Information and Coding

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Contents

Principles

- Let $x^n = x_1 x_2 \ldots x_n$ be the sequence of values (scalars or vectors) produced by an information source until time n.
- Predictive coding is based on encoding sequence $r^n = r_1 r_2 \ldots r_n$, instead of the original sequence x^n, where

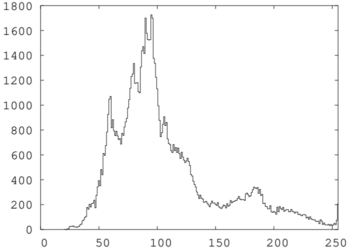
$$r_n = x_n - \hat{x}_n$$

and

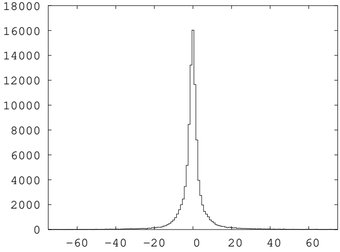
$$\hat{x}_n = p(x^{n-1}) = p(x_1 x_2 \ldots x_{n-1})$$

- The $\hat{x}_n$ are the estimates and the values of the sequence r^n are the residuals.
- Function $p()$ is the estimator or predictor.
- The aim of predictive coding is to have $H(r^n) < H(x^n)$.

Example



Original
$H = 7.26$ bits/symbol



Predictor 1 JPEG
$H = 4.49$ bits/symbol

Simple 1D prediction

- Simple polynomial predictors used in some audio encoders:

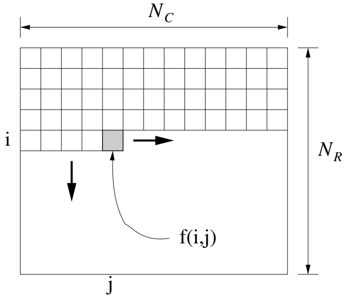
$$\begin{cases} \hat{x}_n^{(0)} = 0 \\ \hat{x}_n^{(1)} = x_{n-1} \\ \hat{x}_n^{(2)} = 2x_{n-1} - x_{n-2} \\ \hat{x}_n^{(3)} = 3x_{n-1} - 3x_{n-2} + x_{n-3} \end{cases}$$

and the corresponding residuals, computed efficiently:

$$\begin{cases} \hat{r}_n^{(0)} = x_n \\ \hat{r}_n^{(1)} = r_n^{(0)} - r_{n-1}^{(0)} \\ \hat{r}_n^{(2)} = r_n^{(1)} - r_{n-1}^{(1)} \\ \hat{r}_n^{(3)} = r_n^{(2)} - r_{n-1}^{(2)} \end{cases}$$

Predictive image coding techniques

- Typically, images are encoded from left to right, top to bottom, i.e., in raster-scan order:



- In this case, the sequence x^n is obtained by concatenating the first $\lfloor n/N_c \rfloor$ image rows, plus the $n \bmod N_c$ pixels from row number $\lfloor n/N_c \rfloor + 1$.

Predictive image coding techniques

- Other approaches use hierarchical decompositions (or multi-resolution).
- This is the case of the HINT method (Hierarchical INTerpolation):

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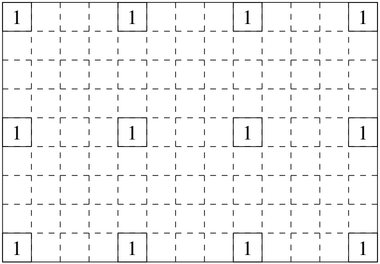

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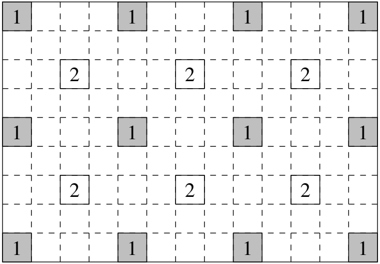

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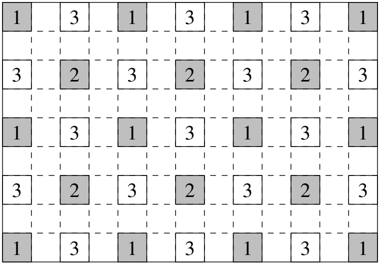


Predictive image coding techniques

- Other approaches use hierarchical decompositions (or multi-resolution).
- This is the case of the HINT method (Hierarchical INTerpolation):

1		3		1		3		1		3		1
	4		4		4		4		4		4	
3		2		3		2		3		2		3
	4		4		4		4		4		4	
1		3		1		3		1		3		1
	4		4		4		4		4		4	
3		2		3		2		3		2		3
	4		4		4		4		4		4	
1		3		1		3		1		3		1

Predictive image coding techniques

- Other approaches use hierarchical decompositions (or multi-resolution).
- This is the case of the HINT method (Hierarchical INTerpolation):

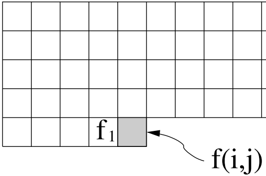
1	5	3	5	1	5	3	5	1	5	3	5	1
5	4	5	4	5	4	5	4	5	4	5	4	5
3	5	2	5	3	5	2	5	3	5	2	5	3
5	4	5	4	5	4	5	4	5	4	5	4	5
1	5	3	5	1	5	3	5	1	5	3	5	1
5	4	5	4	5	4	5	4	5	4	5	4	5
3	5	2	5	3	5	2	5	3	5	2	5	3
5	4	5	4	5	4	5	4	5	4	5	4	5
1	5	3	5	1	5	3	5	1	5	3	5	1

Predictors

- For efficient encoding, the estimated values should be as close as possible to the real values, i.e., the r_k values should be small.
- The decoder must be able to generate the same sequence, $\hat{x}^n$, of estimated values, i.e., the predictor cannot introduce any error during encoding / decoding.
- Therefore, the predictor must be causal, and, in lossy coding, the predictor at the encoder must use the reconstructed values, $\tilde{x}^{n-1}$, instead of the original values, x^{n-1}.

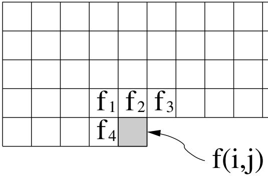
Predictors

- Generally, the complexity of the predictor depends on two aspects:
  - The number of values used for calculating the estimates (the order of the predictor).
  - The spatial (or temporal) configuration of these values.
- Consider the example of a spatial predictor of order 1, where the estimated value is given by the immediately preceding value:



Predictors

- This type of predictor can be easily extended to higher orders, using the last *k* processed pixels of the image.
- However, for orders higher than 3 or 4, the efficiency does not increase significantly.
- This happens because images are 2D signals, not 1D sequences of data.
- Therefore, generally, the spatial configurations used for predictive image coding have a 2D shape:

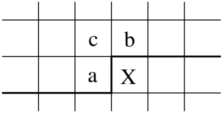


Lossless predictive coding

- One of the main advantages of predictive coding is allowing a simple design of lossless encoders.
- In fact, most lossless encoders for audio and image rely on predictive coding techniques.
- However, for lossless coding, there is an additional constraint regarding the predictor: the estimates generated must be platform independent.
- Generally, this constraint implies that the predictor can use only integer arithmetic.

Linear prediction: the lossless mode of JPEG

- The lossless mode of JPEG (ISO/IEC 10918-1, ITU-T T.81, 1992) provides seven linear predictors:



Mode	Predictor
1	a
2	b
3	c
4	$a + b - c$
5	$a + (b - c)/2$
6	$b + (a - c)/2$
7	$(a + b)/2$

- Generally, the performance of the several predictors may vary considerably from image to image.
- If encoding time is not a problem, then all of them can be tested and the one with the best compression rate chosen.

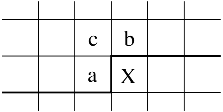
Linear prediction: the lossless mode of JPEG

Example:

Predictor	1	2	3	4	5	6	7
Entropy	4.49	4.21	4.74	4.17	4.16	**4.04**	4.10

The nonlinear predictor of JPEG-LS

- JPEG-LS (ISO/IEC 14495-1, ITU-T T.87, 1999) uses a predictor based on the same spatial configuration as that of JPEG:



- However, instead of a linear predictor, it uses the nonlinear predictor

$$\hat{x} = \begin{cases} \min(a, b) & \text{if} \quad c \geq \max(a, b) \\ \max(a, b) & \text{if} \quad c \leq \min(a, b) \\ a + b - c & \text{otherwise} \end{cases}$$

- Note that the linear part of this predictor ($a + b - c$) is the same as predictor number 4 of JPEG.

The nonlinear predictor of JPEG-LS

Example:

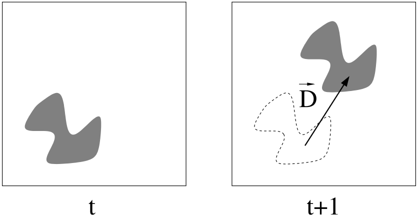
(a)

(b)

Predictor	1	2	3	4	5	6	7	JLS
Entropy (a)	4.49	4.21	4.74	4.18	4.16	4.04	4.10	**3.98**
Entropy (b)	5.60	5.05	5.82	5.19	5.23	4.97	5.15	**4.93**

Motion compensation

- Typically, the differences between consecutive frames of a video sequence are due to motion of the scene objects.
- Exceptions occur when there are scene changes, zoom-in / zoom-out operations and camera translation.



- To explore this redundancy, it is frequent to use temporal prediction (interframe compression), which relies on motion compensation.

Motion compensation

- Conditional replenishment video coding:
  - Finds zones in the video frame where there were changes with respect to the previous frame.
  - Only those zones are encoded.
  - This technique does not use motion compensation. It just performs a detection of temporal activity.
- Video coding based on motion compensation involves the following steps:
  - Estimation of the motion vectors.
  - Compensation, i.e., temporal prediction.
  - Encoding of the motion vectors.
  - Encoding of the prediction residuals.

Motion compensation

- There are a large number of techniques for motion detection, but one of them is clearly the most common approach for video coding.
- For each frame block (for example, of $N \times N$ pixels), it seeks the position where it minimizes some measure in relation to the previous frame (the reference frame).
- Note that this approach tries to find the position that minimizes a measure of interest, which might not correspond to the true motion in the scene…

Motion compensation

- Typically, we want to minimize some measure $C(i,j)$, such as

$$C(i,j) = \sum_{r=1}^{N} \sum_{c=1}^{N} d\Big(g(r,c,t) - g(i+r, j+c, t+1)\Big)$$

  where $d(\cdot)$ is, for example, $(\cdot)^2$ or $|\cdot|$.
- Due to complexity constraints, searching is limited to a neighborhood of $(N+2\Delta) \times (N+2\Delta)$ pixels around the block, i.e., $-\Delta \leq i,j \leq \Delta$.
- If exhaustive search is used, it is guaranteed to find the minimum of $C(i,j)\ldots$
- This approach is generally computationally too demanding, hence other sub-optimal techniques have been proposed.

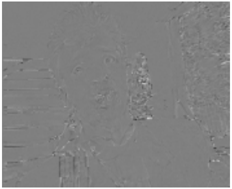
Motion compensation

- Example:

Frame 200

Frame 201

Direct difference
$H = 5.23$ bpp

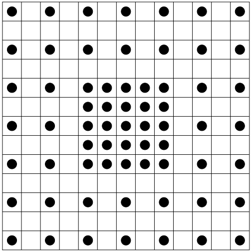


Motion compensation
$H = 4.38$ bpp

Motion compensation

- Several of the sup-optimal approaches for finding the best reference block rely on spatial sub-sampling.
- For example, considering that the most probable zone for finding the reference block is in the near neighborhood of the block, then we may use the following scheme:

Total: 169 blocks

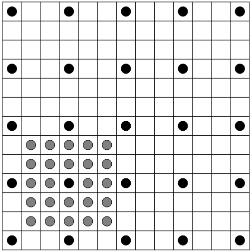


Sub-optimal: 65 blocks

Motion compensation

- If we consider that after finding a reasonably good reference block it is probable that others better than itself can be found in the near neighborhood, then we can use a greedy search:

Total: 169 blocks

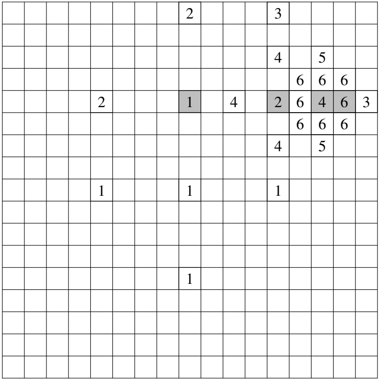


Sub-optimal: 49 blocks

- A number of other variants of local search have been proposed…

Motion compensation

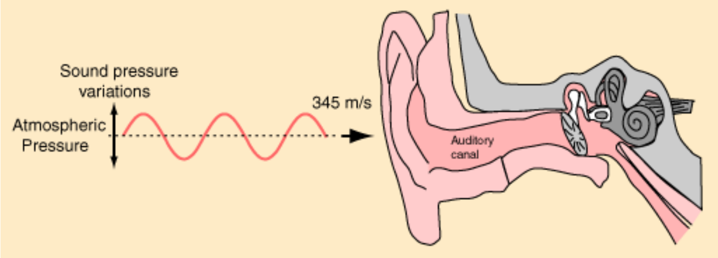
- For example, the logarithmic search:



Contents

The human auditory system

- Humans perceive **sound** by the sense of hearing. By sound, we commonly mean the vibrations that travel through air and are audible to humans.



The human auditory system

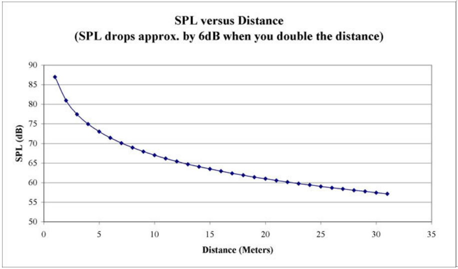
- **Audio** is the electrical representation of sound.
- Generally, humans can perceive variations in sound pressure from 16-20 Hz to 20-22 kHz.
- However, our capacity for perceiving sounds of very small amplitude varies according to frequency, being maximum between 2 and 4 kHz.
- The human voice produces frequencies approximately between 200 Hz and 8 kHz. Telephone communications limit this range from 300 Hz to 3.4 kHz (200 Hz to 3.2 kHz in the USA) .

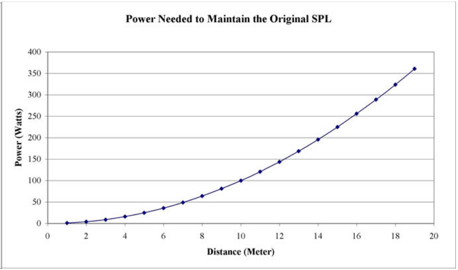
The human auditory system

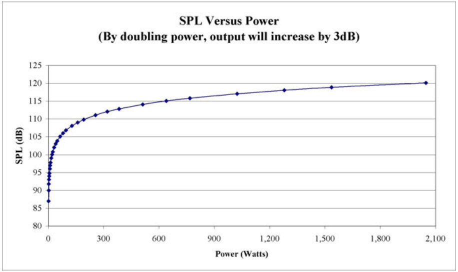
- Normally, the amplitude range that we can hear is about 100 dB:

Source of sound	Sound pressure level (dB)
Jet engine at 30 m	150
Jet engine at 100 m	140
Threshold of pain	125–130
Hearing damage (short-term exposure)	120
Maximum output of some MP3 players	110
Hearing damage (long-term exposure)	100
Major road at 10 m	80–90
TV (at home level) at 1 m	60
Normal talking at 1 m	40–60
Very calm room	20–30
Calm breathing	10
Auditory threshold at 2 kHz	0

The human auditory system

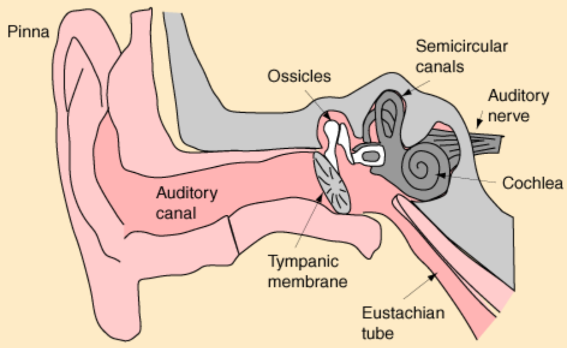






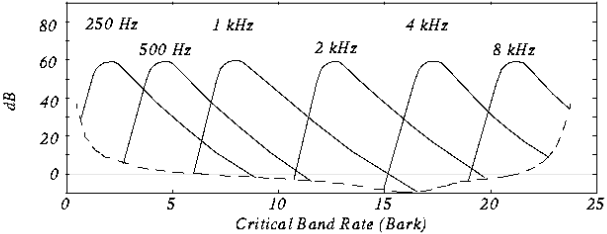
The human auditory system

- The auditory system can roughly be described as a **bandpass** filter-bank, consisting of strongly overlapping bandpass filters.



The human auditory system

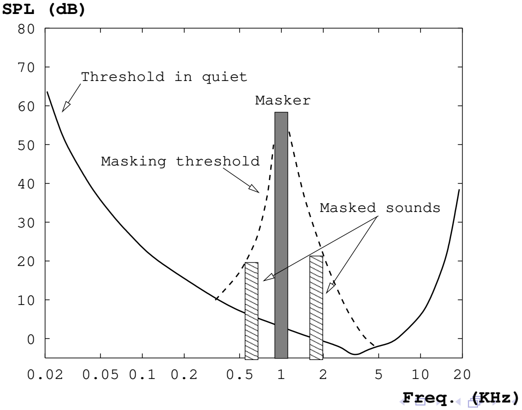
- These “filters” have bandwidths in the order of 50 to 100 Hz for signals below 500 Hz and up to 5000 Hz for signals at high frequencies.



- They are called **critical bands.**
- Twenty-five critical bands, covering frequencies of up to 20 kHz, are normally taken into account.

The human auditory system

- **Simultaneous masking** is a frequency domain phenomenon where a low-level signal (maskee) can be made inaudible (masked) by a simultaneously and close in frequency stronger signal (masker).

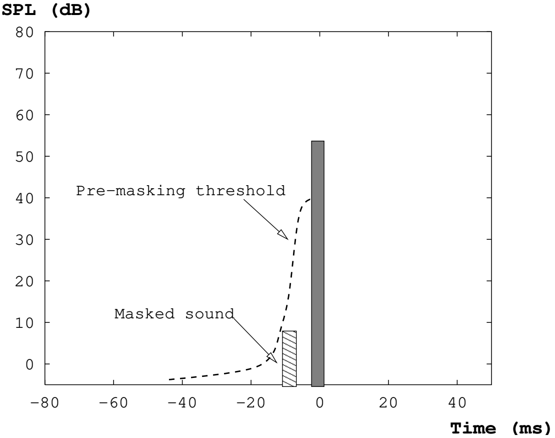


The human auditory system

- In addition to simultaneous masking, the time domain phenomenon of **temporal masking** plays an important role in human auditory perception.
- Temporal masking may occur when two sounds appear within a small interval of time.
- Depending on the individual sound pressure levels, the stronger sound may mask the weaker one, even if the maskee precedes the masker…

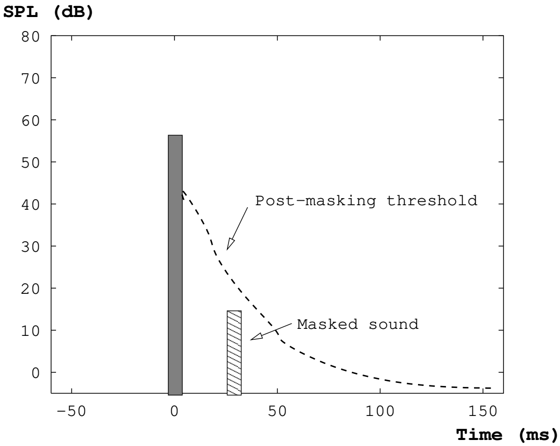
The human auditory system

- The **pre-masking** has a duration of about 5 to 20 ms:



The human auditory system

- The **post-masking** has a duration of about 50 to 200 ms:



Contents

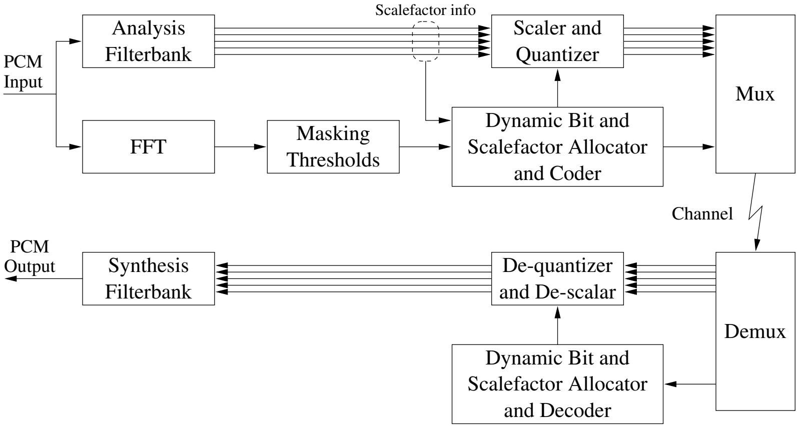
MPEG–1

- MPEG–1 audio coding is organized in three layers, I, II and III, with increasing performance, but also complexity and delay.
- It allows sampling frequencies of 32, 44.1 and 48 kHz, and bitrates between 32 kb/s (mono) and 448 kb/s (Layer I), 384 kb/s (Layer II) and 320 kb/s (Layer III).
- In terms of transparent CD (stereo) quality, the bitrates and compression rates are, approximately,

Layer	Bitrate	Compression rate
I	384 kb/s	4
II	192 kb/s	8
III	128 kb/s (VBR)	12

MPEG–1

MPEG–1 layer I and II



MPEG–1

- The analysis filterbank has 32 subbands, equally spaced in frequency.
- Each block is formed of 384 audio samples (8 ms for $f_s = 48$ kHz), meaning that each subband contains 12 samples.
- For $f_s = 48$ kHz, the width of each subband is 750 Hz.
- Usually, the bits are dynamically assigned to the coefficients of the subbands according to a psychoacoustic model (how to do this is not part of the standard).
- For each block of 12 coefficients (subband), an uniform quantizer (from 15 available) is selected, according to predefined levels of quality and compression.

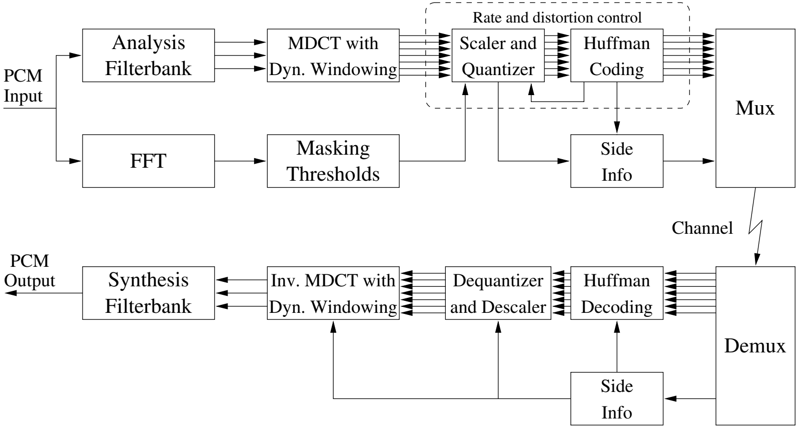
MPEG–1

MPEG–1 layer III

- MPEG–1 layer III has several differences in relation to the other two layers, being much more complex.
- It is based on hybrid coding: subband and transform.
- It allows **variable bitrate coding (VBR)**.
- It relies on a technique designated analysis by synthesis for dynamic bit assignment.
- It uses an advanced **pre-echo control**.
- It uses non-uniform quantization and statistical coding.

MPEG–1

MPEG–1 layer III



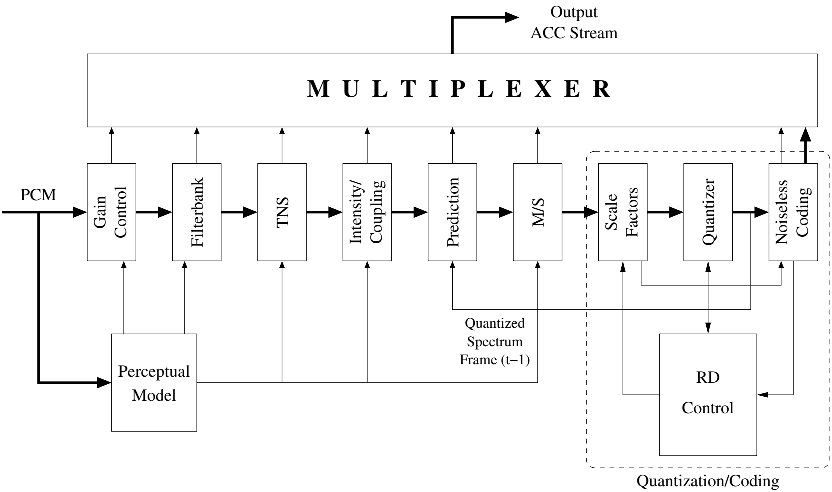
MPEG–2

- The MPEG–2 audio coding standard includes the MPEG–1 audio and introduces extensions for multi-channel configurations.
- MPEG–2 provides two audio coding standards: one forward and backward compatible with MPEG–1, the other incompatible.
- Forward compatibility means that a multi-channel decoder understands MPEG–1 mono and stereo streams.
- Backward compatibility means that a MPEG–1 decoder is able to extract stereo audio from a MPEG–2 multi-channel stream.
- The second standard is MPEG–2 AAC (Advanced Audio Coding).

MPEG–2 AAC

- Some parts are identical to those of MPEG–1/2 layer III:
  - Filterbank with dynamic windowing.
  - Non-uniform quantizers.
  - Huffman coding.
- MPEG–2 AAC defines 3 coding profiles:
  - The low complexity profile.
  - The main profile.
  - The scalable sampling rate profile.

MPEG–2 AAC


Output
ACC Stream
M U L T I P L E X E R
PCM
Gain
Control
Filterbank
TNS
Intensity/
Coupling
Prediction
M/S
Scale
Factors
Quantizer
Noiseless
Coding
Perceptual
Model
Quantized
Spectrum
Frame (t−1)
RD
Control
Quantization/Coding

MPEG–4

- The MPEG–4 coding standard provides tools for coding audio objects, such as **natural audio** (for example, speech and music) and **synthetic audio**, aiming several applications:
  - Telephone over the Internet.
  - High quality music.
  - Text-to-speech conversion.
  - Synthesized music.
  - ...
- The synthesized audio can be obtained through text (TTS) or instrumental descriptions.

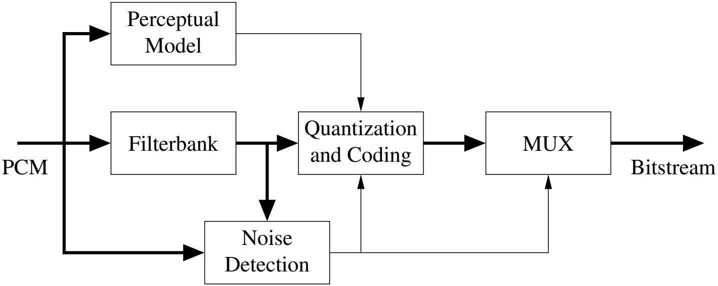
MPEG–4

- The encoding of the **natural audio** relies on several techniques:
  - Harmonic vector excitation coding (HVXC), for $f_s \leq 8$ kHz, and bitrates between 2 and 4 kbp/s (until 1.2 kbp/s, for VBR).
  - Code excited linear predictive (CELP), for $8 \leq f_s \leq 16$ kHz, and bitrates between 4 and 24 kbp/s.
  - Transform-domain weighted interleave vector quantization (TwinVQ) and AAC, for $f_s \geq 8$ kHz, and bitrates greater than 6 kbp/s.

MPEG–4

Perceptual noise substitution

- Produces a perceptual equivalent signal, instead of trying to reproduce the original waveform.
- It is used for audio components similar to noise.
- If this type of signal is detected in a certain band, then instead of coding the coefficients, it is encoded their total power.



MPEG–4

Low-delay coding

- Mode used in bi-directional, real-time, communications, where long delays are not acceptable (the “standard” encoder may introduce delays of several hundreds of ms).
- It uses windows with half the normal size.
- It does not use dynamic window adaptation (this implies long delays).
- The bit reservoir is minimized or even eliminated.

MPEG–4

Scalable coding

- The MPEG–4 allows a limited number of layers (typically 2 to 4). This is the **large-step scalability**.