

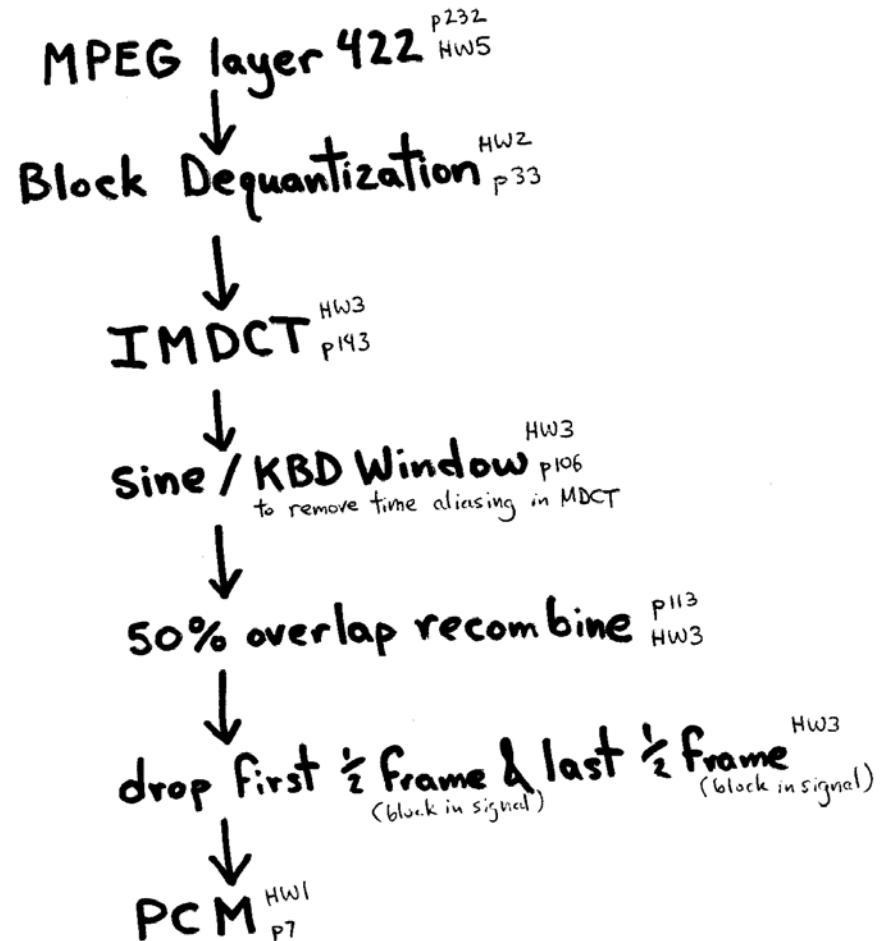
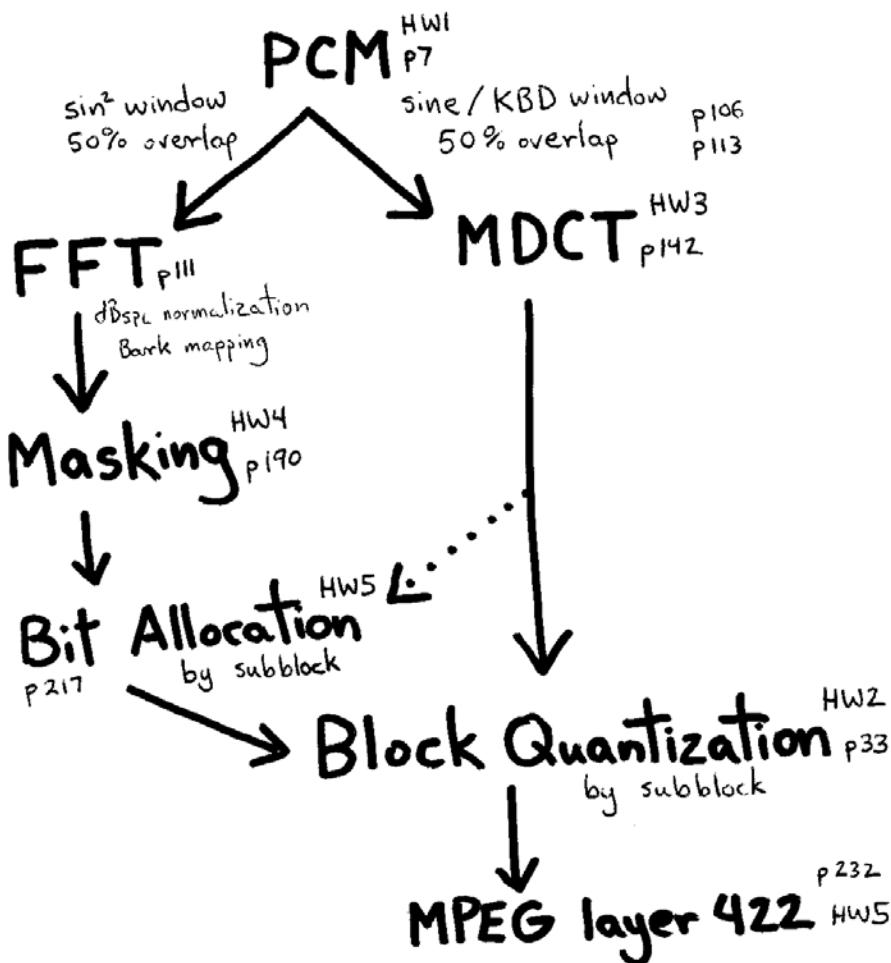
Music 422

Perceptual Audio Coding

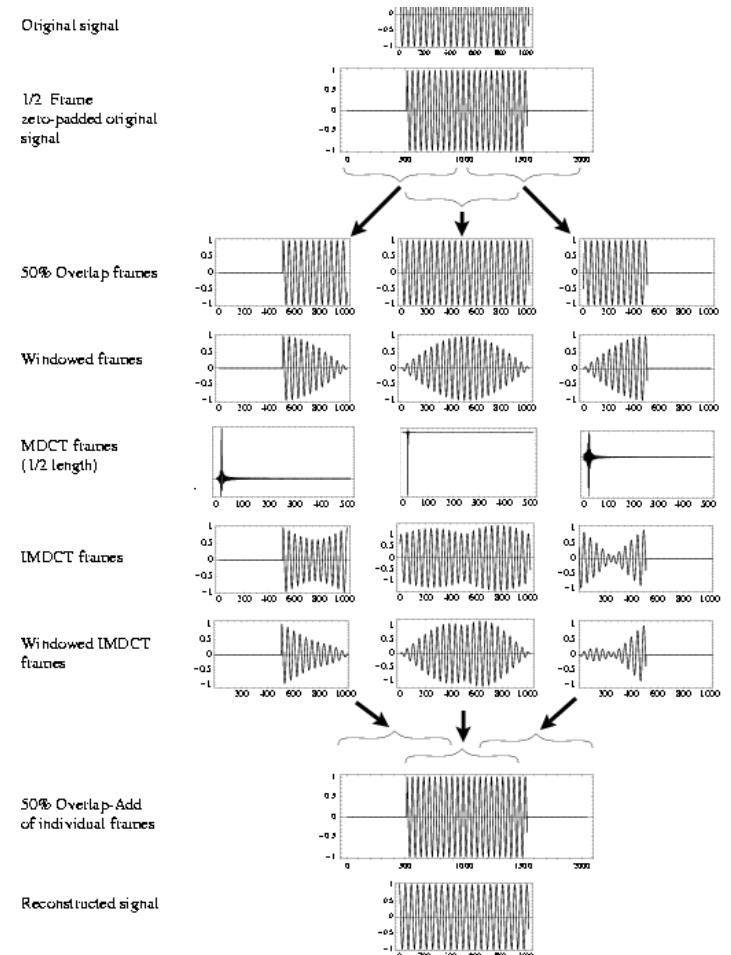
Review Lecture

Craig Stuart Sapp
19 February 2010
Stanford University

Encoding/Decoding Architecture

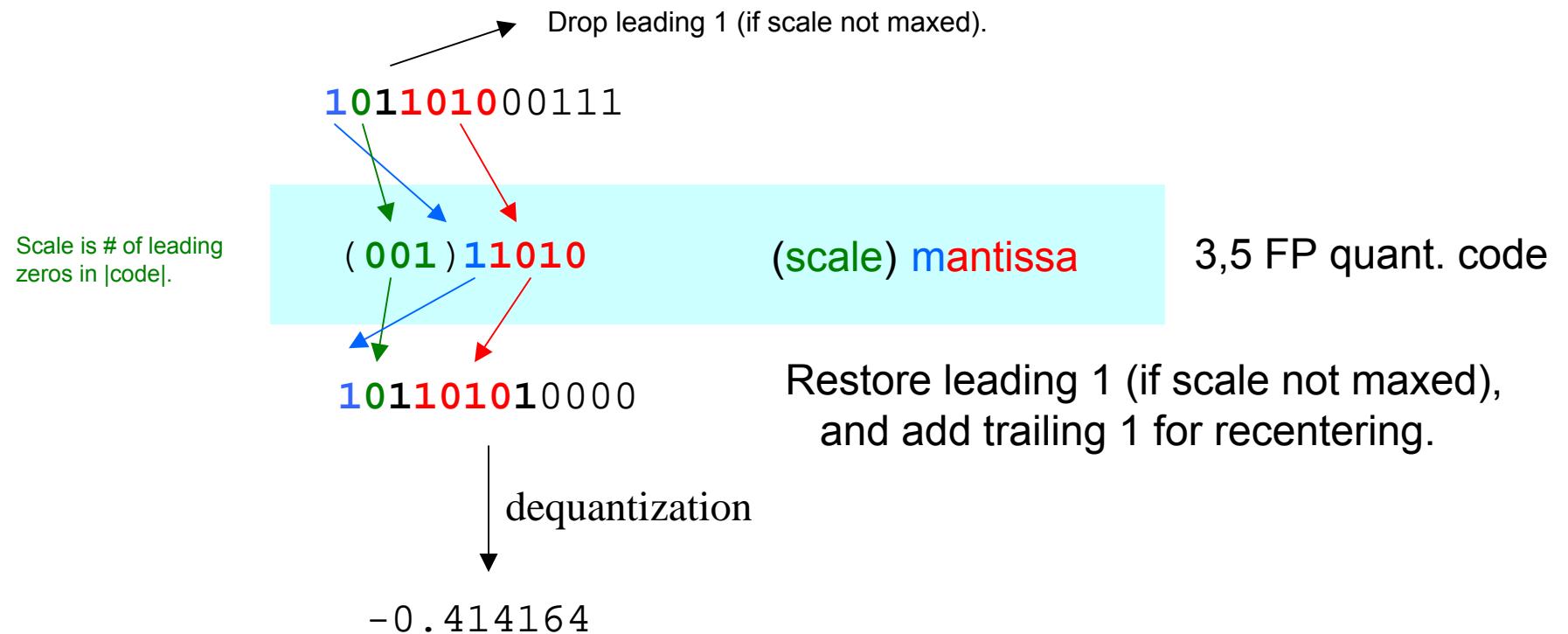


Time Domain Alias Cancellation



Floating Point Quantization

-0.41 in 12-bit mid-tread quantization: 1011 0100 0111
sign bit



Block Floating-Point Quantization

- Share a single **scale** value with multiple **mantissas**.
- Scale is derived from maximum absolute value in block.
- Must store leading 1s (so 6 dB noisier than plain Floating-Point).

-1.00	1111 1 111 1111	(000) 11111	1111 1 100 0000
-0.97	1111 1 100 0010	11111	1111 1 100 0000
-0.41	1011 0 100 0111	10110	1011 0 100 0000
0.00	0000 0 000 0000	00000	0000 0 000 0000}
0.008	0000 0 001 0000	00000	0000 0 000 0000}
0.51	0100 0 001 0100	01000	0100 0 100 0000
0.82	0110 1 000 1111	01101	0110 1 100 0000
0.999	0111 1 111 1101	01111	0111 1 100 0000
1.0	0111 1 111 1111	01111	0111 1 100 0000

↓ scale shared for all mantissas

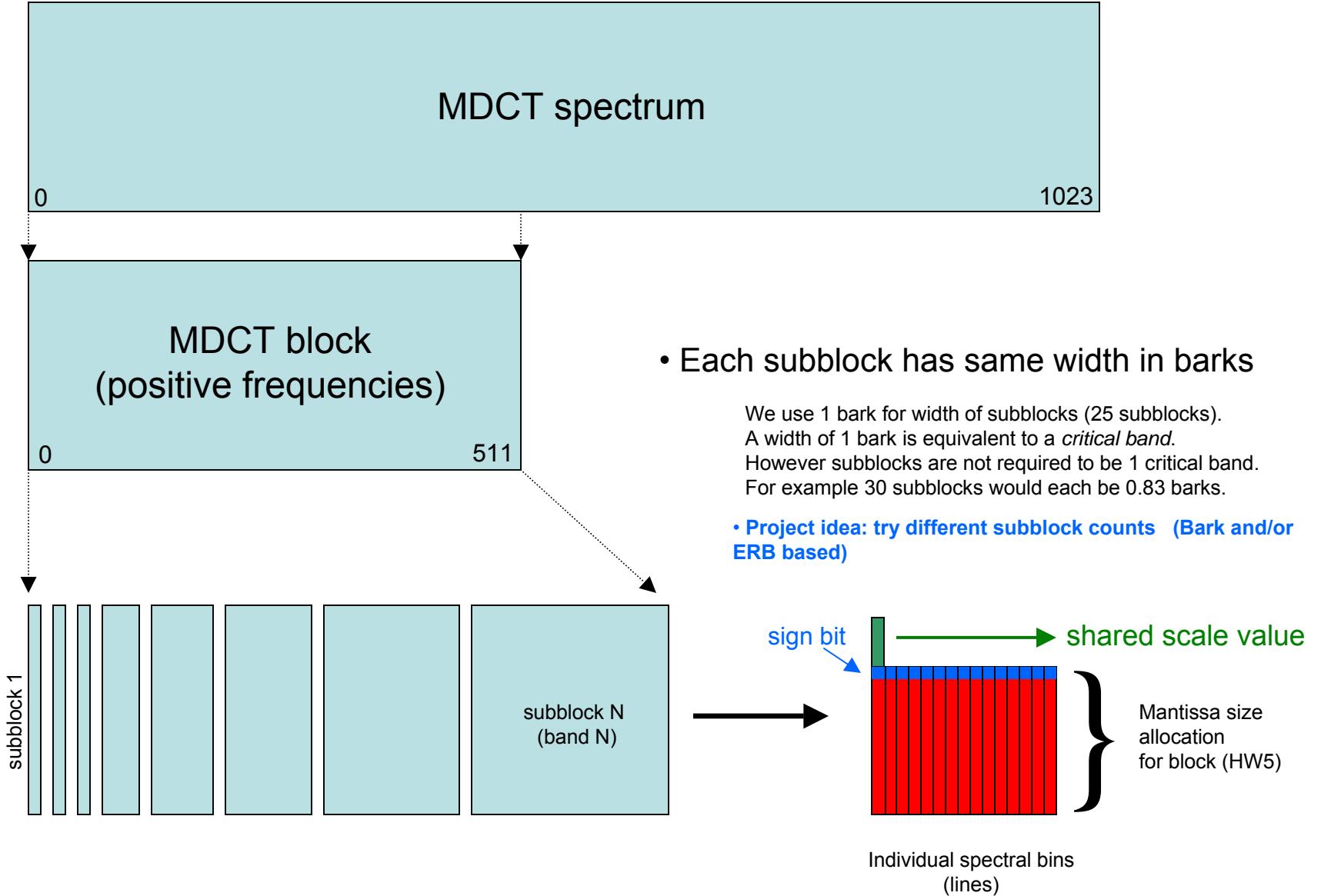
↑ Don't remove any leading 1s

↑ Don't add leading ones (because they were not taken out)

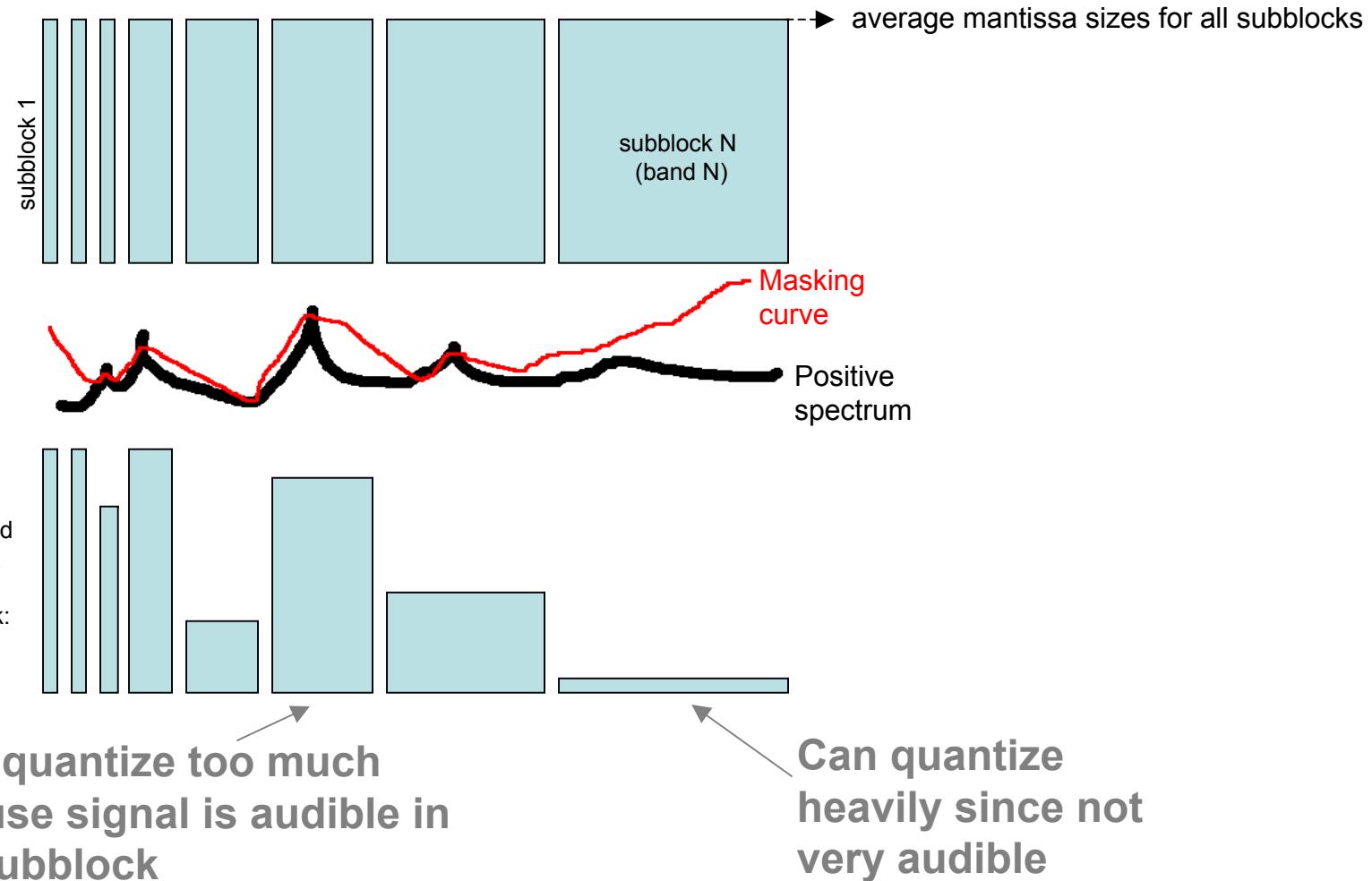
If |mantissa| is zero, then don't add trailing 1!

- Example BFP uses 48 bits ($3*1+5*9$) compared to FP at 72 bits ($3*9+5*9$).
- Block floating-point used in quantizing sub-block spectral amplitudes (HW5).

Subblock Block Floating Point



Bit-Allocation Optimization



Bit-Allocation Optimization (2)

$$R_b^{opt} = \frac{P}{K_p} + \frac{\log_2(10)}{20} \left[\text{SMR}_b - \frac{1}{K_p} \sum_{R_c \neq 0} K_c \text{SMR}_c \right]$$

Bit pool: number of bits available to all mantissas.

Signal-to-mask ratio for subblock b .

of spectral bins in subblock c .

Optimized mantissa bit allocation for each subblock b .

Number of non-zero spectral bins in block.

Summation over all subblocks c which are not pre-assigned an allocation of 0 mantissa bits due to a poor SMR.

A constant

Without pre-screening by subblock SMR:

$$R_b^{opt} = R_{avg} + \frac{\log_2(10)}{20} \left[\text{SMR}_b - \frac{1}{K} \sum_c K_c \text{SMR}_c \right]$$

Average bits per mantissa in block

Total bins in block.

Signal-To-Mask Ratio

$$\text{SMR}_b = 20 (\log_{10} |x_{\max_b}| - \log_{10} M_b)$$

Maximum spectral bin
amplitude in subblock

Masking amplitude for
subblock

- Project idea: compare FFT/MDCT as `x_max_b`

Equivalent:

$$\text{SMR}_b = \text{dB}_{(\text{SPL})} \text{ of max amplitude in subblock} - \text{dB}_{(\text{SPL})} \text{ of mask amplitude for subblock}$$

Alternative:

$$\begin{aligned}\text{SMR}_b &= \max_n [\text{dB}_{(\text{SPL})} \text{ of amplitude}_n \text{ in subblock} - \text{dB}_{(\text{SPL})} \text{ of mask amplitude}_n \text{ in subblock}] \\ &= \max_n [SMR_n]\end{aligned}$$

Mantissa Bit-Allocation (3)

SMR's Expressed in amplitudes:

$$R_b = R_{\text{avg}} + \log_2 \left(\frac{x_{\max_b}}{M_b} \right) - \frac{1}{K} \sum_c K_c \log_2 \left(\frac{x_{\max_c}}{M_c} \right).$$

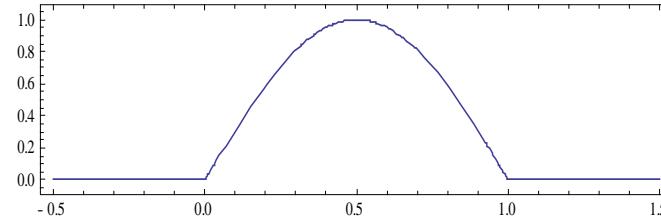
Bit-allocation optimization, ignoring masking:

$$R_b = R_{\text{avg}} + \log_2 x_{\max_b} - \frac{1}{K} \sum_c K_c \log_2 x_{\max_c}.$$

Even/Odd Window Definition

**Continuous normalized window
(causal definition):**

$$\sin(\pi * t) \quad \text{for } t >= 0 \text{ and } t <= 1$$

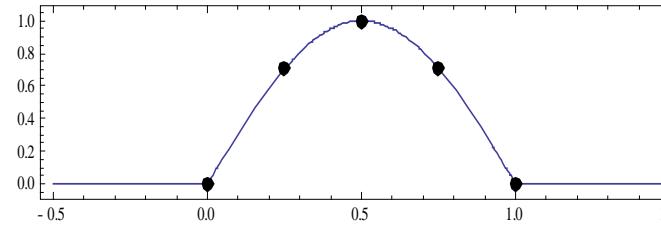


Odd-length sampled window:

$$N=5$$

$$\sin(arange(N)*\pi/(N-1))$$

- **N-1** for stretching sample points to center of window (otherwise not symmetric: {0, .2, .4, .6, .8}).



$$\left\{ 0, \frac{1}{4}, \frac{1}{2}, \frac{3}{4}, 1 \right\} \rightarrow \left\{ 0, \frac{1}{\sqrt{2}}, 1, \frac{1}{\sqrt{2}}, 0 \right\}$$

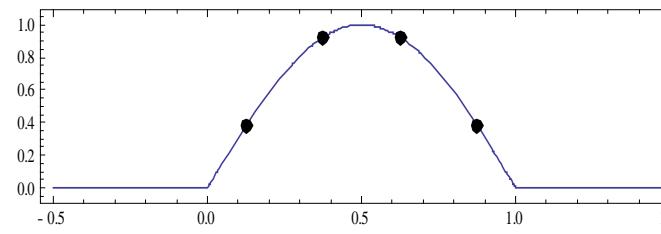
Even-length sampled window:

$$N=4$$

$$\sin((arange(N)+0.5)*\pi/(N))$$

(c.f. p. 107)

- **0.5** offset for sampling from center of window, not left justified in window range (otherwise {0, 1/4, 1/2, 3/4}).
- not $\sin(arange(N)*\pi/(N-1))$ since COLA of window² @50% overlap would not occur (otherwise range would be {0, 1/3, 2/3, 1}).



$$\left\{ \frac{1}{8}, \frac{3}{8}, \frac{5}{8}, \frac{7}{8} \right\} \rightarrow \{0.382683, 0.92388, 0.92388, 0.382683\}$$

c.f. Parallels to DFT and MDCT

Even & Odd Window Lengths

Optimized for 50% (25%, 12.5%...) overlap add systems

```
import numpy as np

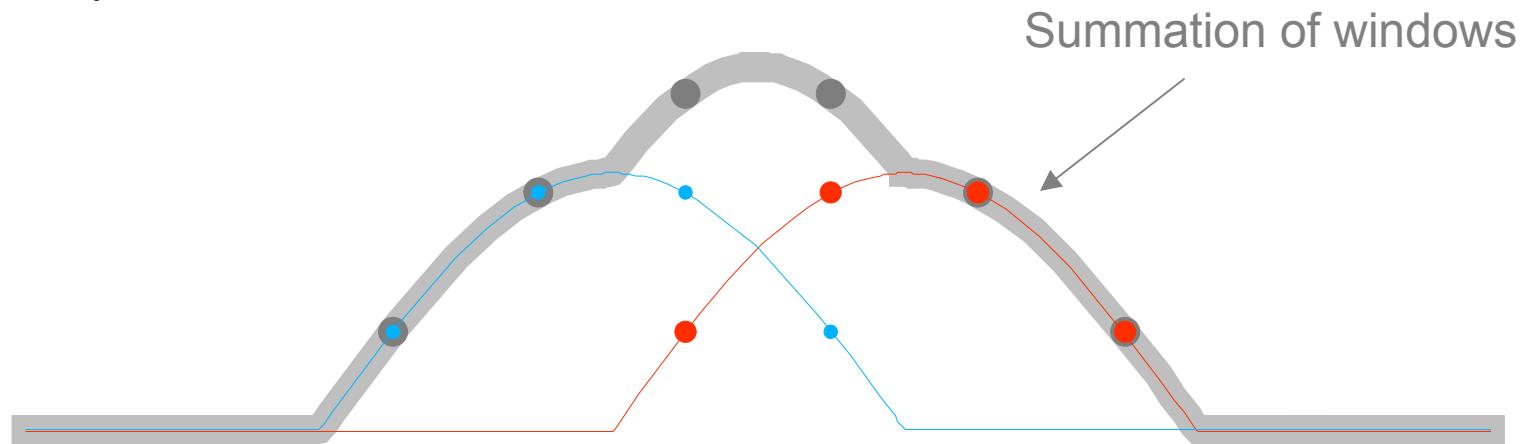
def SineWindow(length):
    N = int(length)
    assert N > 1
    if (N%2 == 0):      # even sampling of continuous window definition
        halfwindow = np.sin( np.arange(N/2)+0.5 ) * np.pi / N
        return np.concatenate([halfwindow, np.flipud(halfwindow)])
    else:                # odd sampling of continuous window definition
        halfwindow = np.sin( np.arange(N/2+1) * np.pi / (N-1) )
        # don't repeat the middle sample of an odd window:
        return np.concatenate([halfwindow, np.flipud(halfwindow[0:-1])])

def HanningWindow(length, power=2):
    return SineWindow(length)**power
```

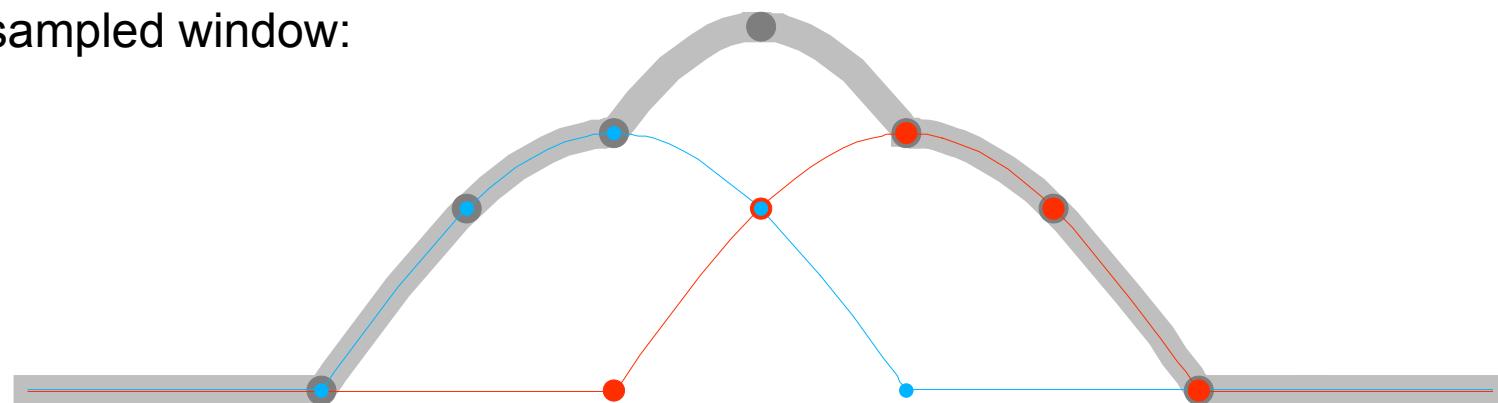
50% Overlap Add

(Sine Window)

Evenly sampled window:



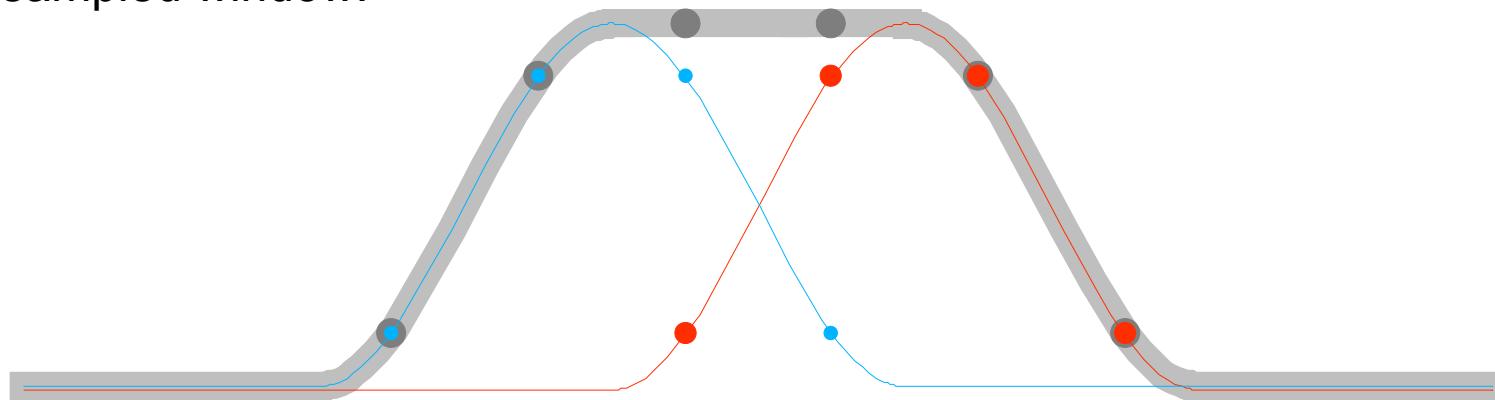
Oddly sampled window:



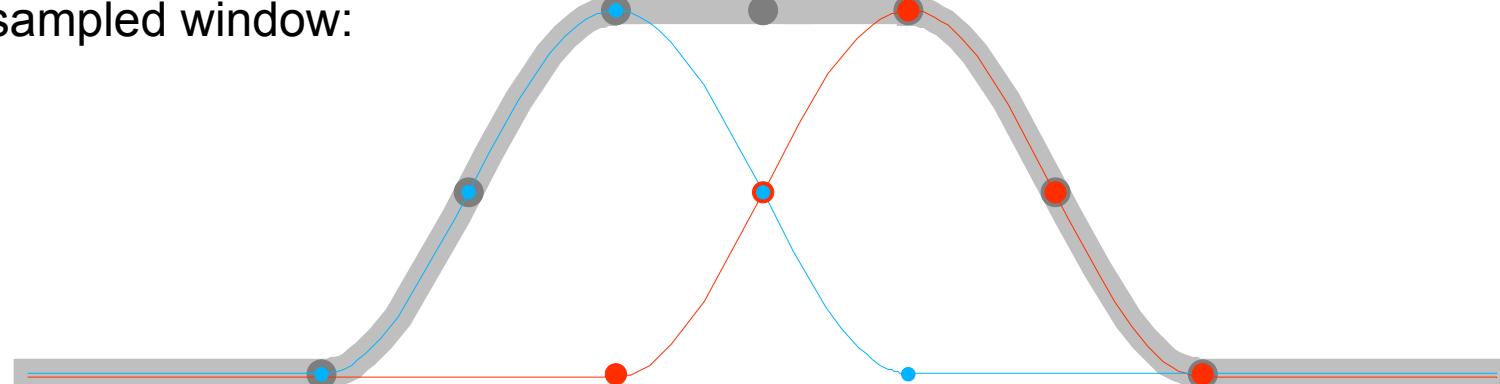
Constant OverLap Add (COLA)

(Hanning Window = Sine² Window)

Evenly sampled window:

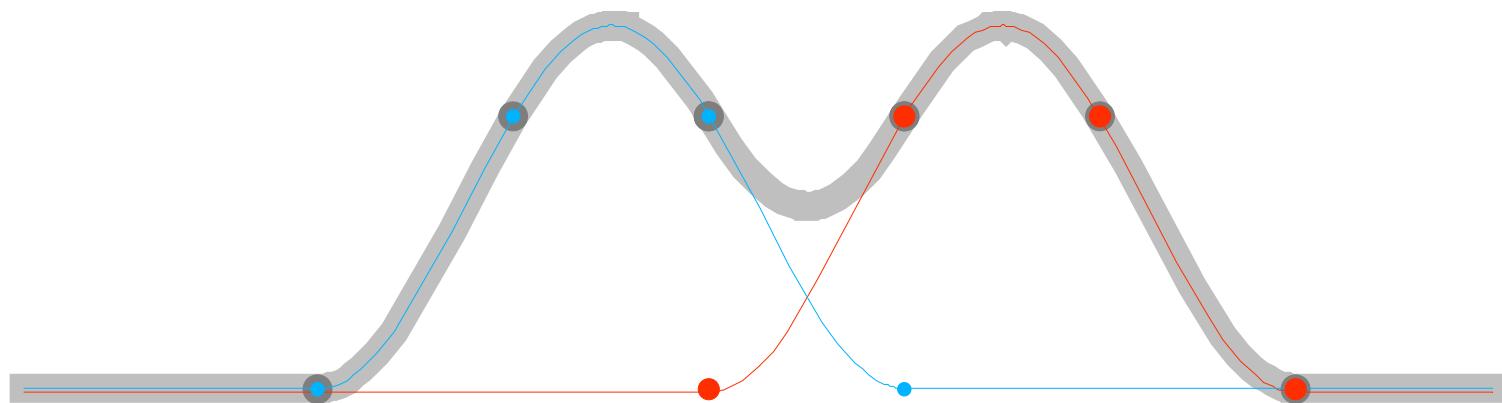


Oddly sampled window:



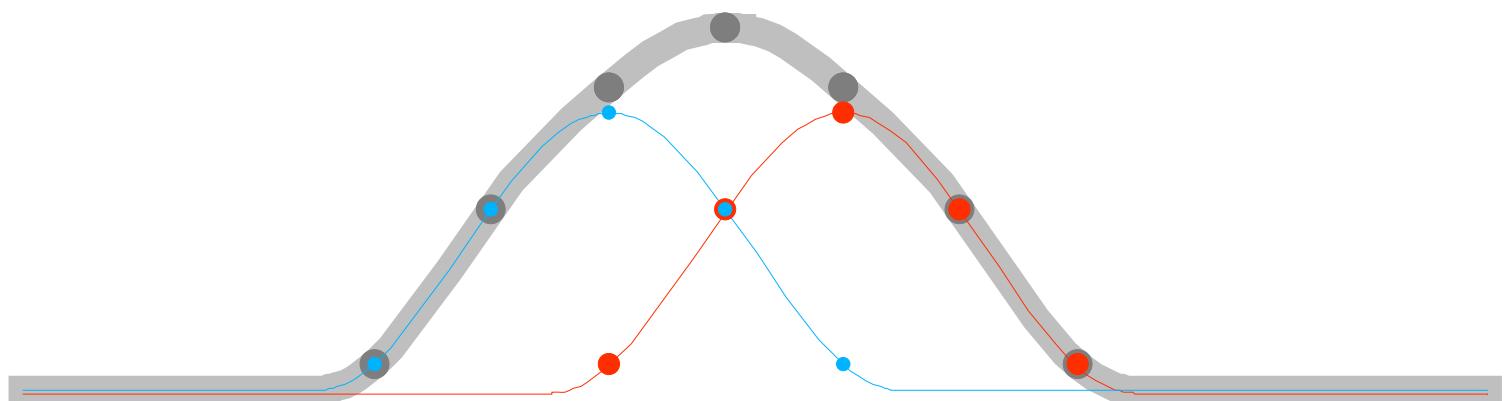
Bubbly COLA

Using even length with odd window sampling definition (less than 50% overlap):

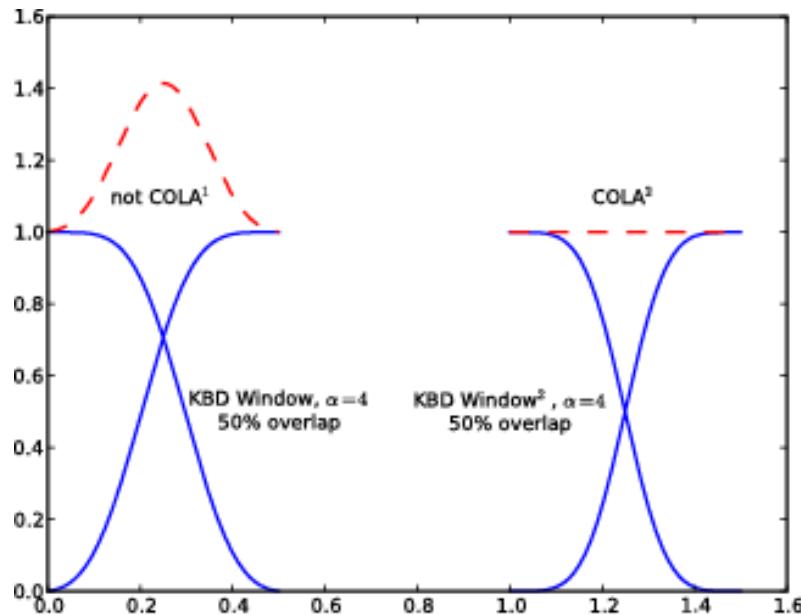


N.B.: `numpy.hanning(4)` → [0, 0.75, 0.75, 0]

Using odd length with even window sampling definition (more than 50% overlap):



Kaiser-Bessel Derived Window



```
import pylab as pl
N = 1024
window = KBDWindow(1024, 4.0)
winl = window[0:N/2]; winr = window[N/2:N]
overlapsum = winl + winr
win2l = winl**2; win2r = winr**2
overlap2sum = win2l + win2r
n = np.arange(N)/(N-1.0) # normalized range
nl = n[0:N/2]; nr = n[N/2:N]
pl.plot(nl, winl, 'b-'); pl.plot(nr, winr, 'b-')
pl.plot(nl, overlapsum, 'r-')
pl.plot(nl+1, win2l, 'b-'); pl.plot(nl+1, win2r, 'b-')
pl.plot(nl+1, overlap2sum, 'r--')
pl.text(0.25, 1.1, 'not COLA$^1$', \
        horizontalalignment='center', verticalalignment='center')
pl.text(1.25, 1.1, 'COLA$^2$', \
        horizontalalignment='center', verticalalignment='center')
pl.text(0.5, 0.5, 'KBD Window, $\alpha=4$\n50% overlap', \
        horizontalalignment='center', verticalalignment='center')
pl.text(1.0, 0.5, 'KBD Window$^2$, $\alpha=4$\n50% overlap', \
        horizontalalignment='center', verticalalignment='center')
pl.show()
```

KBD Window in Python

```
import numpy as np

def KBDWindowKVH(length, beta):
    """Kaiser Bessel Derived Window
    Arguments:
        length: Length of output window array. Must be positive & even.
        beta:  Kaiser window parameter alpha times pi.
    """
    N      = int(length)
    assert N%2 == 0
    assert N > 0
    M      = N/2
    w      = np.kaiser(M+1, beta)
    wMsum = 1.0 / np.sqrt(w[0:M+1].sum())
    wnsuml = np.sqrt(np.cumsum(w[0:M]))
    wl     = wnsuml * wMsum
    return np.concatenate([wl, np.flipud(wl)])
```

Note: $\beta = \pi \alpha$
c.f. sinc()

```
import numpy as np
import scipy.special as sps

def KBDWindowCSS(length, alpha):
    """Kaiser Bessel Derived Window
    Arguments:
        length: Length of output window array. Must be positive & even.
        alpha:  Kaiser window parameter.
    """
    N      = int(length)
    assert N % 2 == 0
    assert N > 0
    halfN   = N/2
    beta    = np.pi * alpha
    bessels = sps.i0(beta * np.sqrt(1.0-(4.0*np.arange(halfN)/N-1.0)**2.0))
    halfwin = np.cumsum(bessels)
    normalize = halfwin[-1] + sps.i0(beta*np.sqrt(1.0-(4.0*halfN/N-1.0)**2.0))
    halfwin = np.sqrt(halfwin/normalize)
    return np.concatenate([halfwin, np.flipud(halfwin)])
```

KBD Window in Python (2)

```
import numpy as np

def kaiserWindow(length, alpha): # Colin Raffel's implementation
    N = int(length)
    n = np.arange(N)
    w = (n - (N-1)/2.0)/((N-1)/2.0)
    w = w**2
    w = np.sqrt(1-w)
    w = np.pi * alpha * w
    w = np.i0(w)
    w = w/np.i0(np.pi*alpha)
    return w

def KBDWindowCAR(length, alpha): # Colin Raffel's implementation
    N      = int(length)
    assert N%2 == 0
    assert N > 0
    wKai  = kaiserWindow(N/2+1, alpha)
    wHalf = np.sqrt(np.cumsum(wKai[0:-1])/np.sum(wKai))
    return np.concatenate([wHalf, np.flipud(wHalf)])
```

Decibels

10 decibels = 1bel

decibel = $10 \log_{10}(\text{amplitude}^2 / \text{reference}^2)$ (reference is an amplitude)

decibel = $20 \log_{10}(|\text{amplitude}|/\text{reference})$ (extracting square from log innards)

- amplitude² is proportional to Energy, Intensity, Power:

decibel = $10 \log_{10}(\text{intensity}/\text{reference})$ (reference is an intensity)

decibel = $10 \log_{10}(\text{energy}/\text{reference})$ (reference is in energy units)

decibel = $10 \log_{10}(\text{power}/\text{reference})$ (reference is in power units)

- amplitude² is proportional to Pressure², RMS² and Voltage²:

decibel = $10 \log_{10}(\text{pressure}^2/\text{reference}^2)$ (reference is in pressure units)

decibel = $20 \log_{10}(|\text{pressure}|/\text{reference})$ (reference is in pressure units)

decibel = $10 \log_{10}(\text{voltage}^2/\text{reference}^2)$ (reference is in voltage units)

decibel = $20 \log_{10}(|\text{voltage}|/\text{reference})$ (reference is in voltage units)

decibel = $10 \log_{10}(\text{rms}^2/\text{reference}^2)$ (reference is in some type of rms amplitude)

decibel = $20 \log_{10}(|\text{rms}|/\text{reference})$ (reference is in some type of rms amplitude)

- units of reference and number being compared have to **match**

Purpose of Decibels

- to relate numbers on a wide range from very small and very large.

Reference: 1 meter (distances are proportional to amplitude)

- Diameter of the Universe: 2.60×10^{26} m → 528 dB
 - Diameter of the Milky Way: 9.5×10^{20} m → 419 dB
 - Distance to Proxima Centauri: 4.1×10^{16} m → 332 dB
 - Distance to the Sun: 1.478×10^{11} m → 223 dB
 - Stanford to South Africa: 1.7×10^7 m → 144 dB
 - Diameter of Earth: 1.276×10^7 m → 142 dB
 - Blue Whale: 27 m → 29 dB
 - Giraffe: 5.5 m → 15 dB
 - Human: 2 m → 6 dB
 - Mouse: 0.03 m → -30 dB
 - Virus: 2×10^{-5} m → -93 dB
 - Diameter of hydrogen atom: 1.11×10^{-11} m → -199 dB
 - Diameter of proton: 10^{-15} m → -300 dB
 - Diameter of electron: $< 10^{-16}$ m → -320 dB
 - Plank length: 1.616×10^{-35} m → -695 dB
- Universe has a dynamic range of 1223 dB (203 bits)

Sound Pressure Level in Decibels

$$Db_{SPL} = 20 \log_{10} (\text{pascals}/0.000020)$$

- *Reference pressure:* $20 \mu\text{Pa} = 20 \times 10^{-6} \text{ kg}/(\text{m}\cdot\text{s}^2)$
- 1 pound per square inch = 6,894.76 pascals
 $20 \mu\text{Pa} = 2.9 \times 10^{-9} \text{ p.s.i.}$
1 atmosphere = 101,325 pascals = 194 dB_{SPL}
- My bike tires are inflated to 150 p.s.i.:
 $20 \log(150 / (2.9 \times 10^{-9})) = 214 \text{ dB}_{SPL}$
- Noise from blood flow in ears is about 0 dB_{SPL}
- Noise from air molecules randomly hitting eardrum is very roughly -6 dB_{SPL}

http://www.silcom.com/~aludwig/Physics/Noise_floor/Molecular_noise_and_blackbody.htm

Other Decibels

- Decibel **differences** are the same regardless of the reference.
 - For example, 6 dB difference when doubling amplitude for any reference:
 $20 \log(2/r) - 20 \log(1/r) = 20 \log[(2/r)/(1/r)] = 20 \log(2) = 6.0206$ dB difference
- But **don't** subtract decibels that use different references.

- Other references:

dBm: 1 mW (power)

dBW: 1 W (power)

dBV: 1 V_{RMS}

dBu: 0.775 V_{RMS}

dB_{SIL}: 10^{-12} W/m²

dBfs: max

Loudness
(perceptual weighting):

dB(A)

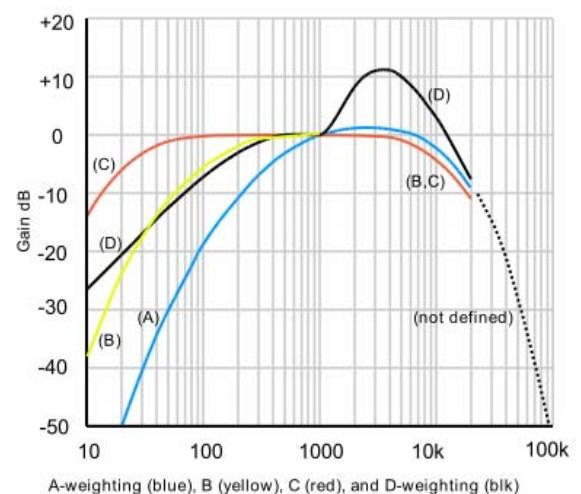
dB(B)

dB(C)

dB(D)

dB(Z)

en.wikipedia.org/wiki/A-weighting



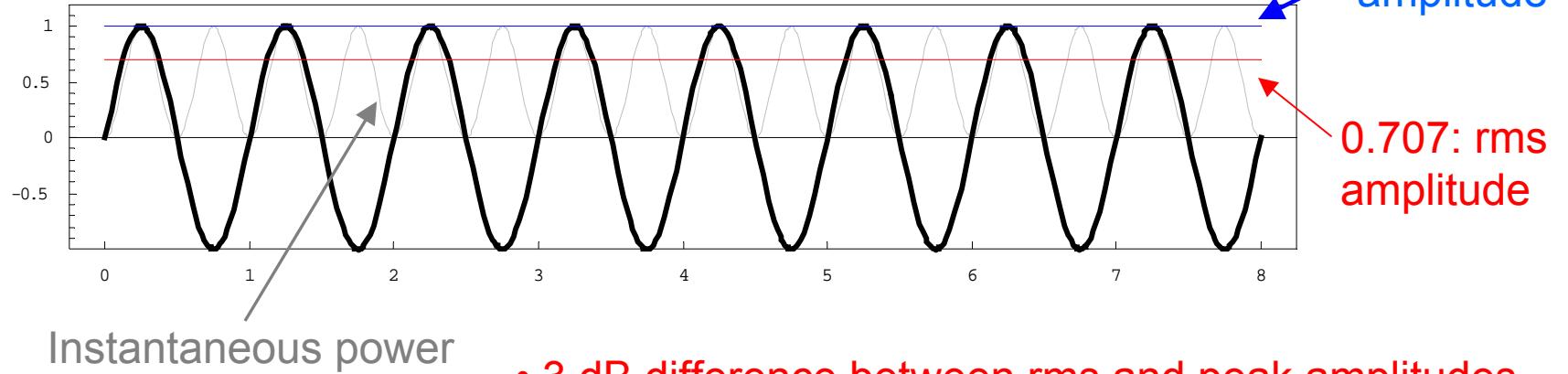
RMS

- **Root Mean Squared** usually the measured pressure value of a physical sound:

$$x_{\text{rms}} = \sqrt{\langle x^2 \rangle} = \sqrt{\frac{x_1^2 + x_2^2 + \dots + x_n^2}{n}}$$

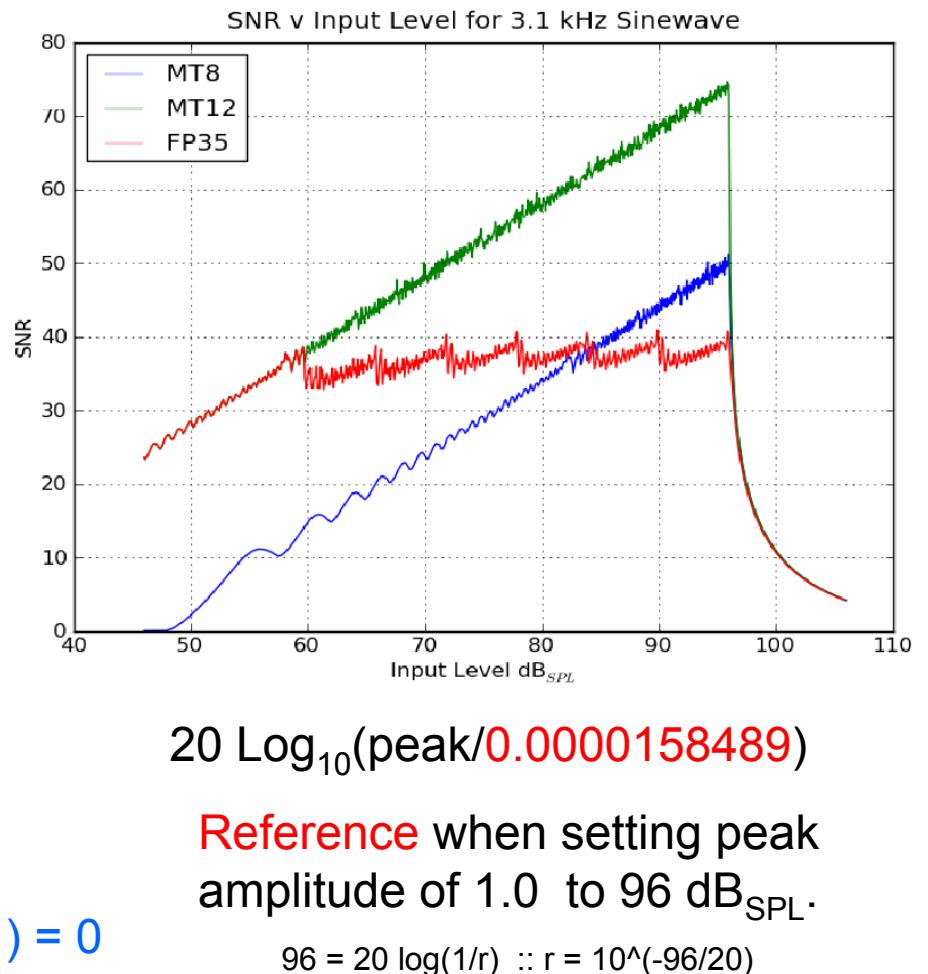
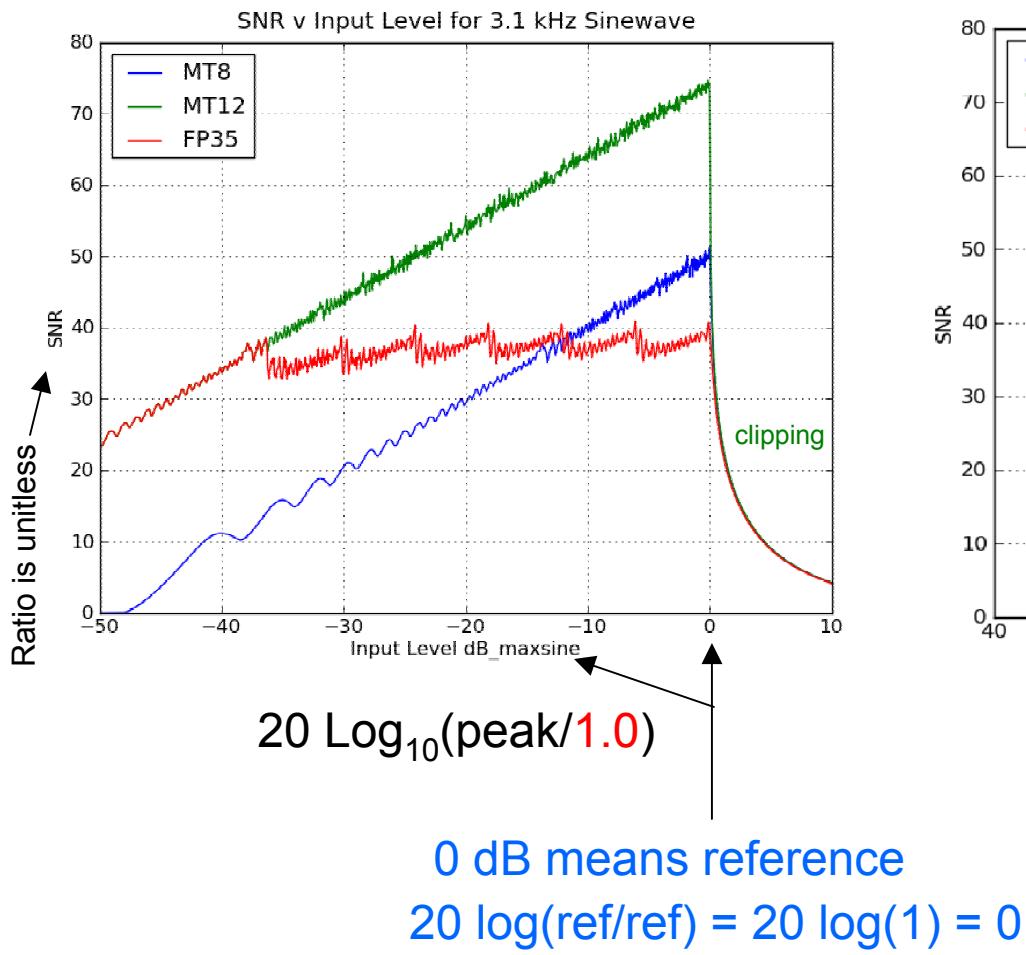
- Square root of the average power in a waveform (units in amplitude or pressure).

$\sin(2\pi t)$:

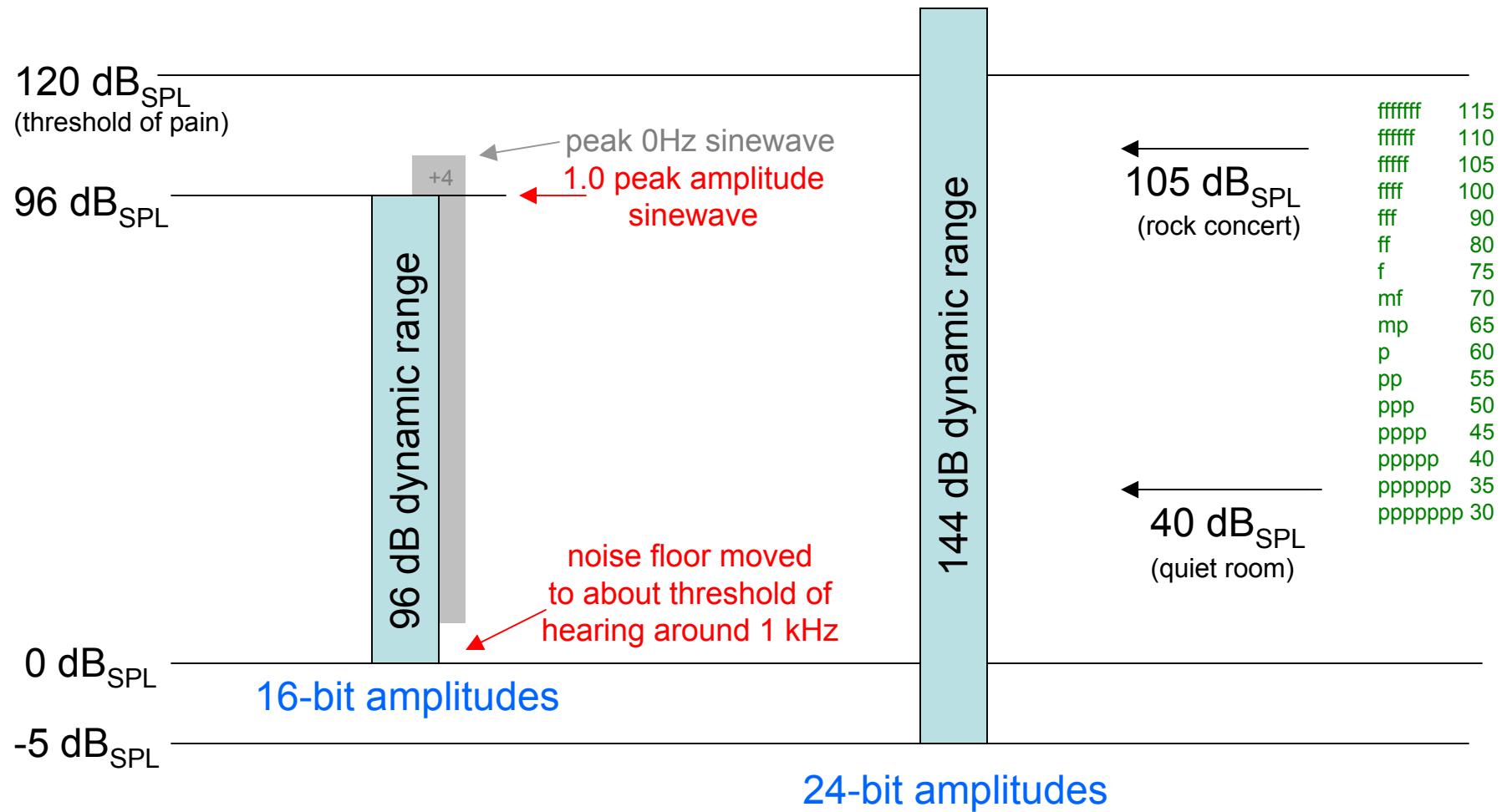


dB always in relation to a reference

- Good practice to identify reference as a subscript after “dB”.

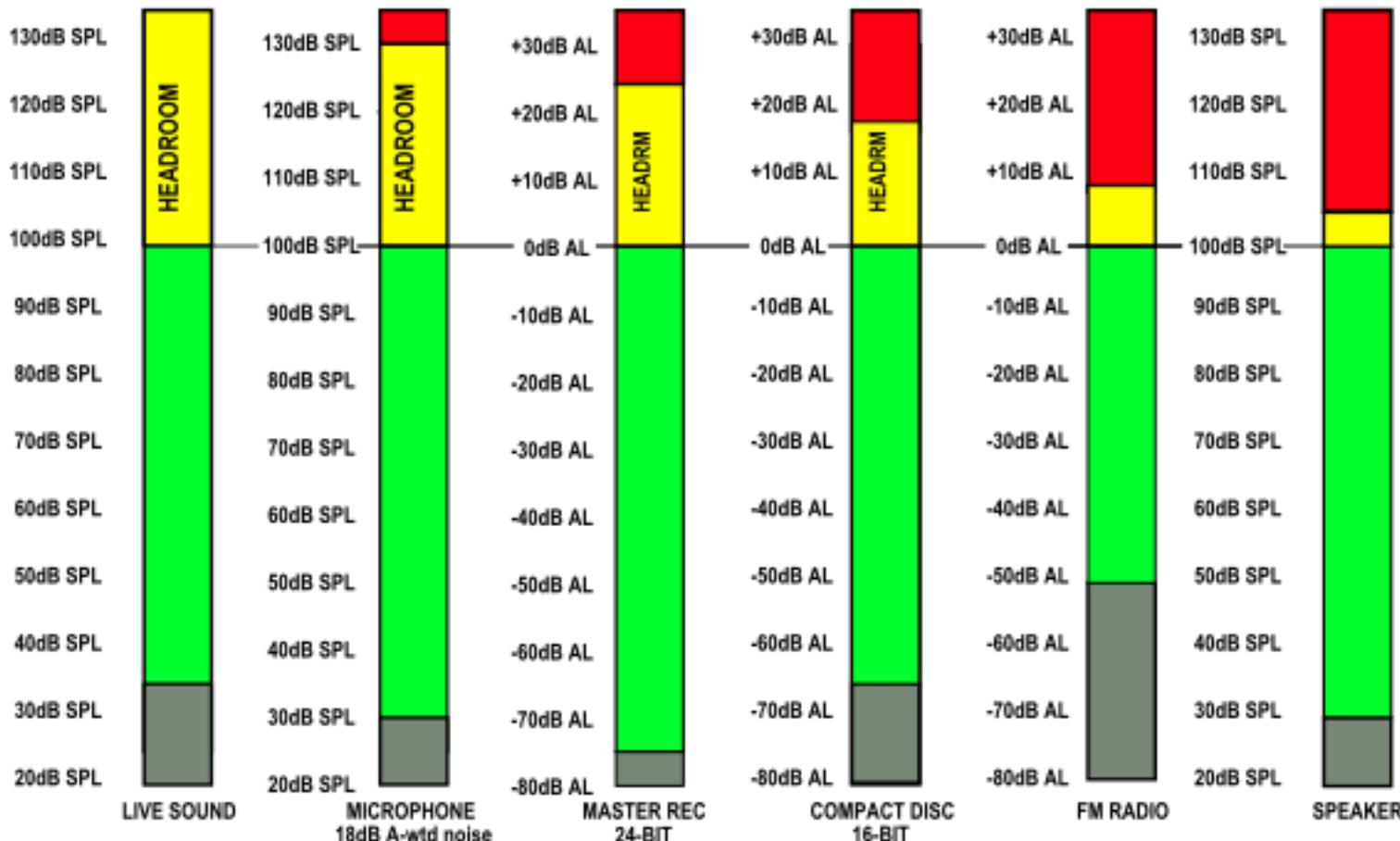


16-bit vs 24-bit dynamic range



Dynamic Ranges

http://en.wikipedia.org/wiki/Headroom_%28audio_signal_processing%29



PROGRAMME LEVEL CAPABILITY AT TYPICAL STAGES OF THE AUDIO PROCESS

LEVELS are Peak RMS Equivalent

NOISE is measured ITU-R 468 weighted for true subjective validity

MASTER recording assumes 24-bit with 24dB of headroom assigned (alternative EBU standard)

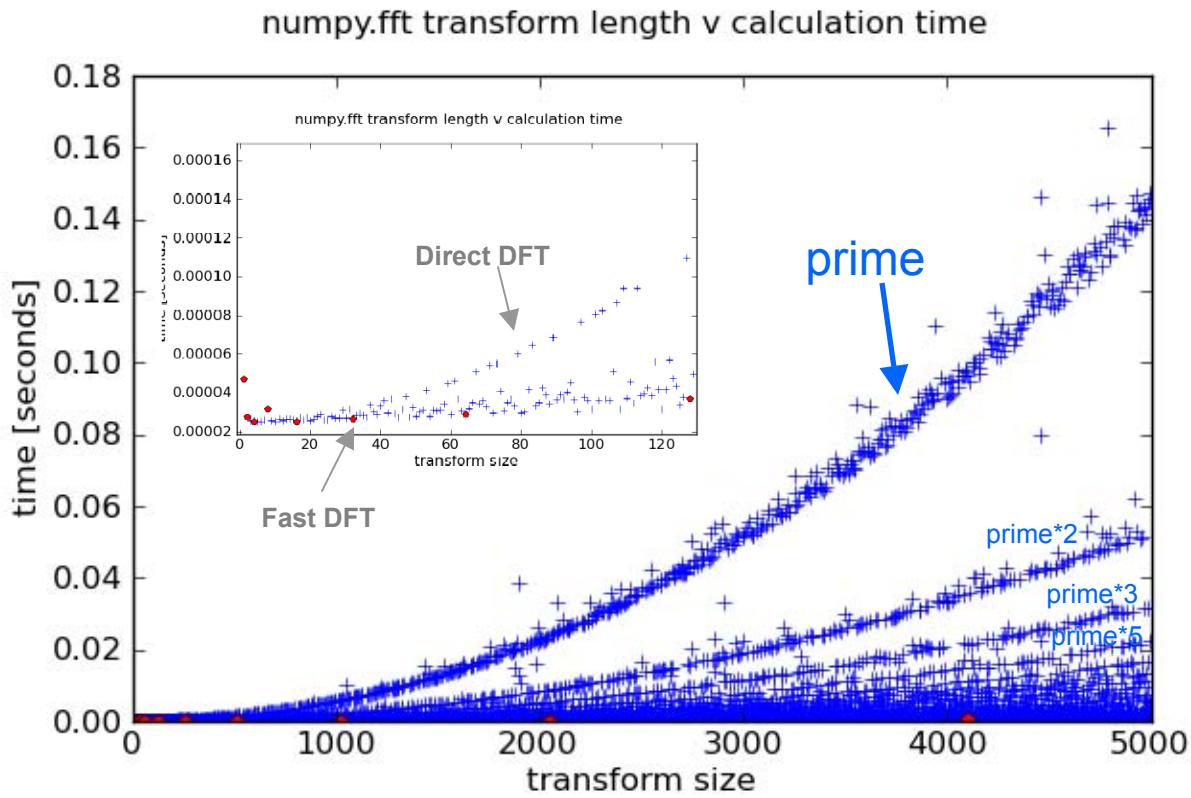
SPEAKERS are typical hi-end 100W per channel 86dB for 1W sensitivity, pair at 2m in listening room



NOT AVAILABLE
HEADROOM
NORMAL RANGE
NOISE

NOTE HOW, WITH AN ASSUMED ALIGNMENT LEVEL OF 100dB SPL, (NEEDED FOR REALISTIC LEVELS) MOST SYSTEMS CANNOT COPE

FFT speed

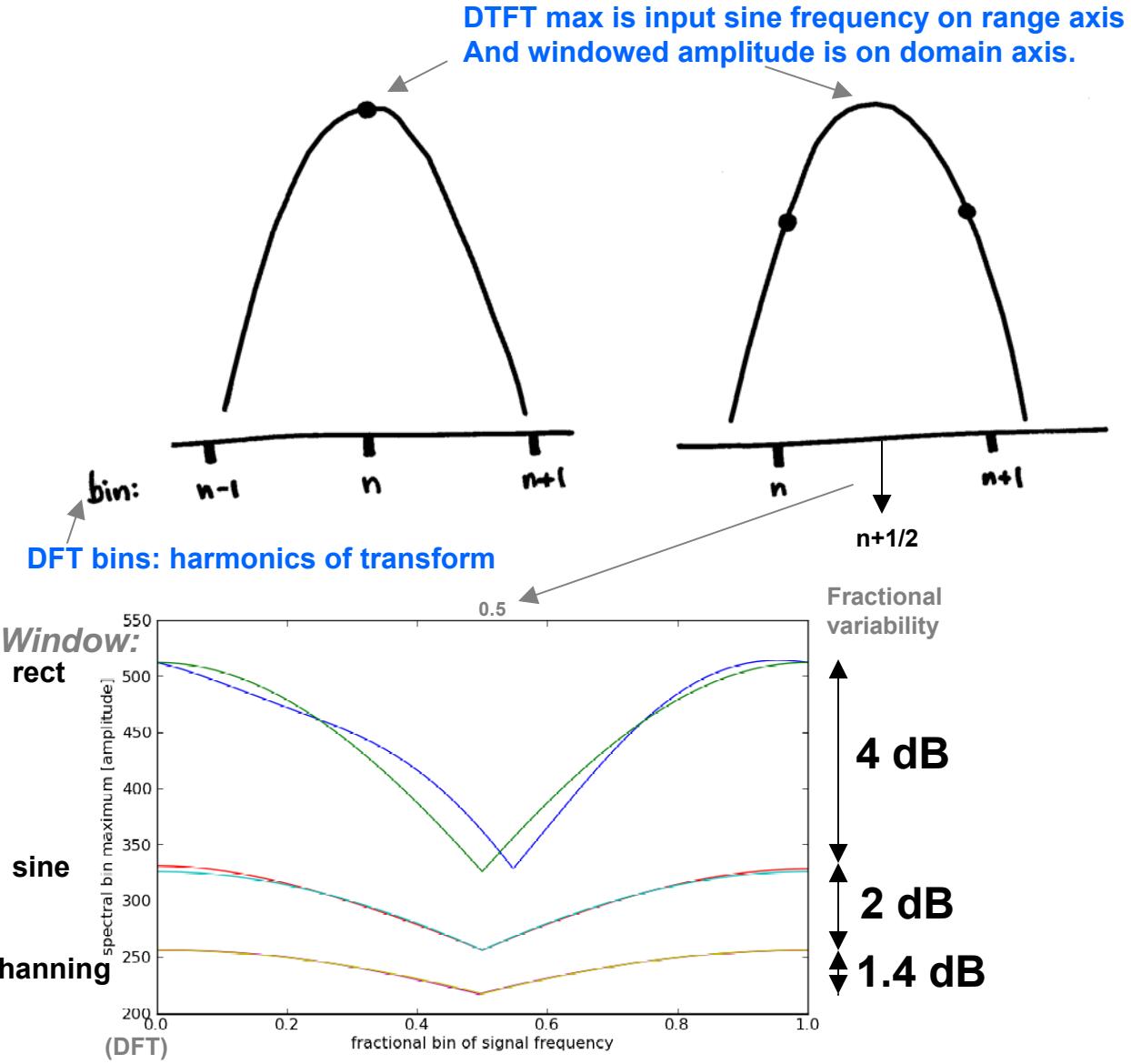


```
import pylab as pl
import numpy as np
import numpy.fft as npf
import time
N = 5000
domain = np.arange(1,N+1)
range = np.zeros(N)
dummy = np.zeros(N)
pow2i = [2**n-1 for n in xrange(1 \
+int(np.log2(N)))]
```

```
for i in xrange(N):
    time1 = time.clock()
    dummy = npf.fft(np.arange(domain[i]))
    time2 = time.clock()
    range[i] = time2 - time1
```

```
pl.plot(domain, range, 'b+')
pl.plot(domain[pow2i], range[pow2i], 'rp')
pl.xlabel('transform size')
pl.ylabel('time [seconds]')
pl.suptitle('numpy.fft length v calculation time')
pl.show()
```

Fractional bin sinewave



```

import pylab as pl
import numpy as np
import numpy.fft as npf
N = 1024
from craigwindow import *
sinwin = SineWindow(N)
hanwin = HanningWindow(N)

lowrange = np.zeros(N+1)
lowrangesin = np.zeros(N+1)
lowrangehan = np.zeros(N+1)
for i in xrange(N+1):
    sig = np.sin(2 * np.pi * (2.0+1.0*i/N) * np.arange(N) / N)
    spec = npf.fft(sig)
    absspec = np.abs(spec)
    lowrange[i] = max(absspec)
    spec = npf.fft(sig*sinwin)
    absspec = np.abs(spec)
    lowrangesin[i] = max(absspec)
    spec = npf.fft(sig*hanwin)
    absspec = np.abs(spec)
    lowrangehan[i] = max(absspec)

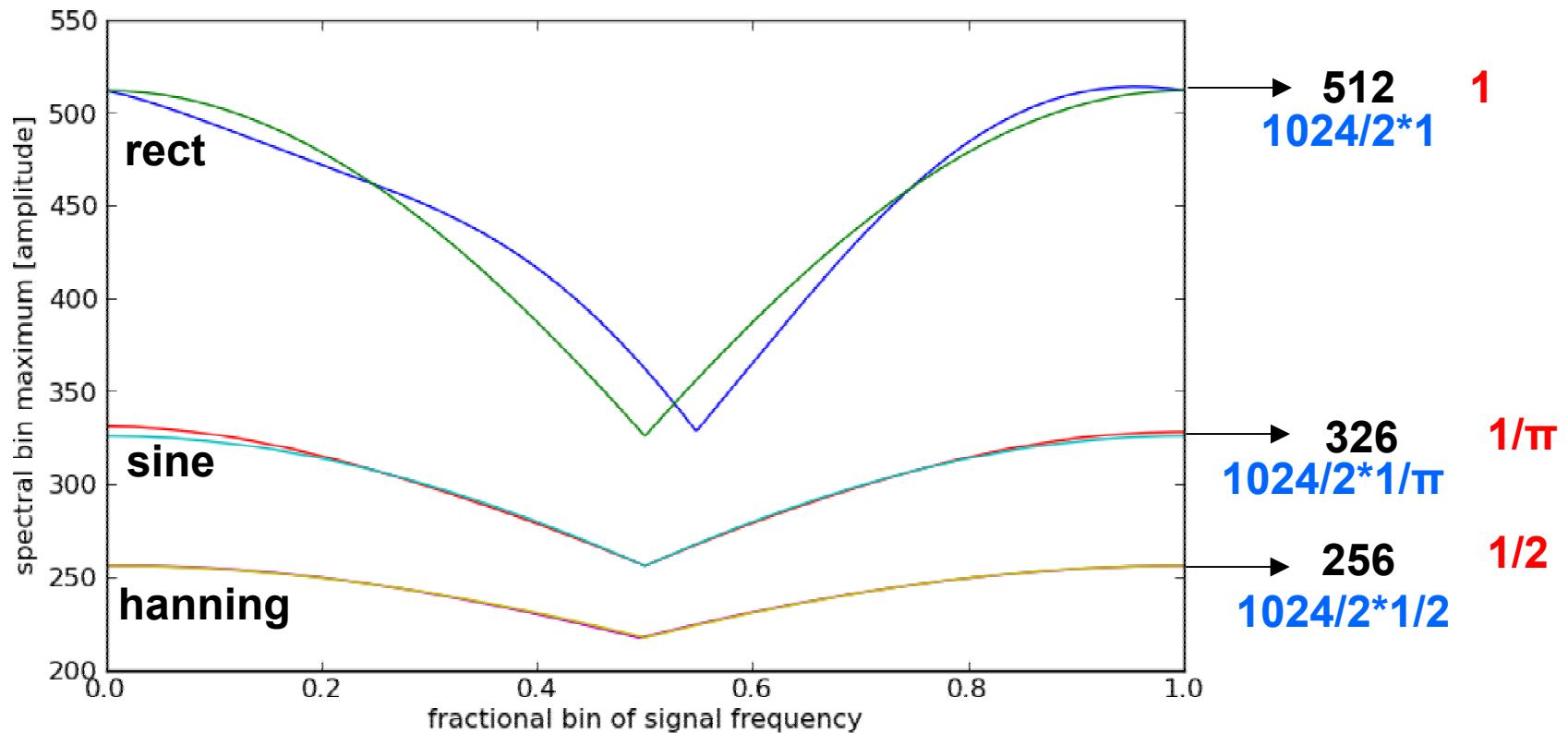
hirange = np.zeros(N+1)
hirangesin = np.zeros(N+1)
hirangehan = np.zeros(N+1)
for i in xrange(N+1):
    sig = np.sin(2 * np.pi * (N/4+1.0*i/N) * np.arange(N) / N)
    spec = npf.fft(sig)
    absspec = np.abs(spec)
    hirange[i] = max(absspec)
    spec = npf.fft(sig*sinwin)
    absspec = np.abs(spec)
    hirangesin[i] = max(absspec)
    spec = npf.fft(sig*hanwin)
    absspec = np.abs(spec)
    hirangehan[i] = max(absspec)

pl.clf()
pl.plot(np.arange(N+1.0)/N,lowrange,
        np.arange(N+1.0)/N,hirange,
        np.arange(N+1.0)/N,lowrangesin,
        np.arange(N+1.0)/N,hirangesin,
        np.arange(N+1.0)/N,lowrangehan,
        np.arange(N+1.0)/N,hirangehan)
pl.xlabel("fractional bin of signal frequency")
pl.ylabel("spectral bin maximum [amplitude]")
pl.show()

```

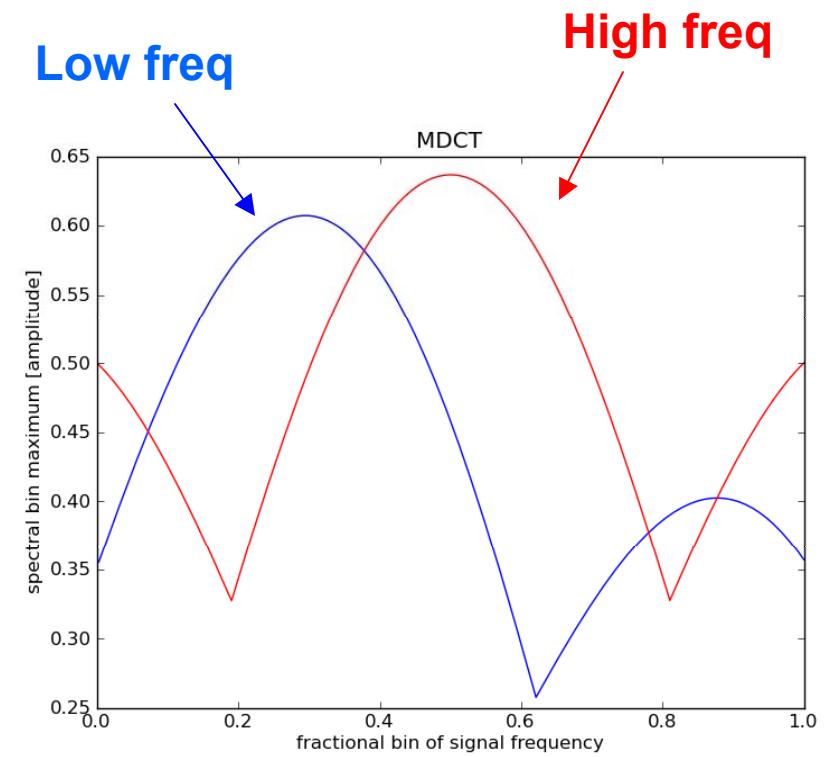
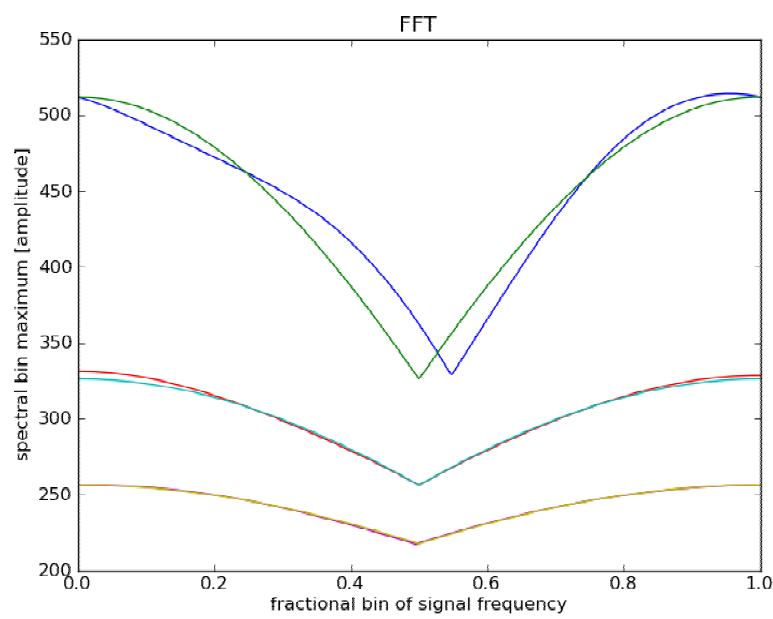
FFT Sine Amplitude v Window

$N=1024$



Spectral |amplitude| is sine peak amplitude/2 * integration of signal window
(signal window normalized by rectangular window)

MDCT Sine Amplitude

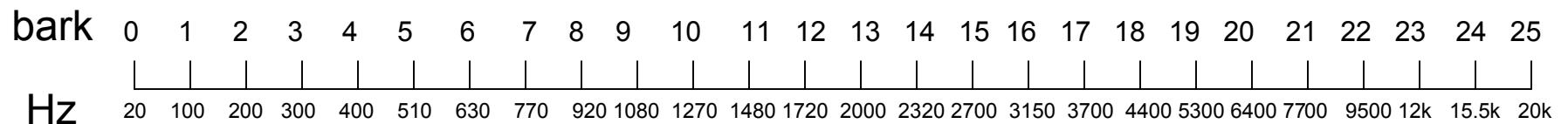
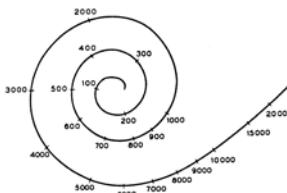


Maximum signal with sinwindow
Generate $1/\sqrt{2}$ magnitude

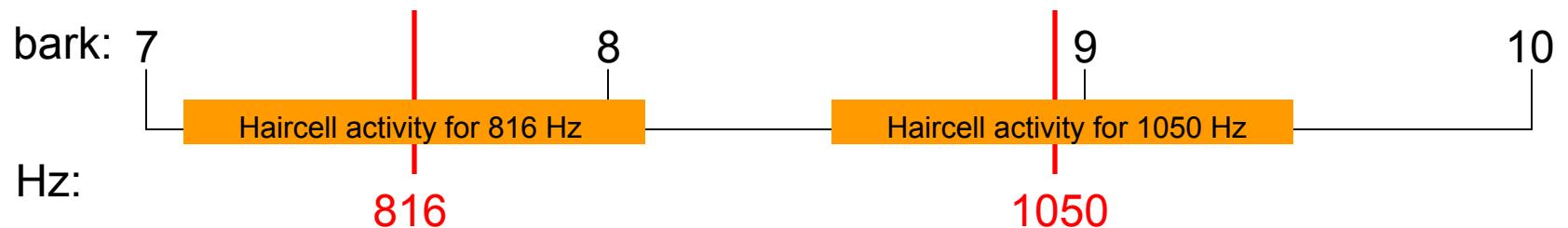
Bark Scale

- Bark scale is a perceptual frequency mapping expressing how humans hear.
- Approximately equivalent to the spatial position on the basilar membrane.

(note inner hair cells don't have equal density along basilar membrane, so only an approximation of distance)

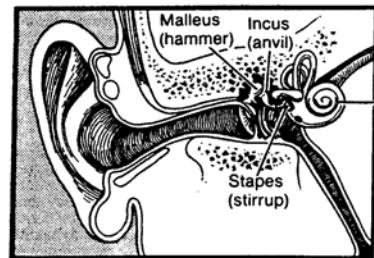


- Difference of one bark is a rough estimate of a critical band which is an estimate how many hair cells are activated by a sinewave (ignoring amplitude variations).

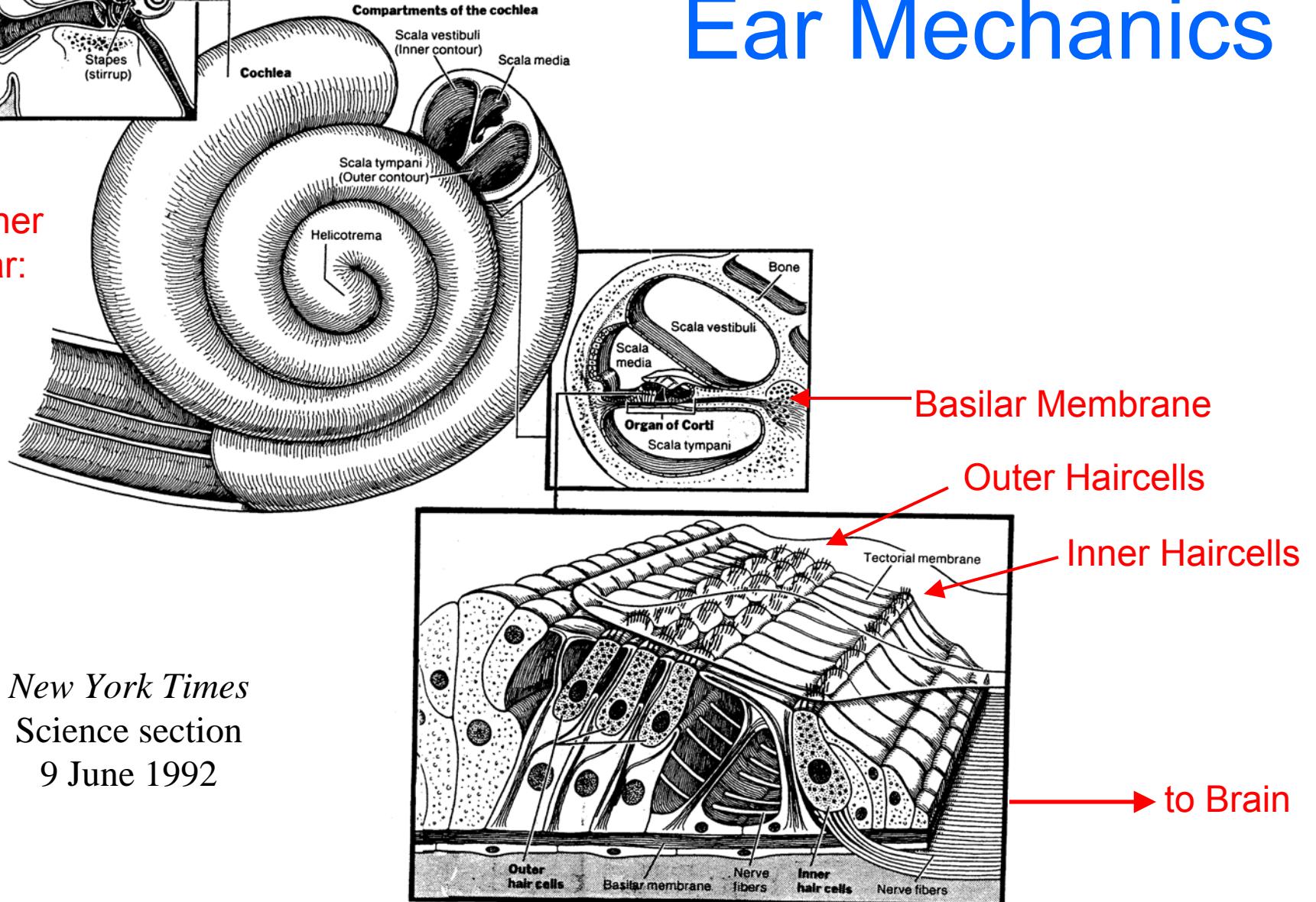


- 3,500 inner haircells = about 140 cells /critical band.

Ear Mechanics

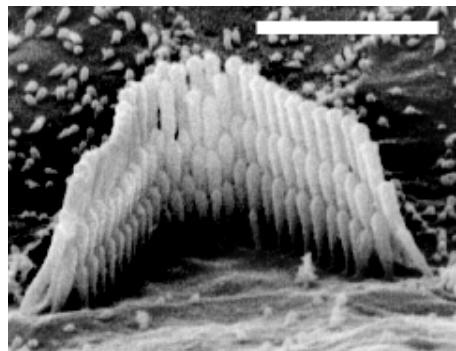


Inner
Ear:

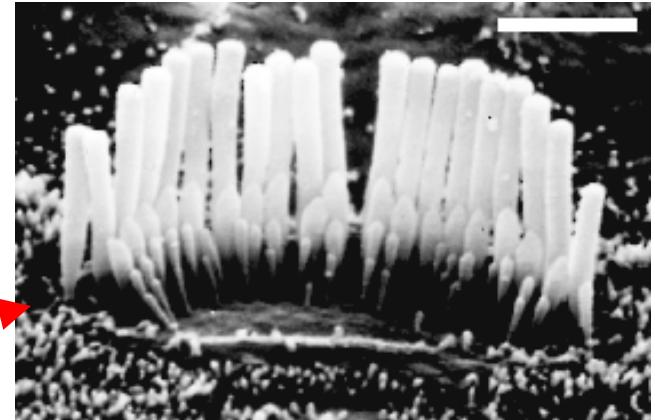


New York Times
Science section
9 June 1992

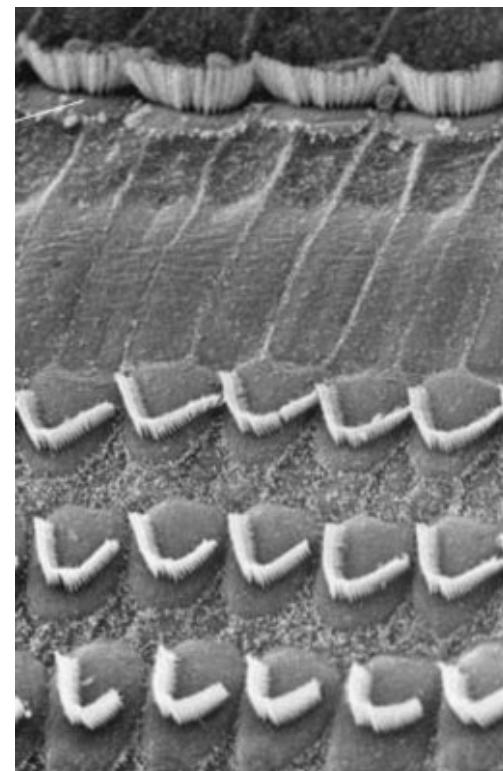
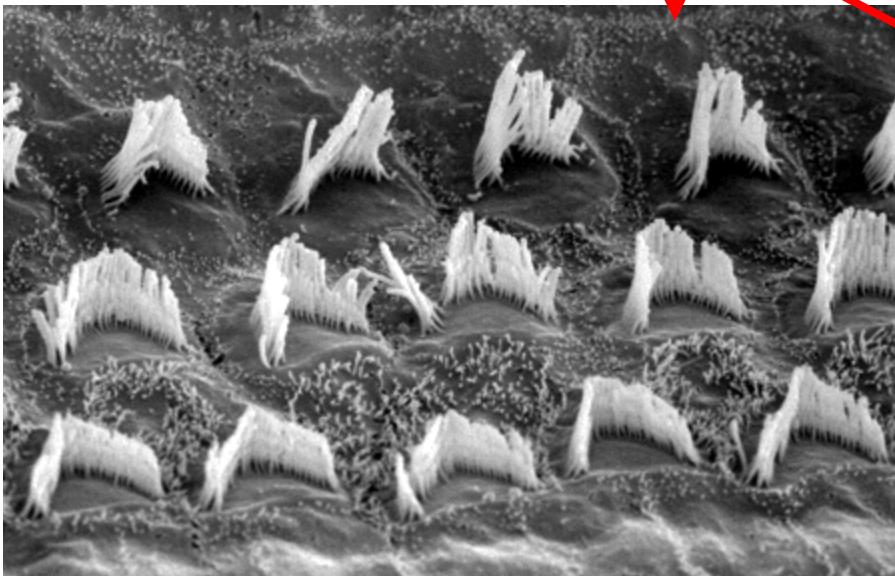
Ear Mechanics (2)



Inner Hair Cells:
each analogous to
FFT/MDCT bin

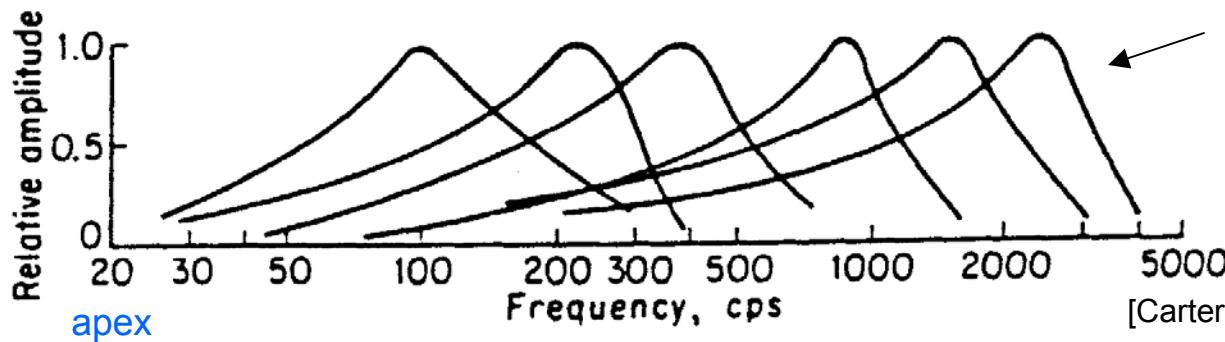
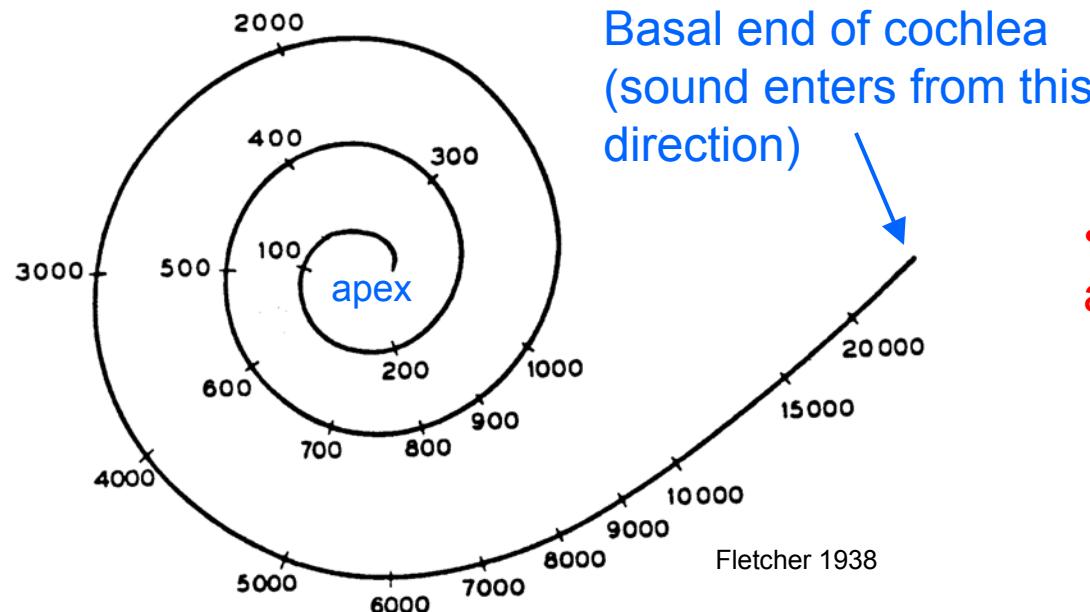


Outer Hair Cells:
resolution
enhancement



Ear Mechanics (3)

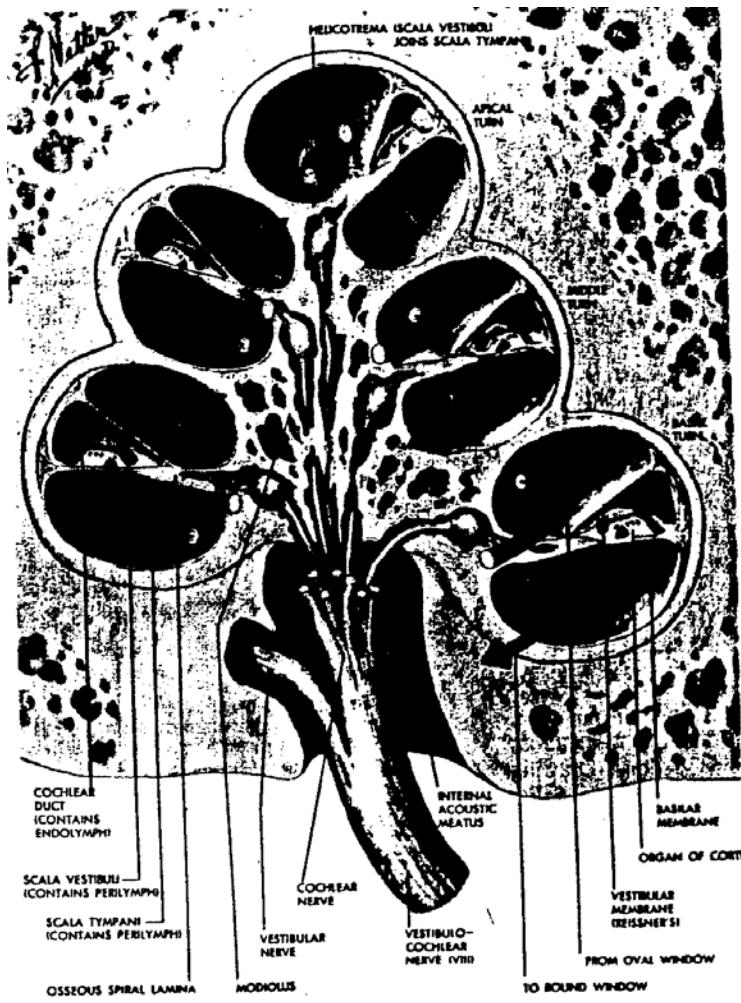
- About 80 inner hair cells per mm at the basal end and 155 cells per millimeter at the apical end [Bredberg 1968]



From *Experiments in Hearing*, Békésy 1960

[Carterette: Handbook of Perception: IV Hearing]

More Schematics of Cochlea



[Frank Netter]

[Stevens p. 1118]

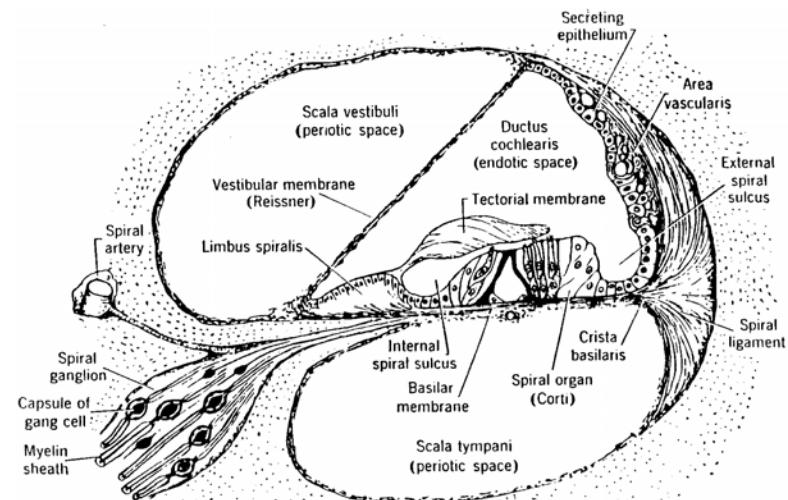


FIG. 1. Diagrammatic cross section of a cochlear canal. The ductus cochlearis (or scala media) contains the organ of Corti with its hair cells, the ultimate end organs of hearing. (From Rasmussen, 1943.)

[Stevens p. 1117]

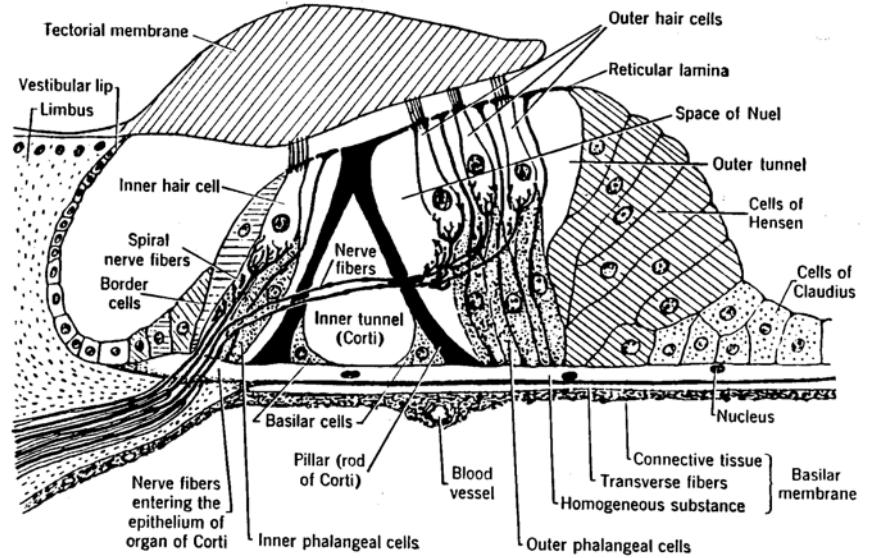
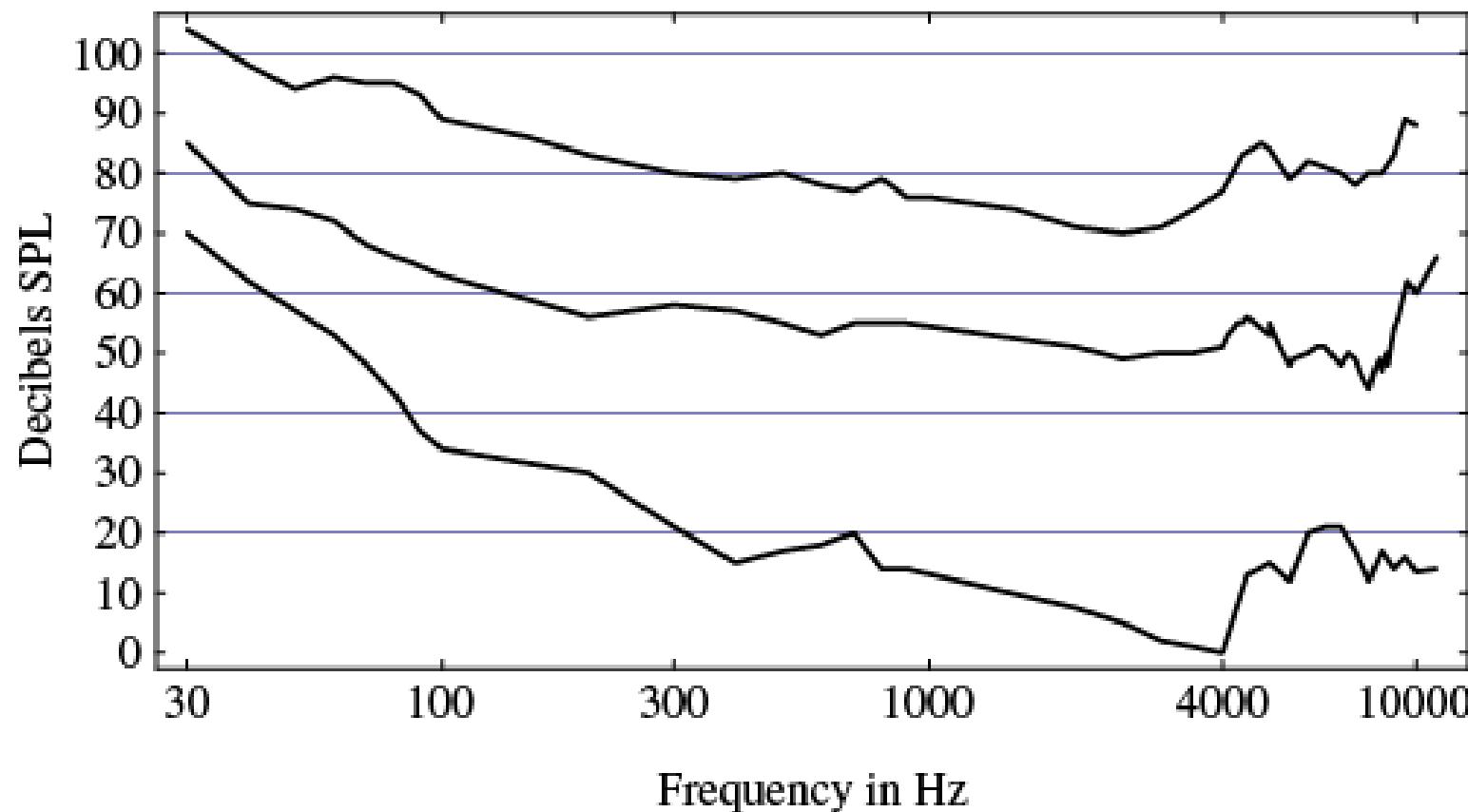


FIG. 2. Diagrammatic cross section of the organ of Corti. The outer hair cells are supported by their respective phalangeal cells, which rest in turn on the movable basilar membrane. The phalangeal cells supporting the inner hair cells rest on bone. Motion of the basilar membrane presumably distorts the hair cells. (From Rasmussen, 1943.)

Equal Loudness Curves

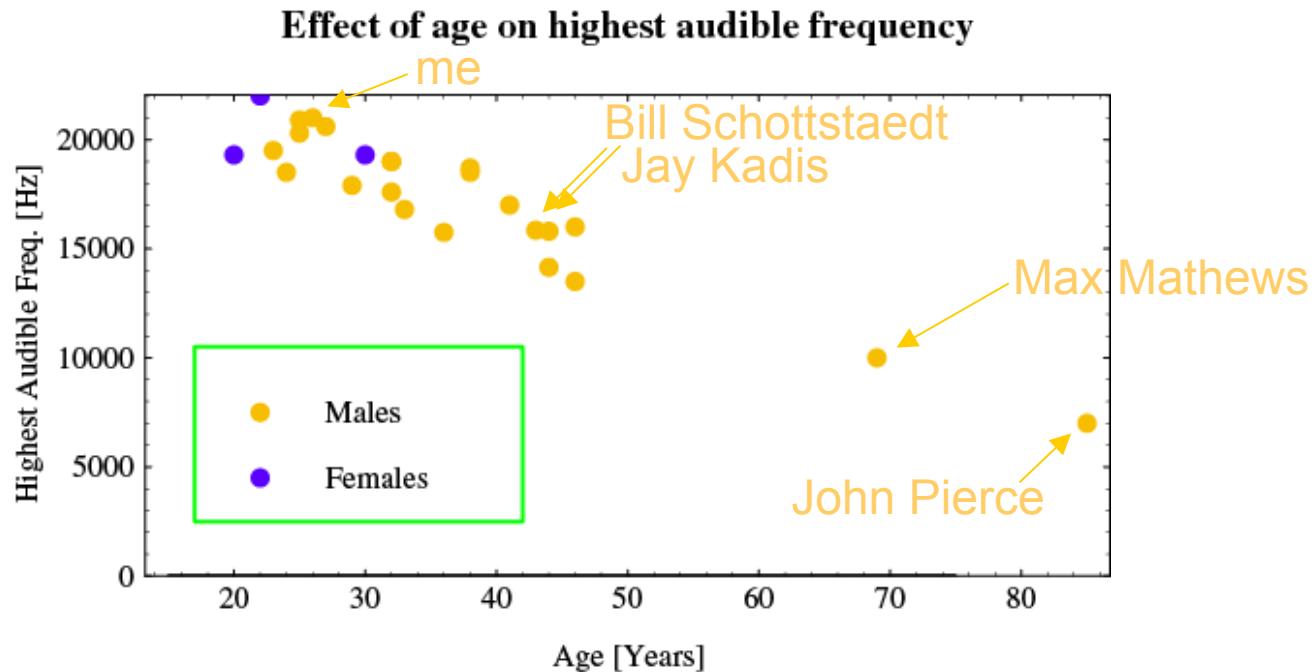
Fletcher-Munson Curve of Equal Loudness for Craig



<http://ccrma.stanford.edu/CCRMA/Courses/SummerWorkshops/96/Psychoacoustics/labs/loudness>

- Project idea: Use your own threshold of hearing in the masking model.

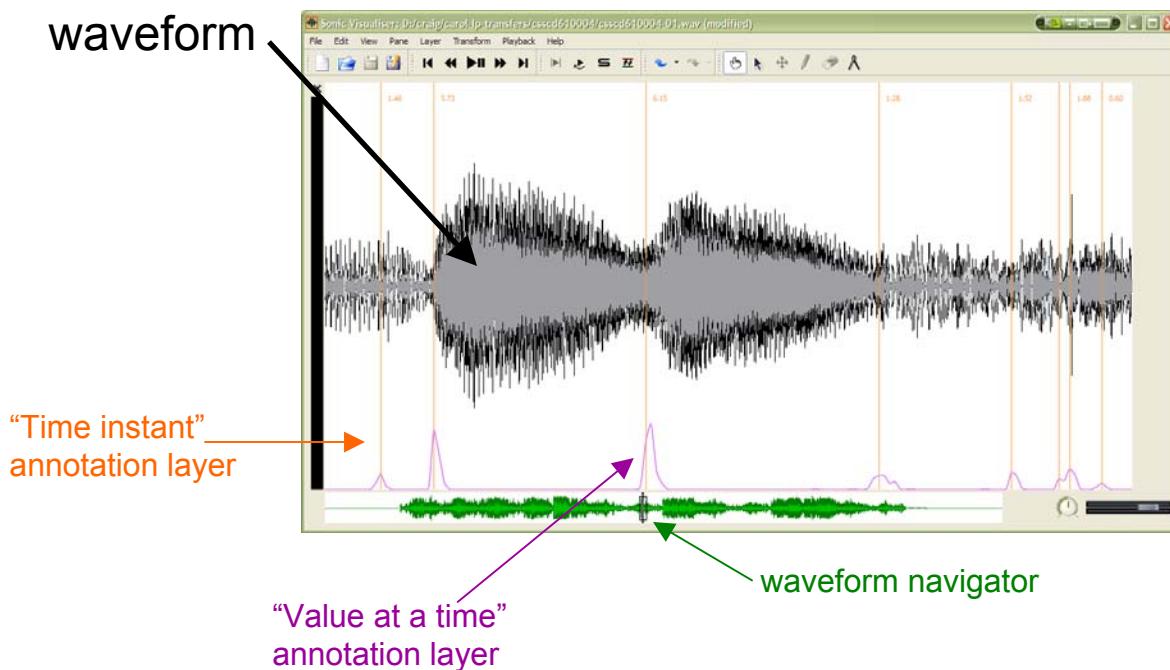
Highest Audible Frequency



- <http://ccrma.stanford.edu/CCRMA/Courses/SummerWorkshops/96/Psychoacoustics/labs/loudness>
- http://www.bbc.co.uk/wiltshire/content/articles/2006/04/04/mosquito_sound_wave_feature.shtml
- About 5% of <20 year olds can hear up to 25 kHz.
- Easy to do high-compression ratios for Senior Citizens.

SONIC VISUALISER

- Audio annotation program: <http://www.sonicvisualiser.org/>

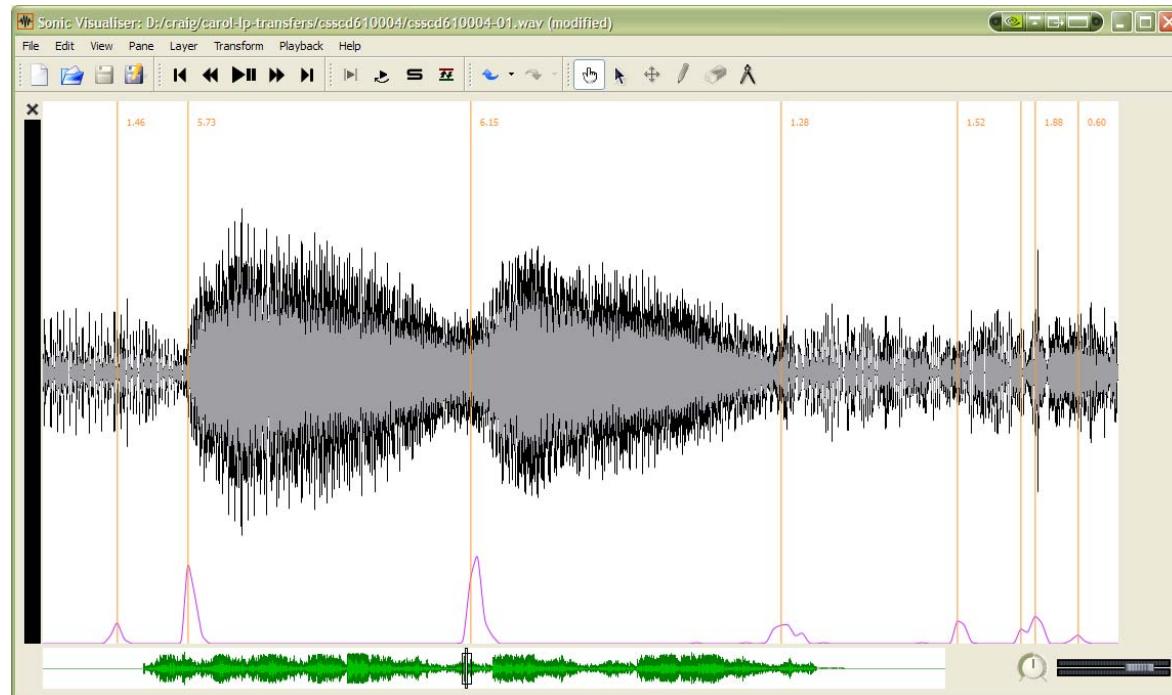


- **Vamp Plugins:** audio analysis plugins in C++ for Sonic Visualiser
<http://www.vamp-plugins.org>
- **VamPy:** Python interface to Vamp plugins for Sonic Visualiser
<http://www.vamp-plugins.org/vampy.html>

Spectral Reflux Plugin

<http://sv.mazurka.org.uk/download>
(linux and windows)

- Useful for a window-switching project



Peaks (onset times):

Peak Onset Time
58.318208616 5.73
58.718208616 6.15
59.158208616 1.28
59.408208616 1.52

Analysis (every 10 ms):

Time	Value
58.178208616	0
58.188208616	0
58.198208616	0
58.208208616	0.522996
58.218208616	1.46201
58.228208616	0.240144
58.238208616	0
58.248208616	0
58.258208616	0
58.268208616	0
58.278208616	0
58.288208616	0
58.298208616	0
58.308208616	0
58.318208616	5.72949
58.328208616	3.21772
58.338208616	0.4205
58.348208616	0
58.358208616	0
58.368208616	0
58.378208616	0
58.388208616	0
58.398208616	0
58.408208616	0
58.418208616	0
58.428208616	0
58.438208616	0
58.448208616	0
58.458208616	0
58.468208616	0
58.478208616	0
58.488208616	0
58.498208616	0
58.508208616	0
58.518208616	0