EE E6820: Speech & Audio Processing & Recognition

Lecture 7: Audio Compression & Coding

1. **Information, compression & quantization**
2. **Speech coding**
3. **Wide bandwidth audio coding**

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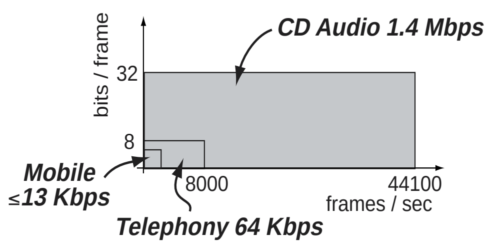
1 Compression & Quantization

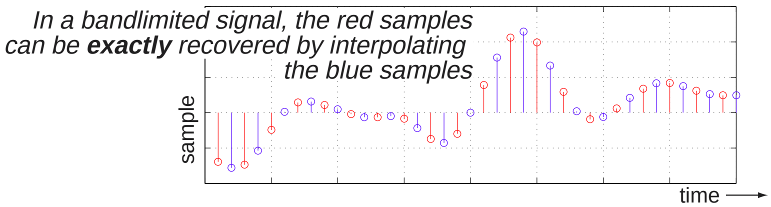
- **How big is audio data? What is the bitrate?**
  - F_S frames/second (e.g. 8000 or 44100)
    x C samples/frame (e.g. 1 or 2 channels)
    x B bits/sample (e.g. 8 or 16)
  - $\rightarrow F_S \cdot C \cdot B$ bits/second (e.g. 64 Kbps or 1.4 Mbps)



- **How to reduce?**
  - lower sampling rate → less bandwidth (muffled)
  - lower channel count → no stereo image
  - lower sample size → quantization noise
- **Or: use data compression**

Data compression: Redundancy vs. Irrelevance

- **Two main principles in compression:**
  - remove redundant information
  - remove irrelevant information
- **Redundant info is implicit in remainder**
  - e.g. signal bandlimited to 20kHz, but sample at 80kHz
  - →can recover every other sample by interpolation:



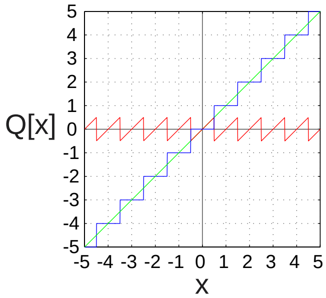
- **Irrelevant info is unique but unnecessary**
  - e.g. recording a microphone signal at 80 kHz sampling rate

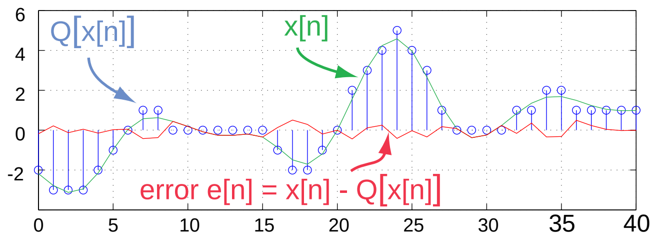
Irrelevant data in audio coding

- **For coding of audio signals, irrelevant means perceptually insignificant**
  - an empirical property
- **Compact Disc standard is adequate:**
  - 44 kHz sampling for 20 kHz bandwidth
  - 16 bit linear samples for ~ 96 dB peak SNR
- **Reflect limits of human sensitivity:**
  - 20 kHz bandwidth, 100 dB intensity
  - sinusoid phase, detail of noise structure
  - dynamic properties - hard to characterize
- **Problem: separating salient & irrelevant**

Quantization

- **Represent waveform with discrete levels**

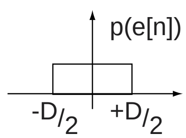




- **Equivalent to adding error e[n]:**

  $$x[n] = Q[x[n]] + e[n]$$

- **e[n] ~ uncorrelated, uniform white noise**



$\rightarrow$ variance $\sigma_e^2 = \dfrac{D^2}{12}$

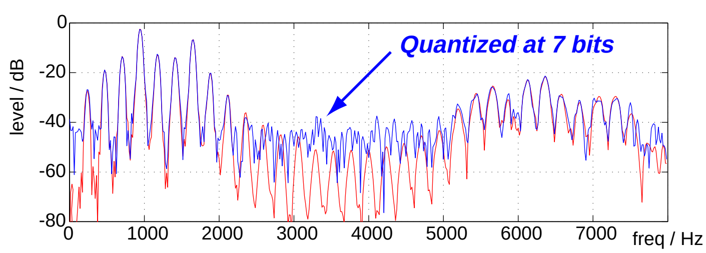
Quantization noise (Q-noise)

- **Uncorrelated noise has flat spectrum**
- **With a *B* bit word and a quantization step *D***
  - max signal range (x) = $-(2^{B-1})\cdot D$ .. $(2^{B-1}-1)\cdot D$
  - quantization noise (e) = $-D/2$ .. $D/2$

$\rightarrow$ **Best signal-to-noise ratio (power)**

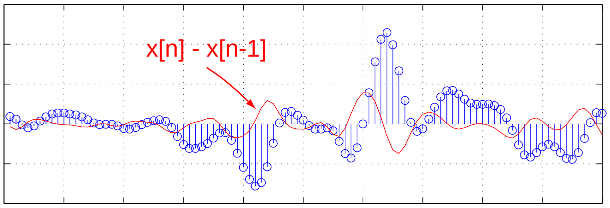
$$SNR = E[x^2]/E[e^2]$$
$$= (2^B)^2$$

.. or, in dB, $20 \cdot \log_{10} 2 \cdot B \approx 6 \cdot B$ **dB**



Redundant information

- **Redundancy removal is lossless**
- **Signal correlation implies redundant information**
  - e.g. if $x[n] = x[n-1] + v[n]$
    $x[n]$ has a greater amplitude range → more bits than $v[n]$
  - sending $v[n] = x[n] - x[n-1]$ can reduce amplitude, hence bitrate



  - 'white noise' sequence has no redundancy
- **Problem: separating unique & redundant**

Optimal coding

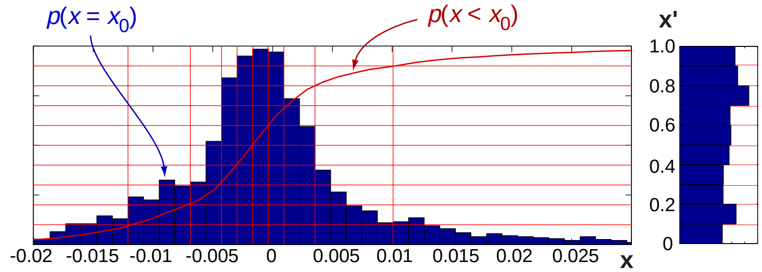
- **Shannon information:**
  An unlikely occurrence is more 'informative'

p(A) = 0.5 p(B) = 0.5	p(A) = 0.9 p(B) = 0.1
ABBBBAAABBABBABBABB	**AAAAABBAAAAAABAAAAB**
A, **B** equiprobable	**A** is expected; **B** is 'big news'

- **Information in bits** $I = -\log_2(probability)$
  - clearly works when all possibilities equiprobable
- **Opt. bitrate → av.token length = entropy** $H = \mathrm{E}[I]$
  - .. equal-length tokens are equally likely
- **How to achieve this?**
  - transform signal to have uniform pdf
  - nonuniform quantization for equiprobable tokens
  - variable-length tokens → Huffman coding

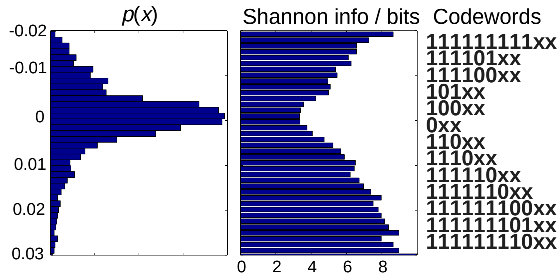
Quantization for optimum bitrate

- **Quantization should reflect pdf of signal:**



  - cumulative pdf $p(x < x_0)$ maps to uniform x'

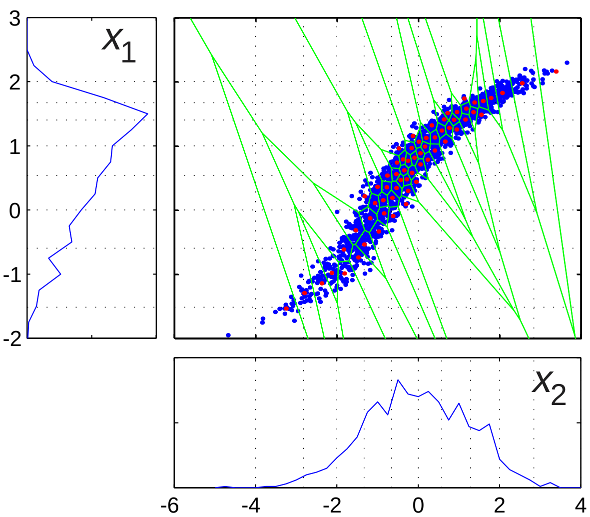
- **Or, codeword length per Shannon** $-\log_2(p(x))$**:**



  - Huffman coding: tree-structured decoder

Vector Quantization

- **Quantize mutually dependent values in joint space:**



- **May help even if values are largely independent**
  - larger space $\{x_1,x_2\}$ is easier for Huffman

Compression & Representation

- **As always, success depends on representation**
- **Appropriate domain may be 'naturally' bandlimited**
  - e.g. vocal-tract-shape coefficients
  - can reduce sampling rate without data loss
- **In right domain, irrelevance is easier to 'get at'**
  - e.g. STFT to separate magnitude and phase

Aside: Coding standards

- **Coding only useful if recipient knows the code!**
- **Standardization efforts are important**
- **Federal Standards: Low bit-rate secure voice:**
  - FS1015e: LPC-10 2.4 Kbps
  - FS1016: 4.8 Kbps CELP
- **ITU G.series**
  - G.726 ADPCM
  - G.729 Low delay CELP
- **MPEG**
  - MPEG-Audio layers 1,2,3
  - MPEG 2 Advanced Audio Codec
  - MPEG 4 Synthetic-Natural Hybrid Codec
- **More recent proprietary 'standards'**
  - RA, WMA, Skype ...

Outline

1. Information, compression & Quantization
2. **Speech coding**
   - General principles
   - CELP & friends
3. Wide bandwidth audio coding

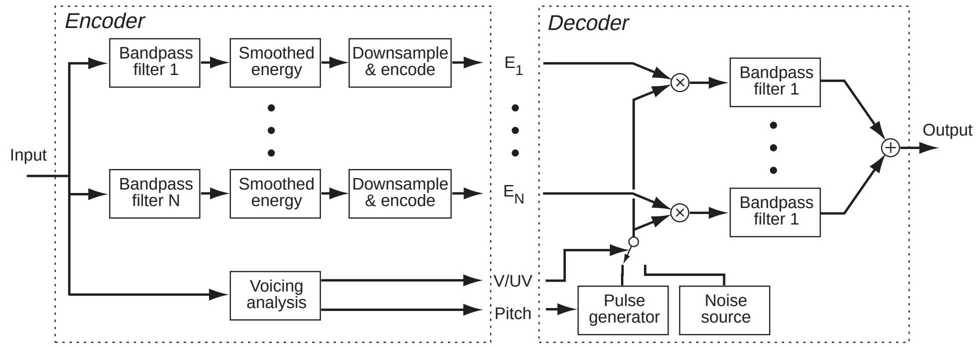
2 Speech coding

- **Standard voice channel:**
  - analog: 4 kHz slot (~ 40 dB SNR)
  - digital: 64 Kbps = 8 bit μ-law x 8 kHz
- **How to compress?**
  Redundant
  - signal assumed to be a single voice, not any possible waveform

  Irrelevant
  - need code only enough for intelligibility, speaker identification (c/w analog channel)
- **Specifically, source-filter decomposition**
  - vocal tract & fund. frequency change slowly
- **Applications:**
  - live communications - offline storage

Channel Vocoder (1940s-1960s)

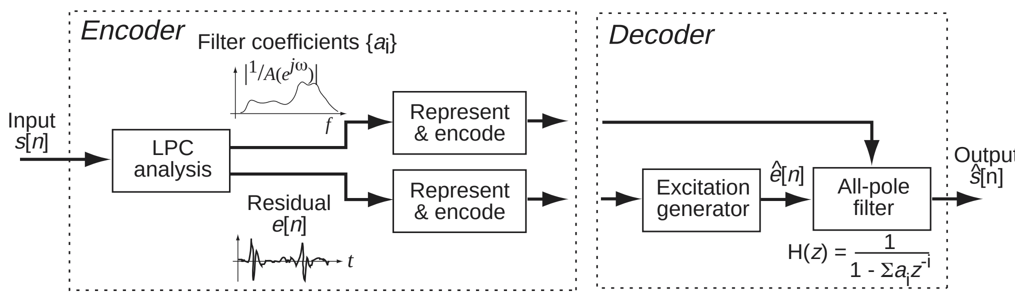
- **Basic source-filter decomposition**
  - filterbank breaks into spectral bands
  - transmit slowly-changing energy in each band



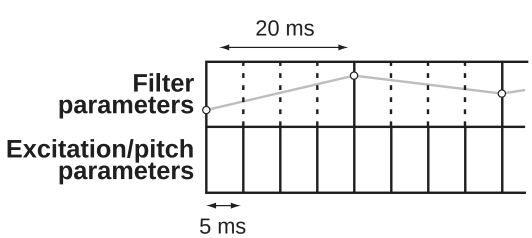
  - 10-20 bands, perceptually spaced
- **Downsampling?**
- **Excitation?**
  - pitch / noise model
  - or: baseband + 'flattening'...

LPC encoding

- **The classic source-filter model**

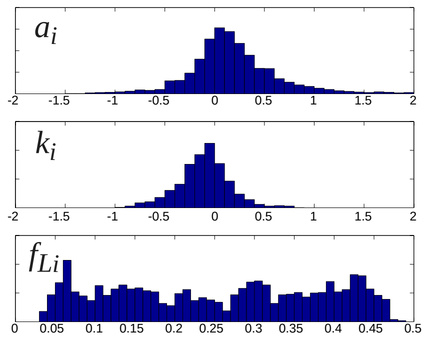


- **Compression gains:**
  - filter parameters are ~slowly changing
  - excitation can be represented many ways



Encoding LPC filter parameters

- **For 'communications quality':**
  - 8 kHz sampling (4 kHz bandwidth)
  - ~10th order LPC (up to 5 pole pairs)
  - update every 20-30 ms → 300 - 500 param/s
- **Representation & quantization**
  - $\{a_i\}$ - poor distribution, can't interpolate
  - reflection coefficients $\{k_i\}$: guaranteed stable
  - LSPs - lovely!



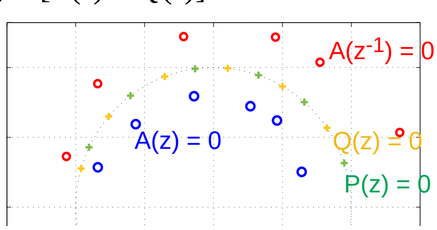
- **Bit allocation (filter):**
  - GSM (13 kbps):
    8 LARs x 3-6 bits / 20 ms = 1.8 Kbps
  - FS1016 (4.8 kbps):
    10 LSPs x 3-4 bits / 30 ms = 1.1 Kbps

Line Spectral Pairs (LSPs)

- **LSPs encode LPC filter by a set of frequencies**
- **Excellent for quantization & interpolation**
- **Definition: zeros of**

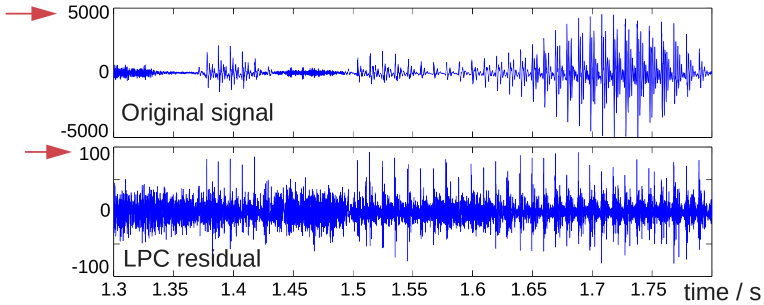
$$P(z) = A(z) + z^{-p-1} \cdot A(z^{-1})$$
$$Q(z) = A(z) - z^{-p-1} \cdot A(z^{-1})$$

  - $z = e^{j\omega} \rightarrow z^{-1} = e^{-j\omega} \rightarrow |A(z)| = |A(z^{-1})|$ on u.circ.
  - $P(z)$, $Q(z)$ have (interleaved) zeros when $\text{angle}\{A(z)\} = \pm\,\text{angle}\{z^{-p-1}A(z^{-1})\}$
  - reconstruct $P(z)$, $Q(z) = \Pi_i(1 - \zeta_i z^{-1})$ etc.
  - $A(z) = [P(z) + Q(z)]/2$



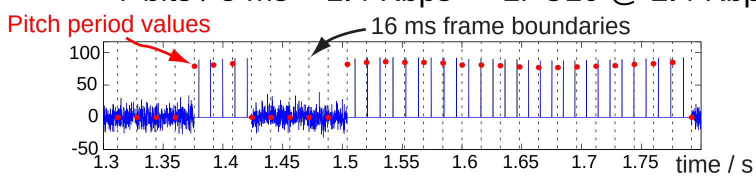
Excitation

- **Excitation already better than raw signal:**



  - save several bits/sample, still > 32 Kbps

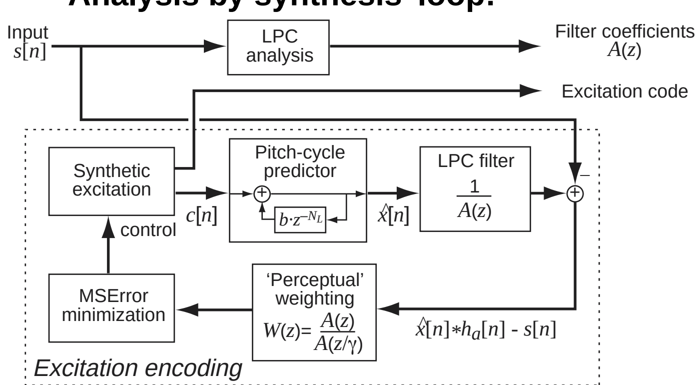
- **Crude model: U/V flag + pitch period**
  - ~ 7 bits / 5 ms = 1.4 Kbps → LPC10 @ 2.4 Kbps



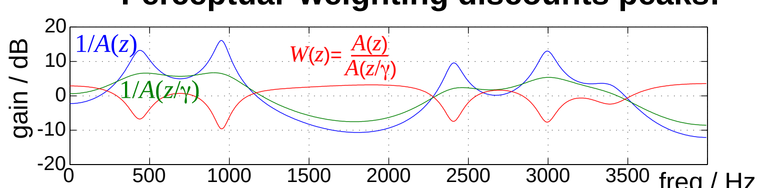
- **Band-limit then re-extend (RELP)**

Encoding excitation

- **Something between full-quality residual (32 Kbps) and pitch parameters (1.4 kbps)?**
- **'Analysis by synthesis' loop:**

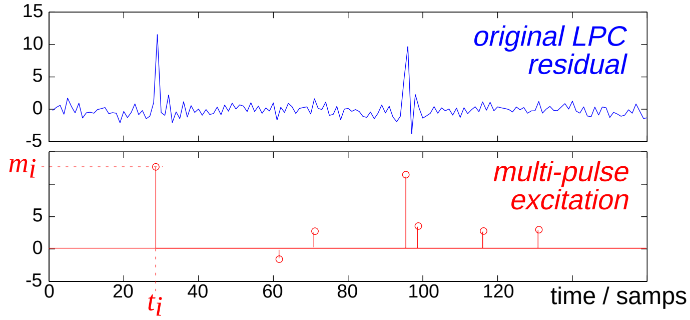


- **'Perceptual' weighting discounts peaks:**



Multi-Pulse Excitation (MPE-LPC)

- **Stylize excitation as *N* discrete pulses**



  - encode as N x (t_i, m_i) pairs

- **Greedy algorithm places one pulse at a time:**

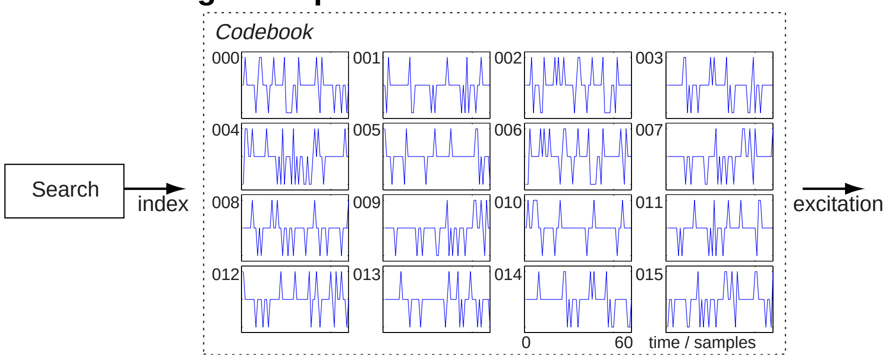
$$E_{pcp} = \frac{A(z)}{A(z/\gamma)}\left[\frac{X(z)}{A(z)} - S(z)\right]$$

$$= \frac{X(z)}{A(z/\gamma)} - \frac{R(z)}{A(z/\gamma)}$$

  - cross-correlate h_γ and $r{*}h_\gamma$, iterate

CELP

- **Represent excitation with codebook e.g. 512 sparse excitation vectors**



  - linear search for minimum weighted error?

- **FS1016 4.8 Kbps CELP (30ms frame = 144 bits):**

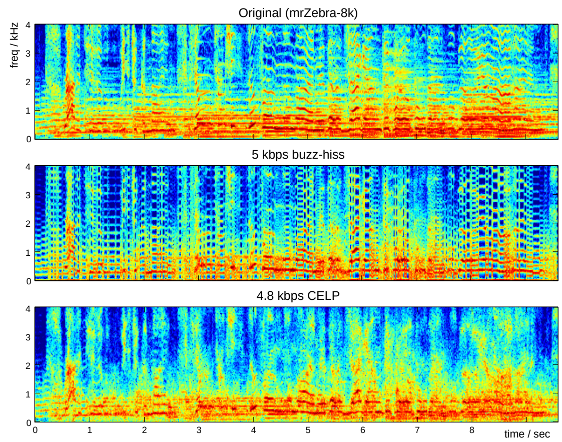
10 LSPs	4x4 + 6x3 bits =	34 bits
Pitch delay	4 x 7 bits =	28 bits
Pitch gain	4 x 5 bits =	20 bits
Codebk index	4 x 9 bits =	36 bits
Codebk gain	4 x 5 bits =	20 bits

138 bits

Aside: CELP for nonspeech?

- **CELP is sometimes called a 'hybrid' coder:**
  - originally based on source-filter voice *model*
  - CELP residual is *waveform* coding (no model)

- **CELP does not *break* with multiple voices etc.**
  - just does the best it can



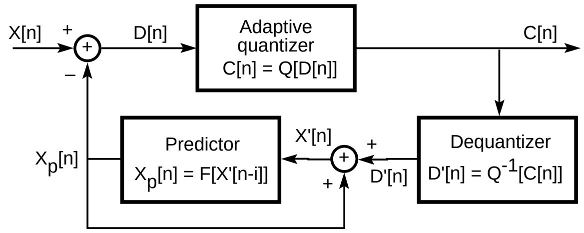
- **LPC filter models vocal tract; also matches auditory system?**
  - i.e. the 'source-filter' separation is good for relevance as well as redundancy?

Outline

1. Information, compression & Quantization
2. Speech coding
3. **Wide bandwidth audio coding**
   - General principles
   - MPEG-Audio

3 Wide-Bandwidth Audio Coding

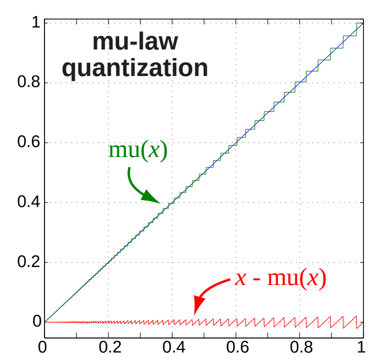
- **Goals:**
  - transparent coding i.e. no perceptible effect
  - general purpose - handles any signal
- **Simple approaches (redundancy removal)**
  - Adaptive Differential PCM (ADPCM)



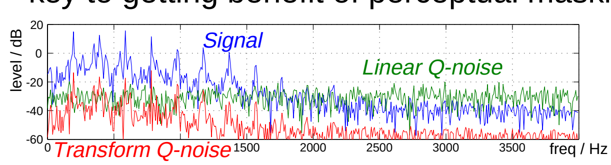
  - as prediction gets smarter, becomes LPC e.g. `shorten` - lossless LPC encoding
- **Larger compression gains needs irrelevance**
  - hide quantization noise with psychoacoustic masking

Noise shaping

- **Plain Q-noise sounds like added white noise**
  - actually, not all that disturbing
  - .. but worst-case for exploiting masking
- **Have Q-noise scale with signal level**
  - i.e. quantizer step gets larger with amplitude
  - minimum distortion for some center-heavy pdf

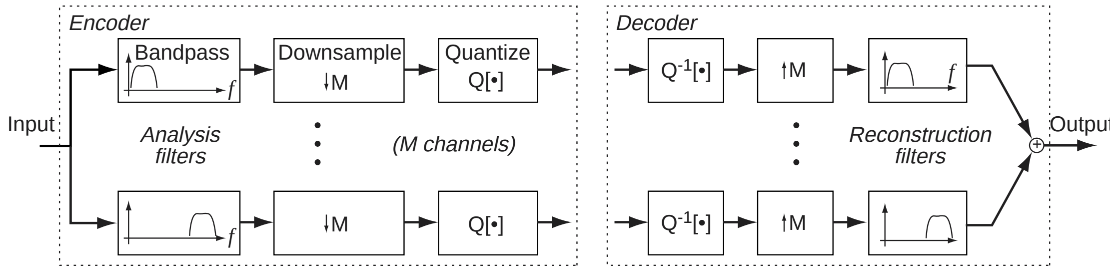


- **Or: put Q-noise around peaks in spectrum**
  - key to getting benefit of perceptual masking



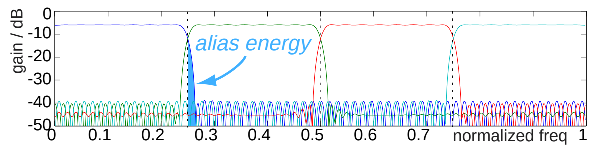
Subband coding

- **Idea: Quantize separately in separate bands**



  - Q-noise stays within band, gets masked

- **'Critical sampling' → 1/M of spectrum per band**

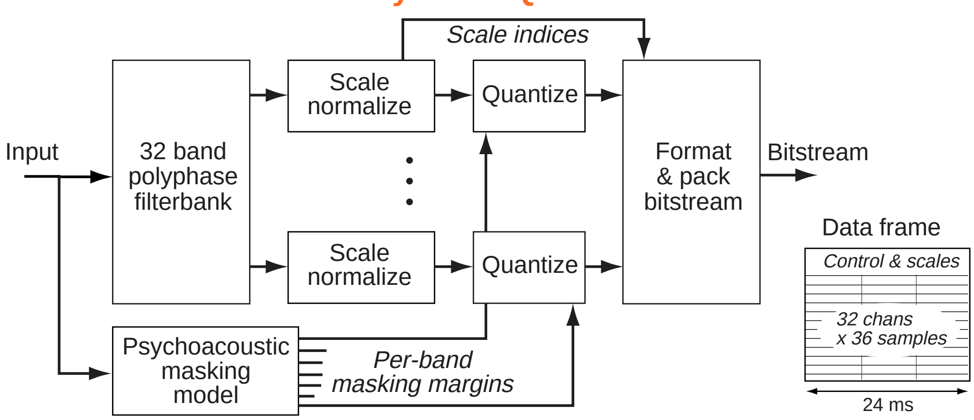


  - some aliasing inevitable

- **Trick is to cancel with alias of adjacent band**
  →'quadrature-mirror' filters

MPEG-Audio

- **Basic idea: Subband coding plus psychoacoustic masking model to choose dynamic Q-levels in subbands**

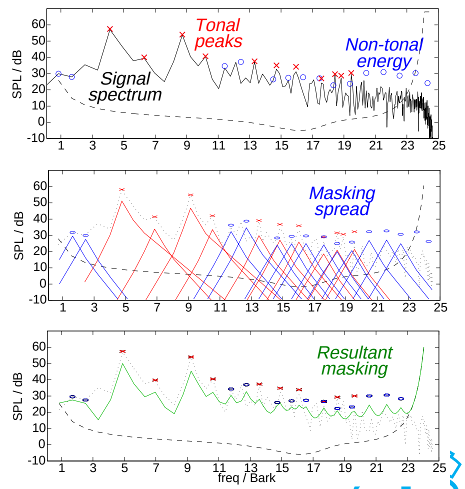


  - 22 kHz ÷ 32 equal bands = 690 Hz bandwidth
  - 8 / 24 ms frames = 12 / 36 subband samples
  - fixed bitrates 32 - 256 Kbps/chan (1-6 bits/samp)
  - scale factors like LPC envelope?

MPEG Psychoacoustic model

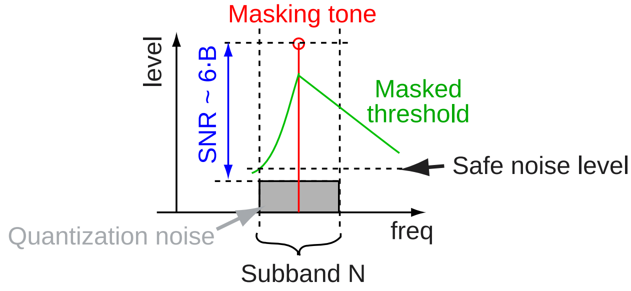
- **Based on simultaneous masking experiments**
- **Difficulties:**
  - noise energy masks ~10 dB better than tones
  - masking level nonlinear in frequency & intensity
  - complex, dynamic sounds not well understood

- **Procedure**
  - pick 'tonal peaks' in NB FFT spectrum
  - remaining energy → 'noisy' peaks
  - apply nonlinear 'spreading function'
  - sum all masking & threshold in power domain



MPEG Bit allocation

- **Result of psychoacoustic model is maximum tolerable noise per subband**



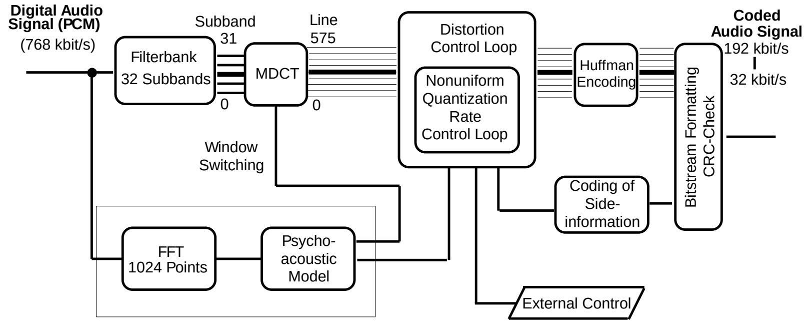
  - safe noise level → required SNR → bits B

- **Bit allocation procedure (fixed bit rate):**
  - pick channel with worst noise-masker ratio
  - improve its quantization by one step
  - repeat while more bits available for this frame

- **Bands with no signal above masking curve can be skipped entirely**

MPEG Audio Layer III

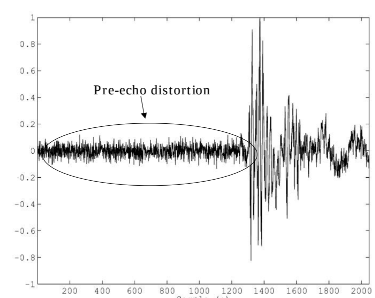
- **'Transform coder' on top of 'subband coder'**



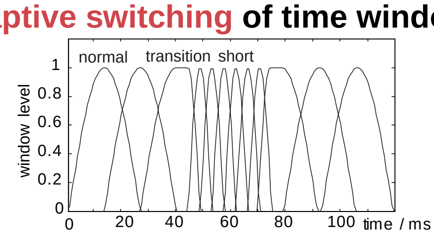
- **Blocks of 36 subband time-domain samples become 18 pairs of frequency-domain samples**
  - more redundancy in spectral domain
  - finer control e.g. of aliasing, masking
  - scale factors now in band-blocks
- **Fixed Huffman tables optimized for audio data**
- **Power-law quantizer**

Adaptive time window

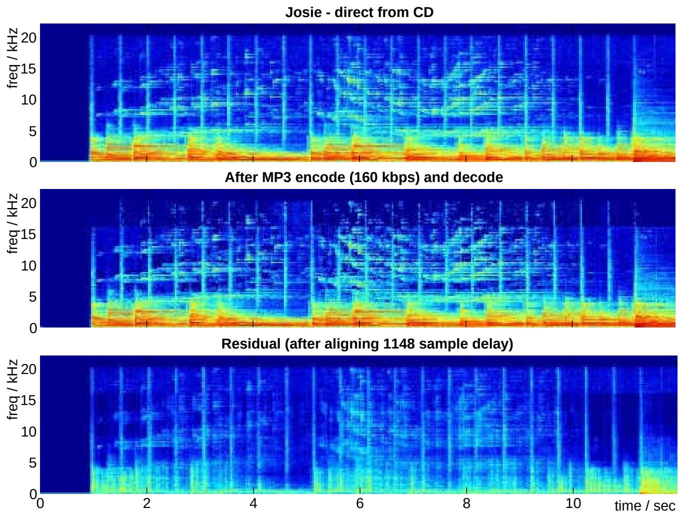
- **Time window relies on temporal masking**
  - single quantization level over 8-24 ms window
- **'Nightmare' scenario:**



  - 'backward masking' saves in most cases
- **Adaptive switching of time window:**



The effects of MP3



- chop off high frequency (above 16 kHz)
- occasional other time-frequency 'holes'
- quantization noise under signal

MP3 & Beyond

- **MP3 is 'transparent' at ~ 128 Kbps for stereo (11x smaller than 1.4 Mbps CD rate)**
  - only decoder is standardized:
    better psych. models → better encoders
- **MPEG2 AAC**
  - rebuild of MP3 without backwards compatibility
  - 30% better (stereo at 96 Kbps?)
  - multichannel etc.
- **MPEG4-Audio**
  - wide range of component encodings
  - MPEG Audio, LSPs, ...
- **SAOL**
  - 'synthetic' component of MPEG-4 Audio
  - complete DSP/computer music language!
  - how to encode into it?

Summary

- **For coding, every bit counts**
  - take care over quantization domain & effects
  - Shannon limits...
- **Speech coding**
  - LPC modeling is old but good
  - CELP residual modeling can go beyond speech
- **Wide-band coding**
  - noise shaping ‘hides’ quantization noise
  - detailed psychoacoustic models are key