Music 499 section ART

Audio Recording Techniques

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Very Basic Concepts

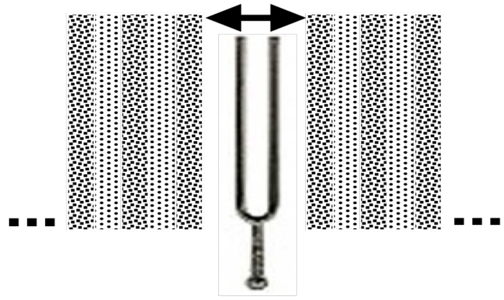
Sound – combinations of frequencies and amplitudes emanating from a vibrating source, transferred through a medium (gaseous, liquid or solid) by means of a series of stronger and weaker areas of molecules (waves of compressions and rarefactions) – eventually received by our inner ear and translated into what we call sound.

Sound has **frequency** (or pitch) and is also heard as having variable **amplitude** (variable volume intensities)(dynamic level). **Frequency** is determined by *how often (how many times per second*) the originating source vibrates back and forth, and **amplitude** (the vibration's intensity) is determined by *how loud or soft the originating source is [or how strong or weak the vibration is]*.

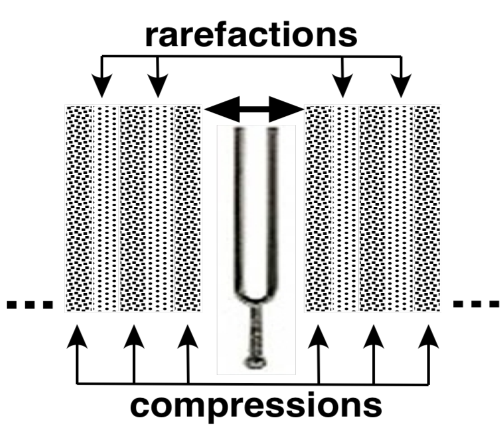
Acoustics - the study of the physical properties of sound;
- the study of the physics and transmission of sound.

Compiled and edited by Scott A. Wyatt

1. Sound is produced by a vibrating source.
2. These vibrations come in contact with air molecules that are in the immediate vicinity of the vibrating source, and create areas of higher air pressure and areas of lower air pressure in synchronization with the vibrations.

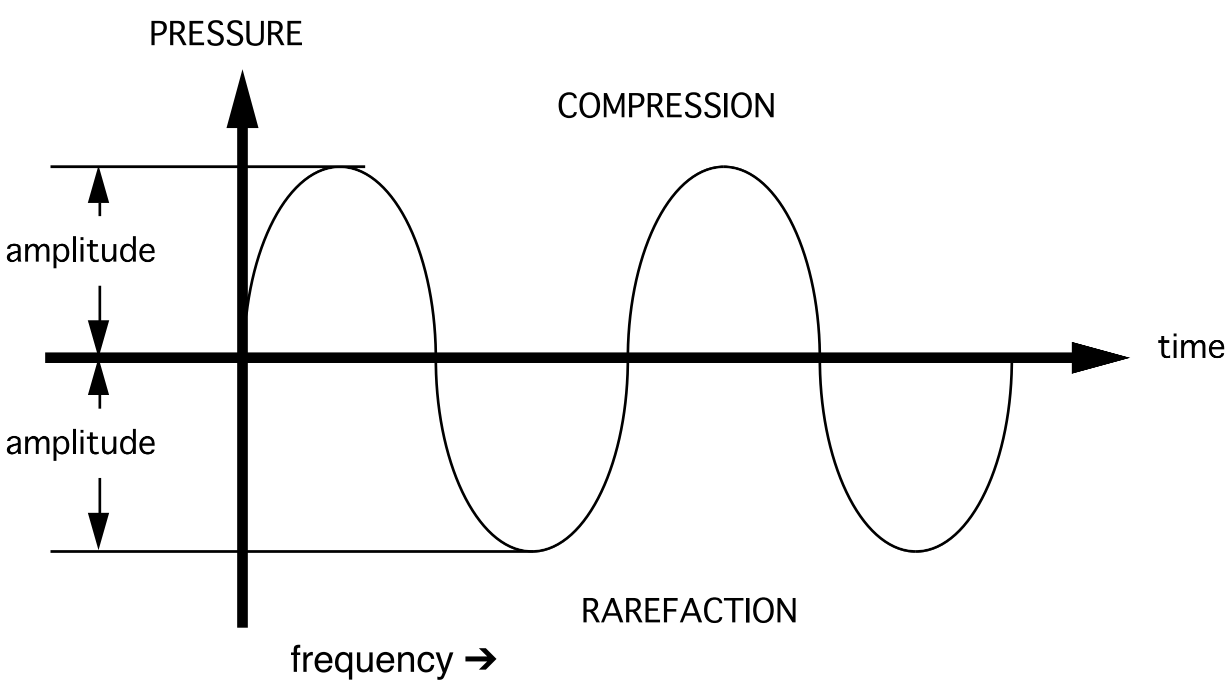


3. The area of higher air pressure is called a "compression".
4. The area of lower air pressure is called a "rarefaction".



5. These areas of compressions and rarefactions continue through the air in the form of a sound wave just like waves on the surface of water.
6. When a sound wave comes in contact with another surface, it causes that surface to vibrate in synchronization with the sound wave. This is how energy is transferred from one source to another while maintaining the characteristic vibration pattern of the original source.
7. The pattern of the pressure variations over time is called the **waveform** of that sound.
8. Waveforms are made up of a repeating pattern.
9. **Frequency** is the number of cycles of vibration that occur in one second. Therefore, the rate at which one full **cycle** of the waveform repeats itself is called the **frequency**.
10. Frequency (the number of cycles per second) is measured in units called **Hertz**. 440 cycles per second is equivalent to 440 Hertz. Most humans are able to hear a frequency range of approximately 30 Hertz to 16,000 Hertz. The range of frequencies 20 Hertz to 20,000 Hertz is often described as **audio frequencies**.

    Frequencies lower than 20 to 30 Hertz and that are inaudible to humans are described as **sub-audio** or **sub-sonic frequencies**. Frequencies higher than what humans are capable of hearing are described as **ultra-sonic frequencies**.



11. **Amplitude** (intensity or strength) is the amount of change, positive or negative, in:
   - atmospheric pressure caused by the compression/rarefaction cycle of a sound wave;
   - the maximum distance that a mass travels away from the equilibrium point;
   - the extreme range/distance of a fluctuating quantity, as an alternating current, swing of a pendulum, vibrating source, waveform, etc., measured from the average or mean (point of equilibrium) to the extreme. The intensity of a sound is directly related to the amplitude of the vibration.

   The **decibel** (**dB**) is used for the purpose of comparing the intensities of two sounds. In acoustics, the reference **sound pressure level** (**SPL**) is the threshold of audibility (0 dB SPL). A sound at the threshold of pain is considered to be 130 dB SPL. When referencing sound pressure levels, the decibel units are always positive, as there is no level below the threshold of hearing.

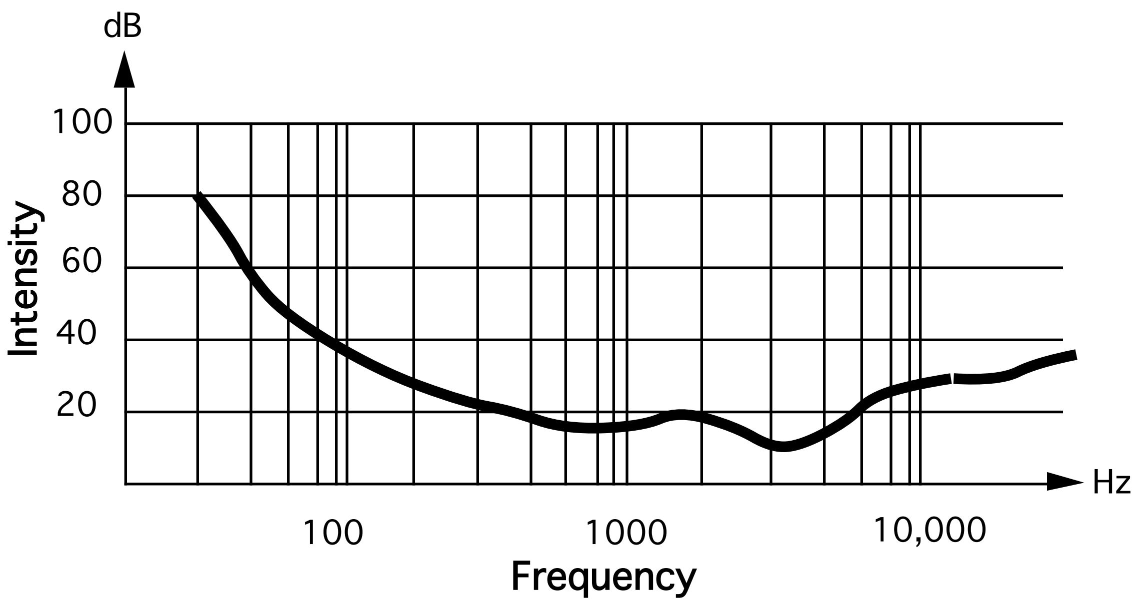
   However, the term decibel is <u>also</u> used as a unit for comparing the intensity of two electrical signal levels with the largest intensity level being 0 dB - all other levels would be below and consequently a negative unit of measurement.

12. All sound is made up of a combination of **sine waves** of various frequencies and amplitudes. This combination of sine waves is what determines **timbre** or the **tone color** of a sound, and is heard as a *composite sound or composite waveform*. A sine wave is a pure tone having no overtones.

Psychoacoustics - the study of the perception of sound.

Pitch is our subjective response to frequency. The frequencies from 25-30 to 4000 Hz. comprise the region of greatest perceptual acuity and sensitivity to change in frequency. Pitch is perceived as a steady-state tone having either a fundamental frequency or center frequency. Although frequencies exist lower than 25 or 30 Hertz, and higher than 20,000 Hertz, those lower frequencies (if heard at all) may be heard as clicks rather than a steady-state tone (or pitch) and those frequencies higher than ~20,000 Hertz would not be heard by humans (therefore not being perceived as pitch or sound).

The sensation of **loudness** is primarily determined by the amount of acoustical energy received by the ear. In addition, research (Fletcher and Munson) has shown that the perception of a sound's loudness is also related to its frequency.



The above graph indicates that human ears are not very sensitive to low frequencies, not very sensitive to those frequencies above 10,000 Hertz, and particularly sensitive to frequencies between 3000 to 5000 Hertz. A sound of 50 Hz must have the intensity of approximately 55 dB if it is to be heard as being just as loud as a 3000 Hz sound that has an intensity of only 15 dB.

There are 2 main types of electrical current:

- Alternating Current (AC): electrical energy that changes direction and amplitude periodically (frequency and amplitude)

- Direct Current (DC): electrical energy that moves in one direction and is of constant amplitude

Electrical current is used to represent sound information within audio equipment.

Analog audio uses Alternating Current to represent sound because its signal shape (waveshape) is *analogous* to sound waveform information (frequency and amplitude).

Digital audio uses Direct Current as a binary coding to represent the frequency and amplitude parameters of sound.

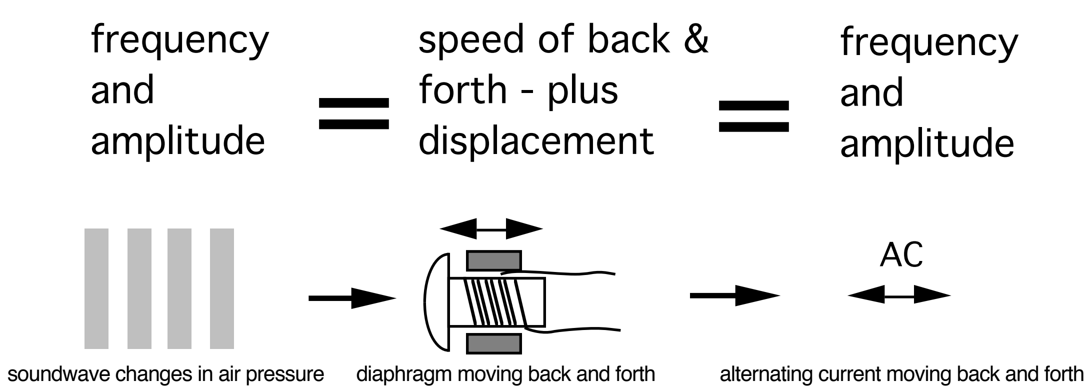
Analog Audio Signal

An analog audio signal is an Alternating Current representation of the sound whose periodic changes in direction and amplitude are directly proportional (analogous) to the frequency and amplitude of the sound - hence the term *analog*.

- Soundwaves consist of periodic alternations (of higher and lower air pressure) that are initiated by a vibrating source. The frequency of the soundwave is determined by how quickly these higher and lower air pressures alternate through a complete cycle. The amount of distance away from the point of average air pressure is called the sound wave's amplitude or intensity or volume. Intensity (or amplitude) is the magnitude of variance in air pressure resulting from the sound.

- When sound waves enter a microphone, the individual changes in air pressure physically move the microphone diaphragm back and forth—at the same rate of frequency and with the same changes of intensity of the sound wave. The diaphragm movement is directly proportional to the frequency and amplitude of the soundwave.

- The speed of the diaphragm movement, as well as the extent of how far the diaphragm moves from its point of equilibrium, is converted (by either electromagnetic or electrostatic conversion) to an electrical signal that changes direction and amplitude strength periodically—in direct proportion to the diaphragm movement. This electrical signal, one that moves forward and backward periodically, and one that changes amplitude periodically, is alternating current or AC.



- Since these periodic fluctuations and intensities of current are *analogous* to the periodic fluctuations and intensities of the original sound wave, the electrical signal generated by this conversion process is called an analog audio signal (a continuous variable, defined with infinite precision).

Standardized Analog Audio Levels

In the analog audio world, there are 4 signal amplitude levels that we deal with: mic, instrument, line, and speaker level.

Mic level is the lowest, or weakest amplitude level signal of the four and requires a preamplifier to bring it up to the standardized line level.

Mic level is generally about 2 mV (0.002 volts) to about .2 V depending on how loud the source is, and on how sensitive the mic is, as well as its output level capability.

Instrument level signals are between mic and line level signals and have the most variation due to a lack of standardization among manufacturers within the industry. You typically see this kind of signal come from keyboards, synths, and from "some" electric guitars. (Many electric guitar outputs are considered to be mic level.) Instrument level requires a form of amplification to come up to line level.

Instrument level is typically 0.1 V to 0.6 V for passive guitar pick-ups and consumer keyboards, and up to 1 V for active guitar pick-ups and professional keyboards.

Generally, instrument level should be connected to a line level input with variable gain capability to bring instrument level up to line level. Sometimes – if you know the voltage output of the transducer or pick-up from the electric guitar is at the lower amplitude output, you can connect the guitar to the mic input of the mixing console, however, EXTREME CAUTION is needed to not overdrive the input of a mic preamplifier or the mic input of a mixing console, as this can cause serious damage.

Line level signals are the highest amplitude level signals before the power amplifier stage, and are higher amplitudes than instrument level.

Typical line level amplitudes range from .32 to .42 V for consumer gear (unbalanced circuitry), and approximately 1.23 V for professional gear (balanced circuitry).

Consumer line level (0 VU) is commonly rated -10 dBV (where reference value (0 dB) = 1 volt. Therefore −10 dBV = 1 x 10^−(10/20) volts = 1 x 10^−0.5 volts = 1 x 0.316 volts = ~ 0.32 volts.) Some manufacturers have pushed this level to .42 volts.

Professional line level (0 VU) is commonly rated +4 dBu (where reference value (0 dB) = 0.775 volts (775 millivolts). The "u" stands for "unloaded", and today is used almost universally in preference to dBv. Therefore +4 dBu = 0.775 x 10^(4/20) volts = 0.775 x 10^0.2 volts = 0.775 x 1.58 volts = 1.23 volts.

The one thing you really need to remember when working in a Professional Line Level Studio is 0 VU = +4 dBu = 1.23 V = -18 dBFS. Be EXTREMELY CAREFUL when interconnecting professional gear with consumer gear as this can cause distortion, noise, and/or significant damage. Proper interconnection of professional gear with consumer gear (and vice versa) requires the use of a **line level shifter**. The use of this box eliminates all problems. Also, be very careful when interconnecting audio devices with different metering systems!! O dB is <u>not</u> equivalent to 0 VU. 0 VU = -18 dB, and 0 dB = ~ +22 VU!!!!!! Know your metering systems!

<u>**Speaker level**</u> signals are post power amplification. After a line level signal enters a power amplifier, speaker level signals are output to your non-powered loudspeakers. These signals are much higher in voltage than line level (measured in watts), require heavier gauge speaker cables for safe signal transfer, and are designed to physically move the output transducer back and forth to convert audio signals to SPL. *You should never plug a speaker level signal into a source expecting anything less than a speaker level signal.*

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Basic Audio Terms

audio signal - an AC (alternating current) signal within the audio frequency range (20 to 20,000 Hz. [hertz/cycles per second]) coming from a generating device (such as a tone generator, synthesizer, microphone, tape deck, phonograph cartridge) which is sent to the input of a sound system.

input - the location, jack, or receptacle provided for the introduction of an audio signal into an audio device.

output - the signal coming out of any audio device or the location, jack, or receptacle from which this audio signal is taken from the audio device.

amplitude - the magnitude of a signal's strength or volume.

amplify - to increase levels, as with a volume control.

attenuate - to decrease the level

channel - a single, complete through-path as from a microphone to a loudspeaker

dynamic range - the span of volume between the loudest and softest sounds in an original signal or recording or playback equipment

frequency - repetition rate of any periodic phenomena, as an electrical signal or sound vibration, expressed in hertz (Hz), formerly cycles per second

frequency range - the span of frequencies, from lowest to highest, that any audio device or system will pass without substantial loss

mic (microphone) level - a low amplitude signal coming from a microphone, phonograph cartridge, or instrument pick-up (ranging from .002 V [2 mVolts] to approximately .2 V). This signal must be amplified up to a standard higher amplitude level (line level) prior to being connected to other audio components.

line level - a standardized reference amplitude level for audio signals (ranging from .42 V (volts) to 1.23 V). This is the signal level put out by audio components after pre-amplification.

speaker level - a standardized amplitude level audio signal (measured in watts) coming from a power amplifier designed to operate loudspeakers.

input transducer - a device which coverts acoustical, magnetic, or optical energy into electrical energy.

instrument level – a semi-standardized reference amplitude level for audio signals coming from electronic keyboards. This level is higher than mic level, but often lower than line level.

noise - any sounds not related to the signal

pre-amplifier - an audio device designed to increase (amplify) weak audio signals (coming from microphones, phonograph cartridges, instrument transducers) up to the required line level amplitude strength needed by other audio components.

power amplifier - an audio device designed to receive a line level audio signal which then amplifies the audio signal up to speaker level.

output transducer - a device that converts electrical energy into acoustical energy.

FIGURE ONE

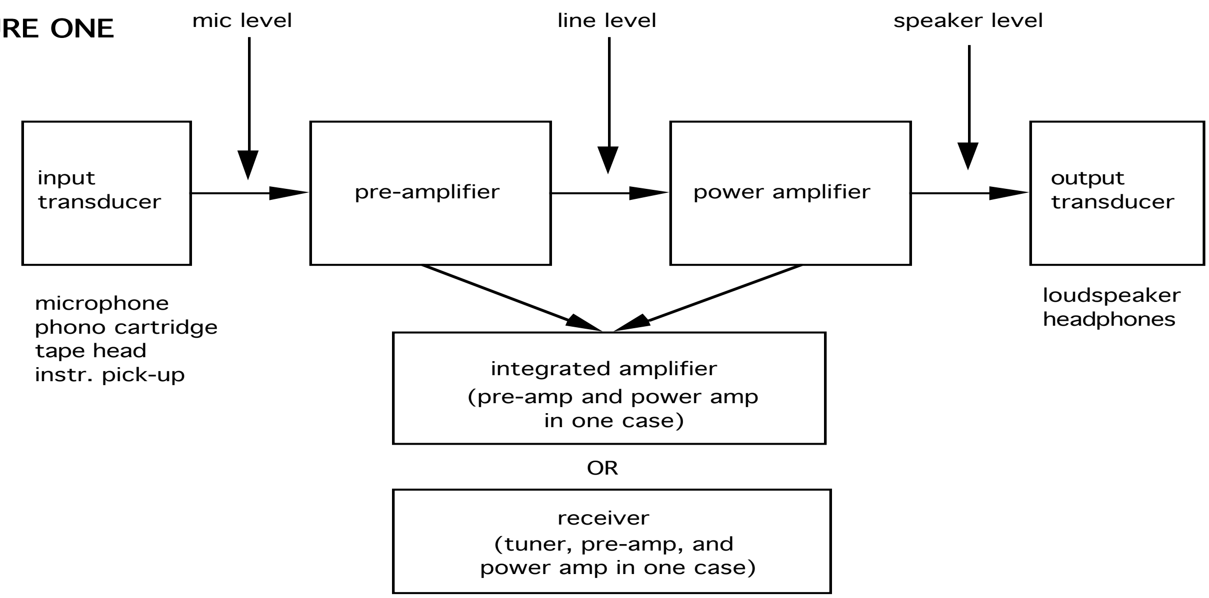


FIGURE TWO

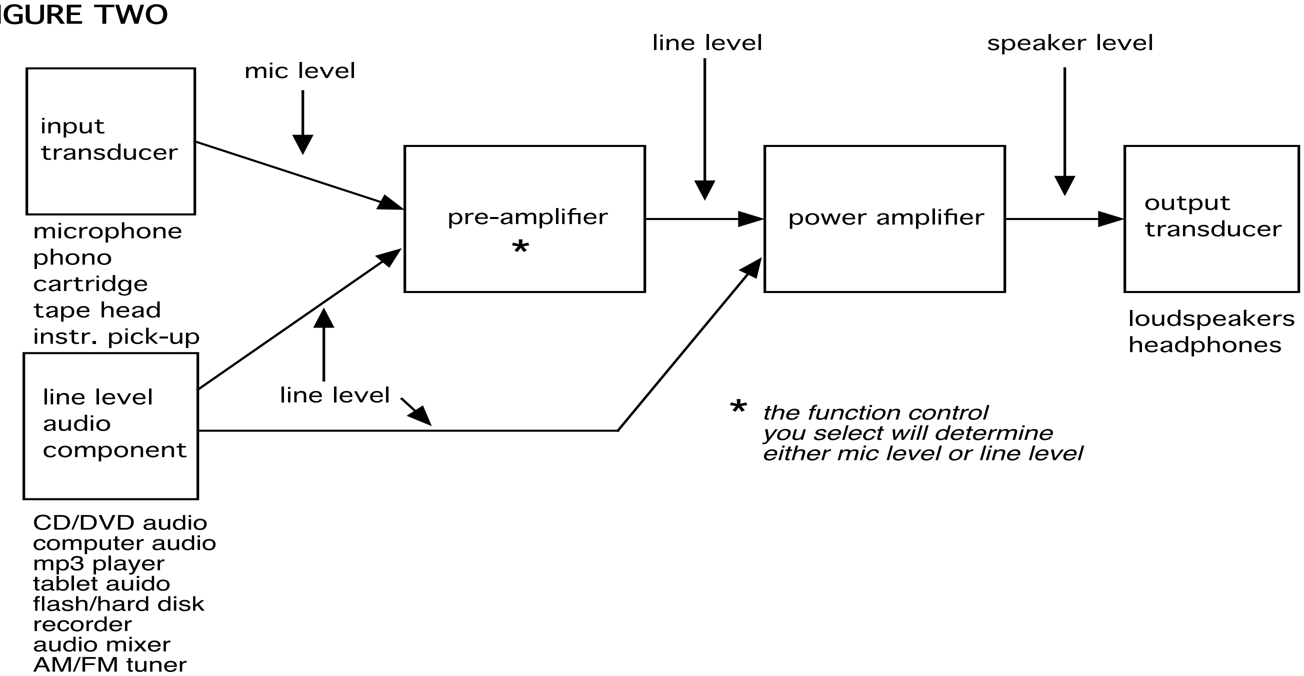
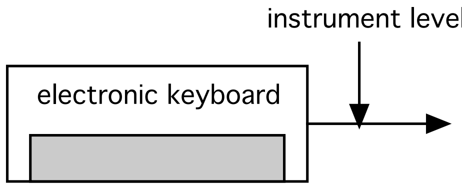


FIGURE THREE



instrument level is higher than mic level
BUT lower than standard line level

FIGURE FOUR

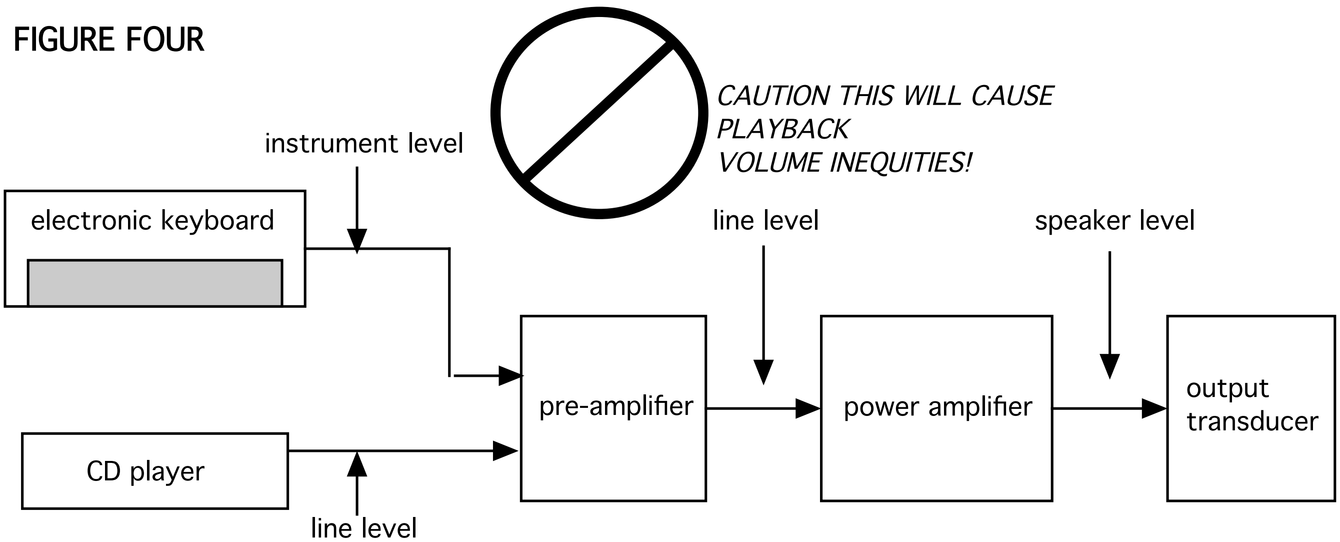
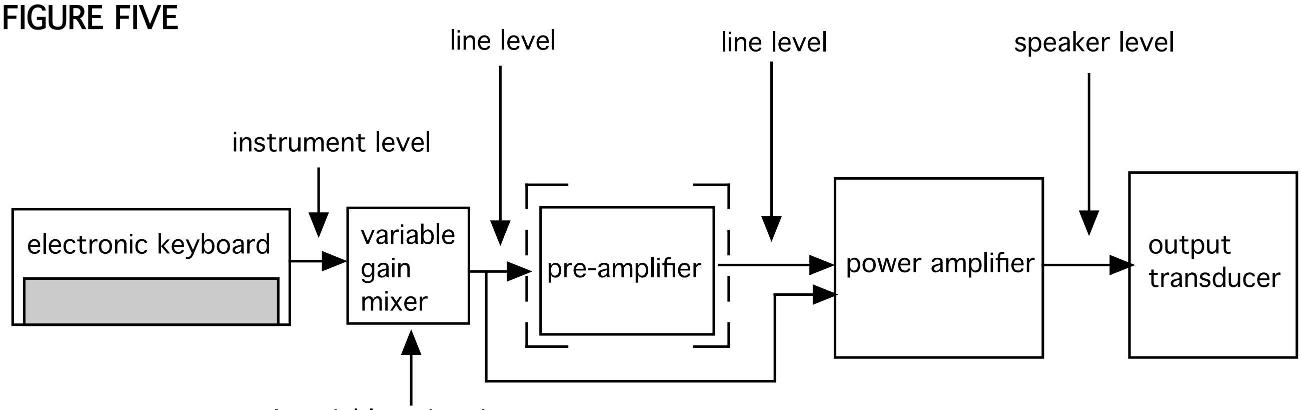


FIGURE FIVE

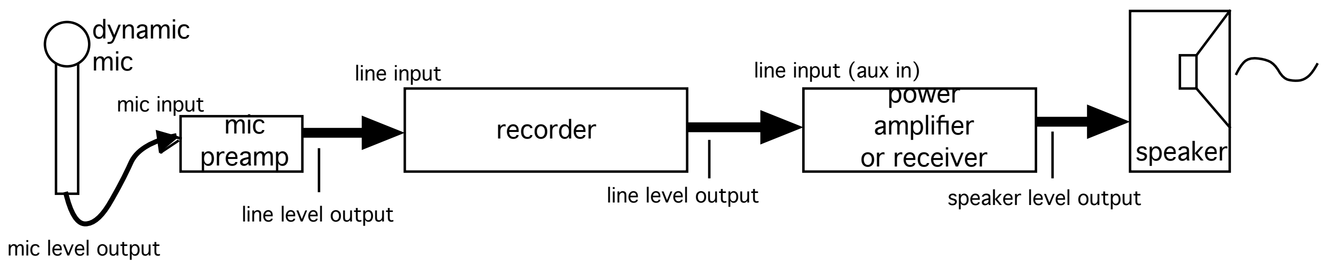


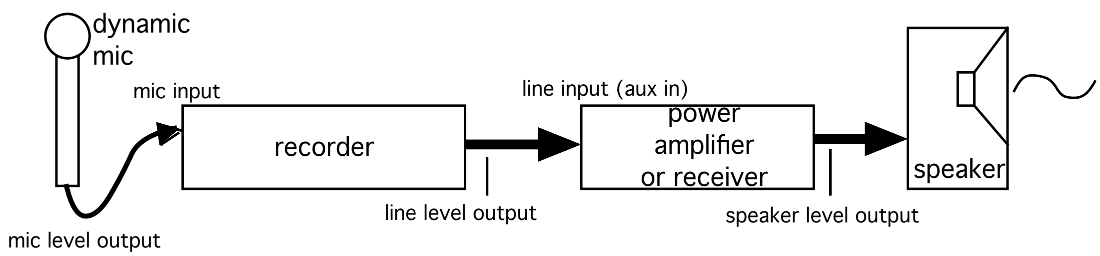
A variable gain mixer
allows one to increase the
amplitude of the instrument level
signal UP to the expected standard
line level signal - which a line level
audio component device requires.

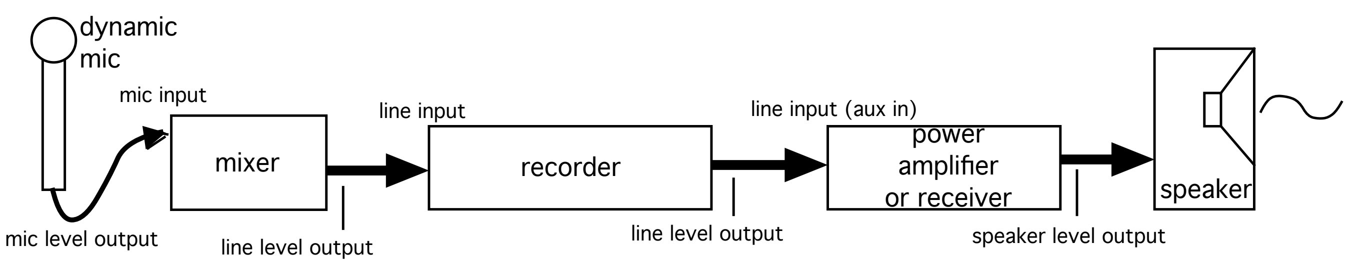
This eliminates the problem of different
playback volume levels when you switch back
and forth among different audio input
devices.

Generally speaking, a microphone (mic level ouput) requires preamplification up to line level by means of a mic preamp. Mic preamps exist as separate devices, or as part of a mixer (mic/line mixer), or as part of a tape recorder (labelled mic inputs). The output of a mic preamp is line level which is what many recorders have as inputs (some recorders have mic inputs and line inputs).

The following diagrams show the correct set-ups and ordering for the audio signal path.

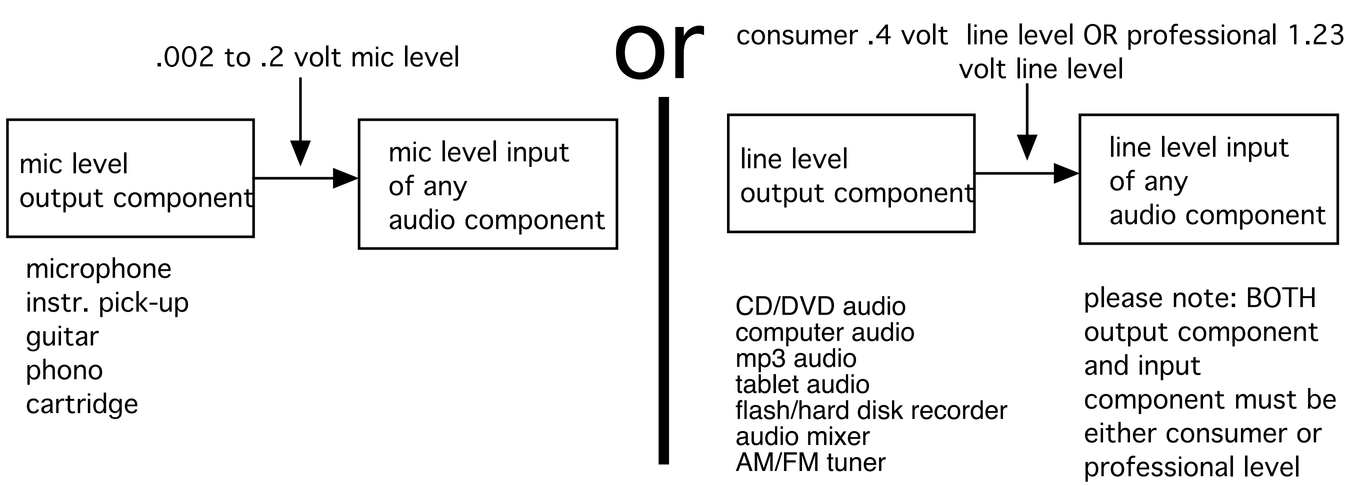


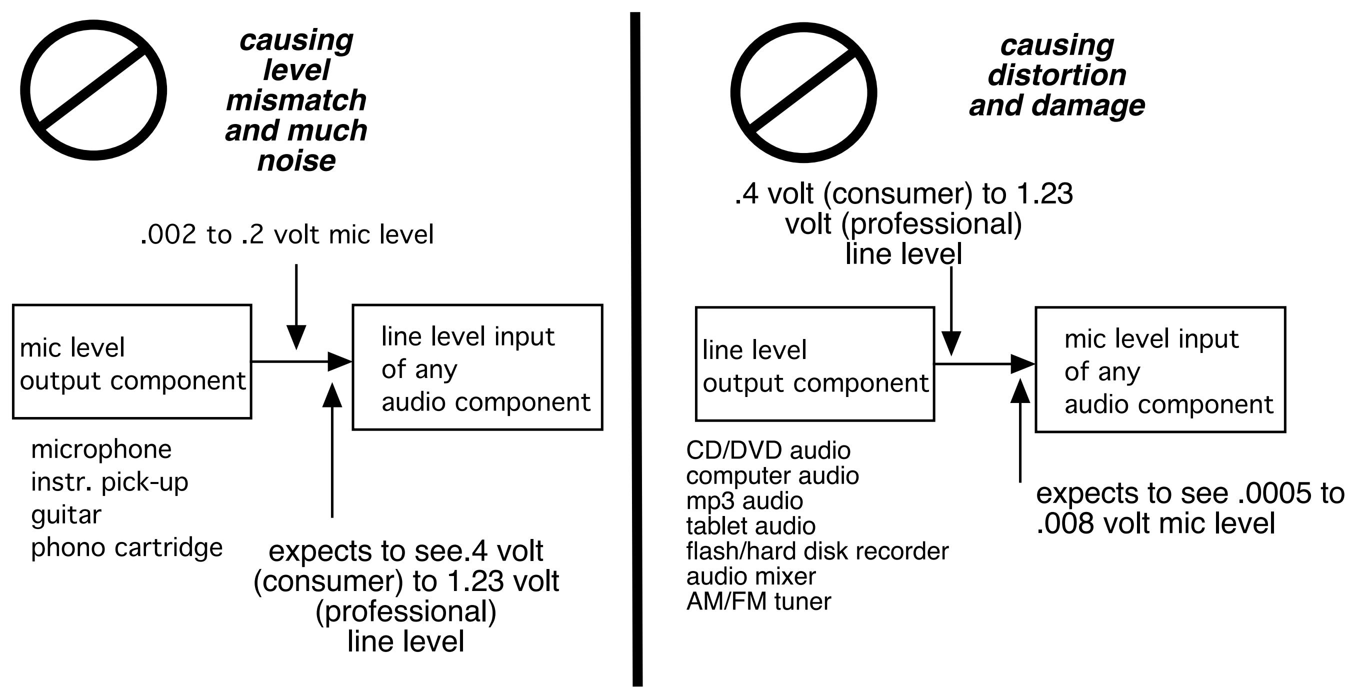




THERE ARE SIGNIFICANT COMPATIBILITY PROBLEMS BETWEEN MIC LEVEL AND LINE LEVEL.

PLEASE USE GREAT CAUTION WHEN CONNECTING AUDIO COMPONENTS.

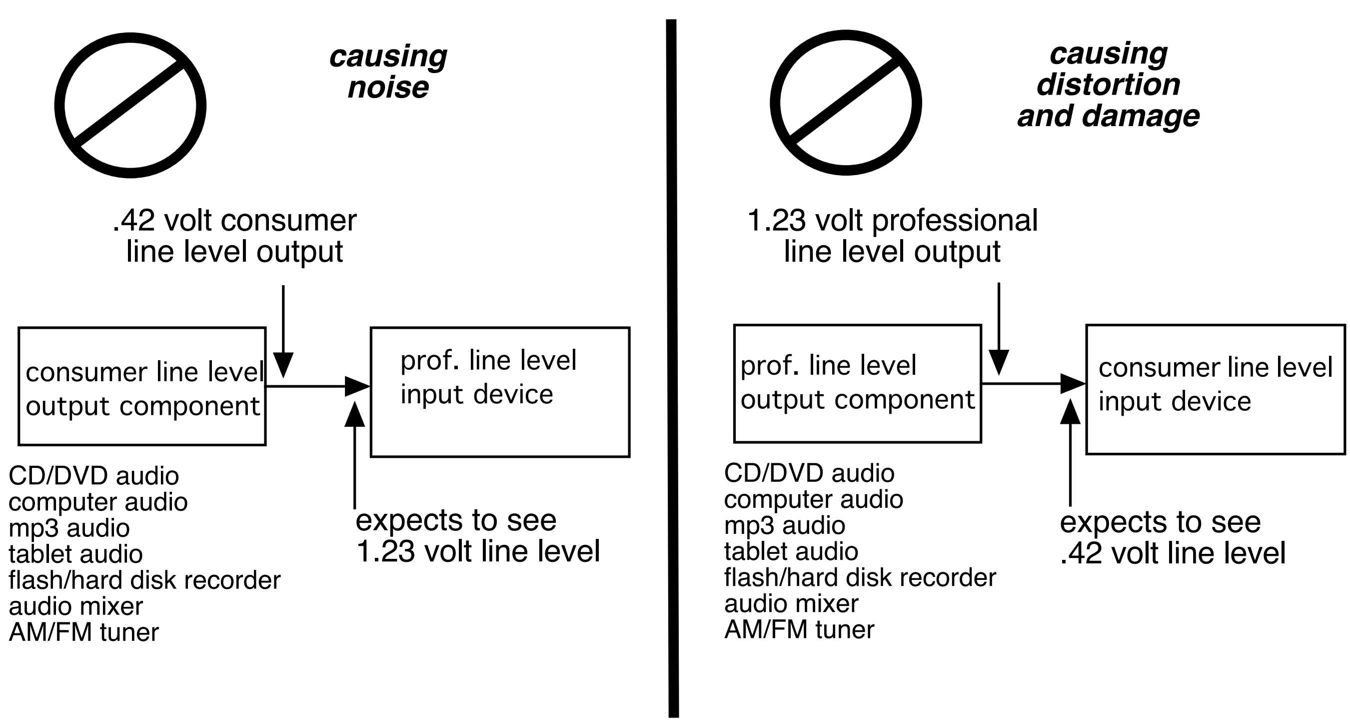




Consumer Audio vs. Professional Audio

Comparison of consumer audio to professional audio

consumer audio	*professional audio*
- reasonably inexpensive	- expensive
- reduced frequency & dynamic range	- expanded frequency & dynamic range
- reduced channel separation	- increased channel separation
- poor crosstalk	- reduced crosstalk
- approx. .42 volt line level	- 1.23 volt line level
- acceptible S/N	- good S/N
- uses RCA or 1/4 inch phone	- uses XLR or 1/4 inch TRS phone
- unbalanced circuitry	- balanced line circuitry
- susceptible to EMI, etc.	- cancels out EMI, etc.
- must use short cable lengths	- may use long cable lengths

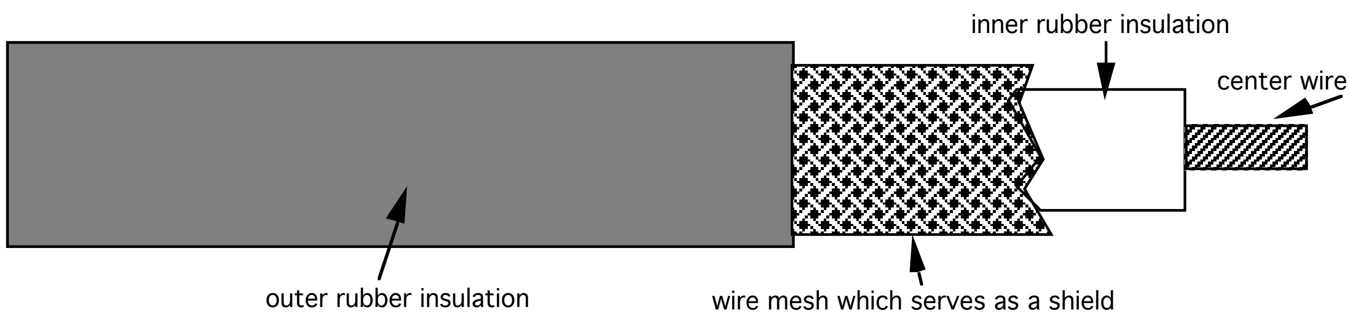


Consumer Audio Line Levels and associated cable types

- Line level signals are approximately .42 volts and utilize what is referred to as *unbalanced circuitry* and cables.

Unbalanced circuitry uses a two-wire cable for interconnecting audio components. This two-wire cable is referred to as either a single conductor shielded cable OR a 2 conductor cable. The center wire or conductor carries the audio signal and the shield or 2nd conductor serves as the ground wire. Unfortunately, unwanted radio frequencies can and usually do enter the cables and are transmitted into the audio signal.

2 conductor cable OR single conductor shielded cable



- Consumer audio components are fairly inexpensive yet are very susceptible to radio frequency interference.

- Short cable lengths for mic level and line level signals must be used to reduce audio signal loss (due to cable resistance) and to reduce unwanted radio frequency interference. The shorter the cables the better! (Do not use cables longer than 4 feet long.)

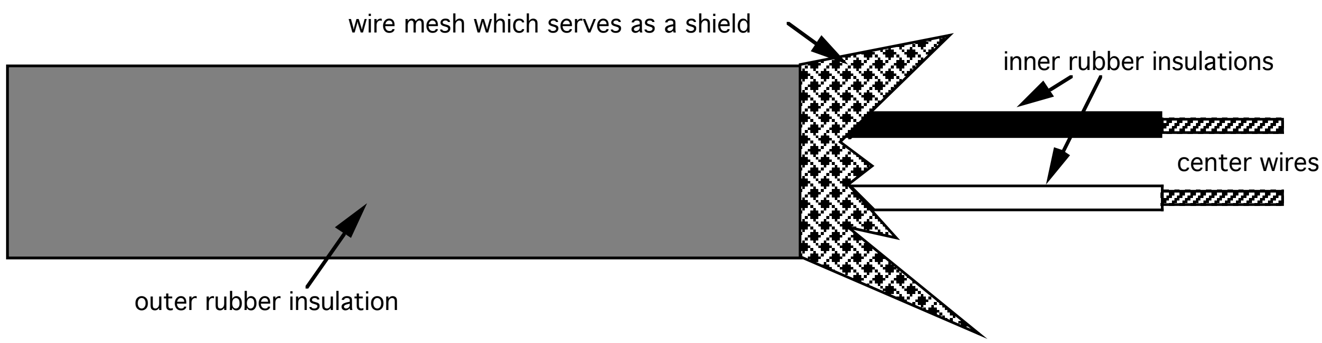
- Consumer audio components offer an acceptable audio frequency range and dynamic range with a detectable noise bed (hiss) in the background.

Professional audio components

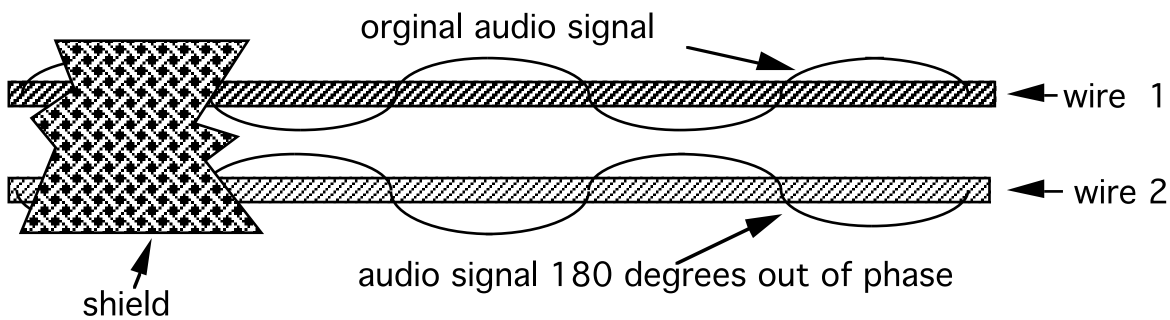
- Line level signals are 1.23 volts and utilize balanced circuitry and cables.

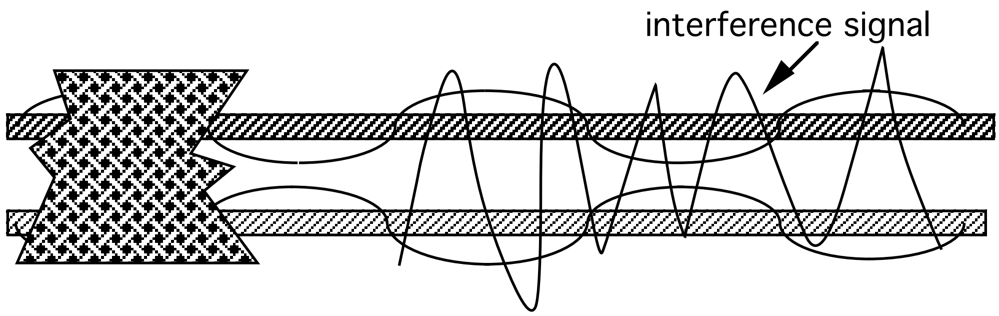
Balanced circuitry uses a three-wire cable for interconnecting audio components. This three-wire cable is referred to as either a two conductor shielded cable OR a 3 conductor cable. The two center wires or conductors carry the audio signal 180 degrees out of phase with each other, and the shield or 3rd conductor serves as an isolating shield (and sometimes serves as a ground). Two signals that are 180 degrees out of phase with each other will cancel each other. Special circuitry is designed at each input of professional audio components to look at the two incoming out of phase signals. Any outside radio frequency interference which have penetrated the cables will be seen by the circuitry as a third signal and labeled as interference or unwanted signal. The circuitry then filters this interference and allows only the original audio signal to pass into the rest of the audio component's circuitry.

3 conductor cable OR 2 conductor shielded cable



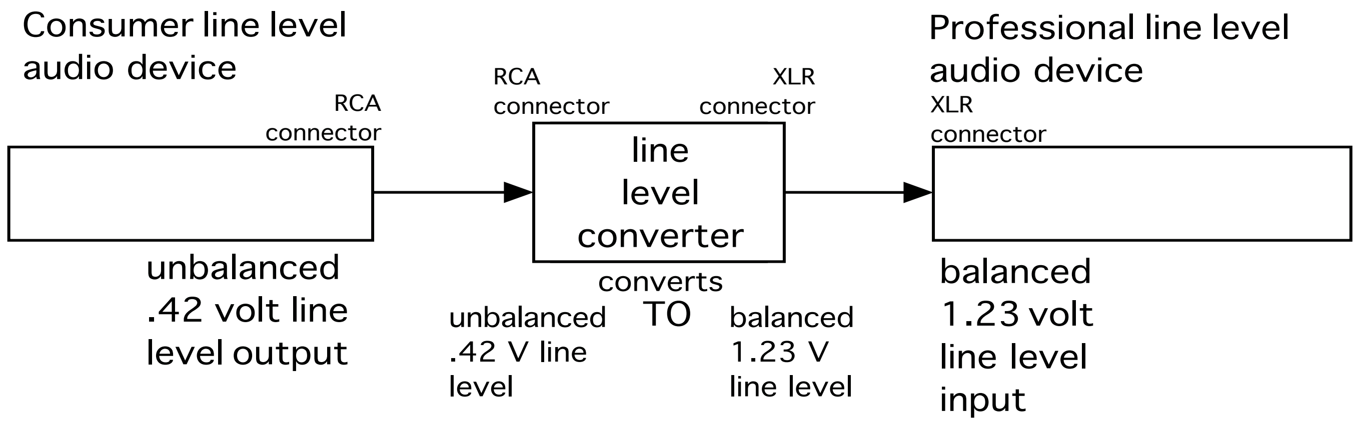
balanced signal (two copies of one audio signal 180 degrees out of phase)



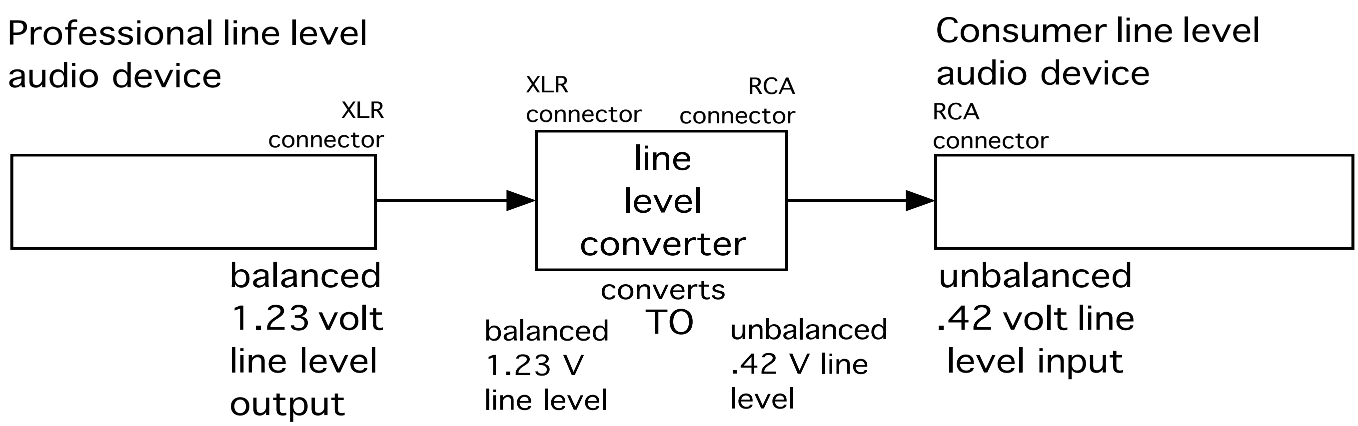


- Professional audio components are expensive and are able to filter out radio frequency interference.
- Long cable lengths for mic level and line level signals may be used without audio signal loss or unwanted radio frequency interference.
- Professional audio components offer a wider frequency reproduction capability, a wider dynamic range, and a much quieter audio signal (significantly reduced hiss in the background).

Properly connecting a Consumer line level device to a Professional line level audio device



Properly connecting a Professional line level device to a Consumer line level audio device



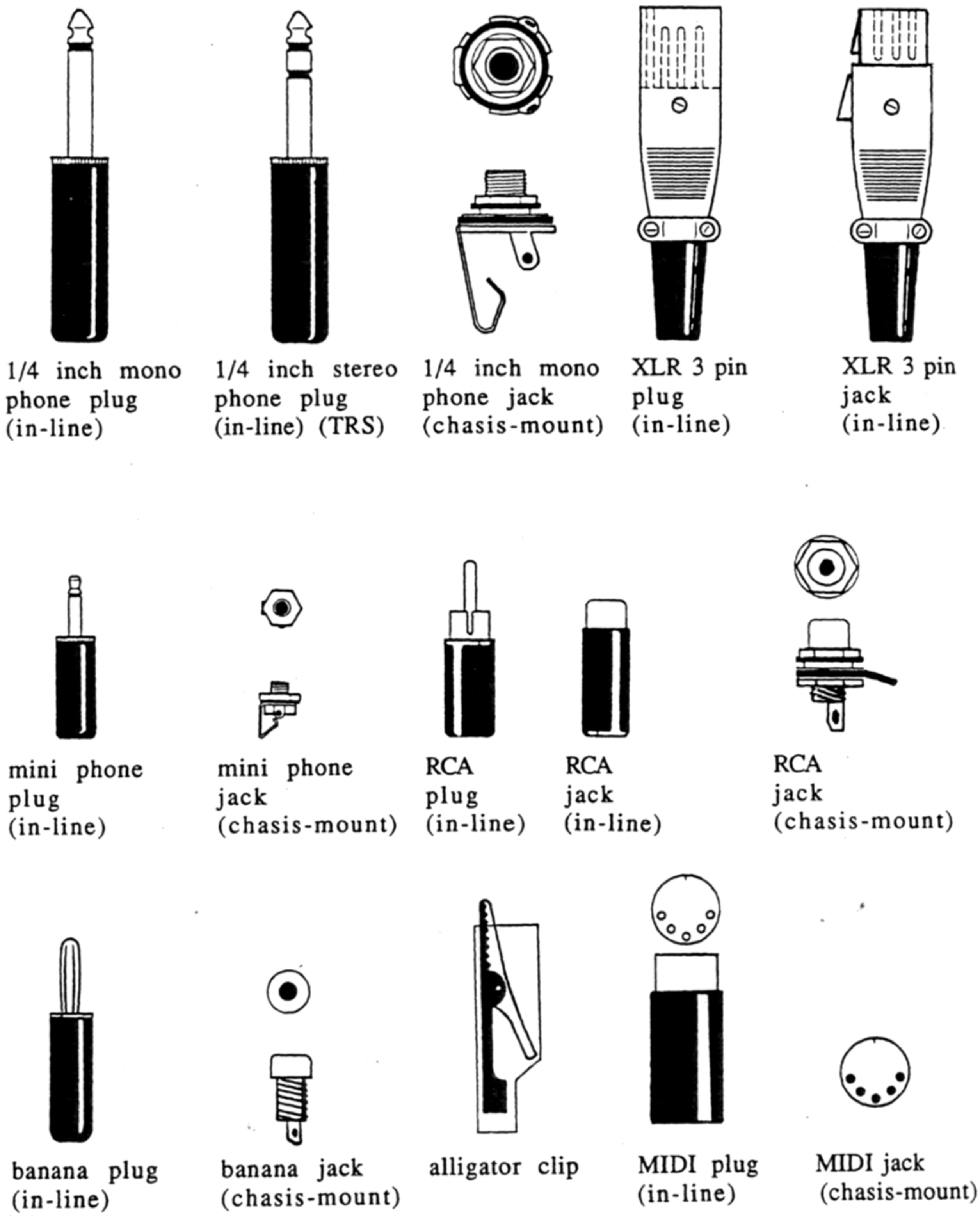
Line level converters (shifters) such as:

- Ebtech LLS-2 2-channel line level shifter ~$80
- Ebteckh LLS-2 XLR to ¼ inch phone ~$90

- Aphex model 124B ~$300 (2-channel XLR to RCA connectors with gain control)

- Behringer MX882 UltraLink Pro ~$100

Common Audio or Audio-related Connectors currently in use



Digital Audio

With digital audio, *many* periodic measurements (samples) of the sound's instantaneous amplitude and frequency are made by an analog-to-digital converter (ADC) that produces a stream of binary numbers (0's and 1's only represented by spurts of DC or direct current) that is the coded representation of the soundwave. These binary numbers can be more easily stored and manipulated than analog signals, and can be stored in the form of magnetic pulses on magnetic tape, magnetic disk, or optical disc. The number of samples per second taken by the ADC is called the ***sampling rate.***

In the late 1920s, Harry Nyquist developed the basic theory that states that the sampling rate must be at least twice the value of the highest audio frequency to be represented. This is called the *sampling theorem* (also known as the *Nyquist theorem*). As long as the sampling rate is at least twice the value of the highest audio frequency desired, the signal recorded will be reasonably accurate. The two common use-sampling rates that the audio industry has agreed upon are 44.1 **kHz** and **48 kHz** (although 96 kHz has recently been introduced as a potential third professional standard).

The analog audio output of digital audio components that is connected to an audio playback system goes through a conversion process device called a digital-to-analog converter (DAC). Digital recorders also have digital inputs and outputs. The two *digital audio transfer* (input/output) *formats* used by the audio industry are: the professional AES/EBU (Audio Engineering Society/European Broadcast Union) format which uses XLR or fibreoptic connectors, and the consumer S/PDIF (Sony/Philips Digital Information Format) approach which typically uses RCA or fibreoptic connectors. Digital duplication requires the identical format to be used by both the transmitting unit and the receiving unit.

Five basic types of standardized two-channel digital recorders exist at the present. <u>DASH recorders</u> (Digital Audio Stationary Head) are expensive and are primarily high-end professional recorders. RDAT recorders (Rotary [head] Digital Audio Tape - now referred to as <u>DAT recorders</u>) that are designed with a rotating cylinder tape head (helical scanning) much like the system used by video recorders. <u>MiniDisc</u> (MD) <u>recorders</u> are designed to record onto a small magnetic rewriteable disc cartridge. <u>Hard</u> <u>Disk</u> (HD) <u>recorders</u> now allow for recording directly to hard disk. <u>Flash</u> <u>recorders</u> are similar to HD recorders however they record directly to flash media cards and no hard disk is utilized.

With pressed CDs (playback-only CDs), the series of 0's and 1's have been pressed into the reflective layer of the disc from a mold. Within a CD player, the information is read by sensing the presence or absence of reflected light from a tightly focused Laser beam pointing at the pits and bumps in the reflective surface within the disc. The digital audio signal is sampled at 16 bits per channel at a rate of 44.1 kHz. The major developers were Philips Electronics of Holland and Sony of Japan.

ADC = analog to digital converter
DAC = digital to analog converter

Sampling rate = the number of samples per second taken by the ADC

Main sampling rates in current use:

44.1 kHz
48 kHz
96 kHz
(192 kHz)

the two current digital transfer input/output formats:

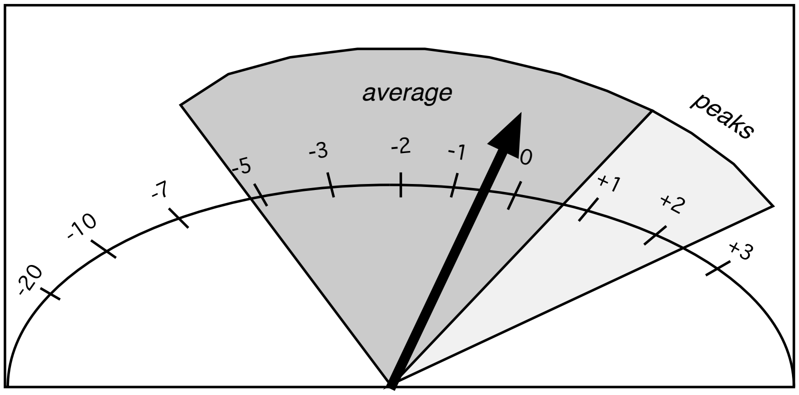
- AES/EBU	**uses XLR or fibreoptic connectors**
- S/PDIF	**uses RCA or fibreoptic connectors**

types of standardized two-channel digital recorders:

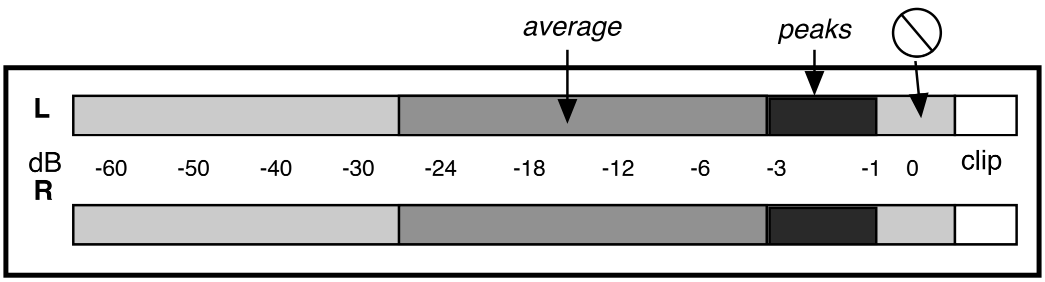
- **DASH**
- **RDAT (DAT)**
- **MD**
- **Hard disk recorders**
- **Flash recorders**

VU meters vs. dB meters

Desired Record Levels



A VU (volume unit) meter is calibrated to express the amplitude intensity of an audio signal. The "0" level supposedly indicates the maximum point the audio signal can be recorded or reproduced without a certain amount of distortion, <u>however</u> today's electronics have been designed so that unwanted distortion does NOT occur until +3 VU.



The decibel meter is found on digital recorders (DAT, flash recorder) and is designed to express the amplitude intensity of an audio signal. The -18 dB point is approximately equal to the "0" VU point of a VU meter. "0" dB is a dangerous point where the amplitude intensity of an audio signal could distort. Any amplitude that registers above "0" dB <u>will</u> distort!

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* Some mixers and recorders make use of a bar graph meter with the label "dB" indicated somewhere on the faceplate. The range is often -40 to +5 or +10. This type of a meter is <u>NOT</u> a true decibel meter - it is actually an expanded VU meter using the terminology dB instead of VU. This is purely a marketing scheme! In this case, use the recommended VU meter record ranges indicated in the first paragraph.

Compact Flash Recorders

A compact flash audio recorder is a digital audio recording device that saves audio onto a compact flash card – the same kind of card used with a digital photo camera. The amount of audio that can be recorded on a compact flash audio recorder thus depends on the size of the compact flash card used. The audio on the compact flash card can be downloaded onto a computer via a USB cable. You can also transfer the audio to a computer using an inexpensive "card reader" that attaches to the computer via a USB cable.

A CompactFlash (CF) card is a popular memory card developed by SanDisk in 1994 that uses flash memory technology to store data on a very small card. It has no moving mechanical parts and does not need a battery to retain data. CF cards allow users to add data to a wide variety of computing devices.

Flash memory is a type of *nonvolatile semiconductor memory* that is widely used for storage and data transfer in consumer devices and enterprise systems. Because flash is nonvolatile memory, stored data is retained when a device's power source is turned off or lost. CF cards feature solid-state construction, which makes them much more rugged than most traditional storage devices.

Examples of reliable line level compact flash recorders

Tascam SS-R200 - 2-channel Removable Media Recorder with WAV or MP3 Formats, XLR/RCA Analog and Coaxial Digital I/O, and RC-20 Direct Play Remote Capability - CF, SD/SDHC, USB – approximately $550.

Installation-friendly Flash Recorder
The TASCAM SS-R200 Solid State Recorder serves up an impressive set of recording and playback features in a compact single-rack unit that's great for installations. Whether you're recording important meetings, capturing audio in your house of worship, or want a great solution for live performances, the SS-R200 gives you the flexibility you need. This unit records in WAV or MP3 formats. You get XLR and RCA analog I/O, as well as S/PDIF or AES/EBU coaxial digital I/O. Get high-quality results when you record with the SS-R200.

TASCAM SS-R200 Solid State Recorder at a Glance:
Convenient
The TASCAM SS-R200 conveniently lets you record WAV or MP3 format to inexpensive, readily-available removable media, including CF, SD/SDHC, or USB.

Flexible feature set
The TASCAM SS-R200 sports a host of cool features that you'll appreciate. You get convenient WAV or MP3 recording, up to 20 tracks of flash start with an optional RC-20 remote control device, PS/2 or USB keyboard connection, and even playback speed control without pitch change.

It's a TASCAM
Whatever you record, the TASCAM SS-R200 Solid State Recorder gives you pro flexibility. From broadcast studios to tour rigs, the SS-R200 delivers rock-solid performance, day-in and day-out. What else wound you expect from TASCAM? And if you're a contractor, you can't go wrong specifying this bullet-proof unit for your client's installation.

TASCAM SS-R200 Solid State Recorder Features:

- Perfect for installations or your road rig!
- Solid state recording to any available Media (CF, SD/SDHC, USB)
- One-rack-space compact design
- Up to 20 tracks of flash start with optional RC-20 remote control device
- PS/2 or USB keyboard connection
- RS-23C and Parallel Control I/O
- Multiple playback modes
- Continuous, single, programmed and random playback
- XLR balanced I/O, RCA unbalanced I/O
- Coaxial S/PDIF, or AES/EBU digital I/O

Tascam SS-R100 - 2-channel Removable Media Recorder with WAV or MP3 Formats, RCA Analog and Coaxial Digital I/O, and RC-20 Direct Play Remote Capability - CF, SD/SDHC, USB – approximately $450.

Installation-friendly Flash Recorder
The TASCAM SS-R100 Solid State Recorder serves up an impressive set of recording and playback features in a compact single-rack unit that's great for installations. Whether you're recording important meetings, capturing audio in your house of worship, or want a great solution for live performances, the SS-R100 gives you the flexibility you need. This unit records in WAV or MP3 formats. You get unbalanced RCA I/O, as well as S/PDIF or AES/EBU coaxial digital I/O. Get high-quality results when you record with the SS-R100.

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- Playback speed control without pitch change
- RCA unbalanced I/O, coaxial S/PDIF, or AES/EBU digital I/O

Audio Recording Techniques
Check Yourself – Exam #1

Part I. Please define the following terms as they apply to this field of study:

1. audio signal

2. input

3. output

4. input transducer

5. output transducer

6. preamplifier

7. power amplifier

8. amplify

9. attenuate

10. amplitude

11. frequency

Part II. Analog audio

1. Analog audio signals are what current type?

2. Mic level is generally about what voltage range?

3. Consumer line level nominal amplitude is what specific voltage?

4. Professional line level nominal amplitude is what specific voltage?

Part III. Sound reproduction systems

1. State the four basic components of a sound reproduction system and the function of each component.

1.

2.

3.

4.

2. Draw a diagram showing the correct placement of each component within a sound reproduction system. Indicate the path of the signal among these components with arrows.

3. List two input transducers.

4. Define integrated amplifier and receiver. Indicate the difference between the two.

5. What is the difference between mic level and line level? Is it advisable to feed a line level audio signal into a mic level input? Why? Is it advisable to feed a mic level audio signal into a line level input? Why?

6. What is needed to amplify a mic level signal up to a line level signal?

7. Does line level require preamplification?

8. For the following components, indicate what category the amplitude output level is:

computer audio -
microphone -
CD/DVD audio -
portable electronic keyboard -
electric guitar -
hard disk recorder -
flash recorder -
mp3 player -

9. Discuss the voltage level differences (benefits and disadvantages), as well as the audible differences between consumer and professional audio systems.

10. Discuss the connector differences, as well as the audible differences due to the use of such connector/circuit systems between consumer and professional audio systems.

11. What is the result if you connect a line level output of a professional level recorder/player into a line level input of a consumer recorder/player or amplifier? Why?

12. What is the result if you connect a line level output of a consumer CD player to a line level input of a professional power amp? Why?

13. If you wish to properly connect the line level output of professional mixer to the line level input of a consumer receiver or integrated amplifier, what is needed and why?

14. If your consumer recorder has line level inputs only, should you connect your consumer microphones to these line level inputs? Explain your answer. What piece of equipment do you need, so that you could correctly connect your consumer microphones to the line level inputs of your recorder?

15. What standard audio connectors are generally used for consumer line level?

16. What standard audio connectors are generally used for professional line level?

Dynamic (moving coil) microphones

A dynamic microphone employs a small diaphragm and a voice coil. Sound waves striking the surface of the diaphragm cause the coil to be moved within the magnetic field of the permanent magnet, thus generating a voltage proportional to the sound pressure at the surface of the diaphragm.

[please see dynamic microphone capsule diagram]

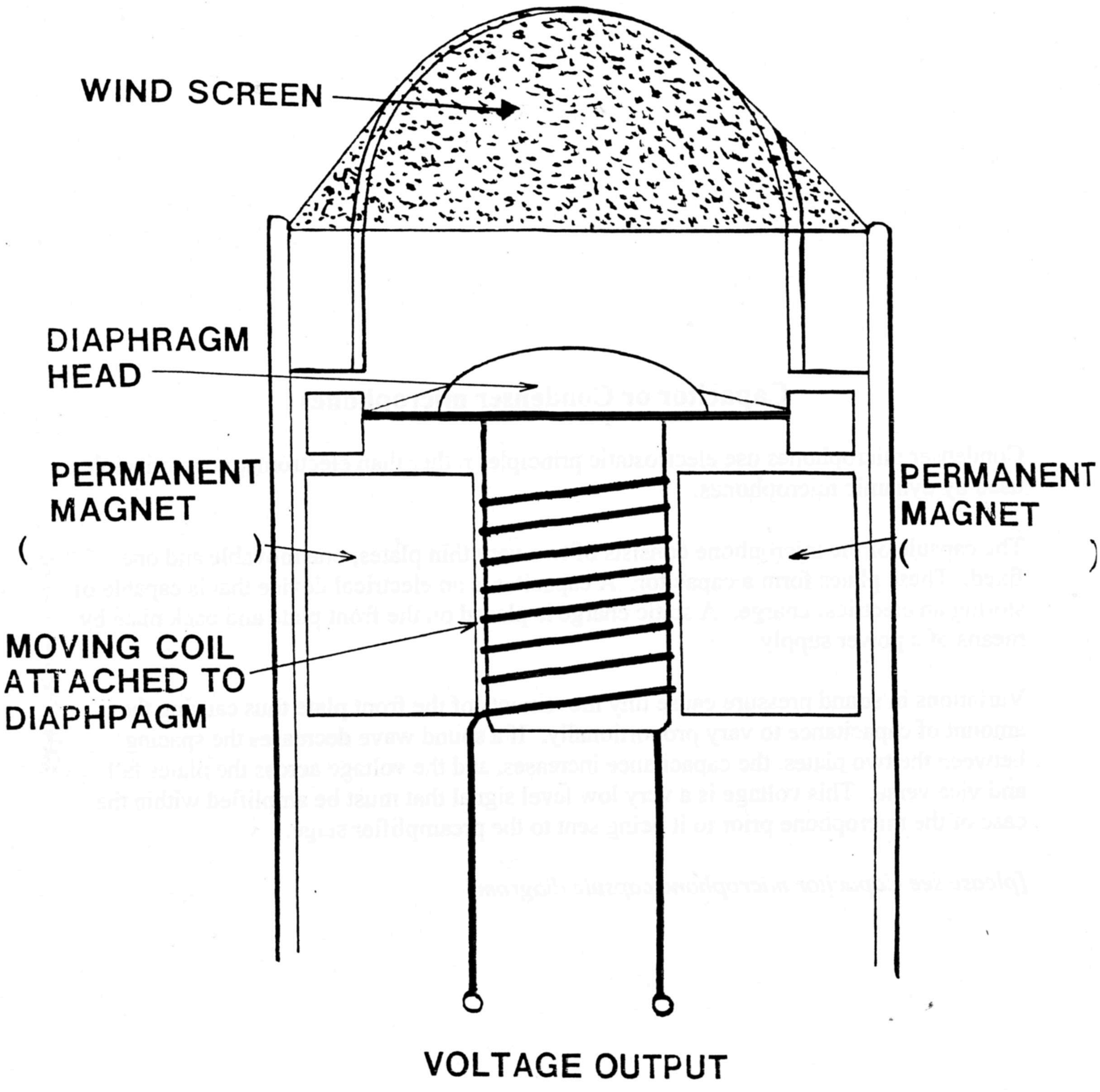
These voltages are large or small depending on the strength of motion of the moving coil, and are therefore an electrical replica of the mechanical sound wave energies. Most dynamic microphones are sensitive from 30 to 15k Hz or better.

Dynamic microphones produce a weak signal that must be preamplified, but most possess good fidelity.

Both the dynamic microphone and the dynamic loudspeaker have a diaphragm (or cone) with a voice coil attached near the apex. Both utilize an electromagnetic system with a coil. The difference is in how they are used. With a loudspeaker, current from the amplifier flows through the coil. The interaction of the magnetic field created as the current flows through the voice coil, with the magnetic field of the speaker's magnet, forces both the coil and the cone to move back and forth, producing sound output.

With the dynamic microphone, the diaphragm is moved by changing sound pressure. This moves the coil that causes current to flow as the magnetic lines of flux are cut or interrupted. So instead of putting electrical energy into the coil (as in a speaker), you get energy out.

Many intercom systems use small dynamic loudspeakers (with lightweight cones) as both speaker and microphone, by switching the same transducer from one end of the amplifier to the other. A good speaker does not make a great microphone, but it is good enough for intercom applications.



Dynamic or Moving-Coil Microphone

Capacitor or Condenser microphones

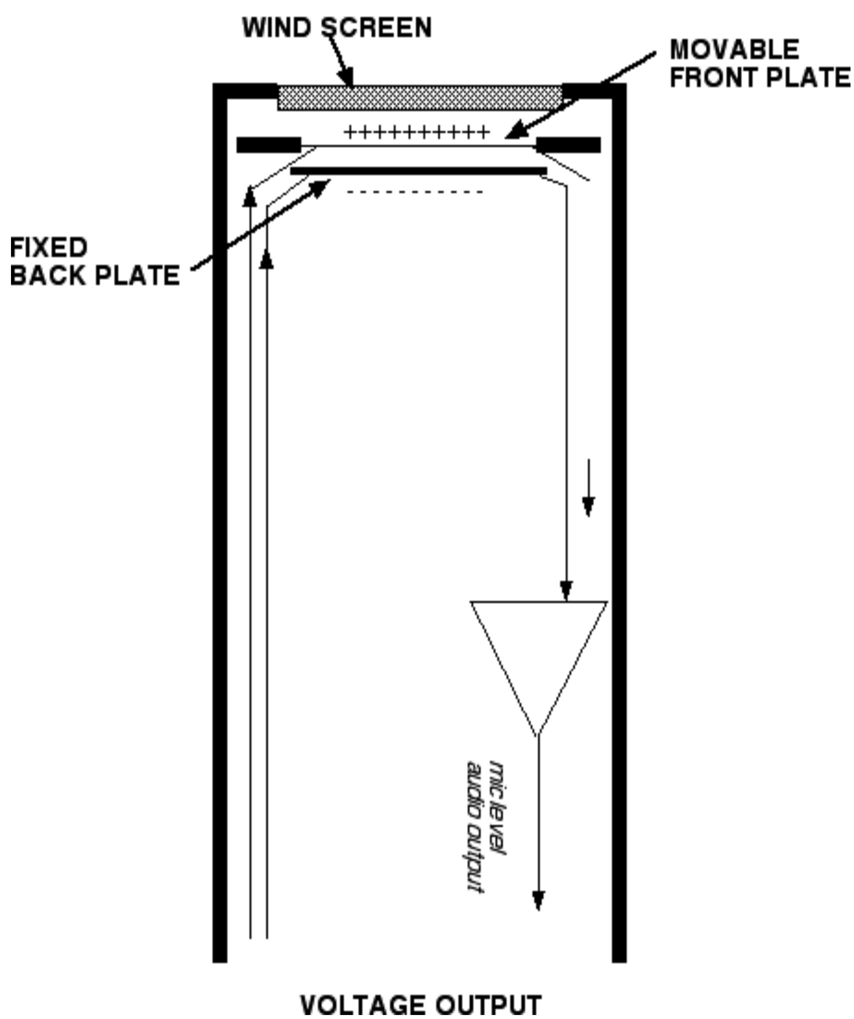
Condenser microphones use electrostatic principles rather than electromagnetic principles used by dynamic microphones.

The capsule of the microphone consists of two very thin plates, one movable and one fixed. These plates form a capacitor. A capacitor is an electrical device that is capable of storing an electrical charge. A static charge is placed on the front plate and back plate by means of a power supply.

Variations in sound pressure cause tiny movements of the front plate thus causing the amount of capacitance to vary proportionally. If a sound wave decreases the spacing between the two plates, the capacitance increases, and the voltage across the plates fall and vice versa. This voltage is a very low level signal that must be amplified within the case of the microphone prior to it being sent to the preamplifier stage.

[please see Capacitor microphone capsule diagram]

Capacitor or Condenser Microphone Capsule



Phantom Power

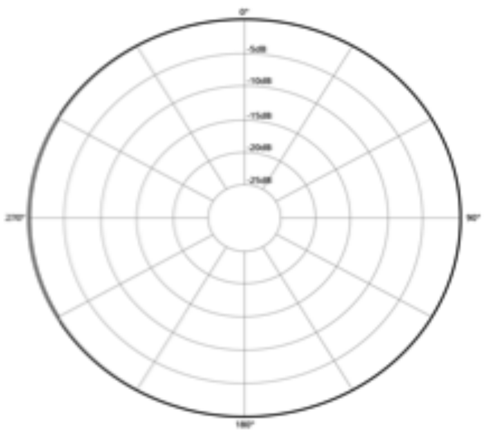
Phantom power supplies a DC voltage sent through a microphone cable to power the electrical circuit inside of the microphone that supplies constant polarized charge to the front and back plates of a condenser microphone or associated device. It is used most commonly with condenser microphones, but is sometimes used to power DI boxes and effects pedals, too. Phantom power is usually 48 volts DC (sometimes less; you'll see it as low as 12v DC), and sent to a condenser microphone through the cable from your mixer or preamplifier. Sometimes, you'll find preamps that don't offer phantom power, so you'll have to provide an external phantom power source.

True condenser microphones require a 48 Volt external power supply (phantom power) to resupply the polar charges to both plates of the capacitor, as well as, supplying power to the mini preamp within the casing of the condenser microphone. Phantom power supplies exist as either self-contained external supplies or as phantom power circuitry found in most mixing consoles. Generally speaking, engineers prefer to work with true condenser microphones for truer sound reproduction in a recording environment.

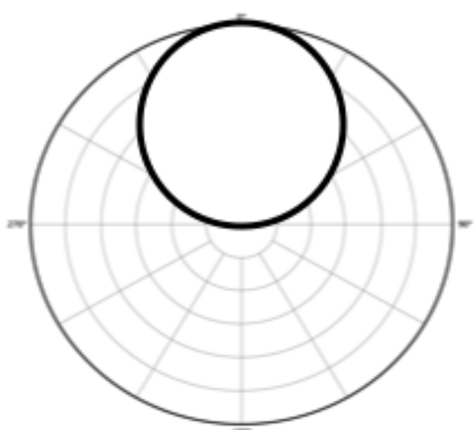
Electret condenser microphones have permanently polarized plates within the capsule of the electret condenser microphone resulting in no need for phantom power. Electret condenser microphones will generally require an AA battery to supply power to the mini preamp within the casing of the electret condenser microphone.

POLAR PATTERNS

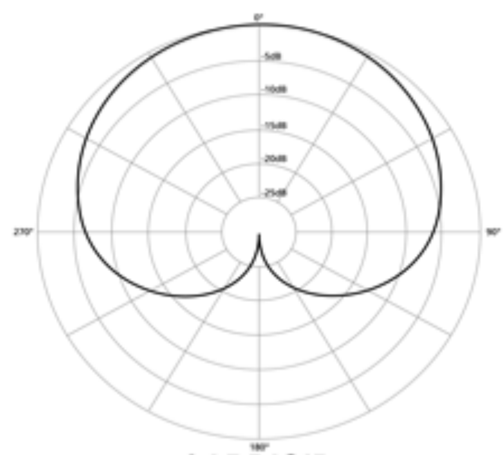
polar pattern – a diagram of microphone capsule's designed field of sensitivity. Such polar patterns include: Omni directional, Uni directional (true and cardioid), and Bi-directional fields.



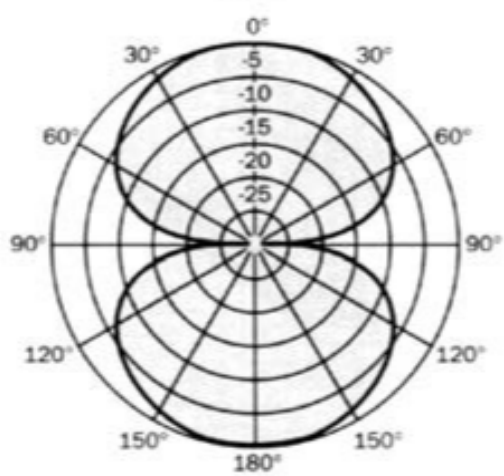
OMNI DIRECTIONAL



TRUE UNI DIRECTIONAL



CARDIOID



BI-DIRECTIONAL or FIGURE 8

Microphone Placement

- close mic position (1 to 6 inches from sound source)

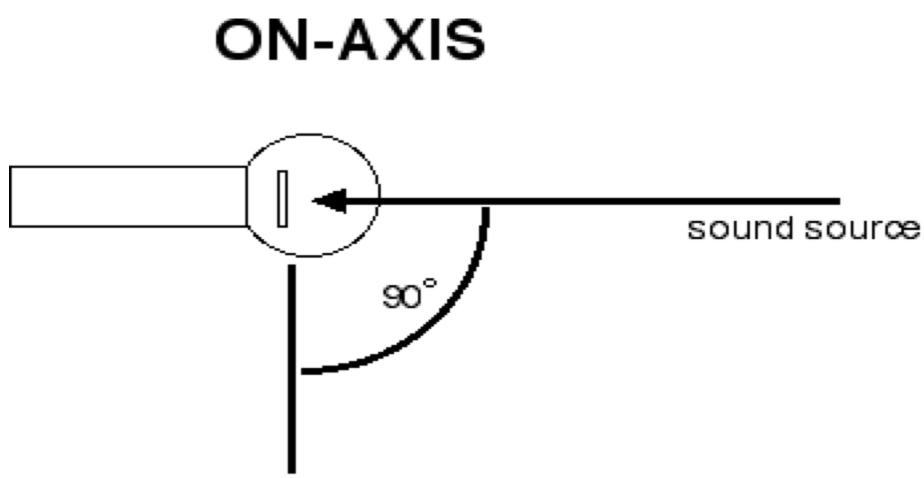
- spot mic position (6 to 48 inches from the sound source)

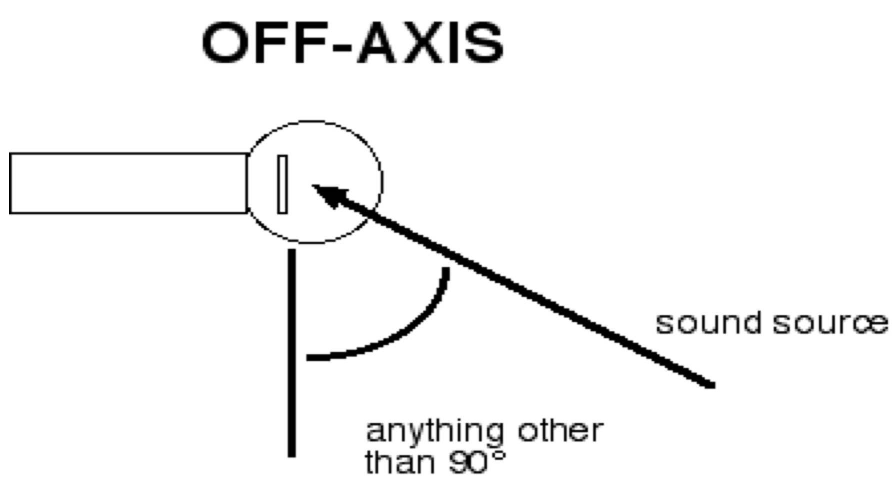
- ambient mic position (6 to 12 feet from the closest sound source)

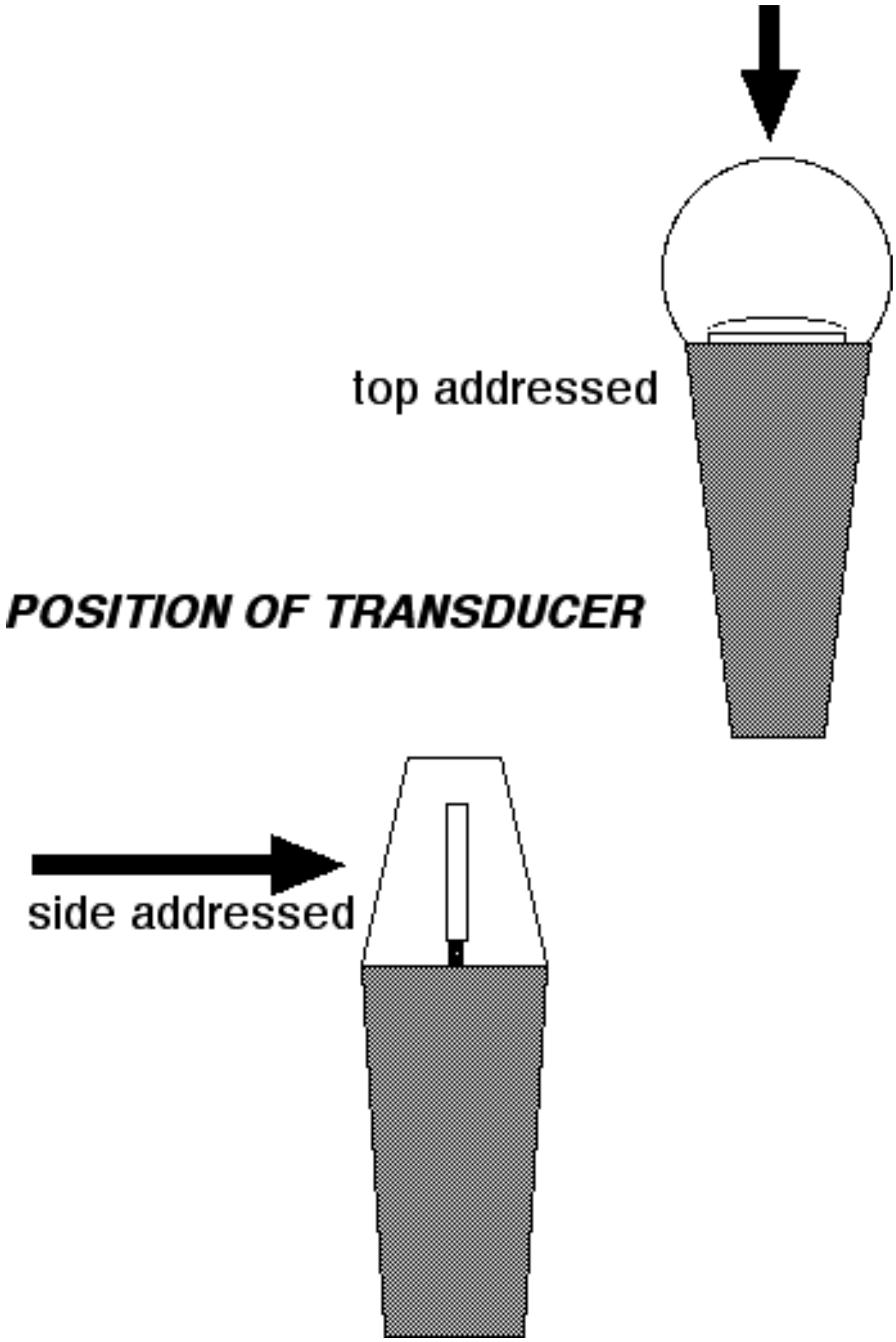
proximity effect - an increased response to low frequencies when the mic is placed within two feet of the source.

on axis - the mic is positioned so that the diaphragm is perpendicular (at 90 degrees) to the sound source

off axis - the mic is positioned so that the diaphragm is 30 to 60 degrees in relationship to the sound source.






top addressed
POSITION OF TRANSDUCER
side addressed

Terms / Distinctions / Differentiations

Monophonic (mono) – (Greek: monos = one, single, alone / phone = sound), a single channel of audio information

2-channel audio – referring to an audio reproduction system or recording having two channels (may or may not be stereo)

2-channel mono – a single channel of audio information recorded or delivered over two channels

Mono (point) source – monophonic audio information – can be panned within two or more audio channels

Stereophonic (stereo) – (Greek: stereos = solid / phone = sound),

- Relating to or constituting a three-dimensional effect of auditory perspective; in audio, a recording involving two or more microphones designed to capture a sound image from two (or more) locations relative to the sound source, thus recording not only frequencies and amplitudes but also time differences (phase differences) of the direct sound, as well as, time difference of reflected sound, based upon the location of the individual microphones. The information from each microphone is separately channeled (panned/positioned) within the 2-channel audio system in an effort to recreate the three-dimensional image of that original sound – in which the timing information, emanating from the two loudspeakers of the two-channel sound reproduction system, is different.

Stereo is different from "binaural." Binaural properly applies to a two-channel audio system designed for headphone reproduction in which the two microphones used to record the sound source are spaced at a distance of about seven inches (normal ear separation). Stereo employs microphone positioning at wider distances than seven inches to account for the positions of loudspeaker delivery.

Pseudo stereo – a monophonic sound source recorded on or delivered over a two-channel audio system with the audio information of one of the channels delayed.

The audible difference between true stereo and pseudo stereo is quite different.

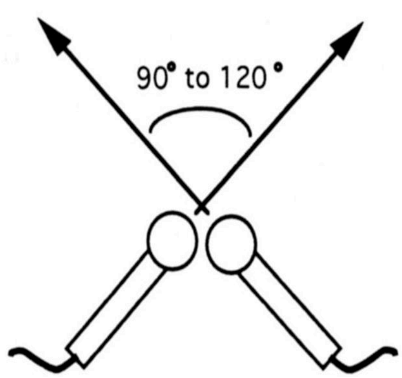
Spaced-Pair

The Spaced-Pair technique is obviously the simplest and most direct approach. The spacing between the microphones is primarily determined by the size of the area to be recorded and the rule that microphones should not be placed more than 12 feet from each other. The logical approach is to place the microphones at the one third and two thirds locations across the front of the area to be recorded.

The minimum distance the mics should be positioned from each other is 3 feet.

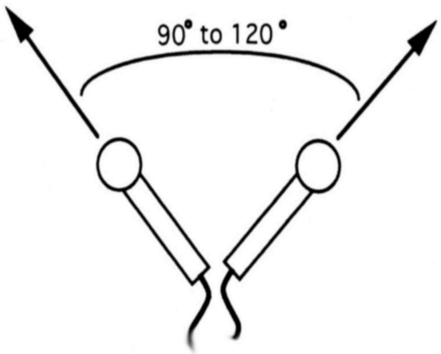
Spaced Pair micing is one of the oldest and simplest ways of stereo miking, but it requires careful listening in order to balance the stereo field presentation of the sound source. Classical recording generally requires a cardioid spaced pair to be placed 8 to 14 feet back from the source, and between 8 and 20 feet up from the floor.

X-Y Coincident-Pair



The X-Y Coincident-Pair technique is a stereophonic miking technique that works well for medium sized coverage down to a single instrument. Sounds will reach both microphones at approximately the same time resulting in a fairly even soundfield. It often reduces the effect of natural reverberation, and can be used to help control unwanted sound pickup in reverberant or poor acoustic environments and to reduce audience noise.

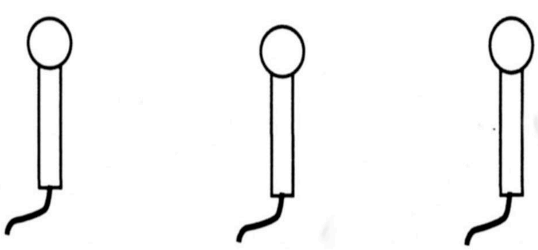
Near Coincident-Pair



The Near Coincident-Pair technique is a stereophonic miking technique that increases the width of the stereophonic image and offers better reproduction of ambient reverberation due to the spacing between the microphone capsules.

A specific version of the Near Coincident-Pair is called ORTF. This technique was developed by the French Broadcasting Organization in 1960. It stands for Office de Radiodiffusion-Télévision Française. A matched pair of cardioid microphones is placed parallel to each other so that their capsules are 17cm (approximately 7 inches). The microphones are aimed away from each other at a 110 degree angle. This is the official ORTF standard.

Three Microphone System



The Three Microphone technique is often used for better coverage of large ensembles but requires the use of a mixer. In this system, the left microphone is fed to the left channel, the right microphone is fed to the right channel, and the middle microphone is sent to both channels equally. This technique provides a more solid central image.

The minimum distance between microphones should be 3 feet.

Some Notes About Classical Recording offered by Frank Horger

Recording:

All recording is an exaggeration of reality. Classical recording techniques were developed to help recreate a believable version of a sound.

Classical Recording Theory:

The classical approach to recording has the goal of capturing and recreating a sound propagating in a space. The recording techniques chosen influence how this sound will be portrayed.

The placement of the sound source, as well as the dimensions of the space, are implied by the **depth**, **width**, and **frequency response** of the final recording. The primary sound of the source is referred to as the direct sound. The secondary reflection of that sound (off of walls, floors, etc.) is referred to as the reverberant sound, or reverb. The ratio of direct sound to reverberant sound (**direct to reverberant ratio**) will play a very important role in determining the believability of a recording.

Size:

The human ear can infer the size of a room by the length of time a sound propagates in it.

A person will first hear the direct sound created, and then hear the reflections of that sound as they bounce back towards the listener. The longer a sound propagates, the longer those reflections will continue to come back to the listener. This determines the length of the **reverb**.

The size and contents of a room will shape the way a sound propagates in that space. They will also determine the quantity and length of reverb that is created.

A soundwave traveling through the air will tend to be absorbed by walls and surfaces. In general, the closer these walls are, (and the smaller the room is) the shorter the sound will propagate. A sound that propagates for a shorter period of time will create shorter reverb.

Alternatively, the farther a sound is from the outer walls and surfaces of a room the less chance it has of being absorbed. In general, the farther away these walls are (and the larger the room is) the longer a sound will propagate. A sound that propagates for a longer period of time will create longer reverb.

Therefore a person can infer the size of a room from the length of the reverb.

Example: A small closet will have a short reverb length of 0.5 second. A school gymnasium may have a reverb length of 3.5 seconds or longer.

Depth:

The human ear can infer its own distance from the sound source by the quantity of reverb it encounters. This **direct to reverberant ratio** is different from the length of the reverb.

If a person stands directly in front of a piano in a recital hall, they will hear a lot of the direct sound from the piano. The reverb created by the piano's sound in the hall will continue to develop past them in the room. Only a portion of this reverb will be reflected back to the listener, so the direct to reverberant ratio will lean heavily on the direct sound.

Conversely, if a person stands far away from the piano in this same recital hall they will hear more of the reverb that is created in the room. In this instance the direct to reverberant ratio will lean heavily on the reverb.

In the case of classical recording you can manipulate the perceived depth of your recording and of individual sound sources by being mindful of the direct to reverberant ratio.

Width:

Stereo recording techniques captures sound with the goal or replaying it on a stereo sound reproduction system. These techniques may need to be altered when sound will be replayed on other systems such as a mono system, quad system, 5.1 system, etc.

Stereo sound reproduction attempts to recreate the natural presentation of multi-directional sound that the human ears use to decode location. This is achieved through the use of two independent audio channels feeding two or more speakers. The **stereo field** describes the implied Left to Right placement of a sound source in this multi-directional image.

Classical recording uses an awareness of the limitations and capabilities of stereo sound reproduction systems in order to manipulate the perceived width of a recording. This includes the perceived distance between multiple sound sources, or the width of a singular source.

Classical Recording Techniques:

All recording is an exaggeration of reality. Microphone placement helps to determine how a sound source will be portrayed in the final recording. The following descriptions or mic techniques serve as a starting point for microphone placement. Moving a microphone just a few inches can sometimes have a profound influence on the final results of a recording. The best practice is to start from a known reference point and then use your own ears to adjust the mics as necessary.

Spot Miking: Placing a single microphone on a sound source. This technique picks up a mono signal that can be placed anywhere in the stereo field. Typically the microphone will be placed between 1 inch and 4 feet in distance away from the source. The closer the microphone is the more this technique favors the articulate direct sound of the source.

Spot miking is common for placing soloists such as vocalists out in front of an accompanying ensemble.

Spaced Pair: This technique involves using two omni or cardioid microphones. The microphones must be a matched pair. These microphones are set parallel to each other, aiming towards the sound source. They are spaced at least 3 feet apart from each other. The width of the final recording can be manipulated by increasing the distance between these two microphones. However, there is a risk that the listener may perceive a center hole in the final stereo image if these microphones are spaced further than 8 – 12 feet from each other. This can be fixed by the addition of another mono mic or stereo pair of mics placed in the center of the spaced pair.

Spaced Pair miking is one of the oldest and simplest ways of stereo miking, but it requires careful listening in order to balance the stereo field presentation of the sound source. <u>Classical recording generally requires a cardioid spaced pair to be placed 8 to 14 feet back from the source, and between 8 and 20 feet up from the floor.</u> An omni spaced pair tends to require the mics to be further apart. The mics can be placed closer to the ensemble, between 4 feet and 12 feet back as well as 4 feet to 16 feet up from the floor depending on the reverb in the room.

Three Across: This technique builds upon the spaced pair technique. A center omni or cardioid microphone is aimed at the center of the sound source. It is placed an appropriate distance from the source in order to achieve the desired direct to reverberant ratio. Two additional microphones are placed on each side of this microphone, at a minimum distance of 3 feet. These microphones are often aimed parallel to the center microphone but this angle can be adjusted in order to favor other sound sources closer to them.

This technique uses similar distances to the Spaced Pair technique. It can be used with either cardioid or omni microphones, or a combination of the two. However, the two outer microphones must be a matched pair. The outside pair of microphones are also called **outriggers**.

X-Y: This technique is also called coincident pair miking. A matched pair of cardioid microphones is placed so that their capsules overlap one on top of the other. Then the backs of the microphones are angled away from each other, usually between 90 degrees and 135 degrees so that one mic effectively points Left and the other Right towards the sound source. This technique creates a stereo recording that is collapsible to MONO later on if desired because the sound that is recorded is naturally time-aligned.

This technique creates a stereo field with a prominent center image. While this can add articulation and detail to a source, it also suffers from a lack of depth and width. This technique will not work with omni microphones. It can be used in place of a Spot mic, or it can be used at a distance of 8 to 14 feet in order to capture the full stereo image of an ensemble.

ORTF (and variations): This technique was developed by the French Broadcasting Organization in 1960. It stands for Office de Radiodiffusion-Télévision Française. A matched pair of cardioid microphones is placed parallel to each other so that their capsules are 17cm. The microphones are aimed away from each other at a 110 degree angle. This is the official ORTF standard.

Variations on the ORTF technique work quite well. The microphones can be angled between 75 and 130 degrees, and the capsules can be spaced between 5 inches and 16 inches apart. ORTF technique captures a believable stereo field. When the microphone angle is decreased it narrows the image width. When the microphone angle is increased, or the distance between the mics is increased, the width of the image increases with generally acceptable changes to the center image. ORTF miking is usually placed between 7 and 12 feet away from the source, and between 8 and 14 feet up from the floor.

The ORTF technique is the most commonly used technique at the University of Illinois.

It is regularly used in the Smith Recital Hall for soloists with piano accompaniment. In this setting the soloist is placed onstage in front of the piano, either center stage or slightly stage left. The piano is placed center stage or slightly stage right. The ORTF mics will be placed center stage, 8 feet back from the soloist and 8 feet up from the stage floor.

In the KCPA Great Hall the ORTF technique is usually used to capture the center image of large ensemble performances. This is flanked by a matched pair of cardioid outriggers. All of these microphones are hung 15 feet above the stage floor. They are spaced between 6 and 12 feet away from the ensemble. The outriggers are spaced between 15 and 25 feet away from the center ORTF, in order to capture the full width of the actual stage. This setup is also augmented by a distant spaced pair of OMNI mics that serve as Hall mics for picking up the reverberant signal closer to the balcony.

Recording Environment Hazards for Concern

poor room acoustics

air circulation noise

lighting noise

noise from the use of risers

noise from the use of music on music stands

noise from performer shoes

noise from observers in the room

extraneous noise from beyond the recording space

When working with stereo miking techniques potential phase cancellation problems can exist

When two or more microphones are mixed together for a stereo recording, phase cancellation can affect the timbre and the significantly reduce the low frequencies of your recording. The simplest thing you can do, therefore, is first set up the stereo pair to your satisfaction, and then quickly switch to mono monitoring to see if there is a loss of low frequencies or if there is a comb filter effect in your sound. IF you notice this undesirable effect, subtly adjust the distance between the mics to massage the timbral balance of the mono sound. Small mic movements will make quite large differences to the mono mix, but without making a huge difference to the stereo sound. The goal is to find mic positions that keep the timbral quality as consistent as possible as you flip between mono and stereo monitoring.

When close miking, it makes sense to keep each mic closer to the instrument it is covering than to sources of spill. This idea is often encapsulated as the '3:1 rule' — namely, in order to keep spill manageably low, the distance between mics on different instruments should be at least three times the distance between each mic and the instrument it is supposed to be covering.

3:1 Rule of Microphone Placement

By Sweetwater *on Jul 29, 1997*

When using two microphones to record a source, normally you will get the best results by placing the second mic three times the distance from the first mic - that the first mic is from the source. This is known as the "3:1 Rule of Microphone Placement." An example: If the first mic is 1 foot from a source, the second mic should be placed 3 feet from the first mic.

While many engineers believe that adhering to the 3:1 Rule will minimize phase cancellation, technically this is not true. The 3:1 Rule works because the level of the signal entering the second mic (the one farther away) is reduced in level compared to the signal entering the first mic; tripling the distance substantially reduces the relative level of the signal in the two mics. This reduces the effects of phase cancellation, since the most cancellation will occur when the two mic signal levels are equal.

Keep in mind that some rules are meant to be broken; you may prefer the sound created by ignoring the 3:1 Rule — experiment and let your ears be your guide! Just be certain your sound is accurately represented.

Audio Recording Techniques
Check Yourself - Exam #2

Part I.

1. Explain what is meant by the term *analog audio signal*.

2. Explain what is meant by the term *digital audio signal*.

3. Define *sampling rate*. What is the sampling rate used for compact disc recording/production?

4. What does *AES/EBU* stand for, and how is this term used in the context of recording? What connectors are used for *AES/EBU*?

5. What does *S/PDIF* stand for, and how is this term used in the context of recording? What connectors are used for *S/PDIF* ?

6. What range of VU's should your average recording be, when using a recording device with VU meters?

7. What range of VU's should your amplitude peaks be, when using a recording device with VU meters?

8. What range of dB's should your average recording be, when using a recording device with dB meters?

9. What range of dB's should your amplitude peaks be, when using a recording device with dB meters?

10. 0 VU is equivalent approximately to what dB meter reading?

Part II.

11. Define the construction/function differences for dynamic microphones and condenser microphones.

12. Although both condenser and dynamic microphones can be used for recording and sound reinforcement applications, what microphone type is generally better for recording (and why)? What microphone type is generally better for sound reinforcement (and why)?

13. Define polar pattern. Define and graph each of the polar patterns of microphones as discussed in class.

Audio Recording Techniques
Check Yourself - Exam #3

1. List and explain any three of the six recording environment hazards you need to be concerned about, prior to and during your recording session? List at least one solution for each problem area that you mentioned.

	Problem	Solution
1.		
2.		
3.		

2. When close miking a vocalist or someone speaking, should the microphone be positioned on-axis OR off-axis to the source? Why?

3. List the names of three stereo recording techniques that use two microphones. Describe what characteristic differences in sound you might expect to receive for each technique.

1.

2.

3.

4. Classical recording generally requires a cardioid spaced pair to be placed ___________ feet back from the source, and between _______________ feet up from the floor.

5. Classical recording ORTF miking is usually placed between ___________ feet away from the source, and between _____________ feet up from the floor.

6. Classical recording ORTF mic position has the two mic capsules positioned at a distance of ______ apart, with both microphones position at a _____ degree angle.

Mixer Basics

An audio mixer or mixing console is an electronic device designed mainly for combining two or more input channels of sound into one or more output channels. The mixer electronically adds or combines the signals together without loss of quality and this process is generally referred to as **waveform summation** since the frequencies and amplitudes of the input signals are summed together. Due to the summation process, one must realize that the amplitude of the summed signal will be higher than the amplitude of the individual input amplitudes. Always pay attention to the VU or decibel meters for the output channels. Be aware of this situation and take great care to not overdrive the output channels of the mixer as this would cause distortion and probable damage to the mixer and other audio components.

Most mixers are divided into specific sections based upon their function. There is an input section followed by the output section. A mixer that has 8 **inputs** and 4 **outputs** is referred to as a 8 X 4 (8 by 4) mixer. Many mixers also have a **submix** section (this section further mixes the main outputs of the mixer to some other group of outputs that usually has a fewer number of output channels). A mixer with 12 inputs, 4 outputs and a submix of 2 outputs would be referred to as a 12 X 4 X 2 (12 by 4 by 2) mixer.

Mixers usually have two different inputs (a **mic level** input and a **line level** input) for each individual mixer input. The output channels of almost all mixers are line level outputs.

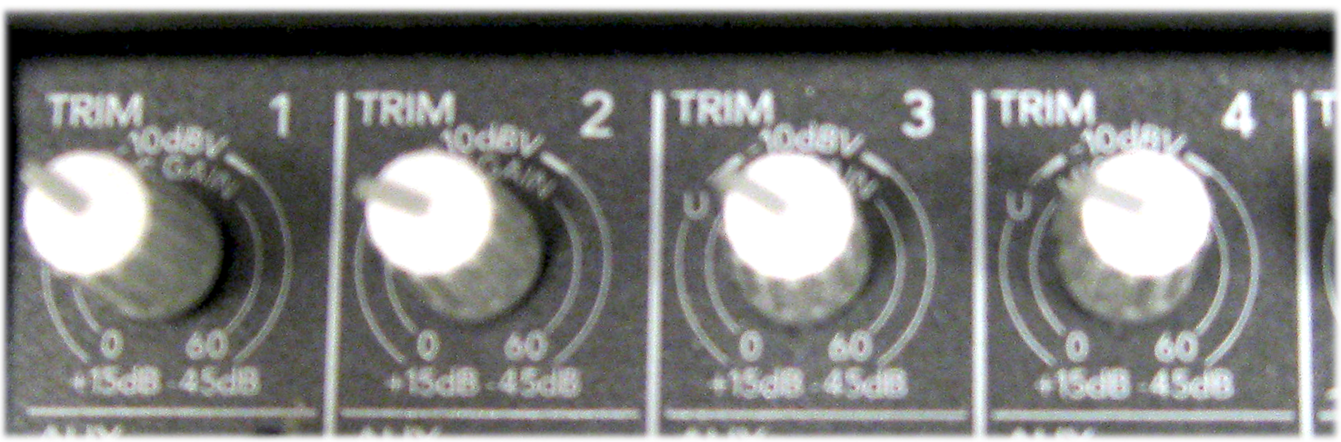
Each individual mixer input channel will have as the basic features: an amplitude input control (rotary knob or vertical **fader**), perhaps an input **channel ON** switch, **output channel selector** switches and a **panpot** that allows you to **pan** (locate or position) the input signal anywhere within the selected stereo output channels (done so via the output channel selector switches).

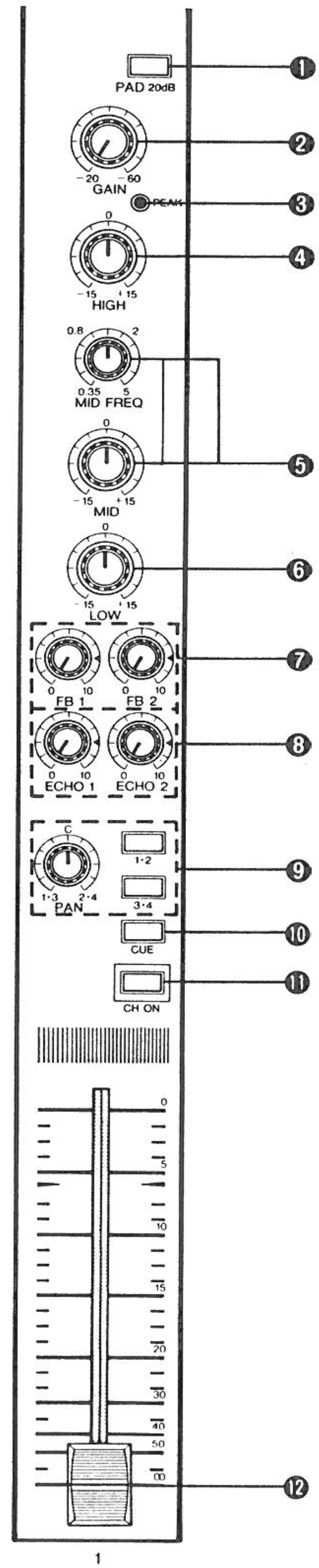
Many mixers also have a set of controls that effect the relative strength of low, midrange and high frequencies of the input signal for each mixer input channel. This timbre control is called **equalization**.

The master output section of the mixer will usually have individual amplitude output control for each **output channel** (rotary knob or vertical fader) (these output channels are sometimes referred to as **group outputs**). Again, care must be taken to not overdrive the output channels of the mixer as this would cause distortion and probable damage to the mixer and other audio components.

Pad – a pad or pad switch inserts a 20 to 26 dB attenuation prior to the mixer input (pre)amplifier that permits the mixer input to correctly accept a line level (or instrument level) input without clipping/distortion. IF the pad is OFF (or not engaged), the mixer input (pre)amplifier is set to correctly preamplify a mic level signal.

Gain/Trim control – the gain or trim control permits you to continuously adjust the sensitivity of the mixer input (pre)amplifier.

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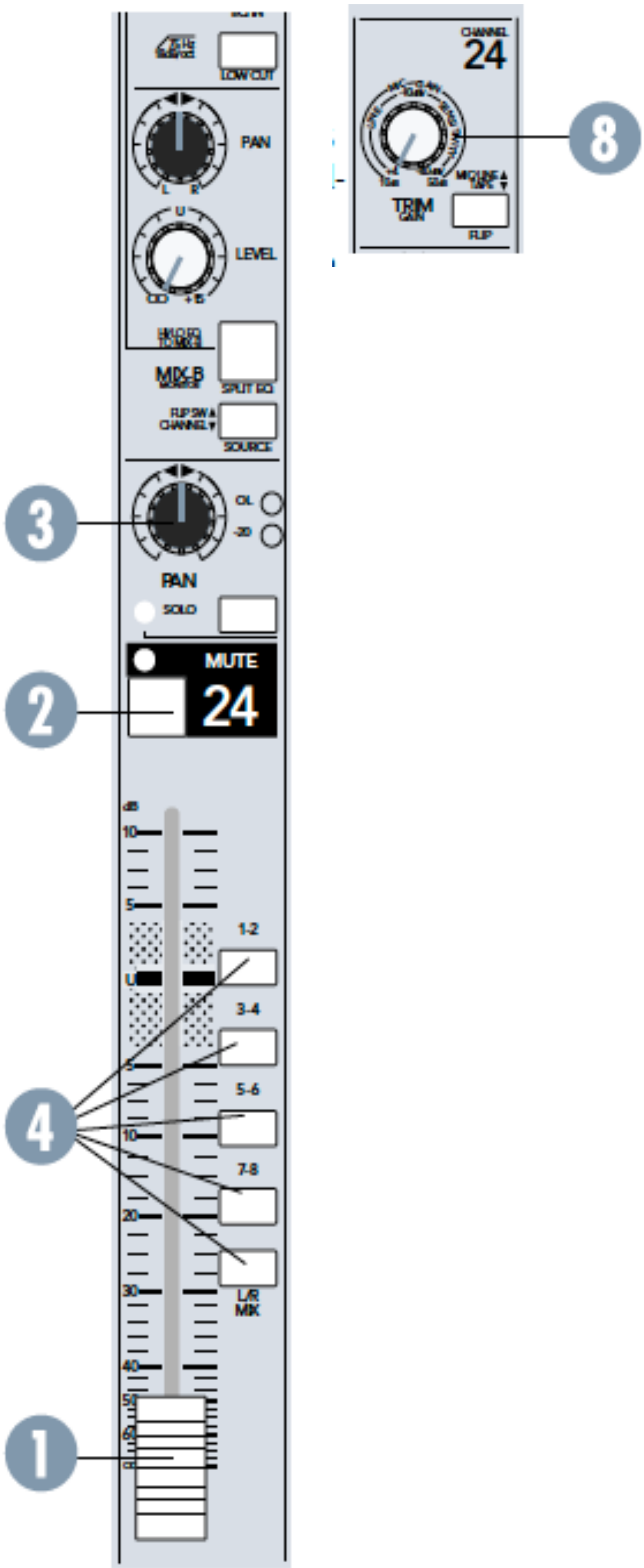
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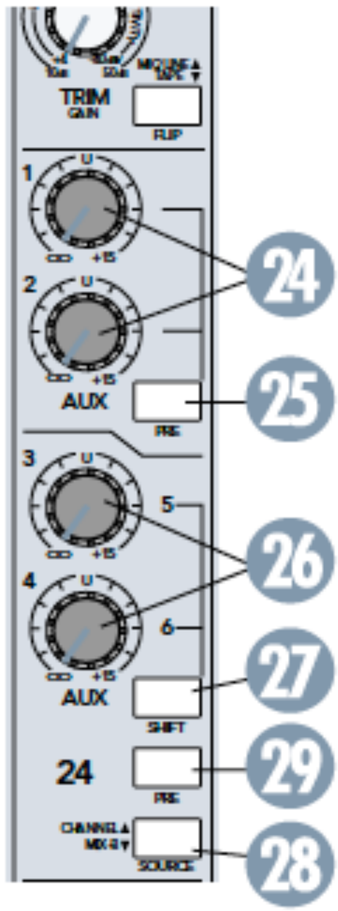
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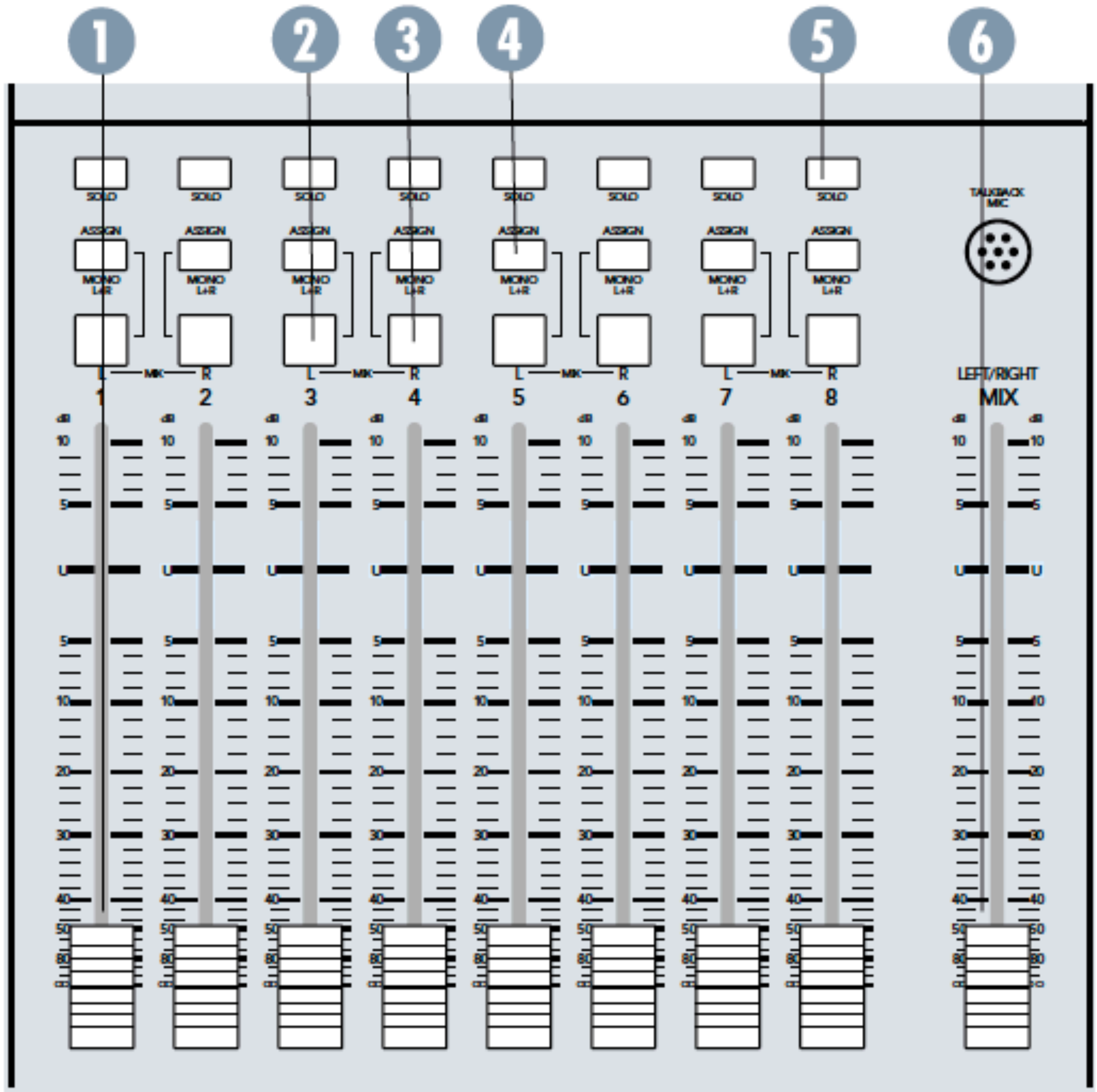
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