KARYN LAUDISI
SUMMER, 2018

LECTURE 01
APRIL 10, 2018

AUDIO BASICS

188.458 AUDIO PRODUCTION

ABOUT THIS COURSE

- not a music production course
- focus will be on small audio projects (e.g. podcasts)
- practice editing, working with digital signal processors, simple recording methods

PROJECT OVERVIEW

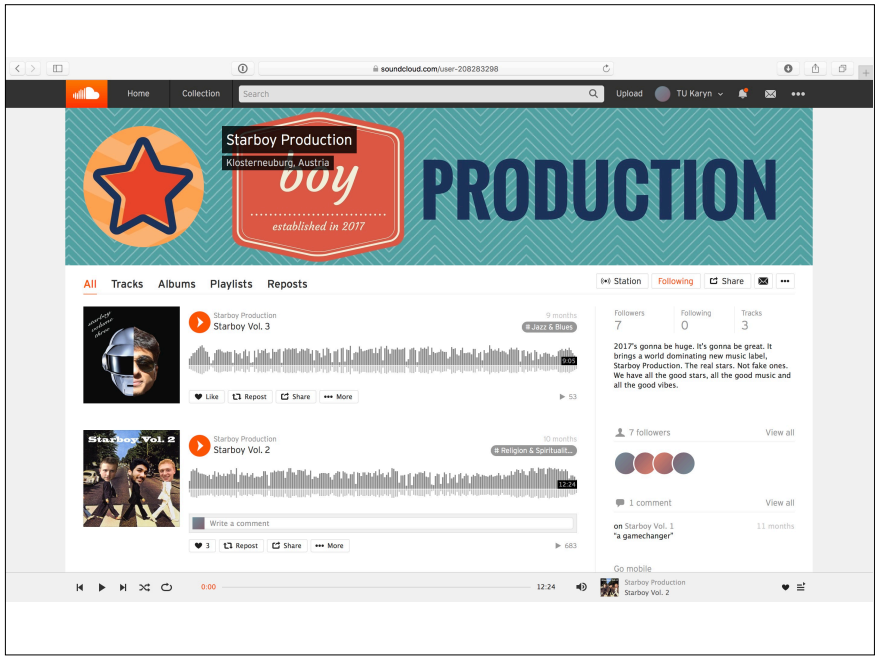
PODCAST REQUIREMENTS

EPISODE 1

DUE APRIL 23RD

Each podcast must contain the following:

- Album art
- 1x 15 second commercial or PSA
- Podcast notes (links, info pertaining to the show, can be listed on SoundCloud)
- Runtime between 10 and 20 minutes
- Musical bumpers (intro and outtro)
- Sound effects
- **Story/Narrative**



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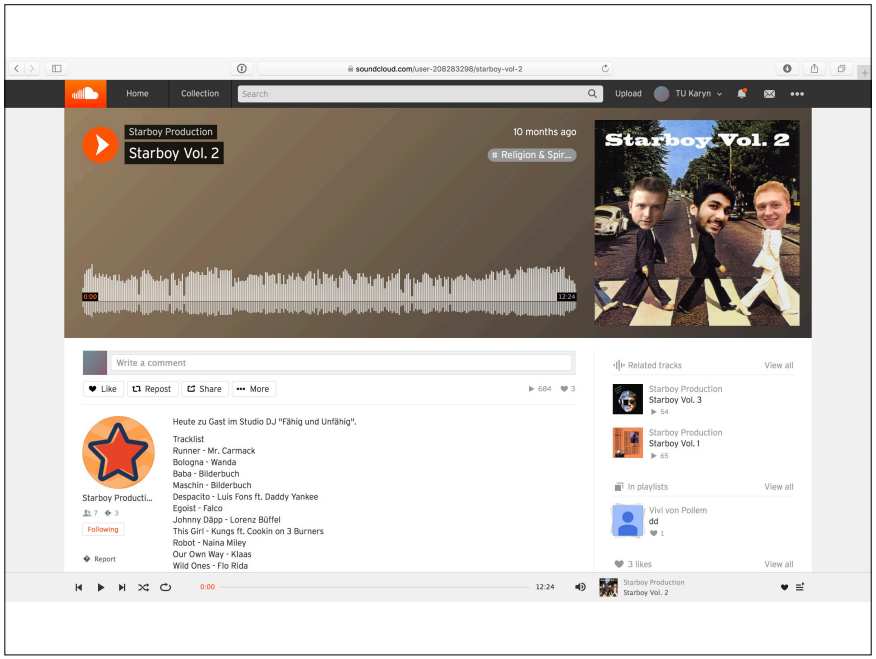

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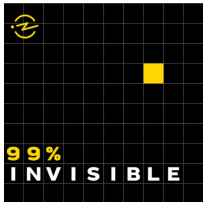

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1 INTRODUCTION TO THE STUDIO

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STUDIO TYPES

PROFESSIONAL RECORDING STUDIO

- designed to capture the best possible sound
- multiple isolated recording rooms
- elaborate & expensive in design

STUDIO TYPES

AUDIO-FOR-VISUAL PRODUCTION STUDIO

- production facility for video, film & game post-production
- scoring, score mix-down, Foley

STUDIO TYPES

PROJECT STUDIO

- majority of studios
- professional to private (home studio)
- music recording, multimedia production, A-V

STUDIO TYPES

PORTABLE STUDIO

- laptop, all-inclusive portable recording system

2 BASICS OF SOUND

WHAT IS SOUND?

- disturbances in the air caused by vibrations
- vibrations produce sound waves
- sound is part physical and part perceptual

WHAT IS SOUND?

PHYSICAL

- energy that travels through a medium
- propagates itself by traveling in waves

PERCEPTUAL

- something we (as humans) perceive as being noise, music, loud, soft, etc.

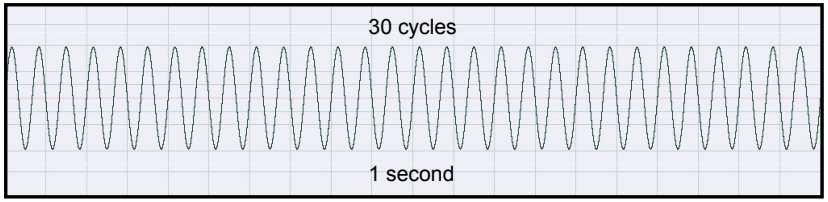
WHAT IS SOUND?

Physical	Perceptual
Frequency	Pitch
Amplitude	Loudness
Harmonics	Timbre
Envelope	Articulation

FREQUENCY

FREQUENCY

- the rate at which a sound wave completes one cycle in one second
- Hertz (Hz) or kiloHertz (kHz)
- 1 cycle/sec = 1 Hz
- 1, 000 cycles/sec = 1 kHz



AUDIO SPECTRUM

- general range of frequencies (frequency bands):

  300 Hz and below = Low

  300 Hz - 3.5 kHz = Mid-range

  3.5 kHz and above = High

- frequencies we perceive to hear are between 20 Hz & 20 kHz; more accurately from about 35 Hz to 16kHz
- human speech can be found as low as about 110 Hz to as high as 10 kHz

AUDIO SPECTRUM

	Range	Description
Low Bass	20 - 80 Hz	power (hum); begin to hear piano, tuba, bass
Upper Bass	80 - 320 Hz	musical structure; fullness
Midrange	320 Hz - 2.5 kHz	intensity; human voice
Upper Midrange	2.5 - 5 kHz	humans most sensitive; intelligibility of human voice; perceive sound source
Treble	5 - 20 kHz	brilliance

PITCH

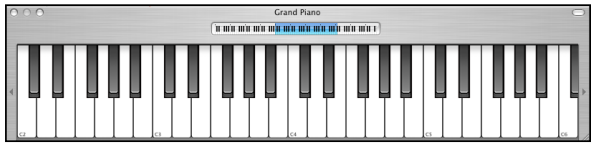
PITCH (TONHÖHE)

- frequency is perceived as pitch
- the highness or lowness of a sound (relative)
- middle C on the piano keyboard vibrates at a frequency of 261.6 Hz, Concert A vibrates at 440 Hz: we hear A as a higher pitch as C
- 1 octave is the doubling of a frequency
- the human ear hears about 10 octaves

PITCH

Middle C	C above Middle C
261.6 Hz	532.2 Hz

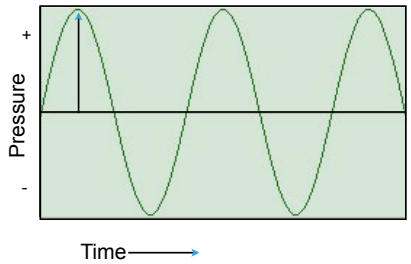
Concert A	Octave above Concert A
440.0 Hz	880.0 Hz



AMPLITUDE

AMPLITUDE

- molecules' maximum displacement from the equilibrium
- dependent on a sound's intensity
- perceived as *loudness*



WAVE PROPERTIES

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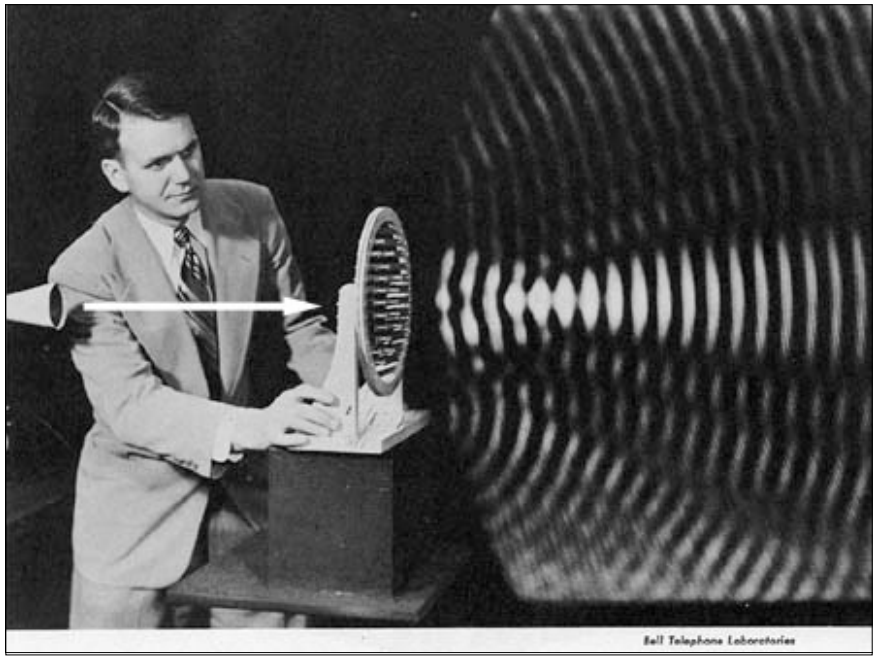

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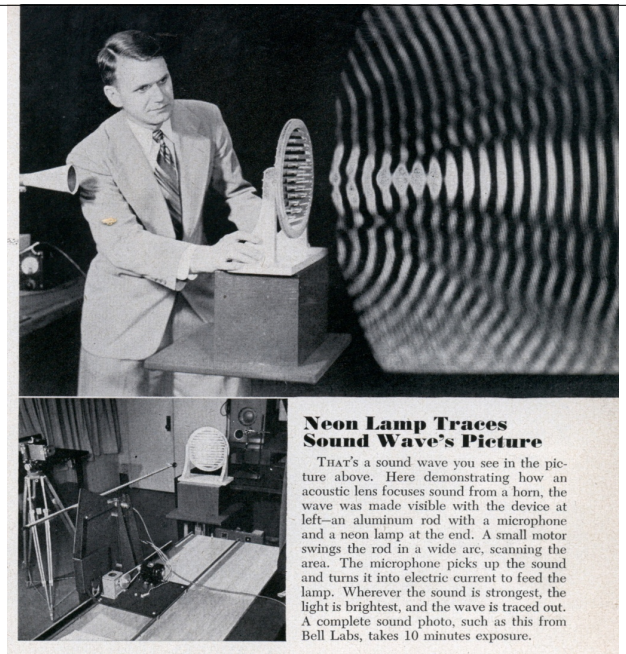

Neon Lamp Traces Sound Wave's Picture

THAT's a sound wave you see in the picture above. Here demonstrating how an acoustic lens focuses sound from a horn, the wave was made visible with the device at left—an aluminum rod with a microphone and a neon lamp at the end. A small motor swings the rod in a wide arc, scanning the area. The microphone picks up the sound and turns it into electric current to feed the lamp. Wherever the sound is strongest, the light is brightest, and the wave is traced out. A complete sound photo, such as this from Bell Labs, takes 10 minutes exposure.

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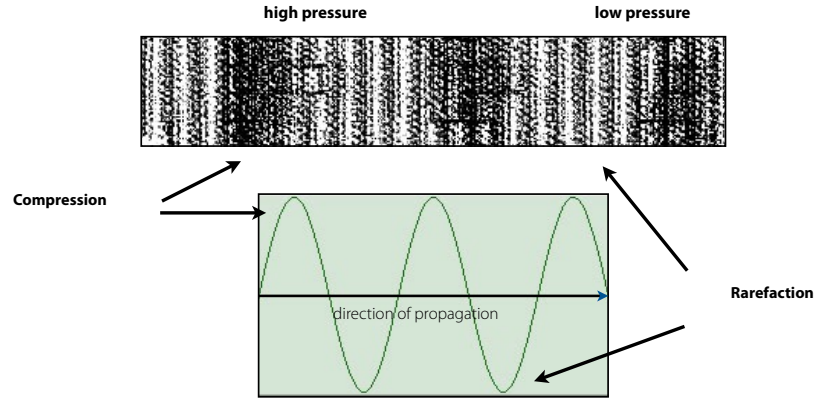
WAVE PROPERTIES



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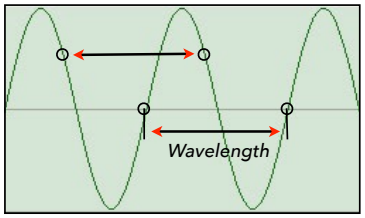
WAVELENGTH

- physical size
- distance between two identical points
- horizontal length of the wave



λ = Tc

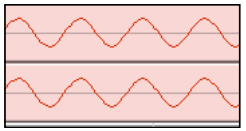
λ = wavelength

T = period (s)

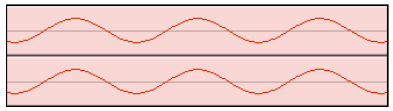
c = propagation speed (ft or m)

WAVELENGTH

- faster something vibrates, the shorter the wavelength
- slower something vibrates, the longer the wavelength



1kHz



Concert A440

WHY IS IT IMPORTANT TO KNOW A WAVE'S SIZE?

ELECTROMAGNETIC WAVES

- need antenna/wire to transmit/receive – need a sufficient cable/ antenna based a wave's length

ACOUSTICAL WAVES

- can cause acoustical problems when the wavelengths are short and radiated in a large room (reflections)

SOUND MOVEMENT

REFLECTIONS

- persistence of a sound after it's source has stopped
- sound bouncing off of hard surfaces

2 PHENOMENON:

- echo
- reverberation

ECHO

- a reflection that has significant time delay
- single or multiple repetitions of sound with a **fixed timing**

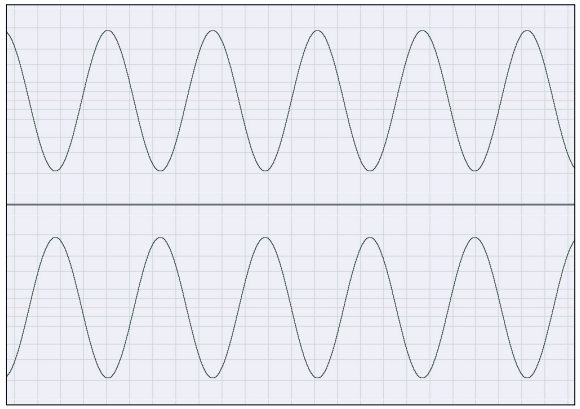
REVERBERATION

- remainder of sound after the source has stopped
- collection of many reflections
- adds spaciousness to a room
- full and partial cancellations of frequencies

PHASE

- a particular point in the time of a cycle
- sound waves: refers to the time relationship of two or more waves at a given point in their cycles
- summing and canceling of waves
- *in phase*: identical waves with their compression and rarefaction cycles coincide with each other
- *out of phase*: identical waves do not coincide with each other causing them to cancel each other out caused by delay

180° out of phase



INVERSE SQUARE LAW

- any point source (sound, light) that spreads equally in all directions without any limits to it's range will obey this law
- the intensity of sound diminishes with the square (√) of the distance
- double the distance from a sound's source, it's sound pressure becomes 6dB less – if in a room/space without echoes (free field)

SOUND AS AN ANALOG SIGNAL

ANALOG SIGNAL

- continuous representation of changes in sound pressure (acoustic sound)
- result of an acoustical signal that has been converted into an electrical current (AC signal)

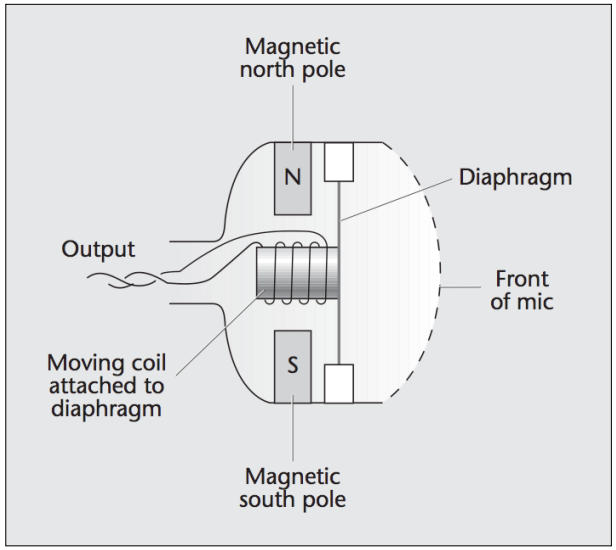
TRANSDUCERS

- convert an acoustic signal to an electrical signal
- the electrical signal creates a changing voltage that correlates directly to the sound wave:
- compression = positive voltage
- rarefaction = negative voltage

TRANSDUCERS

TWO MAIN TYPES

- microphone
- speaker



TRANSDUCERS

ELECTRODYNAMIC DRIVER

- type of transducer used in a loudspeaker
- cone driver – diaphragm – what pushes and pulls air
- coiled wire - voice coil
- magnet

LEVELS

LINE LEVEL

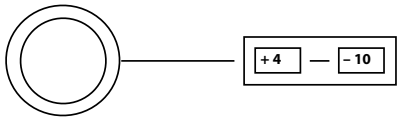
- standard voltage for the signal output of audio equipment
- 2 standards (normal operating voltage)
- + 4 dBu (1.23 V)
- – 10 dBV (.316 V)

LEVELS

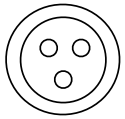
MIC LEVEL

- voltage generated by a microphone
- about 40-80 dB lower than line level

LEVELS



Line



Mic

WIRING

UNBALANCED WIRING

- one conductor
- one shield/ground
- susceptible to noise and hum

WIRING

BALANCED WIRING

- 2 conductors that carry the signal plus a shield (3 conductors)
- reduces noise and interference from power cables, radio signals, etc.

CONNECTORS

XLR

- 3 pins: 2 conductors, 1 shield/ ground = balanced
- connects microphones and certain line signals
- professional equipment

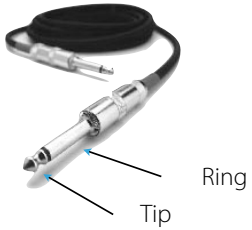
Male XLR

Female XLR

CONNECTORS

PHONE OR ¼" OR 6.5MM - MONO

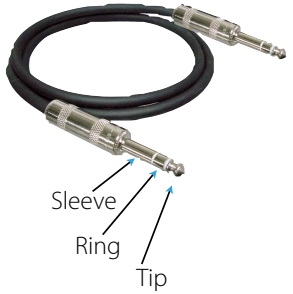
- connects audio equipment, instruments, headphones
- tip/ring or TR
- unbalanced
- connects instruments



CONNECTORS

PHONE OR ¼" OR 6.5MM - STEREO

- tip/ring/sleeve or TRS
- balanced
- connects equipment



CONNECTORS

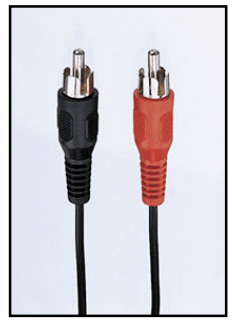
MINI OR 1/8" OR 3.5MM

- most commonly used for headphones
- used for audio in/line out on computers, minidisk players, etc.
- usually stereo

CONNECTORS

RCA

- connects audio and video
- consumer products, i.e. TVs, DVD players,etc.
- color scheme



3 PRODUCTION

DIGTIAL AUDIO WORKSTATION

- integrated computer-based hard-disk recording system
- centralized control over all digital audio processes
- multitrack recording, editing, mixing
- MIDI sequencing
- DSPs = Digital Signal Processors
- video support
- loop-based editors

AUDIO INTERFACE

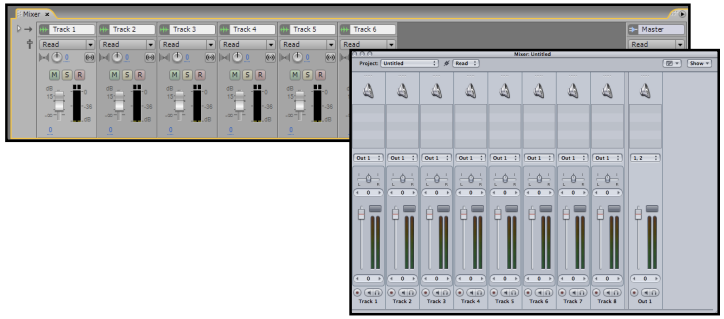
- way to connect analog/acoustic audio to digital (from mic to computer)

CONTROLLER

- mimics a full mixer (faders, pan pots, trim etc.)

VIRTUAL MIXER

- real-time mixing
- same abilities as real mixer – kind-of



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AUDIO BASICS

188.458 AUDIO PRODUCTION

4 MIKING THE INTERVIEW

AGENDA

1. Miking the Interview
2. Editing the Interview
3. Microphones
4. Microphone Characteristics

HANDHELD

- good choice for on-camera interviews or *on the street* interviews
- held under the chin; not in front of the mouth
- use a wind screen (dead cat) (face away from noise when possible)
- usually a dynamic but can also be a shotgun/condensor
- if using for controlled interview, place on mic stand

BOOM/SHOTGUN

- best choice for capturing sound on camera
- allows talent (actors) to move freely
- also good for audio-only interviews
- *audio-only*: use a mic stand, or use as handheld
- *on-camera*: use a boom, keep out of picture

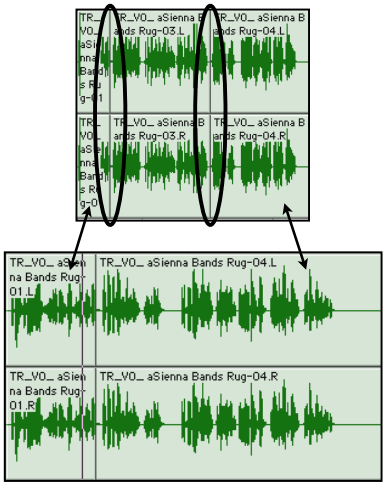
LAVALIERE

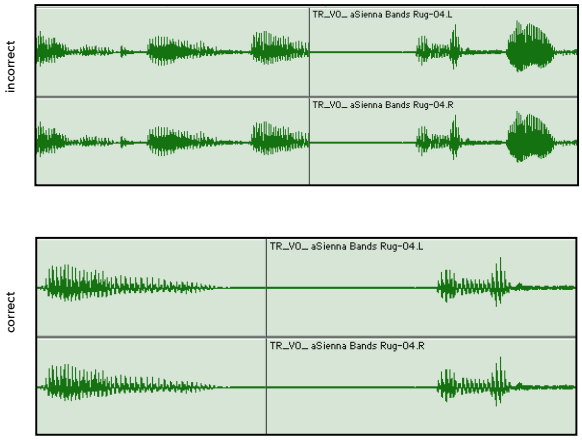
- clip on clothing/tie (less than 25 cm)
- will pick up more voice than ambience - good for noisy environments
- used for dialog on camera - easy to hide
- watch out for rustling
- usually an omnidirectional

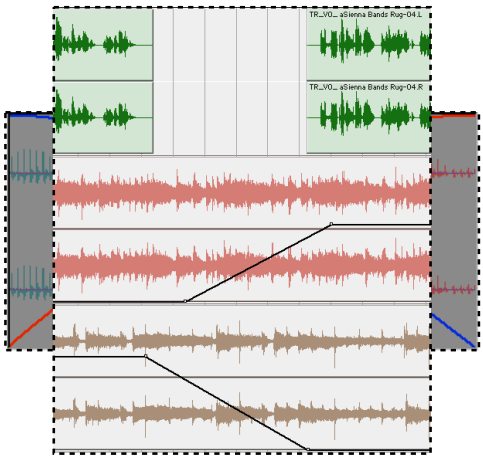
5 EDITING THE INTERVIEW

GUIDELINES

- use multiple tracks (interviewer on track 1; interviewee on track 2)
- edit from silence to silence to avoid clicks/pops
- allow the edit to be natural (breaths, "mhms," natural interruptions)
- scrub (or in video, jog or shuttle)
- zoom in
- crossfade







6 MICROPHONES

MICROPHONES

3 TYPES:

- Dynamic
- Ribbon
- Condenser

DYNAMIC

- magnetic induction
- thicker diaphragm (than ribbon or condenser) due to attached coil (rugged)
- ability to take on greater amounts of sound pressure before distorting
- ability to take on physical abuse
- live & studio environments
- **live**: subjected to abuse, weather, screaming, being dropped
- **studio**: used for close-miking drums, vocals

RIBBON

- magnetic induction
- very thin diaphragm/ribbon suspended between the poles of a magnet - moves in response to sound pressure
- very sensitive due to thin ribbon - great for low signals - not so great for handling
- low-voltage output

CONDENSER

- operates on the electrostatic principle (voltage difference)
- 2 plates:
- one very thin, stretched, electrically conductive plate (diaphragm) and one fixed backplate
- form a capacitor
- change in capacitance, due to the sound wave & difference in distance between the 2 plates, is related to change in voltage

CONDENSER

- requires phantom power
- accurate frequency response and sensitive to transients
- usually used with shock mount – due to the thin diaphragm and sensitivity to handling & noise
- used mostly in a controlled environment: studio

Shure SM58

Neumann U87

RCA 77A
(20's-30's)

AEA R84

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SUMMER, 2018

LECTURE 03
APRIL 25, 2018

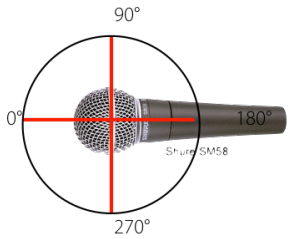
AUDIO BASICS

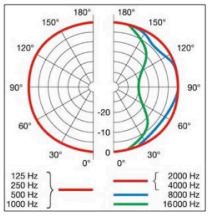
188.458 AUDIO PRODUCTION

7 MICROPHONE CHARACTERISTICS

POLAR PATTERNS

POLAR PATTERNS





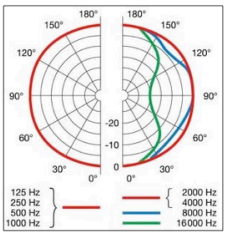
POLAR PATTERNS

OMNIDIRECTIONAL

- responds to sound pressure from all angles

USES

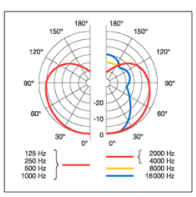
- lavaliere, boundary, stereo
- studio: multiple singers, instruments, etc.
- video: handheld interview mics, background sounds



POLAR PATTERNS

UNIDIRECTIONAL

- greater sensitivity from the front
- most commonly used
- cardioid or directional
- general use, mostly hand-held mics



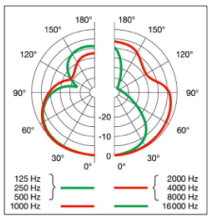
POLAR PATTERNS

HYPERCARDIOID

- variation of the cardioid
- more directional
- boom mics

SUPERCARDIOID

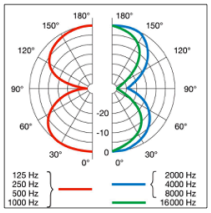
- even more directional
- ideal for when isolation is necessary (bleed)



POLAR PATTERNS

BIDIRECTIONAL

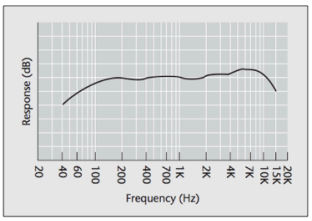
- figure 8 pattern
- picks up sounds equally from the front and back
- good for duets, face-to-face interviews



FREQUENCY RESPONSE

FREQUENCY RESPONSE

- measurement of the mic's output over the audible frequency range when driven by a constant on-axis signal (frequency-response curve)
- flat frequency response = adds little coloration



PROXIMITY EFFECT

PROXIMITY EFFECT

- low-frequency phenomenon – increase in bass response when (typically a directional) mic is in close proximity of the sound source
- "popping" with "p's" & "b's"
- appreciated by vocalists & radio DJs for the warm bass sound

LISTEN/DEMO

8 DIGITAL SIGNAL PROCESSORS

SIGNAL PROCESSORS

EQUALIZERS (SPECTRUM PROCESSOR)

- adjusts a frequency or frequency band's volume
- graphic, parametric

DYNAMIC PROCESSORS

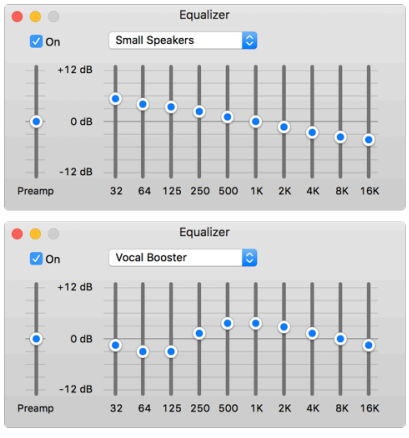
- regulate a sound's dynamic range
- compressors, limiters, expanders, gates

EQUALIZERS

EQUALIZERS

GRAPHIC EQUALIZER

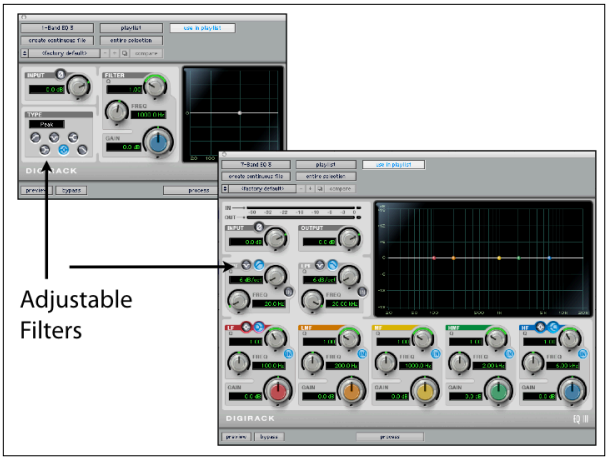
- easiest to use
- used to shape the overall spectrum of a program
- boost & cut over a series of center frequencies



EQUALIZERS

PARAMETRIC

- ability to control several parameters:
  - the amount of boost (+) or cut (-) in dB
  - dial in to a chosen center frequency
  - adjust the bandwidth/range (Q) (quality factor)
- adjustable filters



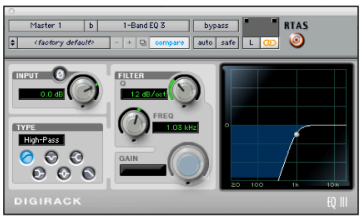
EQUALIZERS

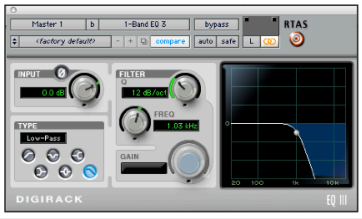
FILTERS

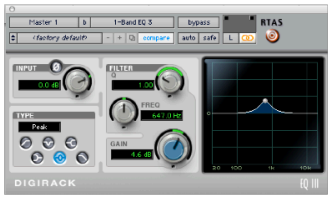
- device or component of an equalizer generally used to pass or reject signals

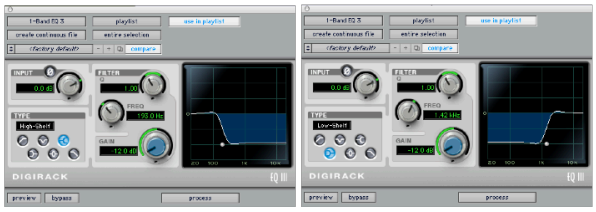
Varieties:

- Hi-Pass: high frequencies pass - lows are cut
- Low-Pass: low frequencies pass -highs are cut
- Shelf: adjusts a range of frequencies above or below a selected target frequency (Hi Shelf/Low Shelf)
- Peak/Notch: cut or boost frequencies around a selected frequency





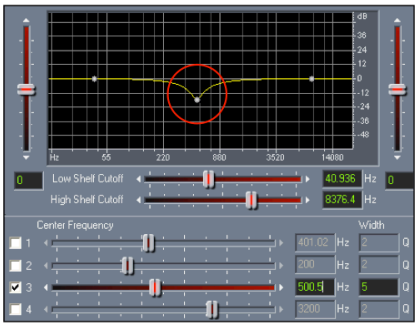




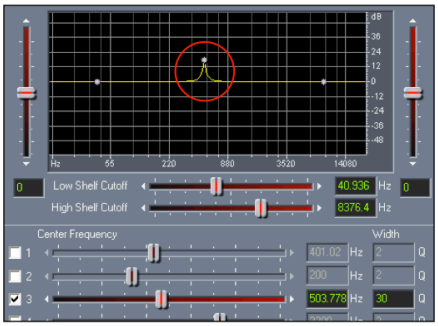
EQUALIZERS

CONTROLS

- Level Control
- Frequency Dial
- Q (Bandwidth) Control
- Low Q: larger band of frequencies is affected
- High Q: smaller band of frequencies is affected



Low Q with about 17dB loss



High Q with about 17dB boost

HARMONICS

HARMONICS

FOURIER'S THEOREM

- states that any periodic waveform can be expressed as a series of sine waves

SINE WAVE

- pure tone - energy at one frequency
- fundamental

FUNDAMENTAL

- initial vibration
- strongest pitch heard

HARMONICS DEFINED

- frequencies above the fundamental that are mathematically related to the fundamental - exact multiples (they are in tune with the fundamental)

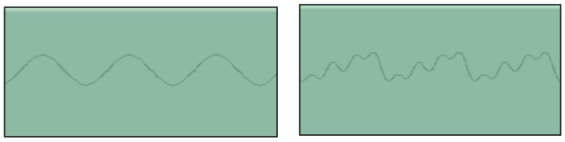
HARMONICS

Pitch	A	A	E	A	C#	E	G	A
Frequency	110	220	330	440	550	660	770	880
Harmonic	1	2	3	4	5	6	7	8

Fundamental

200 HZ FUNDAMENTAL

FUNDAMENTAL PLUS HARMONICS
200 HZ; 400 HZ; 600 HZ



HARMONICS

TIMBRE

- result of a wave's fundamental and harmonics
- describes a sound's character (how the ear distinguishes sounds)
- sound quality or tone color
- how tell the difference between intsruments

EQ TECHNIQUES

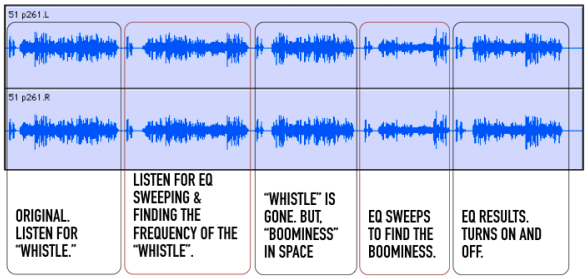
EQ TECHNIQUES

TUNING FOR NOISES

Boost the noise to make it "jump out":

- set the Q as high as possible
- sweep (through the frequencies) till you can the noise comes through
- then cut all the way - noise should be gone (if not lower the Q. If higher frequencies sound, sweep for the higher then cut)

Tuning a Parametric EQ for noises:



EQ TECHNIQUES

- make a voice stand out more (announcer/interviewer, etc.)
  - cut off at 90 Hz (not needed); using a peaking EQ set a gentle slope at 260 Hz & 1.8 kHz (3dB)
- when only music is present, make it "pop" out a bit more
  - boost bass around 100 Hz (6dB) + a high frequency shelving filter at 6dB around 3 kHz

EQUAL LOUDNESS PRINCIPLE

EQUAL LOUDNESS PRINCIPLE

- we are less sensitive to bass and treble frequencies
- hear lower or higher frequencies as a different loudness than a mid-range frequency
- 3 tones: 261.6 Hz, 1 kHz, 6.4 kHz : 1 kHz will sound louder (all played at same level)

THE DECIBEL

THE DECIBEL

- logarithmic measurement that represents how our ears hear the intensities of sound and it's perceived loudness
- 1/10 Bel
- *Bel: logarithm between the power level of 2 sounds or signals*
- measures Sound Pressure Level (SPL)
- *Sound Pressure Level = 20 log (SPL/SPLref)*
- requires a reference
- *threshold of hearing = 0dB at 1 kHz*

THE DECIBEL

DB SPL

- sound pressure level
- reference is the threshold of hearing (0 dB)

DB FS

- full scale
- voltage reference
- digital audio: refers to the maximum voltage level that is possible to record before digital overload/distortion

Typical Sounds	Typical Music	dBSPL
Chest wall vibrates, chok-ing		150
Threshold of pain		130-140
	Very loud rock/classical	110-120
Inside NY subway		100
Noisy traffic	Soft popular music	80
Normal Conversation		60
Library		30
Whisper		20
Recording Studio		10-20
Threshold of hearing		0

DYNAMIC PROCESSORS

DYNAMIC PROCESSORS

- tools that allow control of dynamic range
- **dynamic range**: the difference between the softest sound and the loudest sound of a signal
- help loud sounds from being too loud and soft sounds from getting lost in the mix or by ambient noise

TYPES:

- Compressor/Limiter
- Gate/Expander

COMPRESSOR

- reduces dynamic range of a signal that exceeds a specified volume (threshold)

GENERAL USES:

- use during recording to help reduce transients
- reduce extreme/erratic volume peaks (i.e. the constantly moving singer or electric bass)
- used to boost overall sound in a mix
- to make the louder parts softer; softer parts louder (especially in compensating for a noisy listening environment - like a car)

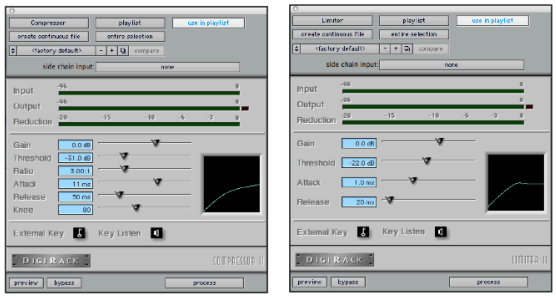
COMPRESSOR

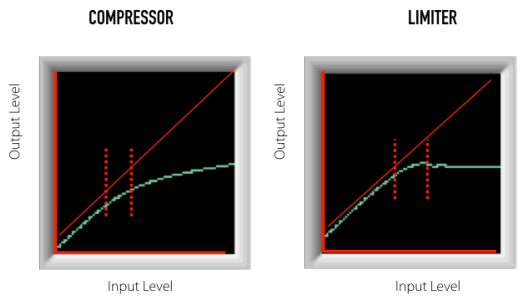
ADJUSTABLE PARAMETERS

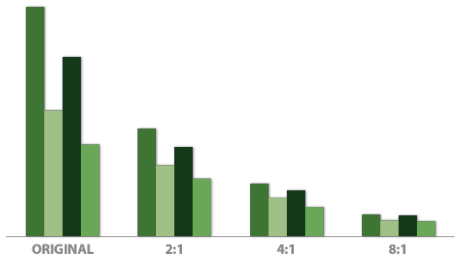
- *Threshold*: level set at which compression kicks in; signal is affected above the threshold
- *Ratio*: amount that a signal's volume is lowered:signal's output gain (2:1 = a 6 dB "peak" over the threshold will have an output increase of 3 dB)
- *Attack*: how fast/slow the compressor reacts to a peaked signal
- *Release*: how fast/slow the compressor lets go of the compression
- *Knee*: how "hard" or "soft" the compressor reaches full compression once the signal reaches the threshold - control to adjust the transition

LIMITER

- type of compressor that limits the loudest sounds (if a ratio is set high enough - 10:1 - on a compressor it will become a limiter)
- prevention of digital clipping or analog overload - prevent levels from increasing beyond a specified level
- ratio may be fixed, threshold is variable
- usually used last in the mix







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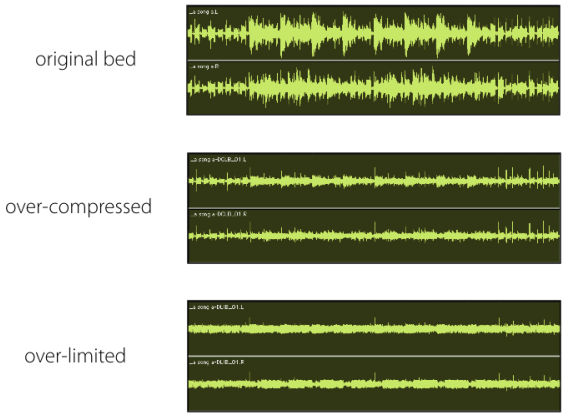

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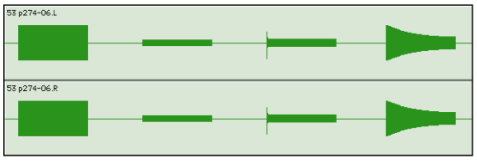
Compression: Attack times



Original .5 sec 1kHz tone

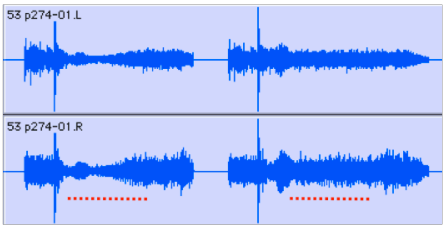
Attack @ .5 ms

Attack @ 1 ms

Attack @ 20 ms

51

Compression: Release times



Slow release time

Faster release time

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GATE

- filters out sounds **below** the threshold: signals above threshold pass while signals below are attenuated
- useful for reducing unwanted noise: **Noise Gate**

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EXPANDER

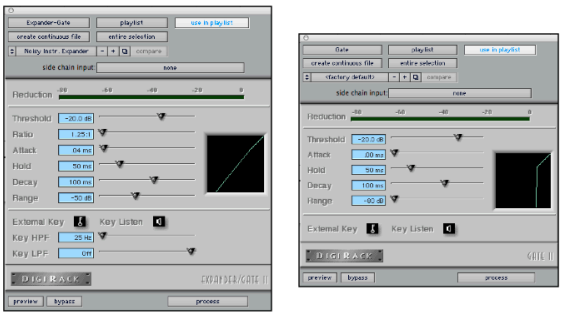
- acts like a gate, except reduces the signal by ratio rather than by volumes: dynamic range is proportionately increased
- ability to increase overall dynamic range while lowering the noise floor

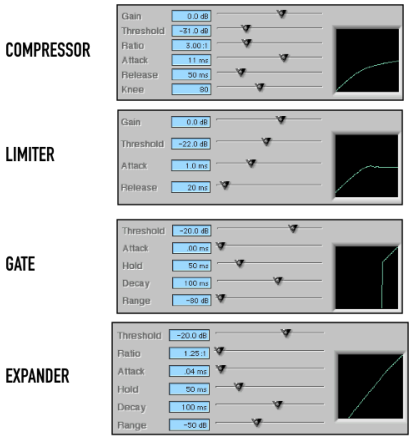
54

GATE/EXPANDER

ADJUSTABLE PARAMETERS:

- Threshold, Ratio (expander), Attack
- *Hold:* length of time gate stays open after the signal falls below the threshold
- *Decay:* rate at which the gate closes after the signal reaches the threshold
- Range*:* how much the signal is attenuated





GATE/EXPANDER EXAMPLES

EXAMPLES

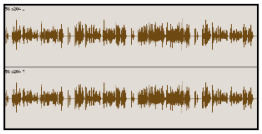
POORLY RECORDED VOICE (FROM CAM MIC)

- Expander to hide room reflections and reduce noise:
- Threshold: just below the softest words (-33dBFS)
- Attack: 3 ms
- Release: 100 ms

EXAMPLES

ENHANCING THE NARRATOR

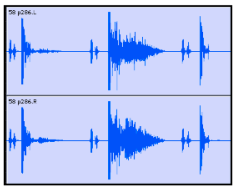
- Part 2: Compressor (Limiter) used to smooth out her heavily stressed words with Threshold: -12dBFS, Ratio: 10:1, Attack: 0.9 ms, Release: 10 ms
- Part 3: extreme compression
- Part 4: *De-esser* filter added to compressor: split signal so lower frequencies are unaffected. High frequencies set with Ratio: 8:1, Threshold -28dBFS (6dB reduction on sibilants), Attack: 1 ms, Release: 5 ms



EXAMPLES

DYNAMICS FOR SOUND EFFECTS

- Part 2: Compressed to make it's reverberation sound longer and hits it target
- Part 3: Expander to rid it of it's reverberation to make it sound more like a drum



9 STUDIO

CONSOLE

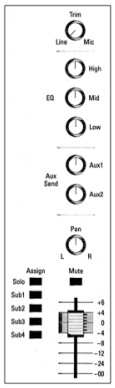
MIXING BOARD/CONSOLE

- main component to a studio and live events
- analog or digital
- virtual or real or as a controller

CONSOLE

CHANNEL STRIP

- (I/O module / input strip)
- mic and line inputs; output to other devices
- signal path runs vertically from top to bottom



CONSOLE

TRIM

- gain – boosts a mic or line signal (preamp)
- pad – attenuate a signal

EQ

- equalization
- used to compensate for any signal discrepancies

AUX SENDS

- route & mix signals to various devices, i.e. effects processor, headphone monitor mix

CONSOLE

PAN

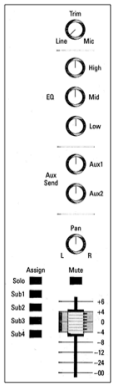
- moves a signal to left, center or right

FADER

- adjusts the overall output level of the signal

SOLO/MUTE

- solo – only the soloed channel(s) can be heard in the overall mix
- mute – channel's signal taken out of the overall mix



LOUDSPEAKERS

LOUDSPEAKER

DYNAMIC LOUDSPEAKER

- most used in pro audio
- electromagnetic induction - electromagnetic driver

BASS-REFLEX BAFFLE

- vented-box / vented loudspeaker
- tuned bass porthole usually in the front of the enclosure
- better bass response due to bass frequencies ejecting from the port in phase with the sound from the cone

LOUDSPEAKER

CROSSOVER NETWORKS

crossover = circuit found in speakers that contain a combinations of filters (high-, low-, bandpass)

at least 2 drivers (3 or more for optimum reproduction)

WOOFER

large diameter speaker

low frequencies

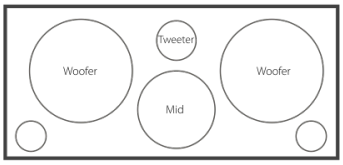
TWEETER

small diameter speaker

high frequencies

MIDS

medium sized - mid frequencies



LOUDSPEAKER

PASSIVE

- receives its power from an external power amp then sends the split frequencies signals to the appropriate driver

ACTIVE

- line level signal split into respective frequency bands - each split signal fed to it's own power amp - in turn drives the respective driver

LOUDSPEAKER

MONITOR/SPEAKER SET-UPS

FARFIELD

- large, multi-driver speakers
- ability to handle high SPLs and to supply high bass levels (due to larger diameter)

NEARFIELD

- small/medium sized speakers usually found on the desktop or above the production console
- placed closer to listener - more direct sound is heard rather than room acoustics

LOUDSPEAKER

SMALL SPEAKERS

- small multimedia speakers - computer
- generally not a good monitoring source for any kind of recording - but good for testing

HEADPHONES

- good headphones can be a great monitoring source - allow to better hear different elements within a stereo field
- used in the field - film/TV
- used for monitoring systems for the performer

10 PSYCHO-ACOUSTICS

PSYCHO-ACOUSTICS

- the study of hearing
- aim of research is to learn how hearing works

OUR EARS

THE HUMAN EAR HAS 3 MAIN PARTS:

- Outer Ear
- Middle Ear
- Inner Ear

THE OUTER EAR

PINNA

- collects and funnels sound to the auditory canal

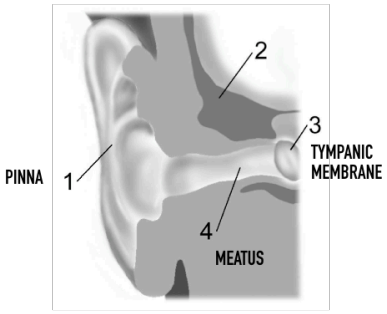
MEATUS (AUDITORY CANAL)

- passageway for sound from the Pinna to the ear drum

TYMPANIC MEMBRANE (EARDRUM)

- vibrates when impacted by sound waves, transfers vibrations to the middle ear

OUTER EAR



THE MIDDLE EAR

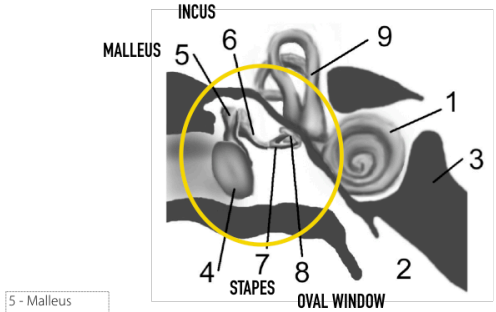
OSSICLES

- 3 tiny bones attached to the ear drum …
  - malleus
  - incus
  - stapes
- … and they transfer vibrations to the …

OVAL WINDOW

- membrane separating the middle ear & cochlea
- vibrations cause a pressure wave to travel in the fluid of the cochlea

MIDDLE EAR

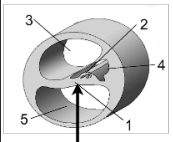


5 - Malleus
6 - Incus
7 - Stapes
8 - Oval Window

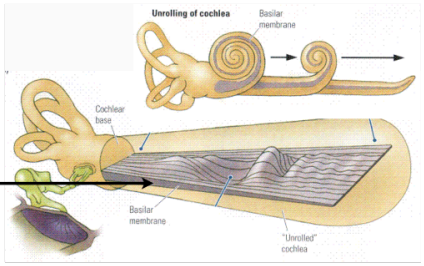
THE INNER EAR

COCHLEA

- mechanical to electrical transducer by converting the pressure waves to electrical nerve impulses to the brain
- divided into 3 parts by the:
  - Reissner's Membrane
  - Basilar Membrane



BASILAR MEMBRANE



ORGAN OF CORTI

- cells closest to the oval window will be excited by higher frequencies
- lower frequencies excite the cells further away
- brain decodes pitch by determining which hair cells are moving on the Basilar Membrane
- brain decodes the level of the sound by how many of the hairs are moving

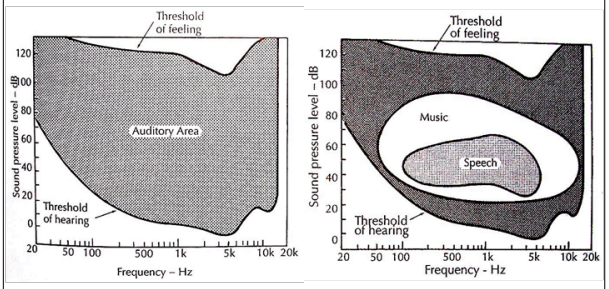
RESPONSE CHARACTERISTICS

RESPONSE CHARACTERISTICS

PHYSICAL

- the construction of the ear limits our frequency range (20Hz - 20kHz) & dynamic range (140dB SPL)
- how the brain processes information
- nonlinearity of the ear - masking, combination tones

FLETCHER MUNSON CURVES



EQUAL LOUDNESS PRINCIPAL

LOUDNESS AND LEVEL

LOUDNESS AND LEVEL

THRESHOLD OF HEARING

- 1 kHz at 0dB

EQUAL LOUDNESS PRINCIPLE

- states that we are less sensitive to bass and treble frequencies and hear lower or higher frequencies as a different loudness than a mid-range frequency

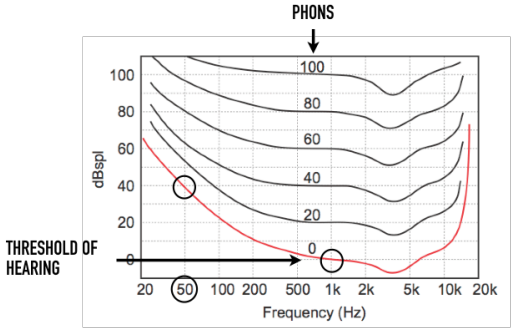
LOUDNESS AND LEVEL

LEVEL

- SPL – physical value measurable by the dB

LOUDNESS

- subjective – perception value
- phons – loudness levels that correspond to sound pressure levels at 1kHz



LOUDNESS AND LEVEL

EQUAL LOUDNESS CURVES

- our ear's frequency response changes with respect to loudness
- our ears have a flat response to louder sounds (reason for loudness controls) – why like to listen at louder levels
- a recording mixed at an excessively high level will sound very light in bass when played back at a normal level
- we use the loudness of sound to determine information about the source, i.e.. distance

NONLINEARITIES OF THE EAR

NONLINEARITIES OF THE EAR

MASKING

- when a softer signal is not heard because of a louder signal (decreased audibility of one sound in the presence of another)
- frequency discrimination caused by the Basiliar Membrane – unable to register energy in a band of frequencies when another band has more energy

NONLINEARITIES OF THE EAR

COMBINATION TONES

- tones our ear create
- ear will hear 2 tones that have a difference of more than 50 Hz as complex set tones:
- equal to the sum & difference of original tones + original tones

1 KHZ & 1.5 KHZ
SUM = 2.5 KHZ
DIFFERENCE = 500 HZ

NONLINEARITIES OF THE EAR

BEATS

- "definite alternating swells and lulls of sound" [Benade]
- result of the ear's inability to separate notes that are close in pitch
- 2 tones very close in frequency played simultaneously, the tone lower in frequency will fall about a 1/2 cycle behind (cancellation), then it continues to fall to a 1 cycle behind (summing)

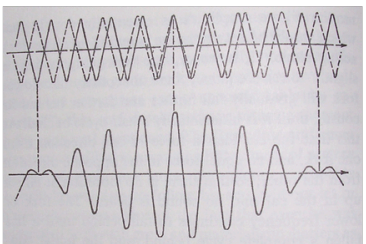


Fig. 18. The beat phenomenon produces these curves. In the upper one, two simple harmonic forces of slightly different frequencies get in and out of step. In the lower graph the resultant of the two forces appears to grow and diminish at the beat frequency.

LOCALIZATION

LOCALIZATION

HOW DO WE KNOW WHERE A SOUND IS COMING FROM?

- 2 ears = Binaural Localization

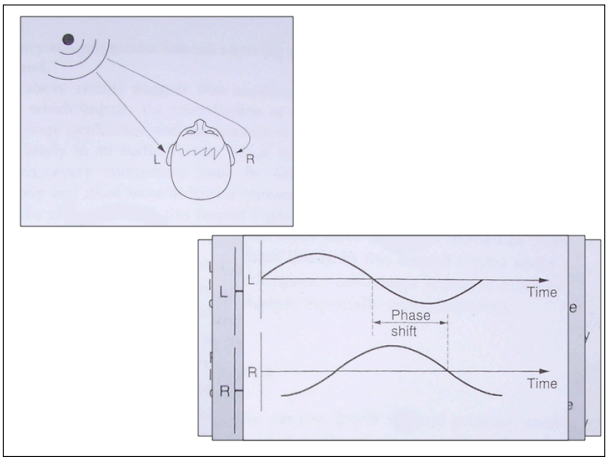
INTER-AURAL INTENSITY DIFFERENCE (IID)

- off-centered sound will reach the closer ear with a higher intensity than the distant ear (intensity difference)
- the distant ear receives mostly reflected sounds (due to the head) that have lost energy, therefore the perceived sound is reduced
- brain takes this information and decides that the sound arrived from the side of the closest ear

LOCALIZATION

INTER-AURAL TIME-ARRIVAL DIFFERENCE

- brain calculates the time delay of sound reaching the left and right ears and determines which sound arrived first
- bumps and ridges of the pinnae reflect the direct sound into the ear causing slight time delays between the reflected and direct sounds



LOCALIZATION

TRANSIENTS

- contains necessary information for localization, size and pitch
- initial transient gives location (clap)
- as the rapid decrease of pressure equalizes, determination of the size of sound and frequency analyzation begins (pitch & timbre)
- if a sound reproduction system impairs the transient, then damage to the ability of the localization and frequency analysis of the sound will occur

LOCALIZATION

PRECEDENCE EFFECT

- Haas Effect, the law of the first wavefront
- refers to how a direct wave and it's reflections give us localization information – the first wave to arrive in our ears we'll interpret as the direction of the sound source
- delay of time plays an important role:
- short delays (0-1 ms) 2 sounds (direct & reflected) will combine & the average will give localization info
- longer delays (> 1 ms) reflections become more audible and are heard as a separate sound
- if the reflection is about 10 dB louder than the direct sound, then we hear the sound source as if it comes from the direction of the reflection