Electronic Equipment Testing

1. Test levels: definition and description of each level's characteristics

Course 1, slides 6-7

There are three levels of test:

- *electronic component (including printed circuit board),*
- *electronic board (mounted components),*
- *equipment.*

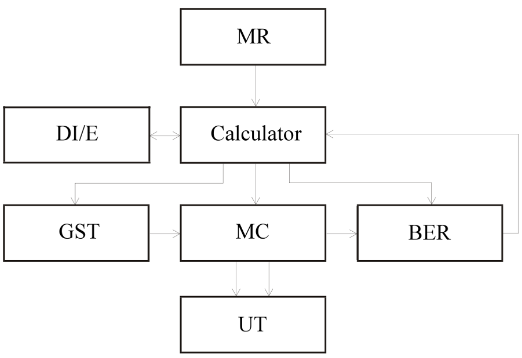
a) *failed test at the component level – the component should not be mounted on the board, economically has been proven that this test (also done by the producer) should be repeated by the user*

b) *can be done through "nail bed testing" (complicated and costly for realizing the test equipment) or through "test stimulus generation" (simple connection, but costly for implementing the test program)*

c) *depends on the equipment: normally involves broken connections between functionally correct boards. Can involve some more sophisticated equipment, such as the signature analyzer.*

2. Architecture of an automatic test equipment. Characteristics of each block's functionality

Course 1, slides 8-9



UT – unit under test

GST – test signal generator

BER – responses evaluation block

MC – connection matrix

MR – external memory

3. Principles of the signature analyzer

Course 2, slides 4-7

The signature analyzer is based on the principle of data information flow compression. The long data sequences are acquired from the tested board/equipment and compressed into fixed-length information, called "signature". The signature should be easy to be recognized, easy to be interpreted by comparing it with a control signature that corresponds to the correct functioning.

According to how the data flow is collected, the signature analyzer can be: serial or parallel.

The heart of the signature analyzer is a Pseudo-Random Sequence Generator, that performs data compression. The signature is provided as a fixed length information, on 16 bits, decoded into groups of 4 bits, displayed as alphanumeric characters on 7 segment LEDs. It avoids using small alphabet letters (b, c, d) and ambiguous capital letters (B, D, G, I), using instead letters like H, P, T, U.

The signature should be simple to be recognized, un-ambiguous and stable.

The signature analyzer should have the auto-test facility, in order to avoid wrong decisions.

4. Principles of testing the static parameters of a digital integrated circuit

Course 3, slides 4-6

The static parameters of an IC are:

- *Input and output voltages*
- *Input and output currents*

They are stable during the test. A time should be given from powering the IC, in order to allow stabilisation of the transition factors.

Test should be carried out under the less favourable conditions: minimum power supply value, maximum circuit's charge, etc.

The Test Signal Generator is formed by a number of Programmable Voltage Sources (PVS) and Programmable Constant Current Generators (PCCG).

The Responses Evaluation Block is a simple measuring instrument.

The Connection Matrix is formed by a number of relays or commuting transistors.

5. Principles of testing the dynamic parameters of a digital integrated circuit

Course 4, slides 2-5

The dynamic parameters of an IC are:

- *the transition times:* t_{tLH}, t_{tHL}
- *the propagation times:* t_{pLH}, t_{pHL}

Those parameters might be defined by fixed or percentage thresholds. The realisation of the testing structure depends on that definition.

The evaluation should be done in the worst functional case: power supply, charge, etc

The Test Signal Generator uses digital programmable pulse generators: fronts, length, amplitude, polarity, filling factor, etc

The Responses Evaluation Block is typically formed by a counter, with "start" and "stop" commanded by the discriminated fronts of the tested parameter.

6. Principles of the modular activation method for processors' functional tests

Course 6, slides 9-11

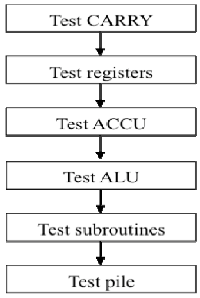
The processor is divided on hierarchical levels (both functional and hardware).

The test is done based on a self-test program.

The program starts with the basic level

Each tested level can be used for testing superior levels

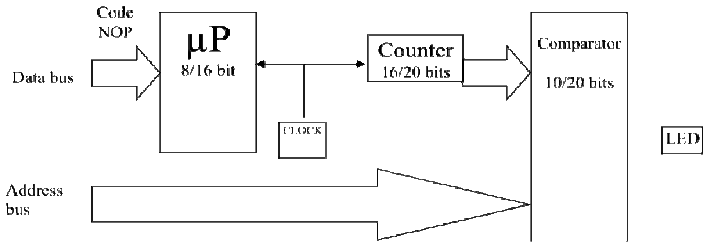
A possible testing strategy:



In order to start, a minimum level of components and functions should be correct! - KERNEL

The kernel should include: initialization circuits, program counter, address bus, data bus, instruction decoder, clock circuit

The hardware test of the kernel might be done through the following strategy:



If the kernel test is passed, the following circuits are functionally corrects:

- *Clock circuit*
- *Initialization circuit*
- *Program counter*
- *Address bus*
- *Data bus – partially*
- *Instruction decoder – partially*
- *Instruction register – partially*

7. Principles of the test stimulus vectors' generation

Course 7

The test stimulus vectors are generated through algorithms that are describing the functions of the schema on a digital electronic board.

The main methods used are:

- *Single Path Activation Method*
- *Poage*
- *Poage McCluskey*

There are two basical principles:

- *forcing the complement of the tested error into the test node, then activation the propagation of the value in that node towards the observable output, where the received value is compared in order to take the decision: correct or erroneous board*
- *describing the functions on the board by using supplementary variables that are describing the functioning status of each line through their activation, then the condition to get complementary values at the output in case of "erroneous" or "correct" board is used in order to calculate the test stimulus vectors.*

8. Principles for testing the Bit Error Rate for a telecommunication digital network

Course 10, slides 4-9

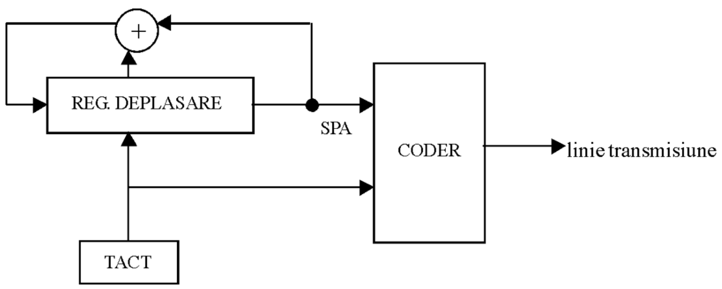
The Bit Error Rate (BER) is the ratio between the number of error bits and the total number of transmitted bits

BER estimation can be done:

- *in-service*
- *out-of-service*

The test signal is a pseudorandom binary sequence, standardized through the 0.151 CCITT recommendation.

The principle of the test generator used at the transmission end is:



The problem is: avoiding the lock on 0 of the shift register.

At the reception end, there is a similar signal generator that provides the same sequence as at the emitter end, that is compared to the one received through the telecom network.

The problem is to ensure synchronization between the two generators.

9. Principles for the realization of a fault tolerant system

Course 11, slides 4-7

Fault tolerance is an architectural attribute of a system, making possible for the system to function properly even when one or more faults appear in its structure.

Implementation is done through "redundancy", but the price to be paid is the high cost.

The use is in very critical applications: nuclear, military, aero-spatial, etc.

Fault = physical problem of one of the system's elements, taking to the permanent, temporary or intermittent erroneous function of the system

Error = symptom of a fault

The used strategies are:

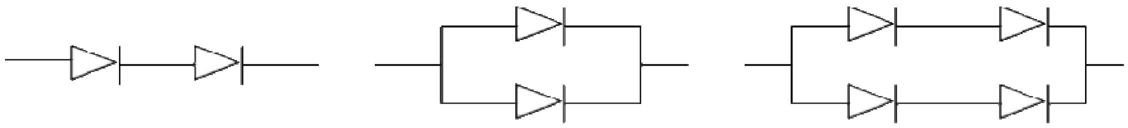
- *Fault diagnosis and faulty elements replacement*
- *Fault masking*
- *Mixed strategies*

The used testing methods are:

- *Initial testing: before normal operation*
- *On-line testing: during normal operation*
- *Off-line testing: for error detection and diagnosis*
- *Redundant modules testing: in order to see if the redundant modules are able to replace the modules that have been detected as faulty*

Redundancy is used in order to reconfigure the system: totally or partially.

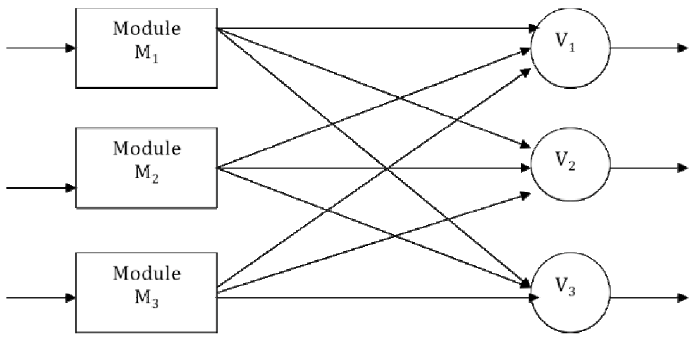
A simple example of using redundancy in order to ensure protection to: shortcircuit, interruption, shortcircuit and interruption, respectively:



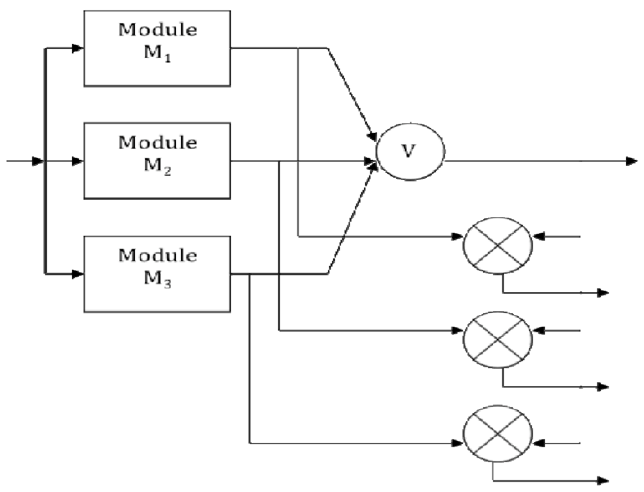
10. Principles for the detection of the faulty module in redundant electronic structures

Course 11, slides 8-10

The fault tolerant systems are implemented based on the use of majority logic redundant structures. The are using a multiple voter configuration:



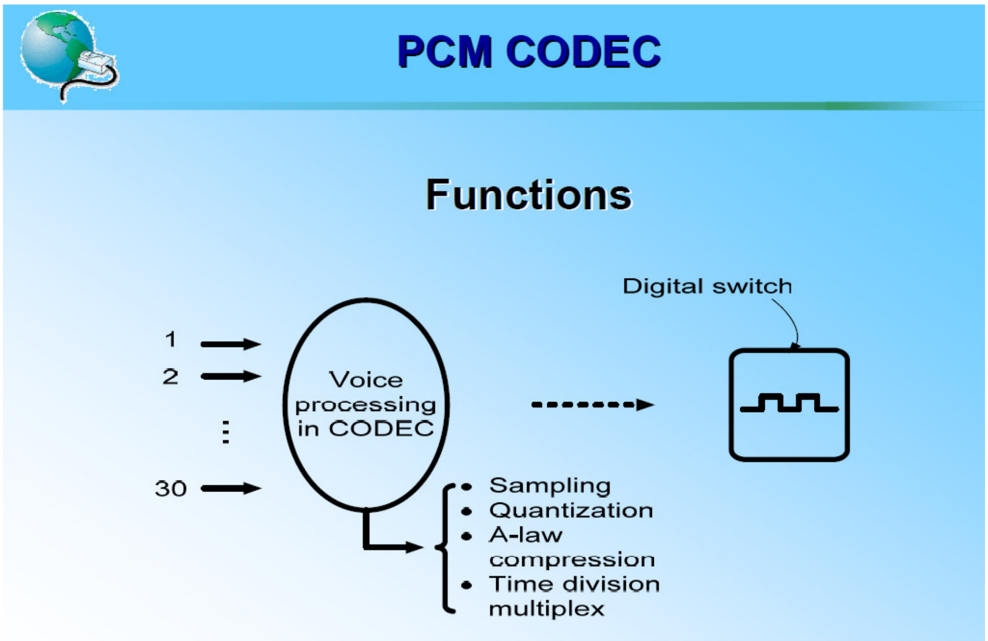
In order to avoid that the error, once appeared in a module, be propagated into the system, the faulty module should be detected and replaced.

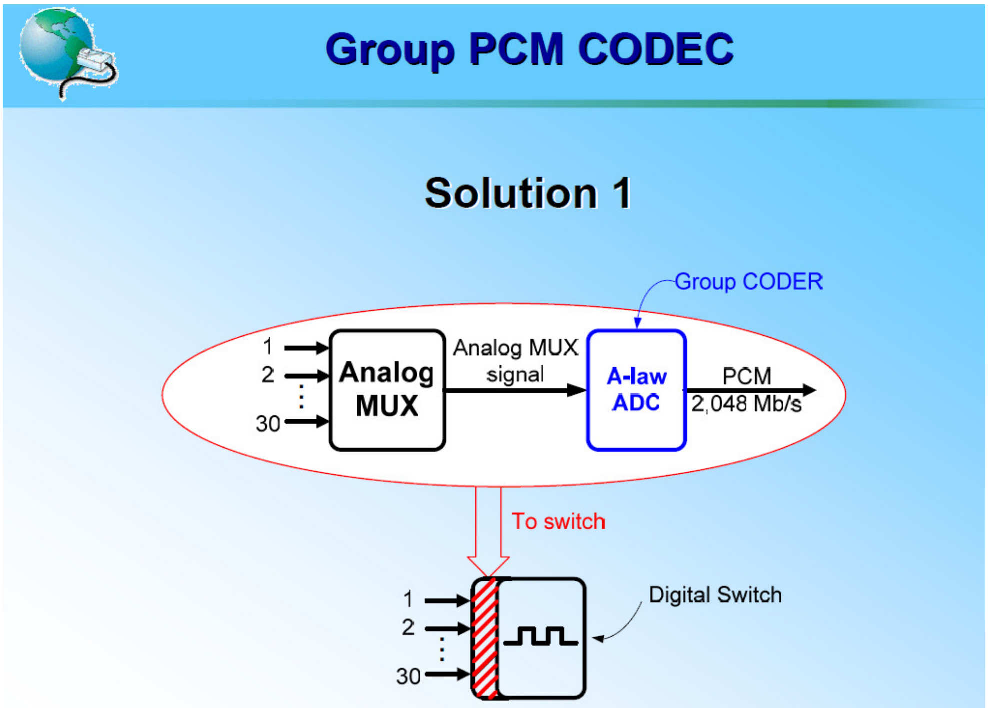


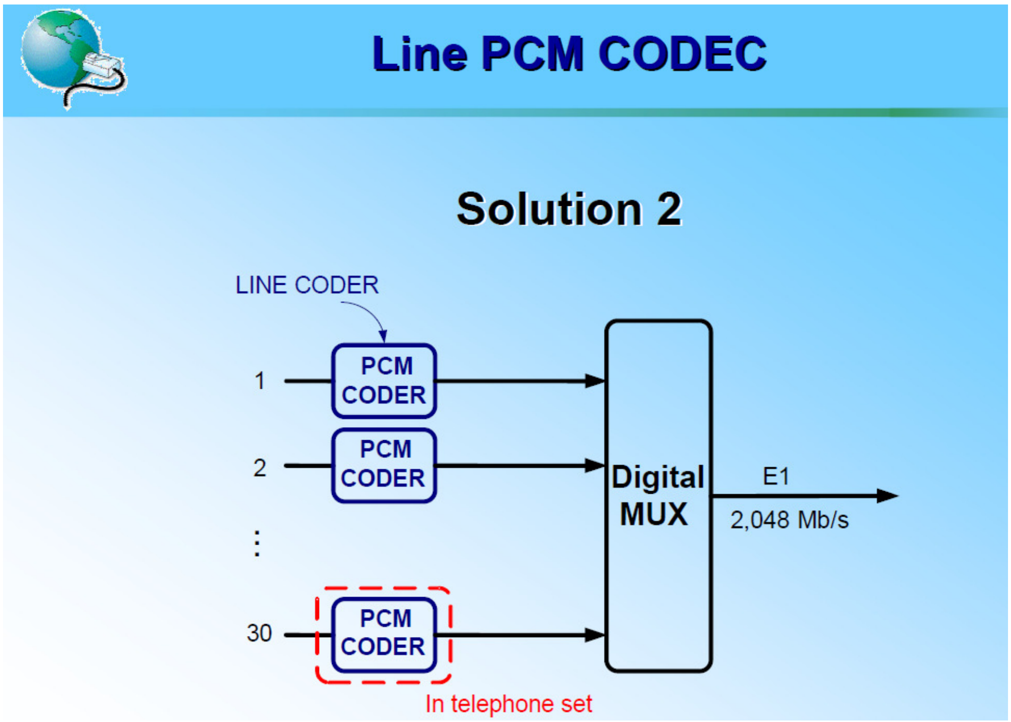
Integrated Digital Networks

1. Compare the PCM codec solutions.

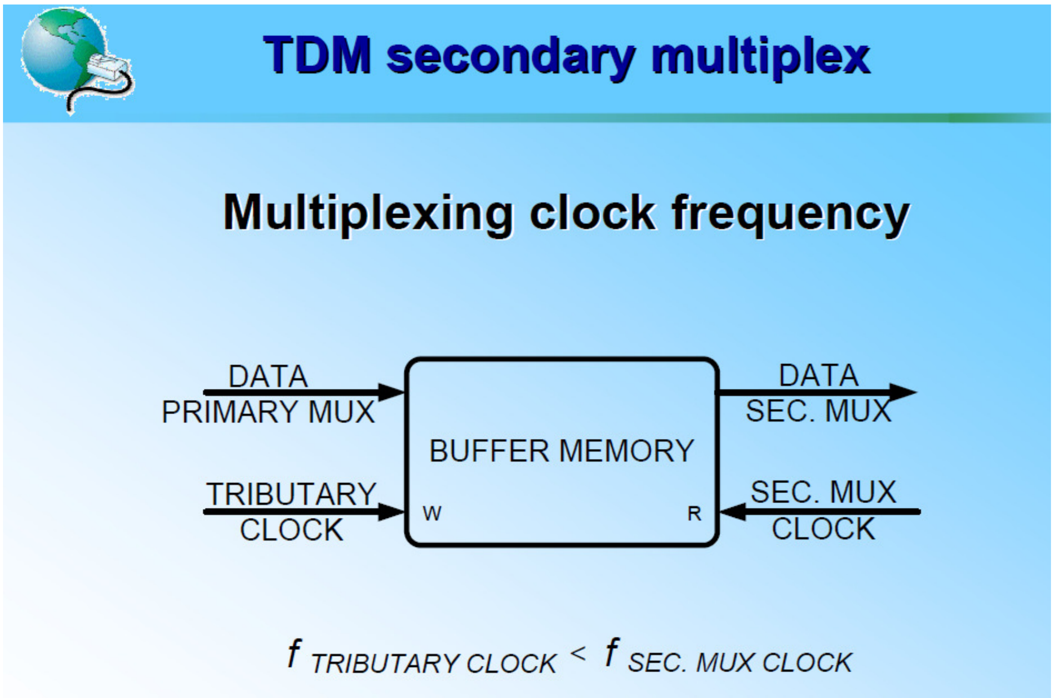
https://intranet.etc.upt.ro/~DIG_INT_NET/course/2_Primary_PCM_CODEC.pdf, 1, 2, 4



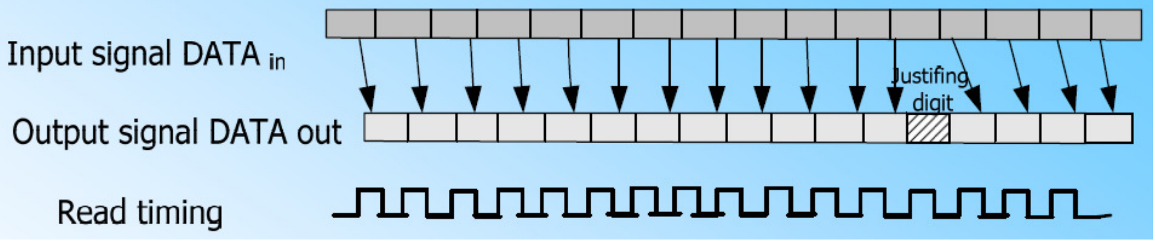




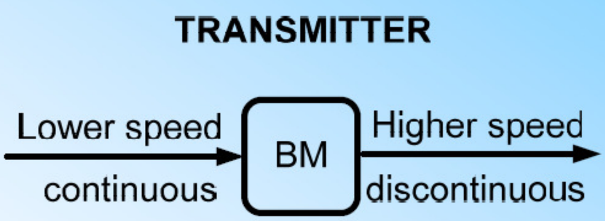
2. **Principle of positive justification**.
https://intranet.etc.upt.ro/~DIG_INT_NET/course/3_Secondary_Digital_TDM.pdf, 5, 7, 9



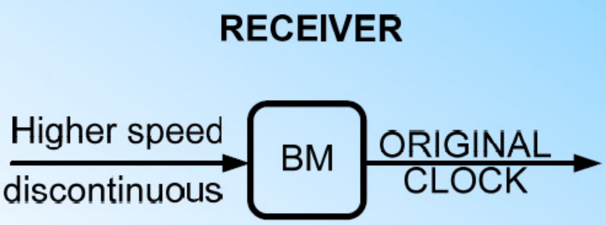
Principle of Positive Justification (III)



Principle of Positive Justification (V)

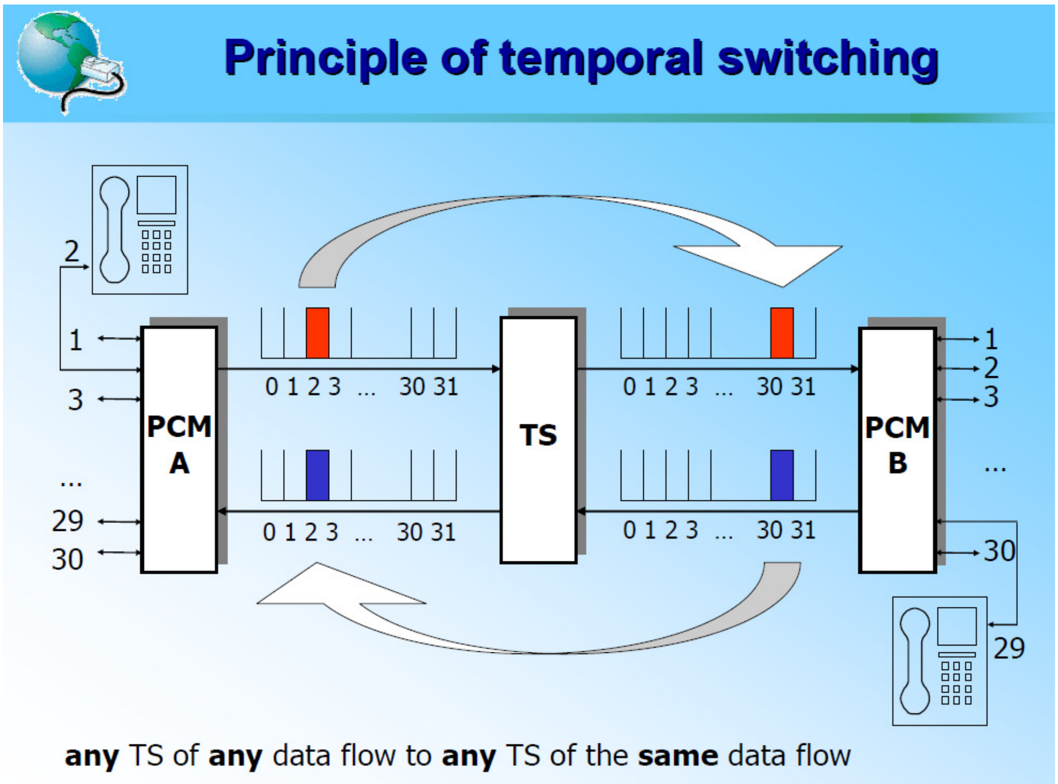


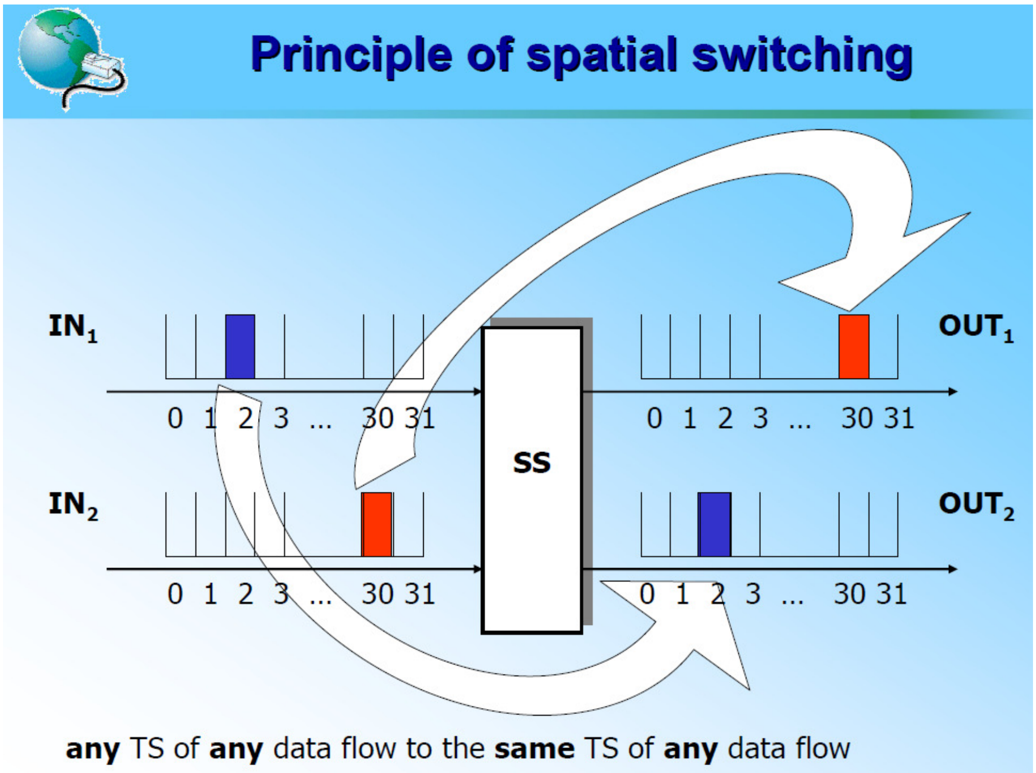
BM = Buffer Memory



3. Digital switching – definition, the principle of the temporal switch, the principle of the spatial switch

https://intranet.etc.upt.ro/~DIG_INT_NET/course/4_Digital_switching.pdf, 1, 3, 8-10





Digital switch

DIGITAL SWITCH

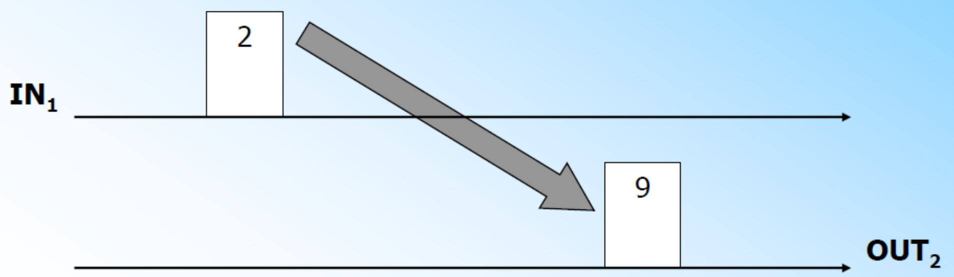
• **any** TS of **any** INPUT (data flow) to **any** TS of **any** OUTPUT (data flow)

TEMPORAL SWITCH

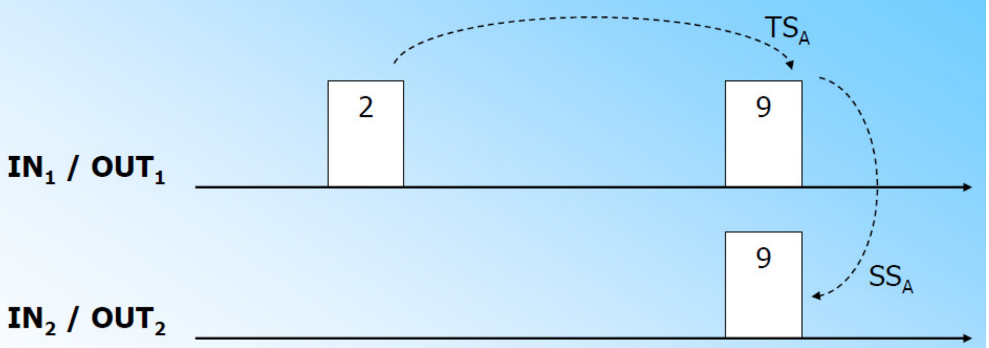
• **any** TS of **any** data flow to **any** TS of the **same** data flow

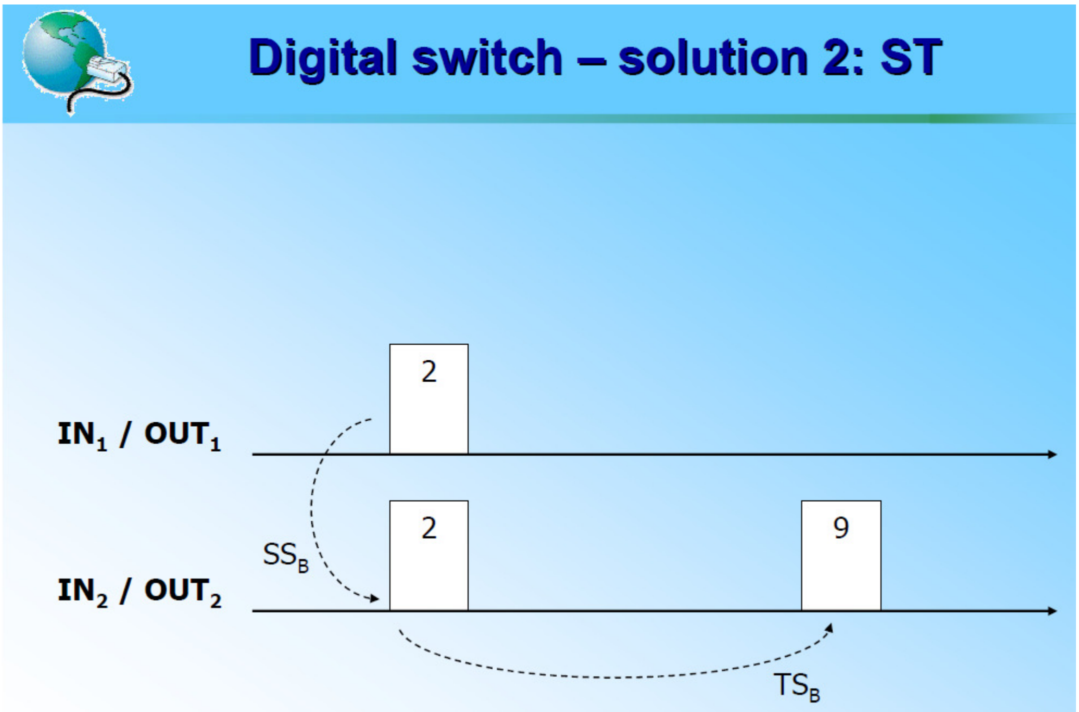
SPATIAL SWITCH

• **any** TS of **any** data flow to the **same** TS of **any** data flow



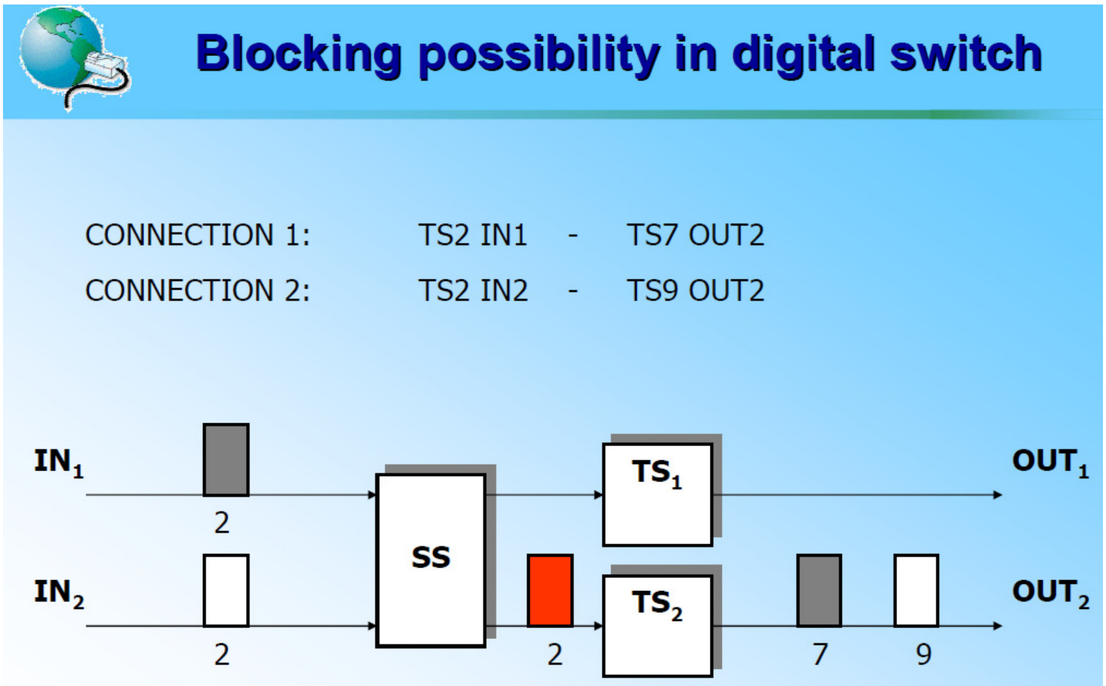
Digital switch – solution 1: TS

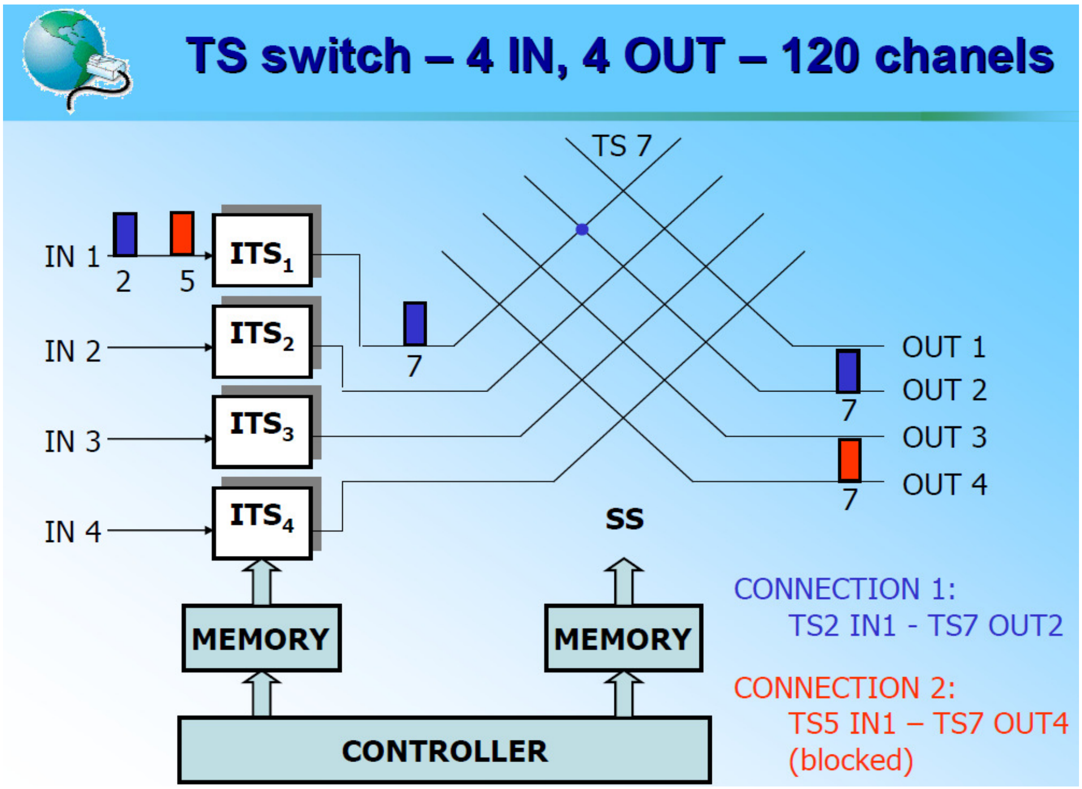


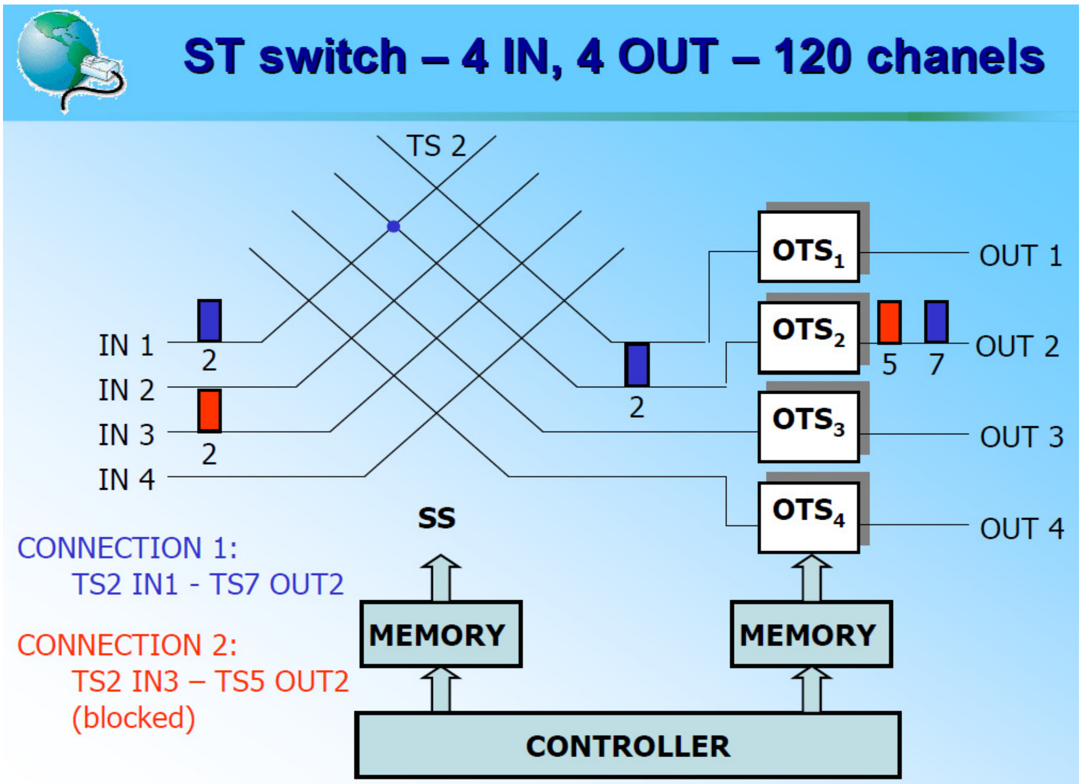


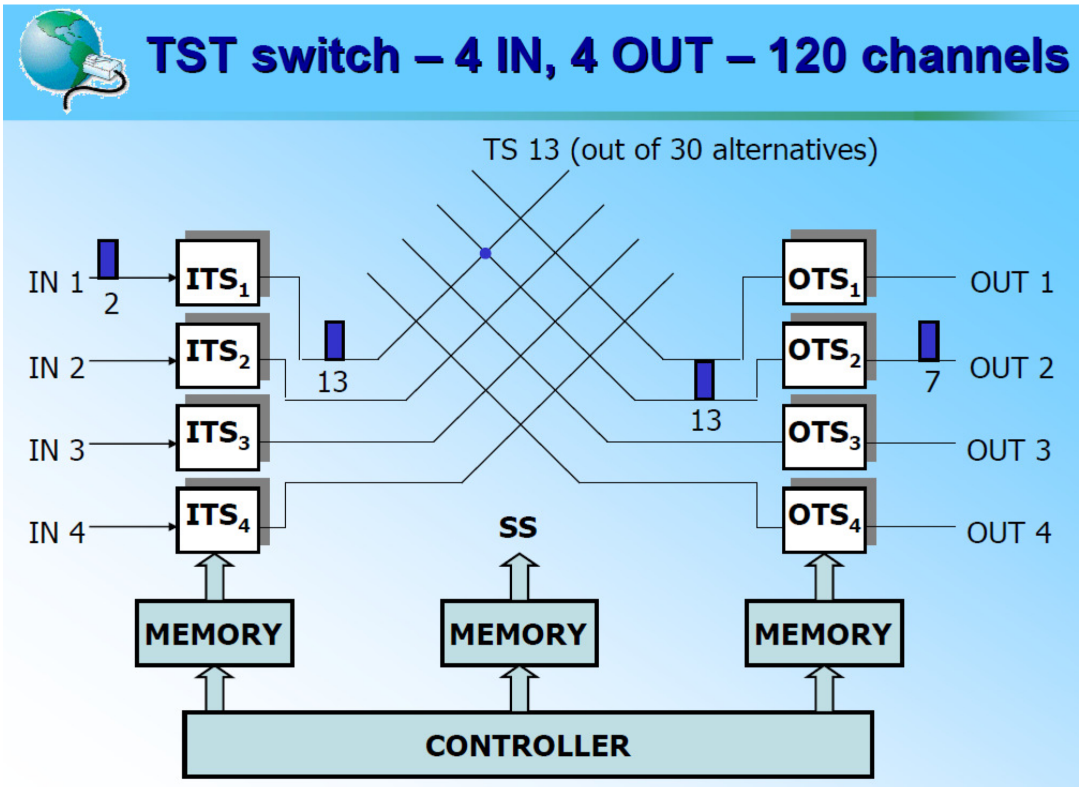
4. Digital switch blocking – definition and example of one switch with blocking and one switch without blocking

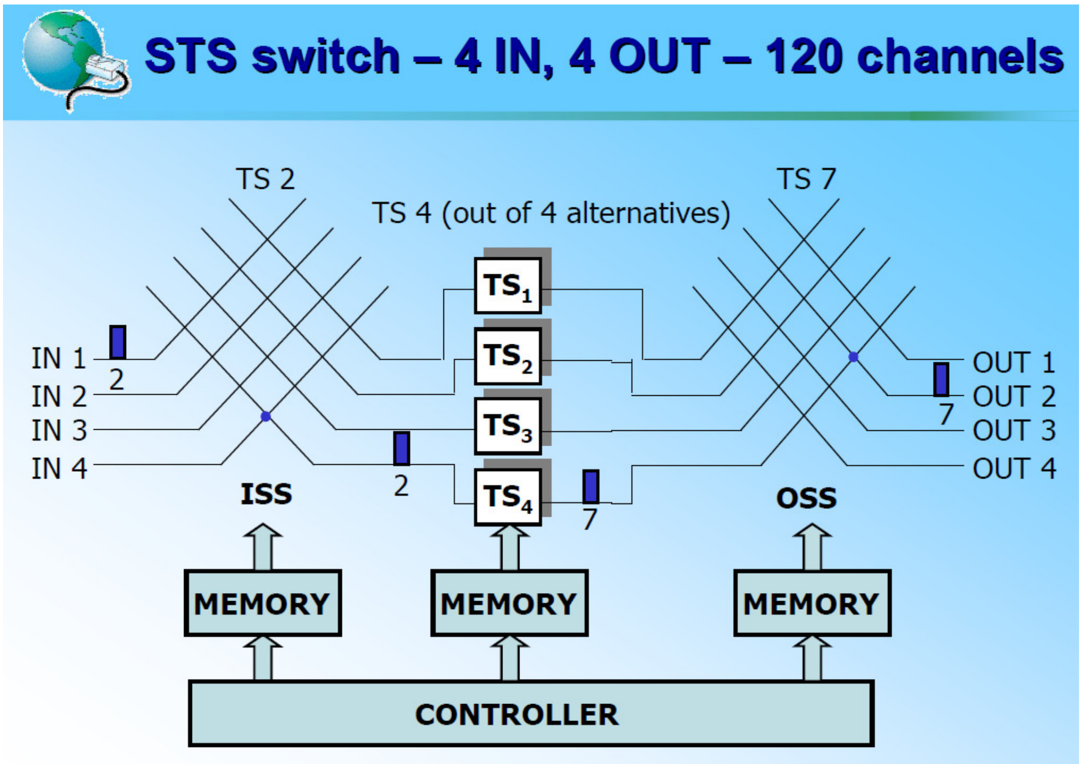
https://intranet.etc.upt.ro/~DIG_INT_NET/course/4_Digital_switching.pdf, 11-16

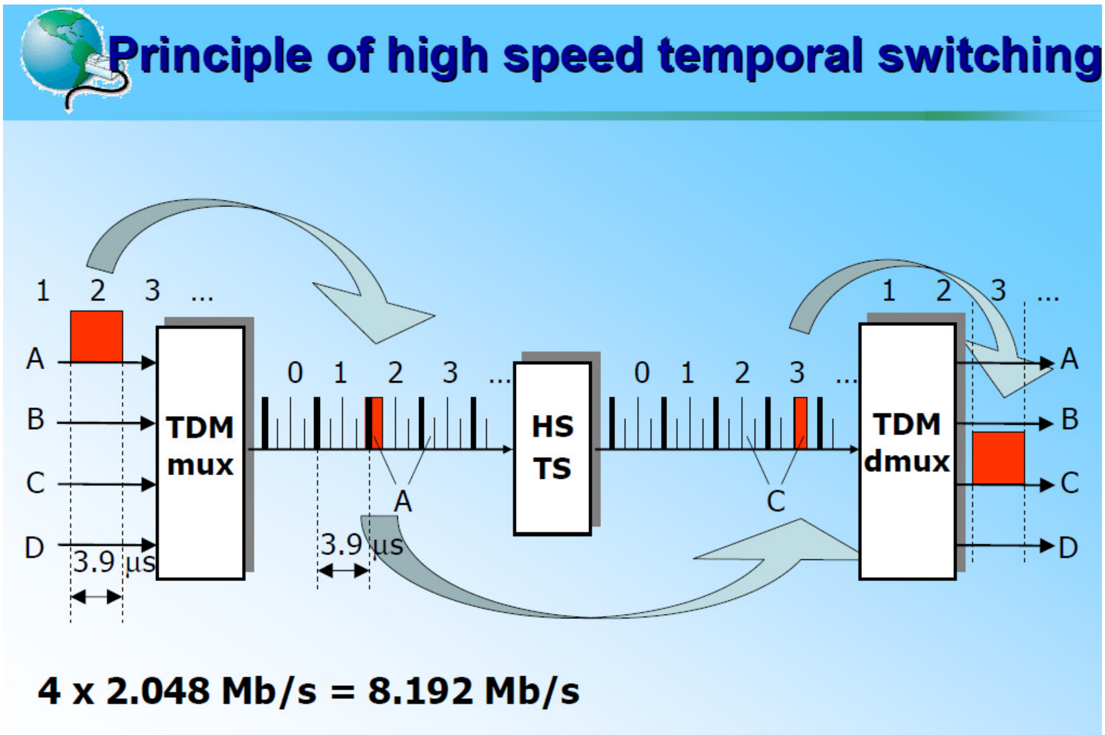



TS switch – 4 IN, 4 OUT – 120 chanels
TS 7
IN 1
2
5
IN 2
IN 3
IN 4
ITS1
ITS2
ITS3
ITS4
7
OUT 1
OUT 2
OUT 3
OUT 4
7
7
SS
MEMORY
MEMORY
CONTROLLER
CONNECTION 1:
TS2 IN1 - TS7 OUT2
CONNECTION 2:
TS5 IN1 – TS7 OUT4
(blocked)


ST switch – 4 IN, 4 OUT – 120 chanels
TS 2
IN 1
2
IN 2
IN 3
2
IN 4
2
OTS1
OTS2
OTS3
OTS4
OUT 1
5
7
OUT 2
OUT 3
OUT 4
SS
MEMORY
MEMORY
CONTROLLER
CONNECTION 1:
TS2 IN1 - TS7 OUT2
CONNECTION 2:
TS2 IN3 – TS5 OUT2
(blocked)


TST switch – 4 IN, 4 OUT – 120 channels
TS 13 (out of 30 alternatives)
IN 1
IN 2
IN 3
IN 4
2
13
13
7
ITS1
ITS2
ITS3
ITS4
OTS1
OTS2
OTS3
OTS4
OUT 1
OUT 2
OUT 3
OUT 4
SS
MEMORY
MEMORY
MEMORY
CONTROLLER


STS switch – 4 IN, 4 OUT – 120 channels
TS 2
TS 4 (out of 4 alternatives)
TS 7
IN 1
IN 2
IN 3
IN 4
2
2
7
7
TS1
TS2
TS3
TS4
OUT 1
OUT 2
OUT 3
OUT 4
ISS
OSS
MEMORY
MEMORY
MEMORY
CONTROLLER



5. Modulation techniques in DSL – list the techniques with full name and explain the principle of one of them

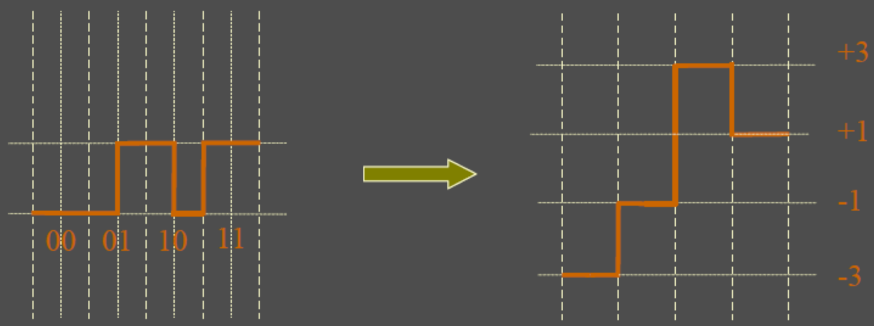
https://intranet.etc.upt.ro/~DIG_INT_NET/course/5_Digital_subscriber_line_xDSL.pdf, 6-14

The most common DSL technologies are: IDSL, HDSL, SDSL, SHDSL, HDSL2, HDSL4, ADSL, RADSL, ADSL.lite, ADSL2, ADSL2+ and VDSL

The emergence of DSL technologies allowed the development of new solutions for the redesign of the subscriber loop

The modulations used in DSL are QAM, 2B1Q, DMT and CAP

❷ DSL Basics
2B1Q

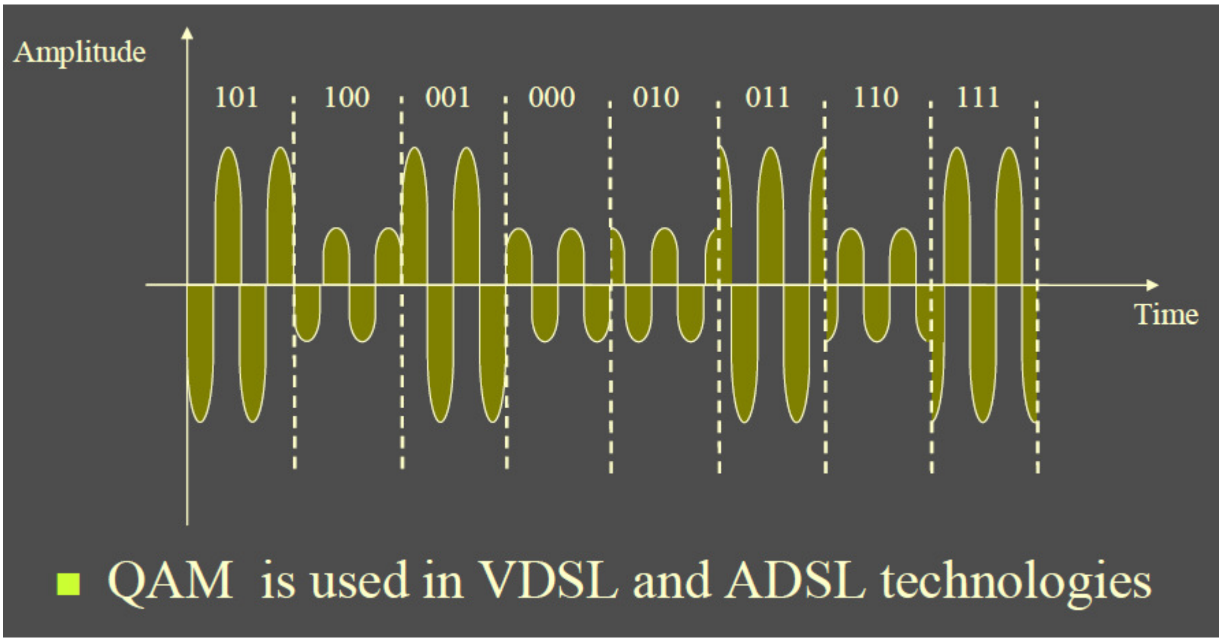


- The 2B1Q (*Two-Binary One-Quaternary*) line code was introduced together with the ISDN standardization, nowadays being used in HDSL, IDSL and SDSL technologies.

QAM:

- QAM (*Quadrature Amplitude Modulation)* is a hybrid modulation
- The signal resulted at the output of the QAM modulator has the following expression:

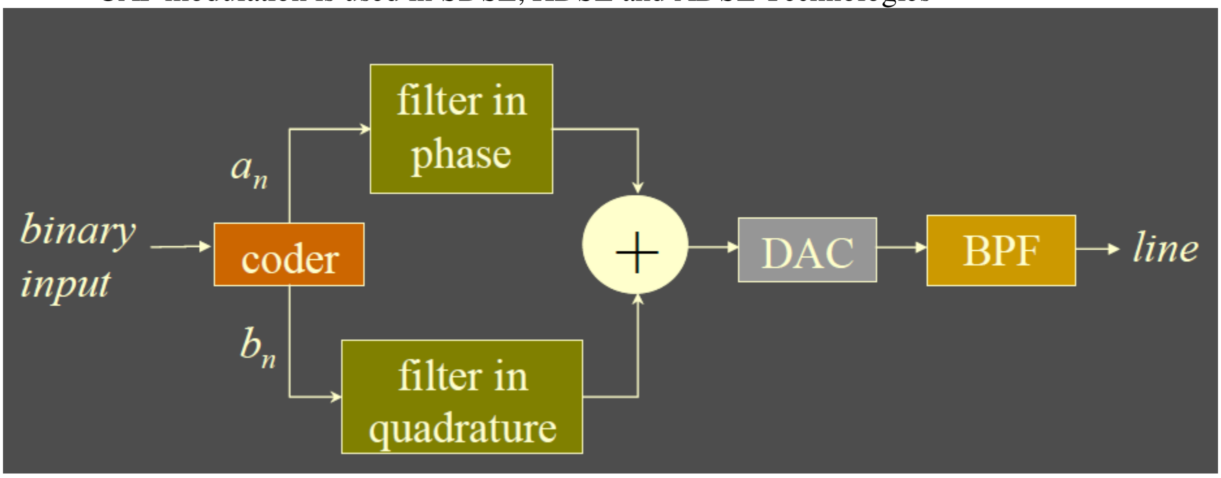
$$s(t) = A\sqrt{\frac{2}{E_g}}g_T(t)\cos(2\pi f_c t + \theta)$$



CAP:

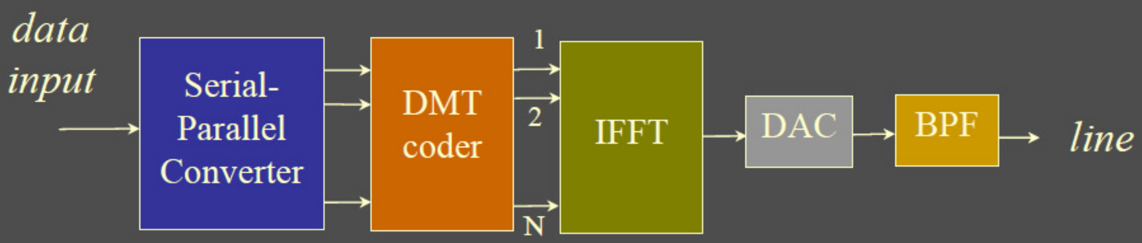
CAP (*Carrierless Amplitude-modulation and Phasemodulation*) is a modulation technique derived from QAM, being, however, more easy to implement

CAP modulation is used in SDSL, HDSL and ADSL Technologies

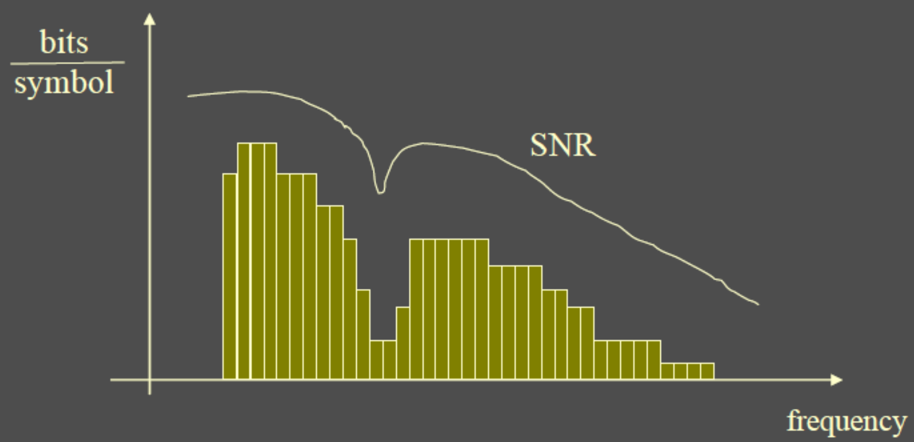


DMT:

- DMT (*Discrete multi-tone modulation*) enables a subchannel division in a single modem



- DMT is a flexible modulation, mainly because of the bits distribution in the subchannel symbols



- According to ANSI, the DMT modulation can theoretically contain up to 255 subchannels centered on m·f frequencies where m = 1 ÷ 255 and f = 4.3125 kHz.
- DMT modulation is used in VDSL and ADSL technologies

6. **ADSL – purpose, characteristics, basic schematic, frequency spectrum**

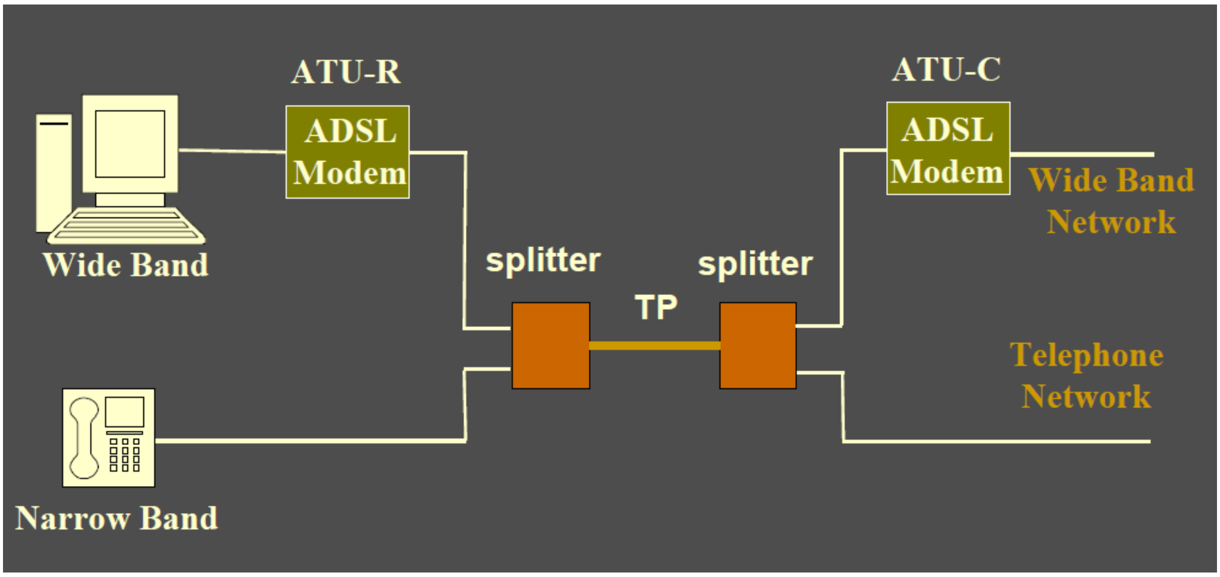
https://intranet.etc.upt.ro/~DIG_INT_NET/course/5_Digital_subscriber_line_xDSL.pdf, 35, 36, 38

Asymmetric Digital Subscriber Line

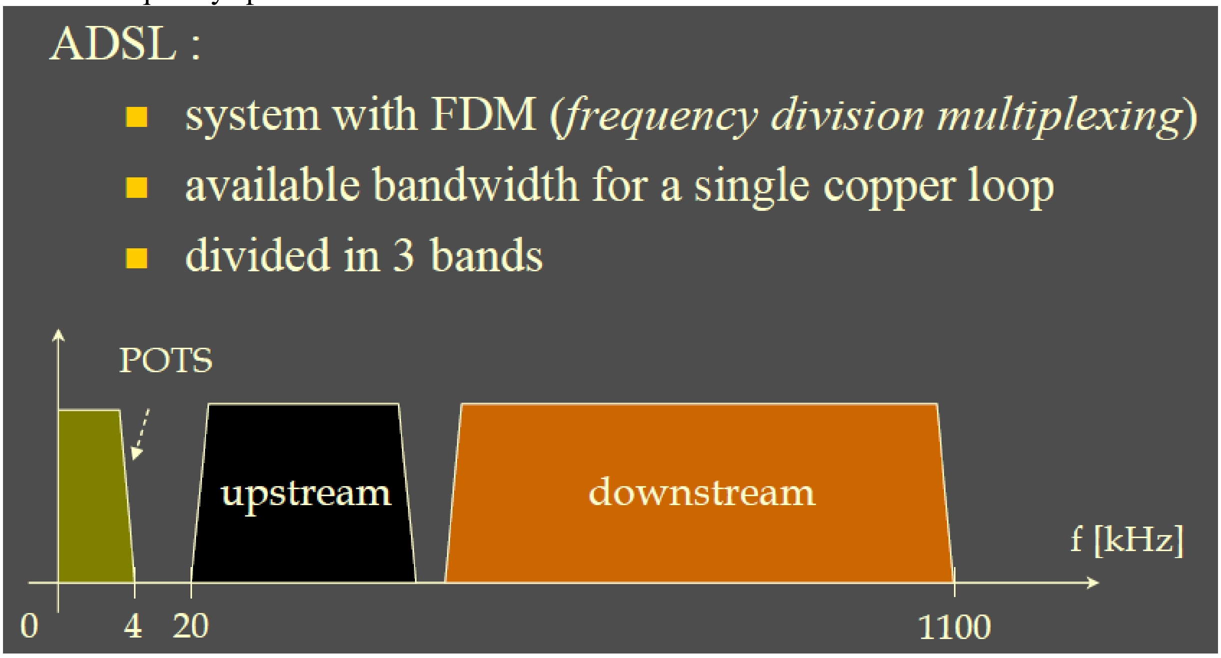
It represents a standard for digital communications, whose purpose is the high rate transmission of data over the telephone lines

This technology was developed in 1989 at Bellcore Labs in order to provide video capabilities over standard telephone lines

Basic schematic:



ADSL: frequency spectrum

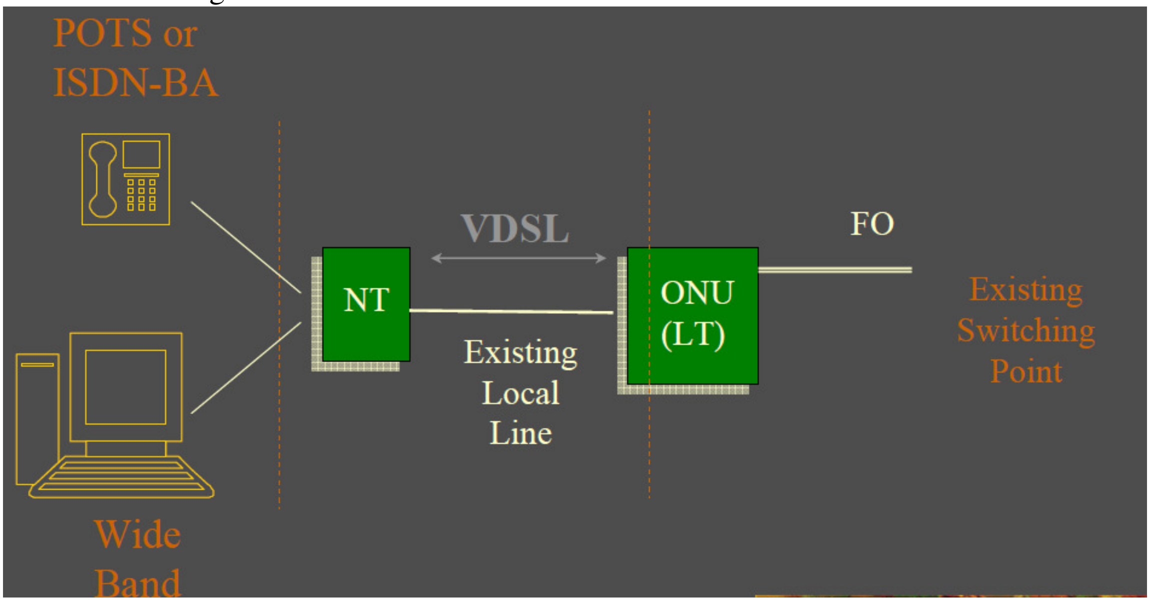


7. VDSL – characteristics, the reference model for the interface, frequency spectrum and data rates

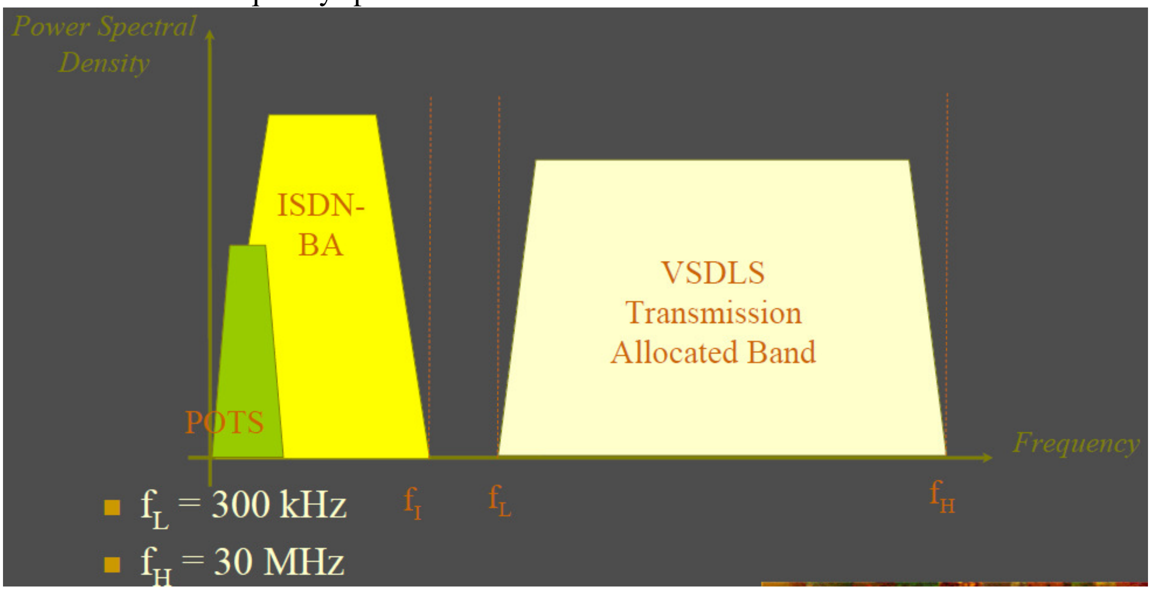
https://intranet.etc.upt.ro/~DIG_INT_NET/course/5_Digital_subscriber_line_xDSL.pdf, 54, 56, 60, 62

Very high bit rate Digital Subscriber Line
_ the DSL technology with the highest transmission rate
_ data rates _ symmetric
_ asymmetric
_ up to tens of Mbps

VDSL – general architecture



VDSL – frequency spectrum



VDSL – bit rates

	Class of Operation	Downstream (kbps)	Upstream (kbps)
ASYM	Class I (A4)	362 x 64 = 23 168	64 x 64 = 4 096
	Class I (A3)	226 x 64 = 14 464	48 x 64 = 3 072
	Class I (A2)	134 x 64 = 8 576	32 x 64 = 2 048
	Class I (A1)	100 x 64 = 6 400	32 x 64 = 2 048
SYM	Class II (S5)	442 x 64 = 28 288	442 x 64 = 28 288
	Class II (S4)	362 x 64 = 23 168	362 x 64 = 23 168
	Class II (S3)	226 x 64 = 14 464	226 x 64 = 14 464
	Class II (S2)	134 x 64 = 8 576	134 x 64 = 8 576
	Class II (S1)	100 x 64 = 6 400	100 x 64 = 6 400

8. Enumerate and explain the ISDN conditions

https://intranet.etc.upt.ro/~DIG_INT_NET/course/6_ISDN.pdf, 1, 4, 6, 7

I.S.D.N. – Integrated Services Digital Network
CONDITIONS

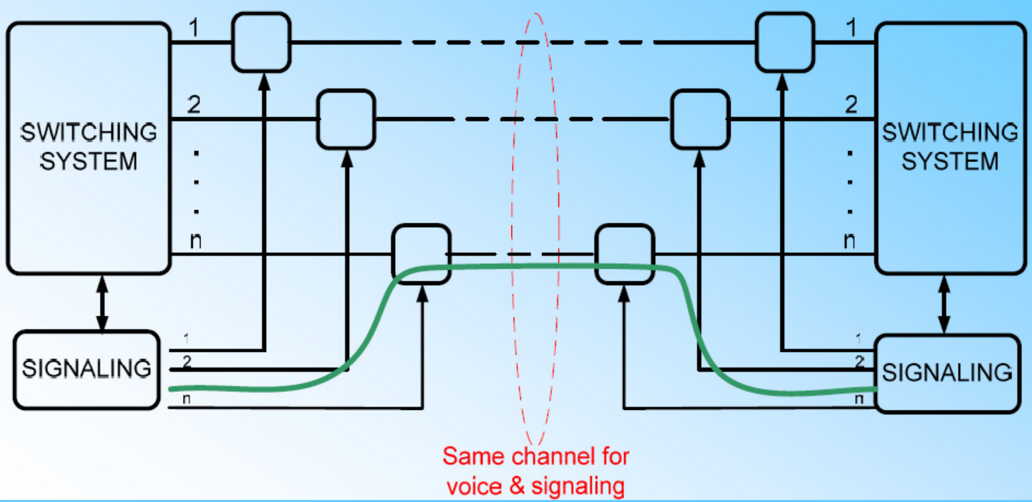
1. Digital connections from **terminal to terminal**
- digital signal
- digital subscriber line - DSL
- digital transmission
- digital switching

2. Common-channel signaling

3. Multiple network **access** for a wide range of voice and non-voice applications

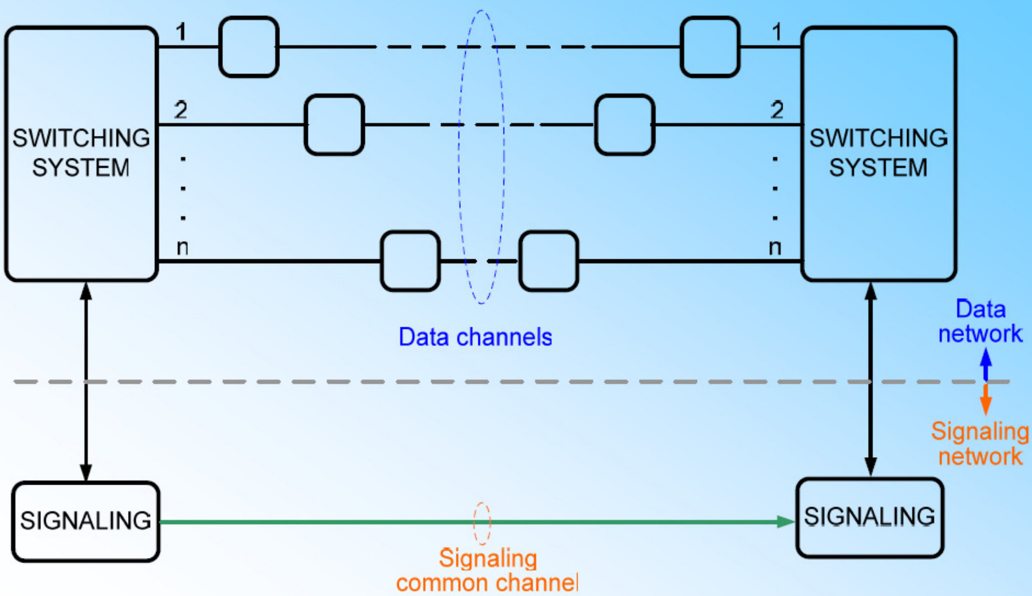
Common channel signaling (II)

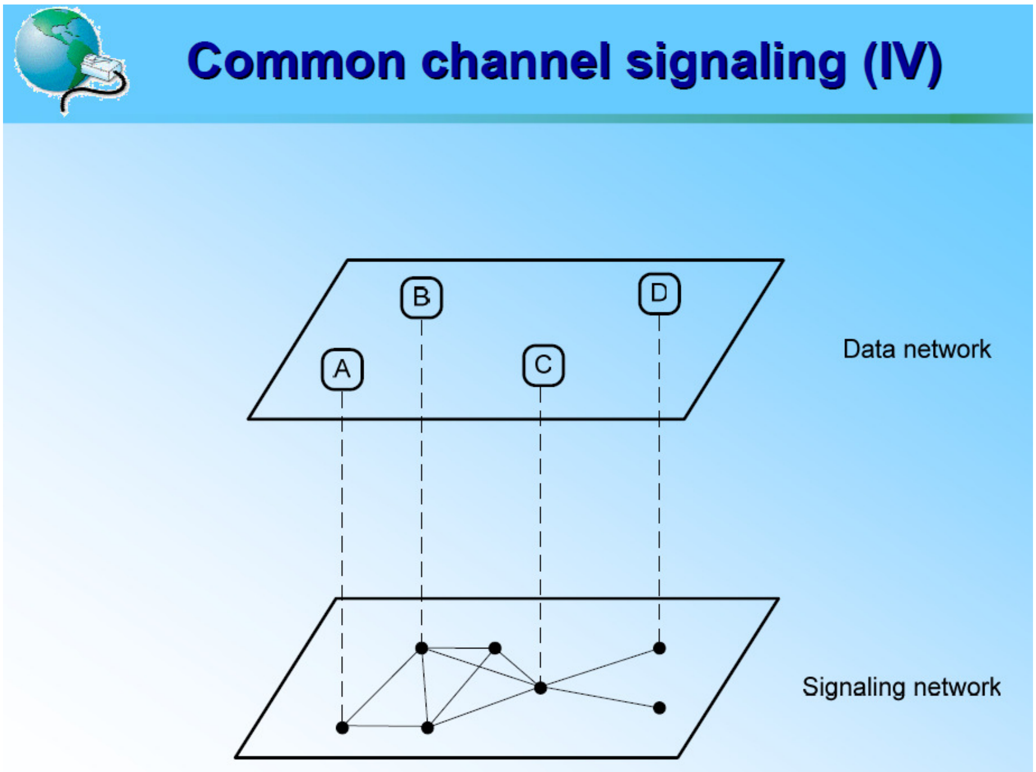
Traditional (POTS)



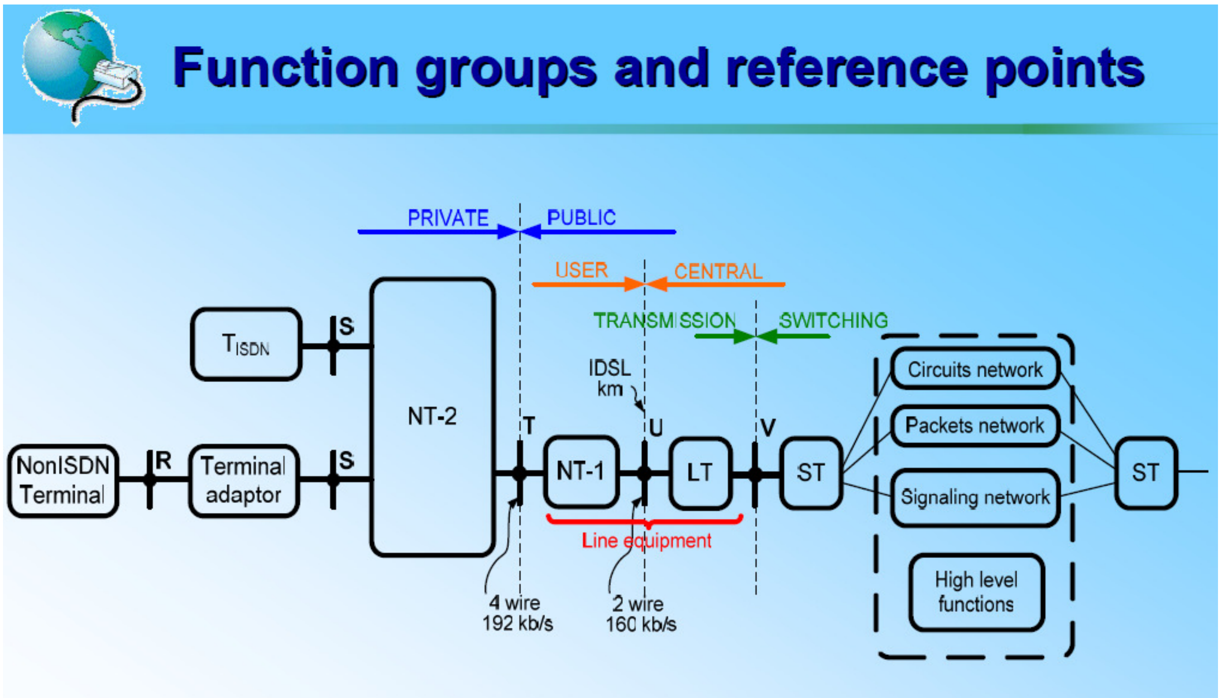
Common channel signaling (III)

Solution for intelligent network (ISDN)



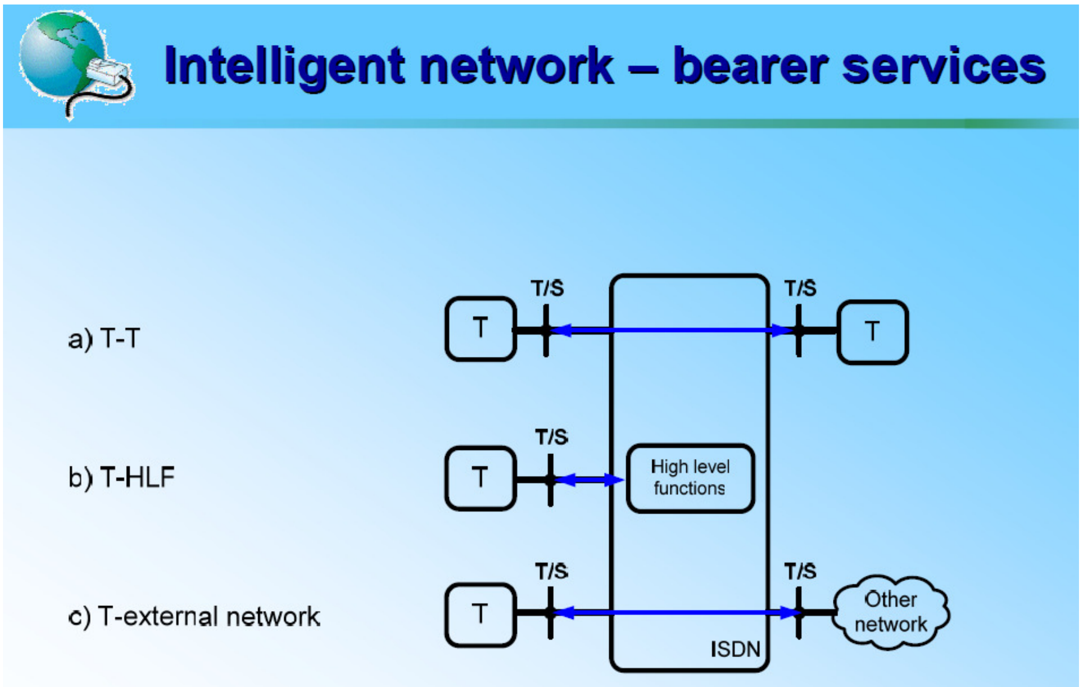


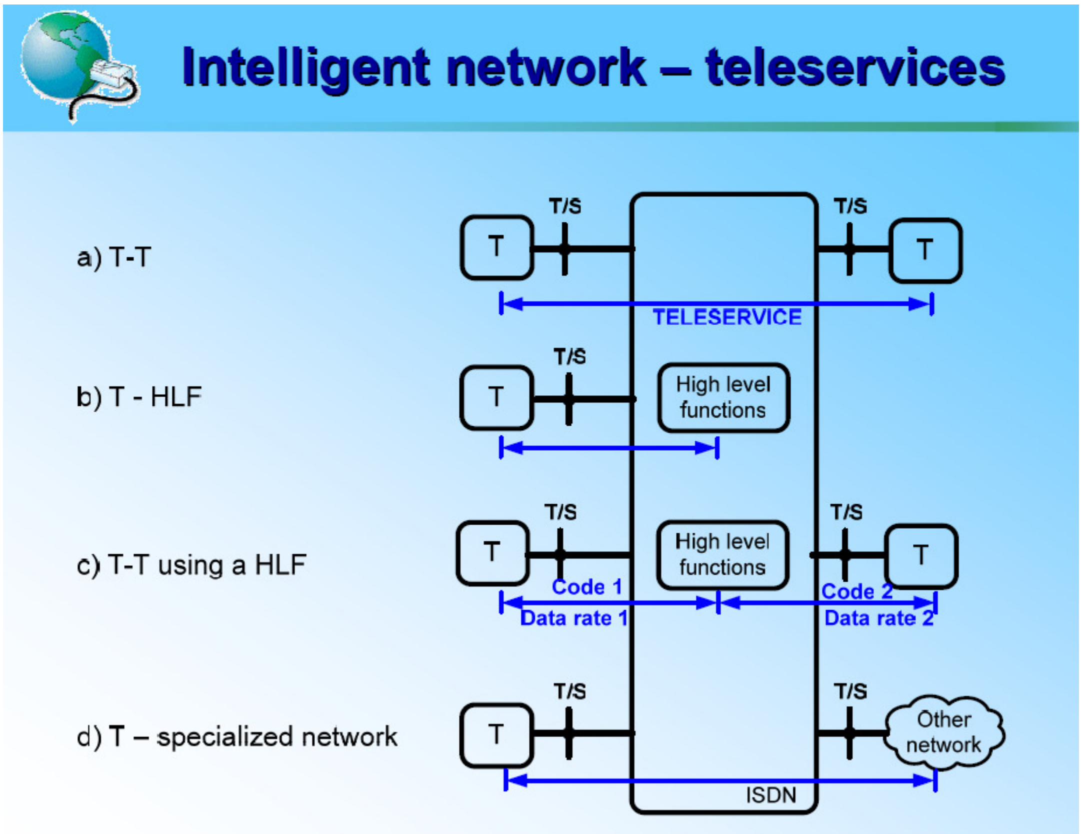
9. Function groups and reference points – definitions, significances and domain limits.
https://intranet.etc.upt.ro/~DIG_INT_NET/course/6_ISDN.pdf, 10



10. Compare the bearer services with teleservices

https://intranet.etc.upt.ro/~DIG_INT_NET/course/6_ISDN.pdf, 11, 12





Data Communications

1. Write down the formula describing the Shannon capacity theorem for the noisy channels and explain the meaning of each parameter.

A: Course 2, slide 20.

THEOREM 2: *Let P be the average transmitter power, and suppose the noise is white thermal noise of power N in the band W. By sufficiently complicated encoding systems it is possible to transmit binary digits at a rate*

$$C = W \log_2 \frac{P+N}{N} \qquad (19)$$

with as small a frequency of errors as desired. It is not possible by any encoding method to send at a higher rate and have an arbitrarily low frequency of errors.

- This is the original Shannon formulation
- The widely used form is:

$$C = W \log_2 \left(1 + \frac{S}{N}\right)$$

- For high values of SNR

$$C = \frac{W}{3} \frac{S}{N} [dB]$$

Signal to noise ratio

2. Give the expression of the power spectral density of a thermal noise, indicating the meaning of each parameter.

A: Course 2, slide 11.

- Thermal noise
  - Generated by the thermal agitation of electrons
  - Uniformly distributed in frequency
  - Generally modeled as white noise
- The amount of thermal noise in 1Hz

$$N_0 = kT \qquad k = 1.3803 \cdot 10^{-23} \quad [J/^{o}K]$$

N_0 is the power spectral density [Watts/Hz]

- The amount of thermal noise in W Hz

$$N = kTW$$

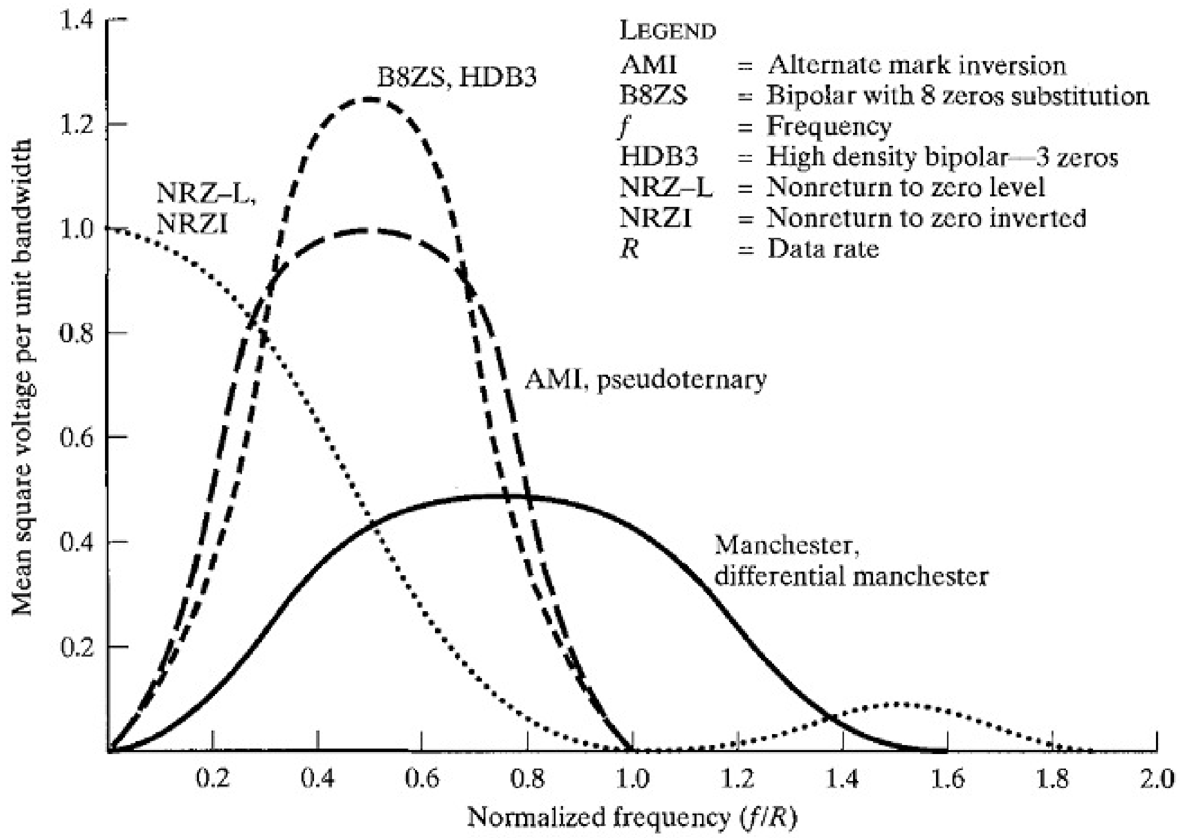
The thermal noise is present in all electronic circuits and transmission media. Generally, it is mathematically described as a “white noise”. i.e. a noise whose power is uniformly distributed over the whole frequency bandwidth. This is the most common type of noise considered when modeling systems, and there is a huge amount of literature (information theory, coding, data transmission) built on the hypothesis that the noise that affects the signal is a white noise.

The power spectral density (PSD) of a white noise is constant and equals $N_0/2$ (if the representation considers the negative part of the frequency axis too), where N_0 depends on the temperature.

Notice that, larger the bandwidth, larger will be the amount of thermal noise “seen” by the receiver. We may therefore say that larger bandwidth transmissions are more affected by the thermal noise, compared to the narrower bandwidth transmissions.

3. Encode the bit sequence {0 1 0 0 1 1} using the following digital encoding techniques (NRZ-L, Binary AMI, Manchester). Draw the signal so obtained (Voltage levels are +A and -A).

A: Course 3, slide 26.



4. Indicate one advantage and one disadvantage of Manchester codes versus NRZ codes.

A: Course 3, slide 19.

Biphase codes pros and cons

- Pros
  - Predictable transition at the middle of each bit, which can be used for synchronization ("self-clocking" codes)
  - Error detection possible due to the same feature
  - No DC component
- Cons
  - Bandwidth efficiency issues (half the bandwidth efficiency of NRZ)
  - Modulation rate can be two times the data rate

Clearly, Manchester encoded signal has a larger number of transitions than NRZ or AMI signals, thus line synchronization is easier. But, in the same time, the larger number of transitions leads to high frequency components of important energy, meaning that the spectral efficiency of the Manchester encoded signal is lower than in the above mentioned cases (notice the wider PSD curve from slide 26). Practically, Manchester coding is a good choice in those applications where the couple transmission throughput – physical available bandwidth guaranteed that the signal is not distorted by the transmission environment.

Example: in the 10Mbps Ethernet (let'it be on coaxial or twisted pairs cable), the bandwidth of the physical medium is large enough such as to allow Manchester coding. By the other hand, this bandwidth becomes insufficient in 100Mbps Fast Ethernet, where a different strategy is adopted.

5. Give the expression of the Nyquist rate for ISI-less transmission, indicating the meaning of each term.

Course 4, slide 12.

Ideal case: a closer look

- The impulse response of the ideal low-pass filter is:

$$g(t) = \frac{\sin(\Omega t)}{\pi t} = \frac{\sin\frac{2\pi}{T_0}t}{\pi t} \quad (11)$$

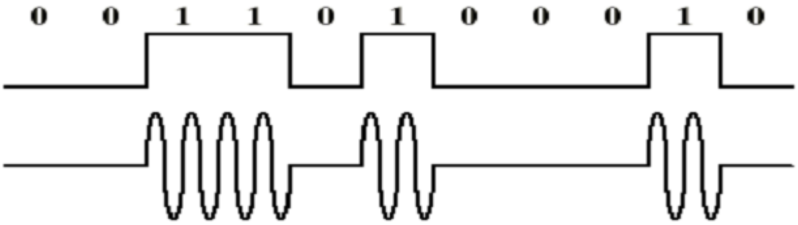
- Such a waveform crosses zero every $T_0/2$ seconds
- In frequency, this corresponds to an ideal low-pass (brick-wall) filter, with the cut-off frequency $F=1/T_0$
- If a symbol is issued every $T_0/2$ seconds, transmission can be made without ISI
- The rate in this case will be 2F symbols/s (Nyquist rate, ideal rate)

Although, unlike in the digital baseband transmission (e.g.:NRZ), a single information symbol is represented by a waveform with infinite duration (a cardinal sine), we can transmit the next information symbol at the first zero crossing of the sinc waveform (i.e.: at $T_0/2$); this will prevent ISI to occur and will allow a transmission at a rate of $2/T_0$ symbols/sec. The later statement is identical with Nyquist's theorem, formulated on the previous slide.

6. Give one graphical example illustrating a signal obtained by digital AM.

A: Course 5, page 3, first slide.

- AM is referred to as linear modulation
- The amplitude of the carrier is changed by the signal to be transmitted
- When the modulating signal is digital, the Amplitude Shift Keying (ASK) case is obtained
- The simplest form of ASK is called On/Off Keying (OOK)



(a) Amplitude-shift keying

7. What do the following terms mean: DSB-AM, SSB-AM, VSB-AM?

A: Course 5, pp. 5-7.

Suppressing…and un-suppressing the carrier

- If the carrier has a DC component:

$$c(t) = c_0 + \cos(\omega_0 t + \varphi_0) \quad (2)$$

- The modulated signal is:

$$s_a(t) = (s(t) + c_0)\cos(\omega_0 t + \phi_0) \quad (3)$$

- In the spectrum, we will retrieve the Dirac pulses corresponding to the sine carrier

$$S(\omega) = H(\omega)\left\{\pi c_0[\delta(\omega+\omega_0)+\delta(\omega-\omega_0)] + \left[\frac{1}{2}G(\omega+\omega_0)+\frac{1}{2}G(\omega-\omega_0)\right]\right\} \quad (4)$$

Remarks

- If c_0=0, then we have AM with suppressed carrier
  - The figures from the next slide correspond to the above case
- The later is also called product modulation, because it consists on a simple multiplication
- All the information of the AM signal is carried by EACH side-band
- From effectiveness reasons, one side-band can be suppressed

Graphical illustration

- The filter selects the lower (inferior) or the upper (superior) side-band, leading to Single Side Band (SSB) AM

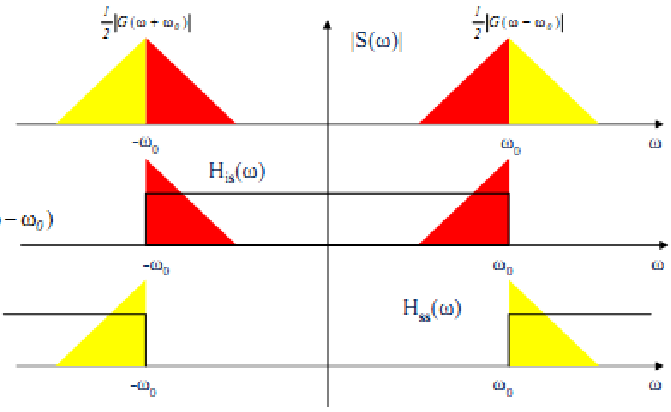
a. Double side-band AM

b. Inferior SB AM

$H_{IS}(\omega) = \sigma(\omega + \omega_0) - \sigma(\omega - \omega_0)$

c. Superior SB AM

$H_{ss}(\omega) = 1 - H_{IS}(\omega)$



SSB-AM

- SSB signal can be expressed as:

$$s_{si}(t) = \frac{1}{2} g(t)\cos(\omega_0 t) + \frac{1}{2} \hbar\{g(t)\}\sin(\omega_0 t) \quad (5)$$

- $\hbar\{g(t)\}$ is the Hilbert transform of g, which can be obtain by passing g(t) through the filter:

$$F(\omega) = -j\,\mathrm{sgn}(\omega) \quad (6)$$

- SSB-AM is spectrally efficient
- Difficult to implement in practice: the filters which separate the side-band must be very selective

AM with Vestigial Side-Band (VSB)

- Only part of the side-bands is suppressed
- Lower frequencies transmitted with both Side-Bands, upper frequencies with one side-band
- This allows easier filtering to separate the bands (frequencies near the carrier must not be filtered)
- 25% more bandwidth required than in SSB, but easier to implement
- Example: NTSC TV system: all upper sideband of bandwidth W2 = 4 MHz, but only W1 = 1.25 MHz of the lower sideband are transmitted

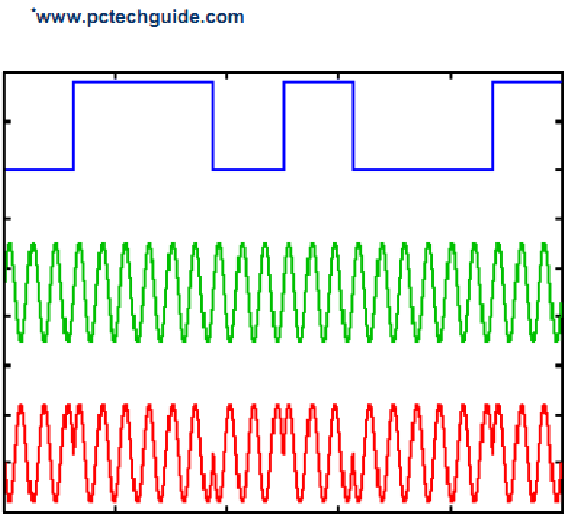
AM signals demodulation

- AM modulation with suppressed carrier is discussed
- Two types of detection for AM: coherent and non-coherent
- Even if simpler, the non-coherent detection is very sensitive to noise
- Coherent detection is oftentimes preferred

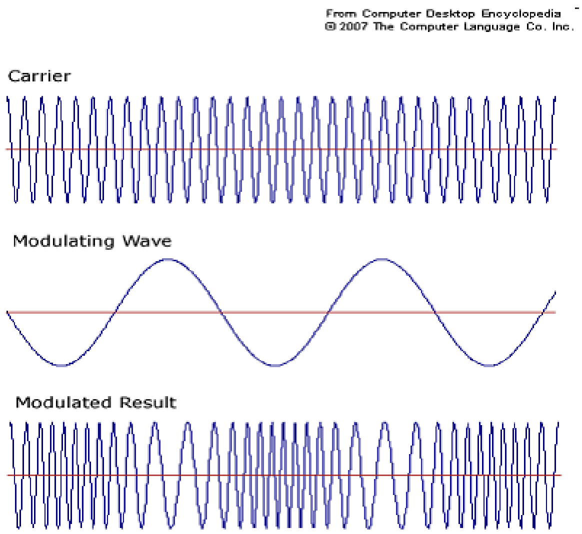
8. Explain the meaning of the term PSK and represent graphically a PSK signal.

A: Course 7, slides 4-5.

- Definition: PM is a method used to transmit analog or digital signals, in which the information is carried by the initial phase of a high-frequency carrier
- PM is not so popular as FM, especially for analog signal modulators (because of its complexity)
- When the modulator is digital, the frequency modulation is referred to as phase shift keying (PSK)



PSK



Analog PM

9. What are the four channel types in USB and which of them has guaranteed bandwidth?

A: Course 10, slide 22.

- 4 types of “data channels” for the communication between the device and the USB host controller
- Control data transfer- uses pipe 0: commands towards the device or status reporting
- Isochronous transfer- for the transmission at a guaranteed speed of real time data (e.g. voice, video)
- Interruption data transfer- from some peripherals to the host (e.g. keyboard, mouse, joystick), requires small delay guarantees
- Bulk transfers: they may use the whole bandwidth, but they are not time-critical (e.g transfers to/from a storage device)
- Prioritization of the real-time traffic possible

10. Write down the expression that describes the orthogonality of the OFDM carriers. What is the relation between the OFDM symbol duration (T) and the fundamental frequency (f0)?

A: Course 11, slide 10.

Orthogonal carriers

- The OFDM carriers are orthogonal, their frequencies being f_0, $2f_0$, $3f_0$ etc.

$$\frac{2}{T}\int_{kT}^{(k+1)T} \sin(mf_0t)\cdot\sin(nf_0t)\cdot dt = \begin{cases} 1,\ if\ m=n \\ 0,\ if\ m\neq n \end{cases} \qquad (1)$$

- Complex exponentials of limited duration used in practice
  - Their duration equals OFDM's symbol time (T)
- The orthogonality is met if: $f_0=1/T$

Audio and Video Systems

1. Digitization parameters and data rates for voice and hi-fi audio

https://intranet.etc.upt.ro/~AVS/Course/1_MULTIMEDIA.PDF, 15,16

- **High-quality stereo standard**
  - CD standard, hi-fi music, 20 kHz audio bandwidth
  - 2 channels
    - for **stereo** recording and transmission
  - f_E = 44.1 kHz
    - **sampling rate**, according to Shannon's theorem
  - n = 16 bits
    - for **quantization** with SNR = 96 dB

  ⇨ data rate: 2 × 44.100 × 16 = 1 411 200 bits/s

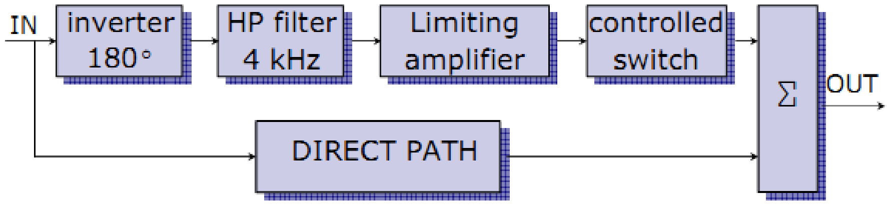
- **Speech-quality standard**
  - telephony standard, voice, 3,4 kHz audio bandwidth
  - 1 channel
    - for voice **recognition**
  - f_E = 8 kHz
    - **sampling rate**, according to Shannon's theorem
  - n = 8 bits
    - for **quantization** with SNR = 48 dB

  ⇨ data rate: 1 × 8.000 × 8 = 64.000 bps

2. Noise reduction principles

https://intranet.etc.upt.ro/~AVS/Course/2_SOUND.PDF, 23-28

- **NOISE** ⇔ signal with low level and middle to high **frequency**
  ⇨ such a signal can be identified and **rejected** (noise gate)
- **Example**: Philips **DNL** (*Dynamic Noise Limiter*)

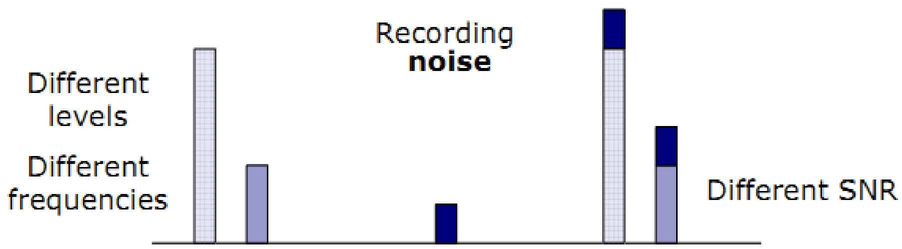


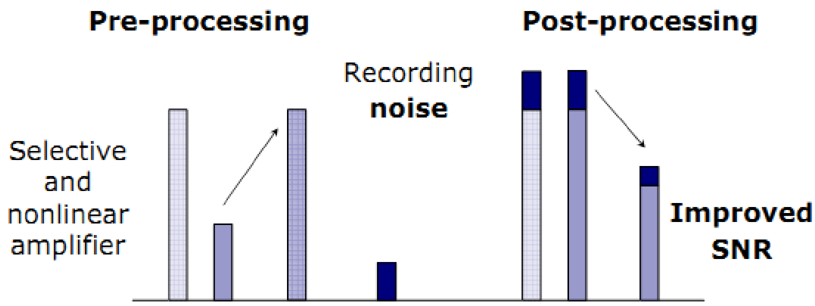
IN: signal with noise
OUT: signal with **improved SNR** with **8 dB**

- **DNL advantage**:
  - works with any recording system on **any playback system**
- **How DNL works in different situations:**
  - during the **pause** between melodies
  - high level recorded music
  - low level recorded music
- **DNL disadvantage**:
  - it cannot make the difference between noise and the real signal

- The systems perform:
  - signal processing before recording
  - opposite processing after playing back

Normal recording **Normal playback**





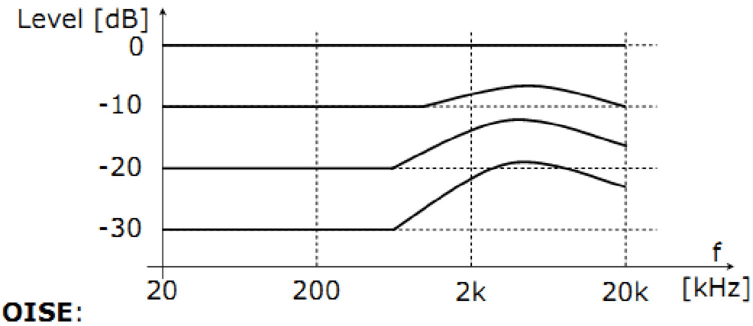
Advantage:

- the real signal is **not altered** and obtained with a **high SNR**

Disadvantage:

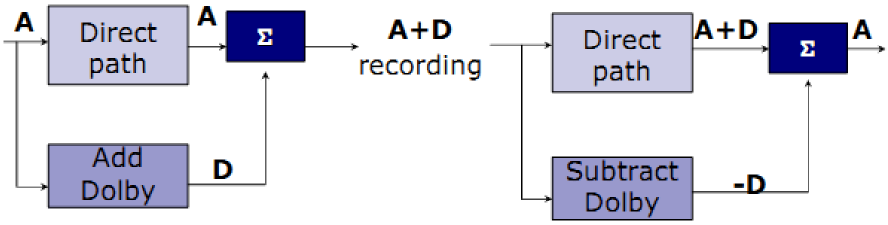
- it only works on the same system (record and playback)

Dolby system (I)



- **NOISE**:
  - high frequency (1 ÷ 15 kHz)
  - low level (-20 ÷ -40 dB)
- **DOLBY** circuits:
  - amplify **nonlinearly** and **selectively** when recording
  - performs opposite processing on playing the signal
  - **Increases SNR** with 9 dB

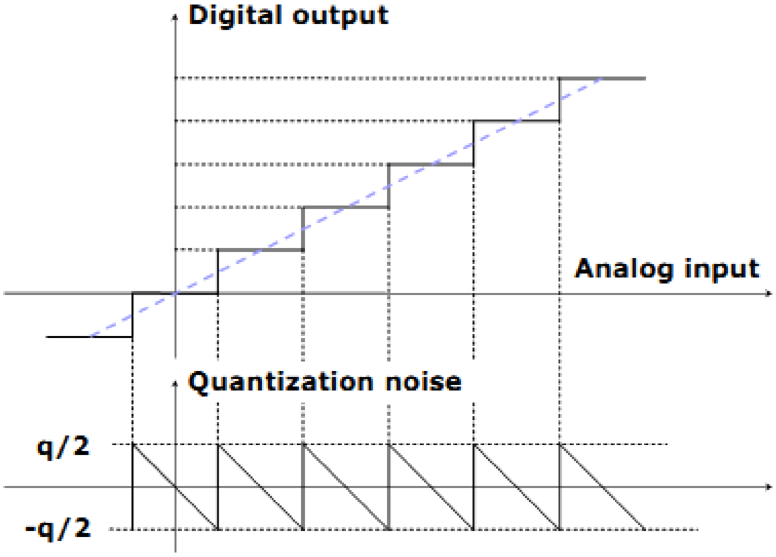
Dolby system (II)



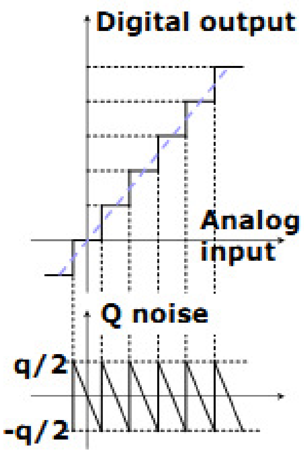
3. Quantization techniques

https://intranet.etc.upt.ro/~AVS/Course/2_SOUND.PDF, 36, 37, 41, 42

Uniform quantization (I)



Uniform quantization (II)



- **decision levels** (analog input)
  - **uniform**
- **quantization levels** (digital output)
  - **uniform**
- **quantization steps** (q) are **constant**:
  - for low level signal
  - for high level signal
- **quantization noise** (error): -q/2 ÷ q/2

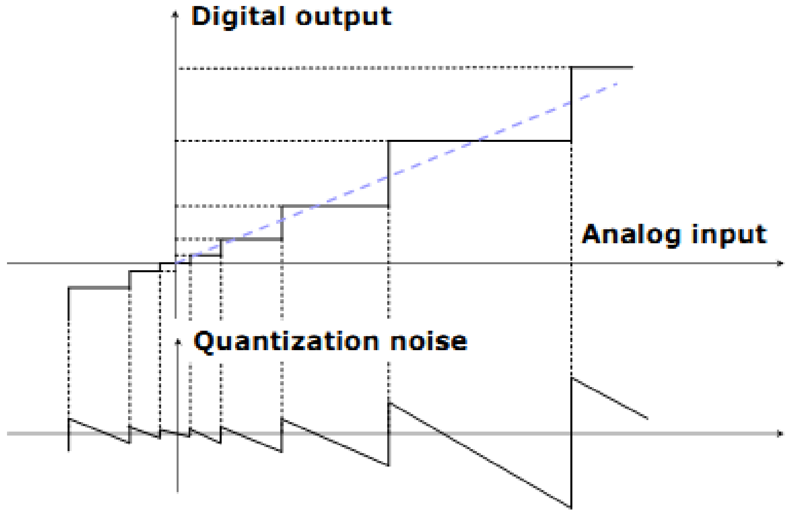
RESULT

- **low level signal with** constant quantization error ⇨ **low SNR**
- **high level signal with** constant quantization error ⇨ **high SNR**

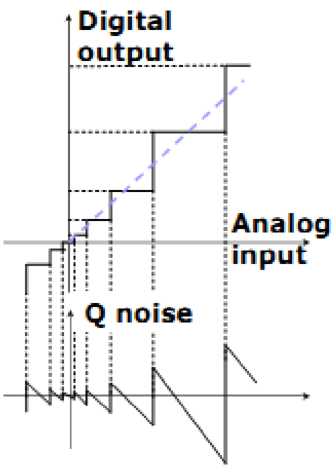
CONCLUSION

- **low general SNR (⇔ low quality)**

Non-uniform quantization (I)



Non-uniform quantization (II)



- **decision levels** (analog input)
  - **non-uniform**
- **quantization levels** (digital output)
  - **non-uniform**
- **quantization steps** are **different**:
  - low for low level signal
  - high for high level signal
- **quantization noise** (error):
  - non-constant

RESULT

- **low level signal with** low quantization error ⇨ **high SNR**
- **high level signal with** high quantization error ⇨ **high SNR**

CONCLUSION

- **high general SNR (⇔ high quality)**

4. The digital photo camera – adjustments, structure

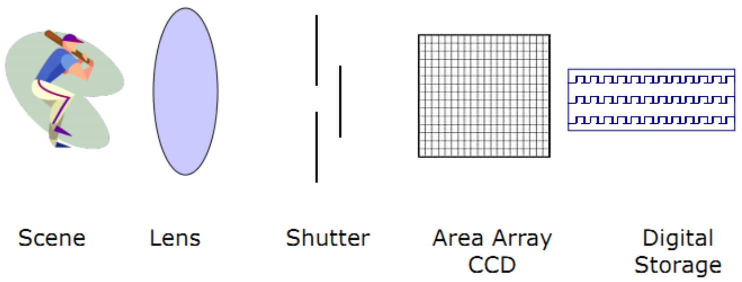
https://intranet.etc.upt.ro/~AVS/Course/3_EC_image_web.pdf, 6-9

- **Conventional** image **capture** needs the following main components:
  - LENS
    - to **focus the light** from a scene onto a photosensitive film (silver)
  - IRIS
    - to control the **amount of light** which hits the film
  - SHUTTER
    - to control the **timing** of the light exposure of the film

Electronic image acquisition (I)

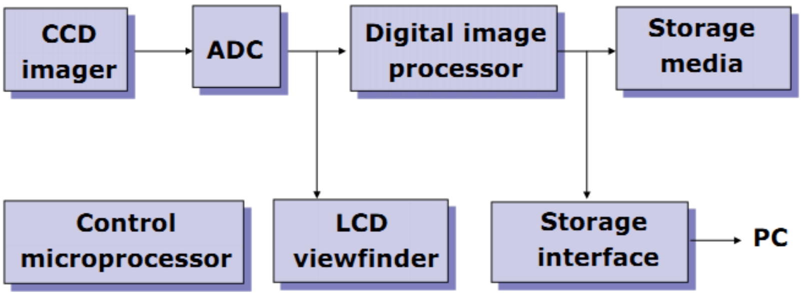
- The **electronic** image is obtained using:
  - traditional elements: lens, iris, shutter
  - additional components:
    - **CCD** (Charge Coupled Device)
      - image **scanning** and **photo-electric** conversion
    - **ADC** (Analog to Digital Converter)
      - delivering the **digital format** of the image
    - **Digital storage media**
      - **electronic** memory, **magnetic** support (disk or tape), optical support

Electronic image acquisition (II)



Digital photo camera (I)

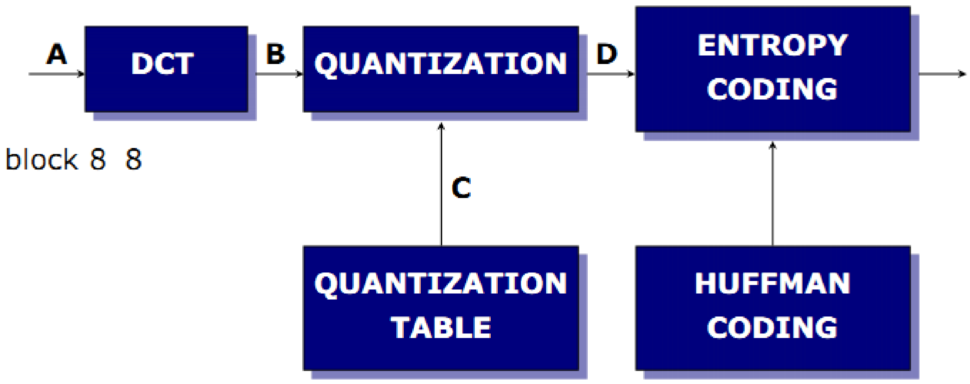
- A portable still image camera has the following **electronic components**:



5. The principle of JPEG compression

https://intranet.etc.upt.ro/~AVS/Course/3_EC_image_web.pdf, 62-66

JPEG Methodology (I)



JPEG Methodology (II)

- **DCT**
  - transforms **time** representation block **A** (lot of data points)
  - in **frequency** representation block **B** (few data points – few frequency components)
- **QUANTIZATION**
  - reduces **non-uniformly** the accuracy of coefficients, **D**, according to the quantization table **C** (**4 tables** implemented in JPEG algorithm):
    - **low frequency** with higher accuracy (small steps, non-zero values)
    - **high frequency** with lower accuracy (big steps, most values equal to zero)

JPEG Methodology (III)

- **ENTROPY CODING**
  - is used to obtain **data compression**
  - zig-zag scanning is used to obtain **long sequences of "zero"**
    - **RLE** (Run-Length Encoding) offers an excellent compression
    - **Huffman coding** is used to obtain higher compression factor

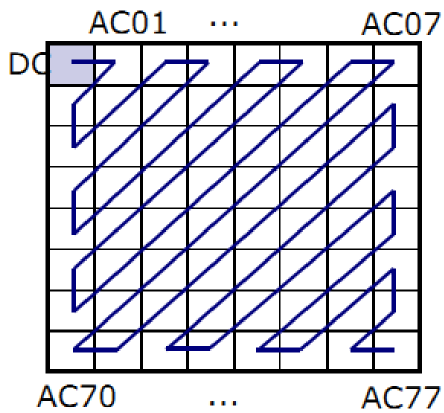
Discrete Cosine Transform (I)

- **DCT** (similar to Fourier transform) converts data from
- from **time domain**
  - 8×8 pixels block:
    - rows 0   7
    - columns 0   7
- to **frequency domain**
  - 8×8 coefficient matrix
  - 00 position
    - DC coefficient
      - average of the 8×8 block
  - 01   77 positions
    - AC coefficients
      - low frequency in the upper left corner
      - high frequency elsewhere

Discrete Cosine Transform (II)

Zig-zag sequencing

- starts with low frequency coefficients (non-zero),
- then high frequency coefficients (zero);
- results a long sequence of zeros, after a few significant values, easy entropy coding (RLE, Huffman)

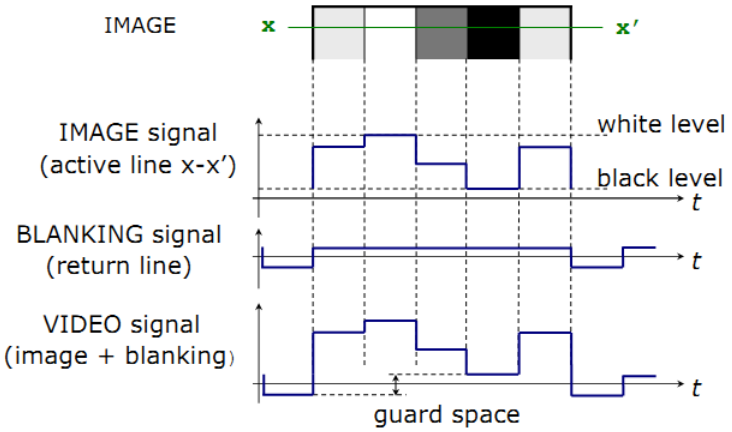


6. The composite video signal (components, parameters, TV line oscillogram)

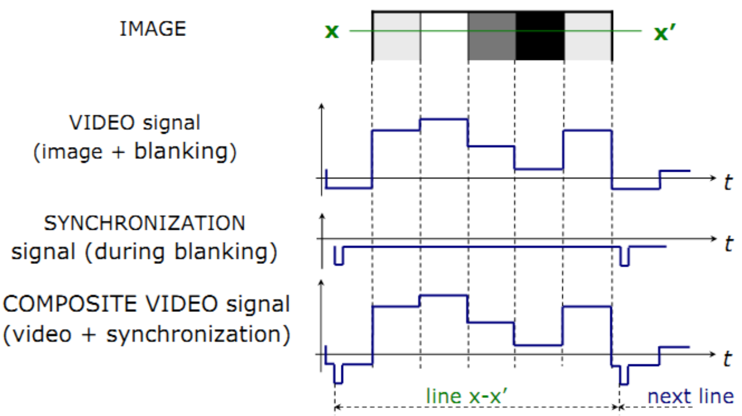
https://intranet.etc.upt.ro/~AVS/Course/4.1_EC_TV_web.pdf, 9, 10, 12

Composite video signal (I)

Is obtained during linear scanning (x-x')



Composite video signal (II)



Frequency of composite video signal (II)

- Aspect ratio
  - 4 × 3
- Vertical resolution
  - 575 visible lines (out of 625)
- Horizontal resolution

  For best resolution perception, the pixel must be **square**
  - 4 / 3 × 575 = 766 pixels

7. Color TV signals

https://intranet.etc.upt.ro/~AVS/Course/4.1_EC_TV_web.pdf, 21-23

Color TV signals (I)

- Luminance of a (color) image is used in black-and-white television:

  $$\mathbf{Y = 0.3 \times R + 0.59 \times G + 0.11 \times B}$$

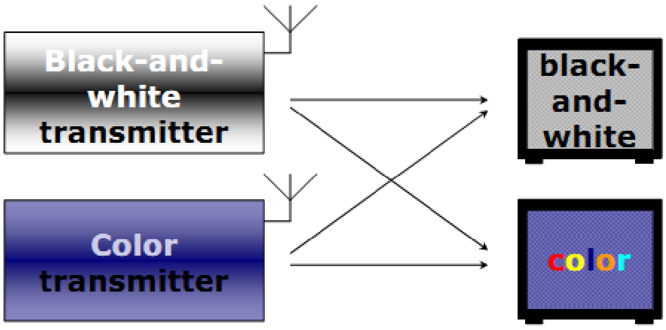
- Using R, G, B signals would be incompatible with the old TV system.

- Compatible color TV systems use:
  - Y – **luminance**
    (for correct processing by black-and-white TV sets)
  - C – **chrominance**
    (color information only, no brightness information)
    ⇨ **color difference** signals: **R-Y, G-Y, B-Y**

Color TV signals (II)

From the 4 signals, only 3 are used:

- luminance
  - $Y = 0.3 \times R + 0.59 \times G + 0.11 \times B$
- chrominance (2 color difference)
  - $R-Y = 0.7 \times R - 0.59 \times G - 0.11 \times B$
  - $B-Y = -0.3 \times R - 0.59 \times G + 0.89 \times B$



Compatible TV signals (I)

- **Luminance**

  $E_Y = 0.3 \times E_R + 0.59 \times E_G + 0.11 \times E_B \qquad = 0 \div 1$

- **Color difference**

  $E_{R-Y} = 0.7 \times E_R - 0.59 \times E_G - 0.11 \times E_B \qquad = -0.7 \div 0.7$

  $E_{G-Y} = -0.3 \times E_R + 0.41 \times E_G - 0.11 \times E_B \qquad = -0.41 \div 0.41$
  (not transmitted)

  $E_{B-Y} = -0.3 \times E_R - 0.59 \times E_G + 0.89 \times E_B \qquad = -0.89 \div 0.89$

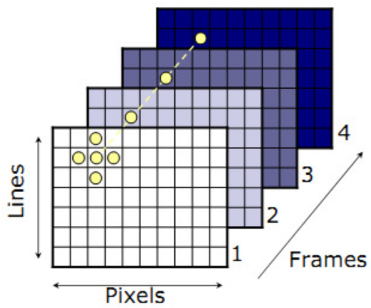
8.Digitization parameters, basic sampling formats and corresponding data rates for the TV signal

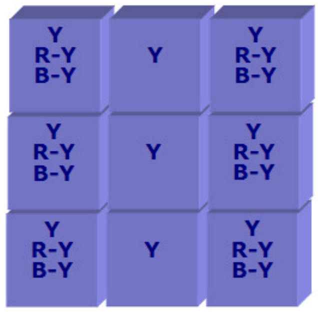
https://intranet.etc.upt.ro/~AVS/Course/4.5_E_DTV_web.pdf, 3-5, 12, 13

Digital TV studio standard (I)

- 1982, CCIR Rec.601 – **USA/Europe** standard
    - NTSC / SECAM / PAL
    - 525 / 625 lines
  - common **digital TV line**
    - same bit **rate**
    - same quality
    - easy system conversion
- TV components (**Y**, **R-Y**, **B-Y**)
  - **orthogonal** sampling
  - standard sampling frequency
    - $\boldsymbol{f_S}$ **= 13,5 MHz**
  - PCM format
    - **8 bits** / component sample

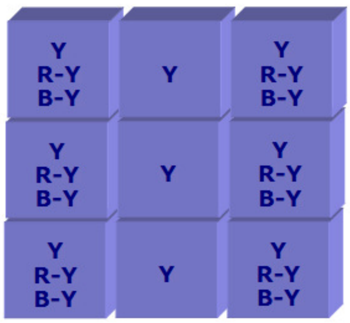
Digital TV studio standard (II)



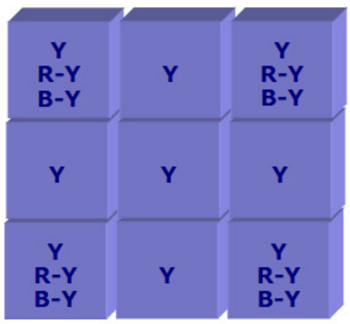


4:2:2 format
STUDIO quality

Digital TV studio standard (III)



4:2:2 format
STUDIO quality



4:2:0 format
BROADCASTING quality

Digital television signal bit rate

- Digital signal bit rate
  - $D = f_S \times n$ [bits/s]
- TV signal bit rate
  - $D = D_Y + D_{R\text{-}Y} + D_{B\text{-}Y}$
    $= f_{S\,Y} \times n_Y + f_{S\,R\text{-}Y} \times n_{R\text{-}Y} + f_{S\,B\text{-}Y} \times n_{B\text{-}Y}$
- 4:2:2 format - TV signal bit rate
  - $\boldsymbol{D_{TV}}$ =
    = 13.5 MHz × 8 b + 6.75 MHz × 8 b + 5.75 MHz × 8 b =
    = 108 Mbits/s + 54 Mbits/s + 54 Mbits/s =
    = **216 Mbits/s**

Digital television standard family

	Standard	Parameters	D_Y	$D_{R-Y}+D_{B-Y}$	D [Mbps]
HIGHER ORDER FORMATS	4:4:4 progressive	f_H = 31.250 Hz f_S = 27 MHz	216	216+216	648
	4:4:4 interlaced	f_H = 15.625 Hz f_S = 13,5 MHz	108	108+108	324
BASIC FORMAT	4:2:2 studio	f_{SY} = 13,5 MHz f_{SC} = 6,75 MHz	108	54 + 54	216
LOWER ORDER FORMATS	4:1:1	f_{SY} = 13,5 MHz f_{SC} = 3,375 MHz	108	27 + 27	162
	4:2:0 broadcast	alternative on lines: 4:2:2 4:0:0	108	54 + 54 0 + 0	162
	2:1:1	f_{SY} = 6,75 MHz f_{SC} = 3,375 MHz	54	27 + 27	108

9. The principle of MPEG compression

https://intranet.etc.upt.ro/~AVS/Course/4.6_E_MPEG_web.pdf, 7-9

MPEG coding (I)

- **Spatial** redundancy removal
  - DCT
- **Temporal** redundancy removal
  - Motion-compensated forward
  - Bidirectional prediction (interpolation)

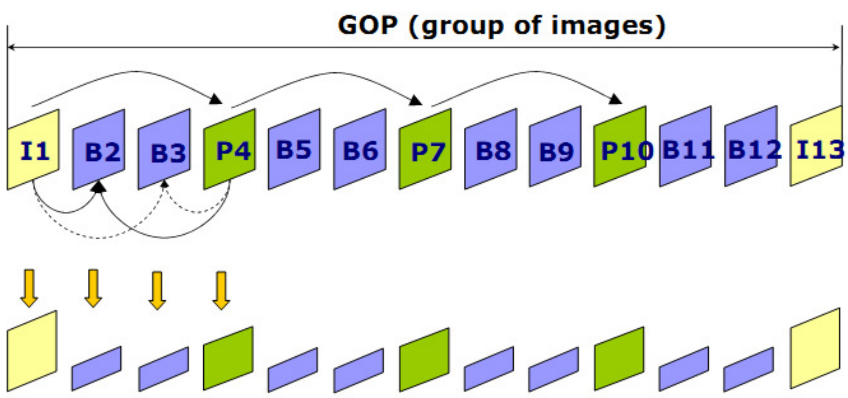
- MPEG uses three **types of images**
  - **Image I**
    - **JPEG** coded
    - independent to the sequence of moving images
    - **robust** coding
    - independent to precedent errors
    - **low** compression factor

MPEG coding (II)

- Image P
  - a **predicted** image is estimated (motion-compensated forward)
  - the **difference** between actual and predicted image is **coded**
  - sequence of predictions may propagate possible **errors**
  - **higher** compression factor

- Image B
  - a bidirectional **interpolated** image is calculated, using I and P images
  - very good estimation
  - may propagate **errors**
  - **best** compression factor

Moving pictures digital compression (I)



Compression

- Different techniques are used
- Resulting different factors

10. The structure and the parameters of a TV channel

https://intranet.etc.upt.ro/~AVS/Course/4.4_E_RF_web.pdf, 3, 5

Modulation methods

- Amplitude modulation
  - □ **AM**

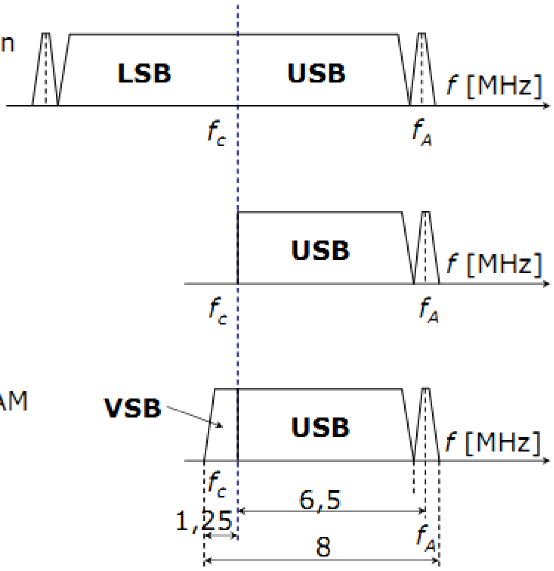
⇨low power efficiency

- Single Side Band AM
  - □ **SSB-AM**

⇨impossible to filter

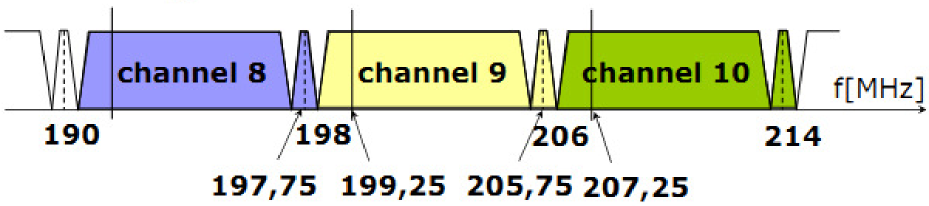
- Vestigial Side Band AM
  - □ **VSB-AM**

⇨standard solution



Intermediate frequency processing

Antenna signal in TV receiver



A **single filter** for:

- channel **extraction**
- adjacent channel **rejection**
- **VSB** rejection