

Overview

- # A little science acoustics, anatomy, Heinrich and Harry
- # A little technology recording, analog/digital, disks
- # A little history Needles and magnets and tracks. Oh my!

* A little production – Produce a little audio (a Ringtone!)

Natural Sound (Acoustics)

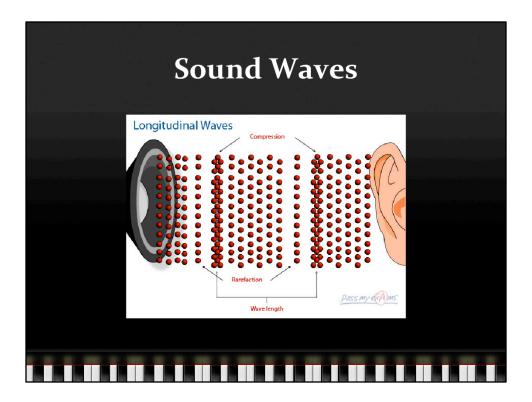
- # Sound = waves in a medium (air, water, bone)
- # Longitudinal (sound) vs. transverse (rock in pond)
- **♯** Wave types
 - # Impulse single pulse
 - * Noise random wave, ongoing but no repetition
 - ₱ Pitched repeating waveform
 - **♯** Sine waves simplest, building blocks for more complex

.

Single pulse: slap 2x4s together

Noise: rub two pieces of sandpaper together Pitched: play a couple of notes on trumpet

Sine: strike a tuning fork?

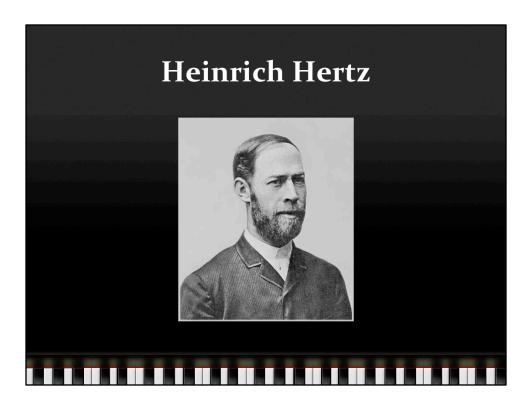


From http://passmyexams.co.uk/GCSE/physics/Basic%20Waves%20Theory.html

Wave Attributes # Amplitude – How compressed the peaks are # Wavelength (λ) – How far apart the peaks are # Speed (e) – How fast the peaks move through the medium # In air – 343 meters/second, 768 mph, about 5 seconds per mile # In water – 1484 m/s, about 4.32 times faster # In bone – 2650 m/s (Why bone?) # Frequency (f = c / λ) in Hertz (cycles/sec) – related to "pitch" # Concert A = 440 Hz (to tune an orchestra) # λ = c / f = (343 meters/sec) / (440 1/sec) = 0.78 meter # f stays put in different mediums => λ changes with c # Why Hertz?

Why bone? Because that's a big part of how you hear yourself talk, which is one reason why recordings of yourself sound so weird.

Wavelengths in feet for frequencies in the audible sweet spot become an issue for multiple mic placements in recording (phase effects) – Don't go there for this lecture...



From http://www.scientific-web.com/en/Physics/Biographies/HeinrichRudolfHertz01.html

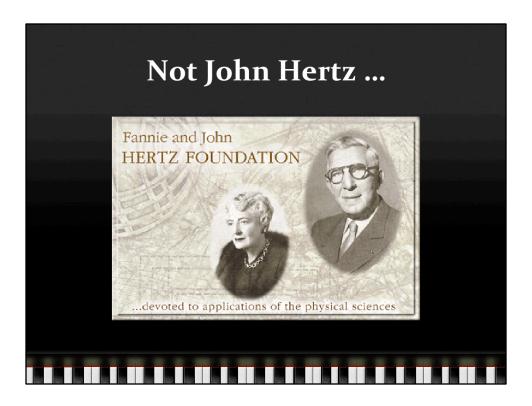


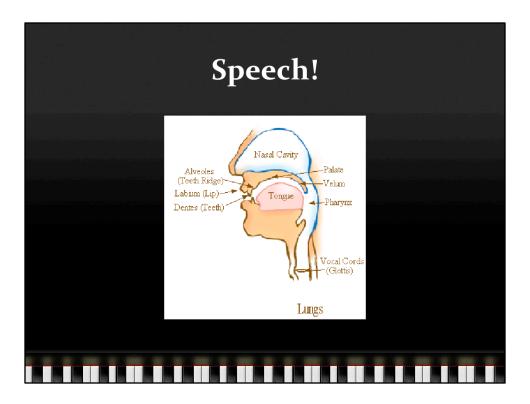
Image from http://www.jhu.edu/advising/scholarships/infopages/hertz.htm



Image from http://gustomobil.wordpress.com/2009/07/11/asyiik-bisa-sewa-lotus/

Making (Sound) Waves * Set up a disturbance in the air * Lots of things do this (listen right now) * Source-filter theory – useful for modeling most sounds * Source generates a waveform * Filter shapes it * Trumpet * Vibrating lips = source * Resonance of the tube = filter

Do resonant tuning demo with trumpet



All this stuff from http://home.hib.no/AL/engelsk/seksjon/SOFF-MASTER/Organs.htm

The speech sounds often have their names from the Latin name of the vocal organ:

Nasal sounds: through nose (velum down)
Oral sounds: through mouth (velum up)

Stops: full oral closure

Fricatives: partial oral closure (friction) **Approximants**: narrowing (no friction)

Labial: from labium, lip(s) active Dental: from dentes, teeth active Alveolar: Alveoles, teeth ridge active Palatal: Palate, hard palate active Velar: Velum, soft palate active Glottal: Glottis, vocal cords active

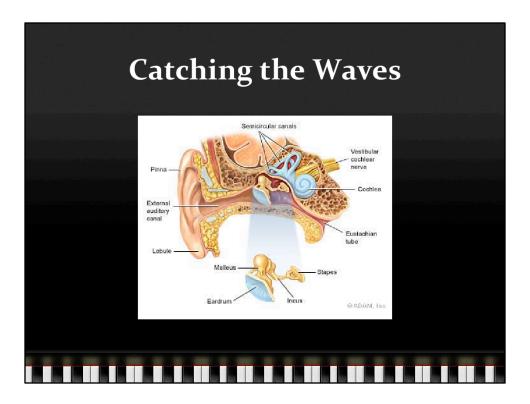
Lots of Frequencies * Real sound has lots of frequencies stacked up * Called harmonics, overtones, partials, formants (speech) * Each harmonic is a sine wave, which means ... * ... can represent any sound by a collection of sine waves

♯ Can see harmonics in frequency spectrogram

- # Trumpet
- # Speech

Bring up audacity to record stuff and display spectrograms

Waveform is in time domain: X axis is time, Y axis is amplitude Spectrogram is in frequency domain: X is time, Y is frequency, darkness is amplitude



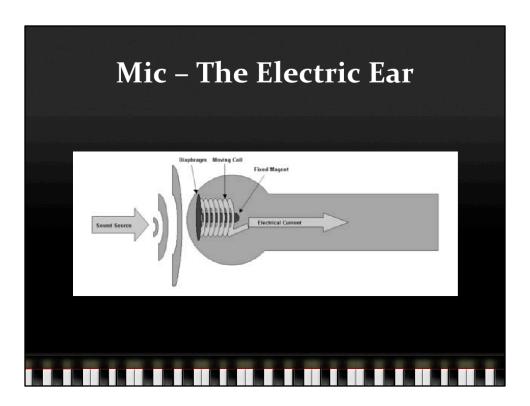
From http://www.nytimes.com/imagepages/2007/08/01/health/adam/1092Earanatomy.html

Waves get to your head, bounce around your head/ear flap, work down ear canal Waves reach ear drum and excite it in sympathy Moves the bones (hammer, anvil, stirrup)

Stirrup kicks the cochlea, which moves the liquid inside

Liquid moves hair cells on wall of cohchlea, which fire and send signals down cochlear nerve to brain

Brain interprets, etc., etc., etc.



From http://www.soundware.co.uk/pages/microphones

Mic, basically takes waves to the stirrup step in the ear – Sets up continuous (analog) current flow in sympathy with waveform much as the stirrup sets up sympathetic vibrations of fluid in the cochlea.

The nerve pulses through the cochlear nerve are more in the frequency domain, while the current coming from the mic is strictly in the time domain.

Recording

- * Capture and store the waves on something ...
- # ... that can be used to (re)generate them at a future time
- # Basically a time machine for a time-based medium
- * Mechanical (Analog) Needles making scratches on moving tinfoil, wax, shellac, vinyl
- # Electric (Analog) Continuous voltage curves stored as magnetic fields on moving wire, cylinder, tape
- * Digital Voltages sampled at discrete times and stored on tape, disk, flash memory as sequence of discrete numbers

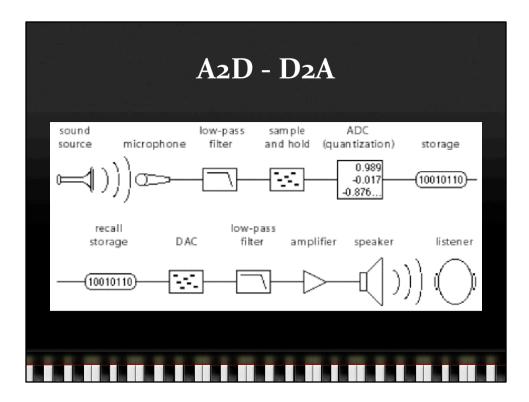


From http://pagesperso-orange.fr/jlf/enphonos.htm

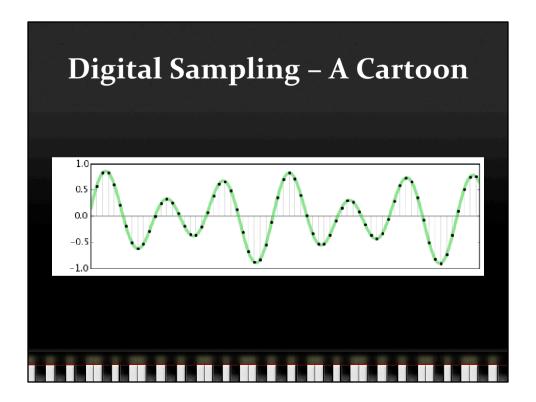
Go to this website to hear modern recordings of the old records played on these machines.



From http://www.recording-history.org/HTML/musictech6.php



From http://www.cycling74.com/docs/max5/tutorials/msp-tut/mspdigitalaudio.html Low pass filter eliminates frequencies that would be aliased on A2D Quantization maps the floating point sample values to n-bit integers DAC generates smooth voltage curves from discrete samples

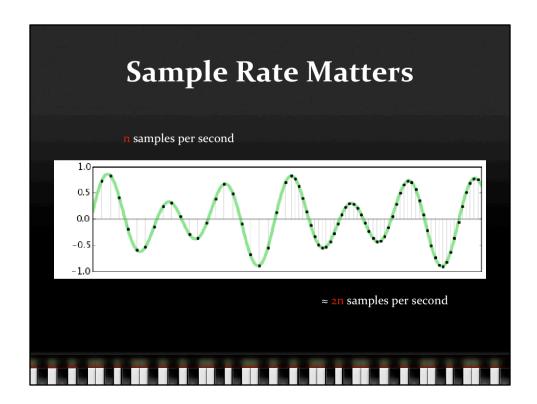


From http://manual.audacityteam.org/index.php?title=Digital Audio

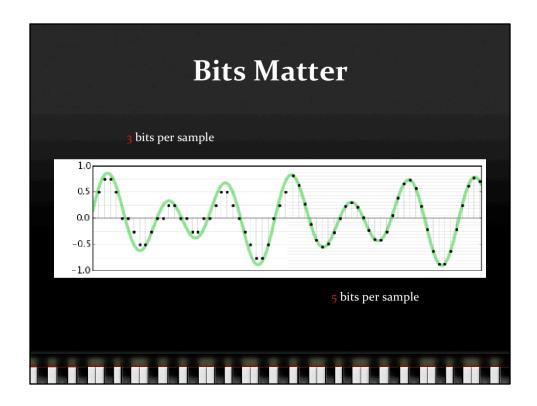
Smooth green voltage curve sampled at discrete points in time, results in a discrete (digital) approximation of the continuous waveform, basically a histogram

Quality!

- # How the sampling is done matters
 - * Recording environment (ambience, noise floor, etc.)
 - # Microphone quality (frequency response, etc.)
 - # How often we take a reading (sample rate)
 - # How accurately we measure (quantization)
 - ∦ How far we twist the volume knob (headroom)
 - # How high the frequencies we're sampling are
- **♯** Lots of opportunities to screw up
- # Once recorded, quality can only decrease

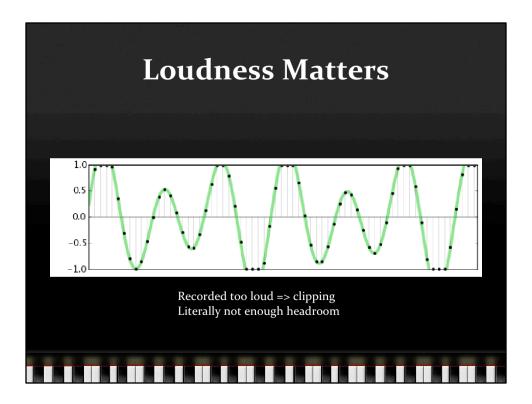


From http://manual.audacityteam.org/index.php?title=Digital_Audio



From http://manual.audacityteam.org/index.php?title=Digital_Audio

8 amplitude levels on the left, 32 on the right

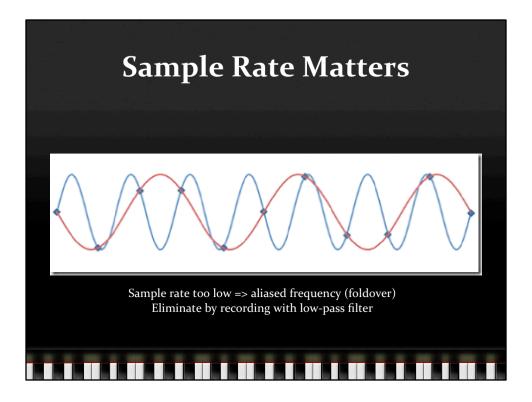


From http://manual.audacityteam.org/index.php?title=Digital_Audio

Literally hitting your head on a ceiling (not enough headroom)

If recorded too soft, you're not taking advantage of all the bits (not using the available headroom)

Any clipping is bad, especially if it's with repeating waveforms because it imposes a square wave over the waveform at high amplitudes



From http://blogs.msdn.com/audiofool/archive/2007/02/15/digital-audio-aliasing.aspx

Any frequency above the Nyquist frequency (half the sampling frequency) will be aliased down into the sub-Nyquist-frequency range

Basically it bounces linearly off the Nyquist frequency ceiling.

For real frequencies between the Nyquist frequency and the sampling frequency (S), the aliased frequency is the sampling frequency – real frequency or ...

f(aliased) = S - f(real), for S/2 < f(real) <= S

Frequencies above S get aliased too, bouncing off 0 (the floor), but we don't see this in typical audio.

Nyquist Theorem (1929)

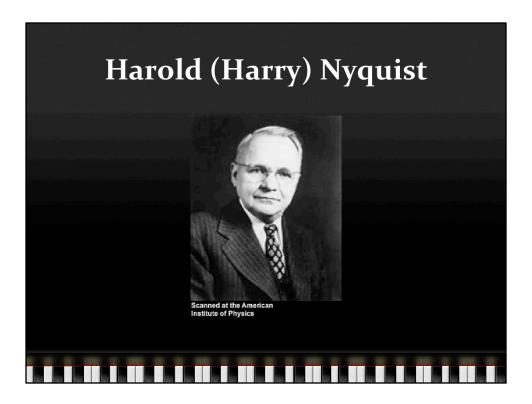
- * Need at least 2 samples of a waveform to represent it
- * So, highest represented frequency = half the sampling rate
- # Nyquist frequency, then = half the sampling rate (S/2)
- * Above that, frequencies "fold over" @ Nyquist frequency
- $\# f(aliased) = S/2 (f(real) S/2), S/2 < f(real) \le S$
- # f(aliased) = S f(real)
- # Named for Harold Nyquist

For sampling rate S and S/2 < f(real) <= S: f(aliased) = S/2 - (f(real) - S/2) = S/2 + S/2 - f(real)= S - f(real)

Draw diagram on board:

X axis = real f from 0 to S

Y axis = aliased frequency from 0 to S/2, draw ceiling at S/2 (horizontal line) Graph is a tent that touches the S/2 line



From http://www.geocities.com/bioelectrochemistry/nyquist.htm

Harold (Harry to his friends and everyone else) Nyquist

Choices for "CD Quality"

- # Sampling rate = 44,100 Hz (44.1 kHz)
 - **♯** Nyquist frequency = 22.05 kHz
 - # Humans can hear from 20 Hz to 20 kHz
- # Quantization = 16 bits
 - **#** 65,536 different amplitude values
 - ***** Roughly FM radio quality
- # Can increase both: ≤ 128 kHz ≤ 32 bits
 - # Useful while processing sound to avoid round-off errors
 - # Once you're finished, master to CD quality
- # Stereo: 2 ears => 2 channels, could be more (5.1 +)

Music Production Time Line

- # Sheet Music (2000 BC, 1300's) Somebody has to perform it
- **♯** Stored performance "Record" performance gestures
 - # Piano Rolls (1895 1929) Player piano generates sound
 - # MIDI (1983) Electronic synthesizer generates sound
- * Recording (1887) Record actual sound of performance
 - # Mechanical (1887 1925) Horn picks up sound
 - # Electrical (1925) Mic picks up sound
 - # Multi-track Recording (1950's) Record parts separately, then mix
 - * Stereo imaging (1955) Use 3 mics for localization (3.0)
 - # Multi-track Production (1960's) Beyond fidelity to imagination
 - # Digital multi-track (1980's) More tracks, better quality
 - # Home recording (1990's) Digital multi-track @ home

Sheet music:

2000 BC Cuneiform tablet in Nippur (Sumerian city)

 6^{th} century BC Greeks had notation for pitches, durations, limited harmonies Mid 1000's Guido D'Arezzo invented stave (four-line)

1300's Today's 5-line staff introduced

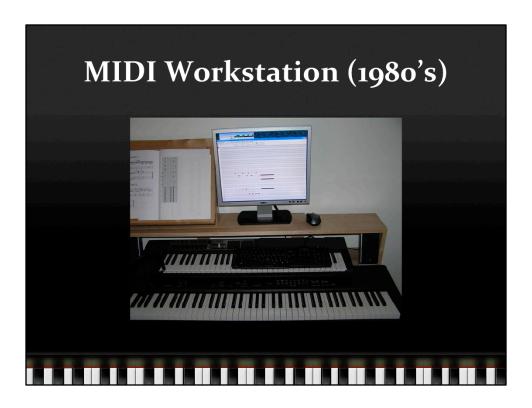
Player piano companies: largely killed off by 1929 crash and great depression MIDI = Musical Instrument Digital Interface

Recording history: long and convoluted ...



http://www.immortalpiano.com/?page=player-pianos

Pneumatic technology – Have to pump foot pedals to play it Killed off by 1929 crash and subsequent depression



http://shazware.com/piano/rig.html

Shows "piano roll notation" on screen in MIDI sequencer software.



http://www.mainspringpress.com/studio_photos.html

Tenor Jacques Urlus and the boys recording a disc in Edison's Fifth Avenue studio (New York), 1916. (ENHS)



http://www.mainspringpress.com/studio_photos.html

The International Novelty Orchestra, a Victor house group under the direction of Nat Shilkret (center, holding baton), prepare to get down and get funky, after the introduction of Western Electric equipment. (LOC)



Les Paul and Mary Ford layering who knows how many tracks with the old Ampex 8-track recorder

http://www.classicaldomainli.com/lespaul2.html

Timeline from http://en.wikipedia.org/wiki/History_of_multitrack_recording 1947 Paul releases first multi track recording (8 guitar parts layered one track at a time with shellac disks!)

1948 Paul got first Ampex reel-to-reel tape recorder as gift from Bing Crosby, modified it with multiple play and record heads to facilitate layering (sound on sound)

1953 Paul commissioned Ampex to build first 8-track recorder (custom)

1955 First commercially available 2 and 3 track analog tape machines available

1958 Atlantic first studio to install 8-track

1968 Beatles use two 4-track recorders for Sgt. Pepper, 8-track for White Album



http://www.soundfountain.com/amb/mercury.html

Really three channels (3.0 in today's parlance), probably because the best tape machines at that time were 3-track Ampex machines



http://www.bbc.co.uk/programmes/b00ml584

They weren't the first, but they exploited it the most by using multi track as a compositional technique, not just for recording

Before 1964, they recorded 2-track mono

With Sgt. Pepper, used two 4-track recorders and lots of effects (pitch shifting, ADT (automatic double tracking), flanging, reverse, etc.

Treated recorded track as source for effects that altered the content in artistic ways



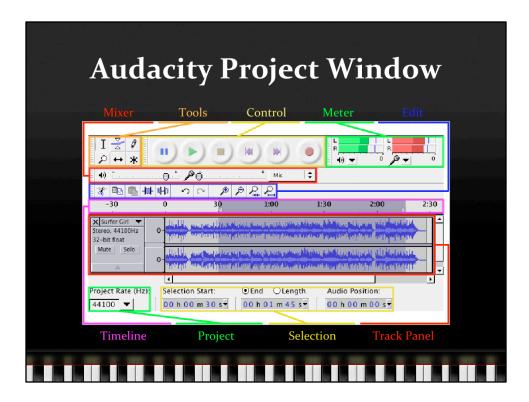
http://www.seymourduncan.com/forum/showthread.php?t=156523

Bring the studio to the artist, instead of the other way around Once recording went digital, it became an "IT" problem, subject to Moore's law, with predictable results.

These days, who needs to pay a recording studio? Just put that money into gear!

audacity - Free Stuff!

- # Audacity open source digital audio production tool
- * Available from http://audacity.sourceforge.net/
- # Cross platform (Mac, Windows, Linux)
- ♯ Record and playback using computer's built-in soundcard
- # Import and export sound using lots of formats
- # Multi-track editing and mixing
- # Lots of effects
- # Bounce to stereo or monaural master



http://manual.audacityteam.org/index.php?title=Quick_Guide

Make Your Own Ringtone!

- # A fun, constrained mini-project
- # Time limited Up to 15 seconds or so
- # Monaural (1 track) Not stereo for most cell phones
- # Play on tiny speakers
 - # Narrow frequency range (no bass) => EQ
 - ★ Narrow dynamic range => Compressed loudness
- # Smallish file (< 300 kb) => Compressed file format (.mp3)
- # Should catch attention Not subtle
- ♯ Suitable for public listening G-rated content

Make Your Own Ringtone!

- ♯ In-class exercise using audacity
- # Use at least 3 sound source files (multi-track editing)
- * No longer than 15 seconds (Ignite-slide length)
- # Mix down to monaural (one track)
- # Maybe make it loopable?
- # Generate, record or download source sounds
 - # http://www.aifreesoundeffects.com/
 - # http://www.pacdv.com/sounds/

Uploading to your Phone

- * Different for different phones/providers YMMV
- **♯** Supported file formats vary
 - # Wav, mp3, MIDI, etc.
 - # Monaural vs. stereo
 - ***** Sample rates & bit depths vary
- ♯ Supported file sizes vary tiny to few hundred kb
- ♯ Supported playback variations vary loopable/not
- # Up to you to figure out how after you create your audio file
- # Websites to help uploading like http://www.myxer.com/make/

A Quick Demo * 3 sources: 1 generated, 1 recorded, 1 downloaded * Collapse stereo tracks to monaural * Use Time Shift Tool to arrange them in sequence