Lecture 4 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

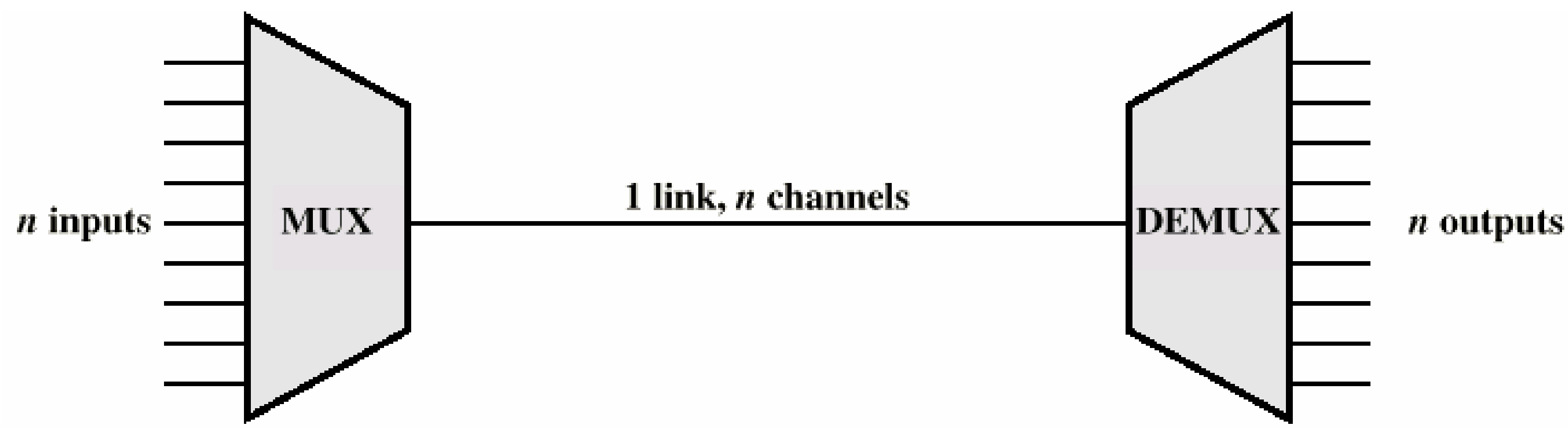
S/N: signal per noise ratio, expressed usually as $\mathbf{10 \bullet \log_{10} S/N}$ and measured in dB (also usually understood as attenuation).

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

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).

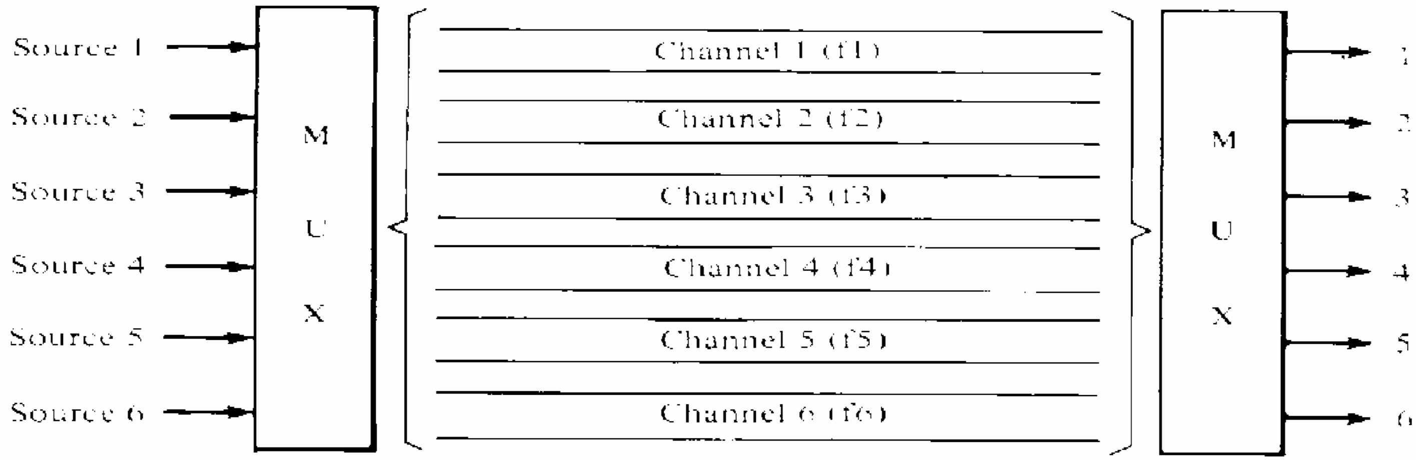
Input signals are analog or digital, converted to analog, multiplexed onto an analog composite signal.

Relevant example: broadcast television, using RF propagation or CATV (Cable Antenna TV)

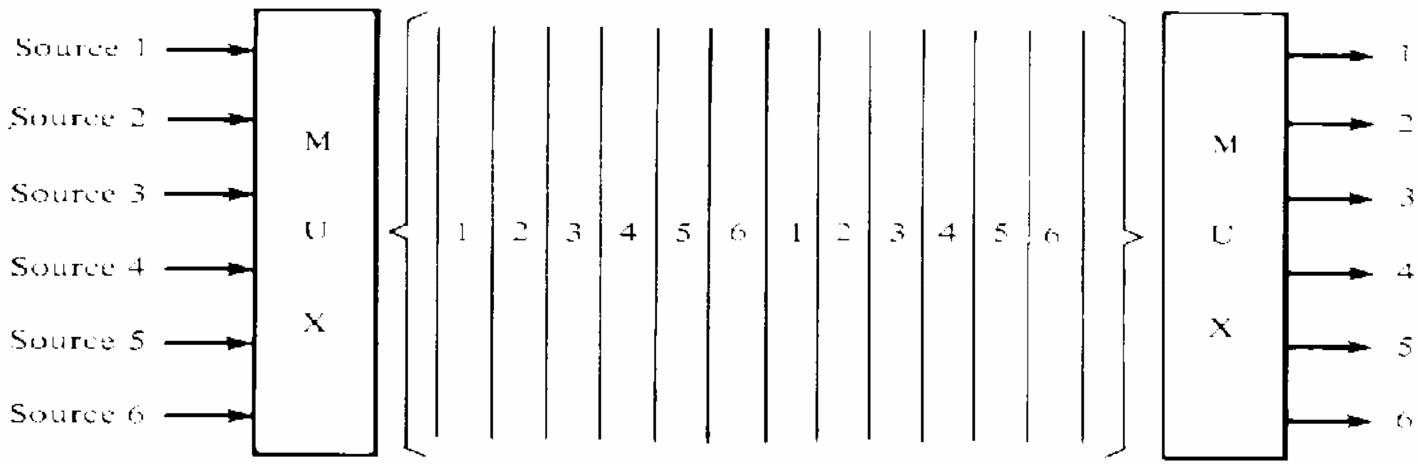
TV signal (B/Wvideo + audio + colour) fits into 6MHz bandwidth

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

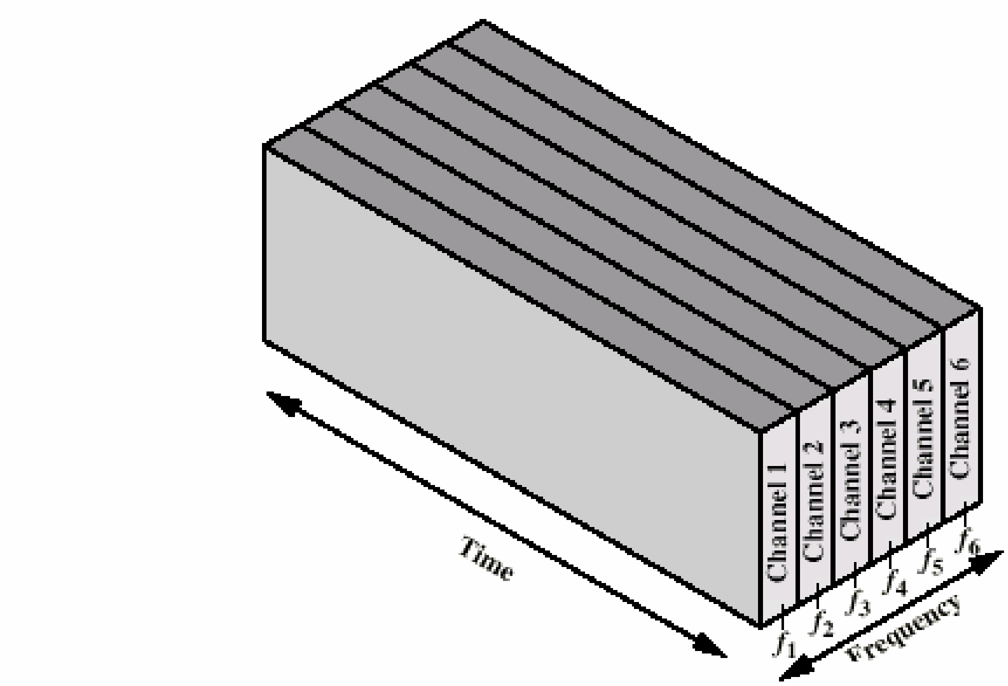


(a) Frequency-Division Multiplexing

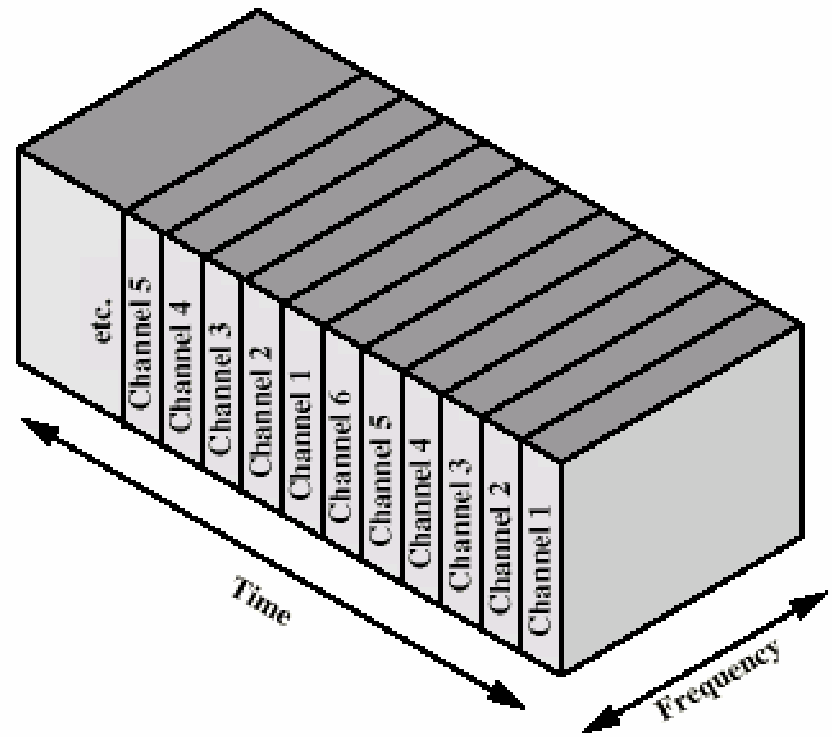


(b) Time Division Multiplexing

Multiplexing techniques


Channel 1
Channel 2
Channel 3
Channel 4
Channel 5
Channel 6
Time
Frequency

FDM


etc.
Channel 5
Channel 4
Channel 3
Channel 2
Channel 1
Channel 6
Channel 5
Channel 4
Channel 3
Channel 2
Channel 1
Time
Frequency

TDM

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:

Nr.Voice Channels	Bandwidth	Spectrum	AT&T	ITU-T
12	48kHz	60-108kHz	Group	Group
60	240kHz	312-552kHz	Supergroup	Supergroup
300	1.232MHz	812-2044kHz		Mastergroup
600 …………..	2.52MHz ………….	564-3084kHz …………….	Mastergroup …………….	 ……………
10800	57,442MHz	3.124-60.566 MHz	Jumbogroup multiplex	

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

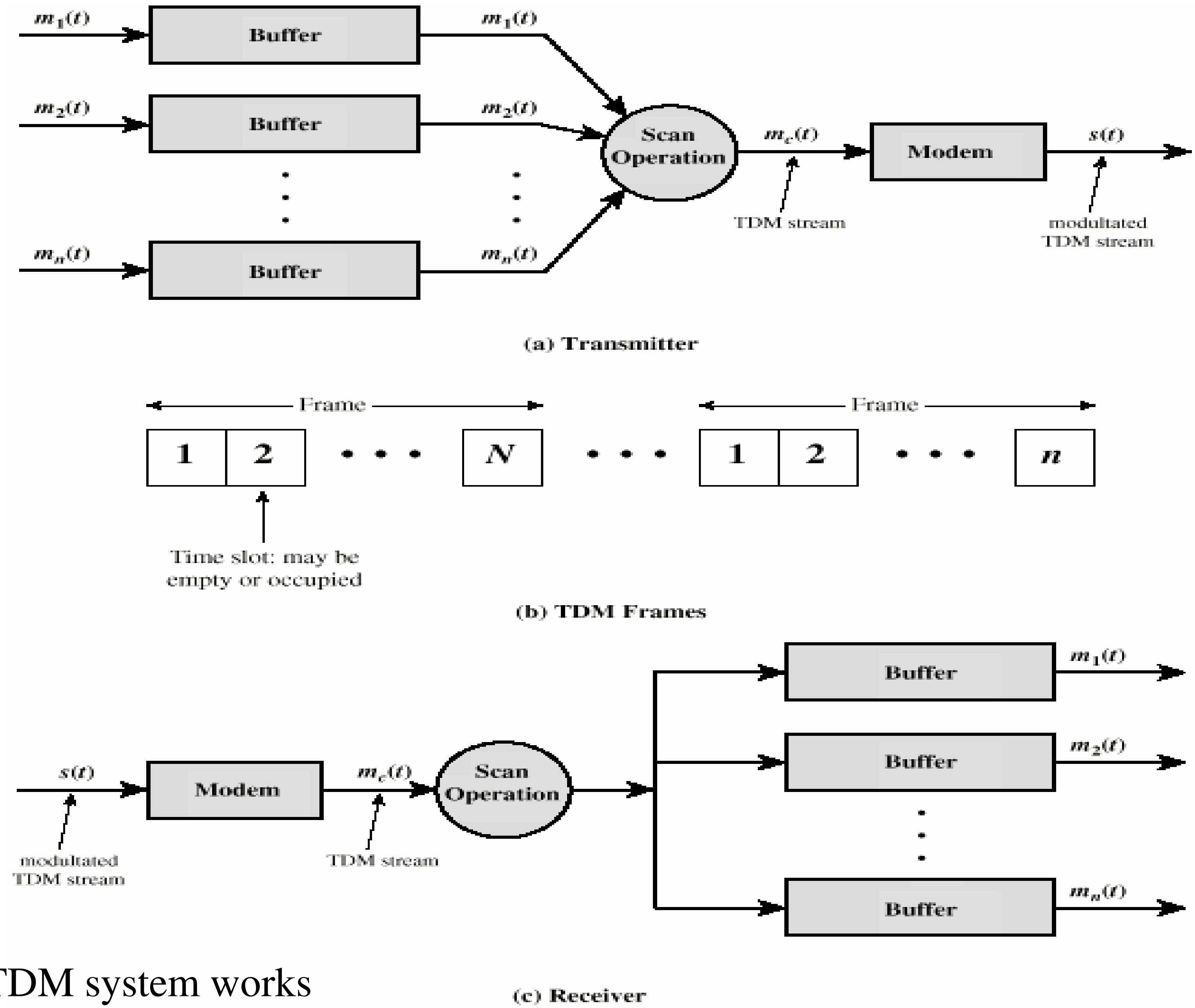
No headers and trailers; Data link control protocols not needed

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



How a TDM system works

(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

Legend: F = flag field d = one octet of data field
A = address field f = one octet of FCS field
C = control field

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).

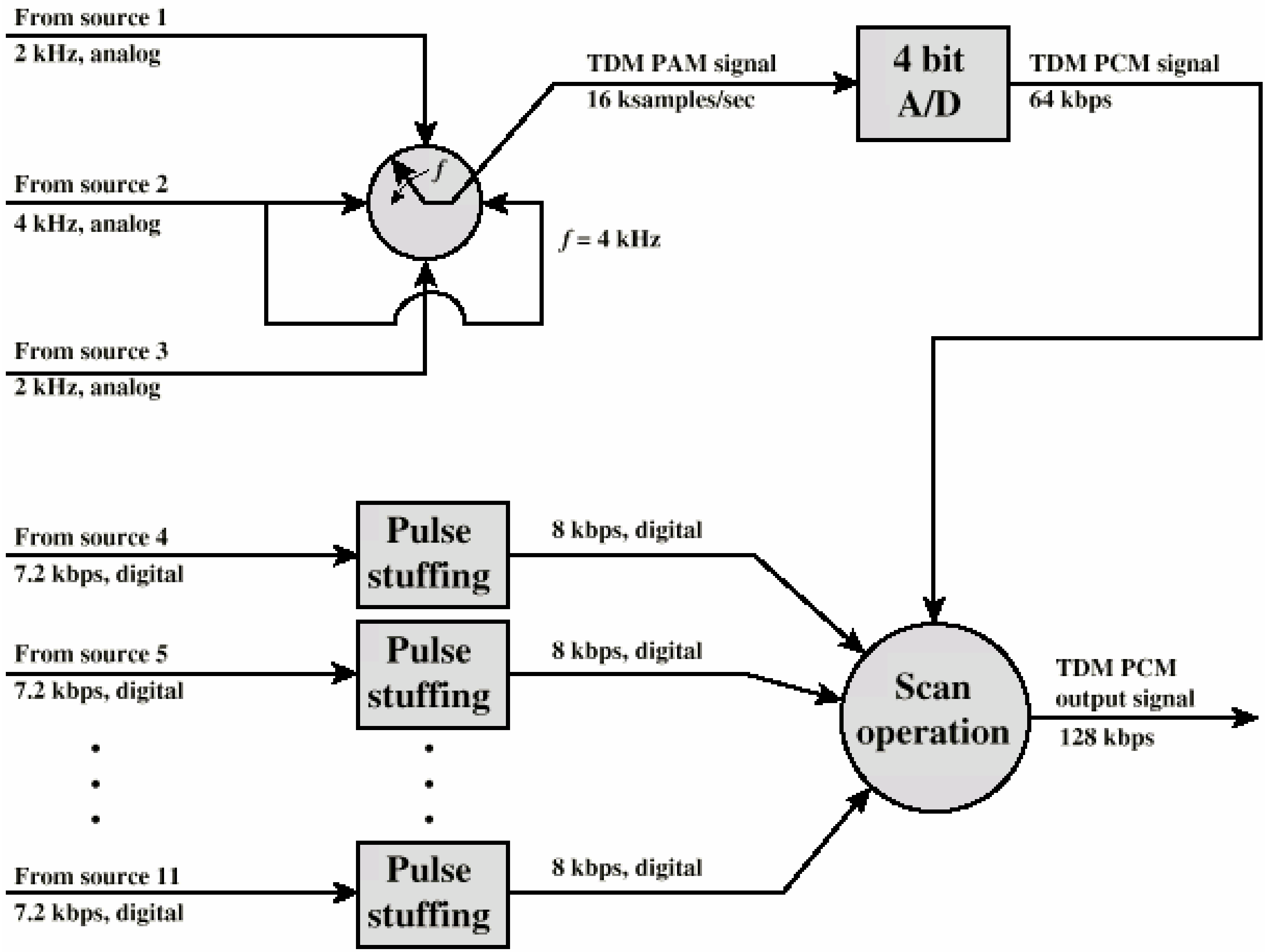
Clocks in different sources drifting

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)

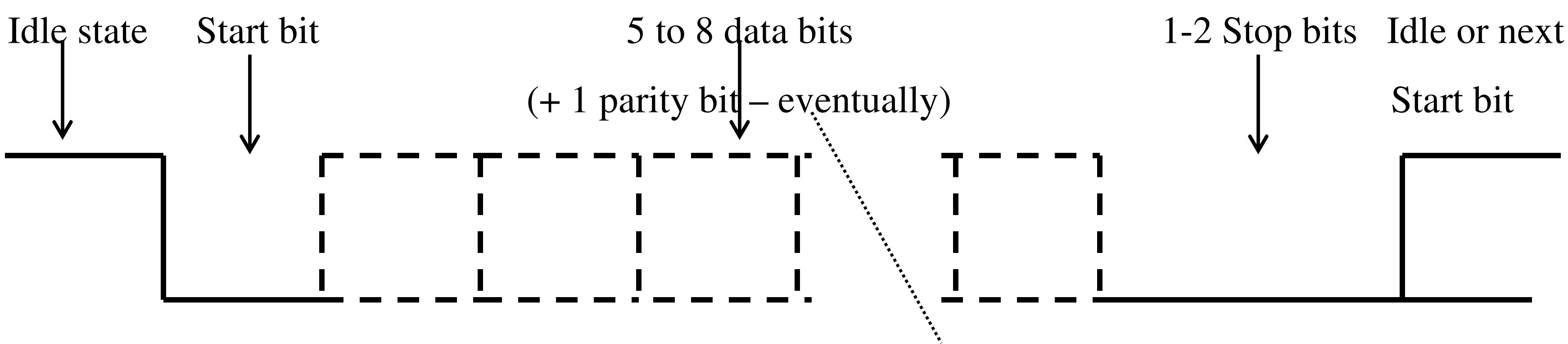
-channel associated signaling: an extra signaling subchannel provided

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).

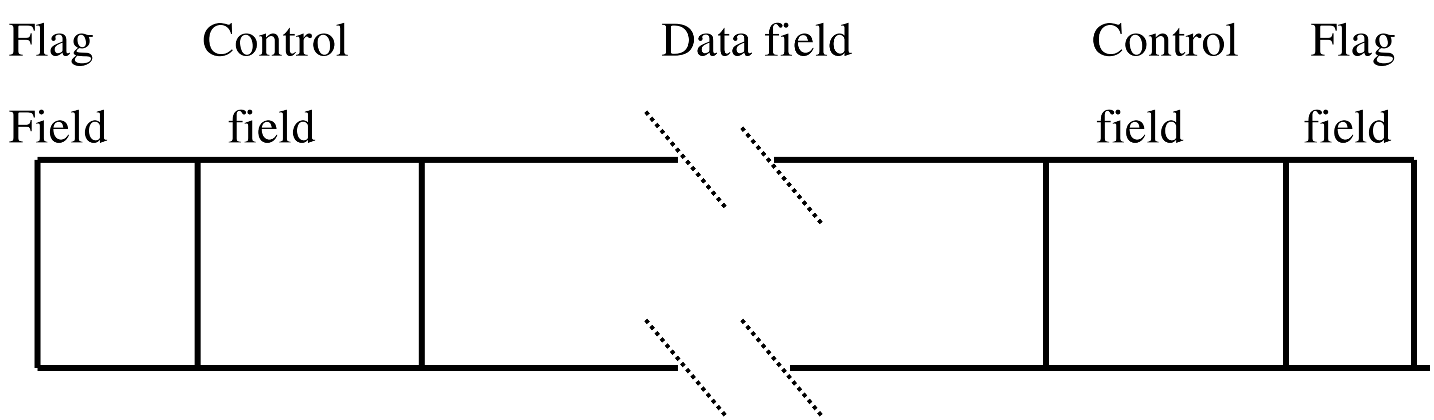
Inter-clock synchronization:

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

Switching techniques

Traditionally the telephonic system is based on circuit switching; is the main infrastructure for communications (computer) networks => the switching term remains.

Switching techniques used in information transfer are:

-**circuit switching**

-message switching

-packet switching

Circuit switching

Physical path between communicating parts, achieved using circuit switching –switches (relays)-in the networks nodes.

Three phase communication:

-circuit establishment (setup), establish a (optimum) path between parts; both parts agree communication

-effective data transmission (signal transfer), on this route

-circuit release (disconnection); initiative of one part.

Drawbacks:

-not efficient due to existence of the first phase (it will exist even if there's no data transfer)

-need for covering bandwidth allocation

-important amount of cabling

-no buffers in switches for transmission equalization

Today use of digital **PBX** (Private Branch Exchange)

First circuit-switching: space-division switching (separated signal paths – divided in space): crossbar matrix of I/O full duplex lines
An improvement: multiple-stage switches
Today all telephony: digital time-division techniques (synchronous TDM)
Signaling in digital telephony:
-inchannel
  -in-band: signals using the same band as the voice channel (as payload)
  -out-band: (voice signals do not use whole 4kHz bandwidth)
-common channel – a common signal channel for a number of voice channels
Signaling may use the same (or not) path as the payload (associated/nonassociated modes)
What's signaling? **Signal = control** Examples:
  -connection setup request = off-hook signal from telephone to switch
  -connection setup acknowledge = dial tone
  -destination address = pulse or tone dialing
  -destination busy = busy tone
  -destination available = ringing tone
Other signaling functions: transmission of: dialed number between switches, information about a call not completed, about billing, diagnose and failure isolation

Message switching

Data transfer using **messages** (independent data units, with diff. lengths but similar structures). Types: control and data (embedding control)

Need for addressing (source & destination of message)

Communications nodes are not physical switches, but computing systems (with memory and processing units).

Philosophy is: message *store & forward* .

Not more dedicated communications path; established in an optimum way (cost, network status) by nodes (using routing tables).

Advantages:

-improvement in efficiency (path multiplexing)

-introduces message priority

-equilibrated transmissions.

Drawbacks:

-messages are too long, memory waste and difficult error recovery

Packet switching

Combines the advantages of previous methods. The **packet** has similar message structure but a lower length, up to 1000octets.

Two methods:

-use of **datagrams** (close to message switching)-more speedy and flexible method

-use or not of transmission acknowledgments (ACK)

-use of **virtual circuits** (close to circuit switching)-use of the three phases (connection request, data transfer, disconnect) for a logical connection activation; use of special control packets for that. Also embedding of control information (piggybacking).

A logical connection may be implemented with more different physical connections.

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 ($v=H \cdot \log_2(1+S/N)$); 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: $50 \cdot (24 + 1000) = 51{,}200$ 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 %

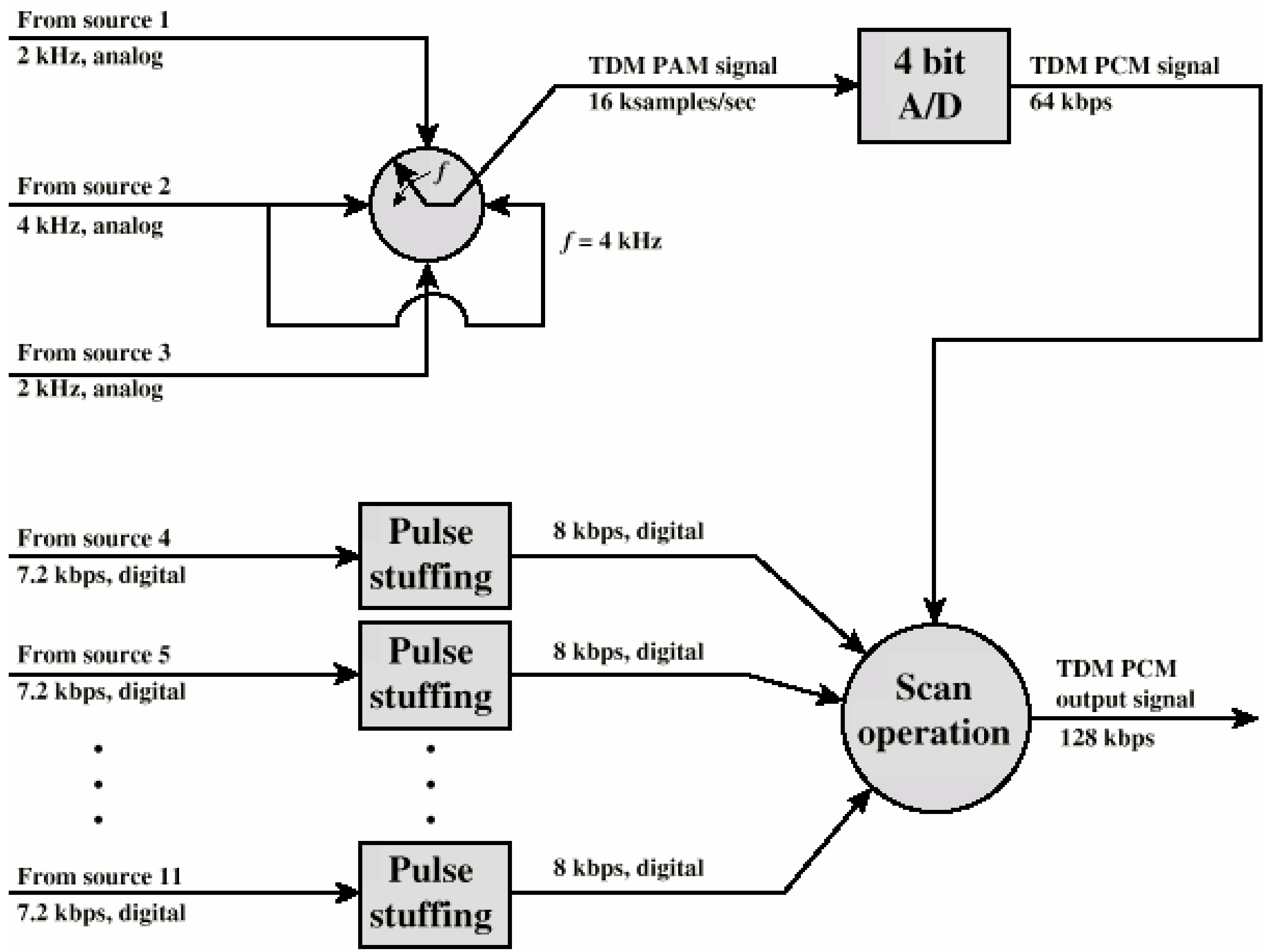
#5. Consider that there are 11 sources to be multiplexed on a single link:

-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
Pulse stuffing
8 kbps, digital
From source 5
7.2 kbps, digital
Pulse stuffing
8 kbps, digital
From source 11
7.2 kbps, digital
Pulse stuffing
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.