Audio coding hints

TLC Networks Group
firstname.lastname@polito.it
http://www.telematica.polito.it/

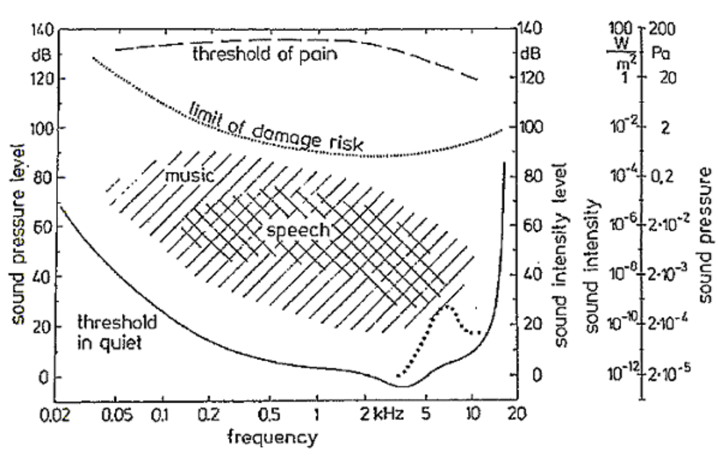
Sound perception

- Bandwidth: 20 Hz – 20 kHz
- Dynamics: ~96 dB
- Frequency resolution is not constant: as the frequency increases, the ability to distinguish among sounds at close frequencies decrease
- Also the amplitude resolution is not constant
- In both cases, very often logarithmic scales are used

Sound perception

- The perceived amplitude is a function of the frequency
  - For the same intensity a sound is perceived as more or less strong depending on the frequency composing it
- Critical bands are frequency bands over which the perceived amplitude is more or less uniform
- Critical band size range between 50 Hz (at the minimum audible frequencies) and 4 kHz (at the maximum audible frequencies)
- As a first approximation, the audible spectrum can be subdivided in 26 critical bands

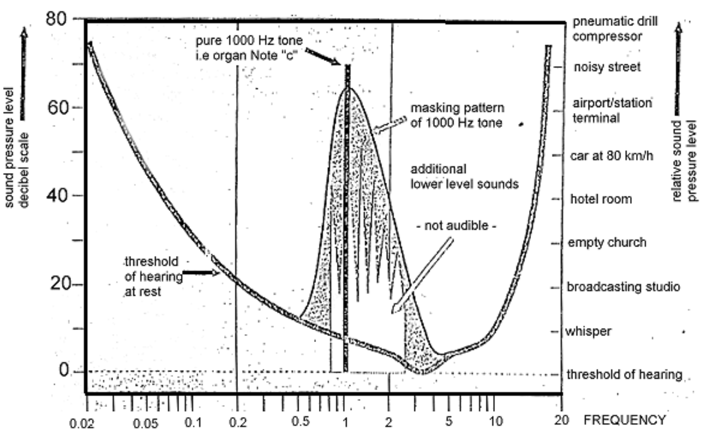
Sound preception – audible area



Masking

- A signal masks (making non audible) signals of smaller amplitude which are close in time or frequency
- The masking effect depends on the time or the frequency distance between the two signals
- Further dependencies from amplitude, frequency, type (tone or noise) of the masking signal

Frequency masking



Audio coding

- Non compressed signal, CD quality:
  - 44100 Hz, 16 bit per sample, two channels
  - ➔ Rate: 1.4 Mbit/s
- ADPCM techniques may reduce the bit rate
- LPC or CELP cannot be used due to the difficulty in source modelling
  - Too diverse sources
- Best results obtained via psicoacustic coders that exploits the ear charateristics and perceptual limits

MPEG coding

- MPEG (Moving Pictures Experts Group) is a ISO working group that defines standard for audio (and video) signals
- The standard specifies the bitstream format, the coding/decoding and the conformity test
- Details on how to implement the coder/decoder are not specified
  - Any designer can pursue its own solution, in the framework defined by the standard
  - Interoperabilty guaranteed by the standard

MPEG coding

- Lossy compression
  - Part of the information contained in the original bitstream is lost
- Exploiting the ear characteristics and perceptive limitations ear the compression becomes “perceptually lossless”
- Group of expert listeners, in optimal hearing conditions, were not able to distinguish between the original bitstream and the coded bitstream with a 6:1 compression ratio

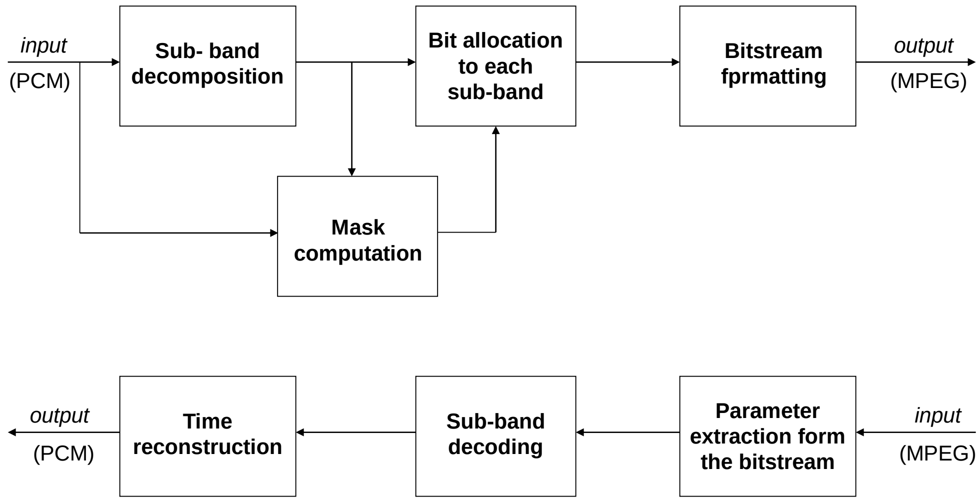
Coding algorithm

- Input signals are the PCM samples
- Frequency transform
- Spectrum divided in 32 sub-band of the same amplitude
- On the basis of the psicoacustic model the masking effects are defined in each sub-band
- As a consequence, the proper number of bits to be used in each sub-band is defined

Coding algorithm

- More precisely:
  - If the amplitude of the signaling in the sub-band is below the masking threshold, no coding is adopted
  - Otherwise, the number of bit is enough to ensure that the quantization noise σ_x^2 is below the masking threshold (for each additional bit σ_x^2 decreases by 6 dB)
- The frame in the bit stream contains header, number of bit in each sub-band, sample values and some auxiliary info (e.g. CRC)

Coding and decoding



Observation

- To simplify system implementation, the filter bank used to decompose the signal in sub-bands is not optimal:
  - The 32 sub-bands have the same size; better results could be obtained if the sub-band were corresponding to the critical band
  - Decomposing the signal spectrum in sub-band is not exactly reversible; the inverse operation introduces a (small) error
  - Filter band are not exactly disjoint. Thus, some influence from signals in adjacent sub-band exists

MPEG-1 Audio (1992)

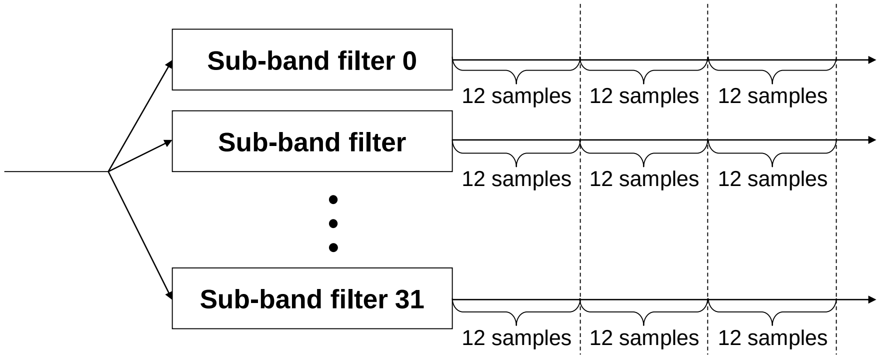
- Three sampling frequences: 32kHz, 44.1kHz e 48kHz
- Channel:
  - Mono (single channel)
  - Dual mono (2 independent channels, e.g. two languages)
  - Stereo (L and R channels independently coded)
  - Joint-stereo (exploit L and R channels correlation and perceptual properties to improve compression ratio)
- Bitrate:
  - Constant and predefined in the range 32 – 224 kbit/s
  - Different bitrate, possibly variable, are supported

Layers

- Three compression layers are envisioned
- Layer 1 is the base layer
- Layer 2 and layer 3 enhance system performance exploiting more complex blocks
- Roughly speaking, the three layers target applications whose bit rate is larger, equal or smaller than 128 kbit/s
- MPEG-1 Layer III is the MP3 format

Layer I

- For each sub-band 12 sample blocks are considered



Layer I

- For each block, a number of bit ranging from 0 to 15 is determined, plus a scaling factor
- The scaling factor is a multiplicative term that enhances the quantization (as in the APCM), represented with 6 bit
- Frame size in bit

Header (32)	CRC (0,16)	Bit allocation (128-256)	Scale factors (0-384)	Samples	Ancillary data

Layer II

- Enhances Layer I performance, by considering groups of 3 blocks of 12 samples each
- The same scaling factor for two/three consecutive blocks is used if the difference in dynamics are small enough, or not perceivable due to the time masking effect
- More efficient coding for samples, scaling factor and bit allocation

Layer III

- Much more complex
- Much better performance
- Compensates for non optimal filter characteristic with a “Modified Discrete Cosine Transform” (MDCT):
  - Sub-bands are further decomposed to enhance spectral resolution
  - Improve filter quality to reduce the aliasing

Layer III

- Block size dynamically modified (12 or 36 samples) depending on whether it is better to enhance the resolutionin time (transient) or frequency (stationary signals)
- More efficient non uniform sample quantization
- Entropic coding for quantized values
- Enhances the choice of the number of bit for each sub-band with a more sophisticated algorithm

MPEG-2 Audio (1994)

- Multichannel support: up to 5 HI-FI channels plus a low frequency channel (5.1 scheme)
- Up to 7 audio channel in several languages
- Three new sampling frequencies (16, 22.05 e 24 kHz)
- Support also for reduce bitrate, down to 8 kbit/s
- Partly compatible with MPEG-1
  - MPEG-2 decoder are able to decode MPEG-1 stream
  - MPEG-2 stream can be formatted to ensure that MPEG-1 decoder are able to extract two channels from the stream

Bibliografia

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- ISO/IEC International Standard IS 11172-3 *“Information Technology – Coding of Moving Pictures and Associated Audio for Digital Storage Media at up to 1.5 Mbits/s – Part 3: Audio”*