the Experimental Music Studios

School of Music

University of Illinois at Urbana-Champaign

Very Basic Concepts

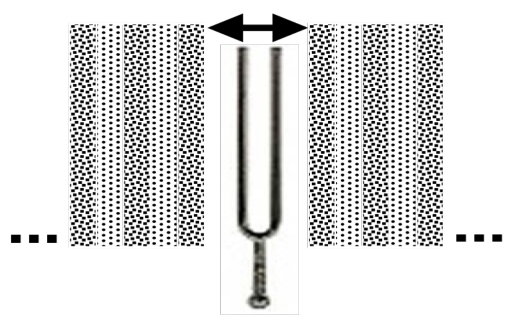
Sound – combinations of frequencies and amplitudes emanating from a vibrating source, transferred through a medium (gaseous, liquid or solid) by means of a series of stronger and weaker areas of molecules (waves of compressions and rarefactions) – eventually received by our inner ear and translated into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

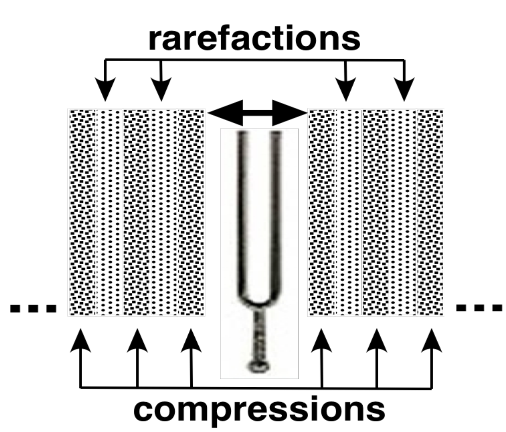
Acoustics - the study of the physical properties of sound;
- the study of the physics and transmission of sound.

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.

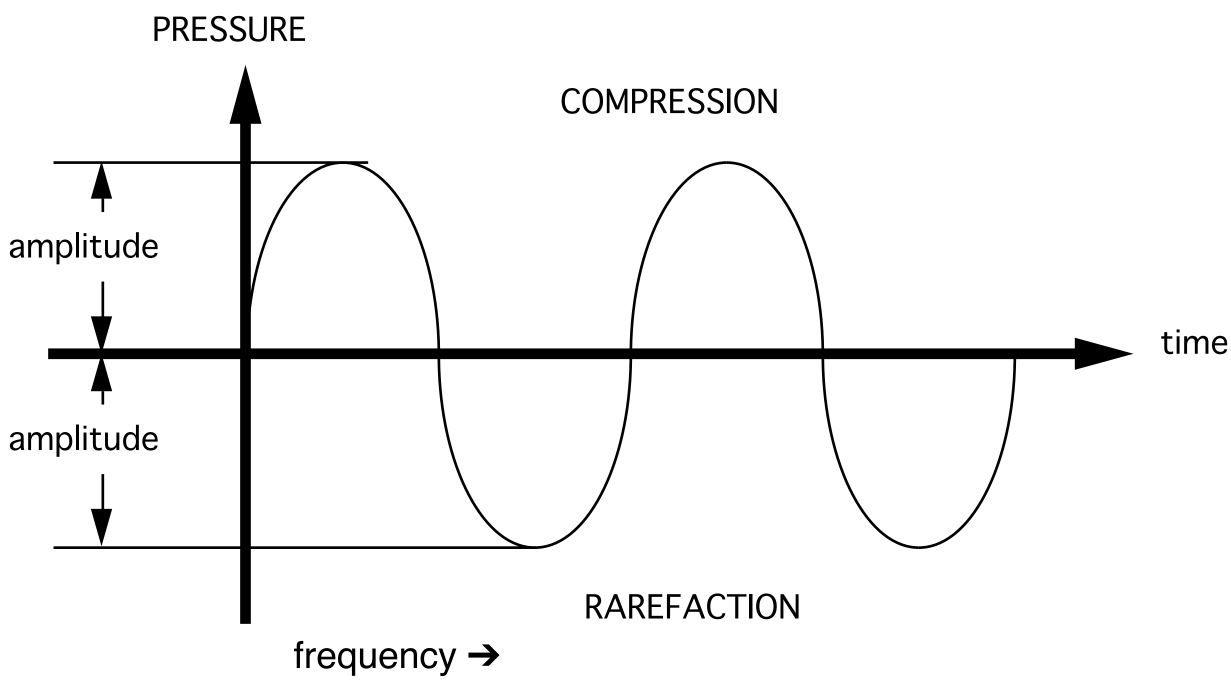


3. The area of higher air pressure is called a "compression".
4. The area of lower air pressure is called a "rarefaction".



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.

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.

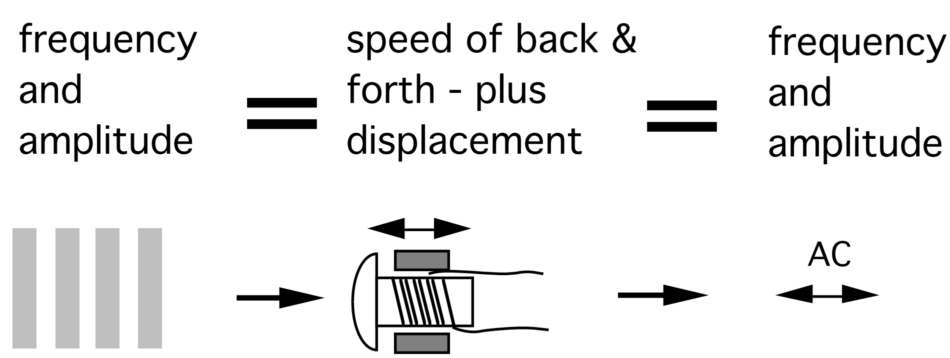
Analog Audio Signal

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.



soundwave changes in air pressure   diaphragm moving back and forth   alternating current moving back and forth

- 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).

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 milliVolts] 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

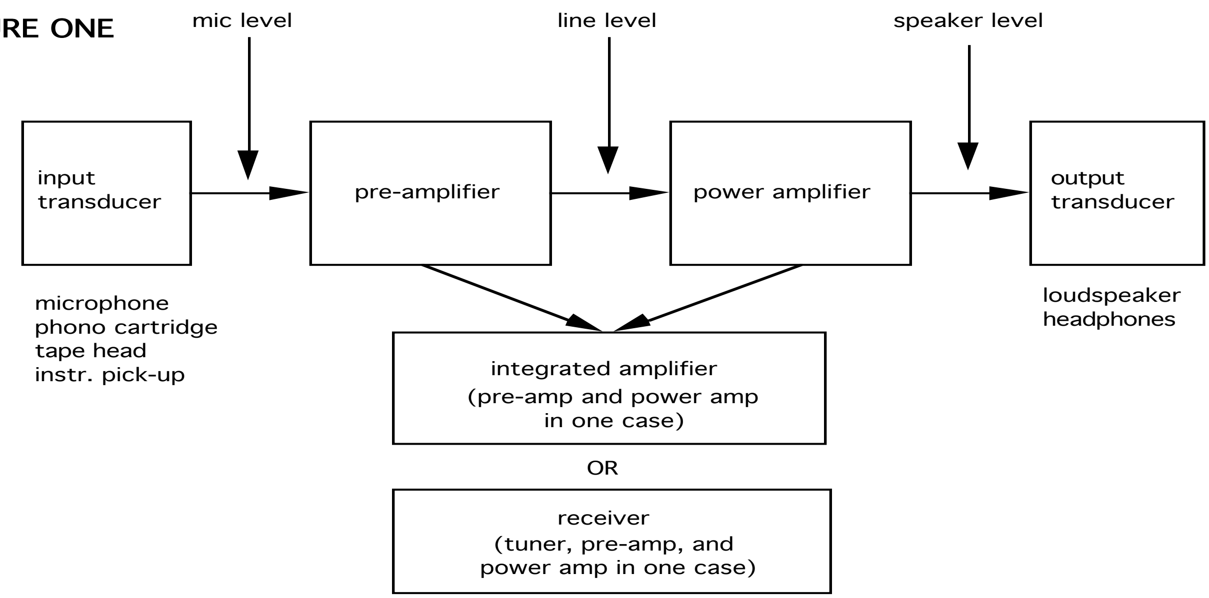


FIGURE TWO

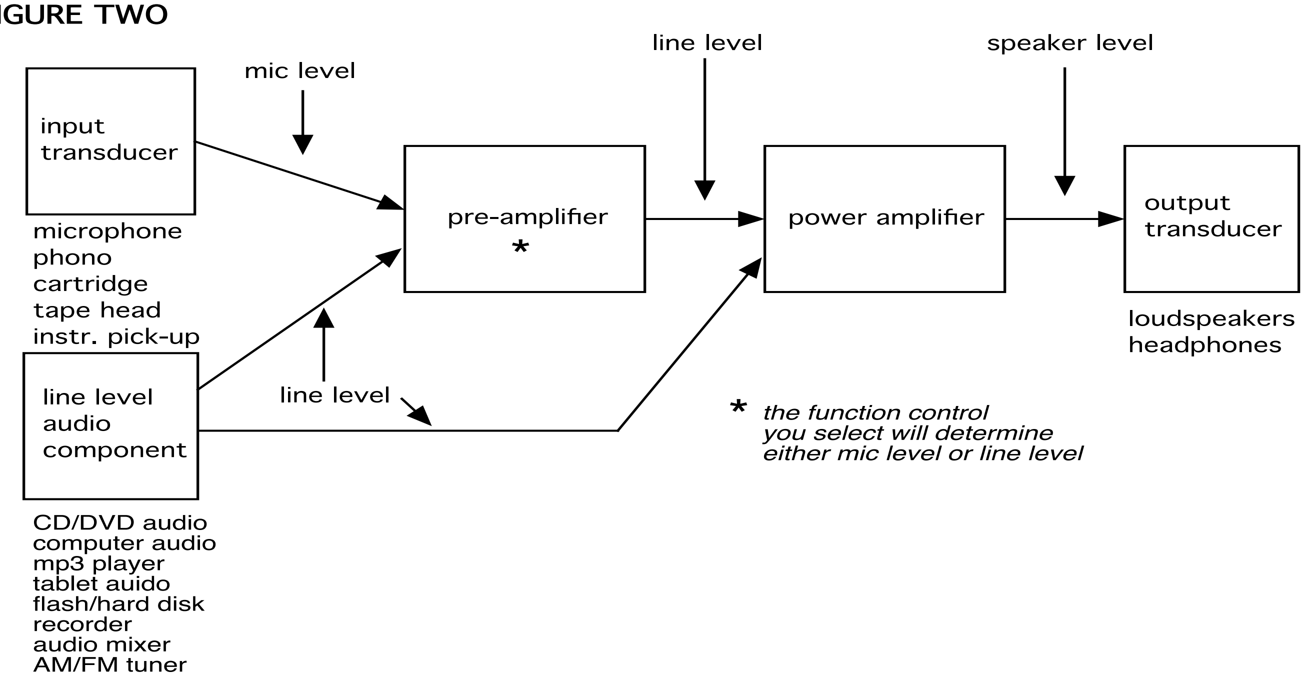
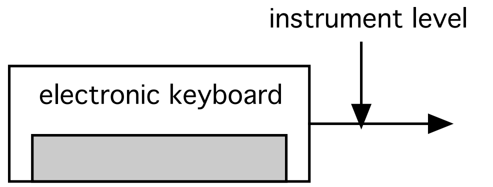


FIGURE THREE



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

FIGURE FOUR

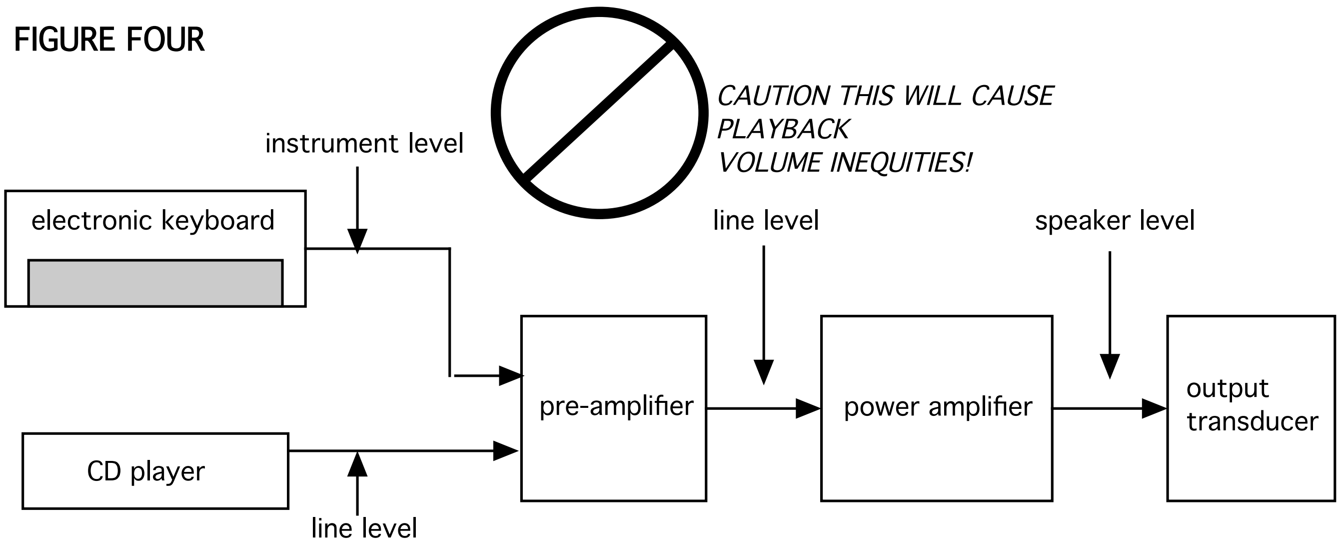
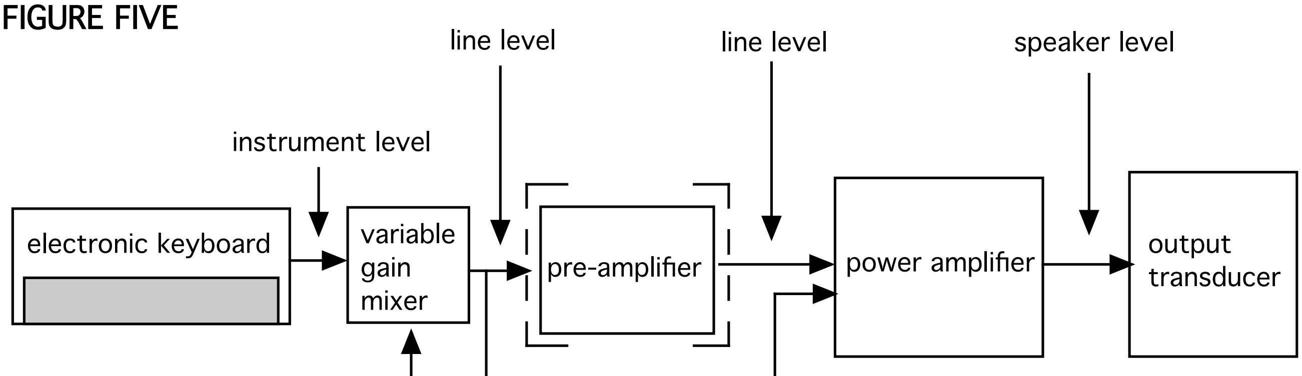


FIGURE FIVE

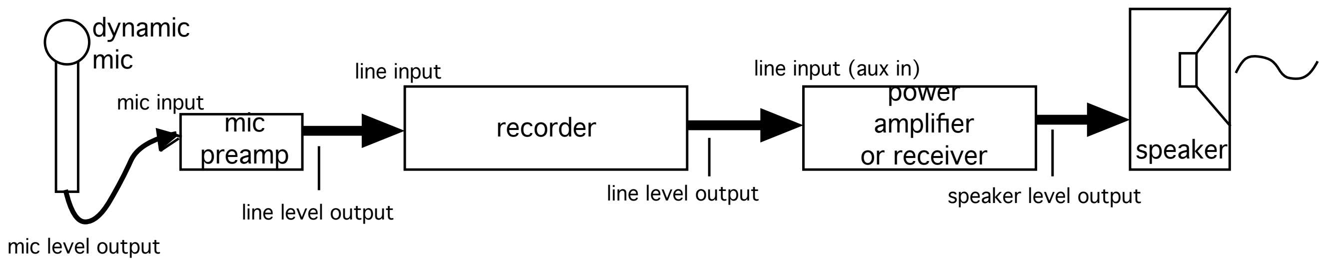


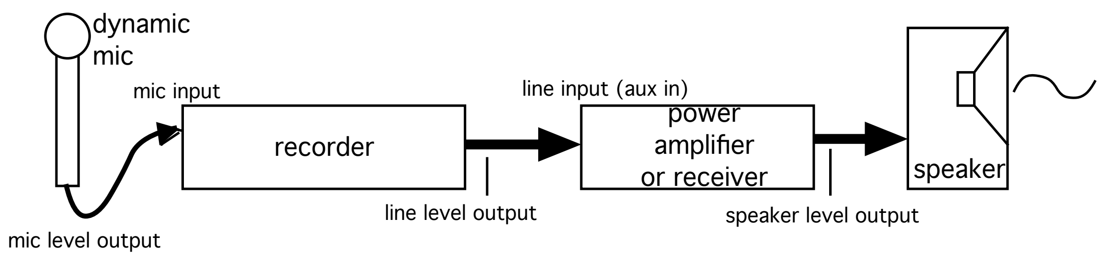
A variable gain mixer
allows one to increase the
amplitude of the instrument level
signal UP to the expected standard
line level signal - which a line level
audio component device requires.

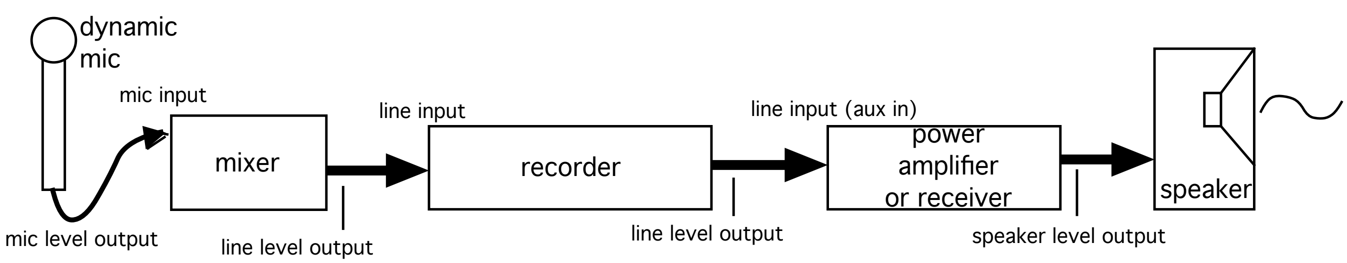
This eliminates the problem of different
playback volume levels when you switch back
and forth among different audio input
devices.

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).

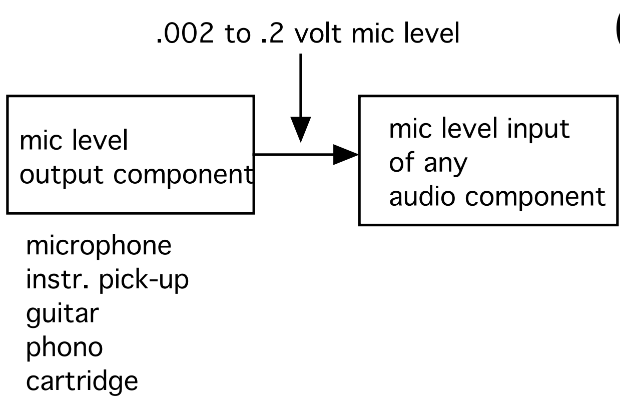
The following diagrams show the correct set-ups and ordering for the audio signal path.



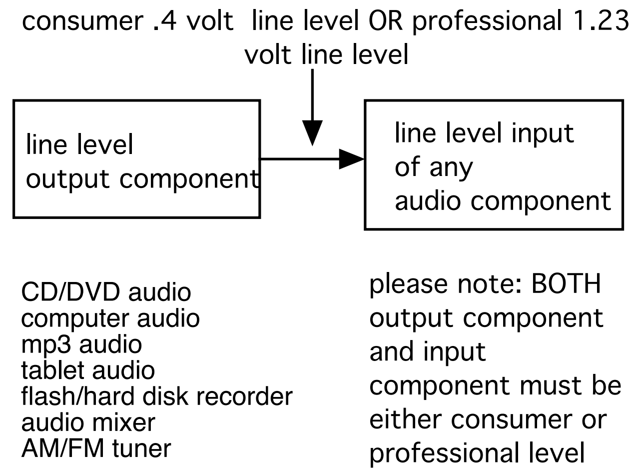


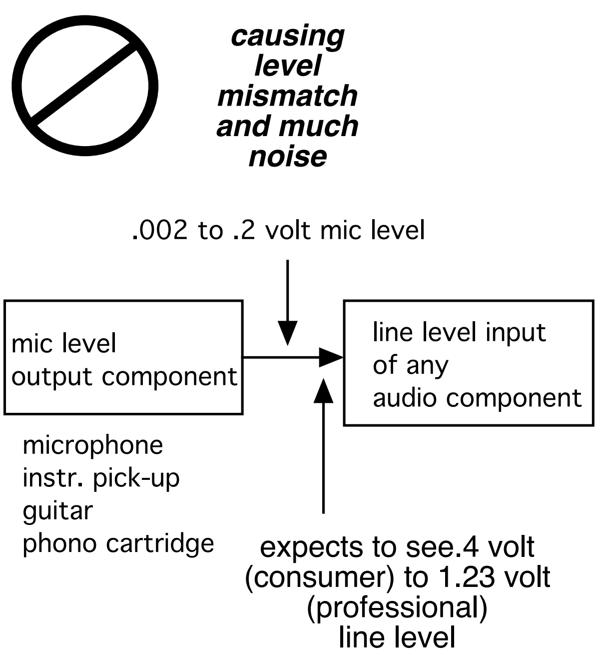


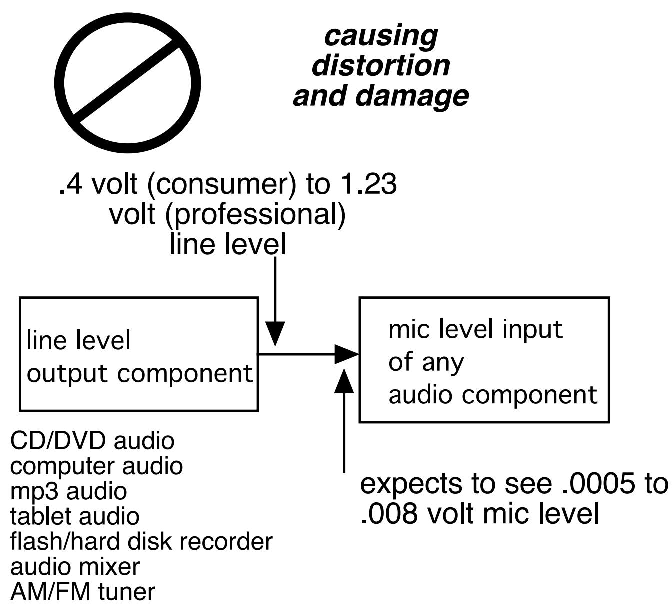
THERE ARE SIGNIFICANT COMPATIBILITY PROBLEMS BETWEEN MIC LEVEL AND LINE LEVEL.
PLEASE USE GREAT CAUTION WHEN CONNECTING AUDIO COMPONENTS.



or



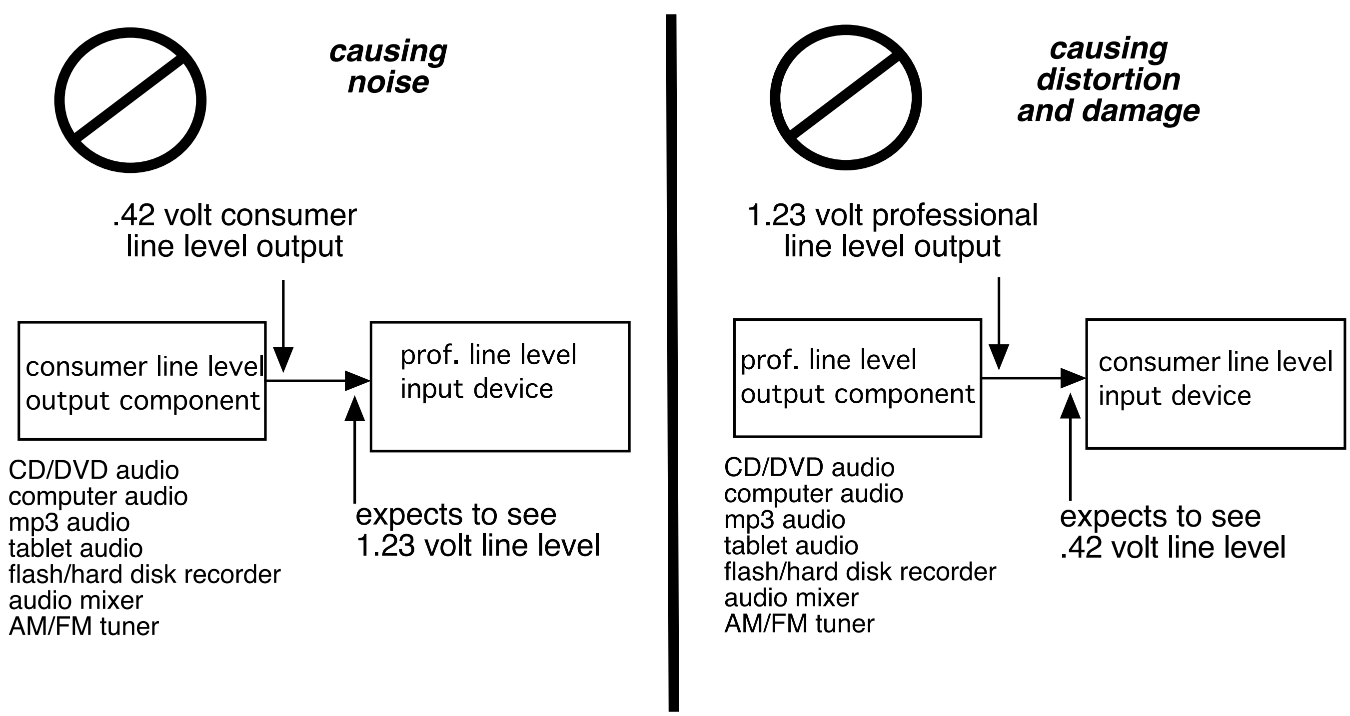




Consumer Audio vs. Professional Audio

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

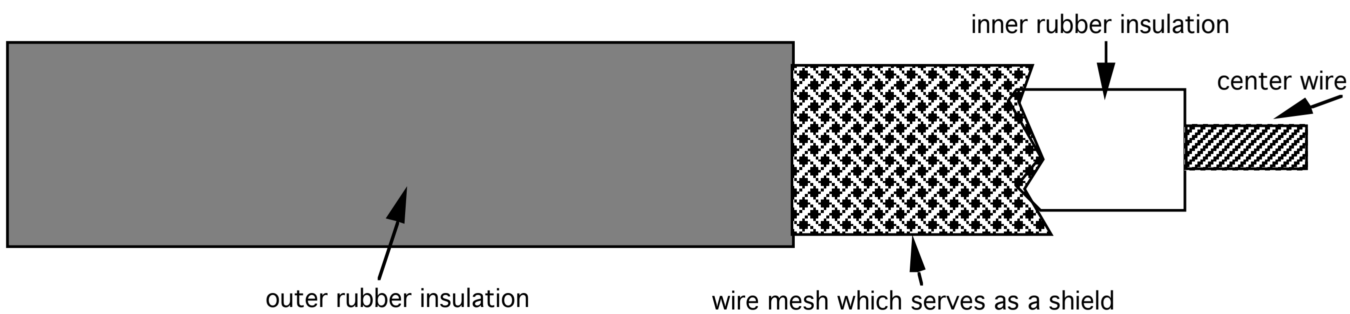


Consumer Audio Line Levels and associated cable types

- 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.)

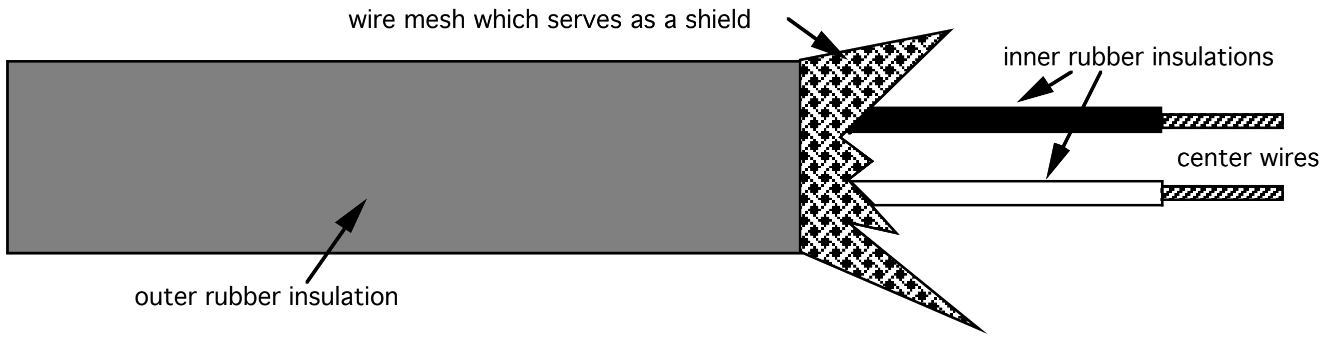
- Consumer audio components offer an acceptable audio frequency range and dynamic range with a detectable noise bed (hiss) in the background.

Professional audio components

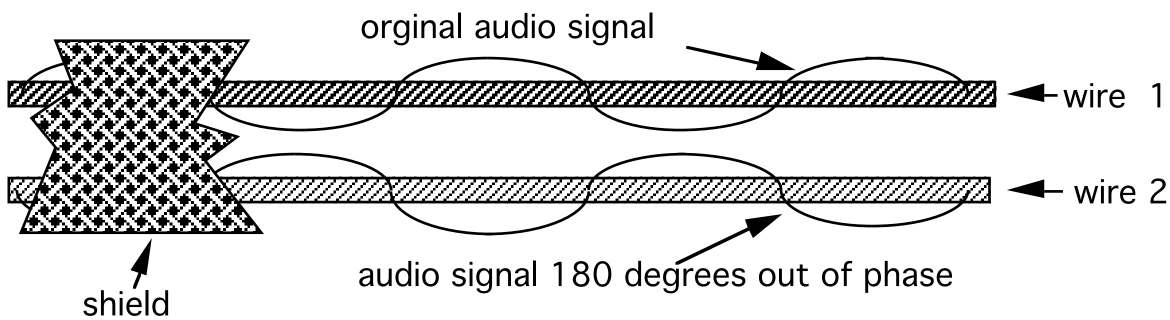
- Line level signals are 1.23 volts and utilize balanced circuitry and cables.

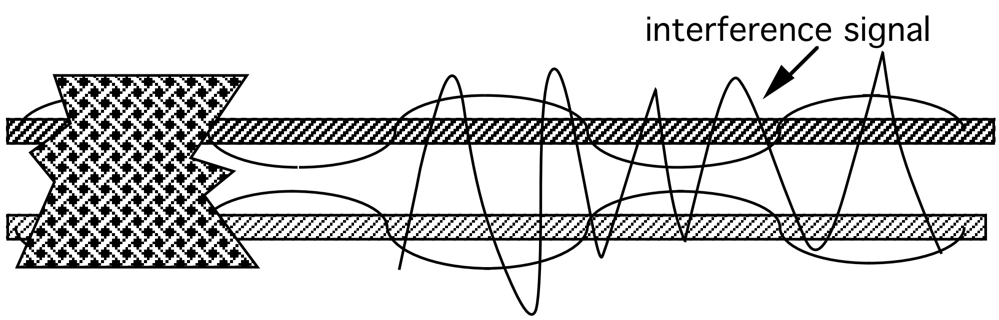
Balanced circuitry uses a three-wire cable for interconnecting audio components. This three-wire cable is referred to as either a two conductor shielded cable OR a 3 conductor cable. The two center wires or conductors carry the audio signal 180 degrees out of phase with each other, and the shield or 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



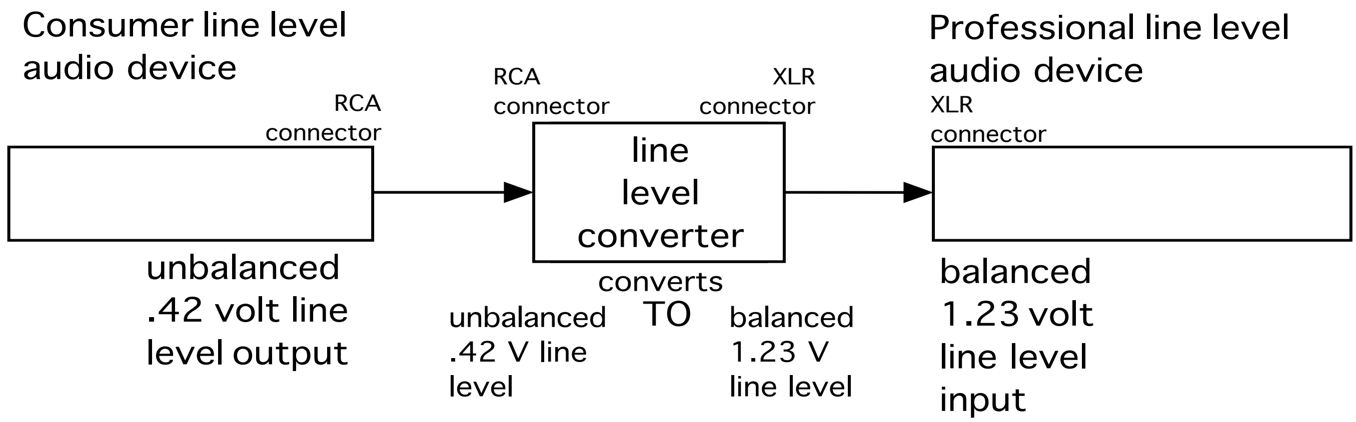
balanced signal (two copies of one audio signal 180 degrees out of phase)



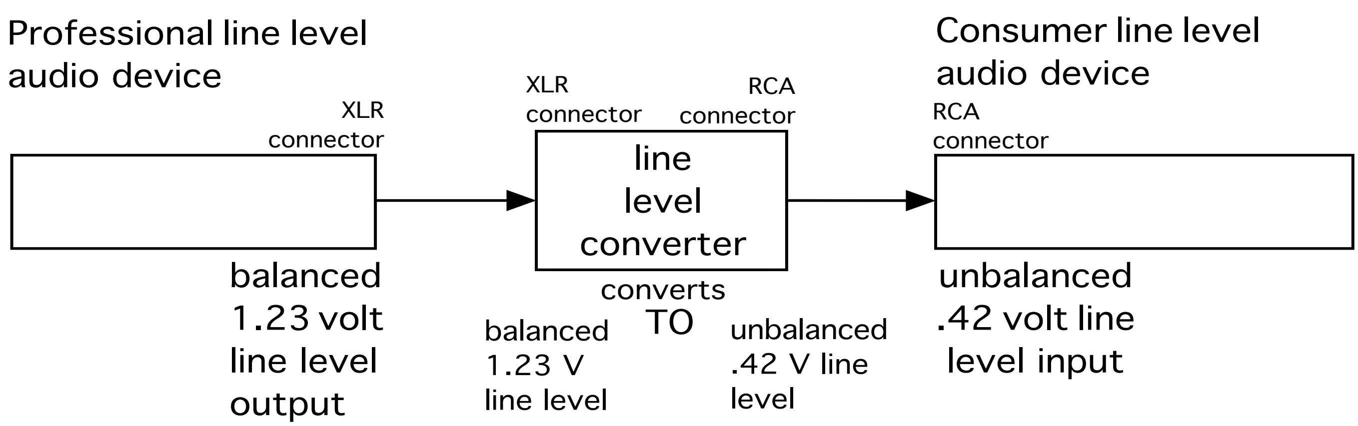


- Professional audio components are expensive and are able to filter out radio frequency interference.
- Long cable lengths for mic level and line level signals may be used without audio signal loss or unwanted radio frequency interference.
- Professional audio components offer a wider frequency reproduction 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



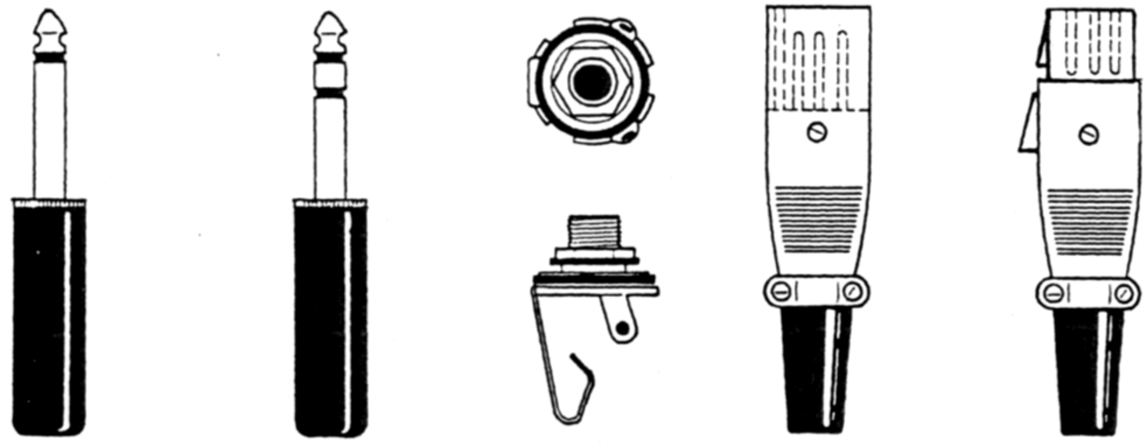
Line level converters (shifters) such as:

- 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



1/4 inch mono phone plug (in-line)

1/4 inch stereo phone plug (in-line) (TRS)

1/4 inch mono phone jack (chasis-mount)

XLR 3 pin plug (in-line)

XLR 3 pin jack (in-line)

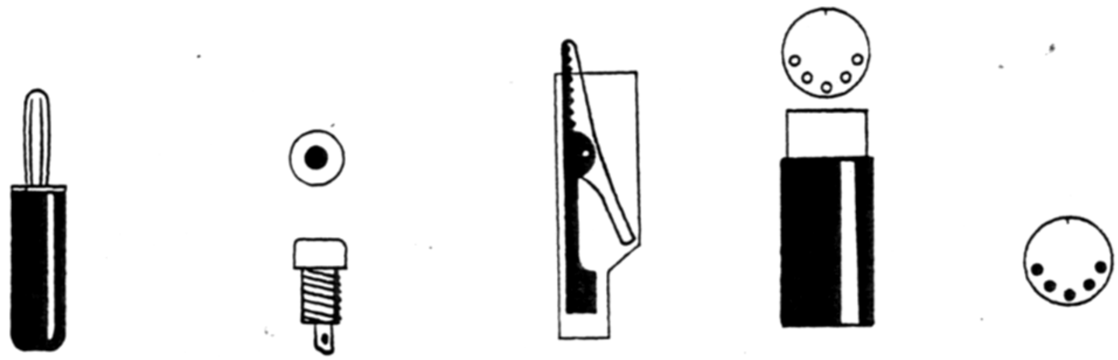
mini phone plug (in-line)

mini phone jack (chasis-mount)

RCA plug (in-line)

RCA jack (in-line)

RCA jack (chasis-mount)



banana plug (in-line)

banana jack (chasis-mount)

alligator clip

MIDI plug (in-line)

MIDI jack (chasis-mount)

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:

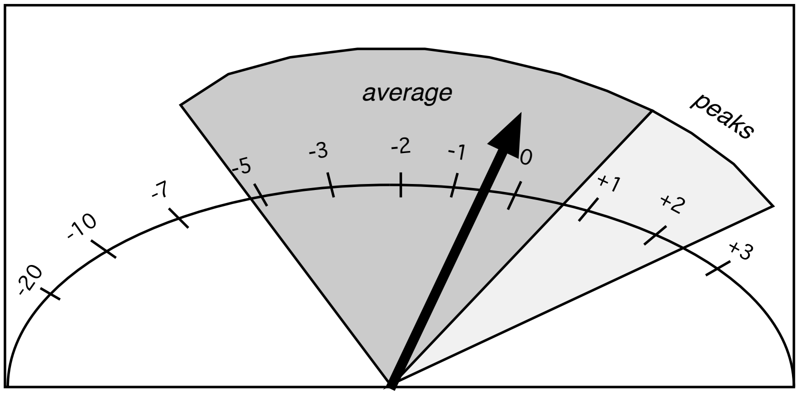
- AES/EBU	**uses XLR or fibreoptic connectors**
- S/PDIF	**uses RCA or fibreoptic connectors**

types of standardized two-channel digital recorders:

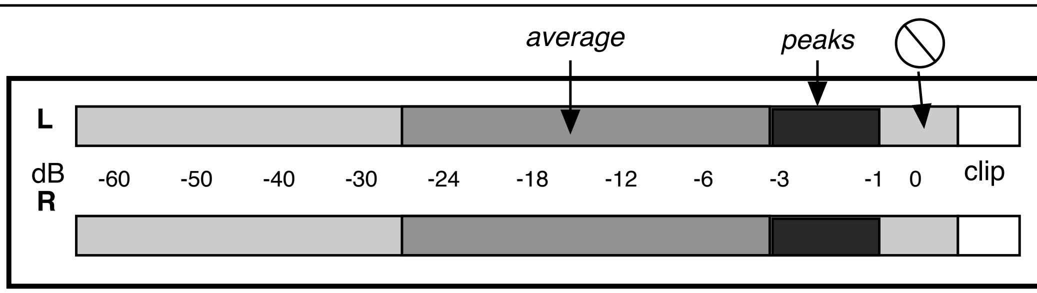
- **DASH**
- **RDAT (DAT)**
- **MD**
- **Hard disk recorders**
- **Flash recorders**

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.