Parametric Coding of High-Quality Audio

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Waveform vs Parametric

- Waveform
  - Filter-bank approach
  - Mainly exploits limitations of human auditory system
  - Mature technology
- Parametric
  - Source model approach
  - Exploits both source model as well as limitations of human auditory system

⟶most audio coders use a combination of both

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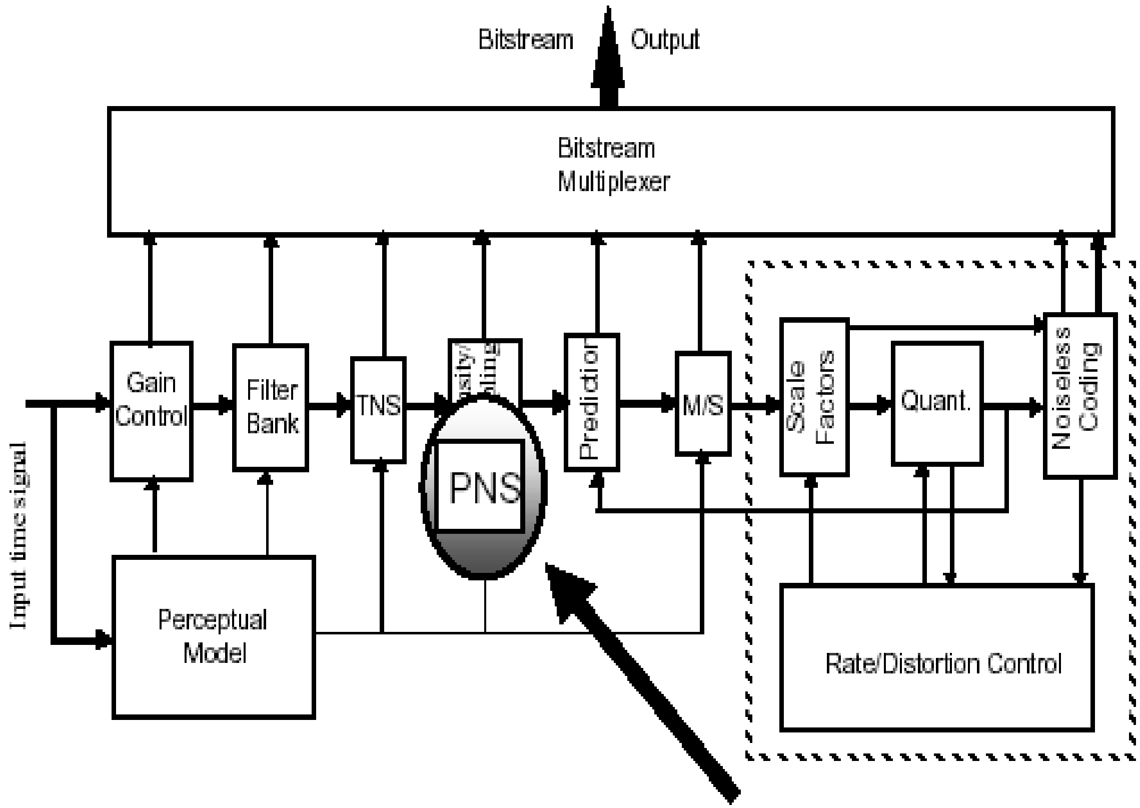
1. Perceptual Noise Substitution (PNS)



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

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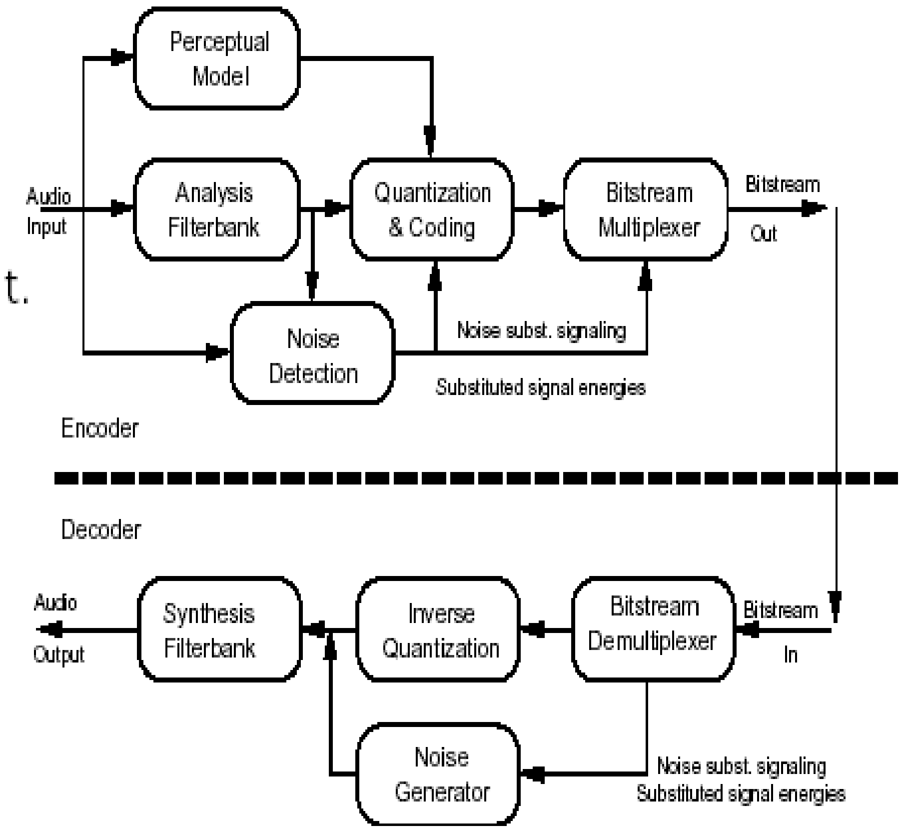
Perceptual Noise Substitution (2)

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



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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

2. Spectral Band Replication

Bandwidth Extension (1)

Background

- Audio coding at very low bitrates ⇒ artifacts
- To avoid excessive artifacts, bandwidth is usually sacrificed at low bitrates (<40kbit/s/ch)
  ⇒ Signal sounds unattractive (muffled)

Concept

- “re-generate” HF signal content at decoder end from LF part (and some helper information)

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Bandwidth Extension (2)

Idea of Spectral Band Replication (SBR)

- Re-generate HF-part of signal spectrum by means of *transposition* of transmitted spectrum
  $\Rightarrow$ ensures preservation of harmonic structure
- Subsequent shaping of signal towards original time/spectral envelope by an adaptive filter (*envelope adjuster*)
- Some more provisions for handling special situations

Compatibility

- SBR bitstream elements (ca. 2 kbit/s/ch) can be stored in AAC bitstream in a compatible way
  - Standard AAC decoders decode AAC part only

- “MPEG-4 Audio Extension #1”
- SBR is used e.g. in High-Efficiency AAC (aacPlus) and MP3Pro

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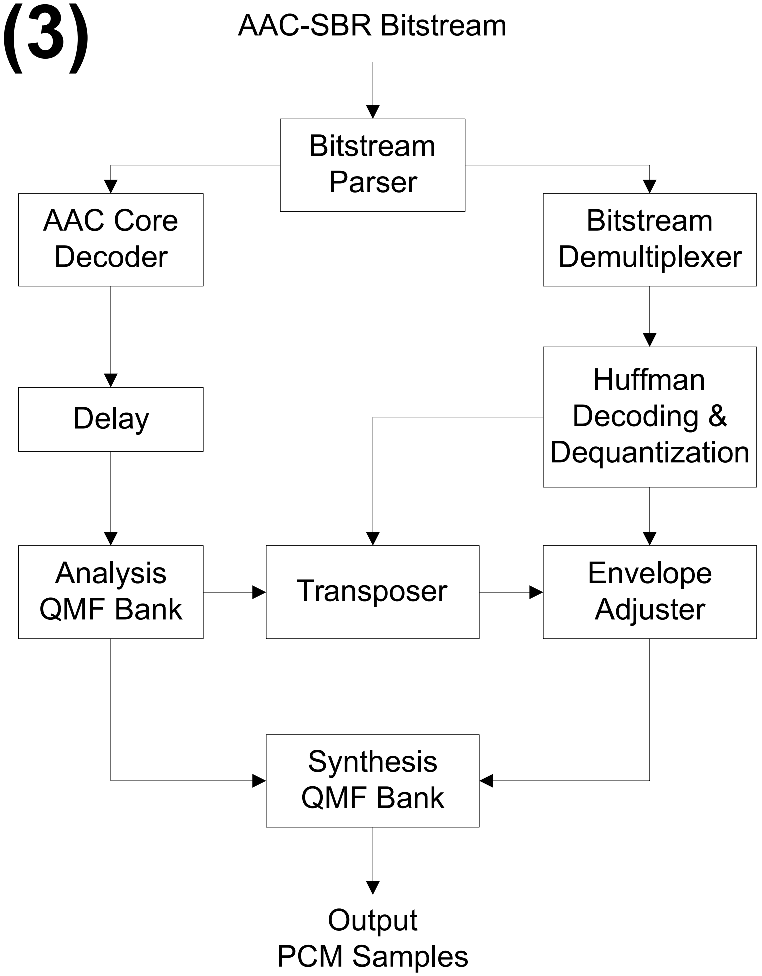
Bandwidth Extension (3)

Spectral Band Replication Scheme (Principle)



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Bandwidth Extension (4)

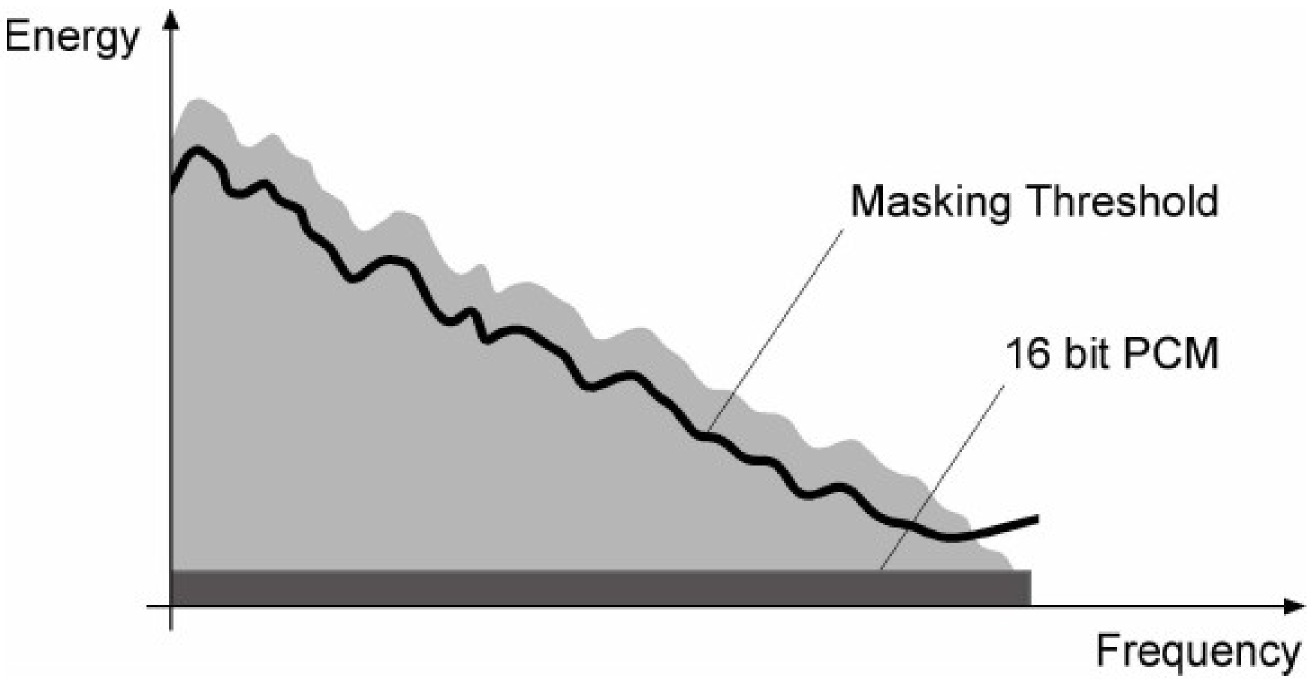

Figure 1: Spectrum and Masking Threshold

Dietz e.a, „Spectral Band Replication, a novel aproach in audio coding“

Bandwidth Extension (5)

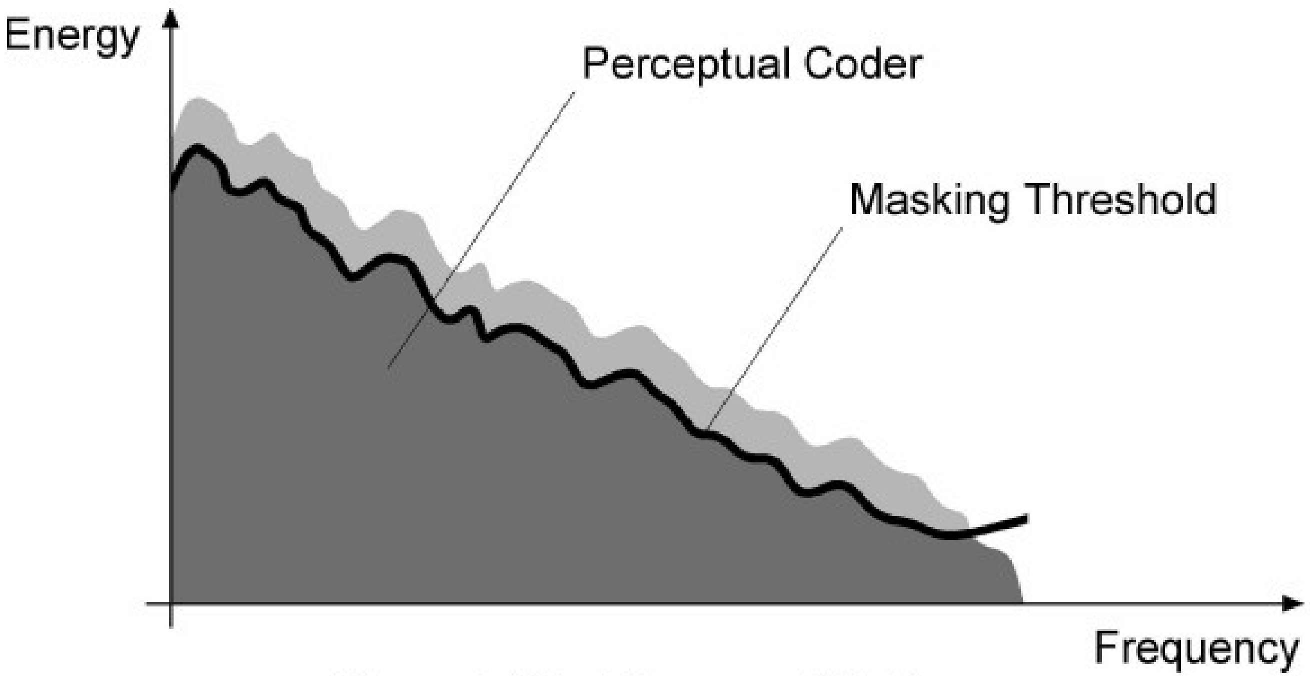

Figure 2: Ideal Perceptual Coding

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Bandwidth Extension (6)

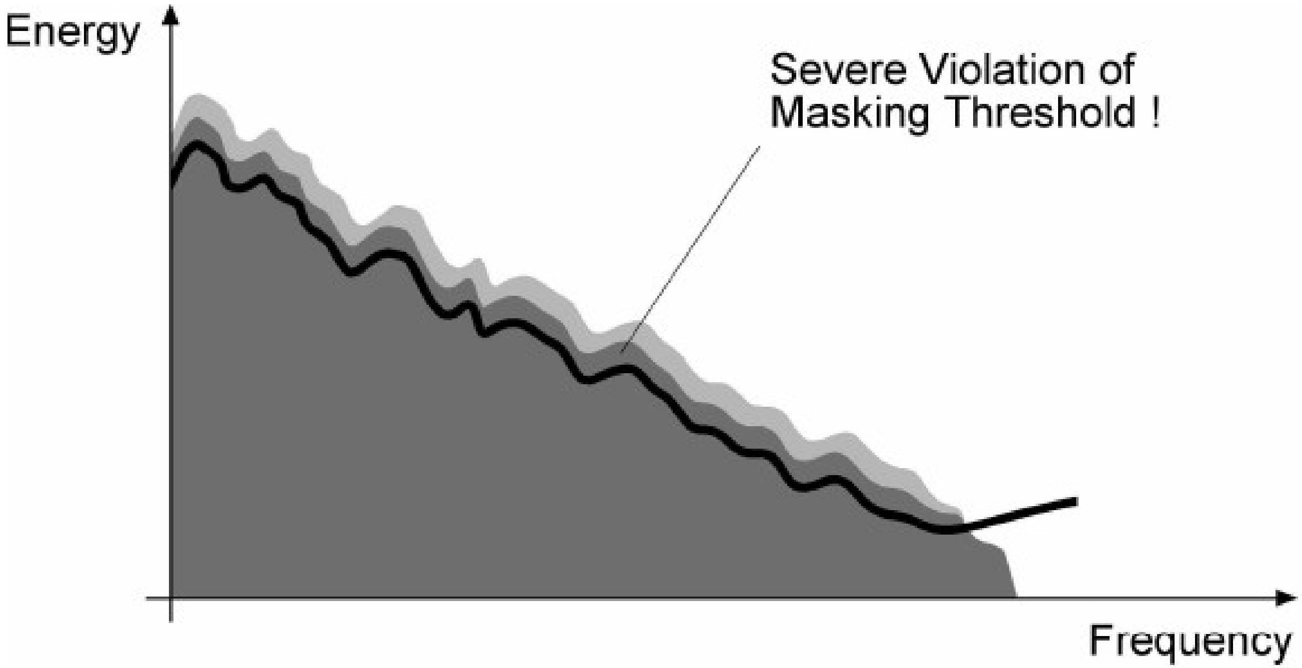

Figure 4: Waveform coding beyond its limits

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Bandwidth Extension (7)

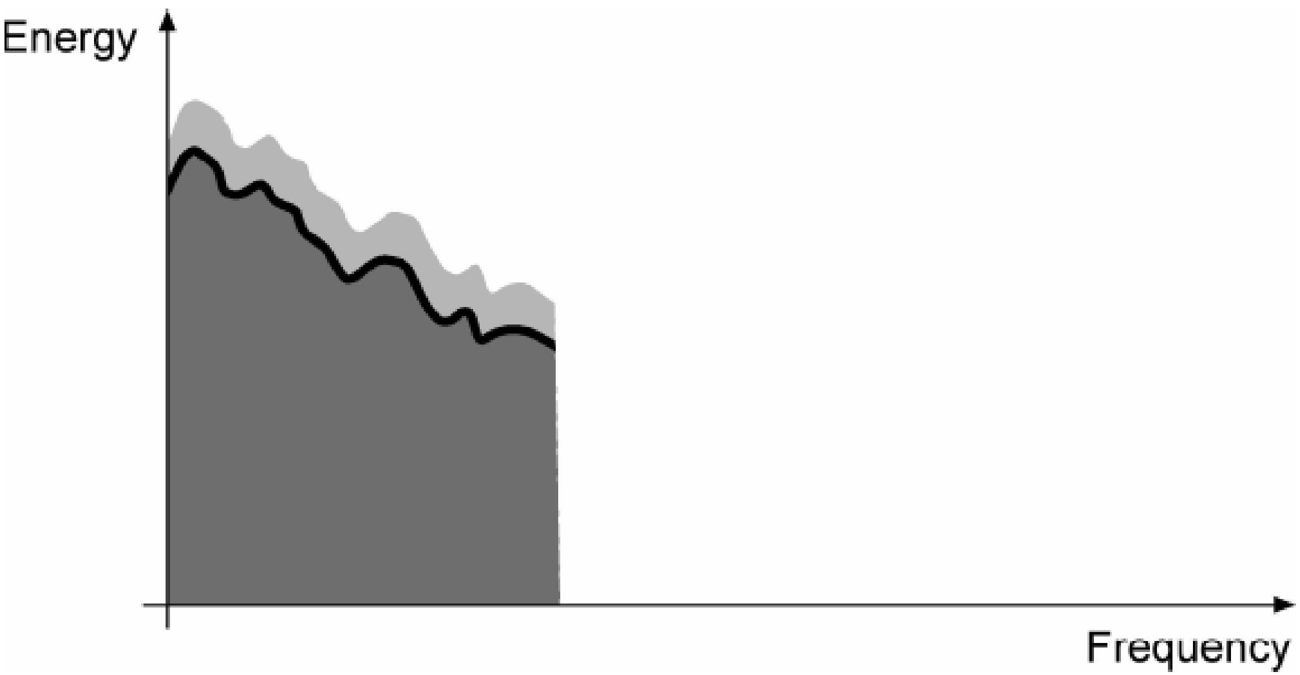

Figure 5: Limiting the audio bandwidth

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Bandwidth Extension (8)

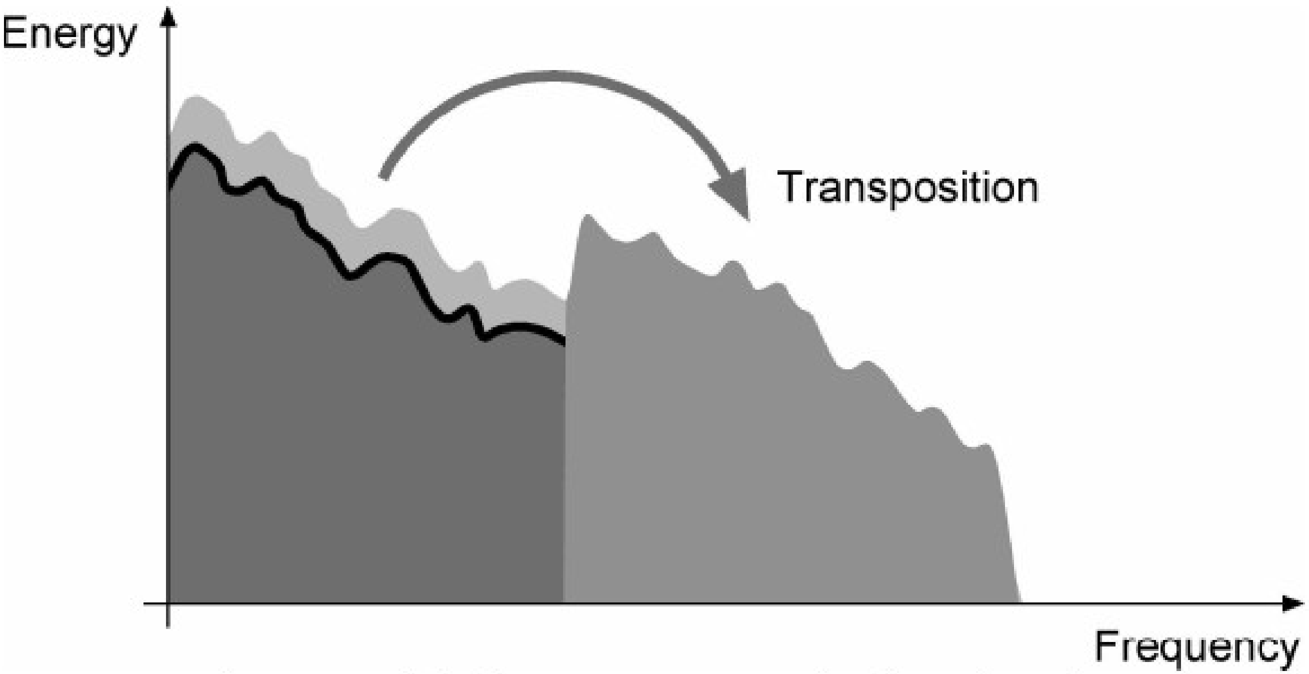

Figure 6: High frequency generation based on the waveform coded low frequency part

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Bandwidth Extension (9)

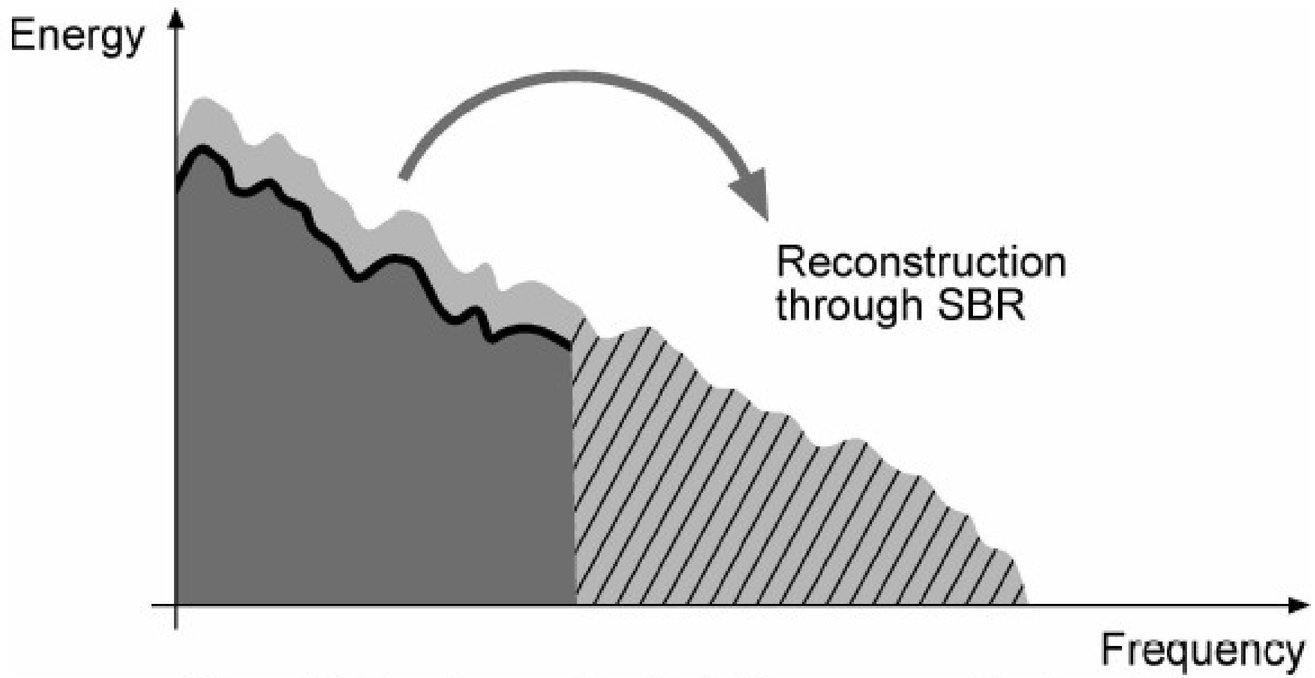

Figure 7: Spectrum after high frequency adjustment

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Audio Coding for Communication Applications

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Introduction (1)

- Audio coding/compression is used in:
  - Multimedia applications
  - Storage
  - Digital broadcast: Digital radio
    - Sirius, XM-Radio, iBiquity, DAB
- Examples:
  - MP3 (MPEG-1 Layer 3)
  - MPEG-2/4
  - AAC
  - AC-3
  - PAC

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Introduction (2)

- New networks:
  - Higher rate wireless services (for instance with

    space-time block coding)
  - Quality of service (->low delay)
  - In-home or local networks

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New Communication Applications

- High quality teleconferencing
- Reporting for radio or TV stations, using wireless networks
- → Delay wise most critical:
  - Virtual presence (for instance musicians playing together over long distance)
  - Concerts with wireless microphones and speakers (data compression for transmitted power and bandwidth)
  - Desired delay < 10 ms (Ultra Low Delay)

Goals

- Audio coding for communications:
  - Ultra low encoding/decoding delay (<10ms)
  - High quality for music and speech
  - Bit-rates about 50-100 kb/s

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Problems

- Conventional audio coders:
  - Good audio quality
  - But very high encoding/decoding delay (>100 ms)

- Speech coders:
  - Low encoding/decoding delay (order of 10…50ms), suitable for communications applications
  - But not high quality audio, don't perform well on non-speech signals like music or room noise

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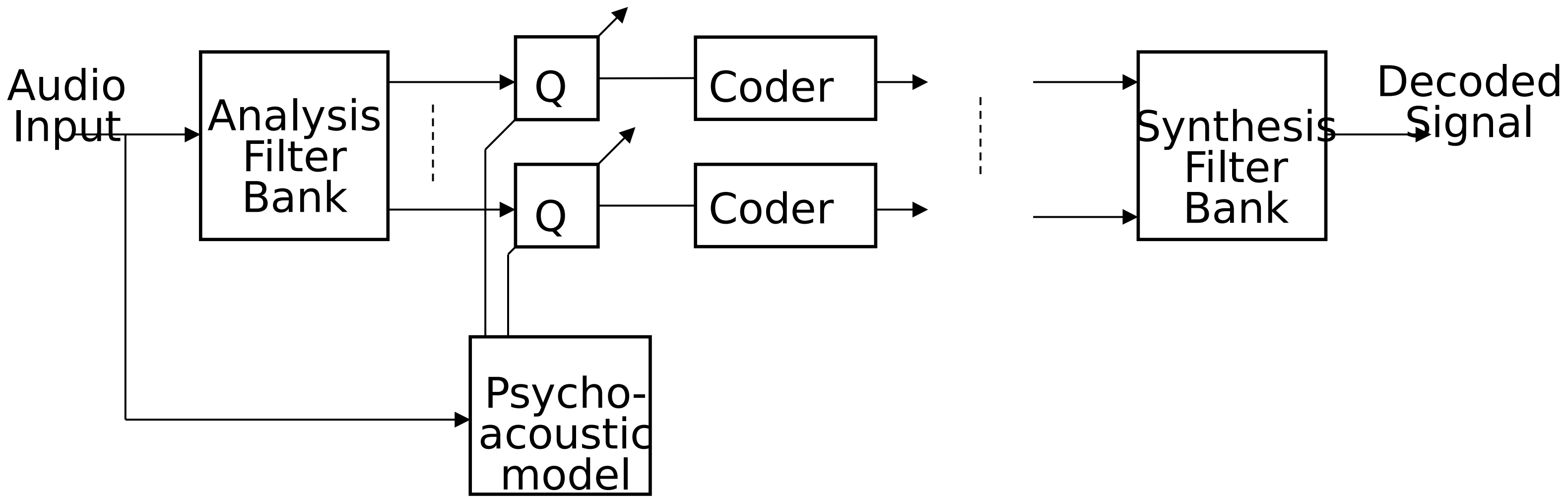
Conventional Audio Coders



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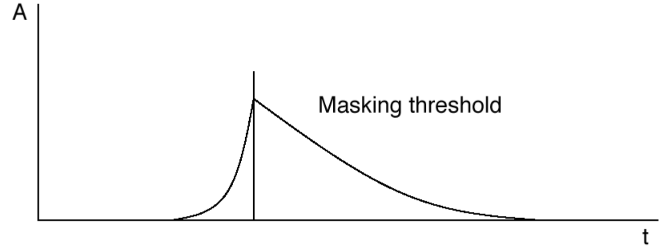
Compression

- Irrelevance (Psycho Acoustics)
  - What the receiver (ear) cannot detect
  - Sound below the threshold of hearing
  - In general sound below the psycho-acoustic “masking threshold”

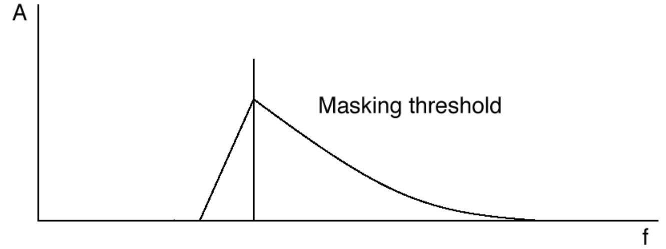
- Redundancy
  - The predictability or statistical dependencies in a signal

Basics of Psychoacoustics (Irrelevance)

- Temporal masking threshold



- Spectral masking threshold



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Major Sources of Delay

- System delay of analysis and synthesis filter bank
- Buffering for bit-rate smoothing
- Previous Approaches:
  - Low delay filter banks
  - MPEG-4 low delay coder: reduced number of subbands

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Limitations of previous Approaches

- High coding gain requires high numbers of subbands
  - Low delay filter banks: delay lower bounded by downsampling factor (= number of bands)
  - MPEG 4 low delay coder: reduced coding efficiency (higher bit-rate), not very low delay (ca. 30 ms at 32 kHz sampling).
  - Reason: subband coding leads to trade-off between coding efficiency and delay.

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New Approach

- Subband coding has same asymptotic gain as predictive coding

  (Jayant, Noll, 1984; Nitadori, 1970).

- But predictive coding has lower delay

  → Replace filter bank by predictor

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How to apply predictive Coding

- Problem: output of psycho-acoustic model is a time/frequency description.
- Approach:
  - Separate stages for application of irrelevance (psycho-acoustics) and redundancy reduction
  - Apply psycho-acoustic quantization noise shaping with linear filters (irrelevance red.)
  - Use lossless predictive coding after quantization (redundancy reduction)

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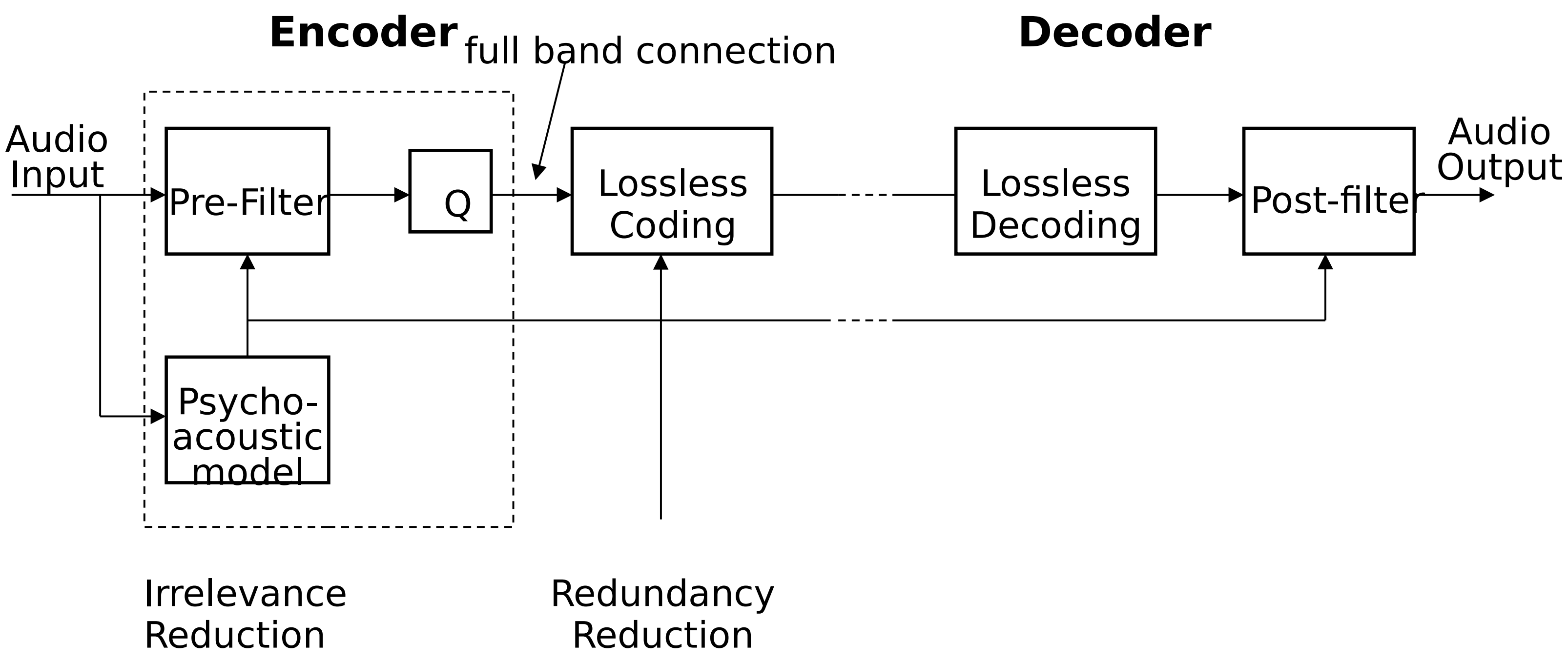
Pre- and Post-Filter Approach



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Function

- Pre- and post-filters form the quantization noise over frequency and time
- Post-filter is inverse of pre-filter
- Pre-filter normalizes the signal to its psychoacoustic masking threshold
- Simple uniform constant step size quantizer is used (rounding operation)
- Added benefit: more precise control over quantization noise shape than conventional approach

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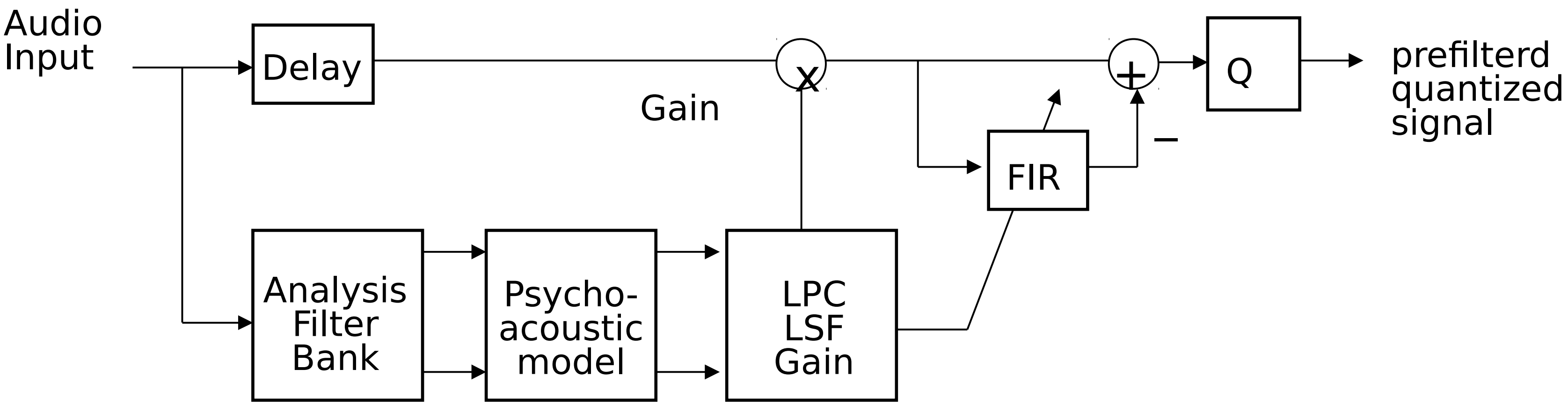
Pre-Filter Structure

Short delay for synchronization with psycho-acoustic model
(our implementation: 128 samples + 128 samples blocking delay)



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Properties

- Disadvantage:
  - Computationally complex structure
- But advantage:
  - No inherent delay, suitable for communications applications

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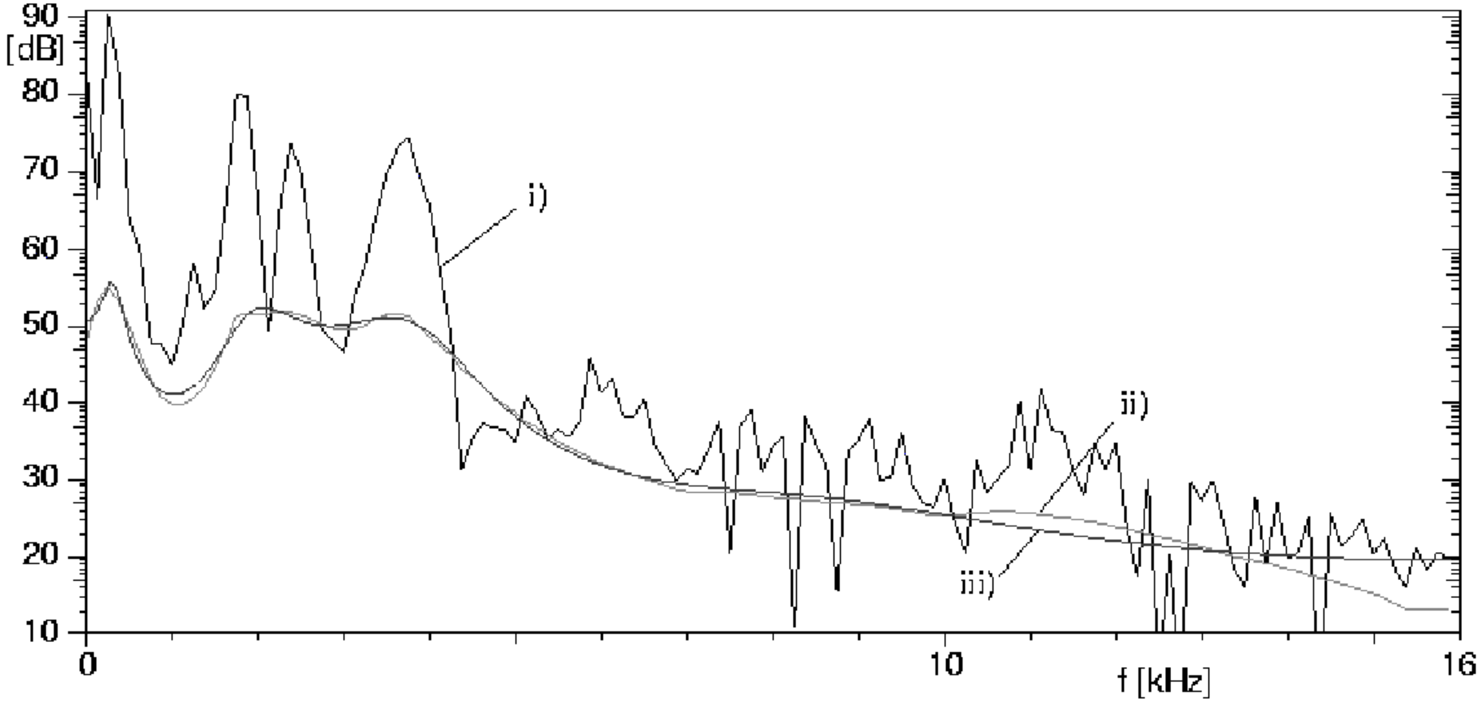
Example Frequency Response



Spectrum of the signal (i) compared to masking threshold (ii) and freq. resp. of post-filter (iii)

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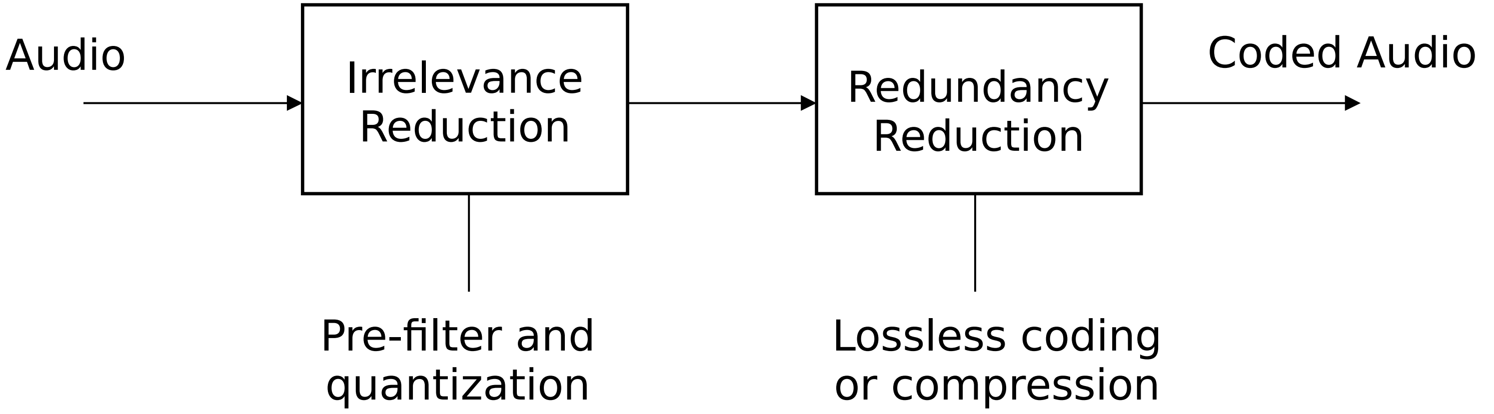
Redundancy Reduction

- Irrelevance removed after pre-filter and quantizer
  - Lossless compression needed for perceptual lossless coding



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Lossless Coding Unit

- Goals:
  - high compression ratios
  - low delay

- Previous approaches:
  - General purpose or text: Lempel-Ziv, PPMZ
  - Audio: Shorten (Softsound, GB), LPAC, LTAC

    (TU-Berlin, Germany), WaveZip (Soundspace,

    CA), MLP (Meridian, GB, for DVD)

Lossless Coding

- Previous approaches:
  - Made for file compression
  - Based on forward (block based) prediction, or transforms
- Problems:
  - High encoding/decoding delay
  - Compression ratio can be improved
- New approach:
  - Backward adaptation (based on past) for low delay
  - Cascading predictors for improved compression

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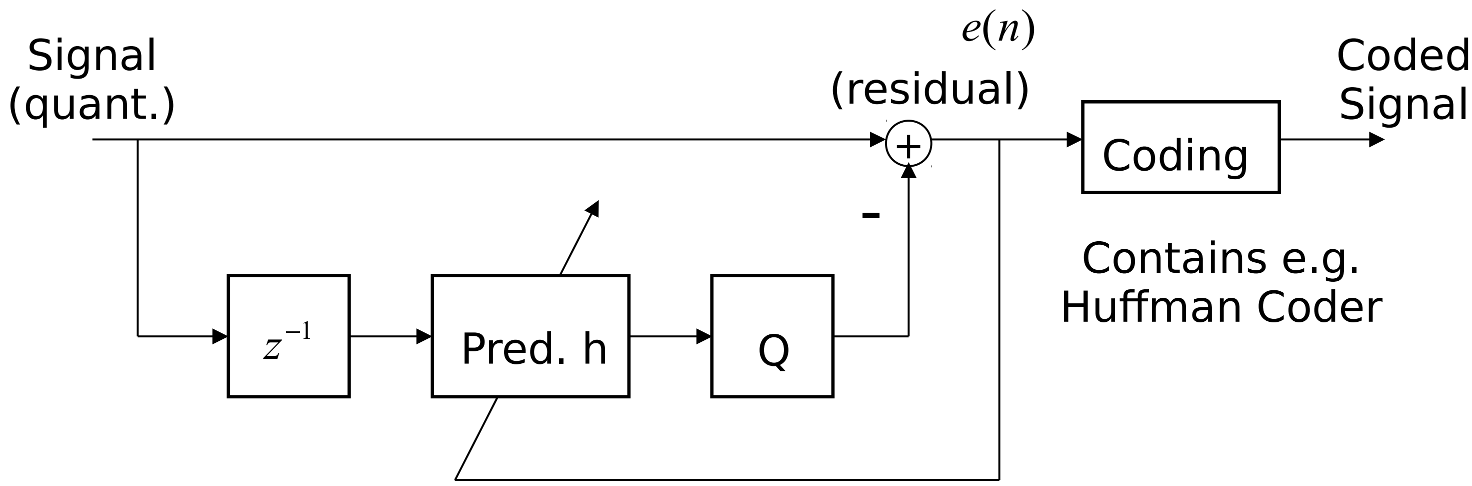
Lossless Predictive Coding – Encoder



For backward adaptation: Predictor coefficient vector **h** updated with LMS algorithm.

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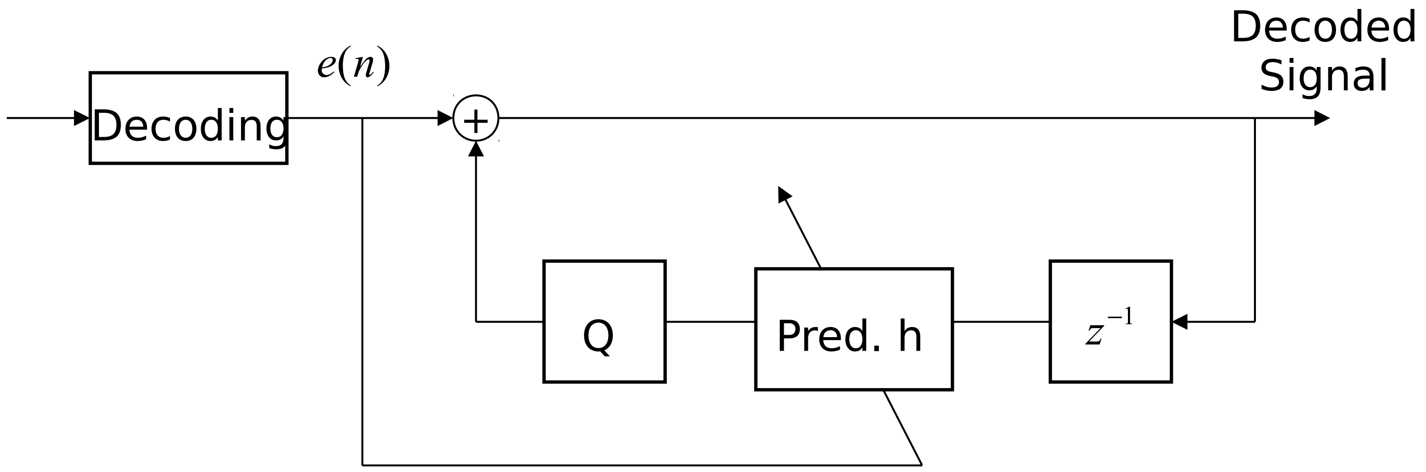
Lossless Predictive Coding – Decoder



Observe: Quantization / rounding of predicted value does not affect lossless property

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Low Coding Delay

- Backward adaptation with LMS algorithm
  - Define a vector of input samples:

$$\mathbf{x}^T(n) = \left[x(n-L+1),\ldots,x(n)\right]$$

  - The predicted value

$$\left[P(n)\right] = round\left(\mathbf{x}^T(n-1)\mathbf{h}(n)\right)$$

  - Update of the predictor coefficient vector h with normalized LMS (Widrow, Hoff, 1960).
    Prediction error:

$$e(n) = x(n) - \left[P(n)\right]$$

$$\mathbf{h}(n+1) = \mathbf{h}(n) + \frac{e(n)}{1+\lambda\left\|\mathbf{x}(n)\right\|^2}\mathbf{x}(n)$$

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Increased Compression Ratio

- Cascading LMS predictors, using the final output,

  has advantages:
  - Increased adaptation speed
  - Improved prediction accuracy
  - Better numerical stability (see Prandoni, Vetterli,

    1998, for a special case)

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Cascading and Combining Predictors

- For us important: Availability of predictors of different orders as additional outputs.
  Reasons:
  - Very non-stationary signals (attacks) require fast adaptation/ short filters
  - Stationary signals require long filters

- Approach:
  - Combine predictors of different orders adaptively (analog to block switching)

Combination of Predictors (1)

- Assume predictors P_1, P_2, P_3 with different orders for different signal statistics. How can they be combined?

- Use predictive minimum description length principle for the “optimal” combination of predictors

$$P = \sum_{i=1}^{3} w_i P_i$$

w_i : probability of P_i being “correct” on past signal.

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Combination of Predictors (2)

- Assume that the prediction error has a Laplacian distribution ($p(x)=e^{-c\cdot|x|}$), with the prediction error $e(n)=\sum_n \left|x(n)-P_i(n)\right|$ and the weights w_i we get:

$$w_i \propto e^{-c\sum_n \left|x(n)-P_i(n)\right|}$$

- Weights “reward” predictors with good past performance
- The weights are normalised such that they add up to 1.

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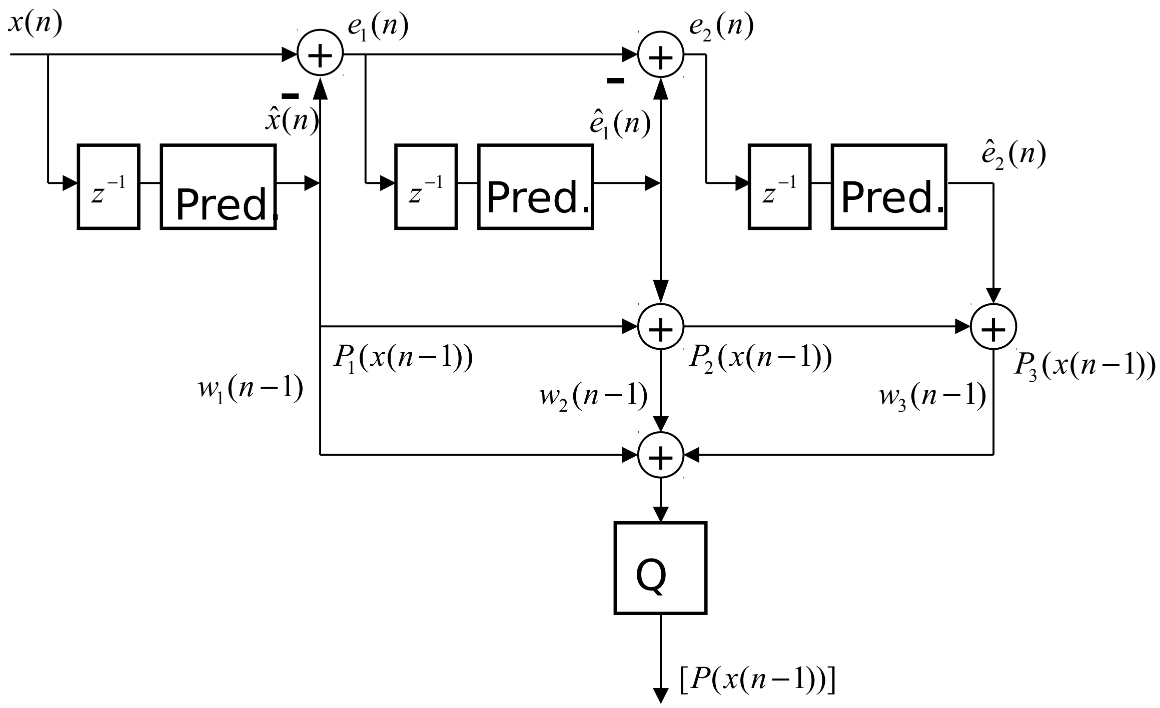
Weighted Cascaded LMS (WCLMS) Prediction



The w’s are adapted based on previous prediction errors, 0<w<1, to adapt to signal statistics (orders, number of coefficients: 120, 80, 40)

Entropy Coding of Residuals

- Take known algorithms, e.g.:
  - Adaptive Golomb-Rice Codes
  - Adaptive Arithmetic Coding
  - Block based Huffman using pre-calculated code books

- No additional delay introduced, because ULD implementation already is block-based (128 samples)

- Inherently variable bit rate

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Comparison of different lossless compression schemes

- Signals are at 32 khz sampling rate, bit-rate in bit/sample. Application after pre-filter and quantization.

Signal	Cascaded LMS	Shorten	Wavezip	LPAC
Pop	1.94	2.52	3.22	2.23
Jazz	1.99	2.67	3.35	2.48
mixed	2.16	2.58	3.19	2.35
Speech	1.96	2.48	3.09	2.12

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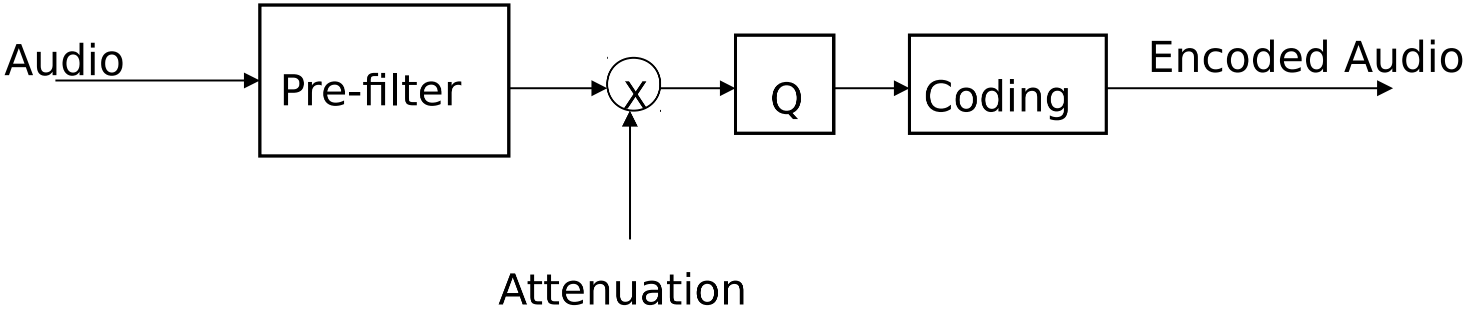
Controlling the Bit-Rate: constant bit rate mode

- A factor of less than 1 leads to quantization noise above threshold of audibility, but a reduced bit-rate

- By adapting the attenuation factor and iterating the lossless coder a target bit rate can be approximated



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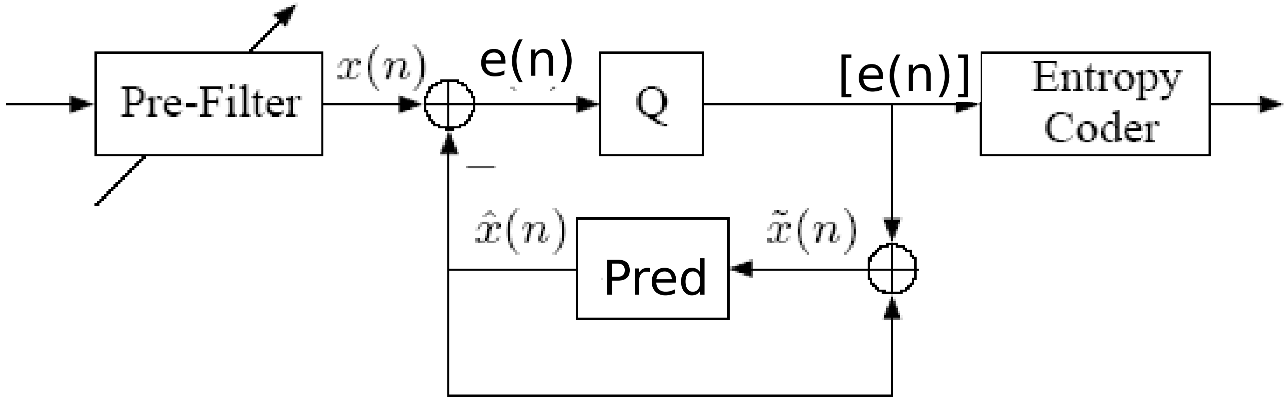
Alternative Predictor Structure

- Closed-Loop Predictor instead of Open-Loop Predictor
- Advantage: only one quantizer
- Disadvantage: Quantization and Prediction not separated anymore (Irrelevance and Redundancy Reduction)



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Conclusions

- Predictive coding can be used to obtain Ultra Low Delay audio coders
- Obtained delay: 6 ms (<10 ms)
- Subjective audio quality comparable to conventional high delay coder at same bitrate
- Price: higher complexity
- Can also be used to obtain high audio quality (unlike speech coders)

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