Sound: • disturbances in the air caused by vibrations
• vibrations produce sound waves

Physical: energy that travels through a medium
• propagates itself by
traveling in waves

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

Wa v e l e n g t h
• physical size
• distance between two identical points
• horizontal length of the wave

• faster something; vibrates, the shorter the wavelength
• slower something; vibrates, the longer the wavelength

ADSR
Attack: The initial strike - time it takes a signal to rise from zero to it's 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

F r e q u e n c y
the rate at which a sound wave completes one cycle in one second

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 as about 110 Hz to as high as 10 kHz

P i t c h (To n h ö h e)
The higher the frequency, the higher the pitch

T h e De c i b e l (d B)
logarithmic measurement that represents how our ears hear the intensities of sound and it's perceived loudness

Transducer
• converts sound from an acoustic signal to an electrical signal

Two main types of transducers: Microphone and Speaker

Electrodynamic Driver
• type of transducer used in a loudspeaker
• cone driver – diaphragm – what pushes and pulls air (coiled wire - voice coil; magnet)

Le ve l s
Line Level
• 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

XLR
• 3 pins: 2 conductors, 1 shield/ground = Balanced
• connects microphones and certain line signals
• professional equipment

Phone or 1/4" or 6 . 5mm, connects audio equipment, instruments, headphones, Mono
• tip/ring or TR
• unbalanced
• connects instruments

Phone or 1/4" or 6 . 5mm, Stereo
• tip/ring/sleeve or TRS
• balanced
• connects equipment

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

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

Dy n ami c Mi c r o p h o n e
• 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

R i b b o n Mi c r o p h o n e
• magnetic induction
• very thin diaphragm/ribbon suspended between the poles of a magnet - moves in response to sound pressure
• low-voltage output
• very sensitive due to thin ribbon - great for low signals - not so great for handling

Condens e r Mi c rophone
• 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

Omnidirectional

responds to sound pressure from all angles, less sensitive to wind noise
Uses:
- lavaliere & 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

Distant Miking

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

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 sound
- audience, presence in a recording studio

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 and 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 & picks up L side with one phase and the R side with inverted phase
• When signals added/subtracted, get a steoreo image as if 2 Cardioids at a 45º 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 90º (to 135º)

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

Lavaliere
• 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 cardioids

Amp l i f i e r s
• 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)

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 Freqencies
Mids
• Medium sized - mid frequencies

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

Controller Interface (Control Surface)
• mimics a full mixer (faders, pan pots, trim etc.)

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Virtual Mixer
• Real-time mixing
• same abilities as real mixer

Prox imi t y e f fe c t
• 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

Tr a n s i e n t R e s p o n s e
• 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

The Nyqui s t Theorem
the highest frequency that can be accurately reproduced has to be less
than half of the sample rate: 44.1 kHz ÷ 2 = 22,050 Hz

Al i a s ing
• "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

Quant i z a t ion
• 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 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

Los s l e s s Forma t s
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

Los s y Forma t s /Pe r ceptua l Coding
MP3 ISO MPEG Layer 3 (MPEG-1 Layer III)
MP3 Pro
MP3 Surround
AAC Advance Audio Coding
• MPEG-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 streaming
• surround-sound encoding

Real Audio (.ra or .ram)
• real-time streaming
• combined with video – Real Media streaming
AC-3
• Dolby Digital
• HDTV (US)
• Home Theater
Ogg Vorbis (open source)

• designed as a substitute for MP3 & WMA

AE S / E BU o r AE S 3
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 / p d i f
Sony/Philips Digital Interface / Format
unbalanced coaxial cable - RCA plugs
uses RCA connectors or optical cable

To s l i n k
consumer fiber optic connector
S/PDIF and ADAT Lightpipe (Alesis)
US B
Universal Serial Bus
medium bandwidth serial digital data
interconnection standard
MIDI
F i r eWi r e / I E E E 1 3 9 4
high speed serial digital 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 format
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 Sy s tem: 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 Conne c t ion: uses a 5-pin DIN connector

Channe l St r ip
(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

Re f l e c t ions
• persistence of a sound after it's source has stopped
• sound bouncing off of hard surfaces
• 2 phenomenon:
Echo
Reverberation

E cho
• a reflection that has significant time delay
• single or multiple repetitions of sound with a fixed timing

Re ve rbe r a t ion
• remainder of sound after the source has stopped
• collection of many reflections
• adds spaciousness to a room
• full and partial cancellations of frequencies

Pha s e
• 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

I nve r s e Squa re L aw
• 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)

Ha rmoni c s
• frequencies above the fundamental that are mathematically related to the 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

S igna l Proce s sor s
Equalizers
• adjusts a frequency or frequency band's volume
• shelf, graphic, parametric
Dynamic Processors
• regulate a sound's dynamic range
• compressors, limiters, expanders, gates

Type s of EQ:
Graphic Equalizer:
• easiest to use
• used to shape the
overall spectrum of
a program
• boost & cut over a
series of center
frequencies

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

Fi l te r s :
• 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 Co n t r o l s
• Level Control
• Frequency Dial
• Q (Bandwidth) Control
Low Q - larger band of frequencies is affected
High Q - smaller band of frequencies is affected

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Some EQ t ips (f rom J a y Ros e)
• 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

Proce s s ing:
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 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)

Adjus t abl e pa r ame te r s
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 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
• same parameters as compressor
• usually used last in the mix

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 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

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

Poorly recorded voice (from cam mic)
Expander to hide room echoes and reduce noise:
• Threshold: just below the softest words (-33dBFS)
• Attack: 3 ms
• Release: 100 ms

Enhancing the announcer/narrator
Part 2: Compressor (Limiter) used to smooth out her heavily stressed words – Theshold: -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

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 E f fe c t s

• 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

Type s of E f fe c t s
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 E a r s
The Human Ear has 3 main parts:
• Outer Ear
• Middle Ear
• Inner Ear

The Oute r E a r
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 Middl e E a r
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 I nne r E a r
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
Basiliar Membrane

The I nne r E a r
Basiliar Membrane
• driven by the fluid pressure waves
• Organ of Corti – thousands of hair cells that respond to the waves - trigger nerve impulses to the brain via the Auditory Nerve
• pitch discrimination, timbre, consonance/ dissonance, masking, precedence effect

Organ of Cor t i
• 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 Basiliar Membrane
• brain decodes the level of the sound by how many of the hairs are moving

L imi t a t ions
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

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

Level
SPL – physical value measurable by the dB
Loudness
subjective – perception value
phons – loudness levels that correspond to sound pressure levels at 1kHz

Equa l Loudne s s Cur ve s

• 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

Ma s k ing

• 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

Combina t ion Tone s

• 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

Be a t s

• "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)

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

Tr ans i ent s
• 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

Pre cedence E f fe c t
• 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