

**21M.380**

**Music and Technology: Recording Techniques and Audio  
Production**

**Christopher Ariza**

**21M.380: Music and Technology: Recording Techniques and Audio Production**  
by Christopher Ariza

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# Chapter 1. Meeting 1, The Tools of Audio Engineering

## 1.1. Announcements

- 21M.380: Music Technology: Recording Techniques and Audio Production
- Foundations, practices, and creative techniques in audio recording and music production, including microphone selection and placement, mixing, mastering, signal processing, automation, and digital audio workstations.

## 1.2. Overview

- Contexts and Tools
- Listening
- About this course

## 1.3. Aural Photography and Aural Sculpture

- Photography and Sculpture: two extremes of what we do when we record and produce audio
- A photograph: capture a “natural” space, time, and sound
- A sculpture: a synthetic re-working of other materials
- A wide range between these two extremes; how do we decide?

## 1.4. Aural Photography and Aural Sculpture Today

- This endeavor used be limited to a few
- Tools are more accurate
- Processors are more powerful
- The necessary technologies are inexpensive
- Relevant to all working in sonic crafts

## **1.5. The Training of an Audio Engineer**

- Listening and ear training
- Musical knowledge and performance experience
- Practical, hands-on experience with hardware and software
- Knowledge of historical and current trends
- Theoretical knowledge of sound, psychoacoustics, and electronics
- Experience working with changing and limited resources
- Willingness to experiment and innovate

## **1.6. Three Large-Scale Audio Production Processes**

- Tracking (recording, overdubs)
- Mixing (editing, processing, producing, composing)
- Mastering

## **1.7. Tools of Tracking**

- Space: acoustics, baffles, absorption, and reflection
- Transducers: microphones and speakers
- Pre-amps: the first stage of amplification, optimize level to the medium
- The recording medium: disc, tape, digital
- Monitoring: providing pre-recorded signals back to the performer

## **1.8. Tools of Mixing**

- The channel strip
  - Adjust level in relation to other channels
  - Adjust panning, or spatial position
  - Apply filters (EQ)

- Process dynamics (compression, limiting, gating, expansion)
- The mixer
  - Combine channels into groups (busses)
  - Route channels or busses for processing (reverb, delay)
  - Automate changes in channel or processor parameters

## **1.9. Tools of Mastering**

- Prepare and optimize mix for the distribution medium
- May use filters, dynamics processing, or speciality processors

## **1.10. Hardware Tools: MOSS**

- MOSS: The MObile Sound Studio for Teaching and Learning at MIT

- Any space is a recording space
- Support for 16 channel recording
- A variety of microphones and preamps
- A variety of monitoring tools

- All will be required to help move before and after class

### **1.11. Software Tools: DAW**

- The digital audio workstation: combines roles of mixing and processing into a single software unit
- Some add feature for MIDI production with virtual instruments and loop based music production
- All students are required to obtain a full-featured DAW immediately

Recommended DAW for Mac and Windows users is Ableton Live Intro, version 8 or better (\$99):

<http://www.ableton.com/live-intro>

- Acceptable alternatives include: ProTools, Logic (Pro or Express), Cubase, Reaper, Digital Performer, Sonar, or FL Studio
- Contact me if you anticipate a problem with this

### **1.12. Software Tools: Waveform Editor**

- The software tool for editing a single audio file
- Usually destructive editing with minimal (if any) mixing functionality
- Free waveform editor for Mac, Windows, and Linux users is Audacity:

<http://audacity.sourceforge.net/>

- Alternatives include: Peak, Adobe Audition, Wavelab
- Download, install, and test ASAP

### **1.13. The Most Important Tool**

- Your ears
- Listening like an audio engineer
- Hearing the production separate from the music

### **1.14. Mix Graphs: Basics**

- Structured listening and analysis of recordings

- Steps
  - Select a piece of music
  - Listen carefully to the music; using headphones is recommended
  - Isolate each audio source (may be more than one within a track)
  - For each audio source, evaluate fundamental attributes
- Report must be posted in the class Forum under the appropriate topic heading
- Students are encouraged to read and comment on others reports

### 1.15. Mix Graphs: Fundamental Attributes of a Mix

- Isolating each audio source
- Tracks, channels, and audio sources
  - A track is one or more channels bundled together for uniform audio processing (represented as a single unit in a DAW)
  - A channel is a single isolated audio stream (one simple or complex waveform)
  - An audio source is a distinct sound or timbre group (which may be captured with one or more channels)
- Perceived relative loudness can be measured between 0 and 1
- Perceived stereo position between -1 and 1
- Estimated frequency response between 20 and 20,000 Hz (more on this next class)

### 1.16. Mix Graph: Example 1

- Blackalicious: Aural Pleasure (audio)

•

Mix Graph 1

Artist: Blackalicious  
 Album: Blazing Arrow  
 Date: 2002  
 Song: Aural Pleasure



SOURCE	VOL	PAN	FQ (Hz)
Kick Drum	.8	0	80-100
Bass	.85	1	60-100
Snare/Clap	.85	-.2, .2	800-2000
Shaker	.7	.8	1000-2000
Tambourine	.85	-.4, .4	2000-4000
Electric Gtr	.75	-.6	800-6000
Horns	.7	-1, 0, 1	900-4000
Female Voice	.9	-1, 1	1000-6000
Male Voice	.95	0	600-2000

Notes: Snap and snare clap double each other. Exact composition of horn section not clear. Tambourine alternates between left and right.

## 1.17. Mix Graph: Example 2

- John Coltrane: Naima (audio)

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Mix Graph 2

Category: jazz or experimental instrumental ensembles

Artist: John Coltrane

Album: Giant Steps

Date: 1959

Song: Naima

SOURCE	VOL	PAN	FQ (Hz)
Sax	.95	-.7	1000-4000
Piano	.8	-.9	600-5000
Bass	.8	.7	60-120
Snare/Cymbals	.7	.9	1000-6000

Notes: All sources seem to be monophonic.

## 1.18. 21M.380: Objectives and Prerequisites

- Gain a critical understanding of, and hands-on experience with, the equipment and practices of modern recording techniques and audio production
- Develop practical and creative approaches to creating, processing, and mixing recordings and improvisation
- Understand the historical, aesthetic, and social contexts of audio recording
- No prerequisites

## 1.19. 21M.380: Course Meetings

- Two types of meetings

- Topic meetings: focused on material in readings, listening, and themes, combining lecture, discussion, demonstration, and listening
- Workshop meetings: hands-on projects, full recording sessions
- Lecture notes

## 1.20. 21M.380: Assignments: Reading

- One book:

Eargle, J. 2004. *The Microphone Book*. 2nd ed. Boston: Focal Press.

- Numerous carefully selected articles and chapters:

1. Dooley, W. L. and R. D. Streicher. 1982. "M-S Stereo: A Powerful Technique for Working in Stereo." *Journal of the Audio Engineering Society* 30(10): pp. 707-718.

2. Horning, S. S. 2002. "From Polka to Punk: Growth of an Independent Recording Studio, 1934-1977." In H. Braun, ed. *Music and Technology in the Twentieth Century*. Baltimore: The Johns Hopkins University Press, pp. 136-147.

3. Katz, B. 2007. "Equalization Techniques." *Mastering Audio: The Art and the Science*. 2nd ed. Burlington: Focal Press, pp. 103-112.

4. Katz, B. 2007. "How to Manipulate Dynamic Range for Fun and Profit." *Mastering Audio: The Art and the Science*. 2nd ed. Burlington: Focal Press, pp. 113-138.

5. Katz, M. 2004. "Aesthetics Out of Exigency: Violin Vibrato and the Phonograph." *Capturing Sound: How Technology Has Changed Music*. Berkeley: University of California Press, pp. 94-108.

6. Lazzarini, V. 2011. "Introduction to Digital Audio Signals." In R. Boulanger and V. Lazzarini, eds. *The Audio Programming Book*. Cambridge, Massachusetts: MIT Press, pp. 431-462.

7. Millard, A. 2002. "Tape Recording and Music Making." In H. Braun, ed. *Music and Technology in the Twentieth Century*. Baltimore: The Johns Hopkins University Press, pp. 158-167.

8. Nielsen, S. H. and T. Lund. 2003. "Overload in Signal Conversion." *AES 23rd International Conference*.
9. Streicher, R. and W. Dooley. 2003. "The Bidirectional Microphone: A Forgotten Patriarch." *Journal of the Audio Engineering Society* 51(3): pp. 211-225.
10. Streicher, R. D. and W. L. Dooley. 1985. "Basic Stereo Microphone Perspectives -- A Review." *Journal of the Audio Engineering Society* 33(7-8): pp. 548-556.

### **1.21. 21M.380: Assignments**

- Mix Graphs (3)
- Processing Reports (2)
- Mix Reports (2)
- Track Sheet Log
- Participation

### **1.22. 21M.380: Assignments: Submission**

- All assignments are submitted digitally via email attachment (or as Forum posts)
- All assignments are due at 11:59:59 PM on due date
- No late assignments will be accepted

### **1.23. 21M.380: Attendance**

- Mandatory and essential
- Always communicate with me about needs for excused absences
- More than one unexcused absence incurs a 3% grade reduction

### **1.24. 21M.380: Exams and Quizzes**

- Quizzes will be announced
- All short written answers
- Quizzes will be based on reading, listening, and course content

- No final exam

### **1.25. 21M.380: Grading**

- See distribution in syllabus
- Emphasis on projects; quizzes and participation are important

### **1.26. 21M.380: Additional Policies**

- Read entire syllabus
- Common courtesies
- Computers in class
- Academic integrity

### **1.27. 21M.380: Contact**

### **1.28. For Next Class**

- Download and read entire syllabus, begin reading in Eargle, begin Mix Graph 1

## Chapter 2. Meeting 2, Measures and Visualizations of Sounds and Signals

### 2.1. Announcements

- Be sure to completely read the syllabus
- Recording opportunities for small ensembles
- Due Wednesday, 15 February: Mix Graph 1
- Quiz next Tuesday (we meet Tuesday, not Monday next week) on material from this and the next class
- Audio examples today will make use of Pd-extended and Martingale

### 2.2. Reading: Eargle: A Short History of the Microphone

- How did the early microphones of Bell, Berliner, and Blake operate?
- In basic terms, how do the electrostatic and electrodynamic microphones developed in the 1920s operate?
- What was the “breakthrough” of the electret microphone?

### 2.3. Basic Measures: Time

- Measured in seconds
- 1 millisecond (ms) is equal to .001 ( $10^{-3}$ ) second
- Example: earLimits.pd
- 1 second is equal to 1000 milliseconds
- 1 microsecond ( $\mu\text{sec}$ ) is equal to .000001 ( $10^{-6}$ ) second, or .001 ms
- 1 second is equal to 1000000 microseconds

## **2.4. Basic Measures: Distance**

- Microphone positioning diagrams may use feet or meters
- 1 foot is .305 meter; 1 meter is 3.28 feet

## **2.5. Sound**

- Variations in pressure through a medium
  - Through air, water, solids
  - As a voltage, as a magnetic flux
- A disturbance in equilibrium
- Vibration: an oscillating disturbance in an elastic medium
- Oscillation offers a special class of sounds: periodic waves

## **2.6. Waves**

- A disturbance transmitted over time
  - Tides
  - Ripples
  - Some waves are periodic (and oscillate) others are non-periodic (random or noise) or a picture of both
- Transverse waves: a ripple in water or a string

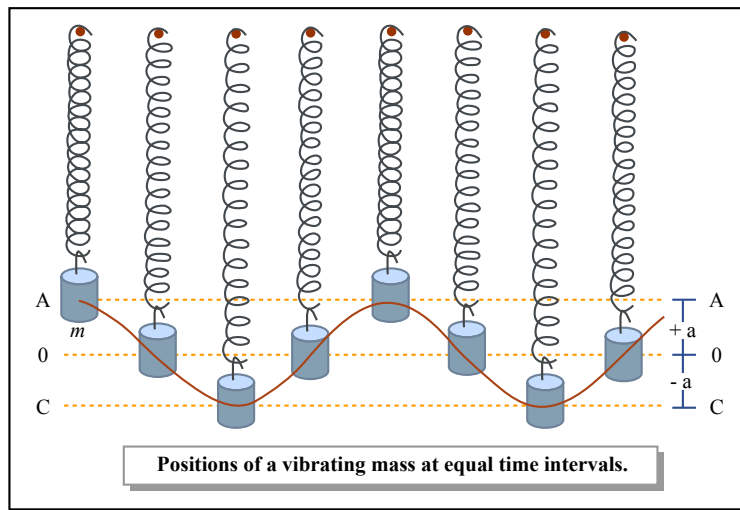


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Image: "Sinusoidal Pressure Waves." From *Sound for Music Technology: An Introduction*  
<http://openlearn.open.ac.uk/mod/resource/view.php?id=285732>. (c) The Open University.

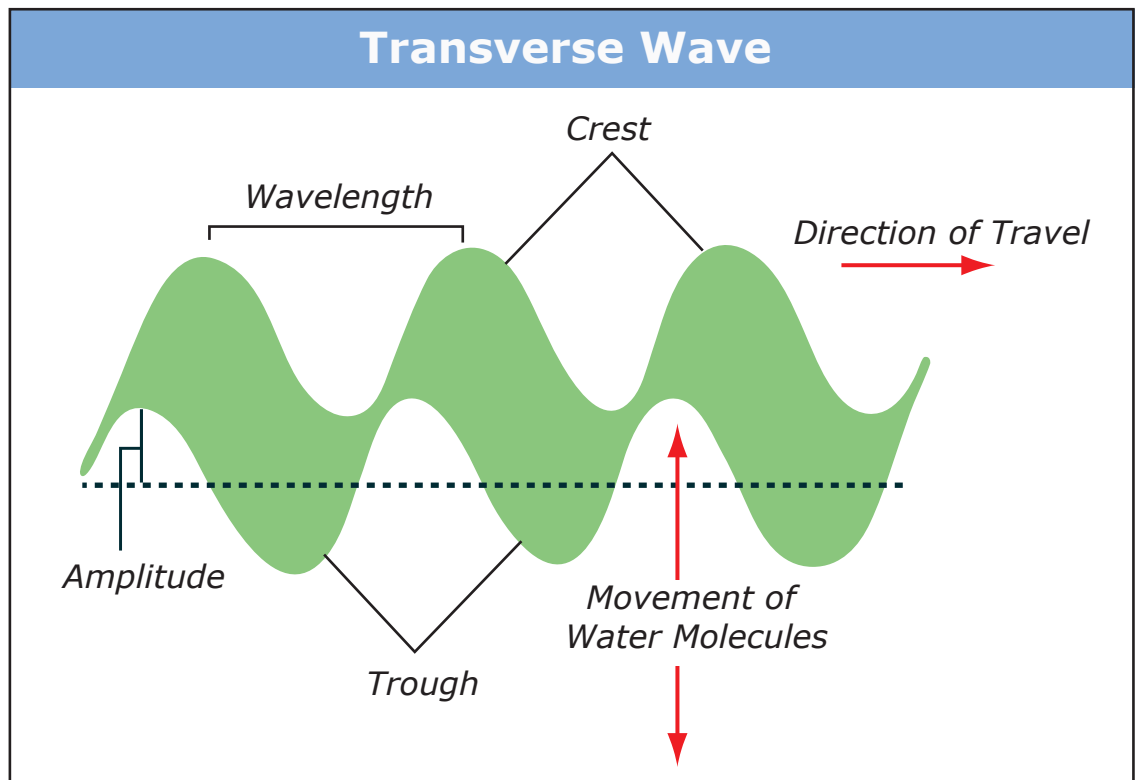


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- Longitudinal sound waves: disturbances in air pressure

# Longitudinal Wave

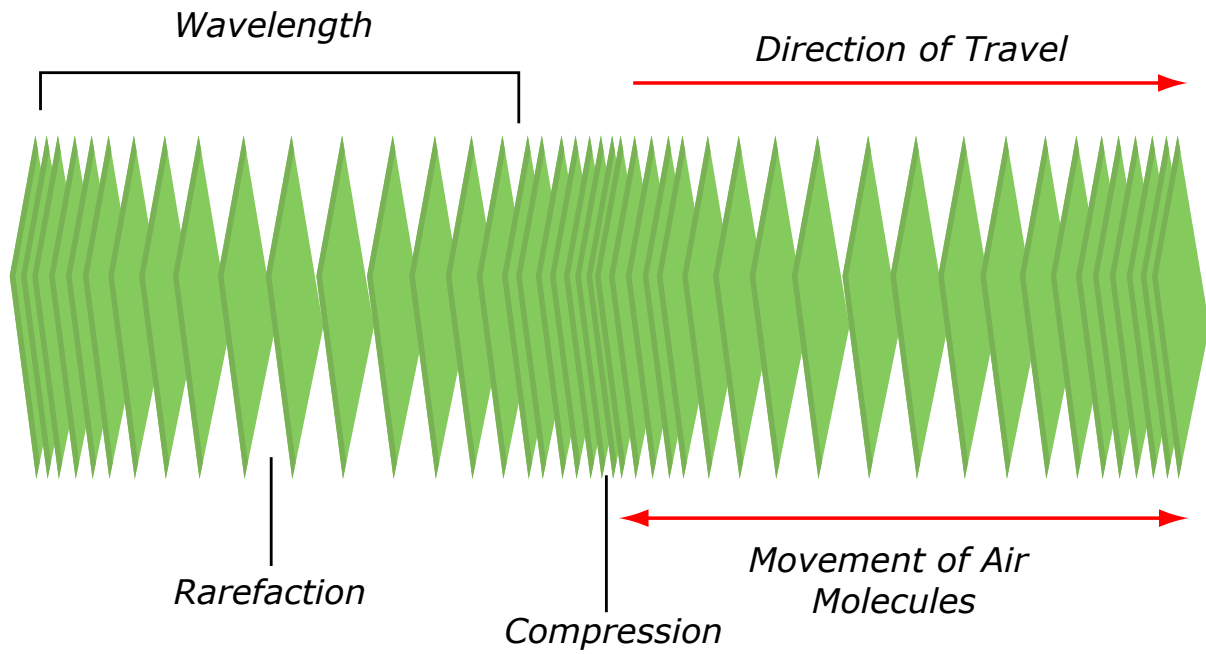


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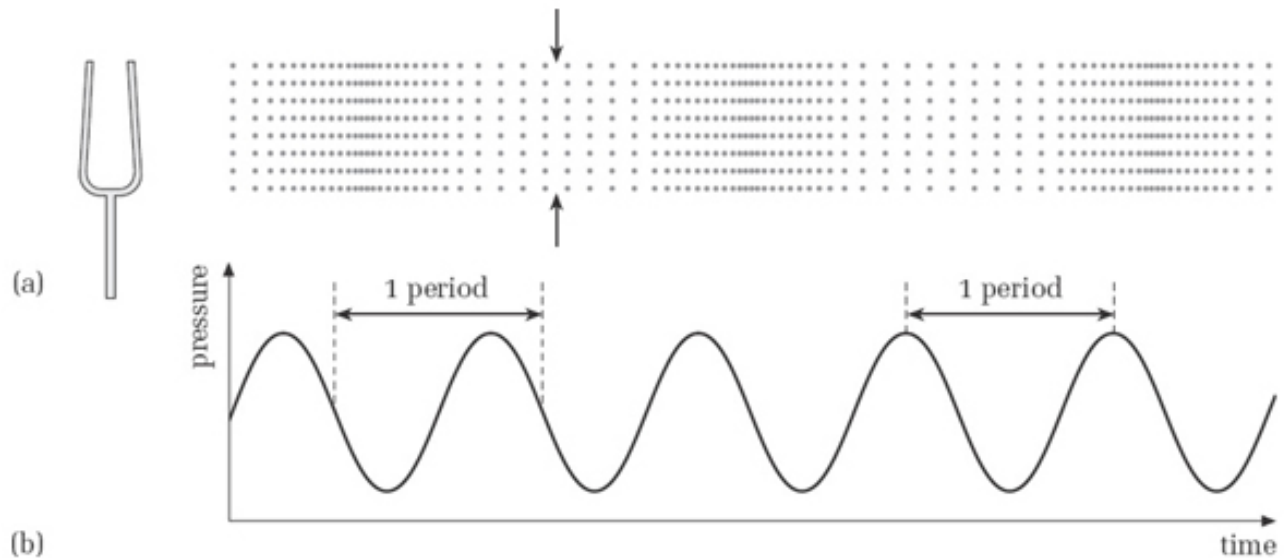


Image: "Sinusoidal Pressure Waves." From *Sound for Music Technology: An Introduction* <http://openlearn.open.ac.uk/mod/resource/view.php?id=285732>. (c) The Open University

## 2.7. The Speed of Sound

- The speed of the sound wave depends on the medium and its temperature
  - Air: 1130 feet per second or 331 meters per second
  - Air: 1.13 feet per millisecond, or .885 ms per foot
  - Sea water: 1533 meters per second
  - Aluminum: 5100 meters per second
  - Diamond: 12000 meters per second
- Always remember how many ms per foot: .885 ms per foot

## 2.8. Natural Oscillation

- Oscillation is the natural motion of many physical objects disturbed from equilibrium

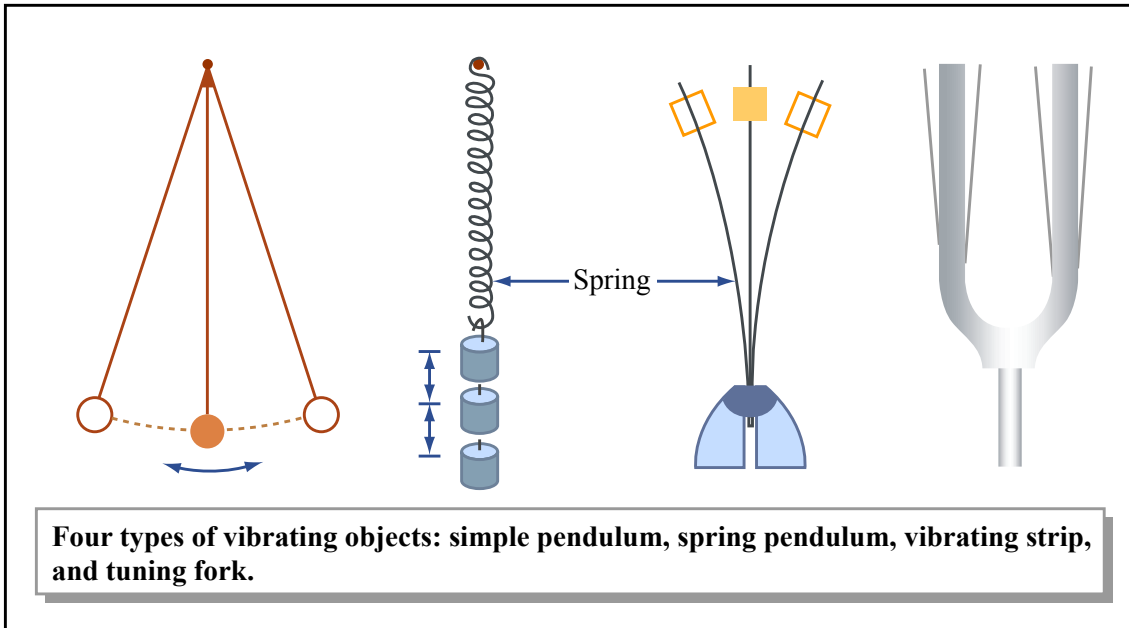


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- Oscillation is a back and forth motion (up and down) over time
  - Pendulums (Swings)
  - Strings
- A natural point of oscillation in an object is a resonance
- Perfect oscillations are impossible in nature
- Noise is everywhere
  - Damping, friction, resistance
  - Mechanical and thermal noise

## 2.9. Perfect Oscillation

- A sine wave is a perfect oscillation
  - Named a sine to describe its shape: a circular motion extended in time

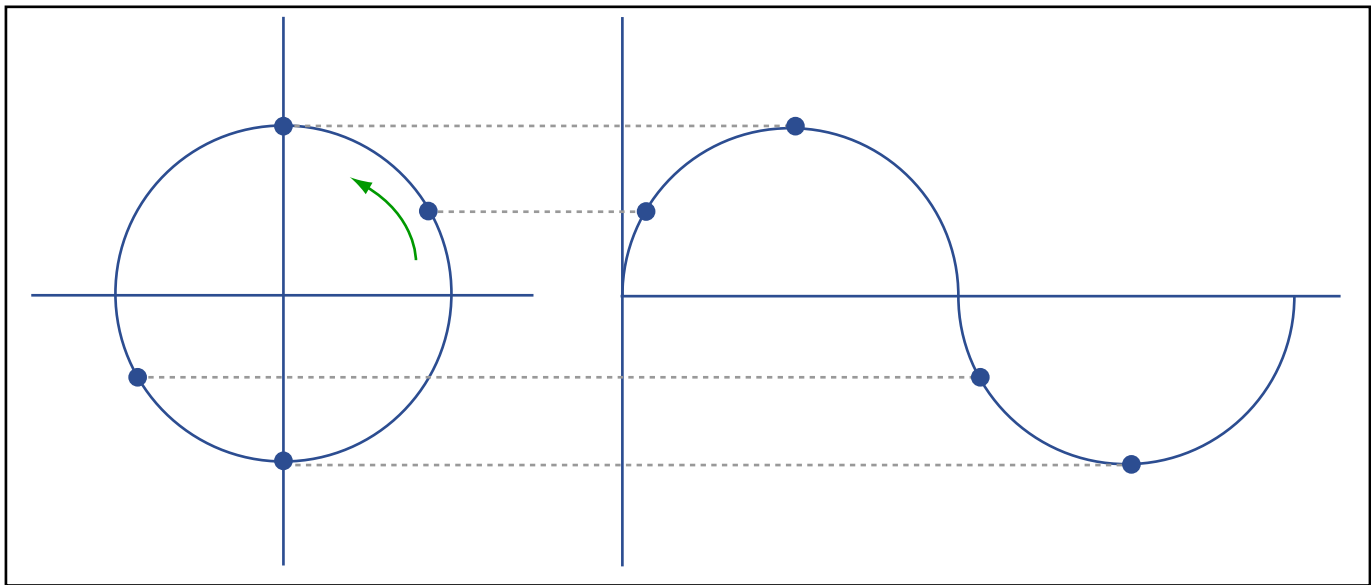


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- No damping or resistance
- No noise
- Machine-made: there are no sine waves in nature
- Example: signalWaveforms.pd
- There are other commonly used perfect oscillations with different shapes
  - Square (Rectangle) wave
  - Triangle wave
  - Sawtooth wave
  - Example: signalWaveforms.pd
  - Complex harmonic waveforms found in nature
- The sine provides a basic building block of sound
  - It is easy to generate mechanically and mathematically
  - It resembles simple harmonic motion: natural resonances in physical objects
  - It sounds as a single isolated tone
  - It provides frequency reference

## 2.10. Measuring a Sine Wave: Frequency

- How often it oscillates: its frequency

- Measured in Cycles Per Second (CPS) or Hertz
- Each cycle is one period, or the distance from crest to crest
- An audible sine wave produces the perception of a single frequency
- Frequency is very similar to pitch, but not the same
- Example: 20 Hz sine wave: 1 period lasts 50 msec (1 cycle / 20 cycle/s)
- Example: 200 Hz sine wave: 1 period lasts 5 ms
- Example: 2000 Hz sine wave: 1 period lasts .5 ms, or 500  $\mu$ sec
- Example: 20000 Hz sine wave: 1 period lasts .05 ms, or 50  $\mu$ sec
- Example: signalWaveforms.pd

### **2.11. Measuring a Sine Wave: Pitch**

- Pitch relates to how the ear interprets frequency
- Pitches are commonly given names: A#, B-, etc
- 12 divisions per octave, each repeating at the octave, is most common
- Register is given with octave specifications as integers following the pitch name: A6, C2
- Middle C on the piano is C4; the range of the piano is from A0 to C8
- MIDI pitch numbers can be used to describe pitch: C4 is 60; C5 is 72; C3 is 48, etc.

### **2.12. Measuring a Sine Wave: Wavelength**

- Distance between crests: wavelength
  - Measured in meters or feet
  - the speed of sound (m/s) divided by the frequency (cycle/s)
- Wavelength considerations are useful in considering how different frequencies interact with spaces and microphones
- Example: Kick drum @ 60 Hz: 18 feet in air ( $331 / 60 \text{ hz} == 5.5 \text{ m}$ )
- Example: Cymbal sizzle @ 16 kHz: .81 inches ( $331 / 16000 \text{ hz} == .02 \text{ m}$ )

## 2.13. Measuring a Sine Wave: Amplitude

- How large are the oscillations: its amplitude
- Intensity: an averaged measure over time
- Acoustic sound: a measure of pressure
- Numerous types of measurements
  - Acoustical power (intensity) as force over area: watts, dynes/cm<sup>2</sup>, pascals
  - In relation to a minimum and a maximum: 0% to 100%, or 0.0 to 1.0
  - In relation to some defined measure: Bels, decibels (dB)
- Decibels: condense a wide range of linear amplitude values into a smaller range
  - A logarithmic measure in relation to amplitude
  - A reference value defines 0 dB
  - $\text{dB} = 20 * \log_{10} * \text{amplitude}$
  - -3 dB change is a factor of .707 amplitude
  - 3 dB change is a good general unit of change
  - -6 dB change is a factor of .5 amplitude
  - Doubling a signal generally results in a 6 dB change
  - Example: ampDbDemo
  - -20 dB is .1 amplitude

## 2.14. Measuring a Sine Wave: Decibels

- Numerous types of dB based on different reference values
- Sound Pressure Levels (dB SPL)
  - Pressure of air measured in reference to human ears
  - 0 dB SPL is equal to .0002 dynes/cm<sup>2</sup>
  - 0 dB SPL is threshold of hearing; 120-130 dB SPL is threshold of pain

- average conversation: 60 dB SPL
- pin-drop: 10 dB SPL
- jet engine: 150 dB SPL
- Visual scale

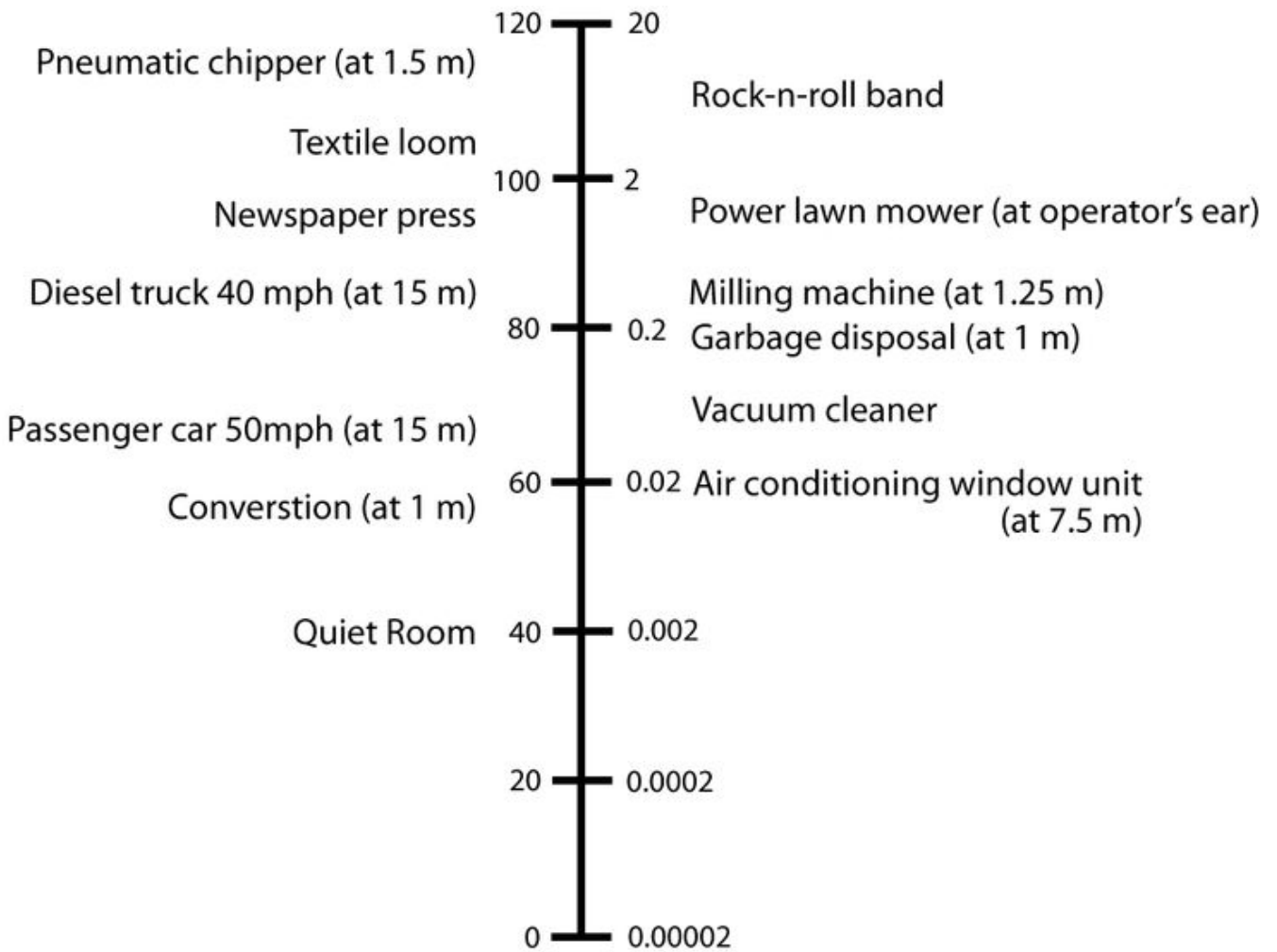


Image: "Sound Pressure Level (SPL) and Sound Pressure (Pa)." From *Principles of Industrial Hygiene*  
 Available at: <http://ocw.jhsph.edu>. Copyright © Johns Hopkins Bloomberg School of Public Health.

- Voltages: dBV, dBu
  - 0 dBV is equal to 1 Volt
  - 0 dBu (or dBv) is equal to .775 Volt
  - Range is generally from -infinity to +20 dBu
- Digital Bits: dBFS (6.0206 dB per bit)
- Amplitude is similar to loudness, but not the same
- A range of amplitudes is called a Dynamic Range

## 2.15. Measuring a Sine Wave: Position

- Phase: relative position of the waveform in its period
- Measured in degrees (360 degrees as a complete cycle) or measured within the unit interval (0 to 1)
- Requires reference to a fixed point or another wave
- 180 degrees is one half-cycle out of phase
- Flipping the phase is the same as multiplying a signal times -1
- Combinations of in-phase signals results in amplitude boosts
- Combinations of out-of-phase signals results in interference or cancellation

Example: phase.pd

## 2.16. Signals Store Simultaneous Information

- Waves can store multiple signals at multiple frequencies in one channel
- Waves can be added (mixed together) to result in more complex waves
- Sometimes these combined waves can be later decomposed into simple waves
- A single wave can store a tremendous amount of complexity

## 2.17. Timbre

- All sounds in nature are more complex than a sine wave (pure frequency)
- Many physical objects (strings, air-columns) have multiple points of resonance

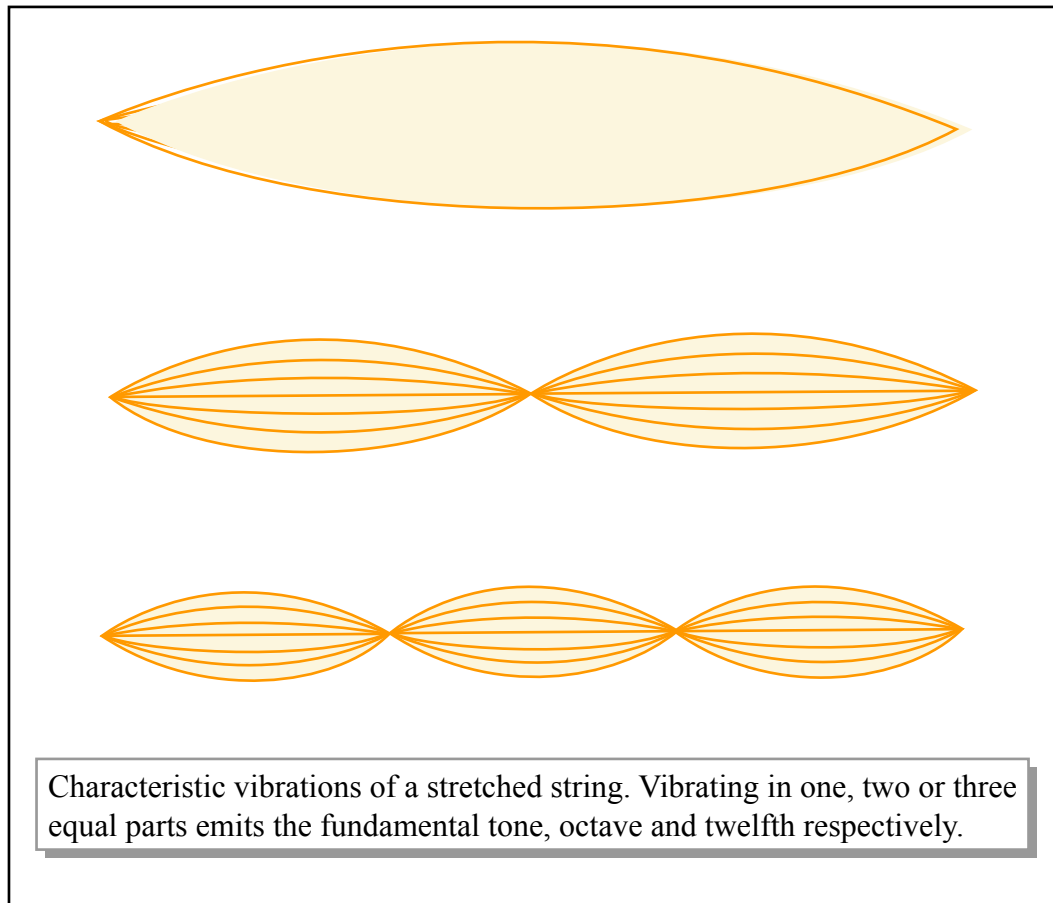


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- The difference in the sound between two instruments has to do with which resonances are prominent
- The lowest resonance is called the fundamental, or the first harmonic ( $f_0$ )
- Higher resonances are called harmonics, partials, or overtones
- Timbre (tone color) refers to the distinctions in sound due to these resonances

## 2.18. Harmonic Spectra

- Some objects resonate in whole-number multiples of the fundamental frequency
- These ratio-specific values are called harmonics



- Example: signalAddition.pd
- Arrangements of common harmonics produce common non-sinusoidal periodic waveforms

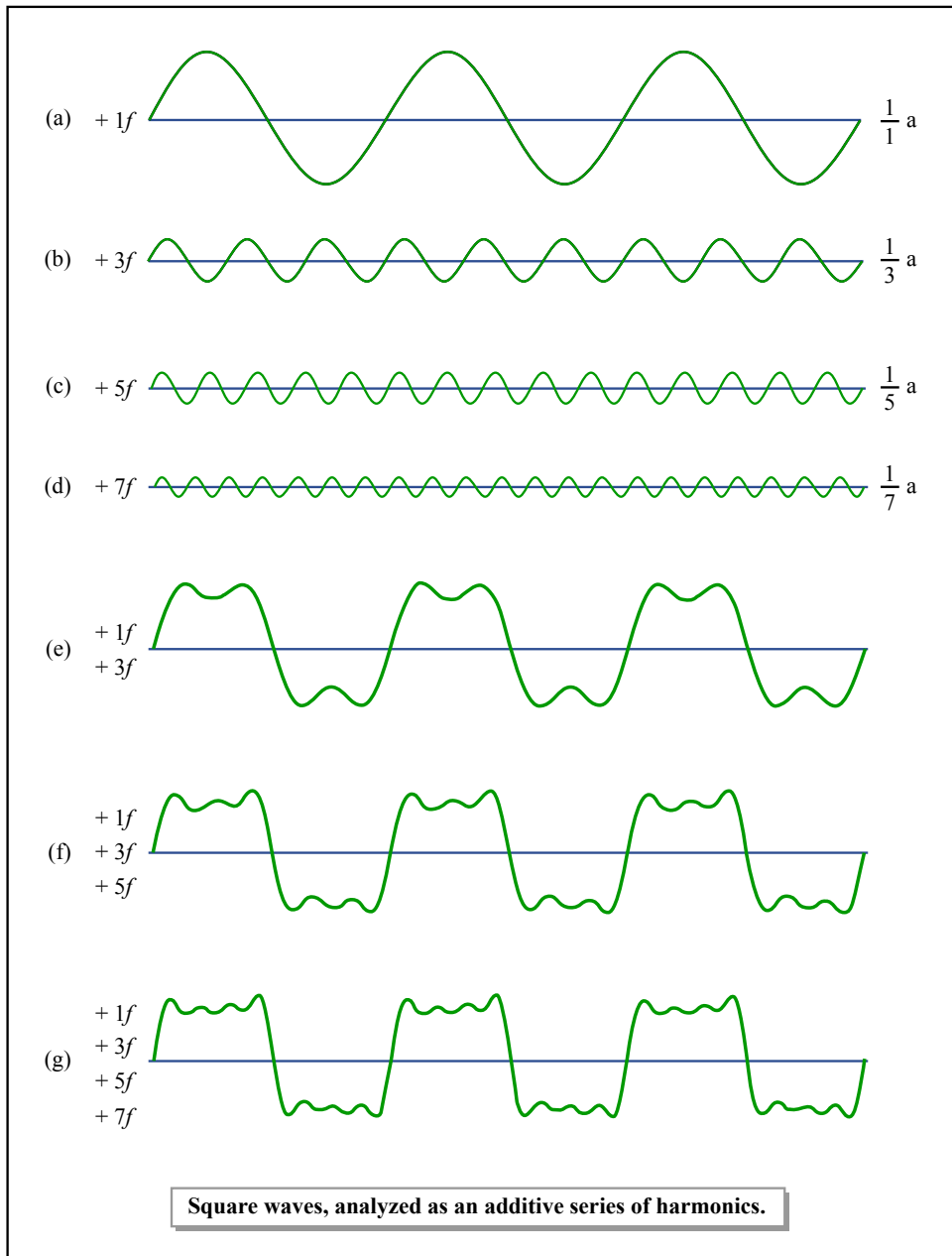


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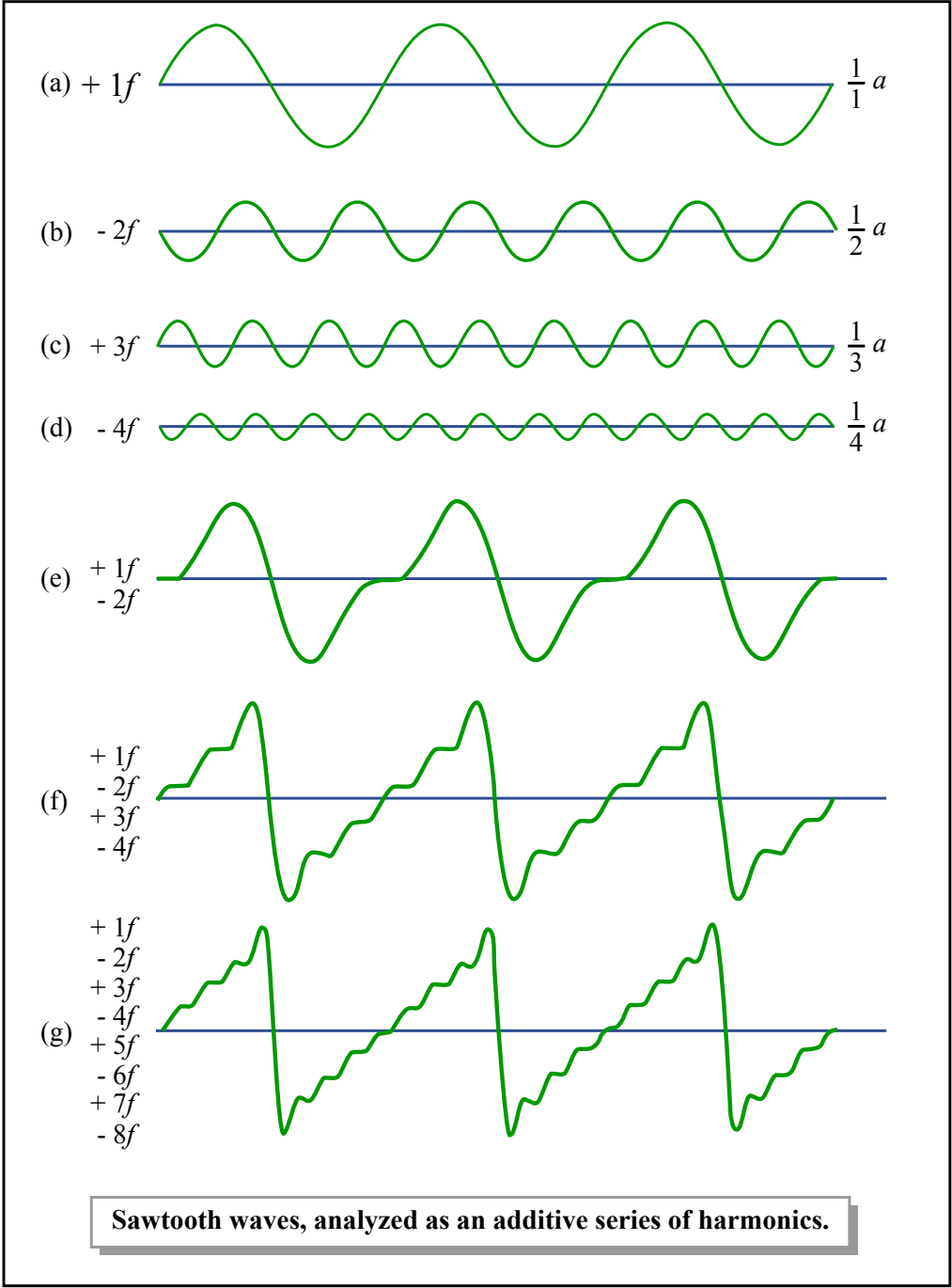


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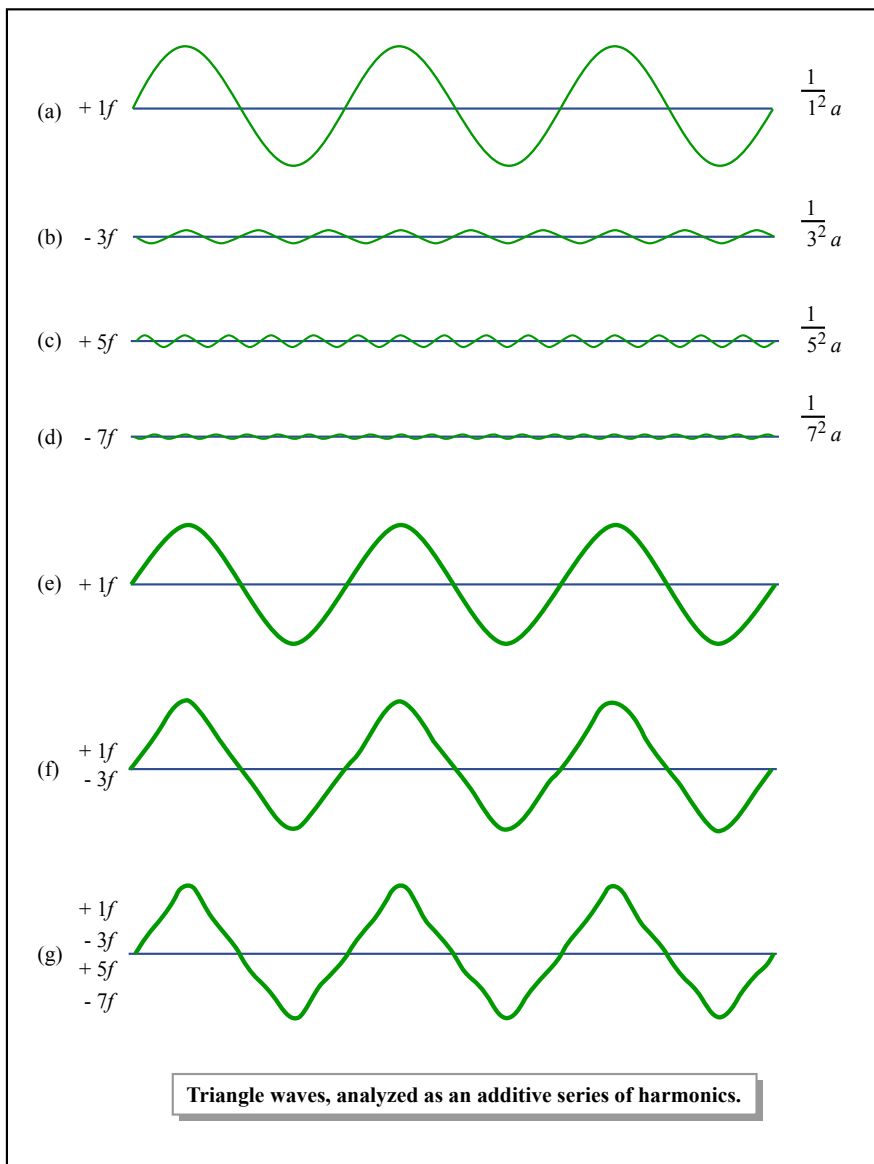


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- Saw: all harmonics w amplitude decreasing by inverse of harmonic number
- Square: odd harmonics with amplitude decreasing by inverse of harmonic number
- Triangle: odd harmonics with amplitude decreasing by inverse of square of harmonic
- Example: sumOfSines.pd

## 2.19. Inharmonic Spectra

- Some objects resonate without a harmonic relation to the fundamental
- Called overtones or partials
- Example: signalAddition.pd

## 2.20. The Duality of Waveforms

- We can look at a waveform to see changes in amplitude over time
- We can look at a spectral analysis and see the amplitude of frequency components (timbre) during a window of time

## 2.21. The Time Domain

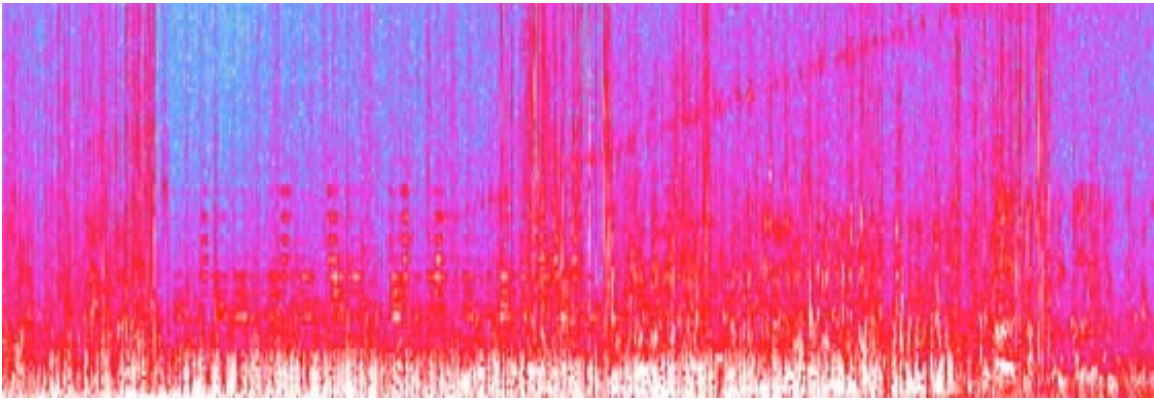
- Graph of displacement over time
- Draw amplitude change (y-axis) over time (x-axis)
- Illustrates the movement of a speaker, microphone, or air pressure
- Digital sound files, DAW waveforms
- Example: adding a track to Ableton Live (drumKitKickMic.aiff)
- Example: opening a file in Audacity (drumKitKickMic.aiff)

## 2.22. The Frequency Domain

- Graph of frequency amplitudes within a single time window
- Draw amplitude (y-axis) over frequency (x-axis)
- Illustrates what the ear hears at a given moment
- Requires mathematical decoding: Fourier Transform
- Reveals the spectrum (timbre) of a sound
- Example: viewing spectrum in Audacity (drumKitKickMic.aiff)
- Example: use of “Spectrum” Live Device (plugin) in Ableton Live (drumKitKickMic.aiff)

## 2.23. Combining Amplitude and Frequency Domains in Three Dimensions

- Two ways
  - Graph of frequency (x-axis), amplitude (color), and time (y-axis)



- Graph of frequency (x-axis), amplitude (y-axis), and time (z-axis)

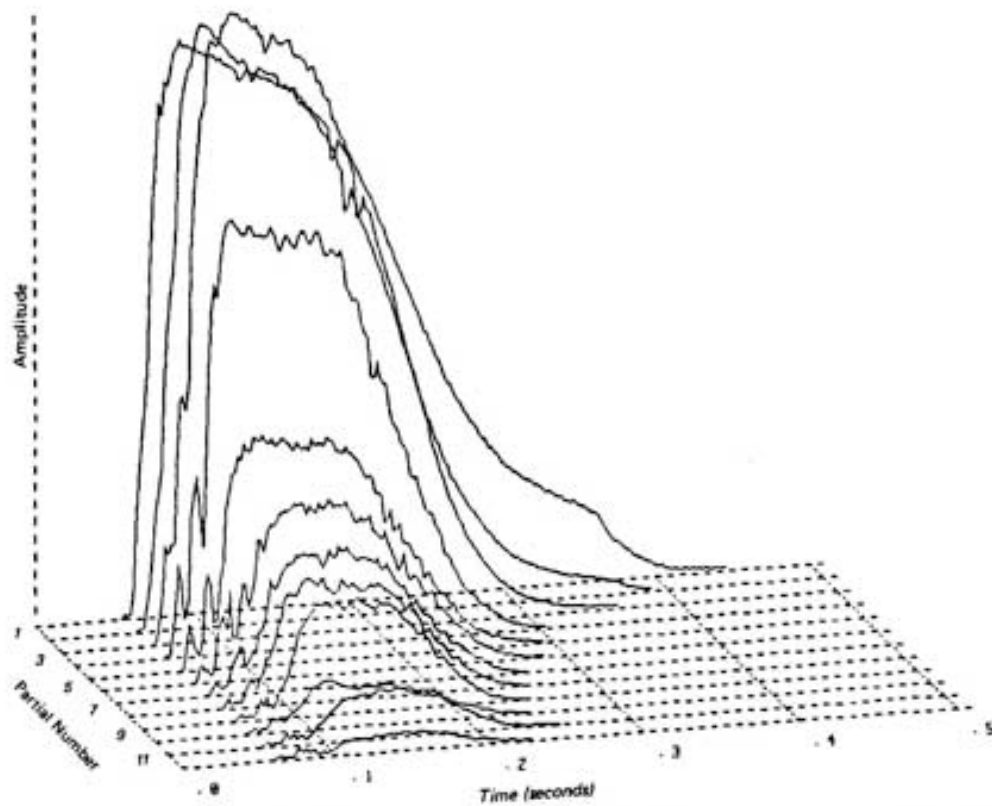


FIGURE 2.30 The amplitude progression of the partials of a trumpet tone as analyzed by Grey and Moorer. (Reprinted with permission of Computer Music Journal.)

Source: Moorer, J., J. Grey, and J. Strawn. "Lexicon of Analyzed Tones (Part 3: The Trumpet)." *Computer Music Journal* 2, no. 2 (1978): 23-31. © ownership uncertain (but not MIT Press). All rights reserved. This content is excluded from our Creative Commons license. For more information, see <http://ocw.mit.edu/fairuse>.

- Sometimes called a spectrogram (or sonogram)
- Closest representation to our experience of sound

- Not perfect for technical and psychoacoustic reasons

## **Chapter 3. Meeting 3, Psychoacoustics, Hearing, and Reflections**

### **3.1. Announcements**

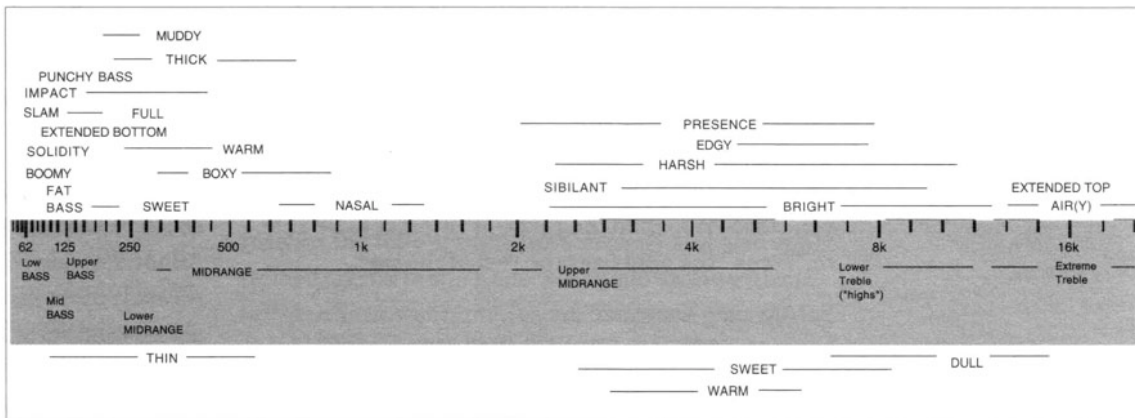
- Need schlep crew for Tuesday (and other days)
- Due Today, 15 February: Mix Graph 1
- Quiz next Tuesday (we meet Tuesday, not Monday next week) on material from this and the next class

### **3.2. Review**

- What is sound?
- How long does it take sound to travel a foot?
- Where can we find sine waves in nature?
- How big is a 60 Hz wave?
- Doubling a signal results in a change of how many dB?
- What are the differences between dB SPL and dBu?
- What is timbre?
- How can we create a saw wave?
- What are inharmonic spectra?
- How do we graph the time domain and the frequency domain?

### **3.3. Qualitative Descriptions of Frequency**

- Talking about sound is an imperfect art
- Descriptive frequency terms



*Subjective Terms we use to describe Excess or Deficiency of the various Frequency Ranges.*

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 Source: Katz, B. *Mastering Audio: The Art and the Science*. 2nd ed. Focal Press, 2007.

### 3.4. Basic DAW Operations and Viewing The Spectrum

- Track orientation and creating tracks
- Adding audio processors
- Setting loop points

### 3.5. Sine and Noise in the Frequency Domain

- A sine produces a single frequency in the frequency domain
- White noise is represented as all frequencies in the frequency domain
- Example: signalWaveforms.pd

### 3.6. Timbre

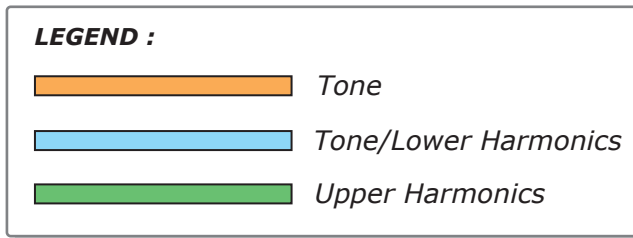
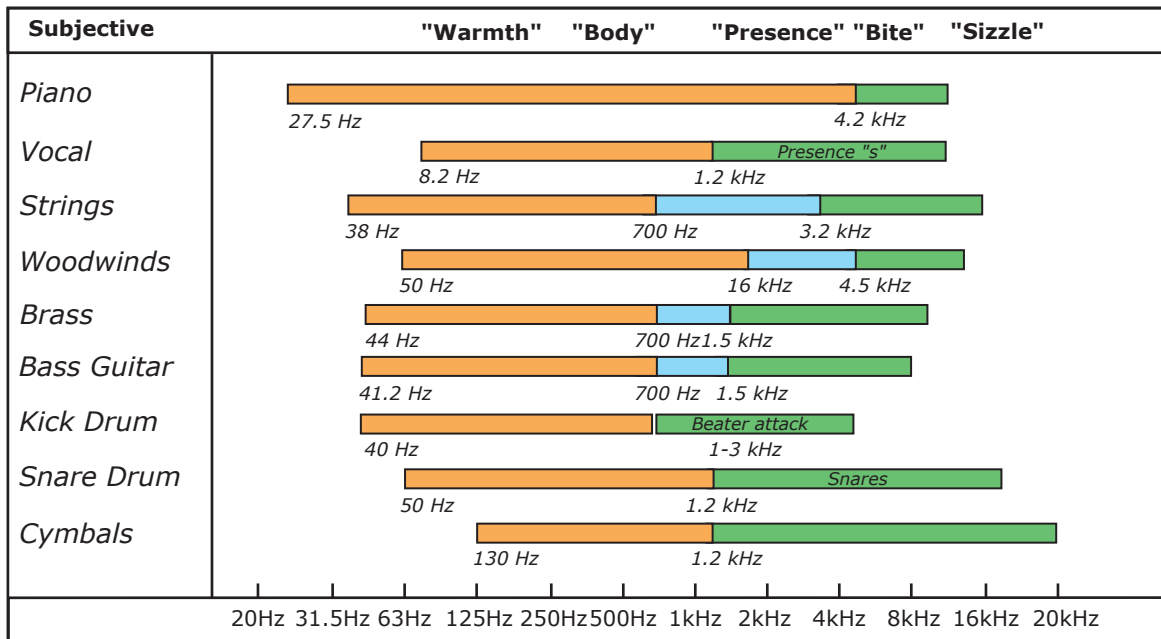
- We hear in the frequency domain
- Our ears are designed to distinguish sounds based on timbre
- We must study the frequency (timbral) range of sound sources



## Frequency Ranges of Musical Instruments and Voices

Instrument	Fundamentals	Harmonics
Flute	261-2349 Hz	3-8 kHz
Oboe	261-1568 Hz	2-12 kHz
Clarinet	165-1568 Hz	2-10 kHz
Bassoon	62-587 Hz	1-7 kHz
Trumpet	165-988 Hz	1-7.5 kHz
French Horn	87-880 Hz	1-6 kHz
Trombone	73-587 Hz	1-7.5 kHz
Tuba	49-587 Hz	1-4 kHz
Snare Drum	100-200 Hz	1-20 kHz
Kick Drum	30-147 Hz	1-6 kHz
Cymbals	300-587 Hz	1-15 kHz
Violin	196-3136 Hz	4-15 kHz
Viola	131-1175 Hz	2-8.5 kHz
Cello	65-698 Hz	1-6.5 kHz
Acoustic Bass	41-294 Hz	700 Hz-5 kHz
Electric Bass	41-294 Hz	700 Hz-7 kHz
Acoustic Guitar	82-988 Hz	1500 Hz-15 kHz
Electric Guitar	82-1319 Hz	1-15 kHz (direct)
Elec. Guitar Amp	82-1319 Hz	1-4 kHz
Piano	28-4196 Hz	5-8 kHz
Bass (Voice)	87-392 Hz	1-12 kHz
Tenor (Voice)	131-494 Hz	1-12 kHz
Alto (Voice)	175-698 Hz	2-12 kHz
Soprano (Voice)	247-1175 Hz	2-12 kHz

Image by MIT OpenCourseWare.



Frequency ranges of instruments, highlighting fundamental tones and harmonics, and how those frequencies contribute to an instrument's subjective character.

Image by MIT OpenCourseWare.

### 3.7. How the Ear Works: Components

- The components of the ear

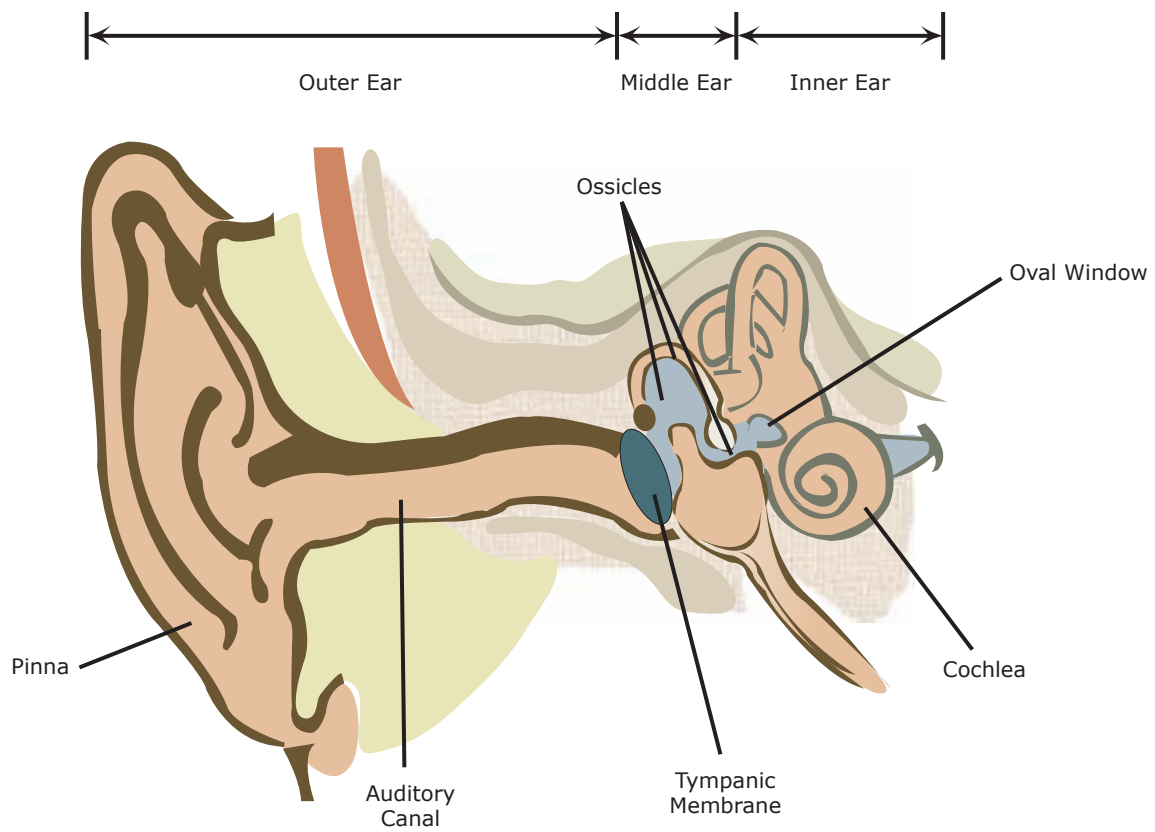


Image by MIT OpenCourseWare. Bear, Mark F., Barry W. Connors, and Michael A. Paradiso. Figure 11.3 in *Neuroscience: Exploring the Brain*. 2nd ed. Baltimore, Md. : Lippincott Williams & Wilkins, 2001. ISBN: 0683305964.

### 3.8. How the Ear Works: The Pathway of Sound

- Sound is transduced from air to skin (tympanic membrane), from skin to bone (ossicles), from bone to skin (oval window), from skin to fluid (perilymph), from fluid to hair (basilar membrane)

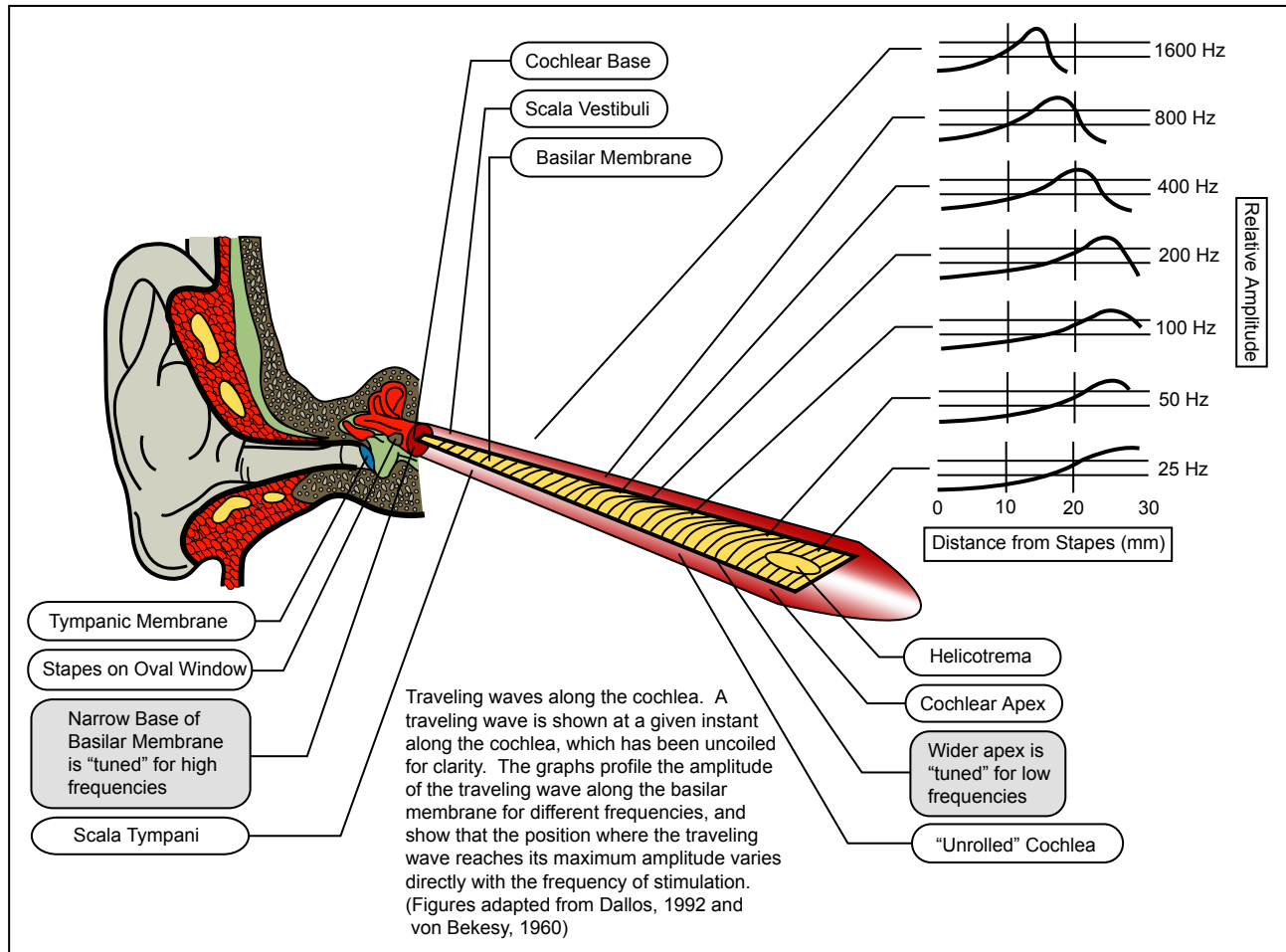


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### 3.9. How the Ear Works: The Cochlea

- The basilar membrane gets more narrow and more thin from base to tip
- Lower frequencies resonate near the tip (least stiff); higher frequencies resonate near the base (most stiff, near the oval window)
- Basilar membrane resonates with component frequencies in the sound
- 20,000 hair cells on the basilar membrane

- The cochlea performs spectral analysis with hair

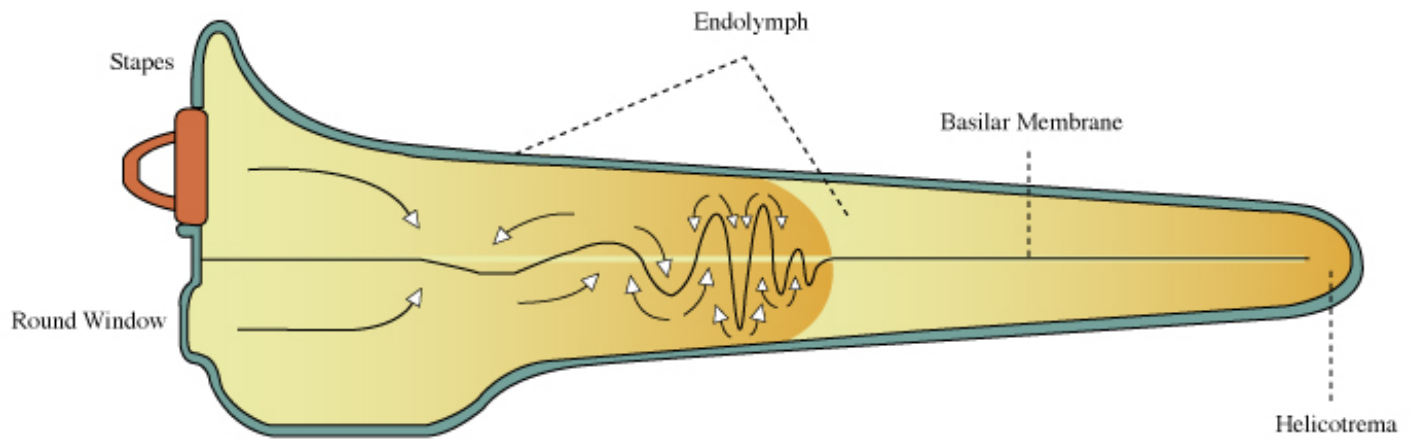


Figure by MIT OCW. After figure 11.9 in *Neuroscience: Exploring the Brain* Mark F. Bear, Barry W. Connors, Michael A. Paradiso. 2nd ed. Baltimore, Md.: Lippincott Williams & Wilkins, 2001. ISBN: 0683305964.

### **3.10. Limits of the Ear**

- Time: 30 milliseconds

Example: earLimits.pd

- Frequencies: 20 to 20,000 Hertz (about 10 octaves)

Example: earLimits.pd

- Amplitudes: from 0 to 120 dB SPL, or 120 dB of dynamic range

### **3.11. Our Ear is Biased**

- Amplitude (dB) is not the same thing as loudness (phons)
- Loudness is frequency dependent
- Fletcher-Munson (Robinson and Dadson/ISO 226:2003) equal loudness curves

# Fletcher-Munson Curves

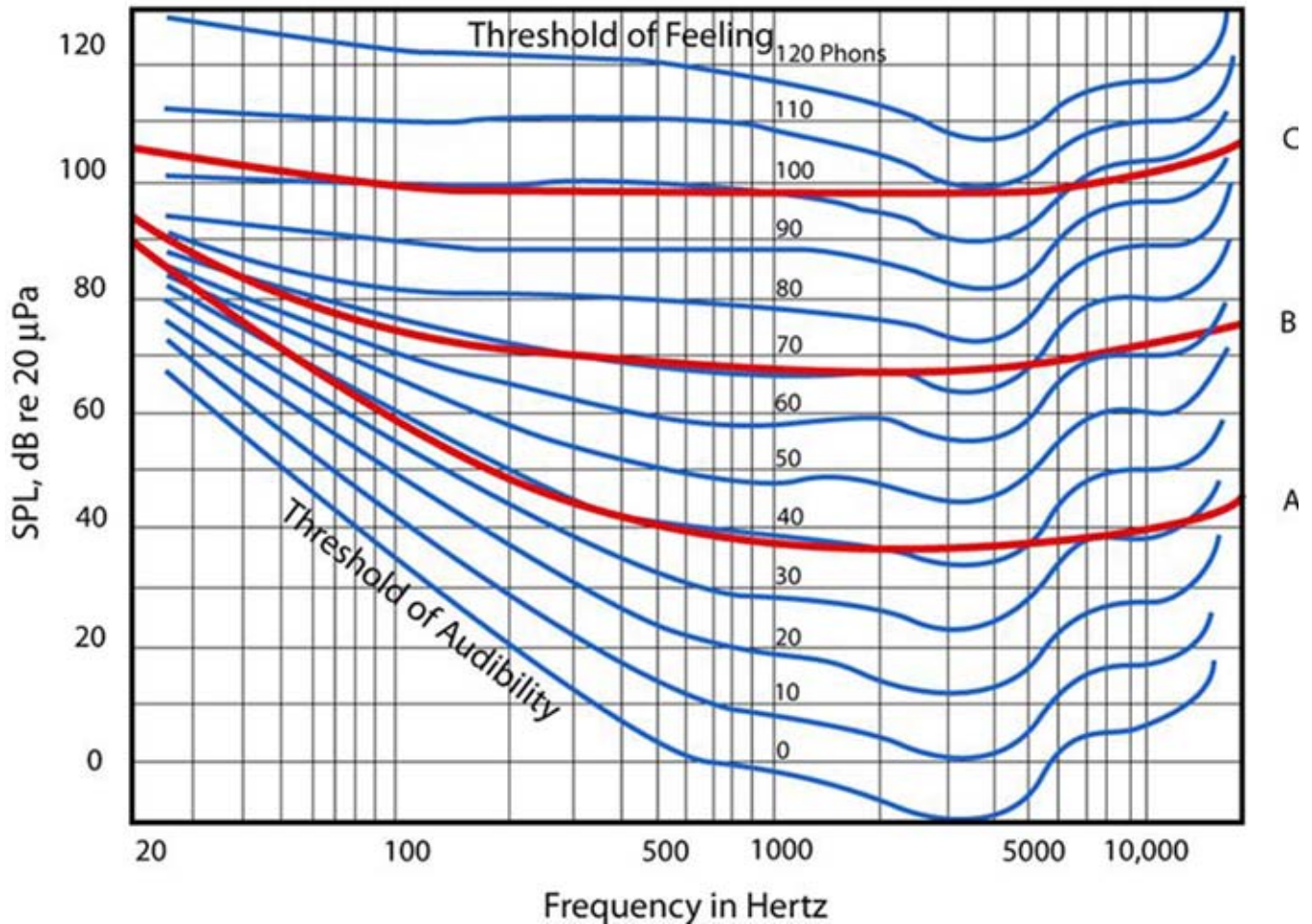
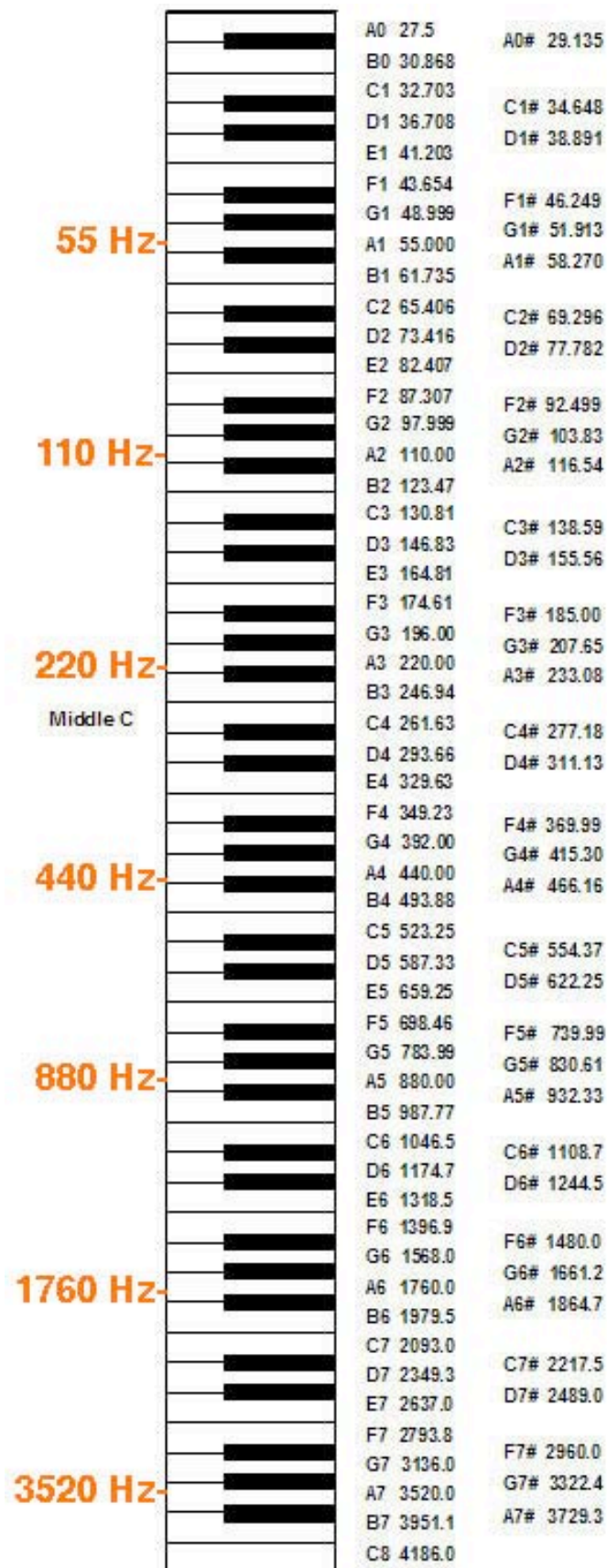


Image: "Fletcher-Munson Curves" from *Principles of Industrial Hygiene*. Available at: <http://ocw.jhsph.edu>. License CC BY-NC-SA, © Johns Hopkins Bloomberg School of Public Health.

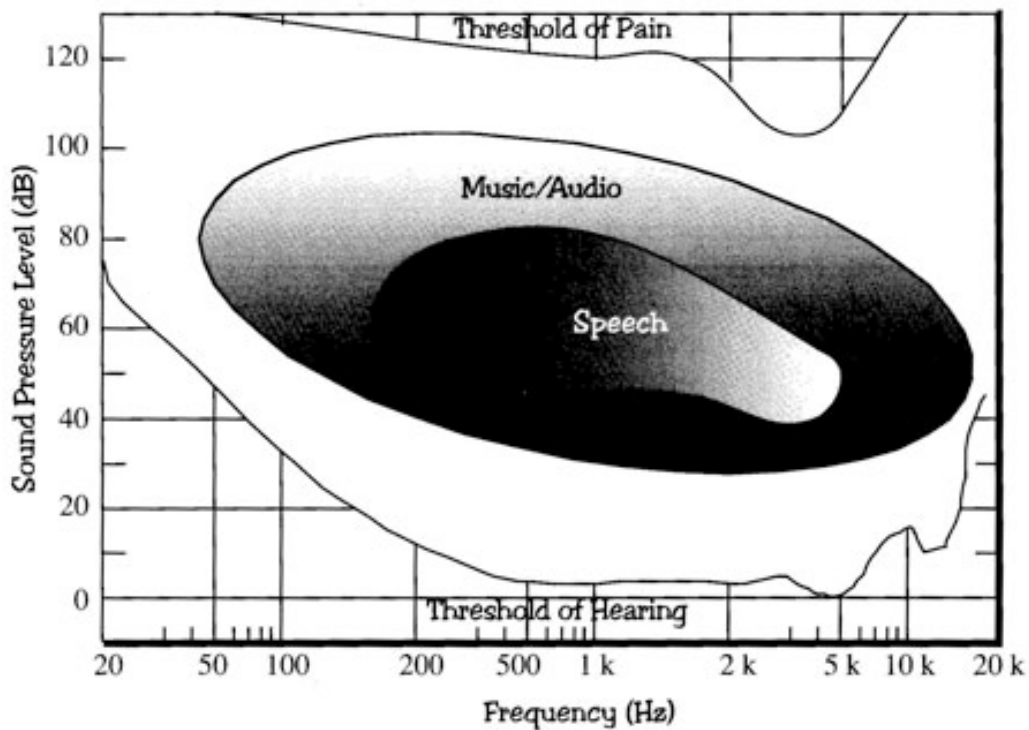
## 3.12. Our Ear Hears Logarithmically: Pitch

- Octave: an equal unit of perceived pitch (not frequency)
- Octaves: a 2:1 ratio of frequencies
- A change from 55 to 110 Hz (a difference of 55 Hz) sounds the same to our ear as a change from 1760 to 3520 Hz (a difference of 1760 Hz)



Courtesy of Tom Irvine. Used with permission.





*Fig. 11.1. A more detailed picture of our audio window. Most music does not exceed a range of level of about 75 dB from softest to loudest. Similarly, most recorded music does not contain much information above 18 kHz.*

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- A 1 Hz change from 70 to 71 Hz is more perceptually much more relevant than a 1 Hz change from 5000 to 50001 Hz
- Example: earLogFrequency.pd
- Some frequency displays are linear, others are logarithmic
- Example: Spectrum in Live: Scale X: Line, Log, ST
- High frequencies are always more accurately displayed

### 3.13. Our Ear Hears Logarithmically: Amplitude

- The ear can handle a range of pressure from .00002 to 1000000 pascals
- Example: earLogAmp.pd
- dB is a logarithmic measure: adding 6 dB doubles the audio power

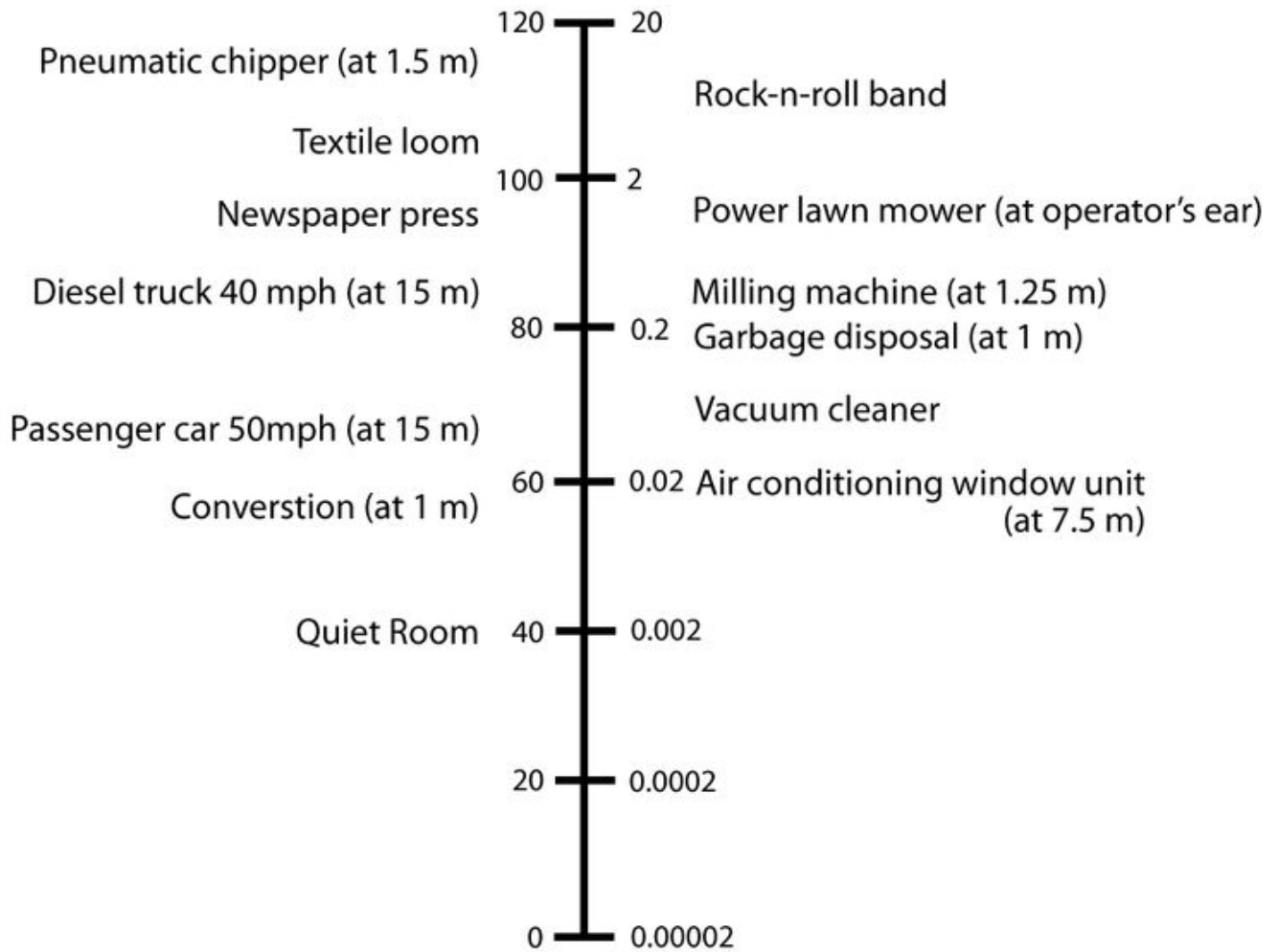


Image: "Sound Pressure Level (SPL) and Sound Pressure (Pa)." from *Principles of Industrial Hygiene*. Available at: <http://ocw.jhsph.edu>. Copyright © Johns Hopkins Bloomberg School of Public Health.

- dB is not the same as perceived loudness (the frequencies matter)

### 3.14. The Limits of Pitch Perception

- Different for different people

- Only relevant on a pitch/logarithmic scale, not a frequency scale
- The smallest conventional unit of pitch change (just noticeable difference [JND]) is 1 cent, or 1/100th of a halfstep, or 1/1200th of an octave
- Most people can probably hear 10 cent pitch changes

Example: jndPitch.pd

- 1 Hz does not always have the same perceptual meaning

### **3.15. The Limits of Amplitude Perception**

- Just noticeable difference (JND) is generally around 1 dB

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See Fig. 10.5 in Thompson, D. M. *Understanding Audio*.  
Hal Leonard Corp., 2005.

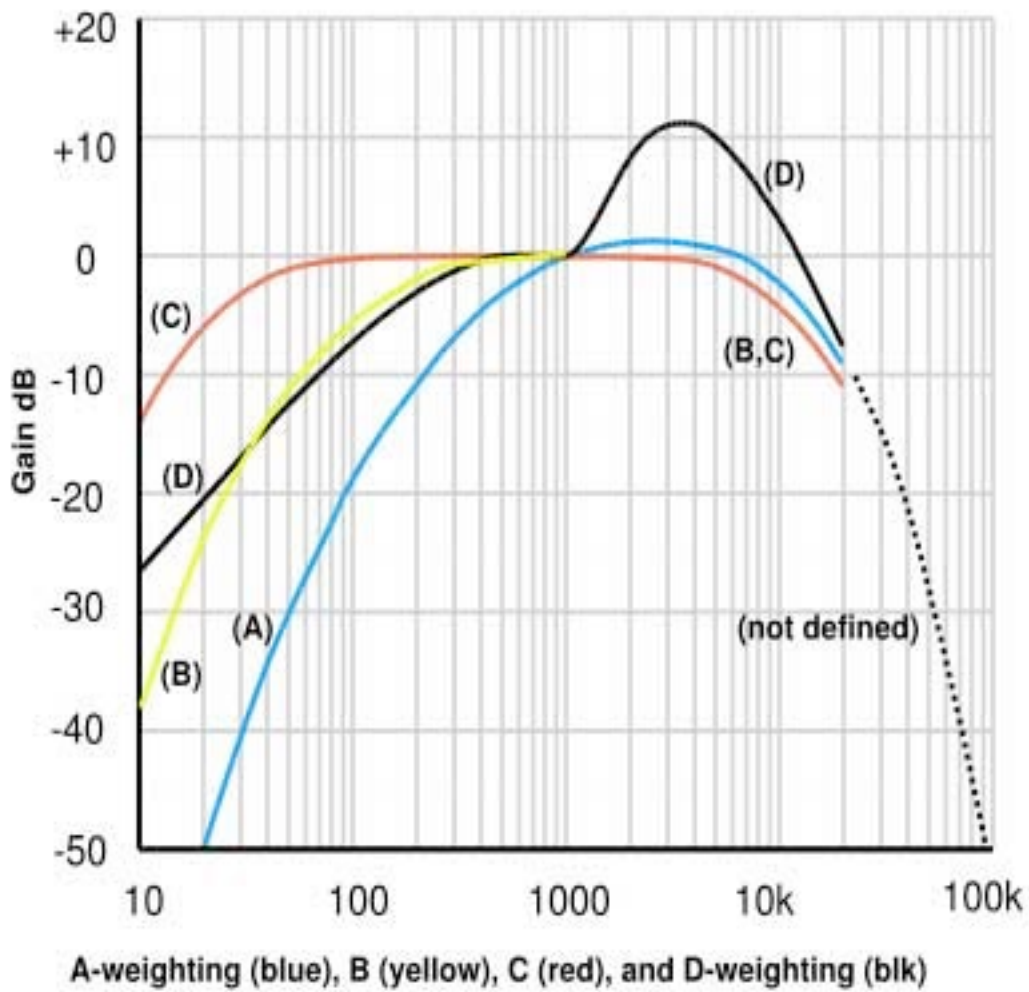
### **3.16. The Limits of Space Perception**

- Minimum audible angle (MAA) is 1 degree along horizontal plane in front
- MAA is about 3 degrees in the vertical plane in front

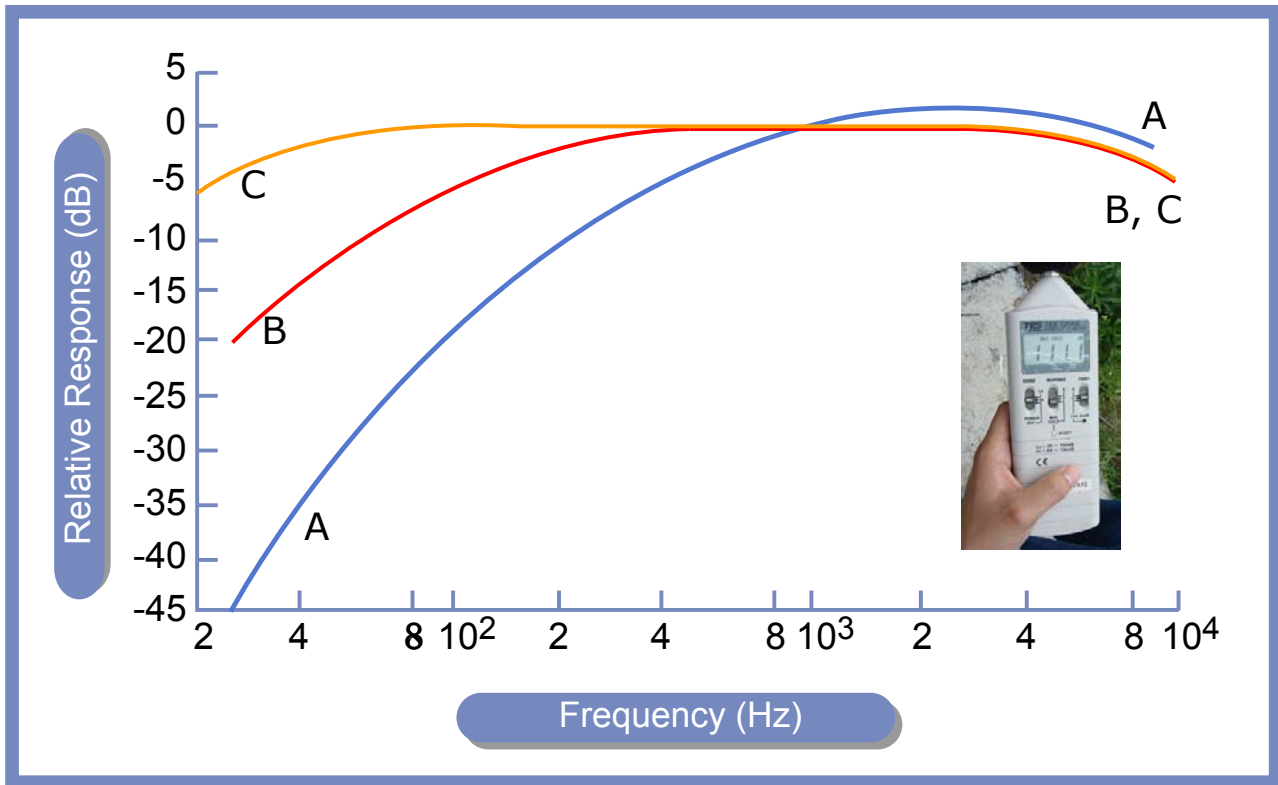
- MAA is greater (our perception is less good) towards side and back

### 3.17. Balancing Amplitude with Frequency Bias

- We can weight amplitude scales to better relate to the ear's frequency bias
- dB-A: A-weighting according to Fletch Munson / ISO 226
- dB-B and db-C: less low frequency offset



Public domain image (Wikipedia).



Graph by MIT OpenCourseWare. SPL meter photo courtesy of [EpicFireworks](#) on Flickr.

- Weights make the dB value closer to perceived loudness
- dB meters include A and C weightings



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- Some spectral analysis tools include weightings

Example: Elemental Audio Systems: IXL Spectrum Analyzer

### 3.18. How the Ear Determines Location

- Methods of determining spatialization
  - Intensity
  - Timing (our ears are separated by distance)

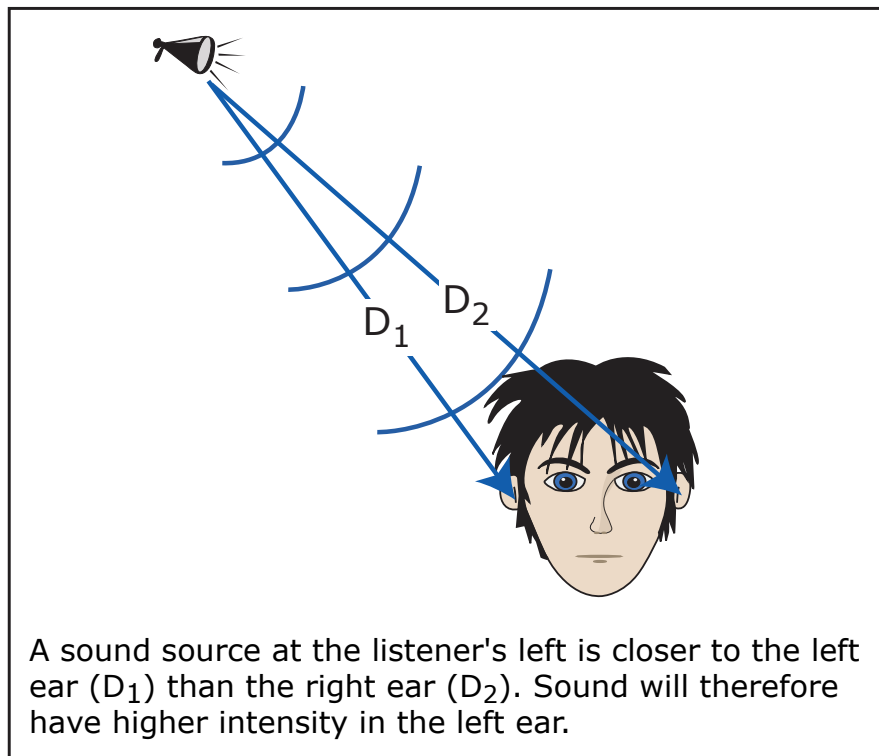


Image by MIT OpenCourseWare.

- Spectral cues
- Reflections off of the Pinna

Graph removed due to copyright restrictions.  
See Fig. 10.15 in Holmes, T. *Electronic and Experimental Music*.  
3rd ed. Routledge, 2008.

- The ear has more directional sensitivity to high frequencies

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See Fig. 10.8 in Thompson, D. M. *Understanding Audio*.  
Hal Leonard Corp., 2005.



Diagram removed due to copyright restrictions.  
See Fig. 10.7 in Thompson, D. M. *Understanding Audio*.  
Hal Leonard Corp., 2005.

- Example: jndPanning.pd
- The ear has more directional sensitivity to sounds in front

### **3.19. Masking**

- Given two sounds at similar frequencies, the loudest wins
- Basilar membrane only registers loudest signal at one place

### **3.20. Reflections**

- Sound reflects (bounces), diffuses, and absorbs off of surfaces
- These factors create ambience or reverb; a space without these features is called anechoic
- Three steps: direct sound, early reflections, reverberations

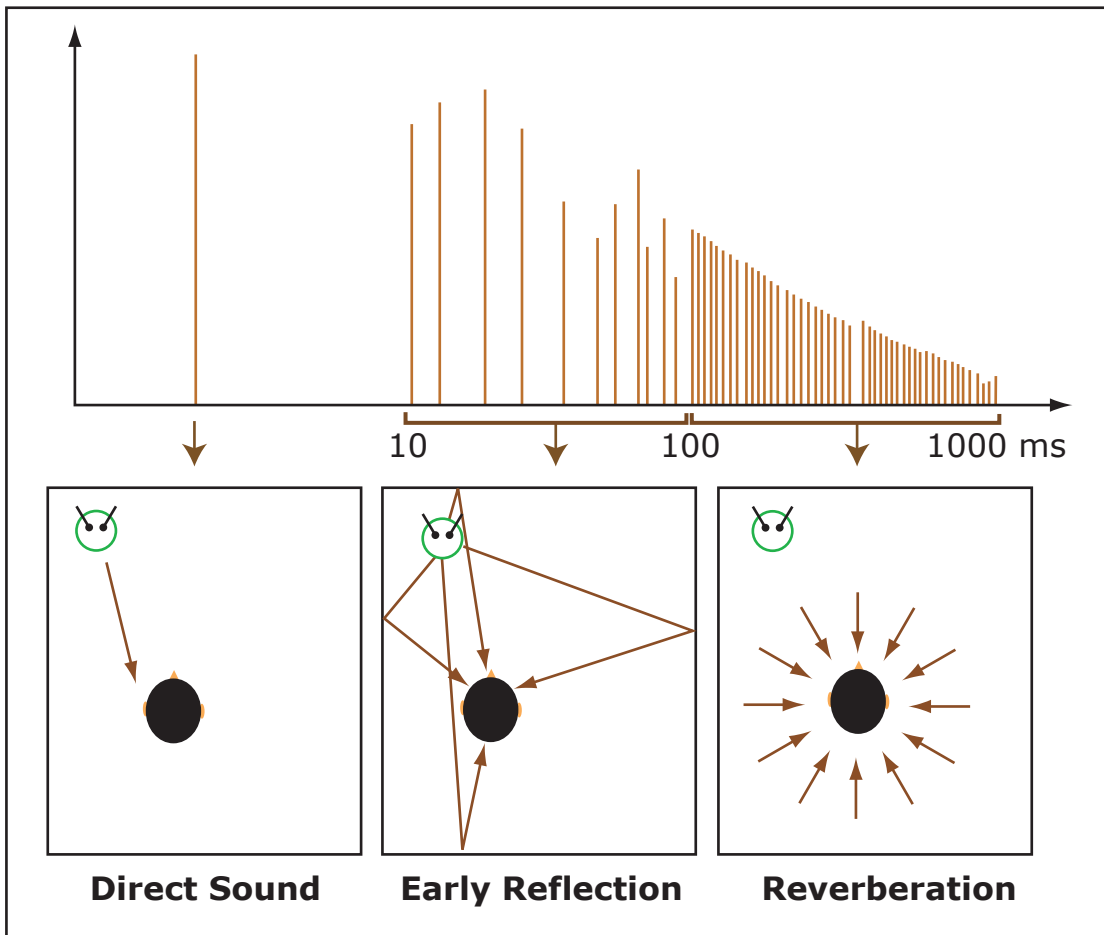


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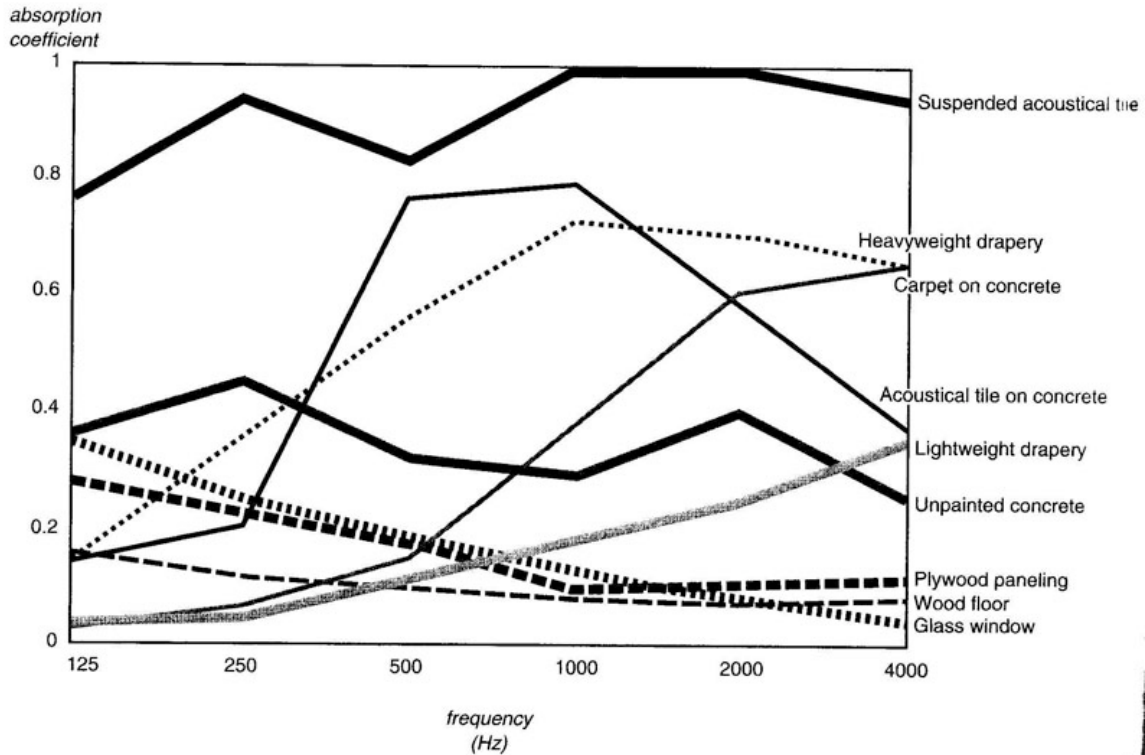
- Early reflections are discrete echos
- Reverberations are echos that are so close to gether (less than 30 msec apart) that they form a continuous sound

### 3.21. Absorption

- Absorption consumes the energy of sound
- Sound does not absorb equally for all frequencies

Figure 11-3

Absorption



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### 3.22. Phase Filtering and Cancellation

- Combining two signals slightly out of phase causes a timbral change: called comb filtering
- Combining two signals 180 degrees out of phase causes signal cancellation
- Combining two signals with delays less than 30 msec results in coloration
- Example: processorsDelay.pd (samples, then noise)
- Always possible when mixing multiple microphone captures

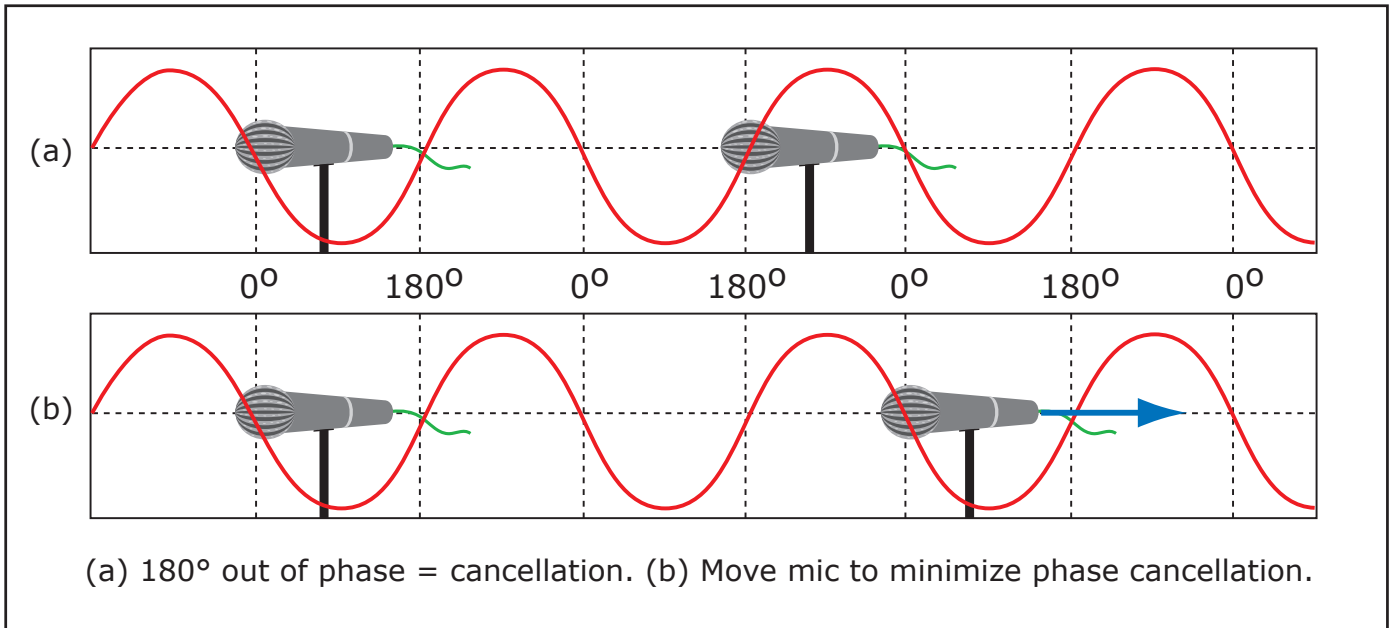
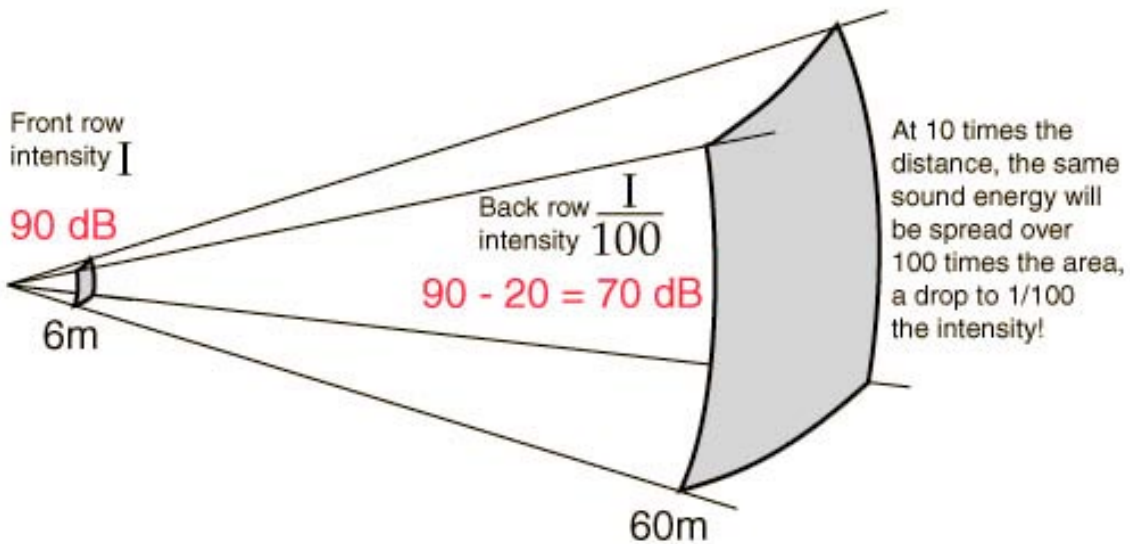
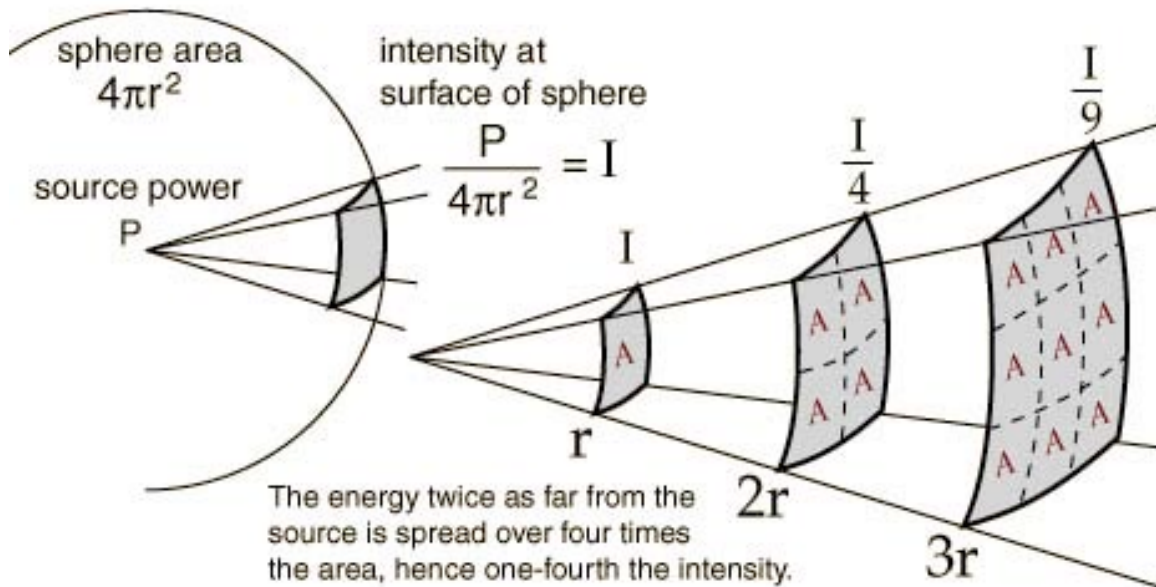


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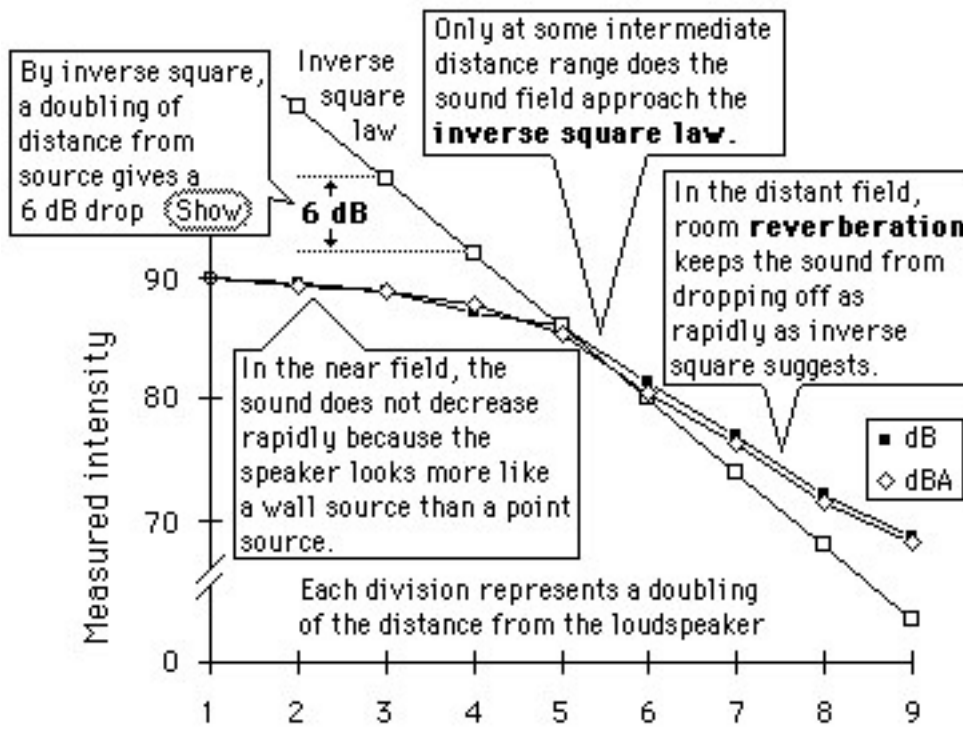
### 3.23. Inverse Square Law

- Amplitude diminishes with distance
- Theoretically, sound in three dimensions diminishes in power according to the inverse square law
- Three-dimensional radiation



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- Doubling the distance from a source reduce the amplitude by 6 dB
- Real-world measures differ



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### 3.24. Reading: Eargle: Basic Sound Transmission and Operational Forces on Microphones

- When comparing the RMS of sine and square waves, what does the difference in values tell us?
- Reverb time measured as the time between the start of the sound and a decrease in how many dB?
- The term “gradient” is used to refer to what?
- Which reduces high frequencies more: dry air or wet air?
- What is diffraction?
- In general, what will happen to sound captured by a directional microphone off axis?

## **Chapter 4. Meeting 4, Workshop: Amplitudes and Recording Hardware**

### **4.1. Announcements**

- Due tomorrow, Wednesday, 22 February: Mix Graph 2

### **4.2. Groups**

- 
- Student names removed for privacy.
- 
- 

### **4.3. Quiz**

- 15 minutes

### **4.4. Using an SPL Meter**

- Ranges
- Fast and Slow response
- A and C weighting

### **4.5. Activity: Amplitudes and Distance**

- Procedure
  - Articulate two different sounds at a distance of 4 feet from the SPL meter; record the peak results with A weighting.
  - Repeat the measurements at 8, 16, and 32 feet

- Repeat the procedure in two different spaces: one reverberant (hallways, large rooms) and one dry (outdoors).
- The two sounds sources can be the shaker and a hand-clap
- Chart results on board in class

#### **4.6. Activity: Recording Hardware: Cables and Stands**

- XLR cables
  - Connectors and gender: male is output, female is input
  - MOSS lengths
  - Wrapping and storing
- Mic stands
  - Two sizes
  - Handling, tightening, and positioning
  - Using the boom

#### **4.7. Activity Schedule**

- 3:50 to 2:15  
 A, B: Amplitude and Distance  
 C + D: Recording hardware: microphone stands and cables
- 2:15 to 2:40  
 A + B: Recording hardware: microphone stands and cables  
 C, D: Amplitude and Distance



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## Chapter 5. Meeting 5

### 5.1. Announcements

- CLASS CANCELLED (Away at conference)
- Due: Mix Graph 2

## Chapter 6. Meeting 6, Controlling Gain and Processing Signals

### 6.1. Announcements

- Mix Graph 3 due Wednesday
- Audio materials for first Processing Report (due 7 March) will be released on Wednesday

### 6.2. Review Quiz 1

- ?

### 6.3. Amplitudes in Nature

- Each overtone has a different dynamic contour in time
- Transients: non-harmonic (non-periodic) attack portion of a sound
- ADSR dynamic contour (envelope)

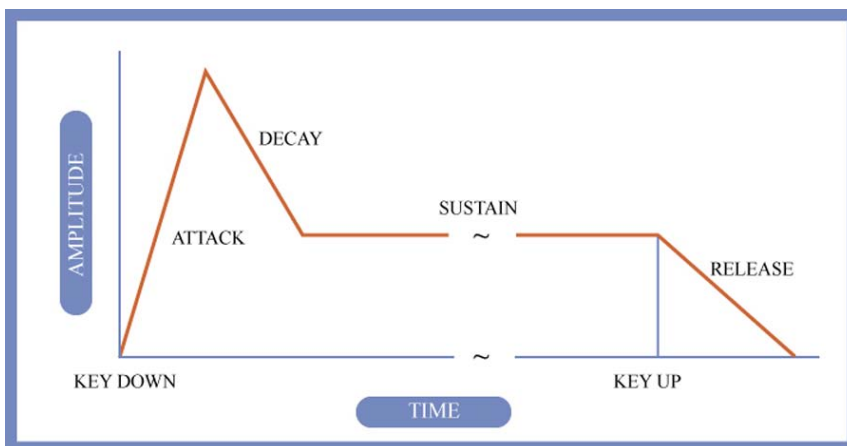


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### 6.4. Dynamic Range

- Dynamic range: range of available amplitudes
- Standard operating level (SOL): optimum average level on a signal
  - Pro-audio: +4 dBu (-20 dBFS)

- Commercial audio: -10 dBV (-7.8 dBu)
- The maximum: peaking, clipping, saturation, overload, distortion, maximum output level (MOL)
  - As a sine wave is clipped, it becomes a square wave
  - Clipping adds harmonics
  - Example: processorsDistortion.pd
- The minimum: noise floor

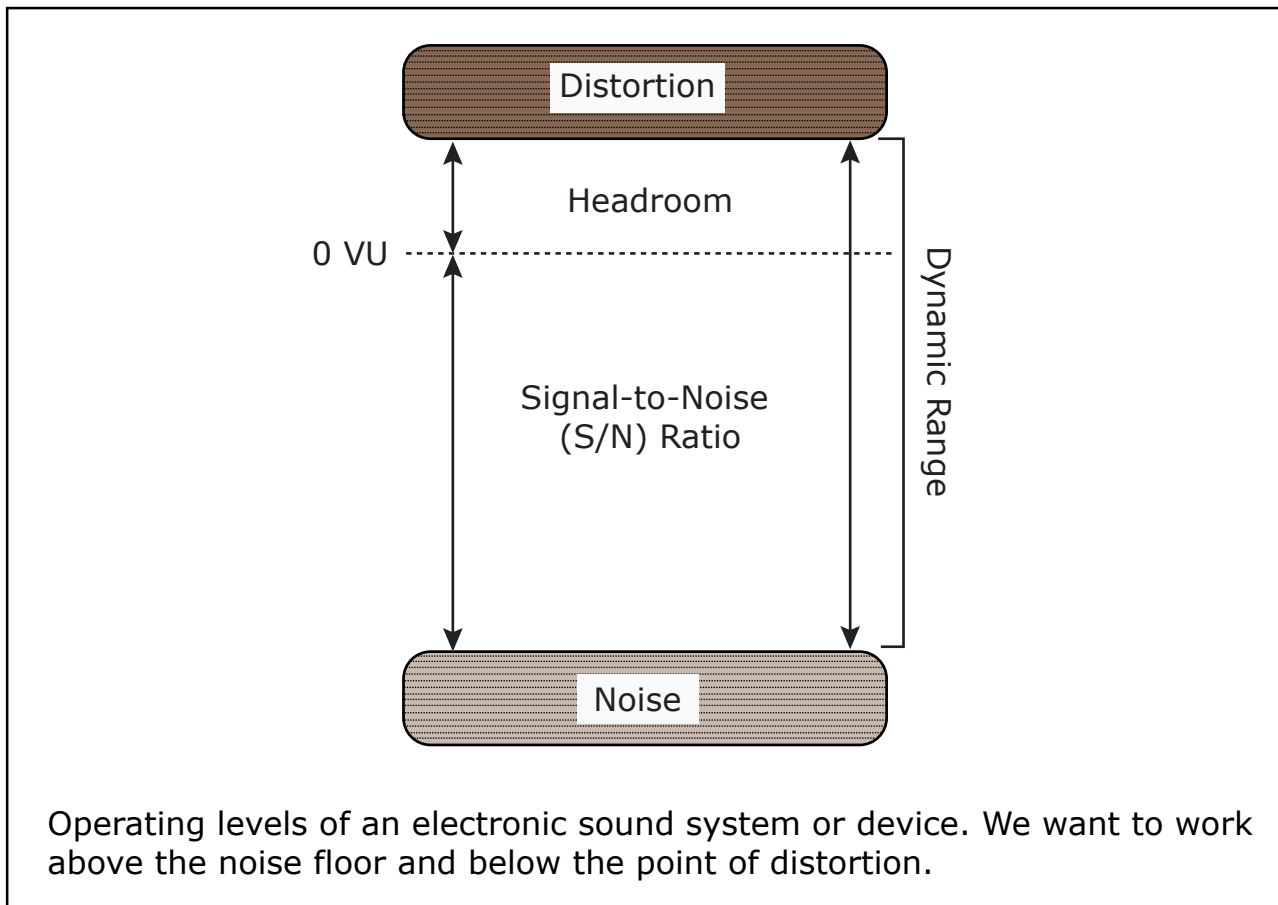


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- Signal to noise ratio
- Peak to average ratio

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Illustration of peak level vs. average level in a sound wave.  
See Fig. 11.4 in Thompson, D. M. *Understanding Audio*.  
Hal Leonard Corp., 2005.

- Headroom: space between SOL and clipping (20 dB is standard)

## 6.5. Amplitude Meters

- A simple measure of signals power
- Potentially misleading
- Many varieties
- Considerations when evaluating amplitude meters
  - Peak or average?
  - Units in dB or something else?
  - Negative and/or positive values?
  - Where is 0 dB and what does it mean?
  - What is negative infinity?

## 6.6. dB Meters

- dBu Meters: negative infinity to +24 dBu (sometimes 20 dBu)
- dBFS Meters: negative infinity to 0 dBFS

- dB SPL Meters: 0 to 120 dB SPL
- Comparisons
  - +4 dBu = -20 dBFS (sometimes -16 to -18 dBFS)
  - -10dBV is equivalent to -7.8 dBu

## 6.7. VU and RMS Meters

- Root Mean Square (RMS): an average
  - Mathematical average

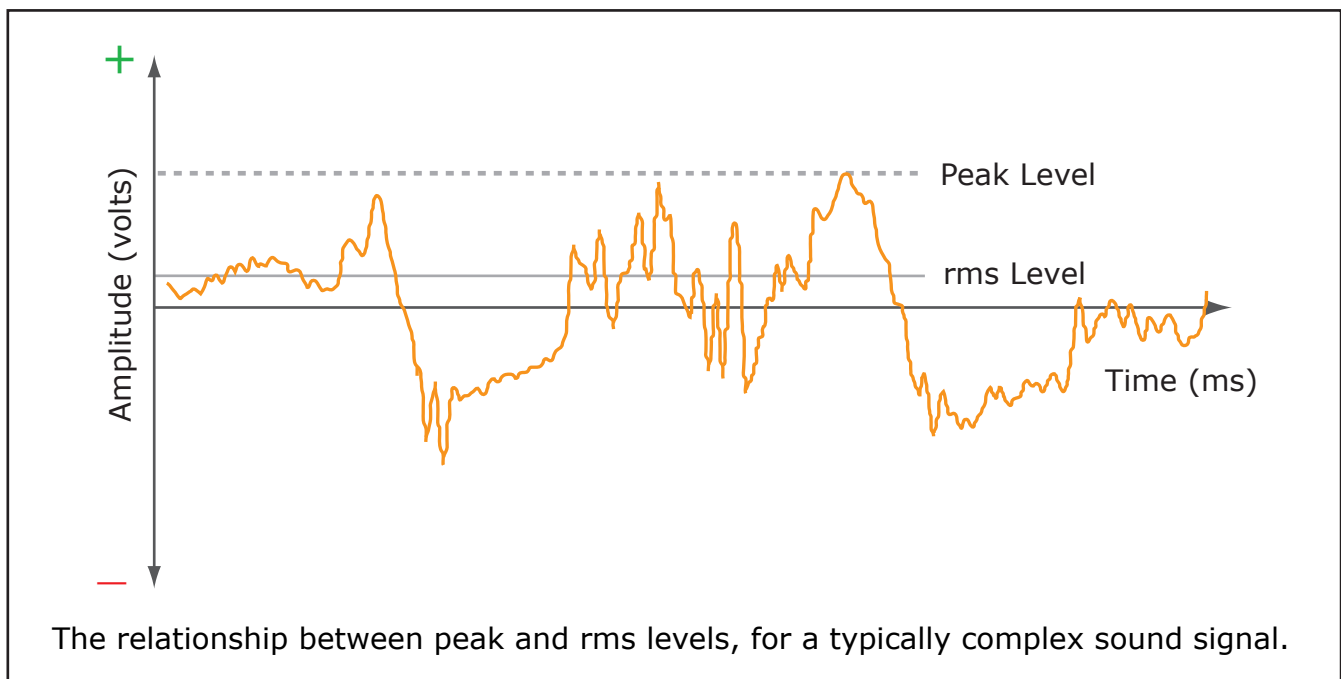


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- Average the square of a number (or a window) of samples, then take the square root
- RMS of a square wave is greater than that of a sine wave

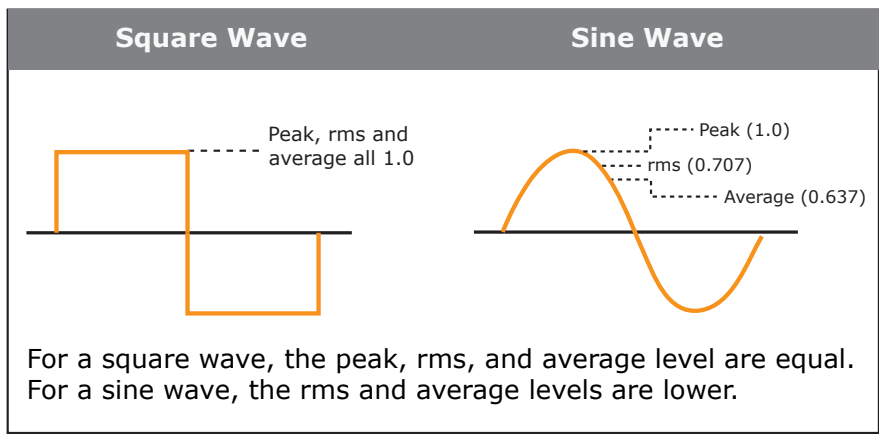
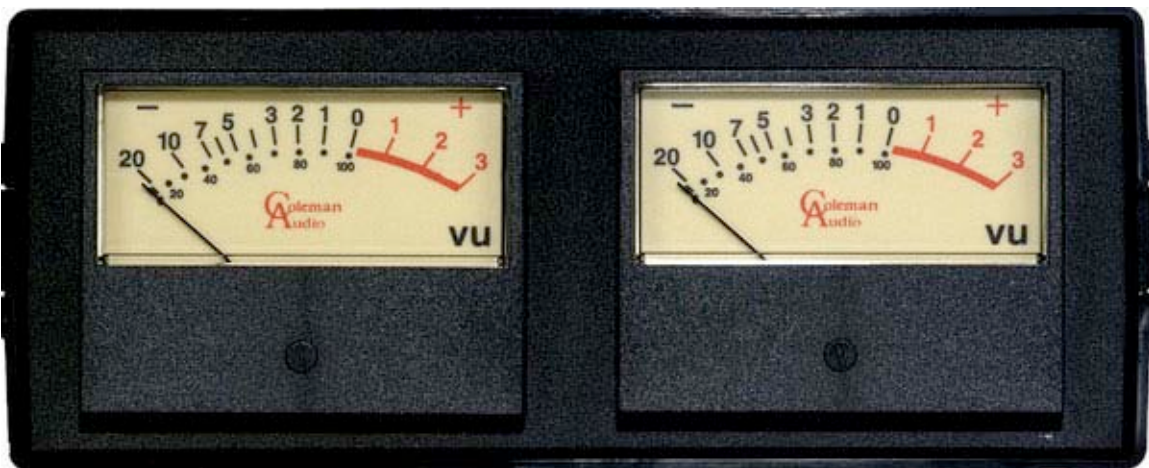


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- Volume Units (VU): an average



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- 0 VU is equal to +4 dBu or 1.228 V RMS for a sine wave
- 0 VU is equal to -20 dBFS (sometimes -18 to -16 dBFS)
- Change in 1 VU may be 1 dB change
- Integrates 300 msec of change
- Peak may be as much as 15 dB (8 to 20 dB) higher than VU reading
- Peak Program Meter (PPM)



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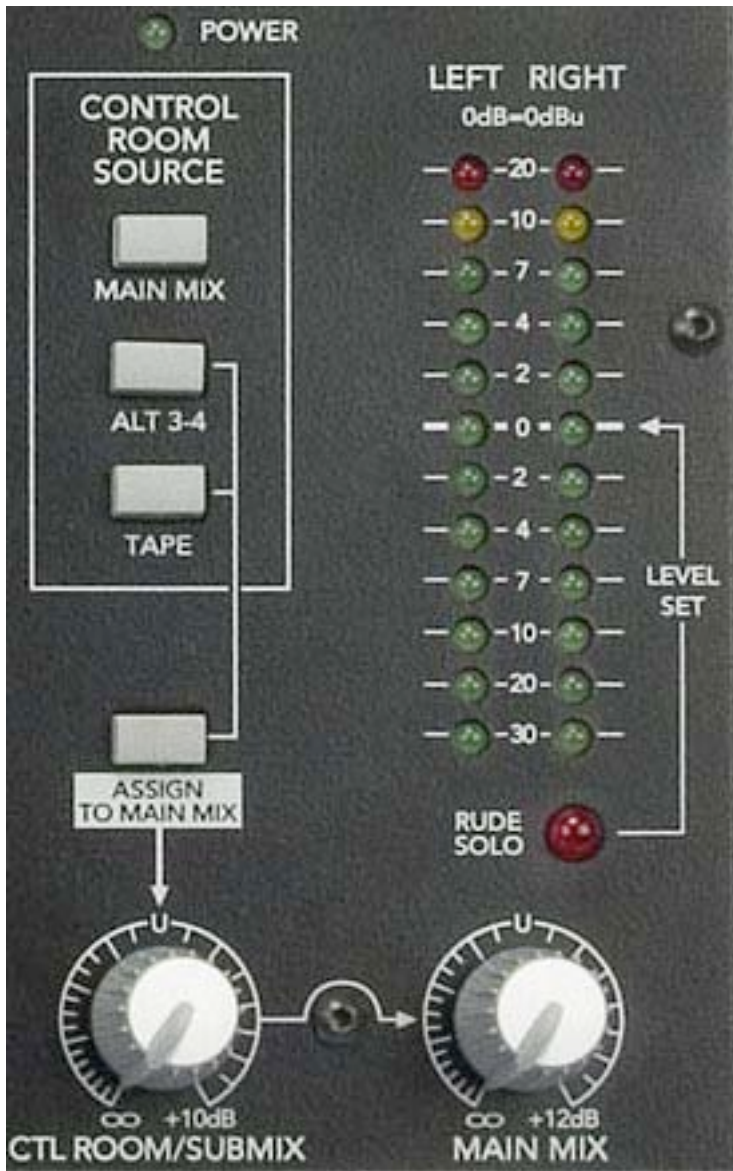
- Scale from 1 to 7; each segment is 4 dB change
- Faster attack time than VU meters (10 ms)
- PPM 6 = 100% reading, +4 dBu = 0 VU
- Adjusts after 10-12 ms

## 6.8. Meter Examples





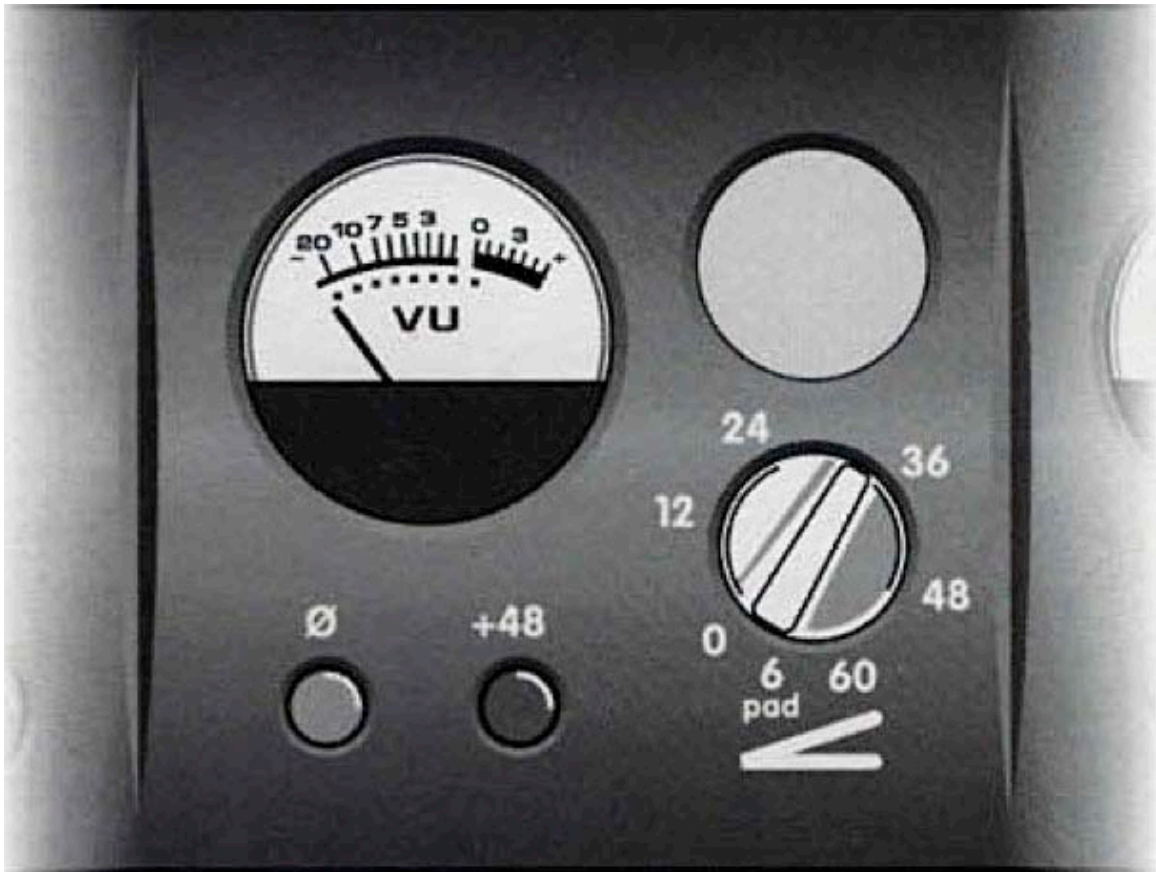
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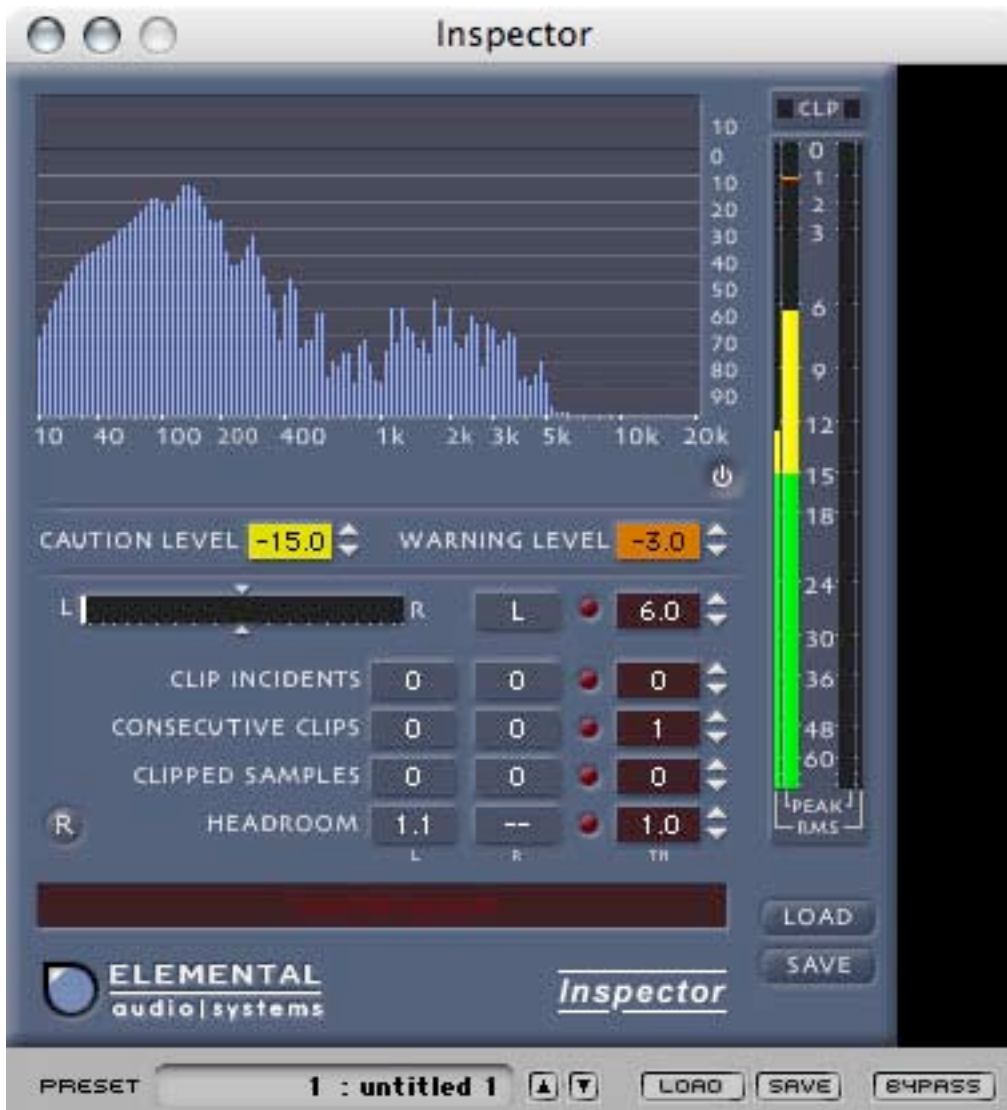
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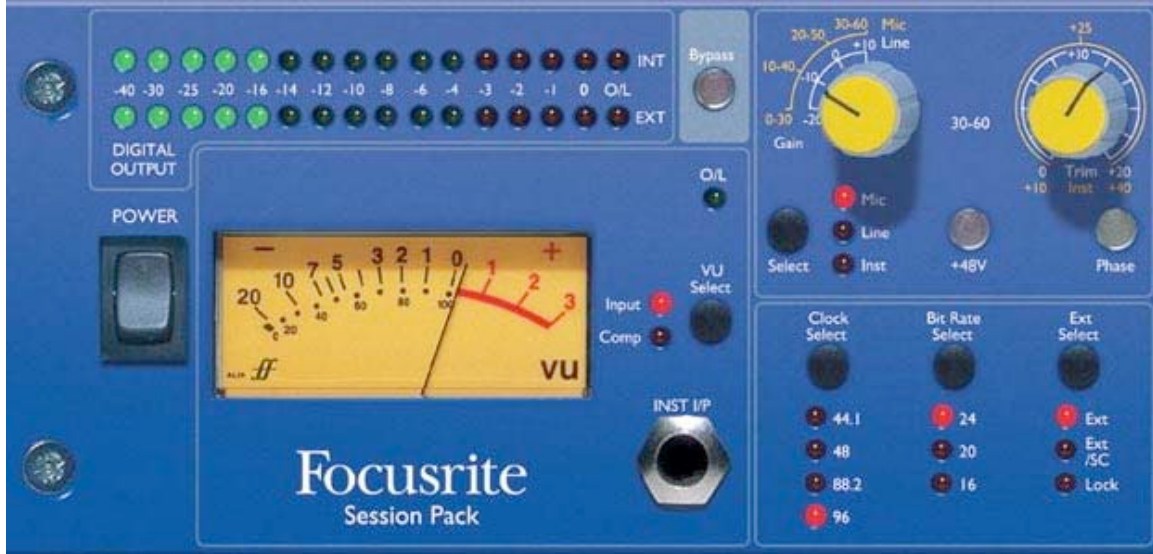
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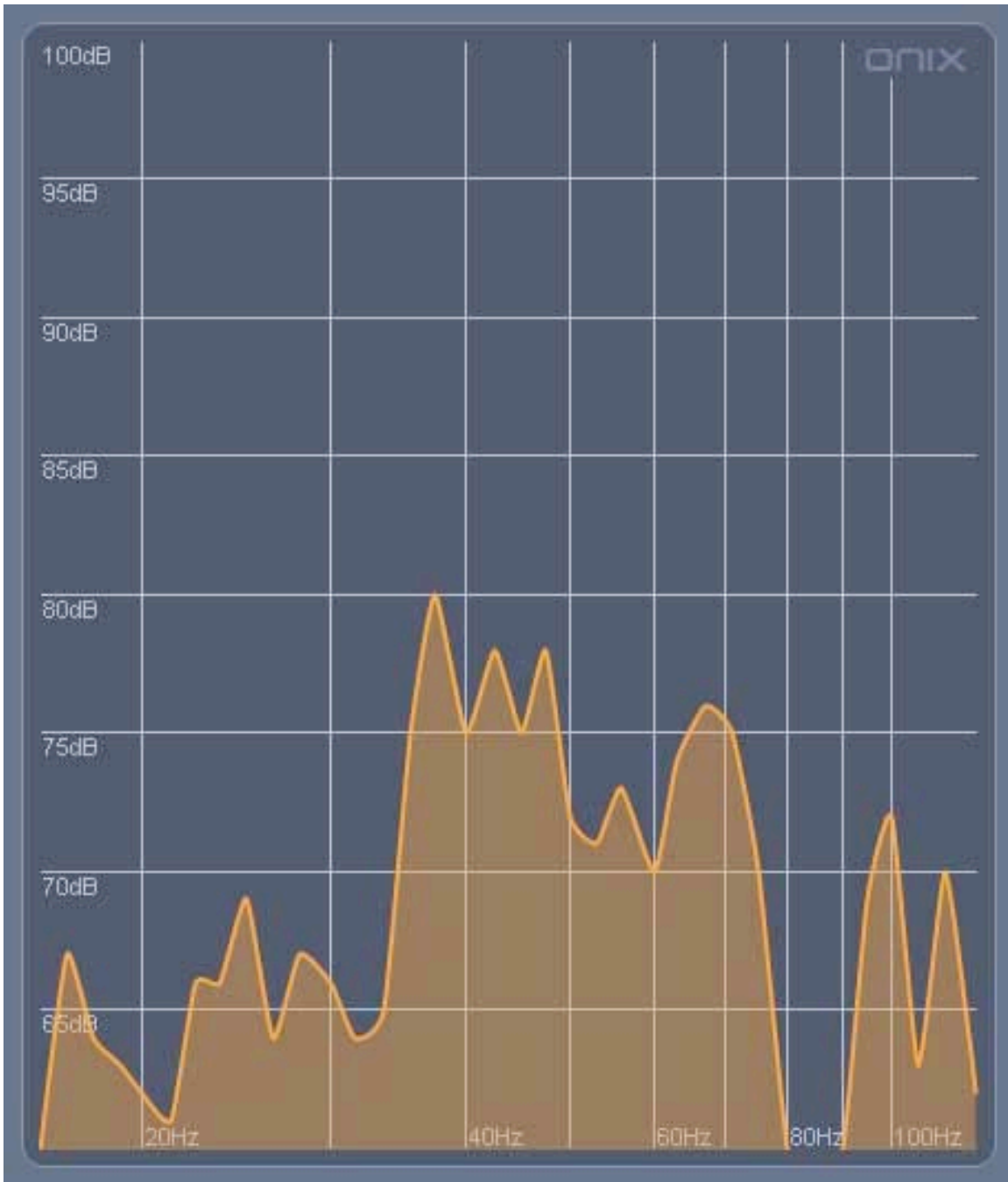
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## 6.9. Changing Amplitudes

- Pre-amp (trim): amplifier with a wide range of gain (0 to 60 dB) designed for bringing very quiet signals up to SOL
- Power amp: amplifier for taking a signal from SOL to a high-powered signal necessary to drive speakers
- Pad (attenuator): reduces gain by a fixed amount with a switch (-6 dB, -20 dB)
- Fader: scales a signal at SOL: unity (no change), boost +10 dB, attenuate to -infinity dB
- Direct Box: convert from -10 dBV to +4 dBu

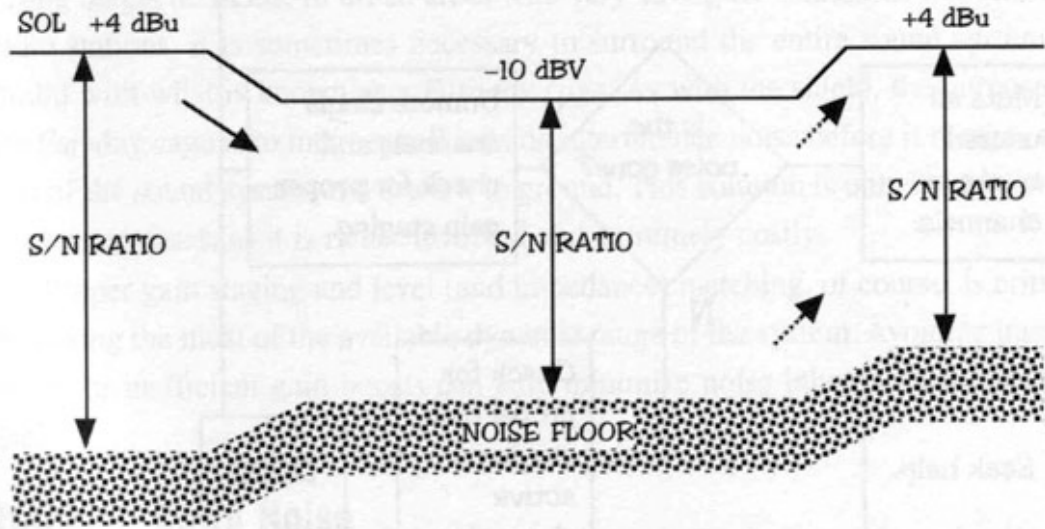
## 6.10. Gain Staging

- Every signal goes through numerous amplifiers from source to destination
- Each amplifier is a gain stage
- Each amplifier (and any process in between) adds noise (has its own noise floor)
- Each gain stage, if above unity, can amplify the last gain stage's noise floor
- Optimal gain staging: first gain stage does all amplification; all subsequent gain stages are at unity
- Optimal gain staging: as much as possible as early as possible

## 6.11. Gain Staging: Example

- Inserting a device with a poor signal to noise ratio can degrade the entire signal path





*Figure 7.21. Inserting semipro gear into the flow reduces available dynamic range by bringing the noise floor up. Semipro gear not only has a lower SOL (standard operating level) but tends to be inherently noisier as well (higher noise floor).*

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## 6.12. Level Setting: Principles

- The essential first step when working with an input
- Mantra: as much as possible as early as possible
- Optimizes signal to noise ratio with ideal gain-staging

### 6.13. Level Setting: Procedure

- Reset, clear, and zero all controls (set trim at minimum)
- Connect or select input
- Set meters (if necessary) to display only the trim gain stage and skip other gain stages
  - On some mixers, this may mean engaging SOLO
  - On some mixers, this may mean engaging Pre-fader listen (PFL) SOLO
- Must get typical material from the source (musician, device, et cetera)
- Raise the trim slowly
- Find amplitude peaks and estimate average peaks with meters
- Continue to raise the trim until average peaks are at +4 dBu (-20 dBFS, 0 VU)

### 6.14. Level Setting: Example

- Tascam HD-P2 portable recorder



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### 6.15. Level Setting: Example

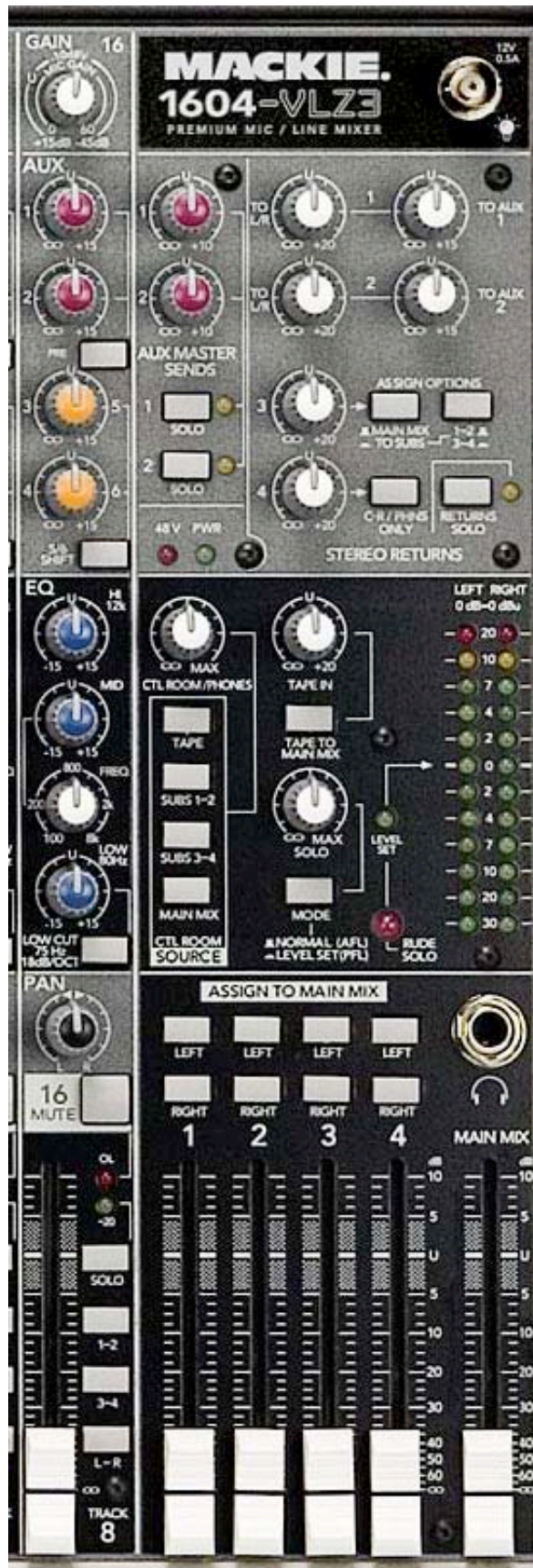
- Avalon AD 2022 preamp



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## 6.16. Level Setting: Example

- Mackie 1604 VLZ3



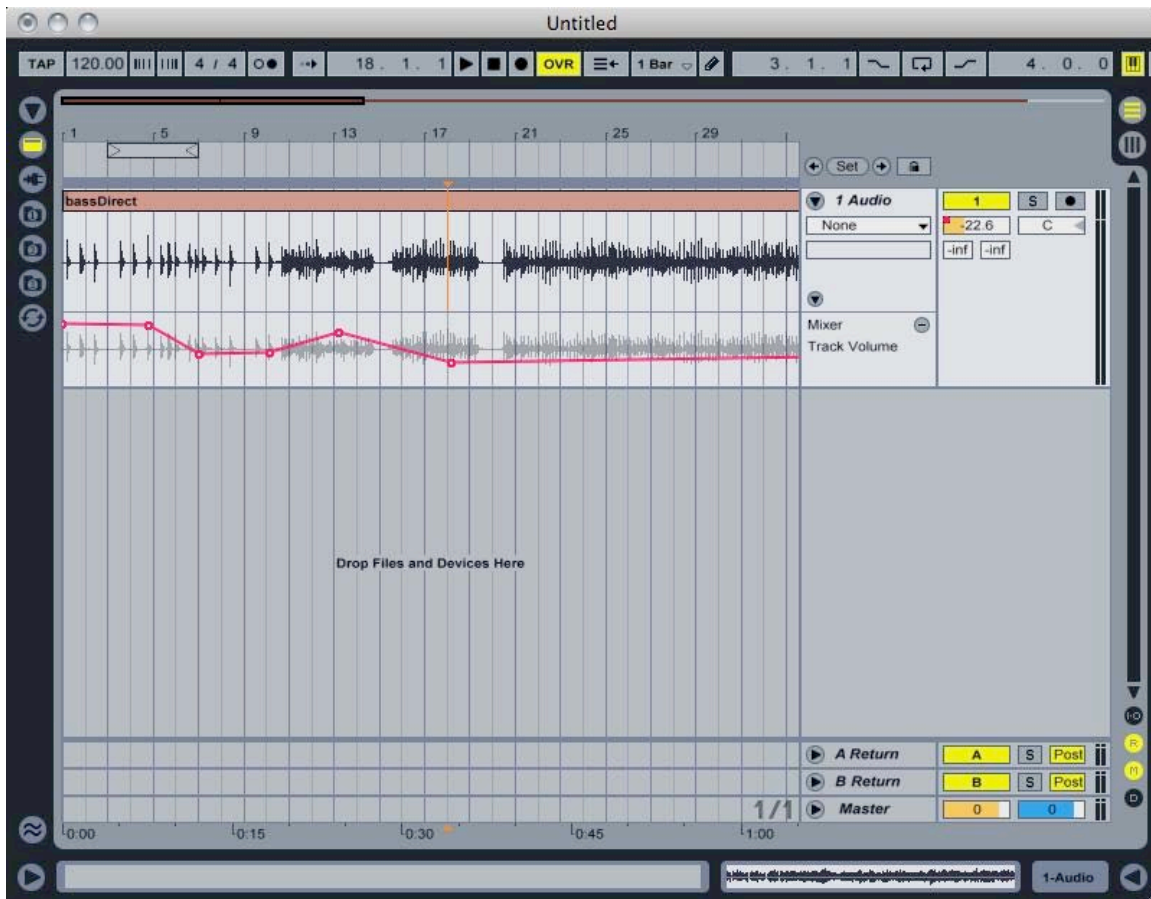
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## 6.17. Automation: Fader Levels

- Automating fader levels in a DAW
- Live: under Mixer, select Track Volume

Double click to add / remove points

Can view view on waveform or in separate lane



Courtesy of Ableton AG. Used with permission.

## 6.18. Panning Amplifiers: Linear

- Take a signal, split into two signals, and inversely vary amplitudes
- A fader that as one turns up, the other turns down
- A bad approach (1 is left, 0.5 is middle, 0 is right)

$$\begin{aligned}L &== (1 - x) \\R &== x\end{aligned}$$

## 6.19. Panning Amplifiers: Non-Linear

- Must reduce amplitude in center to reduce increase in loudness
- Reduction between 3 dB and 4.5 dB

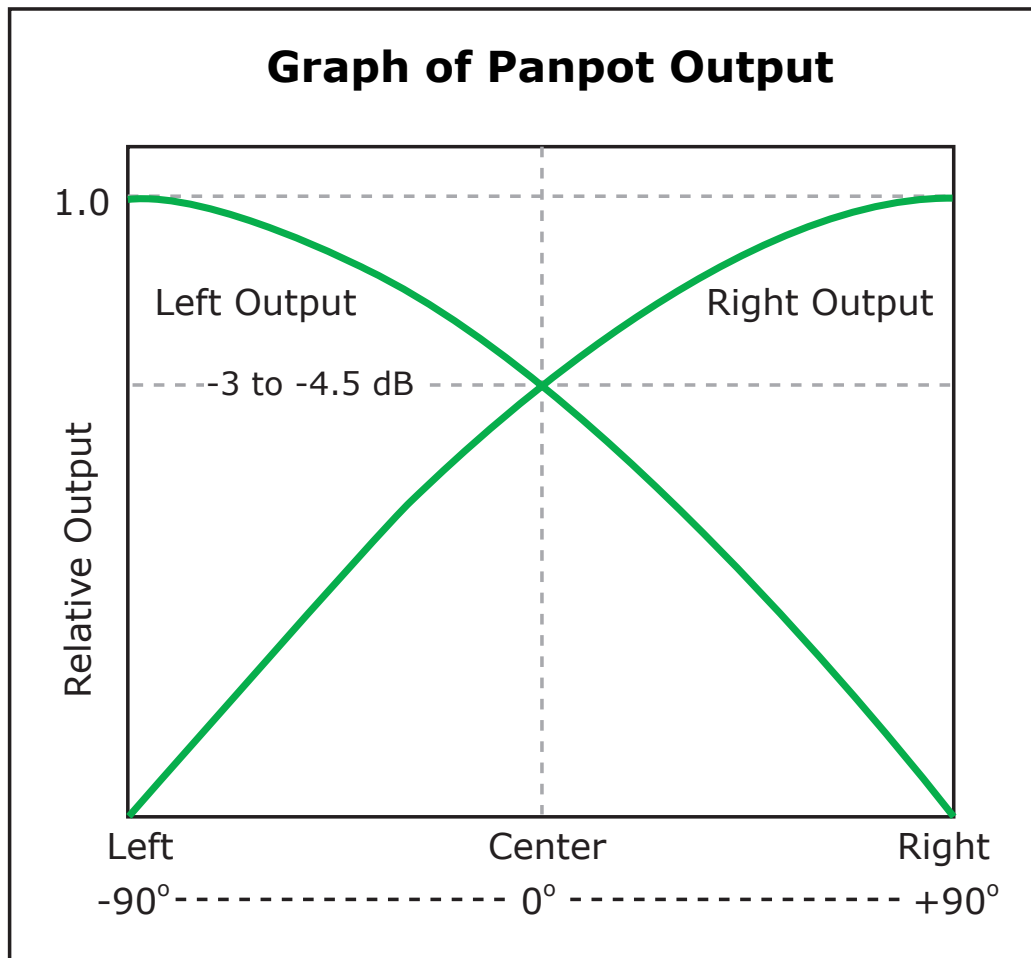


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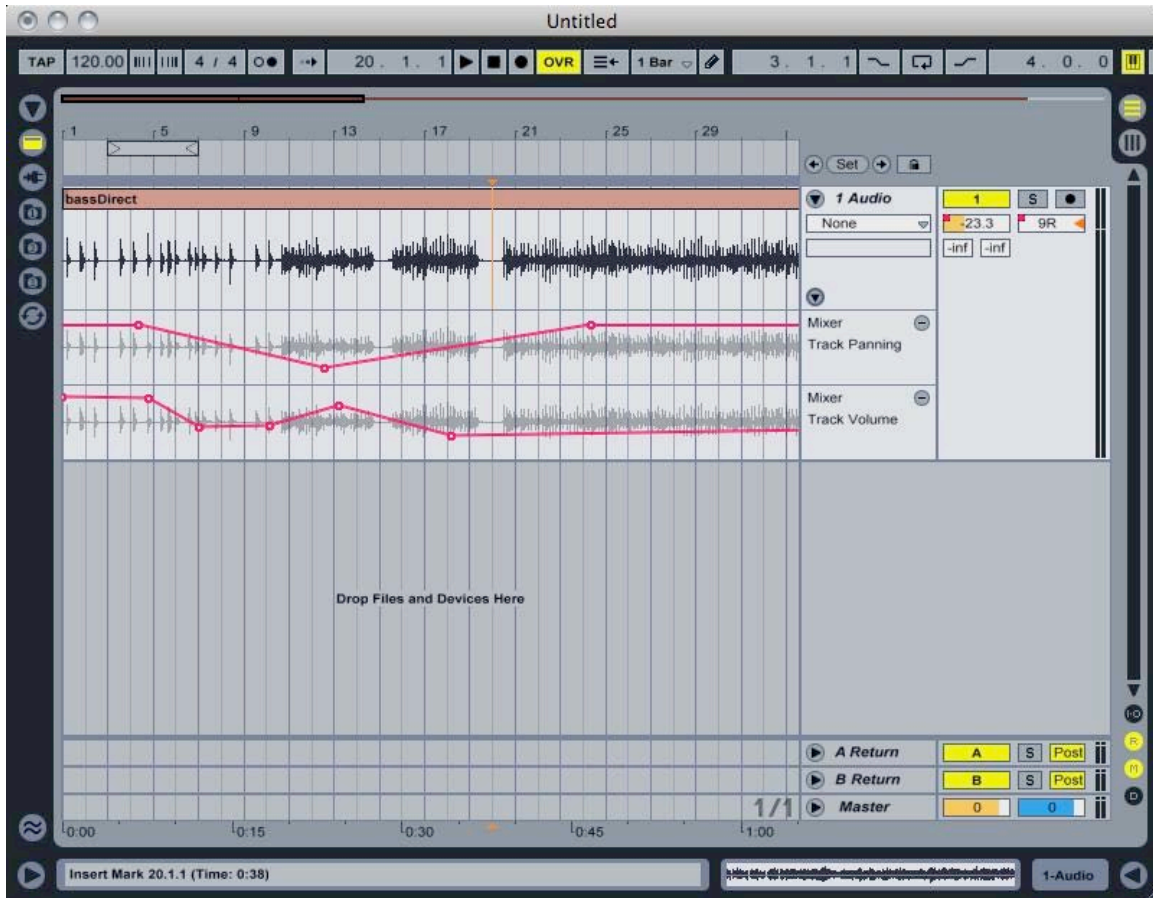
## 6.20. Automation: Stereo Panning

- Automating pan position in a DAW

- Live: under Mixer, select Track Panning

Double click to add / remove points

Can view view on waveform or in separate lane



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## Chapter 7. Meeting 7, Filters and Filter Parameters

### 7.1. Announcements

- Mix Graph 3 due Today
- Quiz on Monday
- Audio materials for first Processing Report 1 (due 7 March):  
audioProcReport01.zip

### 7.2. Equalizers and Filters

- Equalizers are filters (a distinction is not useful)
- Filters selectively boost or attenuate frequency regions
- Filters cannot add frequencies that are not present in the source
- Filter shapes are depicted with frequency-domain graphs and a 0-centered amplitude change
- Filters always manipulate the phase of a signal; mixing out of phase signals can cause filtering

### 7.3. Processors that Shape Frequencies

- Shaping timbre is not the same as transforming timbre
- Shaping timbre
  - Filters
  - Aural exciters and enhancers
  - Bass processors
- Changing and adding frequencies
  - Pitch shifters
  - Harmonizers



## 7.4. Filter Parameters and Units

- Gain: 0 dB is no change (unity); otherwise, may be positive or negative
- Rolloff: slope, a change in gain over frequency measured in dB / octave
- Bands and bandwidth: measured in octaves or Q
- Center and cutoff frequency: Hertz

## 7.5. Filter Types and Parameters

- Low/High Pass, High/Low Cut: cutoff frequency, rolloff
- Low/High shelves: cutoff frequency, gain, Q
- Parametric (peak/notch) filters: center frequency, gain, Q

## 7.6. Filters In Live

- Only use EQ Eight
  - Only use as many filters as you need
- Be sure you have the filter applied to the right channel



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## 7.7. Low/High Pass, High/Low Cut

- The most simple (and extreme) filter
- Parameters: cutoff frequency, rolloff
- Low pass, high cut filter



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- High pass, low cut filter



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- Applications and Cautions
  - Removing high frequency noise or buzz
  - Removing low frequency stage noise, machine noise, hum
  - Isolating one frequency region (combining both low and high pass) to remove leakage
  - Removing super-low frequency signals from digital instruments
  - Removing DC-offset (high-pass)
  - Be careful to avoid removing essential harmonics (low pass)

## 7.8. Low/High Shelves

- Coarse, broad filters
- Parameters: cutoff frequency, gain, Q (sometimes)
- High shelf



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- Low shelf



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- Applications and Cautions
  - Correcting for a general deficiency in a microphone or recording: too little or too much bass/treble
  - Boosting upper harmonics or the air band (high shelf)
  - Avoid boost on low-end; be careful about boosting low frequencies you are not hearing (low shelf)

## 7.9. Parametric Equalizers

- The mother of all eq



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- Parameters: center frequency, bandwidth (Q), gain
- Alternative names: peak / notch filter, peaking filter
- Semi-parametric filters: not all three parameters can be changed
- Two-parameter parametric: bandwidth fixed
- One-parameter parametric: bandwidth and center frequency fixed
- Applications and Cautions
  - Extremely narrow bandwidth boosts will result in pitched-overtones
  - Favor boosting with broad bandwidths to shape regions of harmonics/fundamentals

- Narrow bandwidth cuts can be used to remove noise, feedback, or other undesirable artifacts
- Favor low-frequency boosts with a broad-bandwidth parametric over a low-shelf boost

## 7.10. Graphic Equalizers

- Numerous one-parameter parametric filters
- Distributed across the frequency range in equal octave segments
- Common center frequency spacings: 1/3 octave, 1/6 octave
- Used for live sound engineering, tuning rooms, avoiding feedback
- Example: DBX 1231



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- Example: Mackie Quad Eq



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## 7.11. Filters and Transducers

- Every transducer is a filter
- Every acoustical space is a filter



## 7.12. Examining the Frequency Range with a Filter

- Often we do not know which frequencies need adjustment
- We need a way to scan the frequencies of a sound to find target frequencies
- The parametric filter is the tool to do this
- Sometimes called sweeping the frequency range, or focusing the equalizer

## 7.13. Examining the Frequency Range: Steps

- Instantiate a parametric filter
- Set a very narrow bandwidth (high Q)
- Set a high gain boost
- Slowly sweep the center frequency through the active frequency range
- Listen
- After finding the desired center frequency, adjust gain and bandwidth to appropriate settings

## 7.14. Filter Concepts: Usage

- Use as little dB of gain as needed
- Favor cutting to boosting
- Use as few filter processors as necessary

## 7.15. Filter Concepts: Gain Staging

- The peak amplitude going into the filter should be the same going out
- If frequencies were boosted, may need to reduce output gain
- If frequencies were cut, may need to increase output gain

## 7.16. Filter Concepts: Curves Over Absolutes

- The filter curves can be shifted (vertically) with the same relative result
- The curve is what matters; gain is relative

## **7.17. Filter Applications: Noise Reduction**

- High Pass: low frequency rumble (no likely interaction with overtones)
- Low Pass: high frequency room, tape, and system noises (beware likely interaction with overtones)
- Parametric: remove or reduce particular noises, resonances, or other problems (likely interaction with overtones)

## **7.18. Filter Applications: Isolation**

- Isolate only the frequencies needed from the track
- May remove leakage from other instruments
- May sometimes sacrifice some timbre for greater isolation

## **7.19. Filter Applications: Crossovers and Multiband Processing**

- Crossover: divide a single track into two with low and high-pass filters
- Multiband processors: use numerous filters to isolate the spectrum into regions
- Can process frequency regions separately, then mix back together

## **7.20. Filter Applications: Musical Tunings**

- Emphasize overtones to make different frequencies predominate
- Can tune inharmonic instruments (drums)
- Can alter the timbre of an instrument
- Often use various shapes of parametric filters
- For low frequencies a combination of a shelf and high-pass filter is effective
- High shelf filters can be used to adjust all upper overtones

## **7.21. Filter Applications: Complementary Adjustments**

- For a boost on track A in frequency range X, do a cut on track B in frequency range X
- Can mitigate masking

## 7.22. Processing Report 1

- Process each of the five audio tracks to remove noise, remove leakage, increase isolation, balance timbre, and optimize mixing potential. Only filter/equalization processors can be used.
- For each file, in prose, describe the problems with the audio file and what you are trying to do. Then, describe the specific processors you are using, as well as the specific filter settings used. Be sure to include units for all parameters.
- Send back to me the processed audio files (export or bounce out of the DAW) as well as the written report.
- In Live: select the track, select “Export Audio,” rendering the Master track. Be sure to listen to rendered audio to make sure it has been processed.

## 7.23. Listening: Filters

- Listen to the modification and identify the filter

## 7.24. Reading: Katz: Equalization Techniques

- Why should gentle EQ slopes be favored?
- What are three approaches to focusing the EQ described by Katz?
- What is the “yin and yang” of EQ, as described by Katz?
- Why can EQ generally not be used to fix comb filtering?
- What is a linear phase EQ?
- What is a dynamic EQ?

## 7.25. Acoustical Spaces as Filters

- Lucier: Iterative re-resonance: “I Am Sitting in a Room” (1970)

## **Chapter 8. Meeting 8, Interconnections, Signal Flow, Busses, and Patch Bays**

### **8.1. Announcements**

- Audio materials for first Processing Report (due 7 March):  
audioProcReport01.zip
- About Eargle readings
- Need schlep crew of three for Wednesday at 3:10 at my office
- Need volunteer solo musicians for Wednesday who can bring instruments

### **8.2. Quiz**

- ?

### **8.3. Pro Audio and Consumer Audio**

- Standard operating level and signal-to-noise ratio
- Cables
- Price

### **8.4. Cables**

- Wires (conductors): carry voltages or grounds
- Shielding: meso level of protection
- Insulation: outer level of protection
- Connectors and Jacks: provide easy interface, can be male (M) or female (F)

### **8.5. Signals, Voltages, and Grounds**

- Analog sound can be represented as a changing voltage
- Grounds are a point of zero voltage

- For safety: a path for faulty currents
- Ground loops: grounds with differing electrical potentials on the same connection (not exactly a ground)

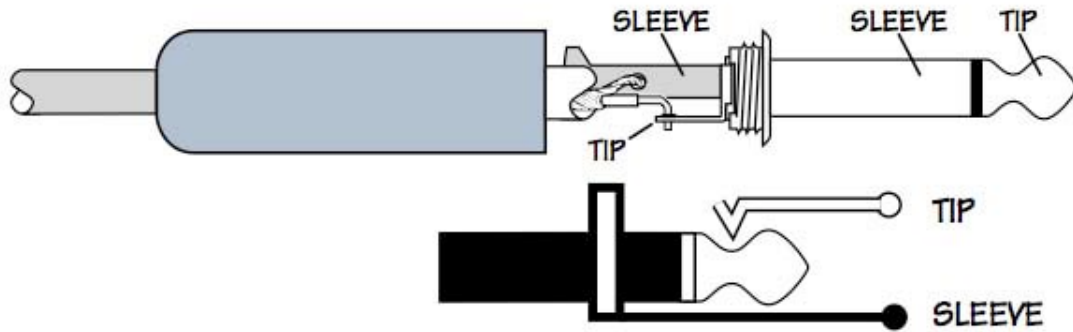
May result in a 60 Hz hum

## 8.6. Analog Cables: Types

- Unbalanced
  - Two conductors: one signal, one ground
  - SOL: -10 dBV
  - High impedance
  - Length Limit: 25 feet
- Balanced
  - Three conductors: two signals, one ground
  - SOL: +4 dBu
  - Low impedance
  - Length Limit: 1000 feet
  - Active and transformer balanced

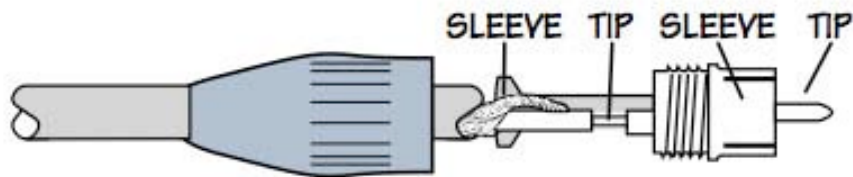
## 8.7. Analog Cables: Connector Examples

- TS



**Figure C: TS Plug**

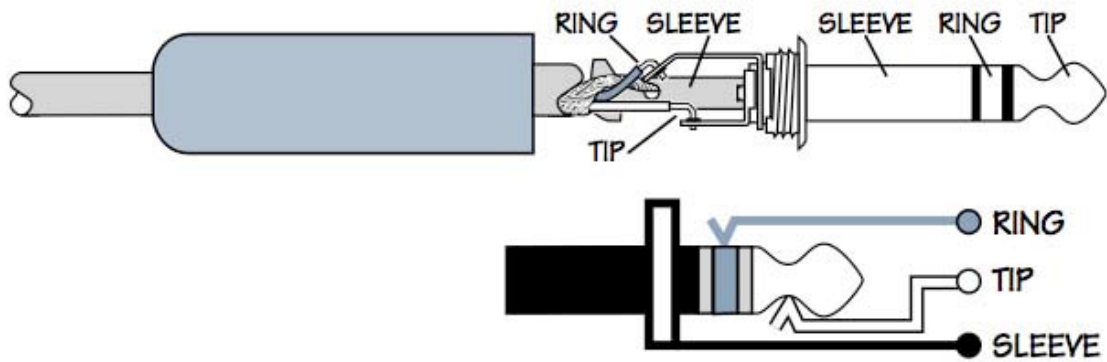
- RCA (Phono)



**Figure D: RCA Plug**

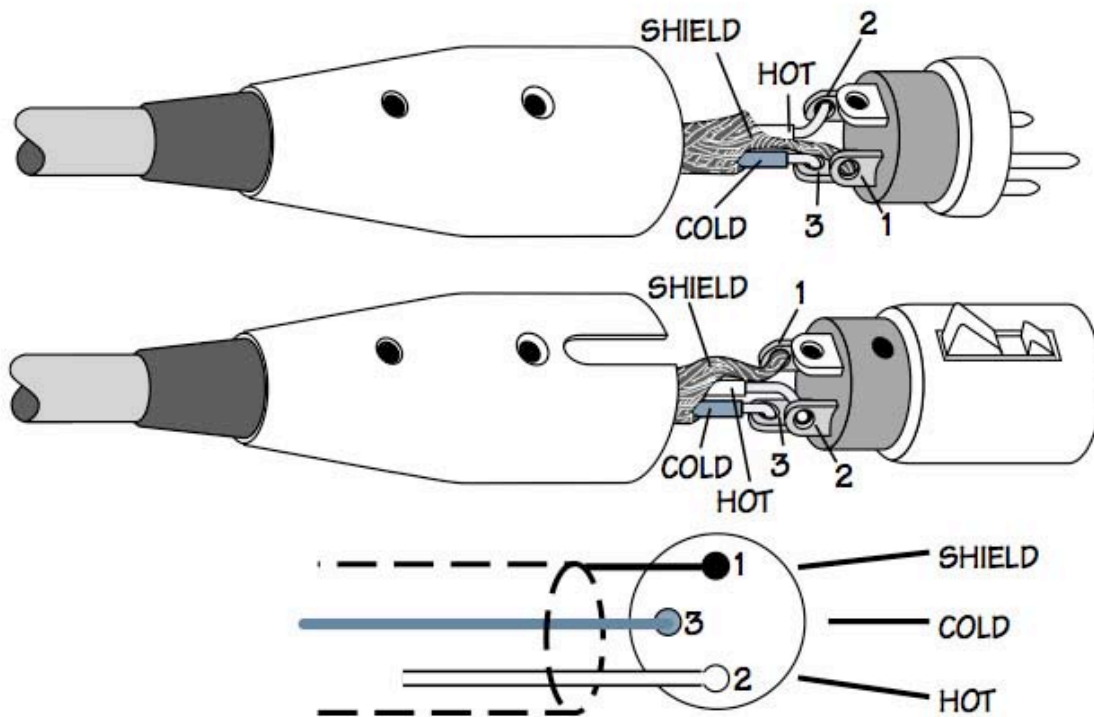
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- TRS



- XLR

Inputs are always XLRF, outputs are always XLRM



**Figure A: XLR Connectors**

- TT (Bantam)

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## 8.8. Balancing a Signal

- Take a positive and negative (180 degree phase inversion) of a signal
- Transmit over a distance
- At the destination, make the negative positive again
- Sum the signals, than divide in half
- As a procedure: (1) signal (2) signal+ | signal- (3) signal+ noise+ | signal- noise+ (4) signal+ noise+ | signal+ noise- (5) signal++ (6) signal

## 8.9. Cable Internals: Conventional Two Conductor

- One braided wire, with shield used as second conductor (ground)





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- Called guitar cable, instrument cable

### 8.10. Cable Internals: Conventional Three Conductor

- Two braided wires, with shield used as second conductor (ground)



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- Called Mic cable, twisted pair

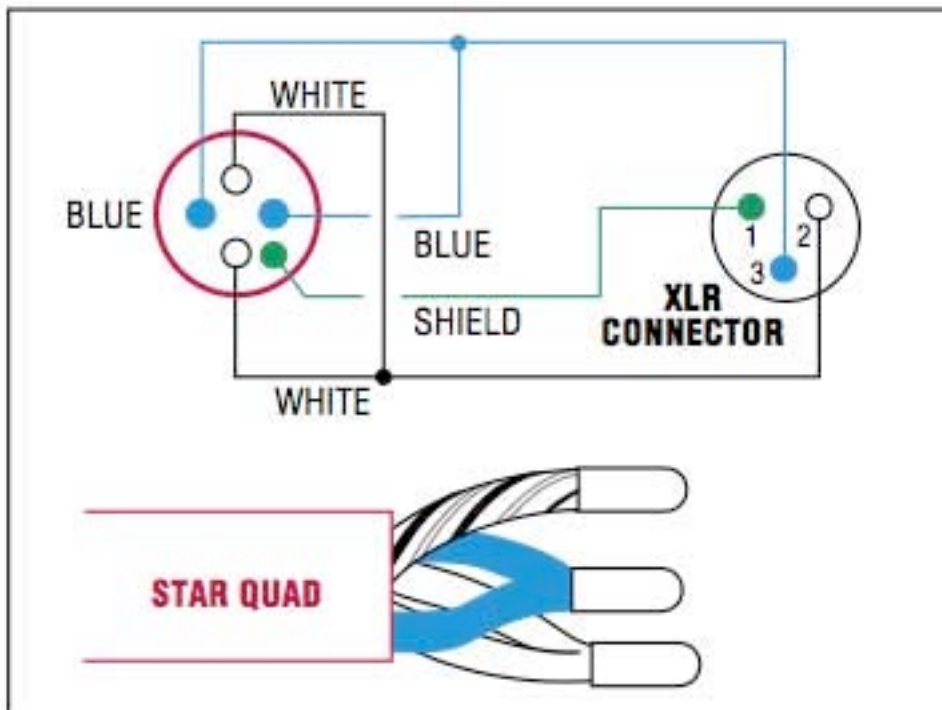
### 8.11. Cable Internals: Star Quad

- 5 conductors: 1 ground, 2 positives, 2 negatives

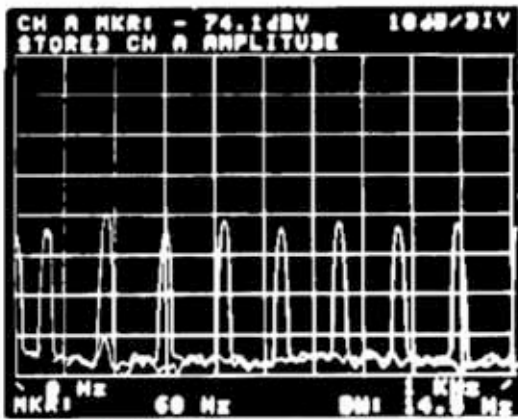
- Four braided wires, with shield as fifth conductor



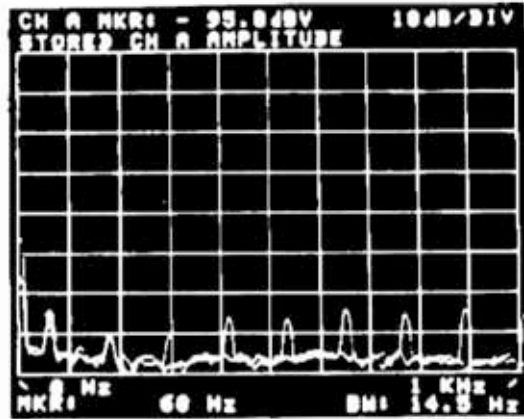
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Typical 2 conductor mic cable.



Canare Star Quad cable.

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## 8.12. Converting from Balanced to Balanced

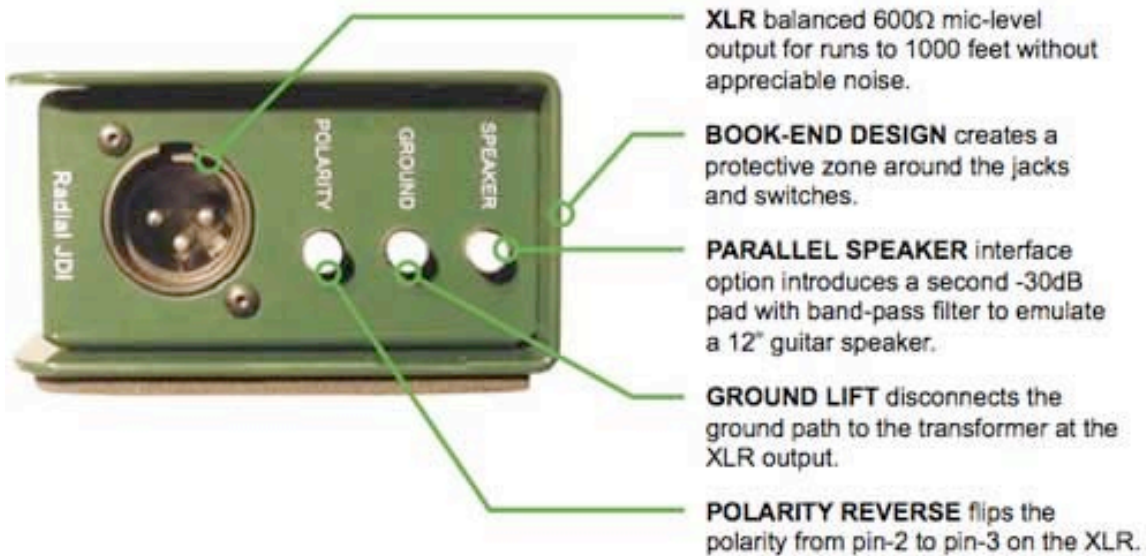
- Use a cable (best) or adapter (not recommended)

## 8.13. Converting from Unbalanced to Balanced: DI Box

- Never use an adapter or a cable
- Direct Injection Box: convert -10 dBu to +4 dBu and balance signal



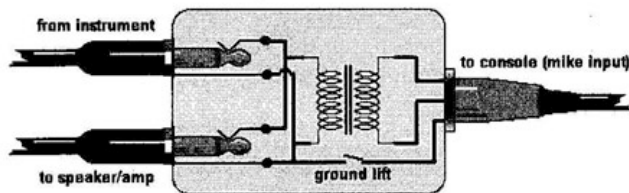
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- Transformer isolation removes ground-hum noise

**Figure 16.4.** Schematic for a direct (DI) box.



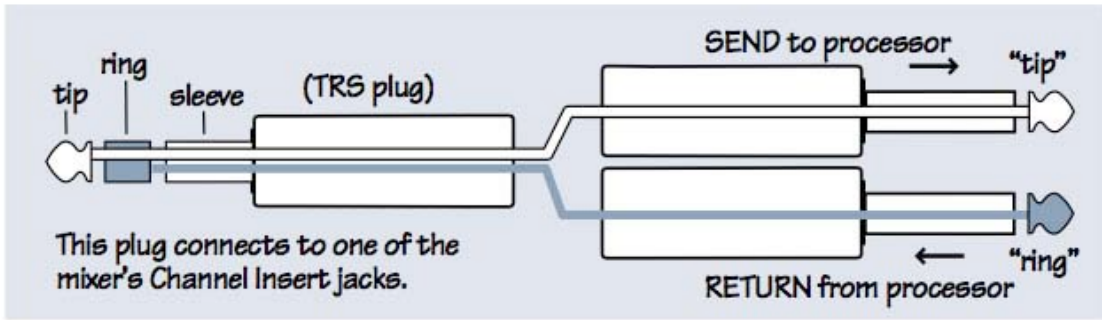
- Used to connect guitars, basses, keyboards, guitar/bass amp direct outs, turntables, drum machines, synths, et cetera into pro-audio inputs
- Can be used in forward and reverse to extend the run of an unbalanced signal

## 8.14. Analog Cables: More Examples

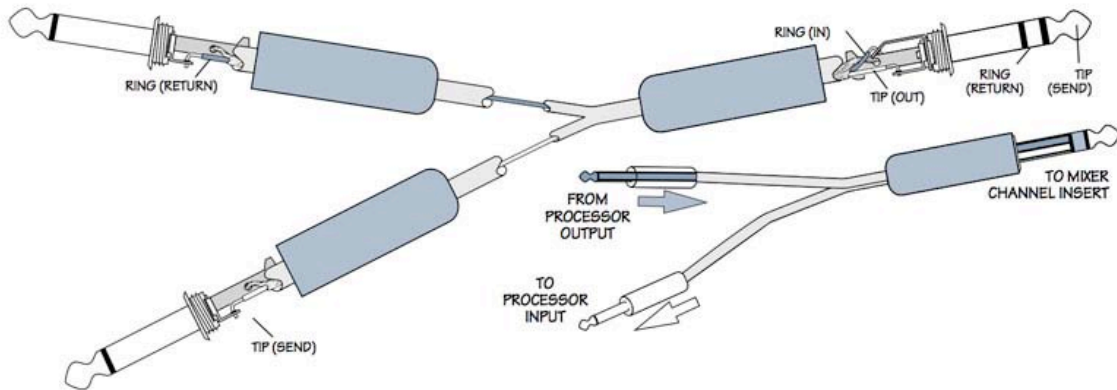
- Mini Stereo: 3 conductors used for 2 unbalanced channels



- Y or insert cable: 3 conductors used for 2 unbalanced signals



**Figure E: Insert Plug**



**Y-cord insert cable**

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- Banana
- Designed for amplified signals



- Speaker Wire

- Speakon
  - Designed for high-wattage, amplified signals



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## 8.15. Digital Cables: Types

- Always handle two or more channels per cable
- Unbalanced
- Balanced
- Fiber Optic

## 8.16. Digital Cables: Examples

- SPDIF (Coaxial): looks like RCA



- AES/EBU: looks like XLR
- Toslink (2 channel optical)





- ADAT/Lightpipe (8 channel optical)
- MADI (optical or coaxial up to 64 channels)

### **8.17. Snakes**

- Bundle cables in a single insulation



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## 8.18. Power: AC and DC, Phantom Power

- Alternating current (AC): 120 volts RMS in a 60 Hz sine wave
- Direct current (DC): not a sine wave
- Transformers: rectifies and smoothes AC into DC
- Phantom power: +48 Volt DC transmitted on +/- signal lines of a balanced cable

## 8.19. The Mixer and the Patchbay

- Mixer: signal control, processing, combination, and routing

- Combines fundamental tools used in almost every signal processing context
- Patchbay: signal routing
- Offers tools that have evolved into conceptual paradigms: may be hardware, may be software

## 8.20. The Mixer: Primary Components

- A mixer can be seen as having two primary components
- Channel strips
  - A number of commonly used routing and processing tools bundled together
  - Should be called a “track strip”: may be applied to one or more channels
  - Physical mixers are made of numerous (4, 12, 16, 32, 64) channel strips
- Busses
  - A signal destination (a repository that signals lead in to, output may go to another channel or physical output)
  - May be called mains or main bus, groups or sub-groups, or auxiliaries, aux sends, aux

## 8.21. Channel Strip: Basics

- Amplifiers, processors, and distributors (bus assignment)
- Common vertical orientation is not the same as signal flow

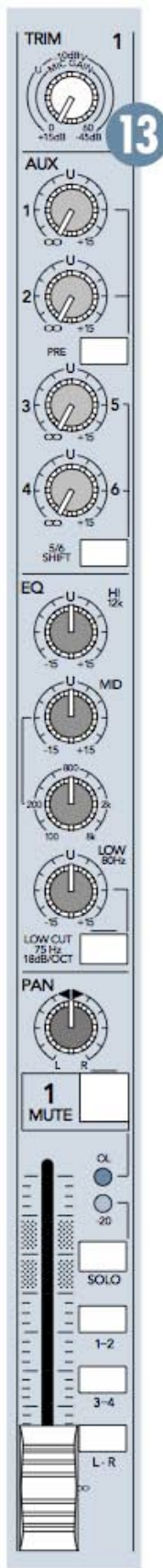
## 8.22. Channel Strip: Components

- Input or input selector
- Preamp, trim, line/mic level switch, pad, phase
- Insert: serial processing slot
- Low cut filter
- Auxiliary sends: for parallel processing or fader-controlled bus assignment
- Eq and dynamics (serial processors)
  - Shelves and parametric eq

- Dynamic effects such as compressors, limiters, gates, and expanders
- Mute and solo control
- Fader
- Panning and bus assignment
  - Bus assignments may be stereo or multichannel
  - May use panning to assign to one channel of a stereo bus

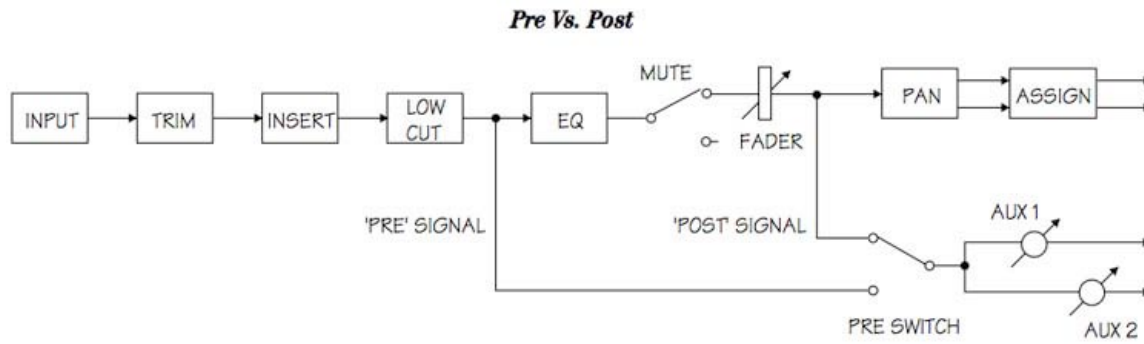
### **8.23. Channel Strip: Example: Mackie 1604**

- Vertical orientation is not the same as signal flow
- Channel strip



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- Signal flow



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## 8.24. Channel Strip: Example: Mackie Onyx 2408

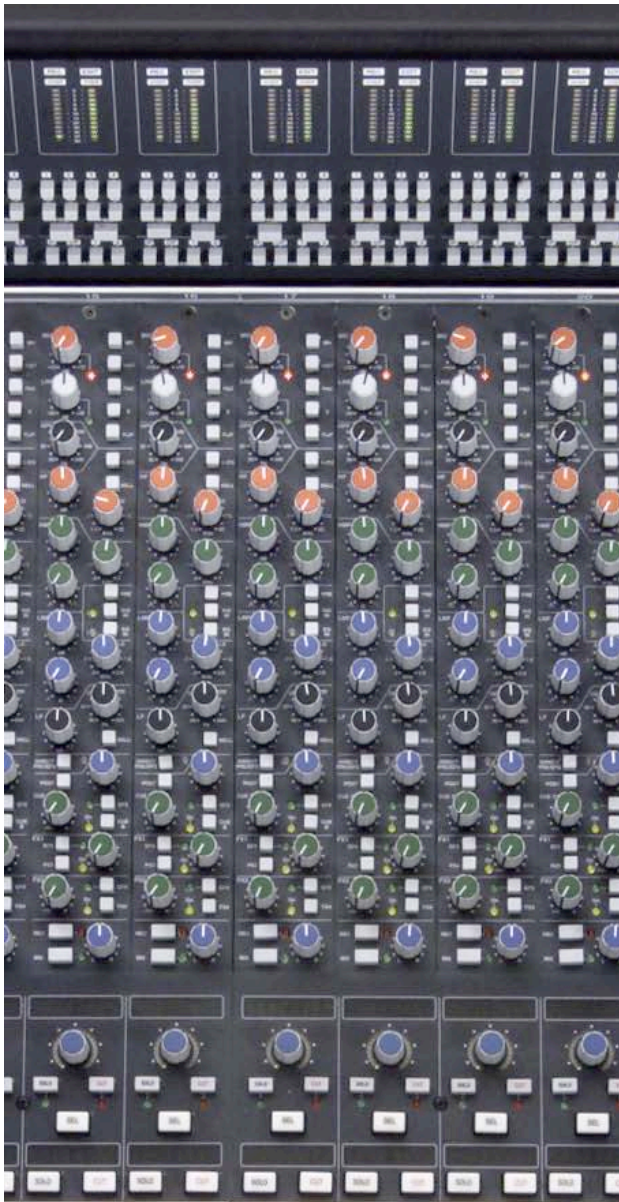
- Channel strip





## 8.25. Channel Strip: Example: SSL AWS 900

- Channel strip



## 8.26. Channel Strip: Example: SSL XLogic

- Channel strip



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## 8.27. Busses

- Channels may output to one or more bus
- Other channels may take a bus as an input
- Used for grouping and processing related channels
- Used for distributing sub-mixes to other processors or outputs

## 8.28. Busses: Main-Outs, Sub-Outs, Control Room

- Main Outs: final output destination to a physical output; may be stereo or multiple channel
- Sub Outs: busses to alternative physical outputs
- Control Room: a bus designed to deliver audio to the engineer, not the main outs

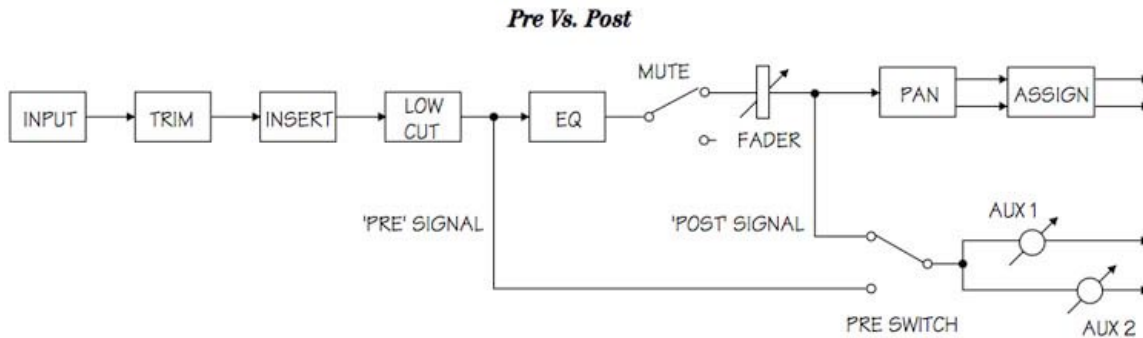
## 8.29. Busses: Grouping

- Assign a number of channels to a group channel
- Use the group channel for shared processing or fader control
- Then, assign the group to the main output

## 8.30. Busses: Auxiliaries

- Channel strip bus assignment with a rotary fader
- Used for creating a sub-mix different from the channel fader position
- On a physical mixer, physical output might be labeled auxiliary or auxiliary send
- On a virtual mixer, auxiliaries are tracks that receive a bus as input

- Used to provide a different mix to monitors or outboard processors
- Can be pre- or post-fader



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### 8.31. Patch Bay

- Expose all inputs and outputs in one place
- Can refer to a stand-alone device, or to the i/o section of a larger device
- Bring i/o from the rear of all devices to a front-panel interface
- Examples



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### **8.32. Patch Bay: Concepts**

- Vertical pairs matter
- Out over in: from front, outputs are represented on top, inputs are represented on bottom
- From rear: connect outputs from other devices on top; connect inputs to other devices on bottom
- Three common figurations: normal, half-normal, and de-normal

### **8.33. Patch Bay: Normal**

- A normal connection is a default connection that does not require a patch
- A normal connection flows from the rear top to the rear bottom; no front-panel patch is necessary
- Can be half normal or full normal: difference is what happens when a cable is inserted into the front top
  - half normal: inserting a cable into front top does not break the normal connection; the signal is sent two places at once
  - full normal: inserting a cable into front breaks the normal connection; the signal is sent one place (out the front top)

### **8.34. Patch Bay: Denormal/Open**

- What you see is what you get
- No internal normal connection; front simply connects to rear
- Outputs are still over inputs

### **8.35. Patch Bay: All formats**

-

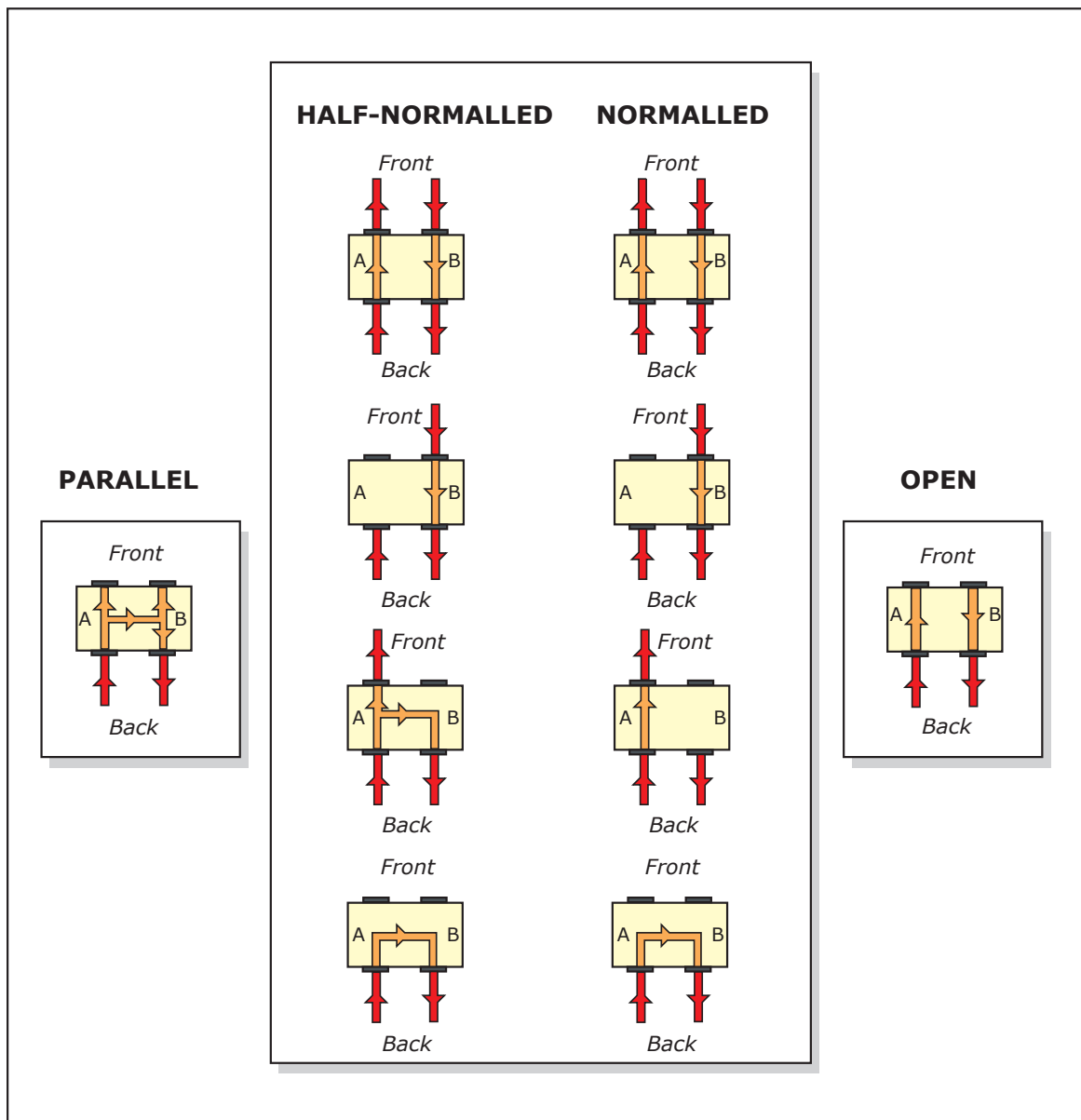
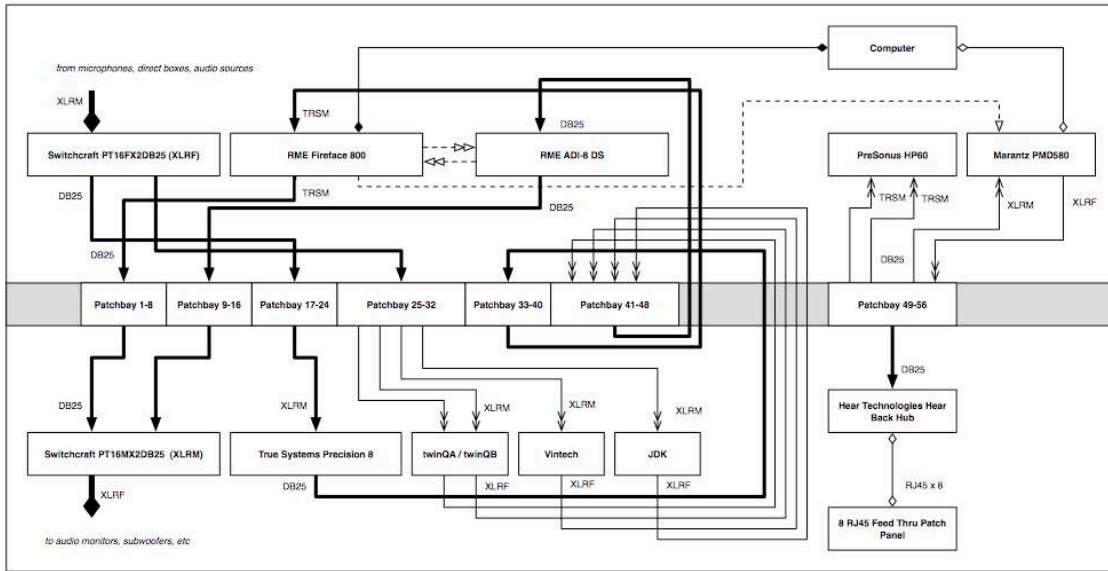


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### 8.36. MOSS: Diagrams

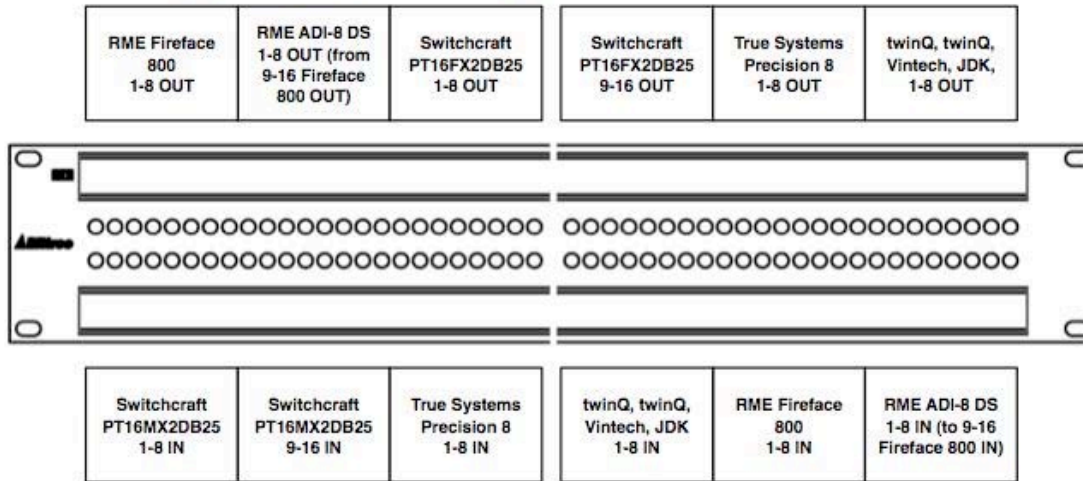
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### Internal Rack Connections

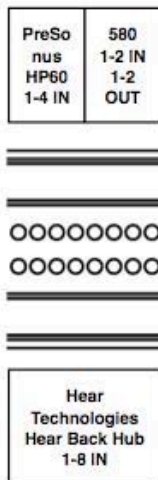


- 16 Channel balanced analog (external)
- 8 Channel balanced analog
- 2 Channel balanced analog
- 1 Channel balanced analog
- 8 Channel double wire digital Toslink
- 2 Channel digital SPDIF
- RJ45
- Firewire

**Primary Patchbay (Points 1-48): all half-normalled**



**Secondary Patchbay (Points 49-56): all isolated**



**8.37. Reading: Eargle: Chapter 3, The Pressure Microphone**

- How are capacitor pressure microphones affected by temperature?
- Is it possible that a microphone pad can change frequency response?
- What cable lengths does Eargle say are possible with a microphone and low capacitance cable?
- Which type of condenser might we expect to have a larger self-noise, a small or a large diaphragm?
- What are the advantages of using an electret material in the design of a capacitor microphone?



- How does a piezoelectric microphone work? What are some applications?

## Chapter 9. Meeting 9, Workshop: Preamps and Level Setting

### 9.1. Announcements

- Processing Report 1 due Today, 7 March  
Based on audioProcReport01.zip

### 9.2. Signal Flow

- Signal flow from mic to computer
  1. Mic front-panel XLR/F jacks
  2. Patch bay mic output (Switchcraft PT16FX2DB25 1-16)
  3. Patch bay Preamp inputs (True 1-8, TwinQ A1-2, TwinQ B1-2, Vintech 1-2, JDK 1-2)
  4. Patch bay Preamp outputs (True 1-8, TwinQ A1-2, TwinQ B1-2, Vintech 1-2, JDK 1-2)
  5. Patch bay computer input / RME IN (RME Fireface 800 / RME ADI-8)

### 9.3. Preamps

- True Systems Precision 8 (True)
  1. Controls: gain, phantom, phase
  2. Meter: dBu, with variable peak reference and hold
- Joemeek TwinQ
  1. Controls: full channel strip with preamp gain, phantom, eq, compression
  2. Meter: VU, with switch for gain reduction meter
- Vintech 1272
  1. Controls: input gain and output gain, phantom power
  2. Meter: 4-segment dBu
- JDK R20
  1. Controls: gain, phantom, pad, phase

2. Meter: VU

## 9.4. Procedure

1. Remove mic from case, attach mic to stand
2. Connect mic cable
3. Patch from mic to pre, pre to RME
4. Engage phantom power, level set
5. Clear levels, disengage phantom power
6. Unpatch
7. Disconnect mic and return to case
8. Wrap cable

## 9.5. Assignments

- Group A
  - Subject: [student name] on guitar
  - Order and assignments
    1. [student name]: Mic I/O 2, TwinQ A1, RME IN 2
    2. [student name]: Mic I/O 10, True 1, RME IN 10
    3. [student name]: Mic I/O 4, Vintech 1, RME IN 4
    4. [student name]: Mic I/O 12, JDK 1, RME IN 12
    5. [student name]: Mic I/O 6, TwinQ A2, RME IN 6
    6. [student name]: Mic I/O 14, True 2, RME IN 14
- Group B
  - Subject: Romi on piano
  - Order and assignments
    1. [student name]: Mic I/O 8, Vintech 2, RME IN 8

2. [student name]: Mic I/O 16, True 7, RME IN 16
  3. [student name]: Mic I/O 1, TwinQ B1, RME IN 1
  4. [student name]: Mic I/O 9, True 3, RME IN 9
  5. [student name]: Mic I/O 3, Vintech 2, RME IN 3
  6. [student name]: Mic I/O 11, True 8, RME IN 11
- Group C
    - Subject: [student name] on Sax
    - Order and assignments
      1. [student name]: Mic I/O 5, TwinQ B2, RME IN 5
      2. [student name]: Mic I/O 13, True 4, RME IN 13
      3. [student name]: Mic I/O 7, Vintech 1, RME IN 7
      4. [student name]: Mic I/O 15, True 5, RME IN 15
      5. [student name]: Mic I/O 2, TwinQ B1, RME IN 2
      6. [student name]: Mic I/O 10, True 6, RME IN 10
      7. [student name]: Mic I/O 4, Vintech 2, RME IN 4

# Chapter 10. Meeting 10, Compression and Limiting

## 10.1. Announcements

- Materials for second processing report will be out on Wednesday
- Next quiz on Monday, 19 March

## 10.2. Review Quiz 2

- ?

## 10.3. Preamps in MOSS

- True 8



- TwinQ



- Vintech

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## 10.6. Dynamics Processors: Input-Output Transformation

- Graph input amplitude to output amplitudes via transfer curve

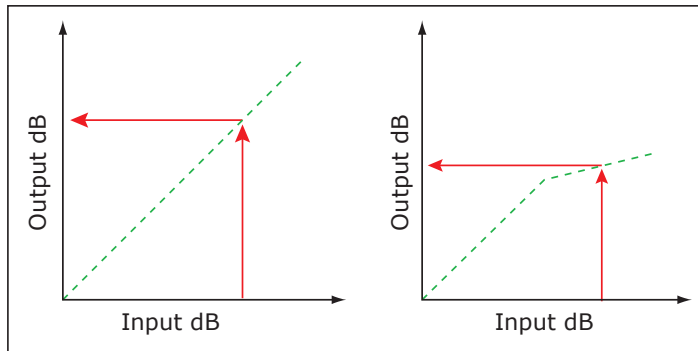


Image by MIT OpenCourseWare.

- A ratio of 1:1 is no change, or bypass
- A shifting the 45 degree line up or down is a boost or cut in amplitude

## 10.7. Dynamics Processors: Two Basic Families

- Processors that reduce amplitudes when amplitudes are above a threshold (downward compression and limiting)
- Processors that reduce amplitudes when amplitudes are below a threshold (downward expansion and gating)
- While amplitudes are reduced, this does not mean that dynamic effects only make sounds more quiet

## 10.8. Gain Reduction Above a Threshold: Compressor

- Reduces (compresses) dynamic range and increases average signal level
- Handles situations where a track needs to be turned up but cannot be turned up without clipping
- Often used to reduce the amplitude volatility of a signal: vocals
- Can raise level of quiet signals: can increase sustain, background, and ambience
- Can increase leakage and noise floor

## 10.9. Compression: Two Steps

- Two steps
  - 1. Reduce gain above a threshold with a ratio
  - 2. Increase gain of the modified signal
- Steps



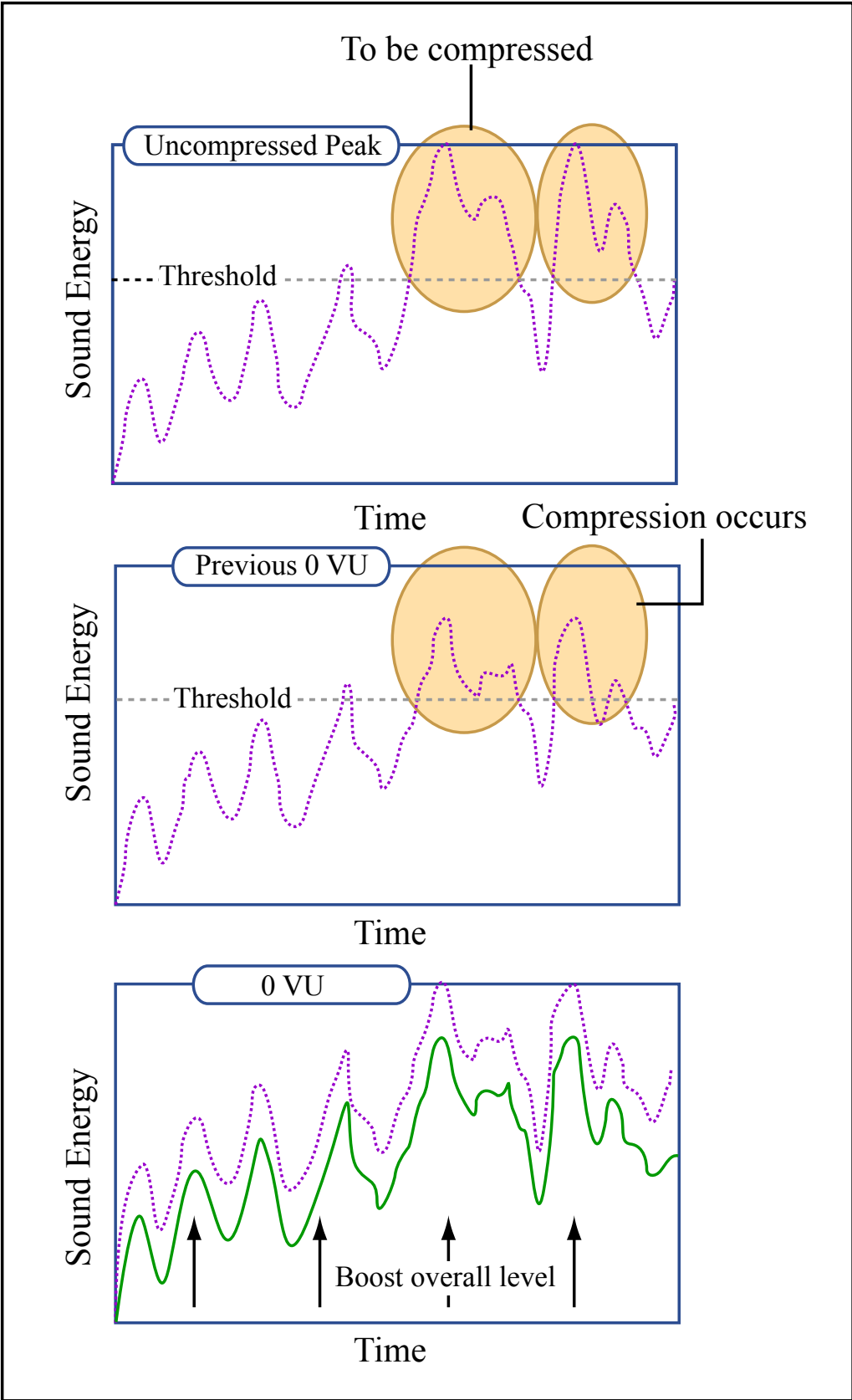


Image by MIT OpenCourseWare.

## 10.10. Compression: Ratio

- Ratio

Image removed due to copyright restrictions.  
"Compression with a 3:1 Ratio" from Gibson, B.  
*Microphones & Mixers*. Hal Leonard Corp., 2007.

## 10.11. Compression: Knees

- Hard and soft knee

Image removed due to copyright restrictions.  
"Hard Knee vs. Soft Knee Compression/Limiting" from Gibson, B.  
*Microphones & Mixers*. Hal Leonard Corp., 2007.

## **10.12. Compression: Attack and Release**

- Attack and release

Image removed due to copyright restrictions.  
"Using the Attack Time Setting to Control Understandability and Punch"  
from Gibson, B. *Microphones & Mixers*. Hal Leonard Corp., 2007.

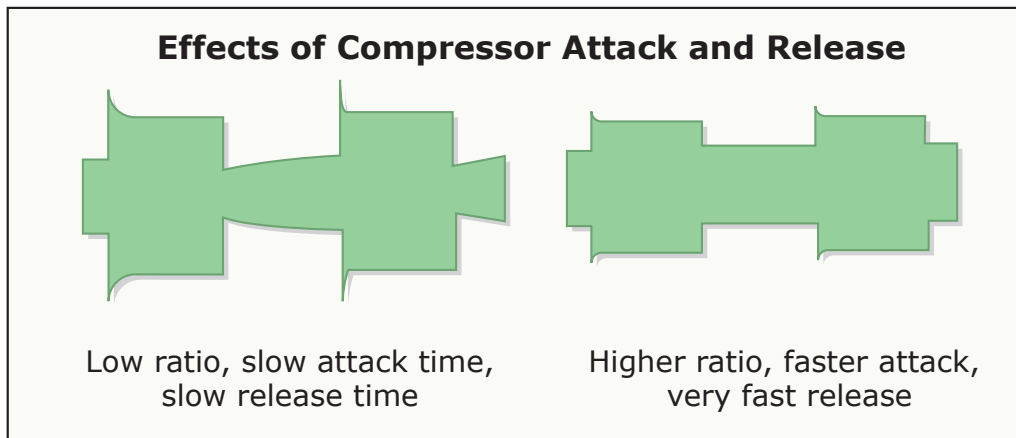


Image by MIT OpenCourseWare.

- Attack times generally around 20-50 ms
- Release times generally around 100-300 ms
- Slower attack times are critical for letting transients pass unaffected: this is often desirable
- Fast attack times can result in lifeless and unnatural percussion sounds
- Slower release times continue to reduce gain of sustain of instruments
- Pumping: attack and release are too fast and compression is audible; sustain of a signal fades in and out after attack of louder signals
- Breathing: hearing the noise floor slowly rise after the signal falls below threshold; remove by decreasing release time

### 10.13. Gain Reduction Above a Threshold: Limiter

- A compressor taken to an extreme ratio
- Ratios are in the range of 10:1 to infinity:1
- Flattens the top of amplitudes (generally) without distortion (depending on attack)
- Often used to protect equipment and limit dynamic ranges

### 10.14. Limiting: Example

- Example

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"Limiting" from Gibson, B.  
*Microphones & Mixers*. Hal Leonard Corp., 2007.

### **10.15. Reading: Katz: How to Manipulate Dynamic Range for Fun and Profit**

- What does Katz say should be the paradigm of sound quality? Why is this often not possible?
- Why was “popcorn noise” necessary for mastering audio for movies?
- According to Katz, what affect does hard-knee compression have?
- Why does Katz state that, in regard to attack and release times, its “probably better to remove all the labels on the knob (except slow and fast) and just listen!” ”

- How is lookahead implemented in digital compressor?
- What is a brick-wall limiter?
- How does Katz describe the release characteristics of an opto-compressor?
- What is soft clipping, as found as a feature on some digital processors?

# Chapter 11. Meeting 11, Expansion, Gating, and Sidechaining

## 11.1. Announcements

- Quiz on Monday
- Audio materials for Processing Report 2 (due 21 March):  
audioProcReport02.zip

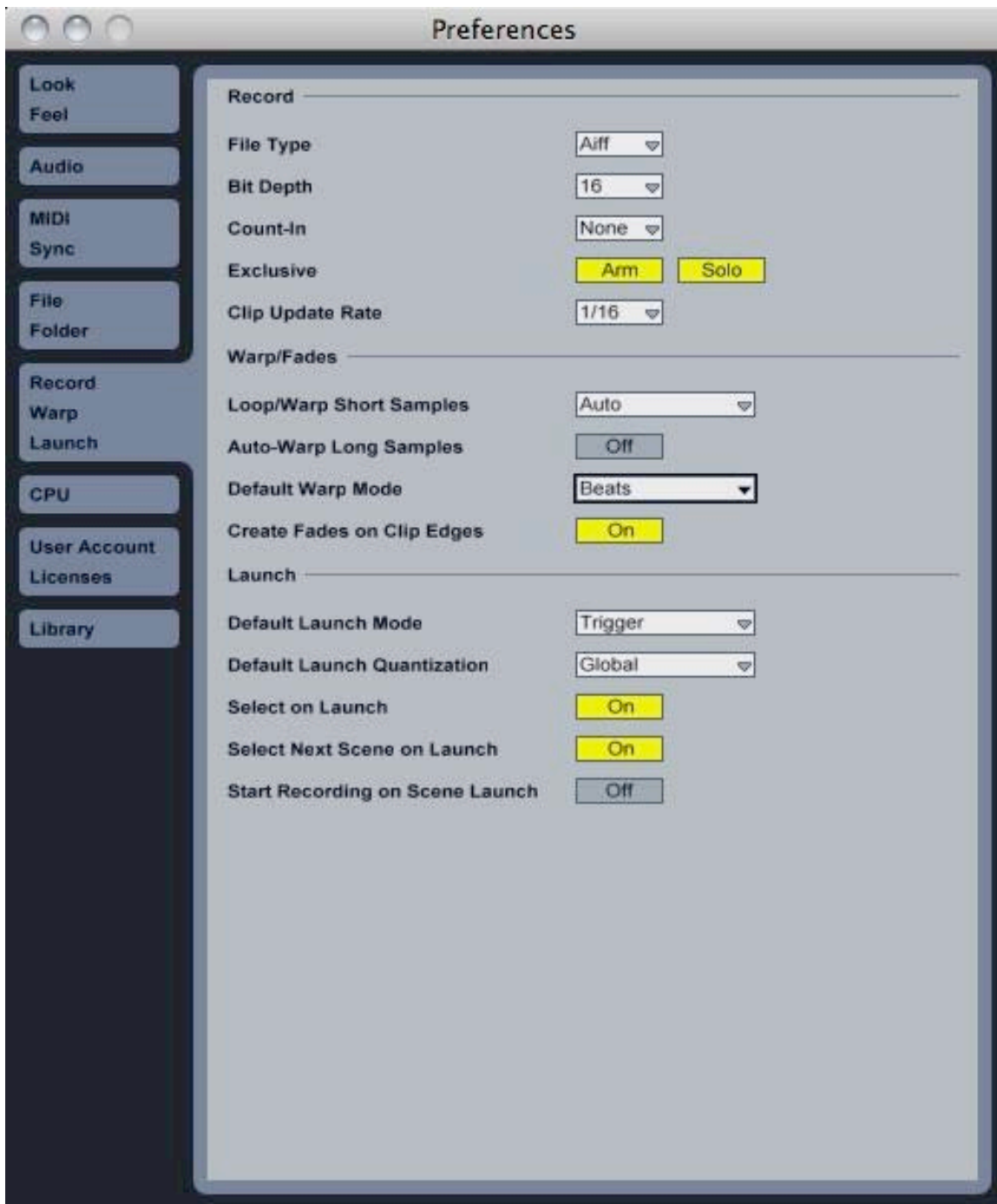
## 11.2. Processing Report 1 Review

- You must specify what DAW and processors/plugin-ins you are using
- You must specify specifics for all relevant parameters
- Describe what you are trying to do
- Audition files before and after processing
- Remove leakage and noise, and find opportunities for improving timbre

## 11.3. Disabling Live's Warp Feature

- Live may automatically attempt to shift the duration (independent of pitch) of a track: we never want this!
- Turn off “Auto-Warp Long Samples” in Warp preferences





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- Always add track in the horizontal view (toggle in upper right corner)
- Double click on sample, view “Clip” and “Sample” controls, and make sure that Warp is *not* selected



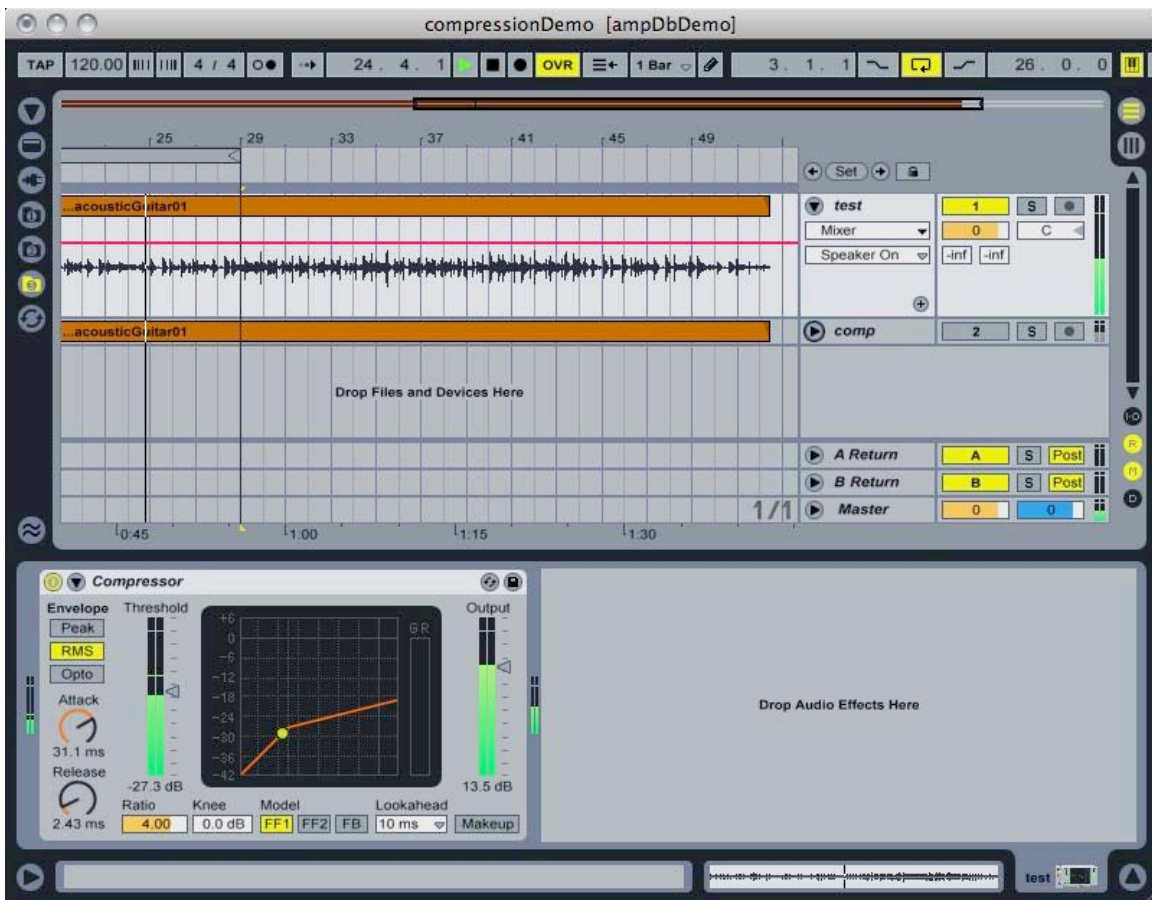
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## 11.4. Compression and Filters

- Dynamic effects and filters are the core of channel-strip processing
- Filters can come before and / or after compression
- If removing gain with a filter, compression best comes after
- If adding gain with a filter, compression best comes before

## 11.5. Compression and Limiting in Live

- Compressor



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- Compressor (one plug-in)
  - Threshold and ratio, knee in units of dB
  - Output (use) and Makeup (do not use)
  - Attack and release, lookahead
  - Envelope follower (triggering) mode: peak, RMS, opto (non-linear release curve: release is faster initially, and slows as GR approaches zero)
  - Feed forward: FF1 and FF2 analyze loudness of incoming signal; FB analyzes output
    - Feed back: FB analyzes output, self-adjusts behavior, results in a smoother sound; lookahead is disabled
- Examples: acousticGuitar01, bassAmp01

## 11.6. Compressors, Peak Limiters, Limiting Amplifiers

- Regular compressors: variable attack, release, ratio, and threshold

- Peak limiters: fast attack, high ratio, high threshold

May be used after compressor

- Leveling amplifiers: medium attack time, a medium to slow release time, a high ratio and a low threshold

Holding the level down in a smooth way

Ratio and threshold may be linked

Early analog tube models were very popular



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## 11.7. Analog Gain Reduction Technologies

- Optical isolators: light and photocell creates inherent lag time

Example: LA-2A



- Field effect transistor (FET): used to more closely emulate tubes; extremely fast and reliable  
 Example: UA 1176 LN (attack 20 microseconds to 800 microseconds, release 50 ms to 1.1 s)



- Voltage controlled amplifier (VCA): versatile, most control  
 Example: DBX 160: the first solid state VCA compressor



## 11.8. Compression in MOSS

- TwinQ optical compressor



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## 11.9. Gain Reduction Below a Threshold: Expander

- Where a compressor reduces dynamic range, an expander increases dynamic range
- Where a (downward) compressor operates above a threshold, (downward) expanders operate below a threshold
- Both reduce gain using a ratio
- Both may have attack and release controls
- Compressors take two steps: expanders take one

## 11.10. Expanders: Applications

- Increase dynamics in a performance
- Reduce or eliminate leakage, reverb, or noise floor

## 11.11. Gain Reduction Below a Threshold: Gate

- An expander taken to an extreme gain-reduction ratio
- Ratios are in the range of 10:1 to infinity:1
- Ratios can be thought of in negative values (below the threshold): -1:-2
- Often used for noise reduction and noise silencing
- Gates

Image removed due to copyright restrictions.  
"The Gate/Expander" from Gibson, B.  
*Microphones & Mixers*. Hal Leonard Corp., 2007.

## 11.12. The Family of Dynamic Effects

- Four (downward) processors

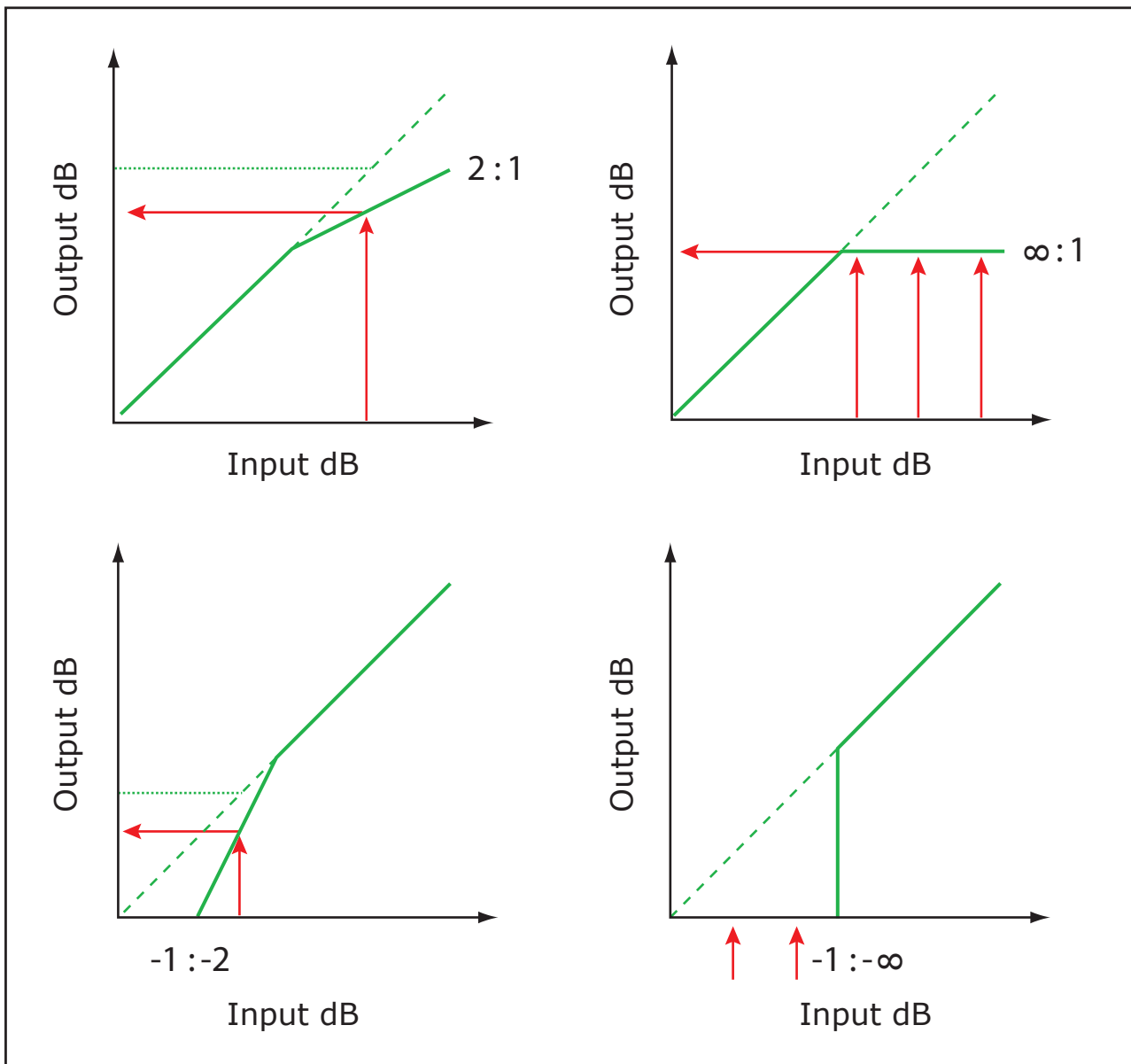


Image by MIT OpenCourseWare.

### 11.13. I/O Map Exercise

- Diagram a compressor with a threshold at -10 dB and a ratio of 2:1
- Diagram a limiter with a threshold at -8 dB and a ratio of 10 to 1
- Diagram an expander with a threshold at -10 dB and a ratio of 2 to 1
- Diagram a gate with a threshold at -16 dB and a ratio of infinity to 1
- Diagram a compressor with a threshold at -6 dB and a ratio of 3:1
- Diagram an expander with a threshold at -20 dB and a ratio of 6 to 1
- Diagram a compressor with a threshold at -8 dB and a ratio of 4:1



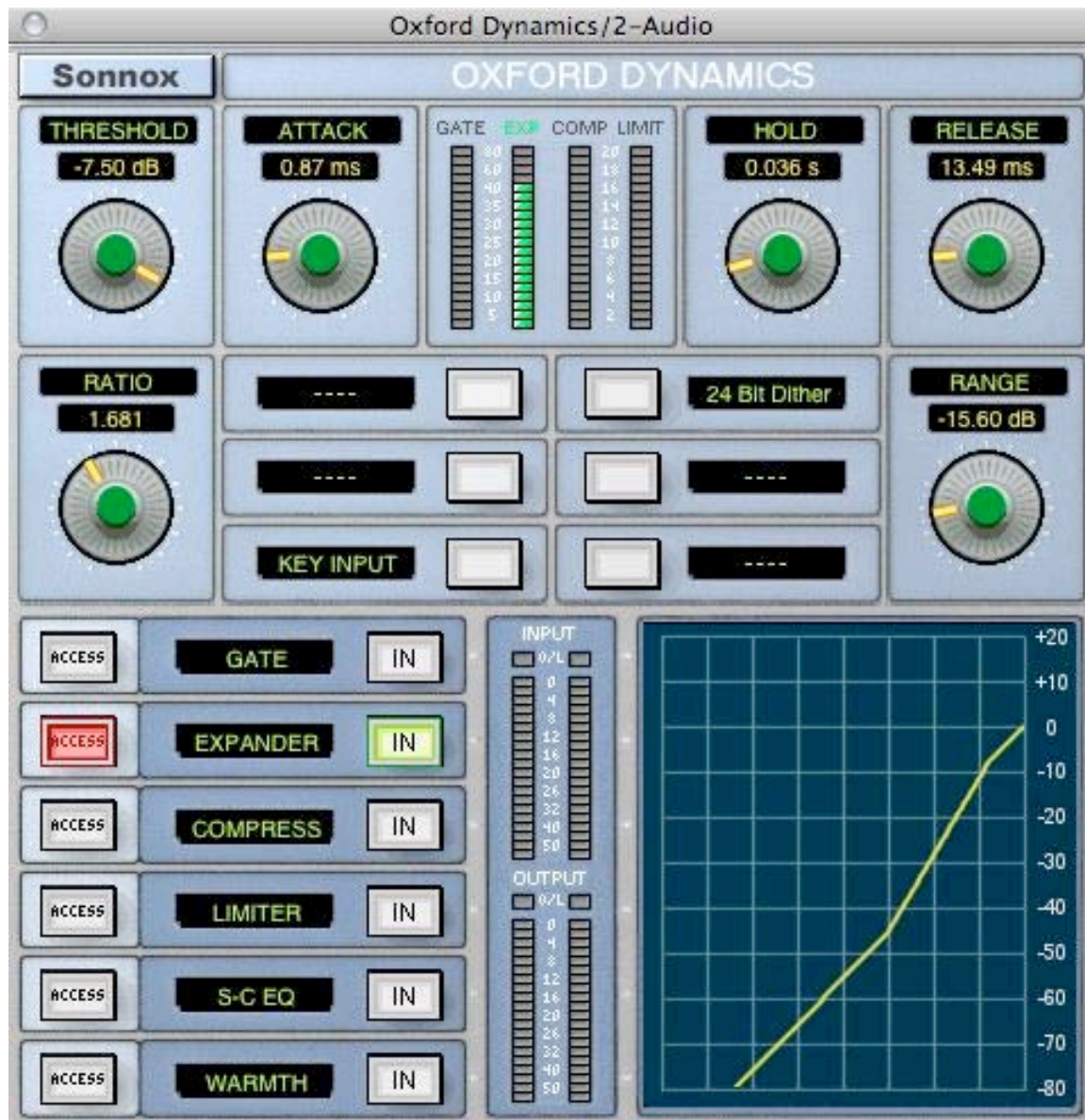
## 11.14. Compression and Limiting in Live

- Gate



Courtesy of Ableton AG. Used with permission.

- Gate (one plug-in)
  - Threshold and attenuation (how much gain is reduced)
  - Attack (time to open) and release (time to close, after hold)
  - Hold (time gate remains opened after cross threshold)
- Examples: epiano01
- More robust expanders can be used to shape internal dynamics



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- Examples: sanshin01

## 11.15. Sidechaining

- The sidechain is the signal used to trigger the compressor
- Amplitude characteristics of one signal can be used to process the amplitude of a different signal
- Compressor in this case is used without makeup gain

## 11.16. Sidechaining: Ducking

- Problem: lower the level of a music track when a spoken voice or musical part enters
- Sidechain is spoken voice or track that needs to be on top of mix (e.g., kick)
- Compressor is used without makeup gain
- Ducking in Live



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## 11.17. Sidechaining: Deessing

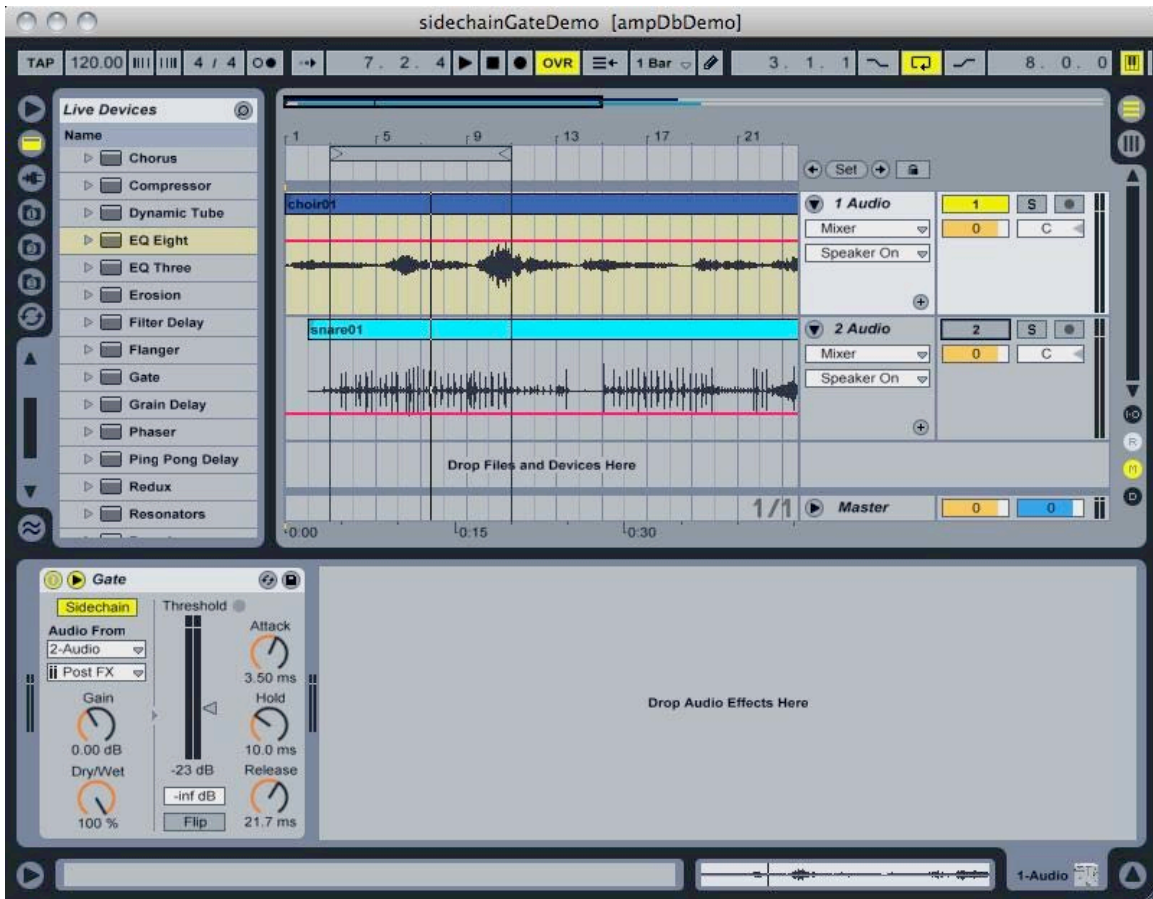
- Problem: vocals s and th sounds (4 to 6 kHz) produce peaks and create extreme presence
- Sidechain signal is filtered version of the source, boosting the problematic frequencies
- Forces the compressor to react more strongly to amplitude activity in that frequency region
- Deessing in Live



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## 11.18. Sidechaining: Gating

- Can use a control track (filtered and processed as necessary) to open and close (gate) on another track
- Sidechain Gating in Live



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## 11.19. Processing Report 2

- Process each of the four audio tracks to remove noise, remove leakage, increase isolation, balance timbre, balance dynamics, and optimize mixing potential. Only filter/equalization and dynamic processors can be used.
- For each file, in prose, describe the problems with the audio file and what you are trying to do. Then, describe the specific processors you are using, as well as the specific filter settings used. Be sure to include units for all parameters.
- Send back to me the processed audio files (export or bounce out of the DAW) as well as the written report.

## 11.20. Reading: Eargle: The Pressure Gradient Microphone

- What polar pattern is found in pressure gradient microphones?
- Why is the polarity of signal entering the back of the ribbon reversed?
- Which has more LF response, figure-8 ribbon or a figure-8 capacitor? Why?

- What is the proximity effect?

## Chapter 12. Meeting 12, Approaching a Mix

### 12.1. Announcements

- Audio materials for Processing Report 2 (due Friday 23 March):  
audioProcReport02.zip
- Mix Report 1 Due Monday 9 April

### 12.2. Quiz 3

- ?

### 12.3. Static, Dynamic, Non-Linear Mixes, and Automation

- A static mix means that fader and panning positions are relatively fixed
- A dynamic mix alters fader and panning positions (and other parameters) during the mix
- Hardware mixers developed ways of recording the movement of faders or knobs: became known as automation
- In DAWs, automation is dynamic parameter data
- Non-linear mixing changes the temporal arrangement of components

### 12.4. Destructive and Non-Destructive Audio Editors and Editing

- Destructive: what you see is what you get
- Destructive: Examples: Peak, Audacity
- Non-Destructive: what you see is one representation of what you have
- Non-Destructive systems are general Digital Audio Workstations (DAWs)
- Non-Destructive: Examples: Pro Tools, Digital Performer, Audacity, Sonar, Cubase
- Non-destructive non-linear editing in Live
- Adjusting the grid in Live

## 12.5. Importing Audio

- Non-Destructive editors provide representations of associated sound files and regions
- Source audio files may or may not be in the project
- Linking versus copying: always copy if possible

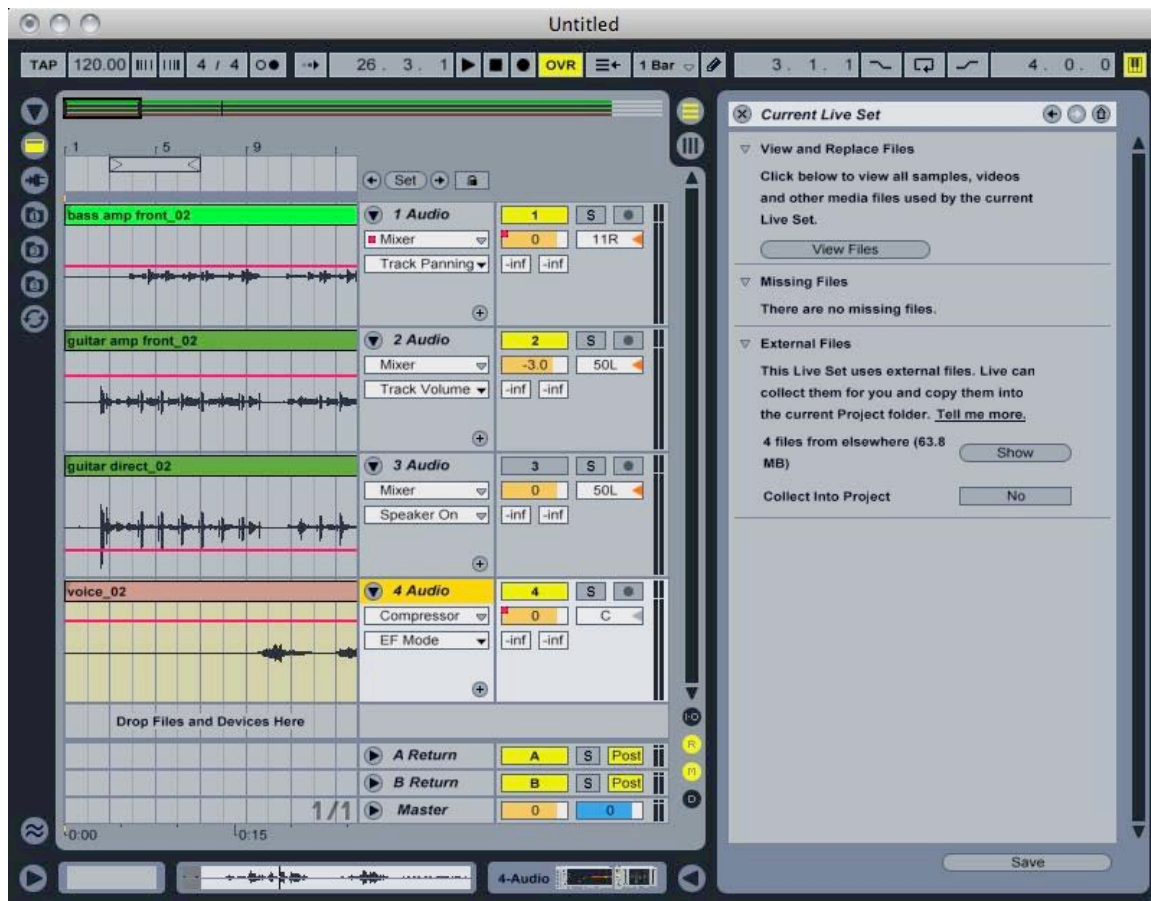
## 12.6. DAW Projects

- Projects are a folder that contain many components
- The project file is small: it only contains control and parameter information
- The project contains subdirectories for audio files and other data
- It is critical to keep all project components together

## 12.7. File Management in Live

- Audio files in Live are not copied to the project unless explicitly forced to be
- The File / Manage Files menu item shows a dialog to Collect Into Project





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## 12.8. DAWs: Tracks

- Many types: audio (mono, stereo, surround), MIDI, auxiliary, master
- Input and output assignments may be busses or physical i/o
- Tracks store information about audio files and how to play them back
- Tracks store multiple parallel parameters that change over time
- Managing automation data in Live

## 12.9. Master Tracks

- Provides a final level control of mix
- Will generally require 3 to 6 dB (or more) gain reduction below unity
- May do 1 to 3 dB (no more) of limiting on master track is sometimes necessary

- Do not compress or do other processing on the master track
- Adding processing to the master track in Live

## 12.10. DAWs: Plugins

- Plugins are serial (insert) processors
- Each plug-in requires hardware processing power
- Always conserve plug-in use as necessary
- Using plug-ins for parallel processing requires an auxiliary track with an insert

## 12.11. Mixing Procedure

- Be sure to time align tracks at beginning
- Can crop tracks at beginning and end
- Listen to each track alone and process
  - Apply channel strip processing
  - Apply fades to remove tacet portions, control start and end positions
  - Set basic pan positions
- Mix groups of instruments organized by microphone capture, ensemble role, or other factors
- Start with loudest instrument and mix downward

## 12.12. Channel Strip Processing

- Optimize each channel or bus-group while maintaining gain staging
- Use filters to isolate necessary frequencies
- Use dynamic effects to remove leakage
- Use moderate to deep compression to raise average level
- Use shallow limiting to control extreme dynamics

### 12.13. Panning Stereo Sources

- If coincident or near coincident, generally pan hard left and hard right
- If not coincident, may explore mixture
- Listen to mono mix to check for phasing distortion
- If combining near and far captures, must pan close microphones to match distant stereo positions

### 12.14. Panning Mono Sources

- Generally avoid 100% hard-panning (95% is a good maximum)
- Low frequencies (bass, kick) are generally toward the center
- Vocals are generally toward the center
- Avoid center build-up with slight offsets out of center (+/- 5%)
- Time keepers (high hat, ride, snare) are often off-center
- Often similar musical roles are balanced left and right (guitars, keyboards)
- Often aim for overall left-right balance

### 12.15. Levels

- Generally 1 dB is the smallest amount of perceivable change
- Always avoid channel, bus, or master clipping
- Amplitudes are relative: find ways to cut rather than boost
- May need to adjust levels by musical sections (boost for a solo)

### 12.16. Timing Offsets

- Measuring distances and calculating millisecond offset
- Use a delay plugin to push back closer microphones
- Listen and adjust: perfect alignment is not required

## 12.17. Double Tracking

- Using two copies of the same audio file panned hard left and hard right
- Hard panning helps removing potential phasing problems
- Delay a second copy by less than 30 ms
- Delay processor must be at 100%
- Alternatively, can use two similar takes of the same material

## 12.18. Bussing: Idiomatic to the DAW

- DAWs can support huge numbers of buses
- Bus design promotes sensible organization of tracks
- Bus design promotes computational efficiency

## 12.19. Bussing: Multiple Captures of the Same Source

- Group multiple microphones of the same source
- Example: drum kits
- Example: pianos
- Example: guitars, amplifiers, direct signals

## 12.20. Bussing: Multiple Instruments Performing Related Roles

- Grouping musical parts or sections
- Example: rhythm sections
- Example: background vocals
- Example: multiple vocal parts

## 12.21. Mixing and Listening

- Listen at multiple output volumes
- Listen on multiple playback devices and headphones

- Take breaks

## 12.22. Case Study: Mackie 1604 VLZ

- One of the most popular small mixers
- Complete manual in Blackboard course documents
- Small collection of everything you need



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## Chapter 13. Meeting 13, Microphones, Directionality, and Monophonic Microphone Techniques

### 13.1. Announcements

- Audio materials for Processing Report 2 (due Friday 23 March):  
audioProcReport02.zip
- Mix Report 1 Due Monday 9 April

### 13.2. Review Quiz 3

- ?

### 13.3. Mix Report 1

- Complete two mixes of two different multi-track studio recordings  
Only one mix can use extensive non-linear editing
- Perform channel strip processing on all channels using only filters and dynamic effects
- Automate only pan and levels
- Bounce a properly trimmed stereo file that has no clipping
- Report requires complete details on all tracks

### 13.4. Mix Materials for Mix Report 1

- C: Jazz quartet  
mix01-c-jazz.zip
- D: Trio of voice and two guitars  
mix01-d-28voxGtr.zip
- E: Duo of voice and percussion  
[file not available for OCW]

- F: Duo of voice and piano

mix01-f-46voxPno.zip

- A: Shimauta

[file not available for OCW]

- G: NIN

[file not available for OCW]

### **13.5. Transducers and Transduction**

- Transduction: conversion of one form of (sound) energy to another form
- Microphones and Speakers
- Transducers always act as a filter
- A frequency domain graph (frequency response curve) is used to show the effect of transduction

### **13.6. Microphones: Numerical Specifications**

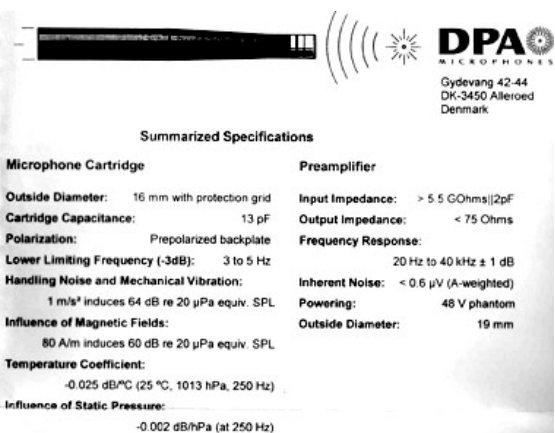
- Frequency response curves
- Transient response
- Self-noise
  1. Identify the microphones internal noise floor
- Sensitivity
  1. Given as negative dB: -57 dB
  2. Amount of boost required to raise input to 0 dBu
  3. A higher number means a more sensitive microphone
- Maximum SPL
- DPA 4006



# Calibration Chart for Studio Microphone

Type: 4006 Serial no.: 2105648 Matched to: 2105649

Versatile Omnidirectional Microphone  
Phantom P48 powered (DIN 45 596)

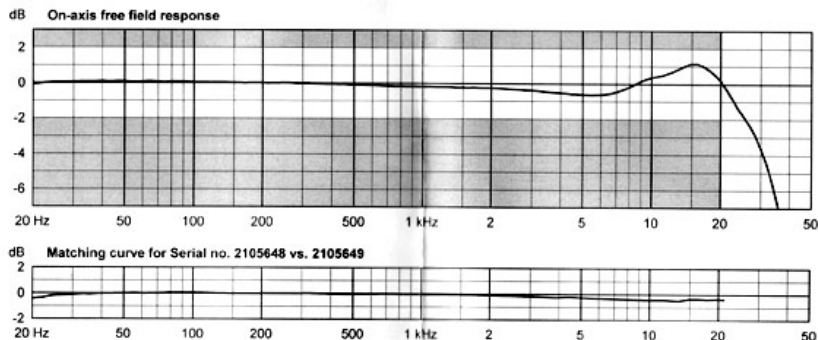


**Summarized Specifications**

Microphone Cartridge		Preamplifier	
Outside Diameter:	16 mm with protection grid	Input Impedance:	> 5 GOhms  2pF
Cartridge Capacitance:	13 pF	Output Impedance:	< 75 Ohms
Polarization:	Prepolarized backplate	Frequency Response:	20 Hz to 40 kHz ± 1 dB
Lower Limiting Frequency (-3dB):	3 to 5 Hz	Inherent Noise:	< 0.6 µV (A-weighted)
Handling Noise and Mechanical Vibration:	1 m/s <sup>2</sup> induces 64 dB re 20 µPa equiv. SPL	Powering:	48 V phantom
Influence of Magnetic Fields:	80 A/m induces 60 dB re 20 µPa equiv. SPL	Outside Diameter:	19 mm
Temperature Coefficient:	-0.025 dB/°C (25 °C, 1013 hPa, 250 Hz)		
Influence of Static Pressure:	-0.002 dB/hPa (at 250 Hz)		

Calibration Data		Calibration Conditions	
Frequency Range:	20 Hz to 20 kHz ±2dB	Barometric Pressure:	1024 hPa
Sensitivity (at 250 Hz):	10,6 mV/Pa	Ambient Temperature:	24 °C
Polarity:	+ V at pin 2 for positive sound pressure	Relative Humidity:	36 %
Equivalent Noise Level:	16 dB(A) re 20 µPa	Date:	24. september 2002
Dynamic Range:	>120 dB	Signature:	<i>Edvin Jensen</i>
< 1% THD at	135 dB re 20 µPa		
< 1% Diff. Freq. Distortion at	135 dB re 20 µPa		

1 Pa = 1 N/m<sup>2</sup> = 10 dynes/cm<sup>2</sup> = 10 µbar = 94 dB SPL (re 20 µPa)



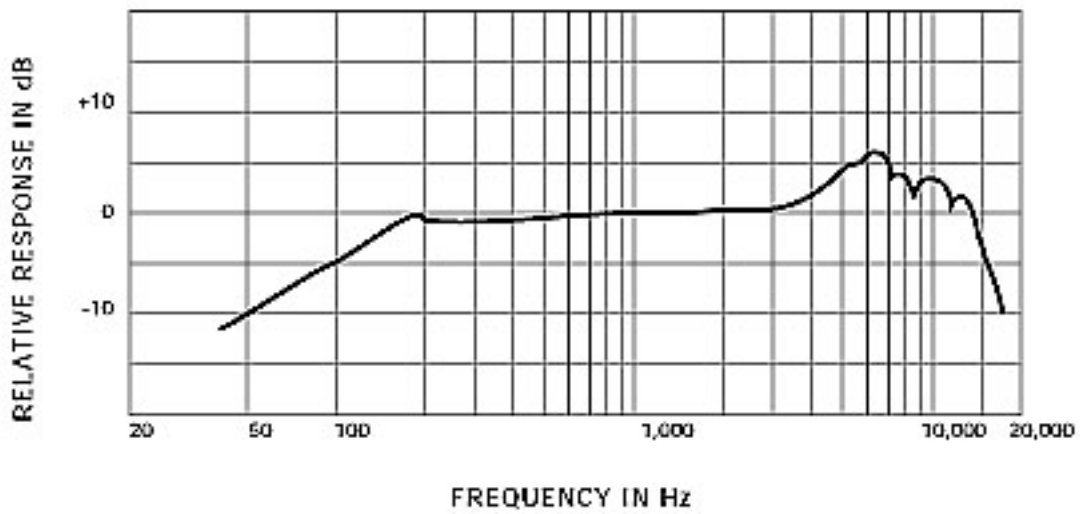
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## 13.7. Visualizing the Affect of Transduction: Examples

- Shure SM-57



Shure SM57

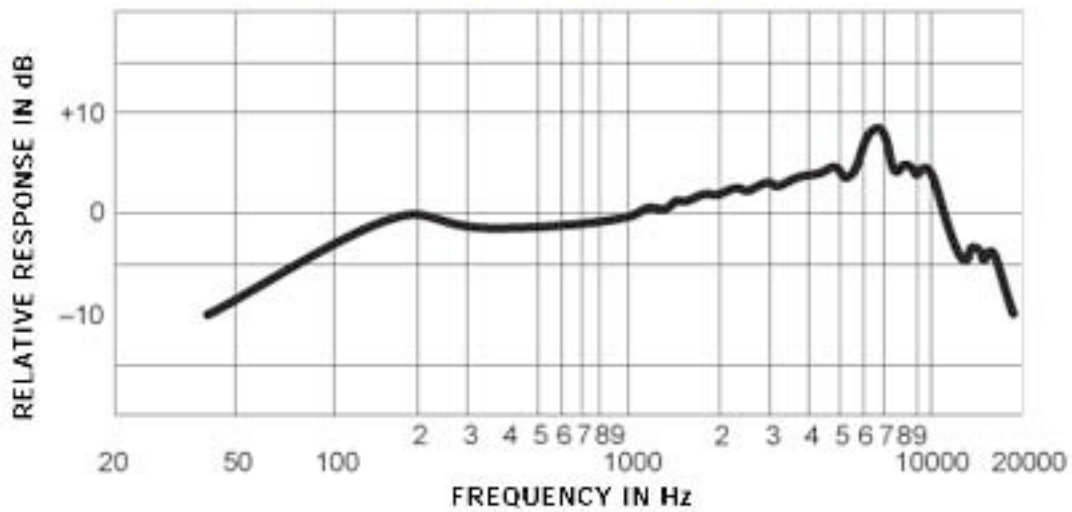


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- Shure 55SH



Shure 55SH Series II



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## 13.8. Microphones

- First stage of transduction
- Permanently alters the sound of the source
- Primary considerations: microphone type, microphone position, acoustical environment

## 13.9. Microphones: Directional Response

- Microphones pick up sound in various patterns (due to pressure or pressure gradient)
- Called polar pattern, pickup pattern, or directional response
- Microphones have a “front” or primary point of address, called on-axis
- Degrees are used to describe off-axis position (reverse is 180 degrees off-axis)
- Pickup patterns are in expanding three-dimensional spaces
- Different pickup patterns have different directional “pull” (sensitivity, or directional response)

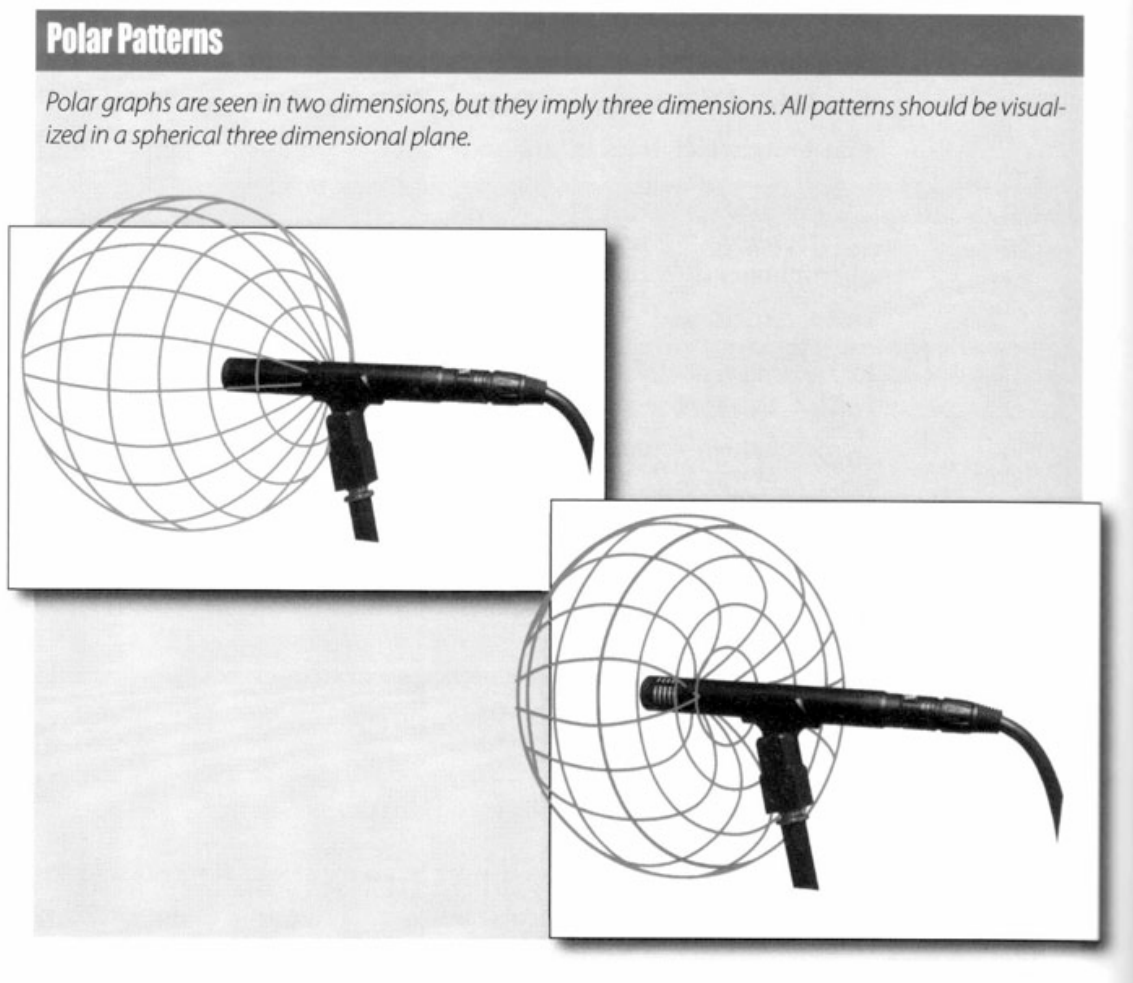
## 13.10. Microphones: Directional Response Types

- Omnidirectional
  1. Gather sound from all around
  2. Called an “omni”
  3. Useful for gather reflections and space of a sound
  4. Not considered a “directional” microphone
  5. No proximity effect
- Bidirectional
  1. Gather sound from two sides
  2. Called a “figure-eight”
  3. Useful for complete side rejection and rejection
  4. Useful for capturing reverse reflections
  5. Useful for getting two sources into one channel
  6. Useful for the sides of a mid/side stereo recording
  7. Common polarity of ribbon microphones (pressure gradient)
  8. Proximity effect
- Unidirectional
  1. Gather sound from one primary direction

2. Useful for focusing in on a singular sound source
  3. Various types of cardioids: reject sound from the rear
  4. Proximity effect
- Some microphones have variable patterns with switches or interchangeable capsules

### 13.11. Directional Response in 2D and 3D

- Three dimensional presentation

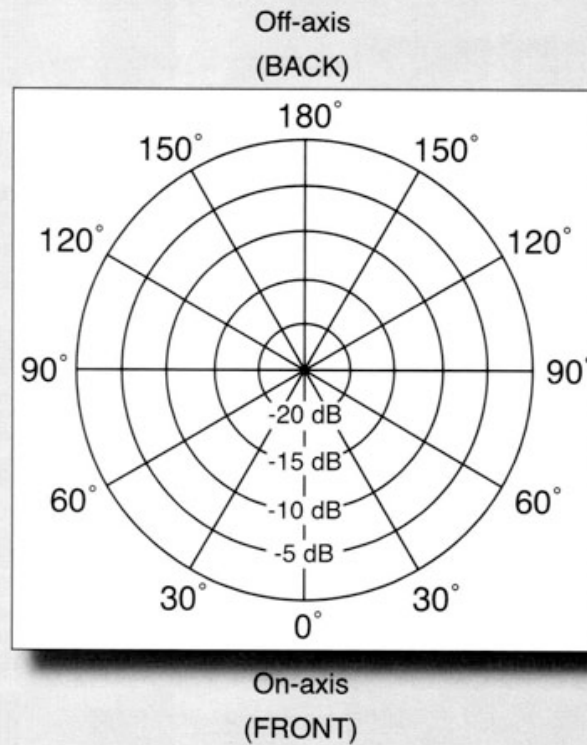


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Source: Gibson, B. *Microphones & Mixers*. 2007.

- Two dimensional presentation

## Polar Response Graph

The polar response graph plots the spacial sensitivity as it relates to the position of the sound source in relation to the microphone capsule. These graphs are considered symmetrical in relation to the plotted sensitivity and, in addition, should be considered three-dimensionally spherical.

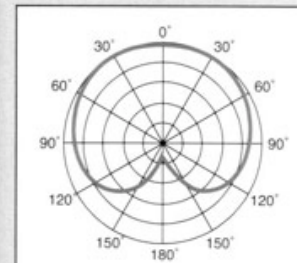
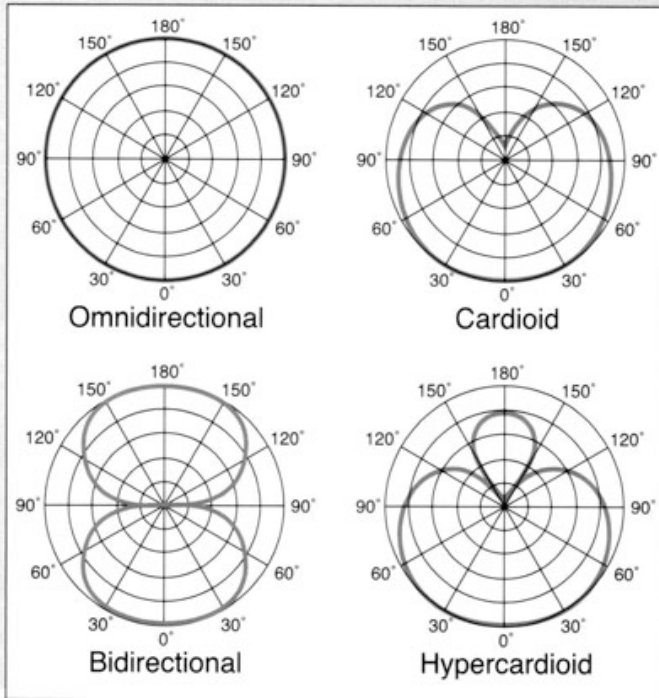


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Source: Gibson, B. *Microphones & Mixers*. 2007.

- Cardioids in two dimensions

## Polar Shapes

The two most basic polar response patterns are omnidirectional (doesn't discriminate against sounds from any direction) and cardioid (discriminates against sound that are 180 degrees off-axis). The other two polar shapes in this illustration are bidirectional (an omnidirectional pattern on each side off the mic) and hypercardioid (a bidirectional pattern with a large half and a small half).



Sometimes the on-axis position is noted at the top of the graph; other times it's noted at the bottom. In either case, the 0° position is always the front of the mic.

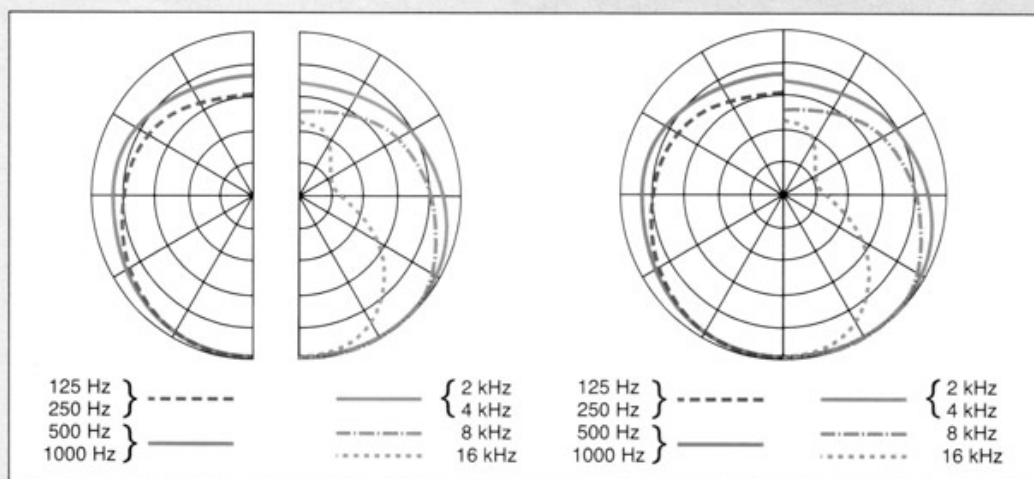
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Source: Gibson, B. *Microphones & Mixers*. 2007.

## 13.12. Directional Response: Frequency Dependence

- Directional response is not the same for all frequencies

## Multiple Frequencies and Symmetry on the Polar Graph

The polar graph often displays the directional characteristic for multiple frequencies. To accomplish this in the least cluttered manner, all patterns are assumed to be symmetrical across the Y axis. In addition, to help clarify the results, various line styles are incorporated on each frequency. Sometimes the polar graph is split, like the graph on the left, to highlight the variations in frequency response; other times the graph is whole, like the graph on the right, with the pattern variations simply changing between left and right.



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Source: Gibson, B. *Microphones & Mixers*. 2007.

### 13.13. Directional Response: Characteristics of Cardioids

- Directional response summarized

Key value is the distance factor



Image removed due to copyright restrictions.  
Characteristics of first-order cardioid microphones, Figure 5-4, in  
Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

- A greater distance factor means a greater directional pull
- Equal-amplitude distance chart

Image removed due to copyright restrictions.

Distance factor illustration for first-order cardioid microphones, Figure 5-5, in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

### **13.14. Proximity Effect**

- Bass frequencies are exaggerated when very close to directional (cardioid or figure-eight) microphones
- Low cut filters are often provided on microphones to mitigate

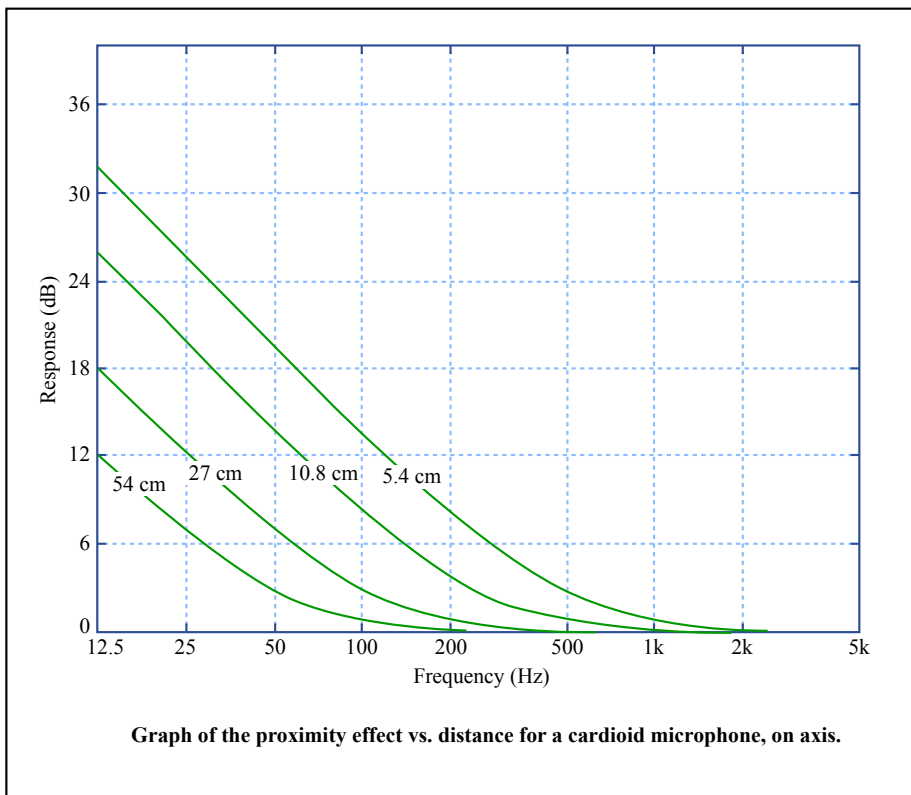


Image by MIT OpenCourseWare.

### 13.15. Microphone Parts and Species

- Diaphragm
  - Large: greater than a few centimeters
  - Small
  - Smaller diaphragms have less off-axis coloration
- Capsule: contains diaphragm as well as mount and possibly a pre-amp
- Transduction Method
  - Magnetic Induction
  - Variable Capacitance
- Transducer Type
  - Condenser (Variable Capacitance)
  - Moving Coil or Dynamic (Magnetic Induction)
  - Ribbon (Magnetic Induction)

### **13.16. Transduction Methods: Magnetic Induction**

- Electromagnetic force
- Moving metal in a magnetic field produces voltages
- Induce a voltage with a magnet
- Used in ribbon and dynamic mics
- Do not require power to operate

### **13.17. Transduction Methods: Variable Capacitance**

- Electrostatic force
- Two closely-spaced, parallel plates: one fixed, one acts as a diaphragm
- Stored charge, between plates, varies due to acoustical pressure
- Requires power to charge plates (usuall 48 V phantom power)
- Output is very small small; must be amplified in microphone

### **13.18. Transducer Type: Dynamic**

- Metal is a coil attached to a diaphragm that moves within a magnetic field

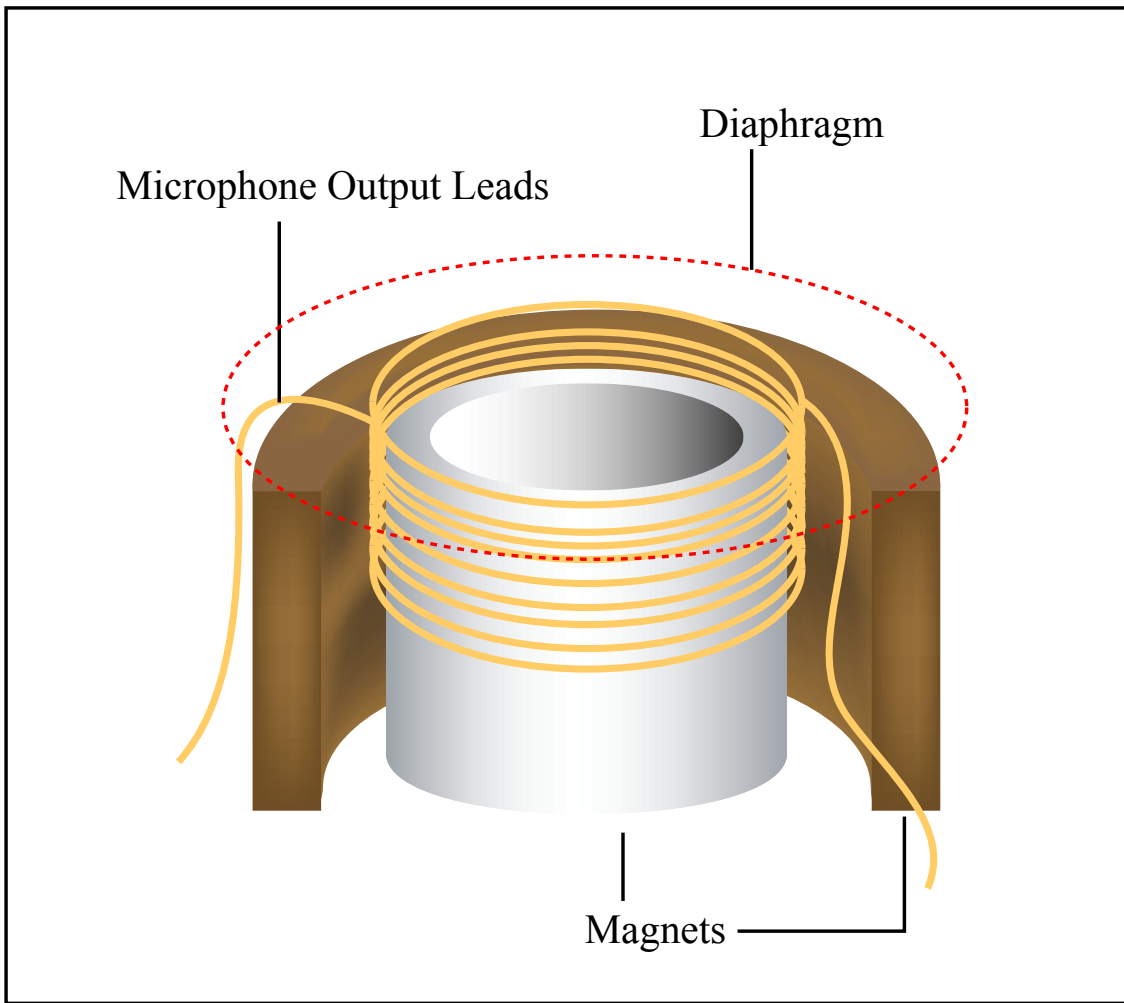


Image by MIT OpenCourseWare.

- Have big magnets: heavy
- Diaphragm must move relatively large distance: slower transient response

- Durable, can handle high SPLs
- May color sound between 5 and 10 kHz
- Often used in close-miking, within a foot of source; can be very close
- Phantom power not necessary, does not hinder performance

### 13.19. Transducer Type: Dynamic: Examples

- Shure SM-57



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- Sennheiser MD-421



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## 13.20. Transducer Type: Ribbon

- Metal is a thin ribbon

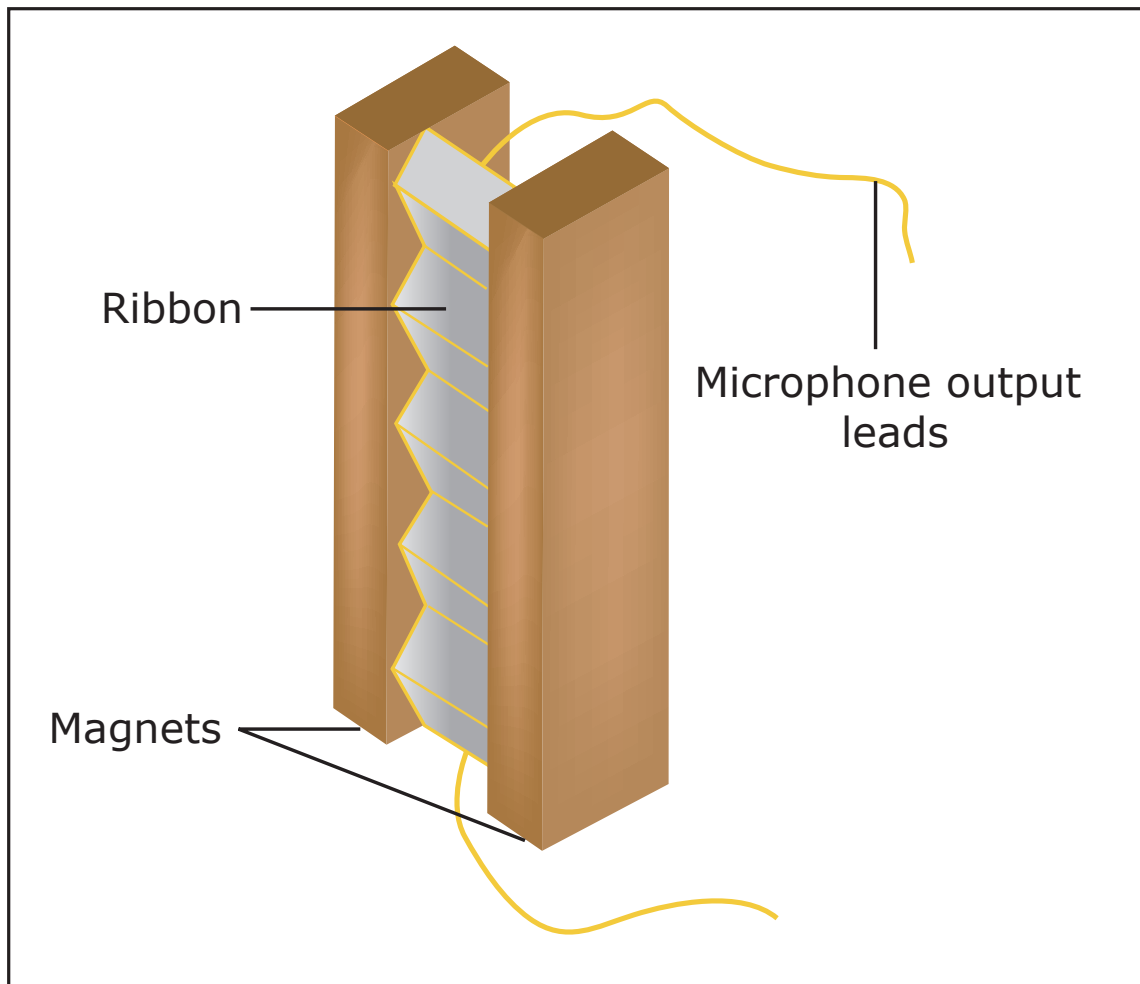


Image by MIT OpenCourseWare.

- Ribbon suspended between poles of a magnet
- Old ribbon mics were very fragile and unreliable
- Newer models are better
- Known for warm sound when used in close proximity
- Phantom power can cause old models to fry

### 13.21. Transducer Type: Ribbon: Examples

- AEA R92





- Royer R-122



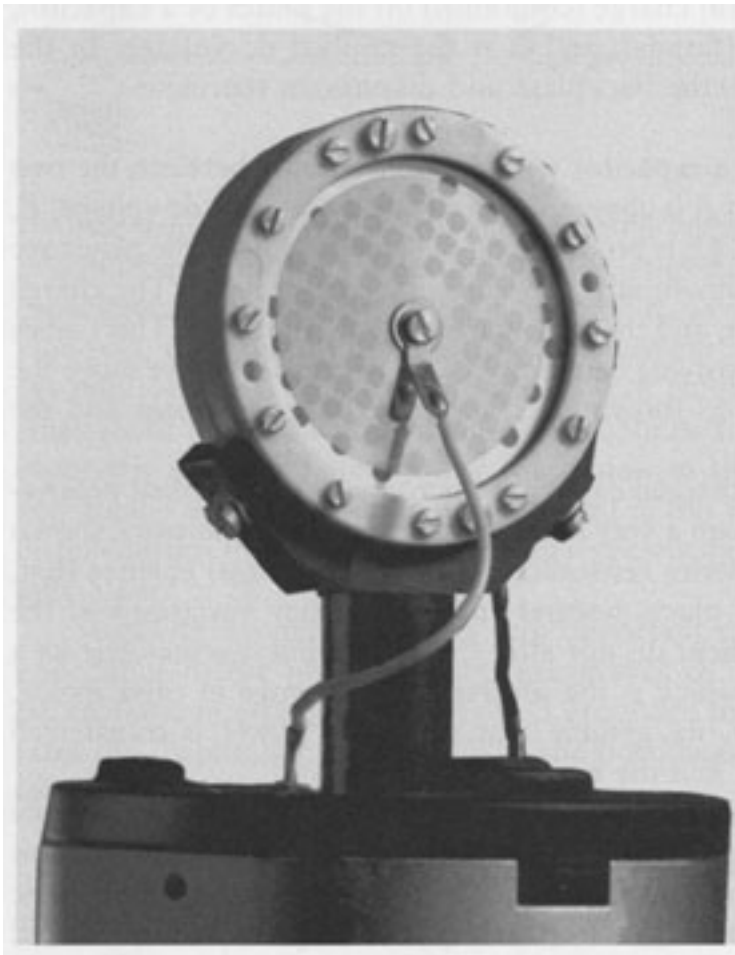
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## 13.22. Transducer Type: Condenser

- Delicate and accurate
- Diaphragm must move relatively small distance: fast transient response



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Neumann center-clamped condenser microphone capsule. © Neumann/USA. All rights reserved. This content is excluded from our Creative Commons license. For more information, see: <http://ocw.mit.edu/fairuse>.

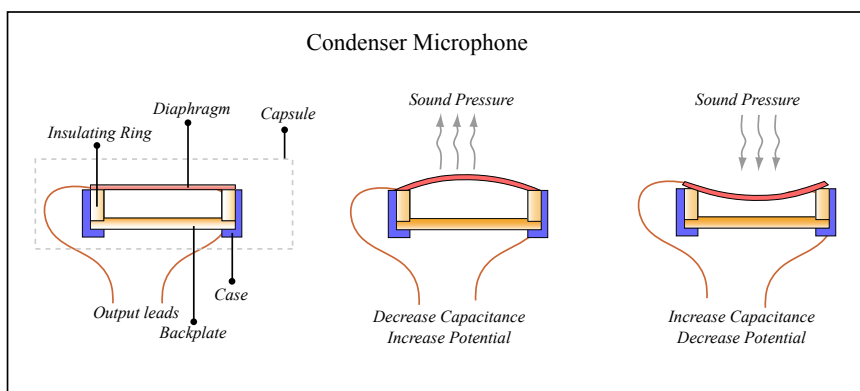


Image by MIT OpenCourseWare.

- Often offers less coloration
- Do not have to be very close to get an intimate sound
- Phantom power necessary
- Internal pre-amp may be transistor- or tube-based

### 13.23. Transducer Type: Condenser: Examples

- AKG C 414 BXL II/ST



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- AudioTechnica AT 4050



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- Neumann M149

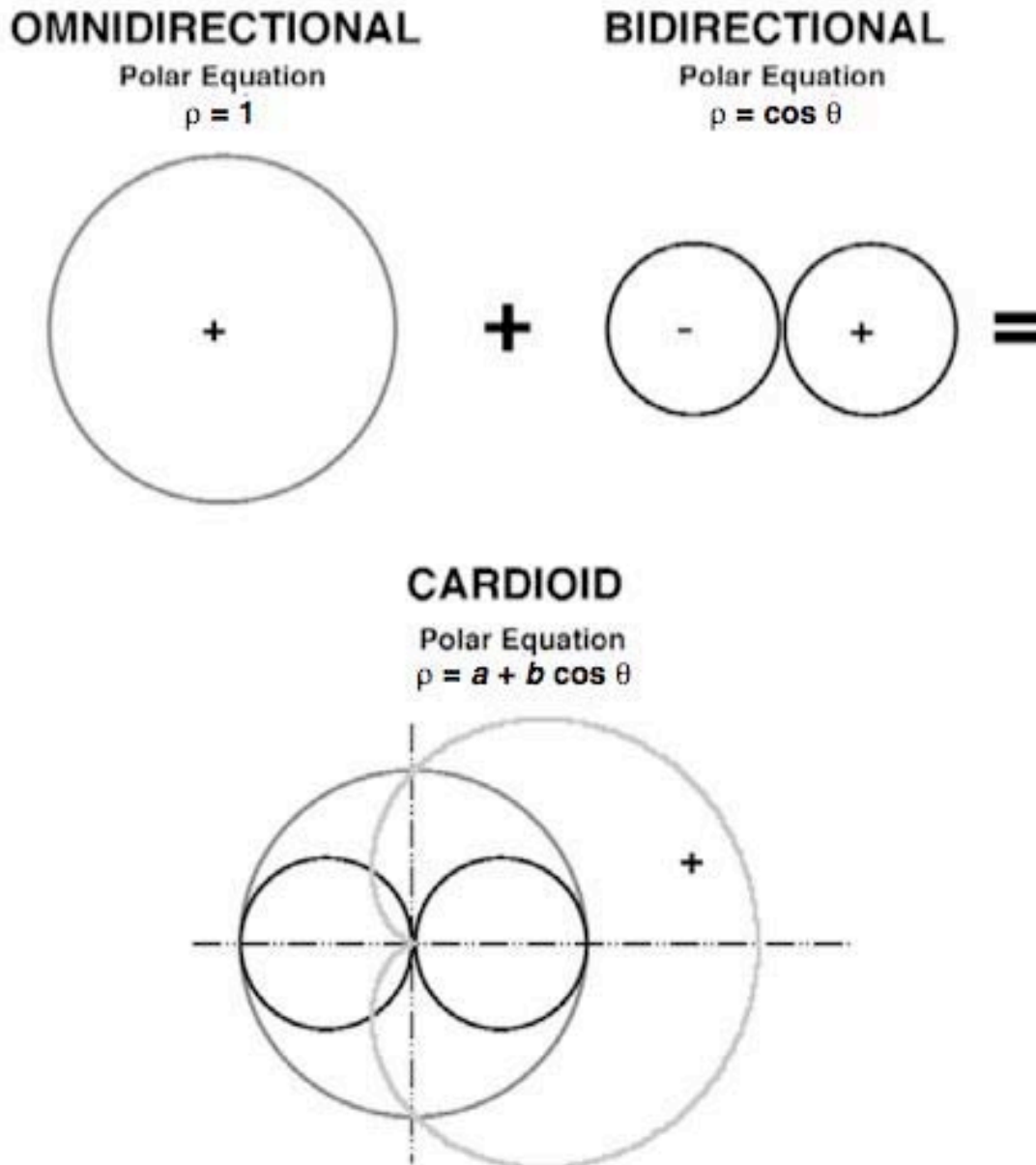


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### 13.24. Reading: Streicher: The Bidirectional Microphone: A Forgotten Patriarch

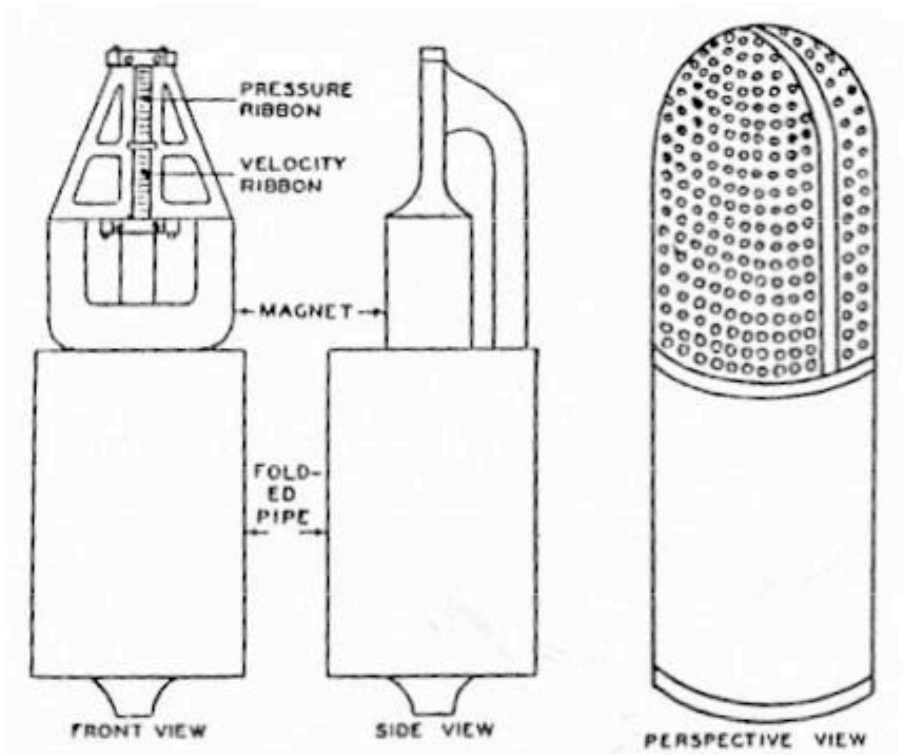
- Omni-directional microphones are pressure microphones: respond only to pressure; diaphragm covers a sealed chamber

- Bi-directional microphones have a diaphragm exposed on both sides: responds to difference (or gradient) in pressure; sometimes called velocity
- A cardioid (directional) pattern can be created by combining omni and bidirectional patterns
- All polar patterns can be derived from combination of omni and bi-directional

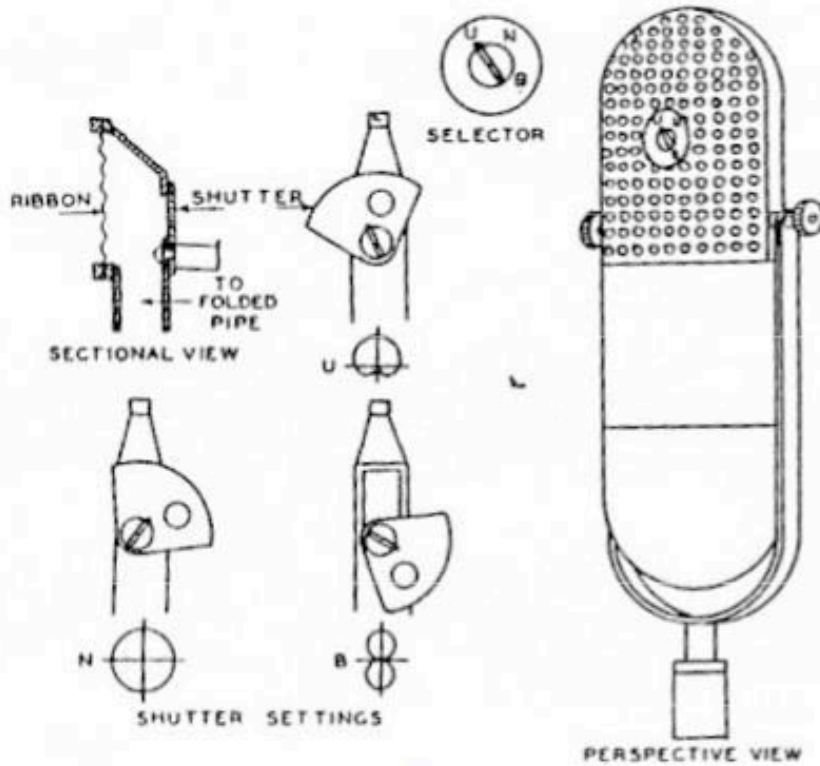


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- Earliest variable polar pattern microphone (RCA 77A) did this mechanically with a diaphragm divided into two parts



(a)



(b)

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 Source: Olson, H. F. "A History of High-Quality Studio Microphones." ) *Convention of the AES* 24 (1976): 862.



- Many modern capacitor mics that offer multiple patterns used two cardioid diaphragms back to back and vary amplitude of components

### **13.25. Recording Instruments: Study, Experience, and Experimentation**

- Conventional approaches based on practice and experience
- Creative approaches based on experimentation
- Walk around and listen
- Thinking of sound in three dimensions
  1. Three dimensional radiation
  2. Sound takes time to travel: 1.13 foot per millisecond (331 m/s)
  3. Sound travels in space: amplitudes diminish with distance
  4. Reflections matter: opportunities for comb filtering / phasing distortion

## Chapter 14. Meeting 14, Stereophonic Microphone Techniques

### 14.1. Announcements

- Need schlep crew for Wednesday: two people  
Meet in my office at 3:10
- Mix Report 1 Due Monday 9 April
- First recording session a week from Wednesday

### 14.2. Stereo versus Mono Microphone Techniques

- Instruments that are large or have diverse points of resonance or movement are often captured in stereo
  1. Piano, harp, percussion keyboard (marimba, xylophone, vibraphone)
  2. Acoustic guitars
  3. Leslie speaker cabinets
- Instruments and sound sources that have a focused output are often captured in mono
  1. Single drums
  2. Brass, woodwinds, and other aerophones
  3. String instruments: violin, bass, cello
  4. Speaker cabinets

### 14.3. Close Captures

- In general, monophonic captures are close
- Closeness offers an intimate sound and good isolation (least leakage)
- Closeness may remove or reduce reflections (ambience, reverb)
- Closeness can lead to unbalanced frequency response or irregular isolation
- Closeness can lead to undesirable air-bursts or physical contact

- Ribbons (figure eight), dynamic (cardioids), and large-diaphragm condensers (cardioids) most often used

## 14.4. Microphone Positioning Charts

- Indicate musician and microphone positions
- Circle + one arrow: cardioid
- Circle + two arrow: figure-eight
- Circle + cross: omni

## 14.5. Strings

- Close captures of strings can be very unnatural
- Often need some space for resonance and smoothing
- *f*-holes and sound holes offer focus of output
- Microphone diaphragm should be aligned to plane of sound board

Image removed due to copyright restrictions.  
Mic placement for violin, Figure 4.55 in  
Huber, D. M., R. E. Runstein, and D. M. Huber.  
*Modern Recording Techniques*. Taylor & Francis, 2001.

- Radiation is in three dimensions

Image removed due to copyright restrictions.  
3 plots of directionality in string instruments, Figure 13-3 in  
Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

- Favor large-diaphragm condensers

## 14.6. Acoustic Bass

- For tone, focus large-diaphragm cardioid at  $f$  holes
- Possibility of too much bass with proximity effect: can increase distance or use an omni to mitigate
- Can use alternative capture of strings (above or below) for more performance articulations
- Can position under strings in bridge or nut

Images removed due to copyright restrictions.  
Two diagrams showing acoustic bass micing.  
Figure 14-12 (mic mounted on bass) and 14-13 (mic on an amplifier).  
In Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

## 14.7. Vocals

- Lots of air, mouth noises, and breathing
- Proximity to nose can increase nasal sound
- Always use pop-screens to avoid plosives
- From 6 to 20 inches recommended
- Large diaphragm condensers always preferred
-

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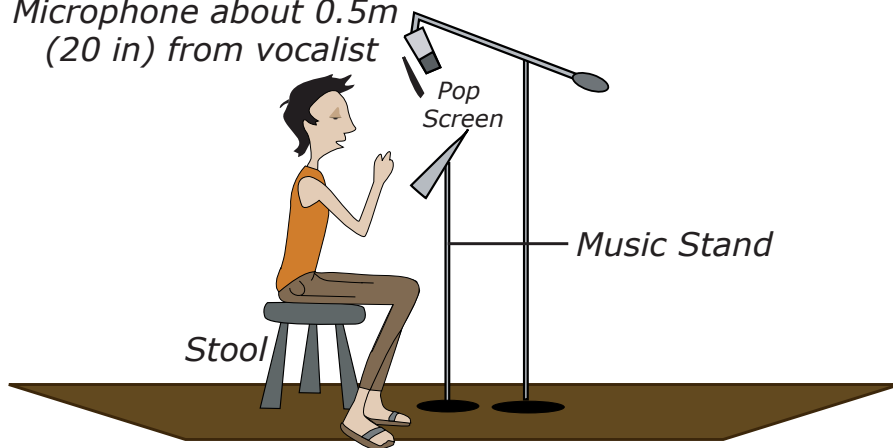
1) Vocalist microphone is normally 0.5 to 1 meter away, Figure 13-9, in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

2) Sound reflecting off a music stand can cause comb-filtered frequency interference with the direct sound - see Fig. 16.10 in Huber, D. M., R. E. Runstein, and D. M. Huber. *Modern Recording Techniques*. Taylor & Francis, 2001.

## Recording a Vocalist

### Side view

Microphone about 0.5m  
(20 in) from vocalist



### Top view

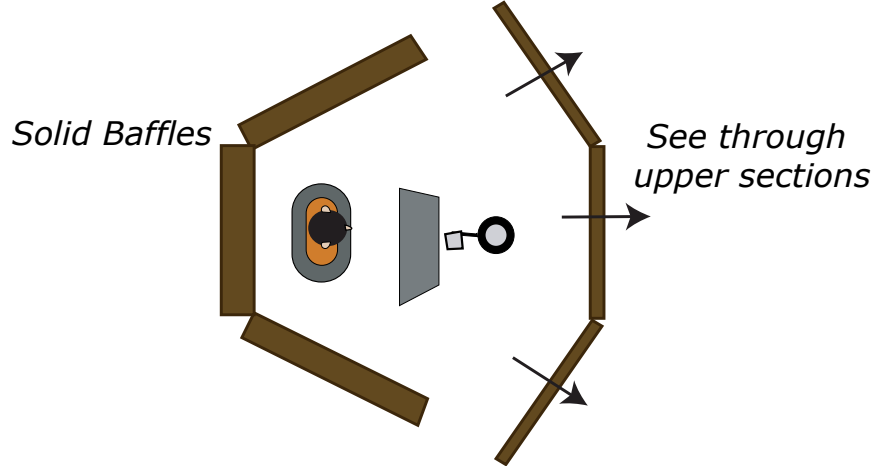
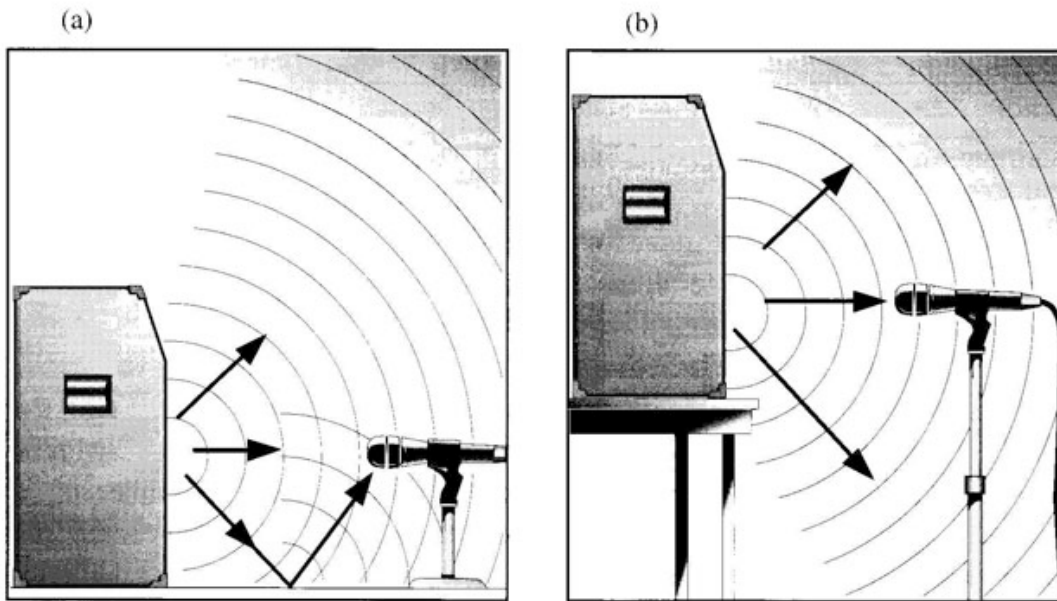


Image by MIT OpenCourseWare.

## 14.8. Amps

- Axis and orientation to speaker makes a big difference
- Often want to be slightly off axis of speaker cone





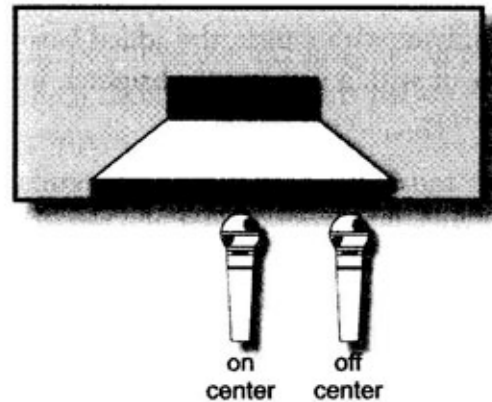
**Figure 4.1** Amplifier on riser

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Source: Crich, T. *Recording Tips for Engineers: For Cleaner, Brighter Tracks*. 2nd ed. Taylor & Francis, 2005.

- The front and back of a speaker are valuable sources
- Position in relation to speaker matters

**Figure 4.42.** Miking an electric guitar cabinet directly in front of and off-center to the cone.



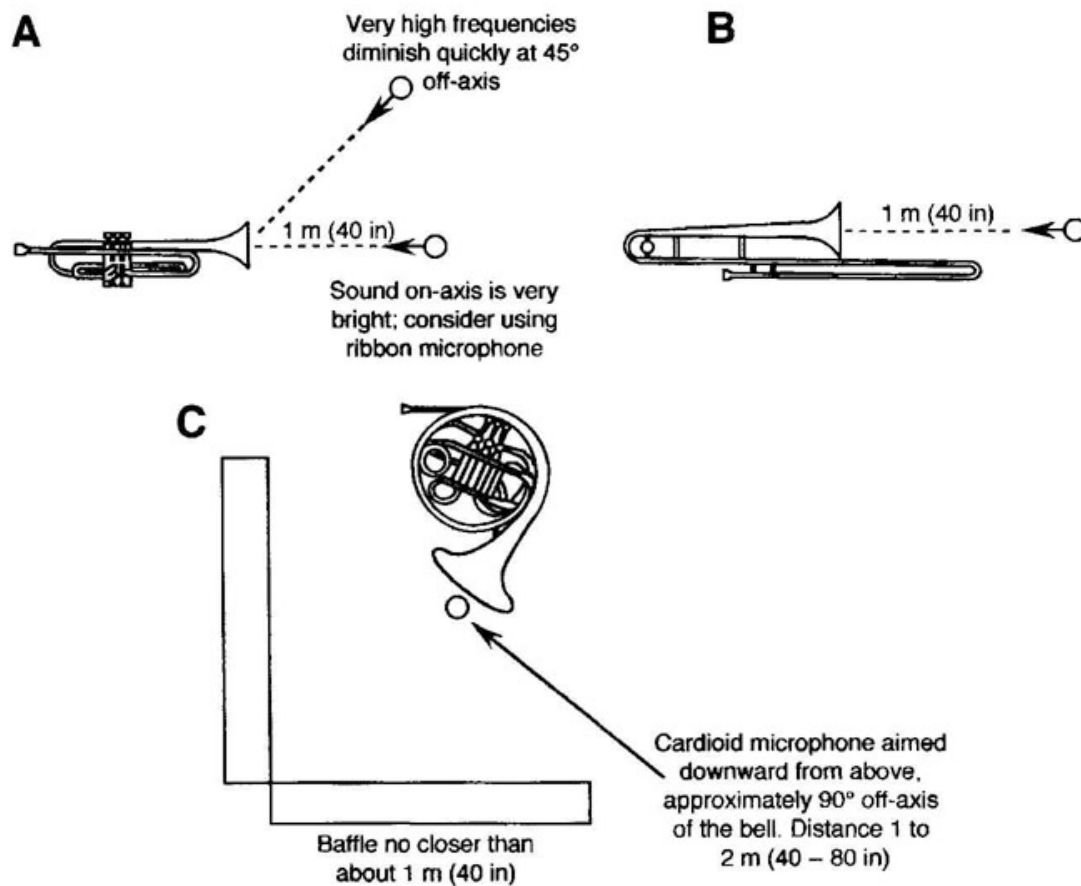
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Source: Crich, T. *Recording Tips for Engineers: For Cleaner, Brighter Tracks*. 2nd ed. Taylor & Francis, 2005.

- Special amps (Leslie speakers, stereo cabinets, diffuse radiating cabinets) require stereo captures
- Favor cardioid dynamics, large diaphragm cardioids (with pad), or ribbon

## 14.9. Brass

- Huge dynamic range, potentially large bursts of air
- Dynamic microphones are effective, safe, and warm
- Favor small diaphragm condensers (with pad) or ribbon
- On-axis captures are common; off-axis (or post recording filtering) may give a warmer sound

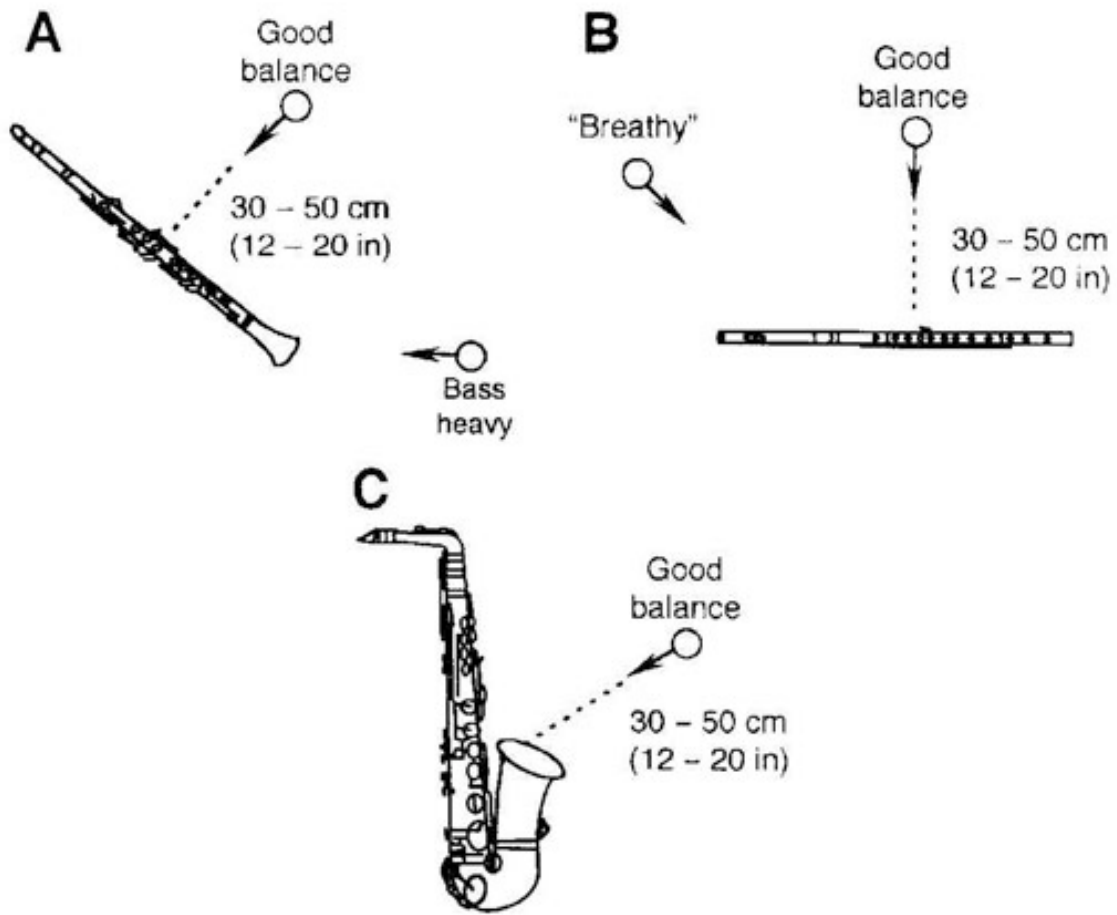


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Source: Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

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Mic placement for trumpet should be slightly off axis.  
See Fig. 4.40 in Huber, D. M., R. E. Runstein, and D. M. Huber.  
*Modern Recording Techniques*. Taylor & Francis, 2001.

## 14.10. Woodwinds

- Clarinets, saxophones, flutes, bassons
- Sound emanates from all around (bell and keys)



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 Source: Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

- Point microphone toward the alignment of the keys and bell

Image removed due to copyright restrictions.  
Mic placement for alto saxophone, points just behind the bell.  
See Huber, D. M., R. E. Runstein, and D. M. Huber.  
*Modern Recording Techniques*. Taylor & Francis, 2001.

Image removed due to copyright restrictions.  
Mic placement for clarinet.  
See Figure 4.56 in Huber, D. M., R. E. Runstein, and D. M. Huber.  
*Modern Recording Techniques*. Taylor & Francis, 2001.

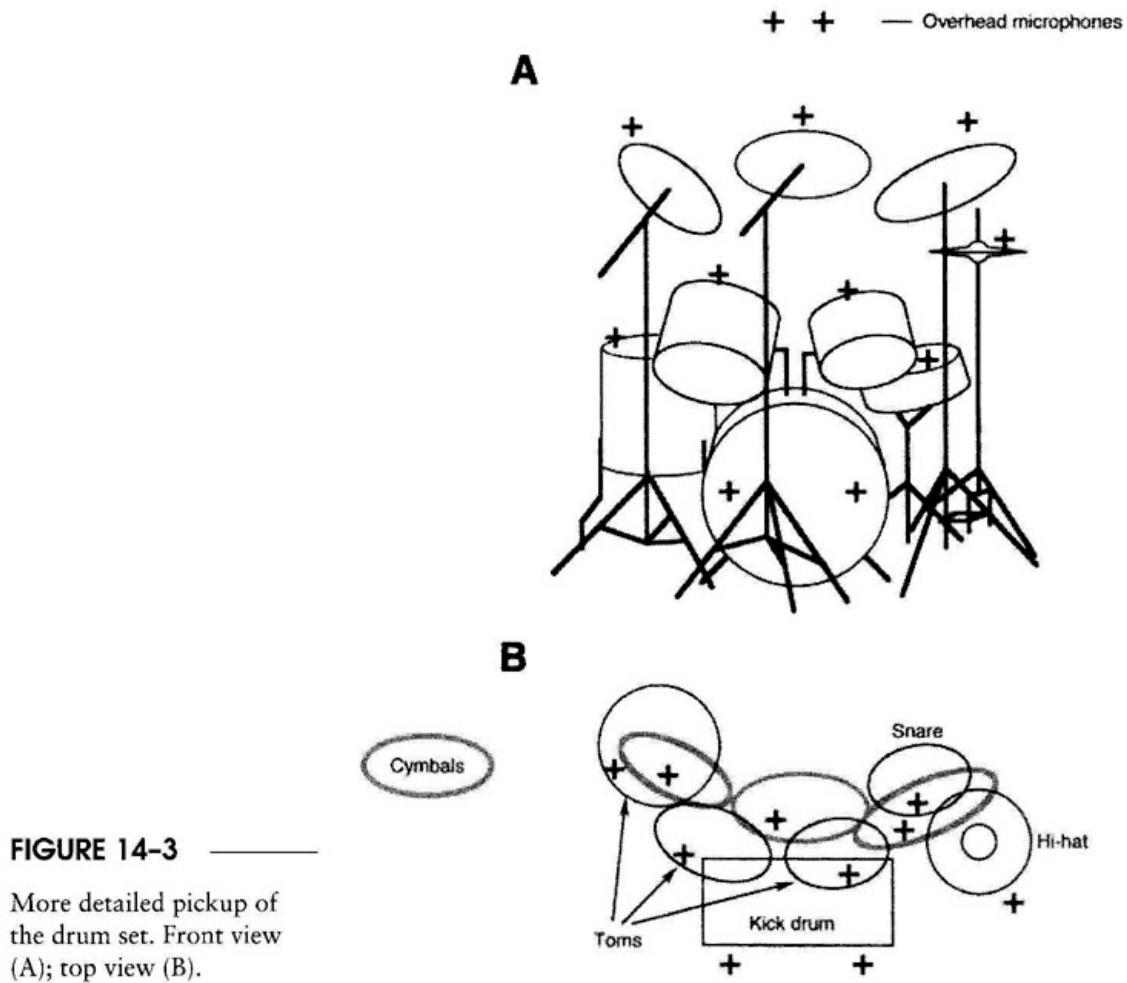
- For flutes, almost all sound out of keys

Image removed due to copyright restrictions.  
Mic placement for flute.  
See Figure 4.57 in Huber, D. M., R. E. Runstein, and D. M. Huber.  
*Modern Recording Techniques*. Taylor & Francis, 2001.

- Favor large or small diaphragm condensators

## 14.11. Drum Kit

- At most: a microphone per drum



**FIGURE 14-3** ———  
More detailed pickup of the drum set. Front view (A); top view (B).

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- At least: two overheads and a kick-drum microphone
- Problem of leakage forces very close captures
- Problem of drummers hitting things with sticks

## 14.12. Drum Kit: Cymbals

- Overhead stereo captures used to gather cymbals and high frequencies (discussed under stereo techniques)
- High-hat often given a close capture: small diaphragm cardioid condenser or dynamic

Image removed due to copyright restrictions. Microphone placement for cymbals: above cymbal out of range of cymbal's rotation, pointing diagonally toward bell. Image 14-5 in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

### 14.13. Drum Kit: Snare drum

- Different sounds from the bottom and the top
- Bottom may offer more snare tone
- Off axis capture preferred



Photo courtesy of [Curt Newton](#) on Flickr.

- Mixing top and bottom may create phasing distortion
- Cardioid or super-cardioid dynamic microphones are common (SM-57 is popular)



- Small diaphragm condenser microphones (with high SPL handling or pads engaged) may be used

#### **14.14. Drum Kit: Toms**

- Top will have more stick sound than bottom
- Mic-per tom gives best isolation and stereo presentation
- One microphone might be used for two rack-toms
- Often more on-axis than snare microphones

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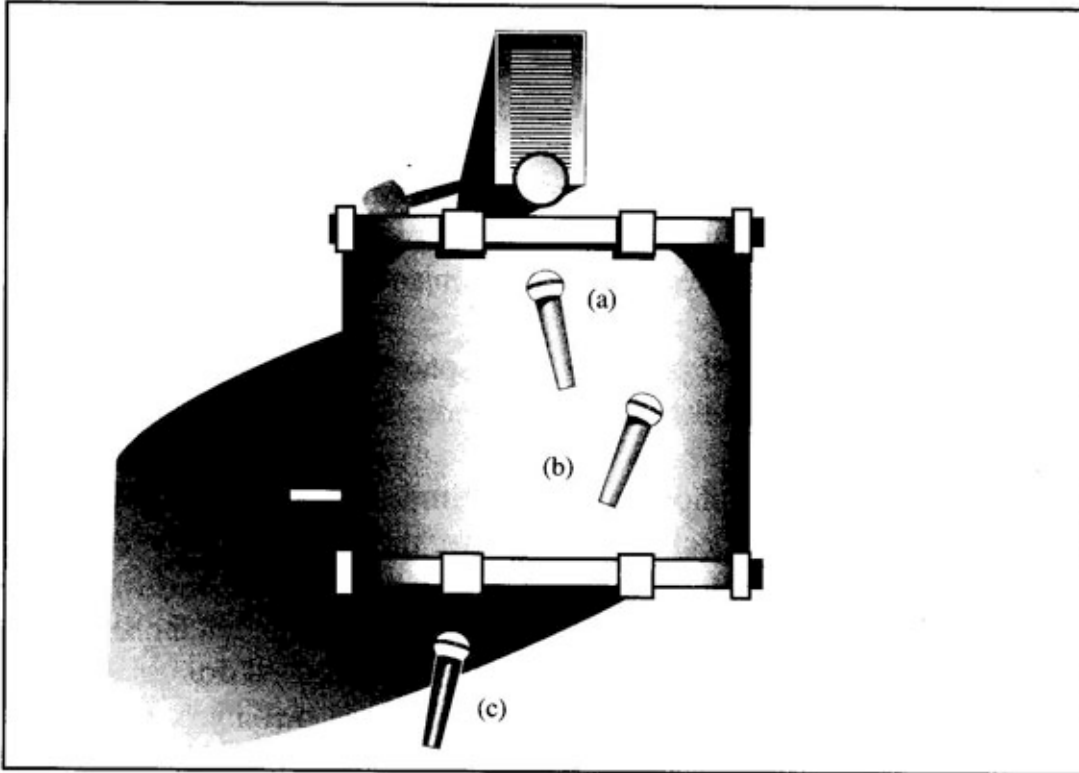
Mic placement for rack tom (aka mounted tom).

See Figures 4.50 and 4.51 in Huber, D. M., R. E. Runstein, and D. M. Huber. *Modern Recording Techniques*. Taylor & Francis, 2001.

- Cardioid dynamic microphones are common
- Small diaphragm condenser microphones (with high SPL handling or pads engaged) may be used

## 14.15. Drum Kit: Kick

- Both low and high frequency ranges are critical
- Inside and outside captures are often used

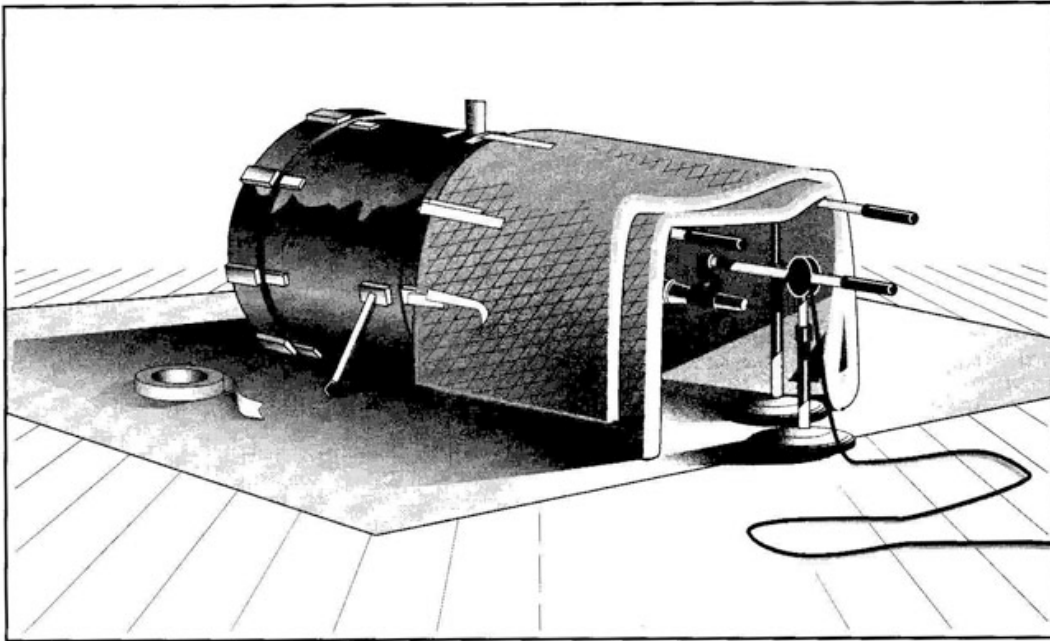


**Figure 3.1** *Microphones inside the kick drum*

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Source: Crich, T. *Recording Tips for Engineers: For Cleaner, Brighter Tracks*. 2nd ed. Taylor & Francis, 2005.

- Can focus sound outside of drum



*Figure 3.2 Kick drum tunnel*

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Source: Crich, T. *Recording Tips for Engineers: For Cleaner, Brighter Tracks*. 2nd ed. Taylor & Francis, 2005.

## 14.16. Stereo Recording: Common Applications

- Can be used for both close and distant captures
- Pianos, acoustic guitars, keyboard percussion, drum set (overheads)
- Ensembles, sections

## 14.17. Localization

- Two sources of localization information
- Timing differences: a single sound arrives to our ears at different times
- Amplitude differences: a single sound arrives to our ears at different amplitudes

## 14.18. Stereo Recording: Common Directional Pairing

- Pairs of cardioids
- Pairs of omnis
- Pairs of figure-eights

- Mid/Side: cardioid and figure-eight (will discuss next meeting)

## **14.19. Stereo Recording: Common Positioning Archetypes**

- Coincident pairs: X-Y, M-S
- Near-coincident pairs (ORTF, NOS, Faulkner)
- Spaced pairs: A-B

## **14.20. Cardioid and Figure-Eight Coincident Pairs**

- Timing is identical; localization is due to amplitude differences
- Common approaches use cardioid or hyper-cardioid pairs; coincident omnis will have minimal differentiation
- Consider which sound sources that are off- and on-axis
- Small diaphragm condensers are preferred if less off-axis coloration is needed (Holman 2008, p. 74)
- Matched pairs (with very similar frequency response) are used to reduce the chance of sound-image movement at different frequencies
- Splay from 60 to 120 degrees

Image removed due to copyright restrictions.

Variable crossed cardioid patterns.

Figure 14-6 in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

- Blumlein array (1931): coincident crossed figure eights

Works best in a wide room, with minimal side-wall reflections (Streicher and Dooley 1985)

Image removed due to copyright restrictions.

Crossed figure-8s and localization.

Figure 11-4 in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

## **14.21. Cardioid and Figure-Eight Near-Coincident Pairs**

- Localization is due to both amplitude and timing differences

Results in good localization with a sense of depth (Streicher and Dooley 1985)

- Front material is less off axis than with coincident pairs
- Not used for close captures: small movements of the sound source can produce large image shifts
- Office de Radio-Television Diffusion Française (ORTF): 6.7 inches (17 cm), 55 degrees from forward

Frequently voted best stereo capture (Holman 2008, p. 85)

- Nederlandsch Omroep Stichting (NOS): 11.8 inches (30 cm), 45 degrees from forward
- Faulkner array: two bi-directional mics, 7.9 inches (20 cm), facing forward
- Diagrams

Images removed due to copyright restrictions.

Several approaches to near-coincident pairs (ORTF, NOS, Faulkner, Stereo 180).

Figures 11-13 and 11-14 in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

## 14.22. Spaced Omni Pairs

- Localization due to both amplitude and timing effects
- Best used for distant captures combined with closer captures

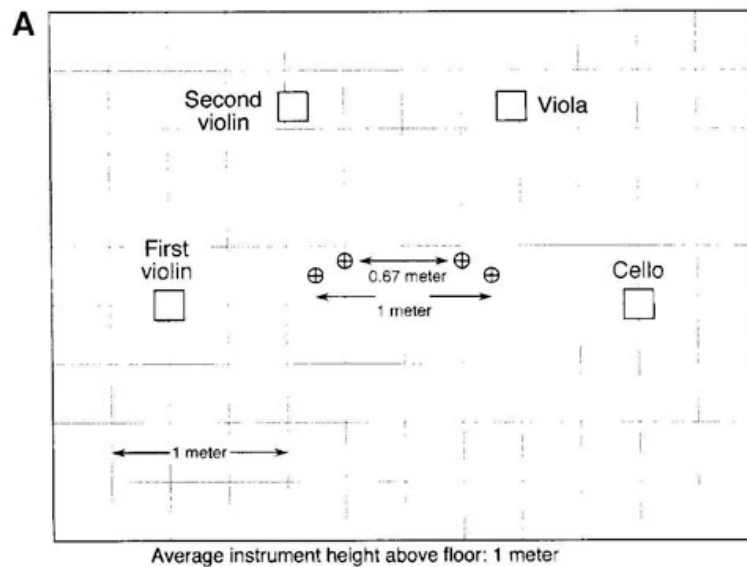
Extremely distant sounds can present negligible directional cues to the listener (Streicher and Dooley 1985)



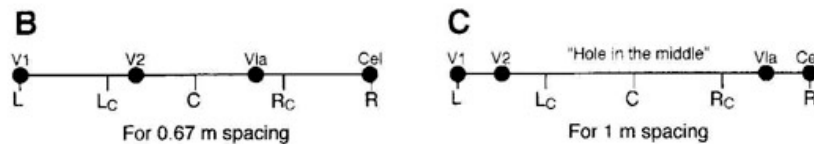
- Omnis have more extended low frequency response and lower noise floor (Homan 2008, pp. 73-74)
- Spacing too close together results too little stereo distinction: coincident omnis are nearly monophonic
- Spacing too far results in audible echos between channels

Vague center imaging (Streicher and Dooley 1985)

- Common approaches: 2 feet (.6 meters) close to performers; 10 to 30 feet used in front of large ensembles (Holman 2008, p. 79)
- Can use subcardioids for a bit more directionality
- Examples



**FIGURE 11-10** Two spaced omnis with a string quartet (A); Franssen analysis for 0.67 m spacing (B); Franssen analysis for 1 m spacing (C).



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# Chapter 15. Meeting 15, Workshop: Microphone Positioning and Recording Sessions

## 15.1. Announcements

- Mix Report 1 Due Monday 9 April
- Processing Report 2 comments and grades out tomorrow

## 15.2. Mid/Side Pairs

- Rather than capturing left and right, capture front and sides
- Combining cardioid and figure eight can result in dual-cardioid equivalence

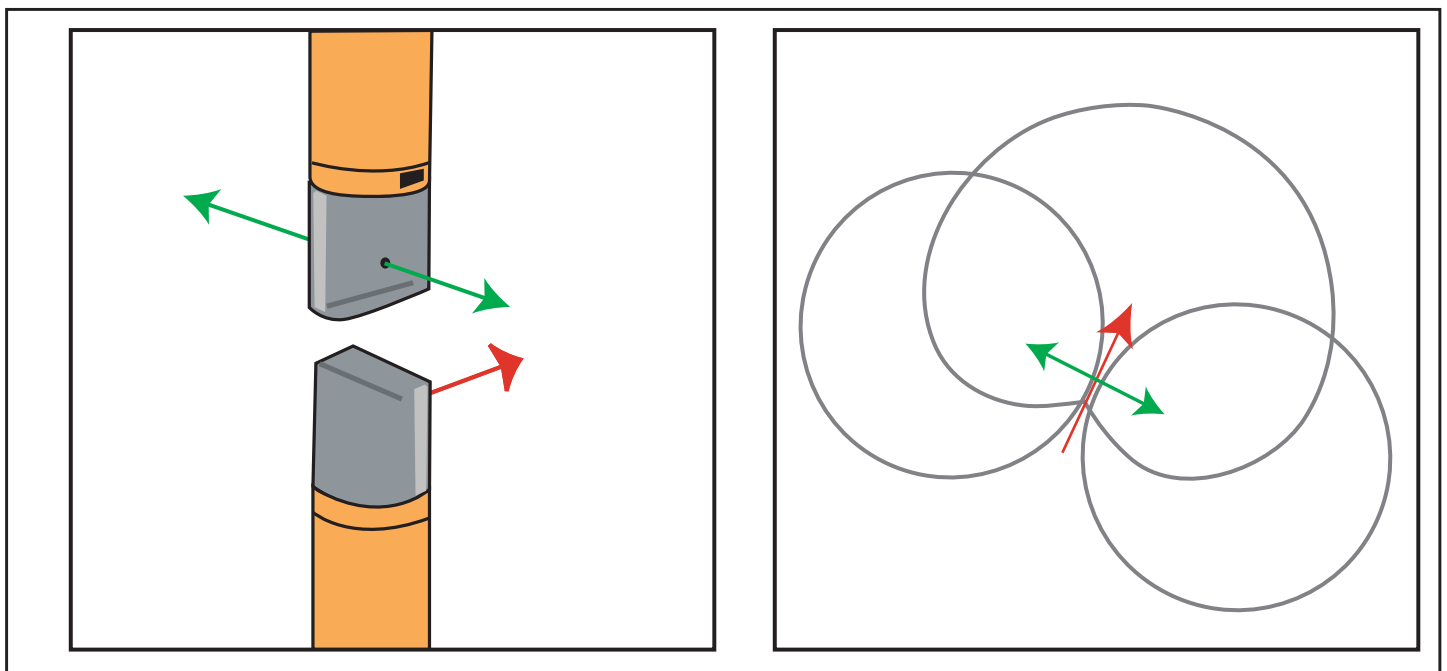
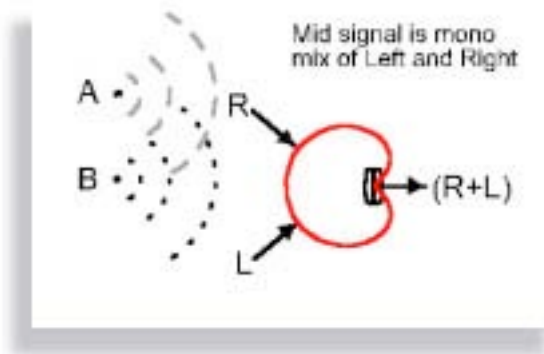


Image by MIT OpenCourseWare.

- Cardioid (M) and Bipolar (S, positive on right)

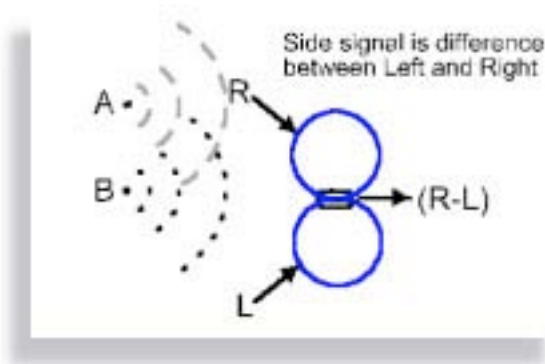
- Cardioid (M): receives coincident  $R + L$



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- Bipolar (S): receives coincident  $R - L$

Sounds arriving to the bipolar microphone are stamped with polarity



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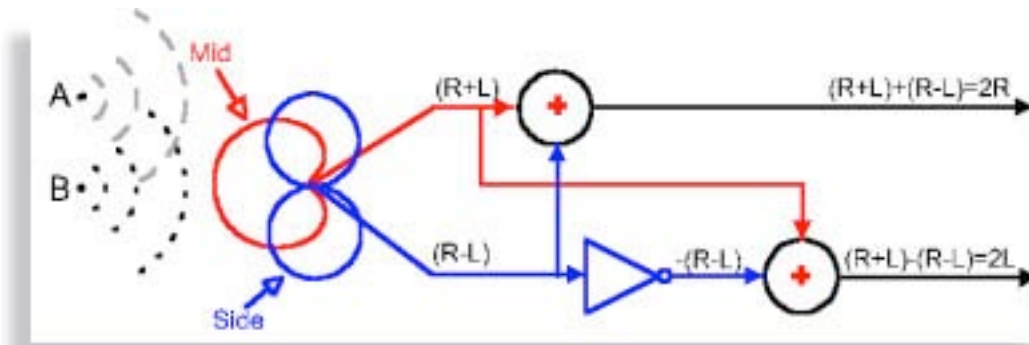
- Converting MS to LR

Take M ( $R+L$ ) and add side ( $R-L$ ): return  $2R$

Or: take front and remove all signals in phase with the left (leaving the right-most capture)

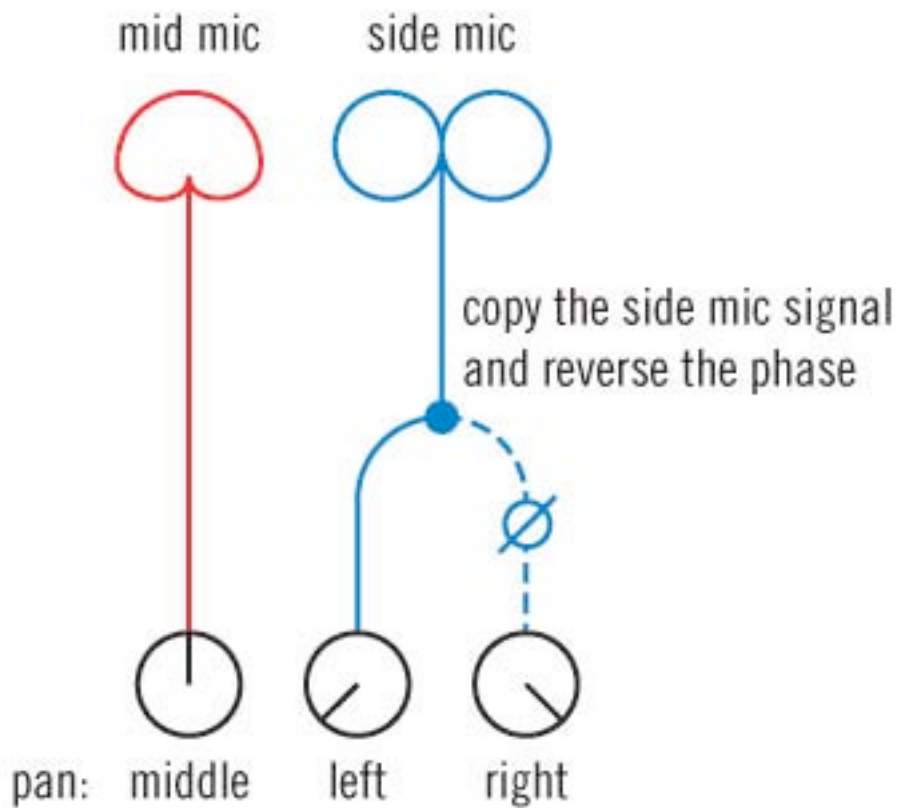
Take M ( $R+L$ ) and add inverse side ( $-R+L$ ): return  $2L$

Or: take front and remove all signals in phase with the right (leaving the left-most capture)



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- Summarized post processing of M/S to L/R signals
  - $L == (M + S) / 2$
  - $R == (M - S) / 2$
- Polarity of figure-eight mic is important: right is positive
- Decoding MS in a DAW requires three tracks
  - Side track is duplicated and panned hard left and hard right
  - Right side track is inverted (use Live Utility plugin)



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Source: <http://www.uaudio.com/blog/mid-side-mic-recording/>

### 15.3. Mid-Side Advantages

- Can easily get an on-axis, mono capture
- Cardioid is on-axis: a potentially better-sound capture
- Can control width of stereo capture in mix

### 15.4. MOSS Microphones

- AKG C414 XL II (4)

Large-diaphragm condenser; multi polar pattern (cardioids, omni, figure-eight)



- Audio-Technica AT4041 (6)

Small-diaphragm condenser; cardioid polar pattern



- TC20mp (2)

Small-diaphragm condenser; omni polar pattern



- Mojave Audio MA-200 (1)

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Large-diaphragm tube condensor; cardioid polar pattern



- Sennheiser MD 421 (2)

Dynamic; cardioid polar pattern



- Shure SM57 (2)

Dynamic; cardioid polar pattern



- Royer R-101 (1)

Ribbon; figure-eight polar pattern

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- AT M250DE (1)

Dual-element instrument microphone



- e604 (1)

Dynamic cardioid w/ more than 160 dB dynamic range

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- Blue enCORE 200 (4)  
Active dynamic cardioid



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## 15.5. Stereo Positioning Review Sheet

- Coincident
  - XY: two cardioids, splay between 60 and 120 degrees
  - MS: one cardioid, one figure-eight, 90 degrees between mid and side
  - Blumlein: two figure eights at 90 degrees
- Near-coincident

- ORTF: two cardioids, 6.7 inches apart, 55 degrees splay from forward
- NOS: two cardioids, 11.8 inches apart, 45 degrees from forward
- Faulkner: two figure-eights, 7.9 inches apart, facing forward
- Spaced
  - AB: two omnis spaced between 2 and 10 feet (or more) apart

## **15.6. Procedure**

1. Review Mics
2. Groups

Each group contains 3 or 4 students (names removed for privacy)

3. Setup stereo configuration based on assignment on card
4. Identify and describe adjacent microphone configuration

## Chapter 16. Meeting 16, Ensemble Microphone Techniques

### 16.1. Announcements

- Mix Report 2 Due Tuesday 10 April (be sure to review requirements)
- Recording session this Wednesday, 11 April, in Killian Hall
- No class next Monday, 16 April
- Next quiz will be Wednesday, 25 April

### 16.2. Recording Session Assignments

- 11 April (Wednesday): Meeting 17, Workshop: Recording Session 1  
Engineering crew: four students [names removed for privacy]  
Instrumentation: 5 singers, including soloist  
Location: Killian Hall
- 23 April (Monday): Meeting 19, Workshop: Recording Session 2  
Engineering crew: four students [names removed for privacy]  
Instrumentation: piano and horn  
Location: Killian Hall
- 2 May (Wednesday): Meeting 22, Workshop: Recording Session 4  
Engineering crew: five students [names removed for privacy]  
Instrumentation: gtr, bs, drum kit, 3 vocal, more  
Location: TBA
- 7 May (Monday): Meeting 23, Workshop: Recording Session 5  
Engineering crew: four students [names removed for privacy]  
Instrumentation: 14 singers, 7 male, 7 female  
Location: Killian Hall

- Need at least four people to move gear before and after

### **16.3. Stereo Applications: Drum-kit overheads**

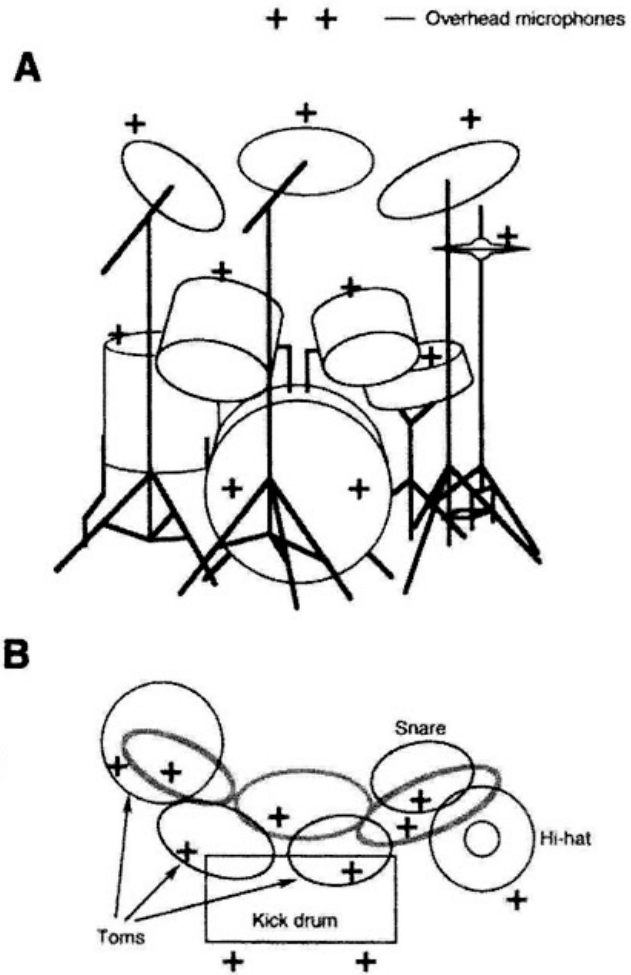
- Given all the microphones, wide coincident might be preferred
- Stereo overheads

Image removed due to copyright restrictions.

"Simple" setup: overhead stereo pair 2 meters above the drumset,  
plus a dynamic mic on the bass drum.

Figure 14-2 in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

- Ensemble mics

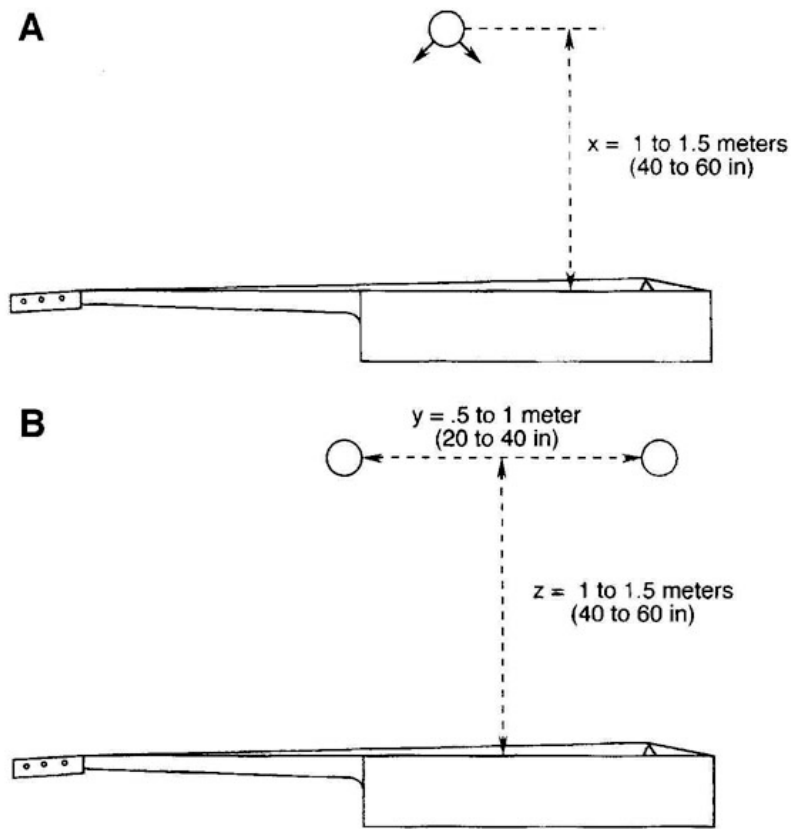


**FIGURE 14-3** —————  
 More detailed pickup of the drum set. Front view (A); top view (B).

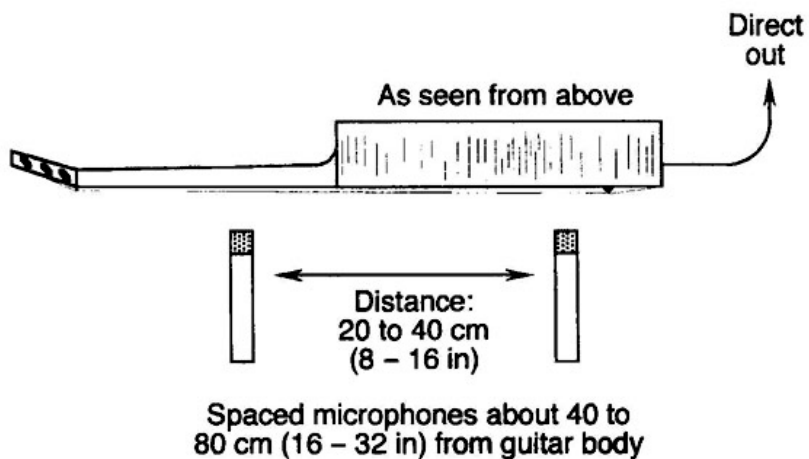
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 Source: Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

## 16.4. Stereo Applications: Acoustic Guitar

.



**FIGURE 13-6** ———  
 Recording the guitar: using coincident or near-coincident microphones (A); using spaced microphones (B).



**FIGURE 14-17** ———  
 Recording the acoustical guitar with microphones and direct input to the console.

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## 16.5. Stereo Applications: Keyboard Percussion

.

Image removed due to copyright restrictions.

"Widely splayed" pair of cardioids for recording vibraphone in stereo

Figure 14-8 in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

## 16.6. Stereo Applications: Piano

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Image removed due to copyright restrictions.  
Recording piano using coincident and near-coincident microphones.  
Figure 13-5 in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.



Image removed due to copyright restrictions.  
Recording piano using mics under the raised lid.  
Figure 14-9 in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

## 16.7. Recording Ensembles

- Goal is often to capture instruments and room
- Need for archival security
- Need for post-production flexibility

## 16.8. Multiple Mics and Comb Filtering

- Combining slightly delayed signals can result in comb filters
- Can mitigate by careful positioning
- Can mitigate by post-production time delays
- Some leakage can be good
- Leakage needs to work with ultimate panning positions

## 16.9. Recording Ensembles: Close Captures, Small Groups

- Using Rejection
- Mixtures of omnis and cardioid

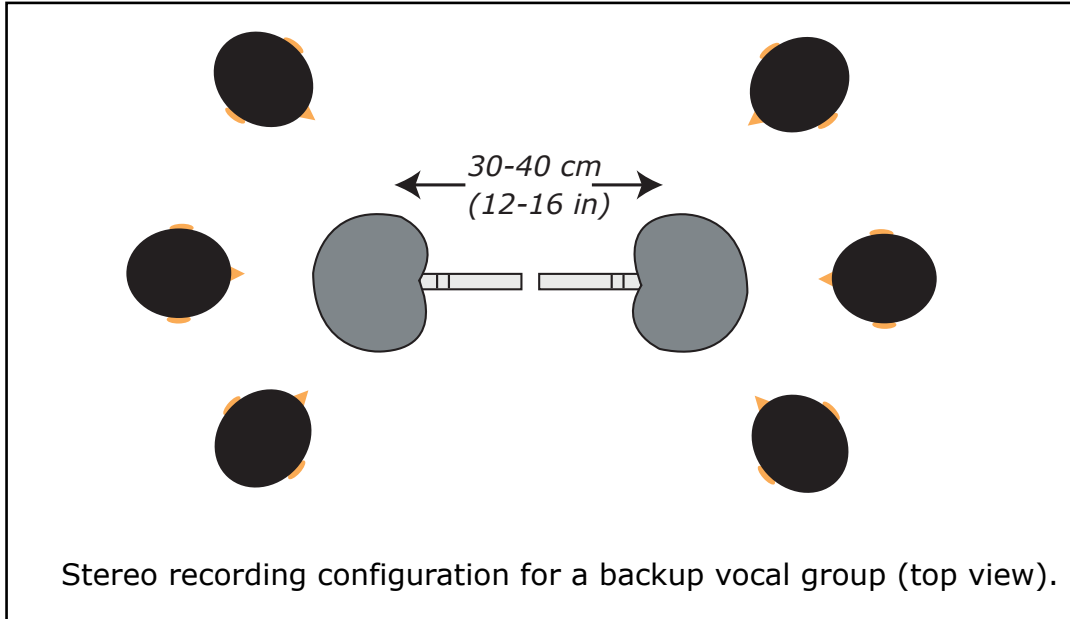
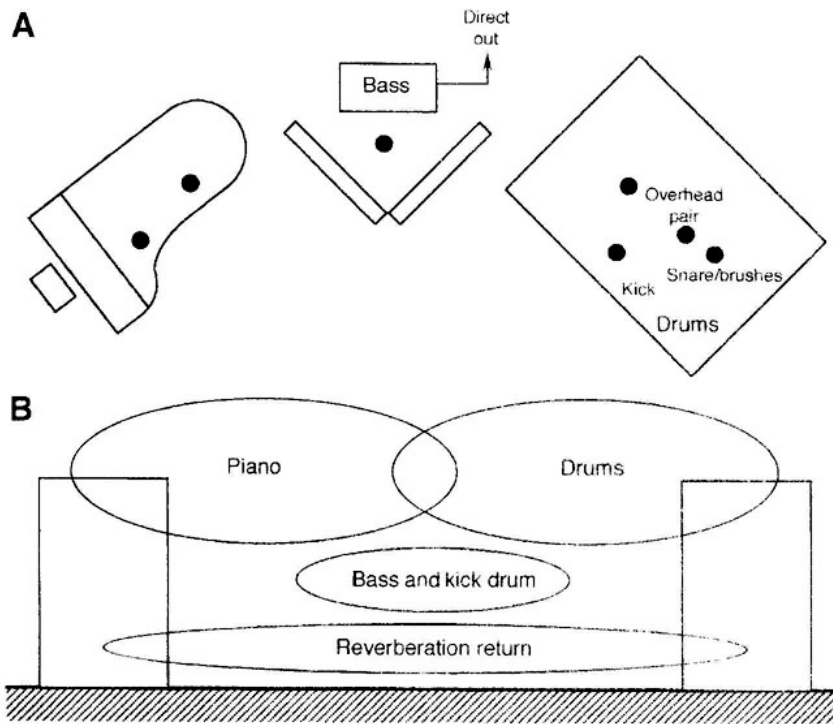


Image by MIT OpenCourseWare. After Eargle.

## 16.10. Recording Ensembles: Close Captures, Considering Panning

- Some isolation, some mixture, with ambiance
- Maintaining stereo field

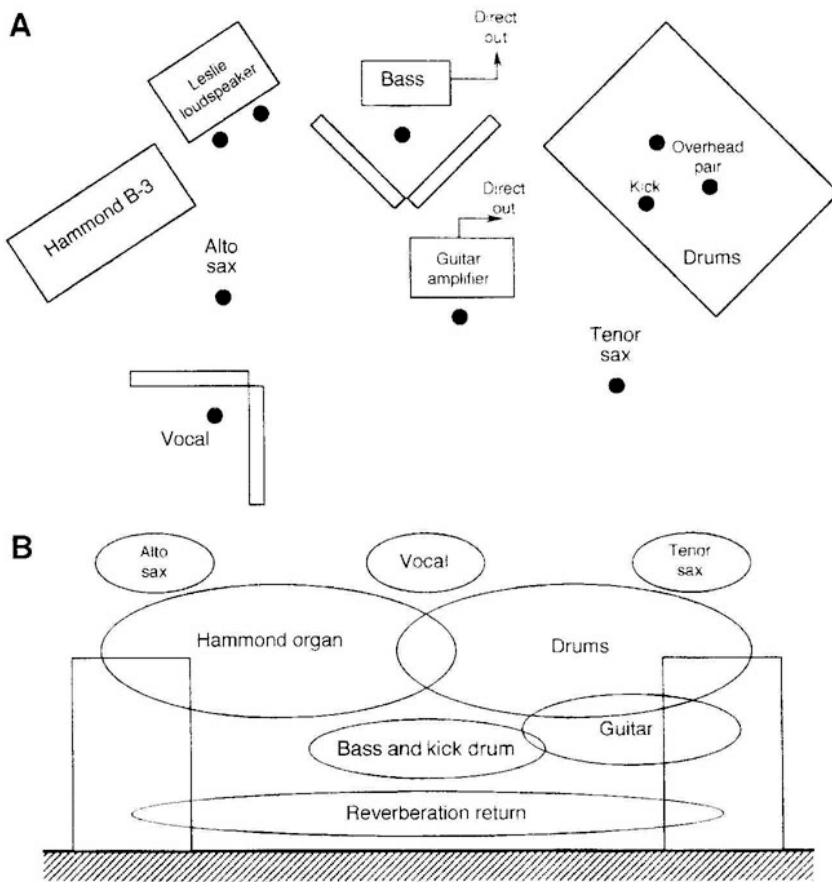


**FIGURE 14-21** ———  
 Recording a jazz trio:  
 studio layout (A); target  
 stereo soundstage (B).

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 Source: Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

## 16.11. Recording Ensembles: Close Captures, Considering Panning

- Maintaining stereo field



**FIGURE 14-22** Recording jazz vocal with small instrumental group: studio setup (A); target stereo soundstage (B).

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 Source: Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004

## 16.12. Recording Ensembles: Comparing Distant and Close Captures

- Notice the direction the musicians are facing:

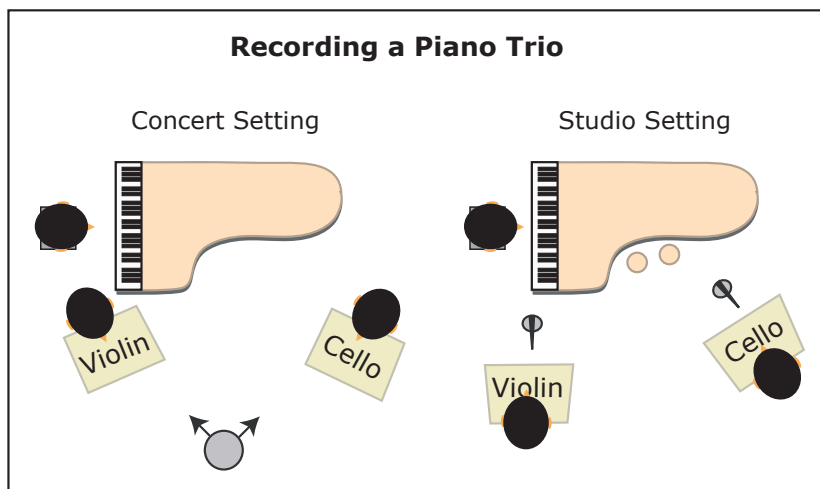


Image by MIT OpenCourseWare. After Eargle.

## 16.13. Recording Ensembles: Concert Recording with Multiple Stereo Captures

- Pair of cardioids and pair of omnis is most common approach

Image removed due to copyright restrictions.

Three examples of mixed arrays: ORTF plus flanking omni mics.

Figure 11-16 in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

## 16.14. Recording Ensembles: Multiple Stereo, Room, and Section Captures

- Can combine stereo captures, room captures, and section captures
- Mixing may require significant time shifting
- The closer the microphone the greater the mixing time shift
- Orchestra example

Image removed due to copyright restrictions.

Large orchestra with chorus example.

Figure 13-13 in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

## 16.15. Recording Ensembles: Multiple Stereo and Section Captures

- Orchestra in a studio example

Image removed due to copyright restrictions.

Figure 14-24 in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

## 16.16. Recording Ensembles: Multiple Stereo and Section Captures

- Orchestra with soloists in a studio

Image removed due to copyright restrictions.  
Figure 13-15 in Eargle, J. *The Microphone Book*. 2nd ed. Focal Press, 2004.

## **16.17. MOSS Track Sheets**

- Must document all aspects of every recording session



MOSS Track List

Date:

	Source	Microphone	Position/Axis/Array	Distance	Input	Preamplifier	48V (+/o)
1							
2							
3							
4							
5							
6							
7							
8							
9							
10							
11							
12							
13							
14							
15							
16							

### 16.18. Microphone Positioning: Exercise

- Exercise: You are to record a trio of piano, acoustic bass, and trumpet. You have 6 AT 4041, 4 AKG 414, 2 Earthworks TC20mp, and 2 Sennheiser MD-421.

### 16.19. Microphone Positioning: Exercise

- Exercise: You are to record a string quartet. You have 6 AT 4041, 4 AKG 414, 2 Earthworks TC20mp, and 2 Sennheiser MD-421.

### 16.20. Microphone Positioning: Exercise

- Exercise: You are to recording a group of 5 singers, including a soloist. You have 6 AT 4041, 4 AKG 414, 2 Earthworks TC20mp, and 2 Sennheiser MD-421.

### 16.21. Studio Practices: Positions

- Lead engineer, LE (1)

Greets performers, runs session, communicates with performers

- Preamp and patch operator, PPO (1)  
Does level setting, patches pre-amps, monitors signal  
Setup stands, microphones, run cables
- Assistants (2 or more)  
Create primary documentation  
Setup stands, microphones, run cables

## **16.22. Recording Sessions**

- If you are working on a session, arrive as early as possible
- Come prepared with a specific microphone plan and position
- All must pay attention and document settings in track sheets; must write own track sheets and turn in at end of term
- Each member of group will be responsible for their own mix

## **16.23. Recording Practices: Procedure**

1. PPO zeros all pre amp levels, disengages all phantom power
2. All begin setup of microphones based on plan and expected instrument positions. LE oversees all microphone installations.
3. Assistants documents all channel assignments (microphone, wall input number, phantom power), tracing cables to ensure accuracy.
4. LE greets musicians, tells them where to set-up, has assistants provide necessary chairs, stands, power, equipment, etc
5. PPO powers phantom power for each channel necessary
6. PPO adds modest gain and checks for signal on each channel, having assistants check each microphone one at a time (snap test)
7. LE directs musicians to provide level-setting information.
8. PPO level-sets, directing the LE to get diverse material as necessary
9. LE initiates recording, tags audio with date and composition titles.

10. LE and PPO, in the case of excessive peaks, can cut the take and re-level set
11. LE directs additional takes as necessary
12. Strike: PPO zeroes preamp and turns off phantom power
13. Strike: LE oversees all microphone removal and storage
14. All cables, stands, and other equipment is stored

## Chapter 17. Meeting 17, Workshop: Recording Session 1

### 17.1. Announcements

- 11 April (Wednesday): Meeting 17, Workshop: Recording Session 1

Engineering crew: four students [names removed for privacy]

Instrumentation: 5 singers, including soloist

Location: Killian Hall

## Chapter 18. Meeting 18, Delay and Reverb

### 18.1. Announcements

- Recording session this Monday, 23 April, in Killian Hall  
Engineering crew: four students [names removed for privacy]  
Instrumentation: piano and horn  
Location: Killian Hall
- Need four person schlep crew for 3:00 PM on Monday
- Next quiz will be Wednesday, 25 April

### 18.2. Recording Session 1 Review

•

### 18.3. Reading: Katz: Aesthetics Out of Exigency: Violin Vibrato and the Phonograph

- What is a phonograph effect?
- What sources of evidence does Katz bring together to demonstrate the changes in vibrato practice?
- Katz offers five alternative theories on why vibrato usage increased. What are they, and why are each of them rejected?
- Why was vibrato useful for violinists making recordings?
- Are there other examples of necessity (or practicality) being the mother of aesthetics?

### 18.4. Processing Signals: Concepts

- Dry (unprocessed) and wet (processed)
- Sometimes replace dry with wet

- Sometimes mix a percent of wet and with dry

## **18.5. Processing Signals: By Replacement**

- Three terms: serial processing, inserts, in-line processing
- Applications: EQ, Dynamic Effects (compression, limiting, expansion, gating), Time Shifting, Spectral Effects

## **18.6. Processing Signals: By Mix**

- Three terms: parallel processing, auxiliaries, side-chain processing
- Applications: Time-based effects
- Side-chain can always be pre or post channel fader

## **18.7. Parallel Processing in Live**

- Use “Insert Return Track” to create (only two are permitted in Live Intro)
- Small, unlabeled boxes appear in each track’s lane to show return level (which can be automated)
- Pre- and post-signal routing selected in the Return track, not the source track



Courtesy of Ableton AG. Used with permission.

## 18.8. Time-Based Processors

- Reverbs
- Delays
- Flangers, chorus, and phasing

## 18.9. Time-Based Processors: Common Attributes

- All employ delays
- All are often processed in parallel (with an auxiliary track or with mix controls)
- All are often best used in stereo rather than mono
- All are easily over-used

## 18.10. Reverb: Goals

- Coherence: reconnecting tracks recorded in isolation or without space
- Recreating an acoustic space
- Special effects

## 18.11. Reverb: Parameters

- Time domain graph

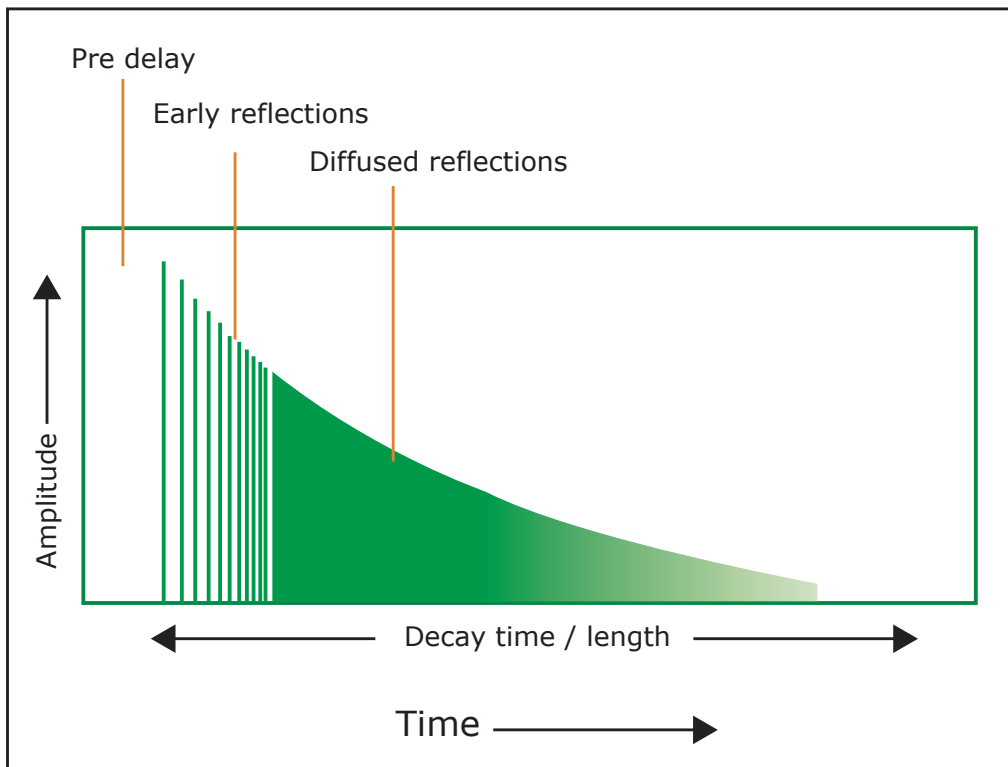


Image by MIT OpenCourseWare.

- Decay: duration of reverberations (time of tail to fall -60 dB)
- Size: color or type of diffusion algorithms
- Pre-Delay: time before reverb starts, a bit (30 ms) is generally needed to get reverb away from dry signal



- Early reflections
- Diffusion
- Wet / dry mix

## 18.12. Reverb in Live

- Basic reverb plugin



Courtesy of Ableton AG. Used with permission.

- Pre-processing filters
- Early reflections controlled by “Shape” parameter: higher values mean faster decay of early reflections
- Spin modulates the early reflections (not recommended)
- High and low frequencies in reverberation can have scaled decays
- Freeze/Flat/Cut: special effect of sustained reverb

- Density and scale: adjust reverberations
- Reflect and Diffuse: level setting for early reflections and reverberations

### **18.13. Reverb: Two Processing Approaches**

- Algorithmic (cheap, fast)
- Sampling or convolution based (expensive, slow)

### **18.14. Reverb: Parallel Processing**

- Reverb plugins should (almost) always be instantiated in auxiliary tracks and used with sends
- When in an aux track, reverb plugins should always be at 100% wet
- Having many tracks share a single reverb gives a sense of cohesion or shared space
- Aux sends permit adjusting how much of each channel will be processed as reverb
- Aux sends should (almost always) be post fader
- Aux track permits global reverb adjustments (level, filtering)
- Aux sends permit using a stereo reverb with a mono channel strip

### **18.15. Reverb: Two Needs**

- Cohesion
  - Decay: under a second; pre-delay: 5 to 10 ms
  - A short reverb to add ambience
  - Can simulate leakage
  - Can help tracks glue together
- Space
  - Decay: over a second; pre-delay: 30 to 70 ms
  - A longer reverb to simulate an acoustic space
  - Places a recording in an environment

## 18.16. Reverb: Algorithm Types

- Often determine arrangement of early reflections and timbre of reverberations
- Good to start with a preset then adjust
- Standard spaces: halls, rooms, chambers, ambience
- Unusual spaces: cathedrals, bathrooms
- Mechanical reverbs: springs and plates

## 18.17. Reverb: Filtering

- All reverbs need filtering
- Carefully shape (and reduce) high frequencies, avoiding metallic sounds
- Avoid extra low frequency reverb
- Use a full-function EQ to shape reverb
- Filtering should be tailored to the music

## 18.18. Reverb: Applications

- Not all tracks need reverb
- Use a shorter decay time than you think necessary
- Use sparingly on low-end tracks (kicks, basses)
- Use less reverb than you think necessary (mastering likely to increase)

## 18.19. Reverb: Auditioning

- Start and stop tracks to listen to reverb alone
- Vary aux channel level to boost level to adjust timbre, then reduce

## 18.20. Microphone Positioning: Exercise

- Exercise: You are to recording a piano and a horn. You have 6 AT 4041, 4 AKG 414, 2 Earthworks TC20mp, and 2 Sennheiser MD-421.

## Chapter 19. Meeting 19, Workshop: Recording Session 2

### 19.1. Announcements

- 23 April (Monday): Meeting 19, Workshop: Recording Session 2

Engineering crew: four students [names removed for privacy]

Instrumentation: piano and horn

Location: Killian Hall

## **Chapter 20. Meeting 20, Workshop: Various Topics**

### **20.1. Announcements**

- Next recording session (#4) a week from today. There will not be a Recording Session 3.
- Last recording a week from Monday, in Killian
- Reading for next class is important (and a little long)

### **20.2. Recording Session 2 Review**

- 

### **20.3. Quiz 4**

- ?

### **20.4. Approaches to Mixing**

- Standard mix examples: vocal/piano, jazz ensemble
- Nonlinear mixes: NIN

### **20.5. Mixing Close and Distant Captures**

- We generally only need a little bit of distant captures: minimize by amplitude and/or frequency
- We generally need to delay close captures to align with distant captures: 10 feet is about 9 msec delay.
- Can use time-domain view to look at alignment

# Chapter 21. Meeting 21, Analog and Digital Audio Fundamentals and Mediums

## 21.1. Announcements

- Recording Session 4: Wednesday: here  
Engineering crew: four students [names removed for privacy]
- Recording Session 5: Monday: Killian  
Engineering crew: four students [names removed for privacy]
- Need four-person shlep crew for each day

## 21.2. Delay: Parameters

- Delay time: time before repeat
- Feedback: gain applied to signal after delay fed back into delay
- Filters
- Wet / dry

## 21.3. Delay: Feedback

- To create one echo use a feedback of zero
- A feedback of 1 will create an infinite number of echos

## 21.4. Delay: Types

- Slapback: single delay, delay about 35 to 100 ms, functioning as a short reverb
- Ping-Pong or stereo: echos change stereo positions

## 21.5. Delay in Live

- Three types: Simple Delay, Filter Delay, Ping Pong Delay
- Toggle Sync/Time button to get direct control independent of tempo

## 21.6. Delay: Tips

- Often use filtering
- Often practical use in an aux track as delay
- Very short single delays can be used for double tracking
- Time delay to musical tempo:  $60,000 / \text{BPM} == \text{beat duration in milliseconds}$

## 21.7. Reading: Lazzarini, Introduction to Digital Audio Signals

- What are the two steps of digital encoding?
- How does the sampling rate limit what frequencies can be encoded?
- How does the quantization (and bit depth) determine what amplitudes can be encoded?
- What is PCM audio? What is not PCM audio?
- What does digital audio aliasing sound like?
- How are mixing, scaling, and offsetting signal implemented in a digital system?
- What are Fourier series?
- What is the difference between FIR and IIR filters?

## 21.8. MOSS: New Microphones

- AT M250DE (1)  
Dual-element instrument microphone



- e604 (1)

Dynamic cardioid w/ more than 160 dB dynamic range



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- Blue enCORE 200 (4)

Active dynamic cardioid





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## 21.9. Microphone Positioning: Exercise

- Exercise: You are recording 14 singers, 7 male and 7 female. You have 6 AT 4041, 4 AKG 414, 2 Earthworks TC20mp, and 2 Sennheiser MD-421.

## 21.10. Microphone Positioning: Exercise

- Exercise: You are recording a large ensemble. You have 6 AT 4041, 4 AKG 414, 2 Earthworks TC20mp, 2 Sennheiser MD-421, 1 AT M250DE, 1 e604, 4 enCORE 200, and 2 mono and 1 stereo direct box

## Chapter 22. Meeting 22, Workshop: Recording Session 4

### 22.1. Announcements

- 2 May (Wednesday): Meeting 22, Workshop: Recording Session 4

Engineering crew: five students [names removed for privacy]

Instrumentation: gtr, bs, drum kit, 3 vocal, tpt, sax, pno, keyboard

Location: TBA

## Chapter 23. Meeting 23, Workshop: Recording Session 5

### 23.1. Announcements

- 7 May (Monday): Meeting 23, Workshop: Recording Session 5

Engineering crew: four students [names removed for privacy]

Instrumentation: 14 singers, 7 male, 7 female

Location: Killian Hall

## Chapter 24. Meeting 24, Dithering and Mastering

### 24.1. Announcements

- Mix Report 2 due Wednesday 16 May (no extensions!)
- Track Sheet Logs: show me after class today or monday
- Subject evaluations!

### 24.2. Review Quiz 4

- ?

### 24.3. Recording Session Review

- From Meeting 17, Workshop: Recording Session 1  
Instrumentation: 5 singers, including soloist
  
- From Meeting 19, Workshop: Recording Session 2  
Instrumentation: piano and horn
  
- From Meeting 22, Workshop: Recording Session 4  
Instrumentation: gtr, bs, drum kit, 3 vocal, more

- From Meeting 23, Workshop: Recording Session 5

Instrumentation: 14 singers, 7 male, 7 female

## **24.4. History**

- Mastering was necessary due to limitation of mediums (records)
- A master specifically referred to an object used to make copies
- Contemporary mastering is really the preparation of a premaster

## **24.5. Motivation**

- Maximize dynamic range
- Sweeten, optimize, and make a mix gel for as many playback systems as possible
- An outside consultant on sonic quality and balance
- A means of competing for attention
- “Mastering is the art of compromise” (Katz 2002, p. 100)

## **24.6. Analog and Digital Mastering**

- Analog mastering is still very popular, but is very expensive
- Many desire to add analog warmth into digitally recorded and/or mixed music
- Digital is cheaper, more repeatable

## **24.7. Training Your Ears**

- Mastering takes experience
- Hearing masters on multiple systems is critical
- High-quality playback systems, and multiple playback systems, are nice, but not required

## 24.8. Metering

- Good, reliable digital meters are critical
- Need to look at peak and average levels levels, possibly with frequency-dependent weighting
- Example: Inspector: IXL Level
- Example: Level Meter (Logic): use two instances, one at Peak, another at RMS

## 24.9. Maximum Peak Levels

- Loudness is not determined by peak level; mastering is not normalization
- No samples in a mix or master should reach 0 dBFS
- Maximum master peak should never be greater than -0.2 dBFS
- Additional head room (-3 dBFS) may be valuable
- Amount of peak movement matters: stuck (pegged) meters are never good
- Peaks should be balanced between L/R channels

## 24.10. RMS

- RMS: root mean square, or the square root of the average of values (from within a window) squared
- Better than VU for evaluating loudness
- May or may not be weighted according to Fletcher Munson
- Mastered audio is generally in the range of -8 to -16 dB RMS

## 24.11. The Loudness War

- Mastering has increased overall loudness of recordings in recent decades
- Statistical Evidence from Nielsen, S. H. and T. Lund. 2003. "Overload in Signal Conversion." In *Proceedings of the AES 23rd International Conference*.

- James Brown (1986): average at -16 dB, peak at -3.4 dBFS
- Back Street Boys (2000): average at -5 dB, peak at 0 dBFS

## **24.12. Loudness War: Waveforms and Listening**

- John Coltrane: My Favorite Things (1961)
- The Roots: Ital (The Univesal Side) (Illadelph Halflife, 1996)
- The Roots: Guns are Drawn (The Tipping Point, 2004)

## **24.13. Mastering Is Not (All) Evil**

- 2006: Hank Plank and the 2x4s: Planks of Grass: Shameful Me
- 2007: Various: RESONANCE: Steel Pan in the 21st Century: Ariza: phanopoeiac
- 2011: Peter Evans Quintet: Ghosts: Ghost
- 2012: Alexander Sigman: Nominal / Noumenal: Entartete
- 2010: Architeuthis Walks on Land: Natura Naturans: Pickup Track
- 2008: Various: SPECTRA: Guitar in the 21st century: Jandek: The World Stops

## 24.14. Basic Steps and Bits

- Mix from 16 or 24 bit sources without master-bus processing
- DAWs mix internally at high bit depths (32 or 64) to offer headroom
- Bounce to disc a 24 bit stereo mix
- Create a new session for 24 bit mastering
- Bounce to disc a 16 bit stereo mix

## 24.15. A Bit of Review

- Bits are discrete data
- 16 bit audio stores 65,536 amplitude positions for 96 dB dynamic range
- 24 bit audio stores 16,777,216 amplitude positions for 144 dB dynamic range
- Bit depth x 6 == dynamic range dB
- Internal DSP processing in DAW is at least 32 bit

## 24.16. Falling Between Bits

- Imagine 2 bit (using only 3 amplitude position) encoding ADC
- Input range is between -1 and 1 volt, encodes amplitude positions at -1, 0, and 1
- If a DC voltage enters at .35, signal will be encoded as zero, and all information is lost
- If a small amount of random samples (white noise with an amplitude of at least .25) is added to the input signal, some samples will be encoded at 0 and others as 1
- The average of many encoded samples will be .35

## 24.17. Dithering

- Adding noise to extend dynamic range downward; often most significant for sounds at bottom of dynamic range
- Exercises, toggles, or modulates lowest bits
- Dithering is always used when moving from a high bit depth to 16 bits



- Dithering should never be performed twice
- Without dither, truncation and poor dynamic range results
- Permits 16 bit audio to extend dynamic range below -96 dBFS; best dithering can result in a perceived dynamic range as great as 115 dB (near 19 bit resolution)

## 24.18. Dithering: Noise Shaping

- EQ the spectrum of the applied noise to reduce perceptibility
- Avoids frequencies that are loud on Fletcher Munson (around 3 kHz)
- Different types of dithers use different noise shapes

## 24.19. Basic Steps and Bits

- Mix from 16 or 24 bit sources without master-bus processing
- Bounce to disc a 24 bit stereo mix *without* dither
- Create a new 24 bit session for mastering processing
- Bounce to disc a 16 bit stereo mix *with* dither and noiseshaping

## 24.20. Dithering Processors

- May be stand alone or coupled with other mastering dynamics processors (limiters and/or compressors)
- Main parameter is bit depth (output) and noise shaping parameters
- Numerous limiters have dithering included
- Be careful to not add dither twice
- Example: Sonnox Oxford Limiter Dither
- Example: Logic Bounce Dither options: Apgee UV22HR, Pow-r #3
- Example: Live Export Audio



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## 24.21. Mastering Setup: Monitor Usage and Calibration

- Listening on multiple monitors is essential in mastering
- 0 dB position of monitors should produce 83 dB SPL with pinknoise
- Pinknoise should output at -20 dBFS RMS
- 83 dB lands at best point in Fletcher Munson

# Fletcher-Munson Curves

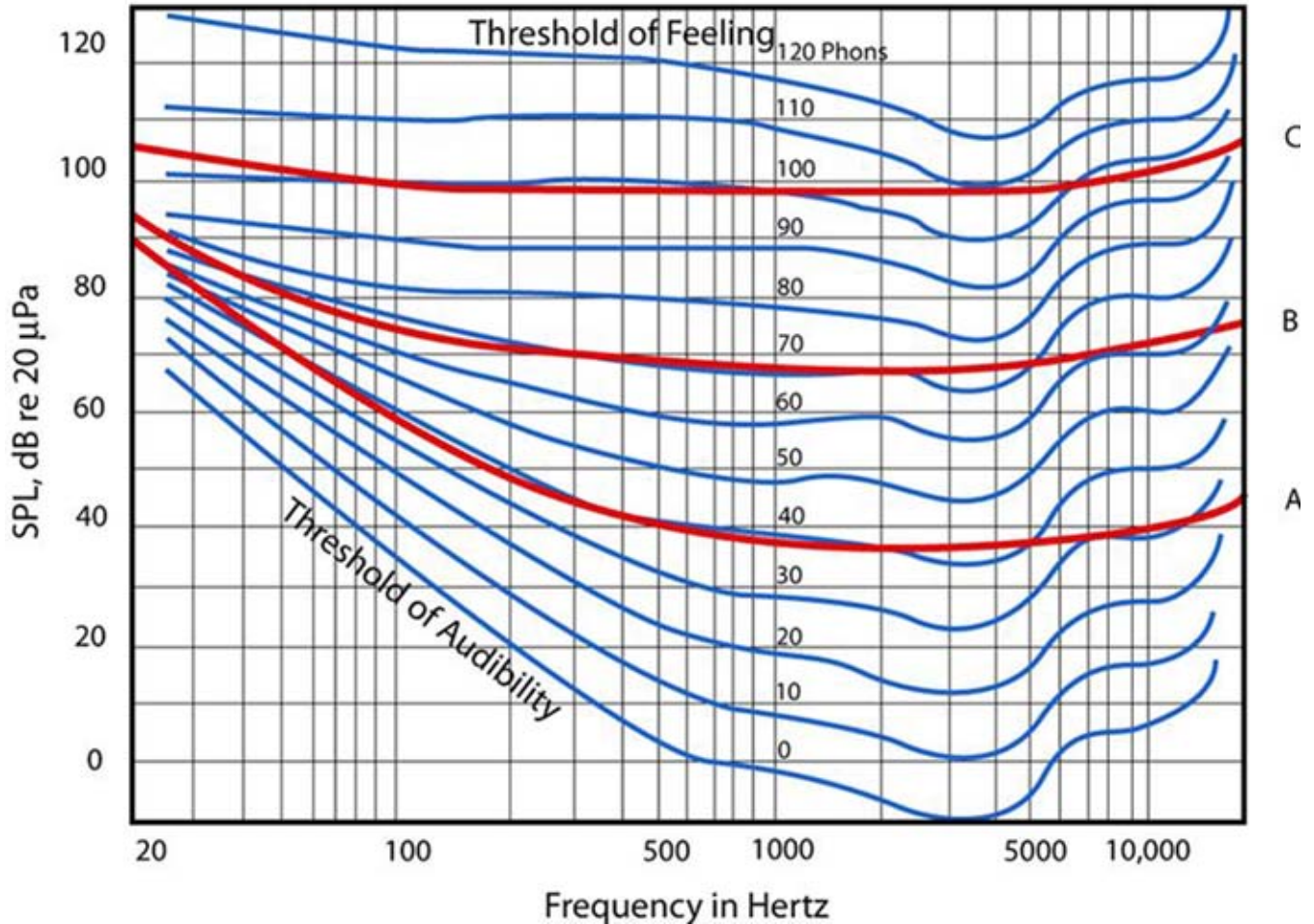


Image: "Fletcher-Munson Curves" from *Principles of Industrial Hygiene*. Available at: <http://ocw.jhsph.edu>. License CC BY-NC-SA, © Johns Hopkins Bloomberg School of Public Health.

## 24.22. Mastering Track Setup

- Create a source track and a master track
- Add mastering inserts to source track
- May have a duplicate clean source track for quick comparison
- Add metering and visualization plugins to master track

## 24.23. Mastering Processors

- Less is more and quality matters
- Limiters: peak limiters, brick wall limiters, mastering limiters
- Filters: parametric filters, linear phase filters, dynamic filters

- Dynamics: manual fade adjustments (macrodynamic manipulation), multi-band compressors/expanders/gates, leveling amplifiers
- Exciters, saturation processors, tube emulators, maximizers
- Noise reduction processors
- Time/phase adjustments, stereo optimization, mid-side adjustments
- There is no standard set of processors to employ: each mix is different

## Chapter 25. Meeting 25, Formats and Distribution

### 25.1. Announcements

- Mix Report 2 due Wednesday 16 May (no extensions!)
- We might listen to mixes if available before class
- Track Sheet Logs: show me after class today
- Subject evaluations!

### 25.2. Reading: Millard: Tape Recording and Music Making

- What form of recording did the earliest electromagnetic recorders replace?
- For what applications did musicians use portable magnetic recorders? What does this suggest about the kinds of music these musicians were interested in?
- Other than duration, what were some other advantages of recording on tape?
- What were some features that led to the success of the Philips compact cassette?
- Why does the author suggest that “rap ... could only have begun on cassette tape”?
- In what ways did the cassette affect the development of world music?
- Has digital technology, like tape, changed the “corporate nexus of independent companies and media conglomerates” in the music industry?

### 25.3. Ordering Mastering Processors

- Exciters should be early in signal path
- EQ might come before or after compression (prefer before with multiband compression)
- A final limiter is often (nearly always) the penultimate processor
- Dither is always last

## 25.4. Sample Setups

- Sample minimal: filter, limiter, dither
- Sample maximal: manual fading, exciter, filter, multi-band compressor, limiter, dither

## 25.5. Mastering Dynamics: Limiting

- Increase loudness with little change to mix
- Remove “unnecessary” transients or spikes
- Will not drastically change relationship between instruments in the mix
- Limiting should be fast and shallow (low threshold around -3 to -6 dBFS); some processors set input level into a fixed (0 dB) threshold
- Output ceiling should be set no higher than -0.2 dBFS
- May create flattened, distorted, and lifeless sound
- Example: Sonnox Oxford Limiter
- Example: Logic Limiter
- Example: Live Compressor as Mastering Limiter



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## 25.6. Mastering Dynamics: Compression

- Increases loudness
- Favor ratios less than 3:1
- Favor long attacks (over 14 ms) and short releases
- Most transparent compression with small ratio (1.01 to 1.1:1) and deep threshold (-30 to -40 dBFS)
- Preserve transients: long (high) attack; Avoid noticeable release times
- May drastically change relationship between instruments in the mix
- May add punch and strength, make tracks gel
- Activity in some frequency ranges may have negative side effects for other frequency ranges
- May create flattened, distorted, and lifeless sound
- Example: Sonnox: Oxford Dynamics

- Example: Logic Compressor
- Example: Live Compressor as Mastering Compressor



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## 25.7. Mastering Dynamics: Multiband Processors

- Benefits: each band is independently optimized
  - Example: a peak in the vocal line will not turn down the bass
  - Example: high frequency overtones and transients remain while mid-range dynamics are controlled
- Detriments: alters balance of mix, alters phase, may phase distort at crossover transitions
- “The multiband device’s virtues permit louder average levels than were previously achievable -- making it the most powerful but also potentially the most deadly audio process that’s ever been invented.” (Katz 2007, p. 128)
- Use as few bands as necessary, from 2 to 5



- Tune bands to the particular material
- Can aim for comparable gain reduction in each band  
Can aim for gain reduction only in specific bands
- Favor slow attacks, low ratios, and deep thresholds
- Favor slow attacks, low ratios, and deep thresholds
- Example: Apple Multiband Compressor
- Example: Izotope Ozone 5 Dynamics

## 25.8. Mastering Filters

- Goal of achieving tonal balance
- Adjustments have secondary / complimentary effects
- Less is more: +/- 3 dB may be sufficient
- Adjustments alter internal balance of mix
- Adjustments should be listened to for long durations and with A/B comparisons

## 25.9. Mastering Filters: Common Applications

- Focusing middle range: using a parametric to boost or cut
- Controlling bass: boosting between 80 and 120 Hz while reducing below 60 Hz
- Sparklies or air band boost: initially seductive but can cause long term fatigue
- DC Offset removal: HPF at 20 Hz
- Filters may have a sound independent of filter settings
- Example: Logic Fat EQ
- Example: Sonnox Oxford Equaliser & Filters

## 25.10. Mastering Exciters

- Generate favorable distortion
- Add harmonics: different processors add different combinations of harmonics

- May lead to unmusical or excessively bright or edgy results
- May add warmth and presence to dry and cold mixes
- May model tube or analog processing or saturation
- Example: Logic Exciter
- Example: Sonnox Oxford Dynamics
- Example: Izotope Ozone 5 Exciter

### **25.11. Mastering Maximizers**

- Psychoacoustic or other perceptual techniques
- Specialized limiting
- Example: Sonnox Inflator
- Example: Izotope Ozone 5 Maximizer

### **25.12. The Complete Processing Chain**

- As little as necessary from each processor
- As few processors as possible
- Example: Dirt Feelin
- Example: Katherine Young's Pretty Monsters: Feldspar

## Chapter 26. Meeting 26, Studios

### 26.1. Announcements

- Mix Report 2 due today (no extensions!)
- Track Sheet Logs: show me after class today

### 26.2. Disc Formats into the 1950s

- 78 RPM discs
  - 1900 to 1925 discs recorded between 74 and 82 rpm
  - 78 rpm based on a 3600 rpm motor with 46:1 gear ratio: 78.26 rpm
  - Covered in shellac
  - Available in 10 inch (3 minutes) and 12 inch (4-6 minutes) formats
- 33.333333 RPM discs
  - Columbia Records: June 1948 releases Long Playing Record
  - Use of more-narrow grooves (microgroove)
  - Use of vinyl offered better sound quality
  - 12 inch diameter, 30 minutes or more per side
- 45 RPM discs
  - RCA Victor introduces in 1949
  - 7 inch diameter, 4 minutes per side
  - Designed to have uniform size, easy distribution, automatic changers (jukebox)
  - Became known as “singles”: one tune per side
  - The B or flip side offered a bonus track

- Extended Play (EP) 45s achieved 7 minutes per side

### **26.3. Early Magnetic Recording Devices**

- 1930s: Magnetophone (AEG, Germany)
- 1940s: Commercially developed in the late 1940s by American Jack Mullin with Bing Crosby
- Reel to reel audio tape recording machines spread in 1950s with companies like Ampex

*The BEST BUY in Recorders!*  
 Complete PUSH-BUTTON CONTROL

- CUT TAPE COSTS IN HALF
- OUT PERFORMS EVERYTHING IN ITS PRICE CLASS
- HIGHEST QUALITY & RESPONSE EVER BUILT INTO A PORTABLE

15,000 cps at 15 & 7 1/2 inches per second

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 AMPEX ELECTRIC CORPORATION  
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**The portable recorder you specified**

now the famous  
**Ampex portable fits your professional recording needs exactly**

**Low Impedance Output . . .**  
All lines, 150 volts, balanced or unbalanced, from tape recorder program level.

**Low Impedance Input . . .**  
Sensitizes to high or low impedance microphones. Low impedance inputs are secured by necessary plug-in transformer.

**Instantaneous Start-Up . . .**  
Cooling to 1/2 full speed in less than 1/10th of a second.

**Illuminating Record Safety Button . . .**  
Provides instant warning of imminent erasure.

**Saddle Tan Case . . .**  
Hand-carried in case of rugged construction for extra portability and safety.

**HIGHEST QUALITY IN THE SMALLEST PACKAGE**

**Superb Fidelity . . .** Frequency response: 30 to 15,000 cps, at 75 db/sec. Down to noise floor 4 db at 15,000 cps. Up to 40 db at 10,000 cps. Speed-tape rate over 55 db. Filter and Equalizer (E.P.T.). Separate magnetic and playback heads.

**Accurate Timing . . .** Precision timing circuit. Timing accuracy within 20 microseconds in 30-second playback.

**Variable Classification . . .** Variable speed tape transport. Variable timing. Large scale meter for precise program level. Control for automatic erasure prior to playback.

**Dependable Service . . .** All parts have been tested in operation of more than 10 years. Full service and repairing organization available for maintenance.



The 300 really fits in your case. It's portable, portable performance. It's the 300, the 300, the 300 professional portable recorder.

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- Multitrack recording on tape, pioneered by Les Paul, developed as early as 1954

## 26.4. Analog Audio Multitracks: Les Paul

- 1940s: Guitarist Les Paul (1915-) experiments with adding and bouncing tracks in direct to wax disk recording
- 1948: produced “Lover (When You're Near Me)” album with this technique, combining up to 8 guitars
- Modifies an Ampex Model 300 mono tape recorder to record multiple individual tracks

FOR

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Tape recording is superior to all other reproduction methods and "AMPEXED TAPE" has the greatest fidelity and range now possible. Simplified operation plus sure results make AMPEX unexcelled for all critical recording uses. Dual tape speeds with automatic speed and equalization change is but one of many exclusive AMPEX features.

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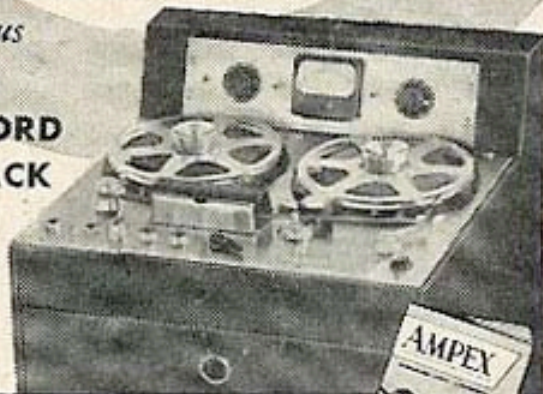
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Panel \$114 Extra



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| <input type="checkbox"/> Laboratory Research     | <input type="checkbox"/> Telemetry             |
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**AMPEX ELECTRIC CORPORATION**  
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**DEALERS IN PRINCIPAL CITIES**

- By 1953 develops first 8 track recorder
- Employed different playback speeds of each track
- In class listening: Les Paul and his wife (audio)

## **26.5. Reading: Horning, From Polka to Punk: Growth of an Independent Recording Studio, 1934-1977**

- Describe the trajectory of recording and mixing equipment used at the Cleveland Recording Company.
- Describe the trajectory of clients that recorded at Cleveland Recording Company.
- What were some of Hamann's technical achievements?

## **26.6. Tom Dowd: Recording Engineering Innovator**

- “Tom pushed those pots like a painter sorting colors. He turned microphone placement into an art” (Atlantic's Jerry Wexler on Dowd; Horning 2002, p. 144)
- Video clip: Tom Dowd: The Language of Music, Chapter 2 (00:02-01:16, 2:42-3:47, 4:10-5:08)
- Video clip: Tom Dowd: The Language of Music, Chapter 7 (3:40-7:05)

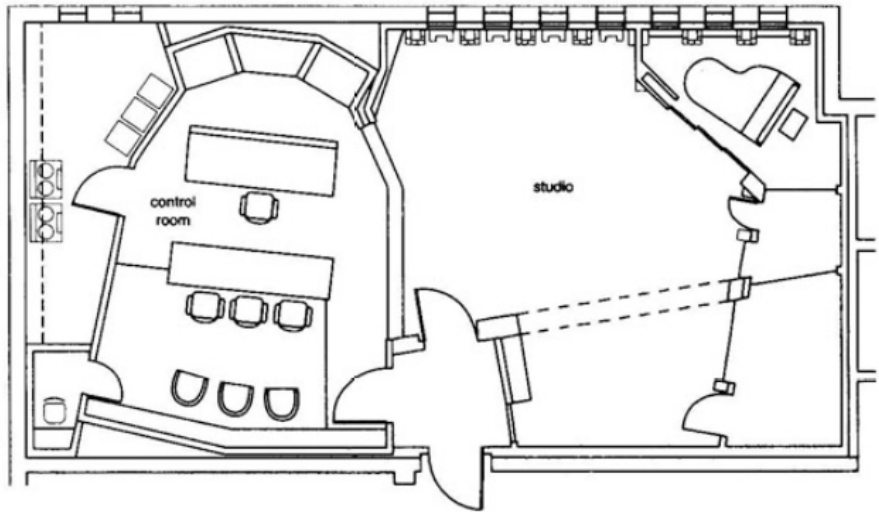
## **26.7. Monitoring and Studios: Simultaneous Recording**

- To improve isolation during simultaneous recording, studios have multiple (isolation) rooms (booths)
- Permit visual contact (windows, video) and aural interconnections
- Requires a monitor feed to be sent from the recording unit to each musician
- Permits musical, expressive performances and great mix flexibility

## **26.8. Studio Design Examples**

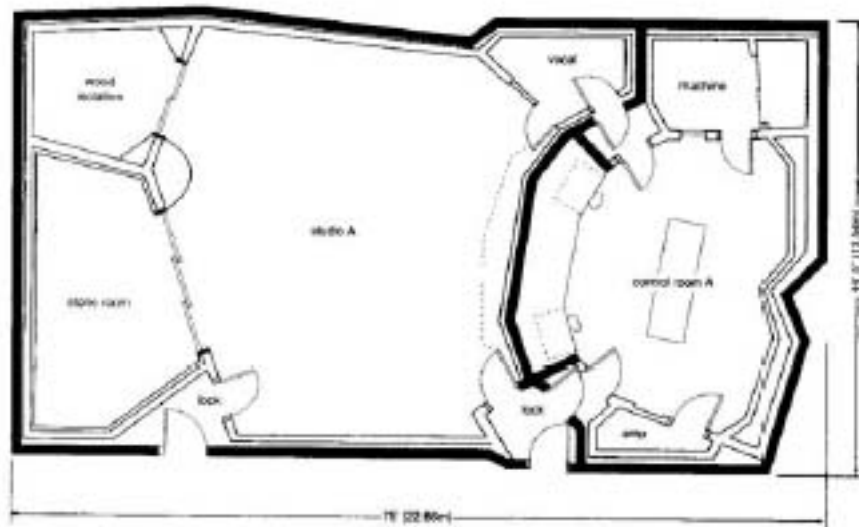
- Sony/Tree's Music Studio, Nashville





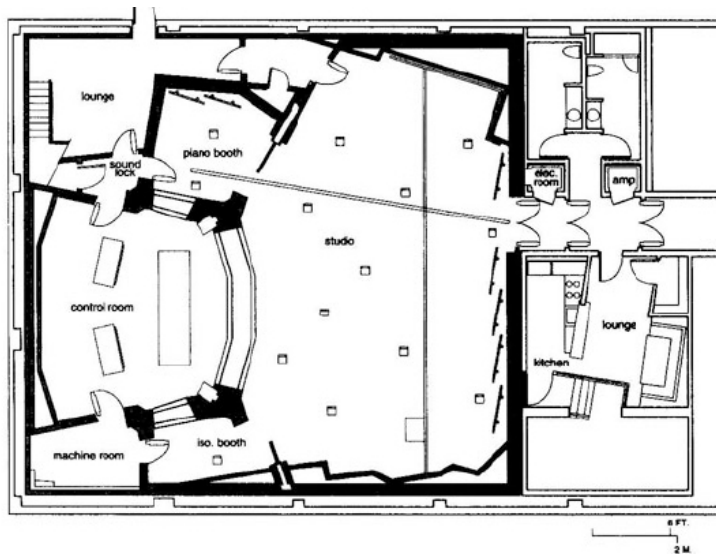
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- Paisley Park's Studio A



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- Studio X, Seattle



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## 26.9. Monitoring and Studios: Non-Simultaneous Recording

- For maximum isolation, can record each part (or section) separately and then overdub
- Requires a monitor feed of previous tracks to be sent to each musician
- Possible benefits using a consistent (or composed) tempo via a click track
- Permits greatest mix flexibility, but sometimes challenging performance contexts

## 26.10. Headphone Monitoring with MOSS

- From within the DAW, create auxiliary channels for each monitor channel
- Use send controls on each channel to send to the appropriate aux channel
- Route output of aux channels to physical outputs on the computer interface (RME)
- Patch up to 8 monitor channels into HearBack Hub inputs
- Distribute personal mixers via CAT-5 cables



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## 26.11. Mix Report 2 Examples

- ?

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