Concepts for the Data Communications and Computer Interconnection

Aim: overview of existing methods and techniques

Terms used:
-*Data* – entities conveying meaning (of information)
-*Signals* – data carrier; electric or electromagnetic representations of data
-*Transmission* – data communications process, using the signal's propagation and processing

Main attributes of data, signals and transmission:
- digital
- analog

Towards *all digital?* Not yet!

Why? Important legacy (old telephone system); everything around (from environment) comes as analog

Today the digital technology offers:

-low cost, due to VLSI technology

-low attenuation, even in the past the analog technology led

-low noise influence

-better capacity utilization

-better data integrity

-security and privacy

-integration of digital and analog data.

Analog Data

Continuous values within some interval; e.g. sound, video.

Digital Data

Discrete values, e.g. text, integers.

Continuous signal

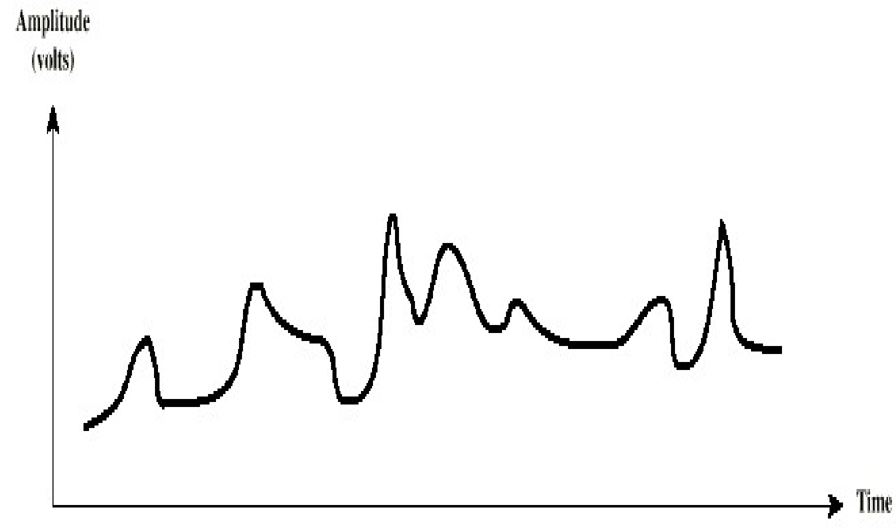
Various in a smooth way over time, may have any values.

Discrete signal

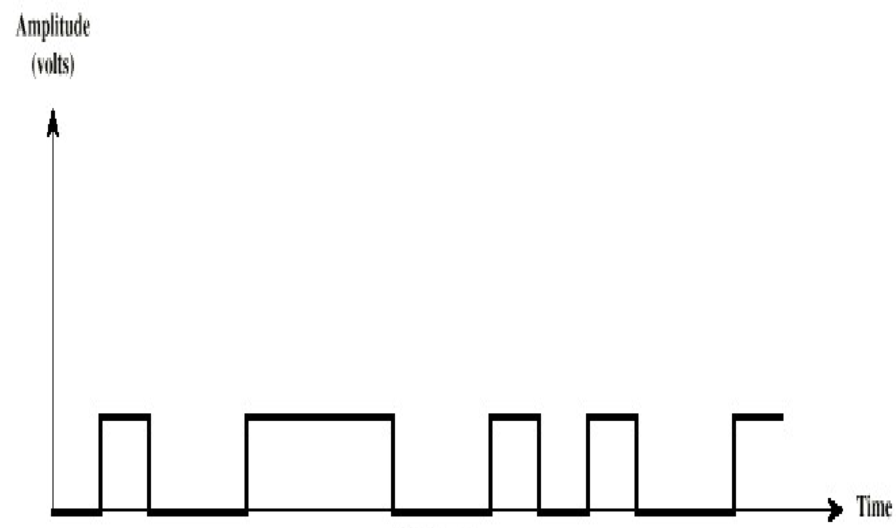
Maintains a constant level, then changes to another constant level. May have one of some (e.g. two) level values.

Mark denotes signal for ‘1’

Space denotes signal for ‘0’ data



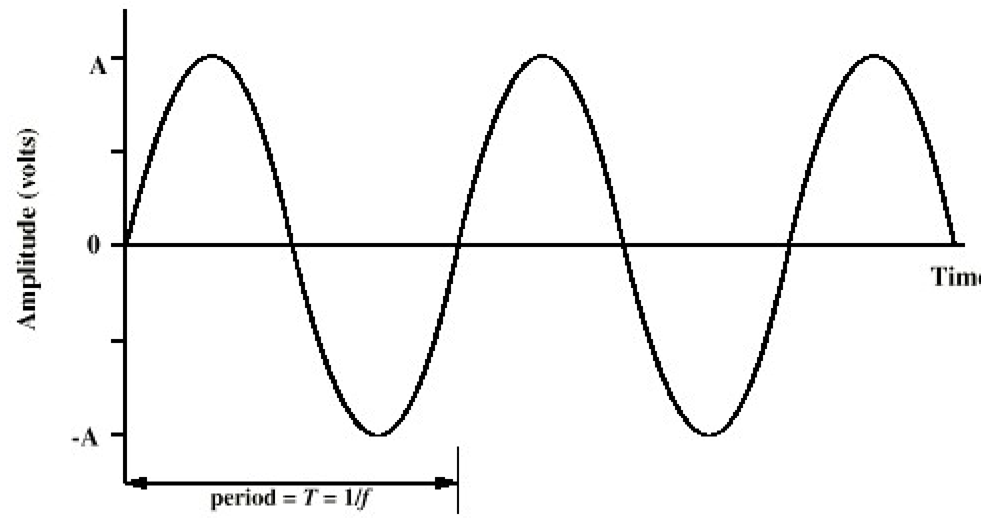
(a) Continuous



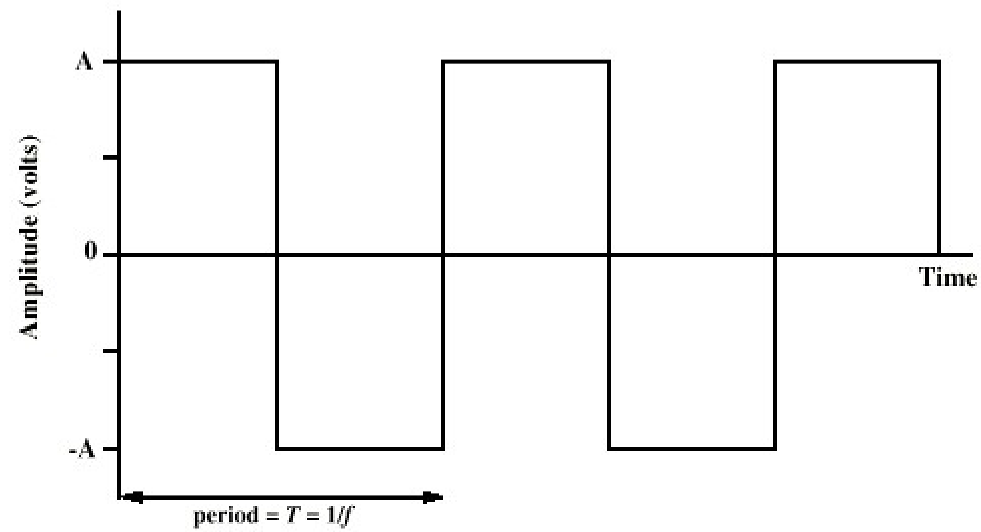
(b) Discrete

Periodic signal

Presents a pattern repeated over time.



(a) Sine wave



(b) Square wave

Parameters of the Sinus Wave (analytical, as function of time):

$$A \cdot \sin(2\pi f t + \phi)$$

Peak Amplitude (A): the maximum strength of signal, expressed in volts

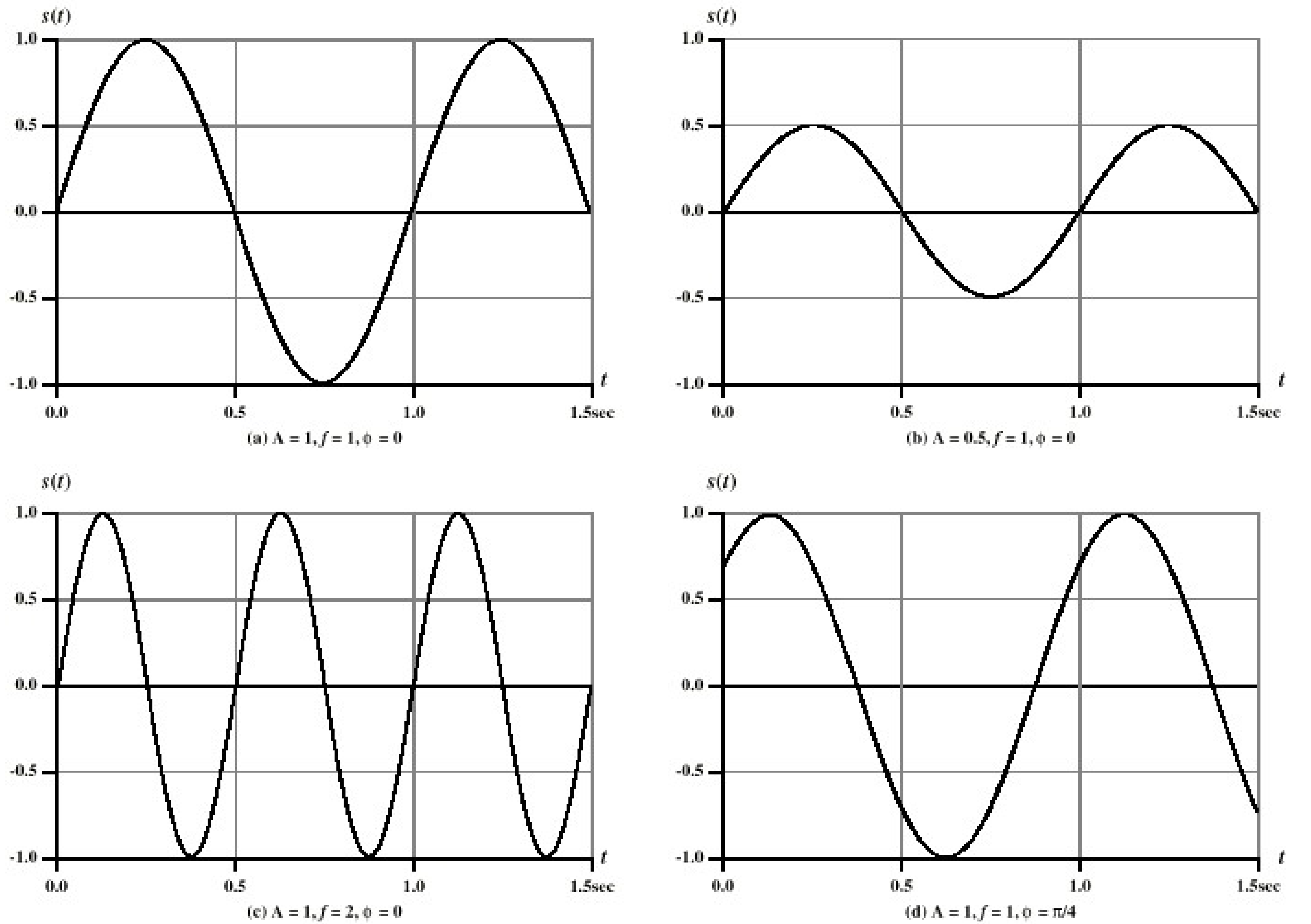
Frequency (f): the rate of change of signal, expressed in Hertz (Hz) or cycles per second

Period of the signal = time for one repetition (T)

$T = 1/f$

Phase (ϕ): means the relative position in time

Wavelength (λ): Distance between two points of corresponding phase in two consecutive cycles. Relations: $\lambda = vT$; $\lambda f = v$, where v: signal speed expressed in m/s.


s(t)
1.0
0.5
0.0
-0.5
-1.0
t
0.0
0.5
1.0
1.5sec
(a) A = 1, f = 1, ϕ = 0
(b) A = 0.5, f = 1, ϕ = 0
(c) A = 1, f = 2, ϕ = 0
(d) A = 1, f = 1, ϕ = π/4

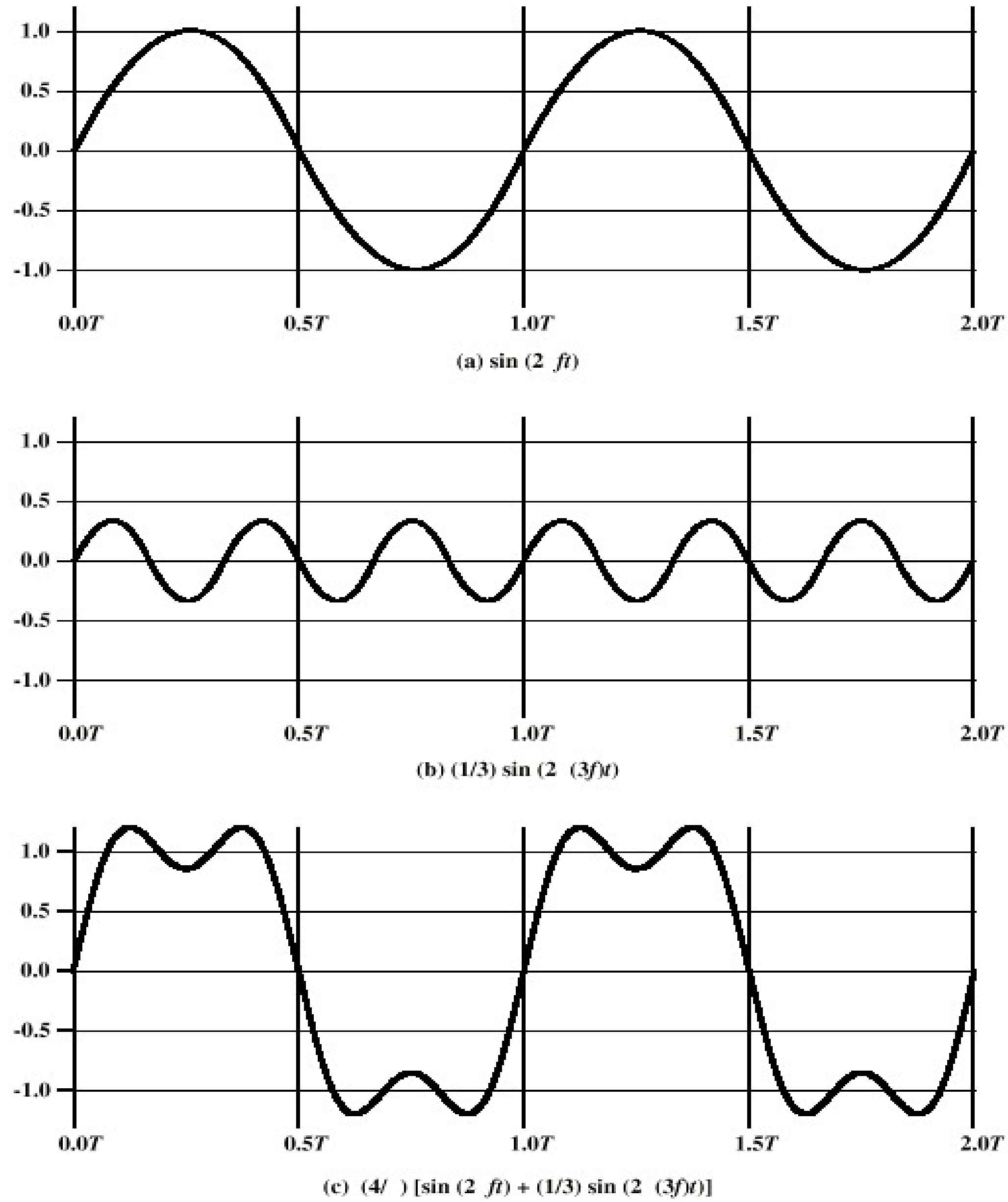
Frequency Domain

In practice, an electromagnetic signal is made up of many frequencies (has sinus components – Fourier analysis); one is the fundamental frequency, others are multiples.

Spectrum – range of frequencies a signal contains.

Bandwidth – signal's width of the spectrum.

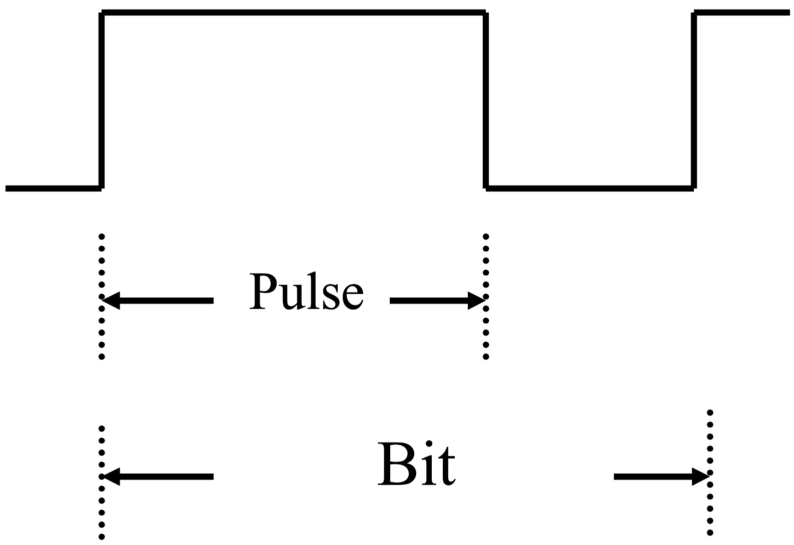
dc Component (continuous component) – component with zero frequency.

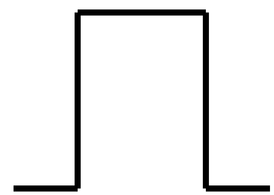


Any signal has a limited bandwidth => limited data rate!!!

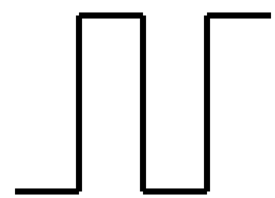
Data Coding terminology

Signal element: Pulse
Modulation rate: 1/Duration of the smallest element or rate at which the signal level changes = Baud rate
Data rate: Number of bits per second (bps)
Data rate = Fn(Bandwidth, signal/noise ratio, encoding technique)





NRZI: 1 bit = 1 signal element



Manchester: 1 bit = 2 signal elements,
Twice modulation rate required than NRZI

How to compare encoding techniques?

Various criteria:

-required bandwidth (lack of higher frequencies => low bandwidth)

-lack of the dc component: allows ac coupling, providing isolation

-how power is spread within the frequency spectrum (main power in the middle of the bandwidth)

-allows error detection (mechanism built in)

-avoid signals interference and allows high noise immunity

-synchronization mechanism built in (no external clock)

-cost and complexity

    -higher signal rate (data rate) => higher costs

    -need for a signal rate greater than data rate

Data Encoding

Digital Data, Digital Signals
Methods:

NRZ (Non Return to Zero-Level) – uses two voltage levels (H,L); may have any polarities
- difficult to find the bit margins
- no transitions between similar bits => dc component, damaging the passive connecting devices

NRZI (Non Return to Zero, Invert on Ones), also known as NRZ-M – codes data using a transition at the beginning of the bit period (‘1’: transition, ‘0’: no transition).

Differential coding – compares polarities of successive signals, not their absolute values => better noise immunity

Multilevel Binary codes

Use more than two voltage levels

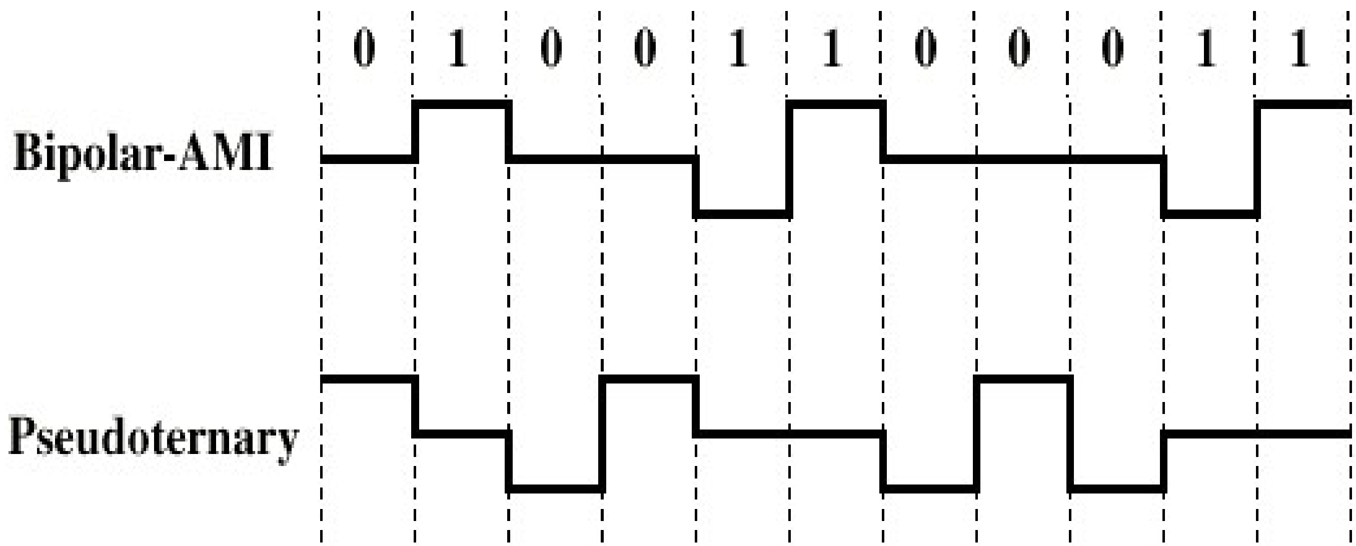
Bipolar-AMI (Alternate Mark Inversion) – ‘0’: no line signal, ‘1’: alternating positive and negative pulses => better synchronization (but avoid long ‘0’ string), lower bandwidth, improved error detection

Pseudo-ternary – reverse coding, ‘1’: no line signal, ‘0’: alternating positive and negative pulses => similar problems as for bipolar-AMI

Drawback:

Receiver must distinguish between three levels: (A, -A, 0)

Requires approx. 3dB more signal power for same probability of bit error

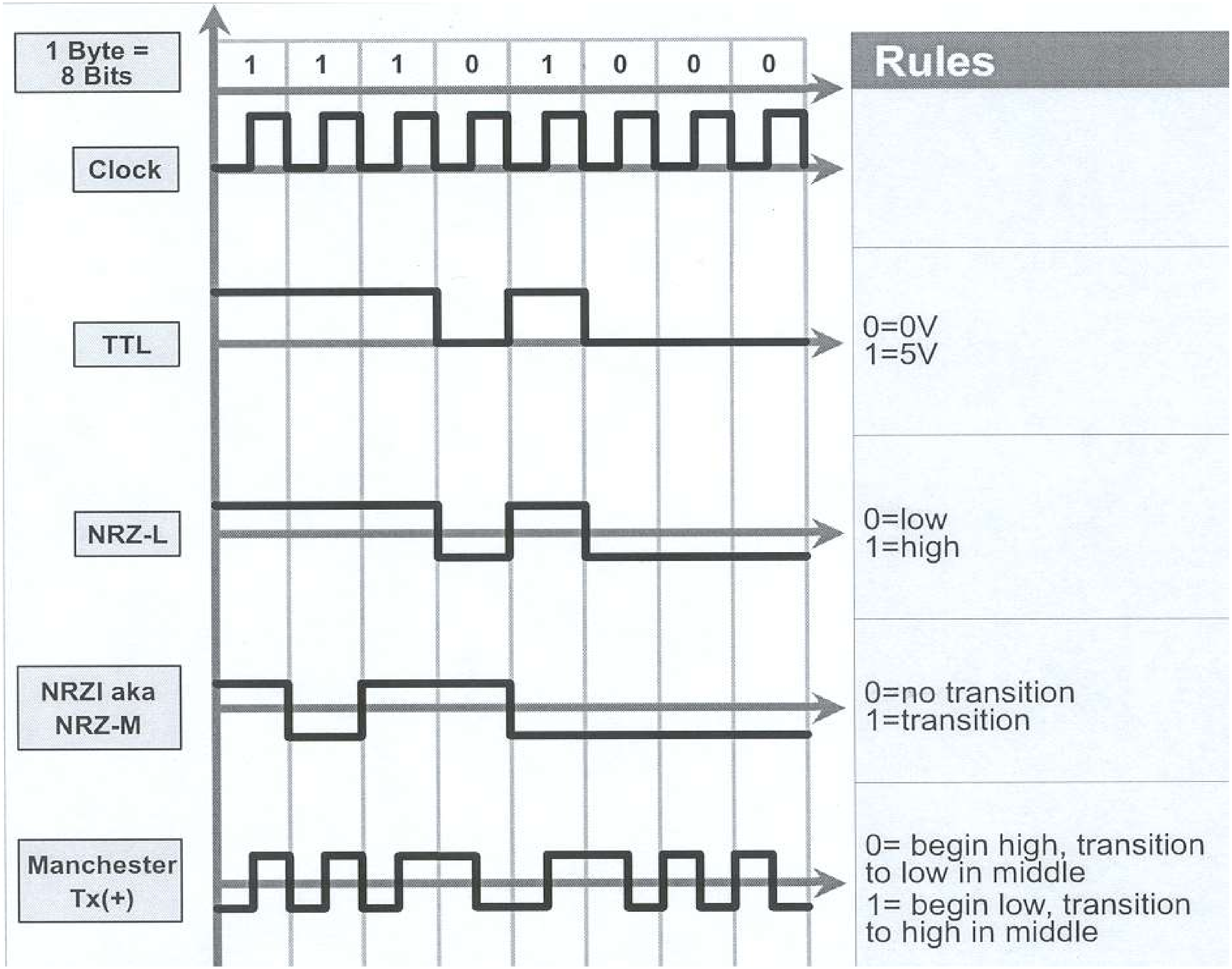


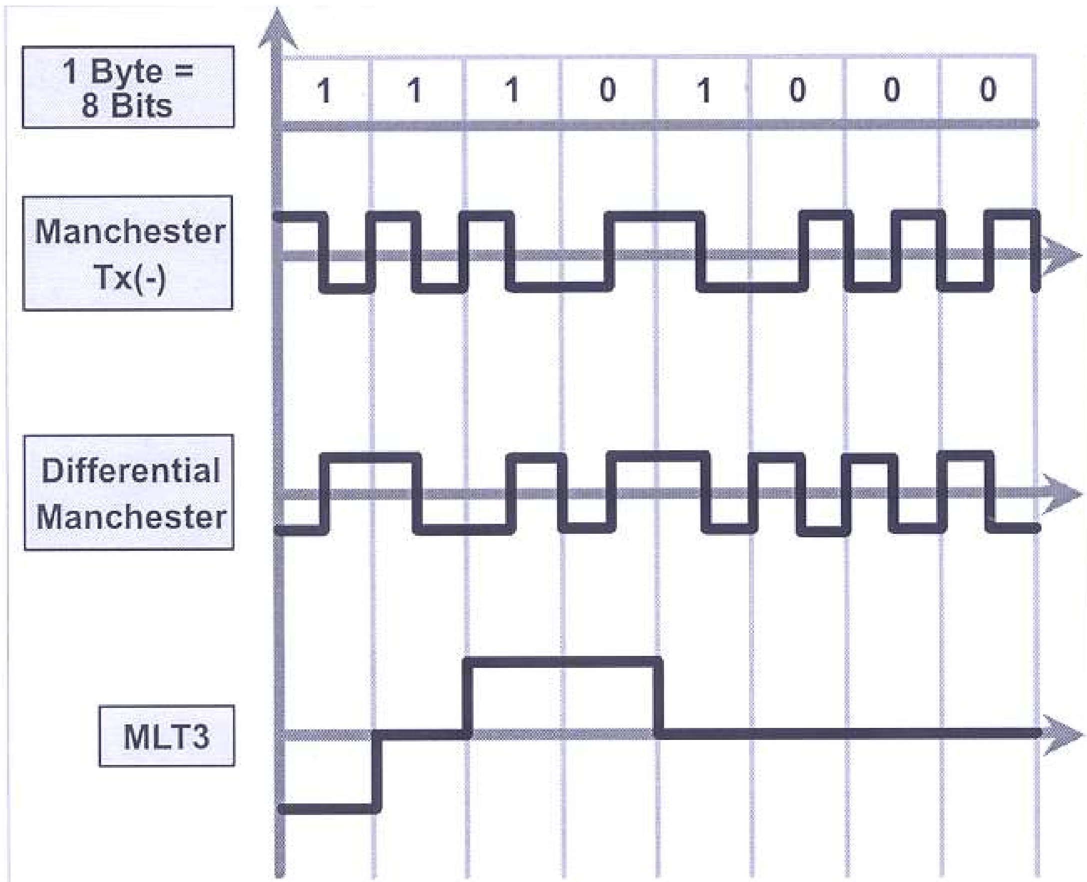
Data Encoding (continued)

Biphasic coding: one or two transitions on a bit period => higher bandwidth, but provides synchronization, better error detection, less noise influence, no dc component

Manchester – always a transition at the middle of the bit period (used as clock signal): data coding by the transition sense (‘0’: Low to High, ‘1’: High to Low for Tx-, and reverse for Tx+)

Differential Manchester – middle transition as clock signal, data coding by a transition at the beginning of bit period (‘0’: transition, ‘1’: no transition). Most used for twisted pair based networks.


1 Byte = 8 Bits
1 1 1 0 1 0 0 0
Clock
TTL
NRZ-L
NRZI aka NRZ-M
Manchester Tx(+)
Rules
0=0V
1=5V
0=low
1=high
0=no transition
1=transition
0= begin high, transition to low in middle
1= begin low, transition to high in middle



Rules

0= begin low, transition to high in middle
1= begin high, transition to low in middle

Always a transition in middle of interval
0 = transition in beginning of interval
1 = no transition in beginning of interval

3 level amplitude,
Level = -1V, 0V, +1V, 0V, -1V, etc...
0 = no transition
1 = transition to next level in cycle

TTL = Transistor-Transistor Logic
NRZ-L = Non-Return to Zero-Level
NRZI = Non-Return to Zero-Inverted
NRZ-M = Non-Return to Zero-Mark
Tx = Transmit
MLT3 = Multi-Level Threshold-3

For WANs, for sake of bandwidth costs: **scrambling techniques** (long constant data streams replaced by filling sequences):

Bipolar With 8 Zeros Substitution B8ZS

Based on bipolar-AMI, but introducing AMI code violation

IF:

Octet of all zeros and last voltage pulse preceding was positive, encode as 000+-0-+

Octet of all zeros and last voltage pulse preceding was negative, encode as 000-+0+-

Causes two violations of AMI code; Unlikely to occur as a result of noise

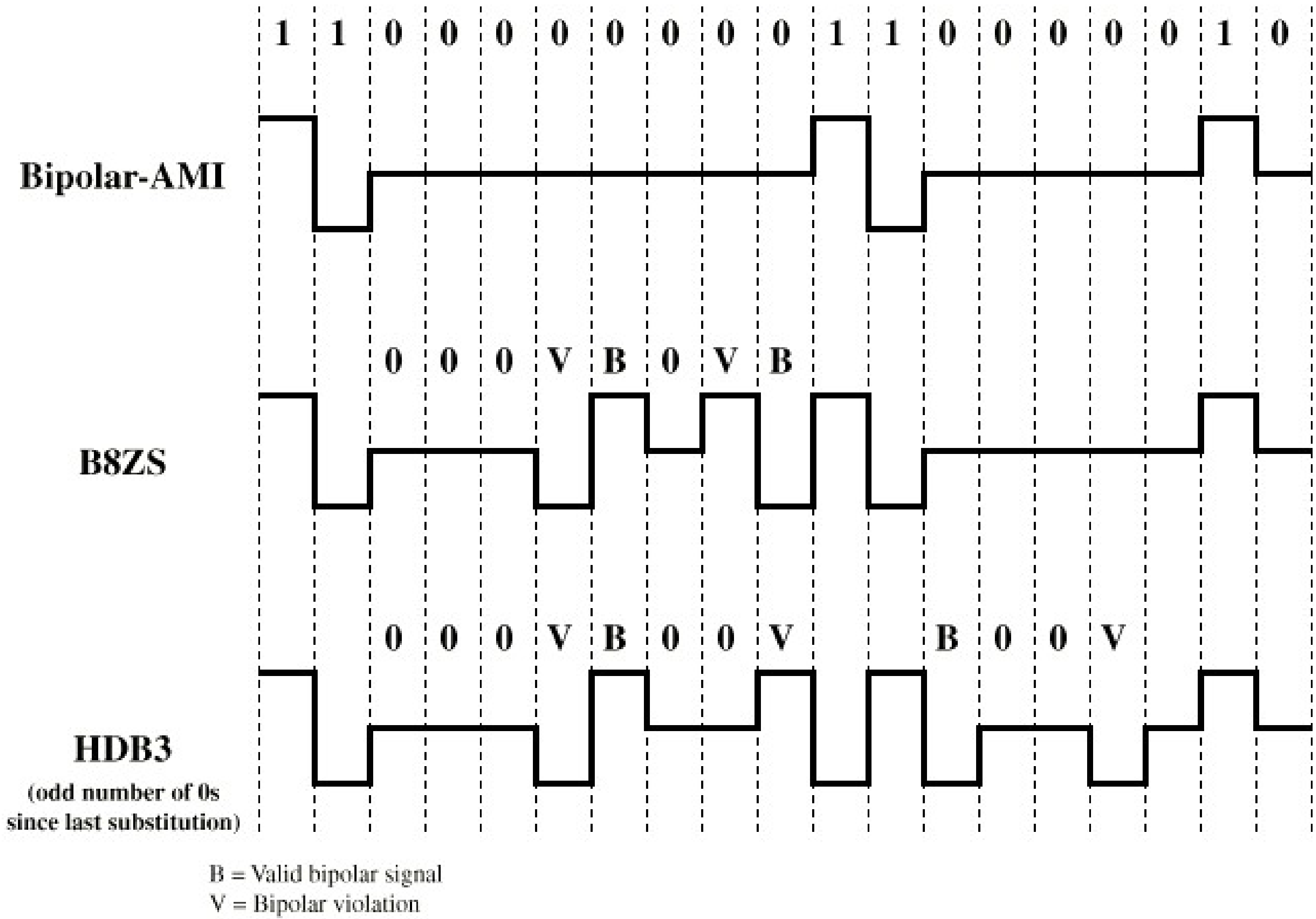
Receiver detects and interprets as octet of all zeros.

High Density Bipolar 3 Zeros HDB3

Based on bipolar-AMI, but introducing code violation (not valid AMI bipolar signal)

String of four zeros replaced with one or two pulses (the AMI code violation sequence).

Also alternation of polarities for the violation codes.


1 1 0 0 0 0 0 0 0 0 1 1 0 0 0 0 0 1 0
Bipolar-AMI
0 0 0 V B 0 V B
B8ZS
0 0 0 V B 0 0 V B 0 0 V
HDB3
(odd number of 0s since last substitution)
B = Valid bipolar signal
V = Bipolar violation

Digital Data, Analog Signals

Use of a constant frequency signal: **data carrier**, modulated conform with the data

Amplitude Shift Keying (ASK) – presence or not of the carrier, at constant amplitude; non efficient for data transmissions; variant for fiber optic transmissions: presence or absence of the light

Frequency Shift Keying (FSK) – two (symmetric) frequencies, near the carrier basic frequency

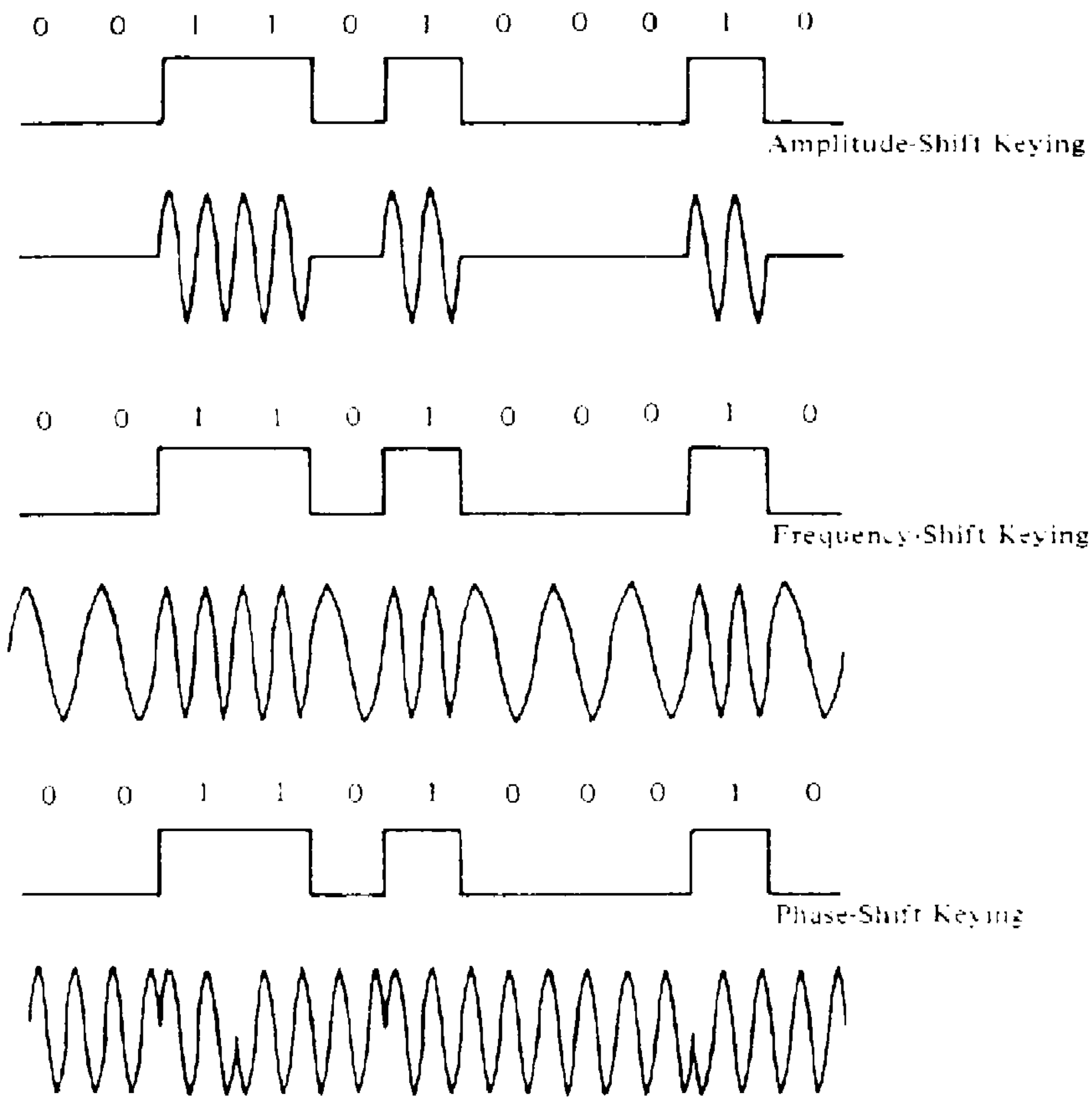
Phase Shift Keying (PSK) – short burst signals

coherent PSK: constant signals having a phase difference of 180º

differential PSK: '0' burst signal with the same phase as the previous (0º shift), '1' burst signal with opposite phase as previous (shift with 0º+ π)

best error resistant, determining the phase shift magnitude, not its absolute value.

Quadrature-PSK coding – codes 2 bits by a burst signal, having more than two phase-shifts per signal: phase shifts of multiples of 90º. Possible extensions…


0 0 1 1 0 1 0 0 0 1 0
Amplitude-Shift Keying
0 0 1 1 0 1 0 0 0 1 0
Frequency-Shift Keying
0 0 1 1 0 1 0 0 0 1 0
Phase-Shift Keying

Analog Data, Digital Signals

Theoretical background: Nyquist sampling theorem: sample at twice the highest signal frequency (for a voice carrying signal with bandwidth of 4kHz, sample at 8kHz, or every 125μsec, having 8000samples/sec)

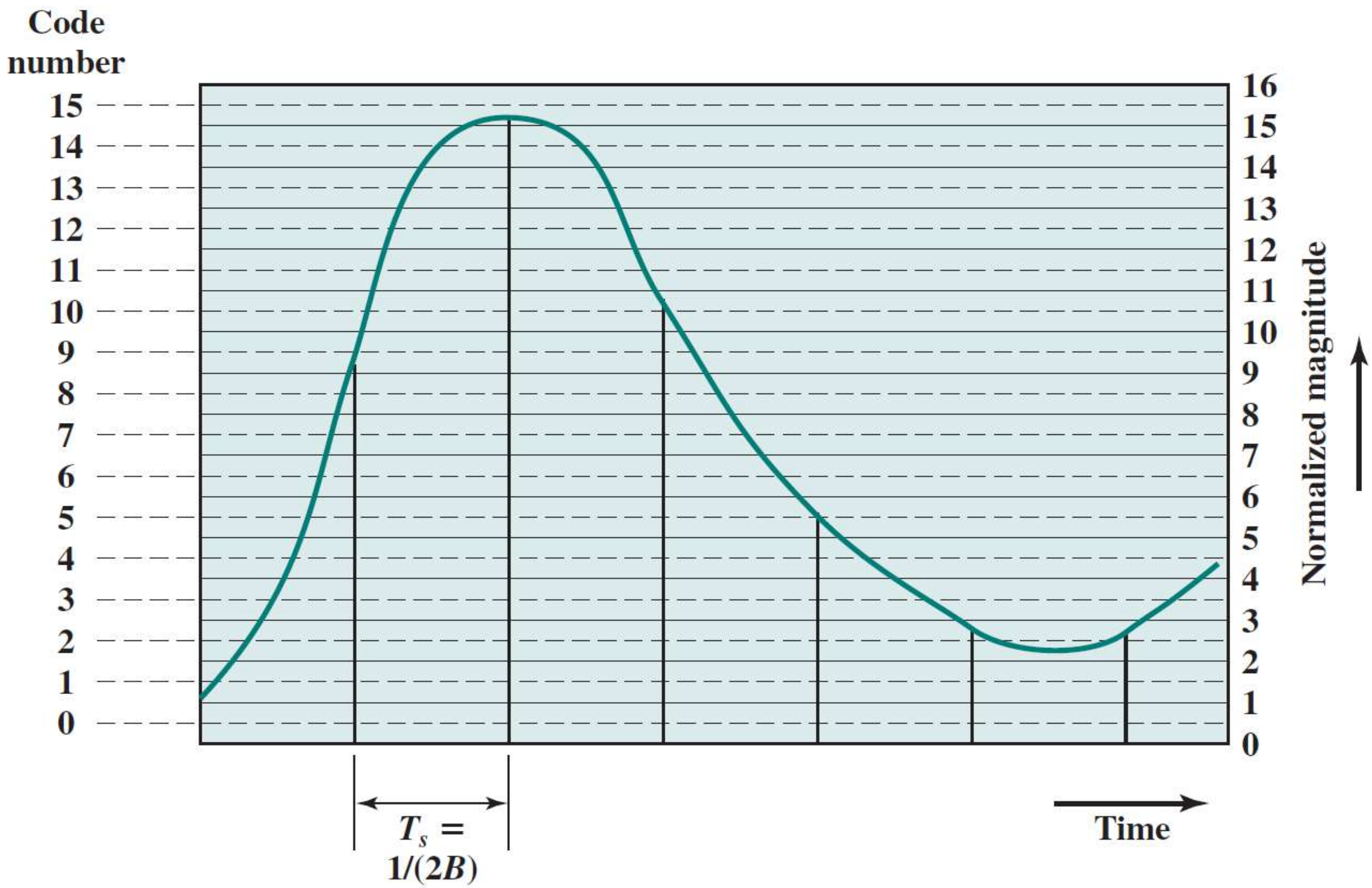
Pulse Code Modulation (PCM), with the following 3 steps:

-signal ***sampling***, using the proper sampling frequency (higher than twice the highest signal frequency); samples represented as PAM (Pulse Amplitude Modulation) pulses

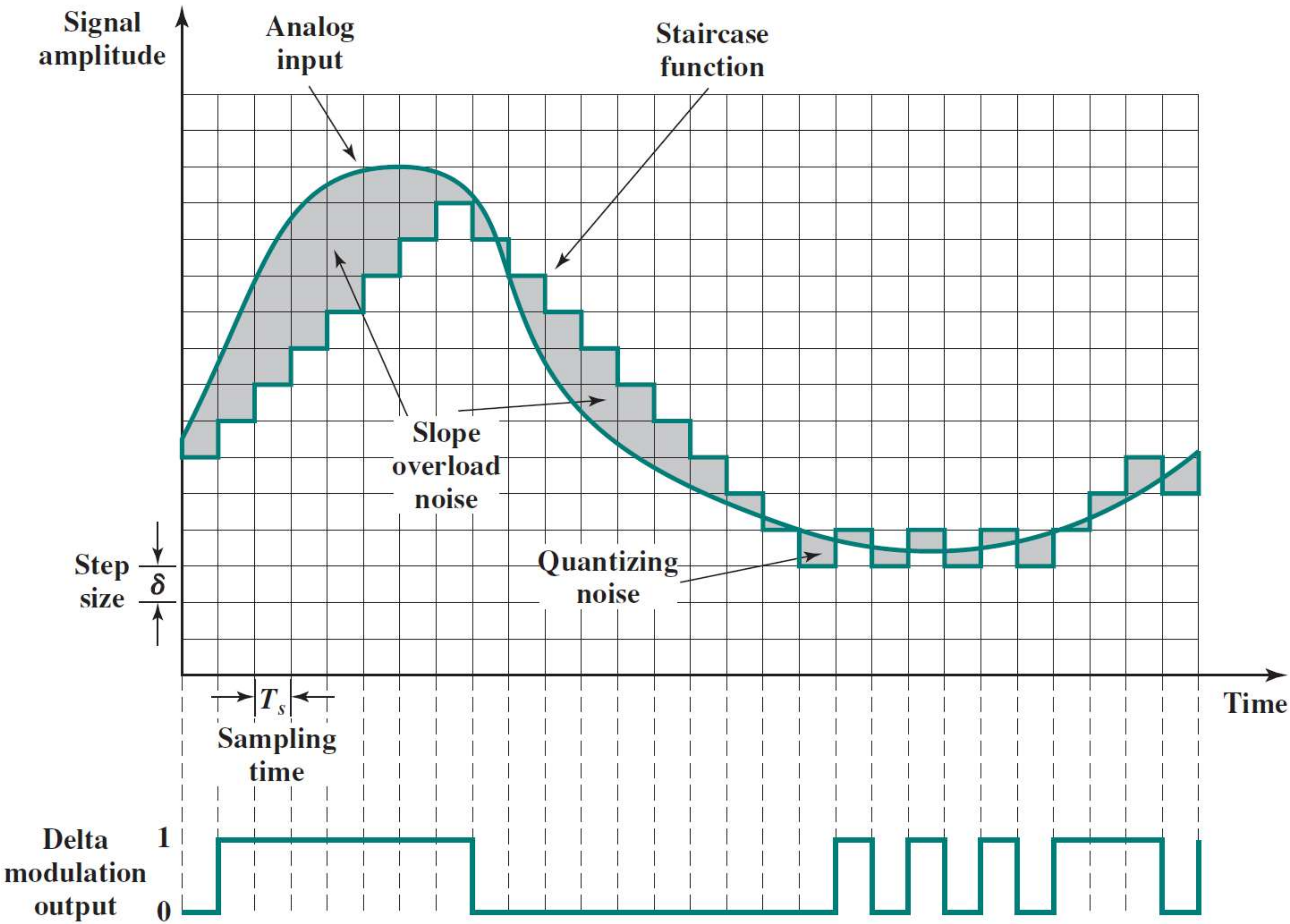
-***quantification*** of the samples, using the available number of digits, obtaining the PCM pulses and their digital values; more digits, more accuracy, greater cost

-digital values representation as pulse trains - ***coding***

Delta Modulation – approximates the analogue signal by a staircase function moving up/down by one quantization level at each sampling interval; output function has a binary behavior (moves up or down at each sample interval); method less used in computer networks

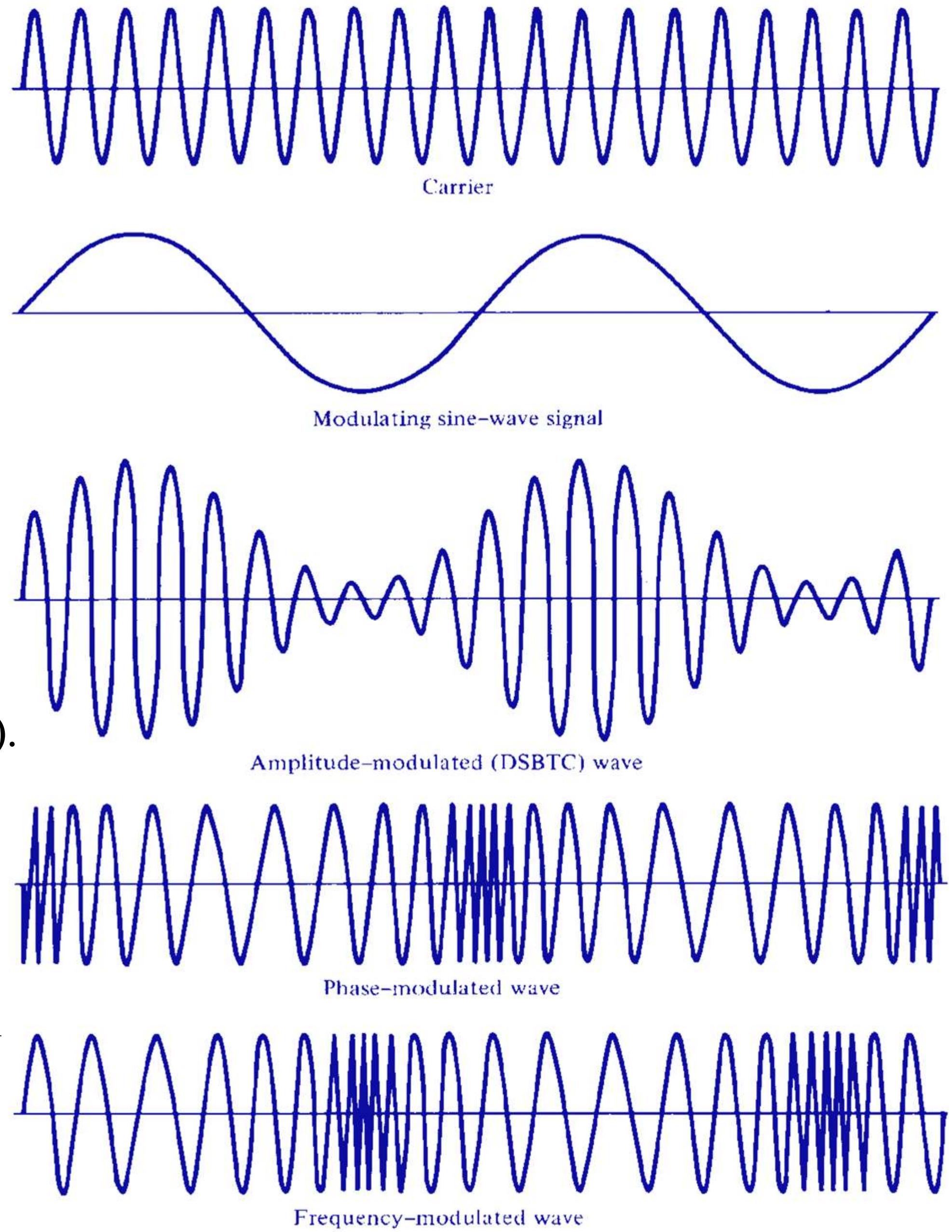


PAM value	1.1	9.2	15.2	10.8	5.6	2.8	2.7
Quantized code number	1	9	15	10	5	2	2
PCM code	0001	1001	1111	1010	0101	0010	0010


Signal amplitude
Analog input
Staircase function
Slope overload noise
Step size
δ
Quantizing noise
T_s
Sampling time
Time
Delta modulation output
1
0

Analog Data, Analog Signals

Used when only analog facilities available.
Why analog data if the voice signals are transmitted in the baseband ?
-higher frequency may be needed for unguided transmission (impossible to transmit baseband signals), or optical
-modulation permits FDM.



Amplitude Modulation

Amplitude of the carrier is varied accord. with some characteristic of the modulating signal (ex: double-sideband transm. carrier).

Phase Modulation

Data carrier's phase is varied linearly according to the data.

Frequency Modulation

Data carrier wave's frequency departs from the center frequency (carrier's) by an amount depending on the value of the modulating signal.

Spread Spectrum

Analog or digital data sent using analog signal (radio transmissions)

Spread data over wide bandwidth

Makes jamming and interception harder

Two schemes:

Frequency hoping

- Signal broadcast over seemingly random series of frequencies
- Hop from one frequency to other at split-second intervals

Direct Sequence

- Each bit is represented by multiple bits in transmitted signal (chipping code)
- Chipping code is obtaining combining original data with pseudorandom bit stream
- Chipping code spreads the signal across a wider frequency band

Transmission impairments

For any communication system, the received signal will differ from the transmitted signal – not an ideal transmission!
Due to various transmission impairments, introducing signal degradation (analog transmissions), bit errors (digital); most encountered transmission impairments are:
Attenuation and attenuation distortion
Delay distortion
Noise

Attenuation

The reduction of signal’s strength (power) with distance.

For guided media attenuation is logarithmic and expressed in dB/m.
For unguided media transmissions, it depends on distance and makeup of atmosphere.

Attenuation = $10 \cdot \log_{10} P_{in}/P_{out}$ [dBel]
Attenuation = $20 \cdot \log_{10} V_{in}/V_{out}$ [dBel]

Received signal strength:

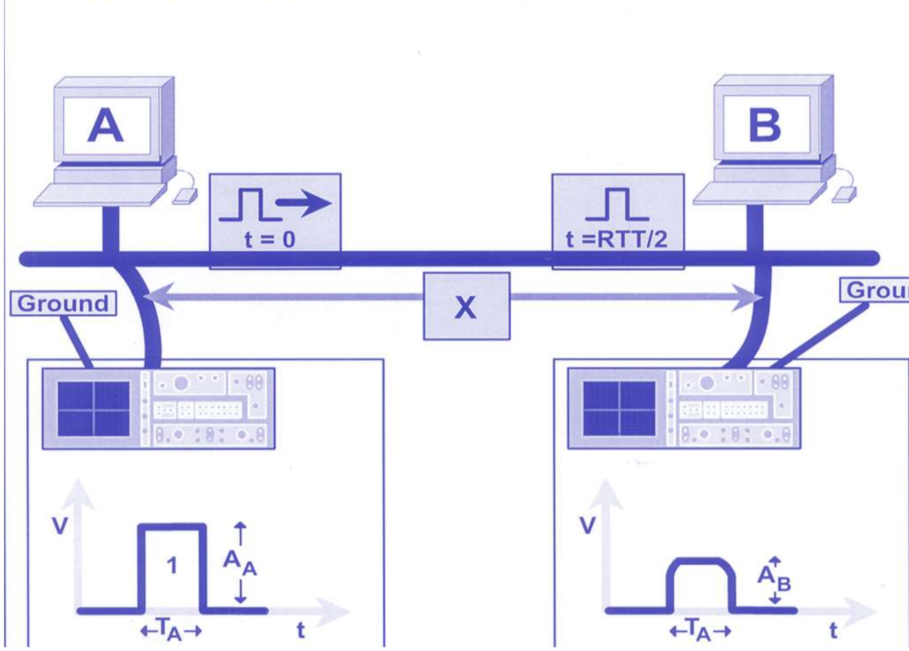
- must be enough to be detected
- must be sufficiently higher than noise, to be received without error.

Use of amplifiers and repeaters for maintaining the signal strength.

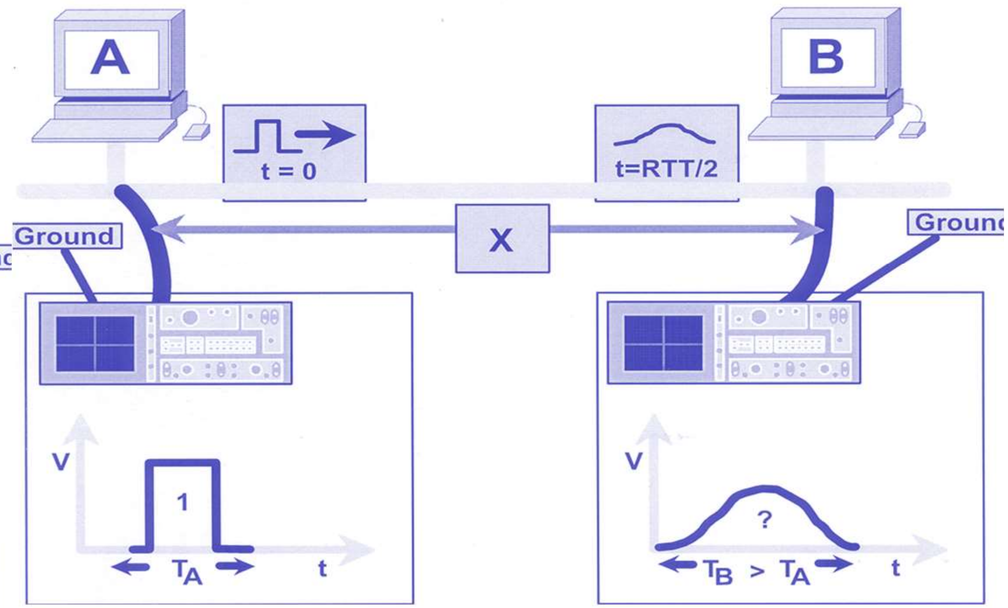
Attenuation depends increasingly of signal frequency =>problems for HF transmissions, but mainly for analog transmissions, resulting signal distortions => techniques for attenuation equalization across the frequency spectrum.

Digital signal concentrates power near the fundamental frequency.

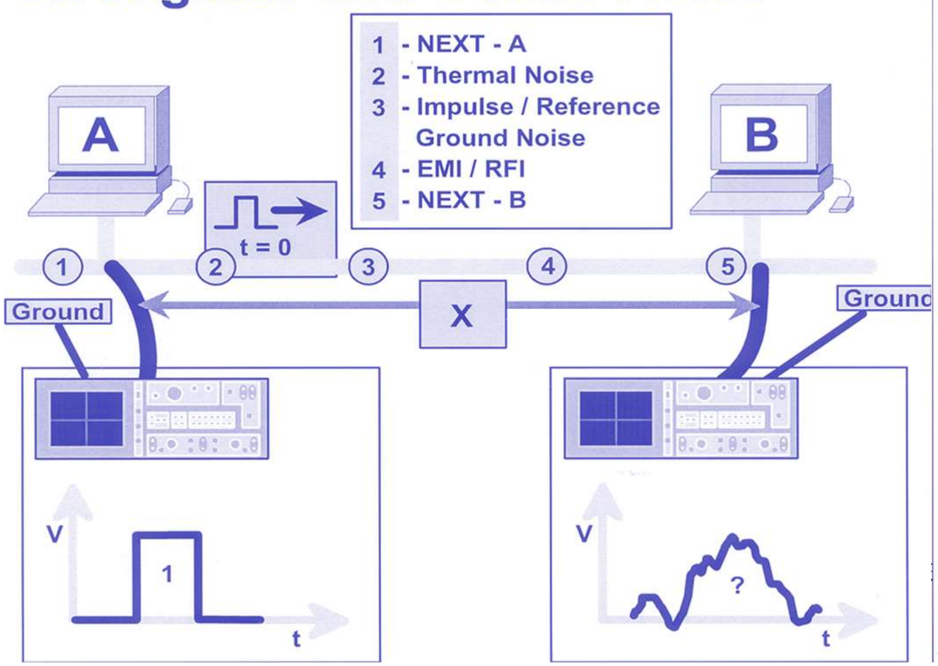
Attenuation



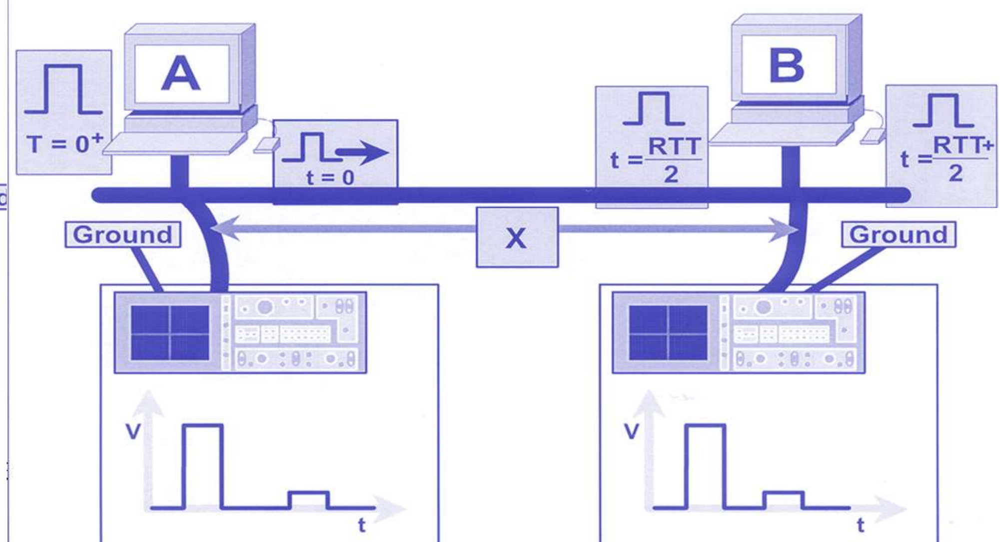
Delay Distortion (Dispersion)



Recognize and Define Noise



Reflection



Delay Distortion

Only in guided media, where the signal's propagation speed depends on frequency => signal distortions, centred frequency components have greater velocity than those from band edges. Use of equalizers.

Noise

Plus to above mentioned distortions: additional signals inserted between transmitter and receiver (generally called **noise**)!

Thermal noise (depends on temperature, not on frequency, intrinsic to structure):

- Due to thermal agitation of electrons
- Uniformly distributed across the spectrum (called white noise)
- Can not be eliminated => an upper bound for communications performances.

Intermodulation noise

Noise signals that are the sum and difference of original frequencies sharing a medium, or multiples of them – due to the nonliniarities of the transmission system.

Crosstalk

A signal from one line is picked up by another (is a coupling between signal paths). Experienced by anyone with the telephone.

Impulse noise

Non predictable, caused by external electromagnetic disturbances, faults and flaws in the system; critical for digital transmissions

Irregular pulses or spikes with short duration, random amplitude (thus may be high), and spectral content.

Communications Channels

Definition: the part that connects a data source to a data sink; based on the transmission media.

Classification criteria:

-**type of the link** (connection):

point-to-point,

point-multipoint (master-slave configuration),

broadcast (common shared medium)

-**information transfer sense**:

simplex: one way

half-duplex: at a moment, data only in a sense, control may be in both

full-duplex: data and control on both ways

- **maximum channel transmission speed** (channel capacity), in junction with the maximum allowed bandwidth

-**type of transmission**

-**baseband**: entire bandwidth of communications media dedicated to one channel; often used for digital transmissions; cheaper, adequate for most LANs

-**broadband**: whole bandwidth divided into multiple independent channels; often used for analog transmissions; multiple transmissions of data, voice, video

Basic theorems used in obtaining the maximum channel speed

Nyquist theorem:

For an ideal channel (without loss, no noise), maximum channel speed (maximum data rate):

$$\mathbf{v=2 \bullet H \bullet \log_2 N}$$

H: frequency bandwidth

N: number of levels used to encode data

(if N = 2, for the bi-level encoding, comes the well known: $\mathbf{v=2 \bullet H}$)

Shannon's theorem:

For a 'more realistic' channel, affected by noise:

$$\mathbf{v=H \bullet \log_2(1+S/N)}$$

S: power of the transmitted signal

N: power of the noise signal

S/N: signal per noise ratio, expressed usually as $\mathbf{10 \bullet \log_{10} S/N}$ and measured in dB (also usually understood as attenuation).

Example: Phone wire bandwidth = 3100Hz (spread between 300Hz and 3400Hz). For an attenuation of 30dB (usual one for that type of wire), what will be the channel capacity?

$10 \cdot \log_{10} S/N = 30$

$\log_{10} S/N = 3$

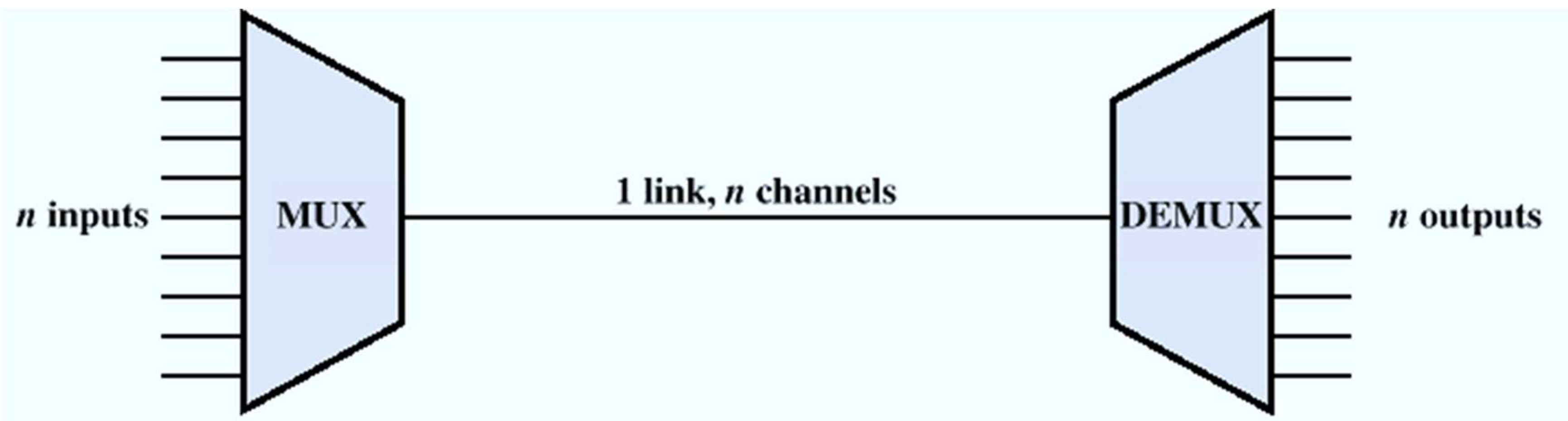
$S/N = 10^3 = 1000$

Channel capacity no more than: $v = 3100 \cdot \log_2(1+1000)$

$=30{,}894\text{bps} < 30\text{kbps}.$

Multiplexing techniques

Used when the total medium **transmission capacity** exceeds the channel's one => channels multiplexing for a better use of medium.
Useful for long-haul comms; trunks are fiber, coaxial, microwave high capacity links.
Higher data rate transmission => better cost-effective transmissions for a given application over a given distance.
Usually data-communicating devices require modest data rate 64kbps

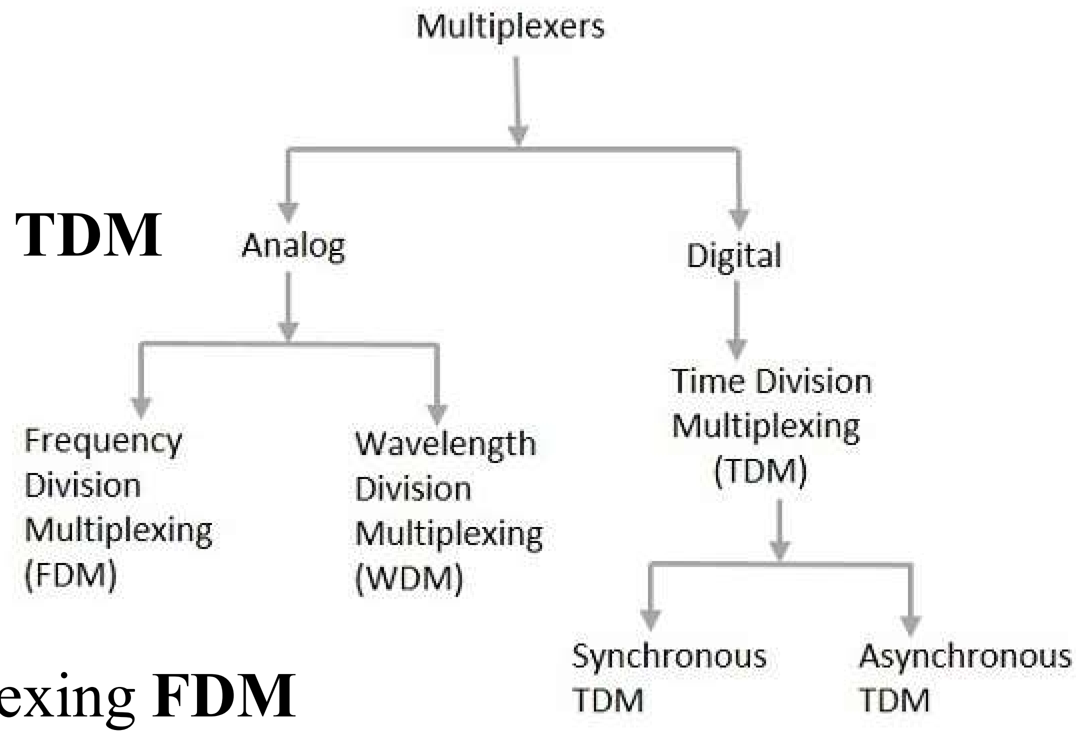


Techniques:

Time Division Multiplexing **TDM**

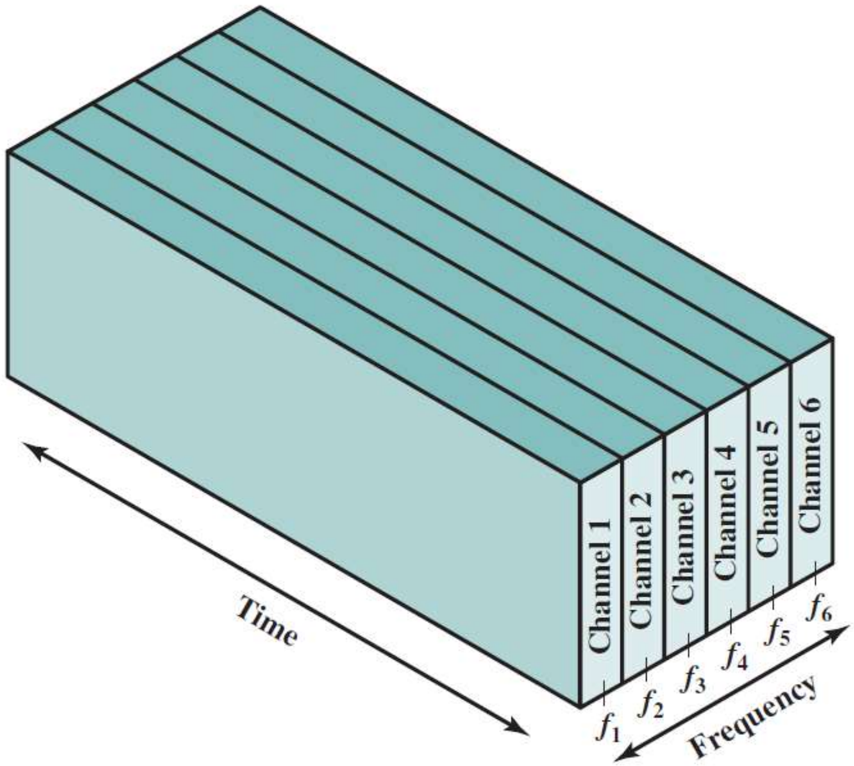
synchronous

statistical

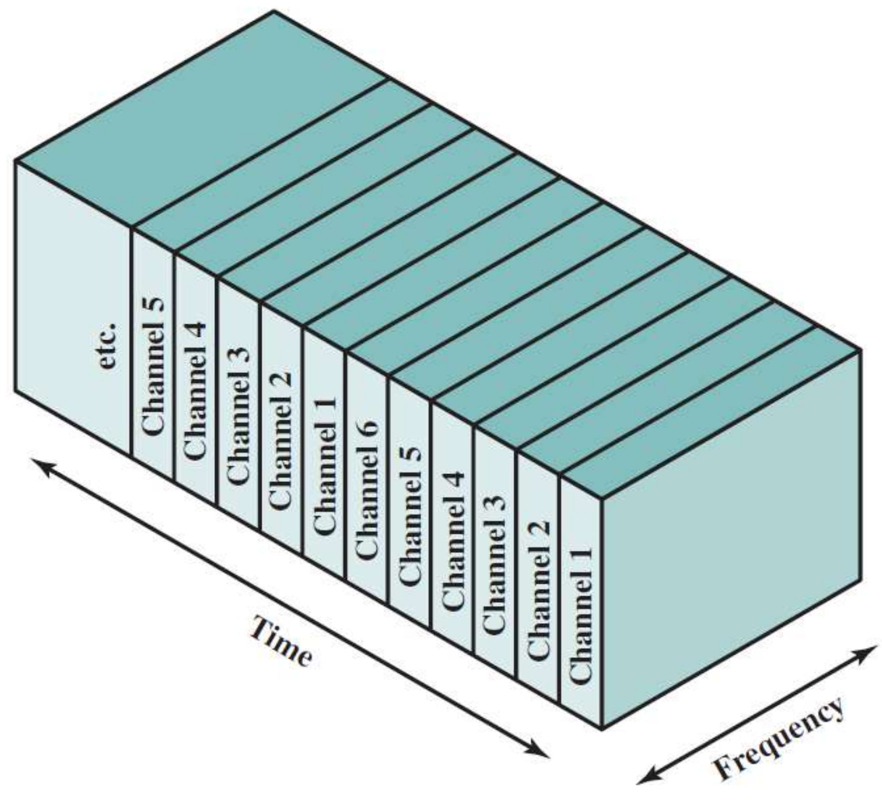


Frequency Division Multiplexing **FDM**

Wavelength Division Multiplexing **WDM** – for optical transmissions



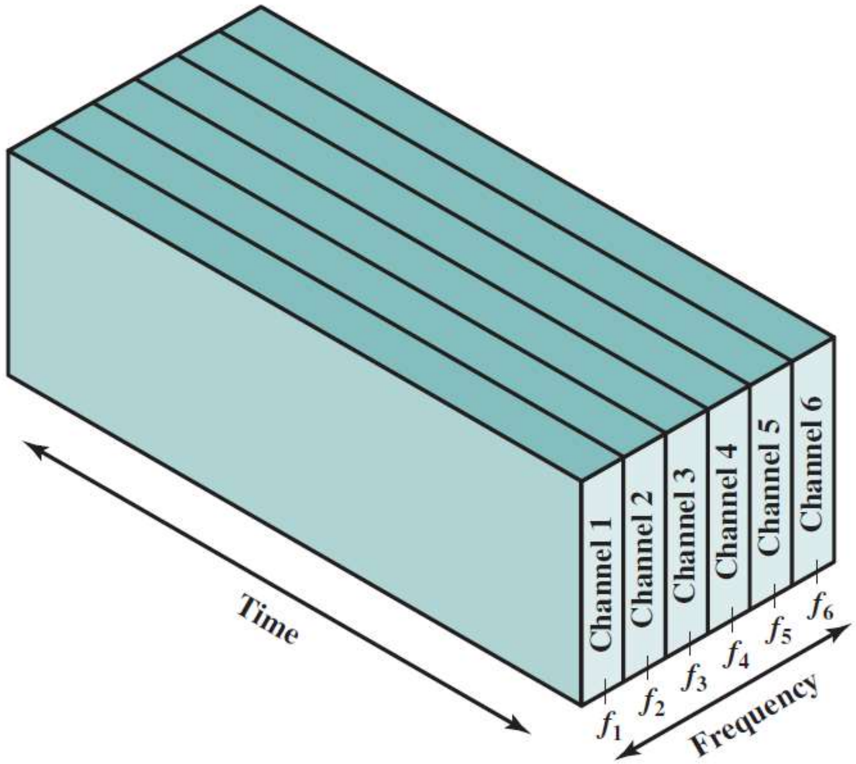
(a) Frequency-division multiplexing



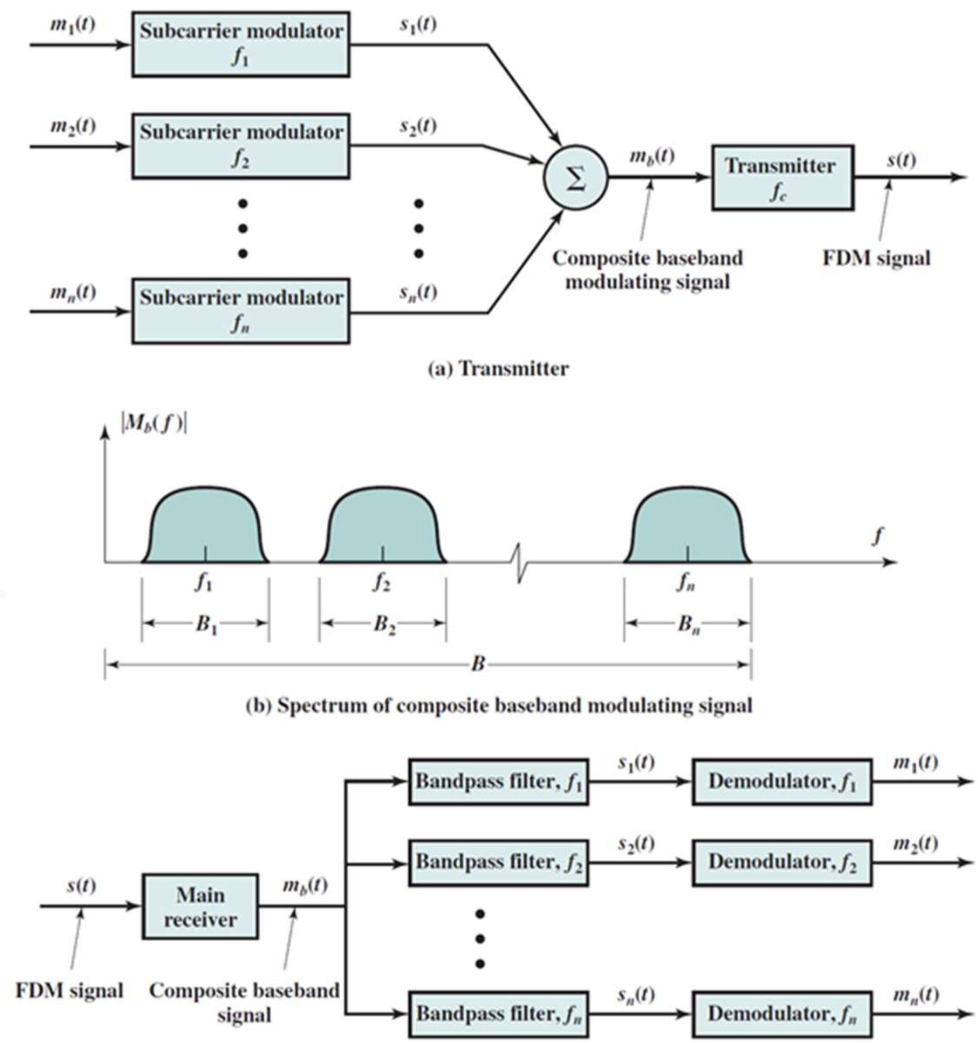
(b) Time-division multiplexing

FDM

Total allocated bandwidth >> that required by a single signal.
A number of signals carried simultaneously, each signal modulated onto a different carrier frequency, which are separated for avoiding signals bandwidths to overlap (use of guard bands).
Input signals are analog or digital, converted to analog, multiplexed onto an analog composite signal.
Relevant example: broadcast television, using RF propagation or CATV (Cable Antenna TV)
TV signal (B/Wvideo + audio + colour) fits into 6MHz bandwidth
For a coaxial cable bandwidth of 500MHz => tens of TV signals
Frequency allocation: from 54-60MHz (first channel) to 800-806MHz (68^{th} channel) – in US



(a) Frequency-division multiplexing



(a) Transmitter

(b) Spectrum of composite baseband modulating signal

(c) Receiver

Analog Carrier System

Provides voice-band signals transmission over high capacity links.
Standard (ITU-T hierarchy) based on AT&T – but not identical!
Some levels from the hierarchy:

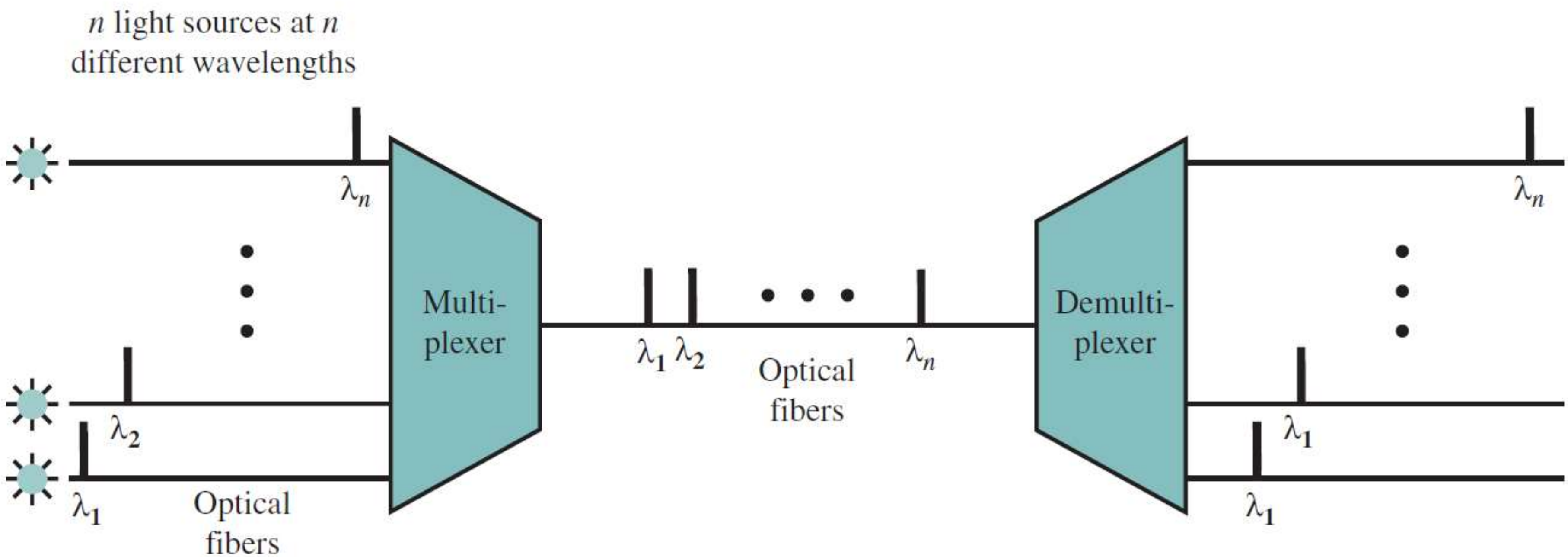
Table 8.1 North American and International FDM Carrier Standards

Number of Voice Channels	Bandwidth	Spectrum	AT&T	ITU-T
12	48 kHz	60–108 kHz	Group	Group
60	240 kHz	312–552 kHz	Supergroup	Supergroup
300	1.232 MHz	812–2044 kHz		Mastergroup
600	2.52 MHz	564–3084 kHz	Mastergroup	
900	3.872 MHz	8.516–12.388 MHz		Supermaster group
$N \times 600$			Mastergroup multiplex	
3,600	16.984 MHz	0.564–17.548 MHz	Jumbogroup	
10,800	57.442 MHz	3.124–60.566 MHz	Jumbogroup multiplex	

WDM(wavelength division multiplexing)

Multiple beams of light at different frequencies - transmitted on the same optical fiber

Dense wavelength division multiplexing (DWDM): the use of more channels, more closely spaced, than ordinary WDM (usualy channel spacing of 200 GHz or less)



Synchronous TDM

Total achievable data rate of the medium >> data rate of the signal (at least equal with the sum of signals data rate).

Method: multiple signals carried on a single path by interleaving in time portions of each (slots).

Interleaving may be at bit level or at blocks.

Time slots pre-assigned to sources and are fixed (some may be empty- slots are wasted) i.e. is synchronous.

Time slots do not have to be equally distributed among sources, depending on their own data rate.

TDM Link Control

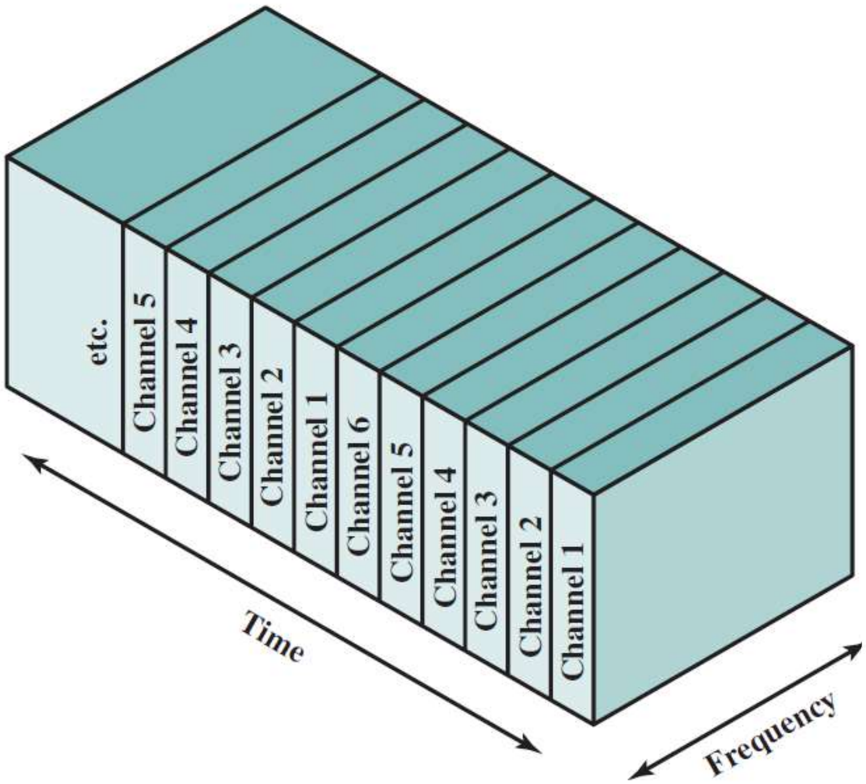
No headers and trailers; Data link control protocols not needed

Flow control

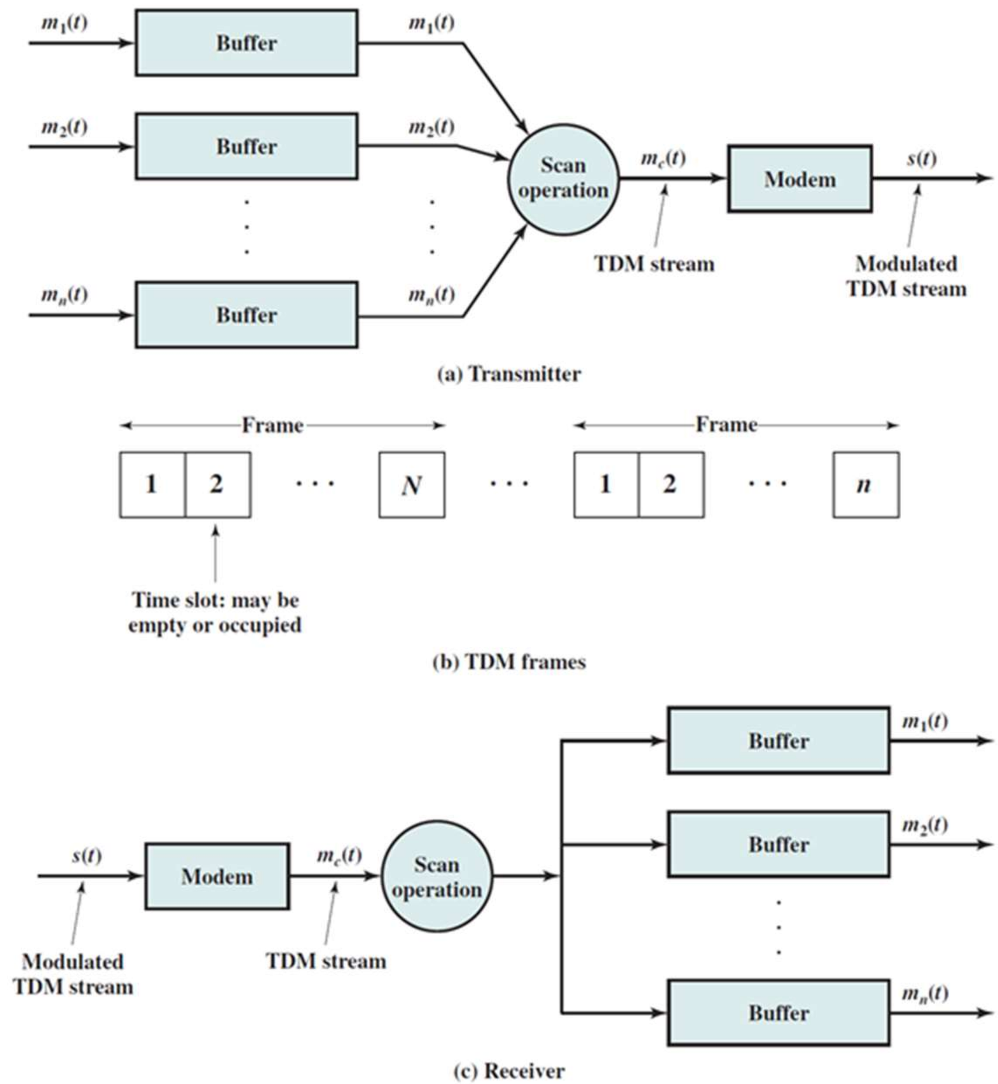
- Data rate of multiplexed line is fixed
- If one channel receiver can not receive data, the others must carry on
- The corresponding source must be quenched; this leaves empty slots

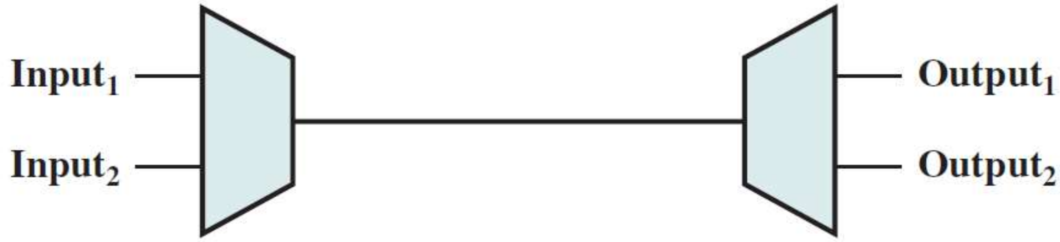
Error control

- Errors are detected and handled by individual channel systems



(b) Time-division multiplexing





(a) Configuration

Input$_1$·········· F_1 f_1 f_1 d_1 d_1 d_1 C_1 A_1 F_1 f_1 f_1 d_1 d_1 d_1 C_1 A_1 F_1

Input$_2$···F_2 f_2 f_2 d_2 d_2 d_2 d_2 C_2 A_2 F_2 f_2 f_2 d_2 d_2 d_2 d_2 C_2 A_2 F_2

(b) Input data streams

··· f_2 F_1 d_2 f_1 d_2 f_1 d_2 d_1 d_2 d_1 C_2 d_1 A_2 C_1 F_2 A_1 f_2 F_1 f_2 f_1 d_2 f_1 d_2 d_1 d_2 d_1 d_2 d_1 C_2 C_1 A_2 A_1 F_2 F_1

(c) Multiplexed data stream

Data Link Control on TDM

New issues:

Framing: synchronization of TDM frames, add of extra control bits per TDM frame

No flag or SYNC characters bracketing TDM frames

Must provide synchronizing mechanism

Added digit framing

- One control bit added to each TDM frame
  - Looks like another channel - "control channel"
- Identifiable bit pattern used on control channel
- e.g. alternating 01010101…unlikely on a data channel

Pulse stuffing: synchronizing various data sources, adding extra bits or pulses, obtaining multiples of a basic data rate (ex. 4kHz).

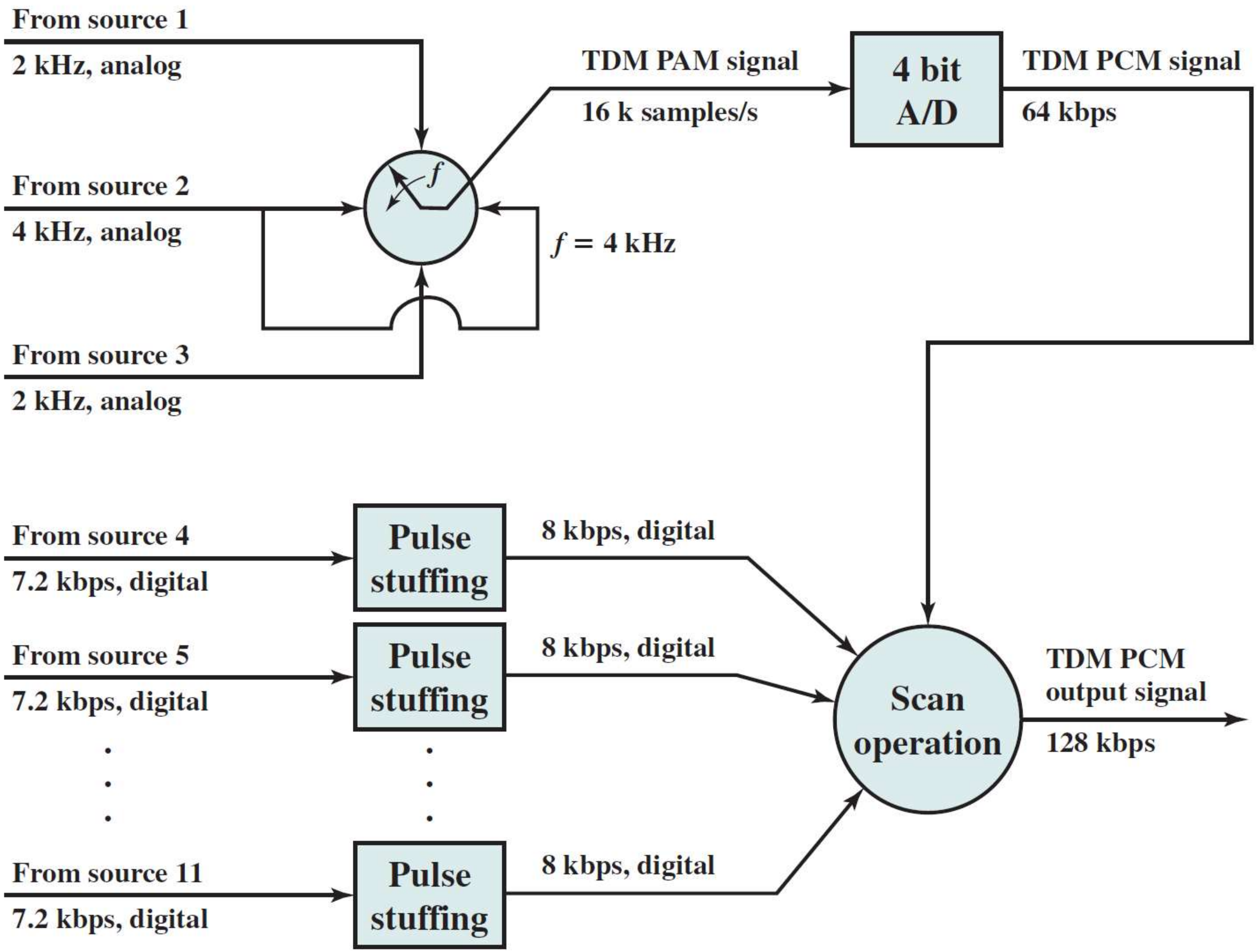
Clocks in different sources drifting

Data rates from different sources not related by simple rational number

Outgoing data rate (excluding framing bits) higher than sum of incoming rates

- Stuff extra dummy bits or pulses into each incoming signal until it matches local clock
- Stuffed pulses inserted at fixed locations in frame and removed at demultiplexer

TDM of Analog and Digital Sources



Digital TDM Hierarchy

Scale of the Digital Services:

North America	Europe
DS0: 64kbps	E0:64kbps
DS1:1.544Mbps	E1:2.048Mbps
DS26.313Mbps	E2:8.448Mbps
DS3:44.736Mbps	E3:34.368Mbps
DS4:274.176Mbps	E4:139.264Mbps
………………………	………………………..

Why 64kbps the basic data rate?

Bandwidth of the voice signal: 4kHz => Sample rate: 8kHz, or one sample every 125μsec

Number of bits for quantification: 8 => Needed data rate:

8bits/sample*8000samples/sec = 64kbps.

History: In 1962 telephone carrier (cable) between Bell System offices carried approx. 1.5Mbps over a mile (distance between amplifiers – manholes in the city)

⇒1500/64 = approx. 24 voice channels TDM multiplexed on that carrier => Telecommunication-1 carrier or T1 carrier, in USA.

T1 – 24 channels = Digital Service 1 = DS1

T1 frame has a format of 193bits, transmitted at 125μsec each.

193 = 24 * 8 data bits + 1 framing (control bit) => gross data rate: 1.544Mbps, from this: 8000bps of signaling information… may be to much?

Control bit is 1 or 0, according to the synchronizing sequence 10101….

An example for **signaling**, transmission of control information.

ITU-T standard for signaling differs from US Bell's one (T versus E !)

Two major signalling methods:

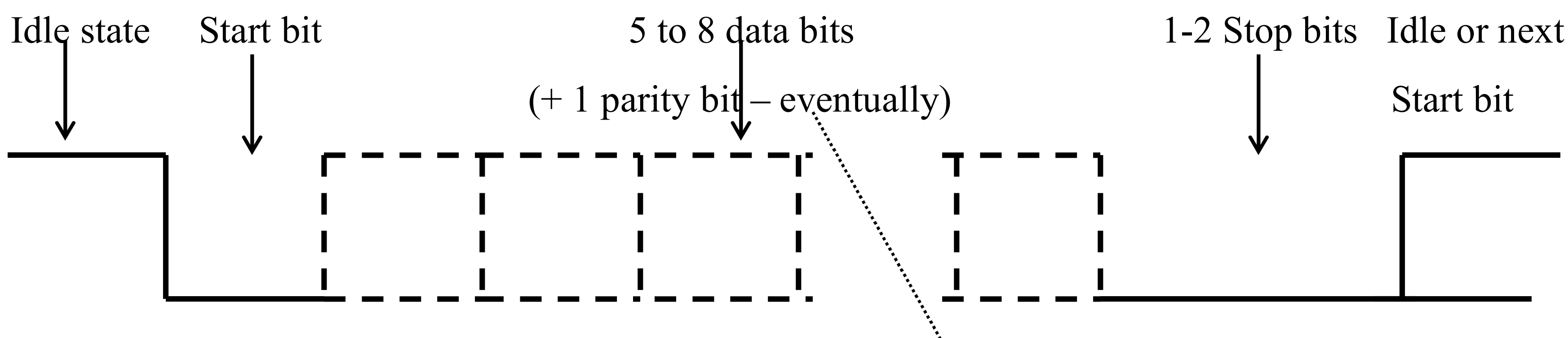
- -common-channel signaling (as above)
- -channel associated signaling: an extra signaling subchannel provided

Synchronization

Asynchronous transmission

Data are transmitted one *character* at a time, where each character is five to eight bits in length (utile data). See ASCII code…

Timing or synchronization must only be maintained within each character; the receiver has the opportunity to resynchronize at the beginning of each new character. Samples are taken in the middle of the bit period.



Synchronous transmission

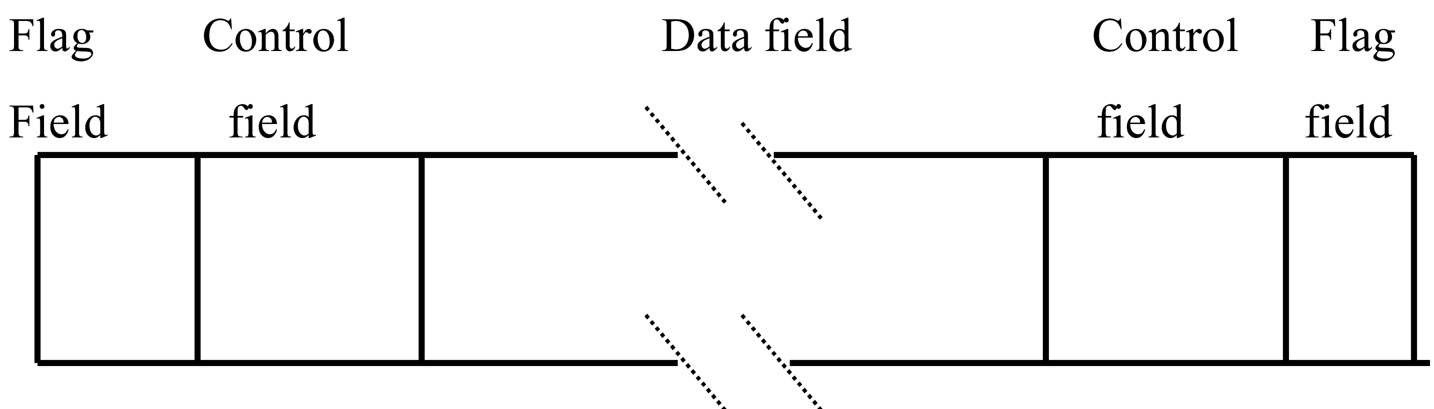
Works with blocks of bits (characters).

Inter-clock synchronization:

- auxiliary clock line
- biphasic coding

+ Synchronization at the block level => extra flag and control fields => data structure of **frame**

Flag fields (synchronization) fields: special bit sequences or *sync* characters; denoted as *preamble-header* and *trailer*



Comparisons

Asynchronous transmission	**Synchronous transmission**
Simple	Complex
Cheap	Expensive
Fixed burden (20%, 30%), depends on Stop bit number	Burden varies with block size
Fits keyboard action	Fits transmissions of blocks of data
1000 bytes takes 10000bits	1000 bytes may take 1003bytes

SOLVED PROBLEMS

#1. The human hearing system operates in the range of 2 – 20,000Hz. What sampling rate will be sufficient to preserve the information content of the signals in this range?

Solution

Cf. Nyquist theory, the sampling rate must be at least twice the bandwidth. Requested bandwidth is: 20000-2 = 19998Hz, so the necessary for sampling is 2•19998 = 39,996 samples/sec.

#2. In order to transmit an uncompressed video stream at 30 frames/second into a quarter size VGA window (160 * 120 pixels), where each pixel requires 24bits for colour, what transmission capacity is required?

Solution

Total number of pixels in a window: 160•120 = 19,200pixels.
Total number of bits requested within a window: 19,200•24 = 460,800bits.
Number of bits for 30 frames (number of bits sent on a second): 460,800•30 = 13,824,000bits so there is a need for a transmission speed of approx. 13.8Mbps.

#3. Given a link with a signal/noise ratio of 1023, what bandwidth is required to support the transmission rate from previous problem?
At the required bandwidth, how many bits will be transmitted per Hertz?

Solution
Cf. Shannon theorem ($v=H \bullet \log_2(1+S/N)$); the transmission rate (channel speed) is 13,824,000bps.
$H = 13{,}824{,}000 / \log_2(1+1023) = 13{,}824{,}000 / 10 = 1{,}382{,}400$Hz, aprox. 1,3MHz.

#4. A full duplex 64,000bps point-to-point data link was observed for sixty seconds. During this observation period, the following were obtained:
-50 original data packets, each containing 24 header bytes and 1000 data bytes
-five additional data bytes observed to be retransmissions
-100 acknowledgements , each containing 24 header bytes and no data
-4 connection management packets, each containing 124 bytes.
What was the channel utilization?

Solution
The bytes sent with data packets: $50 \bullet (24 + 1000) = 51{,}200$ bytes

The bytes sent for retransmission: 5 bytes

The bytes for acknowledgements: 100 • 24 = 2400 bytes

The bytes for connection management: 4 • 124 = 496 bytes

The total of sent bytes: 51,200 + 5 + 2,400 + 496 = 54,101 bytes

1 byte = 8 bits = > a number of: 54101 • 8 = 432,808 bits

Theoretically during 60 sec, channel could carry: 64,000 • 60 = 3,840,000bits.

The utilisation of the channel is: number of sent bits / theoretical number = 432,808 / 3,840,000 ~ 9 %

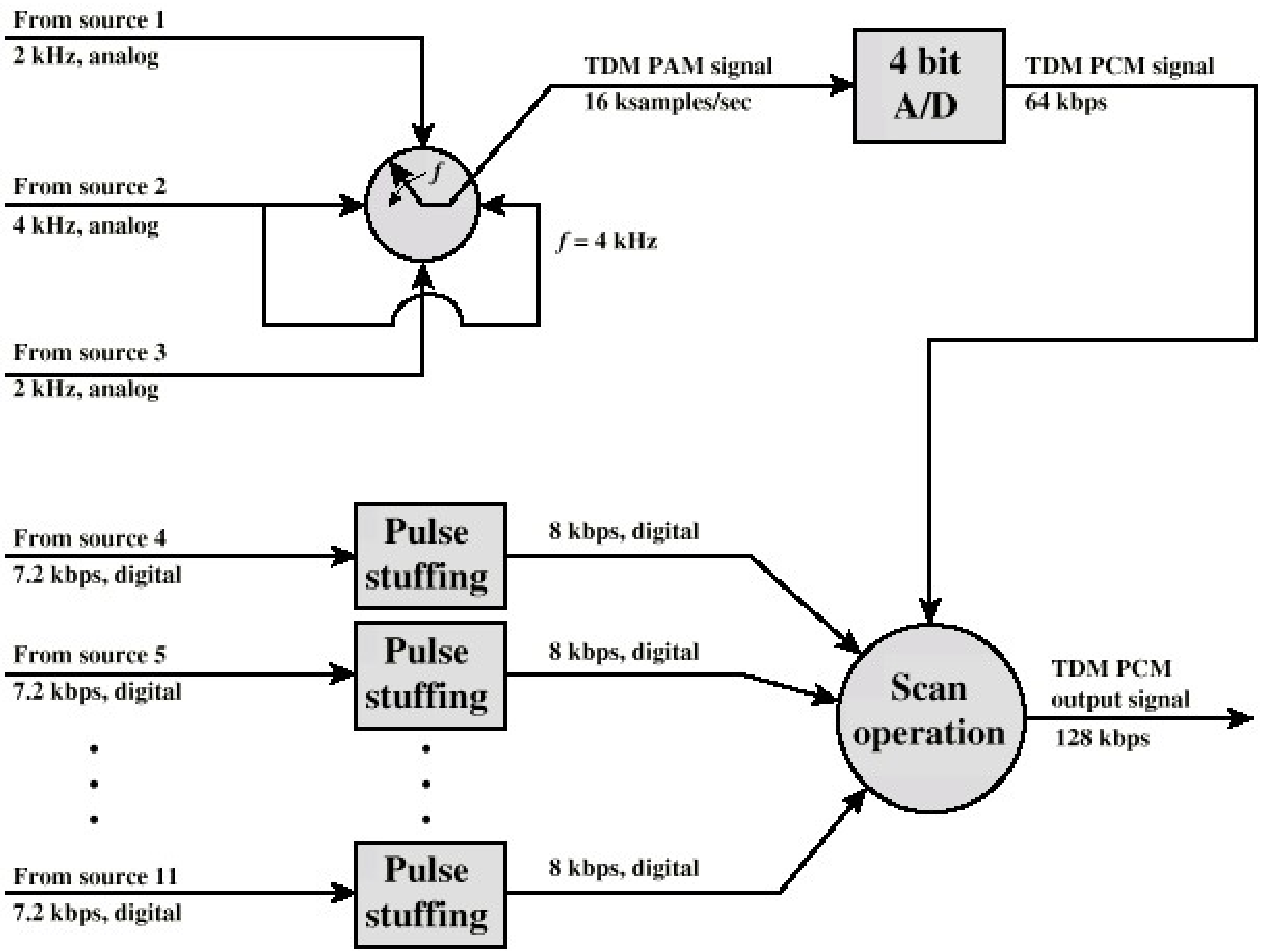
#5. Consider that there are 11 sources to be multiplexed on a single link:

-source 1: analog, 2kHz bandwidth

-Source 2: analog, 4kHz bandwidth

-Source 3: analog, 2kHz bandwidth

-Sources 4-11: digital, 7200bps synchronous.(see next slide)


From source 1
2 kHz, analog
From source 2
4 kHz, analog
From source 3
2 kHz, analog
f
f = 4 kHz
TDM PAM signal
16 ksamples/sec
4 bit A/D
TDM PCM signal
64 kbps
From source 4
7.2 kbps, digital
From source 5
7.2 kbps, digital
From source 11
7.2 kbps, digital
Pulse stuffing
Pulse stuffing
Pulse stuffing
8 kbps, digital
8 kbps, digital
8 kbps, digital
Scan operation
TDM PCM
output signal
128 kbps

Solution

Analog sources converted to digital using PCM;
Cf. Nyquist theorem, the sampling rate needs be at least twice the bandwidth, I.e. 4000samples/sec for sources 1 and 3 and 8000 samples/sec for source 2. Sampling, we obtain analog samples (PAM) => need for quantification (be digitized); assume that 4 bits are enough. If we consider that these 3 sources are multiplexed first, at a scan rate of 4kHz, we will obtain one PAM sample for sources 1 and 3, and 2 PAM samples for source 2. These 4 samples are interleaved and converted to 4-bit PCM samples (digital values). So we need 16bit (16 bit buffer needed) to represent all PCMs, and this item is generated 4000 times/sec, so results a composite rate for the digital signal of 64kbps.
For the digital sources we will use first the *bit stuffing* to reach a rate of 8kbps, so we obtain a aggregate rate of 64kbps. For each digital source we need a 2-bit buffer (because the scan is done every 1/4000sec).
Adding all signals, it results we need a TDM composite signal of 128kbps, and the frame structure will contain 32 bits.

Proposed problems

#1. A modem operates at 1800 baud and can encode each sample using 4 bits. What is the data rate at which the modem can transmit data?

#2. What is the channel capacity for a teleprinter channel with a 300Hz bandwidth and a signal-to-noise ratio of 3dB?

#3. Given a channel with an intended capacity of 20Mbps, the bandwidth of the channel is 3MHz. What signal-to-noise ratio is required to achieve this capacity?

#4. A digital signaling system is required to operate at 9600bps. If a signal element encodes a 8-bit word, what is the minimum required bandwidth of the channel?

#5. a). A digitized TV picture is to be transmitted from a source that uses a matrix of 480 x 500 picture elements (pixels), where each pixel can take one of 32 intensity values. Assume that 30 pictures are sent per second. Find the source data rate.
b). Assume that the TV picture is to be transmitted over a channel with 4.5MHz bandwidth and a 35dB signal-to-noise ratio. Find the capacity of that channel.
c). Assume that a noiseless fiber optic channel is used; how much bandwidth is needed and how many microns of wavelength are needed for this band at 1.30microns?

#6. Deduce the maximum theoretical information rates associated with the following transmissions channels:
a). Telex network with a bandwidth of 500Hz and a signal-to-noise ratio of 5dB
b). Switched telephone network with a bandwidth of 3100Hz and a signal-to-noise ratio of 20dB

#7. A noiseless 4KHz channel is sampled every 1msec. What is the maximum data rate?

#8. Television channels are 6MHz wide. If the channel is noiseless, what data rate may be achieved for a four-level digital signal used?

#9. If a binary signal is sent over a 3kHz channel whose signal-to-noise ratio is 20dB, what is the maximum achievable data rate?

#10. Why has the PCM sampling time been set at 125microsec?

#11. Ten signals, each requiring 4000Hz, are multiplexed onto a single channel using FDM. How much minimum bandwidth is required for the multiplexed channel? Assume that the guard bands are 400Hz wide.

#12. Assuming the velocity of propagation of an electrical signal is equal with 70% of the speed of the light, determine the ratio of the signal propagation delay to the transmission delay, for the following types of data link and 1000 bits of data:

a). 100m of UTP wire and a transmission rate of 1Mbps

b). 0.5km of coaxial cable and a transmission rate of 10Mbps

If the signal propagates with the speed of the light, the same question for :

c). A satellite link and a transmission rate of 512Kbps

d). 2.5km of fiber optic and a transmission rate of 1000Mbps

#13. The maximum distance between two terrestrial microwave stations is given by the expression:

$$d = 7.14\text{sqrt}(K \bullet h)$$

K relates to the curvature of the earth and *h* is the height of the dishes above. Assuming *K* = 4/3 determine *d* for the following values of *h:* 10m, 20m, 50m. 100m.

#14. Draw a block diagram similar to figure in slide#5 for a TDM PCM system that will accommodate 4 digital synchronous inputs at 300bps, and one analog input with a bandwidth of 500Hz. The analog samples will be coded using 4bits.

#15. Find the number of the following devices that could be accommodated by a T1–type TDM line, if 3% of the line capacity is reserved for synchronization purposes:
-110bps teleprinter terminals
-1200bps computer terminals
-64kbps PCM voice frequency lines.