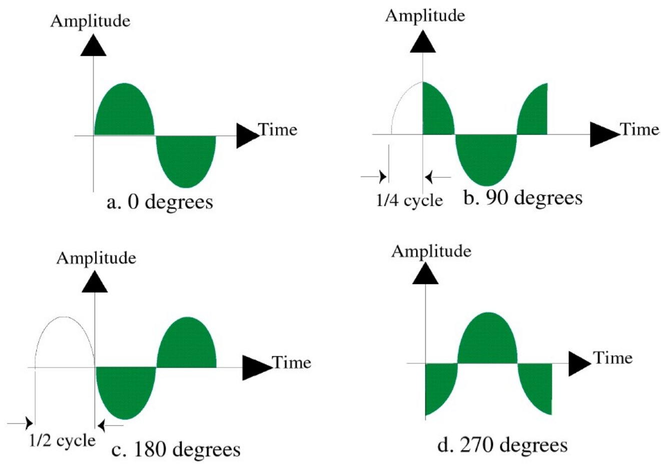
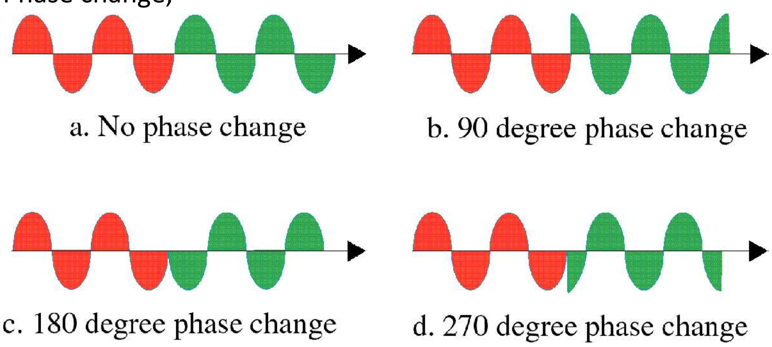
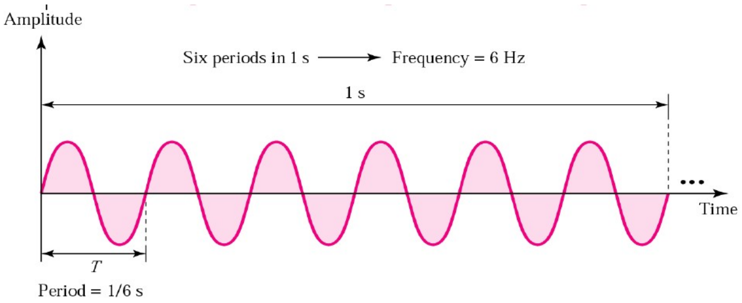
- Periodic signals Characterized by the following parameters
    - Amplitude
        - The value of a signal at a particular point of consideration.
        - Normally measured in volts.
        - 

        - 

    - Phase
        - It gives a measure of the relative positionof 2 signals in time or Phase describes theposition of the waveform relative to time zero.
        - Measured in degree or radian.
        - 

        - Phase change,
        - 

- Frequency
  - Number of periods in 1 second.
  - Inverse of time period.
  - Measured inHertz
  - **Period**: A Period is the amount of time (in second), a signal needs to complete 1 cycle.**Period**: A Period is the amount of time (in second), a signal needs to complete 1 cycle.



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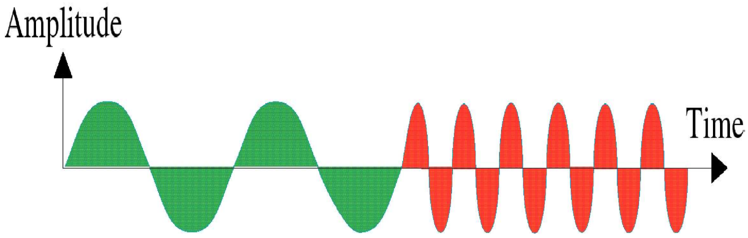
  - Frequency change,



Units of periods and frequencies

Unit	Equivalent	Unit	Equivalent
Seconds (s)	1 s	hertz (Hz)	1 Hz
Milliseconds (ms)	10^{-3} s	kilohertz (KHz)	10^{3} Hz
Microseconds (ms)	10^{-6} s	megahertz (MHz)	10^{6} Hz
Nanoseconds (ns)	10^{-9} s	gigahertz (GHz)	10^{9} Hz
Picoseconds (ps)	10^{-12} s	terahertz (THz)	10^{12} Hz

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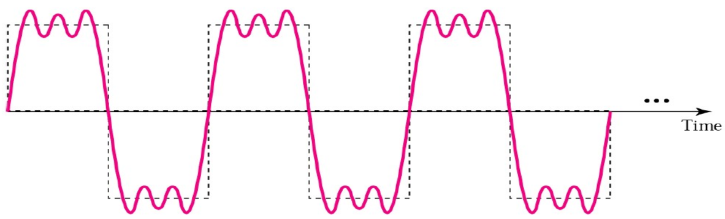
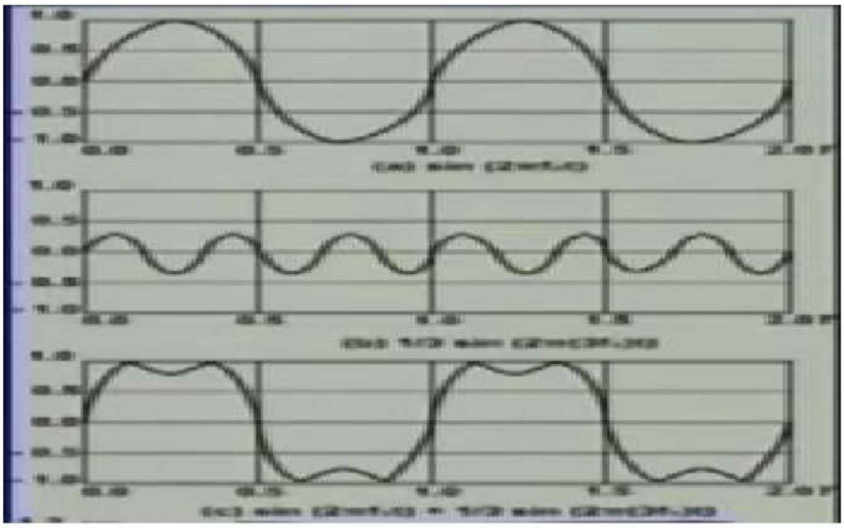
- Wavelength
  - It is the distance travelled by the signal in one period.
  - Usually describes the optical signal.
  - Wavelength= propagation speed x period = propagation speed/frequency

Composite signal

- If we vary the Amplitude ,frequency &Phase we get a composite signal.
- Composite signals cannot be viewed inoscilloscope

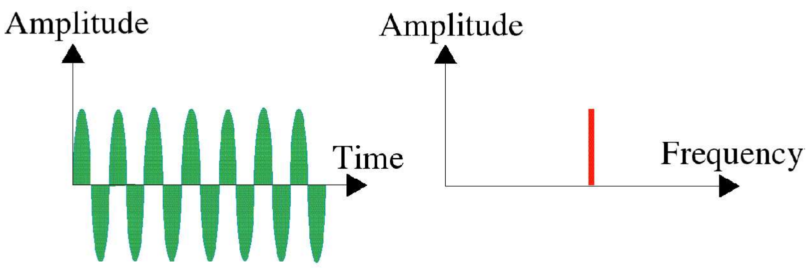


- Example of composite signal,
- Time and Frequency domain
    - An electromagnetic Signal Consists of manycomposite signals made up of many frequency.
    - Fourier analysis: Any composite signal can berepresented or expressed as a combination ofsimple sine wave with different amplitude,frequency and phase
    - S(t)=A1sin(2Πf1t1+Ø1)+ A2sin(2Πf2t2+Ø2)+ …..



$$s(t) = 1\sin(2\Pi f_1 t) + 1/3\sin(2\Pi(3f_1)t)$$

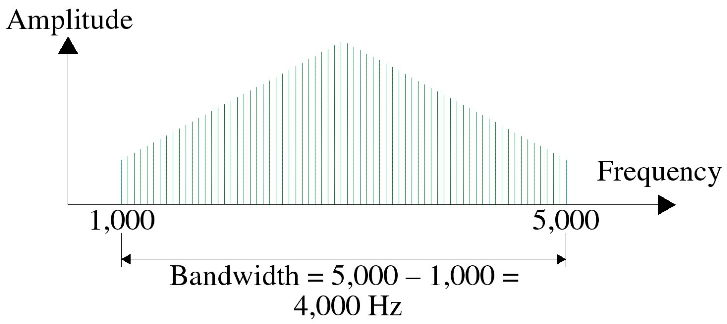
    -
    - Frequency Spectrum : Frequency spectrum is the range offrequencies a signal contains.
    - Time and frequency domain:



a. Time domain b. Frequency domain

Bandwidth

- The bandwidth is a property of a medium.
- Itis the difference between the highest andthe lowest frequencies that the medium cansatisfactorily pass.



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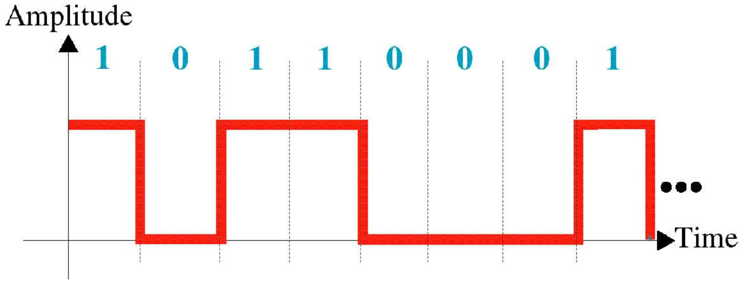
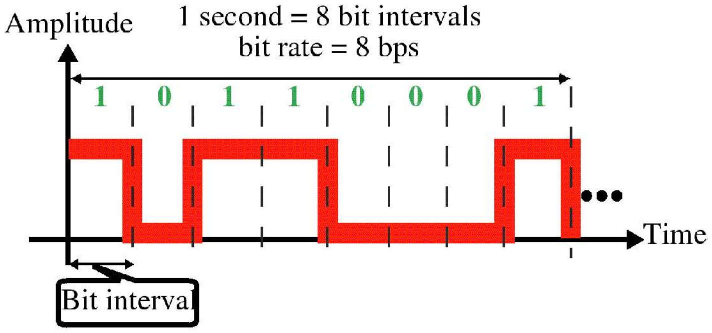
- The analog bandwidth of a medium isexpressed in hertz; the digitalbandwidth, in bits per second.

Analog and Digital Data transmission

- Digital Signals
  - Data can be represented in a digitalform



  - A digital signal is a composite signal with infinite bandwidth.
  - Usually we send 1 bit per level ,we cansend 2 bits per level for which we require 4signal level.
  - In general , if a signal has L levels eachlevel needs $\log_2 L$ bits.
- Bit rate and Bit Interval
  - Bit interval : Time Required to send asingle bit
  - Bit rate :It is the number of bit intervalper second



  - Bit Length : Bit Length is the distance onebit occupies on the transmission medium.
  - Bit length = propagation speed * bitduration
  - Bit interval=1/bitrate

Transmission of Digital Signals

- A digital signal ,periodic, aperiodic is acomposite analog signal with frequencybetween 0 to infinity.
- A digital signal can be transmitted using 2different approach.
  - Baseband
  - Broadband

Baseband Transmission:

- Sending a digital signal over a channel without changing the digital signal to an analog signal.
- Baseband Transmission requires a **low pass channel**, a channel with a bandwidth that starts from '0'.
- This is a case if we have a dedicated medium with a bandwidth constituting only one channel e.g., the entire bandwidth of a cable connecting two computers is one channel.
- **Low pass channel**:
  - Narrow bandwidth
  - Wide bandwidth
- Wide bandwidth:
  - preserves the exact form of a non-periodic digital signal with vertical and horizontal segments.
  - Possible if we have a dedicated medium with an infinity bandwidth between sender and receiver that preserves the exact amplitude of each component of the composite signal e.g., CPU ←→Memory

Broadband Transmission

- Means changing the digital signal to ananalog signal for transmission
- Broadband transmission uses modulation .
- Modulation enable us to use band pass channel
- **Bandpass channel**:
  - It is a channel with a bandwidth thatdo not start with zero .
  - Digital Signals cannot be passed throughbandpass channel(Conversion is required)

Digital signal as CompositeSignal

- Digital signal is nothing but a compositeanalog signal with an infinite bandwidth.
- A digital signal theoretically needs abandwidth between 0 and infinity. Thelower limit 0 is fixed. The upper limit maybe compromised.

Transmission impairments, Channel capacity

Attenuation

- loss of energy (reasonresistance of the medium)
- Measured in decible(db)
- $dB=10 \log_{10} P2/P1$

Distortion

- signal changes shape andform.

Noise

- can be thermal noise , inducednoise etc.

Signal to Noise Ratio (SNR)

- Signal to Noise Ratio= Avg signal power / Avgnoise power
- High SNR means signal is less Corrupted
- Less SNR means signal is high Corrupted

Data Rate Limit

- Data rate Depends on 3 factors
  - Bandwidth
  - Level of signal
  - Quality of signal (level of noise)

- 2 Relations to calculate the data rate
    - Nyquest for noiseless channel
    - Shannon for noisy channel
- Nyquist Bit Rate
    - max bit rate for anoiseless channel
        - Bitrate=2* bandwidth *$\log_2 L$, where L is no. of signallevel
- Shannon capacity
    - determines the highest bit rate in a noisychannel
    - Capacity=bandwidth*log(1+SNR)

Performance

- Bandwidth
    - In networking we use the term bandwidth in two contexts,
        - Bandwidth in ***hertz***, refers to the range of frequencies in a composite signal or the range of frequencies that a channel can pass.
        - Bandwidth in ***bits per second***, refers to the speed of bit transmission in a channel or link.
- Throughput
    - It is a measure of how fast we can actually send data through a network.
    - Bandwidth is the potential measurement of a link, whereas the throughput is the actual measurement of how fast we can send data.
- Latency
    - Latency= propagation time + transmission time + queuing time + processing delay
- Transmission time
    - Transmission time= message size/bandwidth
- Bandwidth Delay Product
    - The number of bits that can fill the link.
- Jitter
    - It is the variation in the packet arrival time.

Transmission media, Guided Transmission, Wireless Transmission

- Assignment (Refer ppt in google classroomTransmission media)

Data Encoding

Data encoding,

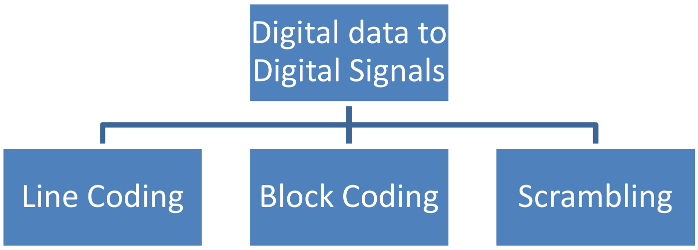
Conversion Technique

- What type of signal we should use it depends on the Situation and available Bandwidth.
- Conversion Technique

Data	Signal	Approach
Digital	Digital	Encoding
Analog	Digital	Encoding
Analog	Analog	Modulation
Digital	Analog	Modulation

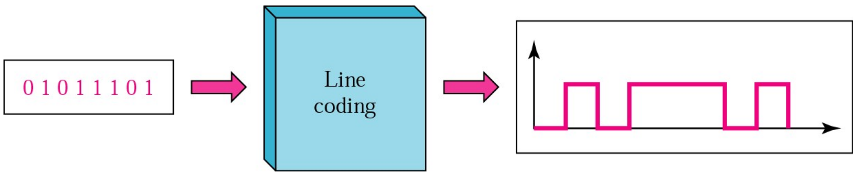
-

Digital data digital signals



Line coding

- Line Coding is a technique for converting digital data into Digital signals



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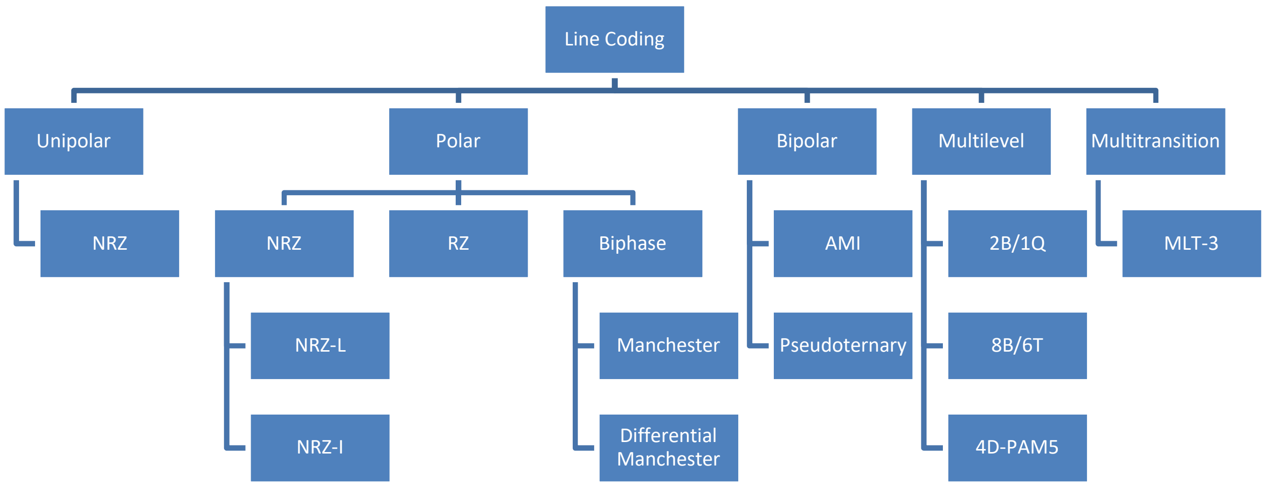
- Types of line coding schemes
    - Unipolar → NRZ (Above diagram)
    - Polar→ NRZ, RZ, and Biphase (Manchester, Differential Manchester)
    - Bipolar → AMI and Pseudoternary
    - Multillevel→ 2B/1Q, 8B/6T, and 4D-PAM5
    - Multitransition→ MLT-3



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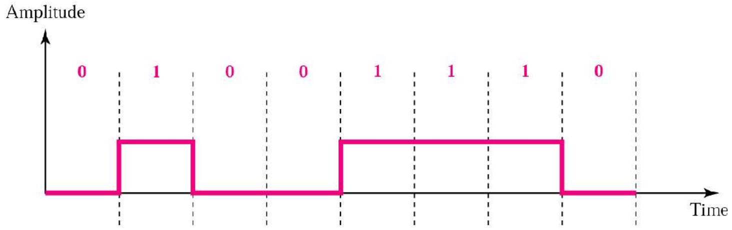
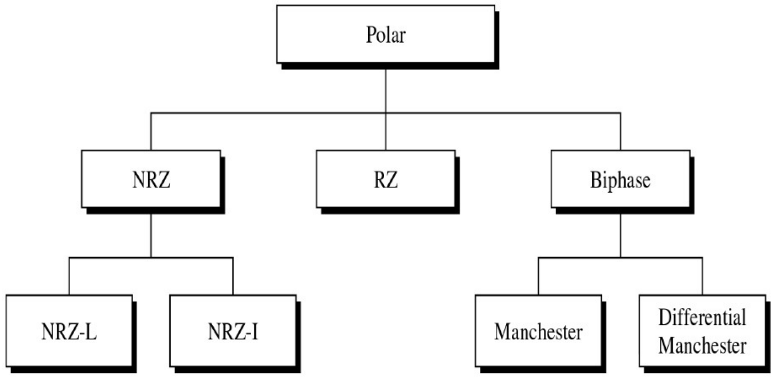
Unipolar (NRZ)

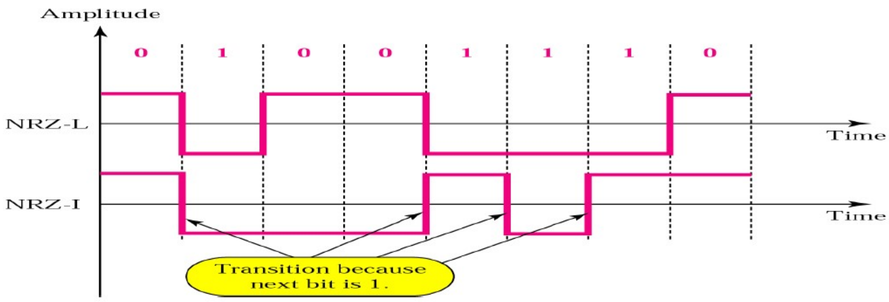
- It uses only one Voltage level
- 

- Positive voltage defines bit 1, zero voltage defines bit 0.
- It is called NRZ because the signal does not return to zero at the middle of the bit.
- Bit rate is same as data rate
- DC Component is Present
- Simple but Obsolete

Polar

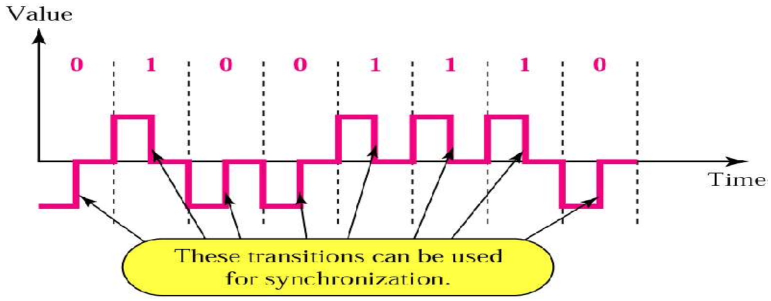
- Polar encoding uses two voltage levels(positive and negative).
- 

NRZ (Non Return To Zero) (NRZ-L, NRZ-I)

- Voltage level is constant during a bit interval.
- There are 2 NRZ scheme
    - NRZ L (L for Low level) (bit 1 –ve Volt, 0 +ve)
    - NRZ I (I for Inversion) (For each 1 in the bitsequence signal level is inverted,Transition of onevoltage level to another represents 1
    - 

RZ (Return to Zero)

- Characteristics
  - 3 signal Levels
  - Baud rate is double the bit rate
  - No DC current
  - Increases the bandwidth is the main limitation



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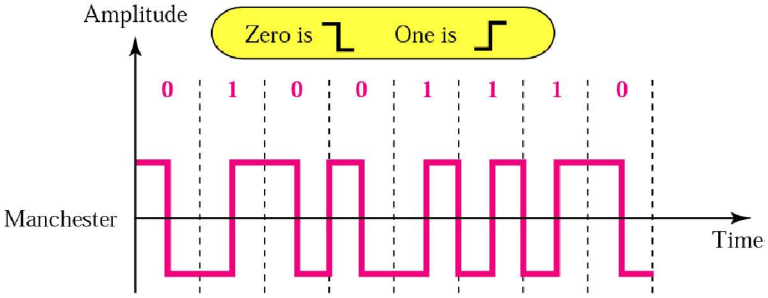
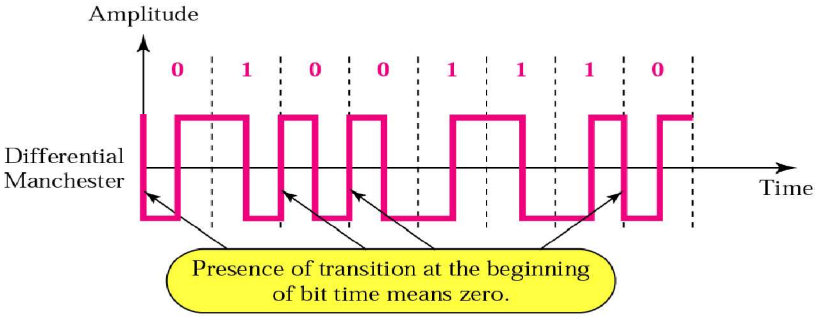
Biphase (Manchester, Differential Manchester)

- Manchester
  - In Manchester encoding, the transition at the middle of the bit is used for bothsynchronization and bit representation.
  - Low to high Represent 1 and high to low represents 0

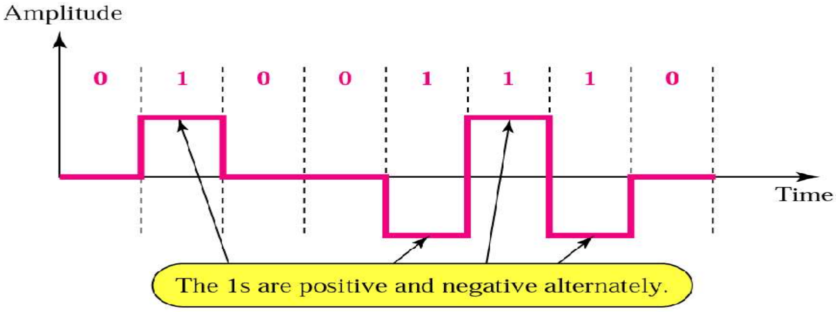


  -
- Differential Manchester
  - Presence of transition at the beginning represents 0
  - Uses inversion in the middle of each bit forsynchronization



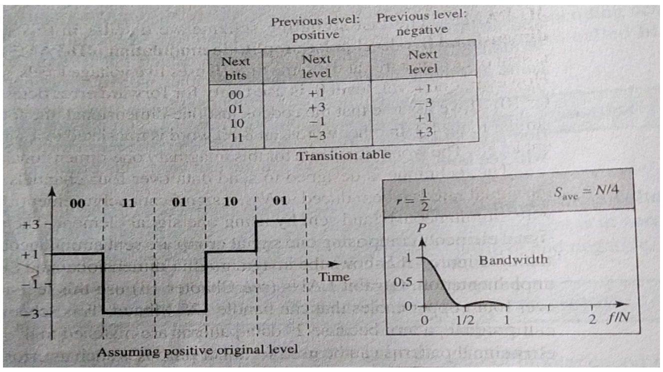
  -

Bipolar

- In bipolar encoding, we use threelevels: positive, zero,and negative.
- **Bipolar AMI (Alternate Mark Inversion)encoding**
    - Uses 3 voltage level.
    - Zero voltage Represent 0.
    - Binary 1 are represented as alternating positive and negative signals.
    - 

    - AMI encoding is good for LAN since itrequires Less Bandwidth.
    - Demerit is it does not provide anysynchronization for large sequence of0000…. And 1…….
    - So **Scrambling** is introduced into the largesequence of 000.. And 1111.. Forsequencing.
- **Pseudoternary**
    - This is a variation of AMI encoding.
    - Zero voltage Represent 1
    - Binary 0 are represented as alternating positive and negative signals

Multilevel

- The goal is to increase the number of bits per baud by encoding a pattern of m data elements into a pattern of n signal elements.
- In **mBnL** schemes, a pattern of **m** data element is encoded as a pattern of **n** signal elements in which **2^m<= L^n**, where **L** is the number of levels.
- **2B1Q**
    - It is the first **mBnL** scheme.
    - Two binary, one quaternary (2B1Q) uses data patterns of size 2 and encodes the 2-bit patterns as one signal element belonging to a four-level signal.
    - Here m=2, n=1 and L=4.
    - The average signal rate is S=N/4. This means we can send data 2 times faster than NRZ-L.
    - It is used in DSL (Digital Subscriber Line) to provide a high-speed connection to the internet by using subscriber telephone lines.
    - 

    -

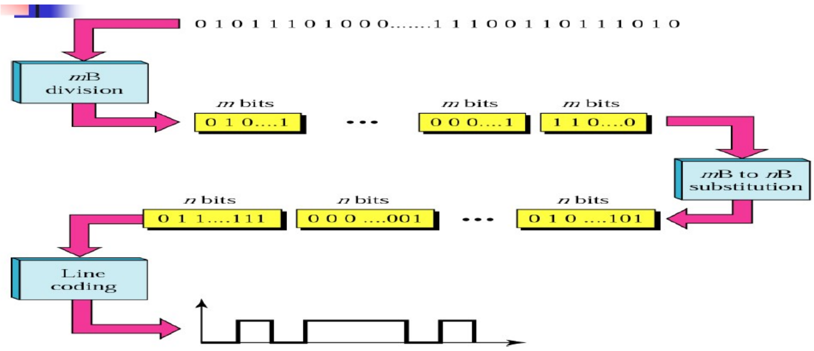
- **8B6T**
    - Assignment
- **4D-PAM5**
    - Assignment

Multitransition

- **MLT-3**
    - Assignment

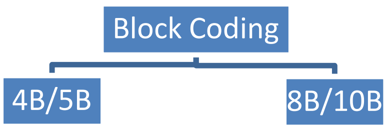
Block Coding

- Block coding is normally referred to as mB/nB coding.
- It replaces each m-bit group with an n-bit group.
- Block Coding was introduced to improve performance of the line coding
- Introduces redundancy to achieve synchronization
- Also allows error detection to some extents



Three Basic Steps : Division , Substitution and Line Coding

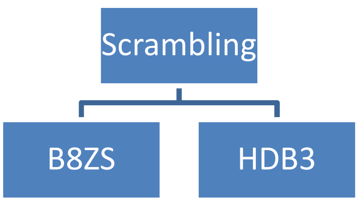
- Two types



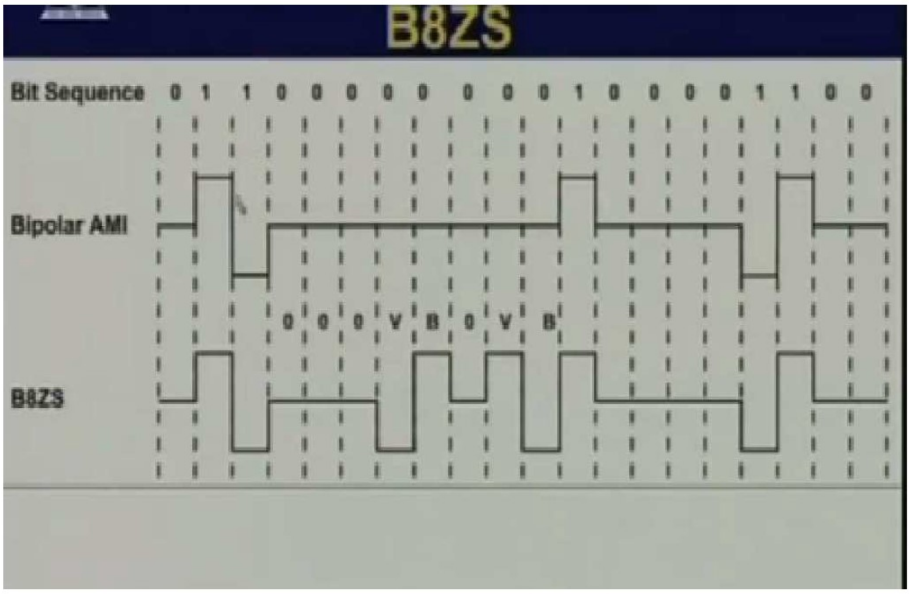
- **4B5B**
    - Assignment
- **8B10B**
    - Assignment

Scrambling

- Two types



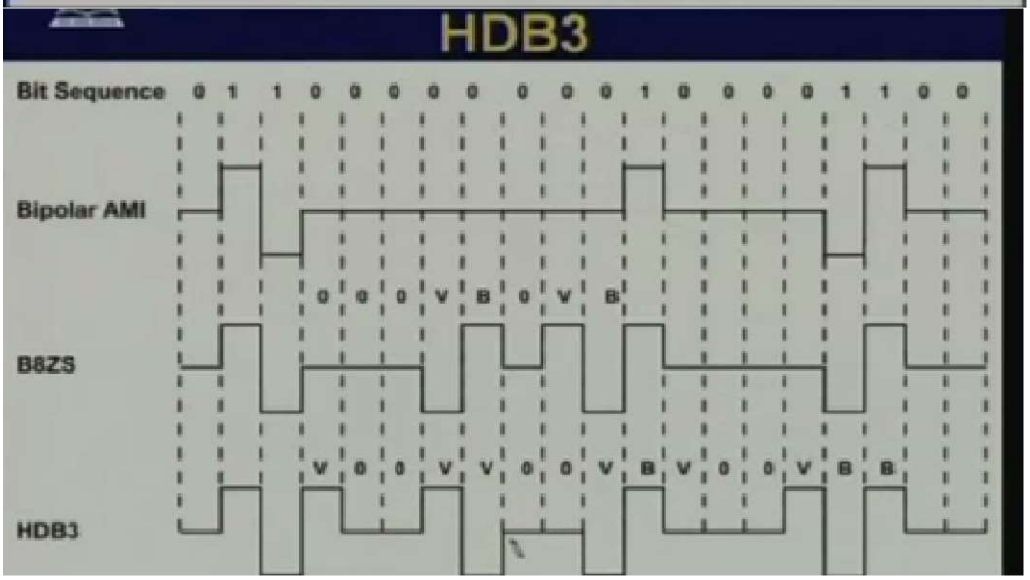
- **B8ZS**
  - The limitation of bipolar AMI overcame inB8ZS ,which is used in North America.
  - A sequence of **8 zero's** is replaced by **000+-0+-** ,if the **previous pulse is positive.**
  - A sequence of **8 zero's** is replaced by **000-+0+-** ,if the **previous pulse was negative.**



  - V: bipolar Violation , B : Valid bipolar Signals

- **HDB3 (High Density bipolar – 3 Zeros)**
  - Another alternatives , which us used in Europe and Japan is HDB3
  - It replaces a sequence of 4 zeros by a code asper the rule given in the table

HDB3 substitution rule		
Polarity of the Preceding pulse	Number of bipolar pulses (ones) since last substitution	
	odd	even
—	000 —	+ 00 +
+	000 +	— 00 —



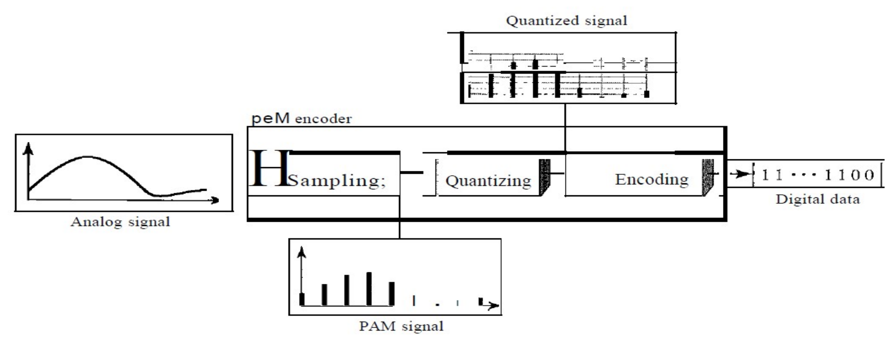
Analog data digital signals

- Most of the time we have an analog signal such as one created by a microphone or camera, and this has to be sent over a medium, which is capable of sending digital signal.
- Primarily two techniques are used,
    - Pulse code modulation (PCM)
    - Delta Modulation (DM)

Pulse Code Modulation (PCM)

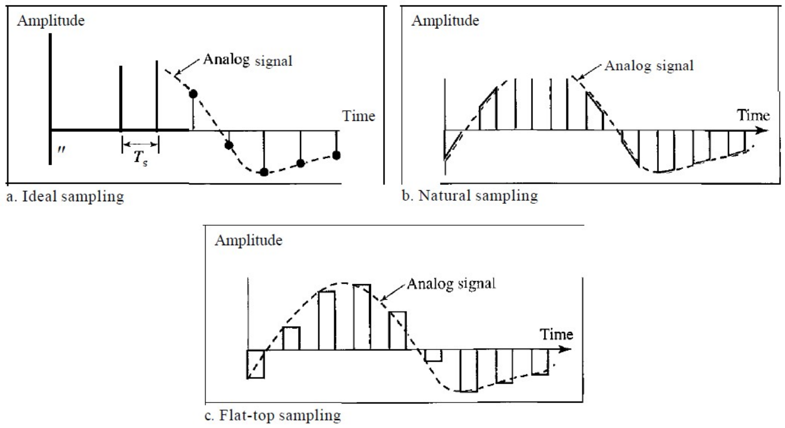
- Assignment
- It is the most common technique to change analog signal to digital data (digitization).

Figure 4.21 *Components of PCM encoder*



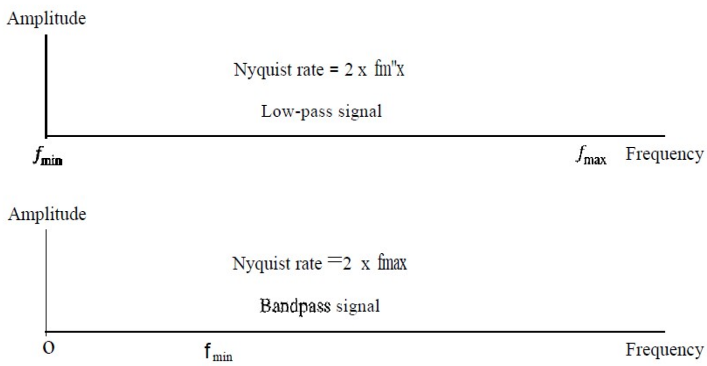
-
- A PCM encoder has three process
    - Sampling : Analog signal is sampled.
    - Quantizing: The sampled signal is quantized.
    - Encoding: The quantized values are encoded as streams of bits.
- **Sampling or Pulse Amplitude Modulation (PAM)**
    - The analog signal is sampled at every T_s sec, where T_s is the sample interval period.
    - **sampling rate** (f_s)**:** The inverse of sample interval (T_s) is called the **sampling rate** or **sampling frequency** (f_s), or f_s=1/ T_s .
    - There are three sampling methods such as,
        - **Ideal** : In ideal sampling, pulses from the analog signal are sampled. This cannot be implemented easily.
        - **Natural:** In natural sampling, a high speed switch is turned on for only the small period of time when the sampling occurs. The result is a sequence of samples that retains the shape of the analog signal.
        - **Flat-top:** The most common sample method, called **sample and hold**, creates flat-top samples by using circuit.

Figure 4.22 *Three different sampling methods for PCM*



- The result of sampling is still an analog signal with non-integral values.
- **Nyquist Theorem:**
    - To reproduce the original analog signal, the sampling rate (f_s) must be at least twice the highest frequency contained in the original signal.
    - A signal can be sampled, only if it is band-limited. A signal with an infinite bandwidth cannot be sampled.
    - The sampling rate must be at least 2 times the highest frequency, not the bandwidth.
    - If the analog signal is **low pass** (which starts from 0), the bandwidth and the highest frequency are same.
    - If the analog signal is **bandpass** (which starts from a particular frequency), the bandwidth value is lower than the value of the maximum frequency.

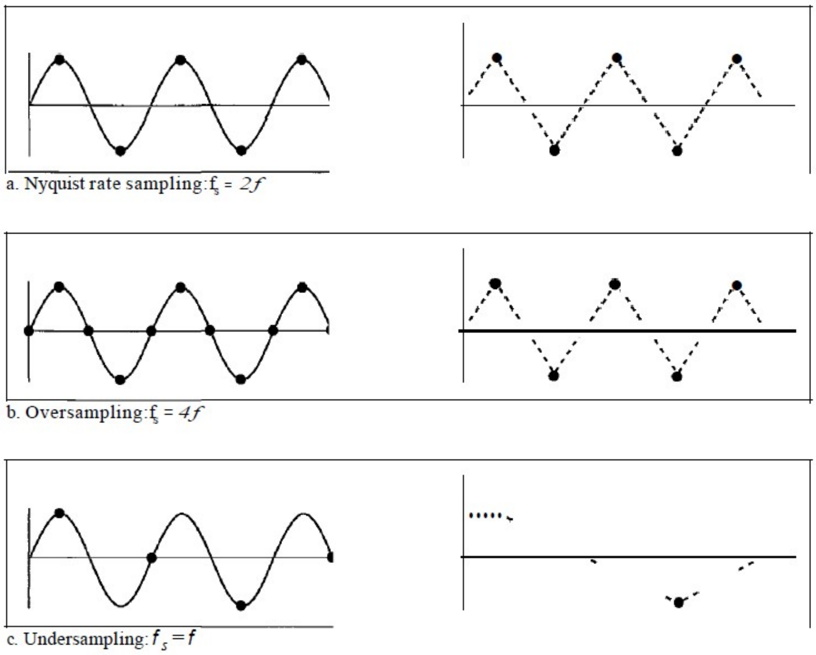
Figure 4.23 *Nyquist sampling rate for low-pass and bandpass signals*



Example 4.6

For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: $fs = 4f$(2 times the Nyquist rate)'/s $= 2f$(Nyquist rate), and $f_s = f$ (one-half the Nyquist rate). Figure 4.24 shows the sampling and the subsequent recovery of the signal.

Figure 4.24 *Recovery of a sampled sine wave for different sampling rates*



It can be seen that sampling at the Nyquist rate can create a good approximation of the original sine wave (part a). Oversampling in part b can also create the same approximation, but it is redundant and unnecessary. Sampling below the Nyquist rate (part c) does not produce a signal that looks like the original sine wave.

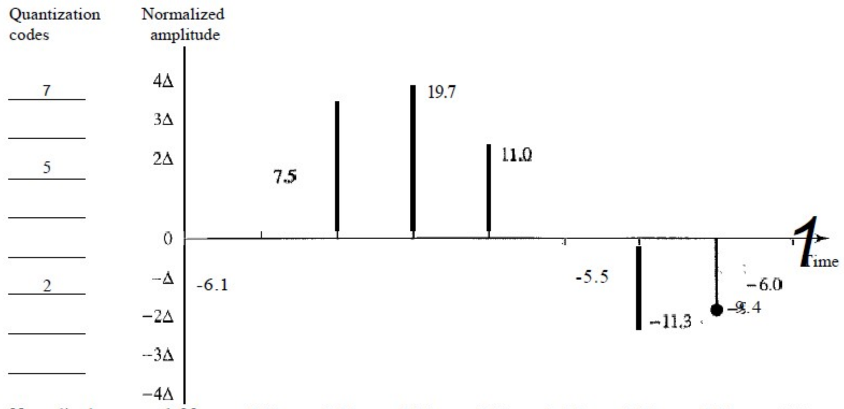
- **Quantization**
  - The result of sampling is a series of pulses with amplitude values between the maximum and minimum amplitude of the signal.
  - The set of amplitudes can be infinite with non-integral values between the two limits. These values cannot be used in the encoding process.
  - The following are the steps in quantization:
    - We assume that the original analog signal has instantaneous amplitudes between Vmin and Vmax.
    - We divide the range into L zones, each of height (delta).

$$\Delta = \frac{V_{max} - V_{min}}{L}$$

    - We assign quantized values of 0 to L - I to the midpoint of each zone.
    - We approximate the value of the sample amplitude to the quantized values.

As a simple example, assume that we have a sampled signal and the sample amplitudes are between -20 and +20 V. We decide to have eight levels (L = 8). This means that Δ =5 V. Figure 4.26 shows this example.

Figure 4.26 *Quantization and encoding ofa sampled signal*



Normalized PAM values	-1.22	1.50	3.24	3.94	2.20	-1.10	-2.26	-1.88	-1.20
Normalized quantized values	-1.50	1.50	3.50	3.50	2.50	-1.50	-2.50	-1.50	-1.50
Normalized error	-0.38	O	+0.26	-0.44	+0.30	-0.40	-0.24	+0.38	-0.30
Quantization code	2	5	7	7	6	2		2	2
Encoded words	010	101	111	111	110	010	001	010	OW

-
- We have shown only nine samples using ideal sampling (for simplicity). The value at the top of each sample in the graph shows the actual amplitude. In the chart, the first row is the normalized value for each sample (actual amplitude/Δ). The quantization process selects the quantization value from the middle of each zone. This means that the normalized quantized values (second row) are different from the normalized amplitudes. The difference is called the normalized error (third row). The fourth row is the quantization code for each sample based on the quantization levels at the left of the graph. The encoded words (fifth row) are the final products of the conversion.

- **Quantization Levels:** In the previous example, we showed eight quantization levels. The choice of L, the number of levels, depends on the range of the amplitudes of the analog signal and how accurately we need to recover the signal. If the amplitude of a signal fluctuates between two values only, we need only two levels; if the signal, like voice, has many amplitude values, we need more quantization levels. In audio digitizing, L is normally chosen to be 256; in video it is normally thousands. Choosing lower values of L increases the quantization error if there is a lot of fluctuation in the signal.
- Quantization Error : One important issue is the error created in the quantization process. (Later, we will see how this affects high-speed modems.) Quantization is an approximation process. The input values to the quantizer are the real values; the output values are the approximated values. The output values are chosen to be the middle value in the zone. If the input value is also at the middle of the zone, there is no quantization error; otherwise, there is an error. In the previous example, the normalized amplitude of the third sample is 3.24, but the normalized quantized

value is 3.50. This means that there is an error of +0.26. The value of the error for any sample is less than Δ/2. In other words, we have -Δ/2 ≤ error ≤ Δ/2.

The quantization error changes the signal-to-noise ratio of the signal, which in turn reduces the upper limit capacity according to Shannon.

It can be proven that the contribution of the quantization error to the SNRdB of the signal depends on the number of quantization levels L, or the bits per sample n_b, as shown in the following formula:

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

- Uniform Versus Nonuniform Quantization

- Encoding
  - $Bit\ rate = sampling\ rate\ \times number\ of\ bits\ per\ sample = f_s \times n_b$

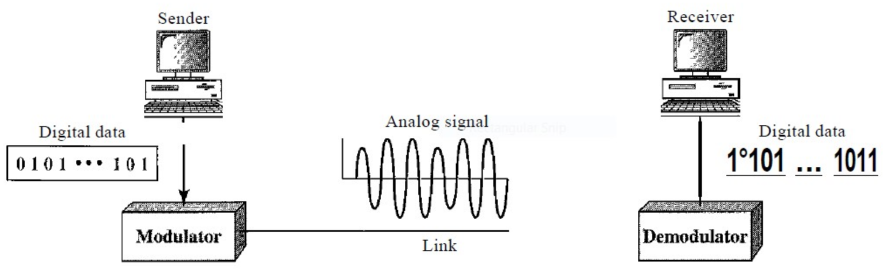
Delta Modulation

- Assignment
- Modulator
- Demodulator
- Adaptive Delta Modulation
- Quantization Error

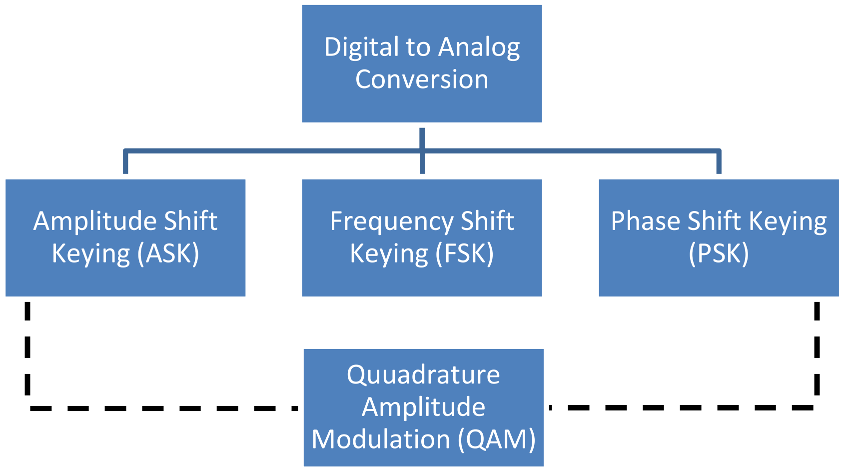
Digital data analog signals

- **Digital-to-**analog **conversion** is the process of changing one of the characteristics of an analog signal based on the information in digital data.

Figure 5.1 *Digital-to-analog conversion*



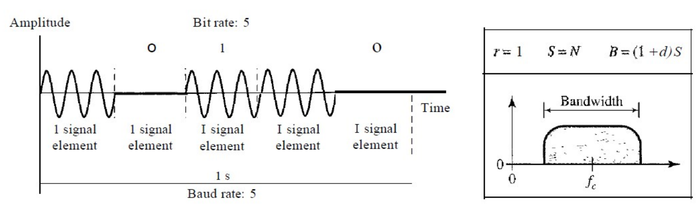
-
- *Data Element Versus Signal Element:* We defined a data element as the smallest piece of information to be exchanged, the bit. We also defined a signal element as the smallest unit of a signal that is constant. Although we continue to use the same terms in this chapter, we will see that the nature of the signal element is a little bit different in analog transmission. [forouzan book]
- *Data Rate Versus Signal Rate:* We can define the data rate (bit rate) and the signal rate (baud rate) as we did for digital transmission. The relationship between them is
  $S=Nx(1/r)$ baud
  where N is the data rate (bps) and r is the number of data elements carried in one signal element. The value of r in analog transmission is $r = \log2\ L$, where L is the type of signal element, not the level. The same nomenclature is used to simplify the comparisons. [forouzan book]
- Bit rate is the number of bits per second. Baud rate is the number of signal elements per second. In the analog transmission of digital data, the baud rate is less than or equal to the bit rate.
- In transportation, a baud is analogous to a vehicle, and a bit is analogous to a passenger. We need to maximize the number of people per car to reduce the traffic.



Amplitude Shift Keying (ASK)

- Both frequency and phase remain constant while the amplitude changes.
- **Binary ASK (BASK)**
    - Otherwise called as *on-off keying* (OOK).
    - Although we can have several levels (kinds) of signal elements, each with a different amplitude, ASK is normally implemented using only two levels.
    - The peak amplitude of one signal level is 0; the other is the same as the amplitude of the carrier frequency.

Figure 5.3 *Binary amplitude shift keying*

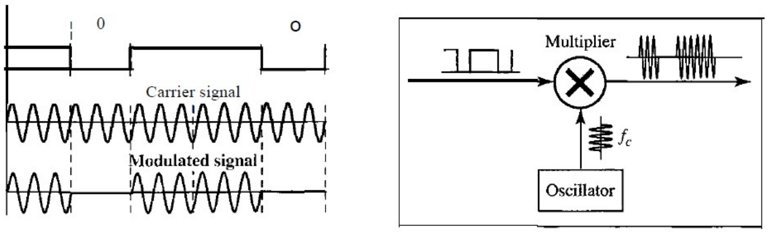


- the bandwidth is proportional to the signal rate (baud rate). However, there is normally another factor involved, called d, which depends on the modulation and filtering process. The value of d is between 0 and 1. This means that the bandwidth can be expressed as shown, where S is the signal rate and the B is the bandwidth.

$$B = (1 + d) \times S$$

- The formula shows that the required bandwidth has a minimum value of S and a maximum value of 2S.
- The most important point here is the location of the bandwidth. The middle of the bandwidth is where fc the carrier frequency, is located. This means if we have a bandpass channel available, we can choose our fc so that the modulated signal occupies that bandwidth. This is in fact the most important advantage of digital-to-analog conversion. We can shift the resulting bandwidth to match what is available.
- **Implimentation**: If digital data are presented as a unipolar NRZ (see Chapter 4) digital signal with a high voltage of I V and a low voltage of 0 V, the implementation can achieved by multiplying the NRZ digital signal by the carrier signal coming from an oscillator. When the amplitude of the NRZ signal is 1, the amplitude of the carrier frequency is held; when the amplitude of the NRZ signal is 0, the amplitude of the carrier frequency is zero.

Figure 5.4 *Implementation of binary ASK*

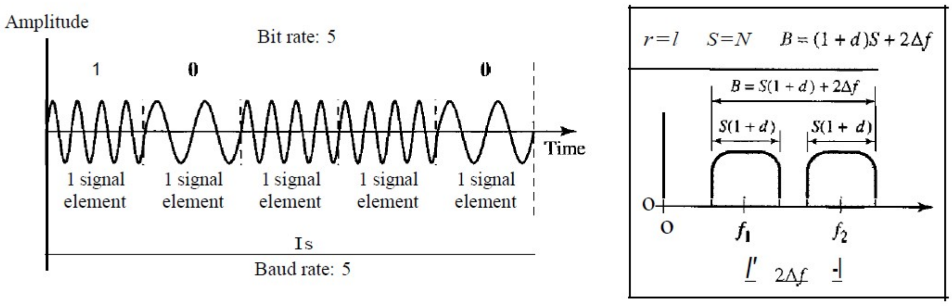


- **Multilevel ASK**
  - The above discussion uses only two amplitude levels. We can have multilevel ASK in which there are more than two levels. We can use 4,8, 16, or more different amplitudes for the signal and modulate the data using 2, 3, 4, or more bits at a time. In these cases, r= 2, r = 3, r =4, and so on. Although this is not implemented with pure ASK, it is
  - implemented with QAM

Frequency Shift Keying (FSK)

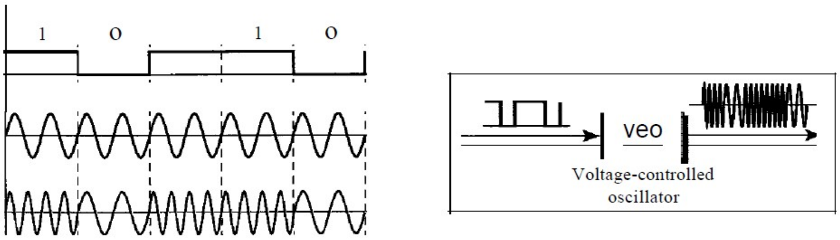
- The frequency of the carrier signal is varied to represent data. Both peak amplitude and phase remain constant for all signal elements.
- **Binary FSK (BFSK)**
  - We have two carrier frequencies, f1 andf2. We use the first carrier if the data element is 0; we use the second if the data element is 1.
  - However, note that this is an unrealistic example used only for demonstration purposes. Normally the carrier frequencies are very high, and the difference between them is very small.

Figure 5.6 *Binary frequency shift keying*



  - Bandwidth $B=(1+d)S+2\Delta f$
  - **Implementation:**
    - There are two implementations of BFSK: noncoherent and coherent.
    - In noncoherent BFSK, there may be discontinuity in the phase when one signal element ends and the next begins. In coherent BFSK, the phase continues through the boundary of two signal elements.
    - Noncoherent BFSK can be implemented by treating BFSK as two ASK modulations and using two carrier frequencies. Coherent BFSK can be implemented by using one *voltage-controlled oscillator* (VCO) that changes its frequency according to the input voltage.

Figure 5.7 *Implementation ofBFSK*



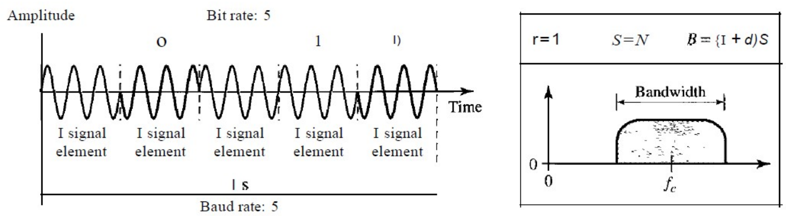
- **Multilevel FSK**
    - Multilevel modulation (MFSK) is not uncommon with the FSK method. We can use more than two frequencies. For example, we can use four different frequenciesf1, f2, f3 and f4 to send 2 bits at a time. To send 3 bits at a time, we can use eight frequencies and so on. However, we need to remember that the frequencies need to be 2Δf apart. For the proper operation of the modulator and demodulator, it can be shown that the minimum value of 2Δf needs to be S. We can show that the bandwidth with $d = 0$ is

$$B = (1 + d) \times S + (L - 1)2\Delta f \quad \rightarrow \quad B = L \times S$$

Phase Shift Keying (PSK)

- Assignment
- The phase of the carrier is varied to represent two or more different signal elements. Both peak amplitude and frequency remain constant as the phase changes.
- Today, PSK is more common than ASK or FSK. However, QAM, which combines ASK and PSK, is the dominant method of digital-to-analog modulation.
- **Binary PSK (BPSK)**
    - We have only two signal elements, one with a phase of 0°, and the other with a phase of 180°.
    - In ASK, the criterion for bit detection is the amplitude of the signal; in PSK, it is the phase. Noise can change the amplitude easier than it can change the phase.
    - PSK is less susceptible to noise than ASK. PSK is superior to FSK because we do not need two carrier signals.

Figure 5.9 *Binary phase shift keying*



- **Quadrature PSK (QPSK)**

Quuadrature Amplitude Modulation (QAM)

- Assignment
- Quuadrature Amplitude Modulation (QAM) is a combination of ASK and PSK.

Analog data analog signals

- Three process
    - Amplitude Modulation (AM)
    - Frequency Modulation (FM)
    - Phase Modulation (PM)

Amplitude Modulation (AM)

- Assignment

Frequency Modulation (FM)

- Assignment

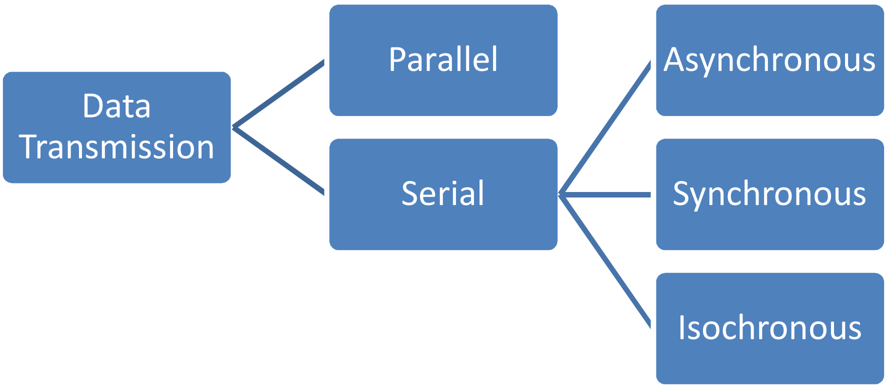
Phase Modulation (PM)

- Assignment

Data Communication & Data link control

Transmission Modes

- Assignment (Refer ppt in google classroom)



Parallel Transmission

- Assignment (Refer ppt in google classroom)

Serial Transmission

- Assignment (Refer ppt in google classroom)

Asynchronous Transmission

- Assignment (Refer ppt in google classroom)

Synchronous Transmission

- Assignment (Refer ppt in google classroom)

Isochronous Transmission

- Assignment (Refer ppt in google classroom)

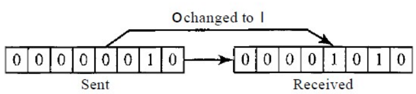
Error Detection

- Data can be corrupted during transmission. Many factors can alter one or more bits of a message.
- Some applications can tolerate a small level of error. For example, random errors in audio or video transmissions may be tolerable, but when we transfer text, we expect a very high level of accuracy.
- Some applications require that errors be detected and corrected.

Types of Errors

- **Single-Bit error**
  - only 1 bit in the data unit has changed.

    :Figure 10.1 *Single-bit error*

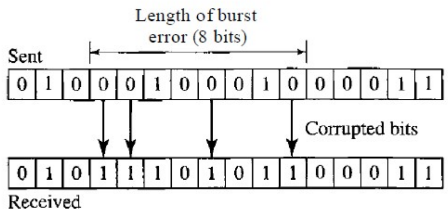
    

  - Here, sent→00000010 (ASCII STX: Start of text), Received→00001010 (ASCII LF: Line feed) (ref: Appendix A of forouzan book)
  - Single-bit errors are the least likely type of error in serial data transmission. To understand this, imagine data sent at 1 Mbps. This means that each bit lasts only

1/1,000,000 s, or 1 µs. For a single-bit error to occur, the noise must have a duration of only 1 µs, which is very rare; noise normally lasts much longer than this.

- **Burst Error**
    - 2 or more bits in the data unit have changed.
    - A burst error does not necessarily mean that the errors occur in consecutive bits.
    - The length of the burst is measured from the first corrupted bit to the last corrupted bit. Some bits in between may not have been corrupted.

Figure 10.2 *Burst error oflength 8*



    - A burst error is more likely to occur than a single-bit error.
    - The duration of noise is normally longer than the duration of 1 bit, which means that when noise affects data, it affects a set of bits. The number of bits affected depends on the data rate and duration of noise.
    - For example, if we are sending data at 1 kbps, a noise of 1/100 s can affect 10 bits; if we are sending data at 1 Mbps, the same noise can affect 10,000 bits.

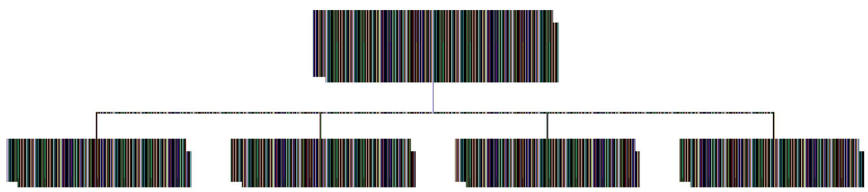
Redundancy

- To be able to detect or correct errors, we need to send some extra bits with our data.
- These redundant bits are added by the sender and removed by the receiver. Their presence allows the receiver to detect or correct corrupted bits.

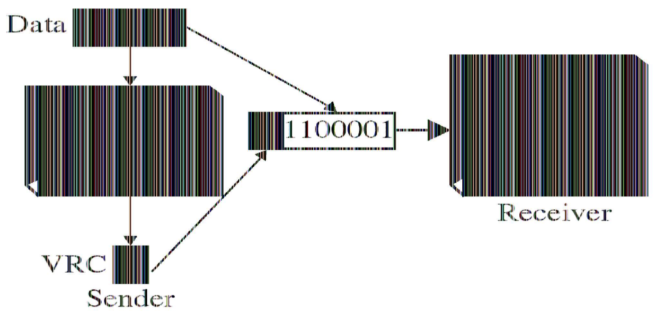
Detection versus Correction

- The correction of errors is more difficult than the detection.
- In error detection, we are looking only to see if any error has occurred. The answer is a simple yes or no. We are not even interested in the number of errors.

Redundancy Checks Method for Error Detection

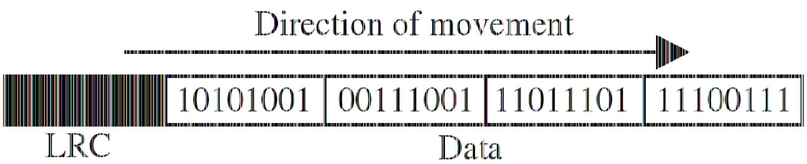


Vertical Redundancy Check (VRC)



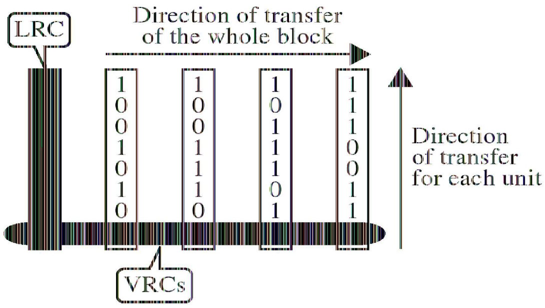
- Vertical redundancy check (VRC) is an error-checking method used on an eight-bit ASCII character.
- A parity bit is attached to each byte of data, which is then tested to determine whether the transmission is correct. VRC is considered an unreliable error-detection method because it only works if an even number of bits is distorted.
- A vertical redundancy check is also called a transverse redundancy check when used in combination with other error-controlling codes such as a longitudinal redundancy check.
- It can detect single bit error
- It can detect burst errors only if the total number of errors is odd.

Longitudinal Redundancy Check (LRC)

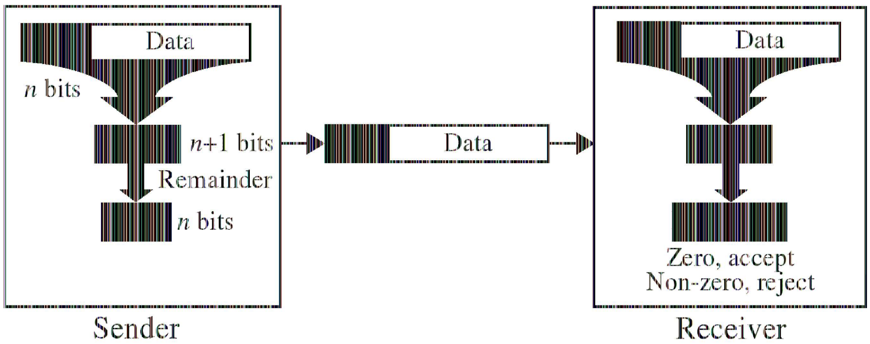


- LRC generally applies to a single parity bit per bit stream. Although simple longitudinal parities only detects errors, a combination with additional error control coding, such as a transverse redundancy check, are capable of correcting errors.
- LRC fields consist of one byte containing an eight bit binary value.
- LRC values are calculated by transmitting devices, which append LRC to messages.
- The device at the receiving end recalculates the LRC on receipt of the message and compares the calculated value to the actual value received in the LRC field.
- If the values are equal, the transmission was successful; if the values are not equal, this indicates an error.
- LCR increases the likelihood of detecting burst errors.
- If two bits in one data units are damaged and two bits in exactly the same positions in another data unit are also damaged, the LRC checker will not detect an error.
- LRC Generation-Steps
    - Add all bytes in messages excluding the starting colon and the ending the carriage return line feed.
    - Add this to the eight-bit field and discard the carries
    - Subtract the final field value from FF hex, producing one’s complement
    - Add one, producing two’s complement

VRC and LRC

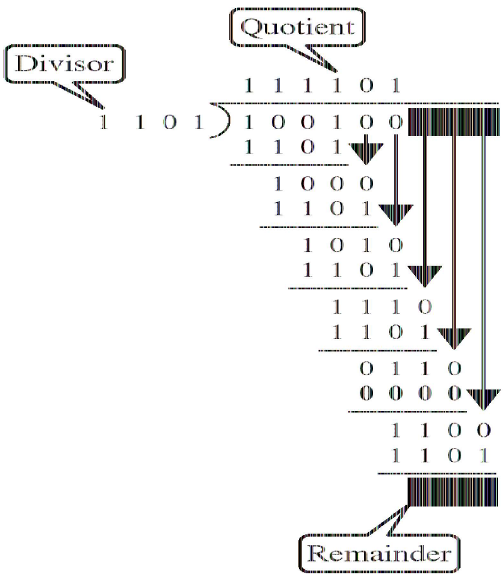


Cyclic Redundancy Check (CRC)



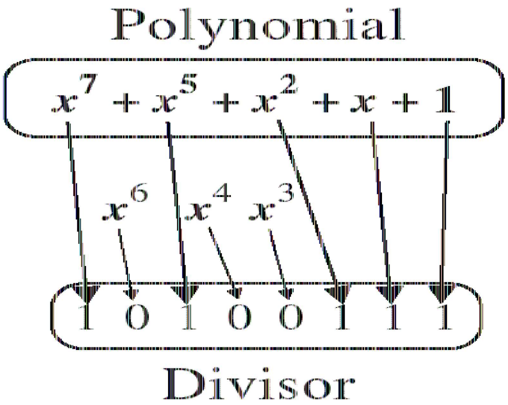
- Given a k-bit frame or message, the transmitter generates an n-bit sequence, known as a frame check sequence (FCS), so that the resulting frame, consisting of (k+n) bits, is exactly divisible by some predetermined number.
- The receiver then divides the incoming frame by the same number and, if there is no remainder, assumes that there was no error.

Binary Division in CRC



Polynomial and Divisor in CRC

$$x^7 + x^5 + x^2 + x + 1$$



Standard Polynomials in CRC

CRC-12

$$x^{12} + x^{11} + x^3 + x + 1$$

CRC-16

$$x^{16} + x^{15} + x^2 + 1$$

CRC-ITU

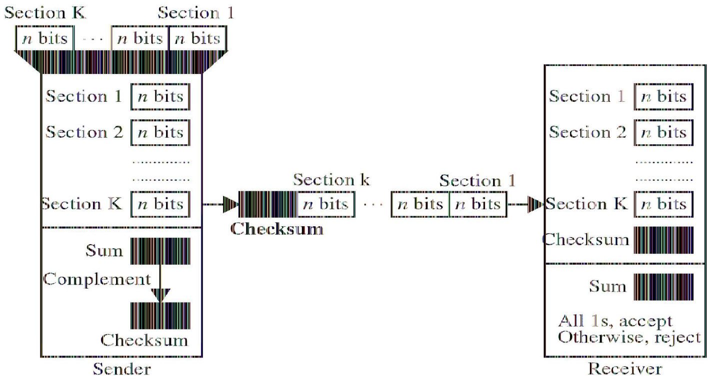
$$x^{16} + x^{12} + x^5 + 1$$

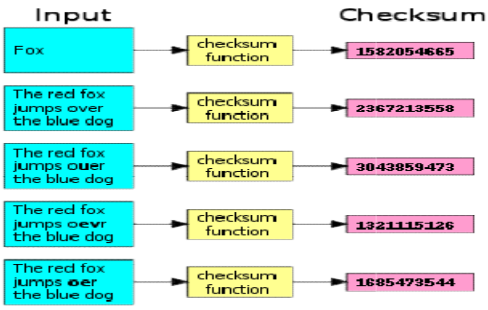
CRC-32

$$x^{32} + x^{26} + x^{23} + x^{22} + x^{16} + x^{12} + x^{11} + x^{10} + x^8 + x^7 + x^5 + x^4 + x^2 + x + 1$$

Checksum

- **A checksum or hash sum is a small-size datum computed from an arbitrary block of digital data for the purpose of detecting errors which may have been introduced during its transmission or storage.**
- **The actual procedure which yields the checksum, given a data input is called a checksum function or checksum algorithm.**
- **Depending on its design goals, a good checksum algorithm will usually output a significantly different value, even for small changes made to the input.**





At the Sender

- The unit is divided into k sections, each of n bits.
- All sections are added together using one's complement to get the sum.
- The sum is complemented and becomes the checksum.
- The checksum is sent with the data

At the Receiver

- The unit is divided into k sections, each of n bits.
- All sections are added together using one's complement to get the sum.
- The sum is complemented.
- If the result is zero, the data are accepted: otherwise, they are rejected.

Performance of Checksum

- The checksum detects all errors involving an odd number of bits.
- It detects most errors involving an even number of bits.
- If one or more bits of a segment are damaged and the corresponding bit or bits of opposite value in a second segment are also damaged, the sums of those columns will not change and the receiver will not detect a problem.

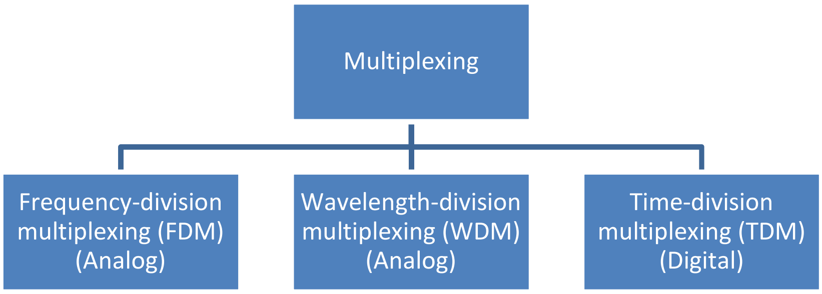
Line configuration

Flow Control,

Error Control

Multiplexing

- Assignment (Refer ppt in google classroom)



FDM synchronous TDM

Statistical TDM

Switching & Routing

- Assignment (Refer ppt in google classroom)

Circuit Switching networks

- Assignment (Refer ppt in google classroom)

Packet Switching principles

- Assignment (Refer ppt in google classroom)

X.25

- Assignment

Routing in Packet switching

- Assignment (Refer ppt in google classroom)

Congestion

- Assignment

Effects of congestion, congestion control

- Assignment

Traffic Management

- Assignment

Congestion Control in Packet Switching Network

- Assignment

LAN Technology

- Assignment

Topology and Transmission Media

- Assignment

LAN protocol architecture

- Assignment

Medium Access control

- Assignment

Bridges, Hub, Switch

- Assignment (Refer material in google classroom)

Ethernet (CSMA/CD), Fibre Channel

- Assignment

Wireless LAN Technology

- Assignment

TCP/IP

- Assignment (Refer ppt in google classroom)

TCP/IP Protocol Suite

Basic Protocol functions

Principles of Internetworking

Internet Protocol operations

Internet Protocol