

**EECS 225D**

# **Audio Signal Processing in Humans and Machines**

## **Lecture 16 – Perceptual Audio Coding**

**2012-3-14**

**Professor Nelson Morgan**  
**today's lecture by John Lazzaro**

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**[www.icsi.berkeley.edu/eecs225d/spr12/](http://www.icsi.berkeley.edu/eecs225d/spr12/)**

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**“Hero” sound**

**Play**



# Today's lecture: Audio Coding

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- \* **Compression: Lossless and Lossy**
- \* **Quantization and Noise**
- \* **Psychoacoustic Masking**
- \* **Time-Frequency Tradeoffs**
- \* **Research Topics**



# OS X System Sound: Hero.aiff

The image shows a screenshot of the OS X Finder application. The window title is "Searching 'Sounds'" and the search term is "Hero". The sidebar on the left shows "FAVORITES" with links to Applications, Utilities, keyboard, and lazzaro. The main pane displays a file named "Hero.aiff" with a music note icon. The file is located in the "Sounds" folder under "Earlier". The status bar at the bottom shows the file path: Macintosh HD > System > Library > Sounds > Hero.aiff, and it indicates "1 of 1 selected".  
  
A large blue annotation text "1 second of 44.1 kHz, 16-bit, stereo audio" is overlaid on the left side of the window. A blue arrow points from this text to the "More Info" section of the file's properties window.  
  
The properties window for "Hero.aiff" is open on the right. It shows the following details:

- Hero.aiff** 186 KB
- Modified: Friday, July 8, 2011 1:36 AM
- General:**
  - Kind: AIFF-C audio
  - Size: 186,450 bytes (123 KB on disk)
  - Where: /System/Library/Sounds
  - Created: Friday, June 24, 2011 10:12 PM
  - Modified: Friday, July 8, 2011 1:36 AM
  - Label:
  - Stationery pad
  - Locked
- More Info:**
  - Duration: 00:01
  - Audio channels: 2
  - Total bit rate: 1,411,200
- Name & Extension:**
- Open with:**
- Preview:**



# How well does gzip work on audio files?

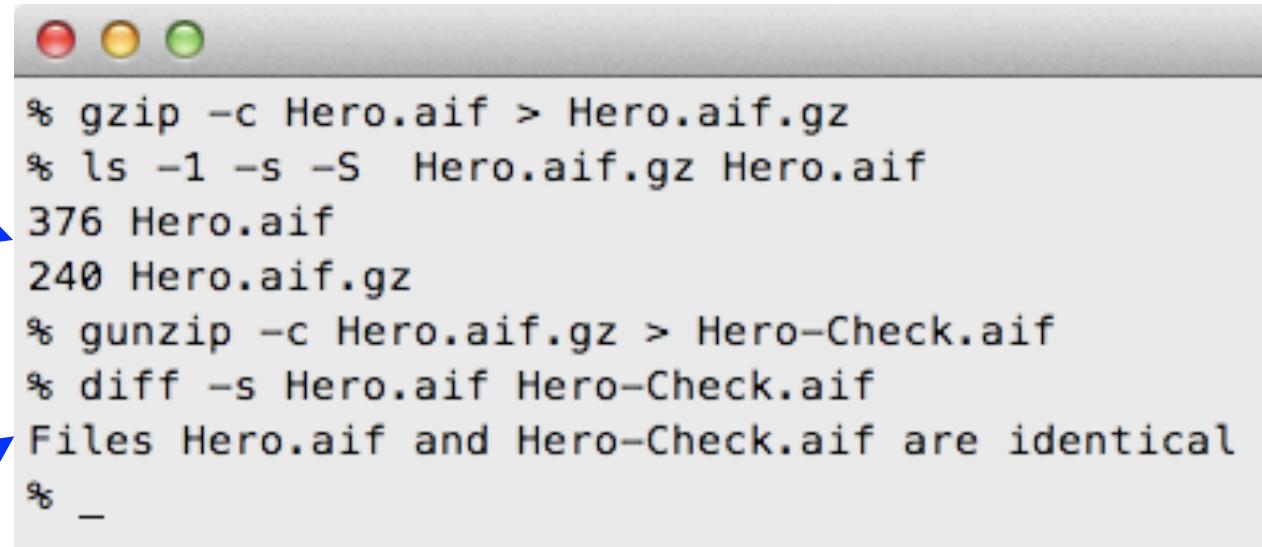
File size reduced  
by a factor  
of 1.56

(measured in units  
of 4KB disk blocks)

“Lossless”  
compression.  
(decompression  
is bit-accurate).

“Shorten”:

Tony Robinson,  
Cambridge, 1992.



```
% gzip -c Hero.aif > Hero.aif.gz
% ls -l -s -S Hero.aif.gz Hero.aif
376 Hero.aif
240 Hero.aif.gz
% gunzip -c Hero.aif.gz > Hero-Check.aif
% diff -s Hero.aif Hero-Check.aif
Files Hero.aif and Hero-Check.aif are identical
%
```

A screenshot of a Mac OS X terminal window. It shows a command-line session where a file 'Hero.aif' is compressed into 'Hero.aif.gz'. The terminal displays the original file size (376 bytes), the compressed file size (240 bytes), and then decompresses it back to 'Hero-Check.aif'. A 'diff' command is run to verify that the original and decompressed files are identical, which they are.

**Lossless algorithms remove redundant bits -- bits that are not needed to exactly reconstruct the original file.**

**Redundancy removal can be improved if the algorithm can be specialized for audio waveforms.**



# Apple Lossless (after shorten, FLAC, ...)

File size reduced  
by a factor of 3.1  
(double the performance  
of gzip on the same file)

Lossless,  
just like gzip.

```
% afconvert -f 'm4af' -d 'alac' Hero.aif Hero-Lossless.m4a
% ls -1 -s -S Hero-Lossless.m4a Hero.aif
376 Hero.aif
120 Hero-Lossless.m4a
% afconvert -f "AIFC" -d 'BEI16' Hero-Lossless.m4a
% diff -s Hero.aif Hero-Check.aif
Files Hero.aif and Hero-Check.aif are identical
%
```

To reduce file size by larger factors, we need  
to go beyond removing **redundancy**.

**One approach:** Remove information that is  
**irrelevant** information for a particular use case.

**Example:** Remove audio information whose  
loss a **human** listener cannot **perceive**.



# MPEG 4 Advanced Audio Codec (AAC)

Request 128 kb/s

Encoder adjusts  
quality to meet  
request

File size reduced  
by a factor of 9.4

```
% afconvert -f 'm4af' -d 'aac' -b 128000 Hero.aif
% ls -l -s -S Hero-AAC.m4a Hero.aif
376 Hero.aif
40 Hero-AAC.m4a
% afconvert -f "AIFC" -d 'BEI16' Hero-AAC.m4a Hero-
% diff -s Hero.aif Hero-Check.aif
Binary files Hero.aif and Hero-Check.aif differ
%
```

↑  
Lossy:

Decompression does not  
restore original file.

Today's Lecture  
How it works

Listening Test: Original: [Play](#)  
128 kb/s (9.4X): [Play](#)  
16 kb/s (23.5X): [Play](#)



# Today's lecture: Audio Coding

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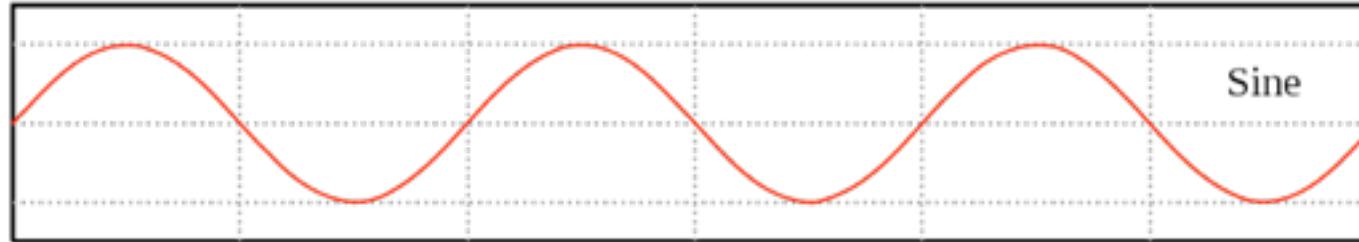
- \* Compression: Lossless and Lossy
- \* Quantization and Noise
- \* Psychoacoustic Masking
- \* Time-Frequency Tradeoffs
- \* Research Topics



# Quantization, noise, and compression ...

To compress a real-valued discrete-time waveform,  
**quantize** the samples to **reduce bits/sample**,  
and then apply lossless compression.

$s(t)$



Play

$s(t) + e(t)$



Play

Quantization corrupts the signal  $s(t)$  with noise term  $e(t)$ .

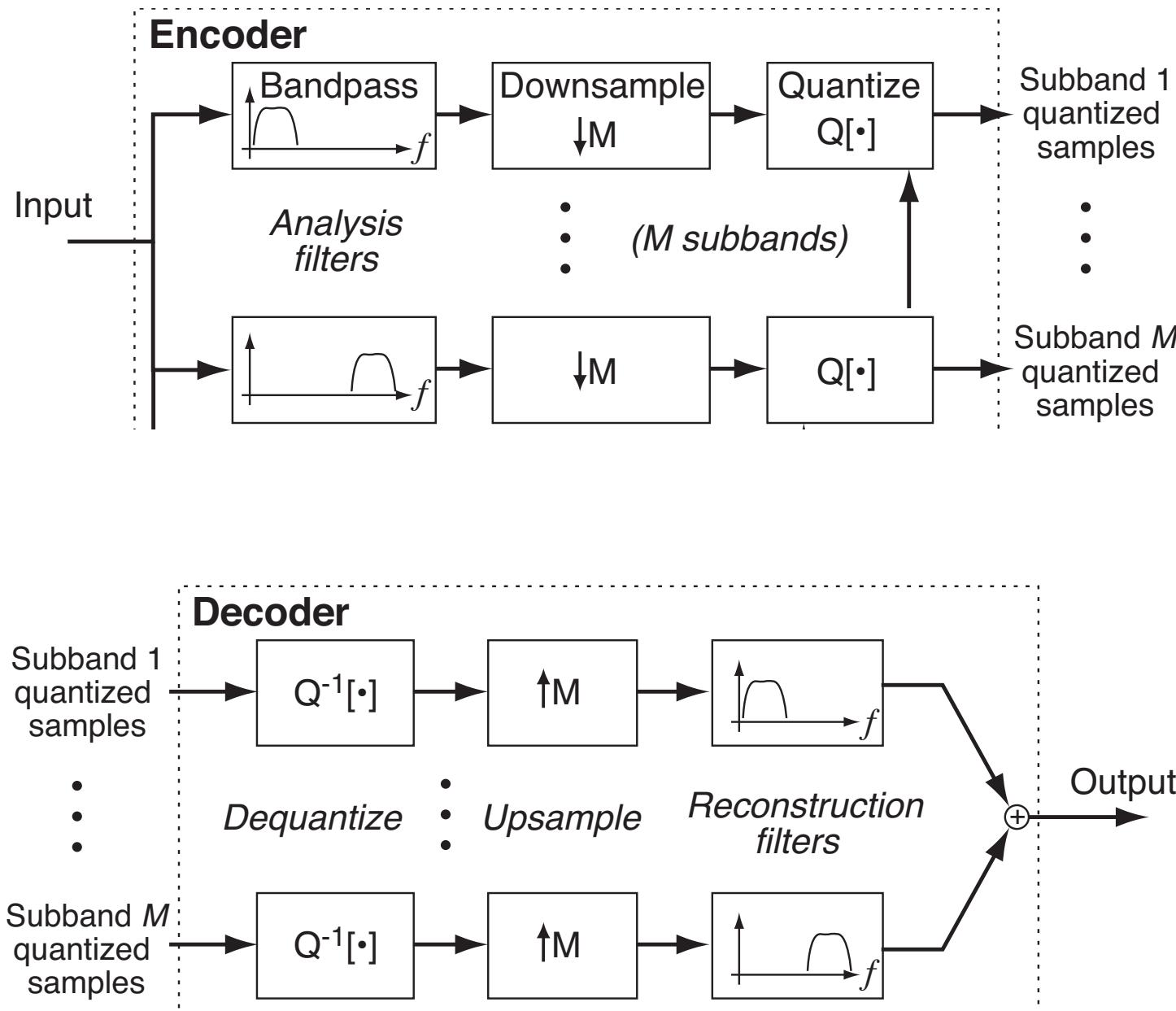
In this example, quantizing to 1 bit is clearly objectionable.  
However, a 40 dB reduction in  $e(t)$  yield a better result.

Quantizing with more bits  
acts to reduce  $e(t)$ .

$s(t) + 0.01 * e(t)$  Play

$s(t) + 0.1 * e(t)$  Play

# Which leads to this architecture ...

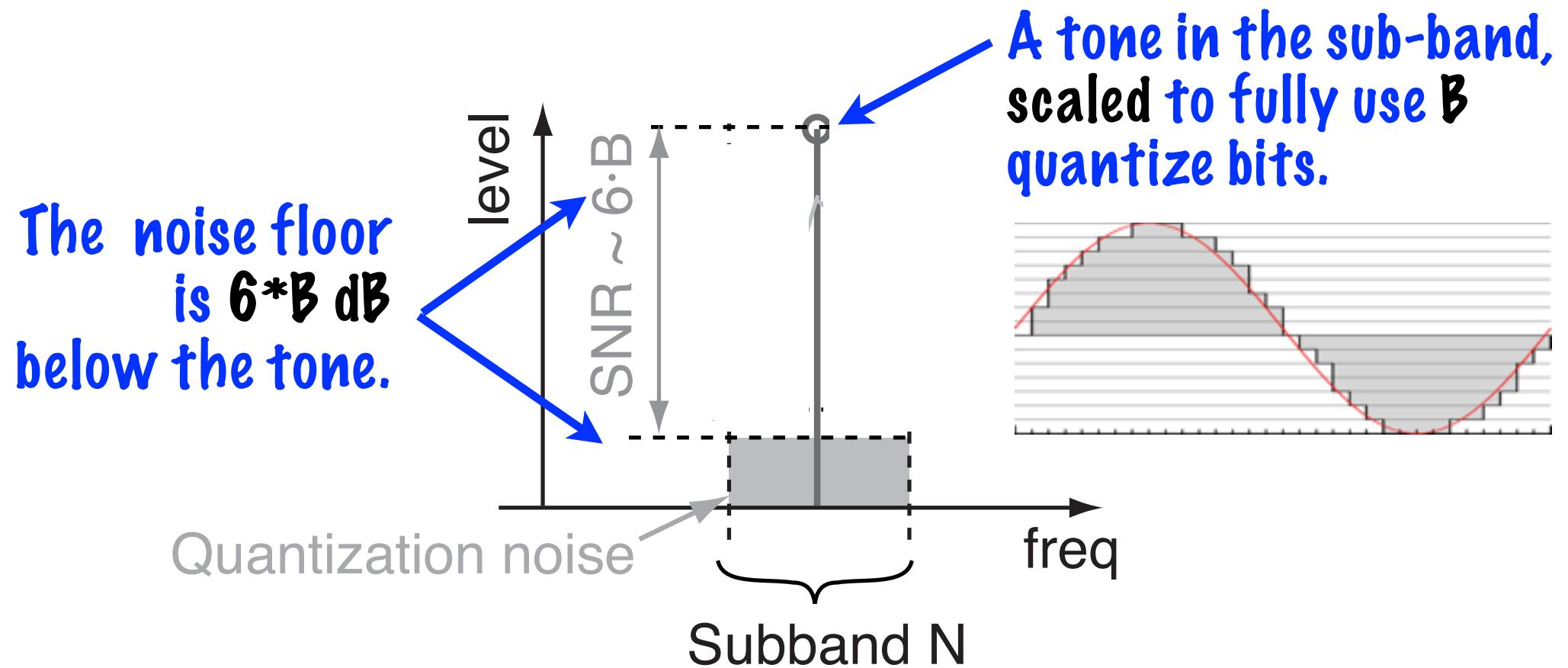
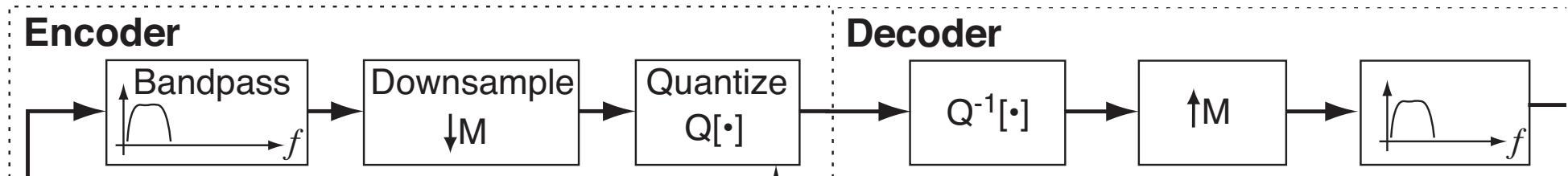


**Filter bank splits audio input into  $M$  sub-bands.**

**Quantize to minimize the number of bits needed across all  $M$  channels.**

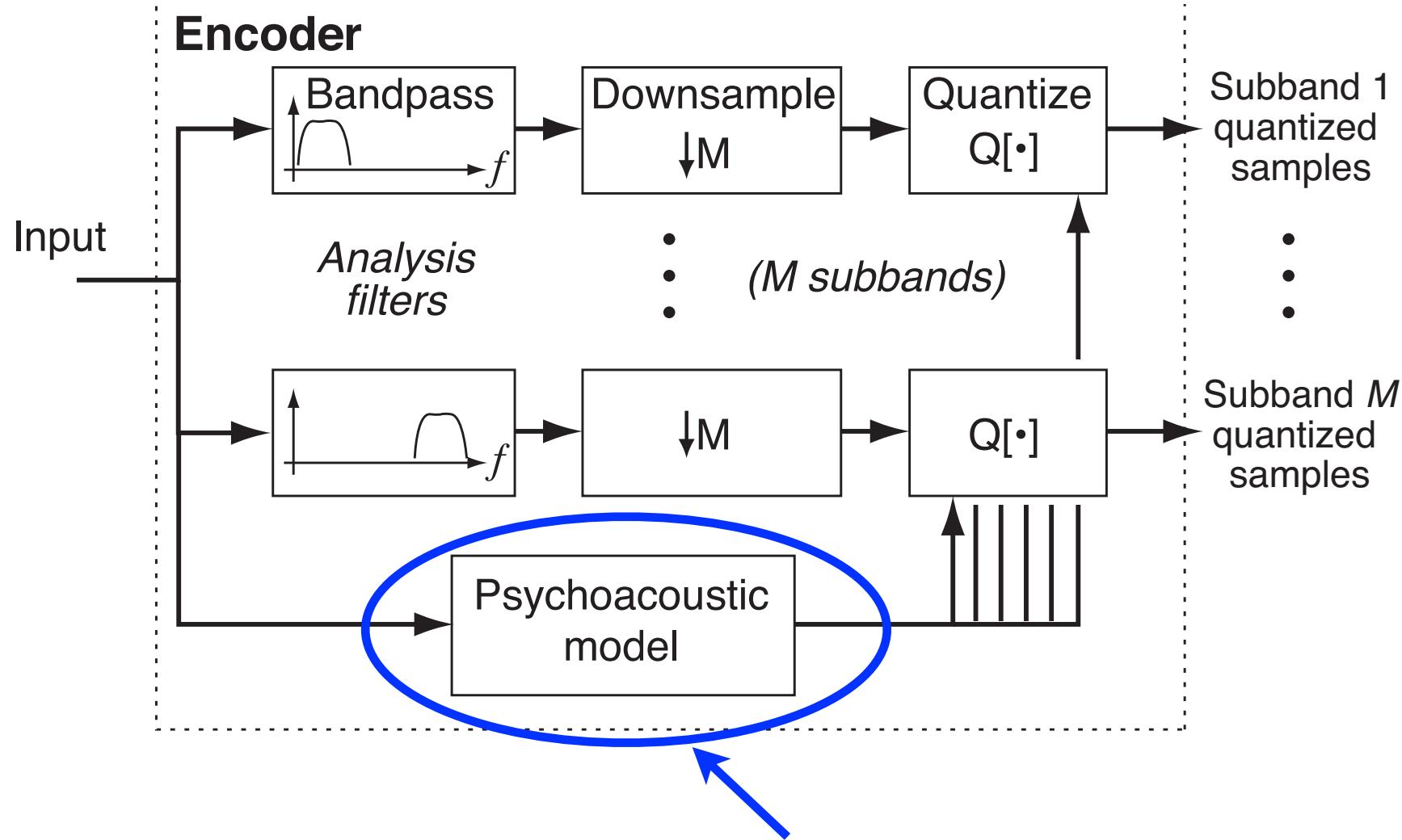
**Constraint:**  
**Human imperceptibility of the encode  $\rightarrow$  decode process.**

# Quantization noise in a sub-band ...



(Approximate result. See the book for the fine print)

# If a B is too small, noise may be audible



Encoder includes a **model** of human perception.  
**Candidate sets** of  $M$  quantizations are **tested** against the model to **check imperceptibility**.

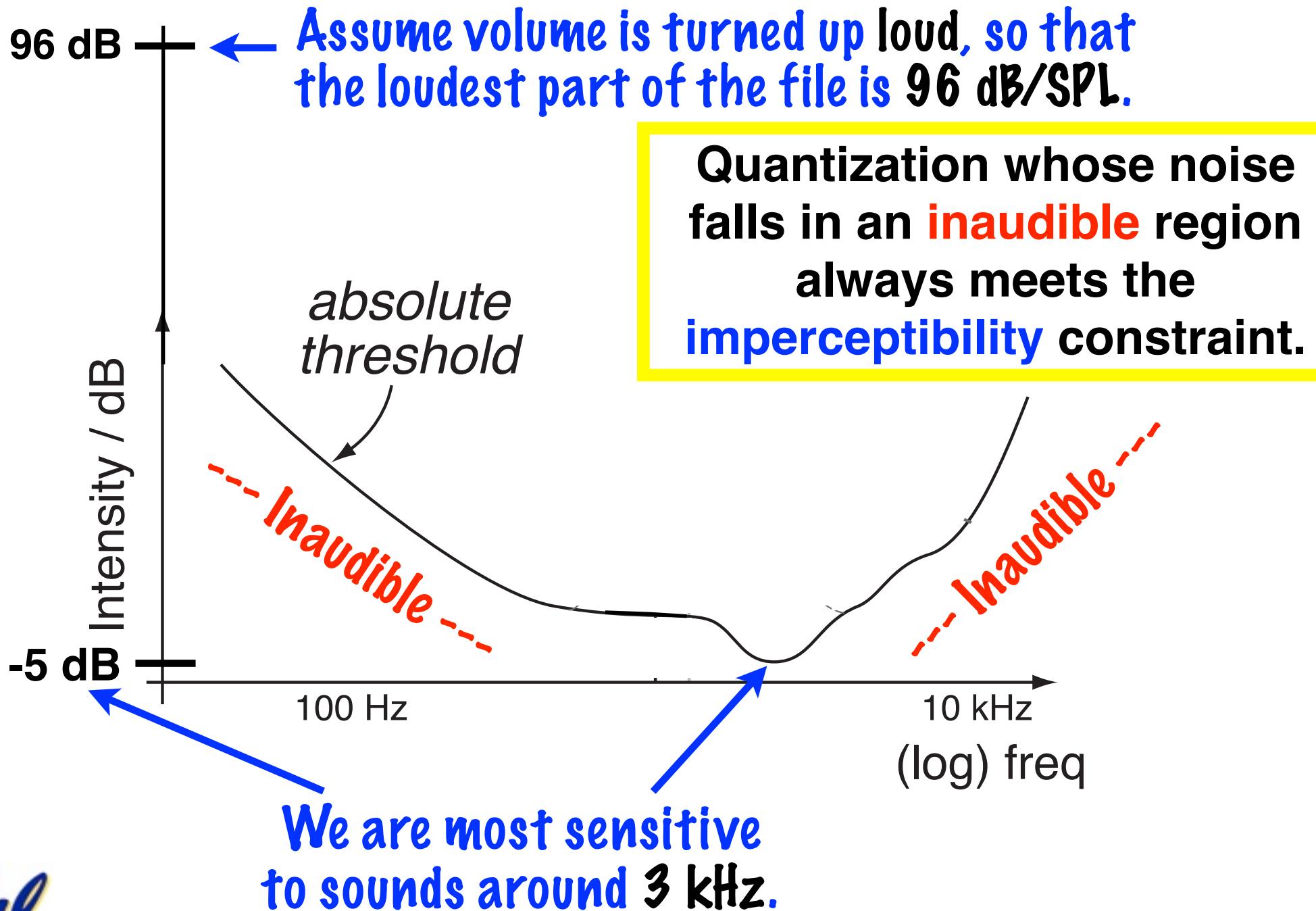
# Today's lecture: Audio Coding

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- \* Compression: Lossless and Lossy
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- \* Time-Frequency Tradeoffs
- \* Research Topics

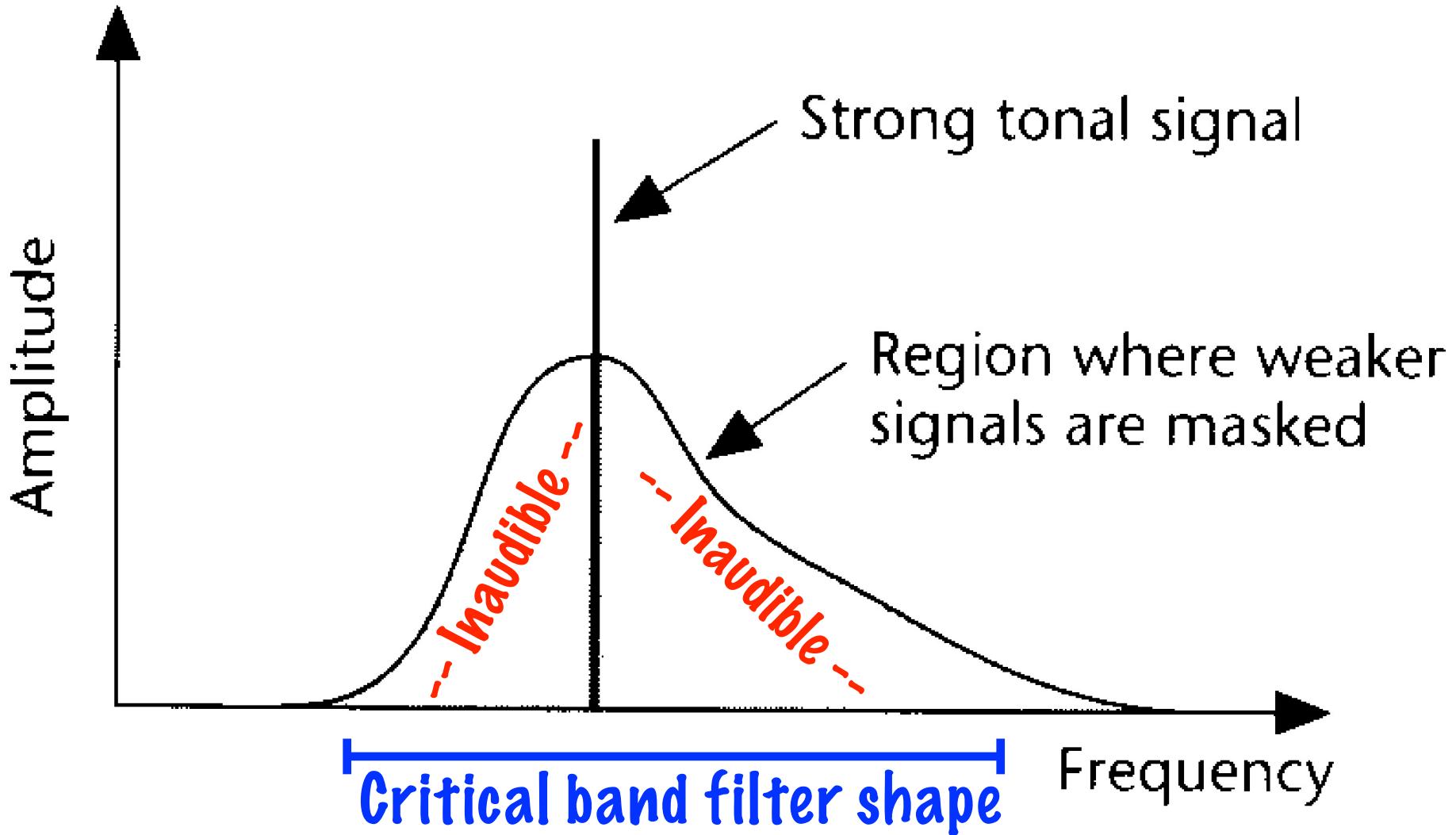


# The absolute threshold of hearing ...



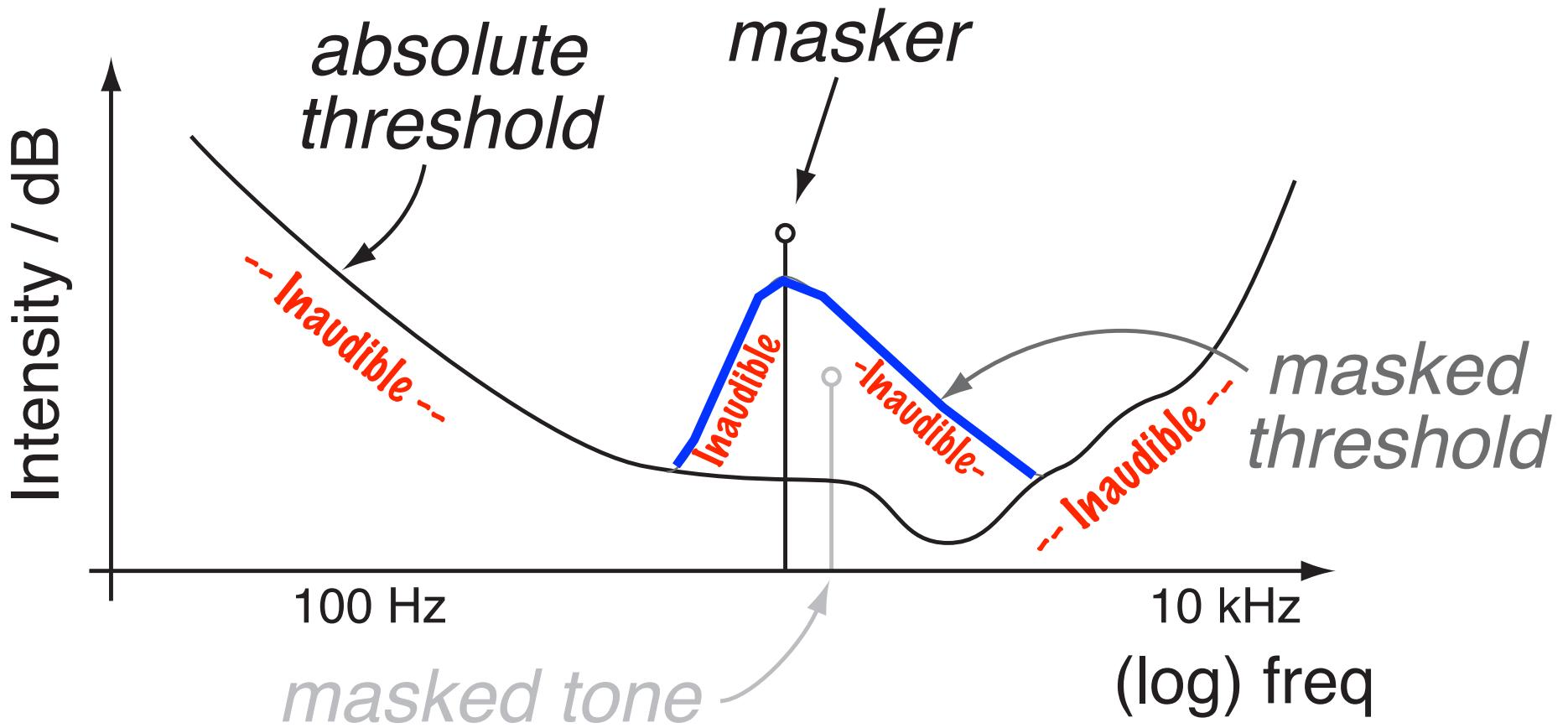
# Tonal masking ....

Quantization noise will be imperceptible if it falls in the **inaudible skirt** surrounding a tonal signal.



# Maskers compose using `max()` function

Given a short segment of **wide-band** audio, we can identify **narrow-band** maskers and compute a **composite masking function** for the audio signal.



Effectively, tonal maskers locally raise the absolute threshold.

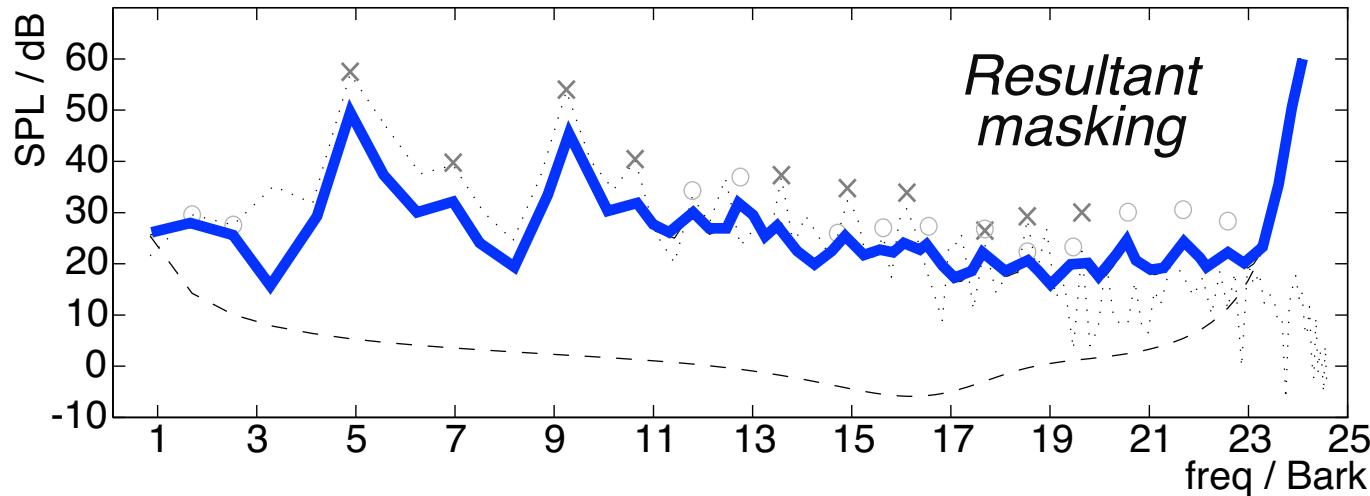
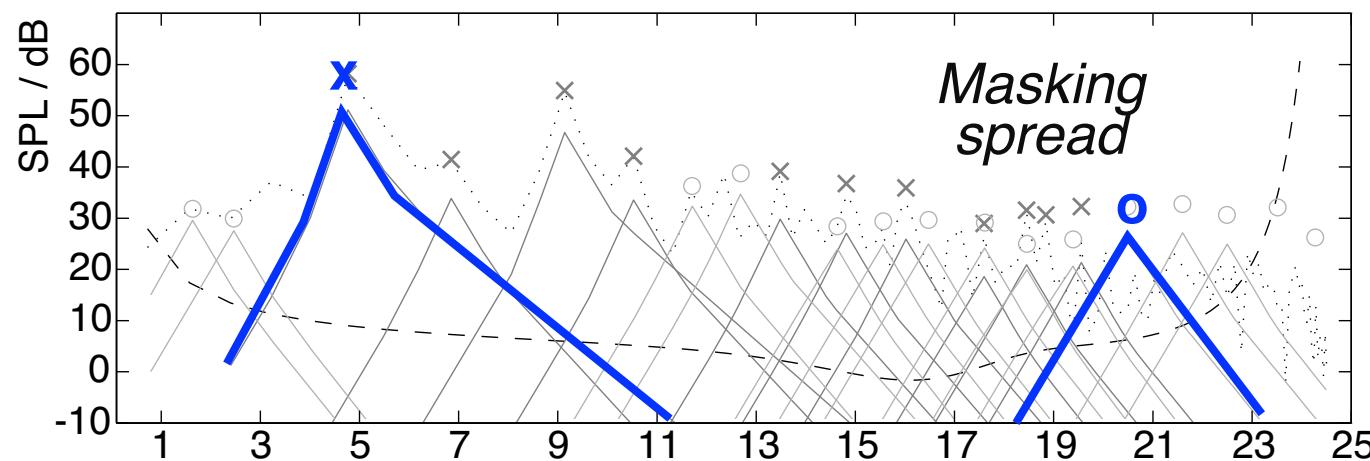
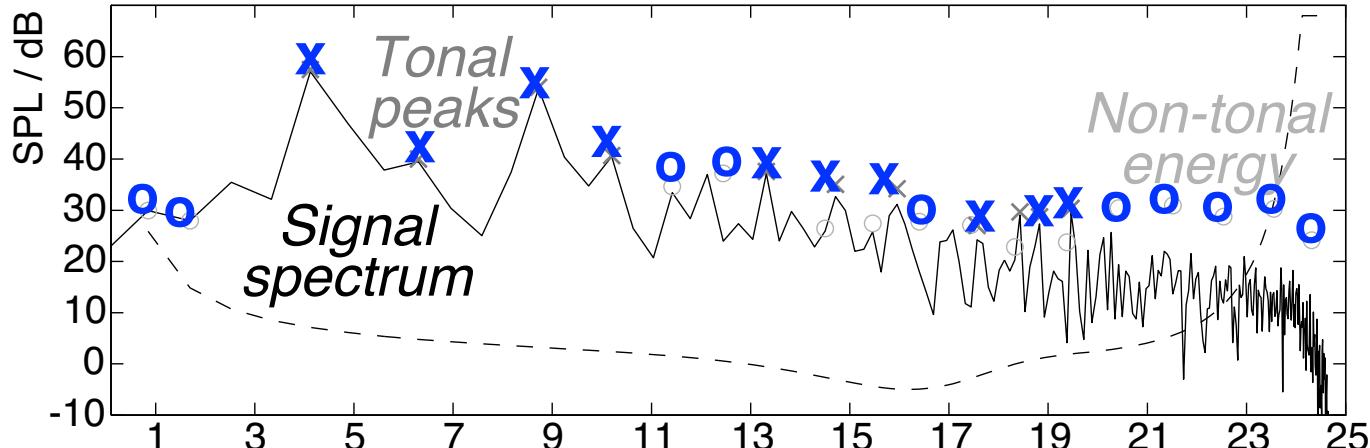
# Computing a mask.

[1] Identify tonal (x) and non-tonal (o) energy peaks.

[2] Place a local masking function for each peak.

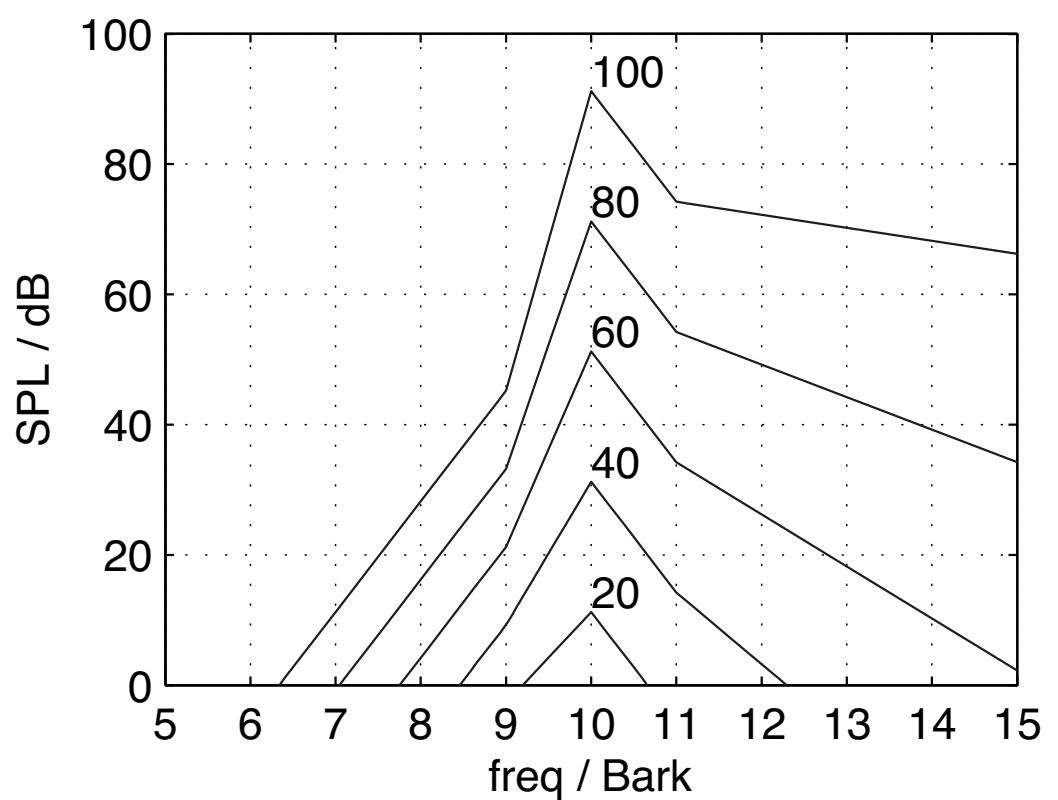
[3] Apply `max()` over frequency to compute the composite masker.

(analysis of a 26 ms audio "frame")

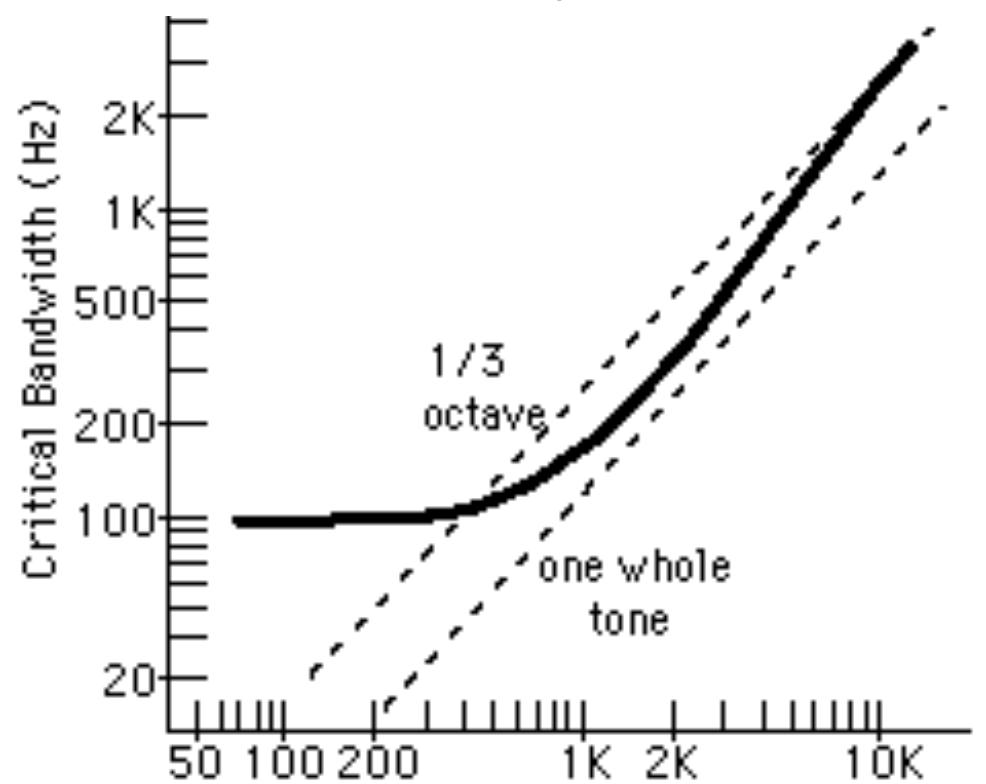


# Masking functions

Masking function **widens** with masker level, following cochlear filter response shapes.



Masking function **widen** at higher channels, following **critical bandwidth**.  
The Bark scale **warping** handles this effect.

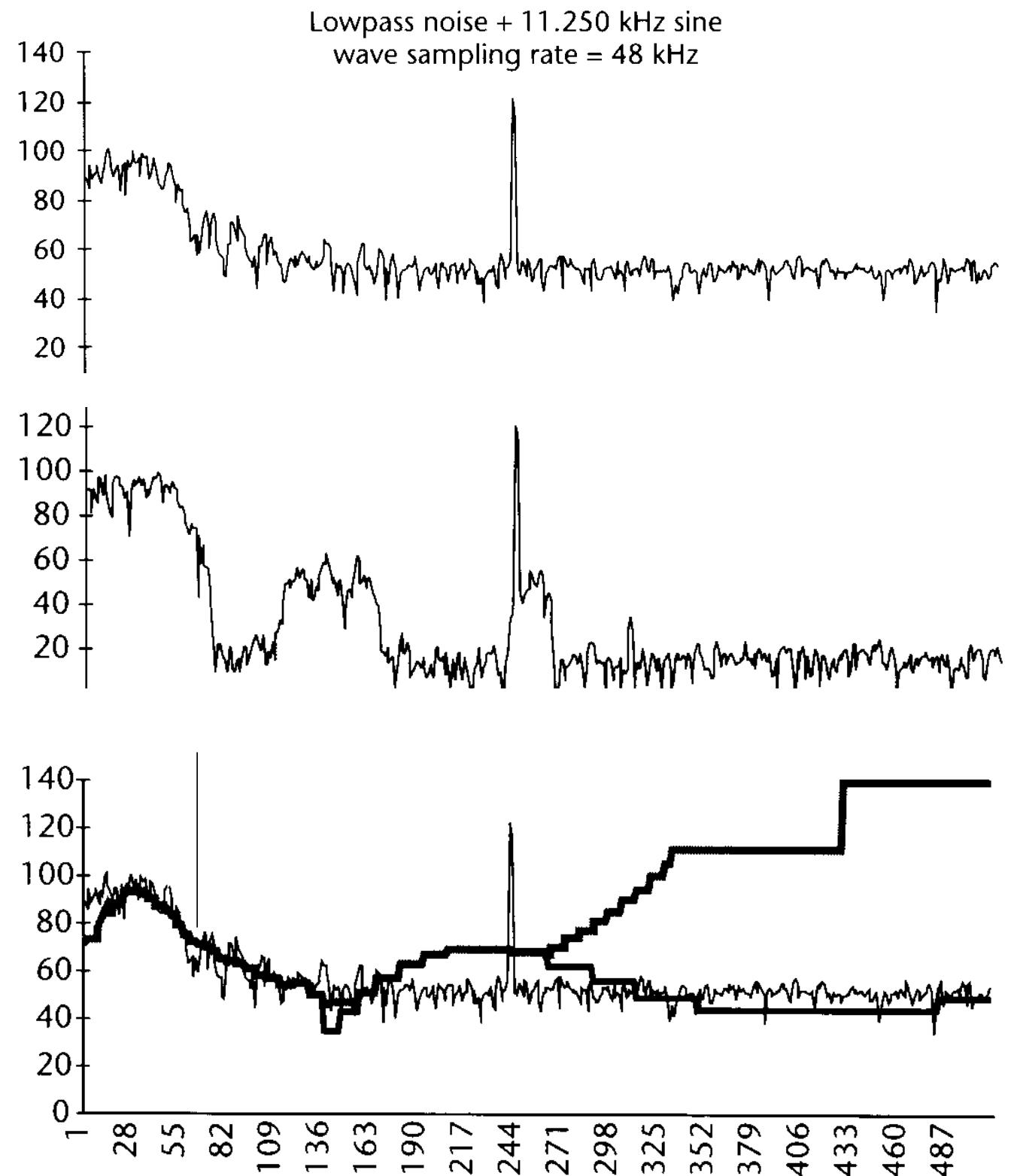


# What is “lost”?

**Input spectrum**  
a mono audio  
frame.

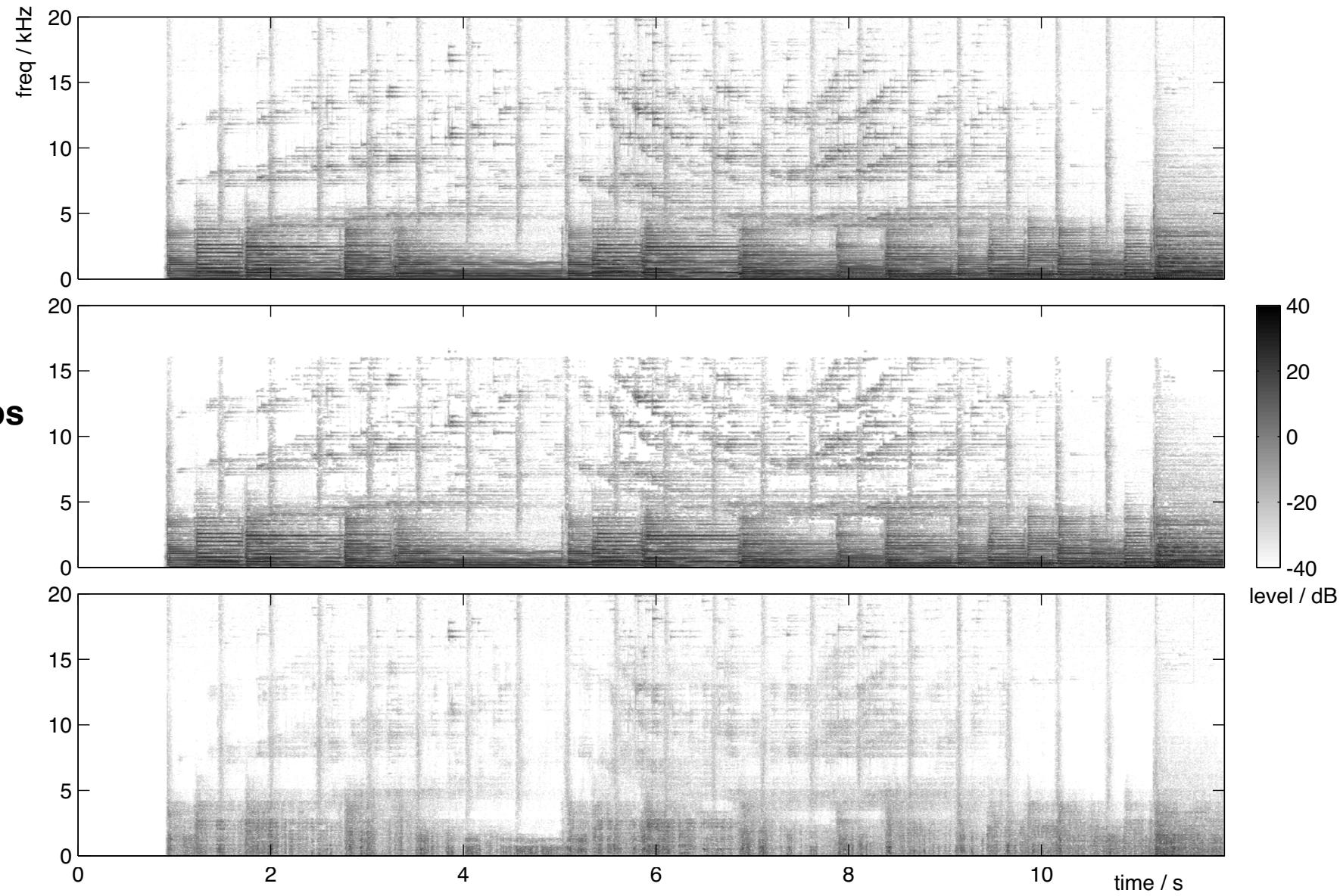
**Spectrum of**  
**encoded audio**  
(64 kb/s).

**Masking profile**  
that guided the  
quantization.



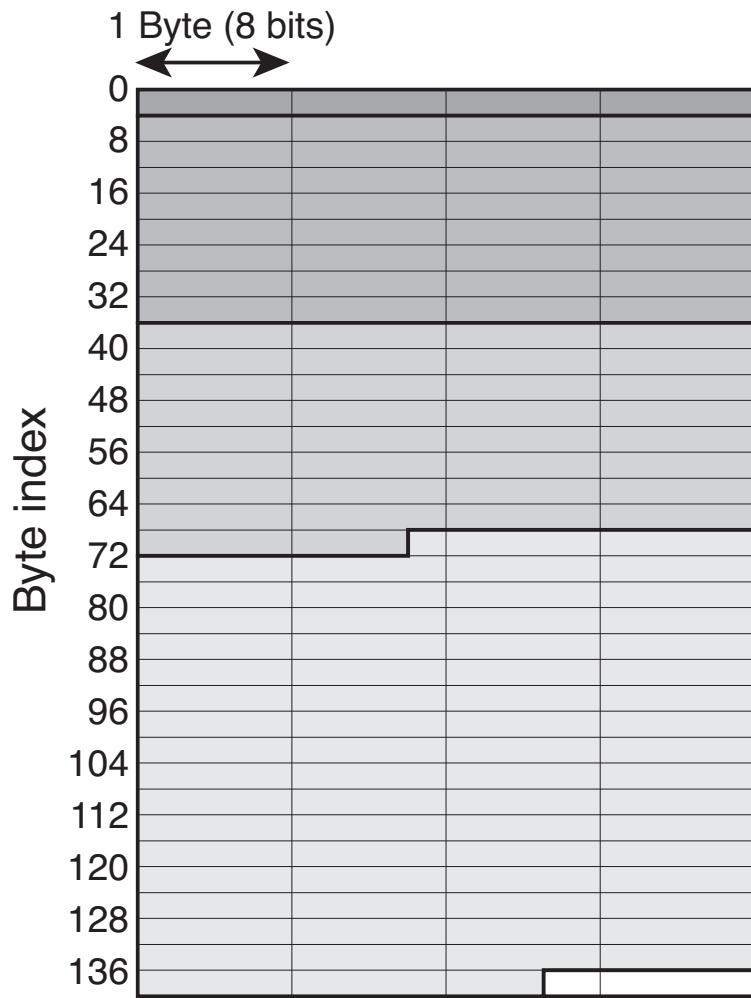
# What is “lost”?

10 seconds of pop music content  
encoded using MP3 @ 128 kb/s.



Noise is only 10 to 20 dB below signal ... but carefully placed!

# The bit level: An encoded frame in a file



**Header:** defines layer, bitrate, channels, etc. (4 bytes)

**Subband bit allocation indices:**

32 subbands x 2 channels x 4 bits  
= 32 bytes

**Subband scale factor indices:**

32 subbands x 2 channels x 6 bits  
(only for subbands with nonzero bit allocation)  
 $\leq 48$  bytes

**Quantized subband samples:**

32 subbands x 2 channels x 12 samples  
 $\times$  2-15 bits / sample (as per bit allocation,  
only for subbands with nonzero bit allocation)

**Padding** to make frame an integer number of 4 byte blocks

**FIGURE 35.12** Bit usage layout in an example MPEG-1 Audio Layer I frame encoding 384 stereo samples in 140 bytes, for a bit rate of 128 kbps.



**MP3: Lossless (Huffman) encoding used on “sample” field.**

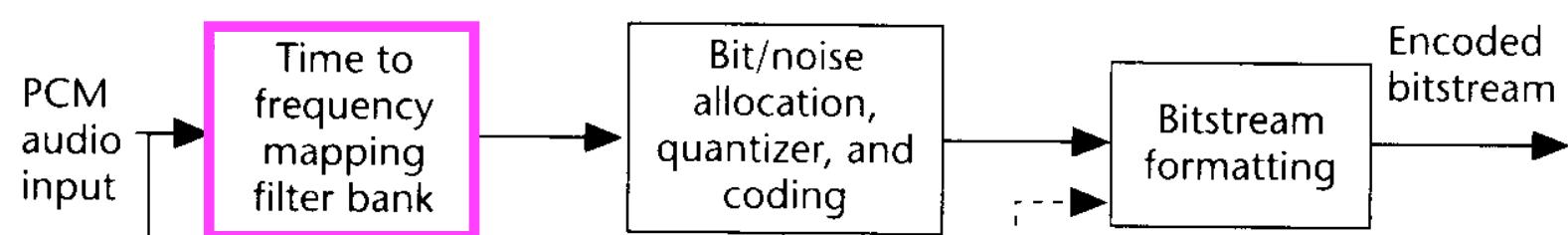
# Today's lecture: Audio Coding

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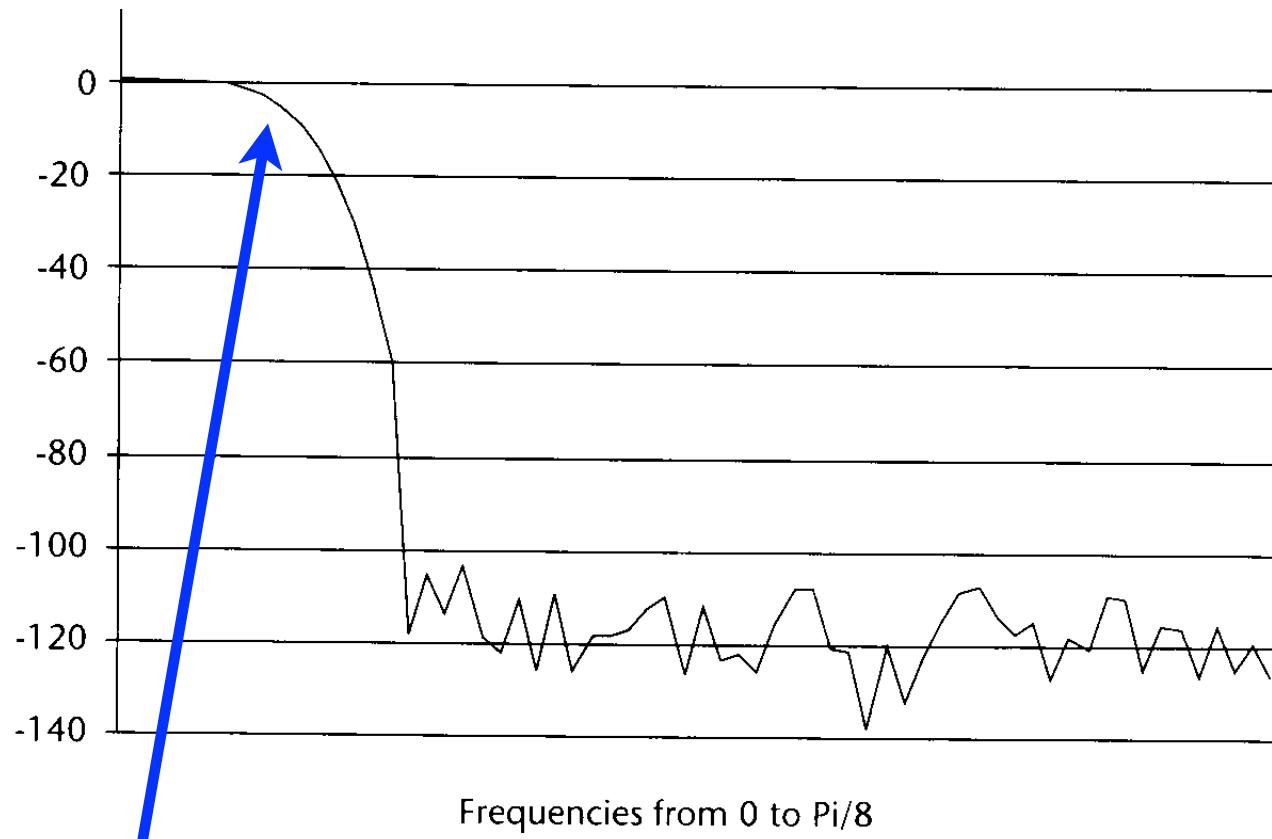
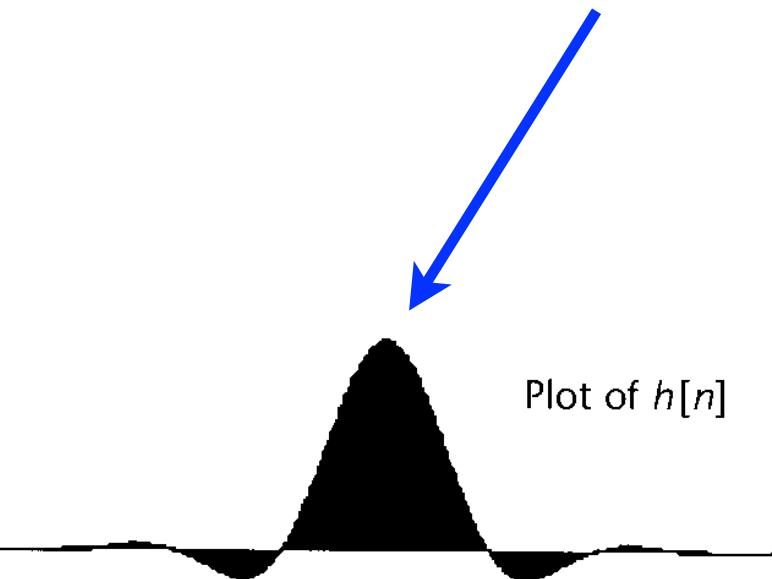
- \* **Compression: Lossless and Lossy**
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# Time-Frequency Tradeoff

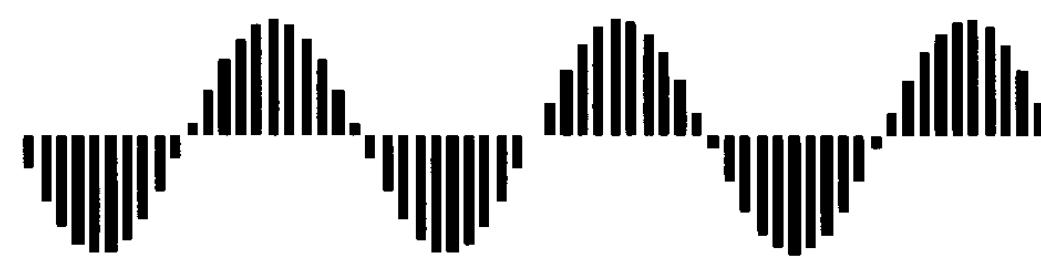


**Good time resolution is required in the filter bank ...**



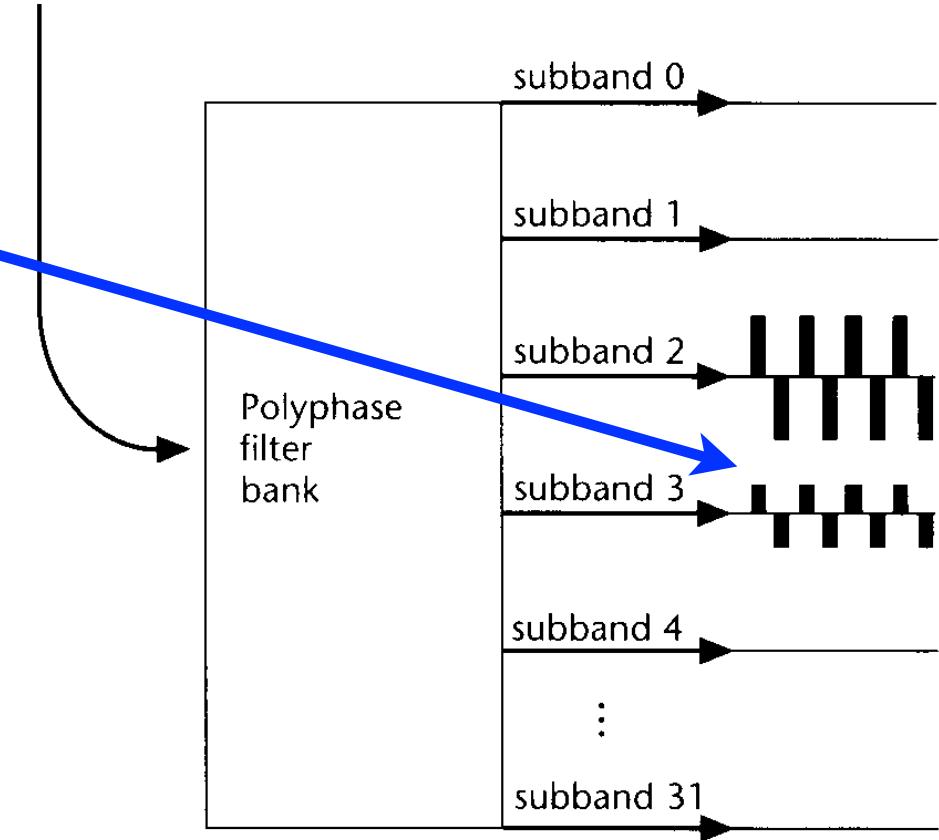
**... which implies a gentle rolloff in frequency.**

# Time-Frequency Tradeoff

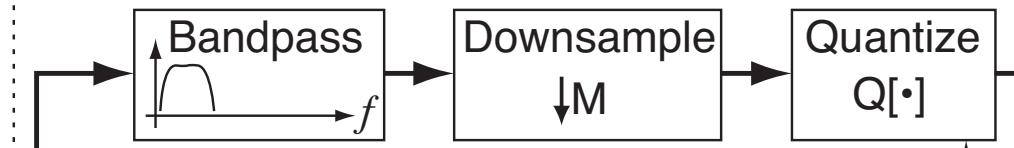


... which results in **aliases** appearing in sub-band outputs ...

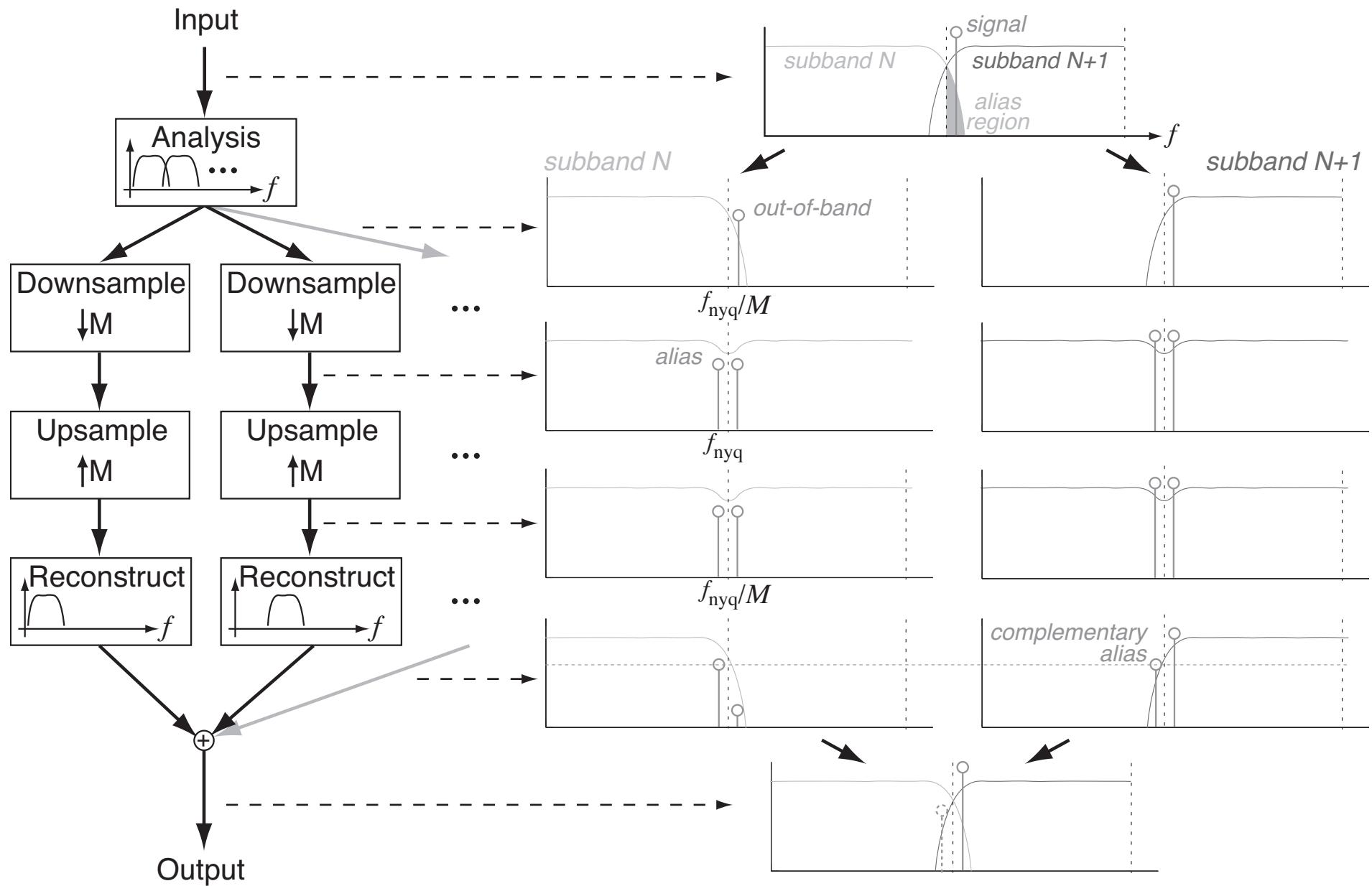
... which **fold over** as we move the sub-band to baseband by **downsampling**.



Encoder



# Solution: Quadrature-mirror filter banks



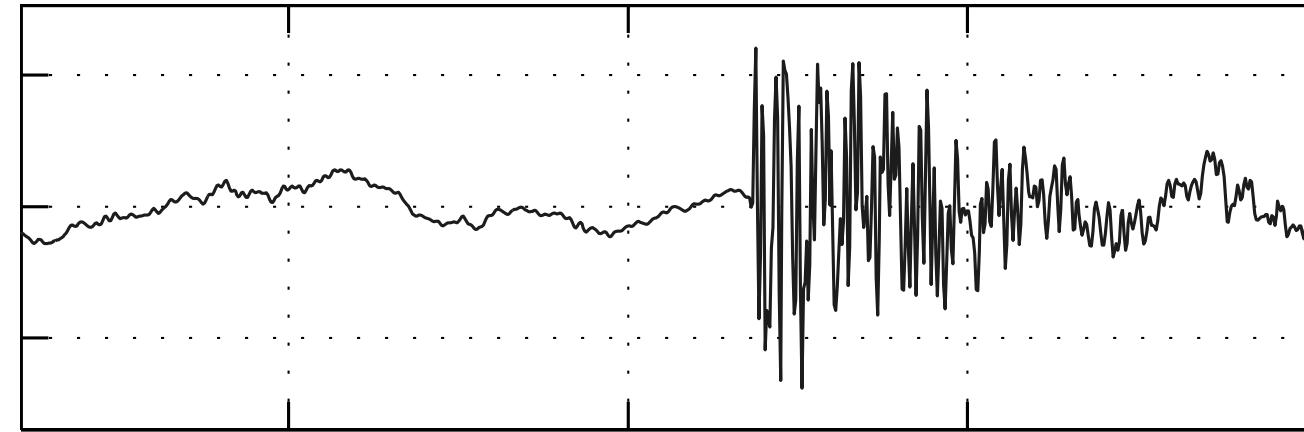
**FIGURE 35.8** Alias cancellation in quadrature-mirror filterbanks.



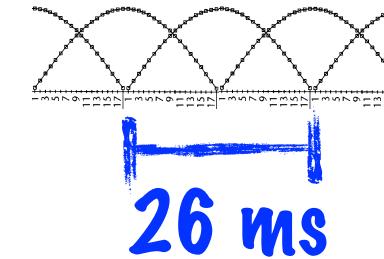
# Frame Windows

Calculated mask yields imperceptible noise once the hit **begins**, but not during the **silence** before the click.

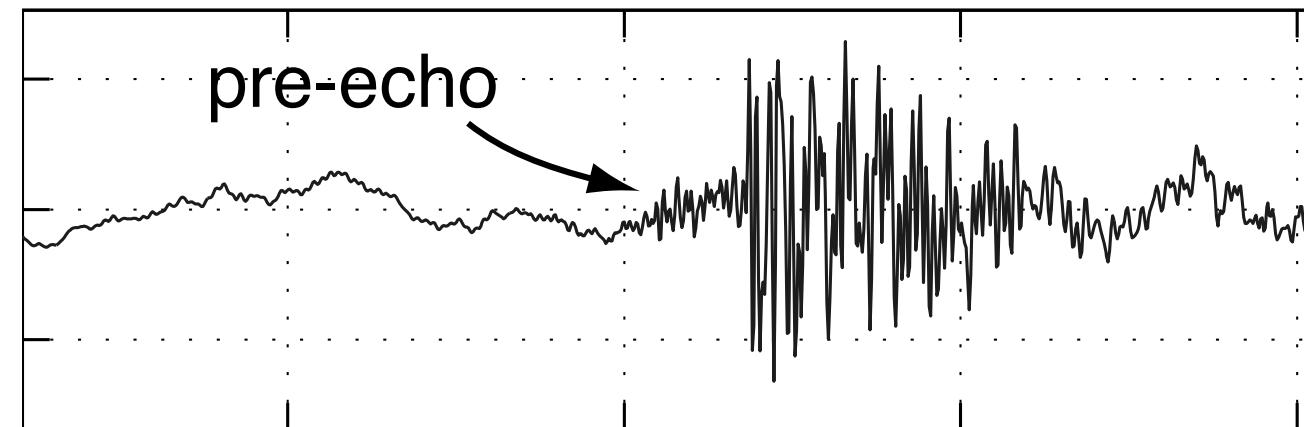
**Input: castanet hit.**



MPEG-2 (Level II)  
Frame Window

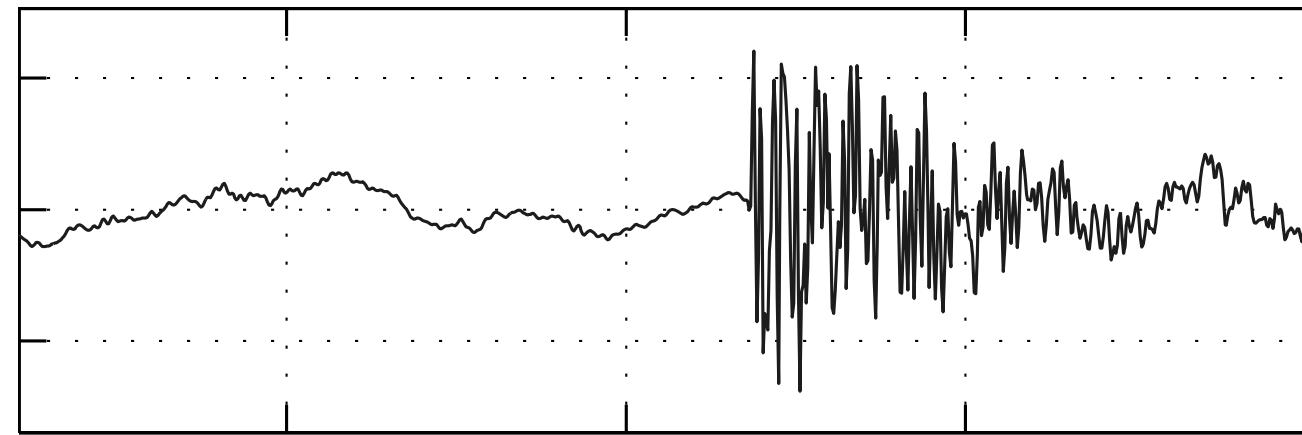


**Decoder output,  
with artifacts.**

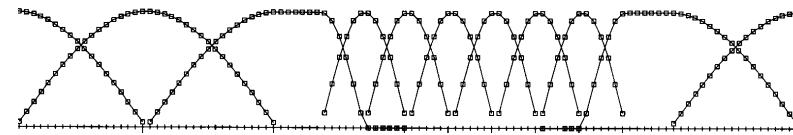


# “MP3” solution: Variable-length frames ...

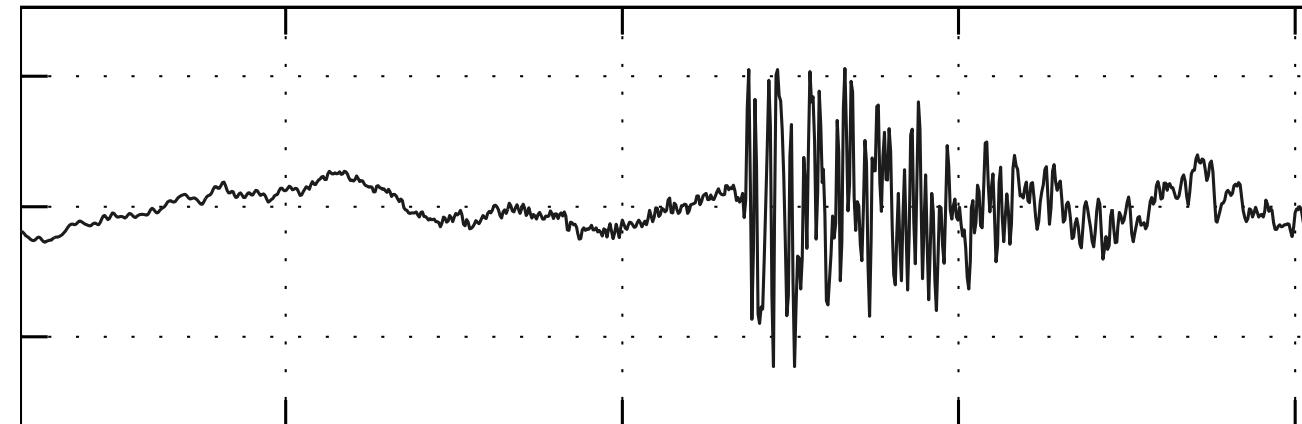
**Input: castanet hit.**



“MP3”  
Frames

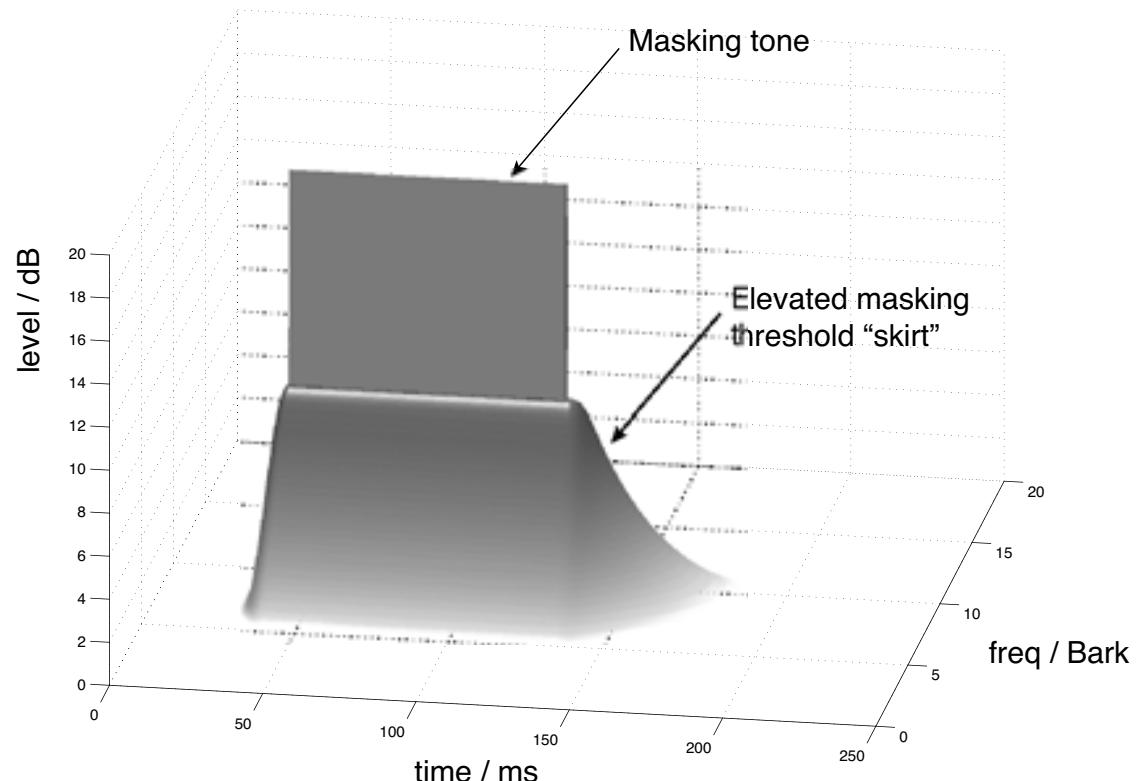
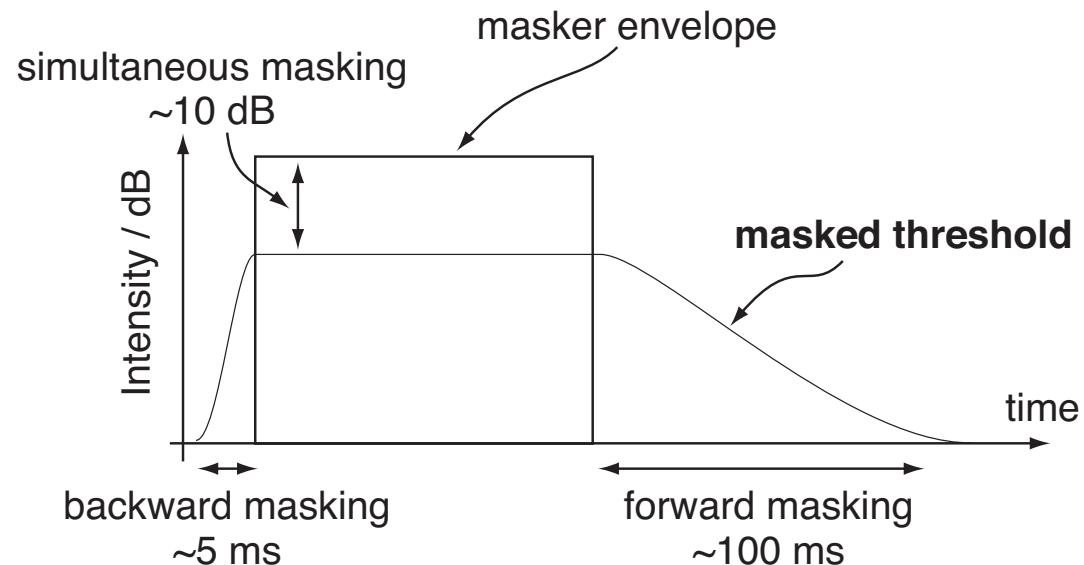


**Decoder output,  
with reduced  
artifacts.**



# Temporal masking effects ...

Masking phenomena have temporal properties which must be considered when encoding examples like “castanets”



# Today's lecture: Audio Coding

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