
MPEG-4 General Audio Coding

Jürgen Herre
Fraunhofer Institute for Integrated Circuits (IIS)



Outline

MPEG-4 General Audio Coding:
The “all-round coder” in MPEG-4 audio

- MPEG-2 Advanced Audio Coding (AAC)
- MPEG-4 Extensions:
 - Perceptual Noise Substitution (PNS)
 - Long Term Prediction
 - TwinVQ Coding Core
- Conclusions



MPEG-2 Advanced Audio Coding (AAC)

History:

- 1994: Official start of AAC development
Goal: Development of a new powerful state-of-the art multi-channel coder without compatibility constraints
- 1997: AAC International standard (IS)
- 1999: AAC part of the MPEG-4 Standard

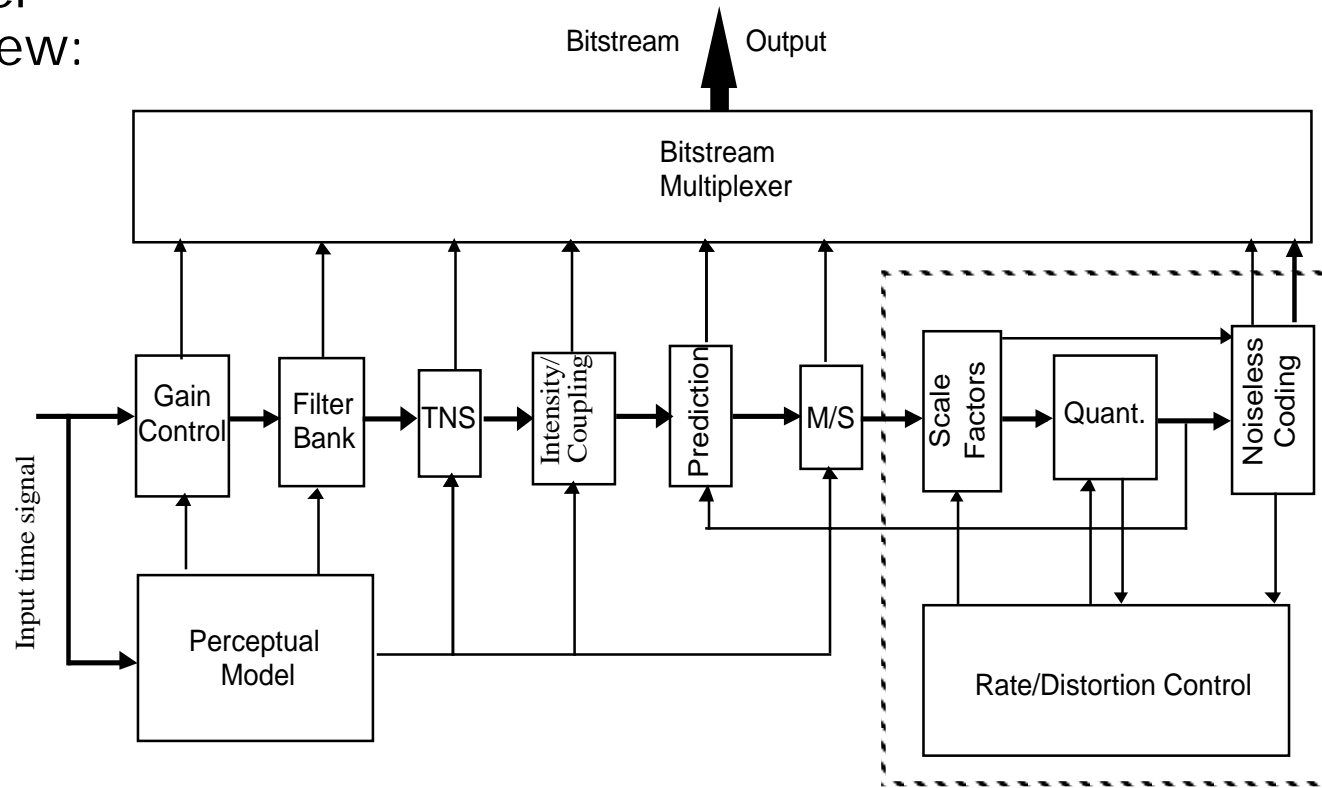
Today:

- Favourite coder for many application areas, like Internet audio, solid state players, ISDN music transmission, High-definition TV (HDTV), satellite & terrestrial digital audio broadcasting, ...



MPEG-2 AAC (2)

Encoder Overview:



MPEG-2 AAC (3)

Basic features:

- High frequency resolution filterbank-based coder (1024 lines MDCT with 50% overlap)
- 1:8 block switching (8 * 128 lines MDCT)
- Non-uniform quantizer
- Noise shaping in half critical bands (scalefactor bands)
- Huffman coding of scalefactors and spectral coefficients



MPEG-2 AAC (4)

Advanced coding tools:

- Window shape adaptation
- Temporal noise shaping (TNS)
- Gain control (SRS profile only)
- Backward adaptive prediction

Joint stereo coding tools:

- Mid/Side stereo (MS) per scalefactor band
- Intensity stereo coding between channel pairs
- Coupling channel(s)

Other features:

- Flexible bitstream format for up to 48 channels
- Low Frequency Enhancement (LFE) channel(s)



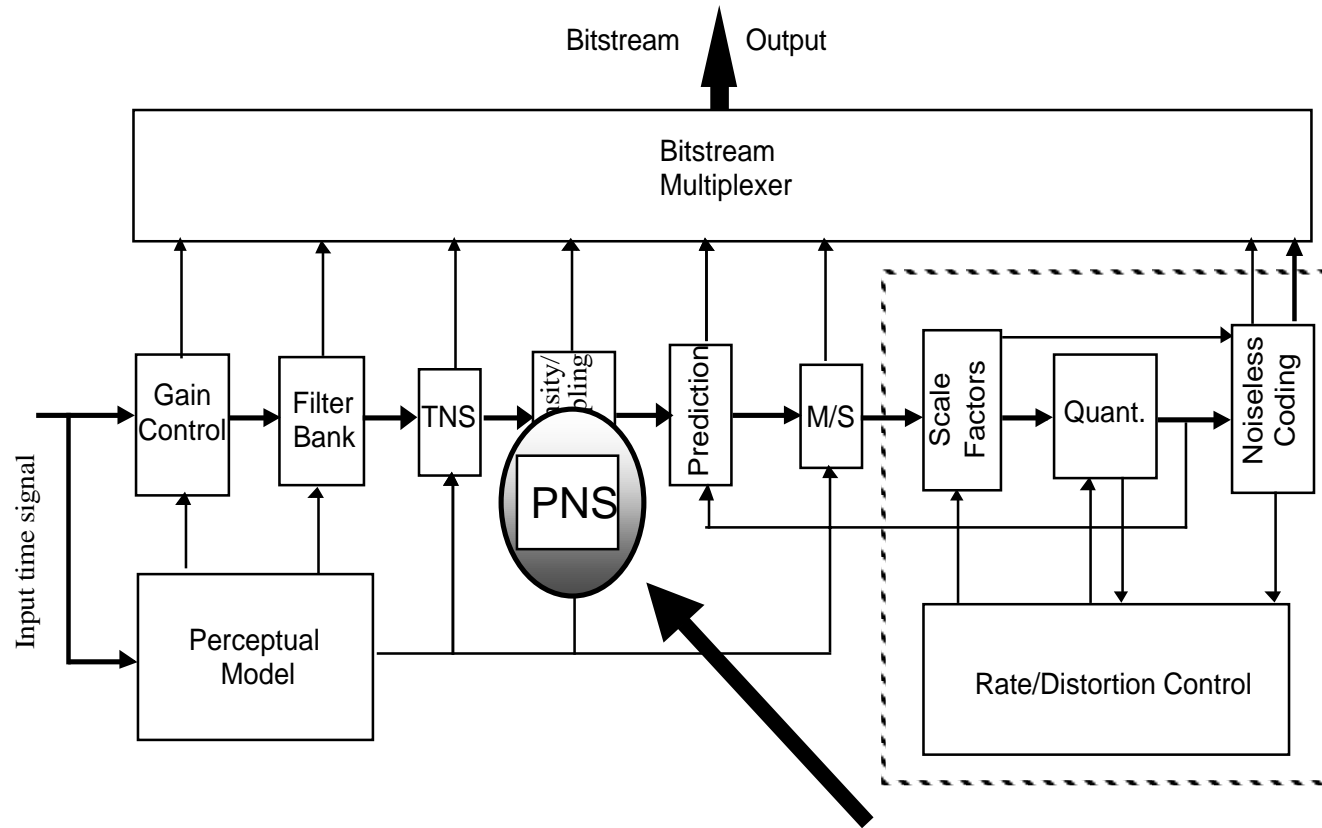
MPEG-2 AAC Performance

Test Results:

- Broadcast quality at 320 kbit/s for 5 channels (better than MPEG-2 Layer II at 640 kbit/s)
- Broadcast quality at 128 kbit/s stereo
- Comparison to other codecs:
AAC 96 kbit/sec stereo comparable to
 - AC-3 at 160 kbit/s
 - Layer II at 192 kbit/s
 - Layer III at 128 kbit/s
- Very low bitrates (comparison within MPEG):
AAC best audio coder at bitrates down to 16 kbit/s for mono and stereo



Extension: Perceptual Noise Substitution (PNS)



Perceptual Noise Substitution (1)

Background:

- Parametric coding of signals gives a very compact signal representation
- Parametric coding of noise-like signal components has been used widely e.g. in speech coding
- Can similar techniques be used in perceptual audio coding ?

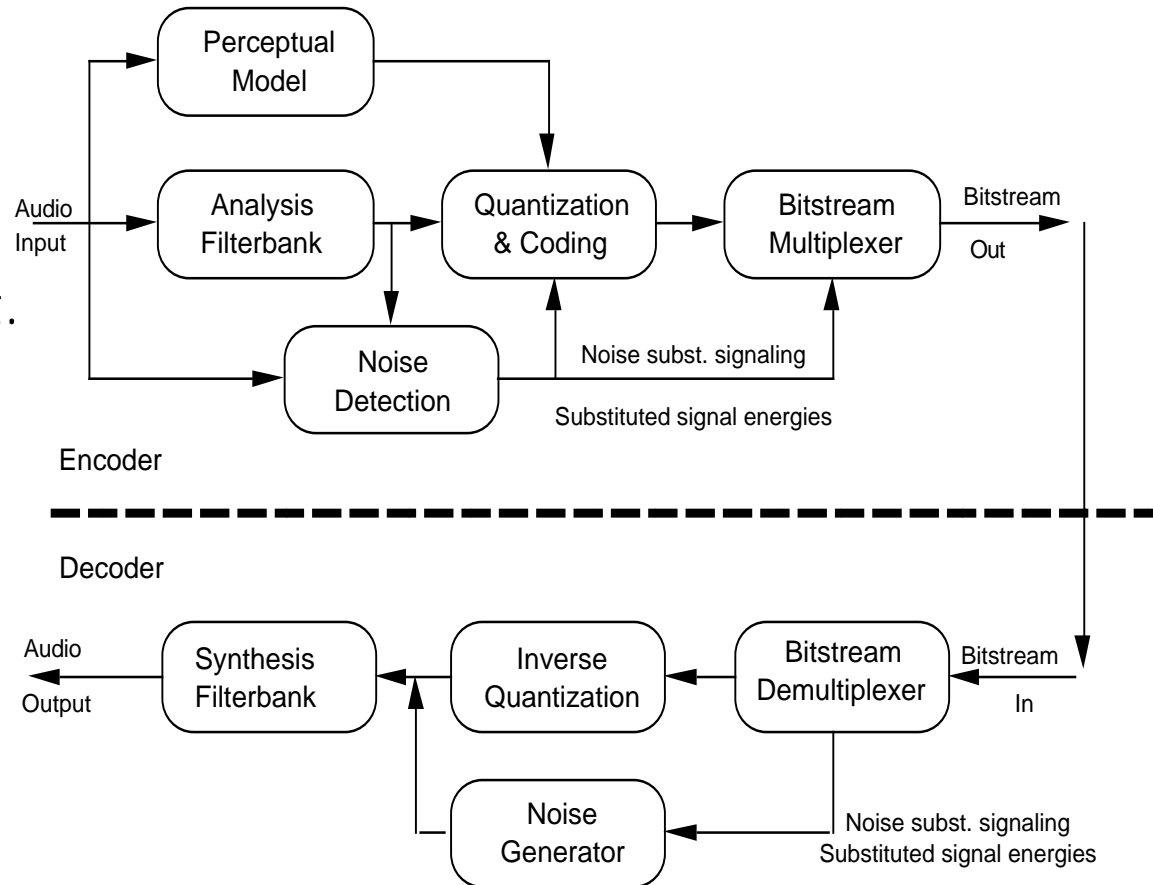
MPEG-4:

- Perceptual Noise Substitution (PNS) permits a frequency selective parametric coding of noise-like signal components



Perceptual Noise Substitution (2)

"Perceptual Noise Substitution" (PNS):
Perceptual coder +
parametric represent.
of noise-like signals



Perceptual Noise Substitution (3)

Principle:

- Noise-like signal components are detected on a scalefactor band basis
- Corresponding groups of spectral coefficients are excluded from quantization/coding
- Instead, only a "noise substitution flag" plus total power of the substituted band is transmitted in the bitstream
- Decoder inserts pseudo random vectors with desired target power as spectral coefficients

⇒ Highly compact representation for noise-like spectral components



Demonstration: Perceptual Noise Substitution

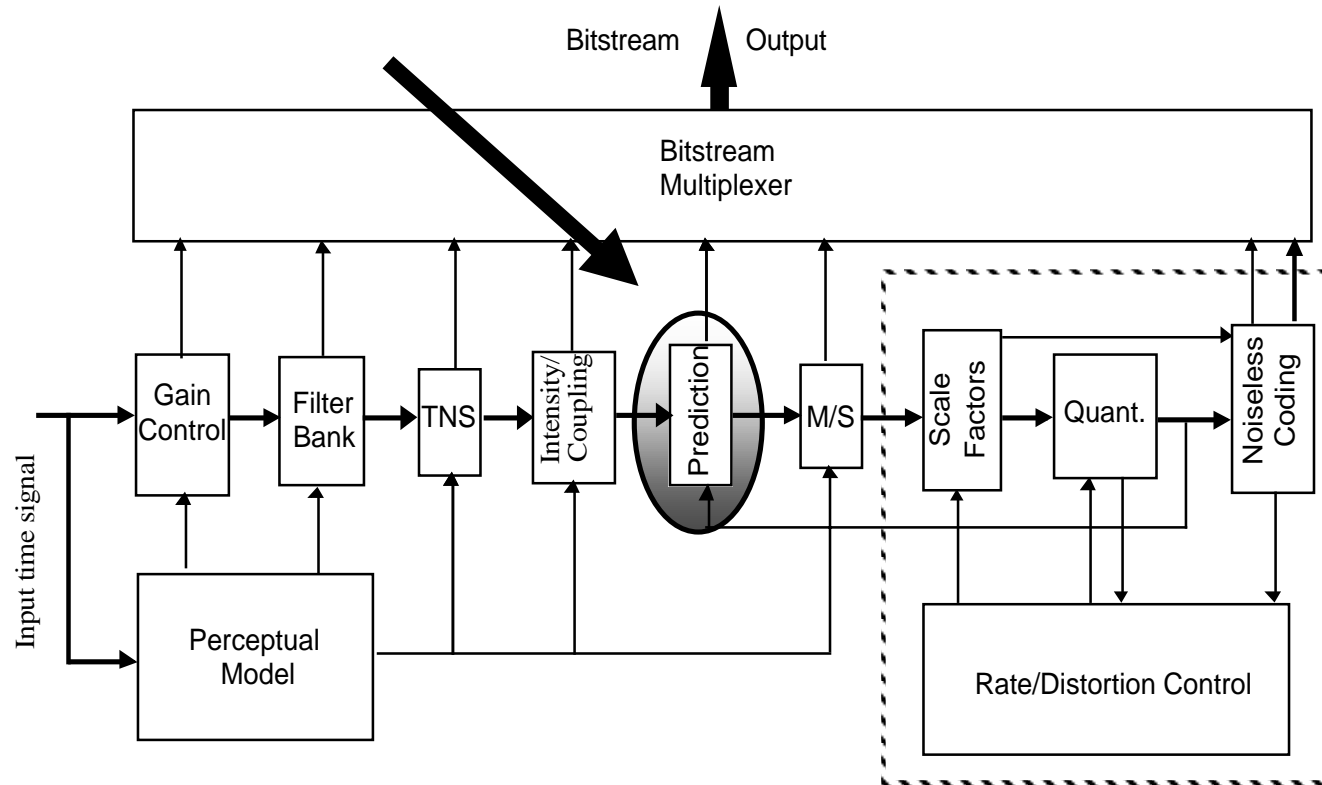
- Suzan Vega / Tom's Diner:
w/o PNS with PNS PNS parts only
- Orchestral Music :
w/o PNS with PNS PNS parts only
- Contemporary Pop Spot:
w/o PNS with PNS PNS parts only

AAC coder @ 32 kbit/s, fs=48 kHz, bandwidth 14 kHz

Note: This is not "normal" operation mode



Extension: Long Term Prediction



Long Term Prediction (1)

Background:

- Tone-like signals require much higher coding precision than noise-like signals (e.g. 20 dB vs. 6 dB)
⇒ High bit rate necessary to code signals with many tonal components (e.g. Harpsichord, Pitch Pipe)
- Tonal signal components are predictable
⇒ Further quality enhancement by predictive coding



Long Term Prediction (2)

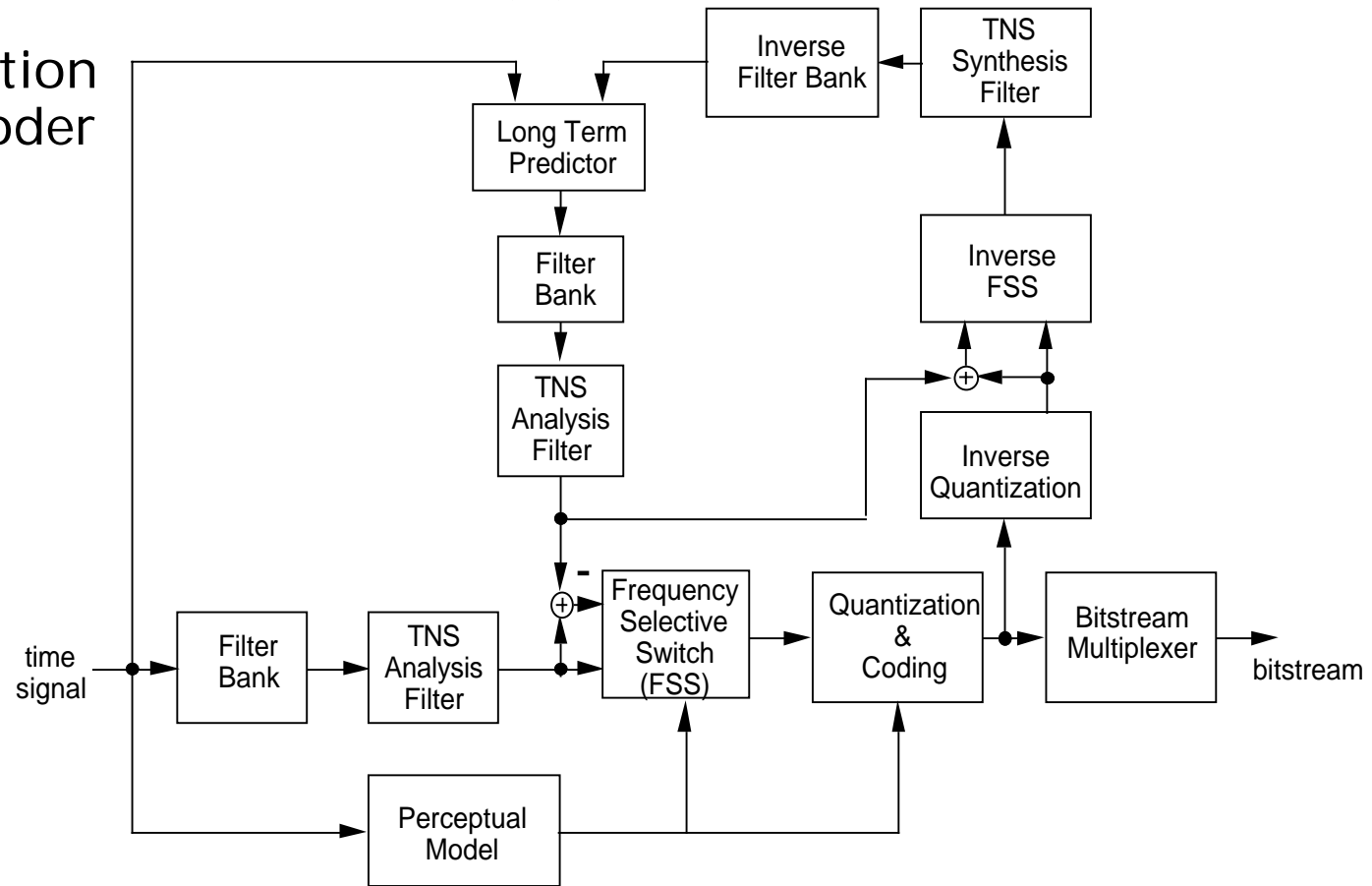
- MPEG-2 AAC:
- Prediction of each spectral coefficient; backward adaptive 2nd order Lattice predictor
 - High complexity (ca. 50% of decoder computation & RAM)
⇒ Prohibitive for cost sensitive applications, used in "MPEG-2 Main Profile" only

- MPEG-4 AAC:
- Long Term Predictor (LTP) as known from speech coding
 - New: Integration into perceptual audio coder
 - Lower complexity: Saving of approx. 50% in terms of computation and memory over MPEG-2 predictors
 - Comparable performance to MPEG-2 predictors



Long Term Prediction (3)

LTP Integration into AAC coder



Transform-Domain Weighted Interleave VQ (1)

- Background:
- Audio coding at extremely low bitrates (≥ 6 kbit/s)
 - CELP speech coders do not perform well for music

- MPEG-4:
- *Transform-Domain Weighted Interleave Vector Quantization (TwinVQ)* as additional coding kernel
 - Fully integrated into MPEG-4 AAC coding system:
 - Uses same spectral representation as AAC coder
 - Makes use of other MPEG-4 tools (e.g. LTP, TNS, joint stereo coding)
 - Possible core coder for \Rightarrow MPEG-4 scalable coding



Transform-Domain Weighted Interleave VQ (2)

Structure:

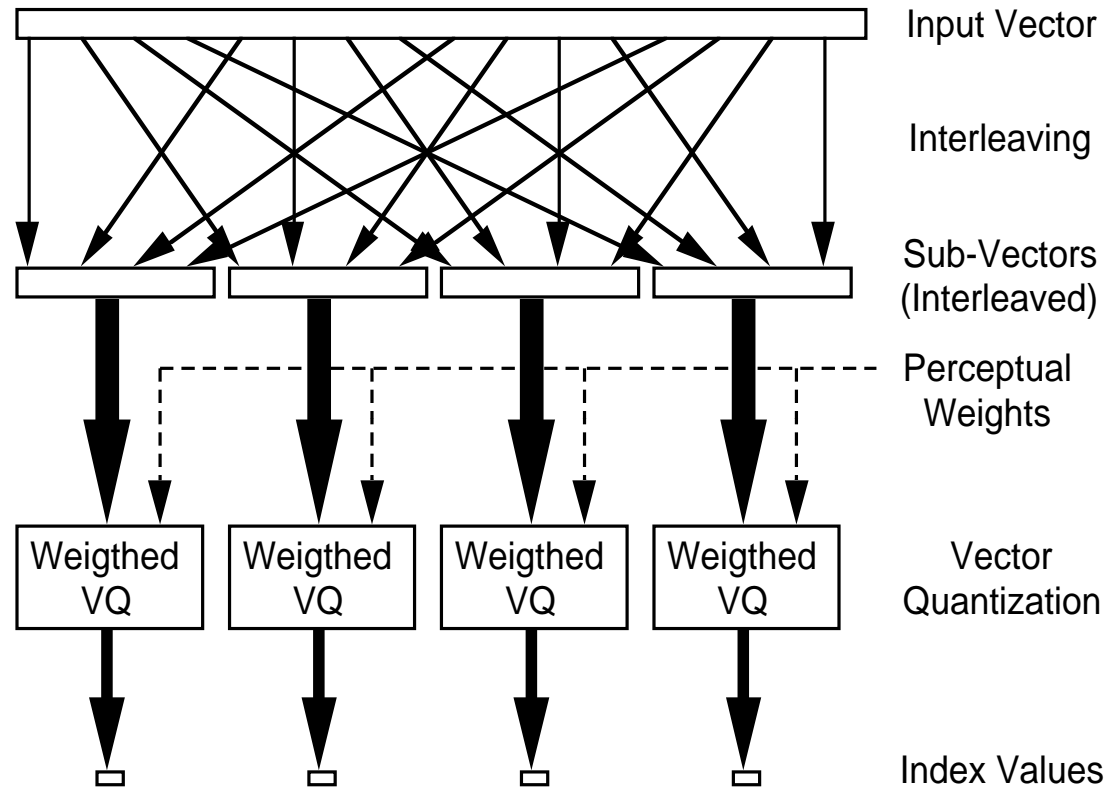
- Normalization of spectral coefficients:
 - LPC envelope (overall spectral shape)
 - Periodic component coding (harmonic components)
 - Bark-scale envelope coding (additional flattening)
- Vector Quantization (VQ) process:
 - Interleaving of spectral coefficients into new sub-vectors
 - Vector quantization
(two sets of codebooks, weighted distortion measure allows distortion control by perceptual model)

⇒ no bit/noise allocation or rate control iteration



Transform-Domain Weighted Interleave VQ (3)

Interleave VQ
Structure:



Conclusions

- MPEG-4 General Audio Coding:
The “all-round” coding system among the MPEG-4 audio schemes, providing a set powerful tools
- Based on MPEG-2 Advanced Audio Coding kernel
- Several enhancements for improved coding efficiency
- Perceptual Noise Substitution (PNS):
Exploiting noise-like components in the signal
- Long Term Prediction (LTP):
Taking advantage of very stationary / tonal signals
- TwinVQ coder kernel supports audio coding at extremely low data rates (\Rightarrow MPEG-4 scalable coding)

