

The AES logo is a shield-shaped emblem. At the top, the word "AUDIO" is written in a bold, sans-serif font. In the center is a large, stylized letter "A" that resembles a speaker or a sound wave. At the bottom, the letters "ES" are written in a similar font. The entire logo is rendered in a golden-yellow color against a dark background.

# AES-Seminarium: Så fungerar MP3

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# AES-Seminarium: Så fungerar MP3

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 DOLBY.

# AES-Seminarium: Så fungerar MP3

*Födelsen av filformatet mp3 blev starten till en total omstrukturering av musikvärlden, där musik nu kunde skickas över internet. Bitströmmen för komprimerade format som mp3 och aac är blott en tiondel av bitströmmen för CD-formatet. I detta seminarium tittar vi på hur komprimeringen går till och vilken information i musikströmmen som har gått förlorad i processen.*

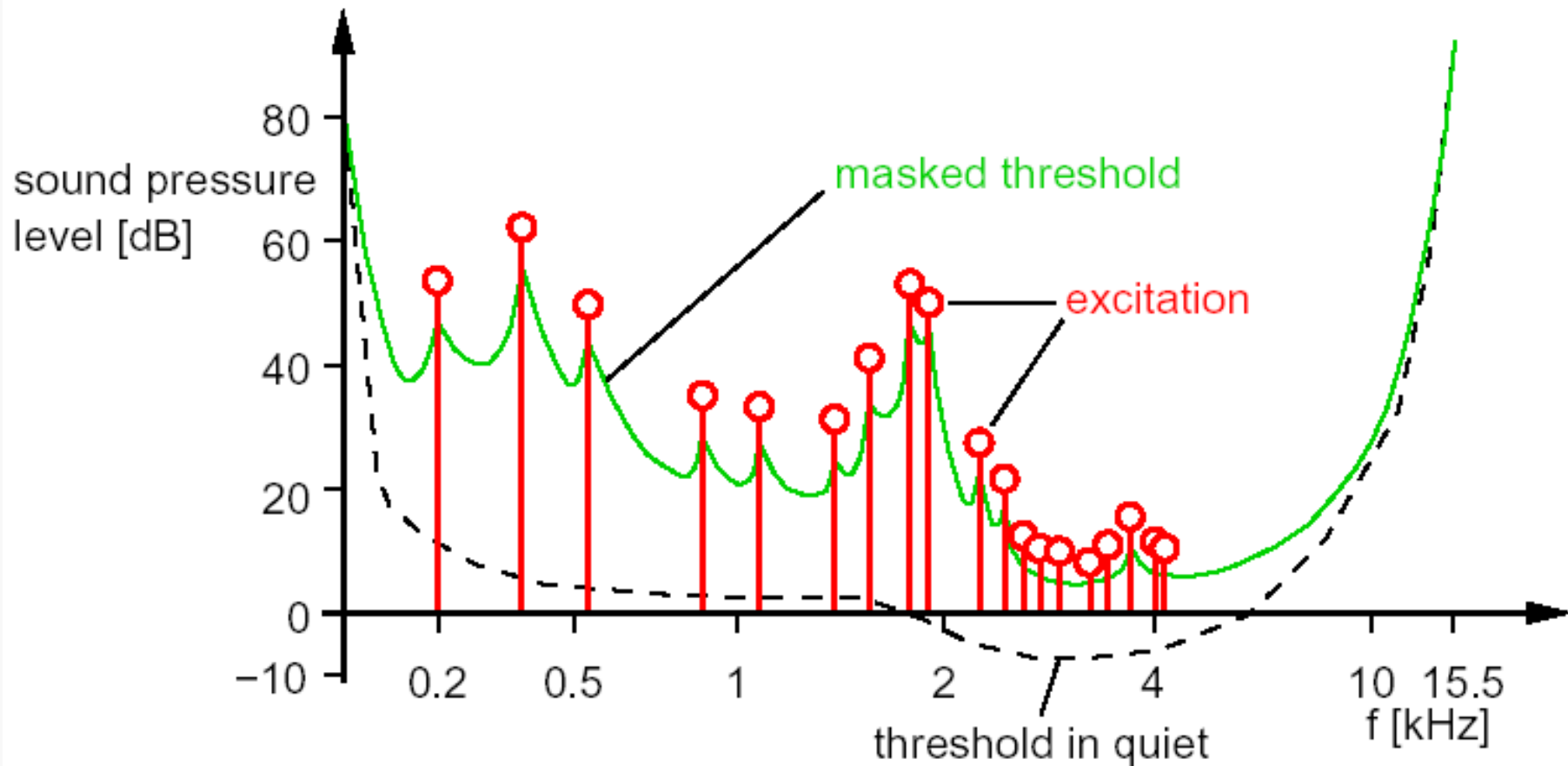
- MP3
  - MPEG-1 Audio, Layer III (ISO/IEC 11172-3, published 1992)
  - ISO/IEC 11172 – Information technology – Coding of moving pictures and associated audio for digital storage media at up to about 1.5 Mbit/s
- AAC
  - MPEG-2 Advanced Audio Coding (ISO/IEC 13818-7, published 1997)

=> Perceptual Audio Coding

# Perceptual Audio Coding – Motivation

- Digital audio requires “high” bit rates...  
CD: 16 bit PCM, 44.1 kHz sampling, stereo => 1.4 Mbit/s
- ...too high for many applications
  - portable players
  - broadcast/streaming
- Perceptual audio coding needed  
=> same (perceived) quality at lower bit rates
  - Exploit signal redundancy
  - Exploit perceptual irrelevance
- Popular codecs introduced 1992 (compression ratio ~1:10)
  - MPEG-1 Layer III (MP3)
  - Dolby Digital (AC-3)

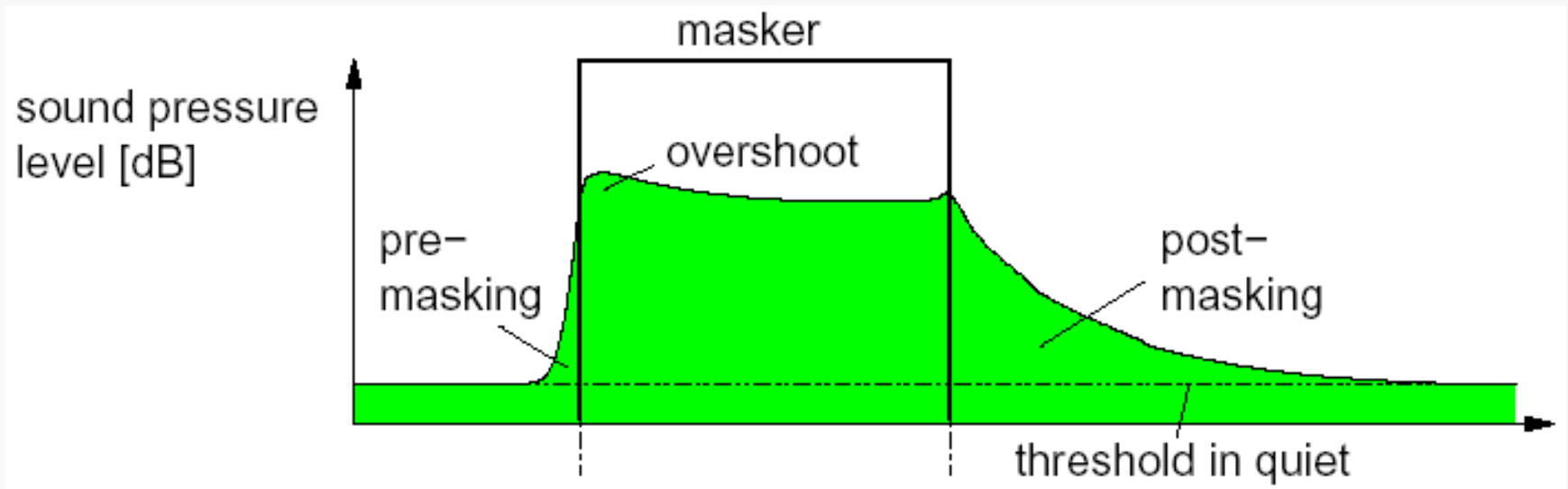
# Spectral Masking (simultaneous)



Masking: test tone (maskee) below threshold is inaudible

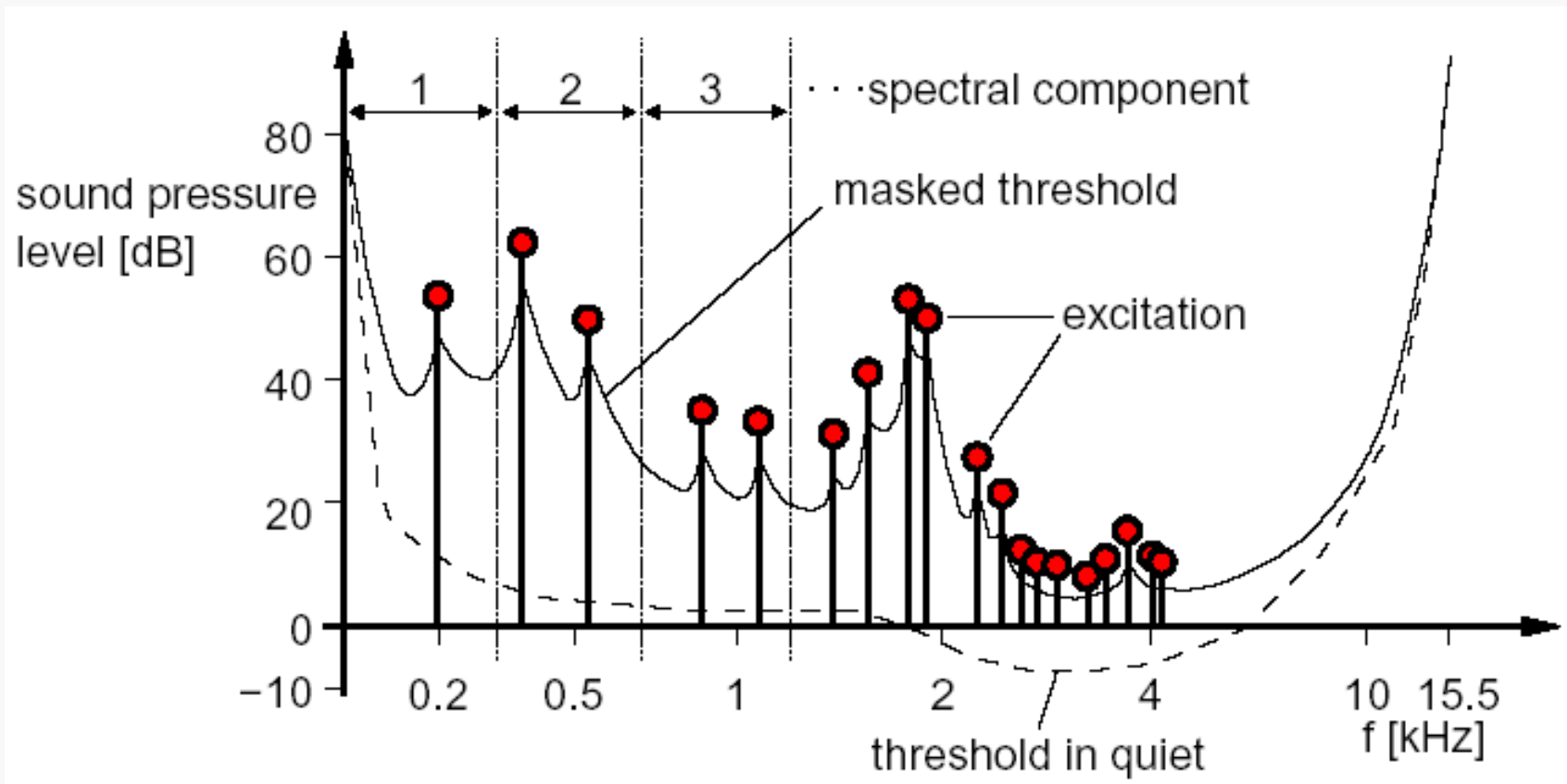
Masked threshold caused by excitation due to masker(s)

# Temporal Masking



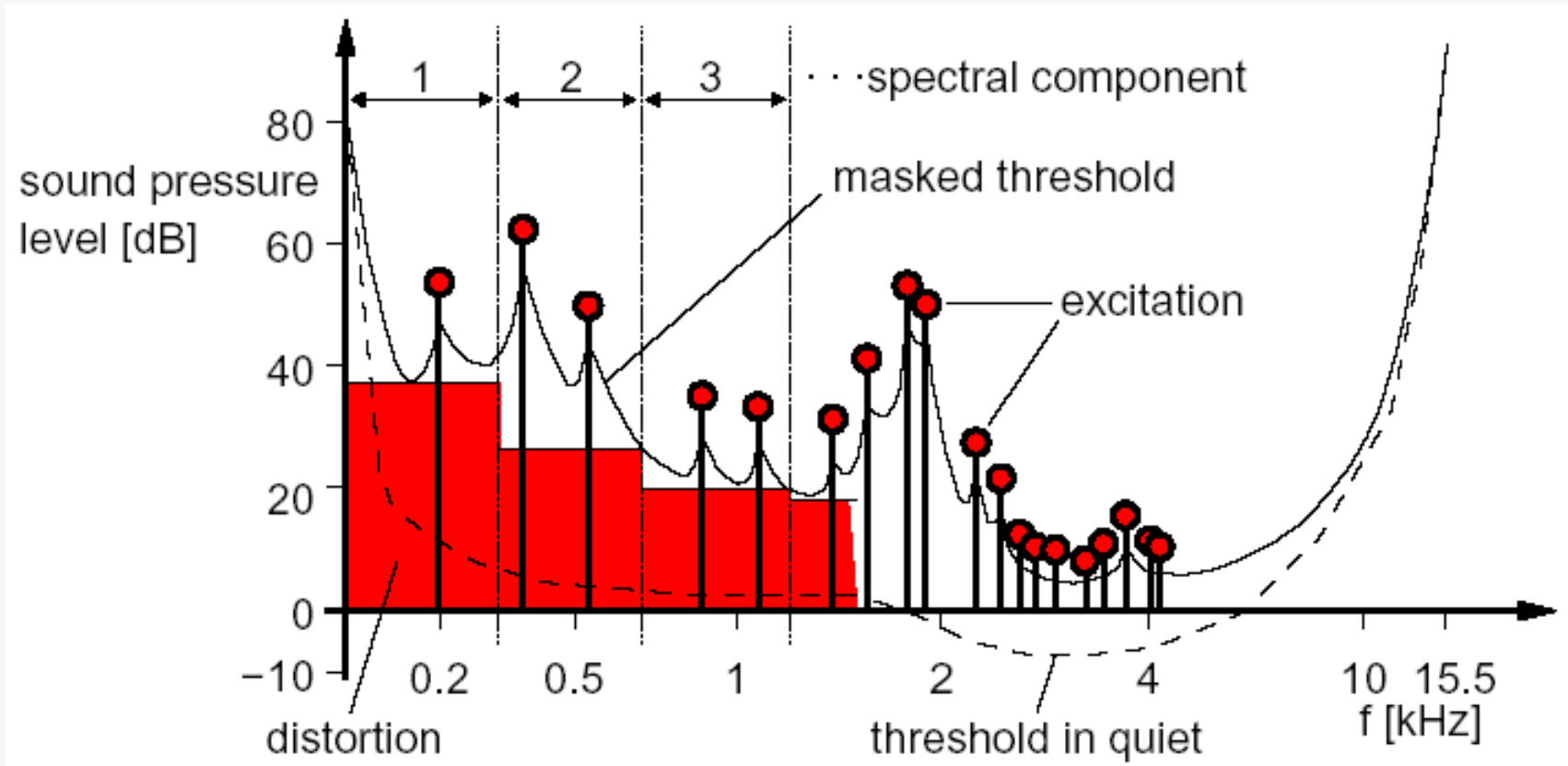
pre-masking: < 5 ms  
post-masking: ~ 20 ms

# Demonstration: Masked Threshold



Original signal

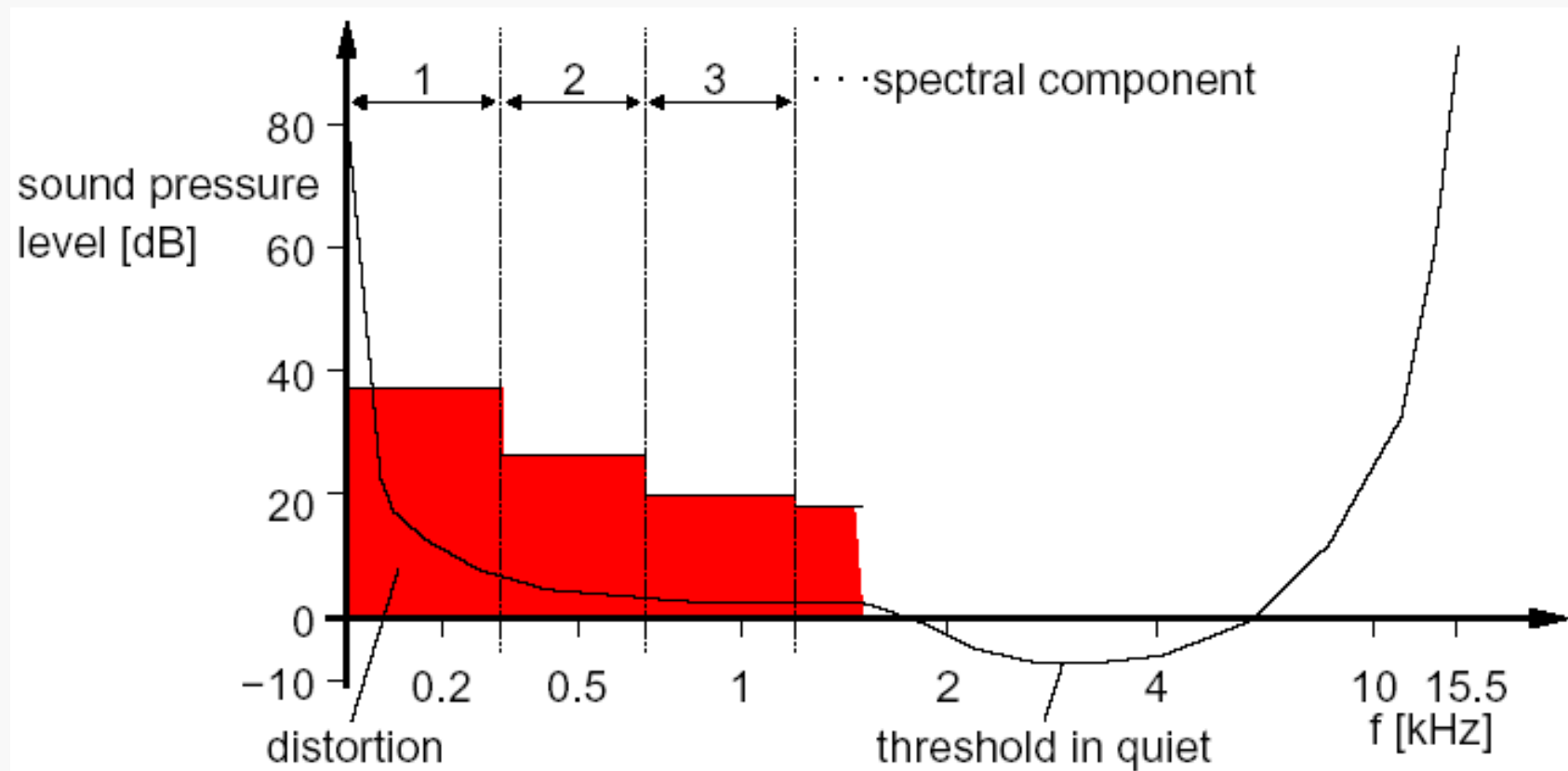
# Demonstration: Masked Threshold



Original signal plus noise at masked threshold (23 dB SNR)

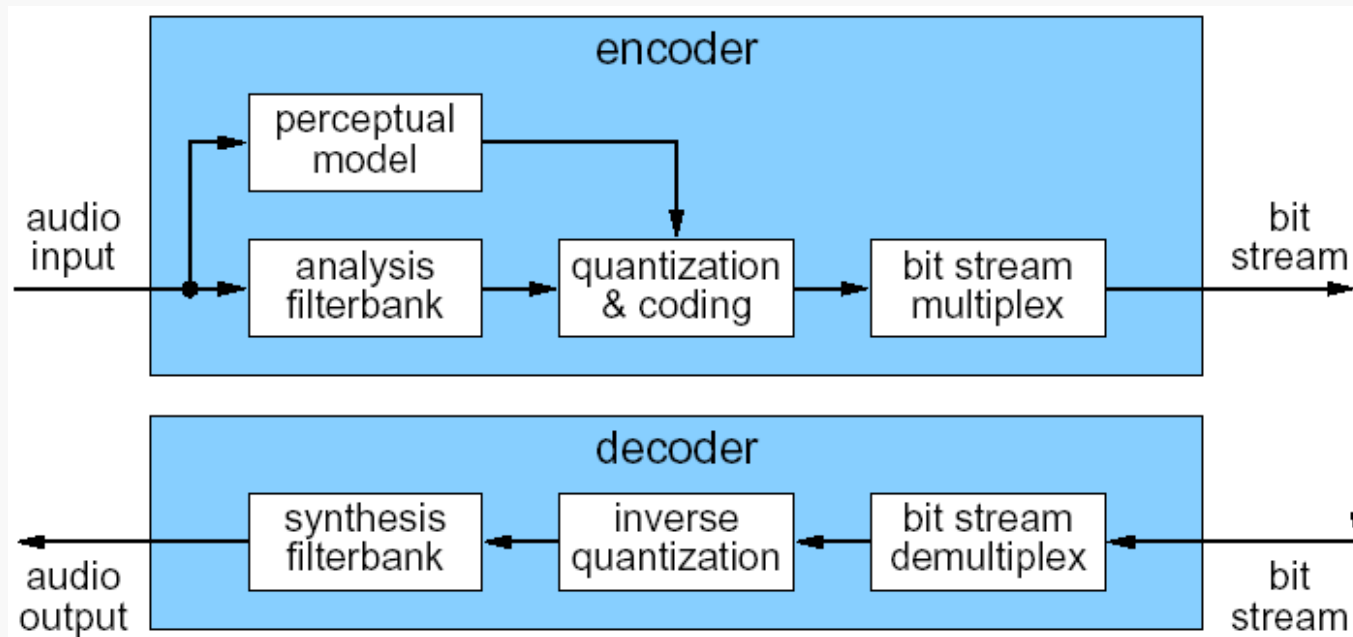


# Demonstration: Masked Threshold



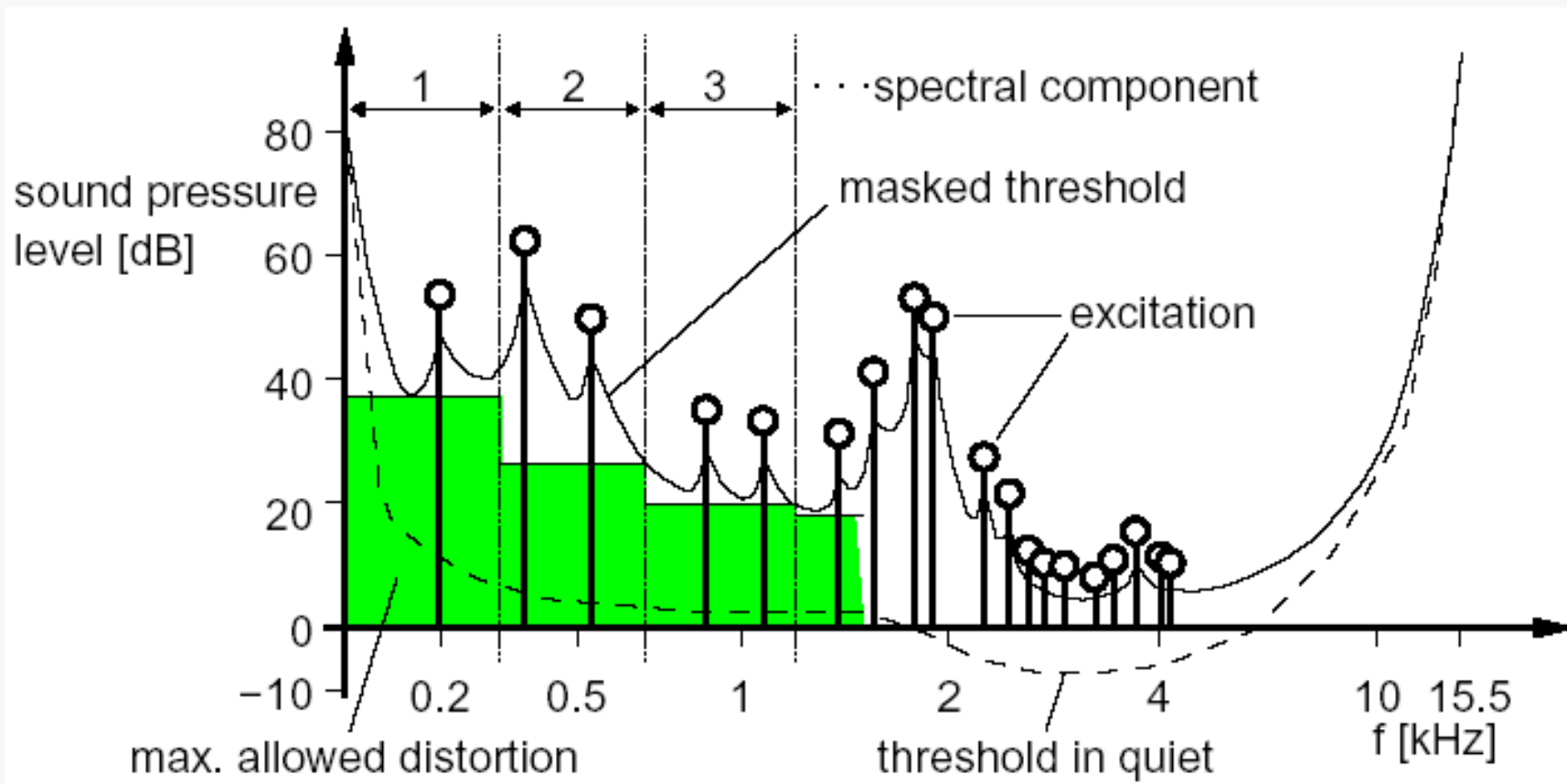
Noise at masked threshold without original signal

# Subband/Transform Audio Codec



- Popular “waveform codecs”: MP3, Dolby Digital, AAC, ...
- Analysis filterbank/transform enables...
  - ...redundancy reduction (spectral decomposition)
  - ...time- and frequency selective quantization

# Psychoacoustic Model controls quantization



Goal: quantization error (distortion) below signal-dependent time and frequency variant masked threshold

# Subband/Transform Audio Coding – Limitations

- Audio Demonstration

- Original PCM stereo                    1.5 Mbit/s
- AAC stereo                                128 kbit/s                    1:12
- Difference signal (23 dB SNR)



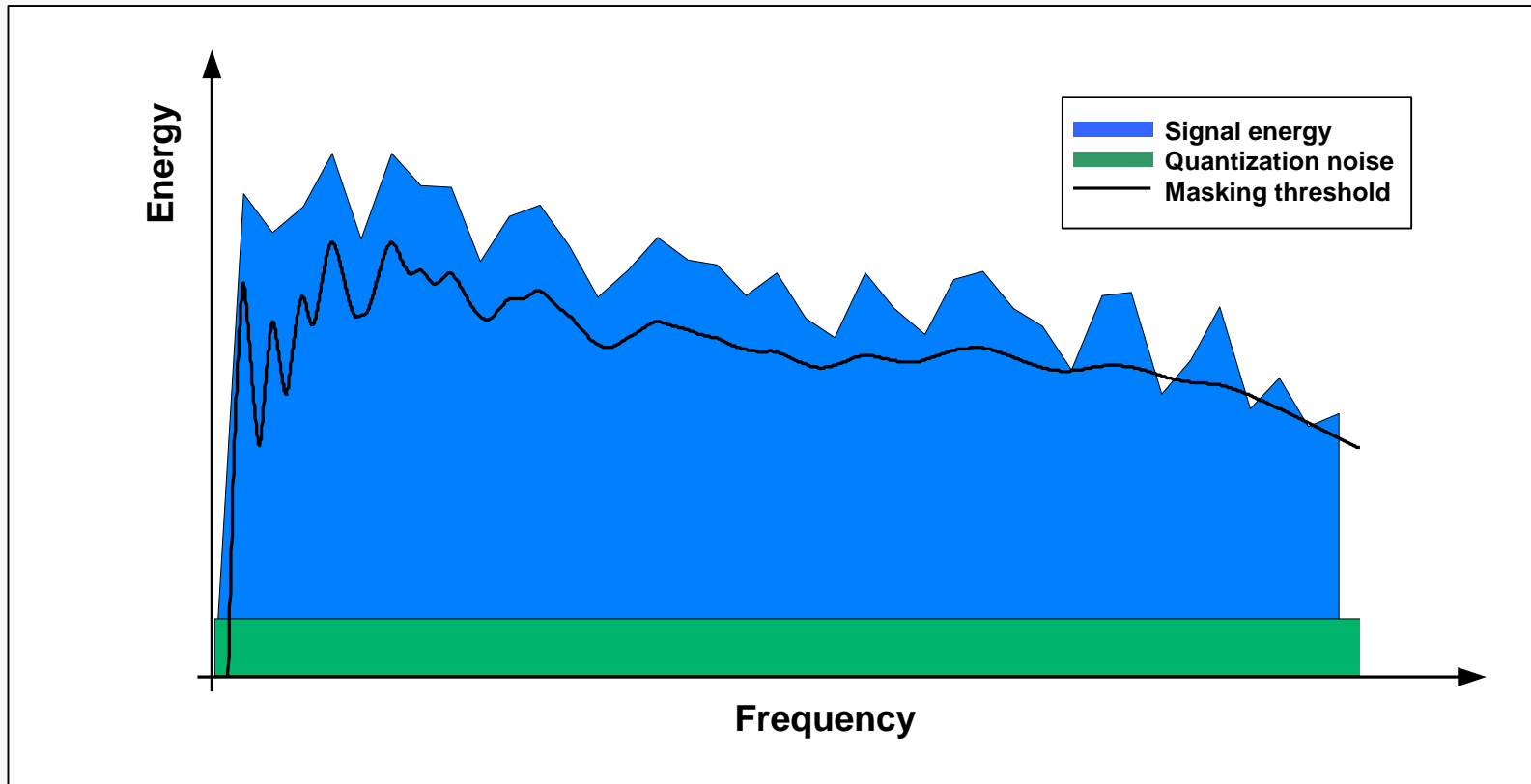
- But ...

- AAC stereo                                48 kbit/s                    1:32
- AAC mono                                 24 kbit/s                    1:64



# Subband/Transform Audio Coding – Limitations

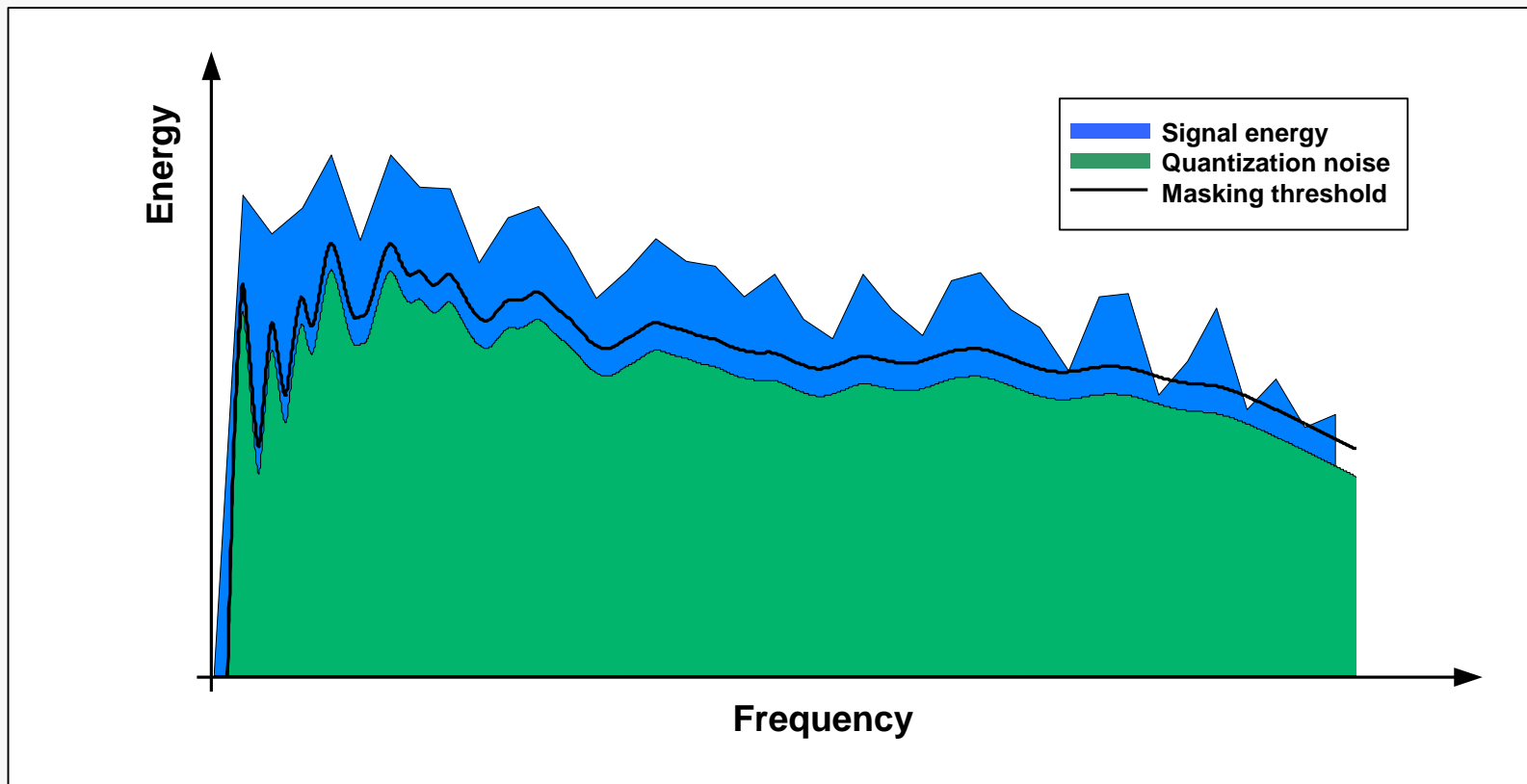
16 bit PCM



Visible blue area indicates bit rate

# Subband/Transform Audio Coding – Limitations

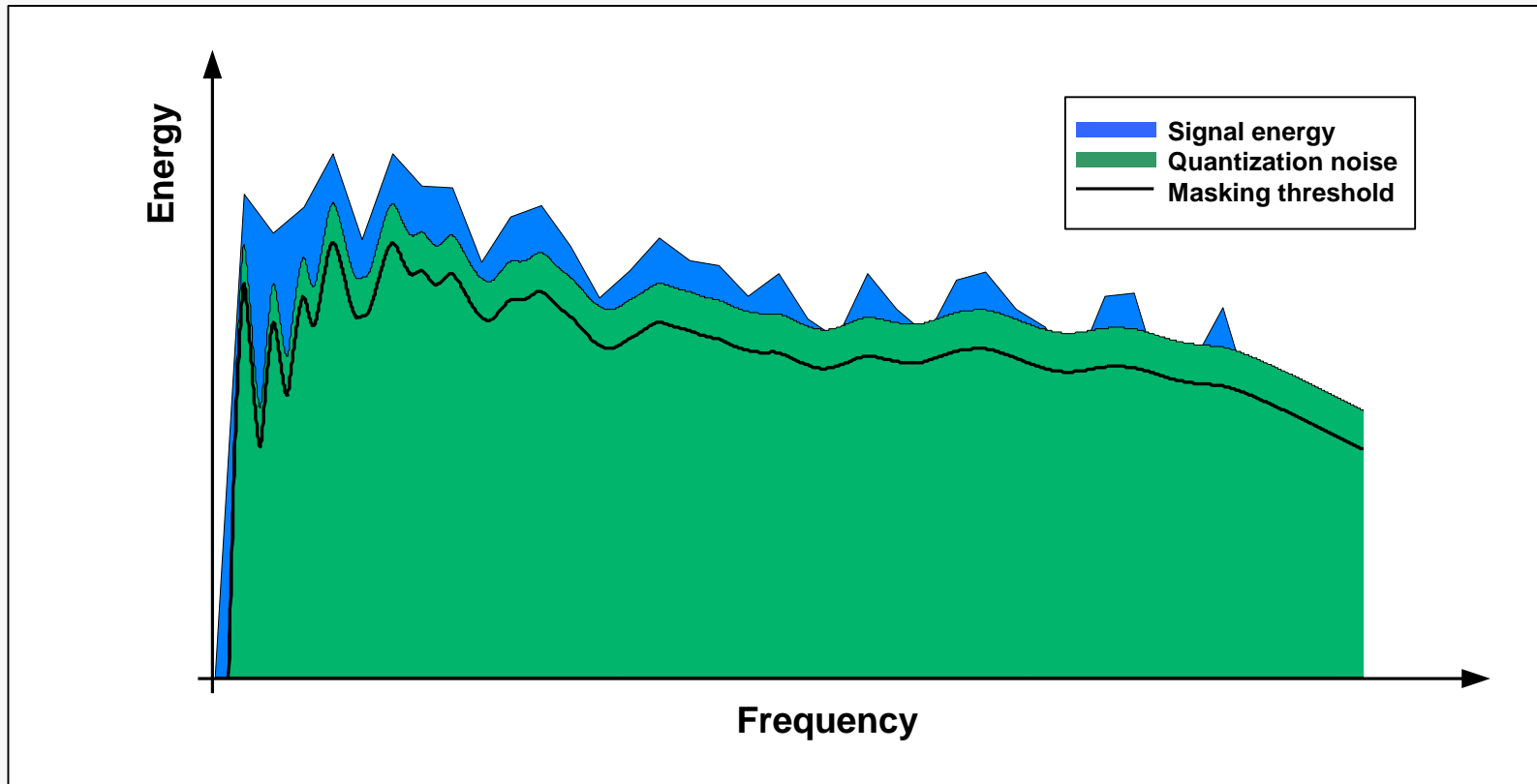
bit rate sufficiently high



Visible blue area indicates bit rate

# Subband/Transform Audio Coding – Limitations

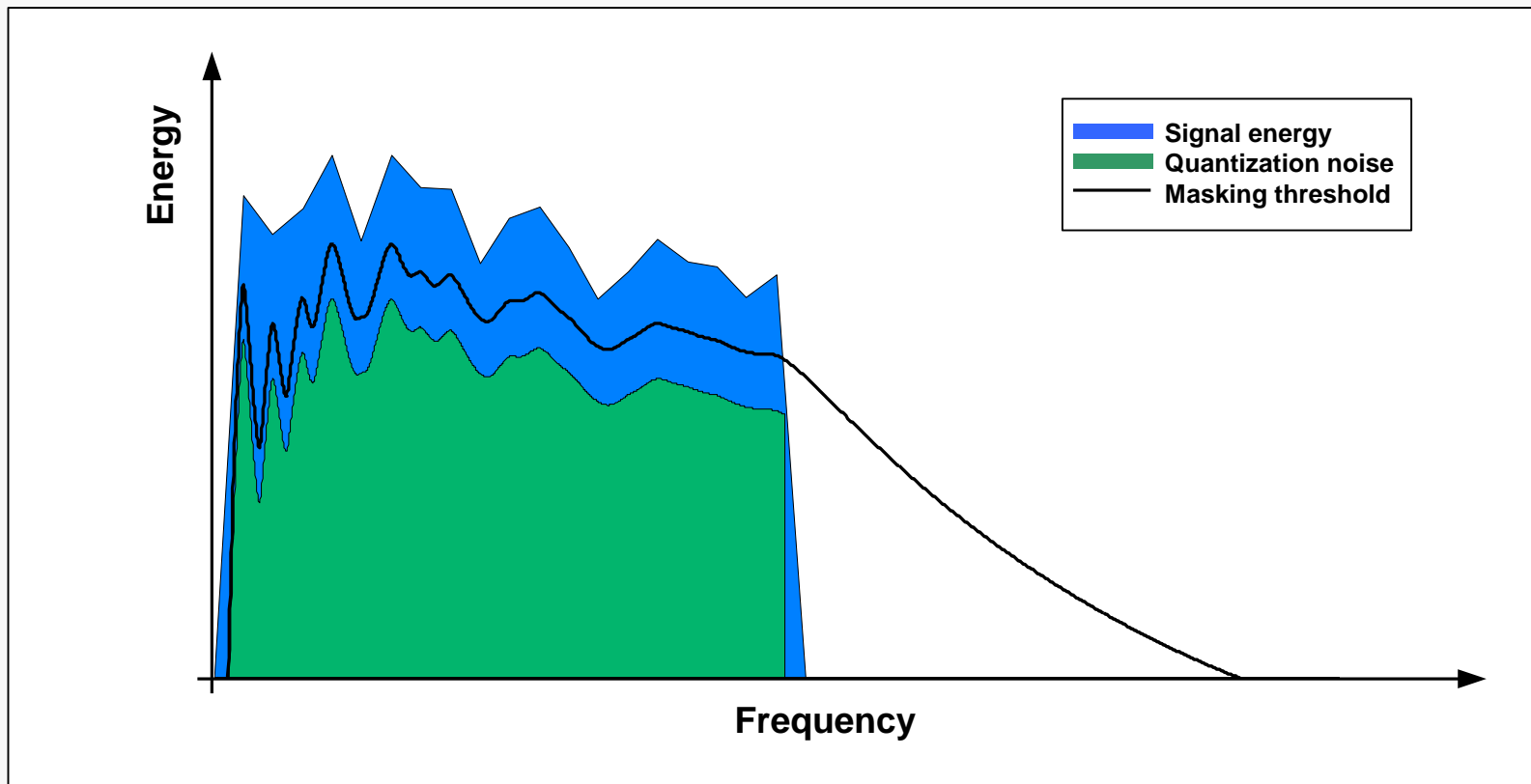
bit rate too low -> quantization noise audible



Visible blue area indicates bit rate

# Subband/Transform Audio Coding – Limitations

- -> limit audio bandwidth to reduce coding artifacts



Visible blue area indicates bit rate

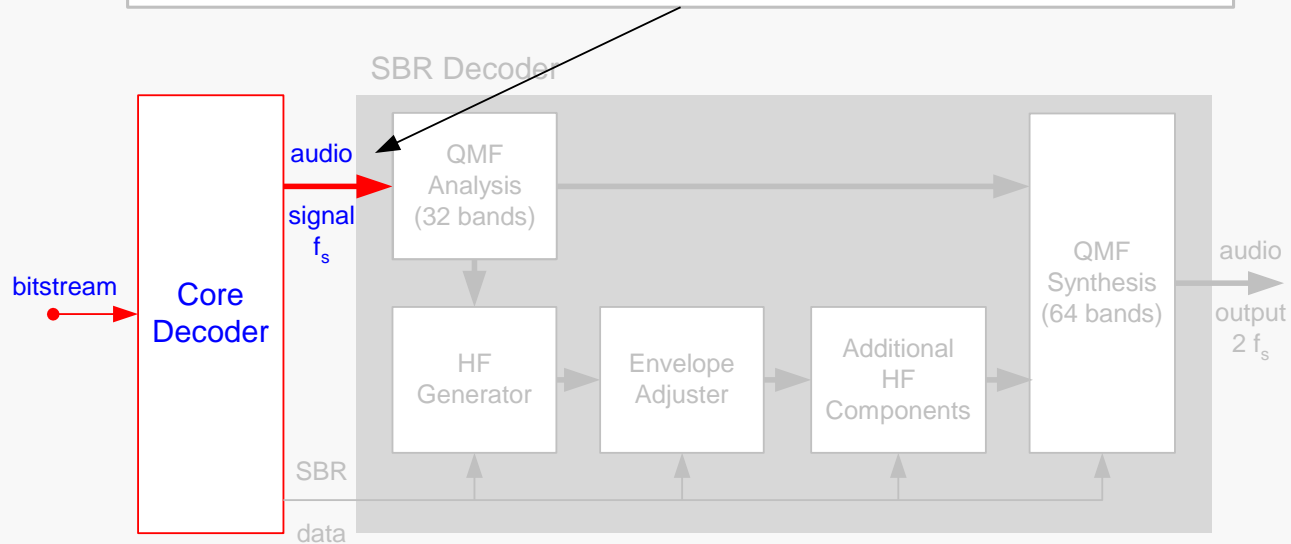
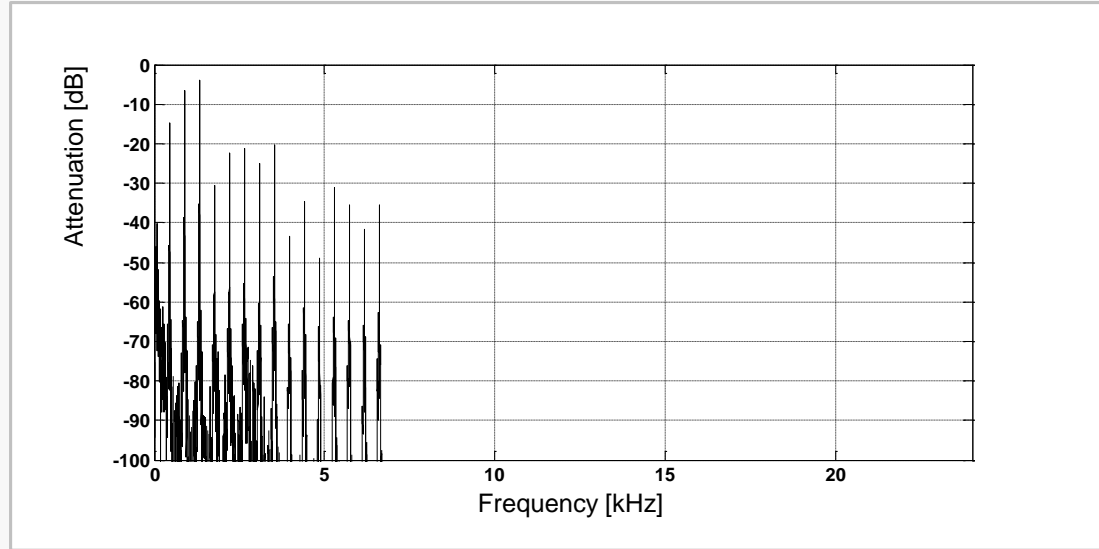


# Spectral Band Replication (SBR)

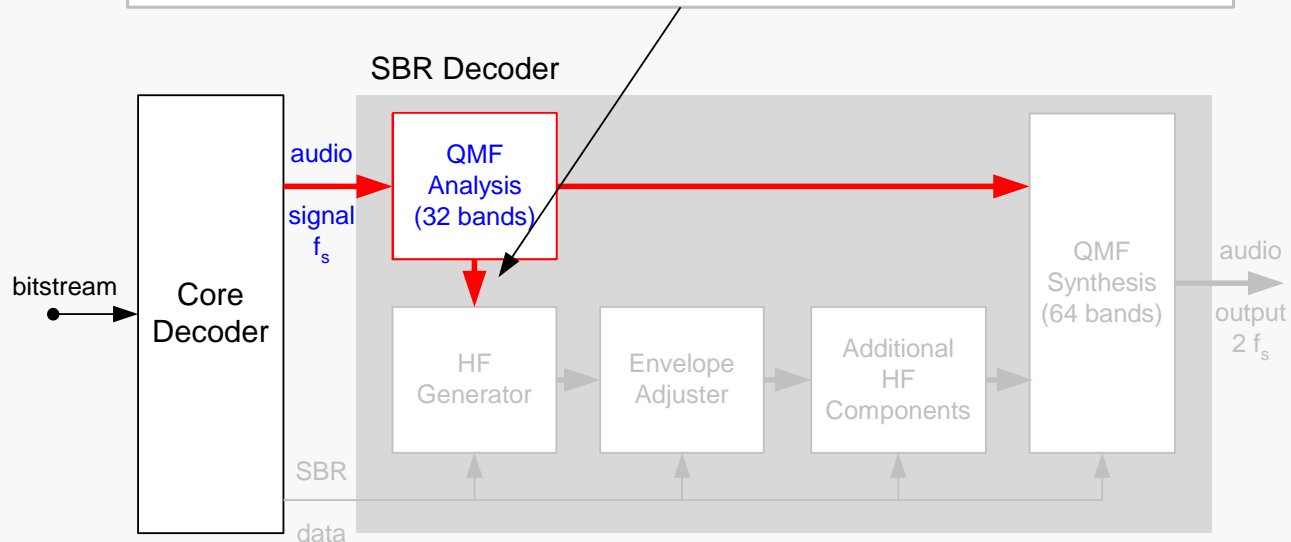
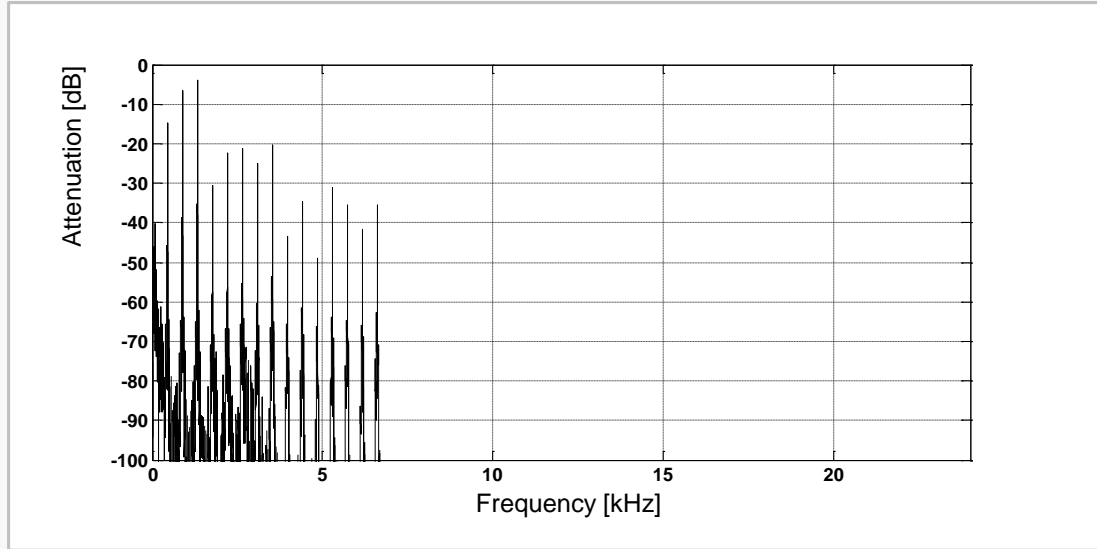
- Problem
  - Conventional coding of high frequency band “expensive” (needs many bits)  
=> limited audio bandwidth at low bit rates
- Approach
  - Reconstruct high frequency band from low frequency band
  - Side information (2 to 3 kbit/s per channel) to control reconstruction  
=> Spectral Band Replication (SBR)
  - “Parametric coding” of high frequency band (not “waveform coding”)
  - AAC+SBR: aacPlus / High-Efficiency AAC
- Developed by Coding Technologies
  - founded 1997 by Lars “Stockis” Liljeryd
  - acquired by Dolby in 2007



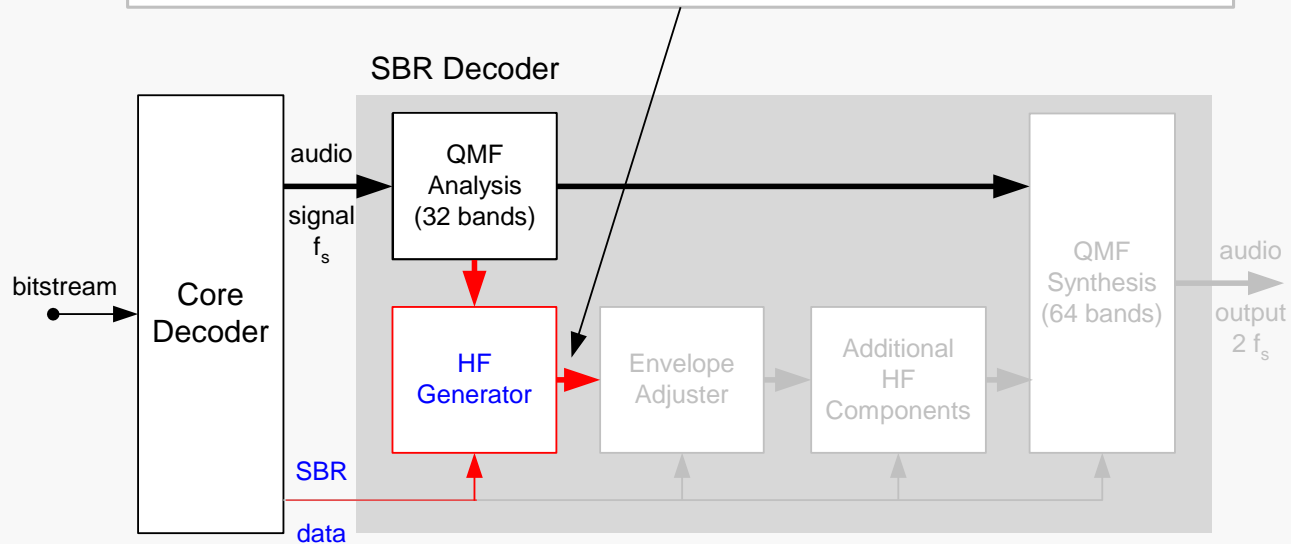
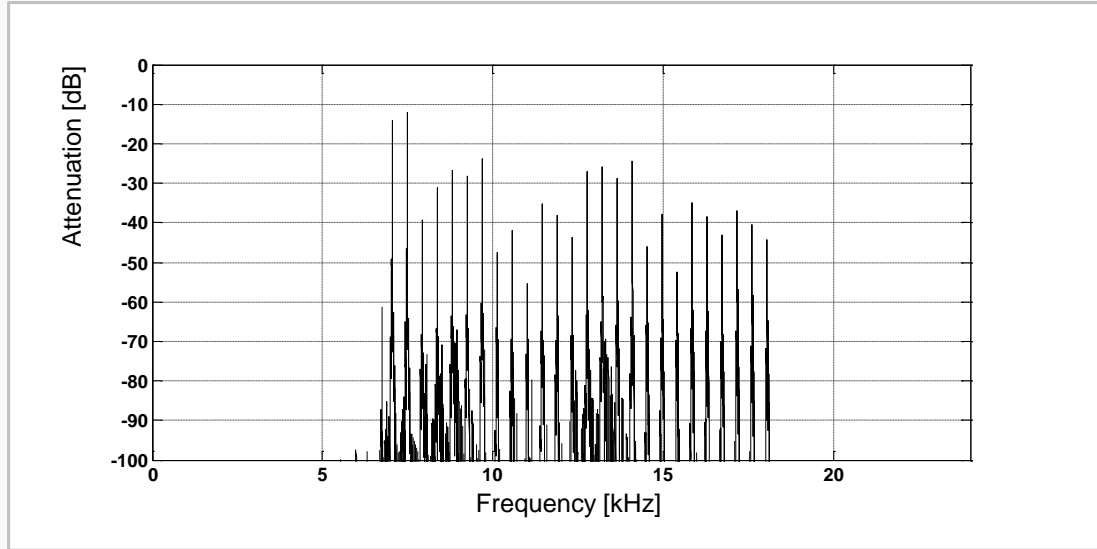
# Audio Decoder with SBR



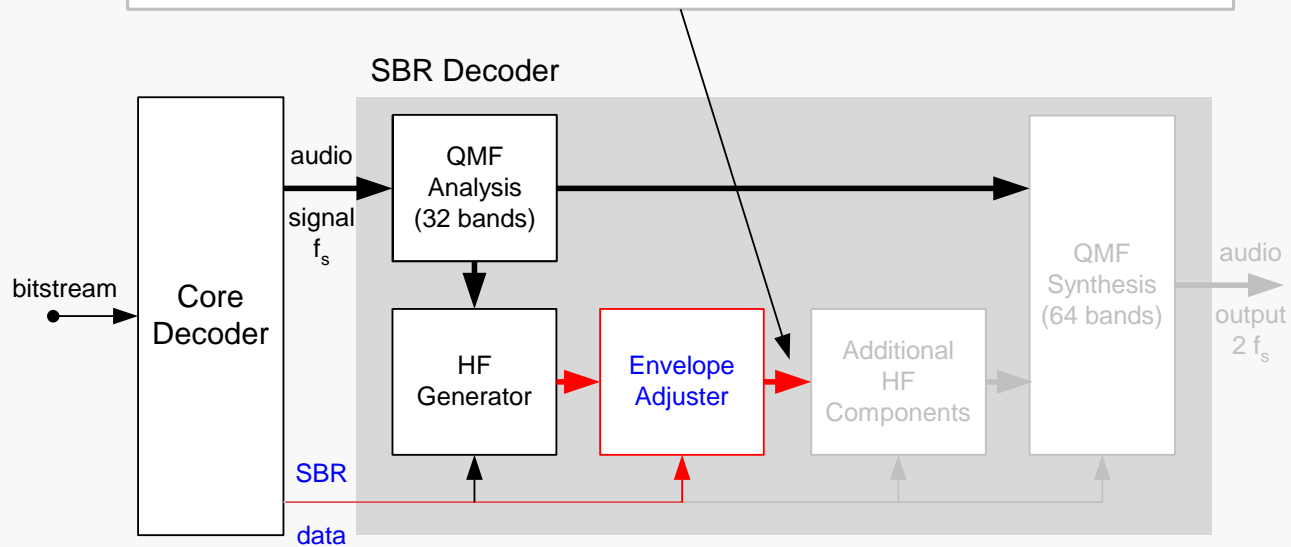
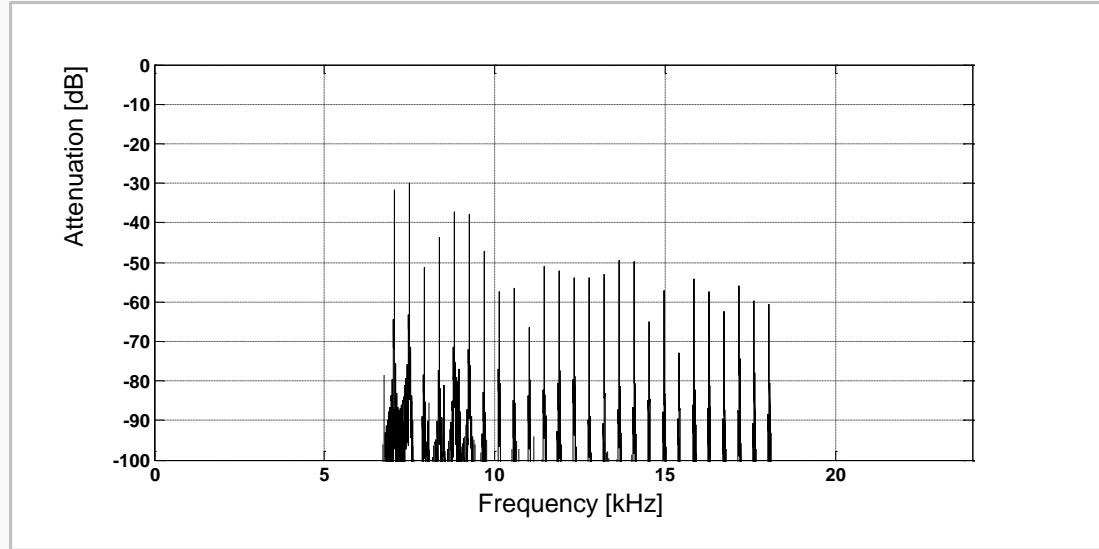
# Audio Decoder with SBR



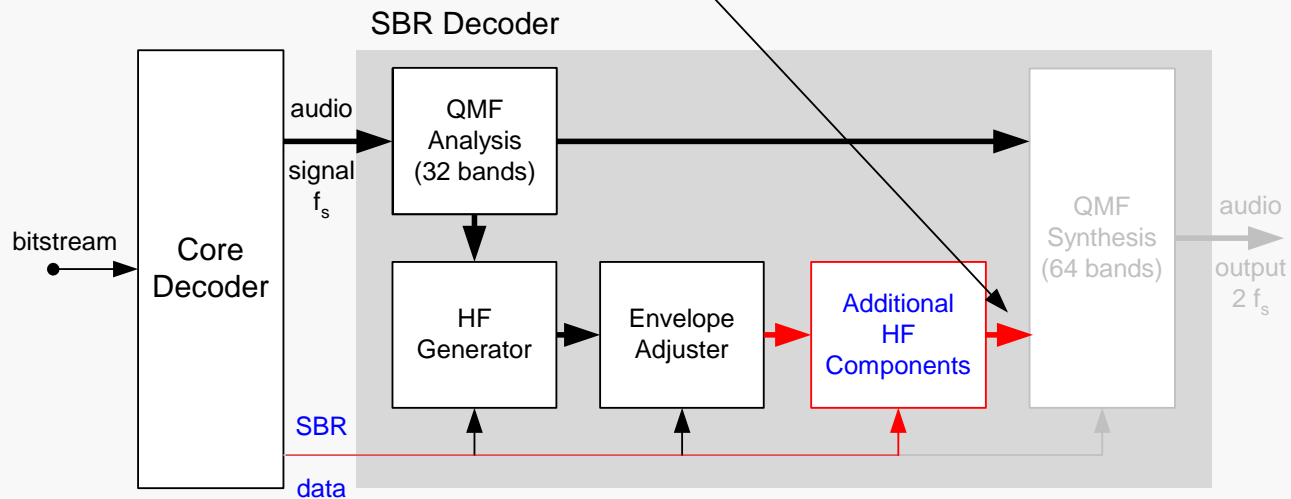
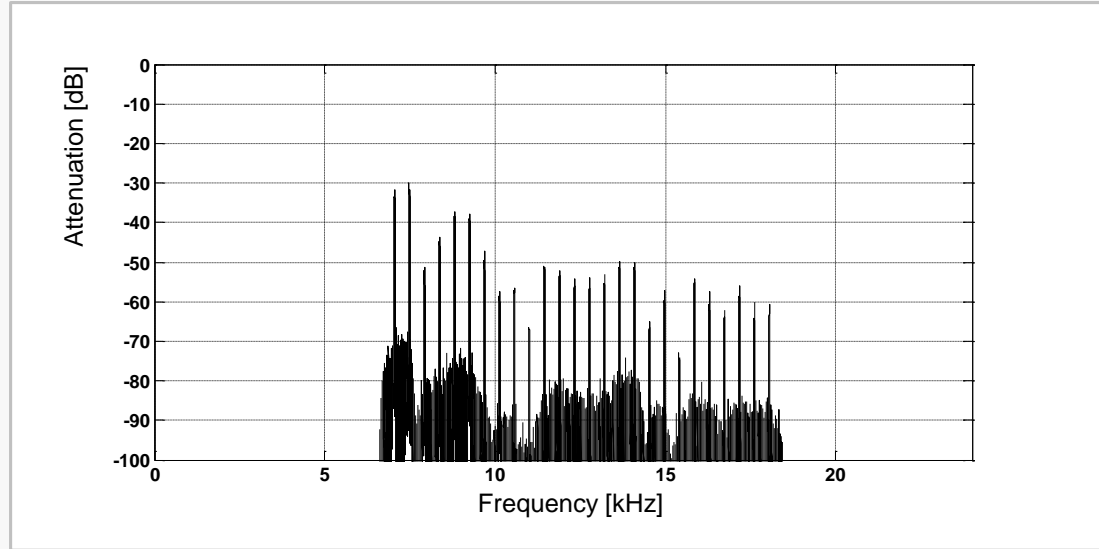
# Audio Decoder with SBR



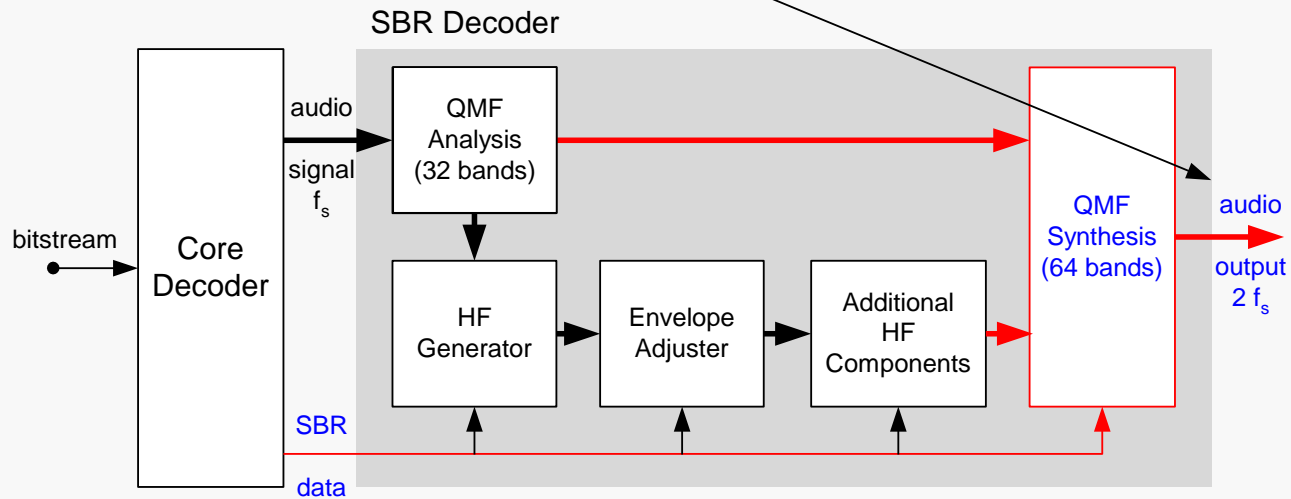
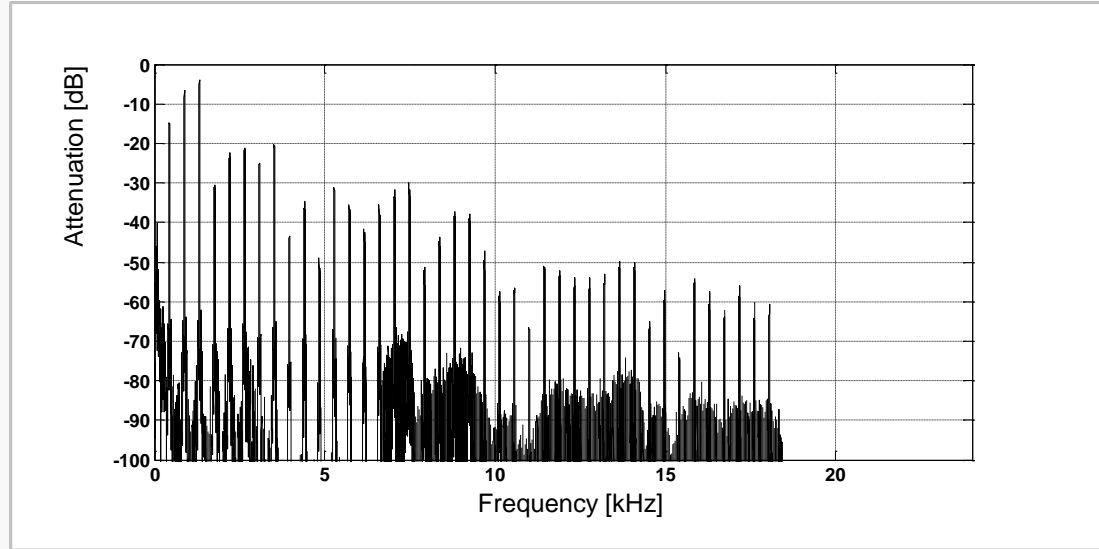
# Audio Decoder with SBR



# Audio Decoder with SBR



# Audio Decoder with SBR

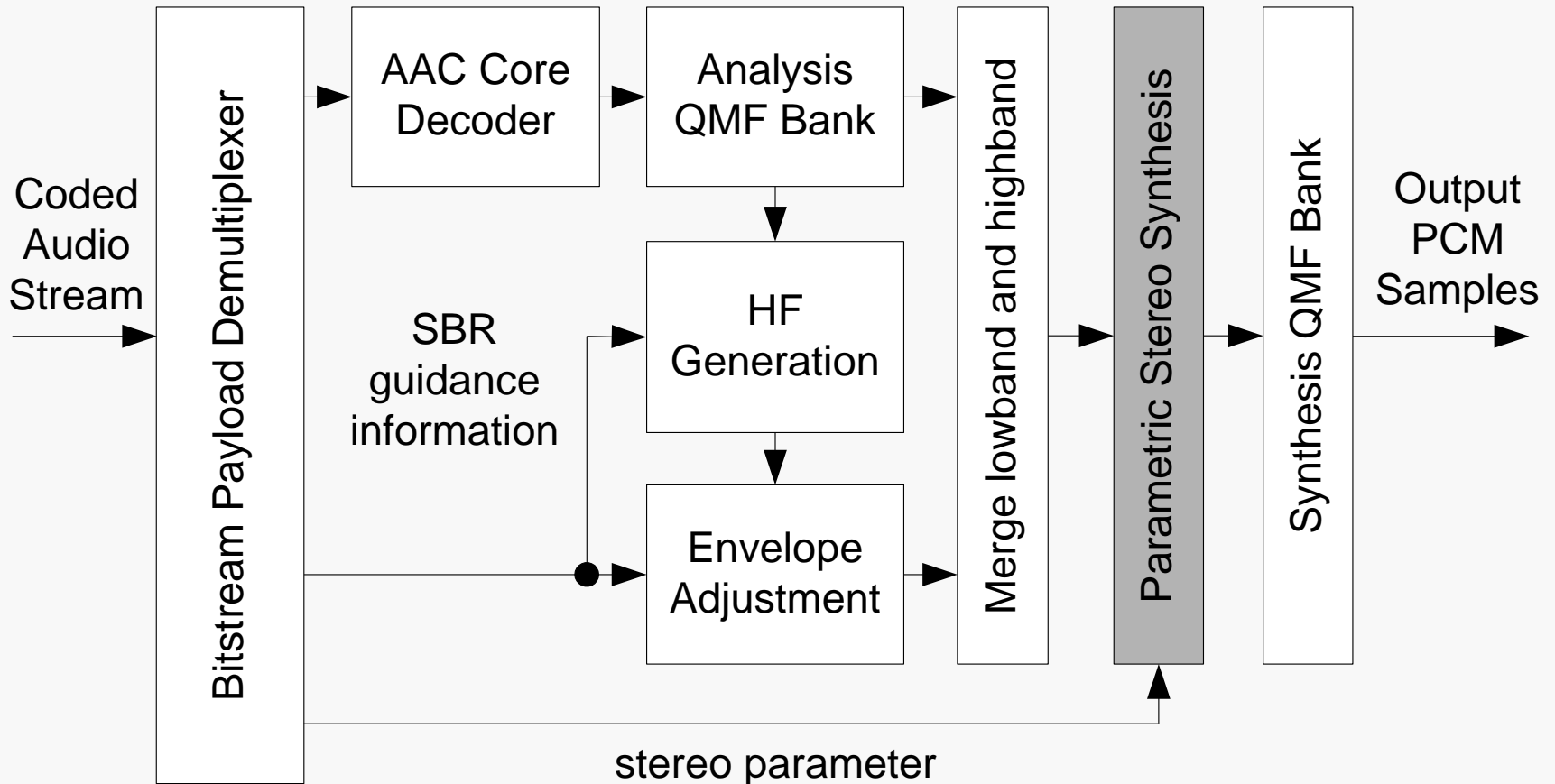


# Parametric Stereo (PS)

- Problem
  - Mono coding better than stereo at low bit rates
- Approach
  - Transmit mono signal + stereo side information
  - Reconstruct stereo signal in decoder
- Stereo Parameter (per subband and time slot)
  - Interchannel Intensity Difference (IID) -> “pan”
  - InterChannel Correlation (ICC) -> “ambience”
- Approx. 2 to 3 kbit/s stereo side information
- “Parametric coding” of spatial/stereo image




# Combining Parametric Stereo with SBR



# Combining waveform and parametric coding

- AAC + SBR + PS => HE-AAC v2
  - MPEG-4 High-Efficiency AAC v2 (ISO/IEC 14496-3, published 2004)
    - Enhanced aacPlus (3GPP Release 6, TS 26.410)

- Audio Demonstration

▪ Original PCM stereo	1.5 Mbit/s		
▪ AAC stereo	48 kbit/s	1:32	
▪ HE-AAC stereo (AAC+SBR)	48 kbit/s	1:32	
▪ HE-AAC v2 stereo (AAC+SBR+PS)	24 kbit/s	1:64	

# Recent developments

- Combination of speech and audio codecs
  - MPEG
    - USAC: Unified Speech and Audio Coding (ISO/IEC 23003-3)
  - IETF
    - Opus (RFC6716)
  - 3GPP/LTE
    - EVS: Enhanced Voice Services (3GPP Release 12, TS 26.445)
- Immersive sound
  - Dolby Atmos
  - NHK 22.2
  - MPEG-H 3D Audio

Thank you!