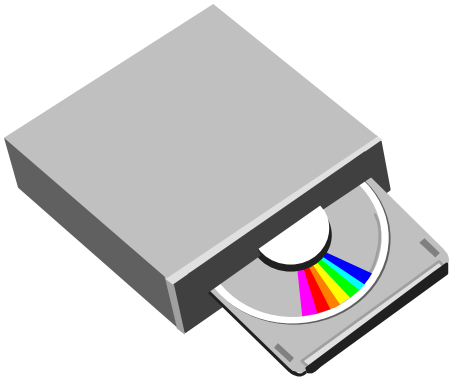
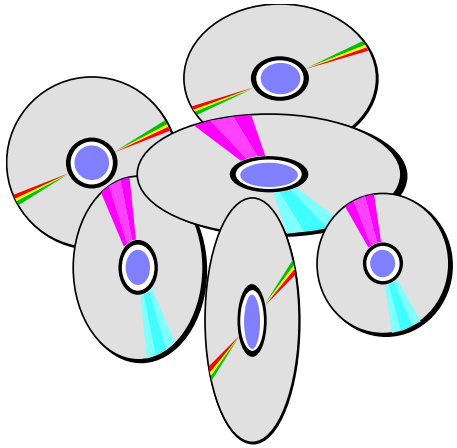
DIGITAL VIDEO STORAGE





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Digital Audio and Video Storage

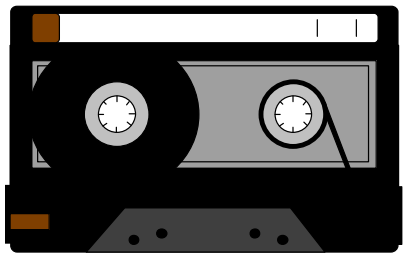
There are several technologies involved in digital audio and video storage, this means in the process of recording a physical support to store the audiovisual (AV) information at hand.

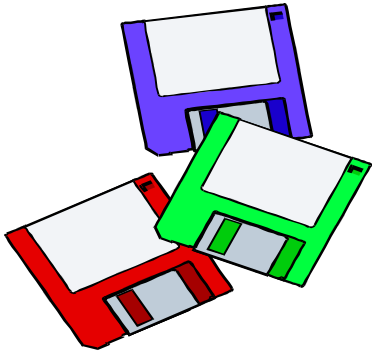
One of the most important technologies for AV storage is audiovisual data coding which should provide the necessary compression efficiency and quality but also other storage functionalities such as random access, already provided in analogue recording.

Recording Functionalities …

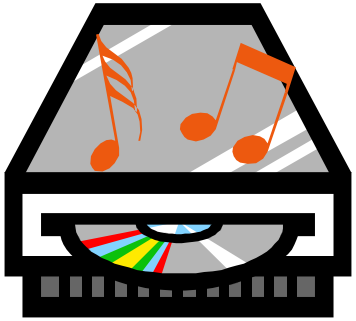
- **Normal video playback – The usual play …**
- **Random access – It shall be possible to access any part of the audiovisual data in a limited amount of time, e.g. 0.5 s.**
- **Reverse playback – Playing at regular speed opposite to the usual temporal direction …**
- **Fast forward and Fast reverse – Faster play (with time compression) in the usual and opposite time directions (more complex form of random access).**
- **Edition – Capability to edit the coded signal in a simple way.**

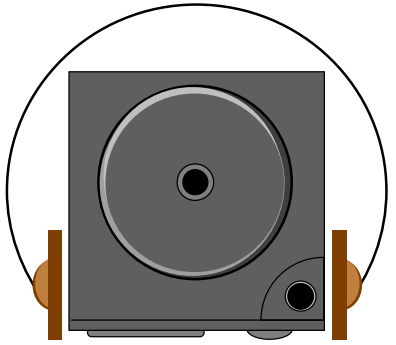
Main Storage Supports





- Magnetic tape
- Magnetic discs
- Optical discs
- …



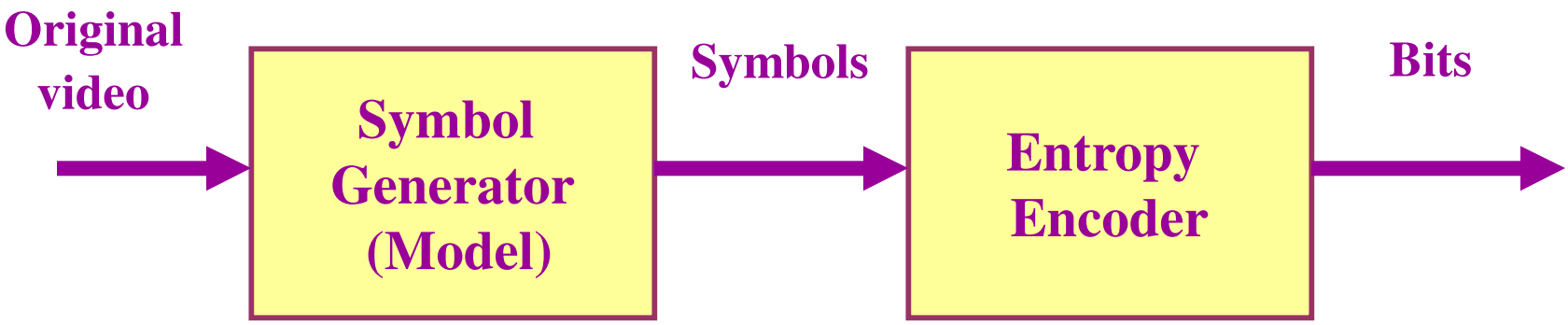


Storage: Which Support ?

Main factors to be taken into account to select an audiovisual storage support:

- **Capacity (in MBytes)**
- **Reading speed (in Mbit/s)**
- **Time and form of access (e.g. sequential or random)**
- **Durability**
- **Mobility**
- **Cost**
- **…**

The Basic Coding Chain

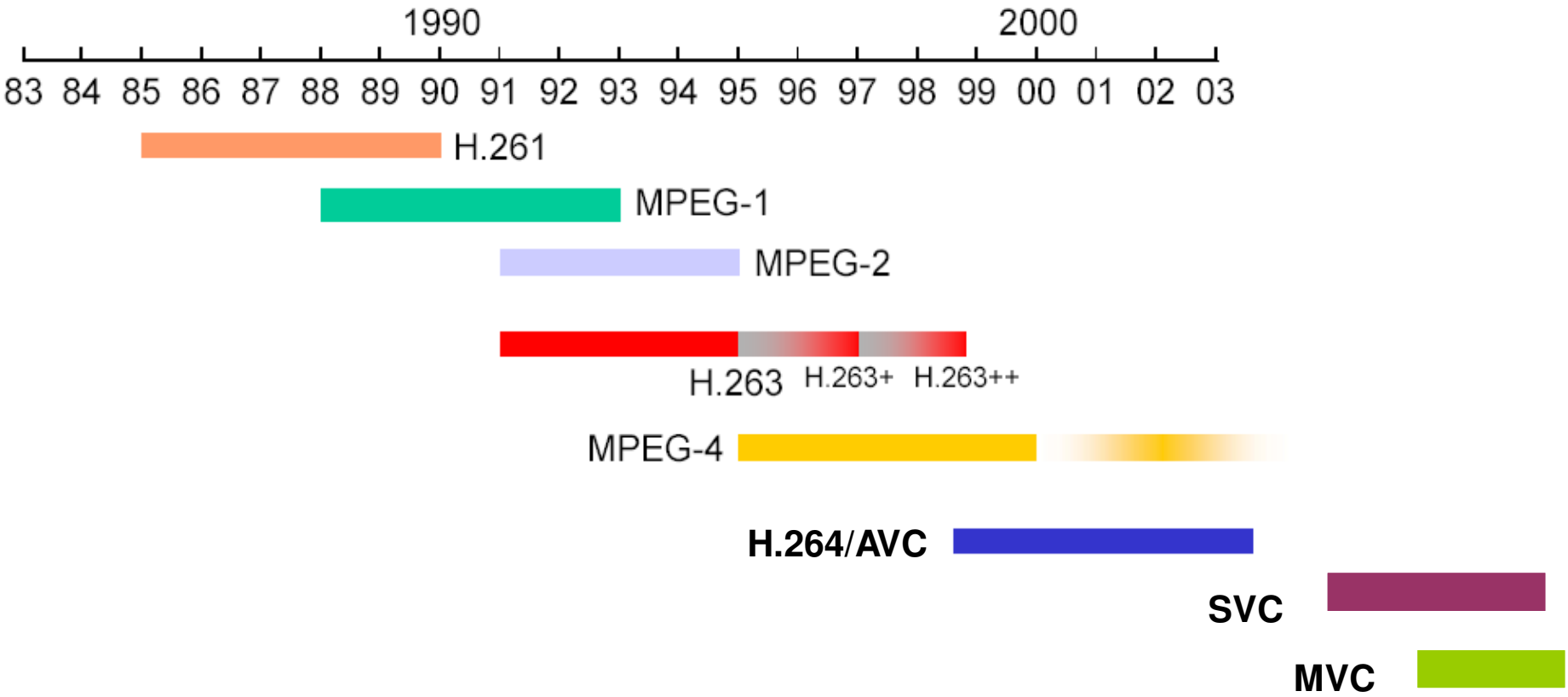


The symbolic coding model depends on the type of data, e.g. audio, video, and on the application requirements, e.g. random access.

The MPEG Family for Source Coding

- **MPEG-1 (1988-1990): Coding of video and associated audio for a target bitrate of 1.5 Mbit/s**
  - **CD Storage (initial target)**
- **MPEG-2 (1990-1993): Coding of video and associated audio (initially for bitrates up to 10 Mbit/s)**
  - **Digital TV (for any transmission channel) and DVD**
- **MPEG-3 (X): Coding of video and associated audio with bitrate up to 60 Mbit/s (finally not defined since MPEG-2 fulfils the needs)**
  - **High Definition TV (HDTV)**
- **MPEG-4 (1994-2008): Coding of video and associated audio (natural and synthetic) based on objects**
  - **Interactive multimedia applications and much more**

Standards Along the Years ...



MPEG-1 Standard

Motivation

- **The emergence of digital storage supports with large capacity and high reading speeds at increasingly lowers costs.**
- **The development of video coding algorithms reaching increasingly higher compression factors for a certain acceptable quality.**
- **The growing electronic integration capability of complex functions in reduced silicon areas (VLSI).**
- **The growing interaction between the telecommunications, computer and consumer electronics industries.**
- **The need to standardize in an area for which the technical development was ready to offer several *de facto* solutions, taking the opportunity to lower the costs and increase the production.**

MPEG-1: Storage Supports (~ 1990)

- **CD-ROM (Compact-Disc Read Only Memory)**
  - **Capacity between 600 MByte and 2 GByte (usually, 700 MB) with a reading speed of about 1.5 Mbit/s (and growing …)**
- **CD-WORM (CD-Write Once Read Many times)**
  - **Reading speed of about 8 Mbit/s**
- **Discos Winchester**
  - **Capacity between 20 and 400 MByte (now above 1 GByte) with a reading speed of about 8 Mbit/s (now above 20 Mbit/s)**
- **DAT (Digital Audio Tape)**
  - **Reading speed of about 7.5 Mbit/s**

Storage: Which Support ?

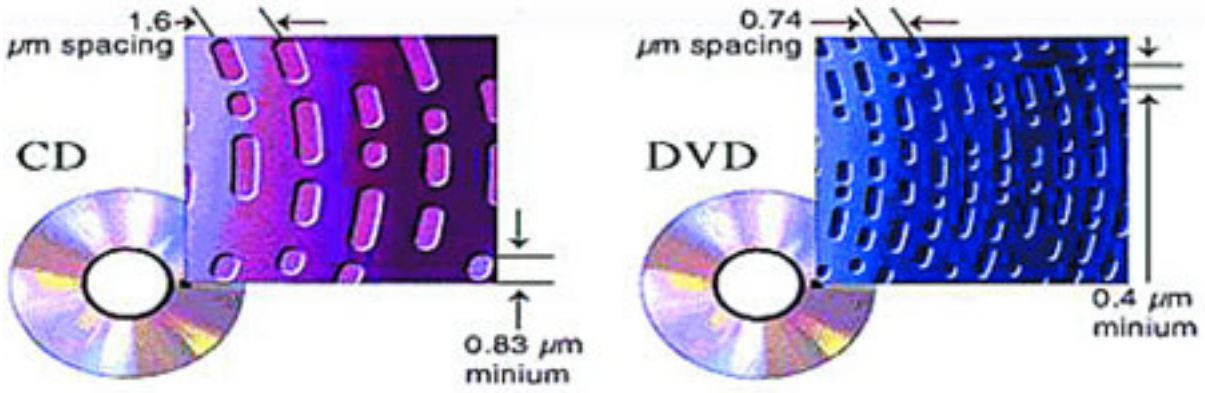
Main factors to be taken into account to select an audiovisual storage support:

- **Capacity (in MBytes)**
- **Reading speed (in Mbit/s)**
- **Time and form of access (e.g. sequential or random)**
- **Durability**
- **Mobility**
- **Cost**
- **…**

The CD-ROM was selected as the most adequate storage support to offer, for the first time in large scale, interactive multimedia signals, mainly due to its large capacity and low cost.

Digital Versatile Disc (DVD): the Support Today !

- The DVD is essentially an evolution of the CD where it is possible to store more video, audio or any other type of digital data.
- There are 2 DVD sizes: 12 and 8 cm both 1.2 mm thick (two 0.6 mm glued subtracts).
- DVDs may be single-sided or double-sided depending on only one or the two sides store data.
- Each side may have one or two storage layers.



HD DVD and Blu-Ray Disc: the Future

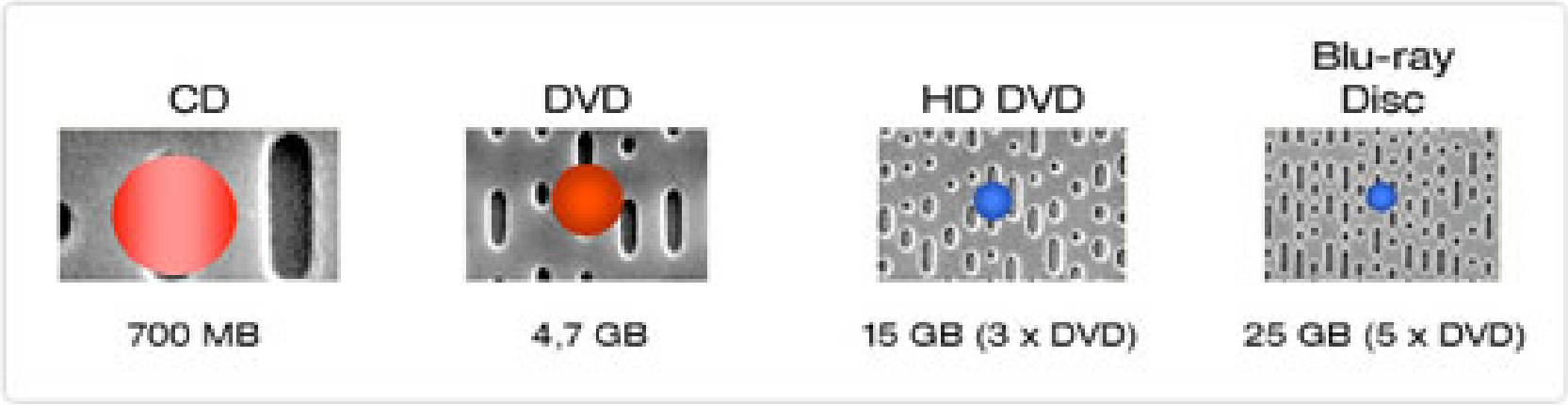
Blu-ray Disc

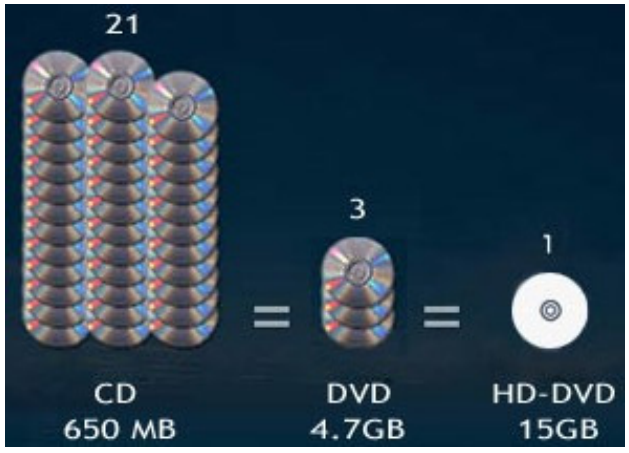
VS

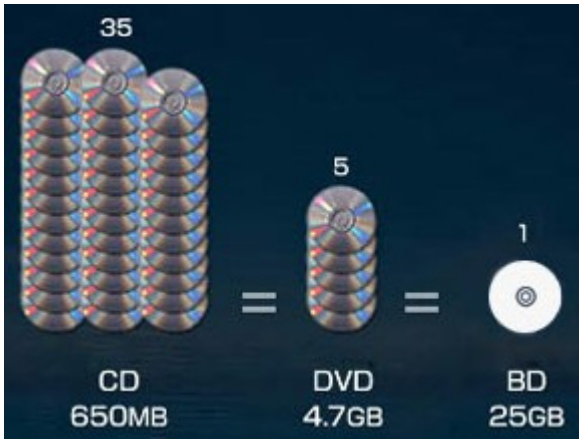
HD DVD

- **A single-layer Blu-ray Disc can store 25 gigabytes (GB), over five times the size of a single layer DVD with 4.7 GB.**
- **A dual-layer Blu-ray Disc can store 50 GB (possibly up to 300GB in the future with 12 layers), almost six times the size of a dual layer DVD with 8.5 GB.**
- **Both disks use the same video and audio codecs, but Blu-ray has a higher maximum bitrate, meaning that multimedia on disks can be stored and delivered with less compression than their HD DVD counterparts.**
- **HD DVD *died* in 2008 …**

Making Comparisons …







Blu-Ray versus DVD

Parameters	Blu-ray	DVD
Storage capacity	25GB (single-layer) 50GB (dual-layer)	4.7GB (single-layer) 8.5GB (dual-layer)
Laser wavelength	405nm (blue laser)	650nm (red laser)
Numerical aperture (NA)	0.85	0.60
Disc diameter	120mm	120mm
Disc thickness	1.2mm	1.2mm
Protection layer	0.1mm	0.6mm
Hard coating	Yes	No
Track pitch	0.32µm	0.74µm
Data transfer rate (data)	36.0Mbps (1x)	11.08Mbps (1x)
Data transfer rate (video/audio)	54.0Mbps (1.5x)	10.08Mbps (<1x)
Video resolution (max)	1920×1080 (1080p)	720×480/720×576 (480i/576i)
Video bit rate (max)	40.0Mbps	9.8Mbps
Video codecs	MPEG-2 MPEG-4 AVC SMPTE VC-1	MPEG-2 - -
Audio codecs	Linear PCM Dolby Digital Dolby Digital Plus Dolby TrueHD DTS Digital Surround DTS-HD	Linear PCM Dolby Digital DTS Digital Surround - - -
Interactivity	BD-J	DVD-Video

MPEG-1: Objectives

Coding of video and associated audio with a total bitrate of about 1.5 Mbit/s with a minimum acceptable (subjective) quality.

- **With MPEG-1, video signals just become another type of (digital) data which may be easily stored and processed, e.g. in a computer.**
- **The video and associated audio information may be stored in any type of digital support or transmitted in any type of digital network.**

MPEG-1: Example Applications

• **Asymmetric applications – These applications involve the repeated usage of the decoding process after a single (or a limited number) of encodings**

- **Movies**
- **Games**
- **Education**
- **Tele-shopping**
- **Tourism**

• **Symmetric applications - These applications involve a similar usage of the encoding and decoding processes**

- **Videotelephony**
- **Videoconference**
- **Video-mail**

MPEG-1 Standard: Structure

Part 1: Systems

Specifies the multiplexing of the several audio and video coded streams in a single stream with synchronization

Part 2: Video

Specifies the video coding solution (bitstream and decoding) for bitrates of about 1.15 Mbit/s

Part 3: Audio

Specifies the audio coding solution (bitstream and decoding) for bitrates of 32-448 kbit/s per channel (mono and stereo)

Part 4: Conformance Testing

Specifies conformance tests for the streams and decoders

Part 5: Reference Software

Software implementation of the parts 1, 2 and 3

MPEG-1 Standard

Part 1: Systems

MPEG-1 Systems: Objectives

The MPEG-1 Systems standard has the objective to combine one or more coded audio and video streams into a single binary stream, called MPEG-1 stream or ISO/IEC 11172 stream.

The MPEG-1 Systems standard defines:

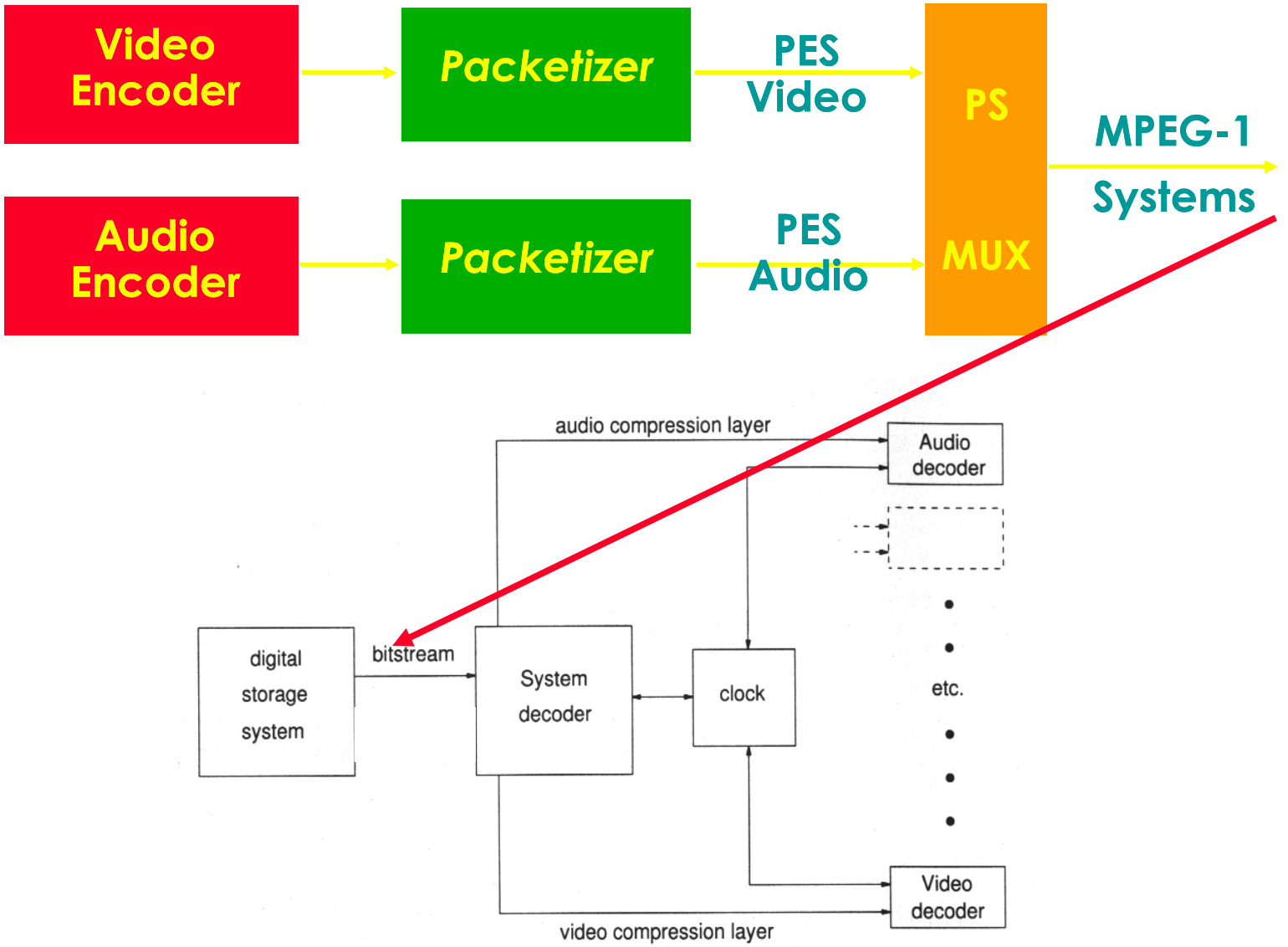
- **Syntax for the streams offering timing control**
- **Multiplexing and synchronization of the audio and video streams**

MPEG-1 Streams

One MPEG-1 stream is formed by two layers:

- **SYSTEM – Serves as envelope for the compression layers; offers the necessary information for the demultiplexing and timing of the compression layers.**
- **COMPRESSION – Includes the coded data that will be given to the audio and video decoders.**

The elementary (coded) audio and video streams are divided into variable size packets – the packets – creating the so called *Packetized Elementary Streams* (PESs).


Video Encoder
Packetizer
PES Video
PS
MUX
MPEG-1 Systems
Audio Encoder
Packetizer
PES Audio
audio compression layer
Audio decoder
digital storage system
bitstream
System decoder
clock
etc.
Video decoder
video compression layer

Packs and Packets

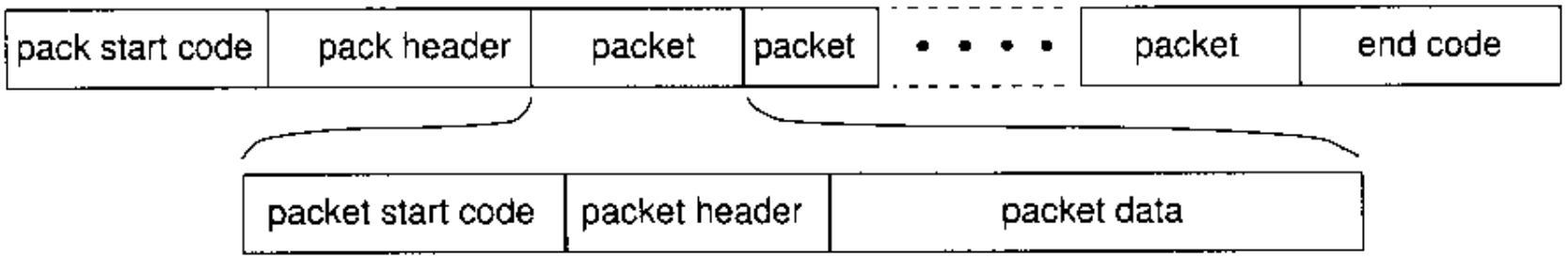
The operations to be performed by the Systems decoder regard the full MPEG-1 stream - *multiplex-wide*- or elementary streams, e.g. audio or video - *stream-specific*.

The MPEG-1 Systems stream is structured in two sub-layers:

- ***PACK sub-layer*** **– Refers to the multiplex-wide operations such as the control of the reading of the stream from the storage support, if possible, the adjustment of the clocks, buffer management, and the definition of the resources needed for decoding.**
- ***PACKET sub-layer*** **– Refers to the stream-specific operations such as demultiplexing and synchronization of the various elementary streams; packets may have a fixed or variable length.**

One *pack* corresponds to a collection of *packets* with additional *multiplex-wide* information.

MPEG-1 Systems Stream Syntax



- **One MPEG-1 Systems stream consists in a sequence of packs, each one containing several packets (with coded audio OR video); one video (or audio) packet may start at any byte of the video (or audio) stream and may have a variable length.**
- **The Systems decoder parses the MPEG-1 stream, giving to the audio and video decoders their respective packets, after inspecting the *packet start codes*.**
- **At most, 32 audio streams, 16 video streams and 2 data streams may be multiplexed in a single MPEG-1 stream.**

MPEG-1 Systems: Synchronization

MPEG-1 Systems synchronization relies on two basic elements:

- ***Systems Target Decoder* (STD) – Hypothetical reference model used to define the ideal decoding process; in this ideal model, the transference, decoding and presentation of the information are instantaneous; in a real system, some delay is inevitable and thus should be accounted for.**

- ***Master Time Base* (MTB) – Reference timing information used to synchronize the presentation of the various elementary streams; it may correspond to the clock of one of the elementary decoders, the Digital Storage Media clock or to an external clock - *Systems Clock Reference (SCR)* derived from a *Systems Time Clock (STC*).**

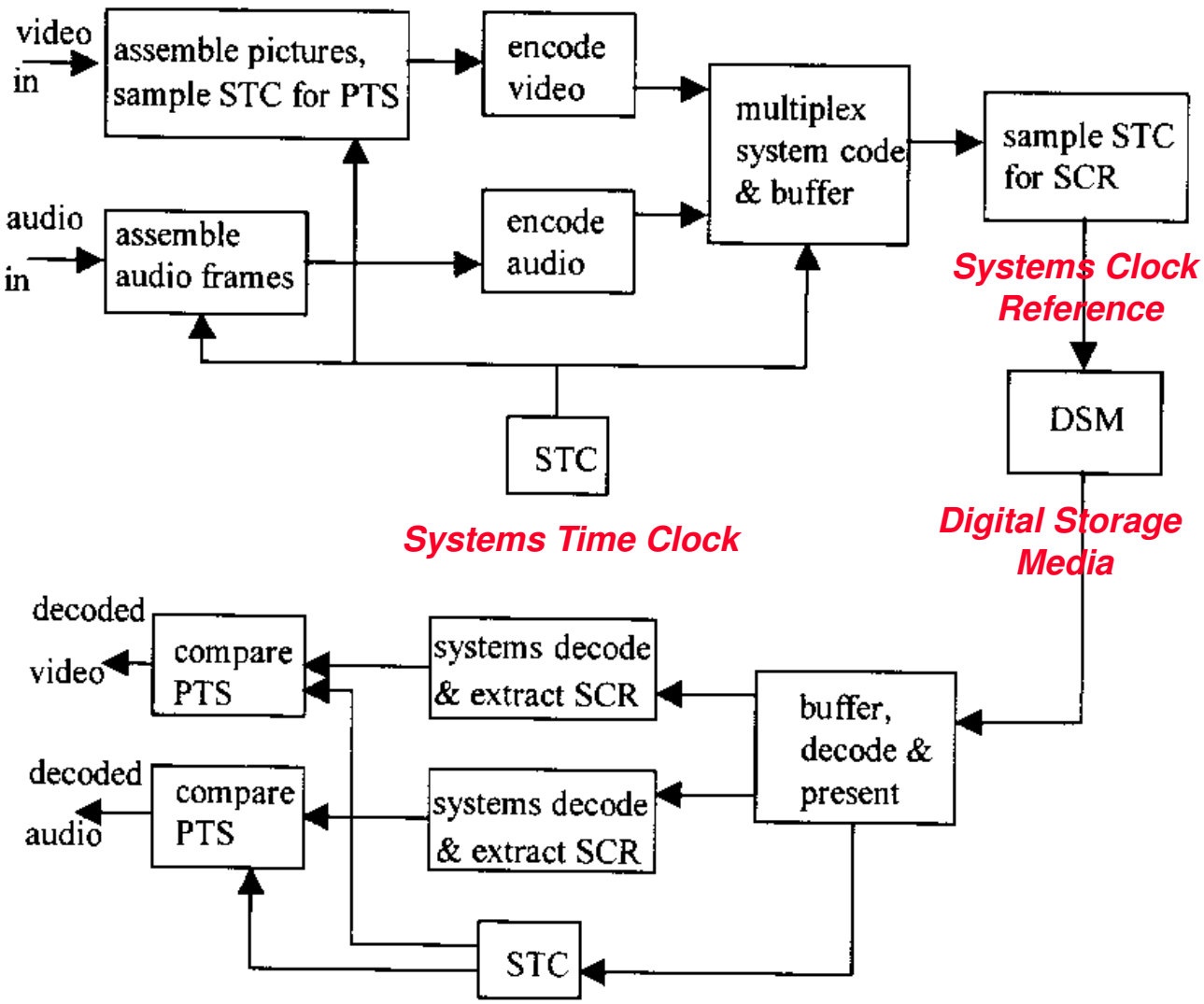
MPEG-1 Systems: Timing

- ***Decoding Time Stamp*** **(DTS) – Timing information that may be present in the packet header to indicate the moment when the corresponding coded information must be decoded in the *Systems Target Decoder* (STD).**
- ***Presentation Time Stamp*** **(PTS) – Timing information that may be present in the packet header to indicate the moment when the corresponding decoded information must be presented in the *Systems Target Decoder* (STD).**

MPEG-1 players use PTS to control the presentation of the decoded information regarding the reference clock.

PTS and DTS are different when the decoding and presentation orders are not the same, such as when using B frames; the STD assumes instantaneous decoding.

MPEG-1 Chain Architecture



MPEG-1 Standard

Part 2: Video

MPEG-1 Video: Requirements

- **Normal video playback – The usual play …**
- **Random access – It shall be possible to access any part of the audiovisual data in a limited amount of time, e.g. 0.5 s.**
- **Reverse playback – Playing at regular speed against the usual temporal direction …**
- **Fast forward and Fast reverse – Faster play (with time compression) in the usual and opposite time directions (more complex form of random access).**
- **Edition – Capability to edit the coded signal in a simple way.**
- **Audiovisual synchronization – Need to guarantee synchronization between audio and video information.**
- **Error resilience – Need to provide some robustness to residual errors.**
- **Total delay – Depends on the applications and may be used to trade-off with quality.**
- **Format flexibility – E.g., it should be possible to use different spatial and temporal resolutions.**
- **Cost – Especially the decoders must have an acceptable (low) cost.**

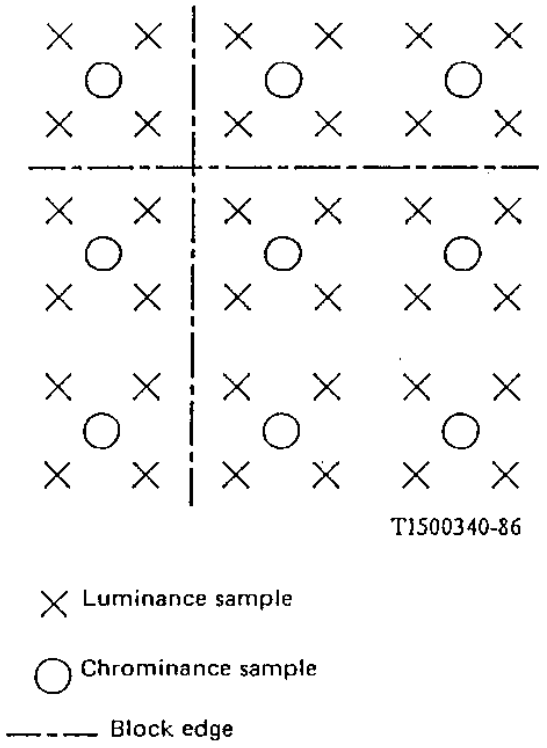
MPEG-1 Video: Objective

Efficient coding of video information with a minimum acceptable quality with bitrates up to about 1.2 Mbit/s (only video); other rates may also be used.

THE FIGHT: The target quality for CD-ROM storage is the quality associated with VHS tapes, targeting the substitution of the analogue storage with digital storage.

MPEG-1 Video: Signals to Code

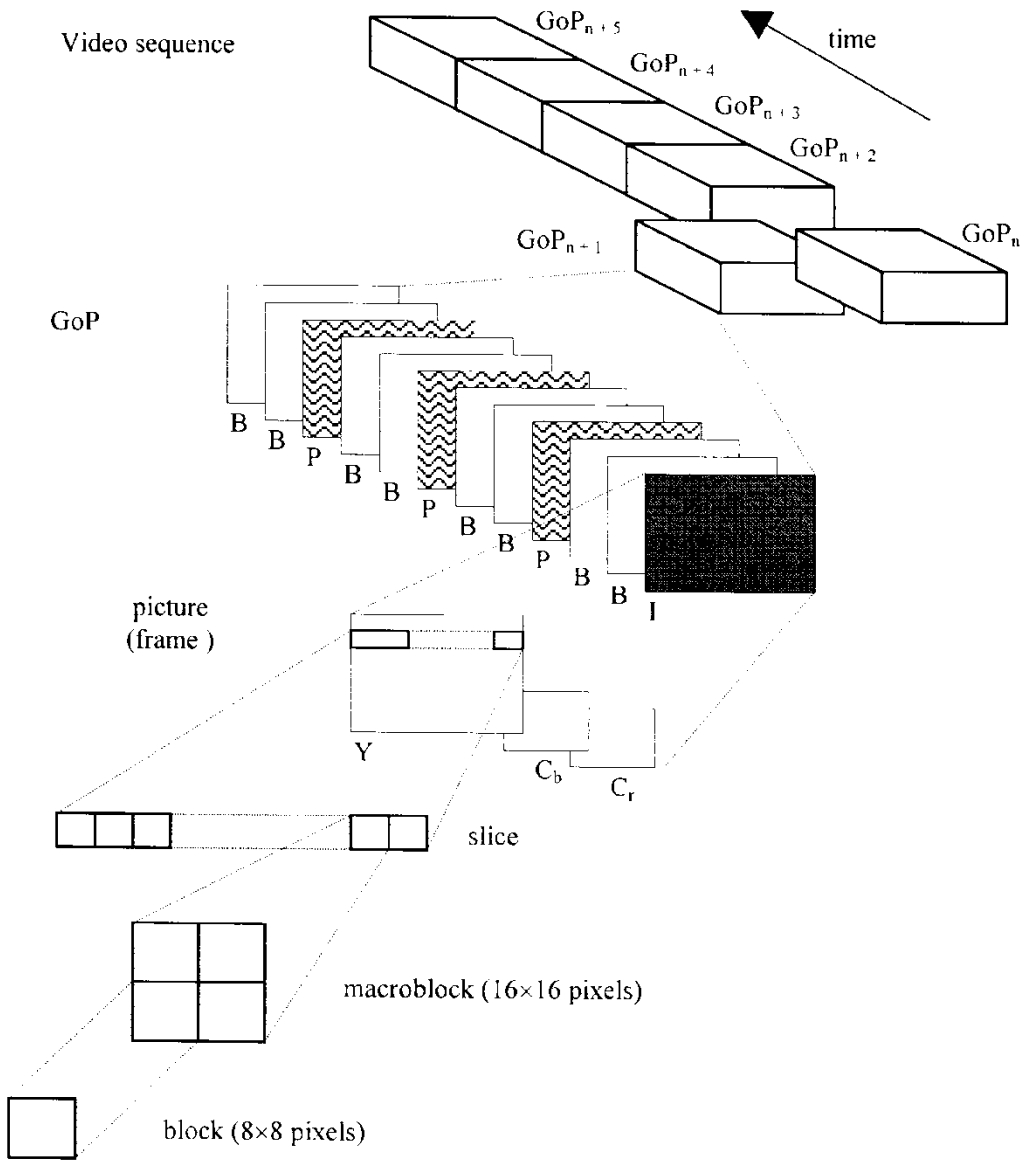
- **Signals -** The signals for each image are the luminance (Y) and two chrominances, designated as C_B and C_R or U and V. The R,G,B primary signals have a gamma correction around 2.2 - 2.8.
- **Resolution -** Luminance as twice the number of rows and columns of the chrominance; this means a 4:2:0 subsampling format is used considering the lower human visual sensibility to colour.
- **Bit depth -** Samples are quantized according to Recommendation ITU-R BT.601 this means with 8 bit/sample.



Video Structure

Spatially, the video data is organized in a hierarchical structure with 5 layers:

- **Sequence**
- **Group of Pictures (GOP)**
- **Picture**
- **Slice**
- **Macroblock (MB)**
- **Block**



MPEG-1 Video: Coding Tools

- **Temporal Redundancy**

Predictive coding: temporal differences and motion compensation (uni and bidirectional; ½ pixel accuracy)

- **Spatial Redundancy**

Discrete Cosine Transform (DCT)

- **Statistical Redundancy**

Huffman entropy coding

- **Irrelevancy**

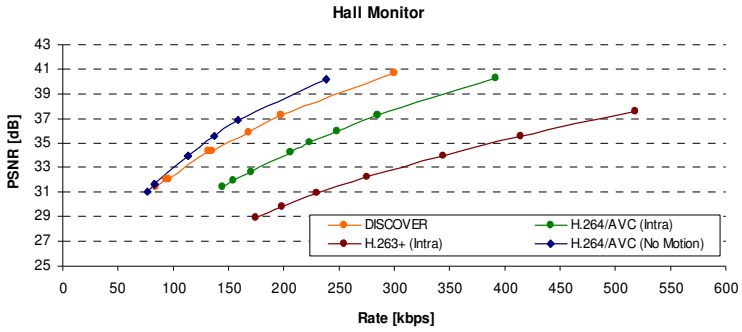
DCT coefficients quantization

Exploiting the Temporal Redundancy

Temporal Prediction with Motion Compensation

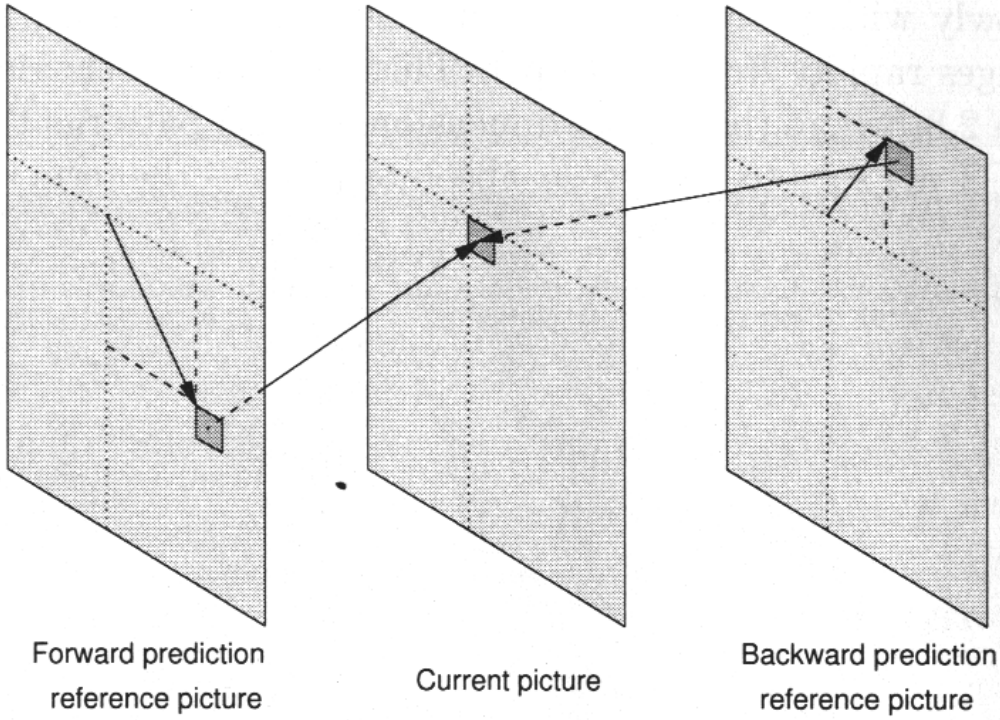
- **FORWARD PREDICTION – It is based on the principle that, locally, each image (or part thereof) may be represented from one or more previous images after a translation.**
- **BACKWARD PREDICTION – It is based on the principle that, locally, each image (or part thereof) may be represented from one or more future images after a translation.**
- **BIDIRECCIONAL PREDICTION – It is based on the principle that, locally, each image (or part thereof) may be represented from a previous image (forward prediction), a future image (backward prediction), or a combination thereof, after corresponding translations.**

As for H.261, the quality of the temporal prediction strongly determines the video codec RD performance since its defines the energy of the difference signal this means the prediction error.



The Future is so Close …

- **Bidirectional predictions 'buy' quality with delay and complexity !**
- **This is possible especially if the application can accept the additional delay …**

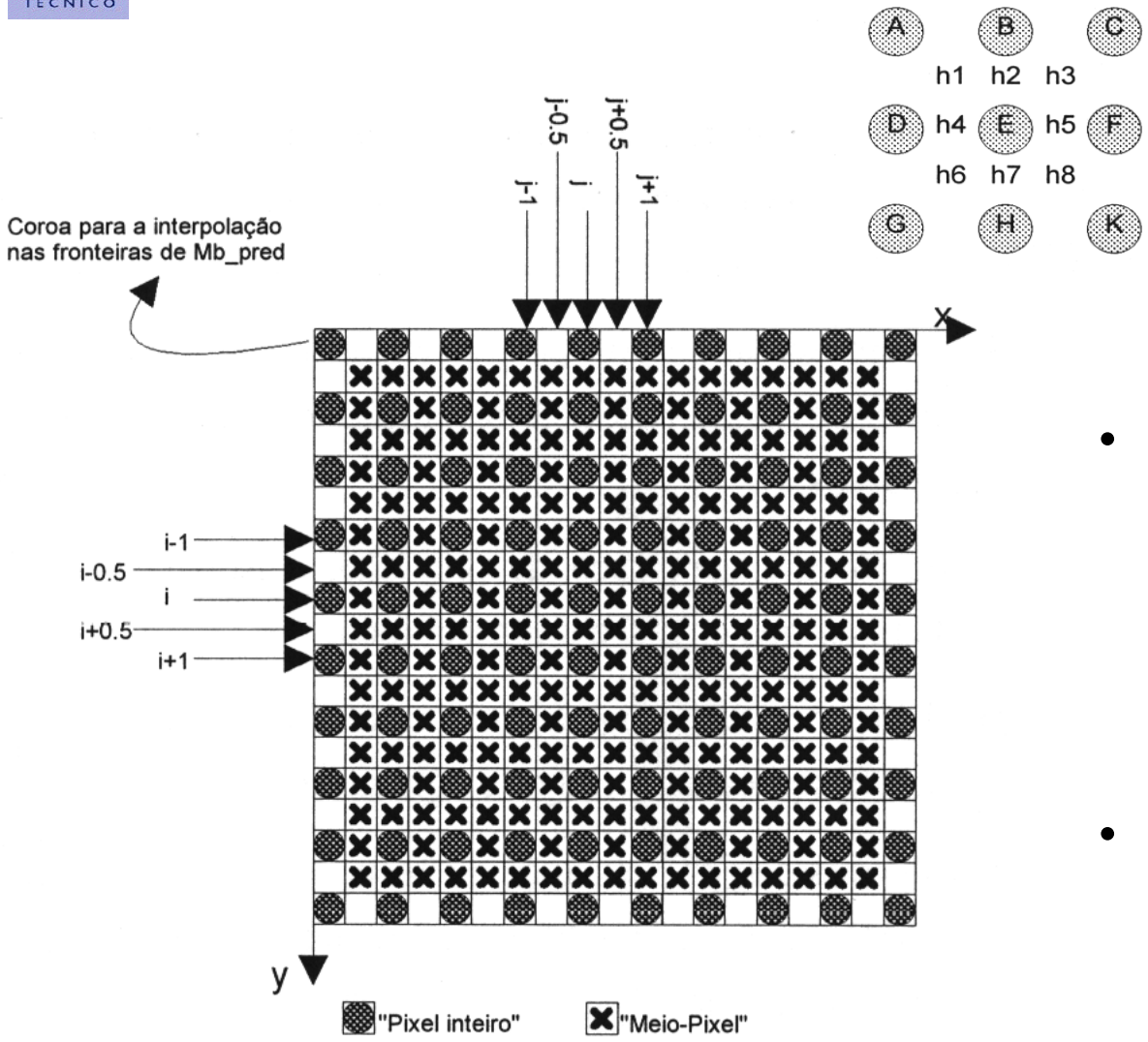


Bidirectional prediction allows to reach better predictions (there is more information available to make the prediction) and, thus, to reach better RD performance for certain conditions; for example, it is useful to deal with uncovered backgrounds.

MPEG-1 Video: Motion Compensation

- **Motion estimation and compensation is performed at the macroblock level.**
- **Motion estimation and compensation are always optional, meaning that the encoder may decide to use it or nor (independently of this being a good or bad decision).**
- **Motion estimation is performed at the encoder and, thus, it is not normative ! There are hundreds of ways of doing motion estimation !**
- **Motion estimation implies a high complexity, thus justifying the need for fast (non full search) motion estimation algorithms.**
- **However, since the bitstream syntax allows to send one or two motion vectors per MB, block matching motion estimation is one of the most used solutions.**
- **Bidirectional prediction cannot be applied to all frames of a sequence since otherwise the delay would be too high; thus, there is a need to define relatively close prediction anchors (which do not predict from the future).**

½ Pixel Motion Estimation and Compensation



- **The ½ pixel motion estimation accuracy allows a more precise estimation of the motion with the consequent reduction of the prediction error.**
- **Its usage is signalled at the frame level.**

Exploiting the Spatial Redundancy and the Irrelevancy

After Time, the Space …

Actual image

Prediction image,
motion compensated

+

+

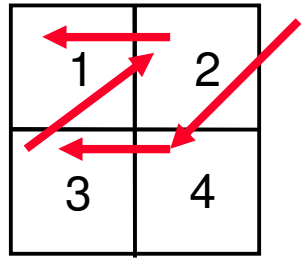
-

Prediction error

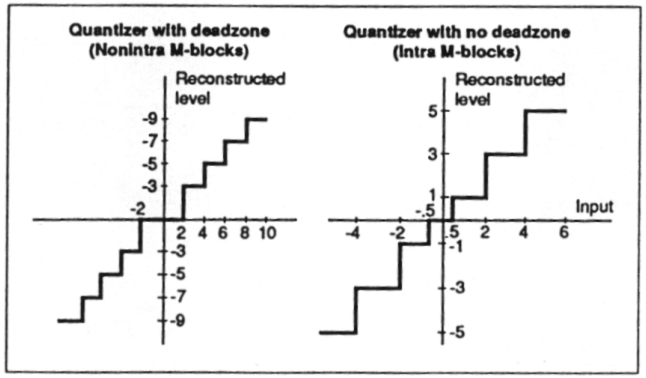
DCT
Transform

MPEG-1 Video: How is the DCT Applied ...

- **The DCT is applied to 8×8 blocks of samples (N=8).**
- **The inverse DCT (IDCT) precision is controlled as in H.261, e.g. the mismatch pixel error has to be always lower or equal to 1.**
- **The DCT coefficients to transmit are selected using non-normative thresholds, allowing the considering of psychovisual criteria to optimize the final subjective impact for different types of content and applications.**
- **The quantized DCT coefficients in each block are zig-zag scanned in order to assure that they are transmitted according to their (decreasing) subjective relevance.**
- **The DC coefficients are differentially coded within each MB and between neighbour MBs (left to right and top-down).**



MPEG-1 Video: Quantization



- **MPEG-1 Video assumes the usage of uniform quantization with dead zone for the Inter MBs and without dead zone for the Intra MBs.**
- **The quantization step is determined through the quantization matrix and the quantization factor; it may be different for each DCT coefficient.**
- **The quantization steps may be changed at any MB.**
- **The standardized quantization matrix is different for Intra and Inter coded MBs. These matrices may be changed to more adequate matrices for the cases at hand, of course paying the necessary bitrate cost.**
- **For Intra coded MBs, the DC coefficient is always quantized with step 8.**

Default Quantization Matrices

8	16	19	22	26	27	29	34
16	16	22	24	27	29	34	37
19	22	26	27	29	34	34	38
22	22	26	27	29	34	37	40
22	26	27	29	32	35	40	48
26	27	29	32	35	40	48	58
26	27	29	34	38	46	56	69
27	29	35	38	46	56	69	83

INTRA

16	16	16	16	16	16	16	16
16	16	16	16	16	16	16	16
16	16	16	16	16	16	16	16
16	16	16	16	16	16	16	16
16	16	16	16	16	16	16	16
16	16	16	16	16	16	16	16
16	16	16	16	16	16	16	16
16	16	16	16	16	16	16	16

INTER

For Inter coding, the high frequency coefficients are not necessarily associated to high frequency image content since they can result from block effect in the reference image, poor motion compensation, or camera noise.

Exploiting the Statistical Redundancy

Combining the Tools ...

MPEG-1 Video: Coding Tools

- **Temporal Redundancy**

Predictive coding: temporal differences and motion compensation (uni and bidirectional; ½ pixel accuracy)

- **Spatial Redundancy**

Discrete Cosine Transform (DCT)

- **Statistical Redundancy**

Huffman entropy coding

- **Irrelevancy**

DCT coefficients quantization

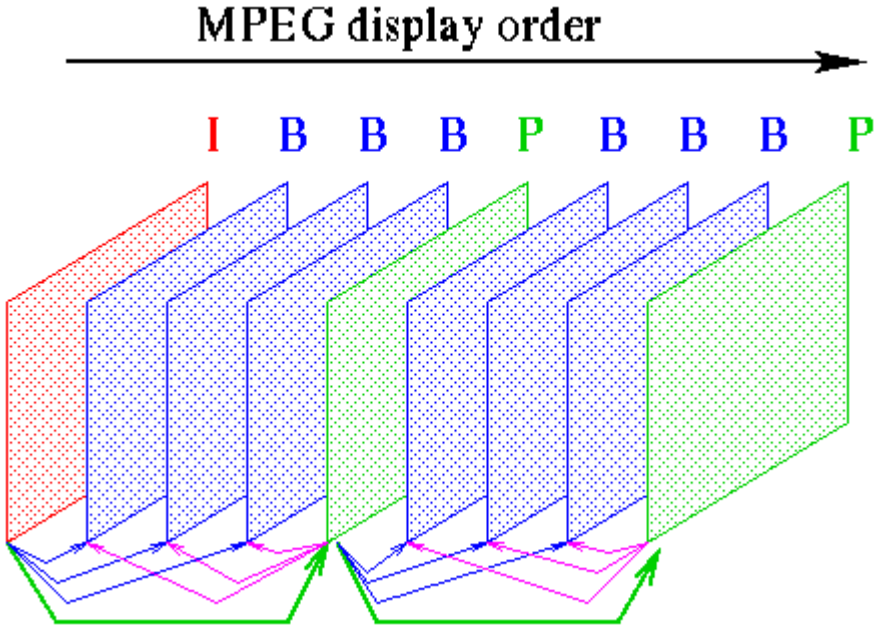
The Functional Sandwich …

Compression Efficiency

The CODEC !

Random Access

Temporal Prediction Structure



Open GOP uses reference pictures from the previous GOP at the current GOP boundary.
There is a flag to signal open and *closed* GOPs !

The "conflict" between coding efficiency and random access led to the definition of 3 frame types depending on the coding tools used:

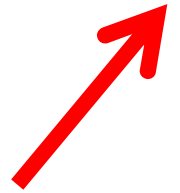
- **Random access: Intra frames (I) – Don't use temporal predictions**
- **Compression efficiency:**
  - **Predicted frames (P) – May only use *forward* prediction from previous I/P frame**
  - **Bidirectionally predicted frames (B) – May use both forward and backward prediction from previous and future I/P frame**

The Frame Level Syntactical Restrictions

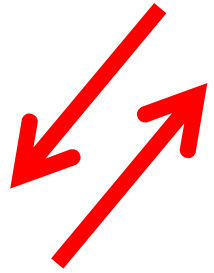
Motion

- **I (INTRA CODED) FRAMES – All MBs in I frames are Intra codec blocks; no temporal predictions are allowed at all and, thus, no temporal redundancy is exploited making these frames rather expensive in rate to achieve a target quality.**

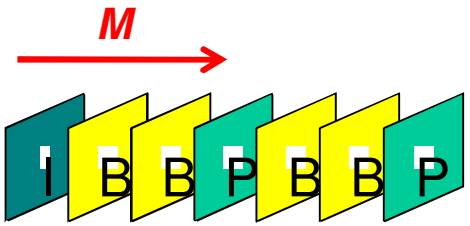
- **P FRAMES – MBs is P frames MAY use backward prediction, this means a prediction from a past frame, with or without a motion vector; for MBs in P frames, only the previous P/I frame may be used as forward prediction.**



- **B FRAMES – MBs in B frames MAY use backward, forward or bidirectional prediction (average prediction from a past and a future frame), with or without motion vector(s); for MBs in B frames, the previous P/I and next P/I frames may be used as forward and backward predictions.**



Temporal Prediction Structure



- The temporal prediction structure is rather flexible and may depend on the content or application.
- A good solution is to insert temporal anchors (I and P frames) about every 0.1 s using a combination like

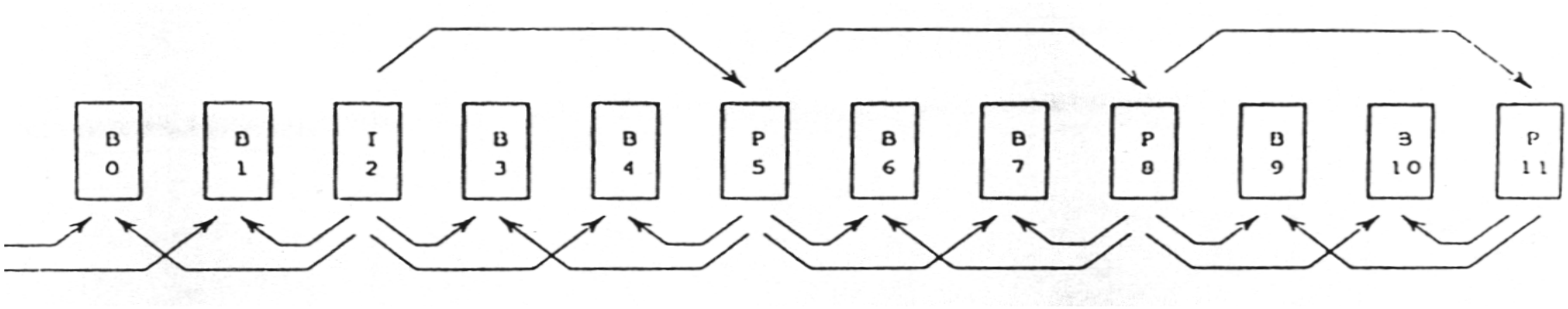
… I B B P B B P B B P B B I B B P B …

- If a regular prediction structure is used
  - N (GOP size) – number of frames between two I frames + 1
  - M – number of B frames between two anchors (I or P frames) + 1

this means that N is always a multiple of M; M and N are not syntactic elements in the MPEG-1 Video bitstream.

The Order is … Out of Order …

Since B frame decoding may only be made after receiving and decoding the corresponding anchor frames, the transmission of I and P frames out of the natural acquisition and visualization orders is inevitable ! This introduces an additional delay …



I	B	B	P	B	B	P	B	B	P	B	B	I	B	B	P	B	B	P
0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18

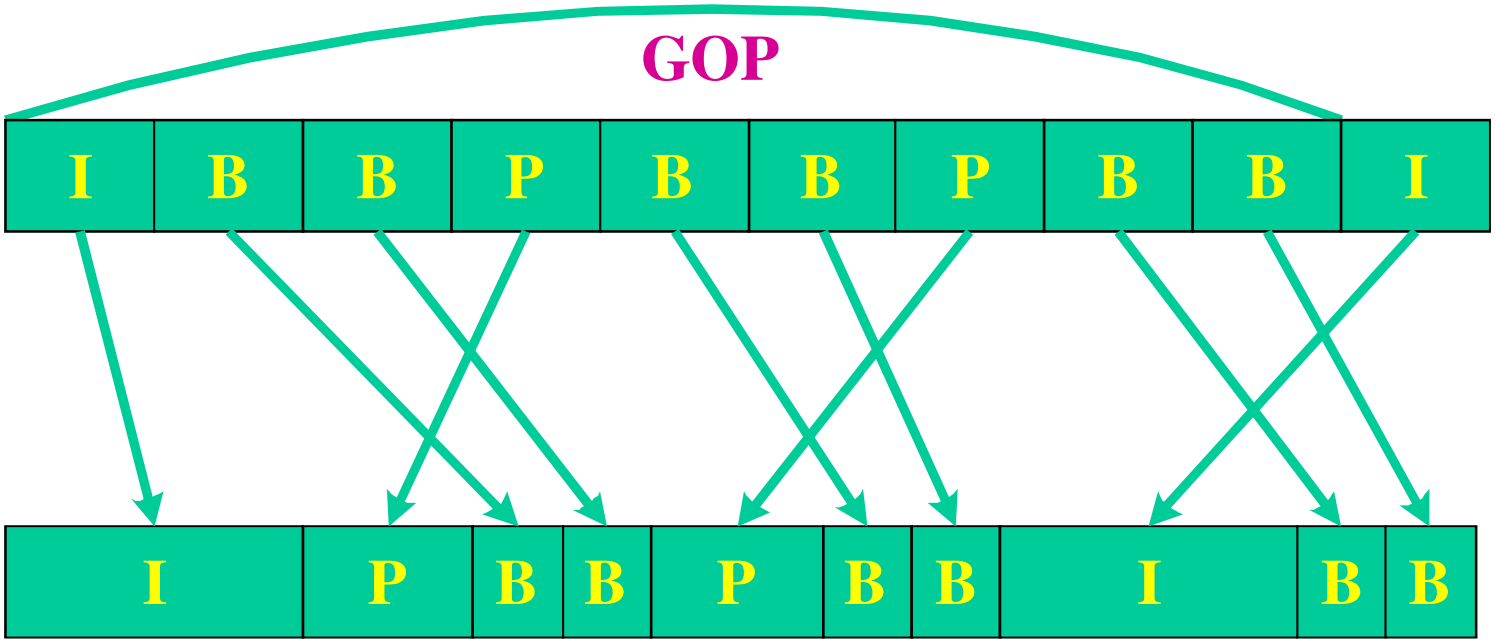
Acquisition and visualization

I	P	B	B	P	B	B	P	B	B	I	B	B	P	B	B	P	B	B
0	3	1	2	6	4	5	9	7	8	12	10	11	15	13	14	18	16	17

Transmission

Putting Order in the Orders …

Acquisition and Visualization Orders



Coding, Transmission and Decoding Orders

Constant Quality: How ?

- Users enjoy content which is shown (and thus coded) with rather constant quality this means without noticeable quality variations along time and space.
- The need for random access led to the definition of 3 frame types depending on the used coding tools which have different compression powers.
- Since the uniform allocation of bitrate resources to the various frames would lead to noticeable quality variations in time, there is the need to non-uniformly allocate the bitrate resources depending on the compression power of the coding tools used for each frame.
- Experience has shown that for M=2-3, good results are achieved attributing similar quality to the I and P frames and a slightly lower quality to the B frames.

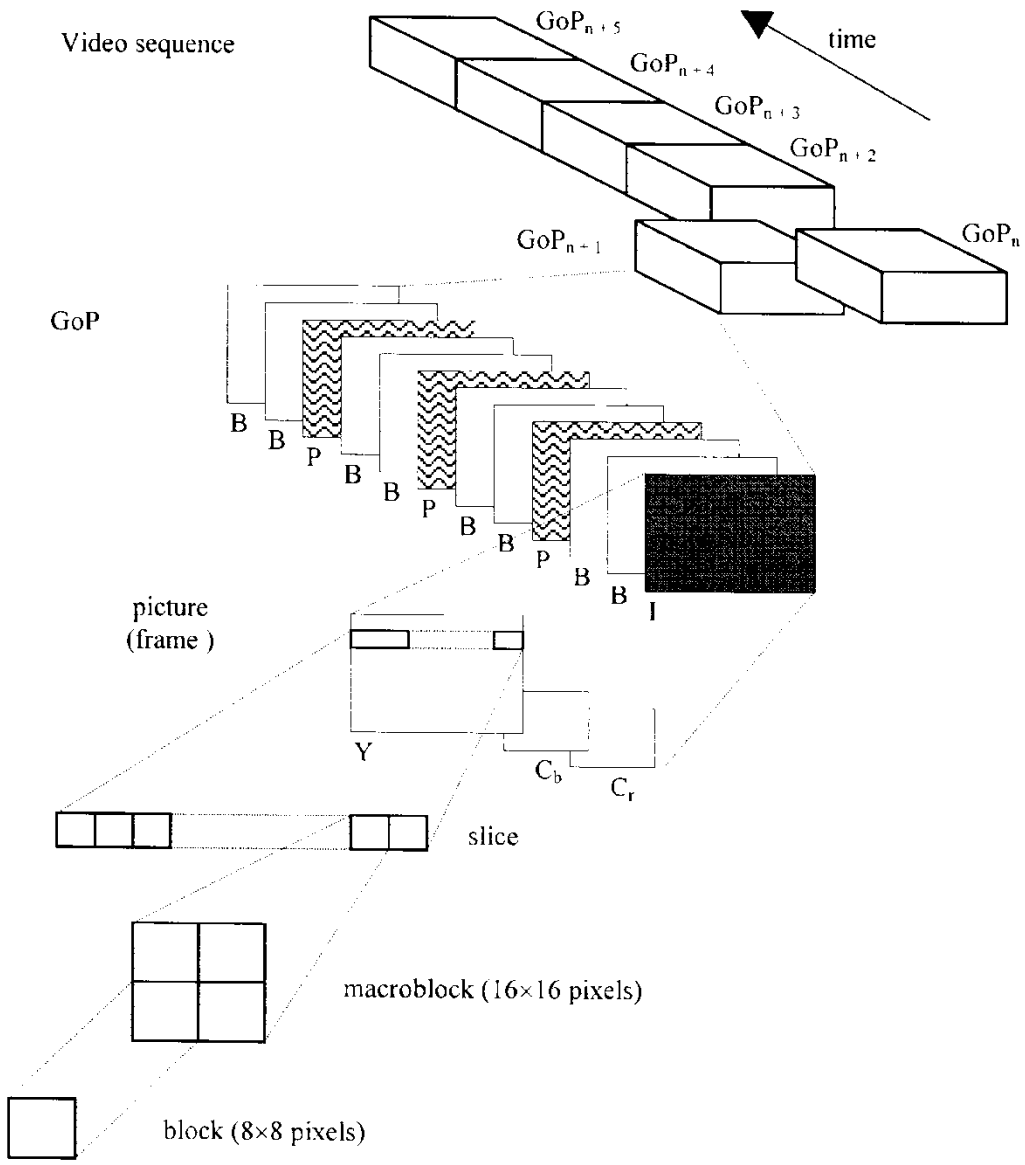
Who Does Take the Best Part ?

- **The ideal allocation of resources among the various frames depends on the specific video content; however, the following type of distribution typically leads to good quality results for natural images:**
  - **P frames with 2-5 times more bits than B frames**
  - **I frames with up to 3 times more bits than P frames**
  - **For low motion, more bits must be allocated to the I frames**
  - **For high motion, the proportion of I frames bits must be reduced passing these savings to the P frames**
- **These rules should only be taken as a starting point; the final bitrate allocation must be performed by the bitrate control method depending on the dynamic characteristics of the video frames.**

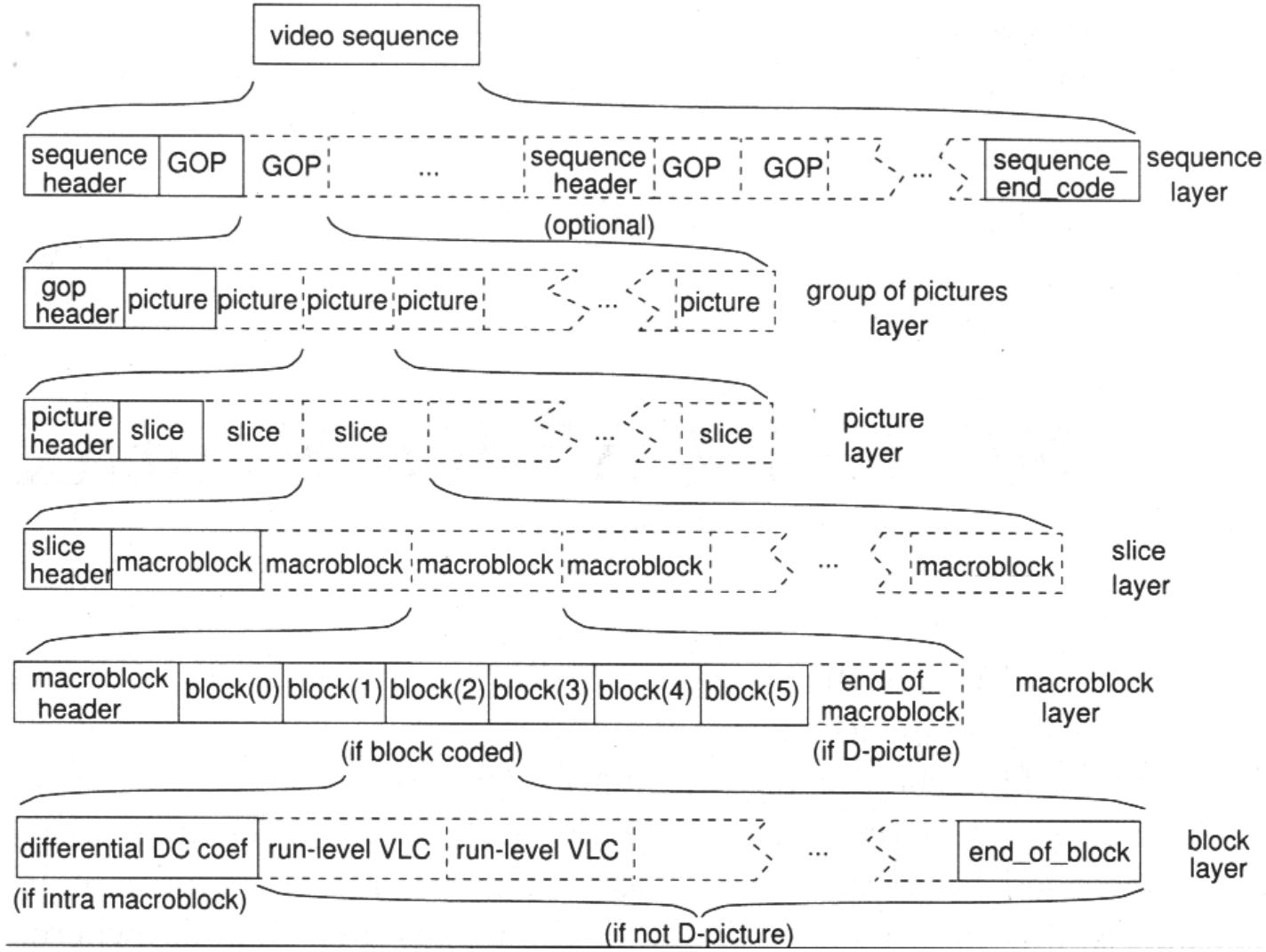
Video Structure

The video data is organized in a hierarchical structure with 5 layers:

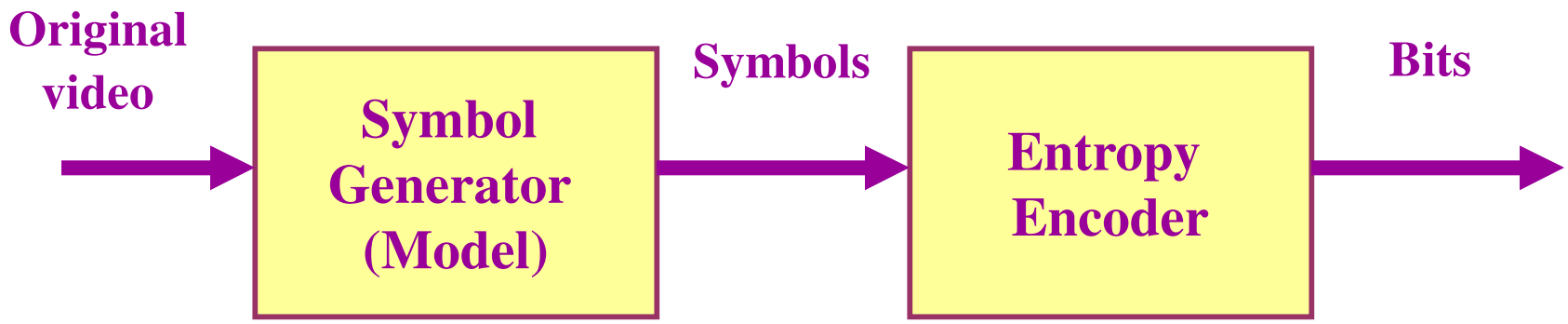
- **Sequence**
- **Group of Pictures (GOP)**
- **Picture**
- **Slice (more flexible than H.261 GOBs)**
- **Macroblock (MB)**
- **Block**



MPEG-1 Video Syntax

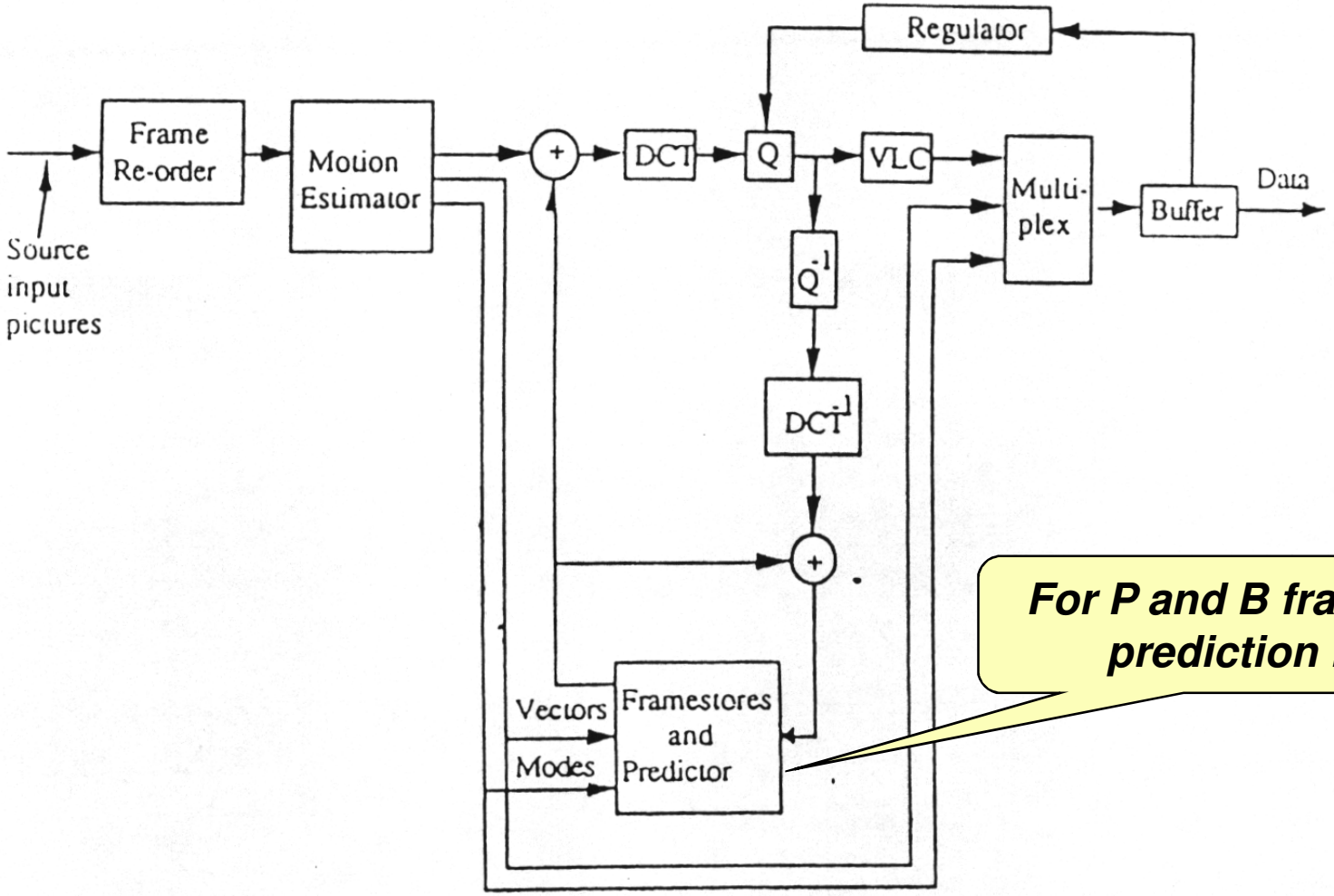


The MPEG-1 Video Symbolic Model

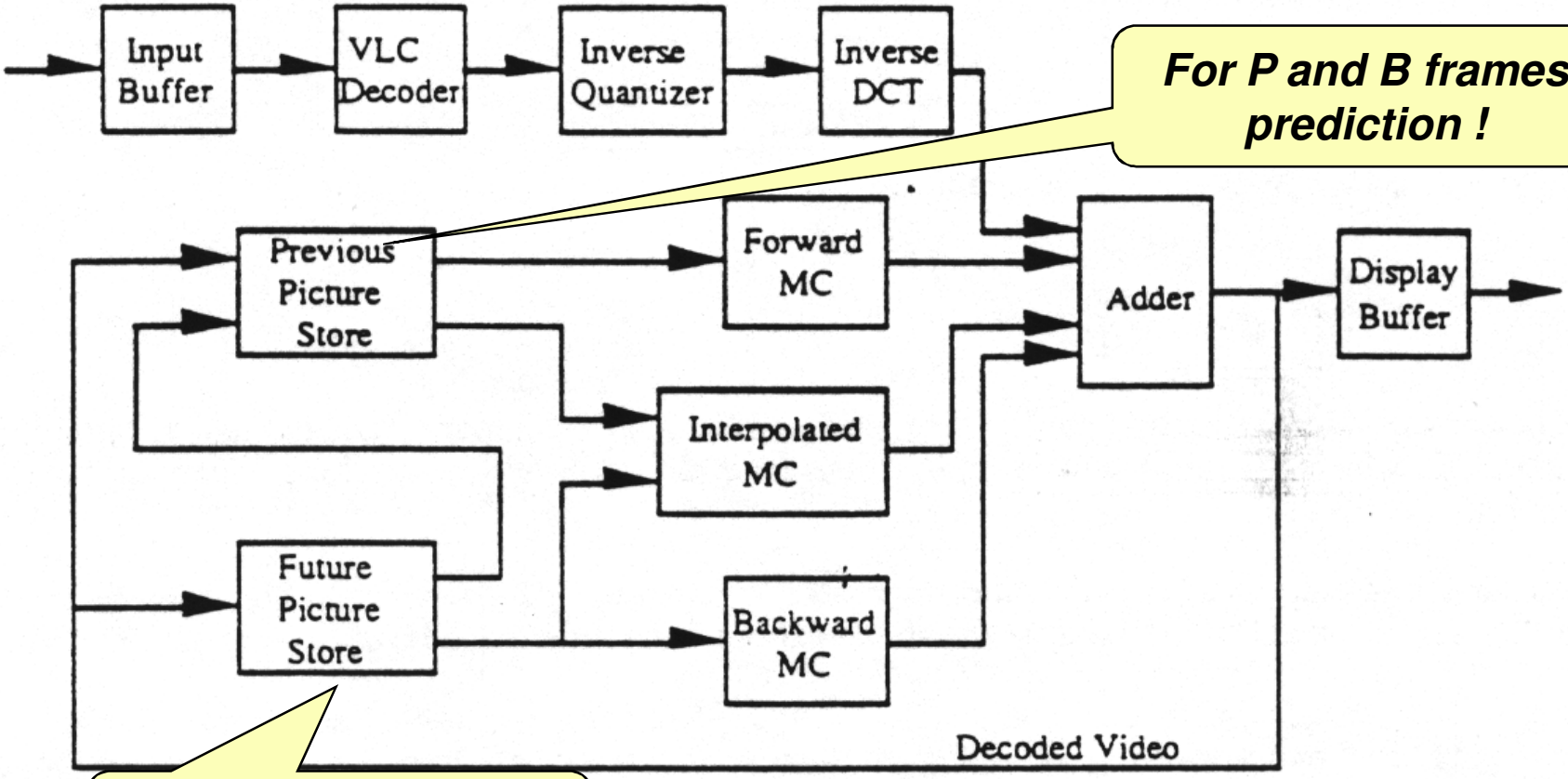


A video sequence is represented as a succession of GOPs, including I, P and B coded frames, each structured in macroblocks, coded using motion vector(s) and/or DCT coefficients, following the constraints set by the frame coding type (I, P or B).

MPEG-1 Video: Encoder



MPEG-1 Video: Decoder



MPEG-1 Video and H.261: Which Relationship ?

VERY INTIMATE … but …

- **H.261 targets real-time applications with a maximum delay around 150-200 ms.**
- **MPEG-1 Video does not have strong delay requirements since it mainly targets storage applications.**
- **MPEG-1 Video must offer all the typical functionalities already available in analogue video storage systems.**
- **MPEG-1 Video is optimized for higher bitrates.**

There is the highest possible technical compatibility between MPEG-1 Video and H.261 to facilitate the simultaneous implementation of both codecs in certain systems.

The Output Buffer …

The bitrate production by the encoder is highly non-uniform in space and time, essentially due to:

- The classification of the frames as I, P and B which implies the usage of coding tools with different compression power
- The allocation of different bitrate resources to every frame to compensate the usage of different coding tools
- The spatial variation of activity within the various image areas
- The temporal variation of activity along time
- The entropy coding of the generated symbols

To make the variable bitrate produced by the encoder 'compatible' with the constant bitrate drained by the channel, an output buffer must be used !

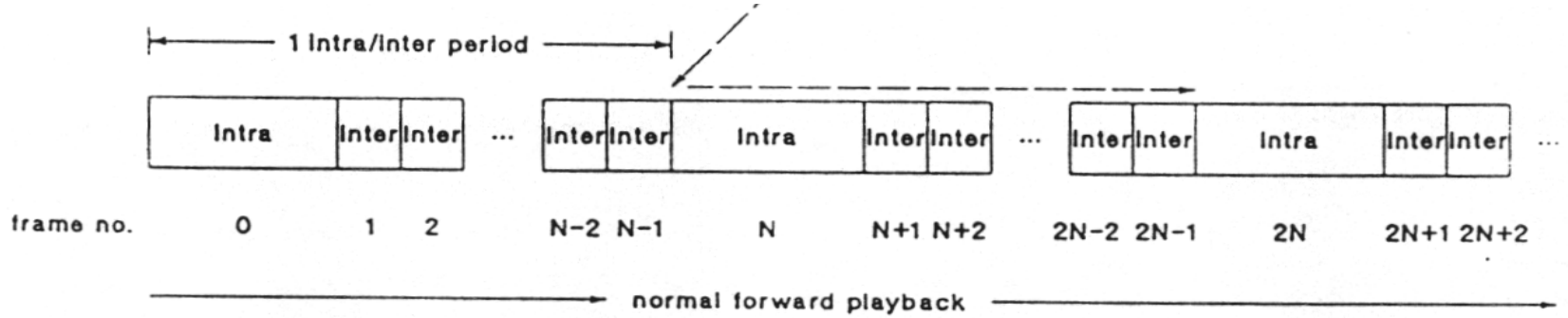
Rate Control … and Offline Encoding ...

- **The encoder must control the produced bitrate along time and within each image in order to reach the best overall subjective quality with the available resources.**
- **While the encoder has the mission to take important decisions, the decoder is a 'slave' limiting itself to follow the 'orders sent by the encoder – the 'boss'.**

For the most important MPEG-1 applications, encoding may be performed off-line (taking whatever time, iterative encoding, multiple passes, etc), thus achieving much higher quality than real-time encoding for similar bitrate resources.

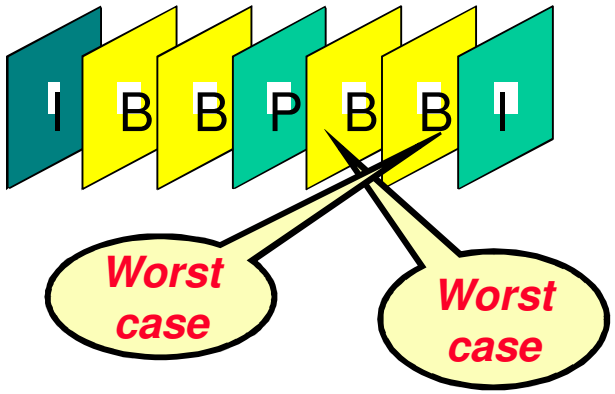
Especial Access Modes

Random Access …



- **The random access facility allows to access – read the bits, decode, and visualize - any video frame within a, as small as possible, limited time, typically around 0.5 s; this imposes the usage of anchor frames like I frames.**
- **The storage device has an address table which allows the fast access to all reference frames, this means I frames; from those I frames, reading proceeds towards the target frame.**

Maximum Random Access Time



For the CD-ROM, the Maximum Random Access Time (MRAT) depends on the allocation of bits among the various types of frames, the frame rate and the time between I frames:

$$MRAT = T_{DSM} + [\ 2\ BF_I + (N/M-1)\ BF_P + 1\ BF_B\ ] \times (1/fs)\ \ (s)$$

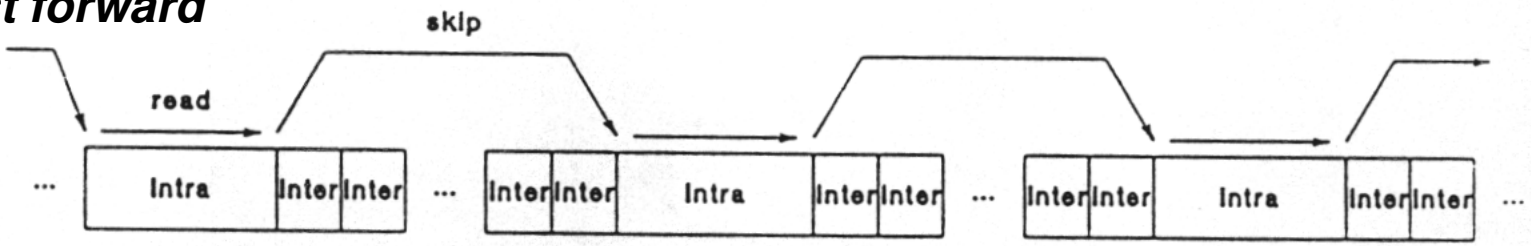
- BF_X **is the maximum frame peak factor for the bits spent in all X type frames**
- T_{DSM} **is the sum of the various access times due to the need to jump in the CD-ROM to read only the strictly necessary bits.**

Fast Forward and Fast Reverse

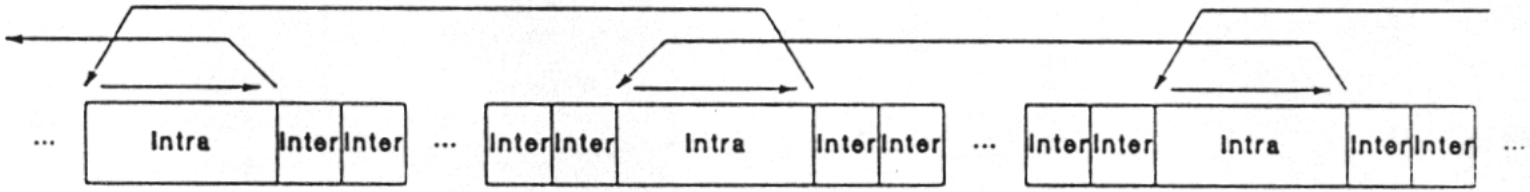
- MPEG-1 offers fast modes with a speed up factor between 8 and 10 which means the video information corresponding to 1 s must be, on average, visualized in 0.1 to 0.125 s.
- This fast modes have to be based on I and/or P frames depending on the temporal prediction structure, notably the M and N values for regular structures.
- The most basic limitation is imposed by the reading speed which limits the number of frames that may be read in a certain time, especially those with the lowest compression factors, this means I and P frames.

Fast Forward and Fast Reverse: Examples

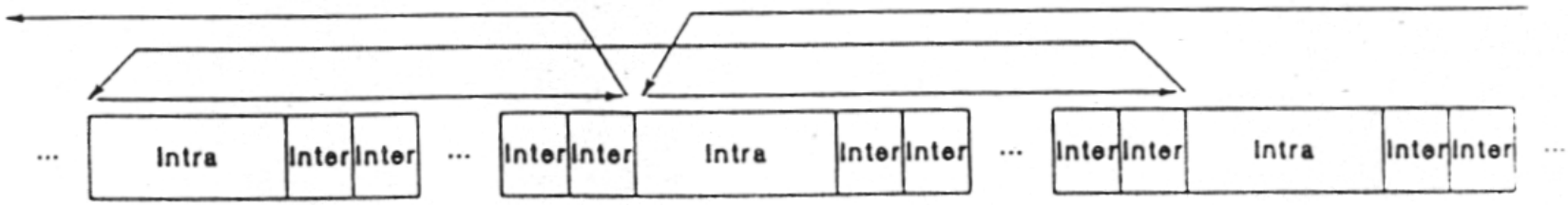
Fast forward



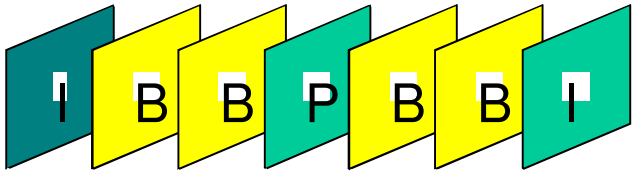
Fast reverse



Normal Reverse: Example



Speed Up Factor



The *Speed Up Factor* (SUF) is computed as the ratio between the 'real' time corresponding to the frames read and their corresponding 'visualization' time.

- **If only I and P frames are read:**

$$SUF = \langle \text{real time} \rangle / \langle \text{visualization time} \rangle = [(K+1)M] / V$$

  - ***K*** **is the number of I and P frames skipped between each one read (if P frames are read, no P frames may be skipped)**
  - ***V*** **is the number of frame periods for the visualization of each frame read**

- **When only I frames are used, it comes K= N/M-1.**

Fast Forward and Reverse: Selecting the Frames

Since the reading speed (constant or peak, if they are different) limits the number of frames that may be read during the time each frame is visualized, there is a minimum value for this visualization time in order the next frame to be visualized may be ready.

- **As the time to visualize a frame has to be higher than the time to access the next frame to be visualized, which is a I frame in the worst case, it comes:**

$$V/f_s \geq BF_I/f_s + T_{DSM} \Rightarrow\Rightarrow V \geq f_d \times (BF_I/f_s + T_{DSM})$$

  - V is the number of frame periods (integer and positive) each frame is visualized
  - T_{DSM} corresponds to the total device access time (for the CD-ROM it includes the time to jump between access points which depends on their number, latency, angular speed, and protocol)

If M= 4, N= 24 and BF_I = 3, it comes V ≥ 6 and SUF= 8, if only I frames are used.

MPEG-1 Video: Constrained Parameters (1)

- **MPEG-1 Video offers great flexibility for the video sequence parameters (included in the bitstream), accepting a large range of spatial and temporal resolutions, aspect ratios and bitrates.**
- **Since it is important to avoid forcing the manufacturers to produce equipment which is unnecessarily complex to guarantee interoperability, a set of values for the basic coding parameters has been defined in the standard, allowing to create the so-called *Constrained Parameters Bitstreams*.**
- **The *constrained parameters* guide (and constraint) the product manufacturers and content producers since all MPEG-1 Video decoders must be able to decode *Constrained Parameters Bitstreams.***
- **However, bitstreams using other parameters may be created: a flag in the bitstream signals if the bitstream follows or not the limitations imposed by the *constrained parameters*.**

MPEG-1 Video: Constrained Parameters (2)

Horizontal picture size	Less than or equal to 768 pels
Vertical picture size	Less than or equal to 576 lines
Picture area	Less than or equal to 396 macroblocks
Pel rate	Less than or equal to 396x25 macroblocks per second
Picture rate	Less than or equal to 30 Hz
Motion vector range	Less than -64 to +63.5 pels (using half-pel vectors) [backward_f_code and forward_f_code <= 4
Input buffer size (in VBV model)	Less than or equal to 327 680 bits
Bitrate	Less than or equal to 1 856 000 bits/second (constant bitrate)

VBV in bytes not bits !

Error Concealment

If MPEG-1 Video coded content is used for transmission over error prone channels, the decoder should:

- **Detect the residual errors at syntactic and semantic levels**
- **Minimize the negative subjective effect of the transmission errors by applying (non-normative) error concealment methods such as:**
  - **Substituting the corrupted image areas with the co-located areas from a previous frame, ☹**
  - **Substituting the corrupted areas with the motion compensated areas from a previous frame, ☺**

MPEG-1 Video: Final Remarks

- **MPEG-1 Video is one of the most common formats for digital video in PCs, e.g. Windows has a MPEG-1 Video player (software).**
- **A large share of the digital video in the Internet is in the MPEG-1 Video format.**
- **There are many products and services based on the MPEG-1 Video standard, notably video cameras; however, MPEG-1 is not anymore the state-of-the-art in terms of video coding for entertainment content …**
- ***Video CD* is based on MPEG-1 and sold hundreds of millions of players in China.**

VICTOR IV

MPEG-1 Standard

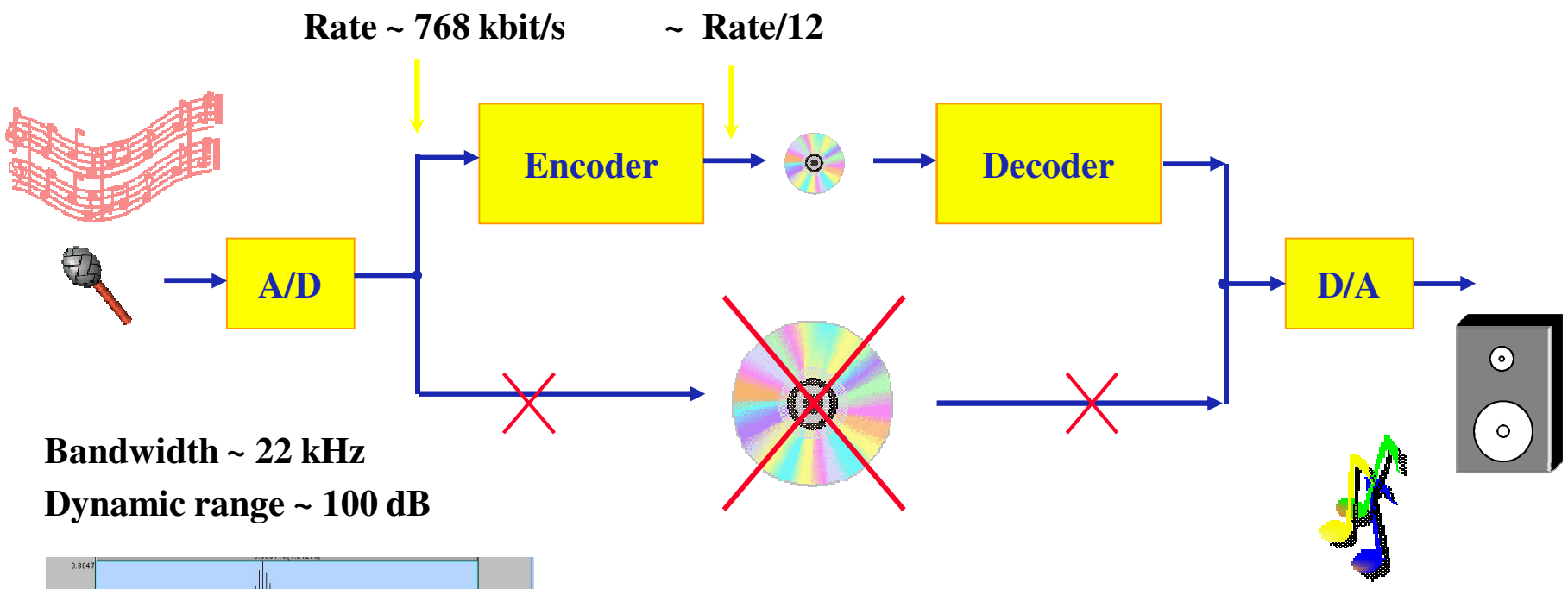
Part 3: Audio

MPEG-1 Audio: Objective

Efficient audio coding, mono and stereo, with high quality, at 32-448 kbit/s per channel, using sampling rates of 32, 44.1 and 48 kHz, and targeting digital audiovisual storage at an overall rate of 1.5 Mbit/s.

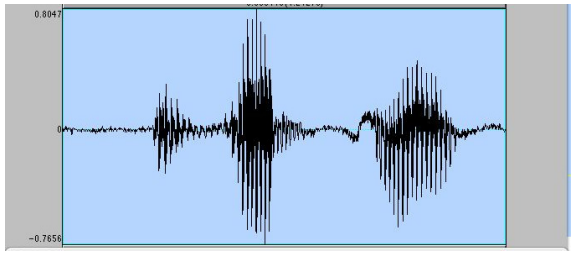
The target audio RD performance is the CD-ROM (PCM) quality at 256 kbit/s, for stereo content.

The Audio Compression Chain …



Bandwidth ~ 22 kHz

Dynamic range ~ 100 dB



MPEG-1 Audio: Applications

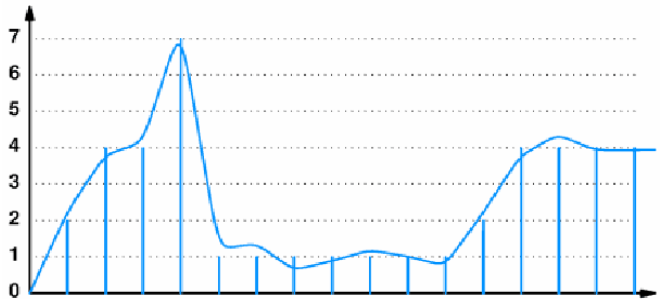
- Audio production
- Audio distribution and sharing
- Internet streaming
- Portable audio
- Audio archival
- Digital radio and television (DAB and DVB)
- Digital audio storage
- Multiple multimedia applications
- ….

PCM for Various Sound Signals …

	Frequency (Hz)	Sampling rate (kHz)	bit/sample (PCM)	PCM bitrate (kbit/s)
Speech (telephone)	300-3400	8	8	64
Speech (wideband)	50-7000	16	8	128
Audio (medium band)	10-11000	24	16	384
Audio (wideband)	10-22000	48	16	768

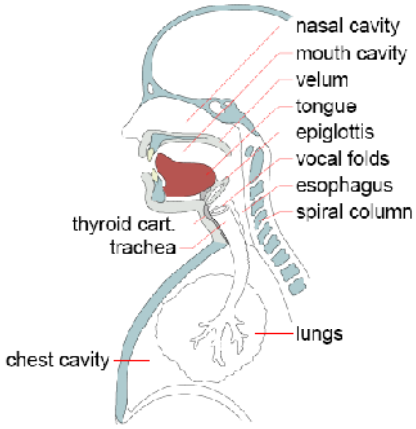
MPEG-1 Audio: Requirements

- High signal quality independently of the spectral and amplitude characteristics of the signal to be coded
- Low encoding and decoding delays
- Spatial integrity for stereo and multichannel signals
- Error resilience to uniform and burst errors and packet losses
- Graceful degradation for higher error probabilities and loss rates
- Resilience to cascading, i.e. successive coding and decoding processes
- Capability to edit, mix, etc.
- Low implementation complexity
- Low energy consumption



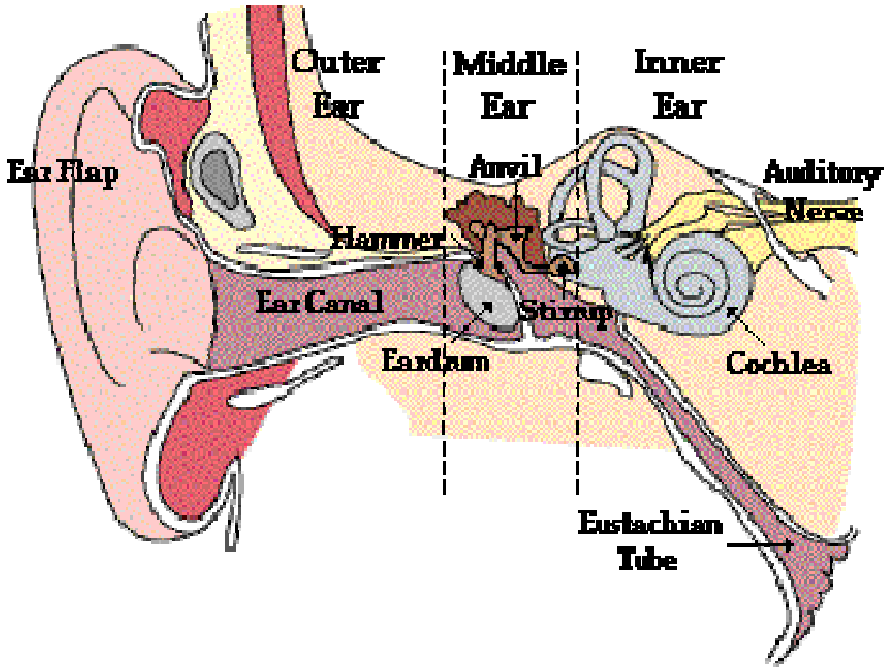
Audio Coding Peculiarities …

- **Very high dynamic range (ratio between the maximum and minimum amplitudes) and larger bandwidth in comparison with speech.**
- **Absence of a universal source production model; the existence for speech of this model allows reaching much higher compression factors.**
- **Certain simplifying assumptions usually adopted for speech coding are not valid anymore such as:**
  - **Gaussianity**
  - **Stationarity**
  - **Spectral smoothness**



Compression gains for high quality audio coding mainly result from irrelevancy reduction (since redundancy is short …).

Human Auditory System

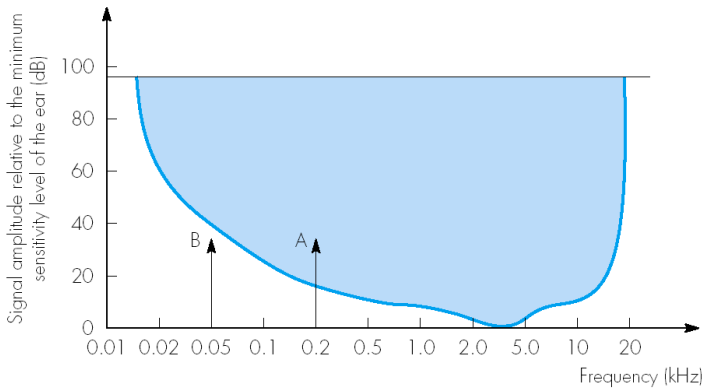


The ear has 3 main sections:
1) Outer ear – Directs the sound to the eardrum.
2) Middle ear – Transforms the sound pressure into mechanical vibration.
3) Inner ear – Converts these mechanical vibrations into excitations of the auditory nerves which send electrical signals to the brain.

- The perception of audio quality depends on the Human Auditory System (HAS).
- The Human Auditory System processing includes physiological and psychological effects.

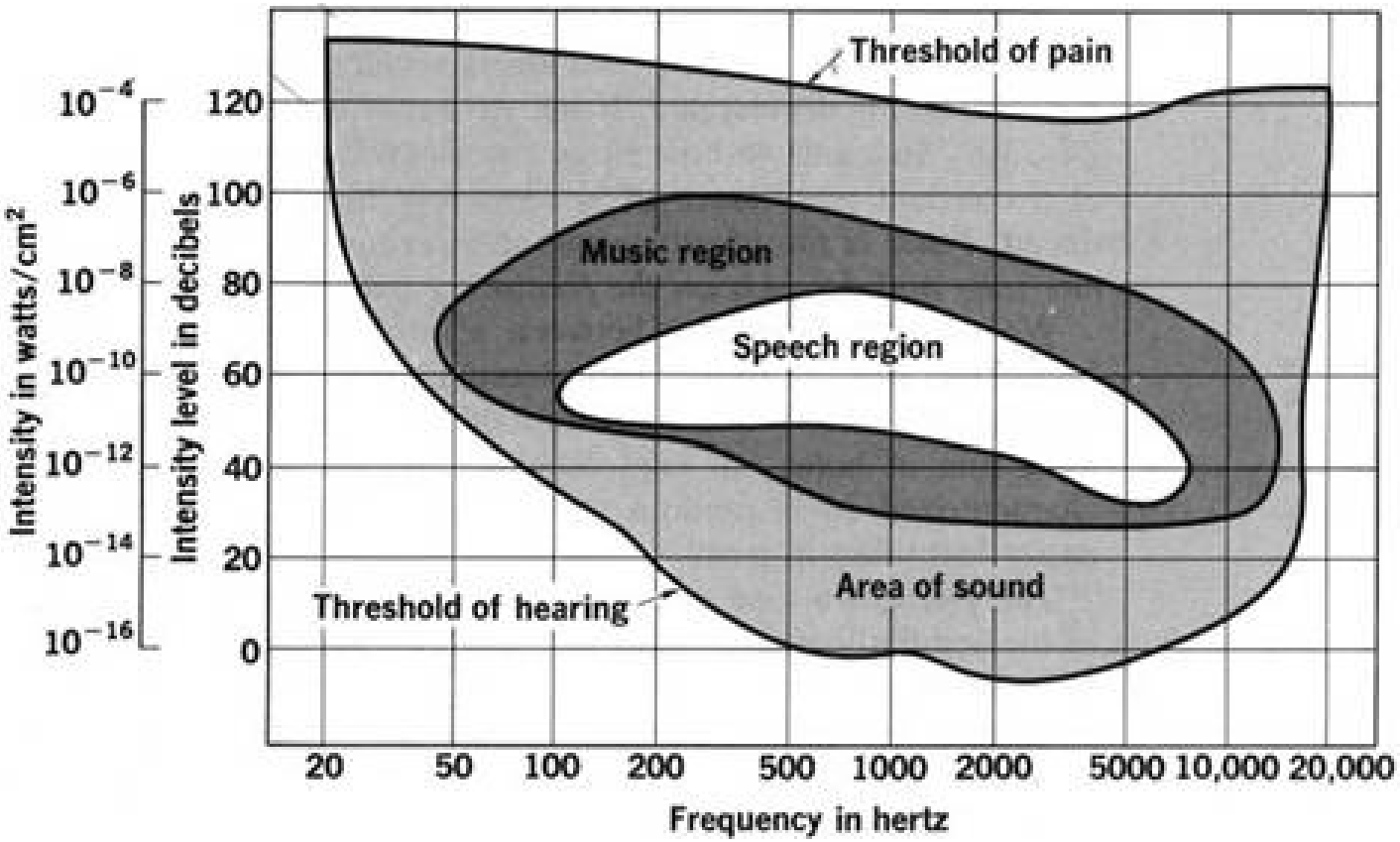
Physiological Effects: the Thresholds

- **Threshold of Hearing – Defines the minimum sound intensity which may be perceived; this threshold varies along the audio band.**
- **Threshold of Feeling or Pain – Defines the sound intensity above which the sounds may cause pain and provoke hearing damages.**



Typically, the threshold of pain is about 120 to 140 dB; sound intensity is measured in terms of Sound Pressure Level relatively to a reference intensity with 10^{-16} W/cm^2 at 1 kHz.

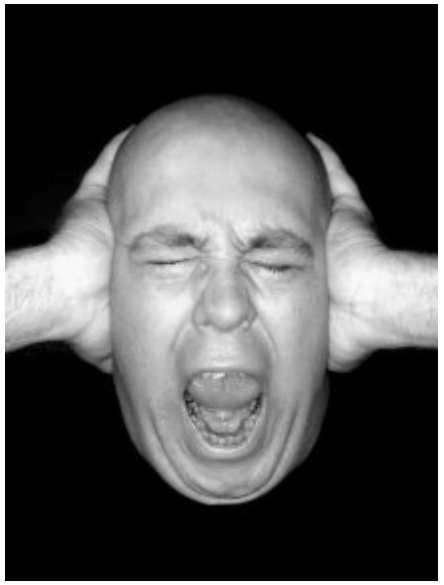
Sound Sensibility …



The human hearing dynamic range is about 100 dB.

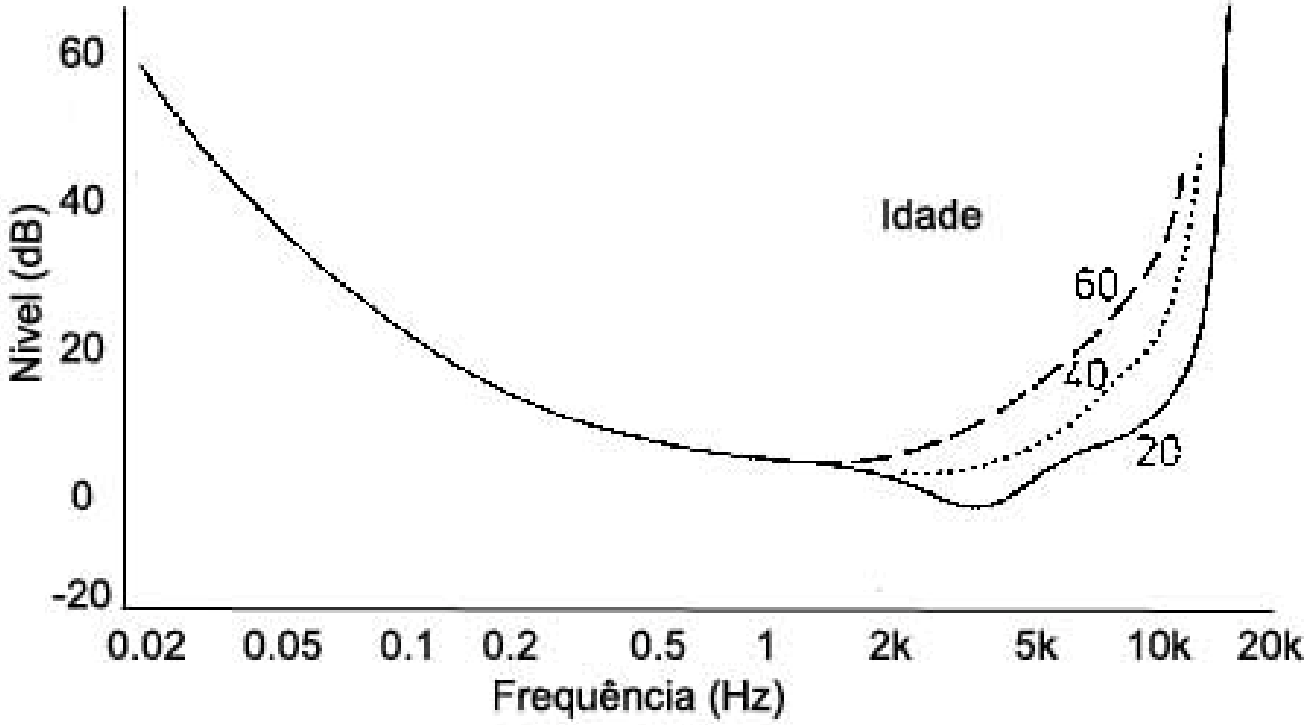
The Attacks …

• Rocket launching	180 dB
• Jet plane flying	140 dB
• Jet plane taking off	120 dB
• Disco	110 dB
• Underground	100 dB
• Urban traffic	90 dB
• Alarm-clock (at 1 meter)	80 dB
• Restaurant	70 dB
• Ar condicioning	60 dB
• Road (at 50 meters)	50 dB
• Living room	40 dB
• Library	30 dB
• Studio	20 dB
• Threshold of hearing	0 dB



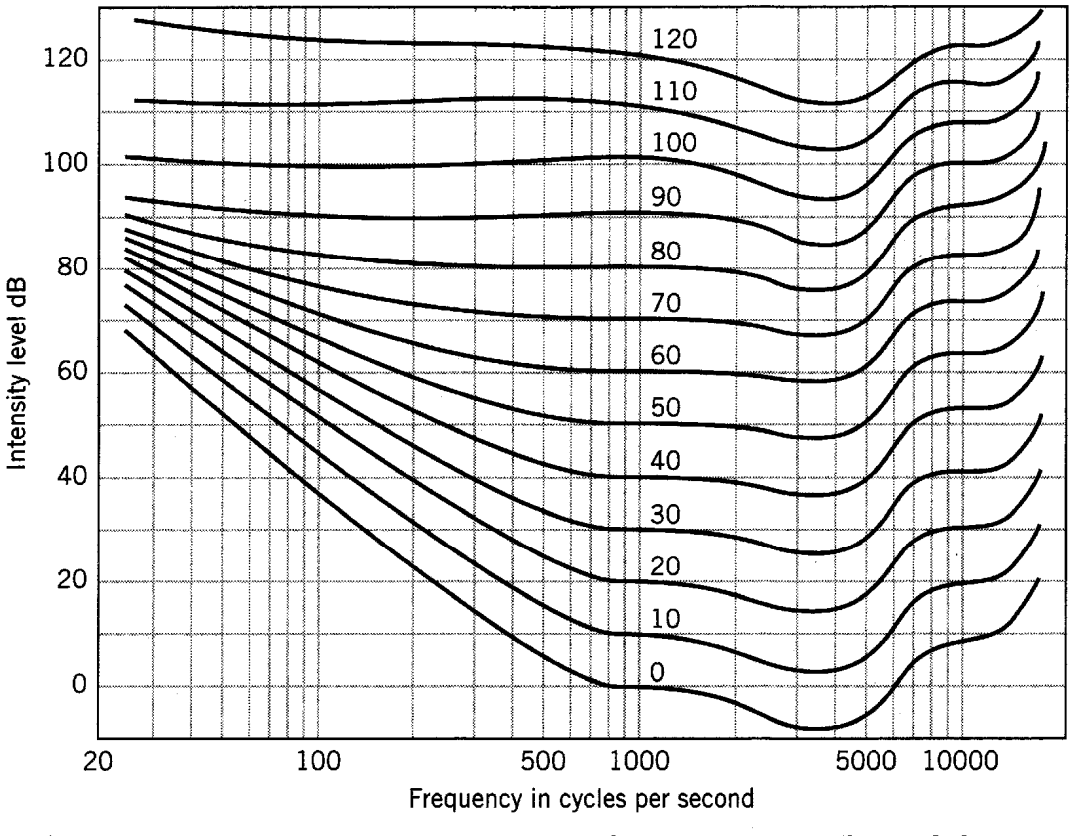
The reference sound intensity (0 dB) is 10^{-16} W/cm^2, for a sound at 1 kHz.

Hearing Threshold Variation with Age …

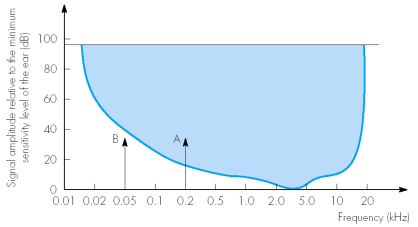


Equal Loudness Curves

- The concept of a threshold of hearing is often extended to produce curves of equal perceived intensity for sounds.
- These 'equal loudness' curves describe the perceived loudness of a sound relative to its actual intensity.
- A similar sensation at the lower frequencies requires a higher intensity since there is less sensibility.

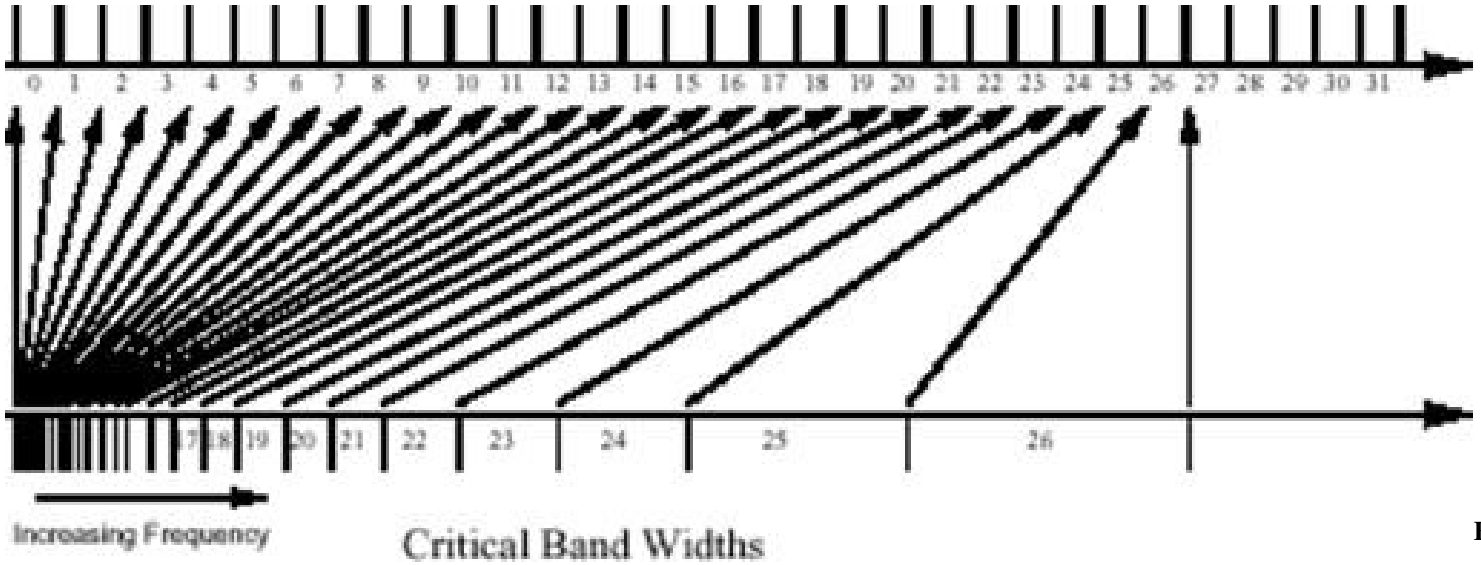


Critical Bands



Critical bands are the bands for which the auditory features are reasonably constant.

- For f ≈ 500 Hz — LBc = 100 Hz
- For f ≈ 1 kHz (2×500 Hz) — LBc = 200 Hz (2×100 Hz)
- For f ≈ 5 kHz (10×500 Hz) — LBc = 1000 Hz (10×100 Hz)



Critical Band Widths

MPEG-1 Audio: Coding Tools

- **Redundancy**

  Frequency coding

  Window switching

- **Statistical Redundancy**

  Huffman entropy coding

- **Irrelevancy**

  Perceptive coding, masking and quantization

  Dynamic allocation of bits

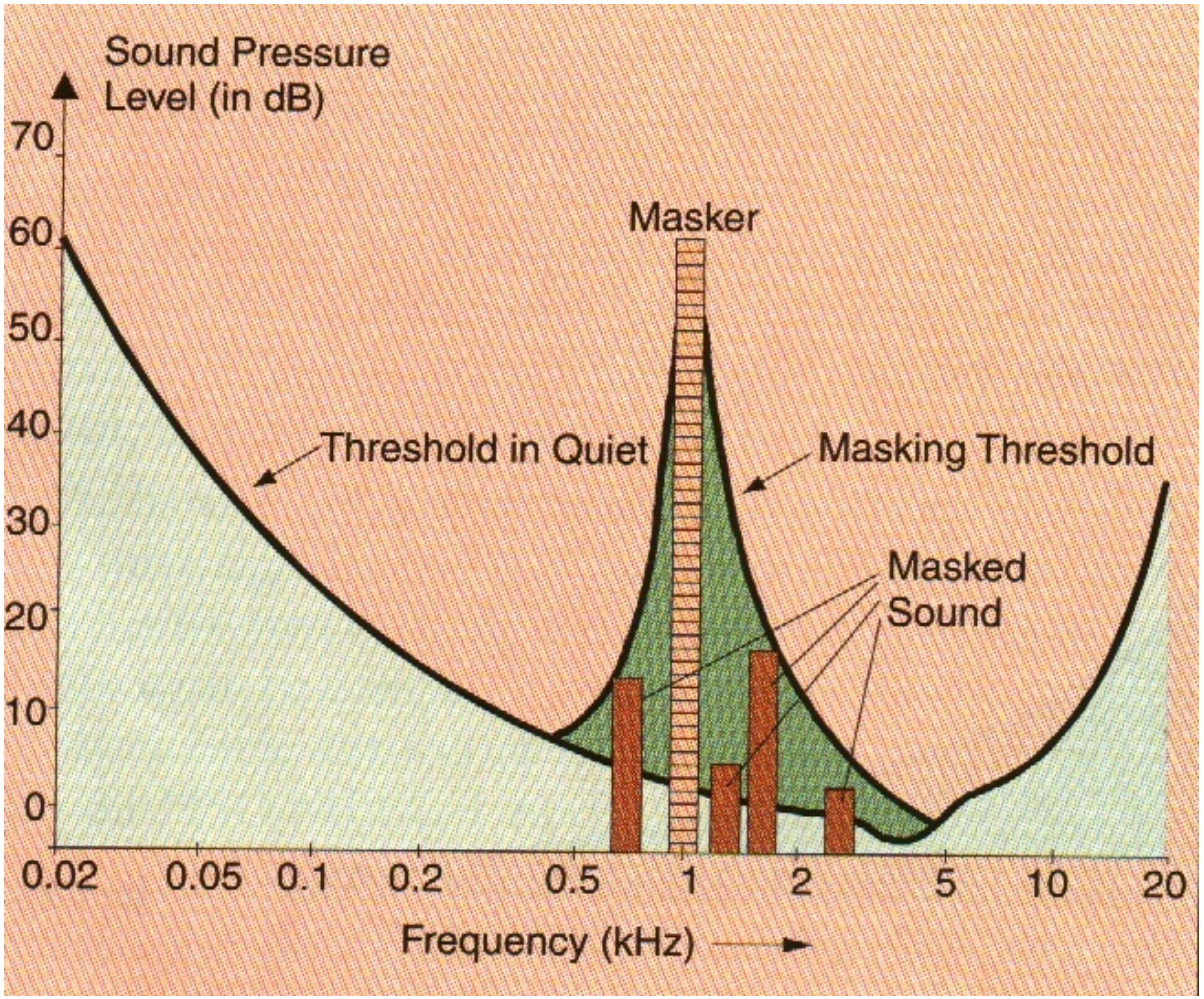
Defining Audio Masking

Auditory masking is the hearing behavior when the perception of one sound is affected by the presence of another sound; in this case, certain sound components may not be partially or totally perceived due to the prominence of other sound components.

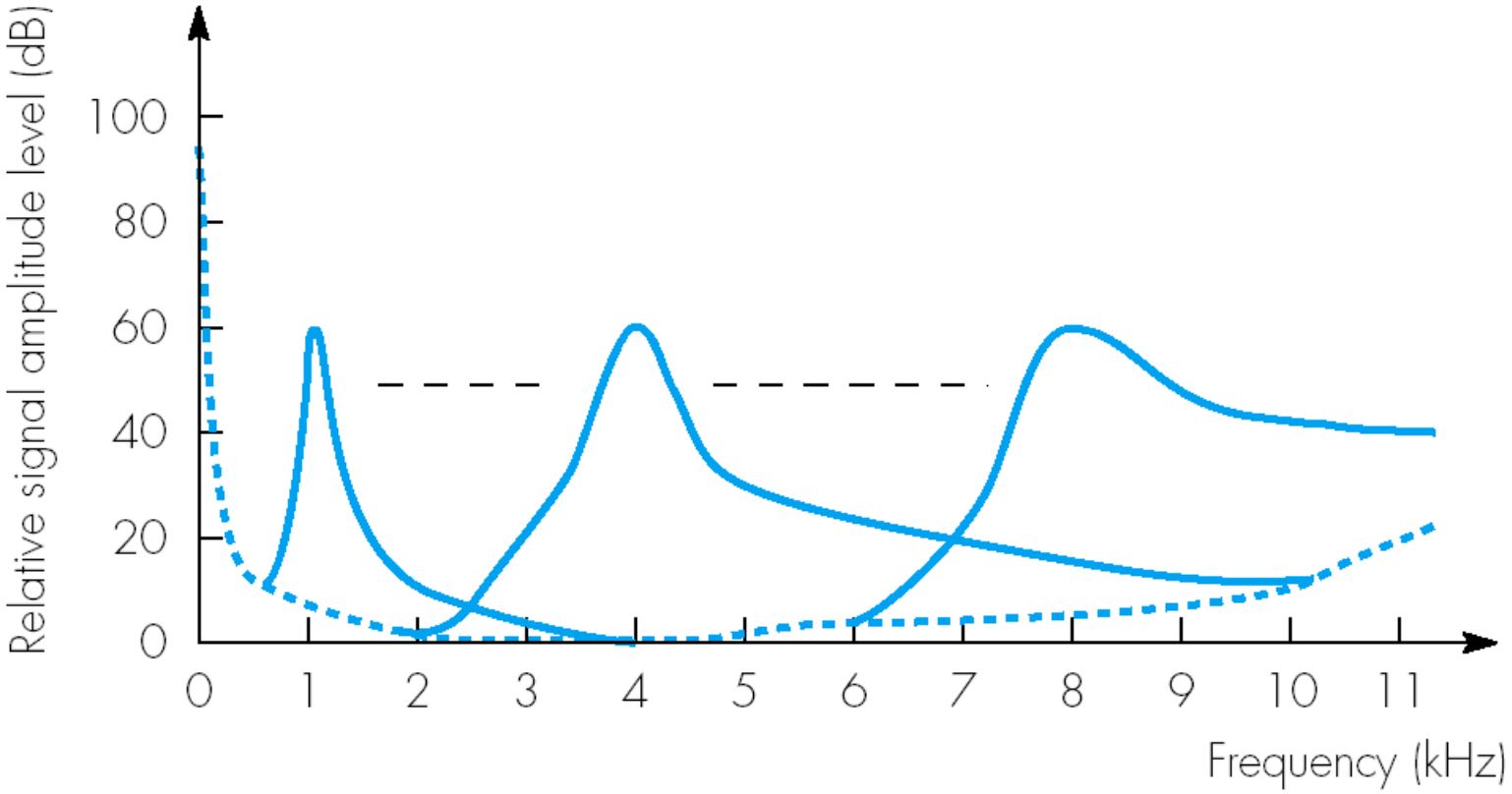
- **One sound may simply obscure another sound or increase its hearing threshold.**
- **The masking sound depends on the circumstances: for example, although it may be possible to speak 'normally' with someone at a party, any distraction may result in the background noise masking the voice of the other person.**

The masking effect is highly non-linear and its effects are very diverse.

Frequency Masking



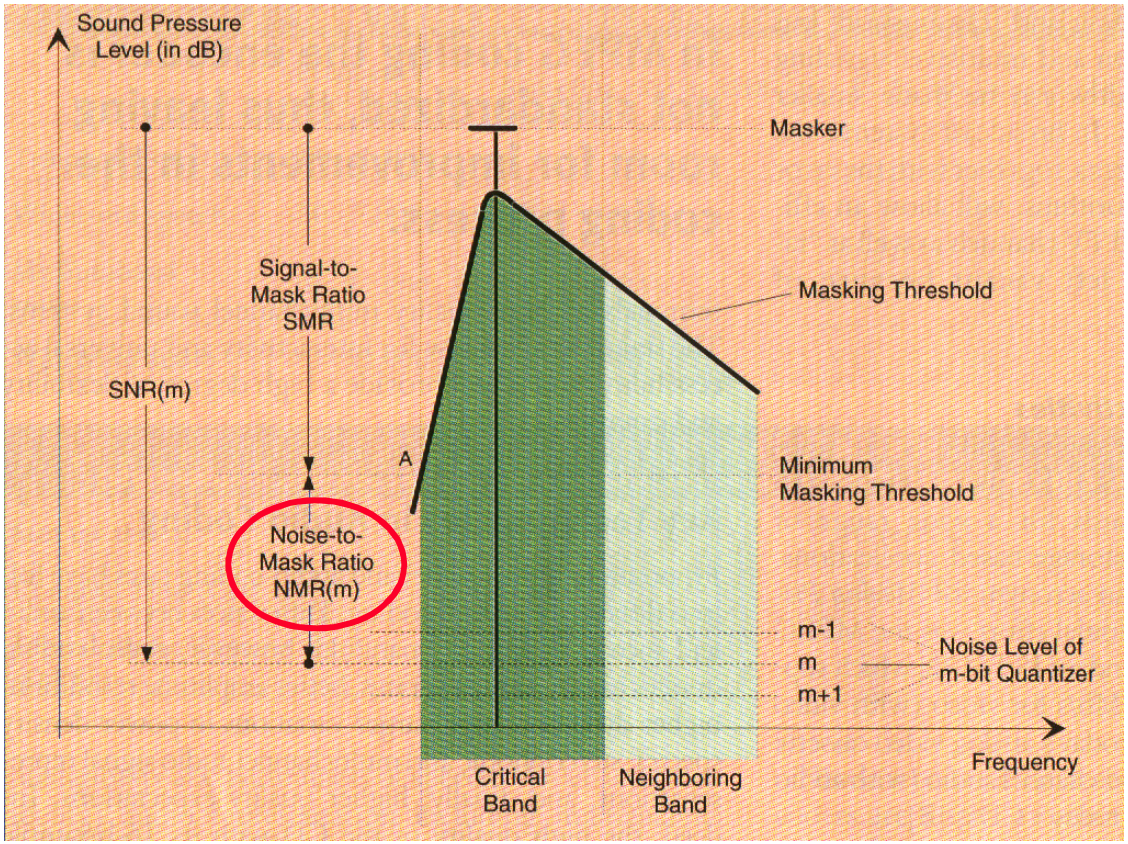
Masking Width Variation with Frequency



Frequency Masking

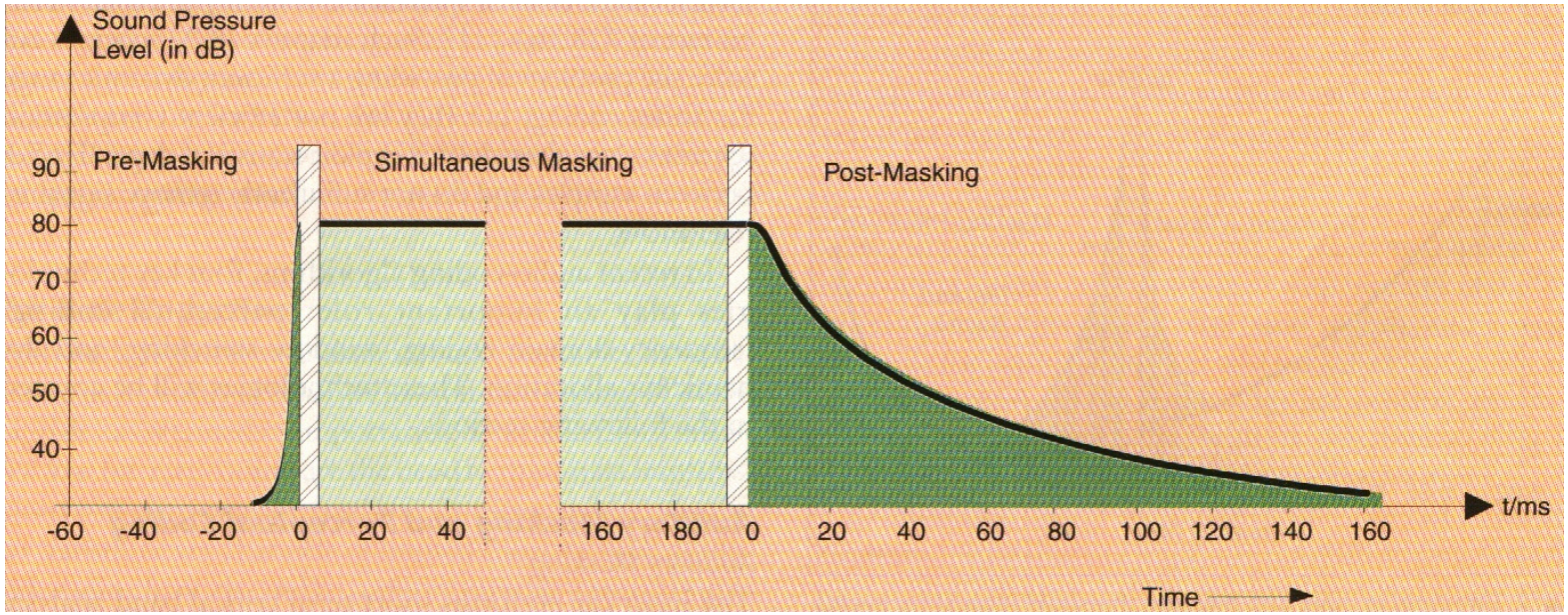
NMR measures the difference between the quantization noise level and the level at which the distortion becomes audible for a certain band.

The coding noise is not relevant while NMR is negative (NMR=SMR-SNR).



Masking and the critical bands shape have been much studied to model the behaviour of the human auditory system.

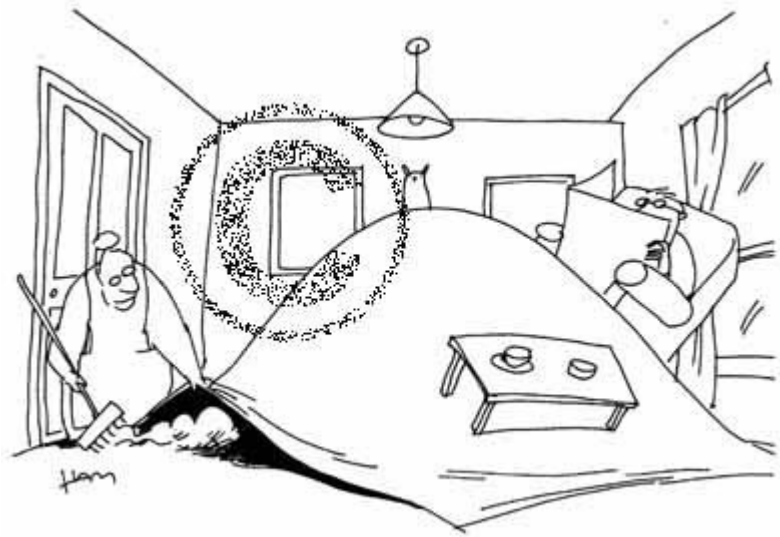
Temporal Masking



- **Temporal masking occurs when a sudden stimulus sound makes inaudible other sounds which are present immediately preceding or following the stimulus.**
- **Masking that obscures a sound immediately preceding the masker is called backwards masking or pre-masking (< 5 ms) and masking that obscures a sound immediately following the masker is called forwards masking or post-masking (≈20 ms).**

Perceptive Coding

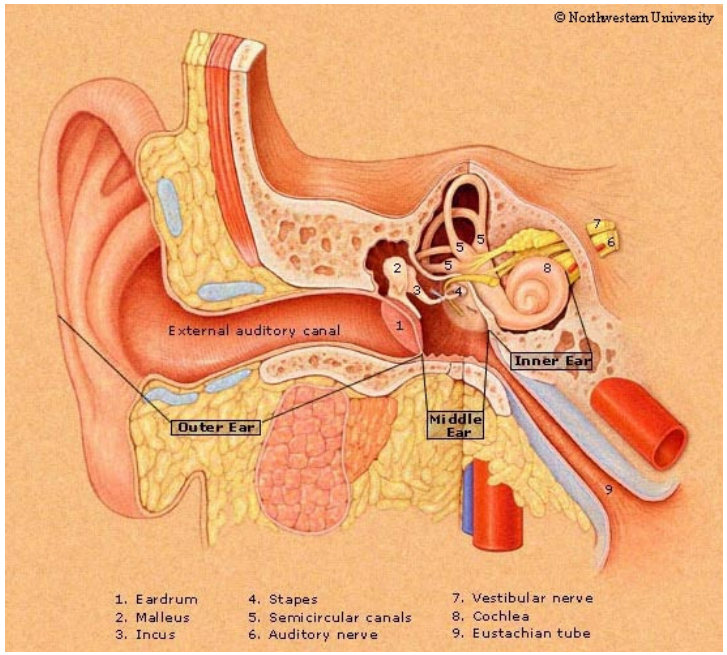
Irrelevancy manifests itself as amplitude or frequency information (resolution, detail) which cannot be perceived by humans. All masked signal components don't need to be coded/transmitted.



Perceptive coding is based on the idea of 'hiding' more noise (coding error) in the frequency zones where that noise is better tolerated, e.g. due to masking, using a psychoacoustic model.

Perceptive coding exploits the characteristics of the receiver and not of the source as in speech coding.

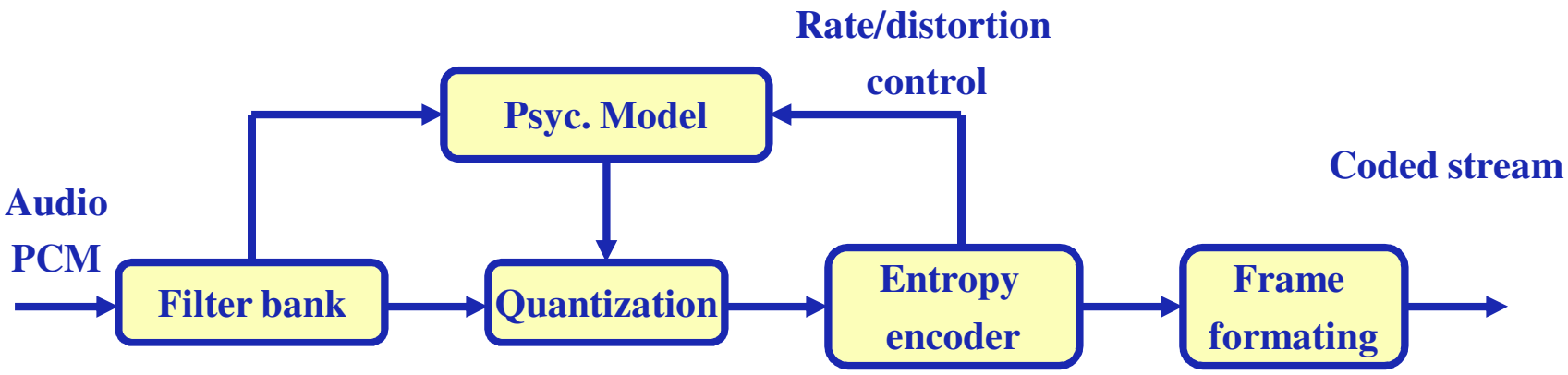
Psychoacoustic Model: the Secret !



A psychoacoustic model is a mathematical model which defines, in a simplified way, the main properties and tolerances of the human auditory model, notably its sound intensity perception, its spectral selectivity and, especially, the masking effect.

It is very useful to dynamically and adaptively estimate the amount and shape of the coding noise that may be injected in the audio signal without becoming perceptible, allowing to reduce the coding rate.

Perceptive Encoder Architecture



The psychoacoustic model controls the quantization noise/error to introduce in each audio band.

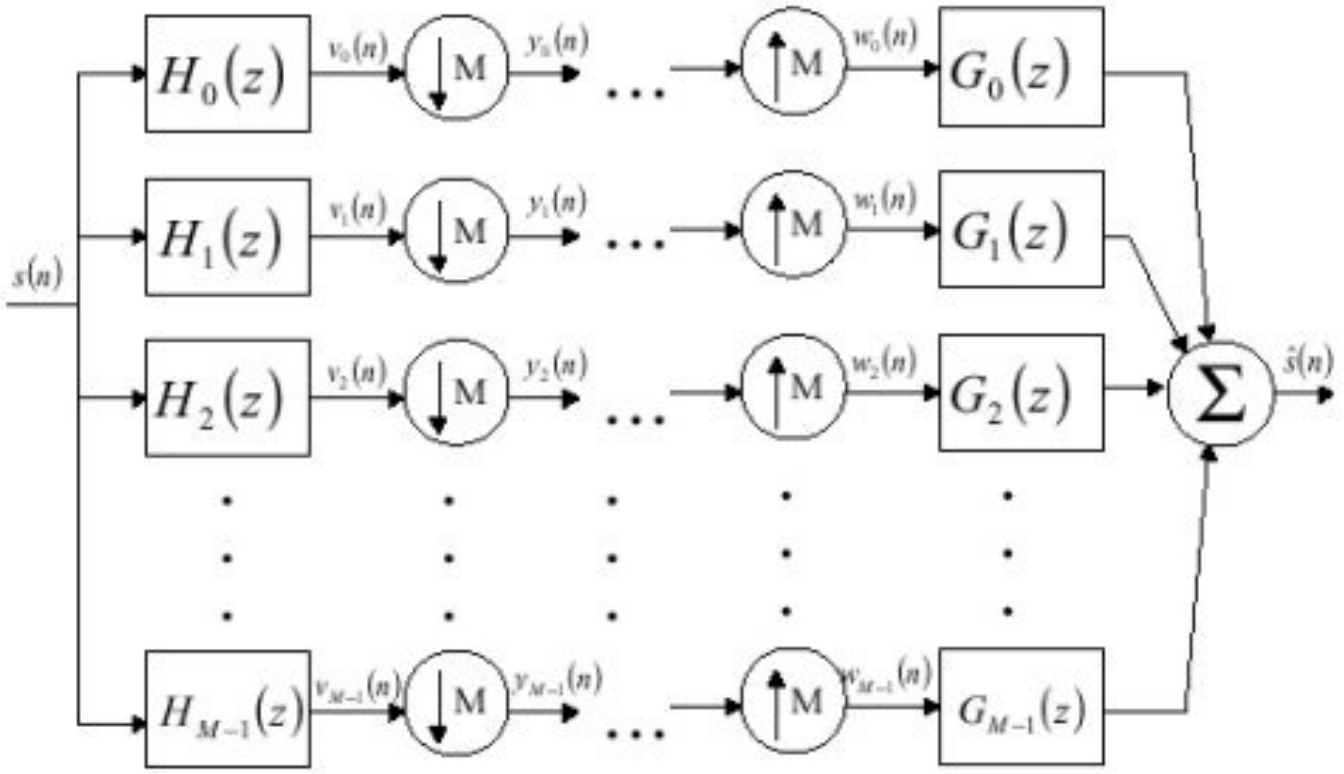
Frequency Coding

Coding in the frequency domain divides the audio signal spectrum in frequency bands; a filter bank is used to generate uncorrelated spectral components and independently quantize those components.

There are two main ways to perform frequency coding:

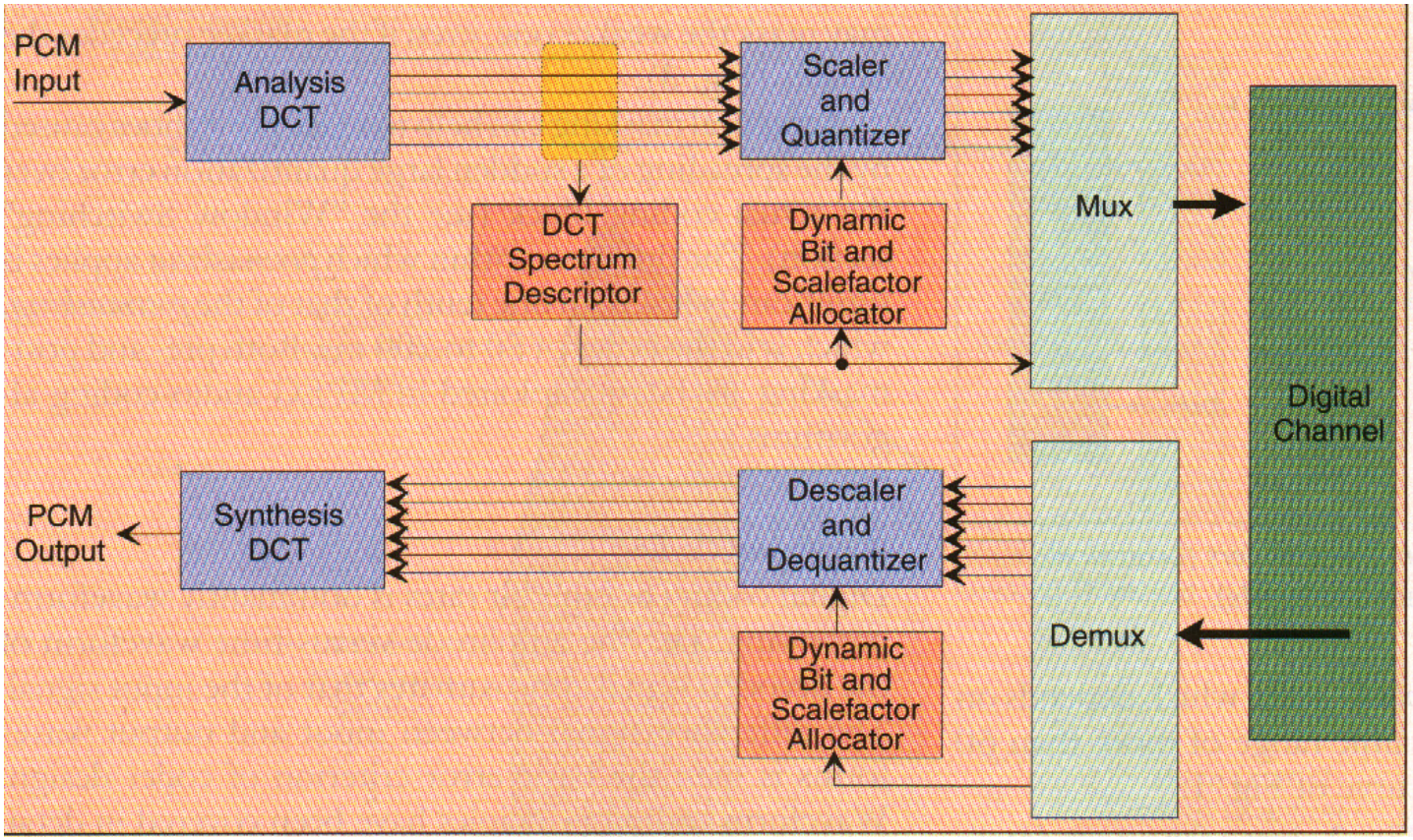
- **TRANSFORM CODING – A samples block is linearly transformed using a discrete transform into a set of quasi-uncorrelated coefficients.**
- **SUBBAND CODING – A samples block is decomposed into several samples subsets using *M* band pass filters, contiguous in frequency, in order the set of generated subbands may be additively recombined to synthesise the original signal.**

The Subband Frequency Decomposition



Filter bank Analysis **Filter bank Synthesis**

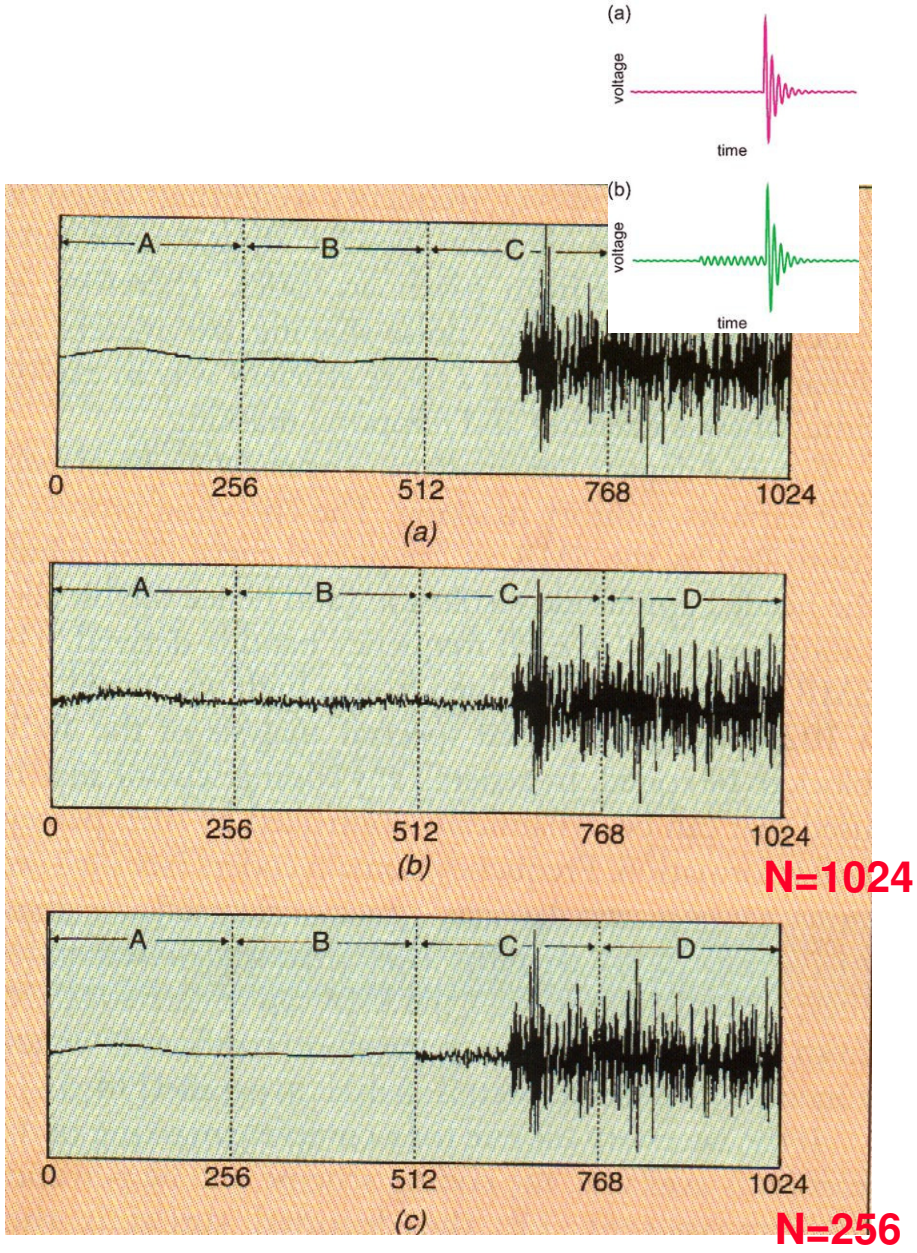
Transform Coding Architecture



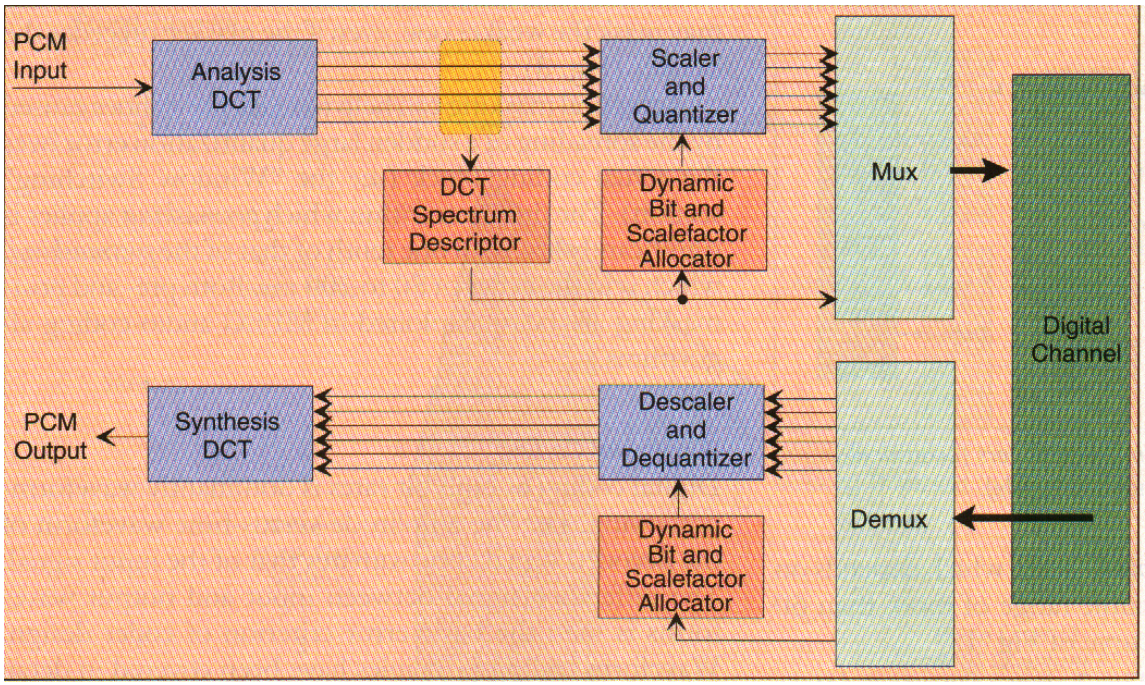
Depending on their frequency band, the various frequency coefficients are differently quantized.

Window Switching

- The usage of frequency coding for blocks of samples where silence is followed by a strong signal creates the so-called pre-echoes since the signal synthesis may significantly change the silent part of the signal (in a more or less stronger way depending on the quantization).
- To limit this phenomenon, variable size transform windows may be used with the encoder selecting the adequate window size depending on the signal characteristics.

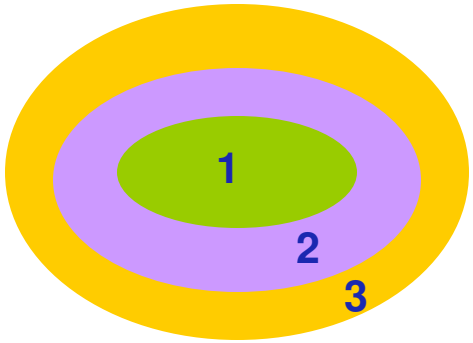


Dynamic Allocation of Bits



- **The quantization of the coefficients (after bit allocation) is performed depending on the spectral characteristics of the samples block to be coded.**
- **It may originate block effect and pre-echoes.**

MPEG-1 Audio: the 3 Layers



MPEG-1 Audio specifies the coded representation and decoding process of audio (mono or stereo pair) signals in three layers where:

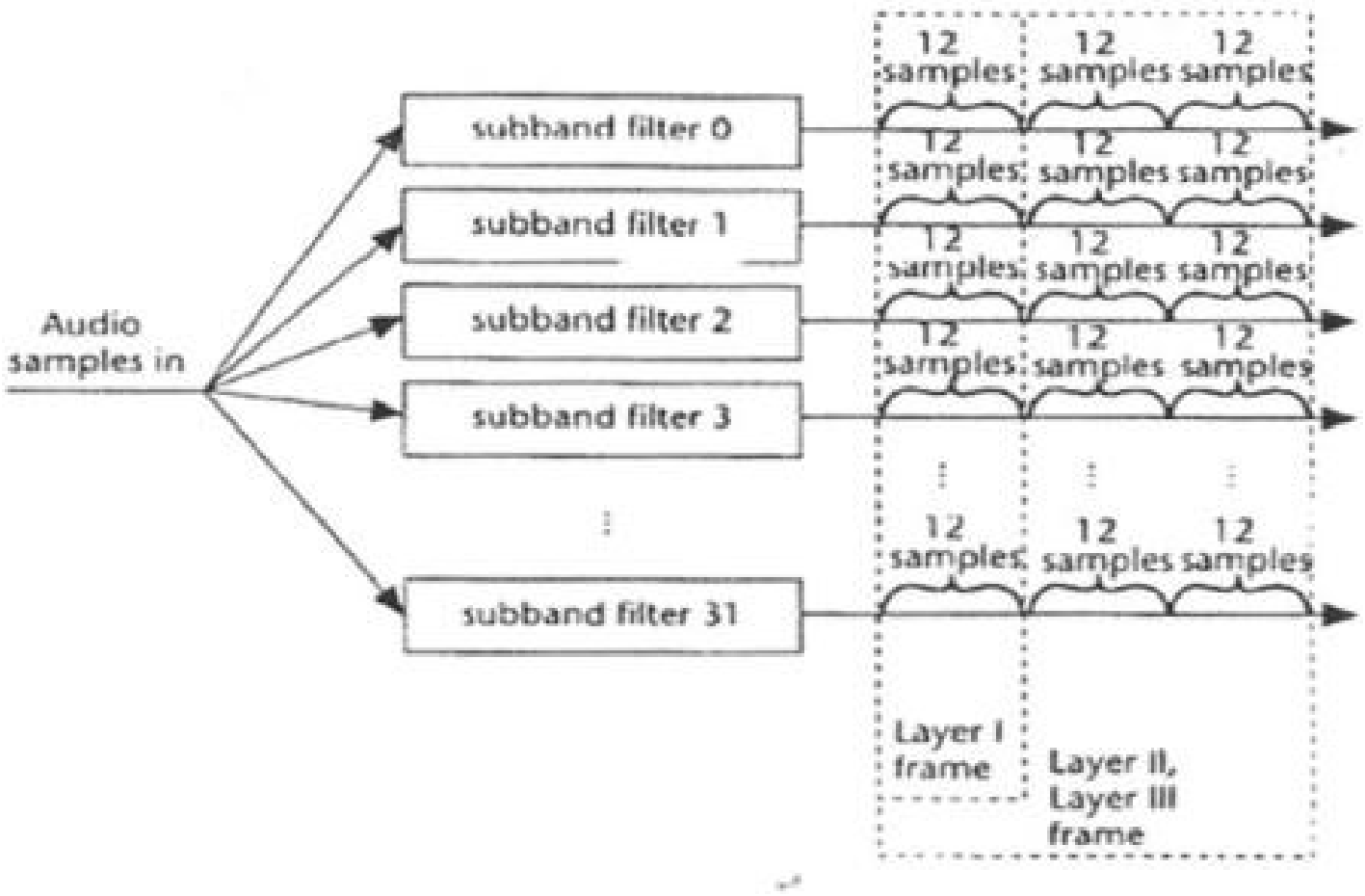
- **Each layer offers a rate/quality/complexity trade-off**
- **Higher layers have higher complexity, delay and coding efficiency**
- **N layer decoders are able to decode N-1 layers coded streams defining a hierarchy of decoders and bitstream syntaxes**

Layer	Typical rate	Minimum coding delay
1	32-448 kbit/s	(256+256+12×32)/48k ≈ 19 ms
2	32-384 kbit/s	(256+256+12×32×3)/48k ≈ 35 ms
3	32-320 kbit/s	(256+256+18×2×32×2)/48k ≈ 59 ms

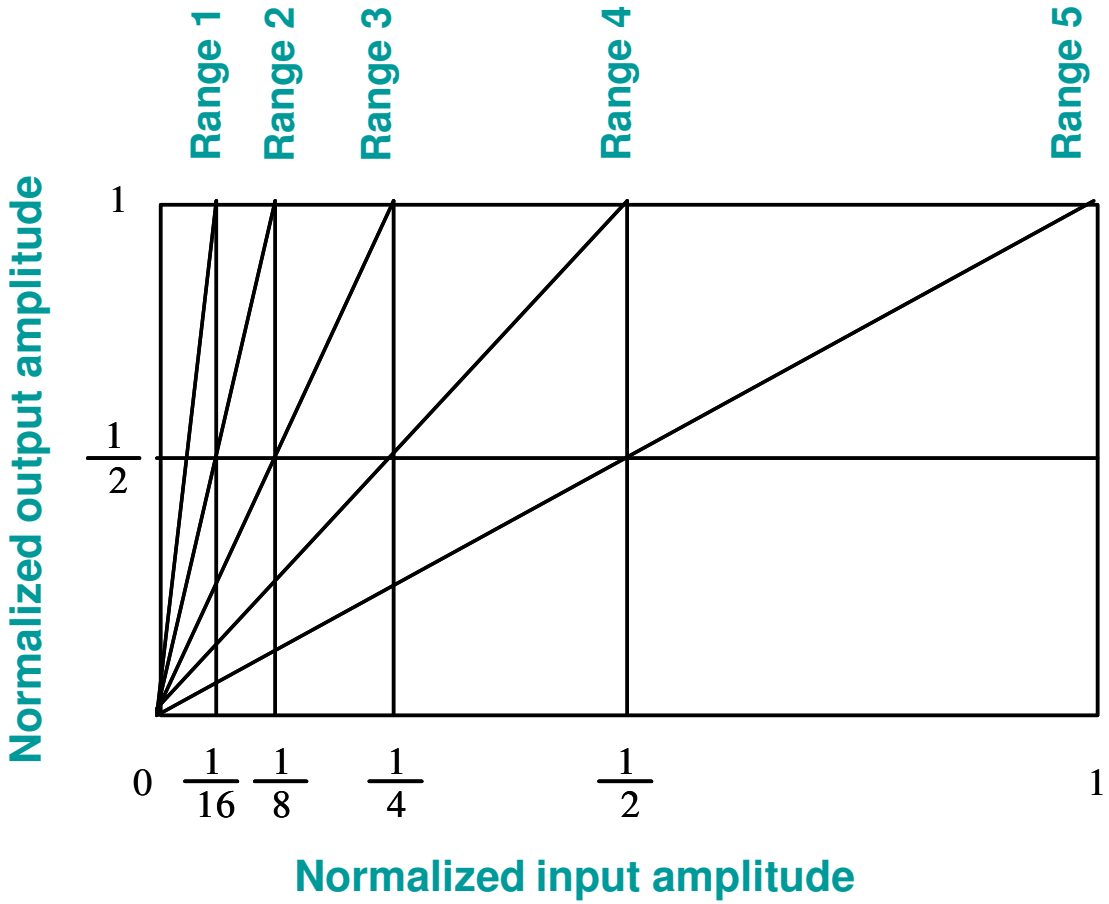
MPEG-1 Audio: Layer 1

- **Blocks with 384 audio samples are coded (corresponding to 8 ms at 48 kHz)**
- **Signal is decomposed into 32 uniform subbands**
- **Fixed segmentation of 12 samples per subband (12×32=384 samples)**
- **APCM type quantization (block companding using a scale factor) for each subband with 0-15 bit/sample; this value may change for each subband; each scale factor 'costs' 6 bits (maximum 6×32=192 bits/frame)**
- **Psychoacoustic models 1 or 2 suggested in the standard (there are 2 in the standard without normative value)**
- **Iterative rate/distortion adjustment to minimize the NMR (*Noise-to-Mask Ratio*) ratio for each subband**
- **Transparent quality regarding the CD quality (PCM) at 384 kbit/s which a typical compression factor of 4**

Samples, Frames and Subbands …



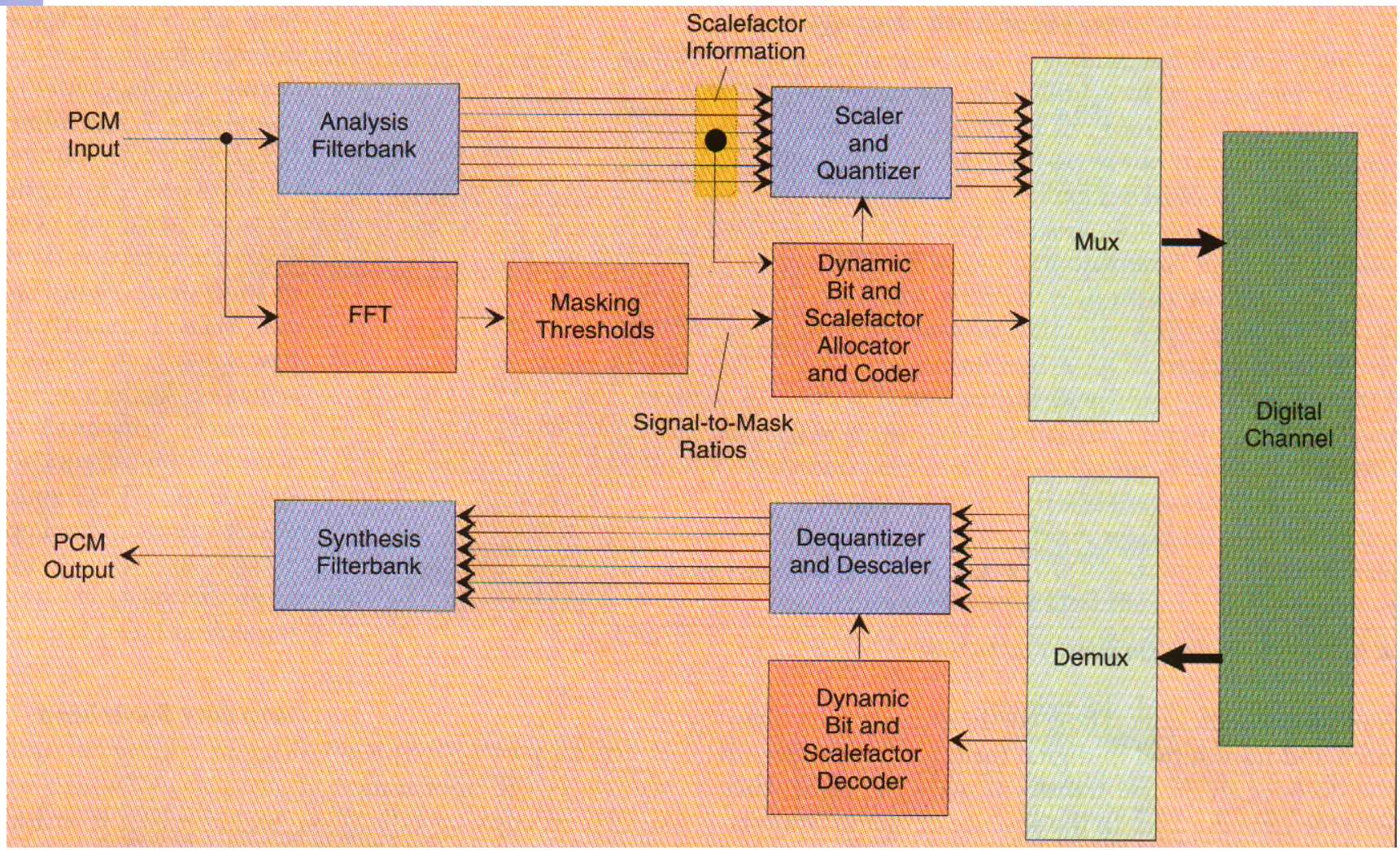
Block Companding or Scale Factors …



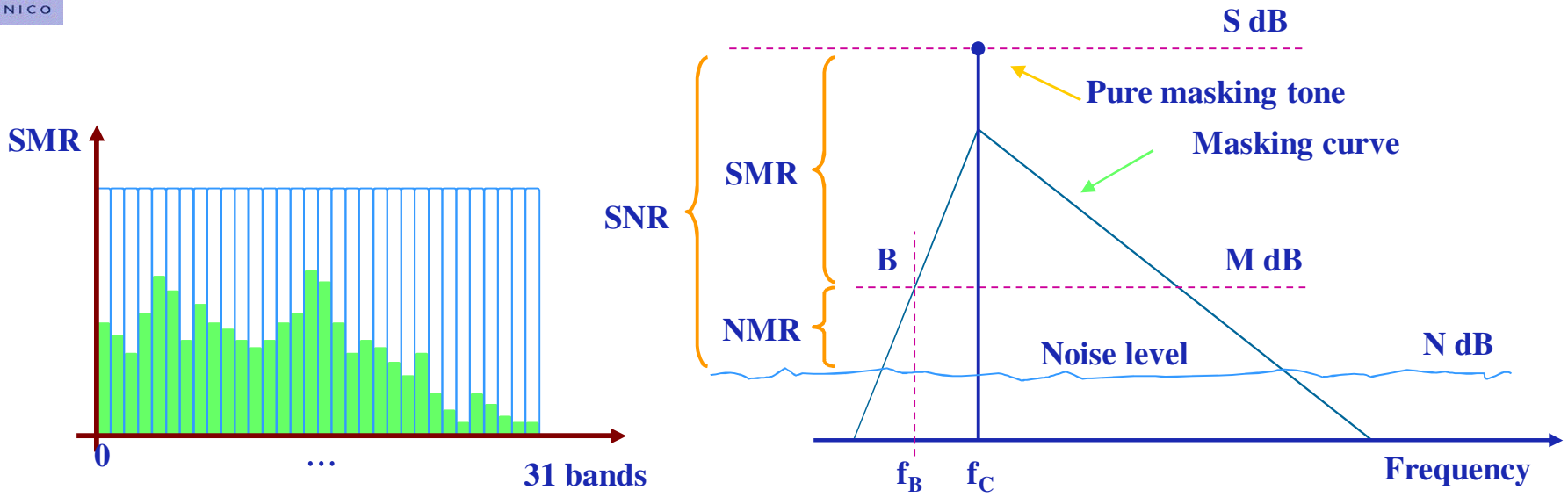
MPEG-1 Audio: Layer 2

- **Blocks with $3 \times 384 = 1152$ audio samples are coded (corresponding to 24 ms at 48 kHz)**
- **Fixed segmentation with $3 \times 12 = 36$ samples per subband**
- **Coding algorithm as for Layer 1 with the exception of using more efficient methods to code the quantization scale factors by exploiting the redundancy between the adjacent scale factors within the 3 sub-blocks of 12 samples.**
  - **Scale factors are shared among the 3 consecutive 'granules' for each sub-band.**
  - **When they are similar or when temporal post-masking can hide the distortion, only one or two scale factors need to be coded.**
- **Transparent quality regarding the CD quality (PCM) at 192 kbit/s which a typical compression factor of 8**

Layers 1 and 2 Encoder Architecture



MPEG-1 Audio: Quantization



The psychoacoustic model is used to define the masking curve for each band, determining the noise allowed for each band and, thus, the number of quantization levels to use for the signal components above the masking threshold.

If the quantization noise remains below the masking threshold, the coded signal is subjectively indistinguishable from the original signal.

MPEG-1 Audio, Layers 1 and 2: Quantization

- **The number of quantization levels for each subband is obtained through the dynamic allocation of bits controlled by the psychoacoustic model.**
- **The number of quantization levels for each subband is the one minimizing the NMR for that band, this means the one allowing 'hiding' more quantization noise and thus spending the minimum rate.**
- **The adoption of block companding determines the coding of samples normalized to the maximum value per subband (using a scale factor by subband), thus allowing to reduce the quantization error for the smaller samples.**
- **The normalized samples per subband (using the scale factors) are PCM coded with a varying number of bits/sample.**
- **When decoding, all samples of all bands without allocated bits are set to zero.**

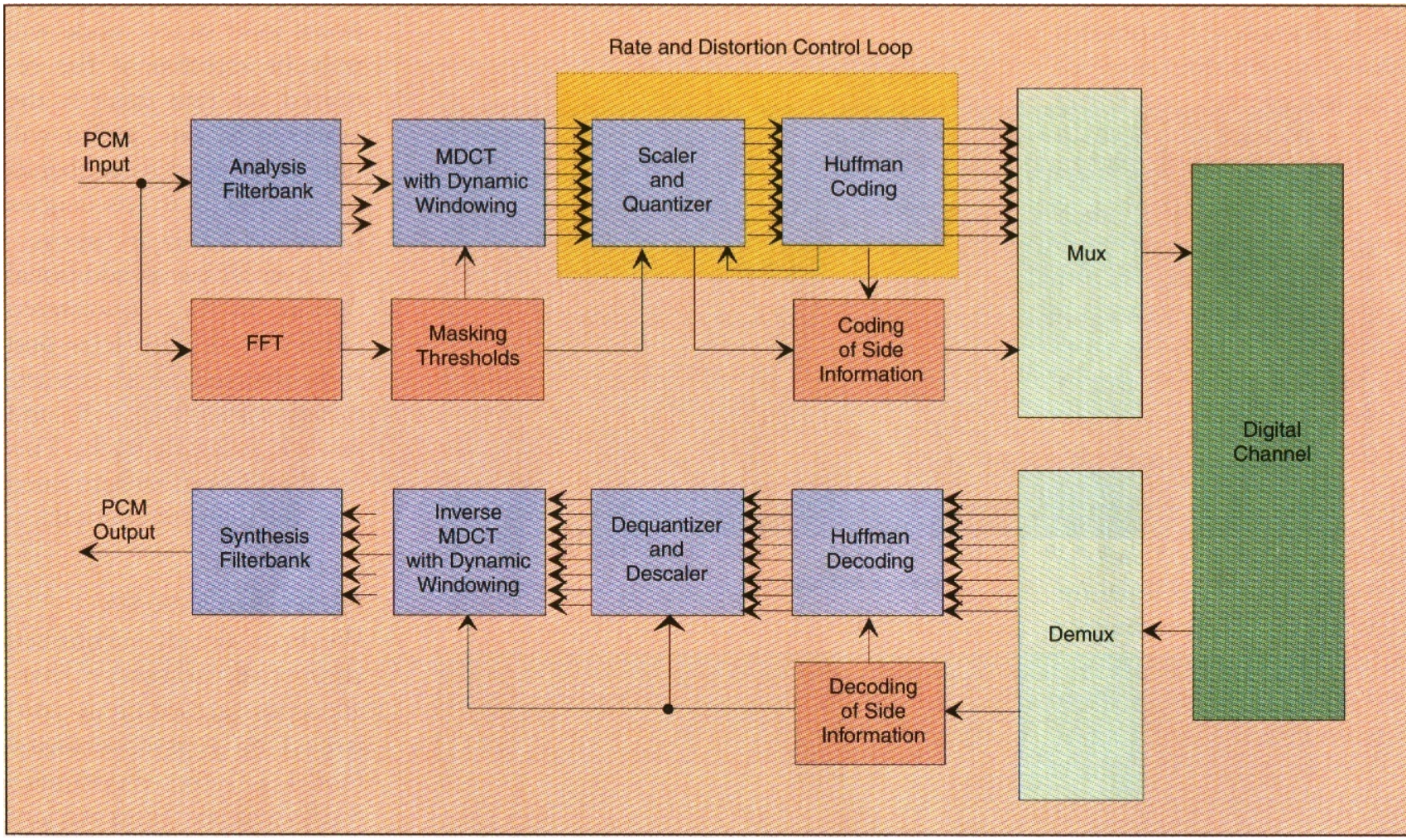
MPEG-1 Audio: Layer 3 (the Famous MP3 !)

- **Blocks with 1152 audio samples are coded (2 groups with 576 samples each)**
- **Hybrid time/frequency coding structure -** The filter bank (subbands) is followed by transform coding (Modified DCT).
- **Dynamic window switching –** To increase the frequency resolution, the 32 subbands are frequency subdivided by applying to each of them a transform with 6 or 18 coefficients; this results into a maximum number of frequency components of 32×18 (6) = 576 (or 192). The smallest window allows to control the temporal resolution and thus to reduce the pre-echoe effect.
- **Overlapping windows -** The MDCT is applied with 50% window overlapping to reduce the block artefacts which means that the MDCT is applied to sets of 12 or 36 subband samples.

MPEG-1 Audio: Layer 3 (the Famous MP3 !)

- **Quantization -** Non-uniform quantization of the MDCT coefficients (exponential like) introducing higher quantization error for the higher amplitude coefficients (where there is lower sensibility to errors); a mechanism with two nested cycles is typically used to control the quantization and coding.
- **Entropy Coding -** Huffman entropy coding of the quantized MDCT coefficients and the scale factors.
- **Psychoacoustic model -** Psychoacoustic model 2 suggested in the standard (more complex than model 1).
- **VBR -** More targeted to variable rate coding (useful for some applications)
- **Target -** Transparent quality regarding the CD quality (PCM) at 128 kbit/s which a typical compression factor of 12

MP3 Encoder Architecture



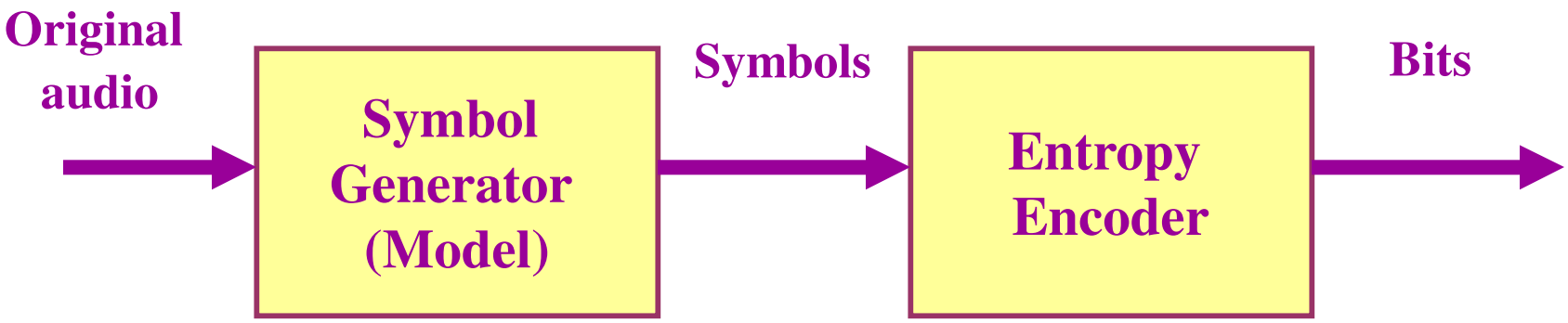
MP3 Encoding Walkthrough …

- **FREQUENCY DECOMPOSITION - Divide the audio spectrum into 32 frequency bands (known as sub-bands) using a filter bank.**
- **DCT TRANSFORM - Calculate 6 or 18 DCT frequency components for each sub-band; use 6 frequency components when there is a need to control time artifacts (pre-echo and post-echo).**
- **MASKING THRESHOLDS COMPUTATION - Use the psychoacoustic model to compute the masking thresholds for the audio (or the allowed noise) for each spectrum partition.**
- **QUANTIZATION - Quantize the DCT components for each band using the defined quantization step and scale factor. If the quantization noise can be kept below the masking threshold, then the compression results should be indistinguishable from the original signal.**
- **ENTROPY CODING – The quantized DCT components for each band are entropy coded.**

MP3 Performance …

Sound quality	Bandwidth	Mode	Bitrate	Compression factor
telephone sound	2.5 kHz	mono	8 kbps *	96:1
better than shortwave	4.5 kHz	mono	16 kbps	48:1
better than AM radio	7.5 kHz	mono	32 kbps	24:1
similar to FM radio	11 kHz	stereo	56...64 kbps	26...24:1
near-CD	15 kHz	stereo	96 kbps	16:1
CD	>15 kHz	stereo	112..128kbps	14..12:1

The MP3 Symbolic Model



An audio sequence is represented as a succession of frames, each with a certain number of audio samples, represented using MDCT coefficients for each subband, quantized based on a psychoacoustic model.

Stereo … and More ….

MPEG-1 Audio: Stereo Coding

iPod Not Included

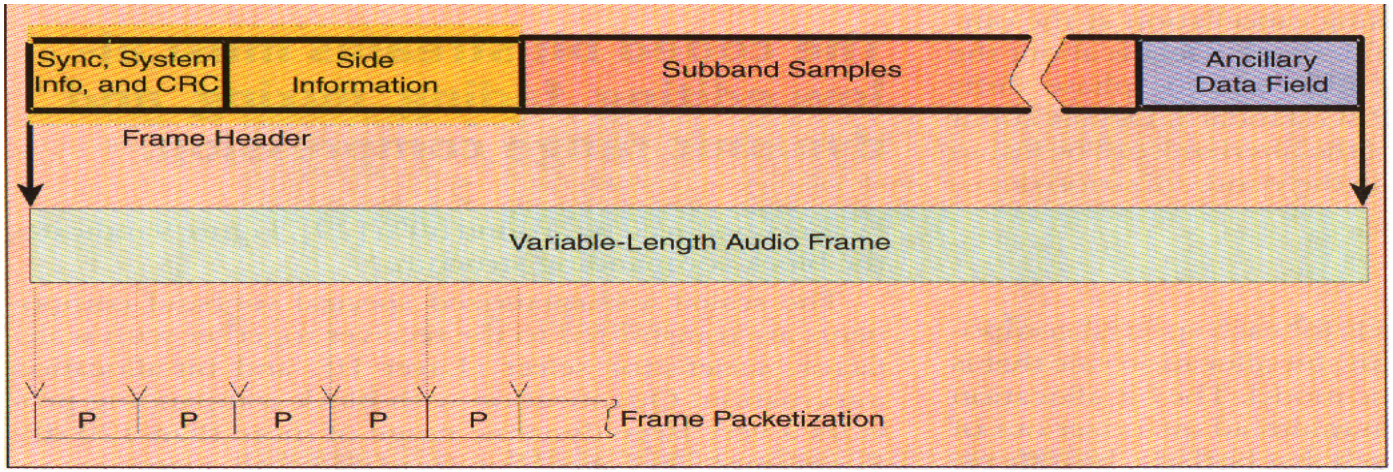
Joint stereo coding takes advantage of the fact that the two channels of a stereo pair contain redundant information. These stereophonic irrelevancies and redundancies are exploited to reduce the total bitrate. Joint stereo is used in cases where only low bitrates are available but stereo signals are desired.

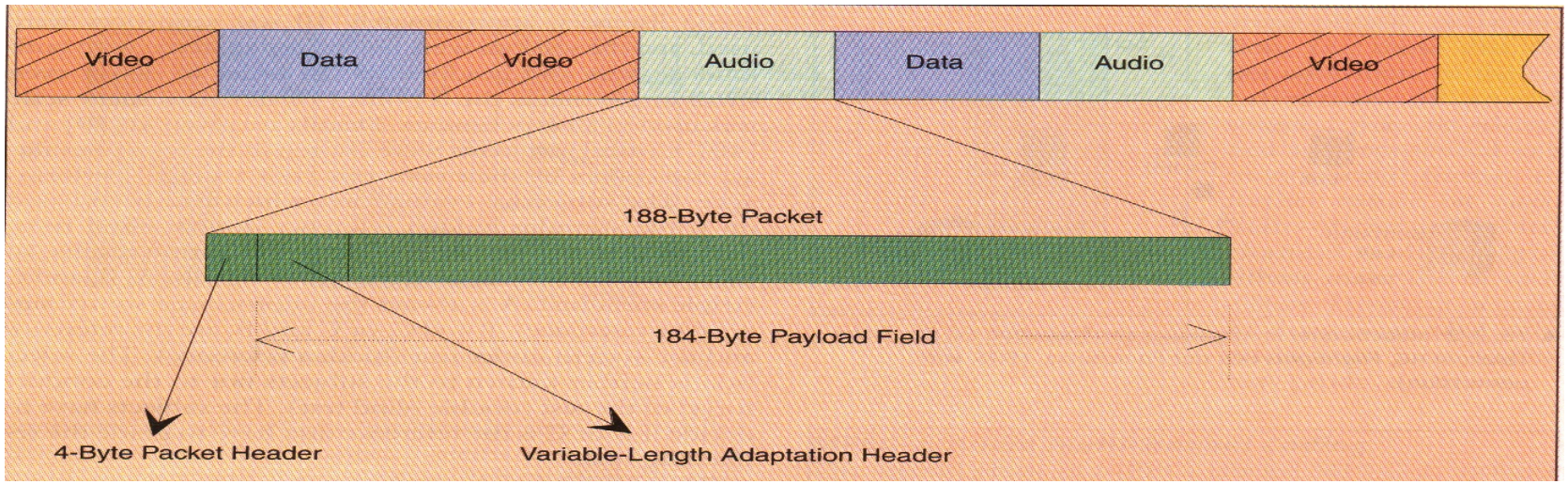
There are 5 audio coding modes:

- **Mono**
- **Dual Stereo – Channels are independently coded, e.g. 2 different languages.**
- **Stereo – Independent coding but sharing certain fields in the coded frame.**
- **Joint Stereo – Channel dependency is exploited through the so-called *intensity stereo technique*; above 2 kHz, the L+R signal is coded together with scale factors for the 2 channels (L and R) since there is lower hearing sensibility.**
- **Mono/Stereo (MS) (only layer 3) – The 2 channels are coded as L+R (middle) and (side) L-R allowing to better control the spatial location of the quantization noise.**

Audio Syntax and Systems Packet Level

CRC is optional !





MP3 Licensing

PC Software Applications		
mp3	Decoder	· **US$ 0.75** per unit or **US$ 50 000.00 - US$ 60 000.00** one-time paid-up
	Codec	· **US$ 2.50 - US$ 5.00** per unit
mp3PRO	Decoder	· **US$ 1.25** per unit or **US$ 90 000.00** one-time paid-up
	Codec	· **US$ 5.00** per unit

Hardware Products		
mp3	Decoder	· **US$ 0.75** per unit
	Codec	· **US$ 1.25** per unit
mp3PRO	Decoder	· **US$ 1.25** per unit
	Codec	· **US$ 5.00** per unit

ICs / DSPs
For available software, supported platforms, porting and licensing options, please contact us at info@mp3licensing.com.

Games	
mp3	· **US$ 2 500.00** per title
mp3PRO	· **US$ 3 750.00** per title

Electronic Music Distribution / Broadcasting / Streaming	
mp3	· **2.0** % of related revenue
mp3PRO	· **3.0** % of related revenue

With MP3, it is effectively easier to 'pirate' music

…

Which does not mean one should do it

…

Or even that it is advantageous to do it, at least in the long term …

Final Remarks

- **There is a growing number of devices, applications, etc, notably portable, including MP3 players.**
- **MPEG-1 Audio Layer (MP3) is commonly used for music in the Web.**
- **Digital Audio Broadcasting (DAB) and Digital Video Broadcasting (DVB) used for a long time only MPEG-1 Audio Layer 2.**
- **MP3 provoked the explosion of one of the biggest current multimedia problem this means digital rights management … and Napster … and peer-to-peer …**

Soon After MPEG-1: ITU-T H.324 Terminals

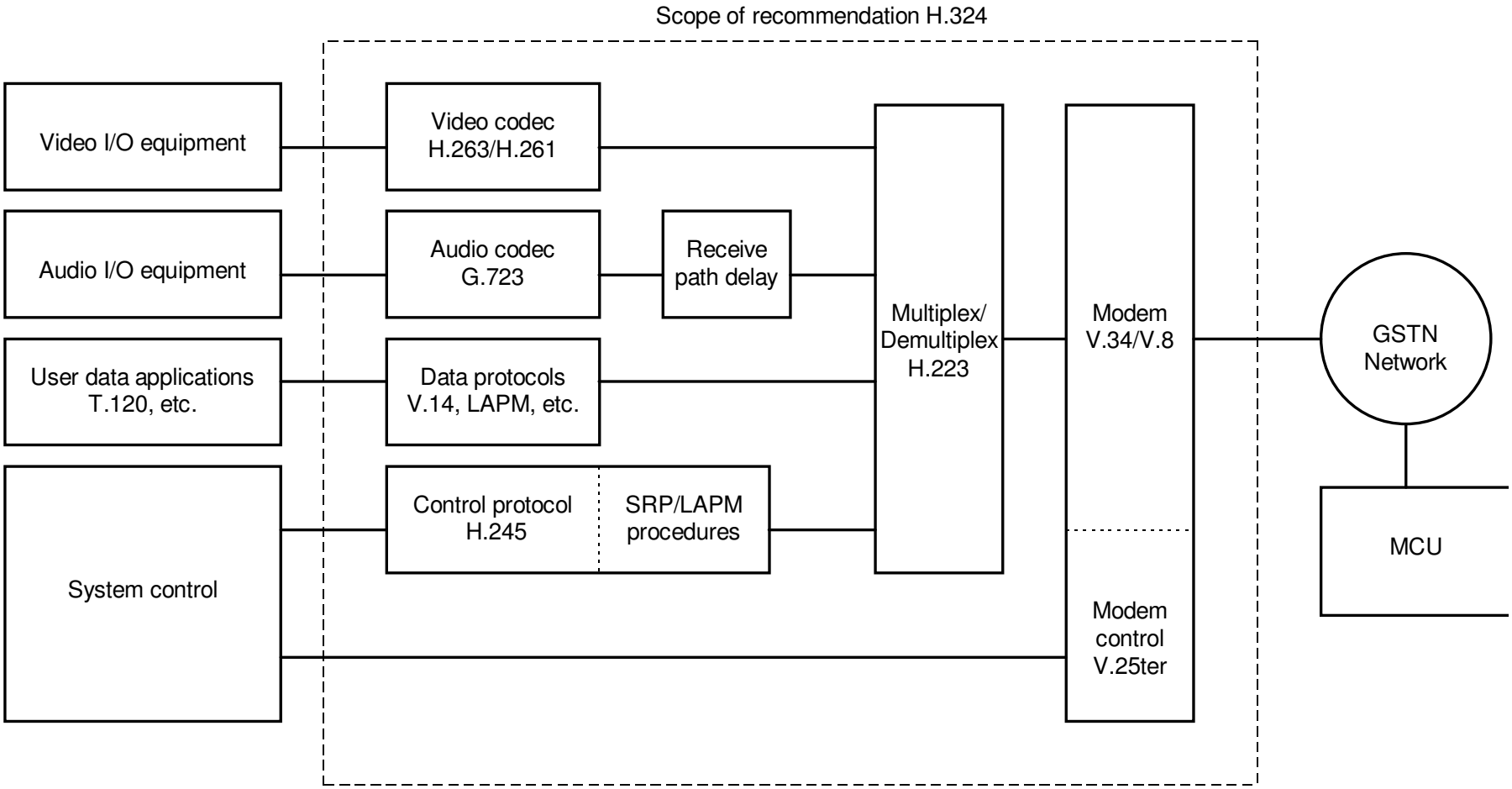
H.324 Terminals: Objectives

H.324 terminals offer real-time, low rate, unidirectional and bidirectional communication with any combination of video, speech, and data, notably over the public telephone network and mobile networks.

- **Multipoint communications are also possible using a MCU (multipoint control unit).**
- **H.324 multimedia telephones may appear as specific equipment (*stand alone*) or integrated in personal computers or workstations..**
- **Both the interoperability with ISDN audiovisual terminals (H.320) and mobile audiovisual terminals (H.324/M) has been considered.**

It was expected the H.324 terminals to have two main types of applications: videotelephony (*domestic*) and PC integrated multimedia applications (*business*).

H.324 Terminal Architecture



H.324 Terminal: Main Modules

- **Video coding - H.263 or H.261**
- **Speech coding - G.723 – Quality similar to analogue quality at 6.4 kbit/s.**
- **Communication control - H.245 – Involves the exchange of 4 types of messages - *Request, Response, Command, Information* – to control the operation of the H.324 terminal, notably regarding capability exchange, and opening and closure of logical channels.**
- **Multiplexer - H.223 – Packet based and allowing the exchange of one or more information streams with speech, video, data and control messages.**
- **Modem - V.34 – Operates up to 28.8 kbit/s (more for later versions).**

Recommendation H.263 (1995): Objective

Video compression with improved quality regarding recommendation H.261 using lower bitrates.

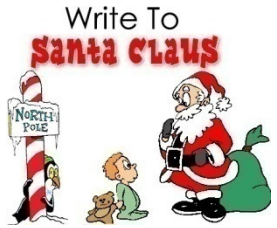
Particular relevance is given to the rates down to 20 kbit/s, targeting videotelephony over the analogue telephone network.

Recommendation H.263: Motivation

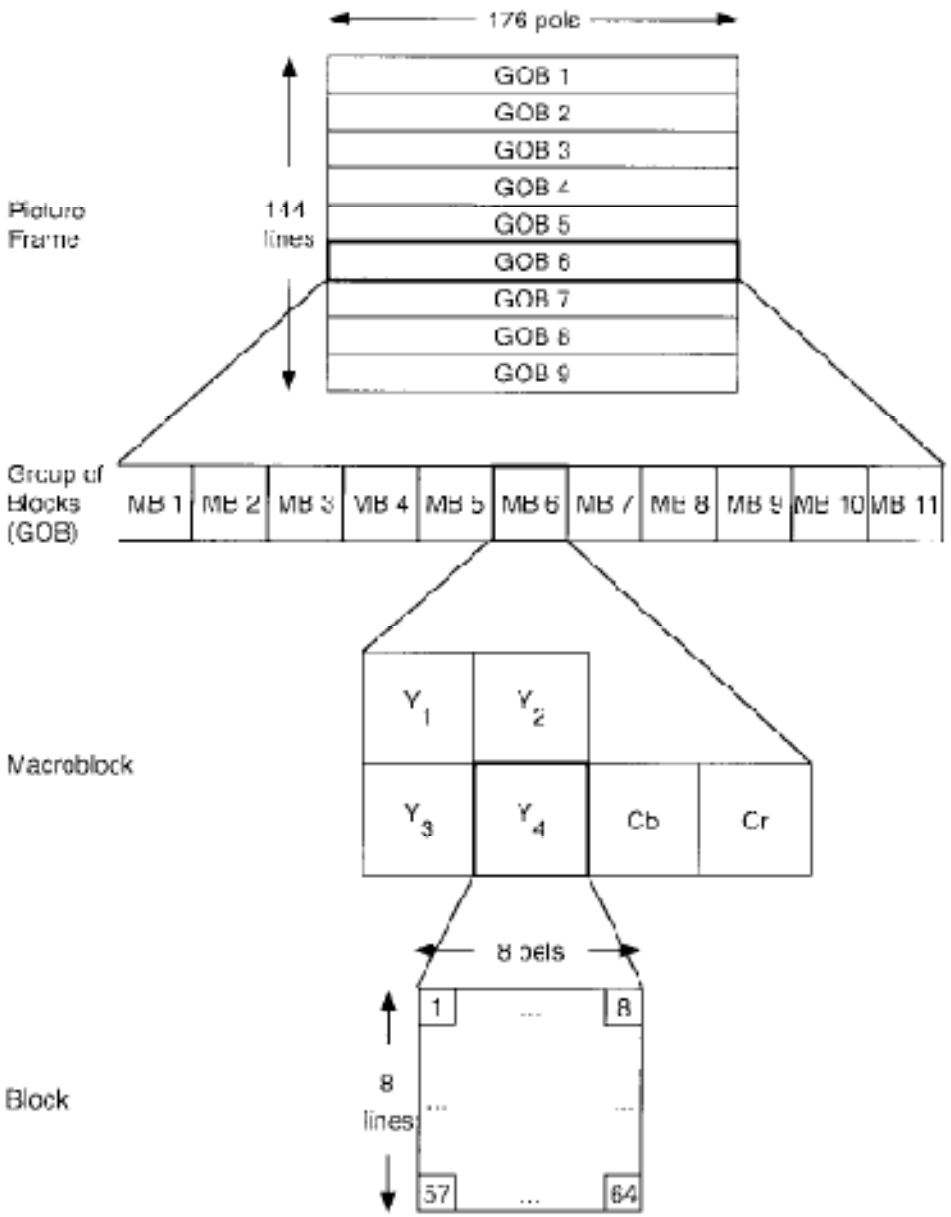
The most important motivation to develop recommendation H.263 was the need to guarantee interoperability between videotelephony terminals for the analogue telephone network, avoiding a normative hole in face of the proprietary developments in the market.

In this context (need to standardize quickly), recommendation H.263 is mostly based on existing technology, notably H.261 and MPEG-1 Video, with some adaptations regarding low rates and low resolutions, targeting the fast deployment of compliant products in the market.

Recommendation H.263: Requirements

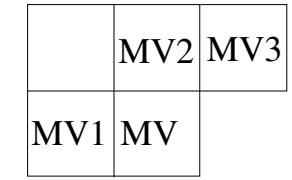


- **Mostly based in existing technology in order it may be rather fast to move to the deployment phase**
- **Low complexity and low cost**
- **Easy interoperability with available services, e.g. H.320/H.261 videotelephony, JPEG images, facsimile**
- **Resilience to transmission errors**
- **Flexibility to accommodate future extension, notably towards higher bitrates and mobile environments**
- **Flexibility in trading spatial and temporal resolutions**
- **Maximization of the video subjective quality**

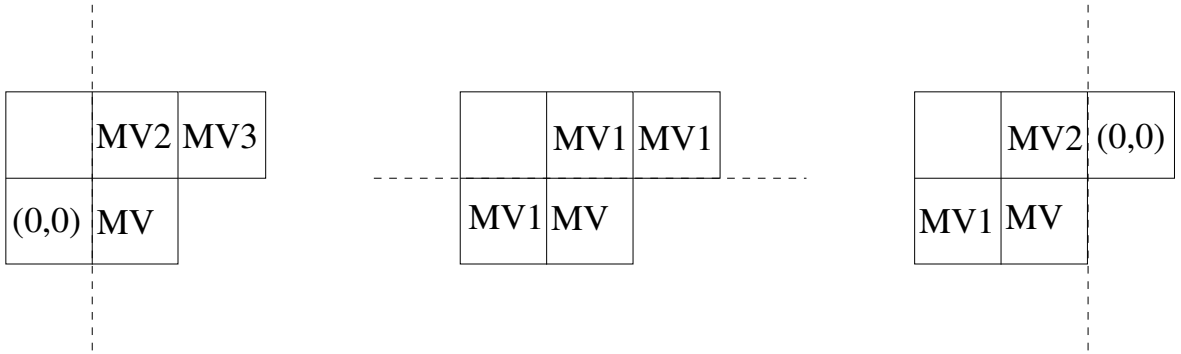


- **Spatially, a video sequence is organized in a hierarchical structure with 4 layers:**
  - **Image (I ou P)**
  - **Group Of Blocks (GOB)**
  - **Macroblock (MB)**
  - **Block**
- **There are 9 GOBs for QCIF and 18 for CIF images (MBs in a row).**

Differential Motion Vectors



MV : Current motion vector
MV1: Previous motion vector
MV2: Above motion vector
MV3: Above right motion vector



---------- : Picture or GOB border

- **Each motion vector is differentially coded regarding its prediction taken as the median, for each component (*x* and *y*), of 3 candidate predictors, MV1, MV2 and MV3.**
- **If any of the candidate predictors was Intra coded or Skipped, its motion vector prediction is taken as 0.**

DCT coefficients are coded as (*last, run, level)* symbols.

- The more frequent symbols are coded with VLC codes while the remaining symbols are coded with a 22-bits fixed length word formed with 7 bits for ESCAP, 1 bit for LAST, 6 bits for RUN and 8 bits for LEVEL.
- EOB in H.261 corresponds to the '10' codeword which could mean in the worst case 99×6×2×10 = 11880 bit/s for a QCIF image.

INDEX	LAST	RUN	LEVEL	BITS	VLC CODE
0	0	0	1	3	10s
1	0	0	2	5	1111 s
2	0	0	3	7	0101 01s
3	0	0	4	8	0010 111s
4	0	0	5	9	0001 1111 s
5	0	0	6	10	0001 0010 1s
6	0	0	7	10	0001 0010 0s
7	0	0	8	11	0000 1000 01s
8	0	0	9	11	0000 1000 00s
9	0	0	10	12	0000 0000 111s
10	0	0	11	12	0000 0000 110s
11	0	0	12	12	0000 0100 000s
12	0	1	1	4	110s
13	0	1	2	7	0101 00s
14	0	1	3	9	0001 1110 s
15	0	1	4	11	0000 0011 11s
16	0	1	5	12	0000 0100 001s
17	0	1	6	13	0000 0101 0000s
18	0	2	1	5	1110 s
19	0	2	2	9	0001 1101 s
20	0	2	3	11	0000 0011 10s
21	0	2	4	13	0000 0101 0001s
22	0	3	1	6	0110 1s
23	0	3	2	10	0001 0001 1s

Bitrate Control

Recommendation H.263 does not precisely state its target bitrate range.

It is well known that H.263 is more efficient (and the only solution until that time) for the lower bitrates; moreover, it may normally compete with advantage with H.261 for higher bitrates.

The optimal trade-off between the spatial and temporal resolutions depends on the video content and it strongly impacts the final subjective quality.

Bitrate Control Methods

The bitrate control for the output coded stream may be made in several (non-normative) ways:

- Information pre-processing, e.g. filtering high frequency information.
- Variation of the DCT coefficients quantization steps.
- Application of a block significance criterion, e.g. only the blocks with a certain activity are considered significant.
- Temporal subsampling, e.g. skipping images**.**

In H.263, temporal subsampling is a rather important tool for bitrate control; this means there are normally no guarantees in terms of a minimum temporal resolution.

H.263 Optional Modes

- **Unrestricted Motion Vector Mode**
  - Motion vectors referencing pixels outside the prediction image
  - Extending the motion vectors range
- **Advanced Prediction Mode**
  - Four motion vectors per macroblock
  - Overlapped block motion compensation (OBMC)
- **Syntax-Based Arithmetic Coding Mode**
- **PB-Frames Mode**

Final Remarks

- **Recommendation H.263 is mainly based on recommendation H.261; some improvements are made to reach better quality, notably for the lower rates, at the cost of some additional complexity.**
- **Typically, H.263 obtains similar qualities as for the H.261 with about 2-2.5 times less bitrate.**
- **Nowadays, many H.263 compliant products and services exist in the market, notably for mobile networks.**
- **However, after the emergence of the H.264/AVC standard, H.263 does not represent anymore the video coding state-of-the-art, notably for low bitrates.**

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