24.963 Linguistic Phonetics
Fall 2005

24.963

Linguistic Phonetics

Basic Audition

- Reading for next week: Keating 1985
- Optional reading: Flemming 2001
- Assignment 1 - basic acoustics. Due next week.

- Independence of source and filter

Audition

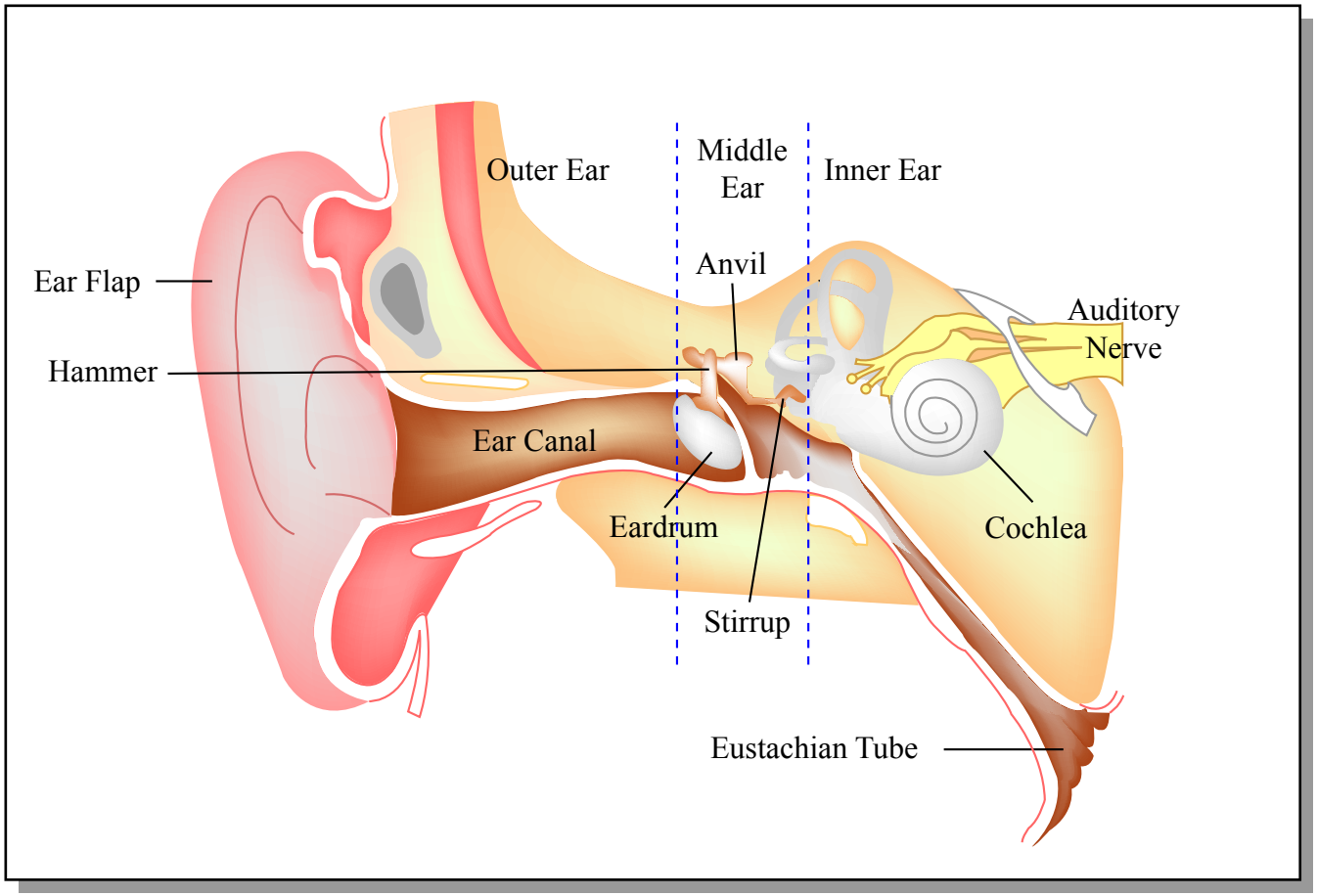


Figure by MIT OpenCourseWare.

Anatomy

Audition

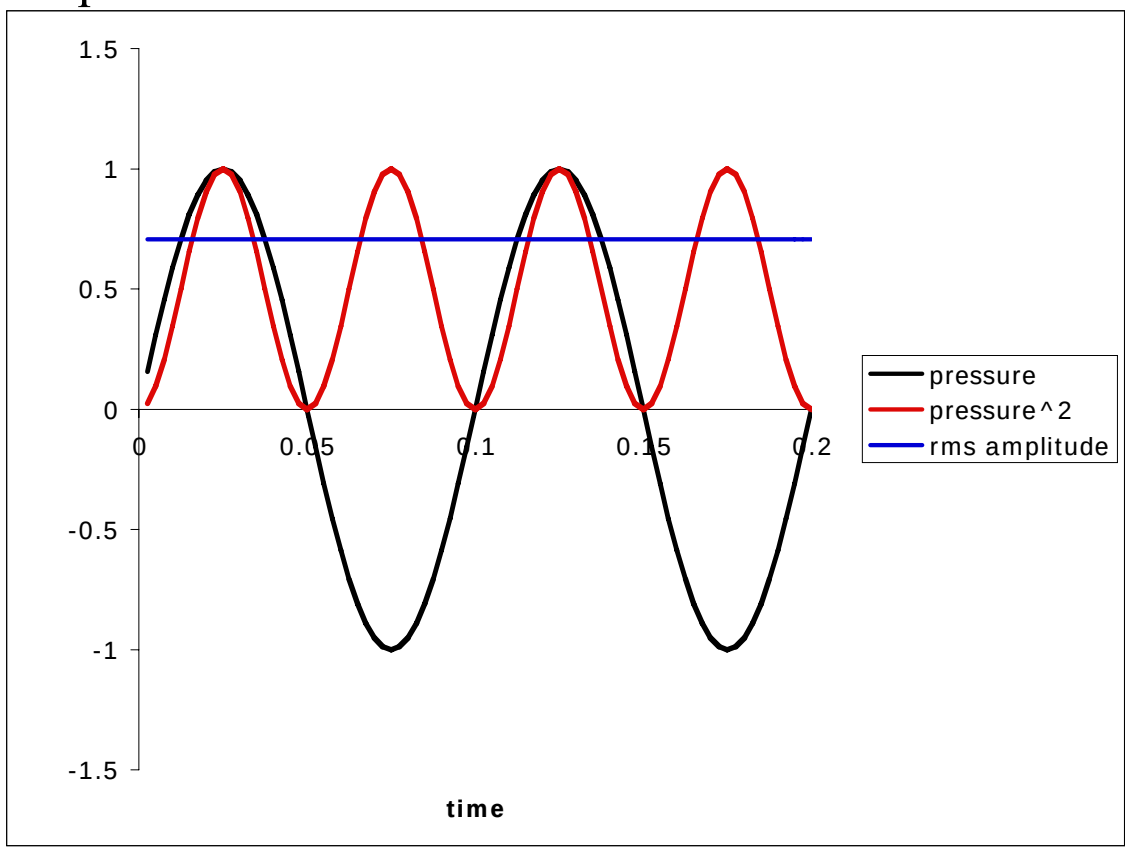
- Loudness
- Pitch
- ‘Auditory spectrograms’

Loudness

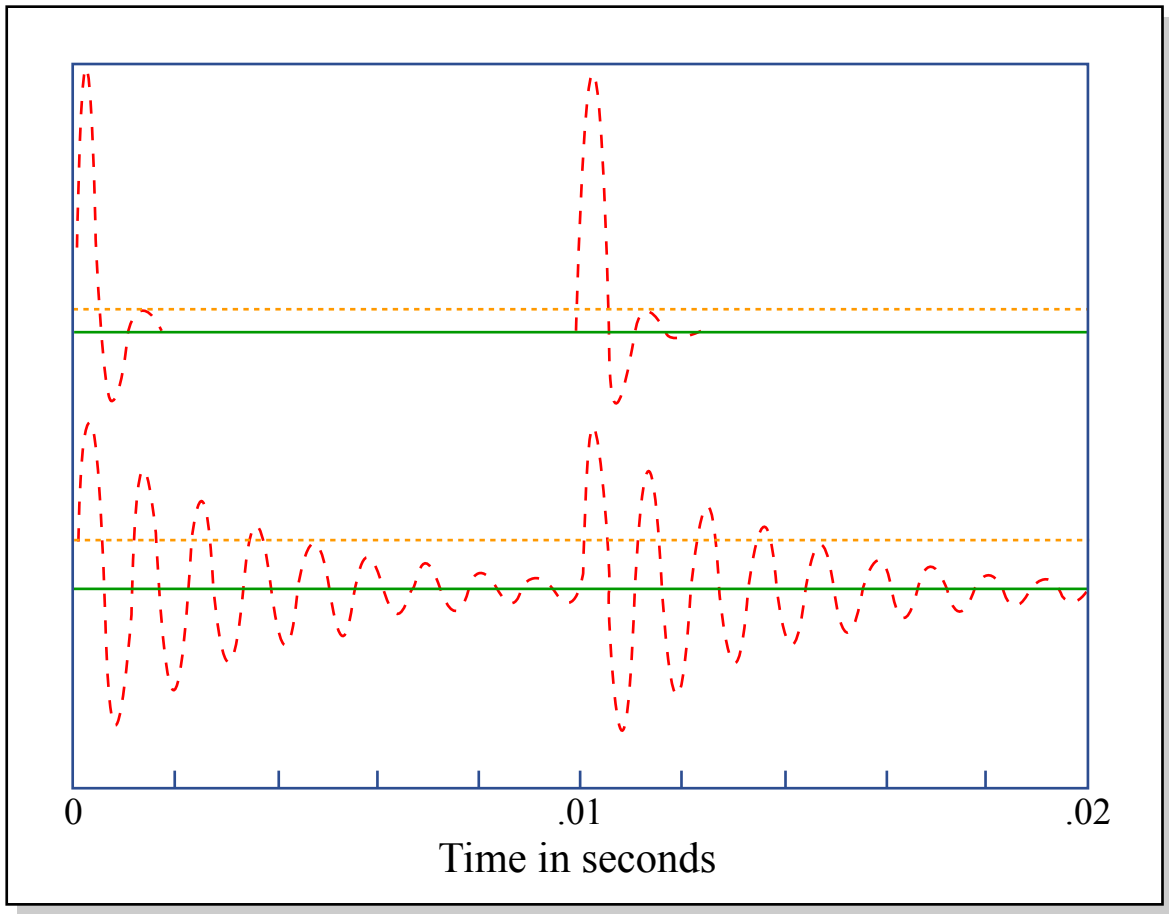
- The perceived loudness of a sound depends on the amplitude of the pressure fluctuations in the sound wave.
- Amplitude is usually measured in terms of root-mean-square (rms amplitude):
  - The square root of the mean of the squared amplitude over some time window.

rms amplitude

- Square each sample in the analysis window.
- Calculate the mean value of the squared waveform:
  - Sum the values of the samples and divide by the number of samples.
- Take the square root of the mean.



rms amplitude



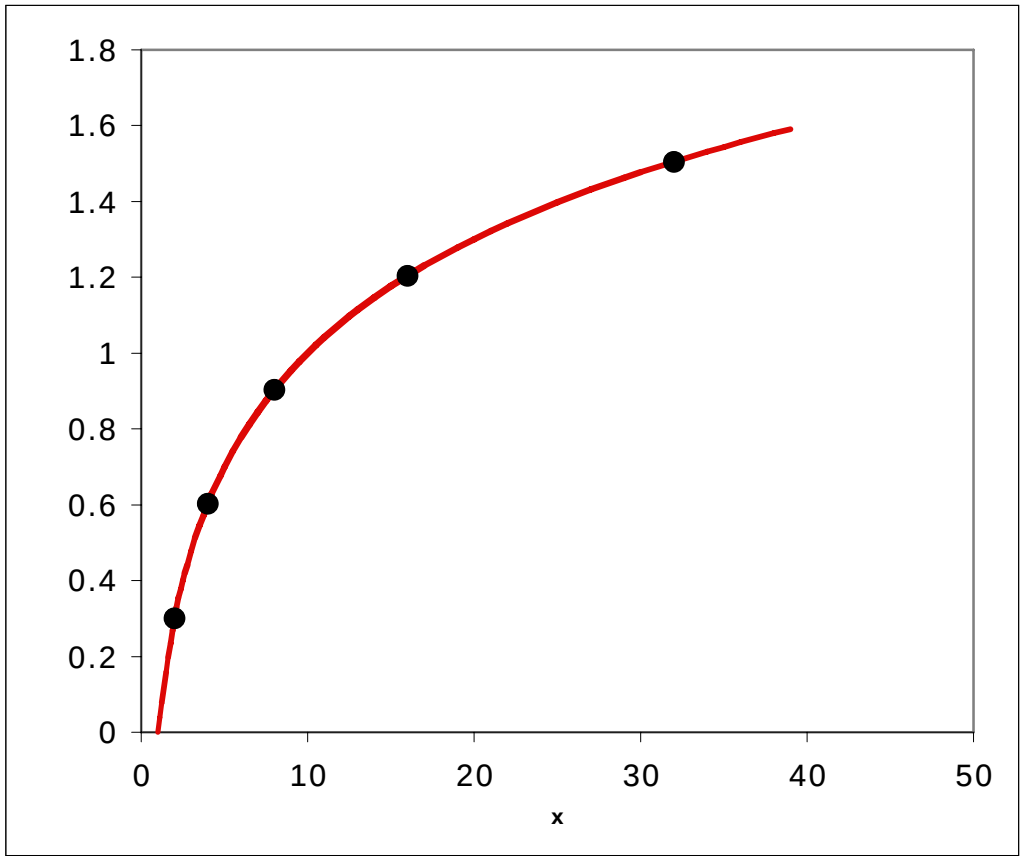
Intensity

- Perceived loudness is more closely related to intensity (power per unit area), which is proportional to the square of the amplitude.
- relative intensity in Bels = $\log_{10}(x^2/r^2)$
- relative intensity in dB = $10 \log_{10}(x^2/r^2)$
  = $20 \log_{10}(x/r)$

- In absolute intensity measurements, the comparison amplitude is usually 20μPa, the lowest audible pressure fluctuation of a 1000 Hz tone (dB SPL).

logarithmic scales

- $\log x^n = n \log x$



Loudness

- The relationship between intensity and perceived loudness is not exactly logarithmic.

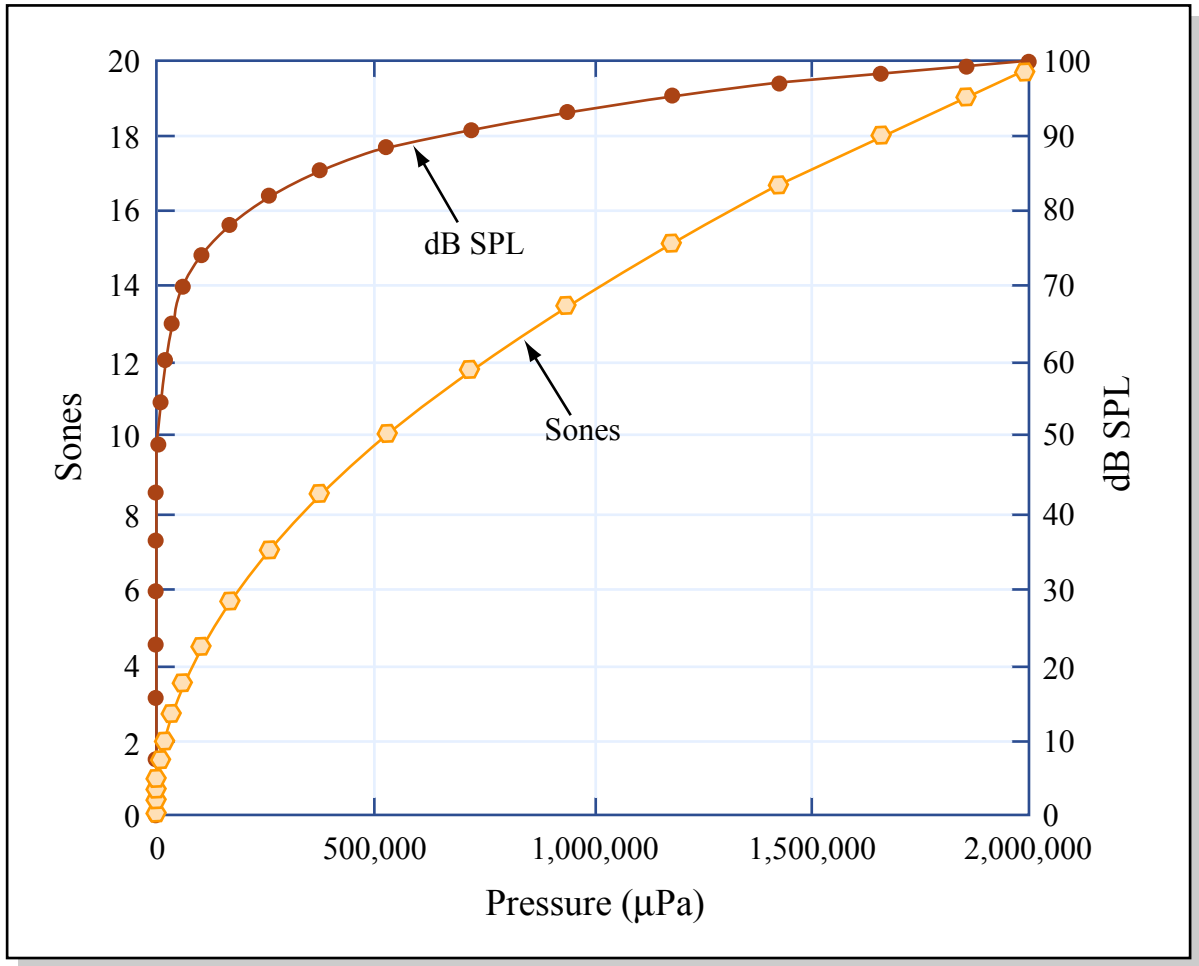
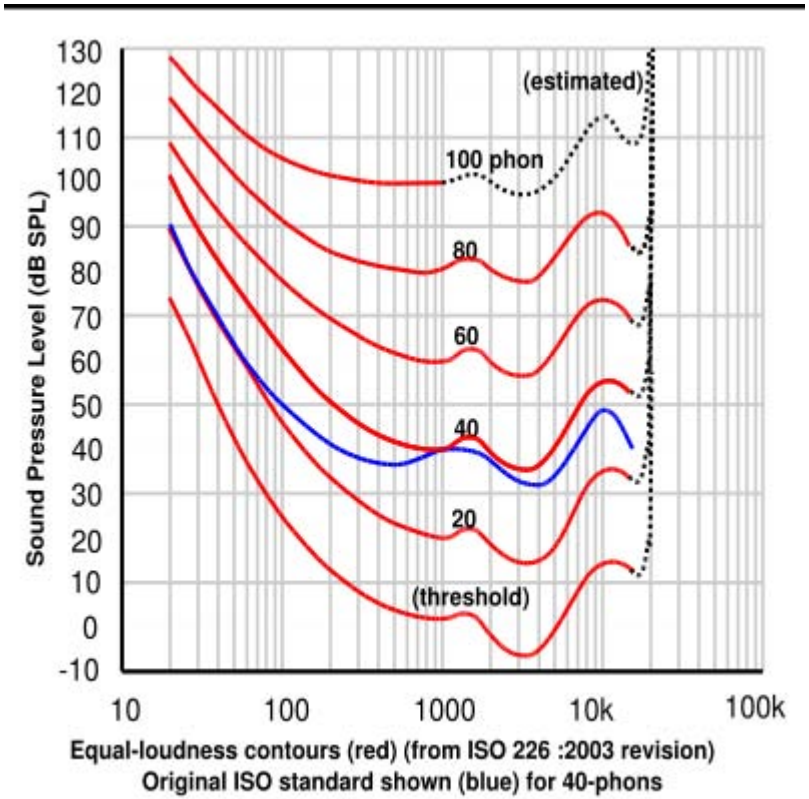


Figure by MIT OpenCourseWare. Adapted from Johnson, Keith. *Acoustic and Auditory Phonetics*. Malden, MA: Blackwell Publishers, 1997. ISBN: 9780631188483.

Loudness

- Loudness also depends on frequency.
- equal loudness contours for pure tones:



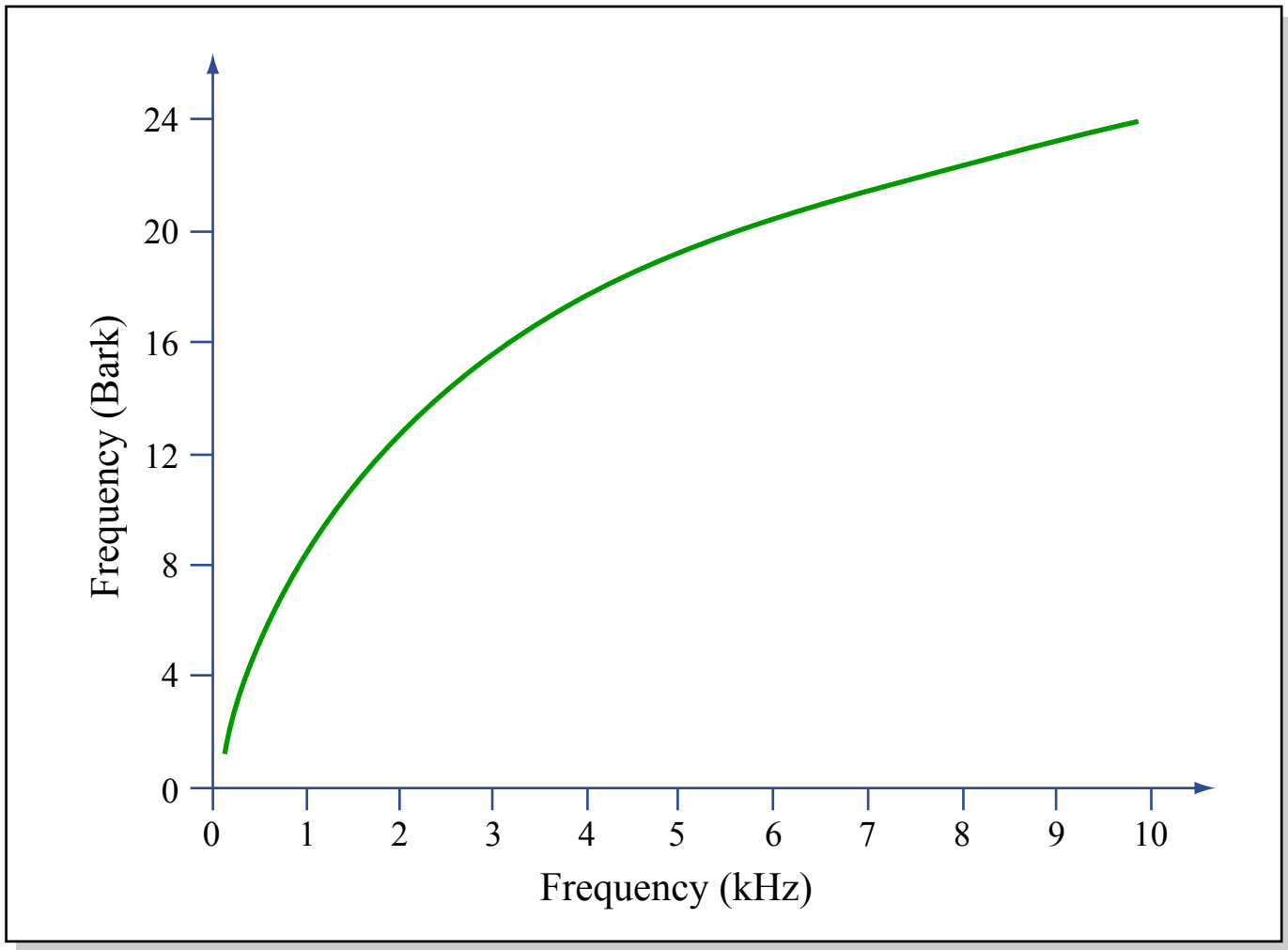
Source: Wikimedia Commons.

Loudness

- At short durations, loudness also depends on duration.
- Temporal integration: loudness depends on energy in the signal, integrated over a time window.
- Duration of integration is often said to be about 200ms, i.e. relevant to the perceived loudness of vowels.

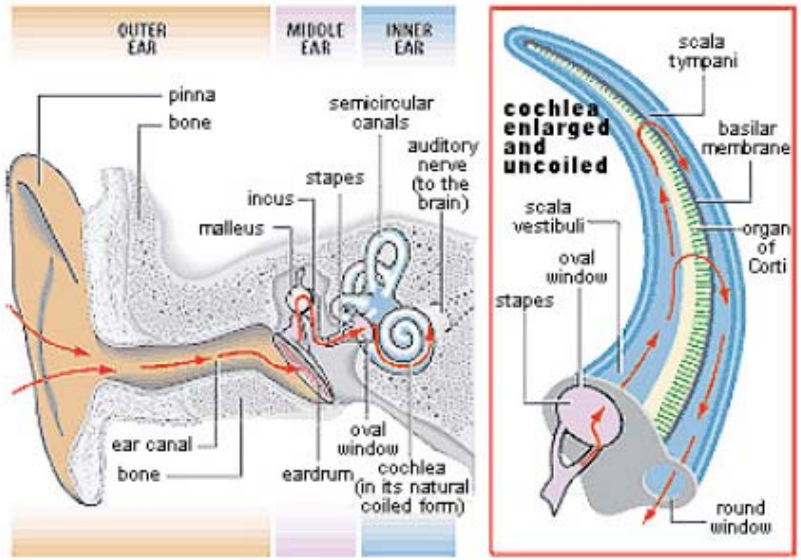
Pitch

- Perceived pitch is approximately linear with respect to frequency from 100-1000 Hz, between 1000-10,000 Hz the relationship is approximately logarithmic.



Pitch

- The non-linear frequency response of the auditory system is related to the physical structure of the basilar membrane.
- basilar membrane 'uncoiled':



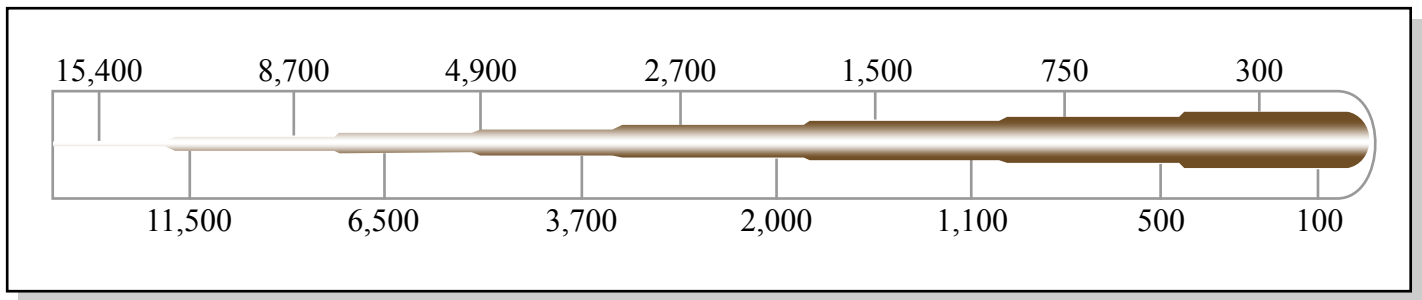
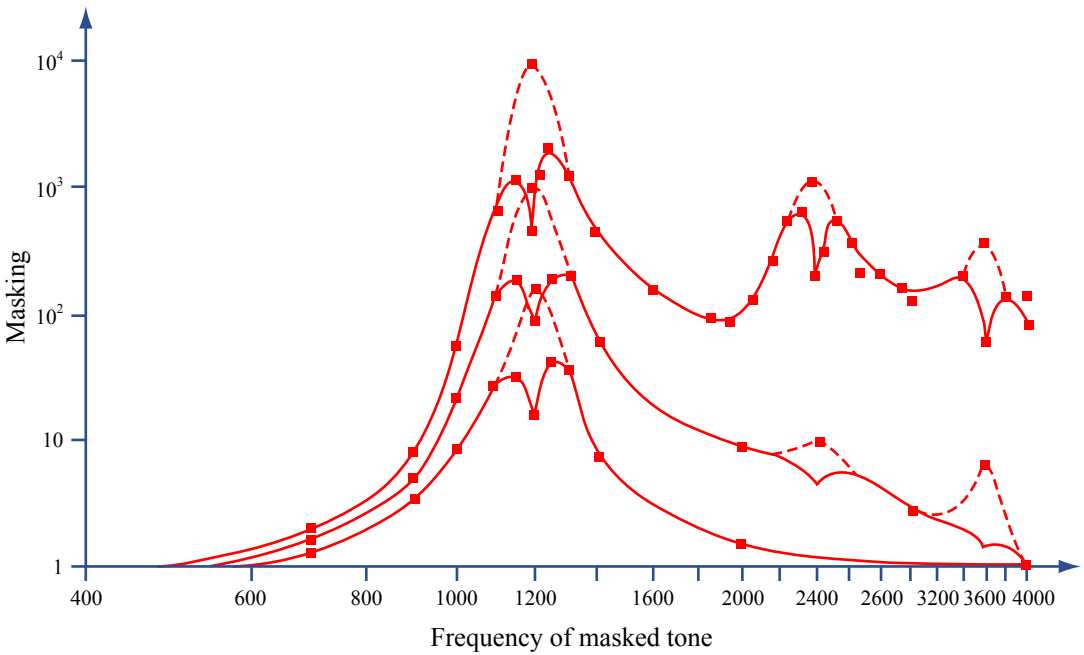


Figure by MIT OpenCourseWare. Adapted from Johnson, Keith. *Acoustic and Auditory Phonetics*. Malden, MA: Blackwell Publishers, 1997. ISBN: 9780631188483.

Masking - simultaneous

- Energy at one frequency can reduce audibility of simultaneous energy at another frequency (masking).
- One sound can also mask a preceding or following sound.



Example of masking of a tone by a tone. The frequency of the masking tone is 1200 Hz. Each curve corresponds to a different masker level, and gives the amount by which the threshold intensity of the masked tone is multiplied in the presence of the masker, relative to its threshold in quiet. The dashed lines near 1200 Hz and its harmonics are estimates of the masking functions in the absence of the effect of beats.

Figure by MIT OpenCourseWare. Adapted from Stevens, Kenneth N. *Acoustic Phonetics*. Cambridge, MA: MIT Press, 1999. ISBN; 9780262194044.

Time course of auditory nerve response

Response to a noise burst:

- Strong initial response
- Rapid adaptation (~5 ms)
- Slow adaptation (>100ms)
- After tone offset, firing rate only gradually returns to spontaneous level.

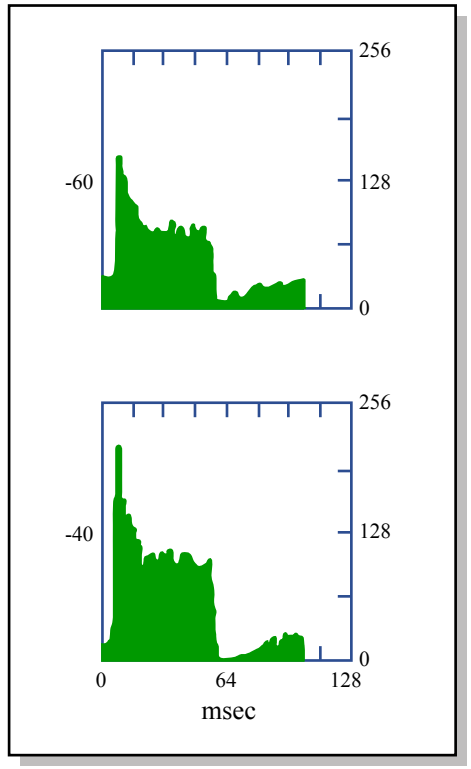


Figure by MIT OpenCourseWare. Adapted from Kiang et al. (1965)

Interactions between sequential sounds

- A preceding sound can affect the auditory nerve response to a following tone (Delgutte 1980).

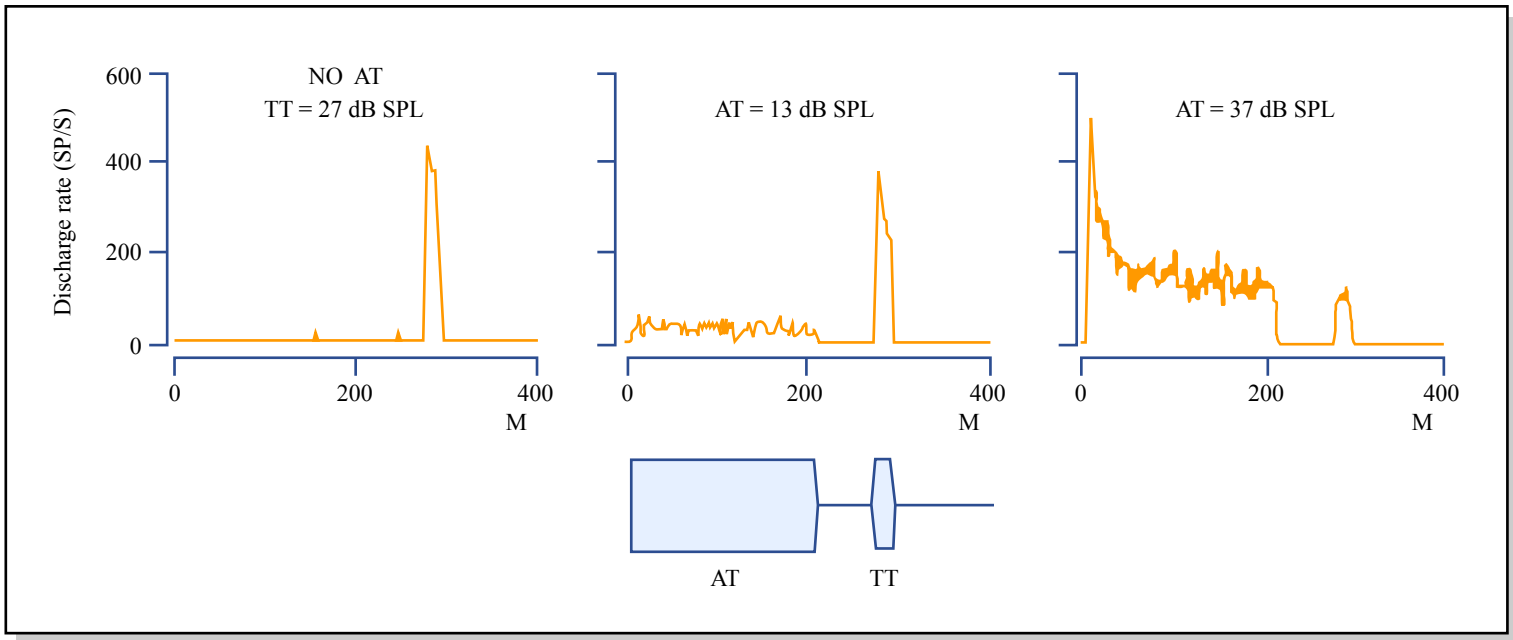
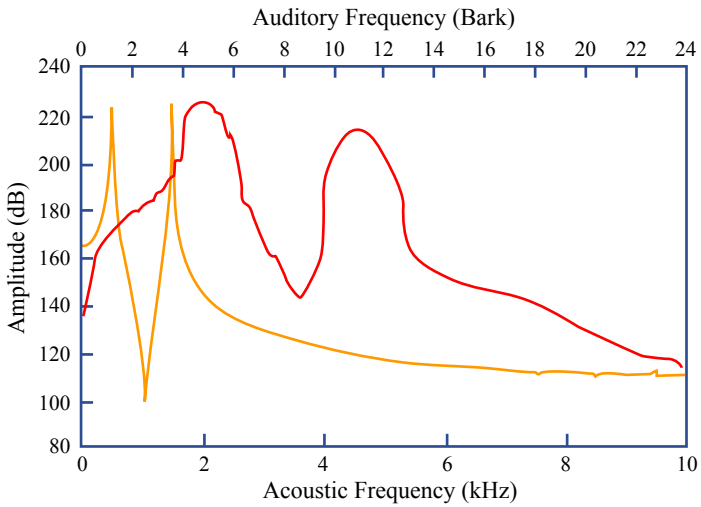


Figure by MIT OpenCourseWare. Adapted from Stevens, Kenneth N. *Acoustic Phonetics*. Cambridge, MA: MIT Press, 1999. ISBN: 9780262194044, after Delgutte, B. "Representation of Speech-like Sounds in the Discharge Patterns of Auditory-nerve Fibers." *Journal of the Acoustical Society of America* 68, no. 3 (1980): 843-857.

Auditory ‘spectrograms’

The auditory system performs a running frequency analysis of acoustic signals - cf. spectrogram.

- A regular spectrogram analyzes frequency of equal widths, but the peripheral auditory system analyzes frequency bands that are wider at higher frequencies.
- Further disparities are introduced by the non-linearities of the peripheral auditory system, e.g.
  - loudness is non-linearly related to intensity
  - masking(simultaneous and nonsimultaneous)



A comparison of acoustic (light line) and auditory (heavy line) spectra of a complex wave composed of sine waves at 500 at 1,500 Hz. Both spectra extend from 0 to 10 kHz, although on different frequency scales. The auditory spectrum was calculated from the acoustic spectrum using the model described in Johnson (1989).

Image by MIT OpenCourseWare. Adapted from Johnson, Keith. *Acoustic and Auditory Phonetics*. Malden, MA: Blackwell Publishers, 1997. ISBN: 9780631188483.

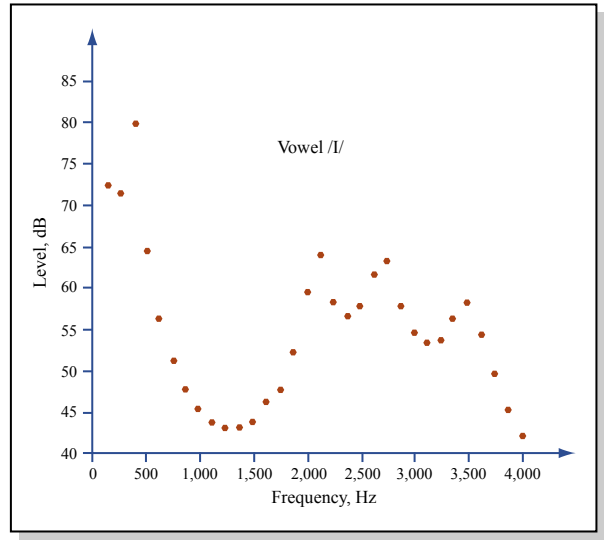
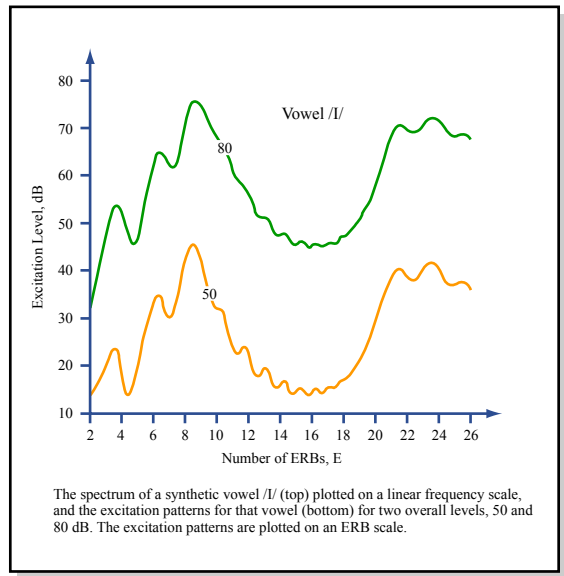


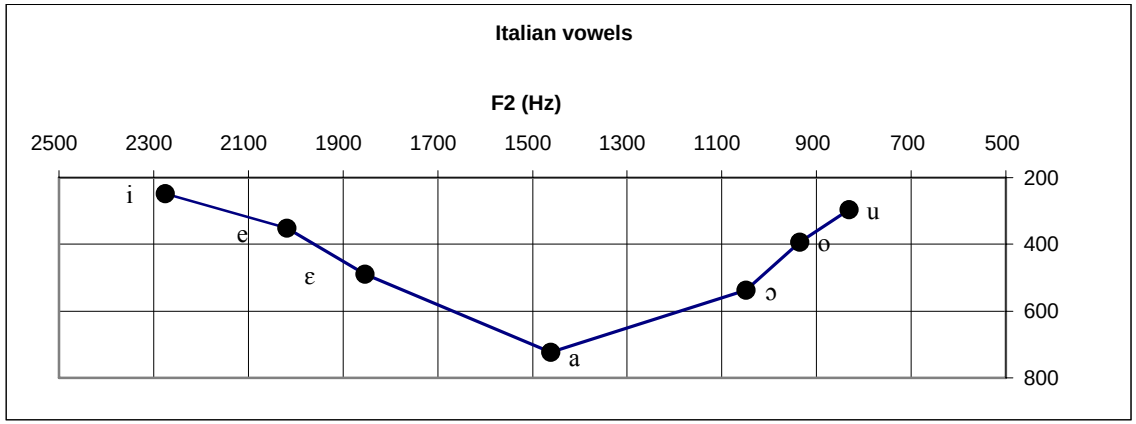
Image by MIT OpenCourseWare. Adapted from Moore, Brian. *The Handbook of Phonetic Science*. Edited by William J. Hardcastle and John Laver. Malden, MA: Blackwell, 1997. ISBN: 9780631188483.

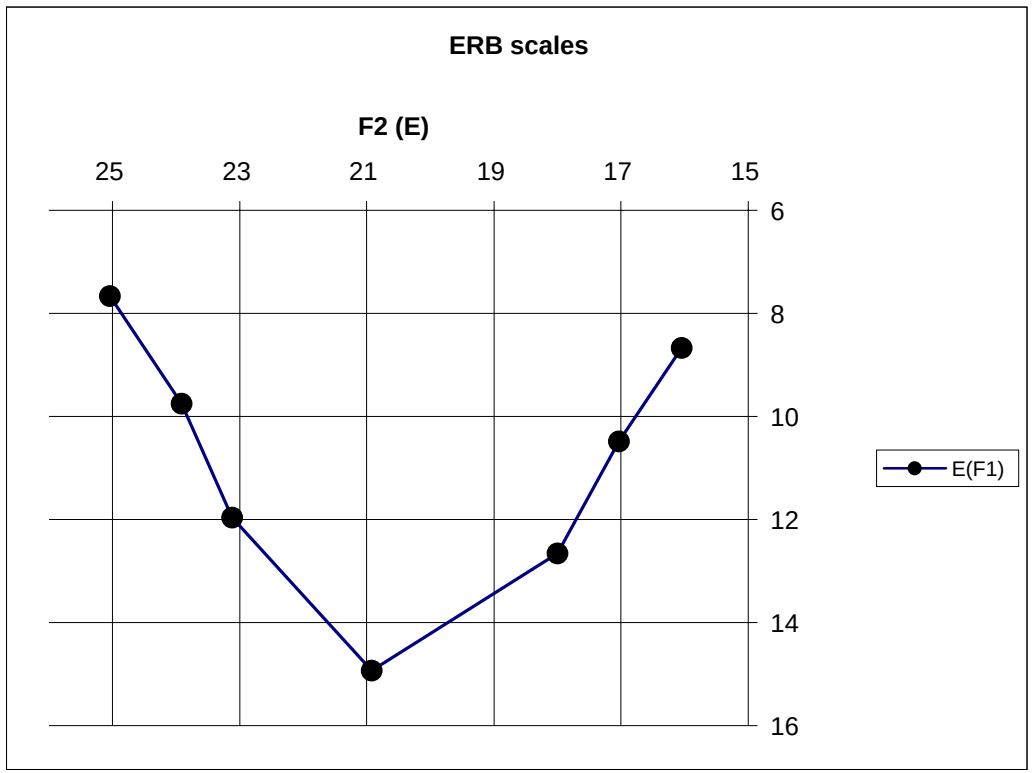


The spectrum of a synthetic vowel /I/ (top) plotted on a linear frequency scale, and the excitation patterns for that vowel (bottom) for two overall levels, 50 and 80 dB. The excitation patterns are plotted on an ERB scale.

Image by MIT OpenCourseWare. Adapted from Moore, Brian. *The Handbook of Phonetic Science*. Edited by William J. Hardcastle and John Laver. Malden, MA: Blackwell, 1997. ISBN: 9780631188483.

Spectrogram images removed due to copyright restrictions.
Figure 3.8 in Johnson, Keith. "Comparison of Normal Acoustic Spectrogram and Auditory Spectrogram or Cochleagram." In *Acoustic and Auditory Phonetics*. Malden, MA: Blackwell Publishers, 1997. ISBN: 9780631188483.


Italian vowels
F2 (Hz)
2500
2300
2100
1900
1700
1500
1300
1100
900
700
500
200
400
600
800
i
e
ɛ
a
ɔ
o
u


ERB scales
F2 (E)
25
23
21
19
17
15
6
8
10
12
14
16
E(F1)

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Linguistic Phonetics

Analog-to-digital conversion of speech signals

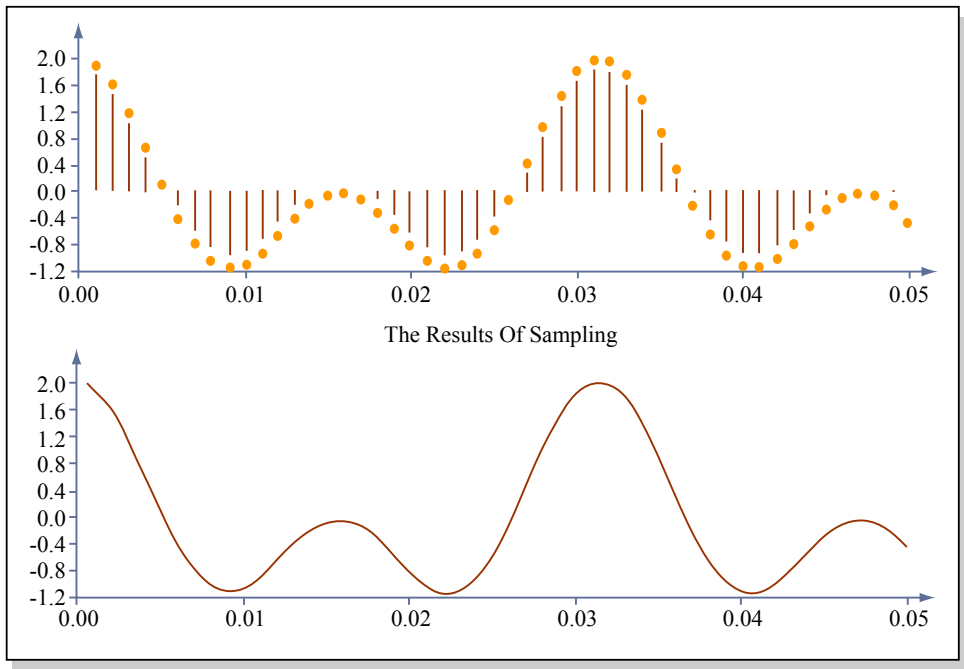
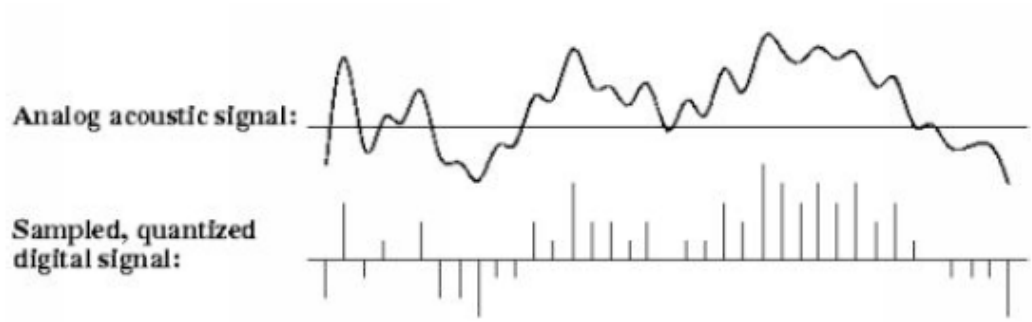


Figure by MIT OpenCourseWare.

Analog-to-digital conversion

- Almost all acoustic analysis is now computer-based.
- Sound waves are analog (or continuous) signals, but digital computers require a digital representation - i.e. a series of numbers, each with a finite number of digits.
- There are two continuous scales that must be divided into discrete steps in analog-to-digital conversion of speech: time and pressure (or voltage).
  - Dividing time into discrete chunks is called **sampling**.
  - Dividing the amplitude scale into discrete steps is called **quantization**.



Sampling

- The amplitude of the analog signal is sampled at regular intervals.
- The sampling rate is measured in Hz (samples per second).
- The higher the sampling rate, the more accurate the digital representation will be.

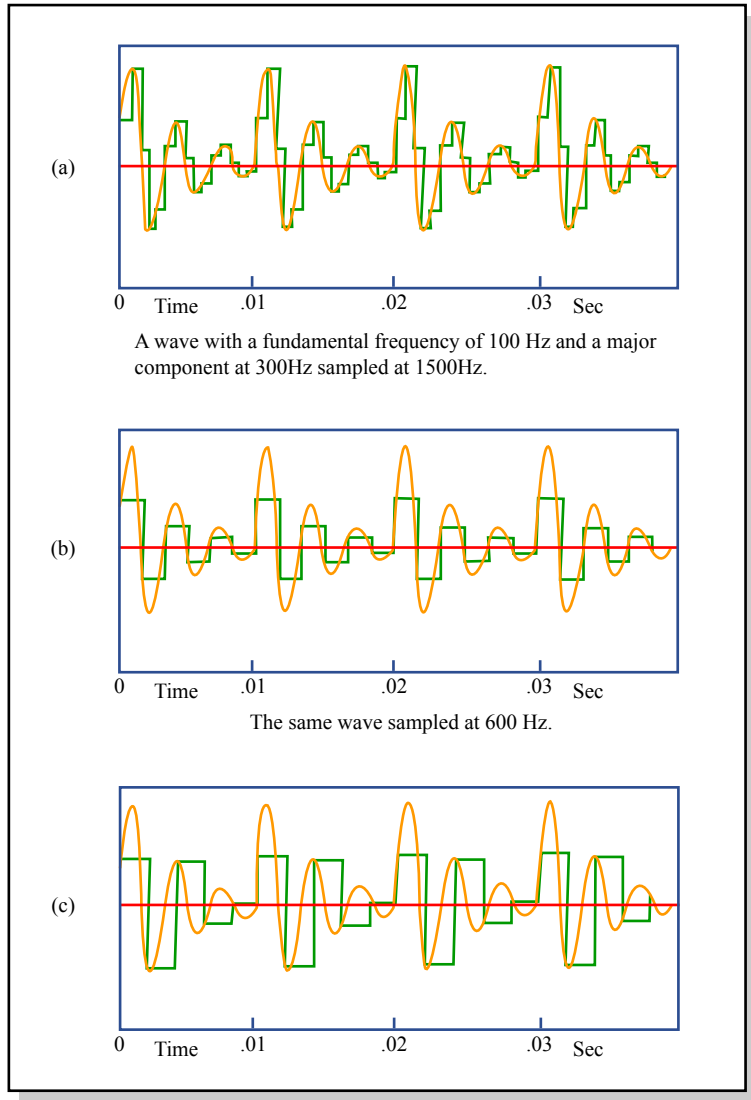


Figure by MIT OpenCourseWare. Adapted from Ladefoged, Peter. L104/204 Phonetic Theory lecture notes, University of California, Los Angeles.

Sampling

- In order to represent a wave component of a given frequency, it is necessary to sample the signal with at least twice that frequency (the Nyquist Theorem).
- The highest frequency that can be represented at a given sampling rate is called the Nyquist frequency.

- The wave at right has a significant harmonic at 300 Hz
  - (a) sampling rate 1500 Hz
  - (b) sampling rate 600 Hz
  - (c) sampling rate 500 Hz

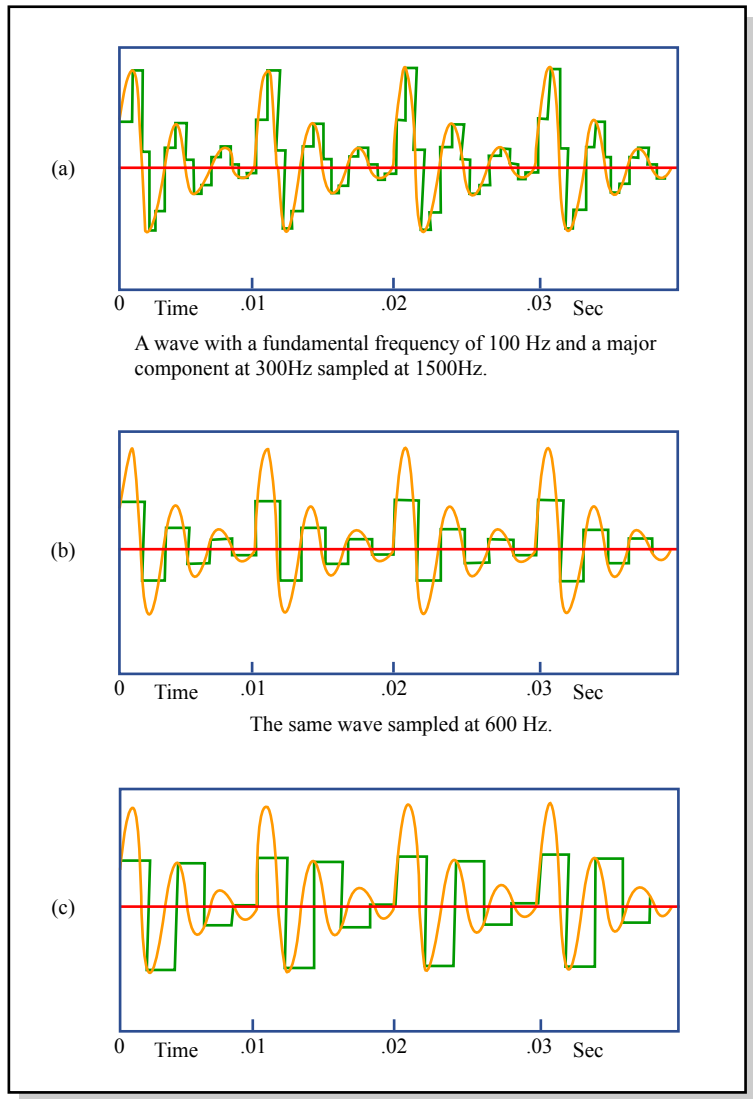


Figure by MIT OpenCourseWare. Adapted from Ladefoged, Peter. L104/204 Phonetic Theory lecture notes, University of California, Los Angeles.

What sampling rate should you use?

- The highest frequency that (young, undamaged) ears can perceive is about 20 kHz, so to ensure that all audible frequencies are represented we must sample at 2×20 kHz = 40 kHz.
- The ear is relatively insensitive to frequencies above 10 kHz, and almost all of the information relevant to speech sounds is below 10 kHz, so high quality sound is still obtained at a sampling rate of 20 kHz.
- There is a practical trade-off between fidelity of the signal and memory, but memory is getting cheaper all the time.

What sampling rate should you use?

- For some purposes (e.g. measuring vowel formants), a high sampling rate can be a liability, but it is always possible to **downsample** before performing an analysis.

- Audio CD uses a sampling rate of 44.1 kHz.
- Many A-to-D systems only operate at fractions of this rate (22050 Hz, 11025 Hz).

Aliasing

- Components if a signal which are above the Nyquist frequency are misrepresented as lower frequency components (**aliasing**).
- To avoid aliasing, a signal must be filtered to eliminate frequencies above the Nyquist frequency.
- Since practical filters are not infinitely sharp, this will attenuate energy near to the Nyquist frequency also.

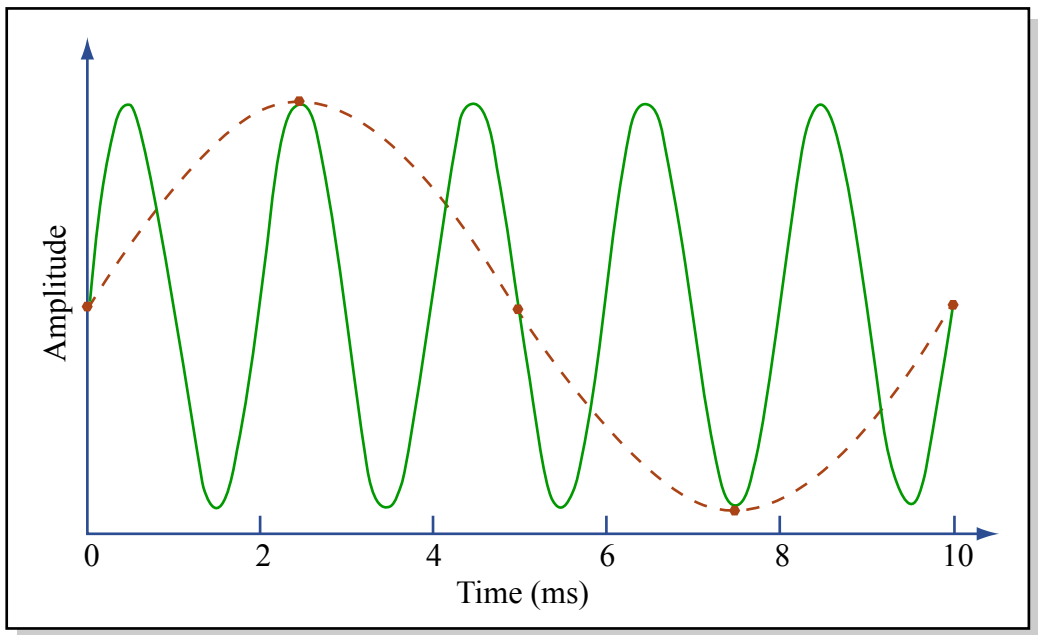
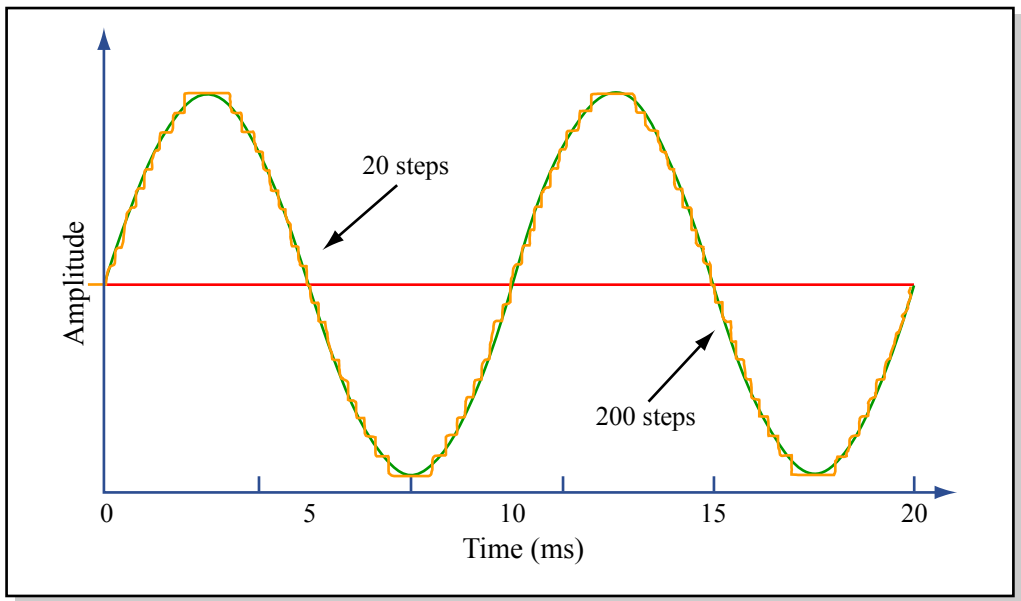


Figure by MIT OpenCourseWare. Adapted from Johnson, Keith.
Acoustic and Auditory Phonetics. Malden, MA: Blackwell Publishers, 1997. ISBN: 9780631188483.

Quantization

- The amplitude of the signal at each sampling point must be specified digitally - quantization.
- Divide the continuous amplitude scale into a finite number of steps. The more levels we use, the more accurately we approximate the analog signal.



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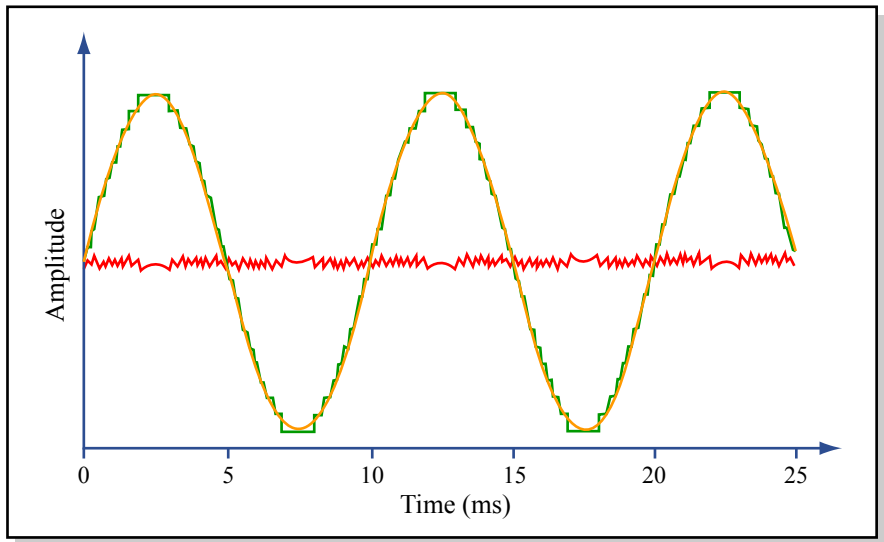
Quantization

- The number of levels is specified in terms of the number of bits used to encode the amplitude at each sample.
  - Using n bits we can distinguish 2^n levels of amplitude.
  - e.g. 8 bits, 256 levels.
  - 16 bits, 65536 levels.
- Now that memory is cheap, speech is almost always digitized at 16 bits (the CD standard).

Quantization

- Quantizing an analog signal necessarily introduces quantization errors.
- If the signal level is lower, the degradation in signal-to-noise ratio introduced by quantization noise will be greater, so digitize recordings at as high a level as possible without exceeding the maximum amplitude that can be represented (clipping).
- On the other hand, it is essential to avoid clipping.



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