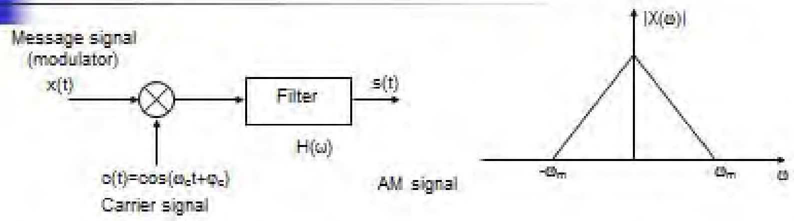
Product modulation- a simple case



- As a result of the modulation, the spectrum of the message signal shifts from "0" to ω_c
- Thus, the signal transmitted through the channel has a spectrum located to the carrier frequency
- Example: "Radio Romania can be listened at 567 KHz" refers to the carrier frequency used

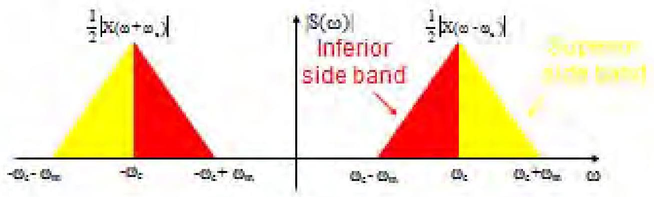
11

AM Frequency domain view [2]

- The spectrum of the modulated signal passed through a filter with $H(\omega)$ transfer function is:

$$S(\omega) = H(\omega)\ [\frac{1}{2}X(\omega_c + \omega) + \frac{1}{2}X(\omega - \omega_c)]$$

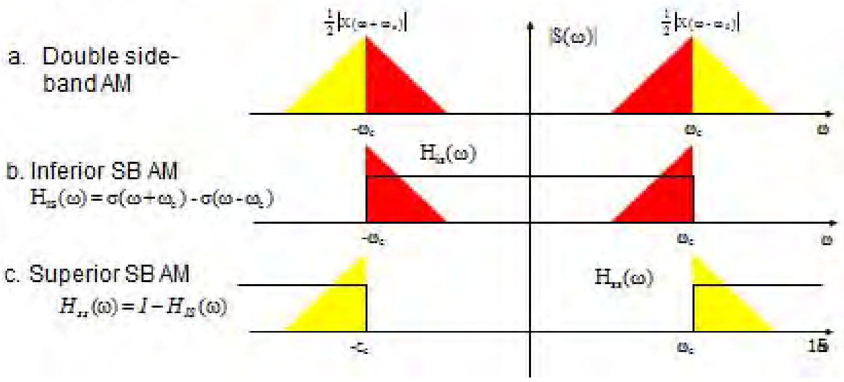
- The filter H selects the type of modulation we want



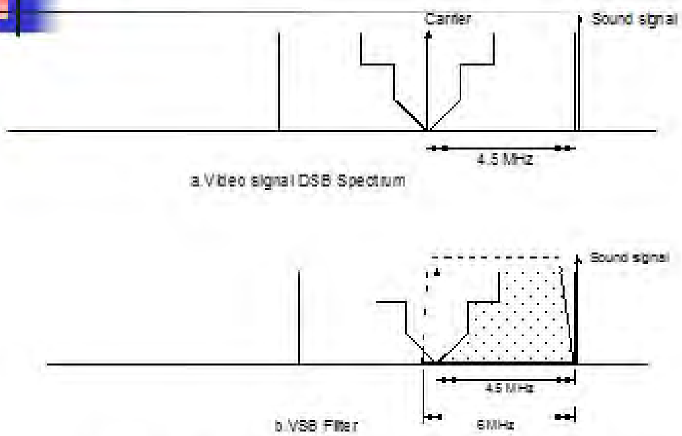
12

SSB selection

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



Vestigial Side Band



Only 6 MHz (standard video signal today) transmitted instead 9 MHz.

20

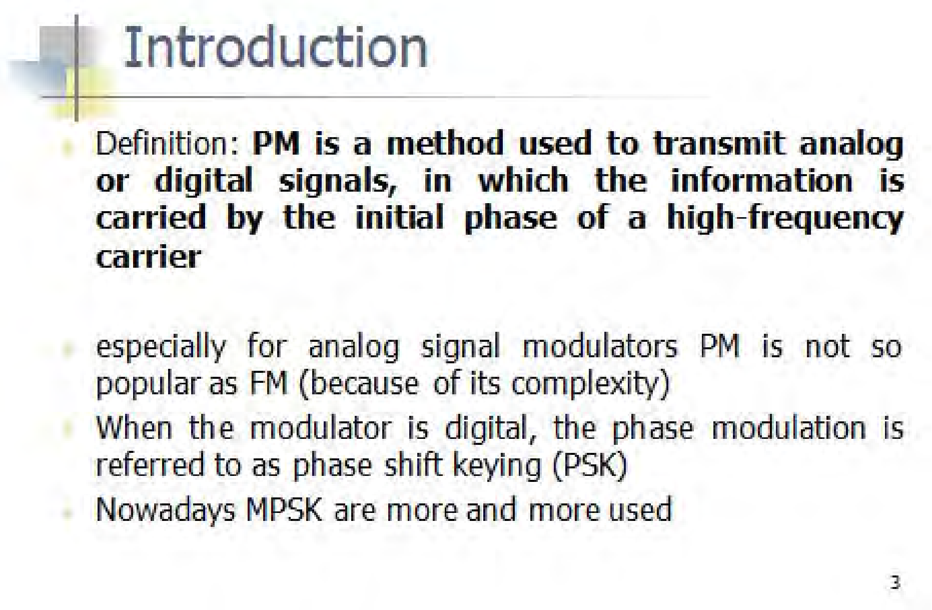
AM with Vestigial Side-Band (VSB)

- 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
- 25% more bandwidth required than in SSB, but easier to implement
- **Example**: NTSC (National Television System Committee) used in most Americas TV system: The signal is attenuated in the range from -1.25 MHz to -.75 MHz. From here on to 4 MHz, the signal is transmitted full strength.
- At 4 MHz it is once again attenuated down to 4.5 MHz so as not to interfere with the sound carrier which is demodulated separately.

22

9. Explain the meaning of terms PSK, BPSK and QPSK. Give the analytical expression of a general PSK signal and particulate in the case of a BPSK and QPSK signal. Graphically represent a BSK signal.

A: Course 7, slides 3, 5, 6, 12.

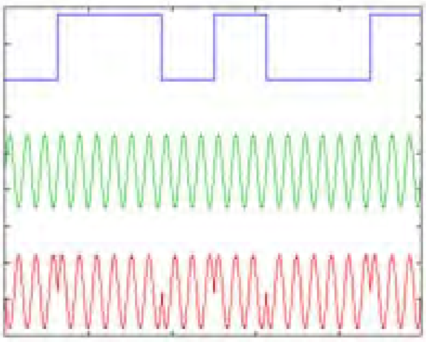


General analitical expession is:

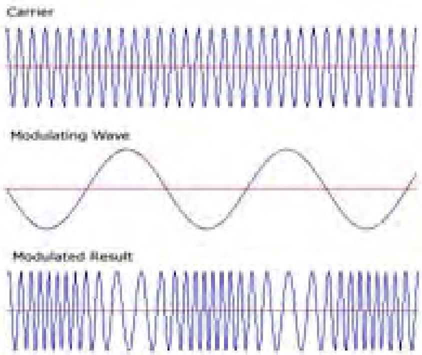
$$s(t) = U_0 \cos[\omega_0 t + k_P x(t) + \theta]$$

Where k_P is phase sensitivity factor , peak phase deviation over one symbol.

Graphical view



Binary Phase Shift Keying (BPSK)



Analog PM (continuous wave phase modulation)

5

Phase-Shift Keying (PSK)

Two-level PSK (Binary PSK) Uses two phases to represent binary digits for instance:

$$s(t) = \begin{cases} A\cos(2\pi f_c t) & \text{binary } 1 \\ A\cos(2\pi f_c t + \pi) & \text{binary } 0 \end{cases}$$

$$= \begin{cases} A\cos(2\pi f_c t) & \text{binary } 1 \\ -A\cos(2\pi f_c t) & \text{binary } 0 \end{cases}$$

6

Quadrature Phase-Shift Keying (QPSK) is a four-level PSK

- Each element represents more than one bit

$$s(t) = \begin{cases} A\cos\left(2\pi f_c t + \frac{\pi}{4}\right) & 11 \\ A\cos\left(2\pi f_c t + \frac{3\pi}{4}\right) & 01 \\ A\cos\left(2\pi f_c t - \frac{3\pi}{4}\right) & 00 \\ A\cos\left(2\pi f_c t - \frac{\pi}{4}\right) & 10 \end{cases}$$

12

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 12, slide 12.

Orthogonal carriers

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

$$\frac{2}{T}\int_{kT}^{(k+1)T} \sin(mf_0 t)\cdot\sin(nf_0 t)\cdot dt = \begin{cases} 1, \text{ if } m = n \\ 0, \text{ if } m \neq n \end{cases} \quad (1)$$

- Which means that there is no interference between the carriers
- **In practice there are used for carriers complex exponentials of limited duration (using a window function)**
- **Their duration equals OFDM's symbol time (T)**
- **The orthogonality is achieved 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

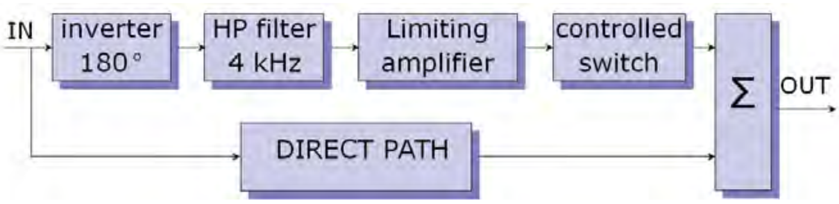
2. Noise reduction principles

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

Playback noise reduction (I+II)

- **NOISE** ⇔ signal with low level and middle to hig**h frequency**

  ⇨ such a signal can be identified and **rejected** (noise gate)
- **Example**: Philips **DNL** (*Dynamic Noise Limiter*)



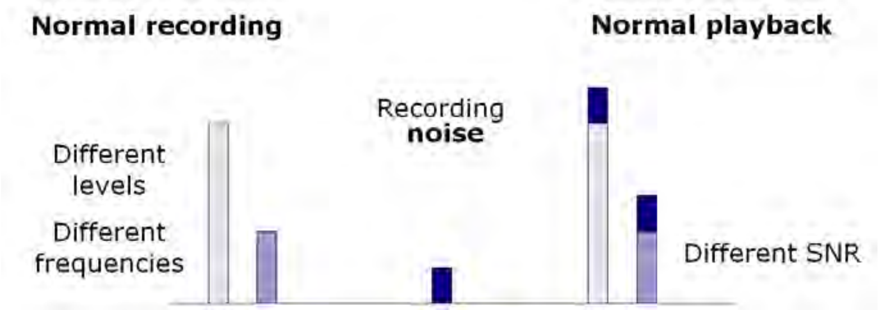
IN: signal with noise

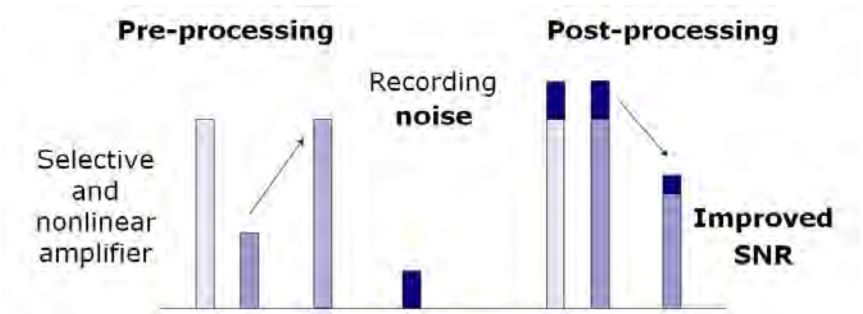
OUT: signal with **improved SNR** with **8 dB**

- **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

__Recording & playback noise reduction systems (I+II)__

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





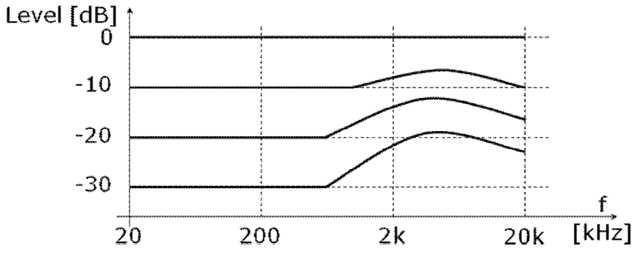
Advantage:

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

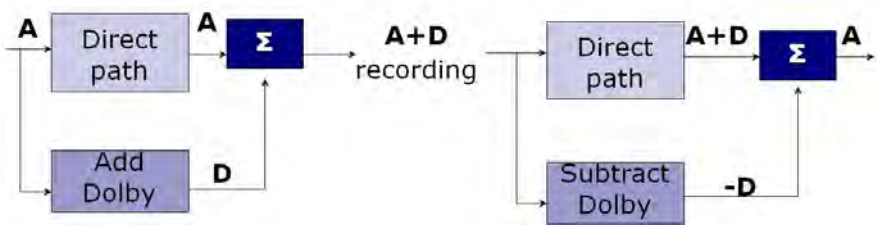
Disadvantage:

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

Dolby system (I+II)



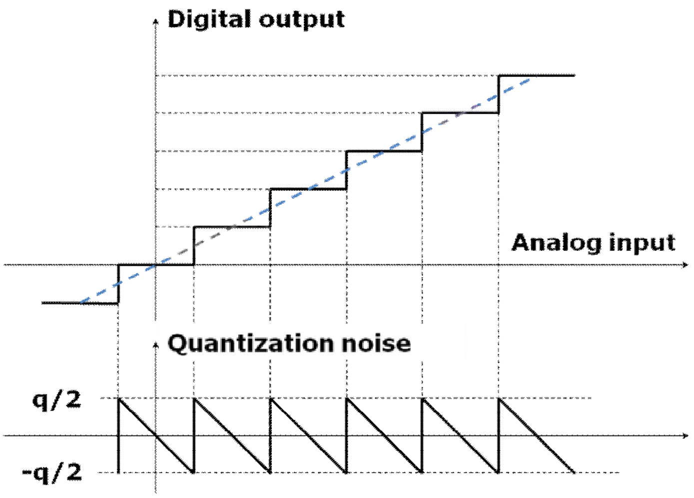
- **NOISE**:
  - high frequency (1 ÷ 15 kHz)
  - low level (-20 ÷ -40 dB)
- **DOLBY** circuits:
  - amplify **nonlinearly** and **selectively** when recording
  - performs opposite processing on playing the signal
  - **Increases SNR** with 9 dB

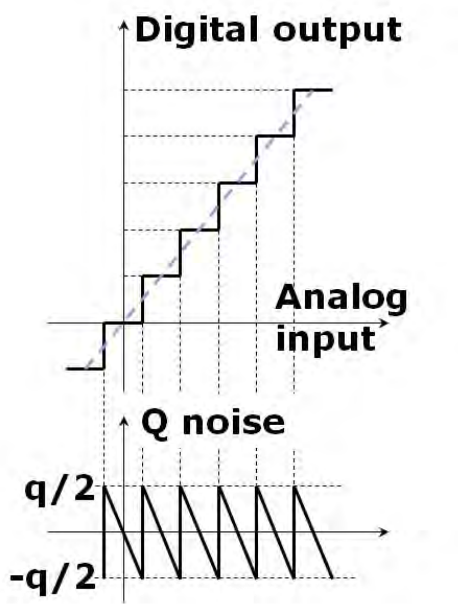


3. Quantization techniques

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

Uniform quantization (I+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

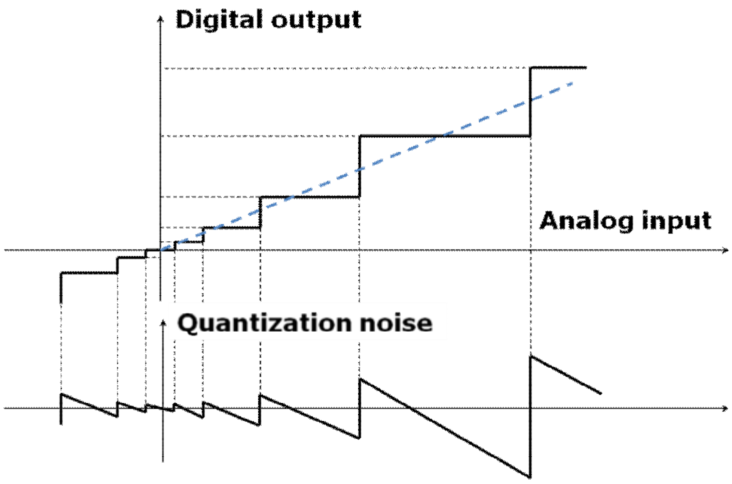
- **low level signal with** constant quantization error ⇨ **low SNR**

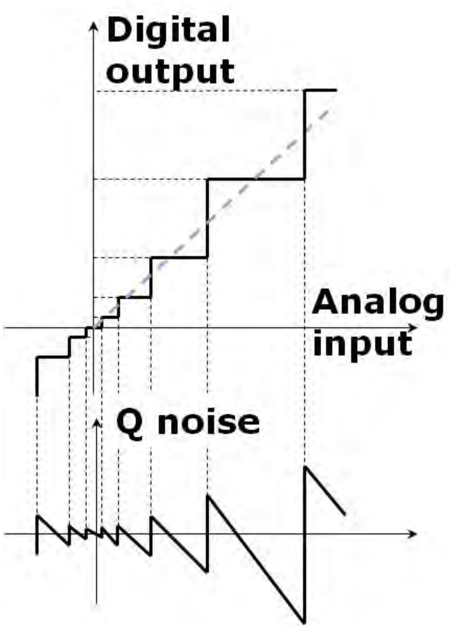
- **high level signal with** constant quantization error ⇨ **high SNR**

CONCLUSION

- **low general SNR (⇔ low quality)**

<u>Non-uniform quantization (I+II)</u>





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

Photographic image acquisition (II)

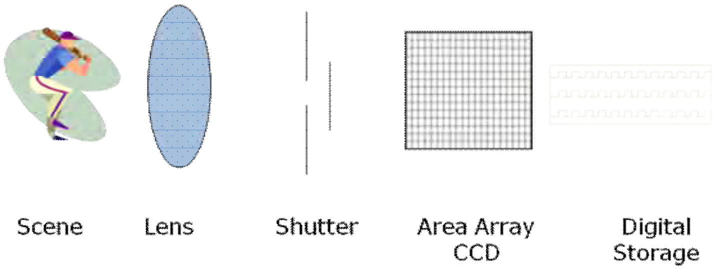
- **Conventional** image **capture** needs the following main components:
  - LENS
    - to **focus the light** from a scene onto a photosensitive film (silver)
  - IRIS
    - to control the **amount of light** which hits the film
  - SHUTTER
    - to control the **timing** of the light exposure of the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

Electronic image acquisition (I+II)

- 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)

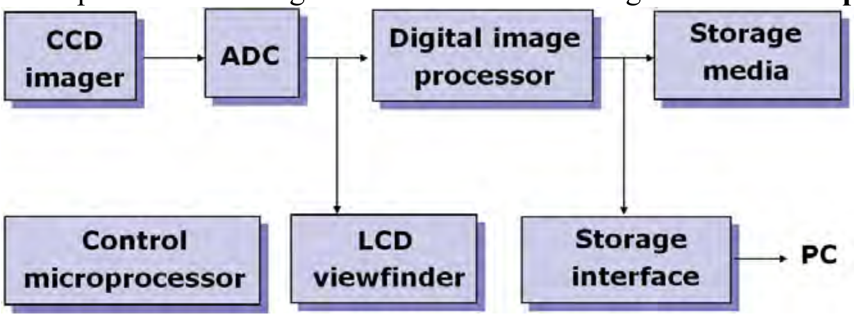
- delivering the **digital format** of the image

- **Digital storage media**
  - **electronic** memory, **magnetic** support (disk or tape), optical support



Digital photo camera (I)

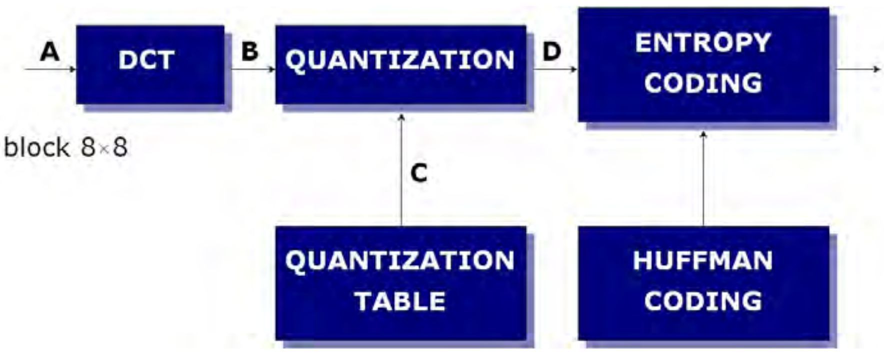
- A portable still image camera has the following **electronic components**:



5. The principle of JPEG compression

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

JPEG Methodology (I+II+III)



- **DCT**
  - transforms **time** representation block **A** (lot of data points)

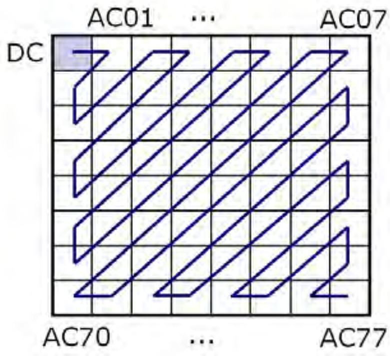
- in **frequency** representation block **B**
  (few data points – few frequency components)
- **QUANTIZATION**
  - reduces **non-uniformly** the accuracy of coefficients, **D**, according to the quantization table **C** (**4 tables** implemented in JPEG algorithm):
    - **low frequency** with higher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

<u>Discrete Cosine Transform (I+II)</u>

- **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

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)

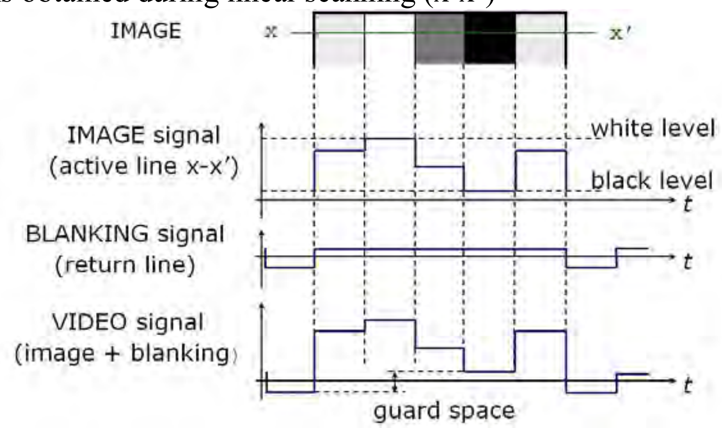


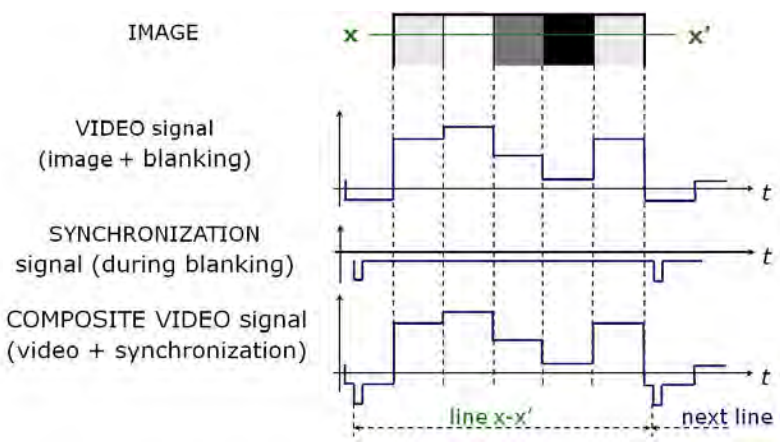
6. The composite video signal (components, parameters, TV line oscillogram)

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

Composite video signal (I+II)

Is obtained during linear scanning (x-x')





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+II)

- 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**
- From the 4 signals, only 3 are used:
  - □ luminance
    - □ Y = 0.3×R + 0.59×G + 0.11×B
  - □ chrominance (2 color difference)
    - □ R-Y = 0.7×R – 0.59×G – 0.11×B
    - □ B-Y = -0.3×R – 0.59×G + 0.89×B

Compatible TV signals (I)

- **Luminance**

  $E_Y = 0.3\times E_R + 0.59\times E_G + 0.11\times E_B = 0 \div 1$
- **Color difference**

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

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

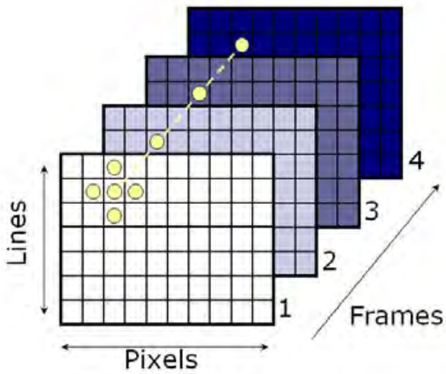
$E_{B-Y} = -0.3\times E_R - 0.59\times E_G + 0.89\times E_B = -0.89 \div 0.89$

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

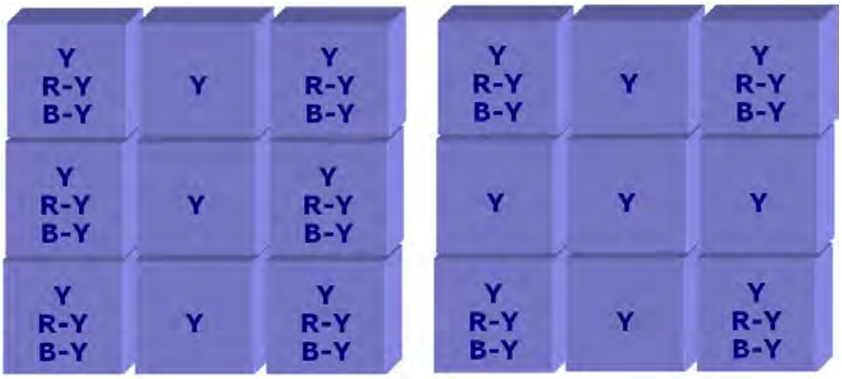
Digital TV studio standard (I+II+III)

- 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
    - f_s **= 13,5 MHz**
  - PCM format
    - **8 bits** / component sample



4:2:2 format

STUDIO quality



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**	$\mathbf{D_Y}$	$\boldsymbol{D_{R\text{-}Y}+D_{B\text{-}Y}}$	$\boldsymbol{D}$ [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	4:1:1	f_{SY} = 13,5 MHz f_{SC} = 3,375 MHz	108	27 + 27	162

FORMATS	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

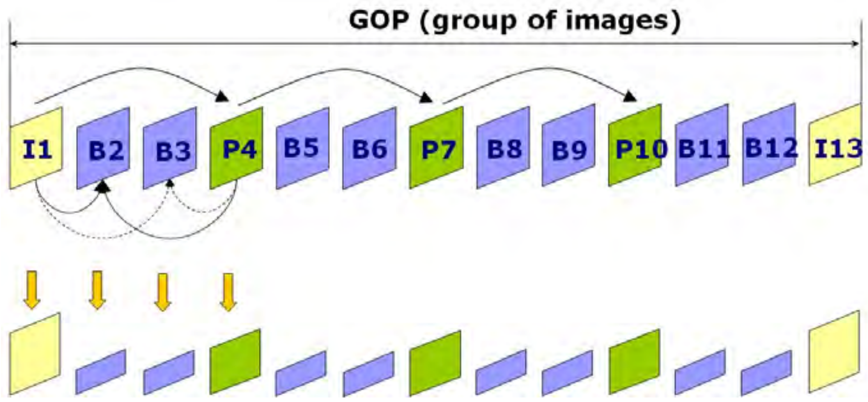
9. The principle of MPEG compression

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

MPEG coding (I+II)

- **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
  - **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

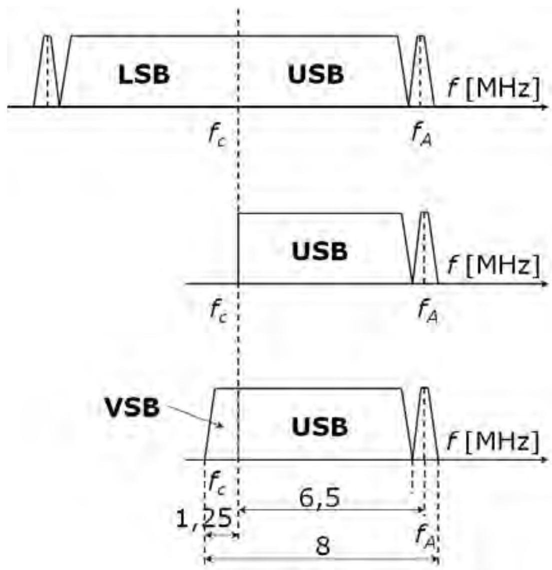
Modulation methods

- Amplitude modulation
  - **AM**

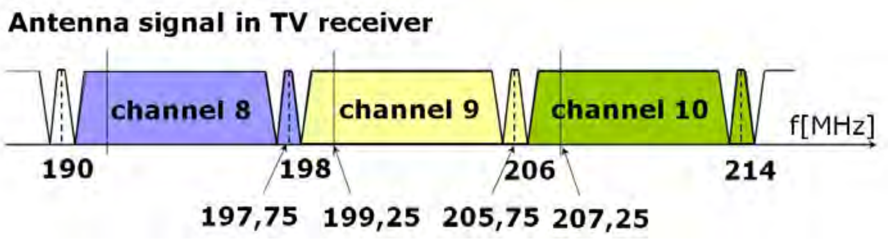
  ⇨low power efficiency
- Single Side Band AM
  - **SSB-AM**

  ⇨impossible to filter
- Vestigial Side Band AM
  - **VSB-AM**

  ⇨standard solution



Intermediate frequency processing



A **single filter** for:

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