Signals, systems, acoustics and the ear

Week 1

Introduction

No eating or drinking in the lab!

Not even water

You may find this course demanding! How to get through it:

- Consult the web site
- Essential to do the reading and suggested exercises
  - Rosen, S., and Howell, P. (2010). *Signals and Systems for Speech and Hearing, 2nd edition*: Brill, Leiden.
- Laboratory sessions go a long way to clarify the material presented
- Bring questions to the tutorial sessions
- Send questions to the staff through Moodle
- Keep up with the work
- If you have problems, *ask for help!*
- If you can do the course work and exercises, you will do well on the exam …
  - if you cannot, you will not!
- You are responsible for bringing printed-out notes to class, if you want to.
  - Made available on the web site on Friday prior to the lecture at the latest
  - Lab sheets will be provided, so you need not print them

People

- Andrew Clark
  - Experimental Officer, will help out in lab, as will …
- Kurt Steinmetzger
  - PhD student
- Dave Cushing
  - Experimental Officer who looks after the lab

Timetable

UCL Week	Date Monday	Lecture topic	Laboratory/other	CW (for following week)
6	5 Oct	Waveforms, signals, sinusoids, frequency, logarithms, dB, digital signals	Pure tone audiometry & decibels	Set I
7	12 Oct	More about waves and dB; Properties of LTI systems (i/o functions, linearity, time-invariance); The BIG idea	Frequency response of an acoustic resonator: two different sizes	Set II
8	19 Oct	Frequency responses & Spectra	Harmonic synthesis (*Esynth*)	Set III
9	26 Oct	Signals through systems; Filter banks	Signals through systems: analogue & digital (*Esystem*)	Set IV
10	2 Nov	The ear as a signal processor	Cochlear simulation	
11	9 Nov	**Reading week: no meeting**		
12	16 Nov	Frequency selectivity in the periphery and in perception	Notched-noise masking	
13	23 Nov	Envelope & temporal fine structure	Adaptive techniques	Choose an appropriate paper and email it to me
14	30 Nov	Binaural hearing; Pitch perception	Measuring F0 in various sounds	Journal article approved
15	7 Dec	Intensity & Loudness; Temporal resolution	Critique journalistic essays	Bring in first draft of essay
16	14 Dec	Psychoacoustics of hearing impairment; perceiving speech-in-noise	Speech-in-noise	Hand in final essay

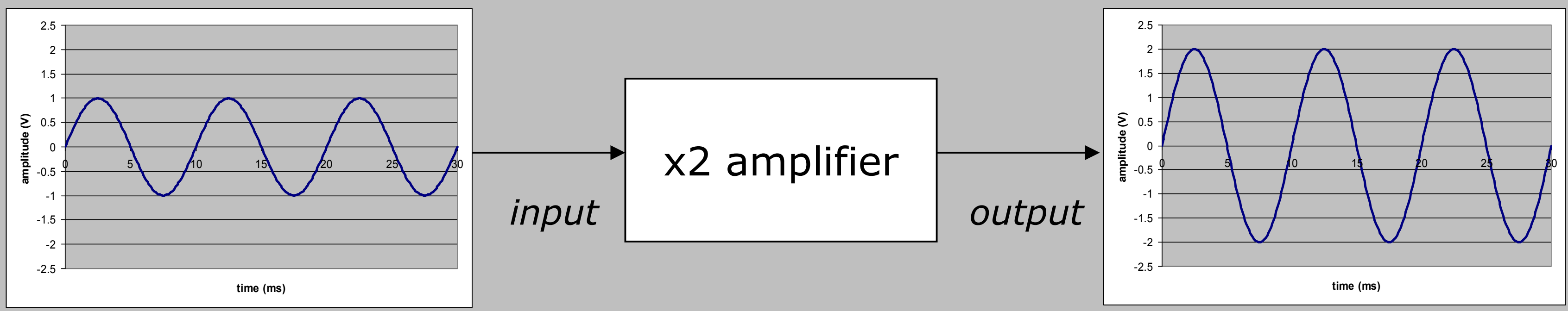
Acoustics, signals & systems for audiology

Week 1

Signals (& Systems)

What are systems & signals?

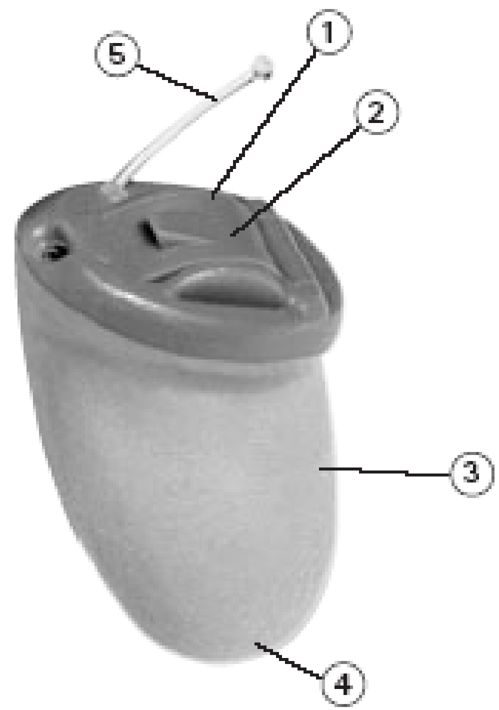
- *Systems* perform an operation on, or transformation of, a *signal* (or *waveform*)
- Concentrate on systems with one input and one output
- Many useful examples in hearing and speech science



System = In-The-Ear Hearing Aid

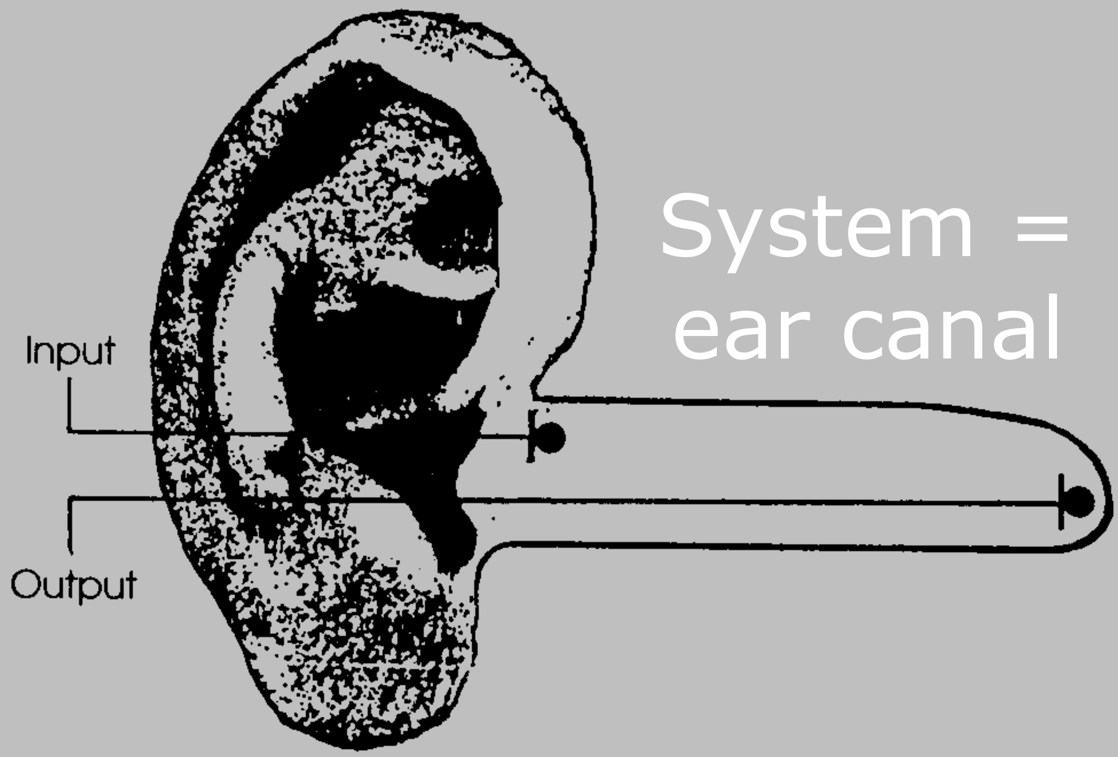
input = sound wave
(variations in pressure)

output = sound wave
(modified in some way)



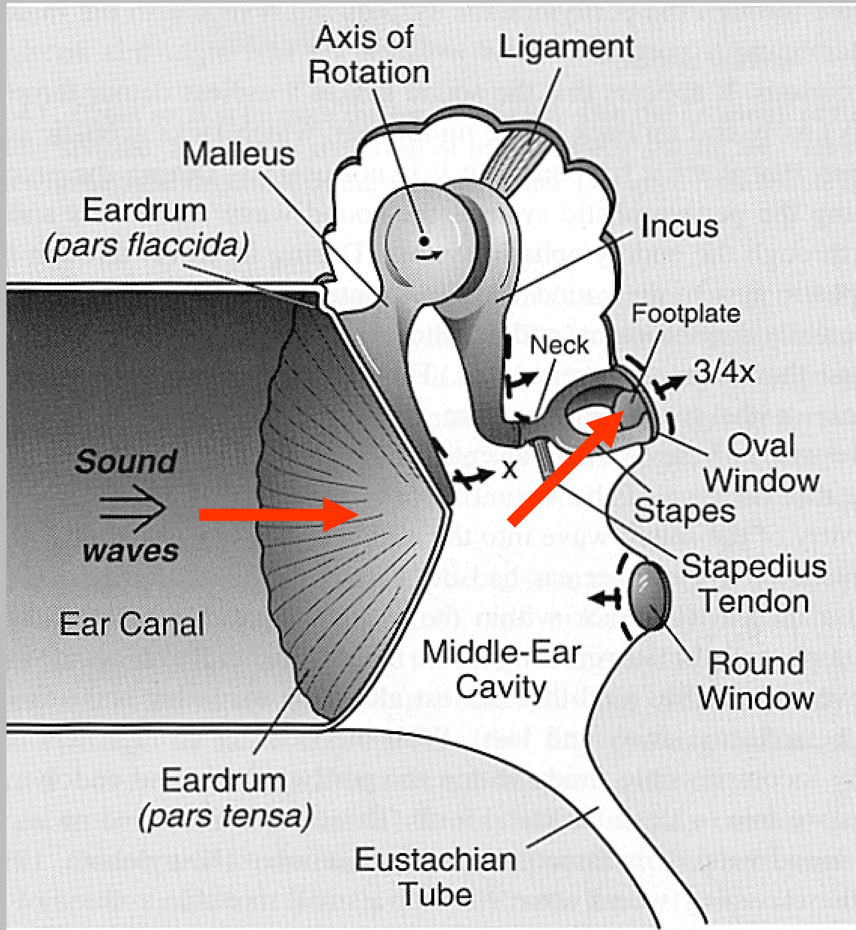
1. Microphone
2. Battery compartment and programming socket
3. Custom made shell
4. Receiver
5. Removal thread

System = ear canal



Input and output signals are both sound waves

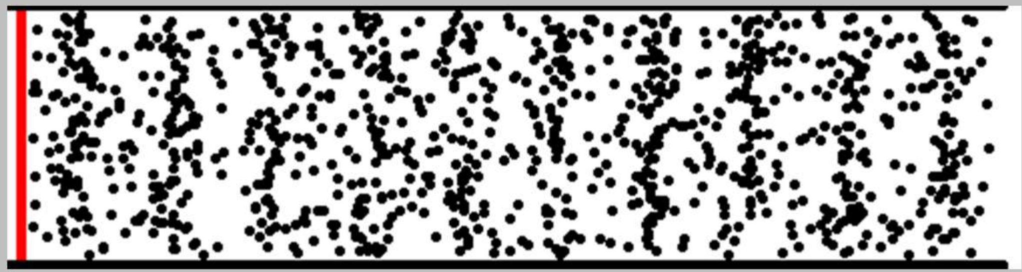
System = middle ear



Input and output signals are both *mechanical* waves (movements)

Tell me some others!

What is sound?



Sound is oscillation of air pressure (pressure wave).

high pressure: air molecules bunched up
low pressure: air molecules spread out

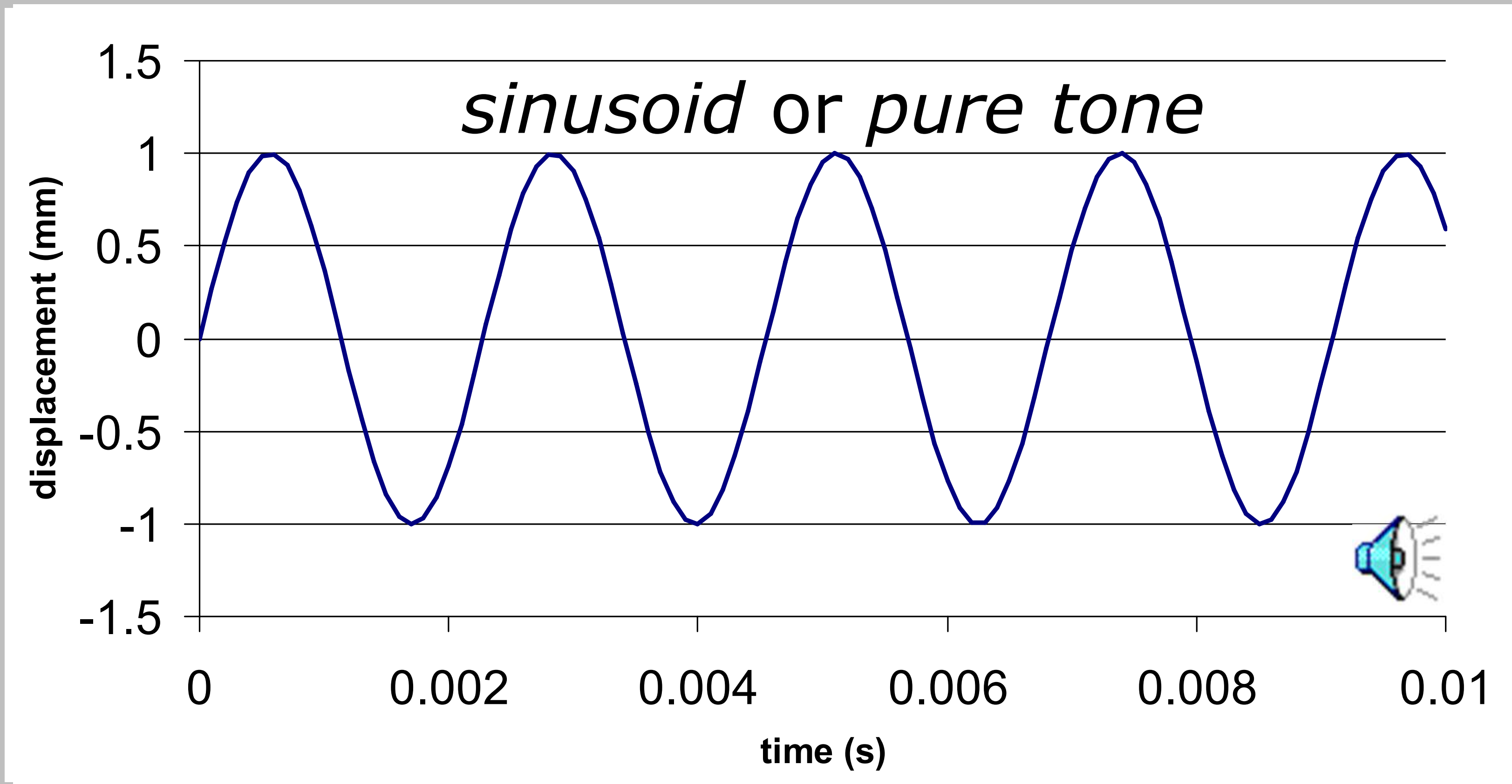
Air molecules do **not** travel through space to carry sound

Sound is one kind of signal

Imagine measuring the instantaneous pressure at a single place

A microphone converts variations in sound pressure to electrical variations in voltage

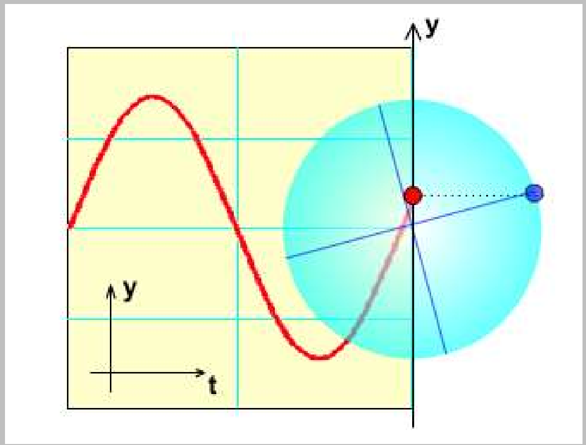
A very simple signal

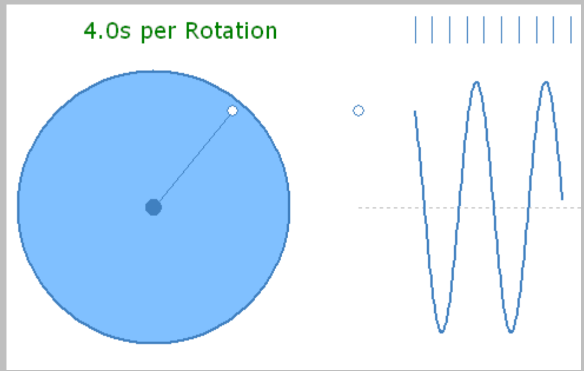


(but every signal we work with can be drawn on the same kind of graph)

Essential characteristics of sinusoids

- Sinusoids are a *unique* shape
  - not just any vaguely regular form
  - are *periodic*
    - a basic *cycle* repeats over and over
  - can be constructed from *uniform circular motion*





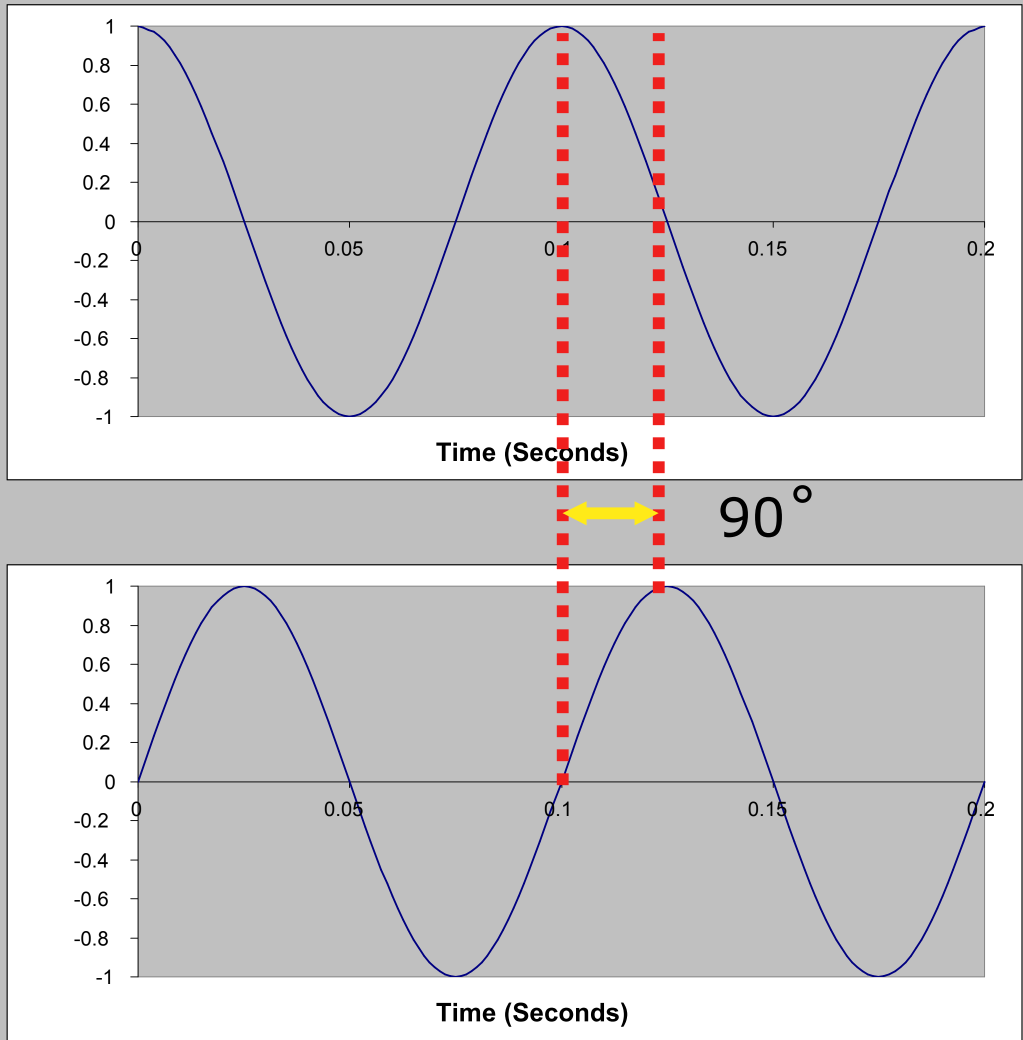
http://www.upscale.utoronto.ca/GeneralInterest/Harrison/Flash/ClassMechanics/Circular2SHM/Circular2SHM.html

Sinusoids can only differ in three ways

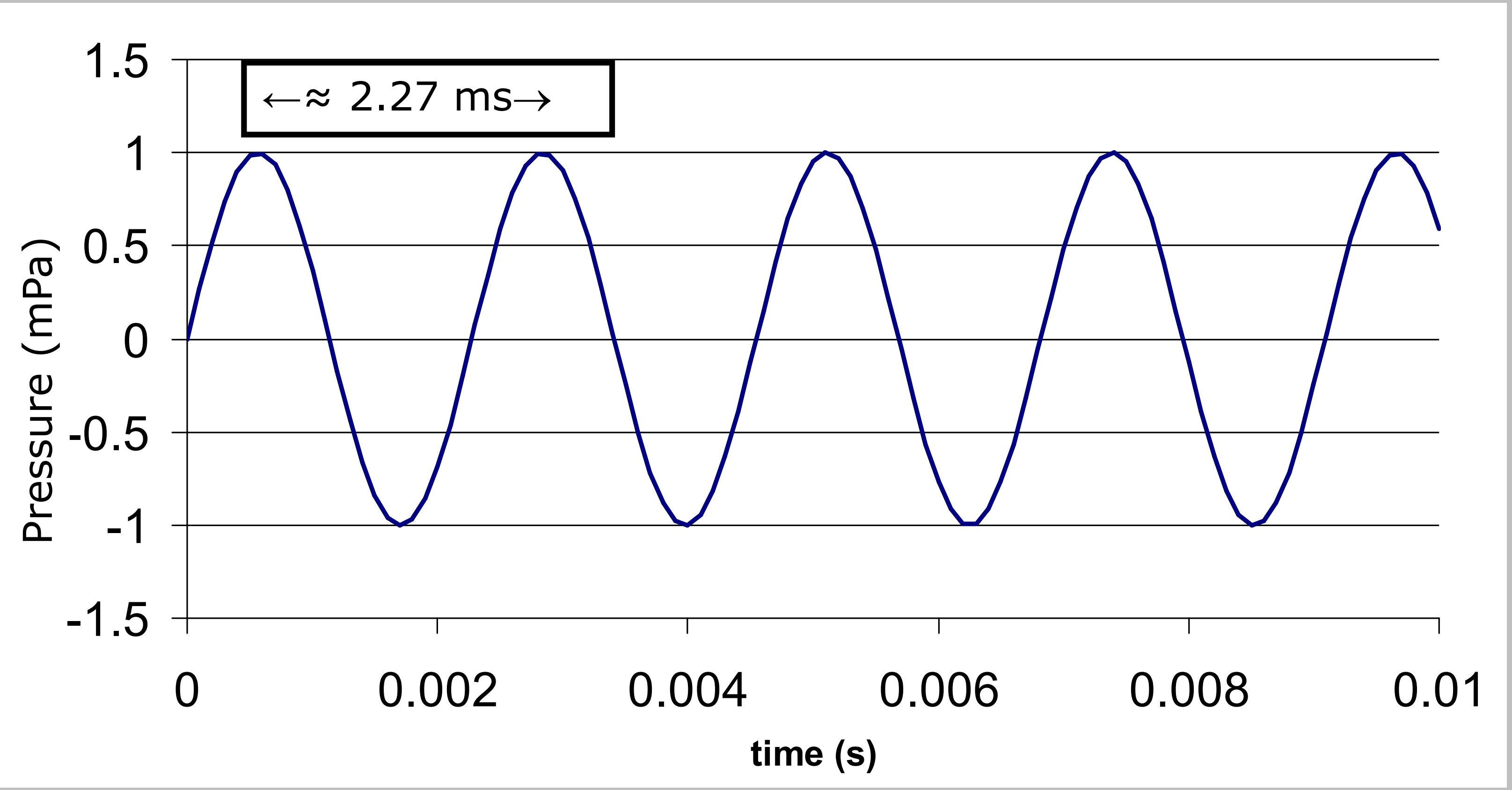
- Once you know a wave is sinusoidal, there are only three things to know about it:
  - frequency
  - amplitude
  - phase
    - generally less important because phase changes are typically not perceived

I: Phase

- Where a sinewave *starts* relative to some arbitrary time
- Measured in cycles or degrees (or radians)
  360° = 1 period = 2π rads
  180° = ½ period = π rads
  90° = ¼ period = $\pi/2$ rads
- Equivalent to a shift in time
- Relatively little effect on perception but still important in many situations



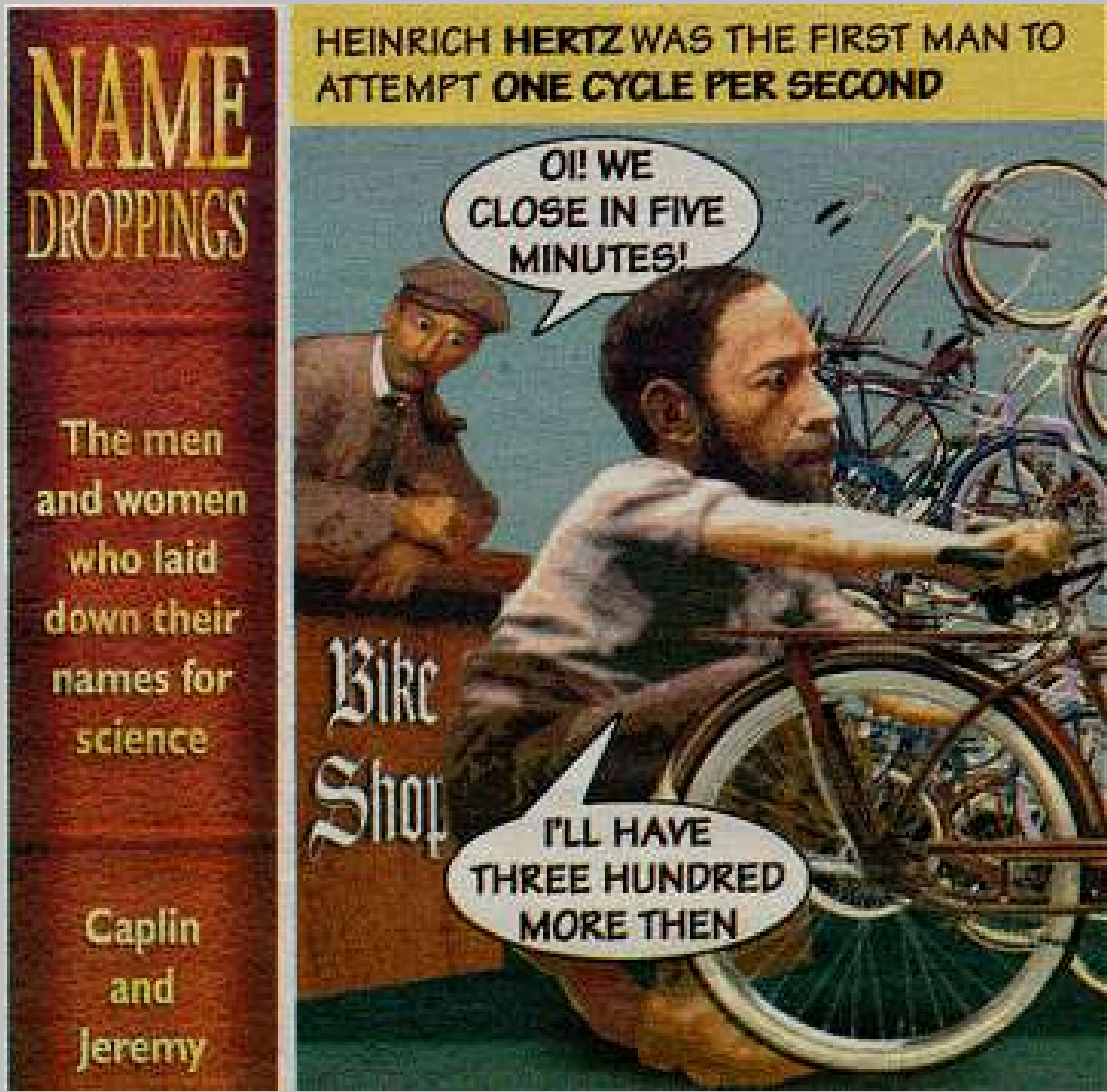
II: Periodicity (frequency)



Specifying periodicity

- The period (*p*) is the time to complete one *cycle* of the wave
- Alternatively, the number of cycles that are completed in one second, is the *frequency (f)*
- $f=1/p$ and $p=1/f$
  - here =1/0.00227 sec = 440 cycles per second (cps)
- But a special unit name is used …

hertz (Hz): a measure of frequency



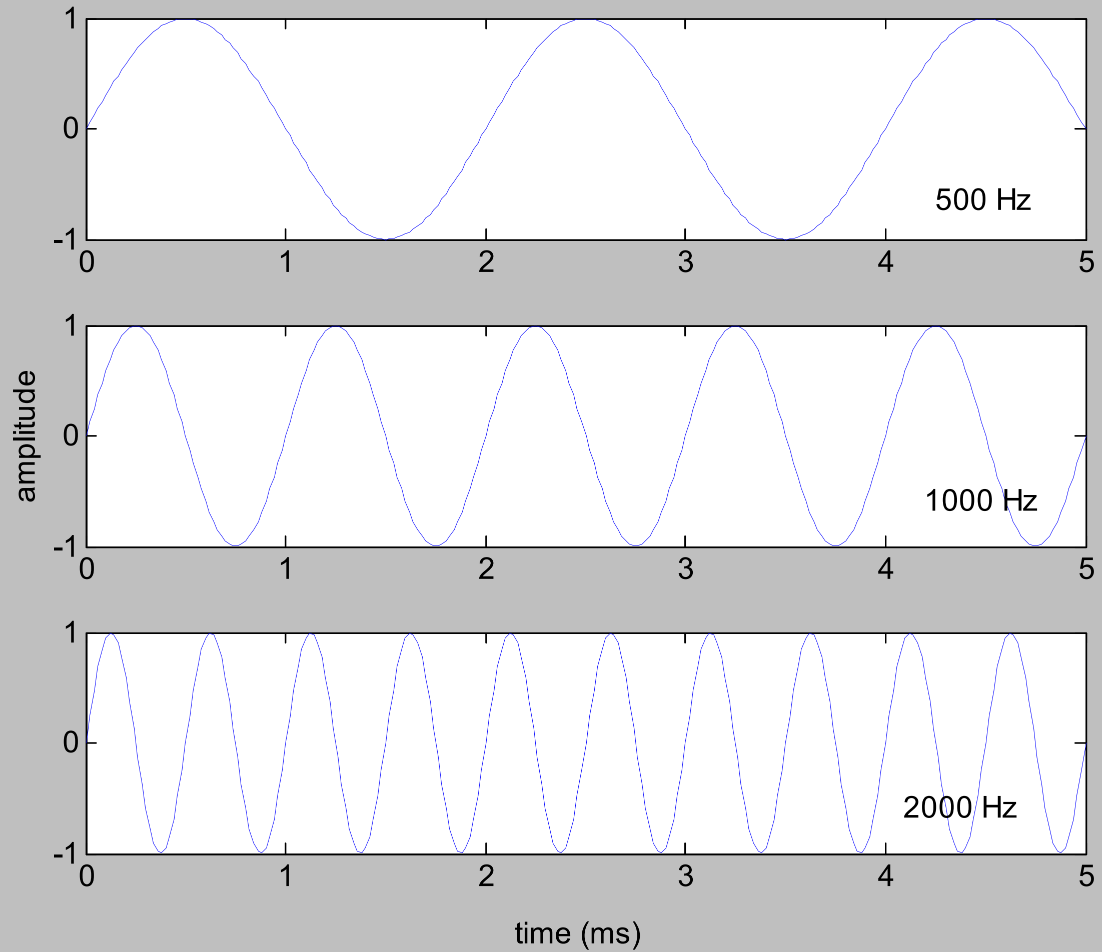
Keep your units consistent!

- period of 0.001 sec = 1 ms (millisecond)

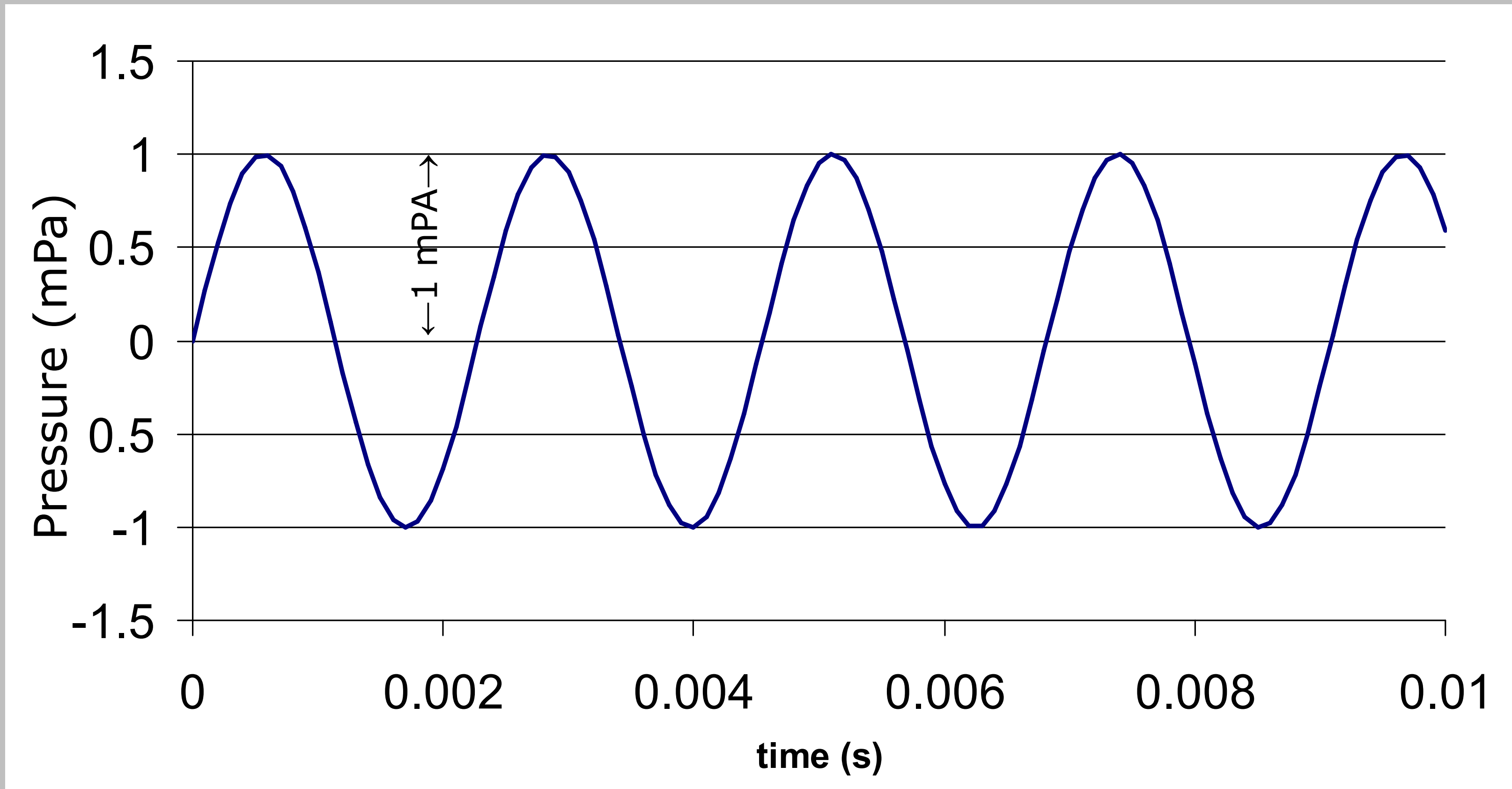
  so:
- period in seconds: f (Hz)=1/p (s)
- period in ms: f (Hz)=1000/p (ms)
- period in ms: f (kHz) =1/p (ms)

- A period of 1 ms = ?? Hz
- A frequency of 100 Hz = ?? ms

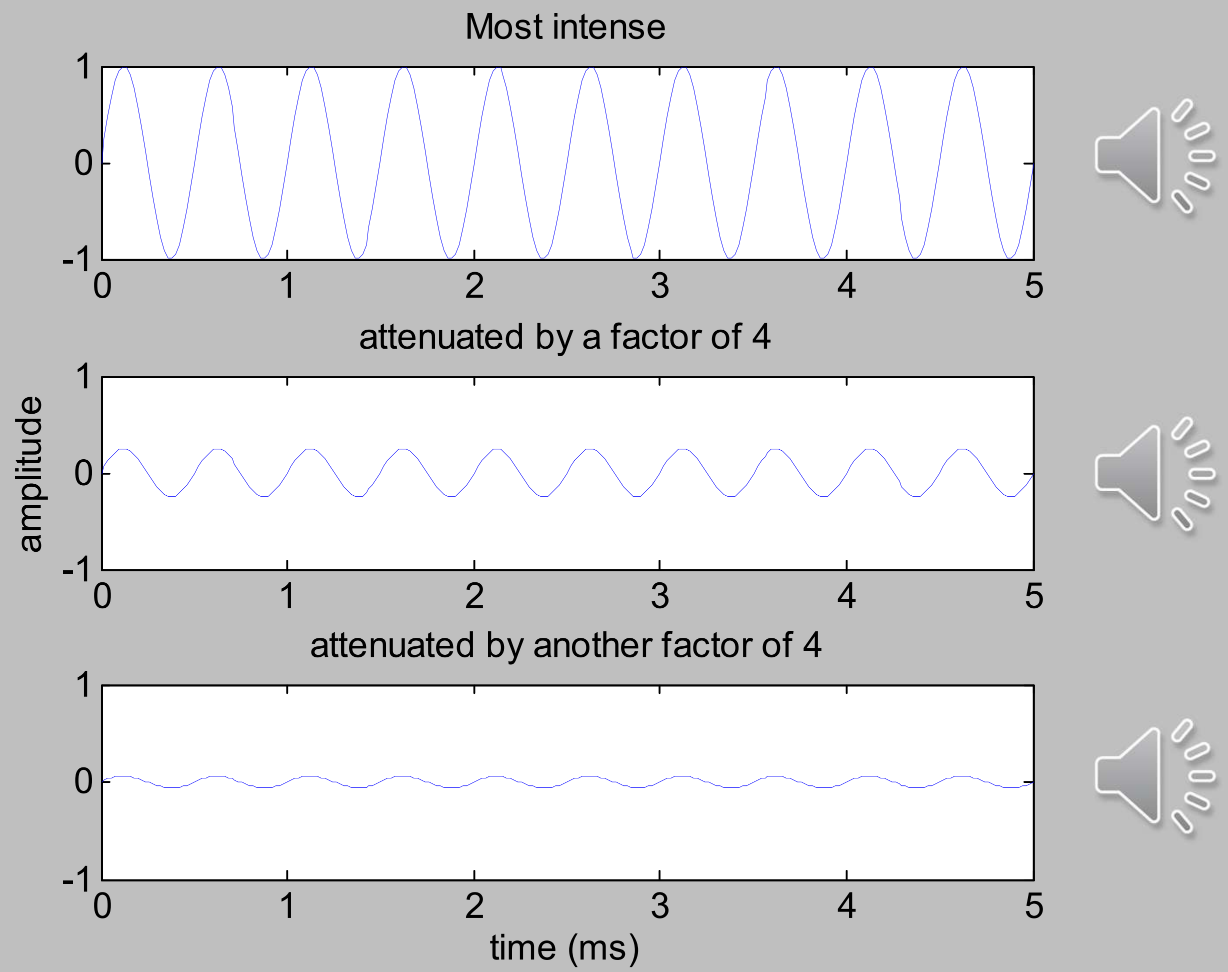
Increases in frequency (decreases in period) lead to increases in subjective pitch



III: Amplitude



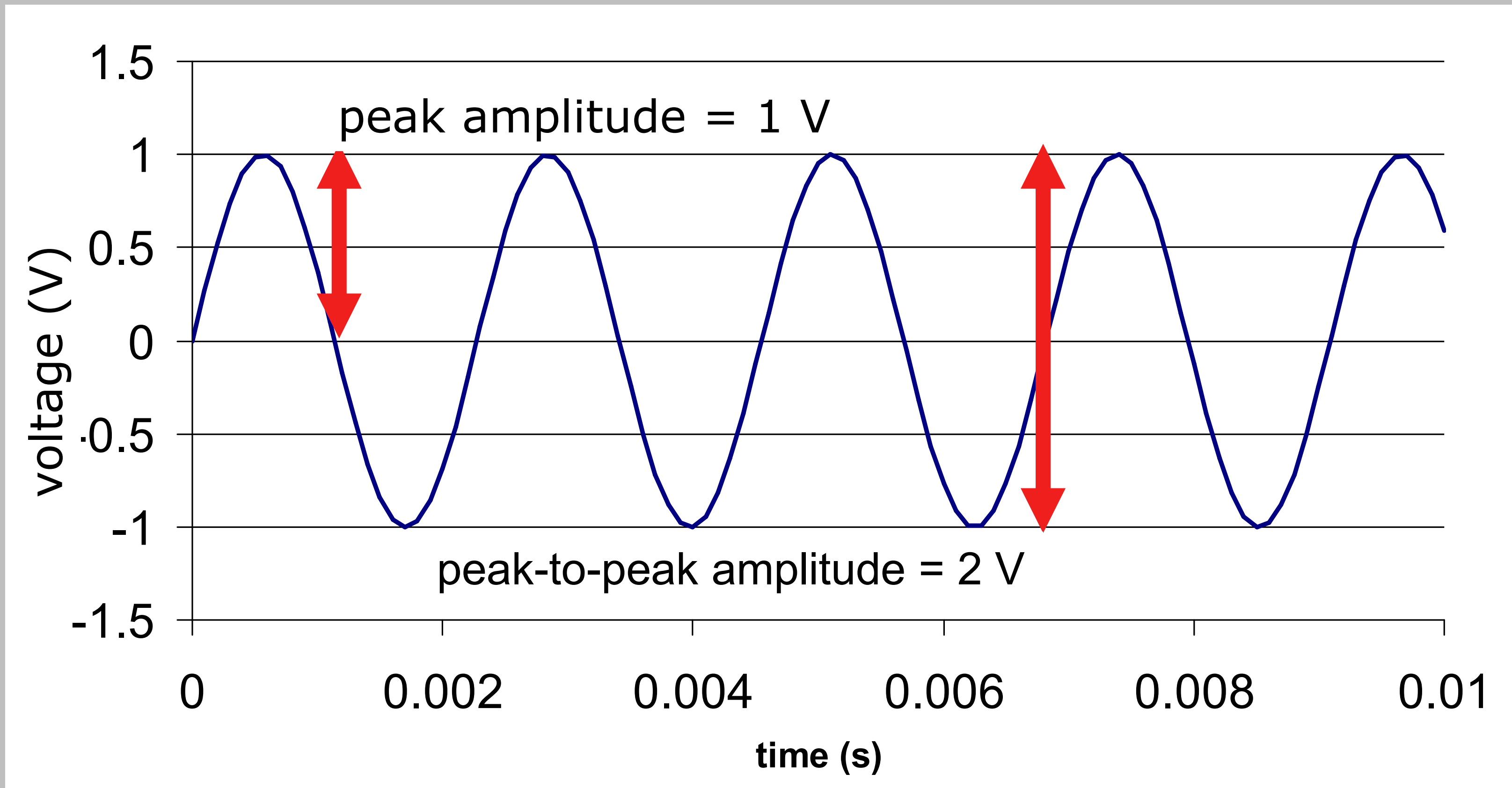
Increases in amplitude lead to increases in perceived loudness



Measures of amplitude

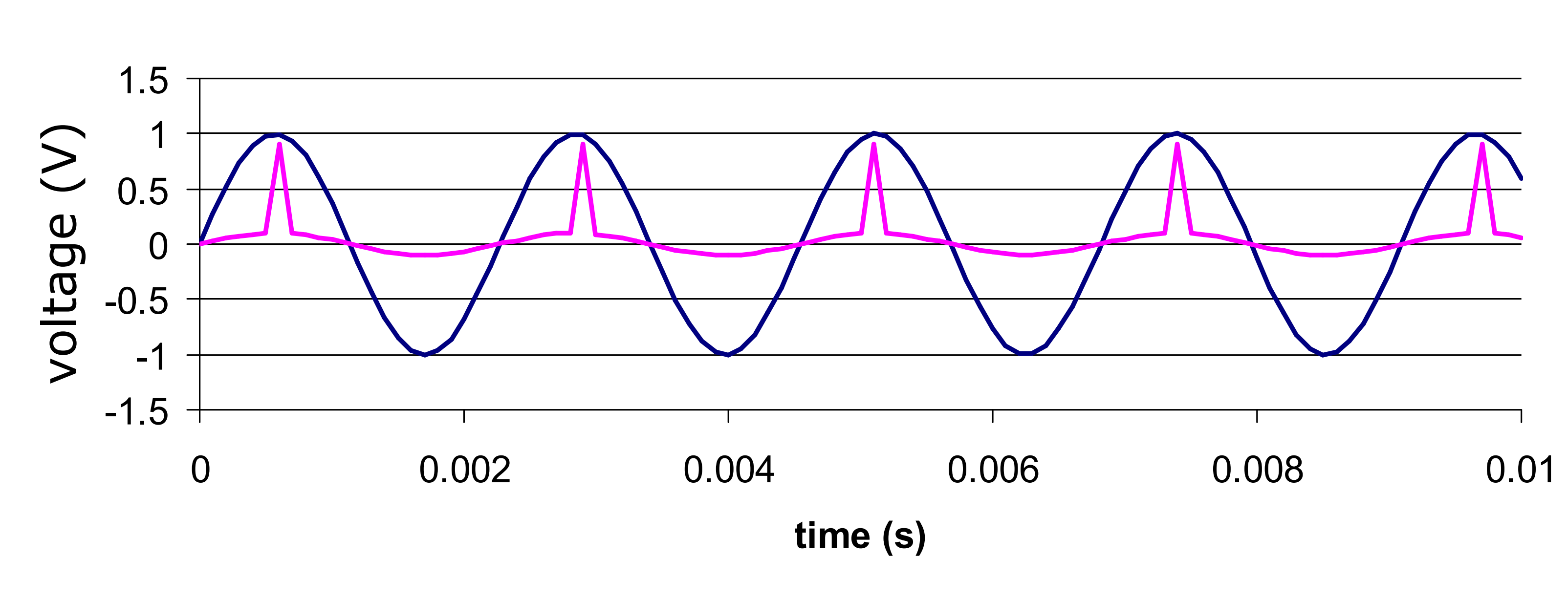
- It is crucial to distinguish instantaneous measures (as in a waveform) from some kind of average
- Instantaneous measures always linear (e.g., pressure in Pa, voltage in V, displacement in metres)
- But also want a single number to be a good summary of the ‘size’ of a wave
- Average measures can be linear or logarithmic (dB)

Simple measures of amplitude



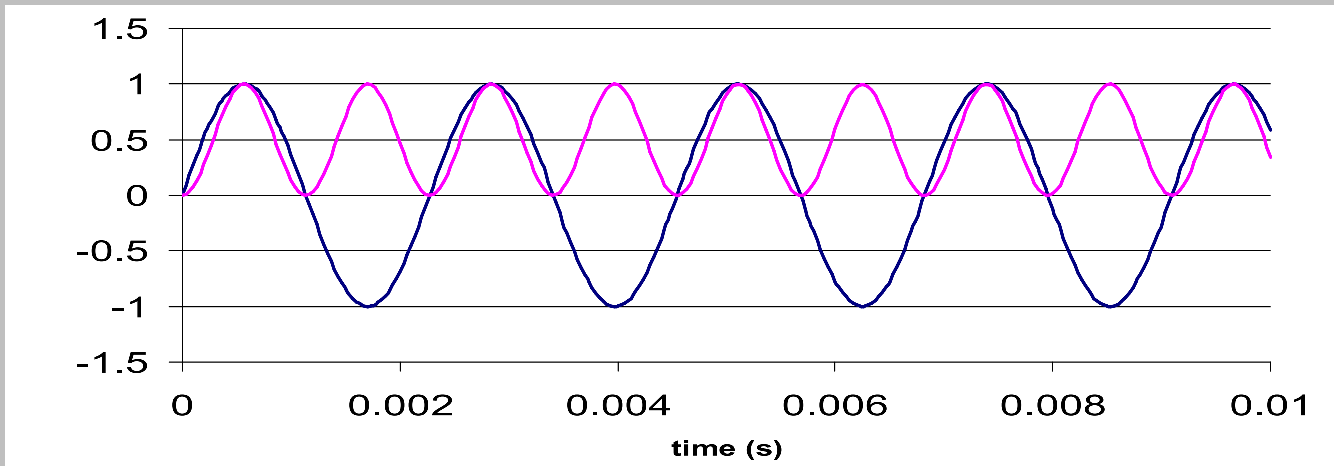
Drawback to peak measures

- Don’t accurately reflect the energy in a waveform



root-mean-square (rms)

- Square all the values of the wave
- Take the average area under the curve
- Take the square root
- A measure of the *energy*, applicable to all waveforms
- Still a *linear* measure (Pa, mm, V)



rms=.707

a sinusoid and its square

Scaling amplitude: The decibel Scale

Idea I:

Define a point of reference and rescale data in terms of that reference

Idea II:

Use a kind of warped scale that relates to perception

Sound Pressure Level

$$\text{Intensity}(dBSPL) = 20\log_{10}\left(\frac{\text{Pressure}(Pa)}{20\mu Pa}\right)$$

- **20µPa** is the standard reference pressure
  - approximately equal to human threshold
- **$\log_{10}$(*ratio*)** turns ratio into power of 10.

Measuring amplitudes with dB

- *Not* a linear unit like pascals
- A logarithmic measure with an arbitrary reference point
  - 0 dB does not mean no sound; it means the same level as the reference
  - Any positive number of dB means greater than the reference (e.g., 10 dB)
  - Any negative number of dB means less than the reference (e.g., -10 dB)
- Many different kinds of dB (SPL, HL, …) which differ essentially in the meaning of 0 dB.

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dB SPL Examples

- Threshold of Hearing (20 μPa)

  $20 \times \log_{10}(20\ \mu Pa / 20\ \mu Pa)$

  $= 20 \times \log_{10}(1) = 20 \times 0$

  $= 0$ dB SPL

- Distinct Pain! (200 Pa)

  $20 \times \log_{10}(200\ Pa / 20\ \mu Pa)$

  $= 20 \times \log_{10}(10000000) = 20 \times 7$

  $= 140$ dB SPL

- An inaudible sound (2 μPa)

  $20 \times \log_{10}(2\ \mu Pa / 20\ \mu Pa)$

  $= 20 \times \log_{10}(0.1) = 20 \times -1$

  $= -20$ dB SPL

Why use a logarithmic unit (dB)?

- Waveform amplitudes *can* be specified in linear rms units and often are,
- But our perception of changes in sound amplitude is more closely related to a logarithmic scale (based on ratios/proportions)
- Compare distinguishing a 1 kHz sinusoid of 50 μPa vs. 100 μPa (obvious change)
- And 1 Pa to (1 Pa + 50 μPa) = 1.00005 Pa (indistinguishable)
- Equal steps on a dB scale correspond to equal *ratios* on the linear scale

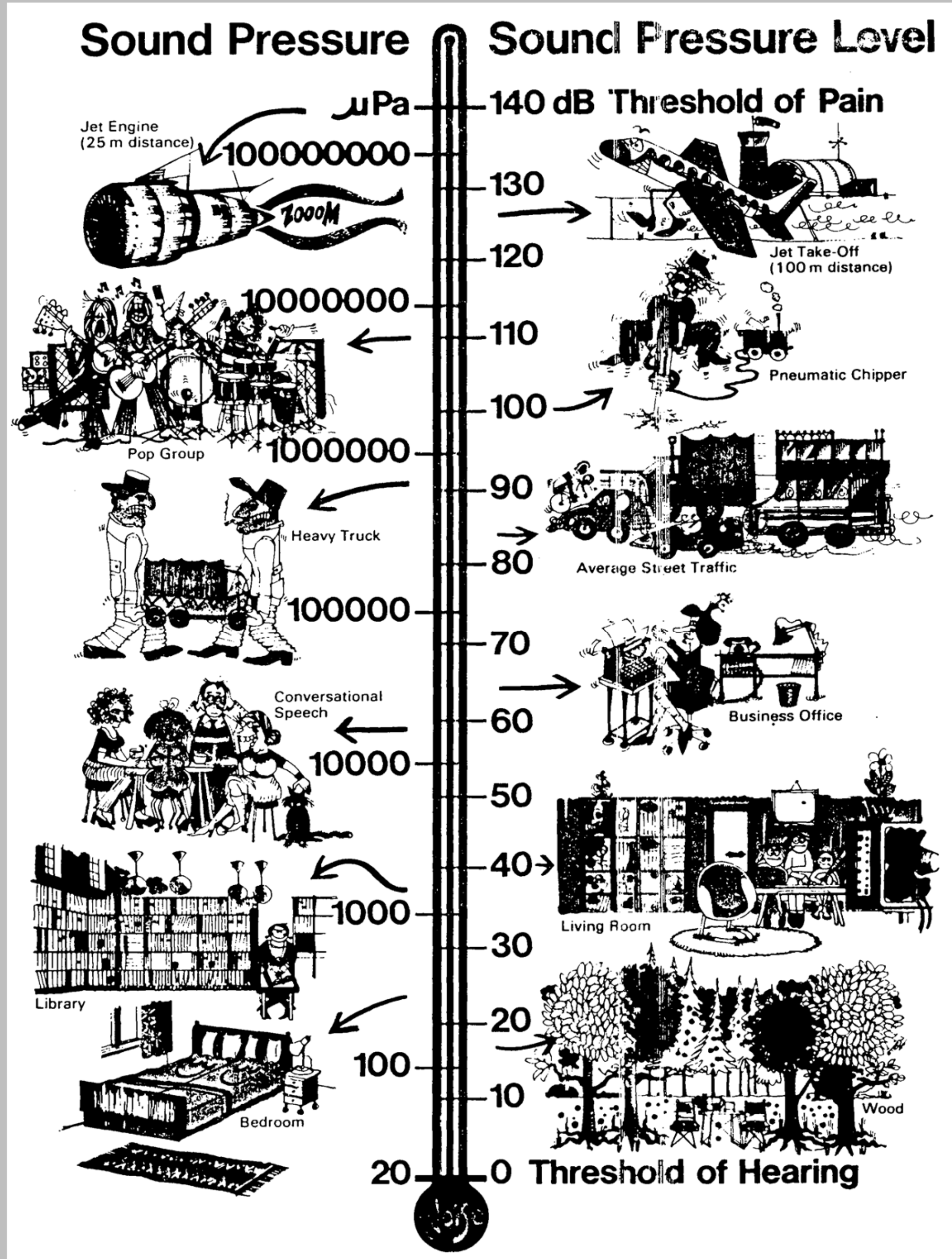
Just-noticeable difference in intensity is about 1 dB

- Standard
- 1-dB less intense
- 3-dB less intense
- 6-dB less intense
- 10-dB less intense

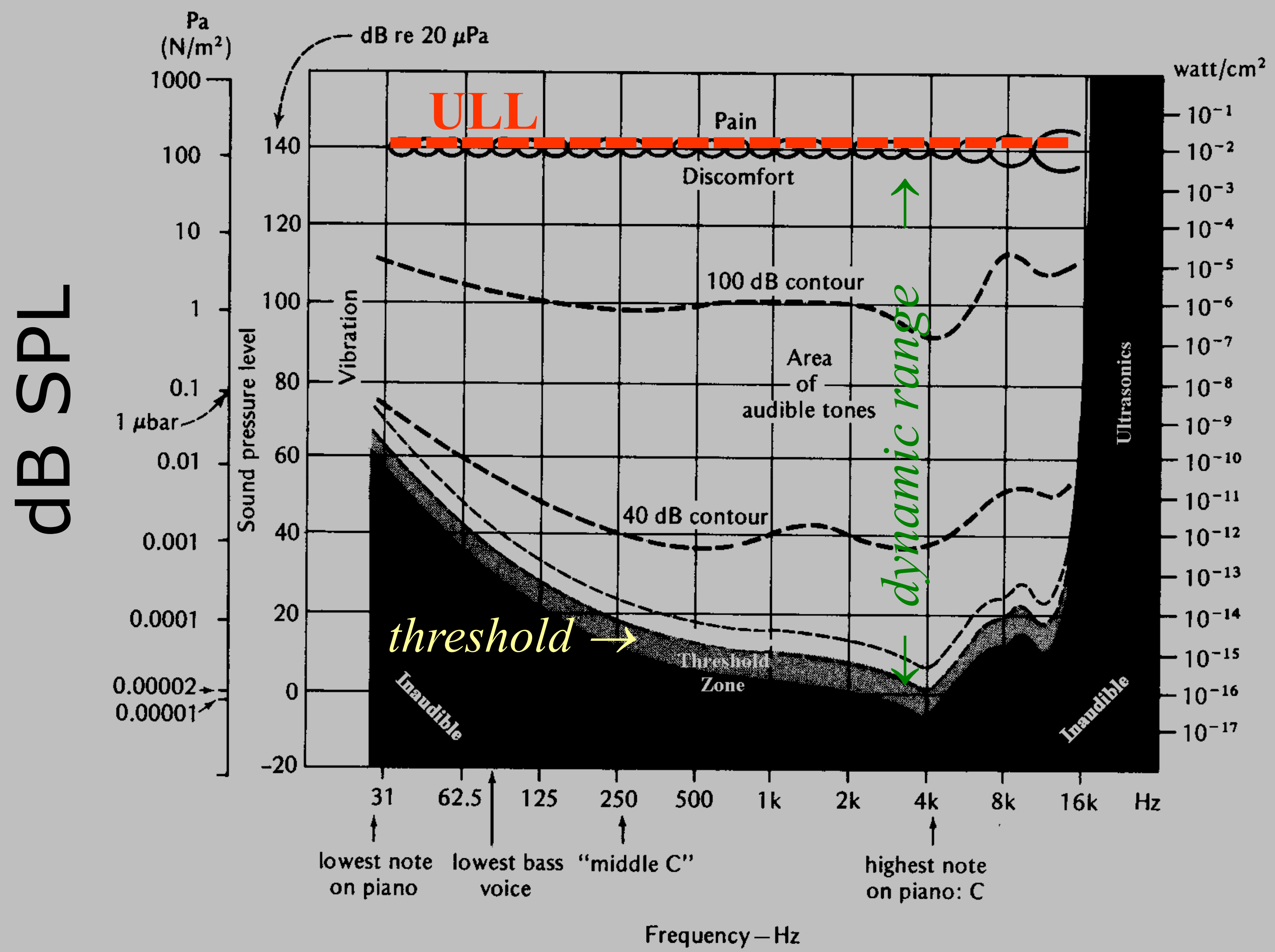
dB scales are used widely

- dB can be used for any amplitude measure as long as a reference is defined.
- dB re 1 mV = 20 * $\log_{10}$(*x* mV/1 mV) where *x* is any number
- 1 V = 20 * $\log_{10}$(1000 mV/1 mV) = 60 dB re 1 mV
- 1 V = 20 * $\log_{10}$(1 V/1 V) = 0 dB re 1 V
- Can use dB for displacement (meters), current (amps), etc.
- Can use dB for sound pressure but a different reference in place of 20 μPa

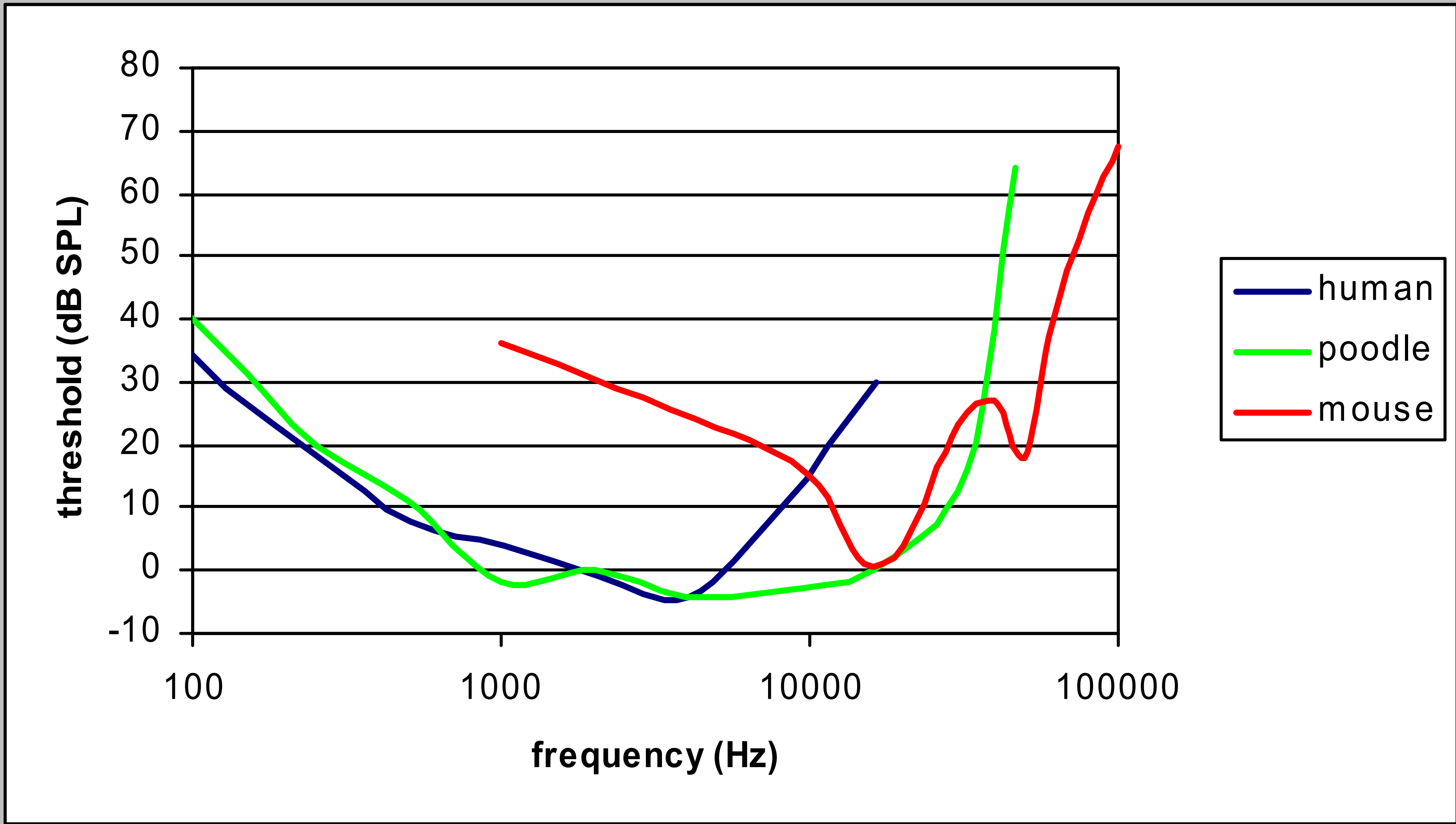
Getting a feel for decibels (dB SPL)



Human hearing for sinusoids



Thresholds for different mammals

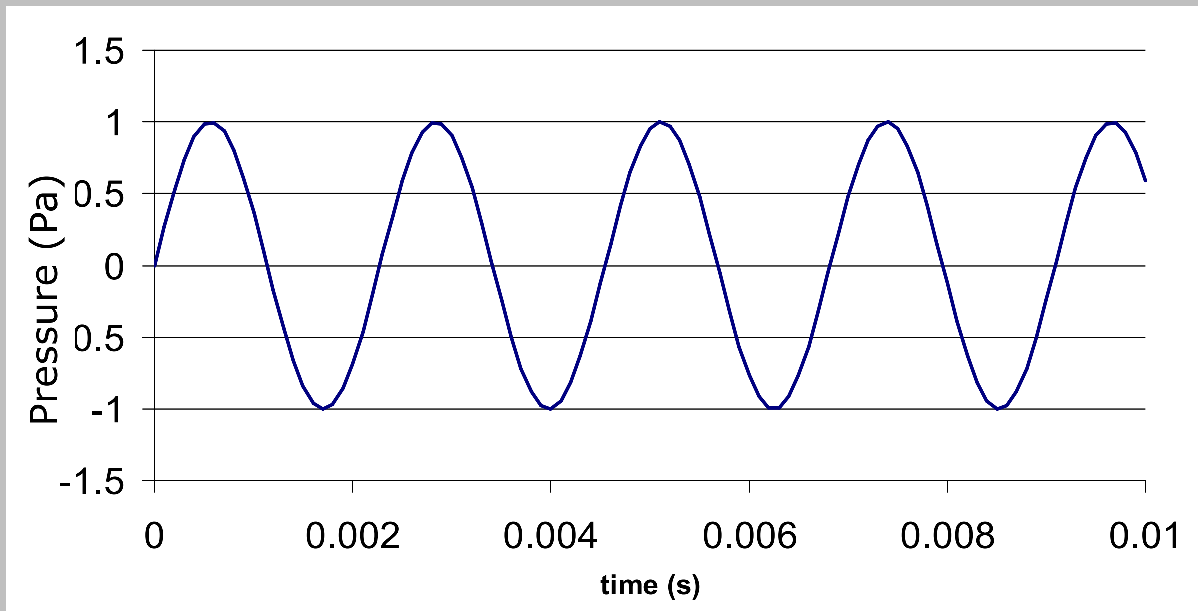


Interlude:
Go back to the lab sheet concerning audiometry

Signals as waveforms

A graph of the *instantaneous* value of amplitude over time

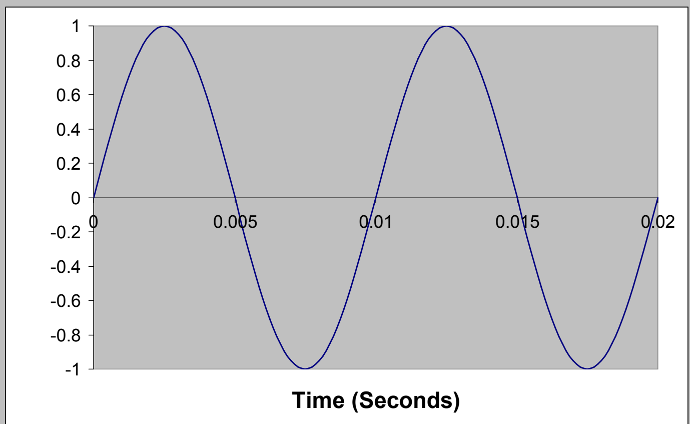
- x-axis is always time (s, ms, μs)
- y-axis always a *linear instantaneous* amplitude measure (Pa, mPa, μPa, V, m)



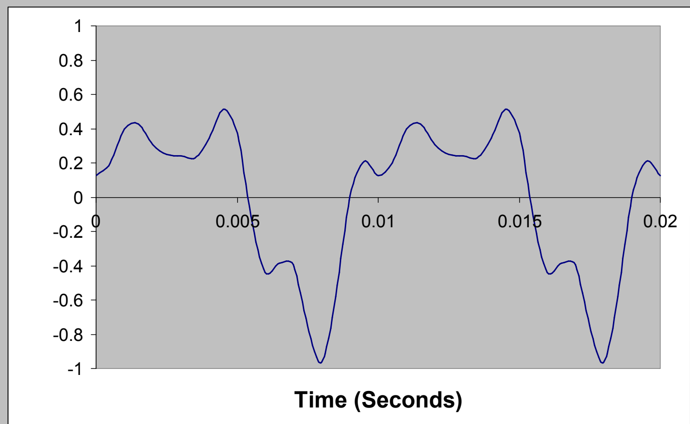
Waveforms are of two major types: periodic and aperiodic

- Periodic waveforms
  - Consist of a basic unit or *cycle* …
  - that repeats in time …
  - typically have a strong pitch …
  - and also come in two types

simple (= sinusoid)



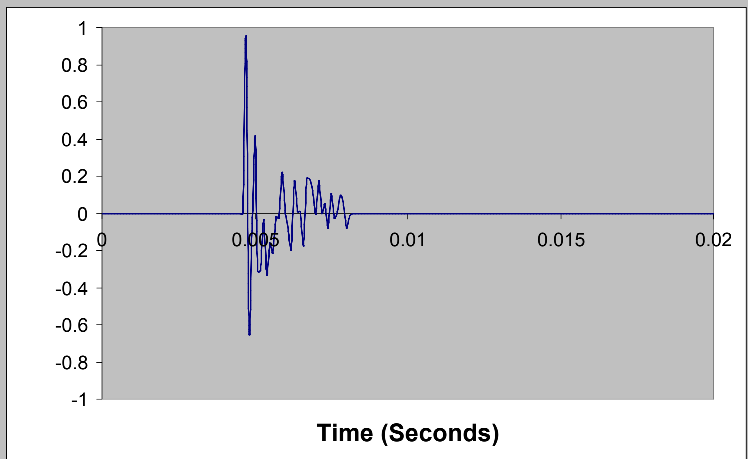
complex



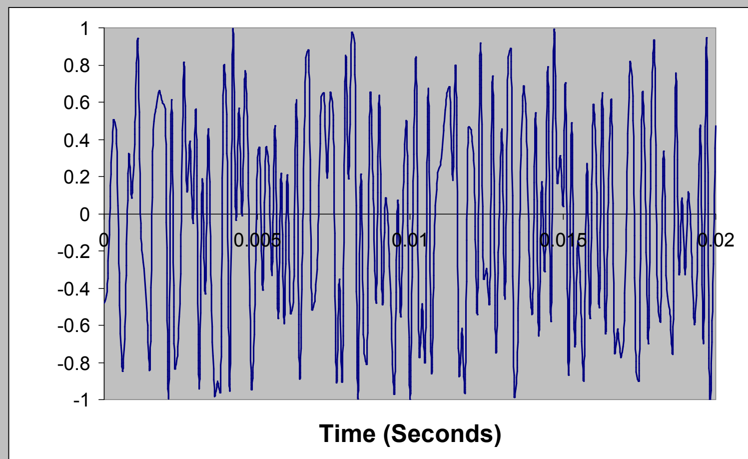
Waveforms are of two major types: periodic and aperiodic

- Aperiodic waveforms
  - do not repeat …
  - and also come in two types (but the distinction is not so important as for periodic waves)

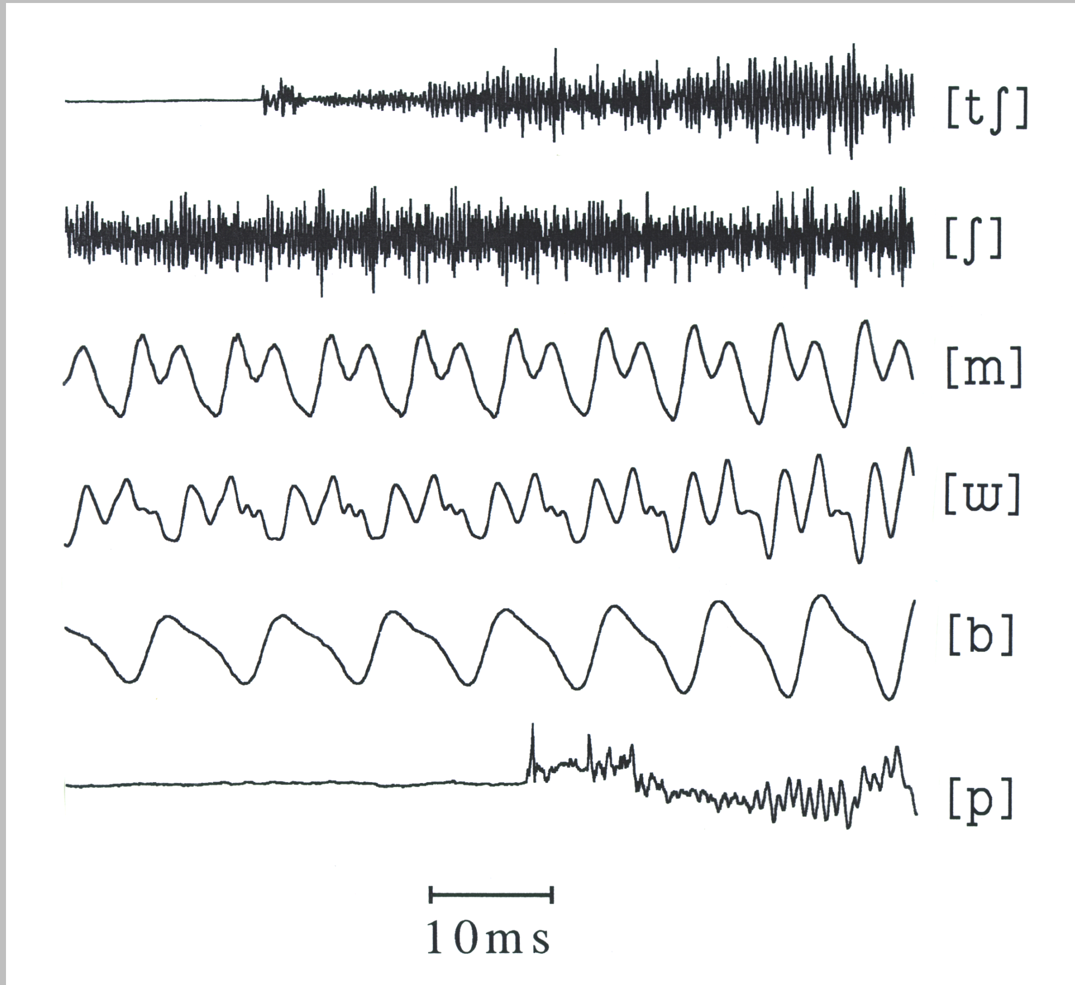
transient



continuous (can be random)



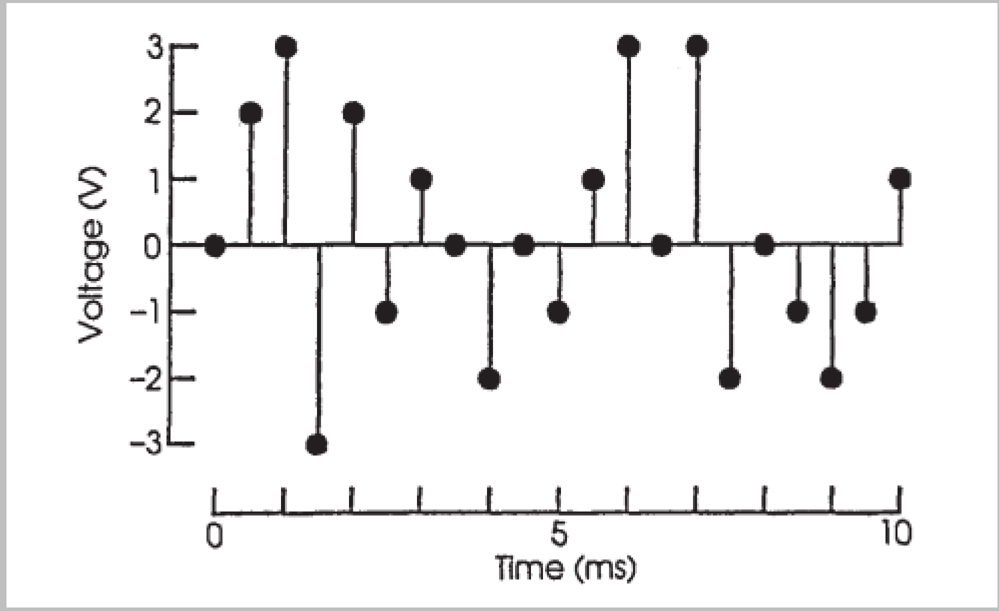
A variety of waveforms



Digital signals

How can we store a waveform on a computer, when a computer can only store a list of numbers?

Time (ms)	Amplitude (V)
0.0	0
0.5	2
1.0	3
1.5	–3
2.0	2
2.5	–1
3.0	1
3.5	0
4.0	–2
4.5	0
5.0	–1
5.5	1
6.0	3
6.5	0
7.0	3
7.5	–2
8.0	0
8.5	–1
9.0	–2
9.5	–1
10.0	1



Digital signals

- How do we go from an *analogue* waveform (like a real sound) to a *digital* one (on a computer)
- Two problems
  - Analogue waveforms exist at infinite points in time (x-axis)
  - Analogue waveforms have amplitude values (usually) with an infinite number of decimal places (y-axis)
- So a waveform has to be converted in some way in order to represent it as a list of numbers
  - On the x-axis : sample
  - On the y-axis : quantise

The rules for sampling and quantisation

- Sampling needs only be twice the rate of the highest frequency component in the signal
  - and going to higher sampling rates doesn’t help
- Quantisation keep improving as more discrete levels (number of bits) is allowed
  - but 16-24 bits is probably enough for audio applications
- Windows standard is 16 bits at 44.1 kHz
- mp3 requires an additional (and complex) processing step!

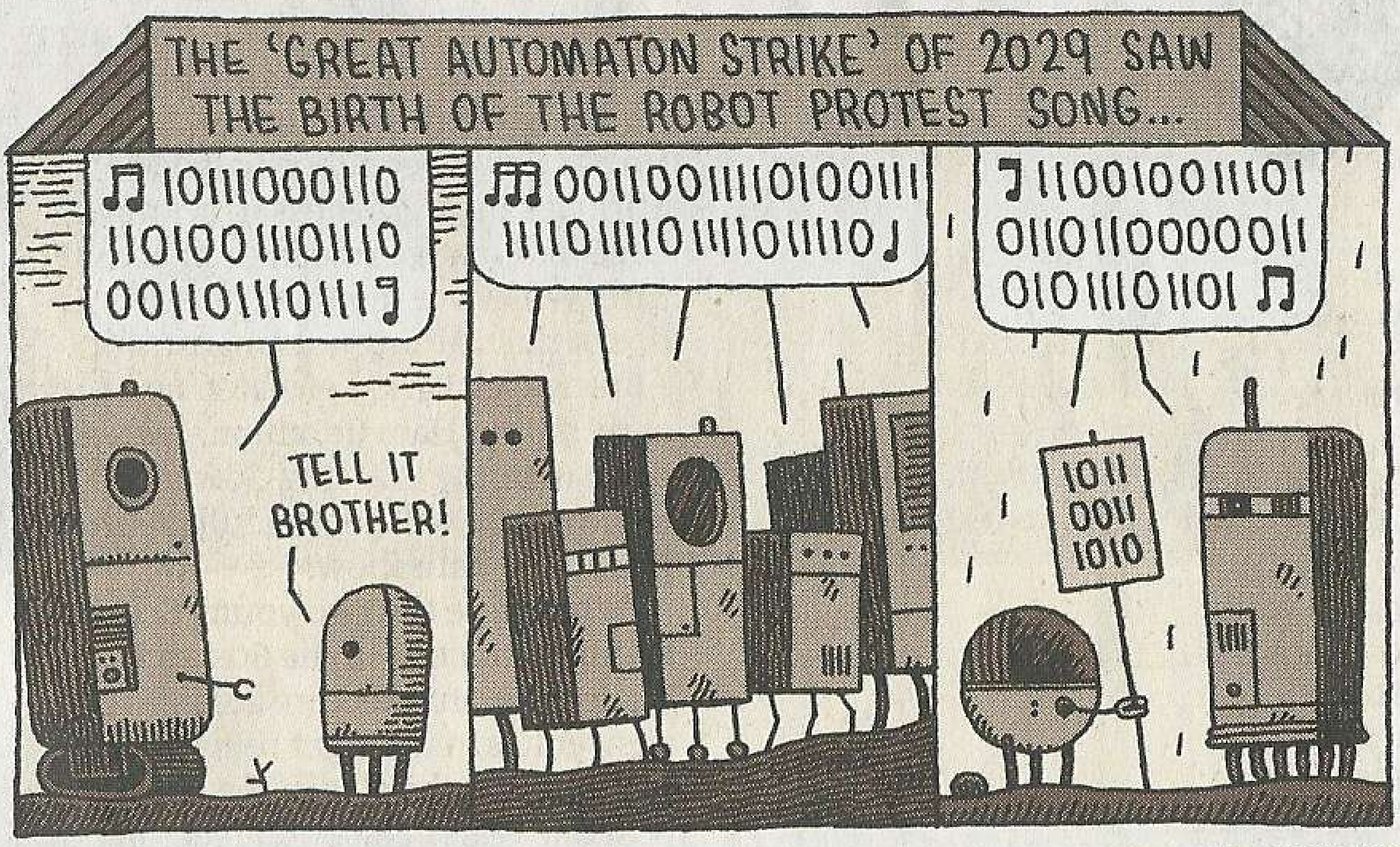


ILLUSTRATION BY TOM GAULD

The End