

# Pro Tools 101

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Adapted from Frank Cook's Training Manuals

What is Pro Tools?

# Digital Audio Workstation

- Pro Tools Has Five Major Components
  - Audio Processing
  - MIDI Processing
  - Notation and Scores
  - Mixing and Automation
  - Audio for Video Post-Production

# Pro Tools History and Evolution

- The reason we are going over this is because it would be on the final exam for PT 101.
- PT was designed by Digidesign and it's founders Peter Gotcher and Evan Brooks
- Early 1980's Gotcher and Brooks devised away to get drum samples on to EPROM computer chips.
- 1984 Digidrums was born and out of that company came a program called Sound Designer. Sound Designer was the first commercial product to combine waveform editing with a graphical display.

# History Continued

- 1985: Digidrums became Digidesign. Digidesign began developing products for MIDI and synthesis on Macintosh computers.
- 1988: Digidesign began selling computer cards for playing back digital audio. The product was Sound Accelerator. It was CD quality 2-channel output card for the Mac II computer.
- 1989: Digidesign released the first “tapeless recording studio” called Sound Tools. This was Sound Designer II and a Sound Accelerator Card and a hardware box called AD that provided two analog-to-digital converters.

# Pro Tools History Cont.

- 1991: Pro Tools was officially released. It only recorded four tracks but eventually software allowed for 16 tracks.
- 1992: Pro Tools Session 8 was released as the first Windows-based version of Pro Tools. Two years later Digidesign introduced Pro Tools TDM. This opened the door for real-time plug-ins.
- 1995: Digidesign merges with Avid Technology
- 1997: Pro Tools |24 came out with 24-bit recording technology.

- 1999: Pro Tools LE provided *host-based or native* audio processing. Also control surfaces were being introduced. This gave touch sensitive mixing control to the engineers
  - ProControl
  - Control|24
- 2002:
  - Pro Tools|HD Hardware Systems were released. These gave higher sample and bit rates to the systems.
  - Mbox was released.
- 2006: Sibelius was bought by Avid.

# Pro Tools in This Decade

- April 2010
  - Avid Acquired Euphonix a company who was known for high-end consoles and EUCON control surface technology. Out of this union came the following consoles.
    - Artist Control
    - Artist Mix
    - Artist Transport
    - Pro Tools|S3 and S6 (Their flagship control surface)
  - New Mbox
  - New HD Interfaces
  - HD Native Platform



# PT in this Decade Continued

- 2012: PT 10 released with new programming. AAX 64 bit audio engine
- 2013: PT 11: 64-bit architecture
- 2015: PT 12: Was released with new functions including
  - Input monitoring
  - VCA controls
  - Track Freeze and Track Bounce
  - Track Commit – This is HUGE. The processed track can be frozen with all the affected audio on it. This frees up a great amount of resources.
  - Cloud saving and Dolby ATMOS

# Pro Tools Ultimate 2018 and First

- Ultimate 2018
  - Renaming of Pro Tools. Pro Tools is now on a subscription basis.
  - All new updates will be pushed out this method.
- First
  - Free Software similar to GarageBand, but allows more functionality than GB.
  - Three Projects saved in the cloud.
  - Subscription based at \$5 a month with 10 Gb of space

Avid Based Systems

# Software Options

- Pro Tools First
- Pro Tools Standard – What most of us will use until we hit the big time.
- Pro Tools HD: This can be stand-alone software or for use with the Pro Tools | HD Native or Pro Tools | HDX hardware.
  - This provides advanced automation, video editing, and surround mixing capabilities.

# Host Based vs. DSP – Accelerated Systems

- Host Based is working solely on the systems in the computer.
- DSP – Digital Signal Processors work on hardware other than the system.

# Audio Interface Options

# Fast Track Family – Way of the Dodo

- Non HD
- USB
- Third Party Interfaces
  - UAD
  - Focusrite
  - and others....



# Mbox Family – Discontinued as Well

- Mbox Family
- Firewire or USB
- Sample rates up to 192K





# Eleven Rack

- Guitar Interface
- Non HD
- USB 2.0
- Standalone Amp
- 8 Channels of Recording
- 96K Highest Sample Rate



# Duet and Quartet – Discontinued as Well

- Non HD
- Originally Apogee Interface
- Allowed for Multiple Inputs 192k



**The Pro Tools PC**  
The Only PC Designed For Pro Tools

Avid Discontinue The Pro Tools  
Duet And Quartet By Apogee

# Pro Tools | HD Series Audio Interfaces: Omni

- Omni
  - 2 XLR Ins
  - 4 Line Ins
  - 8 Outputs
- HD



# Pro Tools HD Interfaces: I/O

- Features
- 8x8x8
- 16x16x16 analog
- 16x16x16 digital



# Pro Tools HD Interface: MADI

- Features
- Co-axial Cable
- 64 channels



# Audio Basics

# Sound Waves – How Do We Hear

- Sound
- Compression/Rarefaction: speaker cone
- Sound travels 1130 feet per second
- Sound waves hit receiver
- Sound waves tend to spread out as they travel away from source, getting weaker over distance (halves each time distance doubles: inverse square law)

# Oscilloscope

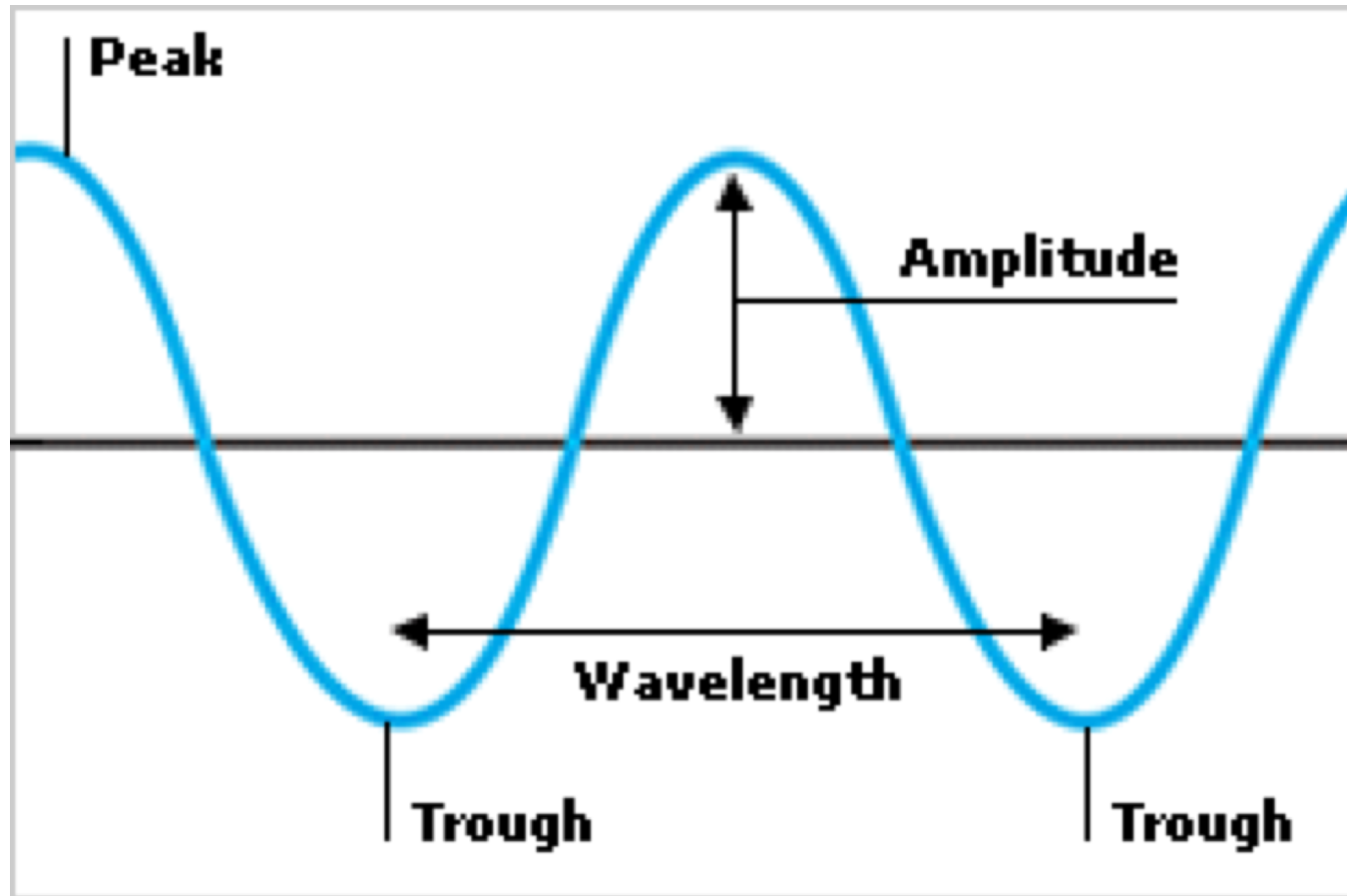
- Let's look at some sound waves.
  - The form that is on the screen is called the wave form.
  - This form can be manipulated depending on the overtone series it produces.
  - The oscilloscope I have is not very good.



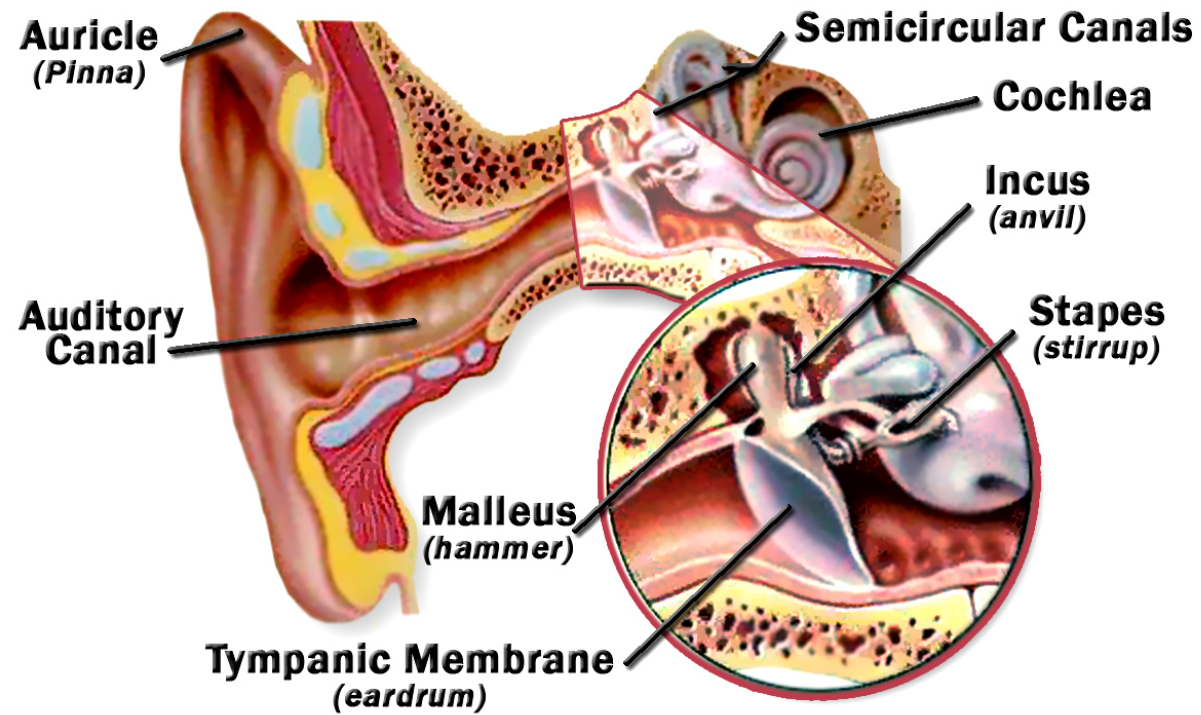
# Waveform

- Waveform is the “shape” of sound – or more accurately, the shape of the vibration that produced the sound. This shaped can be influenced by a number of different factors.
  - Size
  - Shape (String, Vocal Chord, Brass, Woodwind, Percussion)
  - Everything Vibrates Differently – Timbre and the Overtone Series

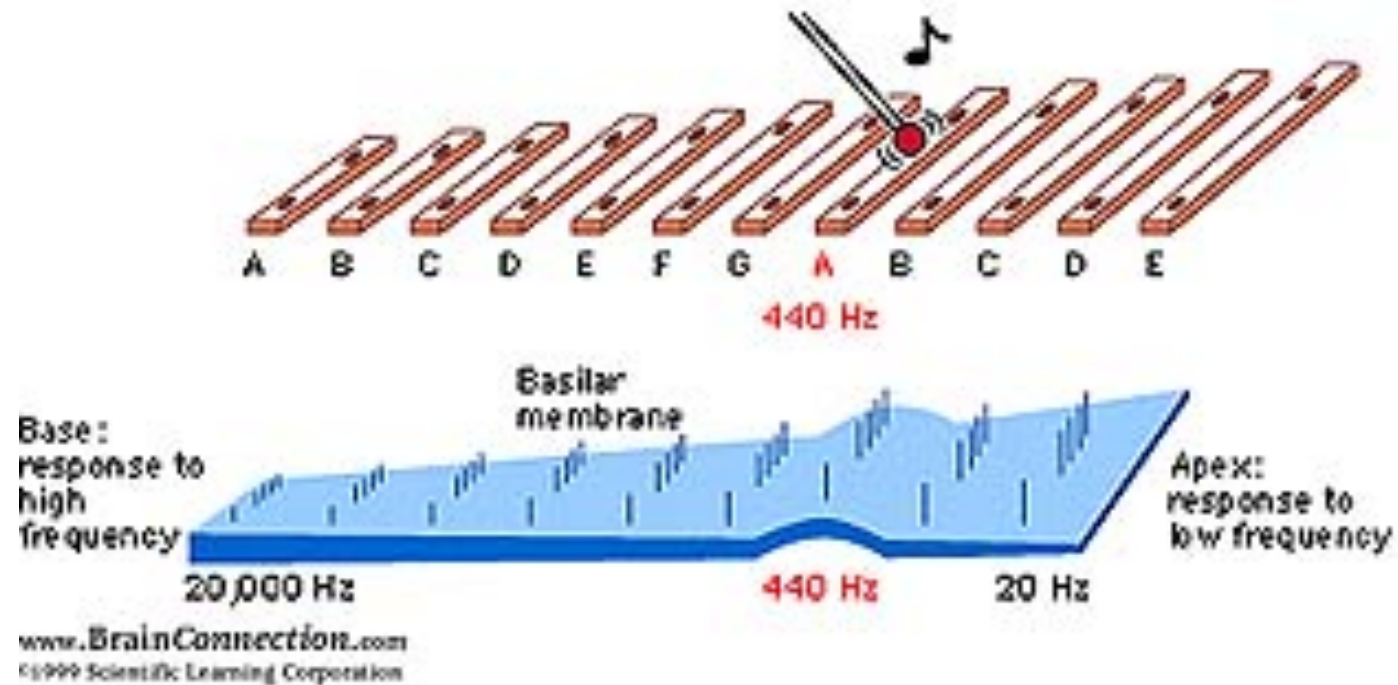
# Sound Waves



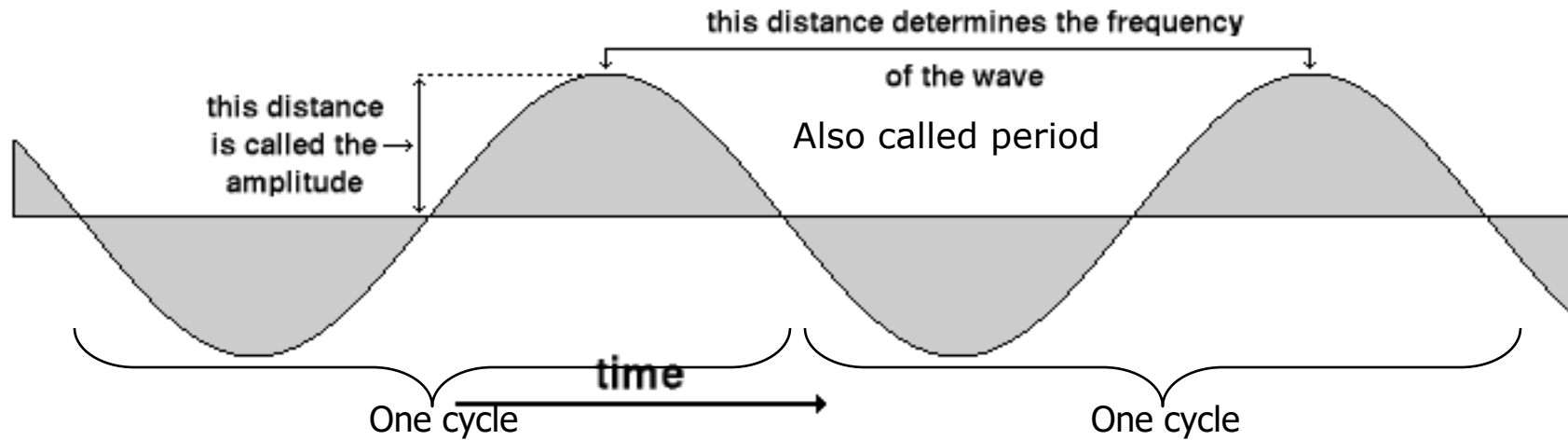
# Inner Ear



# Basilar Membrane



# Sound Waves: Frequency



- Frequency (Hz or kHz) number of cycles per second. Named for Henrich Hertz. The scientist who theorized that sound traveled in waves
- Frequency: highness or lowness of sound
- Human hearing range: 20 Hz – 20kHz (20,000 Hz)
- Doubling Frequency raises the pitch one octave
- [Instrument Frequency Chart](#)

# Audio Basics: Amplitude

- The intensity or of the sound pressure variations that reaches our ears creates our perception of the loudness of the sound.
- Measured in Decibels: dB
- Logarithmic unit that used to describe a ratio of sound pressure. It is not linear.
- As the amplitude increases the sound becomes louder.
  - Doubling intensity of the sound pressure variations produces a 3dB gain.
  - Doubling a sounds loudness results in a 10dB gain.

# Microphones

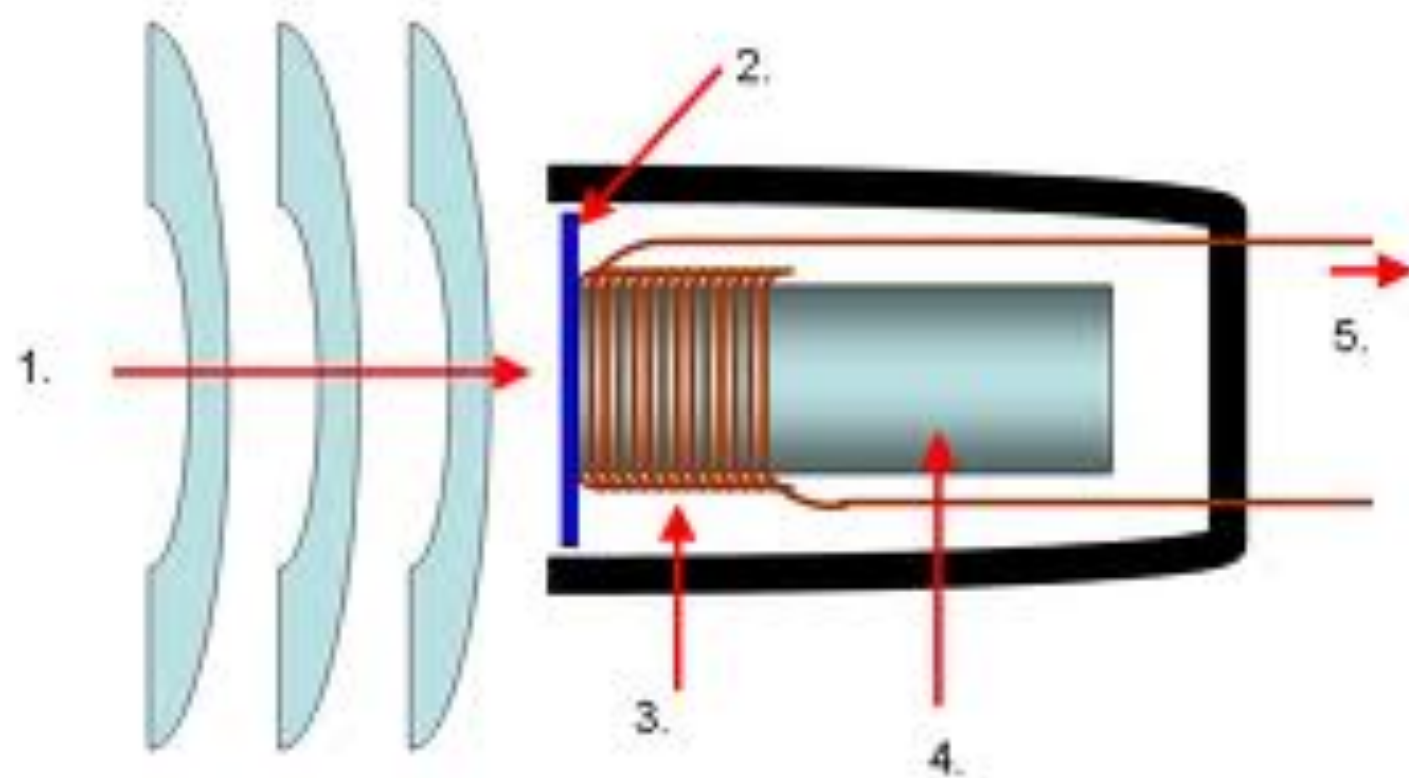
- Transducer: device that changes one form of energy into another
- Mics and speakers: sound to electrical signal

# Microphones: types

- Dynamic: a coil of wire attached to a diaphragm suspended in a magnetic field.
- Sound waves vibrate diaphragm, coil vibrates in magnetic field and generates an electrical signal similar to a sound wave.



# Dynamic Microphone

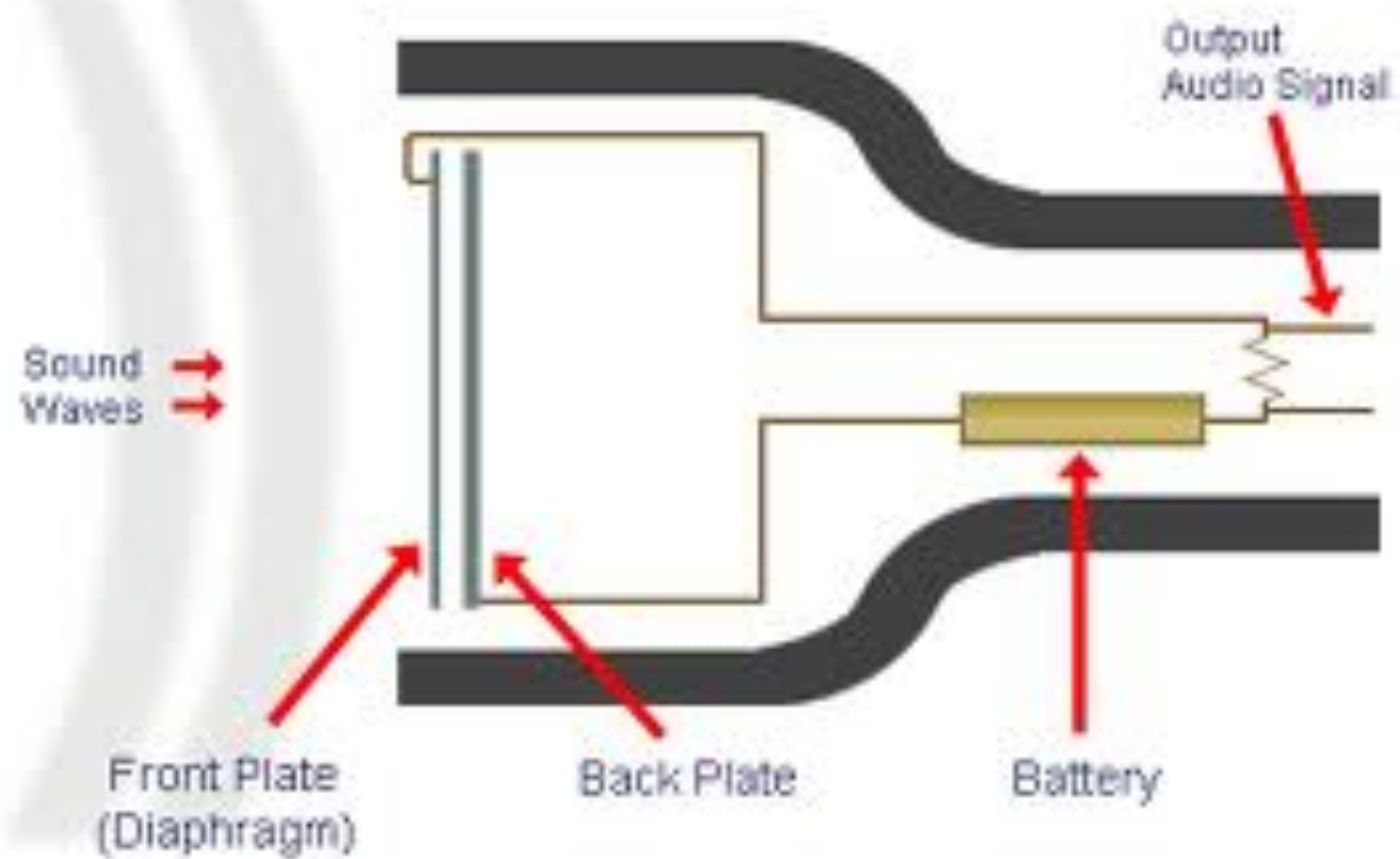


# Microphones: Dynamic

- Tends to have rougher response
- Rugged and reliable
- Handles heat, cold, and high humidity
- Preferred for high SPL: guitar amps, drums
- If flat response, can take the “edge” off sounds (woodwinds, brass)

# Microphones: Condenser

- Sometimes called capacitor mic
- 2 parts: conductive diaphragm and metal backplate, spaced very close together
- Both charged with static electricity
- When sound waves strike diaphragm, it vibrates, varying the spacing between the plates
- This variation generates a similar signal to incoming sound wave
- Diaphragm mass is lower, responds faster to rapidly changing sound waves (transients)



# Microphones: Condensers

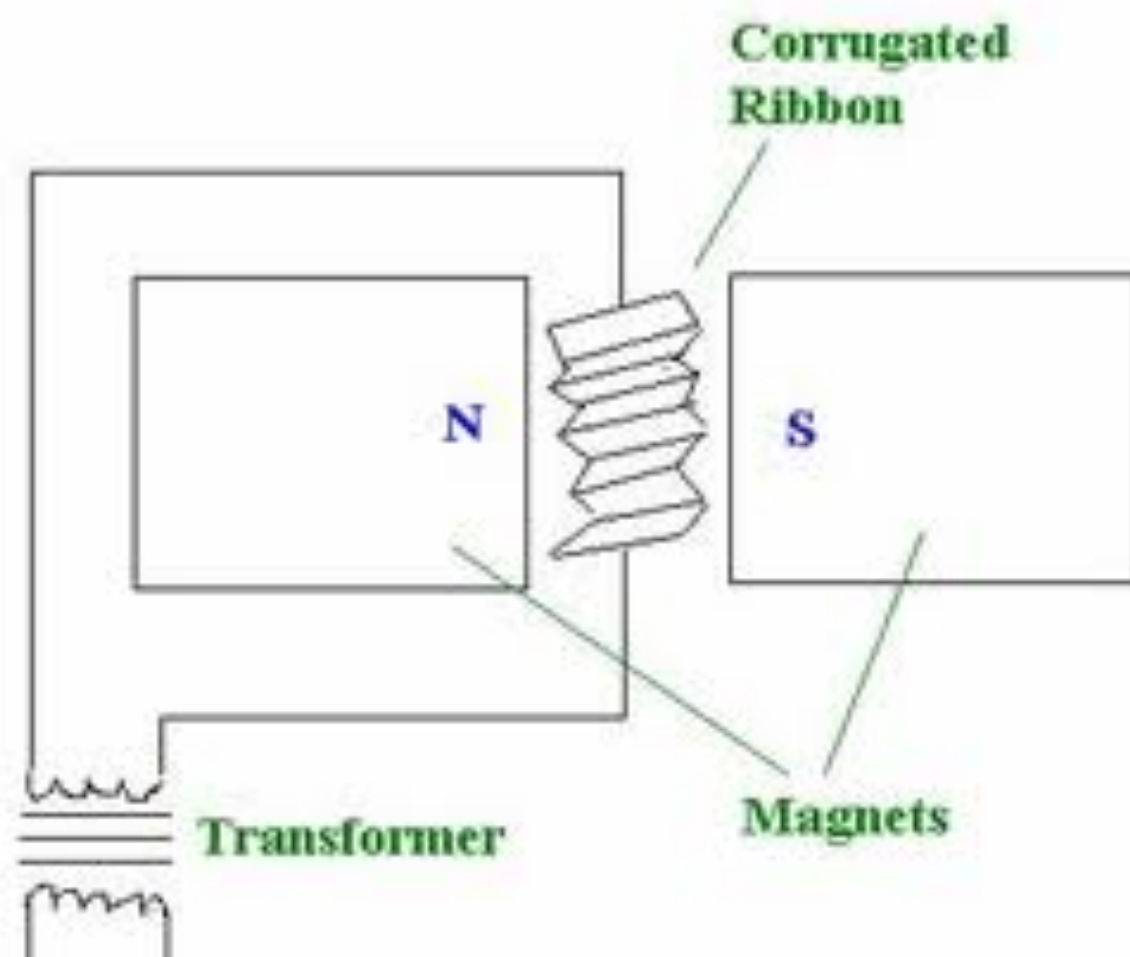
- Condenser mics need external power supply (battery or phantom power)
- Phantom power: 12 to 48 volts supplied through XLR cable
- Most mixers have a switch to turn on phantom power

# Microphones: Condenser traits

- Wide, smooth frequency response
- Detailed sound, Extended Highs
- Omni type has excellent low-frequency response
- Transient attacks sharp and clear
- Preferred for acoustic instruments, cymbals, studio vocals
- Can be miniaturized

# Microphones: Ribbon

- Thin metal foil or ribbon is suspended in a magnetic field
- Sound waves vibrate the ribbon in the field and generate a electrical signal





# Microphones: Ribbon

- Prized for its warm, smooth tone quality
- Delicate, damaged by temperature and humidity changes
- Complements digital recording
- Expensive

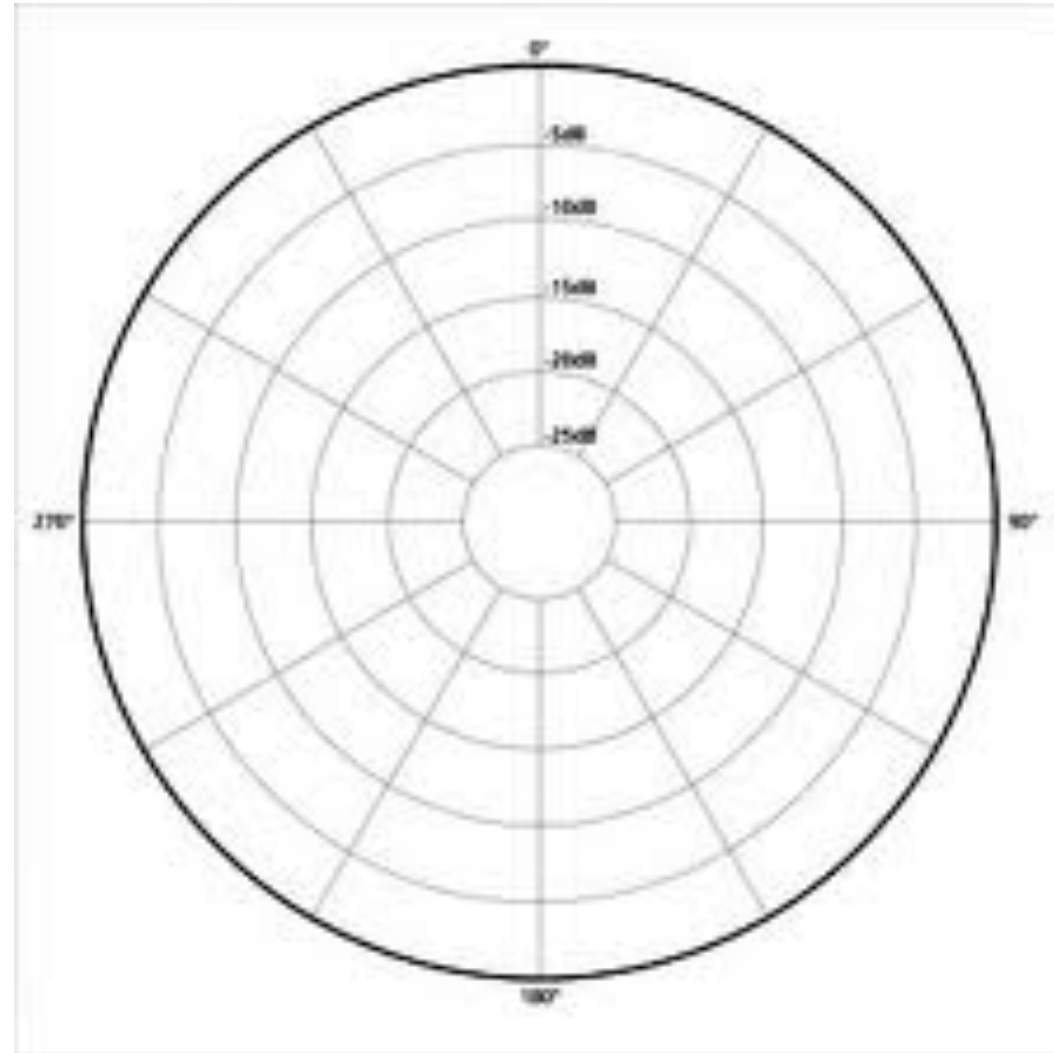
# Microphones: Polar Patterns

- Polar pattern refers to the way a microphone responds from different directions
- Graph based on sensitivity measured in decibels.
- Uni-directional, Omni-directional, Bi-directional (figure-8)
- Three types of unidirectional patterns: cardioid, supercardioid, and hypercardioid

# Microphones: Omni-directional

- All around pickup
- Picks up room reverb
- Not much isolation
- Low sensitivity to pops
- Low handling noise
- No up-close bass boost (proximity effect)
- Extended low frequency response (condensor) great for organ, orchestra, etc...
- Lower cost (in general)

# Microphone: Omni Pattern



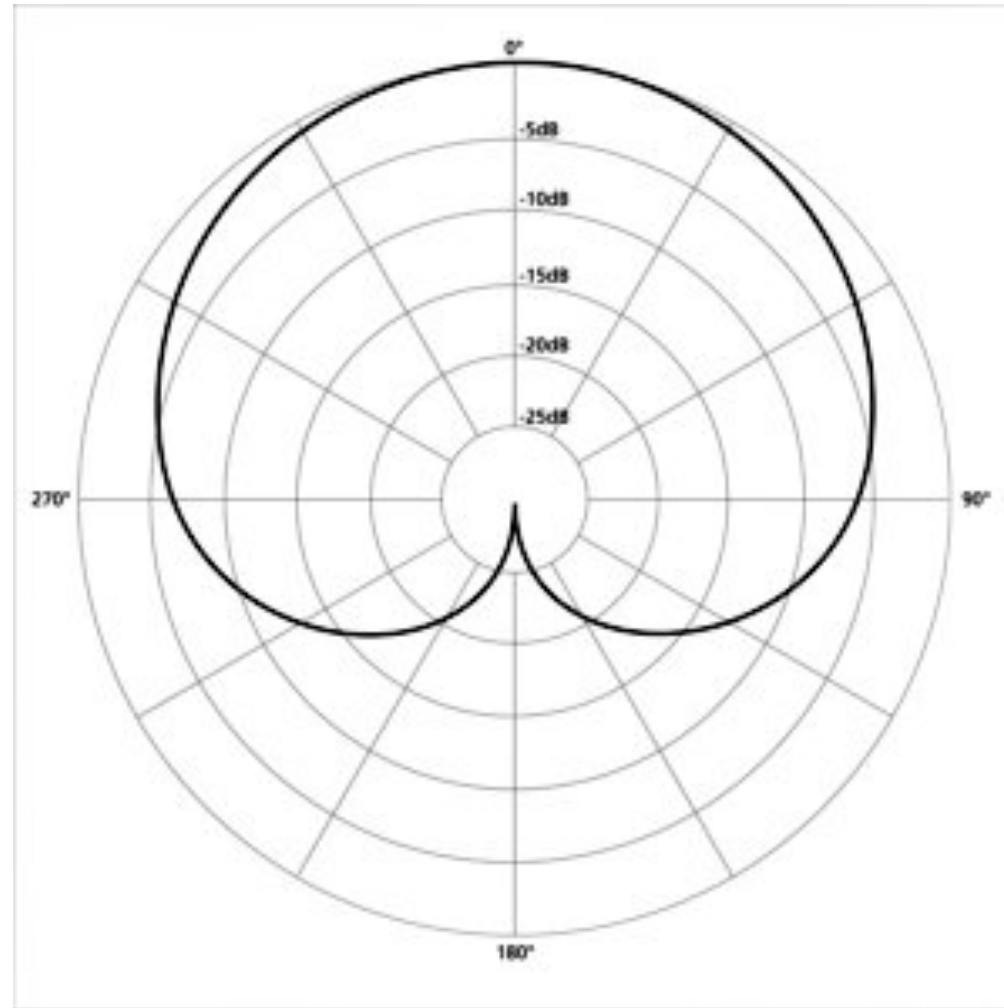
# Microphones: Unidirectional (cardioid, supercardioid, hypercardioid)

- Selective pickup
- Rejection of room acoustics, background noise, and leakage
- Up-close bass boost
- Better gain-before-feedback in live sound

# Microphone: Cardioid

- Broad-angle pickup of sources in front of mic
- Maximum rejection of sound approaching the rear of the mic
- Most popular pattern

# Microphone: Cardioid

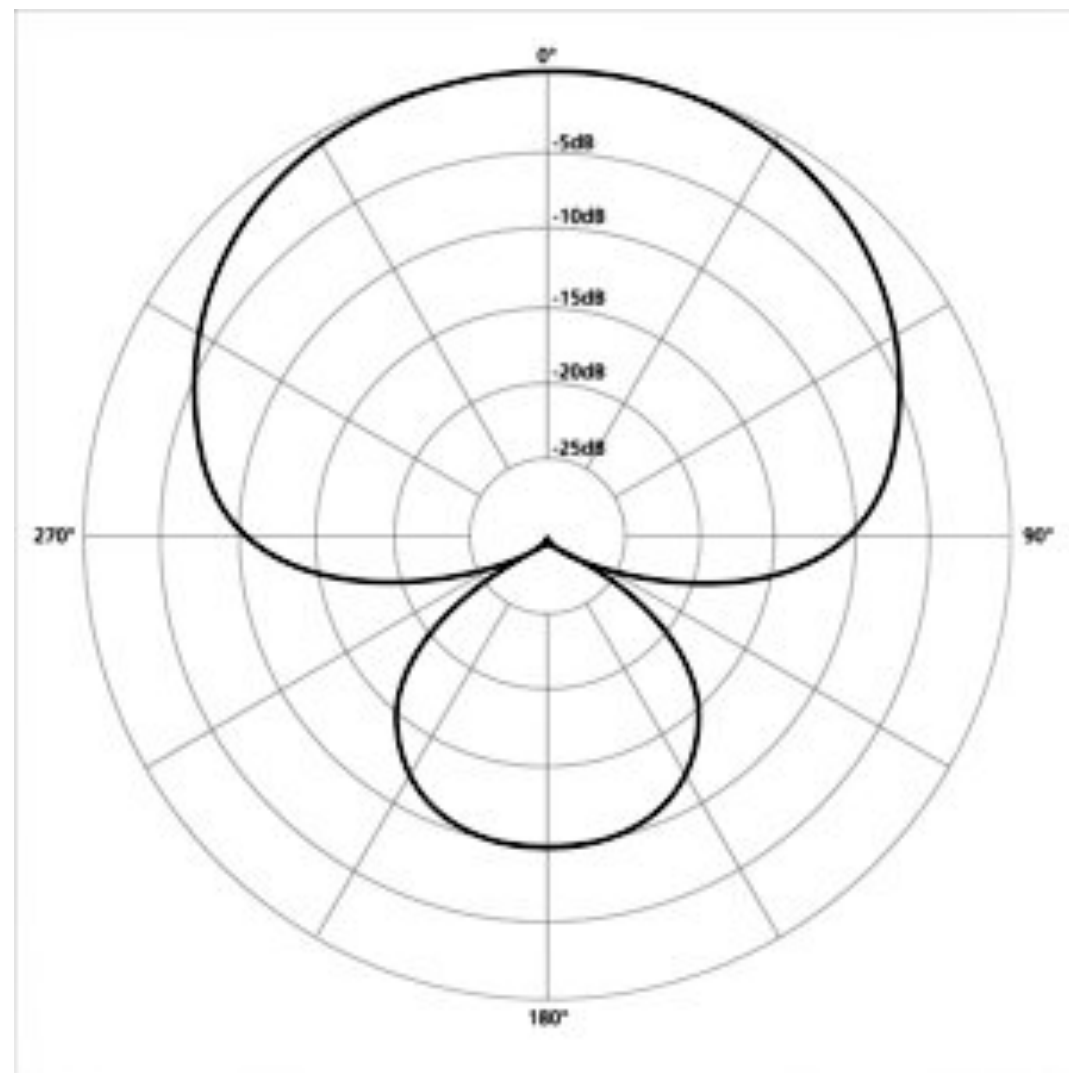


# Microphones: Supercardioid

- Maximum difference between front hemisphere and rear hemisphere
- More isolation than cardioid
- Less reverb than cardioid



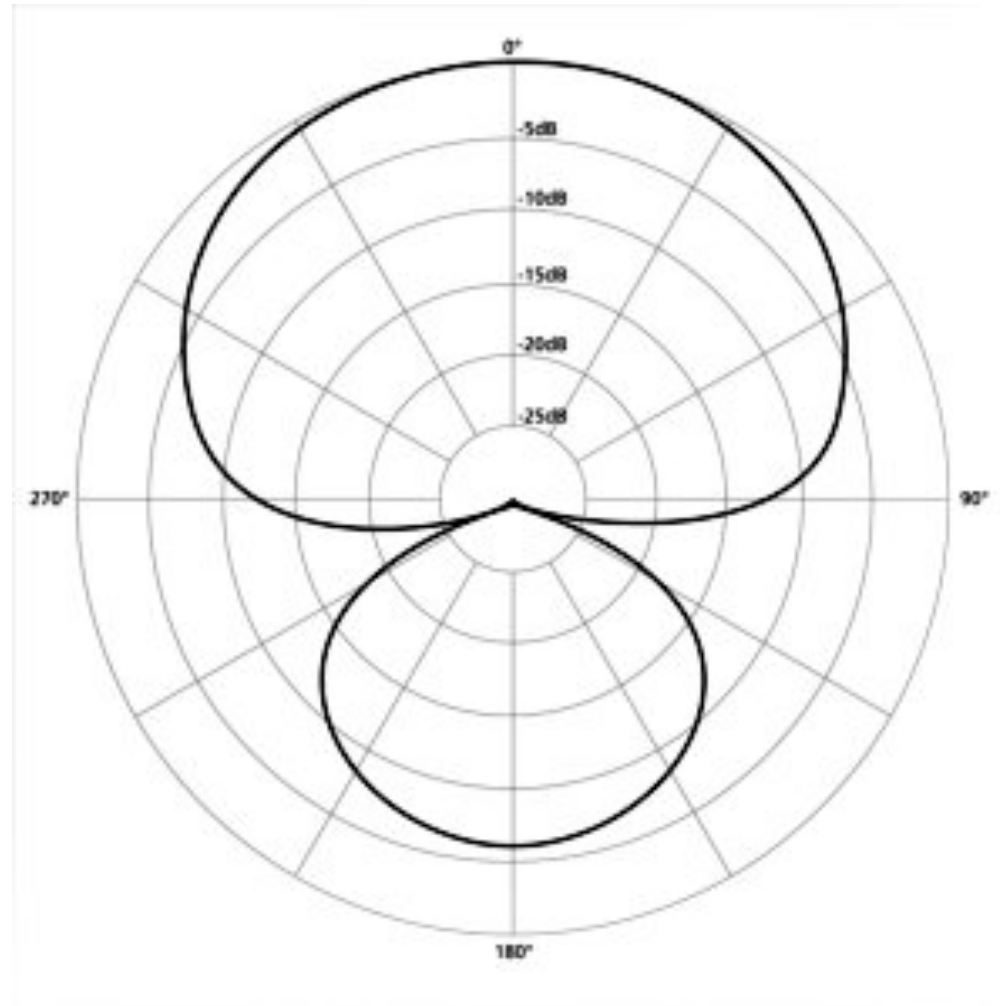
# Microphone: Supercardioid



# Microphone: Hyper-cardioid

- Maximum side rejection
- Maximum isolation – maximum rejection of reverb, leakage, and background noise

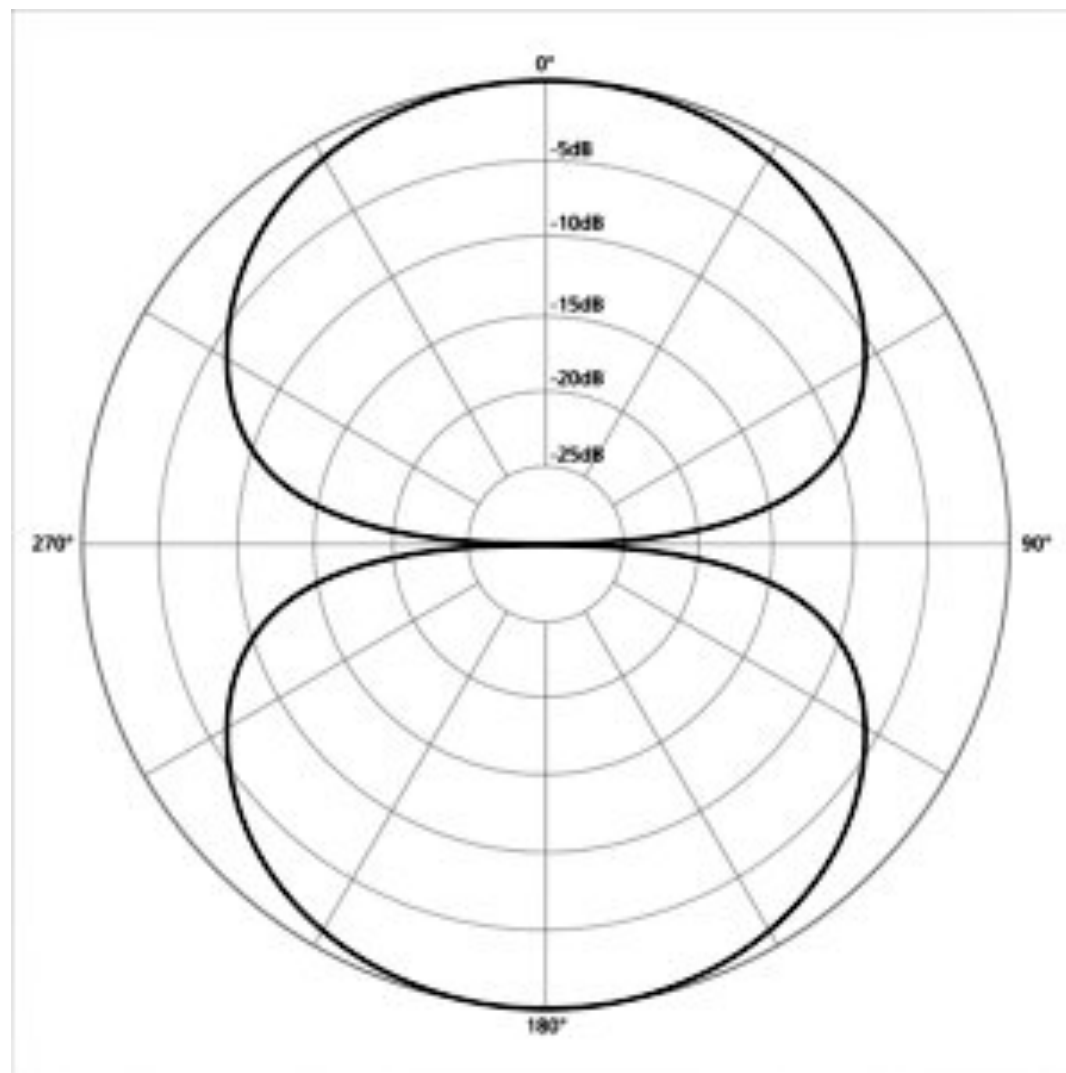
# Microphone: Hypercardioid



# Microphone: Bi-directional

- Also called figure-8
- Front and rear pickup, with sides rejected
- Across table interviews, two-part vocals/instrumentals

# Microphone: Bi-directional



# Microphones

- Frequency response varies from proximity: off axis coloration
- Off axis coloration: mic will have a different tone quality on and off axis

# Microphones:

- Condensers or Dynamics come in all polar patterns (except bi-directional dynamic)
- End-addressed or side addressed
- Switchable polar patterns common with side addressed mics
- Boundary mics: half pattern (usually omni); rejects room acoustics

# Boundary Microphone





# Mic Accessories

- Pop Filter: breath pops “p”, “b”, or “t” sounds
- Stands and booms
- Shock Mount
- Snake

# Microphones: Frequency Response

■ Range of frequencies a mic will reproduce at an equal level (within a tolerance like  $\pm 3\text{db}$ )

- Most instruments: 80Hz – 15kHz
- Bass instruments: 40Hz - 9kHz
- Brass and voice: 80 Hz – 12kHz
- Piano: 40 Hz – 12 kHz
- Cymbals and some percussion: 300 Hz – 15 or 20 kHz
- Orchestra or symphonic band: 40 Hz – 15kHz

# Microphones: Frequency Response

- Roll-off switches: attenuates frequencies below a point below fundamental frequency of instrument you're recording (guitar does not play below 80 Hz)
- Frequency response curve
- Presence peak: rising high end around 5 – 10kHz makes sounds more crisp and articulate (crunchy)
- Proximity effect: a bass boost tendency of a mic when placed too close.

# Microphones: Other ratings

- Impedance
- Maximum SPL: the point at which a mic starts to distort – 125db good, 135db better, 150db excellent
- Sensitivity: how much output voltage a mic produces when driven by a certain SPL
- Self-noise: noise level or hiss a mic produces
- Signal to Noise Ratio: the difference in decibels between mic's sensitivity and it's self noise

# Microphone types

## ■ Large-Diaphragm Condenser

- Good low frequency response and low self-noise

## ■ Small-Diaphragm Condenser

- Stick shaped or pencil, usually cardioid and end addressed, good transient response and detail, good for acoustic instruments

## ■ Dynamic Instrument

- Stick shaped, end addressed, be careful of presence peak, used on drums and guitar amps

# Microphone types

## ■ Live-Vocal

- Ice cream cone shaped large grill to reduce breath pops, can be condenser, ribbon, or dynamic, usually has presence peak and low frequency roll-off

## ■ Boundary Mic

- Used on surfaces like piano lid, wall, picks up direct and reflected sound at the same time for smooth response

# Microphone types

- Miniature Mic
- Stereo Mic: combines two directional mic capsules in a single housing
- Digital (USB) Mic – built-in a/d converter

# Microphone Selection

<b>Natural, smooth tone quality</b>	<b>Flat frequency response</b>
<b>Bright, present tone quality</b>	<b>Presence peak (5kHz)</b>
<b>Extended lows</b>	<b>Omni condenser dynamic with extended low frequency response</b>
<b>Extended highs</b>	<b>condenser</b>



# Choosing a Microphone

<b>Boosted bass up close</b>	<b>Directional mic</b>
<b>Reduced leakage, reduced room acoustics</b>	<b>Directional</b>
<b>Enhanced acoustic</b>	<b>Omni</b>
<b>Miking close to a surface</b>	<b>Boundry (TMZ)</b>

# Choosing a Mic

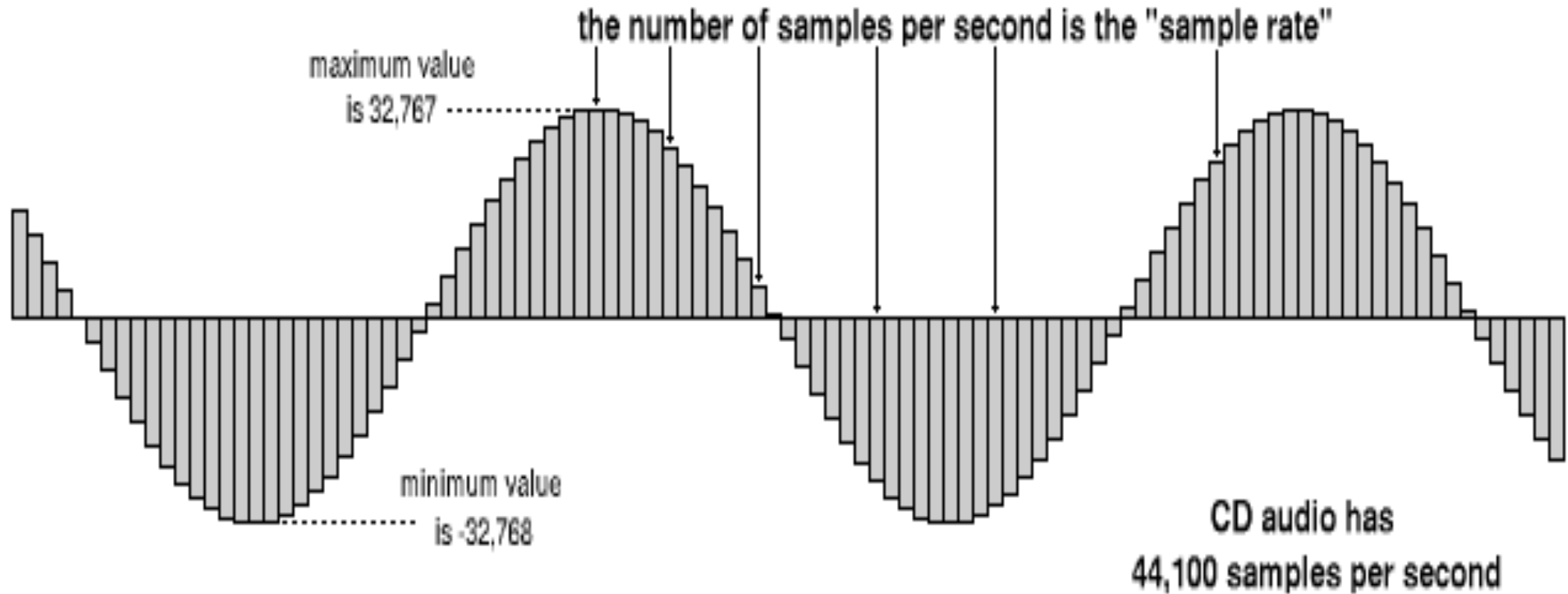
<b>Coincident or near-coincident</b>	<b>Stereo mic</b>
<b>Extra ruggedness</b>	<b>Dynamic</b>
<b>Distortion-free pick-up of loud sounds</b>	<b>Condenser with high SPL spec or dynamic</b>
<b>Low self-noise, high sensitivity, noise-free pickup of quiet sounds</b>	<b>Large-diaphragm condenser mic</b>

Converting Audio to Digital

# What is a Sample?

- Sampling (in this case) is the process of taking discrete measurements of an electrical signal at various moments in time.
- Each measurement, or sample, is a digital representation of the signal voltage at that instant. Played back in succession, these samples approximate the original signal like a flip photobook or a film.

# Sample Rate



# Nyquist Theorem or Sampling Theorem

- In order to produce an accurate representation of a given frequency of sound, each cycle of the sound's vibration must be sampled a minimum of two times.
- If this does not happen alias tones or wrong tones will be produced.

# Standard Sampling Rate

- As stated earlier, humans hear from 20Hz – 20kHz.
- We could in theory record at 40k Sampling Rate and be okay. There are people who hear beyond this.
- So we record at 22.05 Samples per second.
  - If we double that we get 44.1 K as a sample rate. This is the standard sample rate for a CD
  - 48K is used for video
  - 88.2 doubles the original.
  - 96K doubles for video.

# Welcome to Binary

- Computers use binary digits called bits (0's and 1's) to represent data collected. The number of of bits used for each sample is called a *binary word length* or bit depth.
- The more digits the better the accuracy of each sample measurement.
- The relative loudness is rounded is quantized or rounded to the closest available whole-number value within the word length.



# Example

- The range of numeric values available for each sample at a given bit depth is equal to 2 to the nth power ( $2^n$ ), where n is the number of bits in the binary word.
- A 4-bit word can only represent 16 distinct amplitude levels.
- A 16-bit word can represent 65,536 discrete amplitude levels.
- A 24-bit word can represent 16,777,216 discrete amplitude levels.

# Importance of Bit-Depth

- The main thing that you need to know about bit depth is that is related to volume.
  - 8 bit: 48dB
  - 16 bit: 96dB
  - 24 bit: 144dB
  - 32 bit floating point: 32-bit floating point files represent discrete amplitude levels in the same way a 24-bit file does. The extra 8 bits provide exponent biasing and allow for headroom above full-scale 24-bit audio.

# Minimum Bit-Depth

- The music we generally listen to is from 40-105dB. This would require 11 bits in the binary word.
- Pro Tools starts off at a minimum of 16-bits as a minimum bit depth. Plus math really gets funky when one uses odd bit depths.

# Calculating Dynamic Range

- 8-Bit x 6 = 48dB
- 16-Bit x 6 = 96dB
- 24-Bit x 6 = 144dB

Recording in a Digital Format

# Digital Transfers

- If it is digital already, just bring in the file. Converting it to convert it back would only degrade the file.
- Digital to Audio Connections
  - S/PDIF – Sony Phillips Digital Interface.
    - Looks like RCA or Optical Cable.
  - AES/EBU – Looks like an XLR cable
    - Audio Engineering Society or European Broadcast Union
- AES/EBU is more preferred.