

VL Audio Produktion SS07

**Zusammenschrift aus den
VO-Folien**

introduction the studio

Studio Types

Professional Recording Studio

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

Audio-for-Visual Production Studio

production facility for video, film & game post-production

scoring, score mix-down, Foley

Project Studio

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

Portable Studio

gaining popularity
laptop, USB, Firewire, all-inclusive portable recording system (M-Box – ProTools)

basics of sound

What is sound?

- disturbances in the air caused by vibrations
- vibrations produce sound waves

*Sound is part:
Physical
Perceptual*

Physical

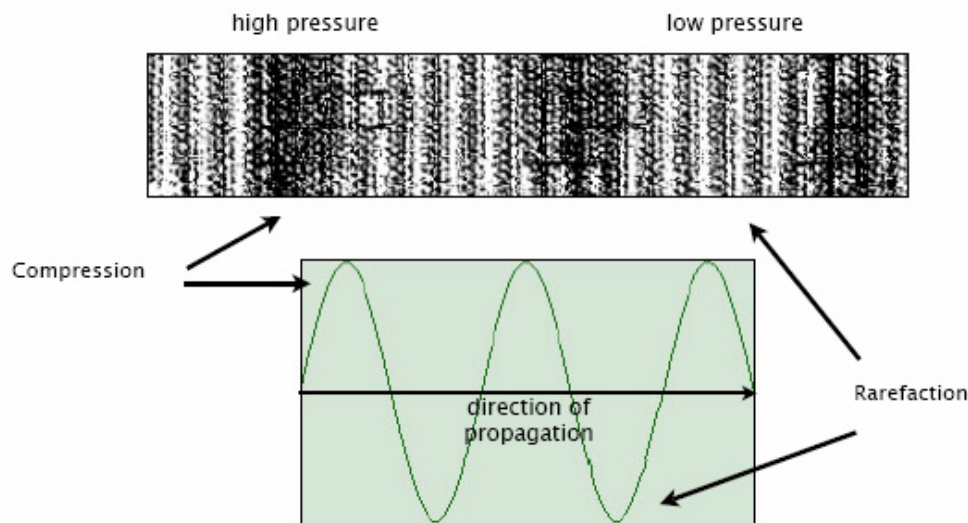
- energy that travels through a medium
- propagates itself by traveling in waves

Perceptual

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

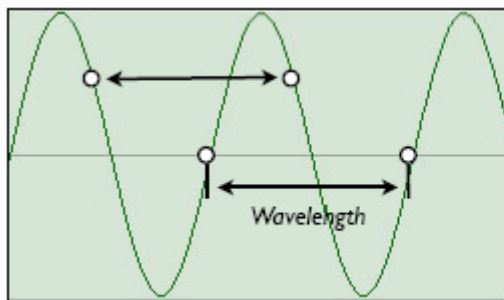
Physical	Perceptual
Frequency	Pitch
Amplitude	Loudness
Harmonics	Timbre
Envelope	Articulation

Wave Properties



Wavelength

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



$$\lambda = Tc$$

λ = wavelength
 T = period (s)
 c = propagation speed (ft or m)

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

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)

The Waveform

Periodic

signal that has a constant (mostly similar) patterns, and repeats indefinitely

Aperiodic

signal that is not constant, and does not repeat

Envelope

visual pattern of amplitude over time

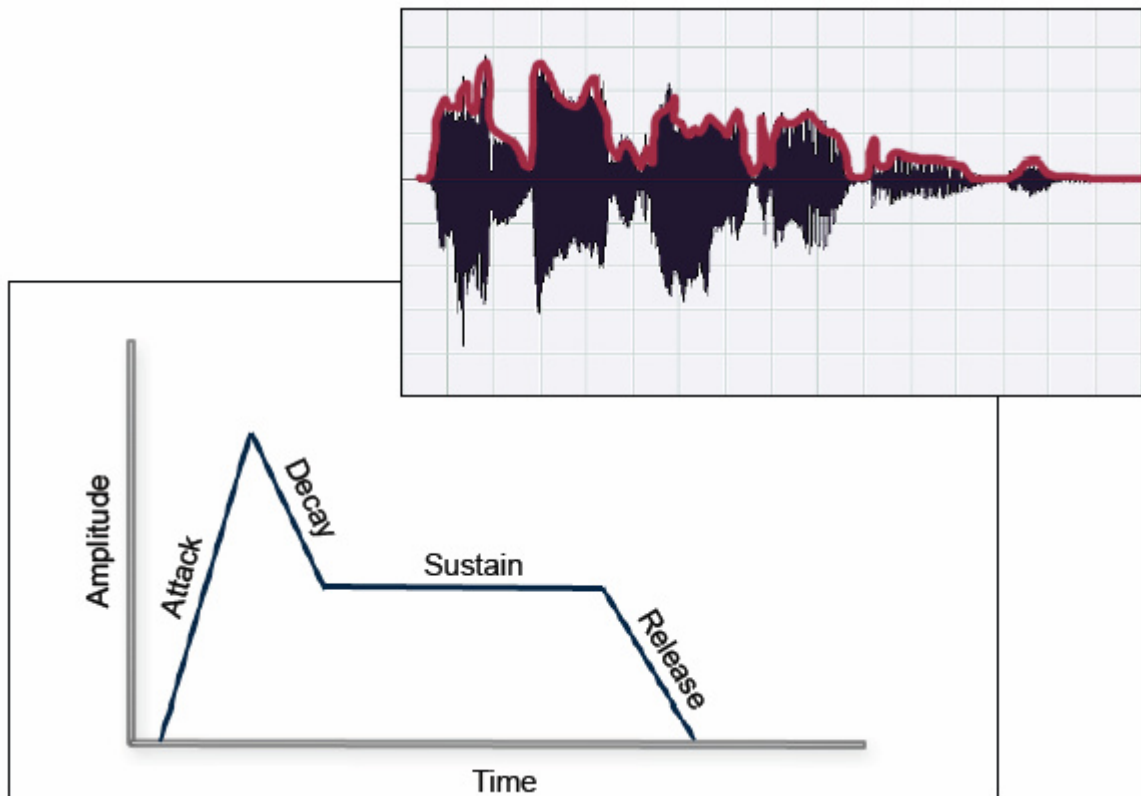
ADSR

Attack: The initial strike – time it takes a signal to rise from zero to its maximum amplitude

Decay: The time it takes for the signal to fall to the sustain value

Sustain: The time the signal remains at a certain level

Release: The time it takes for the level to fall to zero after the elapsed sustain time



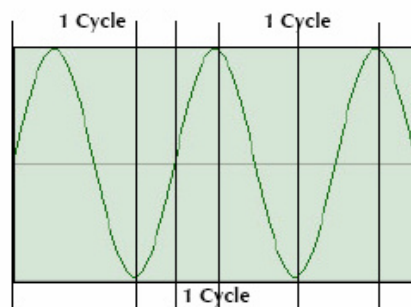
Frequency

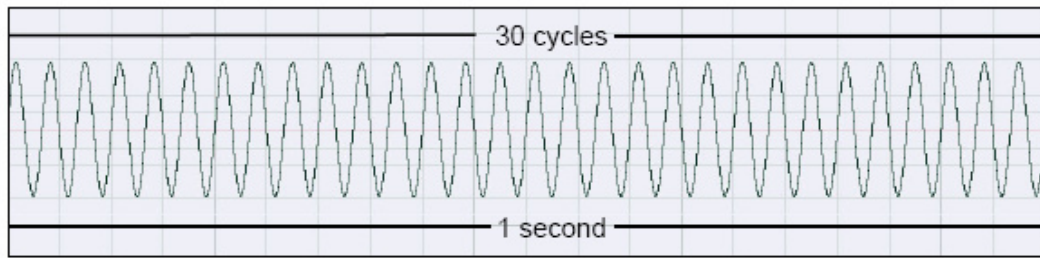
the rate at which a sound wave completes one cycle in one second

Hertz (Hz) or kiloHertz (kHz)

1 cycle/sec = 1 Hz

1,000 cycles/sec = 1 kHz





Audio Spectrum

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 usually (but not limited to) between 20 Hz & 20 kHz
i.e. human speech can be found as low about 110 Hz to as high as 10 kHz

Pitch (Tonhöhe)

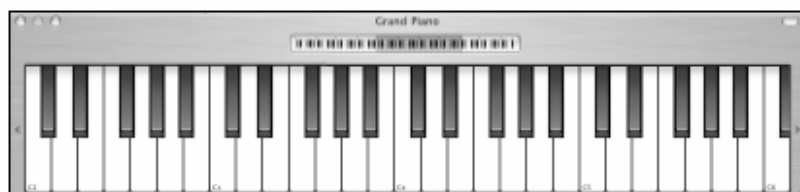
The higher the frequency, the higher the pitch

Middle C on the piano keyboard vibrates at a frequency of 261.6 Hz. We perceive this sound as the pitch.

Heard in octaves – 1 octave is the doubling of a frequency:

Middle C	C above Middle C
261.6 Hz	523.2 Hz

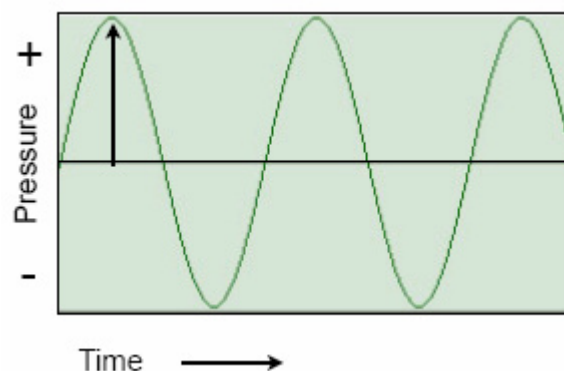
Concert A	Octave above Concert A
440.0 Hz	880.0 Hz



Amplitude

maximum displacement from the equilibrium

Loudness – volume

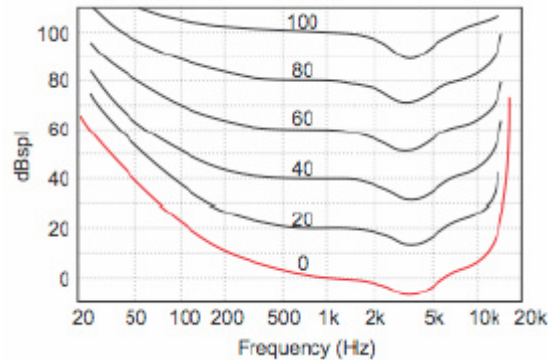


Equal Loudness Principle

we are less sensitive to bass and treble frequencies and 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)

Why is this interesting?



The Decibel (dB)

logarithmic measurement that represents how our ears hear the intensities of sound and its perceived loudness

1/10 Bel

Bel: logarithm between the power level of 2 sound or signals

measures Sound Pressure Level (SPL)

Sound Pressure Level = $20 \log \text{SPL}/\text{SPL}_{\text{ref}}$

requires a reference

threshold of hearing = 0 dB at 1 kHz

dB SPL

sound pressure level
reference is the threshold of hearing

dB FS

Full Scale
voltage reference
digital audio: refers to the maximum voltage level that is possible before digital overload

Typical Sounds	Typical Music	dB SPL
Chest wall vibrates, choking		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
Threshold of hearing		0

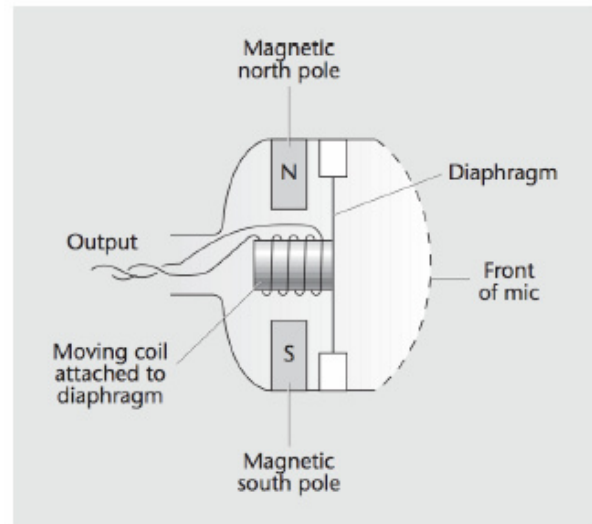
Sound as an Analog Signal

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)
- analogous = similar

Transducer

- converts sound from 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
- Two main types of transducers:
Microphone
Speaker



Electrodynamic Driver

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

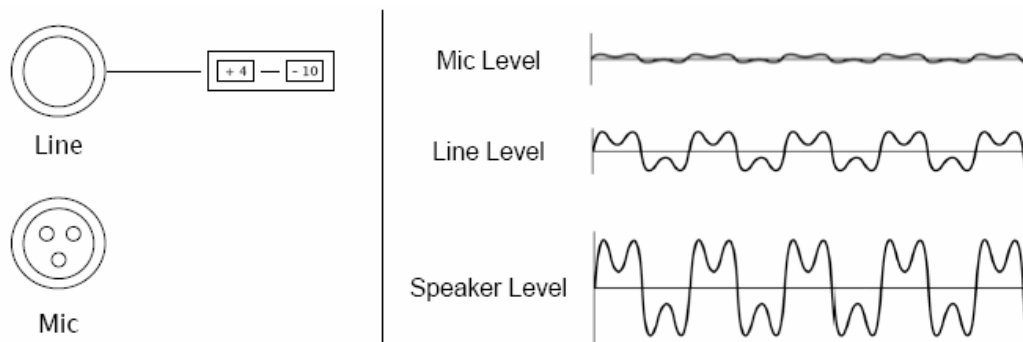
Levels

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)

Mic Level

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



Wiring

Unbalanced Wiring

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

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

Phone or 1/4" or 6.5 mm

connects audio equipment, instruments, headphones

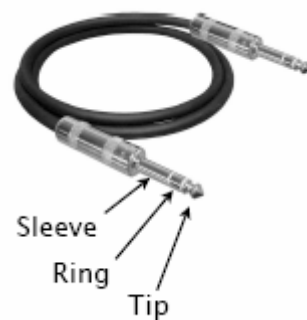
Mono

- tip/ring or TR
- unbalanced
- connects instruments



Stereo

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



Mini or 1/8" or 3.5 mm

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



RCA

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



Microphones

*3 Types:
Dynamic
Ribbon
Condenser*

Dynamic Microphone

- 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 an physical abuse
- live & studio environments
 - live: subjected to abuse, wheather, screaming, being dropped
 - studio: used for close-miking drums, vocals

Ribbon Microphone

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

Condenser Microphone

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 2 plates, is related to change in voltage
-
- 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)

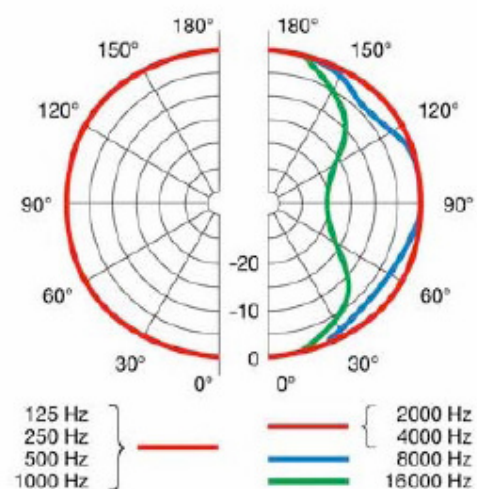


AEA R84

Microphone Characteristics

Polar Patterns

Microphone's response to sound sources from various angles/directions



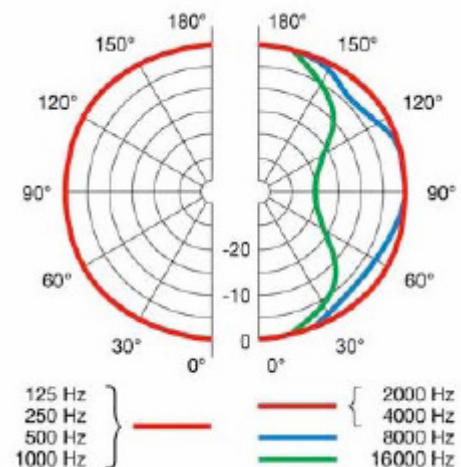
Omnidirectional

responds to sound pressure from all angles

less sensitive to wind noise

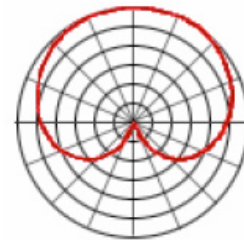
Uses:

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



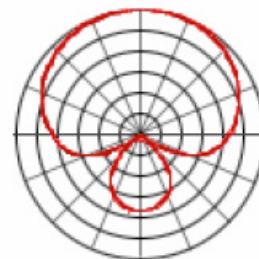
Unidirectional

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



Hypercardioid

- variation of the cardioid
- more directional
- boom mics

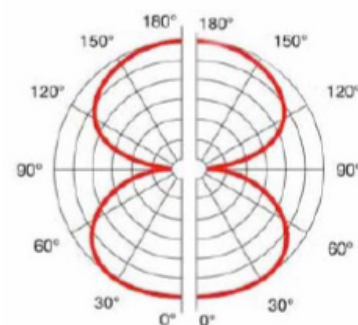


Supercardioid

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

Bidirectional

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



Recording Basics

Techniques

Distant Miking

- 1 or more mics placed approximately 1 m from source (depending on size)
- allows for the room's acoustics, creating a live, open sound
- phase cancellations (excessive reflections)

Close Miking

- miking close to the source – pick up more direct sound
- reduces the possibilities of phase cancellations (no excessive reflections)
- best to use a directional mic
- when miking 2 or more sources close together avoid leakage by keeping mics at a 3:1 distance

Accent Miking

- used to pick up a certain area (ensemble section, solo, etc.)
- mic the target source so it is in balance with the rest of the main source (ensemble)

Ambient Miking

- miking the ambience of an environment, natural

Stereo Miking

Spaced Pair
Coincidence Miking
Generally used to record ensembles or the like

Stereophonic

- illusion of spaciousness or spacial perspective
- stereo image: a well produced spacing of sound sources = source sound natural in a distinct location

Spaced Pair

- 2 microphones placed in front of a sound source – spaced at a certain distance
- mics are of same type, model & manufacturer (omni or cardioid)

Disadvantages:

- depending on size of source – mics are placed at large distance from the source
- if broadcast in mono, phase problems can occur due to sound arriving at different times to each mic

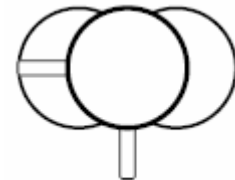
Coincident Miking (Intensity Recording)

- all directional cues are based on differences of loudness
- mics are placed in same spot
- one mic faces left – the other, right
- angle is critical
- mics can be closer to the source
- directional mics

Flavors:
M/S
X/Y

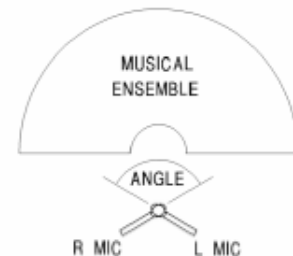
M/S

- Mid-Side Technique
- 2 separate mics or single-unit: Cardioid or Omni and a Bi-directional
- Mid faces middle to directly pick up sound source
- Bidirectional = faces sideways & pick up L side with one phase and the R side with inverted phase
- When signals added/subtracted, get a stereo image as if 2 Cardioids at a 45° angle
- better choice for mixing to mono



X/Y

- 2 mics – same directional pattern
- 2 separate mics or single-unit
- X = Left; Y = Right
- usually 2 cardioids positioned with their fronts very close together (or above one another) at 90° (to 135°)



Field Miking (Outside the studio)

Live concert

Interview for radio, TV, documentary

Theatrical scenes

Handheld Microphone

- interview: radio/TV/documentary style
held under the chin; not in front of the mouth
- Dynamic directional mic (or Omni)
- singer/musician

Lavalier

- clip on clothing/tie (less than 25 cm)
- used for dialog on camera – easy to hide
- will pick up more voice than ambience

Boom

- hand held or floor stand (fishpole)
- best choice for capturing sound on camera
- allows talent (actors) to move freely
- uses a shotgun mic – hyper or super cardioid

Amplifiers

- electronic device used to boost (amplify) an electrical signal (voltage, current, impedance, power)
- many applications: amplify, equalize, match impedances, isolate and distribute signals

Preamplifier (preamp)

- input section of mixers, consoles, etc.
- boosts a mic's input to line level

Power Amplifier

- boost's a line level signal to speaker level
(level with adequate voltage or current to drive a speaker)

Loudspeakers

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

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

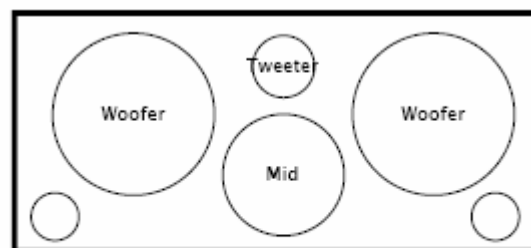
- Large diameter speaker
- Low Frequencies

Tweeter

- Small diameter speaker
- High Frequencies

Mids

- Medium sized – mid frequencies



Ported 3-way system

Crossover frequencies at 600 Hz & 4 kHz

Passive

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

Active

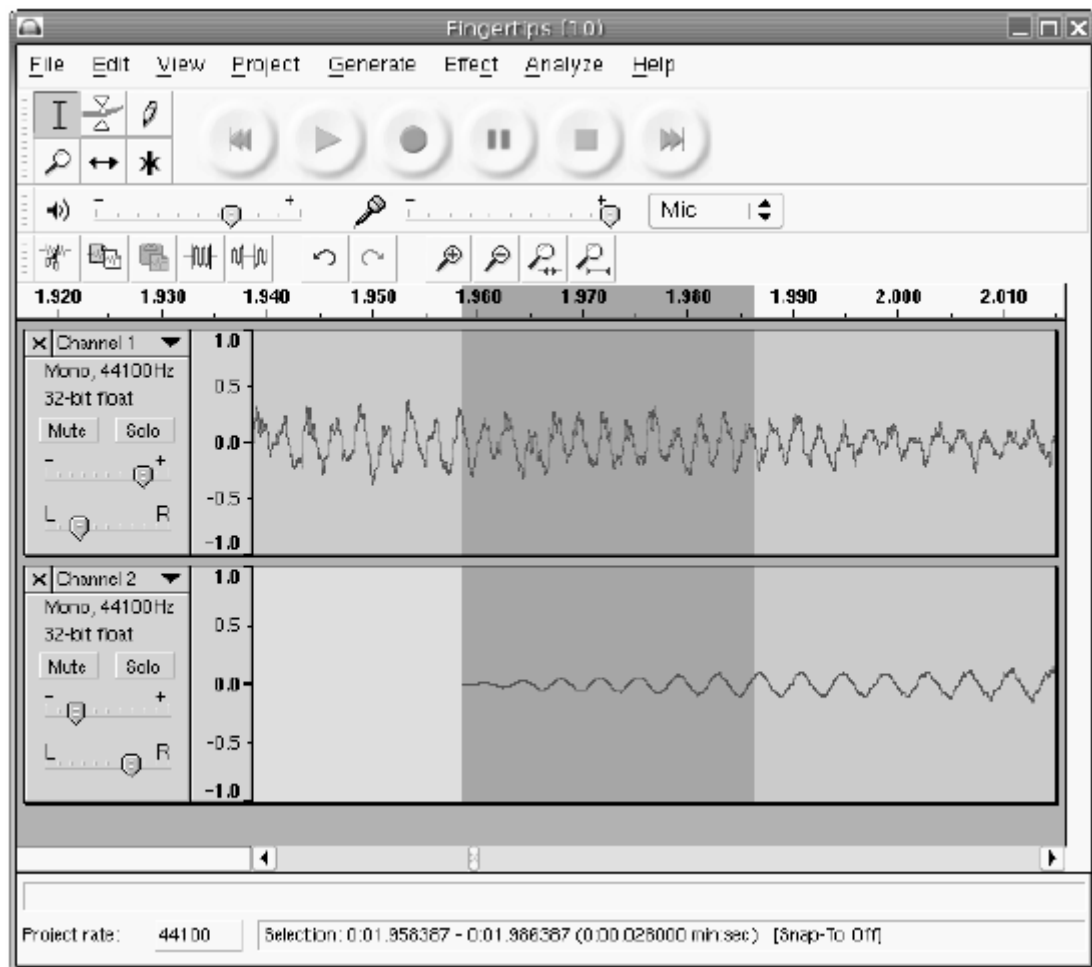
line level signal split into respective frequency bands – each split signal fed to its own power amp – in turn drives the respective driver

Production: The DAW Editing Basics

Digital 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 audio to digital (from mic to computer)

ProTools LE 003 Rack



Controller Interface (Control Surface)

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

Prototools LE 003 Factory, M-Audio
ProjectMix I/O



Virtual Mixer

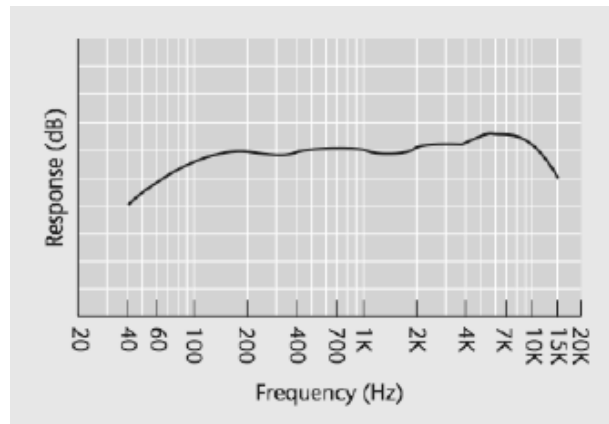
- Real-time mixing
- same abilities as real mixer – kind-of



Microphone Characteristics

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

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

Transient Response

- measure of how quickly the mic's diaphragm will react when hit by the acoustical wave
- usually dependent on the diaphragm's size:

Dynamic

large diaphragm = slow reaction, rugged, less accurate sound

Ribbon

smaller, lighter diaphragm = quicker, more clear & accurate response to the wave

Condenser

very thin, light diaphragm = less resistance, more accuracy

Sound as a Digital Signal

The Nyquist Theorem

the highest frequency that can be accurately reproduced has to be less than half of the sample rate

$$44.1 \text{ kHz} / 2 = 22,050 \text{ Hz}$$

Aliasing

- "the misidentification of a signal frequency, introducing distortion or error" [Oxford English Dictionary]
- distortion that occurs when frequencies higher than the Nyquist frequency are recorded
- anti-aliasing filter: found before the ADC to prevent frequencies above the Nyquist frequency from being recorded

Quantization

- process of choosing whole numbers (discrete) to represent a voltage level (quantization level)
- the more levels (bits), the better:
CD = 16 bits (65,536 quantization levels)
- when the voltage of the analog signal falls between the whole numbers, a quantization error occurs
distortion related to the signal (quantization distortion)

Quantization Error

Why care about it?

- lower the signal – the more audible the error – signal uses a smaller portion of dynamic range

Dither

- added noise to a signal before quantization
- used to counteract the effect of quantization error
- unnecessary when using higher bit depths
- necessary when reducing bit depth

Delivery Formats / Data Compression

reducing the data rate and file size of a digital signal
44.1 kHz stereo, 16 bit = 10 MB/60 sec

Lossless

no data is lost – exact digital clone

Lossy

creates smaller files by discarding bits

Lossless Formats

Wave (.wav)

developed for Microsoft Windows
mono & stereo
8-bit or 16-bit
up to 48 kHz s/r

AIFF (.aif)

Audio Interchange File Format
mono & stereo
8-bit or 16-bit
up to 48 kHz s/r

**FLAC (Free Lossless Audio
Codec) open source**

Lossy Formats/Perceptual Coding

MP3 ISO MPED Layer 3 (MPEG-1 Layer III)

MP3 Pro

MP3 Surround

AAC Advance Audio Coding

- MPED-4 Audio (.m4a)
- up to 5.1 surround sound encoding
- SDMI compliant – allows copyrighted material to be protected from unauthorized copying and/or distribution

WMA Windows Media Audio

- real-time stream
- surround-sound encoding

Real Audio (.ra or .ram)

- real-time streaming
- combined with video – Real Media streaming

AC-3

- Dolby Digital
- HDTV (US)

Connectors

AES/EBU or AES3

a digital audio transmission standard

American Engineering Society & the European Broadcast Union

transmits 2 channels of digital audio data on a balanced line using an XLR connector

s/pdif

Sony/Philips Digital Interface/Format

unbalanced coaxial cable – RCA plugs

uses RCA connectors or optical cable

Toslink

consumer fiber optic connector

S/PDIF and ADAT Lightpipe (Alesis)

USB

Universal Serial Bus

medium bandwidth serial digital data

interconnection standard

MIDI

Firewire/IEEE 1394

high speed data format interface bus – supports

multiple data formats

video

MIDI

- Musical Instrument Digital Interface
- protocol for electrical instruments to communicate with each other – to send/receive data information
"... musical description language in binary form ..."
- combination of hardware and software

MIDI Data

Standard MIDI file formats

MIDI Channels

16 Channels

MIDI Messages

note on, note off, velocity, after-touch, vibrato, pitch bend

MIDI Modes

how the instrument/device responds to the MIDI messages

MIDI System

A device that generates sound
drum machine, synthesizer, etc.

MIDI controller

anything that can control another MIDI device

Sequencer/DAW

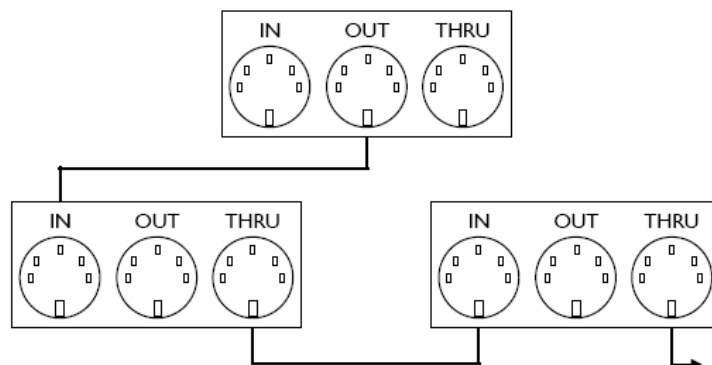
records and plays MIDI

MIDI interface

connects MIDI equipment – allows the computer to send & receive MIDI data

MIDI Connection

uses a 5-pin DIN connector



Consoles

Mixing Board

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

Channel Strip

(I/O module / input strip)

- mic and line inputs; output to other devices
- signal path runs vertically from top to bottom

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

Pan

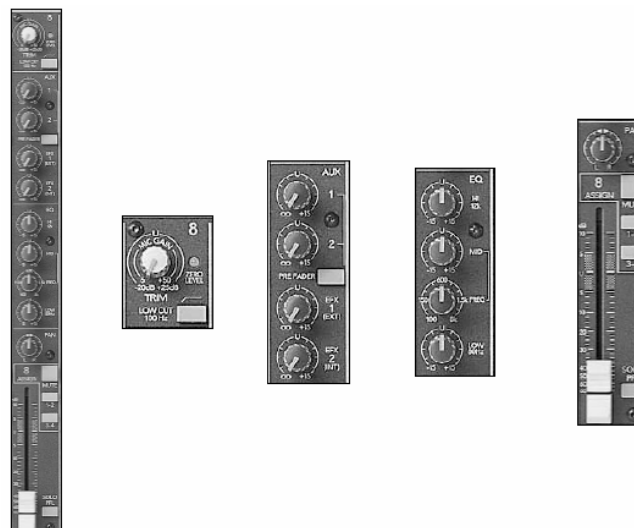
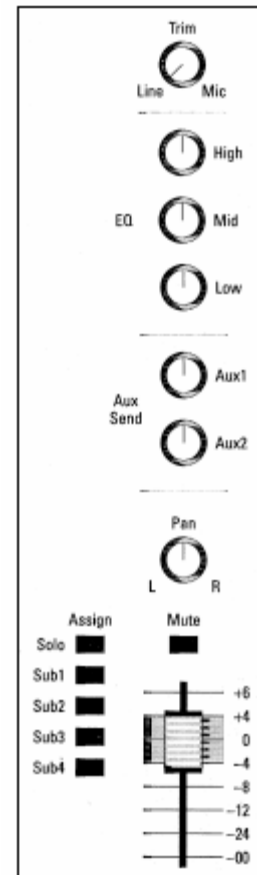
- moves a signal to left, center or right

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



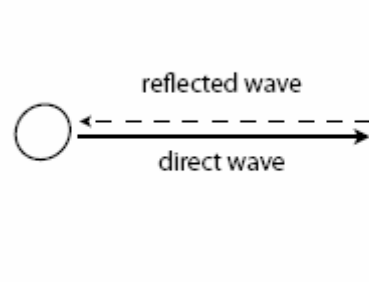
Sound Movement

Reflections

- persistence of a sound after it's source has stopped
- sound bouncing off of hard surfaces
- 2 phenomenon:
 - Echo
 - Reverberation

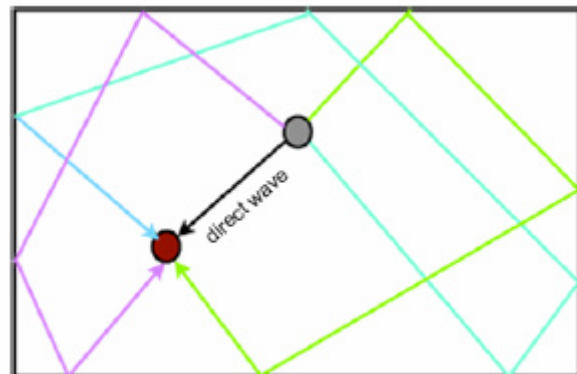
Echo

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



Reverberation

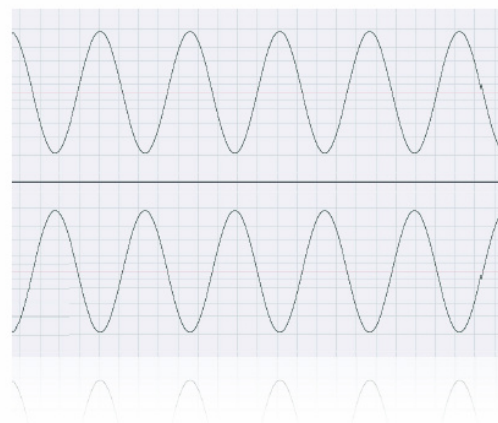
- remainder of sound after the source has stopped
- collection of many reflections
- adds spaciousness to a room
- full and partial cancellations of frequencies



Phase

- a particular point in the time of a cycle
- sound waves: reverts 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

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 ($\sqrt{\quad}$) distance
- double the distance from a sound's source, its sound pressure becomes 6 dB less – if in a room/space without echoes (*free field*)

Harmonics

- frequencies above the fundamental that are mathematically related to the fundamental

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

Fourier's Theorem

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

Sine wave

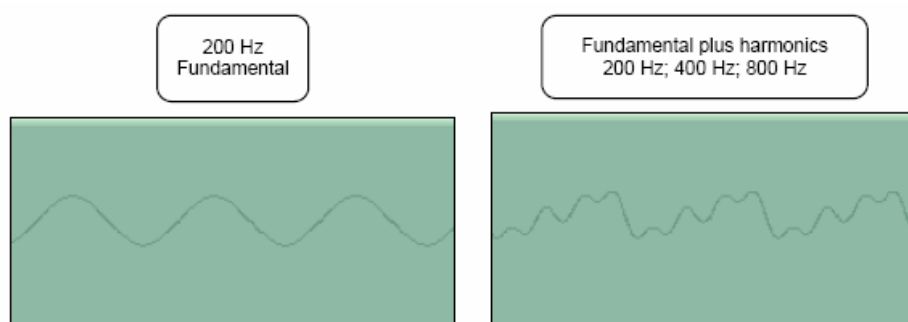
- pure tone
- fundamental

Fundamental

- initial vibration
- strongest pitch heard

Timbre

- result of a wave's fundamental and harmonics
- describes a sound's character (how the ear distinguishes sounds)
- sound quality



Digital Signal Processors

Signal Processors

Equalizers

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

Dynamic Processors

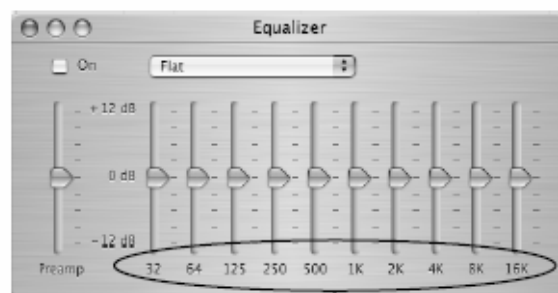
- regulate a sound's dynamic range
- compressors, limiters, expanders, gates

Equalizers

Types of EQ:

Graphic Equalizer:

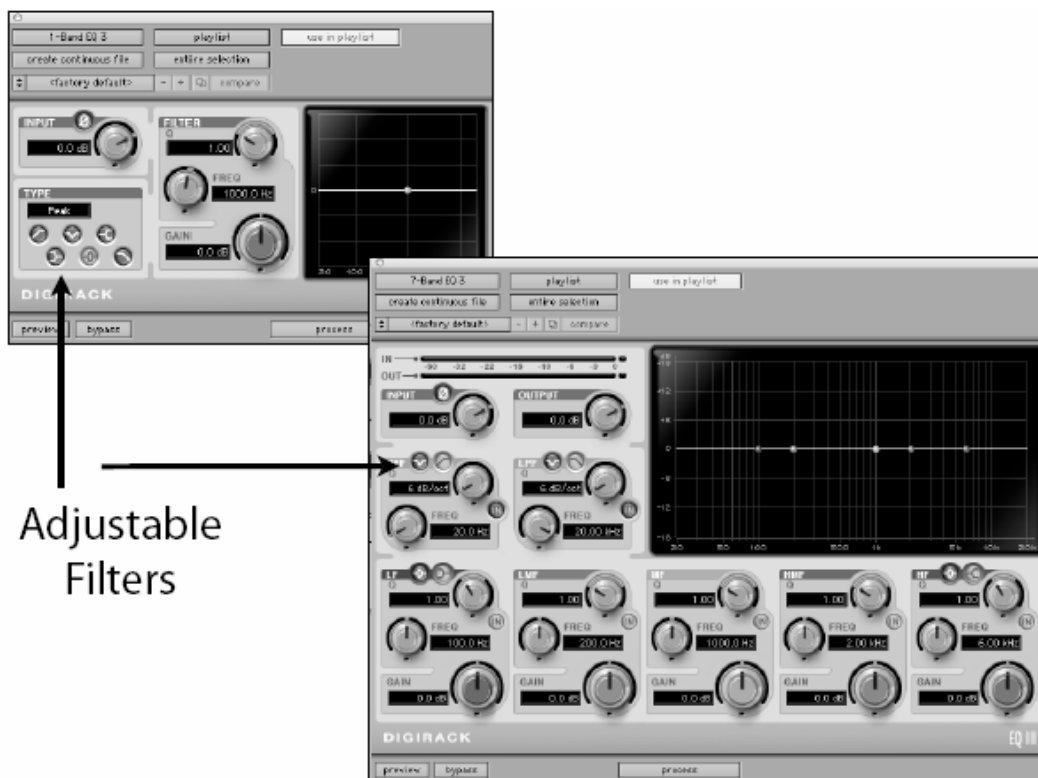
- easiest to use
- used to shape the overall spectrum of a program
- boost & cut over a series of center frequencies



Center Frequencies

Parametric

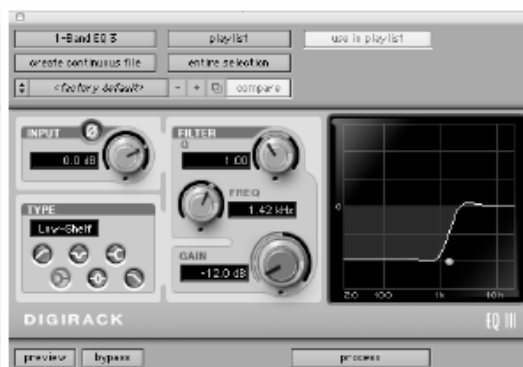
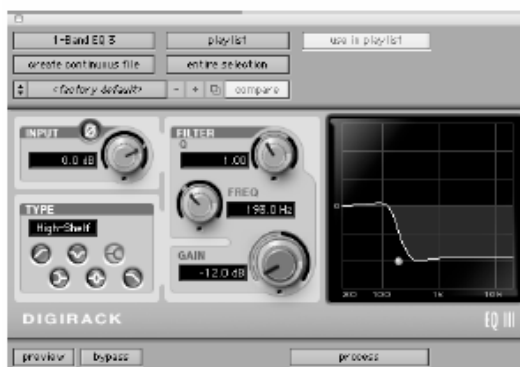
- ability to control several parameters:
 - the amount of boost (x) or cut (-) in dB
 - dial in to a chosen center frequency
 - adjust the bandwidth/range (Q) (Quality factor)
- adjustable filters



Adjustable
Filters

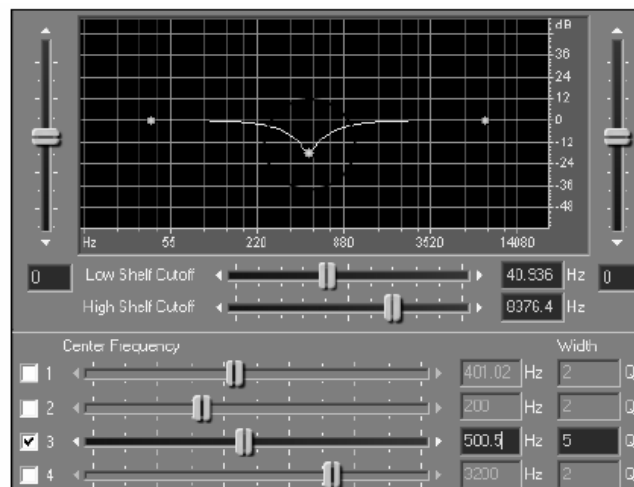
Filters:

- found within 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: cut or boost frequencies around a selected frequency

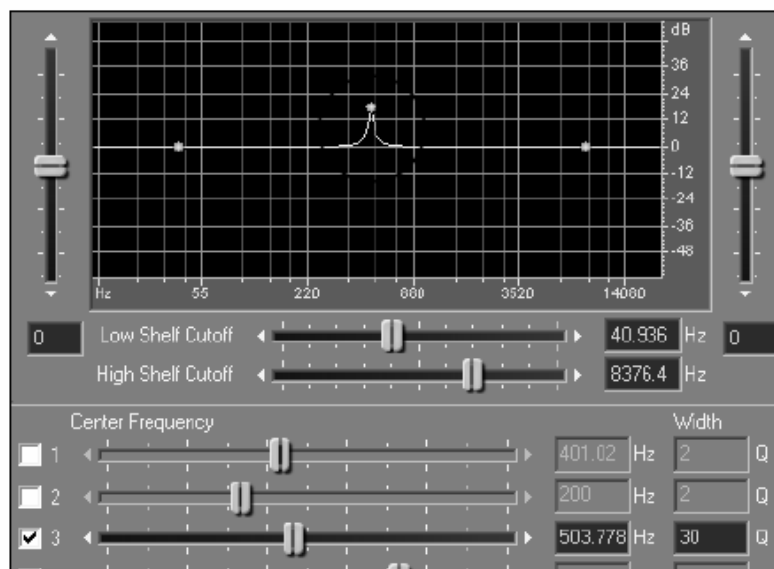


EQ Controls

- 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



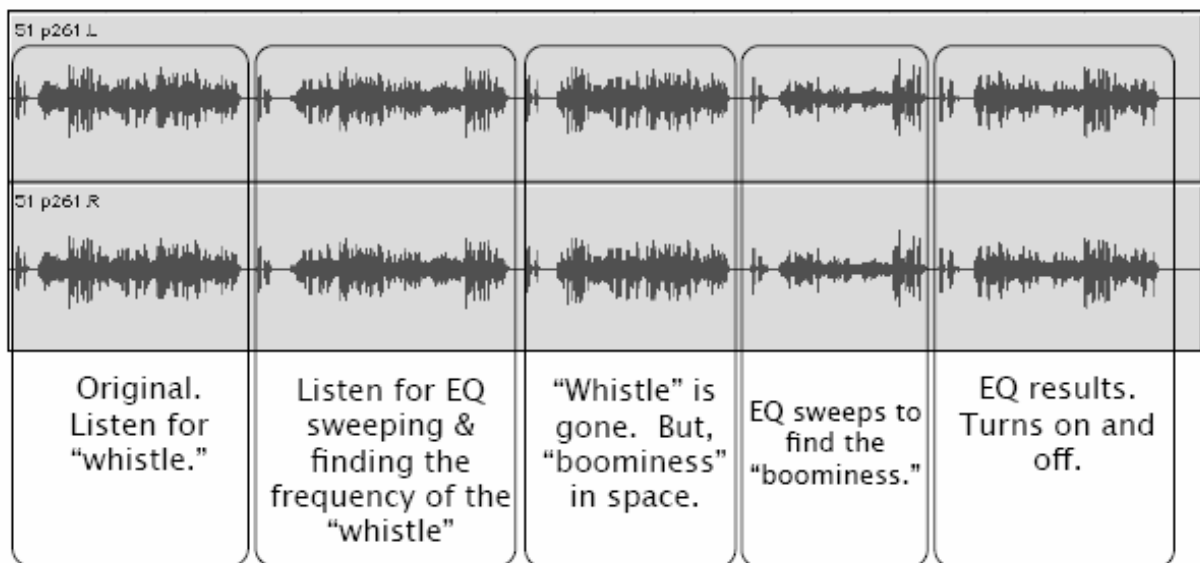
High Q with about 17dB boost

Tuning a Parametric EQ for noises

Boost the noise to make it “jump out:”

- Set the Q and boost 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:



Some EQ tips (from Jay Rose)

- Make a voice stand out more (announcer/interviewer, etc.)
Cut off at 90 Hz (not needed); using a peaking EQ set a gentle slope (Q=7, 3dB) at 260 Hz & 1.8 kHz
- When only music is present, make it “pop” out a bit more
Boost bass around 100 Hz (Q=3, 6dB) + a high frequency shelving filter at 6dB around 3 kHz

Dynamic Processors

Processing:

Dynamic Processors

- tools that allow control of dynamic range
- 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 specified volume (threshold)

General Uses:

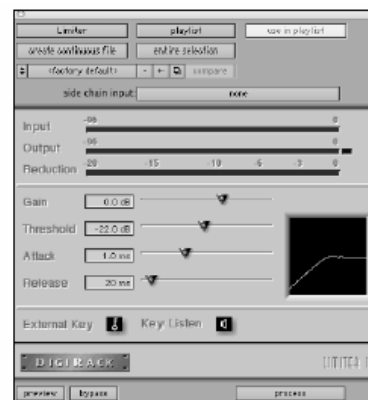
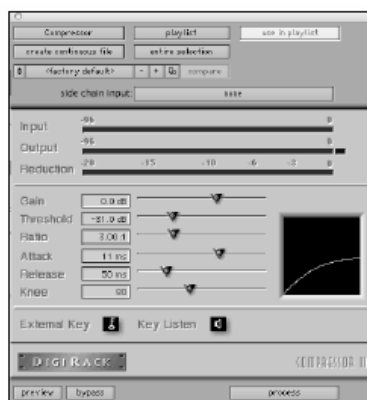
- use during recording to help reduce transients
- reduce extreme/erratic volume peaks (i.e. the constantly moving singer or electric bass)
- used to boost overall sound in a mix
- to make the louder parts softer; softer parts louder (especially in compensating for a noisy listening environment – like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

Adjustable parameters

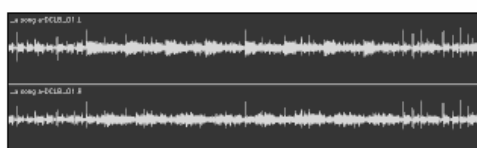
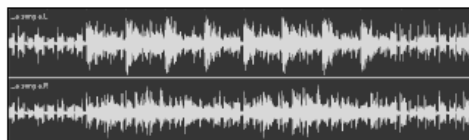
- Threshold:** level set at which compression kicks in
- Ratio:** amount that a signal's volume is lowered: signal's output gain (2:1 = a 2 dB "peak" over the threshold will have an output will increase one dB)
- Attack:** how fast/slow the compressor reacts to a peaked signal
- Release:** how fast/slow 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 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
- same parameters as compressor
- usually used last in the mix



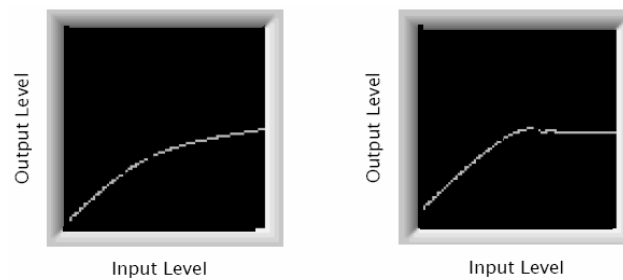
original bed



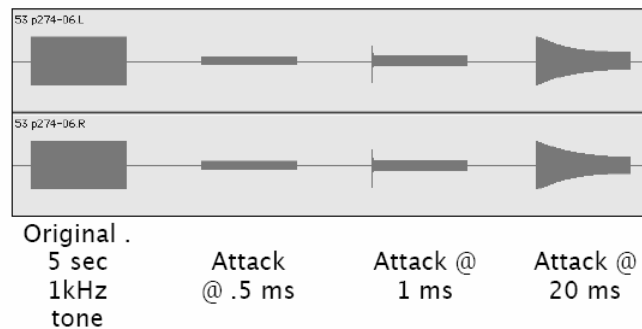
over-compressed



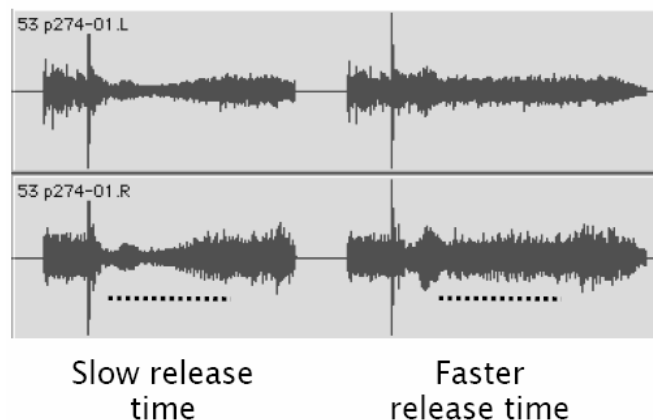
over-limited



Compression: Attack times



Compression: Release times



Gate

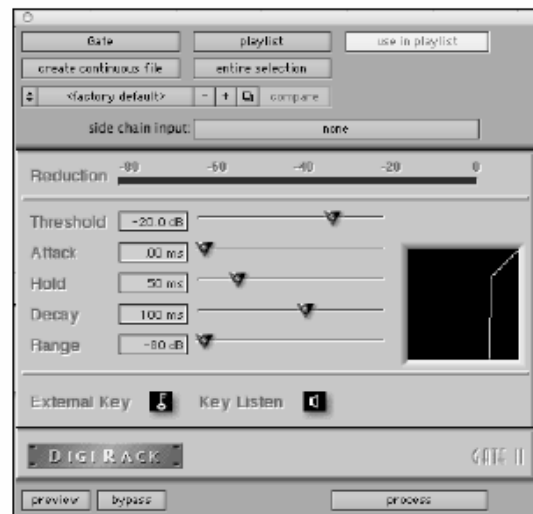
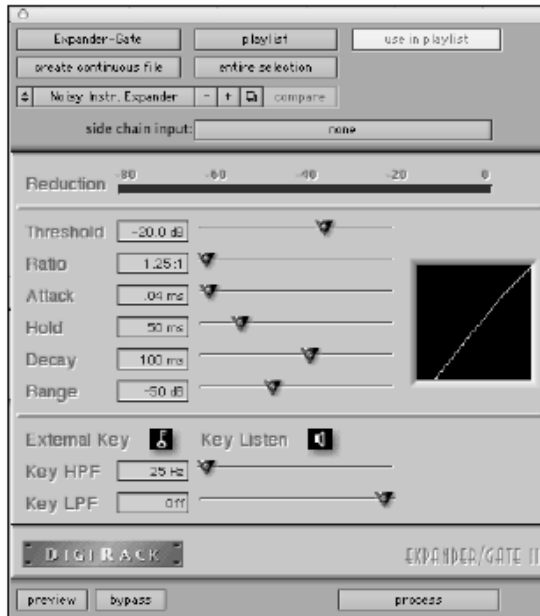
- filters out sounds below the threshold: signals above threshold pass while signals below are attenuated
- useful for reducing unwanted noise: Noise Gate

Expander

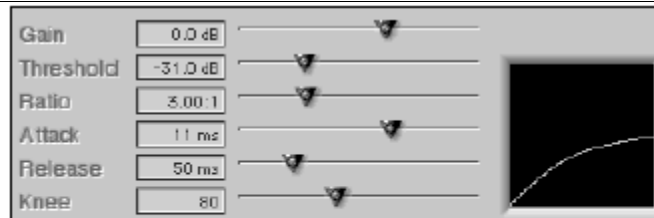
- acts like a gate, except recedes the signal by ratio rather than by volume: dynamic range is proportionally increased
- ability to increase overall dynamic range while lowering the noise floor

Adjustable parameters:

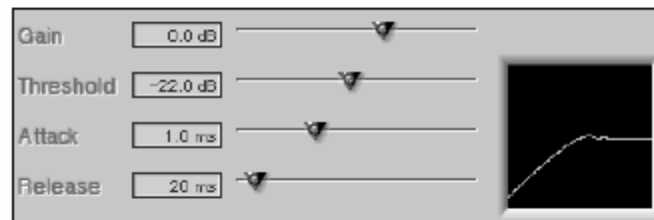
- Threshold, Ratio (expander), Attack
- Hold: length of time gate stays open after the signal falls below the threshold
- Decay: rate at which the gate closes after the signal reaches the threshold
- Range: how much the signal is attenuated



Compressor



Limiter



Gate



Expander



Poorly recorded voice (from cam mic)

Expander to hide room echoes and reduce noise:

- Threshold: just below the softest words (-33 dBFS)
- Attack: 3 ms
- Release: 100 ms

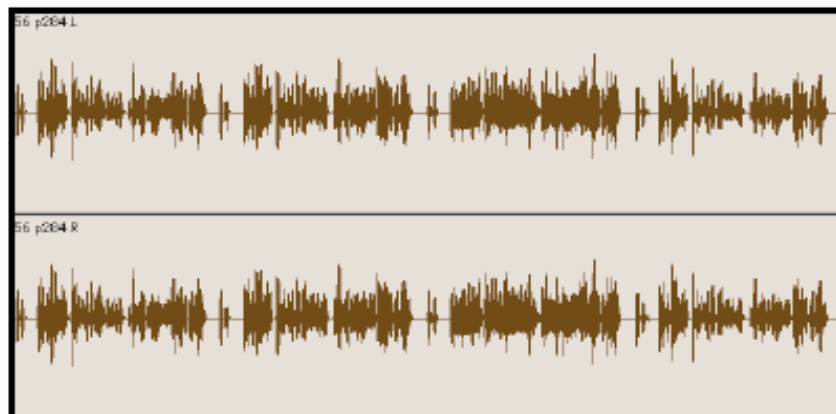


Enhancing the announcer/narrator

Part 2: Compressor (Limiter) used to smooth out her heavily stressed words – Threshold: -12 dBFS, Ratio: 10:1, Attack: 0.9 ms, Release: 10 ms

Part 3: extreme compression

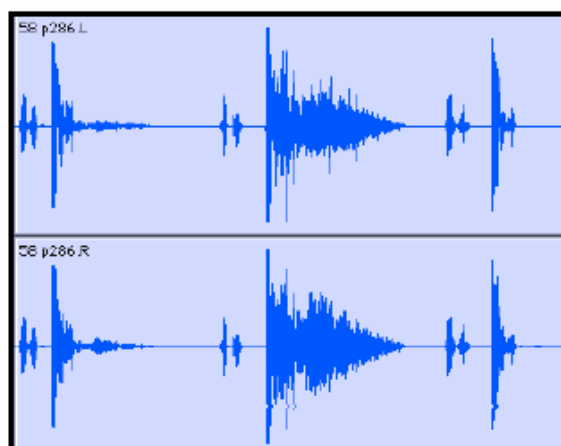
Part 4: De-esser – filter added to compressor: split signal so lower frequencies are unaffected. High frequencies set with Ratio: 8:1, Threshold -28 dBFS (6 dB reduction on sibilants), Attack: 1 ms, Release: 5 ms



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 echo to make it sound more like a drum



Sound Effects

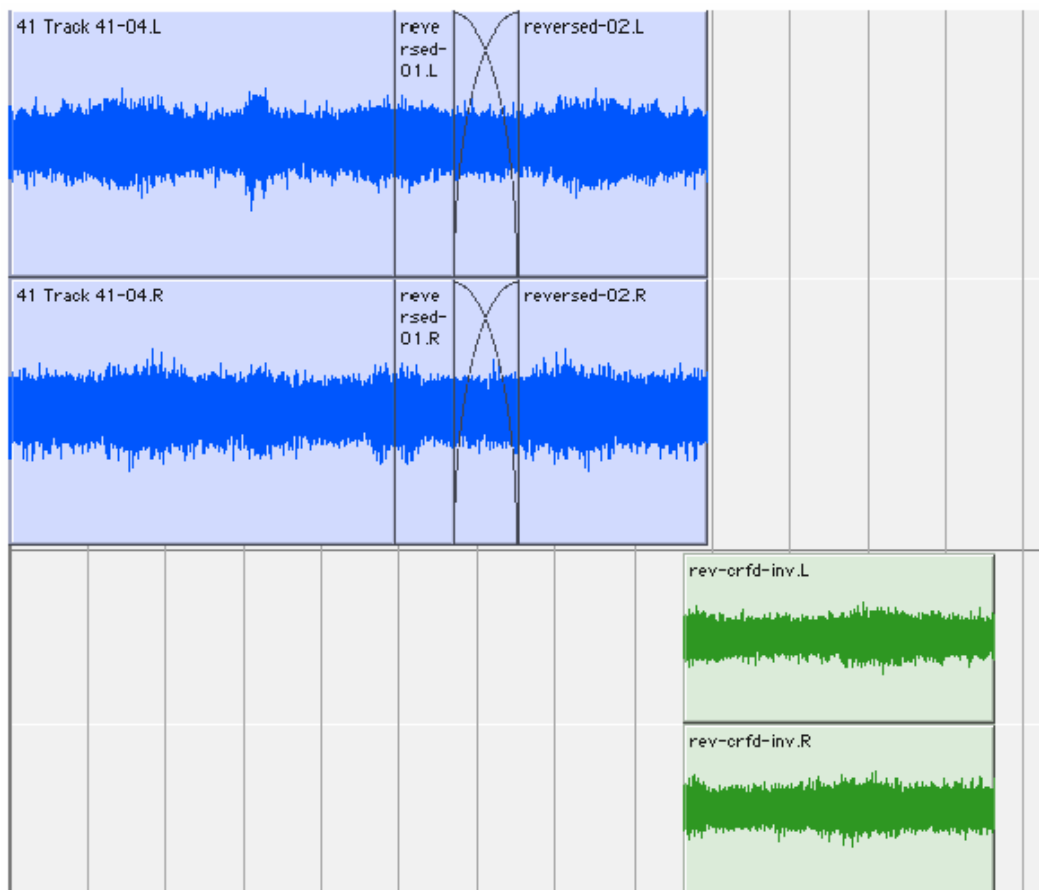
- Add believability to film/TV
- lend to reality – set the environment
- tell the story off screen – see only part of scene while the sound tells us what is happening
- make what looks like a slap in the face, sound like a slap in the face

Types of Effects

- **Hard effects:** sounds linked to an on-screen action
- **Ambience:** background sounds
- **Natural Sound** (nat sound): any sound that is recorded on tape – background (SOT – sound on tape)
- **Foley:** creating sound effects to on-screen actions – use of props

Tips for applying the effects:

- hard effects should be in sync w/picture-starting at the first frame of the action – depending on the action
- background ambience should be present throughout – if not long enough, loop it (drone, hum), layer it (crowd), manipulate



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

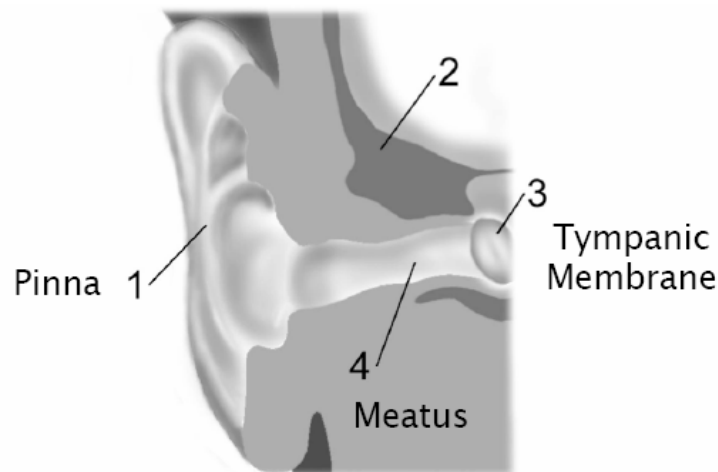
collects and funnels sound to the auditory canal

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



The Middle Ear

Ossicles

3 tiny bones attached to the ear drum ...

malleus

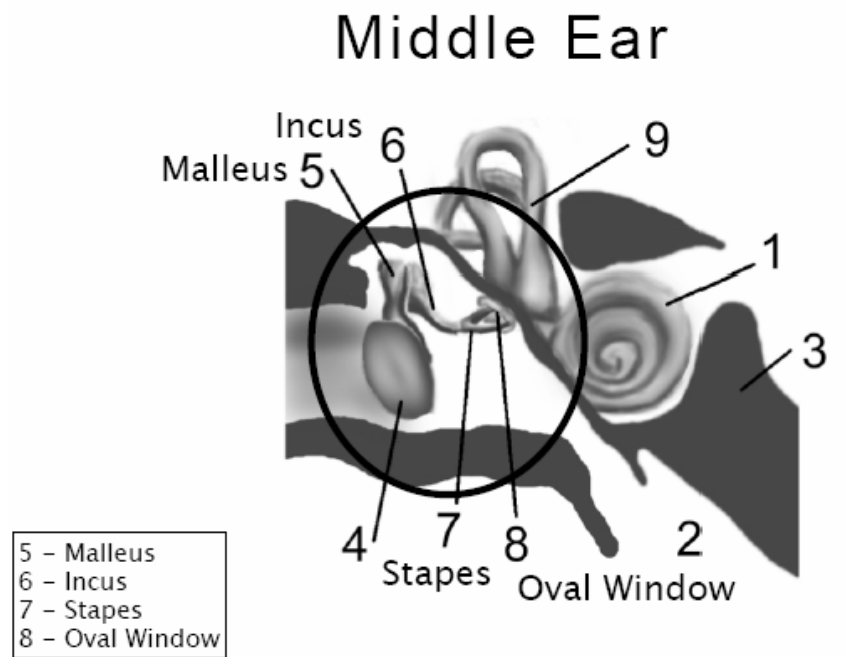
incus

stapes

... and transfer it's vibrations to the ...

Oval Window

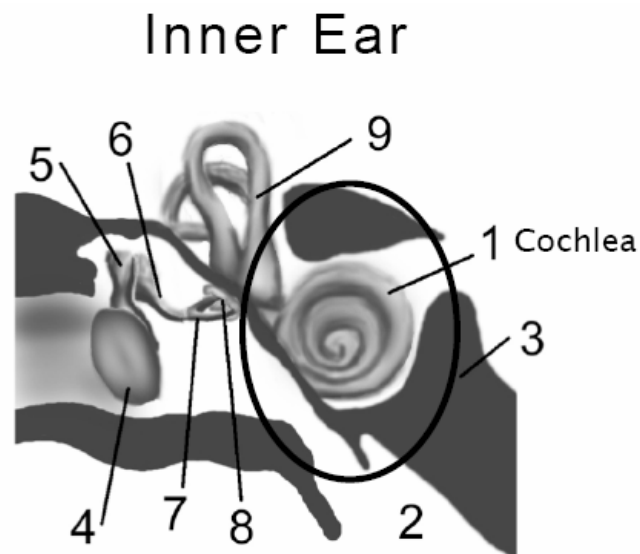
- Membrane separating the middle ear & cochlea
- Vibrations cause a pressure wave to travel in the fluid of the cochlea



The Inner Ear

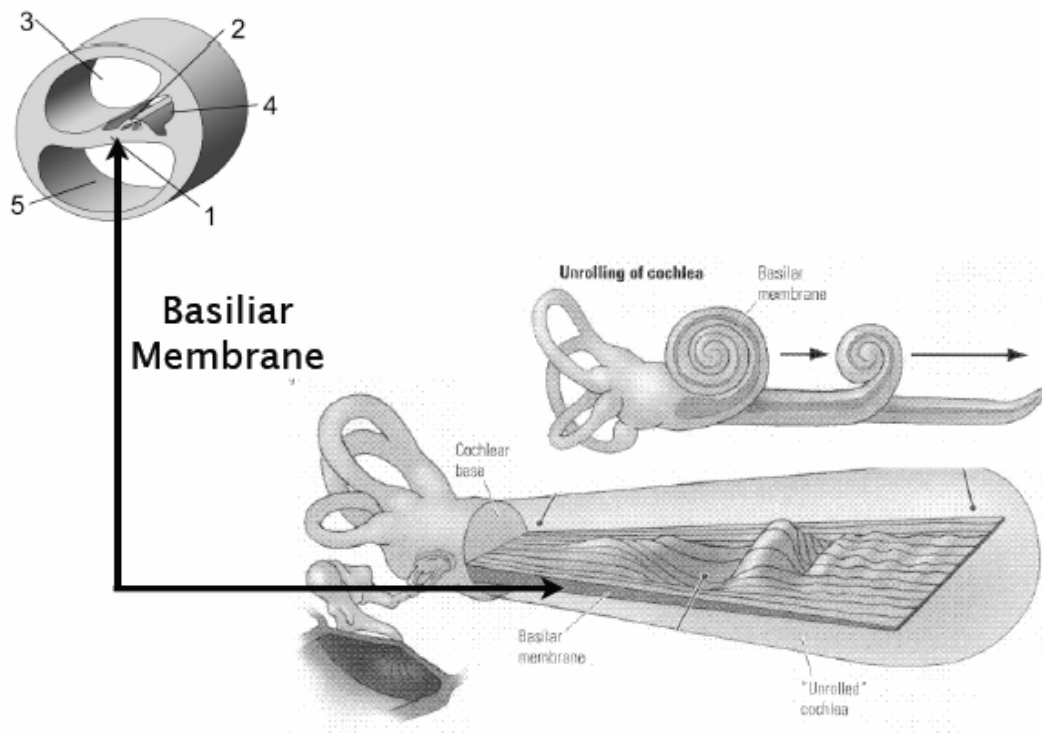
Cochlea

- mechanical to electrical transducer by converting the pressure waves to electrical nerve impulses to the brain
- dividid into 3 party by the:
Reissner's Membrane
Basiliar Membrane



Basiliar Membrane

- driven by the fluid pressure waves
- Organ of Corty – thousands of hair cells that respond to the waves – trigger nerve impulses to the brain via the Auditory Nerve
- pitch discrimination, timbre, consonance/dissonance, maksing, precedence effect



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

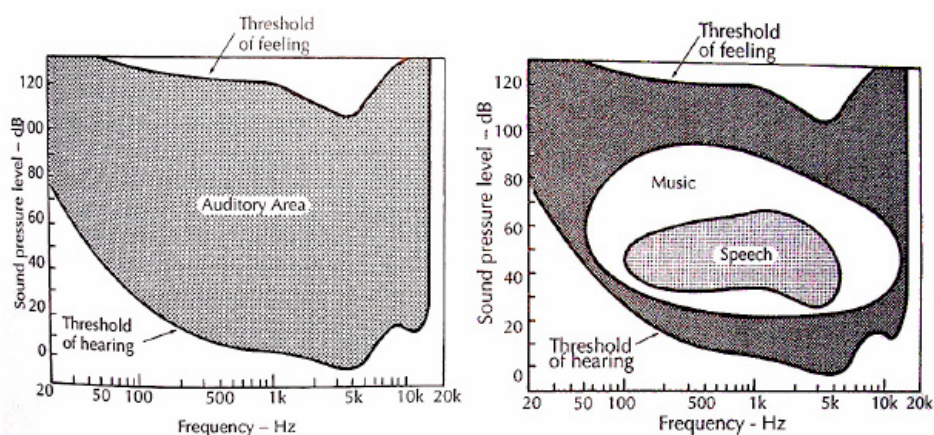
Limitations

Physical

the construction of the ear limits our frequency range (20 Hz – 20 kHz) & dynamic range (140 dB SPL)

How the brain processes information

nonlinearity of the ear – masking, combination tones



Loudness and Level

Threshold of Hearing

1 kHz at 0 dB

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

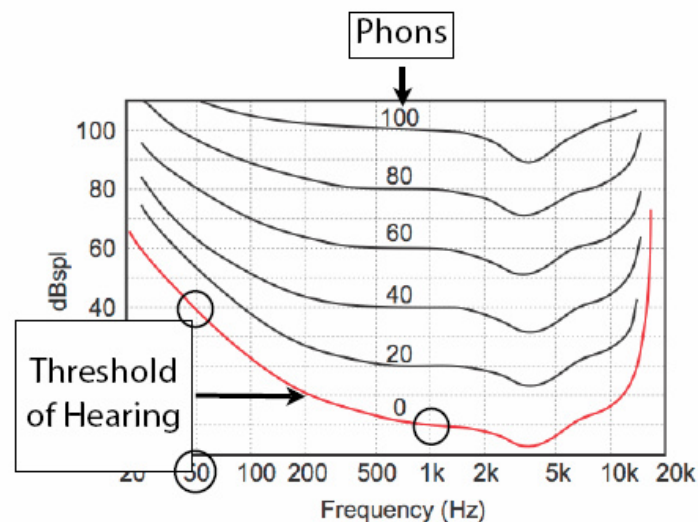
Level

SPL – physical value measurable by the dB

Loudness

subjective – perception value

phons – loudness levels that correspond to sound pressure levels at 1 kHz



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
- recording mixed at an excessively high level will sound very light in bass when played back at a normal level
- use the loudness of sound to determine information about the source, i.e. distance

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 Basilar Membrane – unable to register energy in a band of frequencies when another band has more energy

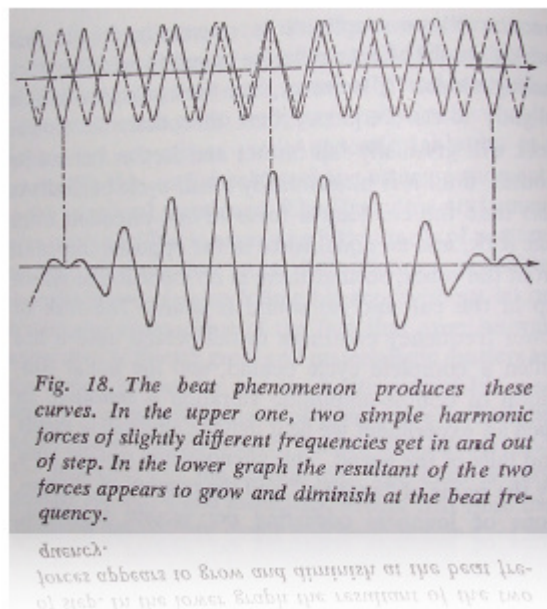
Combination Tones

- 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

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 $\frac{1}{2}$ cycle behind (cancellation), then it continues to fall to a 1 cycle behind (summing)



Localization

How do we know where a sound is coming from?

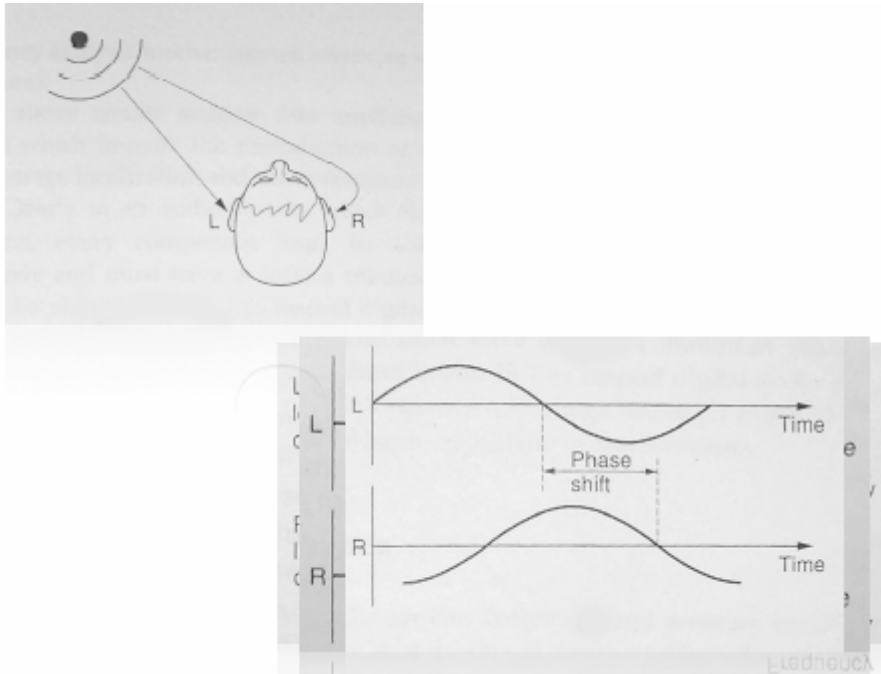
2 ears = Binaural Localization

Inter-aural intensity difference (IID)

- off-centered sound will reach the closer ear with a higher intensity than the distant ear (intensity difference)
- the distance 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

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



Transients

- contains necessary information for localization, size and pitch
- initial transient give location (clap)
- as the rapid decrease of pressure equalizes, determination of the size of sound and frequency analysis 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

Precedence Effect

- Haas Effect, the law of the first wavefront
- refers to how a direct wave and its 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