Parametric Audio Coding in MPEG-4

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The MPEG-4 Audio Standard

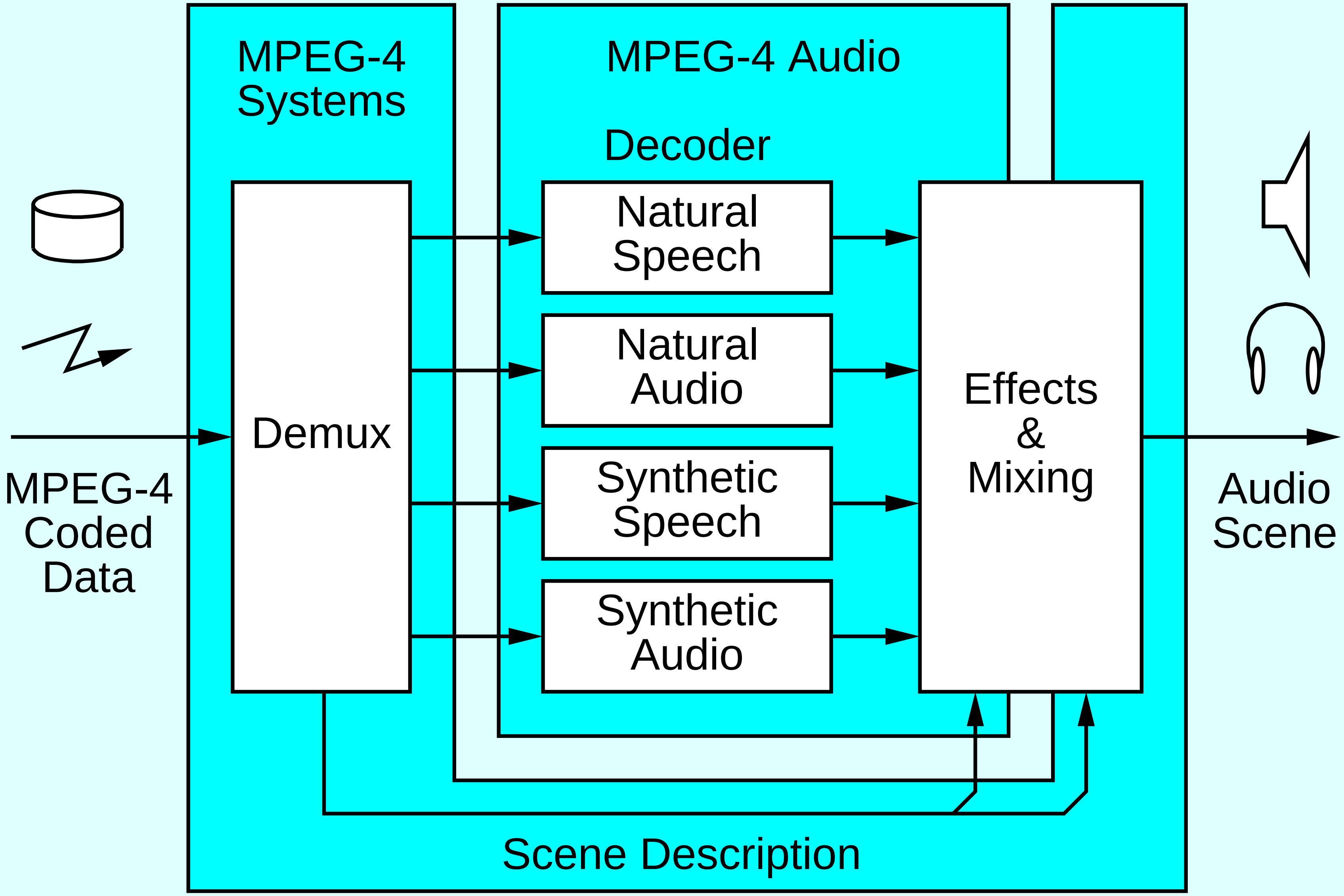
What is MPEG-4 Audio?

- ISO/IEC Standard 14496-3

  $\Rightarrow$ efficient coding of speech and audio objects

- Tools: speech and audio coding (2 .. 64+ kbit/s/ch)
- Additional functionalities
  - bitrate scalability (embedded coding)
  - error robustness
  - natural and synthetic content
  - combination of multiple objects in a scene

The MPEG-4 Audio Standard: Decoder



The MPEG-4 Audio Standard: Tools

Audio Tools: Coding of natural objects

- Speech: Original (4 kHz BW) 2 kbit/s
  - HVXC (parametric, 2 .. 4 kbit/s)
  - CELP (NB + WB, 4 .. 24 kbit/s)
- Audio: Original (Stereo) 48 kbit/s
  - TwinVQ (6 .. 16 kbit/s/ch)
  - AAC-scalable (16 .. 64+ kbit/s/ch)
  - BSAC (fine-step scalability)
  - LowDelay (20 ms delay)
  - HILN (parametric, 4 .. 16 kbit/s)

The MPEG-4 Audio Standard: Tools

Audio Tools: Coding of synthetic objects

- Speech: TTS Interface
- Audio: “Structured Audio” (DSP) incl. MIDI

Systems Tools: Composition of audio objects

- Mixing of audio objects
- Effects: DSP blocks from “Structured Audio”
- 3D Audio: “Environmental Spatialisation”

The MPEG-4 Audio Standard: Tools

3D Audio: “Environmental Spatialisation”

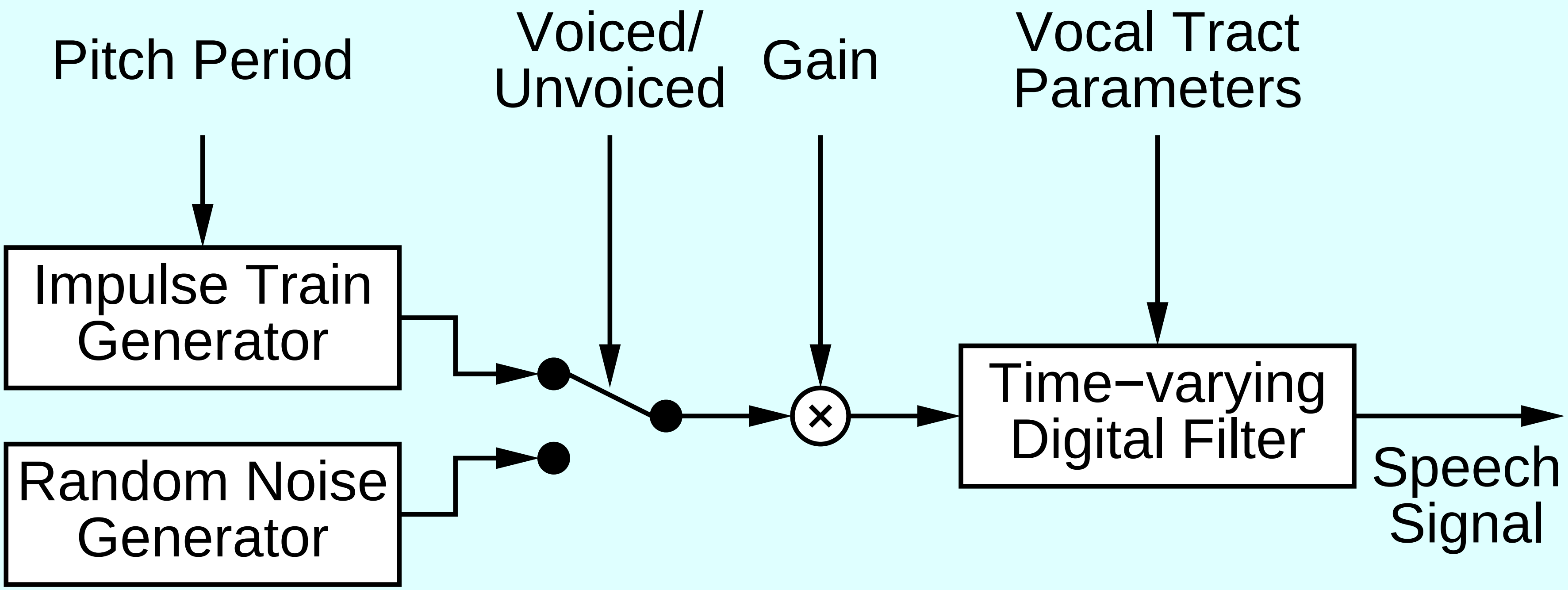
- Physical approach:
  description of acoustical properties of environment
  (room geometry, sound source position, ...)

  $\Rightarrow$ corresponding audio and visual scene

- Perceptual approach:
  high-level perceptual description of “audio scene”
  (room reverberance, source presence, ...)

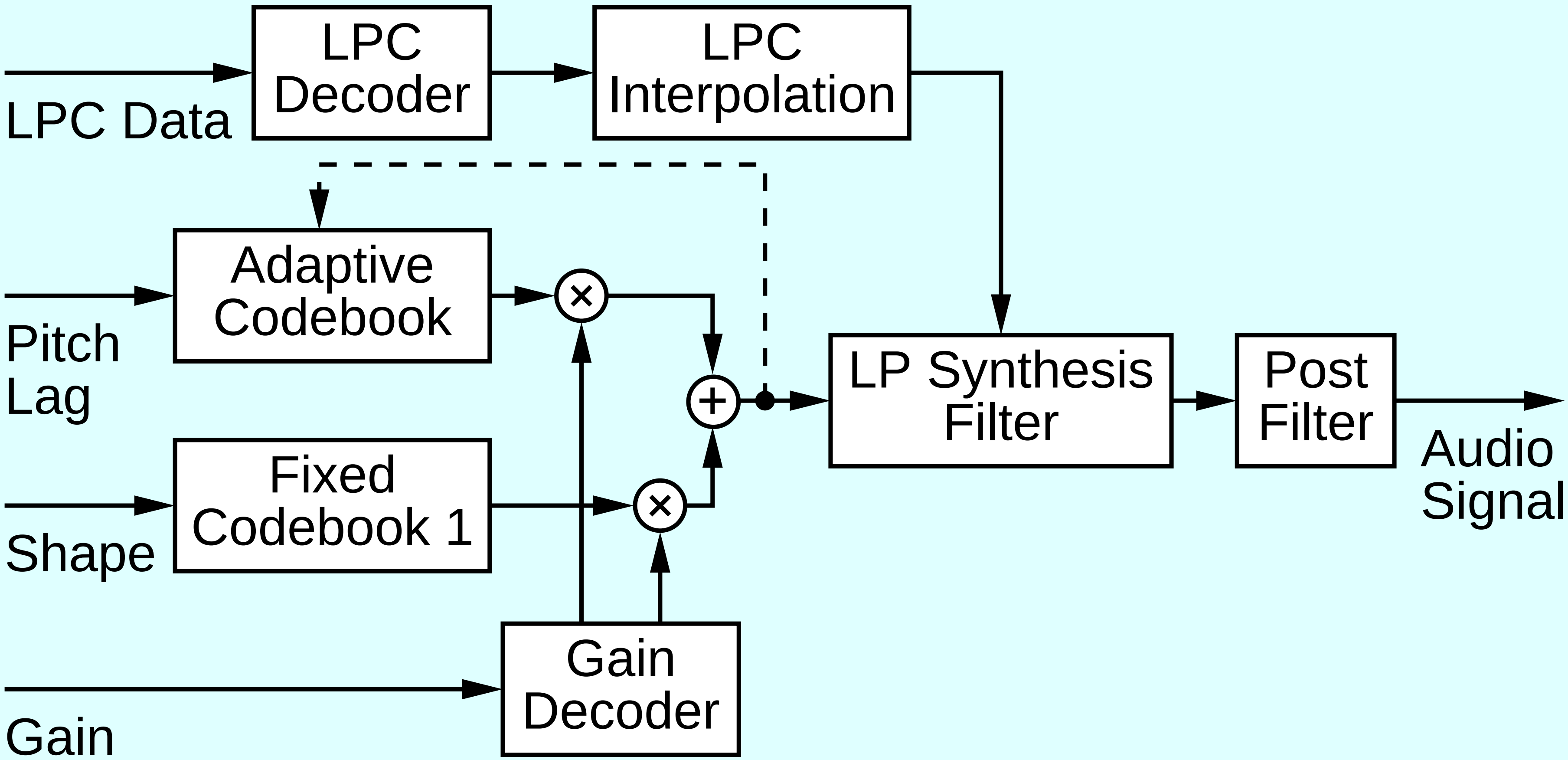
  $\Rightarrow$ audio and visual scene independent

Fundamentals: Speech Coding (CELP)



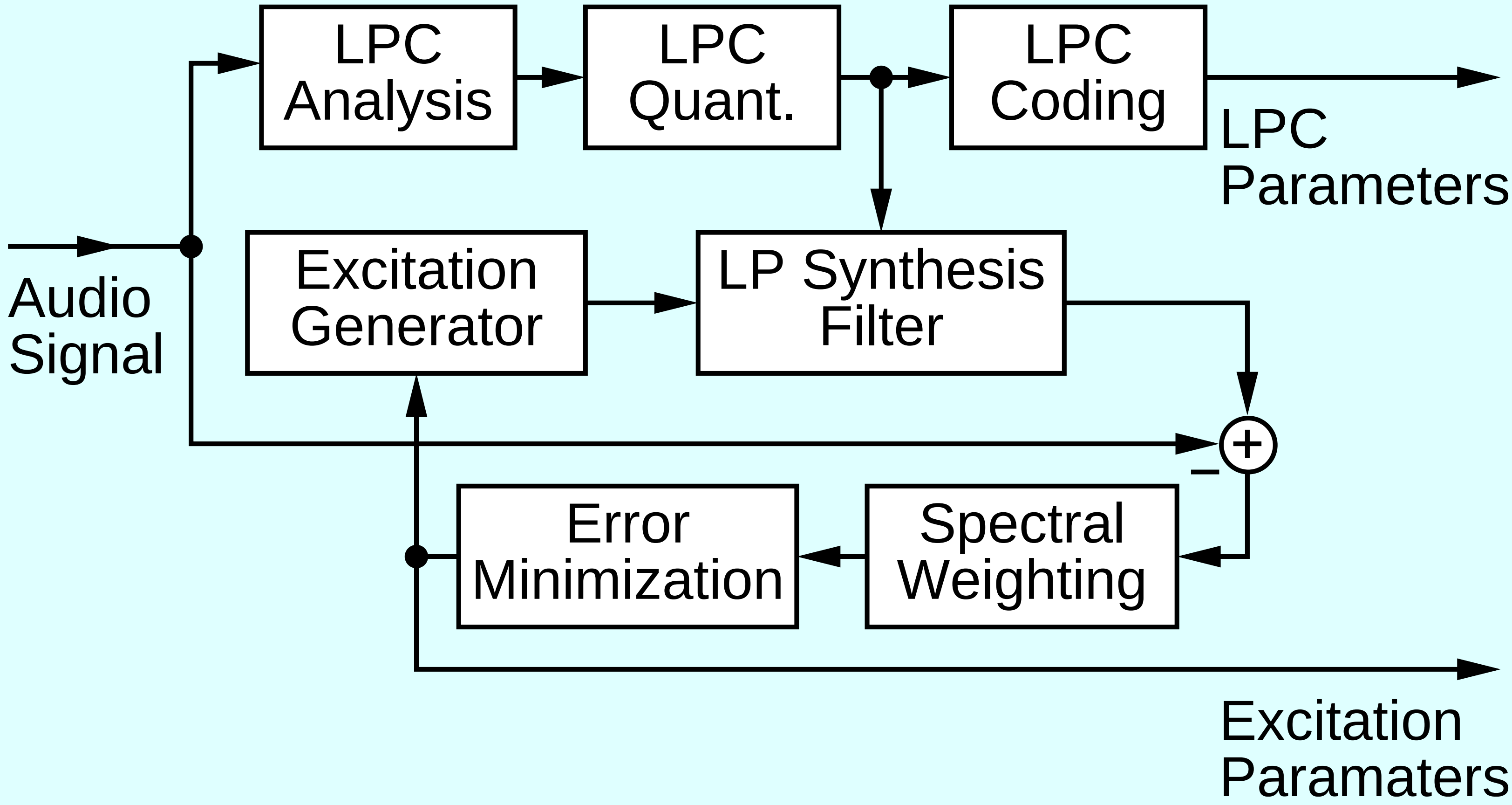
Simplified model of speech generation / synthesis

Fundamentals: Speech Coding (CELP)



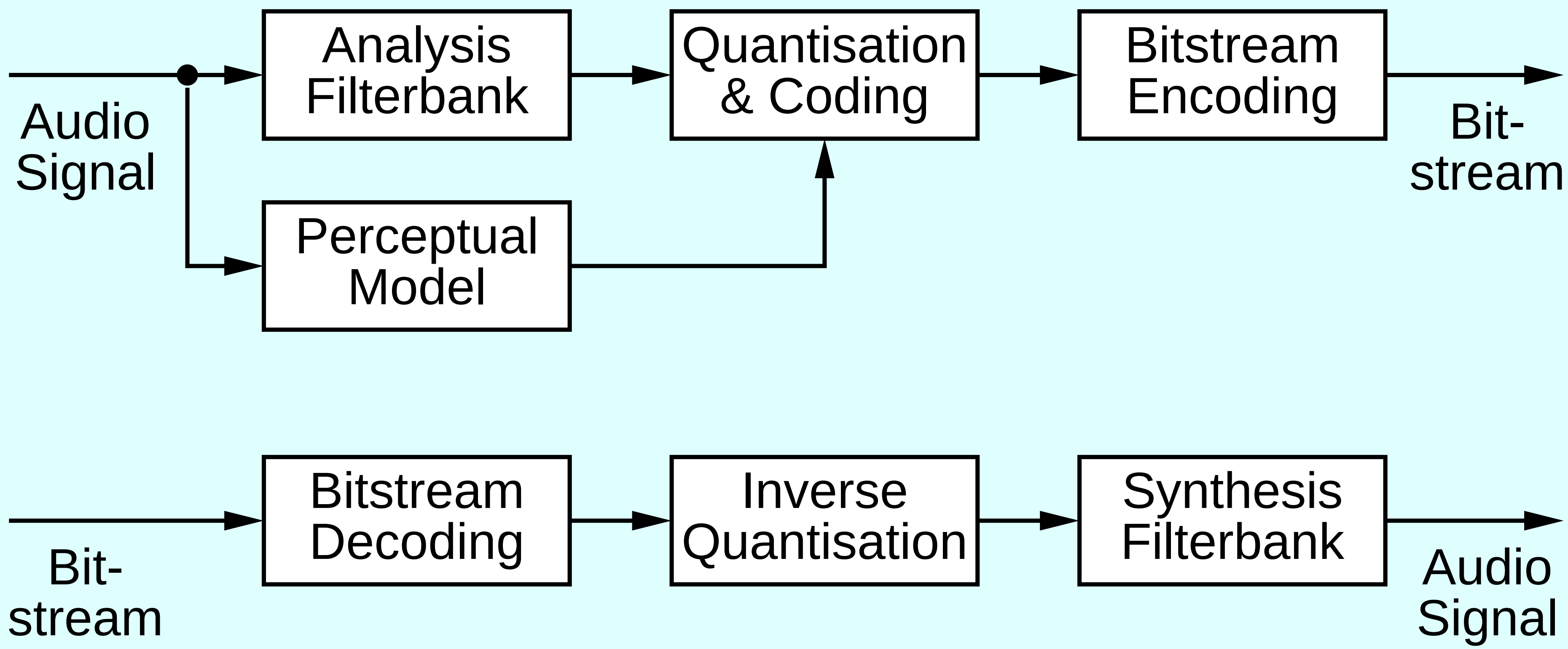
Basic structure of a CELP decoder

Fundamentals: Speech Coding (CELP)



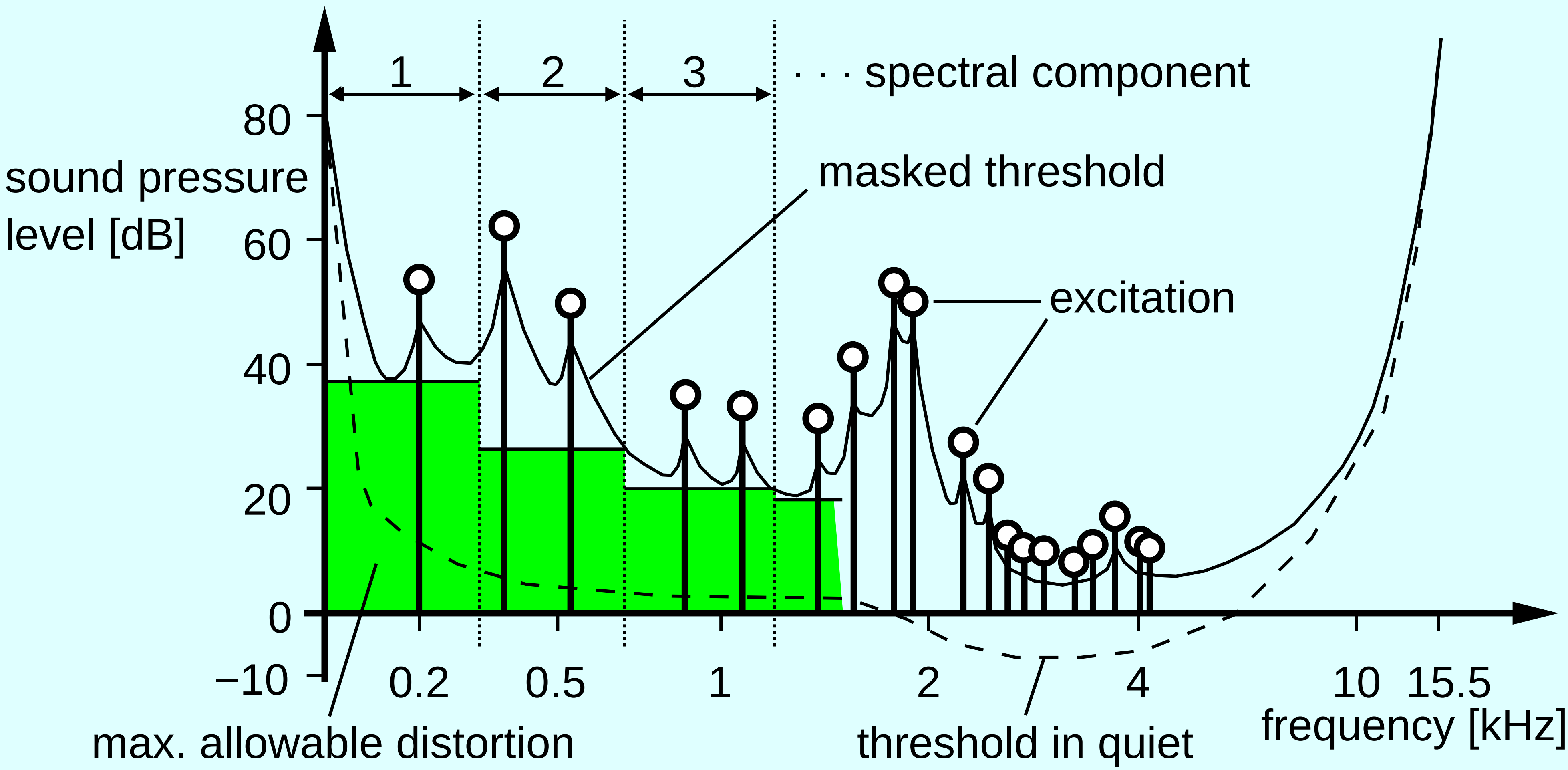
Basic structure of a CELP encoder

Fundamentals: Audio Coding (MPEG-1/2)



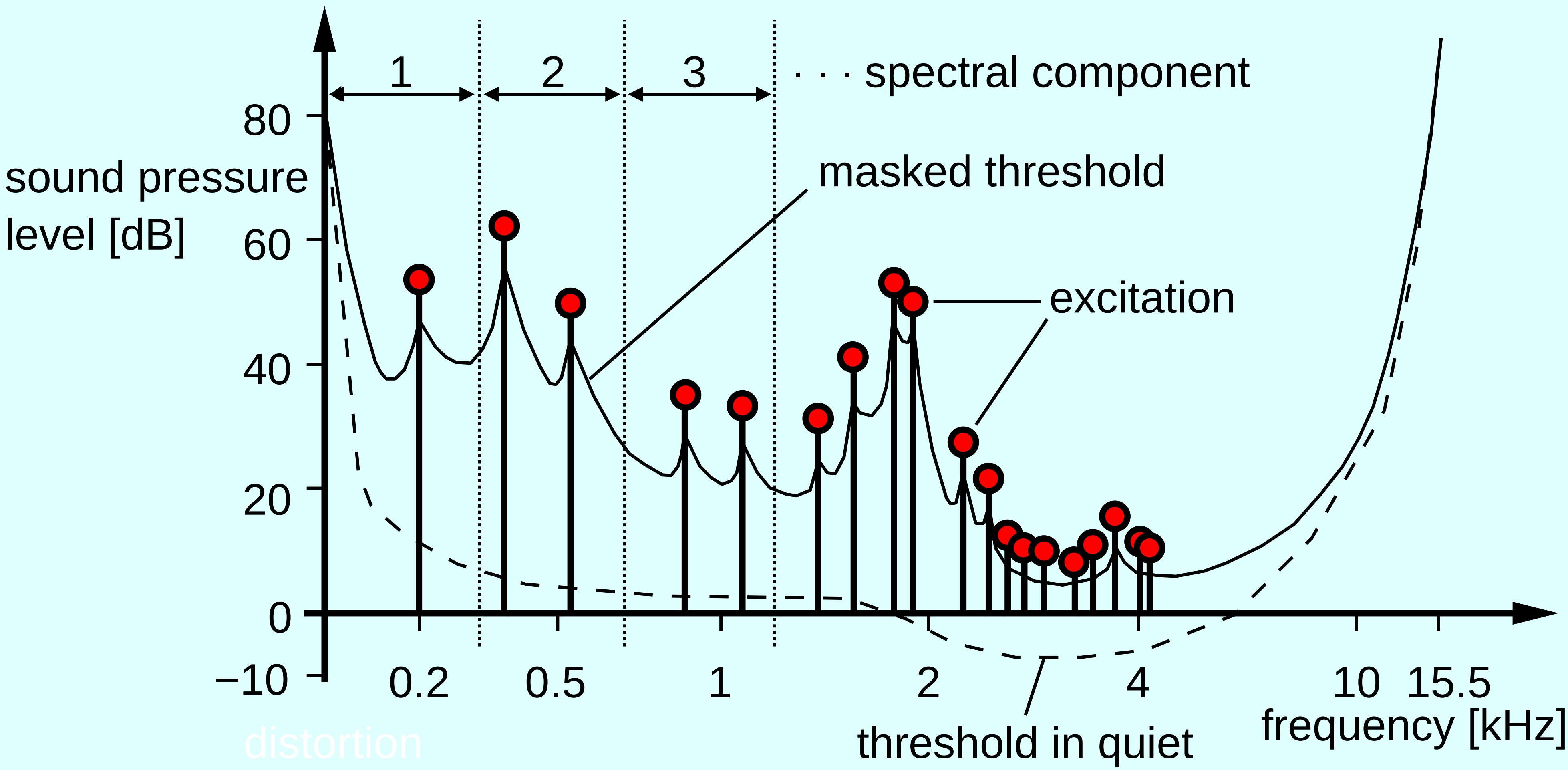
Basic structure of a perceptual audio encoder / decoder
(T/F coder: time/frequency decomposition)

Perception: Simultaneous Masking



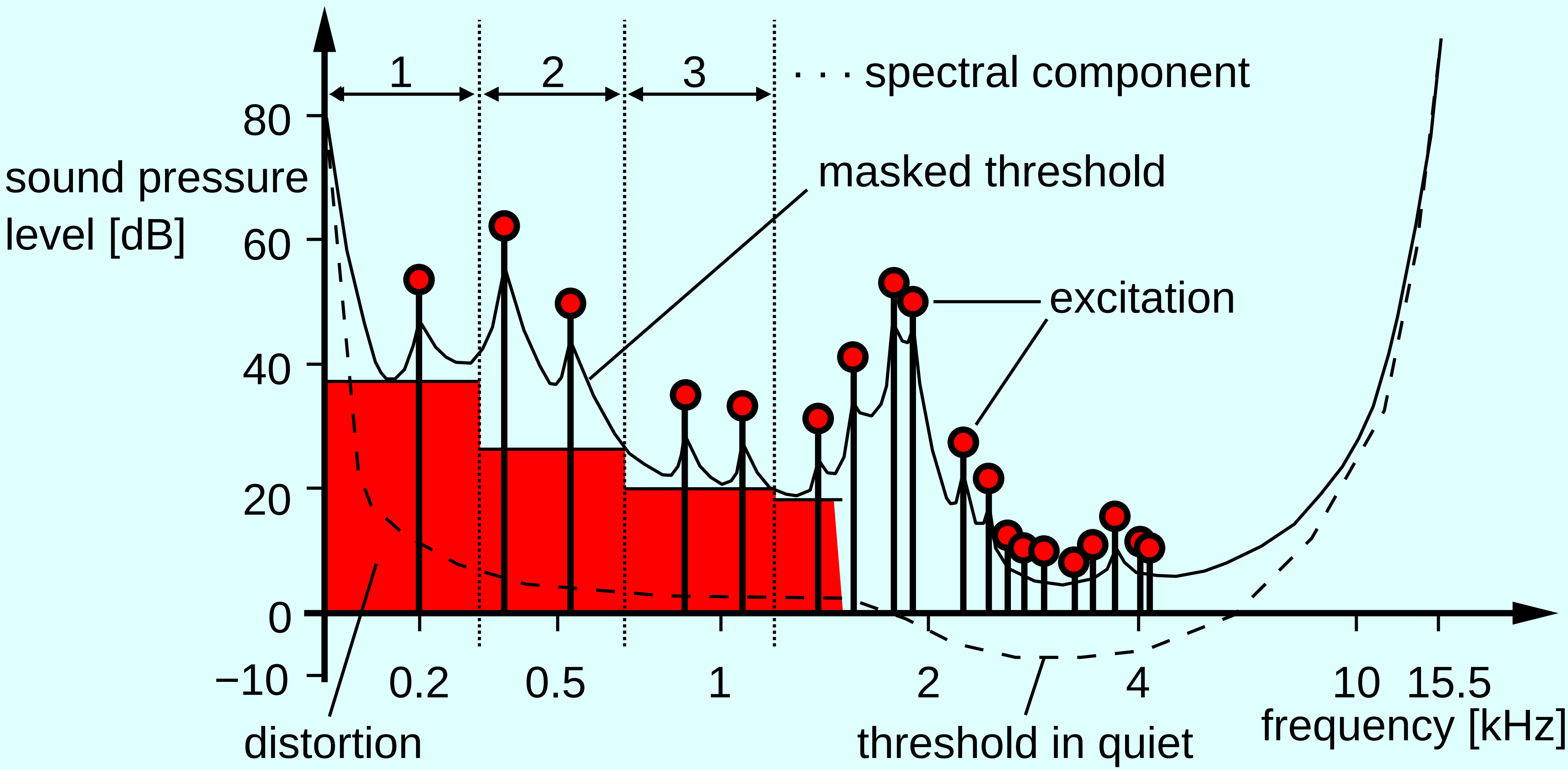
threshold in quiet and masked threshold

Perception: Simultaneous Masking



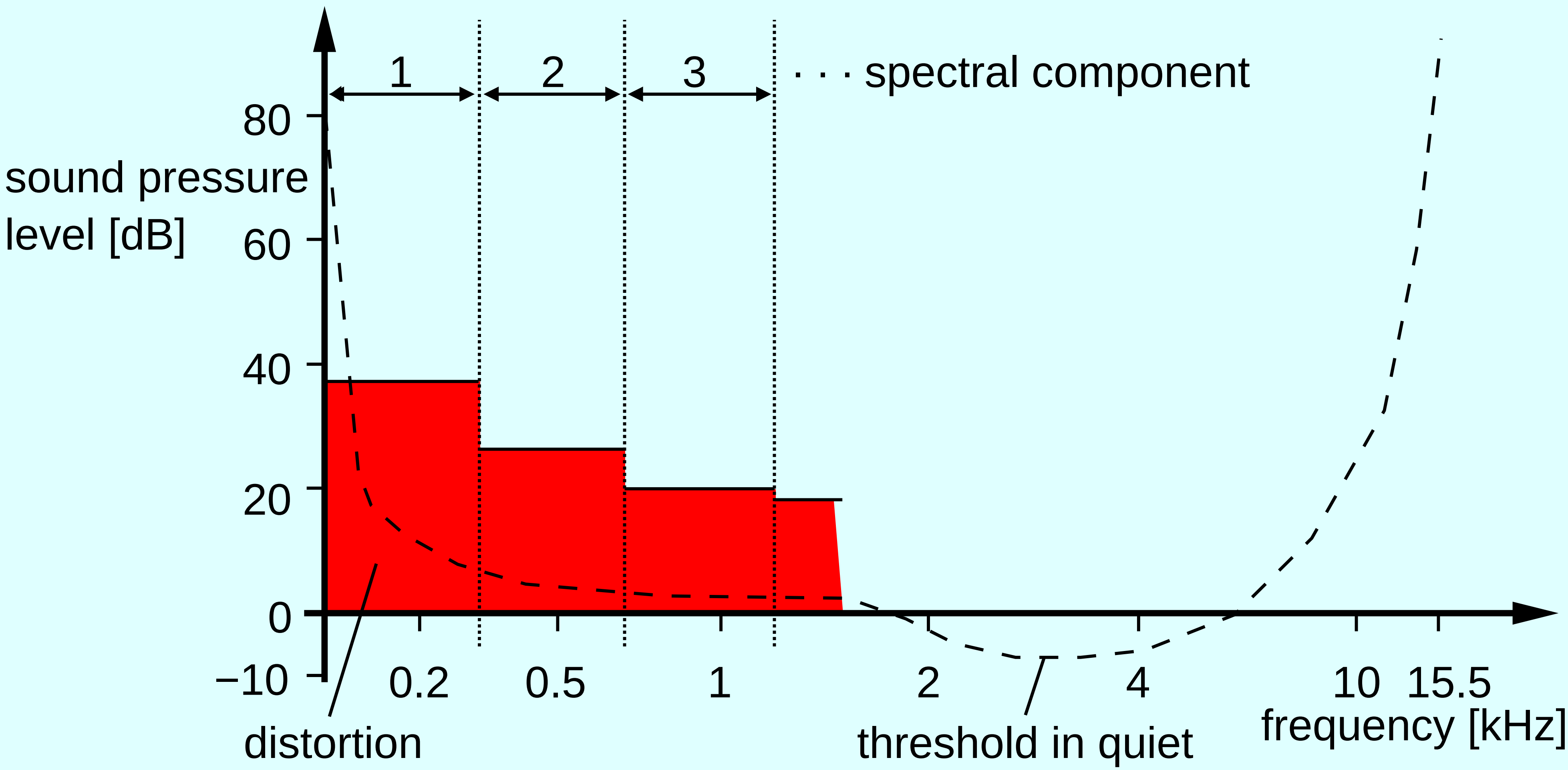
original signal

Perception: Simultaneous Masking



original signal + quantisation noise

Perception: Simultaneous Masking



quantisation noise only

What is “Parameteric Audio Coding” ?

. . . so what is “Parameteric Audio Coding” about ?

- **Problem:**
  MPEG-4 range of applications
  $\Rightarrow$ requires speech <u>and</u> audio coders
- **Idea:**
  generalised approach to audio coding

What is "Parameteric Audio Coding" ?

Representation of audio signal x

- *physical*: waveform $x(t)$
- *abstract*: musical score (compact, ambiguous)
  $\Rightarrow$ promising approach for efficient audio coding

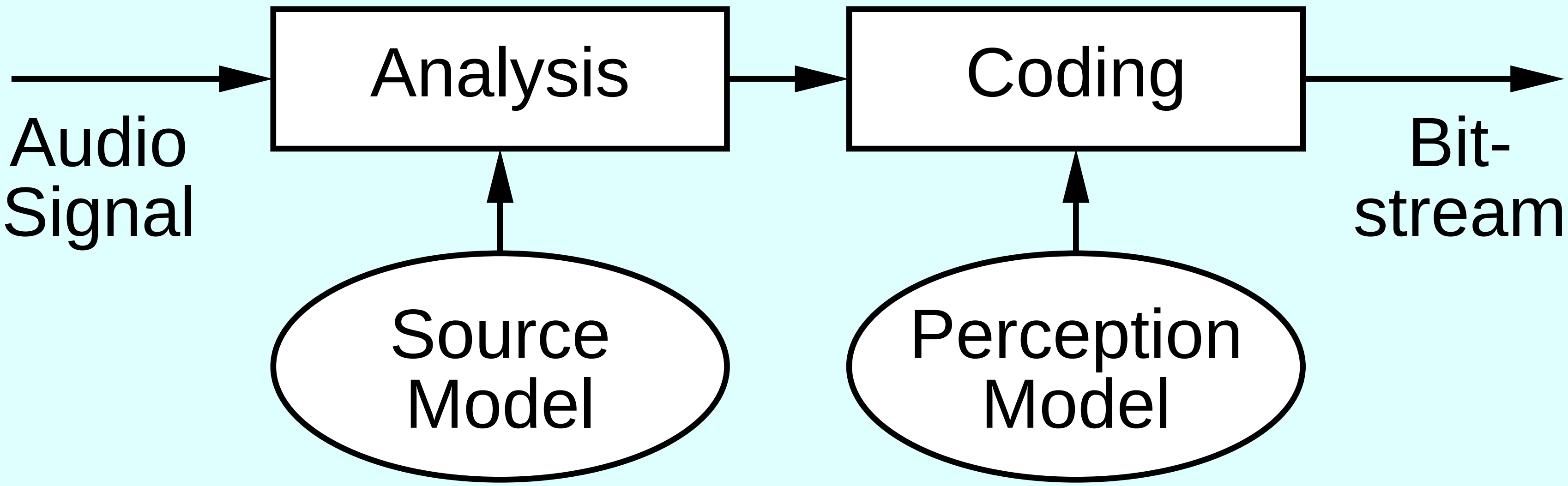
Audio coding with abstract signal representation

- Encoding: *physical* $\Rightarrow$ *abstract representation*
  but: automatic transcription to score very difficult !!!
- $\Rightarrow$ *Compact representation of audio*
  automatically derived from real-world signal

What is “Parameteric Audio Coding” ?

Signal representation: source and perception model

$\Rightarrow$ **Parametric Audio Coding**



- source model $\Rightarrow$ redundancy reduction
- perception model $\Rightarrow$ irrelevancy reduction

Source Models for Audio Signals

Spectral Decomposition

- stationary signal within a frame (duration T)
  $\Rightarrow$ time-to-frequency (T/F) transform (e.g. MDCT)
- signal-adaptive time/frequency resolution

Physical Modelling: Excitation + Resonances

- speech: periodic/random excitation + LPC filter
- music synthesis: e.g. waveguide

Source Models for Audio Signals

Sinusoidal Modelling

$$\hat{x}(t) = \sum_{i=1}^{N} a_i(t) \cdot \sin(\varphi_i + 2\pi \int_0^t f_i(\tau)\, d\tau)$$

- Applications:
  - music instrument analysis/synthesis
  - speech & audio coding
- modelling of "spectral peaks"
- tracking of trajectories / phase continuity
- phase φ_i often perceptually irrelevant

Source Models for Audio Signals

Transient Modelling

- sinusoids with amplitude envelope (attack & decay)
- sinusoidal modelling of DCT spectrum
- T/F coding / wavelets / “matching pursuit”

Noise Modelling

- subband noise models (Bark, ERB)
- MA model: DCT of noise spectrum
- AR model: white noise + LPC filter
- “Bark-warped” LPC

Source Models for Audio Signals

Extended Sinusoidal Models

- set of sinusoids with common fundamental frequency
  $\Rightarrow$ harmonic tone
- "bandwidth enhanced sinusoids":
  sinusoid $\Rightarrow$ narrow-band noise (using AM/FM)

$\Rightarrow$ **Problem: Choice of source model ?**

Efficiency vs. Generality

specialised source model
not suitable for arbitrary signals

Parametric Audio Coding

Concept of Parametric Audio Coding

- combination of different *source models*
  $\Rightarrow$ decompose audio signal into components
- utilise *perception models*
  $\Rightarrow$ "optimal" decomposition (relevant components)

$\Rightarrow$ **Analysis/Synthesis Approach**

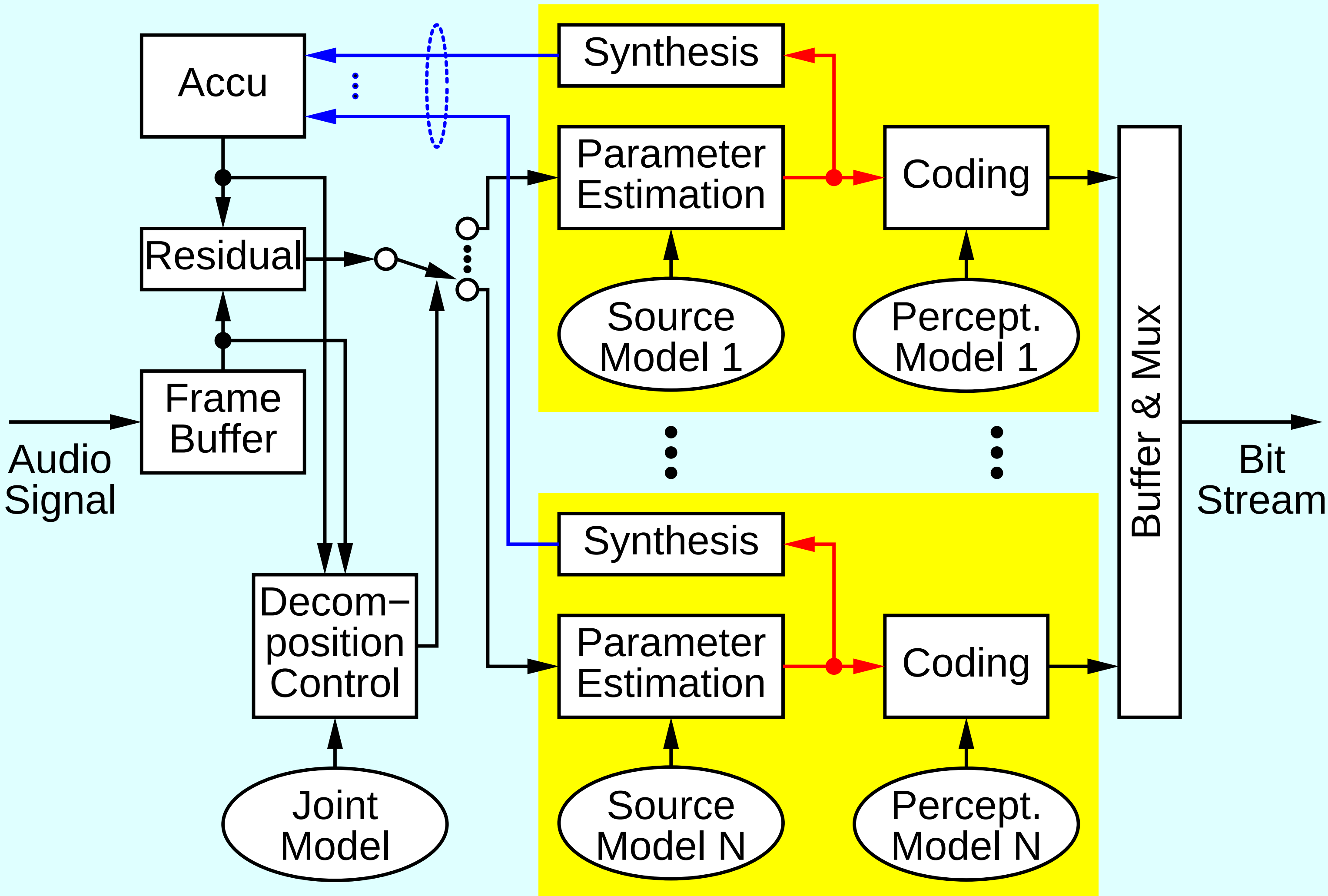
Parameter quantisation and coding

- quant. step size: "just noticeable differences"
- parameter prediction & entropy coding

Parametric Audio Coding: Encoder

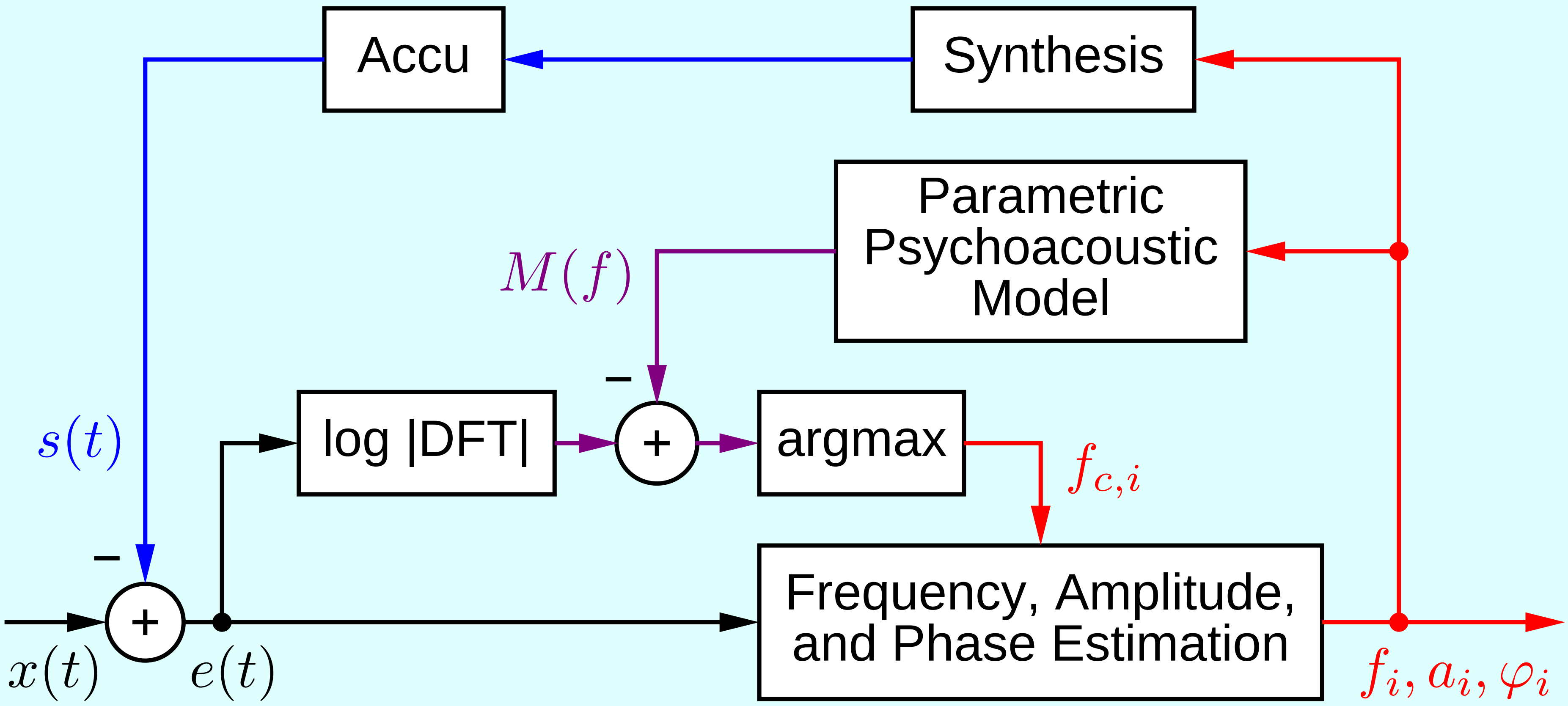
Extracted Signal Components

Model Parameters



Parametric Audio Coding: Encoder

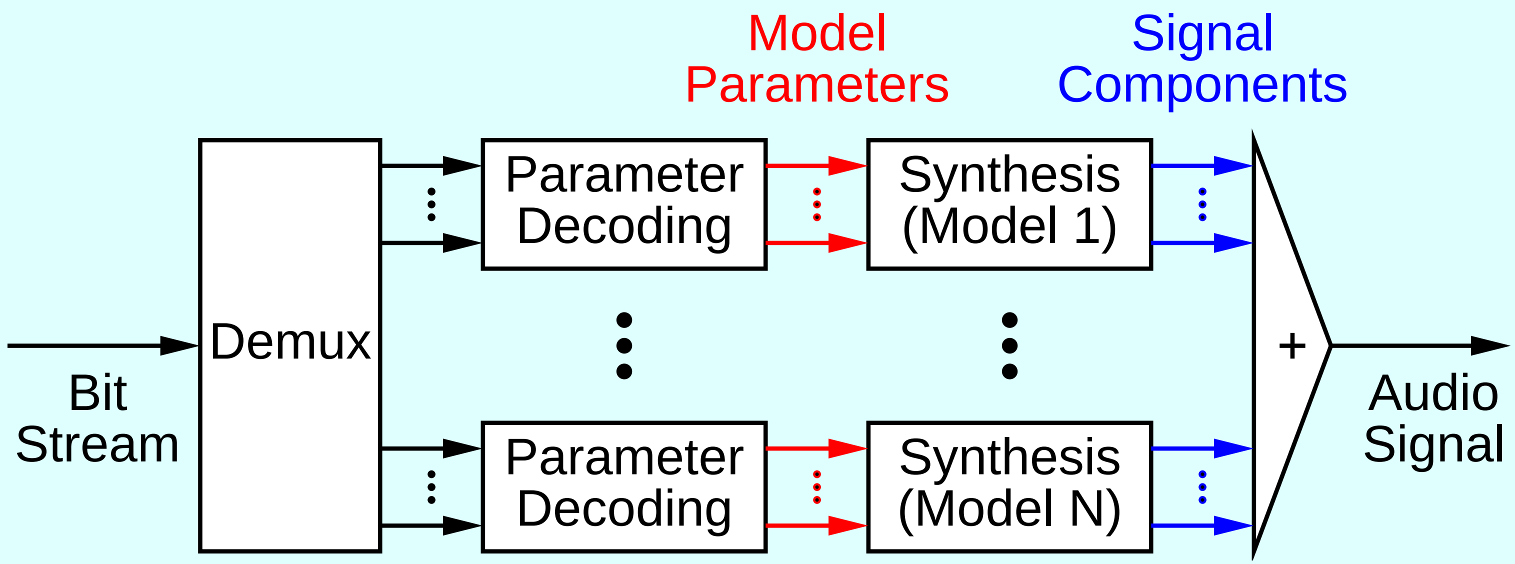
Example: Decomposition into sinusoidal components



Analysis/Synthesis Loop

Parametric Audio Coding: Decoder

Parameter Decoding & Signal Synthesis



Additional functionalities

- scalability: base + enhancement bitstream
- signal modification: time-scaling & pitch-shifting

Comparison: Speech and Audio Coders

Audio Coder (e.g. MPEG-1/2)

- perception model for encoder control
  $\Rightarrow$ efficient for arbitrary signals ($\geq$ 32 kbit/s/ch)

Speech Coder (e.g. CELP)

- specialised source model (vocal tract)
  $\Rightarrow$ efficient for speech signals (4 .. 24 kbit/s)

Parametric Coder (e.g. HVXC, HILN)

- recreate perceived sound
  $\Rightarrow$ no waveform approximation required

Parametric Audio Coding: HILN

Example: MPEG-4 Parametric Audio Coder HILN

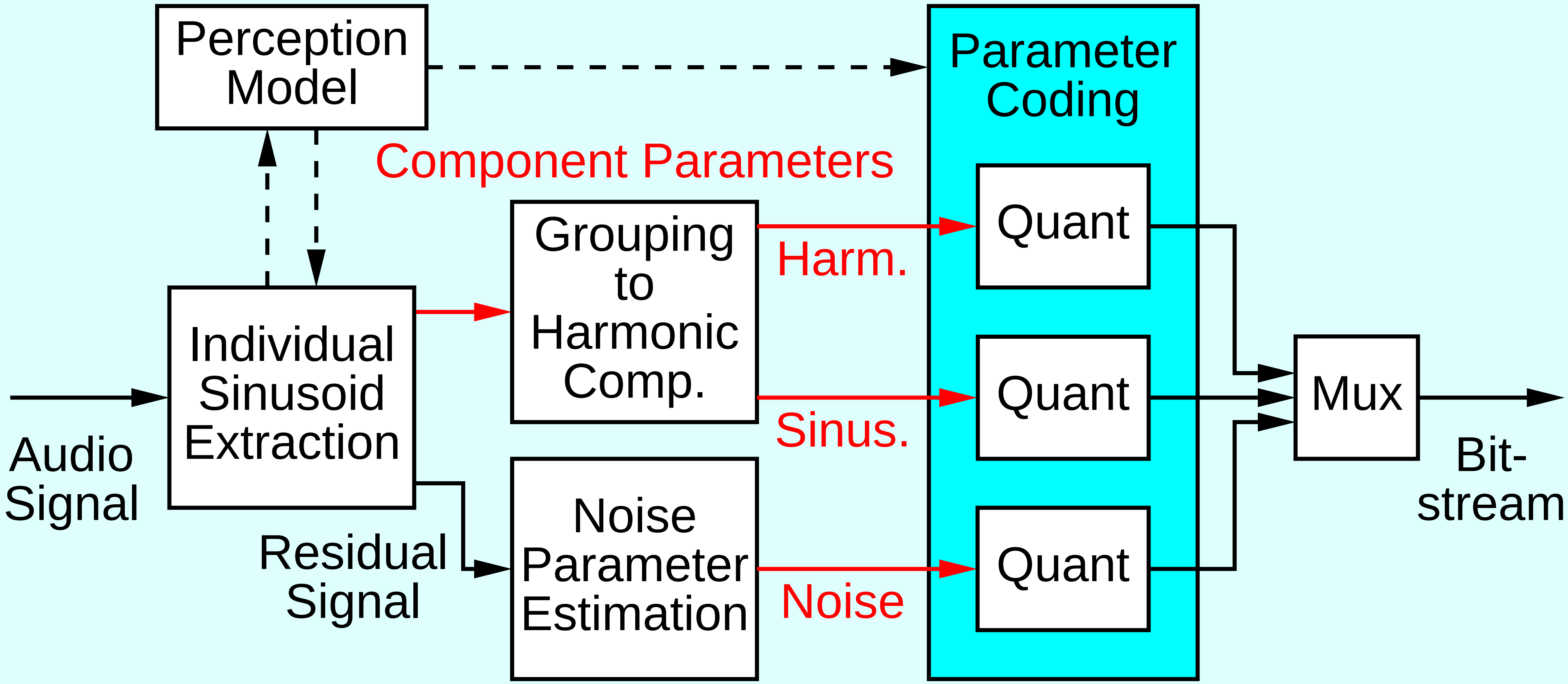
"Harmonic and Individual Lines plus Noise"

Models and parameters in HILN:

harmonic tone:	fundamental freq. & LPC spectrum
sinusoids:	frequency & amplitude [opt.: ampl. envelope, start phase]
noise:	LPC spectrum

- frame size 32 ms (typ.)
  $\Rightarrow$ 4 .. 16 kbit/s @ 8 kHz bandwidth (typ.)

HILN Encoder

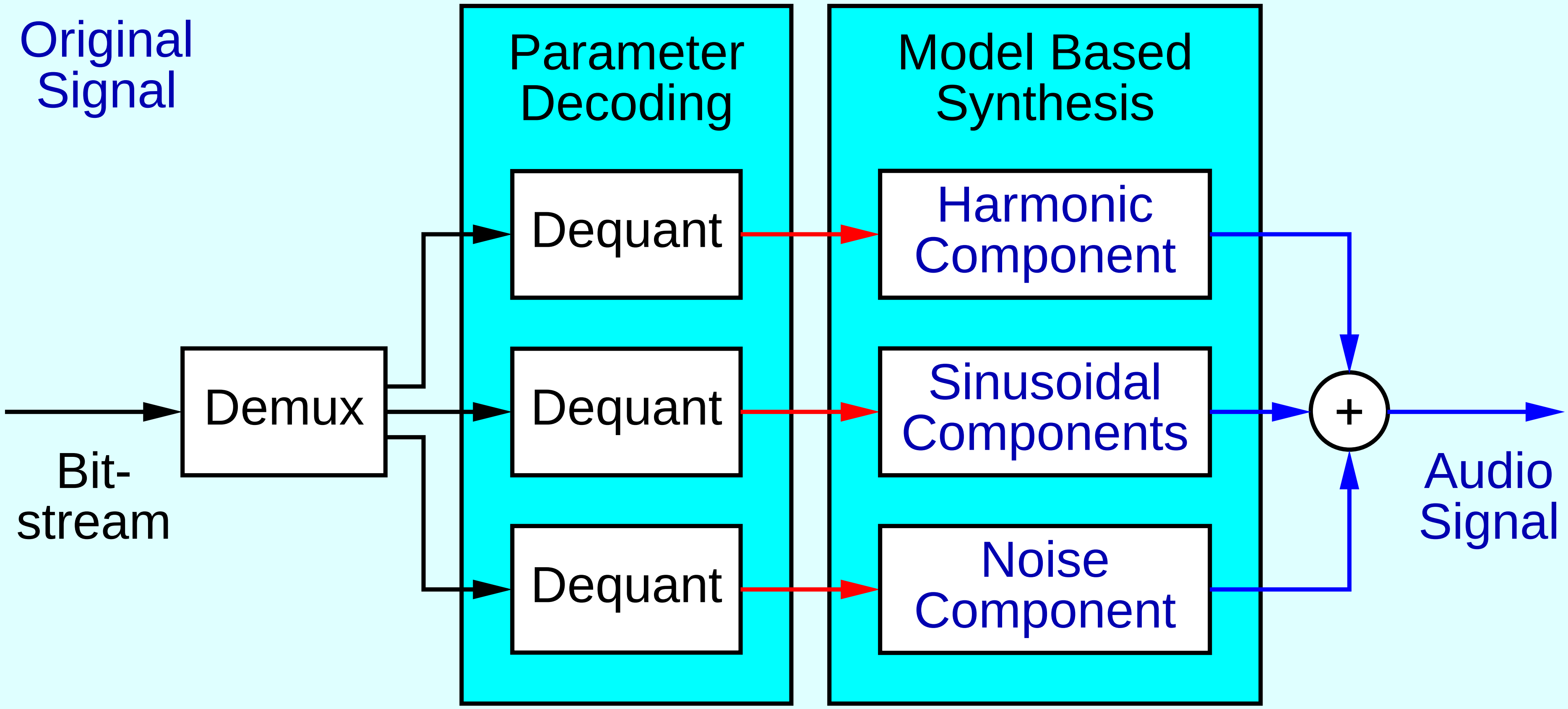


HILN Parametric Audio Encoder

(selection of relevant components by perception model)

HILN Decoder: Audio Demonstration

Example: MPEG-4 HILN @ 6 kbit/s, f_s = 16 kHz



HILN Parametric Audio Decoder

HILN Decoder: Audio Demonstration

Signal modification (HILN, f_s = 16 kHz)

- Interactive pitch and speed control (16 kbit/s)
- Pitch-shifting: +20% (6 kbit/s)
- Time-scaling: -17% (6 kbit/s)

Bitrate scalability (HILN, f_s = 16 kHz)

- base layer: 6 kbit/s
- base + enhancement layer: 6+10 kbit/s
- non-scalable bitstream: 16 kbit/s

HILN Decoder: Audio Demonstration

Comparison of coding techniques: (6 kbit/s)

- original (8 kHz bandwidth)
- speech coding (MPEG-4 CELP)
- T/F coding (MPEG-4 TwinVQ)
- parametric audio coding (MPEG-4 HILN)

HILN Parametric Audio Coding

MPEG-4 Verification Test

- MPEG-4 TwinVQ and HILN comparable at 6 kbit/s
- MPEG-4 AAC and HILN comparable at 16 kbit/s
- Additional functionalities of HILN:
  bitrate scalability, speed & pitch change

Why to “speed up HILN encoding?”

- Reference encoder:
  - only optimised for audio quality
  - very high computational complexity (not real-time)

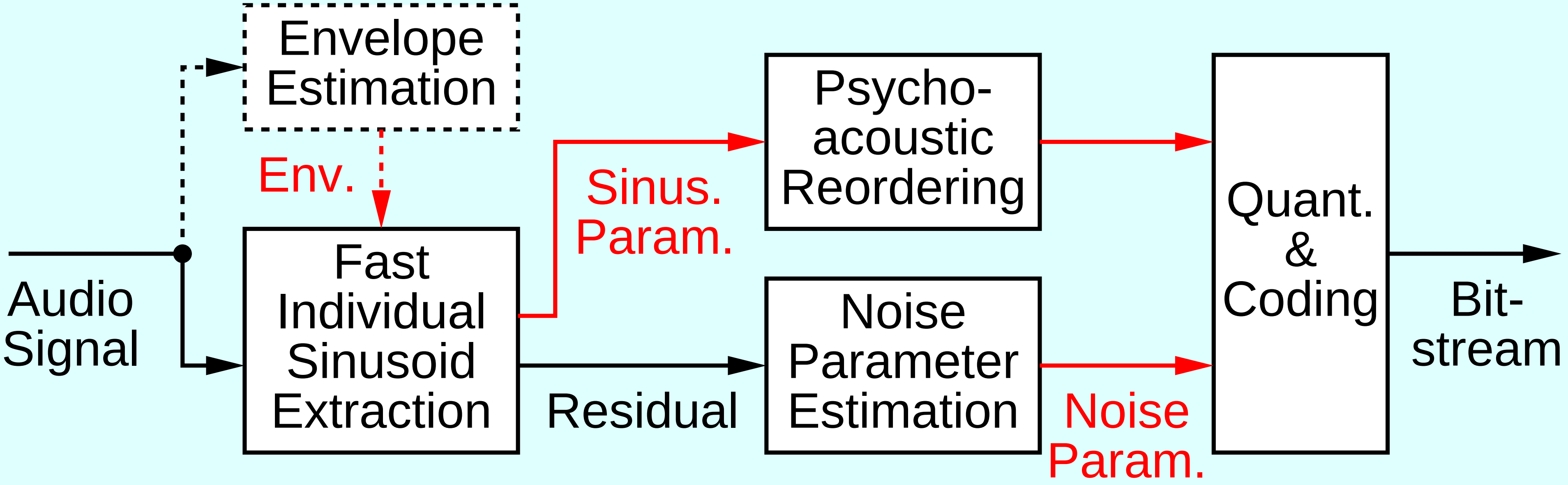
HILN Encoder Optimisation

Goal: Real-time HILN encoding on a normal PC

Possible approaches to reduce complexity:

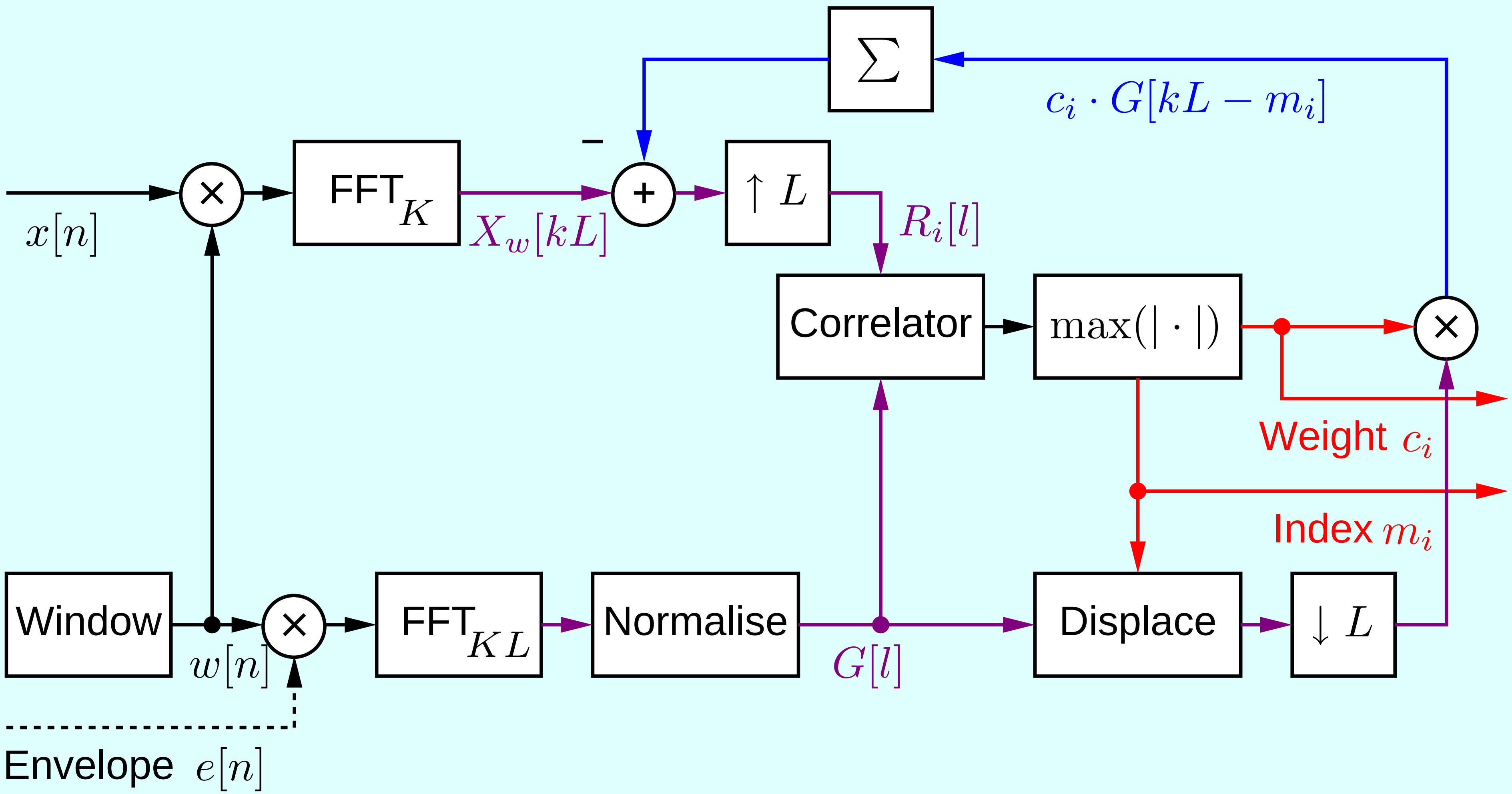
- Sinusoid extraction by matching pursuit [Goodwin 1997, Verma 1999]
- Fast sinusoid extraction in the frequency domain
  - correlation of spectrum with “prototype”
  - stepwise refined frequency search
  - energy metric $\Rightarrow$ needs psychoacoustic reordering
- Fast HILN encoder implemented in frequency domain (optional amplitude envelope supported)

HILN Encoder Optimisation



Block diagram of fast HILN encoder

HILN Encoder Optimisation



Fast sinusoid extraction in the frequency domain

Results: Computational Complexity

Encoder	Bitrate [kbit/s]	CPU Load [MHz]	Rel. Speed
Reference Encoder	6	26000	1
Fast Encoder (Env.)	6	51	510
Fast Encoder	6	24	1080
Reference Encoder	16	31000	1
Fast Encoder (Env.)	16	100	310
Fast Encoder	16	46	680

Encoding speed on Intel Pentium III (500 MHz)

ANSI-C implementation, f_s = 16 kHz

Results: Subjective Quality

Subjective quality of fast encoder

- Informal comparison (6 & 16 kbit/s, 39 items)
  - fast encoder with ampl. envelope
    $\Rightarrow$ 10 .. 20% of items (slightly) worse than ref. enc.
  - fast encoder (no ampl. envelope)
    $\Rightarrow$ percussive items (clearly) worse than fast enc.
- Current limitations of fast encoder
  - simplified or no envelope estimation
  - no sweep estimation
  - no harmonic component grouping

Results: Audio Demonstration

Audio demonstration: Comparison of encoders

Original	f_s = 16 kHz	f_s = 16 kHz
Reference Encoder	6 kbit/s	16 kbit/s
Fast Encoder (Env.)	6 kbit/s	16 kbit/s
Fast Encoder	6 kbit/s	16 kbit/s

Audio demonstration: Real-time encoding+decoding

- real-time encoding (16 kbit/s, f_s = 16 kHz)
- real-time decoding with interactive pitch change

Potential Parametric Coding Artifacts

Potential artifacts related to source models:

- limitations of source models
- bad decomposition (hard decisions are problematic)
- bad parameter estimation

Potential artifacts related to perception models:

- quantisation (consider “just noticeable differences”)
- selection of most relevant components
- is phase information irrelevant?
  (transients, clipping in sinusoidal synthesiser)

Examples of Artifacts

- Parametric coding: no waveform approximation
  $\Rightarrow$ difference signal meaningless
  - original: pop music
  - coded by parametric audio coder
  - difference signal (original-coded)
- Limitations of source models:
  model noise with sinusoids (e.g. applause)
  - original: white noise
  - coded using 0 to 120 sinusoids

Examples of Artifacts

- Limitations of source models:
  no model for transient (percussive) components
  - original: castanets
  - coded using sinusoids + noise
  - same, but with amplitude envelopes enabled

Examples of Artifacts

- Limitations of source models:
  specialised speech model not suitable for music
  - original: speech
  - coded by parametric speech coder
  - original: pop music
  - coded by parametric speech coder

Examples of Artifacts

- Bad signal decomposition:
  many sinusoids forced on harmonic grid
  - original: orchestral music
  - coded (harmonic component too strong)
- Bad signal decomposition:
  many tonal components modelled as noise
  - original: pop music
  - coded (noise component too strong)

Outlook

Improved source models

- “Gap” between tonal signals and noise
- Better transient models
- Combination with speech coder and T/F coder

Encoder optimisation (signal decomposition)

- Improved segmentation (e.g. tonal vs. noise)
- Grouping in time and frequency (trajectory, harmonic)
- Automatic segmentation of speech/music

. . . audio objects are transparent   ; -)

further reading ...

- Parametrische Audio Coding – Bibliographie

  `http://www.tnt.uni-hannover.de/˜purnhage/`

- MPEG Audio Web Page
  (tutorials, test reports, etc.)

  `http://www.tnt.uni-hannover.de/project/mpeg/audio/`

- *Official* MPEG Home Page

  `http://www.cselt.it/mpeg/`