Audio Terminology

A:

absorption To *absorb* is to receive (an impulse) without echo or recoil: *a fabric that absorbs sound; a bumper that absorbs impact*; therefore *absorption* is the act or process of absorbing. The absorption of sound is the process by which sound energy is diminished when passing through a medium or when striking a surface, i.e., *sound is attenuated by absorption*. [AHD] The physical mechanism is usually the conversion of sound into heat, i.e. sound molecules lose energy upon striking the material's atoms, which become agitated, which we characterized as warmth; thus, absorption is literally the changing of sound energy to heat. A material's ability to absorb sound is quantified by its ***absorption coefficient***, whose value ranges between 0 (total reflection) and 1 (total absorption), and just to keep things interesting, varies with sound frequency and the angle of incidence.

acoustic feedback The phenomenon where the sound from a loudspeaker is picked up by the microphone feeding it, and re-amplified out the same loudspeaker only to return to the same microphone to be re-amplified again, forming an acoustic loop. Each time the signal becomes larger until the system runs away and *rings* or *feeds back* on itself producing the all-too-common scream or squeal found in sound systems. These buildups occur at particular frequencies called *feedback frequencies*.

acoustics 1. *Hearing*; from the Greek *akouein*: to hear. 2. The study of sound

active crossover A loudspeaker crossover requiring a power supply to operate. Usually rack-mounted as a separate unit, active crossovers require individual power amplifiers for each output frequency band. Available in configurations known as *stereo 2-way*, *mono 3-way*, and so on. A *stereo 2-way* crossover is a two-channel unit that divides the incoming signal into two segments, labeled *Low* and *High* outputs (***biamped***). A *mono 3-way* unit is a single channel device with three outputs, labeled *Low*, *Mid* and *High* (***triamped***). In this case, the user sets two frequencies: the Low-to-Mid, and the Mid-to-High crossover points. Up to *stereo 5-way* configurations exist for very elaborate systems.

active equalizer A variable equalizer requiring a power supply to operate. Available in many different configurations and designs. Favored for low cost, small size, light weight, loading indifference, good isolation (high input and low output impedances), gain availability (signal boosting possible), and line-driving ability. Disliked for increased noise performance, limited dynamic range, reduced reliability, and RFI susceptibility; however, used everywhere.

ambience 1. *Acoustics*. A perceptual sense of space (Blesser). The acoustic qualities of a listening space (White). 2. *Psychoacoustics* The special atmosphere or mood created by a particular environment; also spelled ***ambiance*** (AHD). Contrast with reverberation.

ampere *Abbr.* **I**, also **A. 1.** A unit of electric current in the International standard meter-kilogram-second (mks) system. It is the steady current that when flowing in straight parallel wires of infinite length and negligible cross section, separated by a distance of one meter in free space, produces a force between the wires of 2E-7 newtons per meter of length. 2. A unit in the International System specified as one International coulomb per second and equal to 0.999835 ampere. [After **André Marie Ampère**.] [AHD]

amplifier An electronic device used to increase an electrical signal. The signal may be voltage, current or both (power). ***Preamplifier*** is the name applied to the first amplifier in the audio chain, accepting inputs from microphones, or other transducers, and low output sources (CD players, tape recorders, turntables, etc.). The preamplifier increases the input signals from mic-level, for instance, to line-level. ***Power amplifier*** is the name applied to the last amplifier in the audio chain, used to increase the line-level signals to whatever is necessary to drive the loudspeakers to the loudness required. See amplifier classes.

audio 1. Of or relating to humanly audible sound, i.e., audio is all the sounds that humans hear (*approximately 20 Hz - 20 kHz*). 2. a. Of or relating to the broadcasting or reception of sound. b. Of or relating to high-fidelity sound reproduction. [Audio traveling through air is vibrations, or cycles of alternating pressure zones. Rarefaction follows each cycle of compression, which produces a wave.] [AHD]

B:

balanced line The IEC standard on amplifiers explains a balanced interface by saying that "The purpose of a balanced interface is to transfer a desired signal as a differential voltage on two signal lines." (IEC 602689-3:2001, page 111). It goes on to explain that "... only the common-mode impedance balance of the driver, line, and receiver play a role in noise or interference rejection. This noise or interference rejection property is independent of the presence of a desired differential signal. Therefore, it can make no difference whether the *desired* signal exists entirely on one line, as a greater voltage on one line than the other, or as equal voltage on both of them."
Balanced lines are the preferred method (for hum free) interconnecting of sound systems using a shielded twisted-pair. Because of its superior noise immunity, balanced lines also find use in interconnecting data signals, e.g., RS-422, and digital audio, e.g., AES/EBU. The principal behind balanced lines is that the signal is transmitted over one wire and received back on another wire. *The shield does not carry any information*, thus it is free to function as a true shield, but must be earth grounded *at each end* to be successful. (For a detailed tutorial on proper grounding practices. This circuit's shining virtue is its great common-mode noise rejection ability. The concept here relies on induced noise showing up equally (or common) on each wire. It is mainly due to EMI (electromagnetic interference: passing through or near magnetic fields), RFI (radio frequency interference: strong broadcast signals), noisy ground references, or a combination of all three. A true balanced line exhibits exactly equal impedance from each line relative to ground, guaranteeing equal noise susceptibility. Since the balanced input stage amplifies only the difference between the lines, it rejects everything else (noise) that is common to the lines

bandpass filter A filter that has a finite passband, neither of the cutoff frequencies being zero or infinite. The bandpass frequencies are normally associated with frequencies that define the half power points, i.e. the -3 dB points.

bandwidth *Abbr.* **BW** 1. *Electronic filters* The numerical difference between the upper and lower -3 dB points of a band of audio frequencies. Used to figure the Q, or quality factor, for a filter. 2. *Telecommunications* The size of the communications channel. In analog communications, bandwidth is measured in Hertz (Hz), while digital communications measures bandwidth (data transfer rate) in bits per second. For example, an analog telephone channel has a bandwidth of 4,000 Hz, while a digitally coded telephone channel has a bandwidth of 64 kilobits/second.

biamp, **biamplified**, or **biamplification** Term used to refer to a 2-way active crossover where the audio signal is split into two paths, and using separate power amplifier channels for each driver.

boost/cut equalizer The most common graphic equalizer. Available with 10 to 31 bands, on 1-octave to 1/3-octave spacing. The flat (*0 dB*) position locates all sliders at the center of the front panel. Comprised of bandpass filters, all controls start at their center *0 dB* position and boost (amplify or make larger) signals by raising the sliders, or cut (attenuate or make smaller) the signal by lowering the sliders on a band-by-band basis. Commonly provide a center-detent feature identifying the *0 dB* position. Proponents of boosting in permanent sound systems argue that cut-only use requires adding make-up gain that runs the same risk of reducing system headroom as boosting.

C:

cables Audio systems use many different types of cables

- **coaxial cable** A single copper conductor, surrounded with a heavy layer of insulation, covered by a thick surrounding copper shield and jacket. A constant-impedance unbalanced transmission line.
- **data cable** Ethernet Data
- **fiber optics** The technology of using glass fibers to convey light and modulated information. Short distances (typically less than 150 feet) use plastic fibers, while long distances must use glass fibers.
- **mic cable** (aka **audio cable**) A shielded twisted-pair, usually designed for low current, high flexibility and low handling noise. The best insulating materials are somewhat inflexible, so most mic cables use rubber, neoprene, PVC, or similar materials, with small gauge wire, and therefore, true mic cables are not intended for long runs. Unfortunately the term "mic cable" has become synonymous with general-purpose audio cable (as distinguished from *speaker cable*) when it can be quite different. The very best audio cable may not be the best mic cable and vice versa.
- **quad mic cable** or **star-quad mic cable** [*a term coined by* ***Canare*** *for the first quad mic cable, but was not trademarked and is now a generic term*]. A four-conductor cable exhibiting very low noise and hum pickup (hum reduction can be 30 dB better than standard mic cable). The four conductors are wound together in a spiral, and then opposite conductors are joined together at the connectors forming a two-conductor balanced line (also called *double balanced*) with superior performance.
- **speaker cable** An unshielded insulated pair, normally not twisted, characterized by heavy (or large) gauge conductors (hence, low-resistance), used to interconnect the output of a power amplifier and the input of a loudspeaker. The coupling between amplifier and loudspeaker may be direct or via transformer (see constant voltage). The *star quad* design described above also makes excellent speaker cables for use in high noise environments.

- **twisted-pair** Standard two-conductor copper cable, with insulation extruded over each conductor and twisted together. Usually operated as a balanced line connection. May be shielded or not, abbreviated **UTP** (*unshielded twisted-pair*), or **STP** (*shielded twisted)*

cardioid A heart-shaped plane curve, the locus of a fixed point on a circle that rolls on the circumference of another circle with the same radius. (AHD)

cardioid microphone A directional microphone with an on-axis response shaped like a cardioid. Different degrees of caridiod-ness exist, termed subcardioid and hypercardioid.

CD-R (*compact disc-recordable*) A compact disc that is recordable at least once

chassis ground 1. The common point on a conducting chassis surrounding the system electronic circuit boards; usually separate from the signal ground but may be tied at one point. 2. The earth grounding connection provided on the chassis for safety reasons

compressor A signal processing device used to *reduce the dynamic range* of the signal passing through it. For instance, an input dynamic range of 110 dB might pass through a compressor and exit with a new dynamic range of 70 dB. This clever bit of skullduggery is normally done through the use of a VCA (voltage controlled amplifier), whose gain is a function of a control voltage applied to it. Thus, the control voltage is made a function of the input signal's dynamic content. [*Long answer*: What "compression" is and does has evolved significantly over the years. Originally compressors were used to reduce the dynamic range of the *entire signal*; with modern advances in audio technology, compressors now are used more sparingly. First the classical case: The history of compressors dates back to the late '20s and '30s (the earliest reference I have located is a 1934 paper in the Bell Labs Journal.) The need arose the very first time anyone tried to record (sound-motion pictures film recording, phonograph recording, etc.) or broadcast audio: *the signal exceeded the medium.* For example, the sound from a live orchestra easily equals 100 dB dynamic range. Yet early recording and broadcasting medium all suffered from limited dynamic range. Typical examples: LP record 65 dB, cassette tape 60 dB (w/noise reduction), analog tape recorder 70 dB, FM broadcast 60 dB, AM broadcast 50 dB. Thus "6 pounds of audio into a 4 pound bag" became the necessity that mothered the invention of the compressor (*sorry*). Early compressors did not have a "threshold" knob, instead, the user set a center ("hinge") point equivalent to the midpoint of the expected dynamic range of the incoming signal. Then a *ratio* was set which determined the amount of dynamic range reduction. The earlier example of reducing 110 dB to 70 dB requires a ratio setting of 1.6:1 (110/70 = 1.6). The key to understanding compressors is to always think in terms of *increasing and decreasing level changes* ***in dB*** *about some set-point*. A compressor makes audio *increases* and *decreases smaller.* From our example, for every input *increase of 1.6 dB above* the hinge point, the output only *increases 1 dB*, and for every input *decrease of 1.6 dB below* the hinge point, the output only *decreases 1 dB*. If the input increases by *x-dB*, the output increases by *y-dB*, and if the input decreases by *x-dB*, the output decreases by *y-dB*, where *x/y* equals the ratio setting. Simple - but not intuitive and not obvious. This concept of *increasing* above the set-point and *decreasing* below the set-point is where this oft-heard phrase comes from: "*compressors make the loud sounds quieter and the quiet sounds louder*." If the sound gets louder by 1.6 dB and the output only increases by 1 dB, then the loud sound has been made quieter; and if the sound gets quieter by 1.6 dB and the output only decreases by 1 dB, then the quiet sound has been made *louder* (it didn't decrease as much). Think about it - it's an important concept. With advances in all aspects of recording, reproduction and broadcasting of audio, the usage of compressors changed from reducing the entire program to just reducing selective portions of the program. Thus was born the *threshold* control. Now sound engineers set a threshold point such that all audio below this point is unaffected, and all audio above this point is compressed by the amount determined by the ratio control. Therefore the modern usage for compressors is to turn down (or reduce the dynamic range of) just the loudest signals. Other applications have evolved where compressors are used in controlling the *creation* of sound. For example when used in conjunction with microphones and musical instrument pick-ups, compressors help determine the final timbre by selectively compressing specific frequencies and waveforms. Common examples are "fattening" drum sounds, increasing guitar sustain, vocal "smoothing," and "bringing up" specific sounds out of the mix, etc.]

condenser microphone [*Also called* ***capacitor microphone*** *but more properly, the correct name is* ***electrostatic microphone***.] A microphone design where a condenser (the original name for *capacitor*) is created by stretching a thin diaphragm in front of a metal disc (the *backplate*). By positioning the two surfaces very close together an electrical capacitor is created whose capacitance varies as a function of sound pressure. Any change in sound pressure causes the diaphragm to move, which changes the distance between the two surfaces. If the capacitor is first given an electrical charge (*polarized*) then this movement changes the capacitance, and if the charge is fixed, then the backplate voltage varies proportionally to the sound pressure. In order to create the fixed charge, condenser microphones require external voltage (*polarizing voltage*) to operate. This is normally supplied in the form of phantom power from the microphone preamp or the mixing console.

connectors Audio equipment uses many types of connectors as follow

- **banana jack** or **banana plug** A single conductor electrical connector with a banana-shaped spring-metal tip most often used on audio power amplifiers for the loudspeaker wiring. Usually configured as a color-coded molded pair (red = hot & black = return) on 3/4" spacing. Also used for test leads and as terminals for plug-in components. The British still refer to these as a GR plug, after General Radio Corporation, the inventor (according to *The Audio Dictionary* by Glenn D. White).
- **binding posts** Alternate name for banana jacks above, derived from the capability to loosen (unscrew) the body and insert a wire through a hole provided in the electrical terminal and tighten the plastic housing down over the wire insulation, holding the wire in place.

- **BNC** A miniature bayonet locking connector for coaxial cable. Used to interconnect S/PDIF digital audio. See BNC for development and name history.

Connectors cont:

- **Cannon connector** or **Cannon plug** Alternate reference for XLR.
- **Elco connector** or **Elco plug** AVX manufactures several connectors used for interconnecting multiple audio channels at once, most often found in recording studios on analog and digital audio tape machines. One of these, a 90-pin version (Vari*con Series 8016), carries 28 shielded pairs of audio channels, allowing 3-wires per channel (positive, negative & shield) for a true balanced system interconnect.
- **Euroblocks** Shortened form of *European style terminal blocks*, a specialized disconnectable, or *plugable* terminal block consisting of two pieces. The receptacle is permanently mounted on the equipment and the plug is used to terminate both balanced and unbalanced audio connections using screw terminals. Differs from regular terminal strips in its plugability, allowing removal of the equipment by disconnecting the plug section rather than having to unscrew each wire terminal.
- **RCA** (aka *phono jack* or *pin jack*) The Radio Corporation of America (RCA) originally developed this type of unbalanced pin connector for internal chassis connections in radios and televisions during the '30s. It became popular for use in the cables that connected phonograph cartridges to preamplifiers because it was inexpensive and easily fitted to the rather small diameter shielded cables used for the cartridge leads (then they were mono cartridges so single conductor shielded cables were adequate -- *now you know.*). The standard connector used in line-level consumer and project studio sound equipment, and most recently to interconnect composite video signals. (excerpted from *Yamaha Sound Reinforcement Handbook*, pp. 297-298).
- **Speakon®** A registered trademark of Neutrik for their original design loudspeaker connector, now considered an industry de facto standard.
- terminal strips or terminal blocks Also called *barrier strips*, a type of wiring connector provided with screwdown posts separated by insulating barrier strips. Used for balanced and unbalanced wiring connections, where each wire is usually terminated with a crimped-on spade- or ring-connector and screwed in place; not disconnectable, or plugable. Has become known as the *U.S. style terminal blocks*. Contrast with Euroblocks.
- **1/4" TRS (*tip-ring-sleeve*)** 1. Stereo 1/4" connector consisting of tip (T), ring (R), and sleeve (S) sections, with T = left, R = right, and S = ground/shield. 2. Balanced interconnect with the positive & negative signal lines tied to T and R respectively and S acting as an overall shield. 3. Insert loop interconnect with T = send, R = return, and S = ground/shield. [*Think: ring, right, return*]

- **1/4" TS (*tip-sleeve*)** Mono 1/4" connector consisting of tip (T) [*signal*] and sleeve (S) [*ground & shield*] for unbalanced wiring.

Connectors cont:

- **XLR** 1. Originally a registered trademark of ITT-Cannon. The original model number series for Cannon's 3-pin circular connectors - invented by them - now an industry generic term. [*Ray A. Rayburn tells the whole story*: "At one time Cannon made a large circular connector series that was popular for microphones called the P series (now known as the EP series). Mics used the 3-pin P3 version. Some loudspeakers use the P4 or P8 versions of this connector to this day (*Neutrik Speakon NL4MPR 4-pole chassis mount and all Speakon 8-pole chassis mount connectors are made to fit the same mounting holes as the Cannon EP series*). In an attempt to make a smaller connector for the microphone market, Cannon came out with the UA series. These were "D" shaped instead of circular and were used on such mics as the Electro-Voice 666 and 654. There was a desire for a smaller yet connector. Someone pointed out the small circular Cannon X series. The problem with this was it had no latch. Cannon rearranged the pins and added a latch, and the XL (X series with Latch) was born. This is the connector others have copied. Later Cannon modified the female end only to put the contacts in a resilient rubber compound. They called this new version the XLR series. No other company has copied this feature."] 2. The standard connector for digital and analog balanced line interconnect between audio equipment.

constant-Q equalizer (also constant-bandwidth) Term applied to graphic and rotary equalizers describing bandwidth behavior as a function of boost/cut levels. Since Q and bandwidth are inverse sides of the same coin, the terms are interchangeable. The bandwidth remains constant for all boost/cut levels. For constant-Q designs, the skirts vary directly proportional to boost/cut amounts. Small boost/cut levels produce narrow skirts and large boost/cut levels produce wide skirts.

constant-voltage The common name given to the general practices begun in the 1920s and 1930s (becoming a U.S. standard in 1949) governing the interface between power amplifiers and loudspeakers used in *distributed sound systems*. Installations employing ceiling-mounted loudspeakers, such as offices, factories and schools are examples of distributed sound systems. The standard was derived from the need to minimize cost and to simplify the design of complex audio systems. One way to minimize cost is to minimize the use of copper, and one way to do that is to devise a scheme that allows the use of smaller gauge wire than normal 8 ohm loudspeakers require. Borrowing from the cross-country power distribution practices of the electric companies, this was done by using a transformer to step-up the amplifier's output voltage (with a corresponding decrease in output current); use this higher voltage to drive the (now smaller gauge due to smaller current) long lines to the loudspeakers; and then use another transformer to step-down the voltage at each loudspeaker. Clever. This scheme became known as the *constant-voltage* distribution method. The term "constant-voltage" is quite misleading

and causes much confusion until understood. Point 1: In electronics, two terms exist to describe two very different power sources: "constant-current" and "constant-voltage." Constant-current is a power source that supplies a fixed amount of current regardless of the load, so the output voltage varies, but the current remains constant. Constant-voltage is just the opposite. The voltage stays constant regardless of the load, so the output current varies but not the voltage. Applied to distributed sound systems, the term is used to describe the action of the system *at full power only*. This is the key point in understanding. *At full power the voltage on the system will not vary as a function of the number of loudspeakers driven*, that is, you may add or remove (subject to the maximum power limits) any number of loudspeakers and the voltage will remain the same, i.e., constant. Point 2: The other thing that is "constant" is the amplifier's output voltage at rated power -- and *it is the same voltage for all power ratings*. Several voltages are used, but the most common in the U.S. is 70.7 volts rms. The standard specifies that all power amplifiers put out 70.7 volts at their rated power. So, whether it is a 100 watt, or 500 watt or 10 watt power amplifier, the maximum output voltage of each must be the same (constant) value of 70.7 volts. This particular number came about from the second way this standard reduced costs: Back in the late '40s, UL safety code specified that all voltages above 100 volts peak created a "shock hazard," and subsequently *must be placed in conduit.* Expensive. Bad. So, working backward from a maximum of 100 volts peak (conduit not required), you get a maximum rms value of 70.7 volts (Vrms = 0.707 Vpeak). [Often "70.7 volts" is shortened to just "70 volts." It's sloppy; it's wrong; but it's common -- accept it.] In Europe, the most common level is 100 volts rms (although 50 V and 70.7 V are used too). This allows use of even smaller wire. Some large U.S. installations used as high as 210 volts rms, with wire runs of over one mile! Remember, the higher the voltage the lower the current, and consequently the smaller the cable and the longer the line can be driven without significant line loss. [The reduction in current exceeds the increase in impedance caused by the smaller wire because of the current-squared nature of power.] In some parts of the U.S., safety regulations regarding conduit use became stricter, forcing distributed systems to adopt a 25 volt rms standard. This still saves conduit, but adds a considerable increase in copper cost, so its use is restricted to small installations. Modern constant-voltage amplifiers either integrate the step-up transformer into the same chassis, or employ a high voltage design to directly drive the line without the need for the transformer. Similarly, constant-voltage loudspeakers have the step-down transformers built-in. Both 70.7 volt amplifiers and loudspeakers need only be rated in watts. An amplifier is rated for so many watts output at 70.7 volts, and a loudspeaker is rated for so many watts input (to give a certain SPL). Designing a system becomes a relatively simple matter of selecting speakers requiring so many watts to achieve the target SPL (quieter zones use lower wattage speakers, etc.), and then adding up the total to obtain the amplifier(s) power. For example, say you need (10) 25 watt, (5) 50 watt and (15) 10 watt loudspeakers, then you need at least 650 watts of amplifier power (actually you need about 1.5 times this due to real world losses, but that's another story).

crossover An electrical circuit (passive or active) consisting of a combination of high-pass, low-pass and bandpass filters used to divide the audio frequency spectrum (20 Hz - 20 kHz) into segments suitable for individual loudspeaker use. Since audio wavelengths vary from over 50 feet at the low frequency end, to less than one inch at the high frequency end, no single loudspeaker driver can reproduce the entire audio range. Therefore, at least two drivers are required, and more often three or more are used for optimum audio reproduction. Named from the fact that audio reproduction *transitions* (or *crosses over*) from one driver to the next as the signal increases in frequency. For example, consider a two driver loudspeaker crossed over at 800 Hz: Here only one driver (the *woofer* - "*woof*, *woof*" = low frequencies) works to reproduce everything below 800 Hz, while both drivers work reproducing the region immediately around 800 Hz (the *crossover region*), and finally, only the last driver (the *tweeter* - "*tweet, tweet*" = high frequencies) works to reproduce everything above 800 Hz. Crossover circuits are characterized by their *type* (Butterworth, Bessel and Linkwitz-Riley being the most popular), and by the steepness of their *roll-off slopes* (the rate of attenuation outside their passbands) as measured in decibels per interval, such as *dB/octave*, or sometimes *dB/decade* [useful rule-of-thumb: 6 dB/octave approximately equals 20 dB/decade].

crosstalk (signal) 1. Undesired capacitive, inductive, or conductive coupling from one circuit, part of a circuit, or channel, to another. 2. Any phenomenon by which a signal transmitted on one circuit or channel of a transmission system creates an undesired effect in another circuit or channel. Note: In telecommunications, crosstalk is usually distinguishable as speech or signaling tones

D:

damping factor Damping is a measure of a power amplifier's ability to control the back-emf motion of the loudspeaker cone after the signal disappears. *The damping factor of a system is the ratio of the loudspeaker's nominal impedance to the total impedance driving it*. Perhaps an example best illustrates this principle: let's say you have a speaker cabinet nominally rated at 8-ohms, and you are driving it with a Rane MA 6S power amp through 50 feet of 12 gauge cable. Checking the MA 6S data sheet (obtained off this web site, of course), you don't find its output impedance, but you do find that its damping factor is 300. What this means is that the ratio of a nominal 8 ohm loudspeaker to the MA 6S's output impedance is 300. Doing the math [8 divided by 300] comes up with an amazing .027 ohms. Pretty low. Looking up 12 gauge wire in your handy Belden Cable Catalog (... *then get one*.) tells you it has .001588 ohms per foot, which sure ain't much, but then again you've got 100 feet of it (that's right: 50 feet out and 50 feet back -- *don't be tricked*), so that's 0.159 ohms, which is six times as much impedance as your amplifier. (Now there's a lesson in itself -- *use big cable*.) Adding these together gives a total driving impedance of 0.186 ohms -- still pretty low -- yielding a very good damping factor of 43 (anything over 10 is enough, so you don't have to get extreme about wire size). [Note that the word is damp-*ing*, not damp-ning as is so often heard -- correct your friends; make enemies.]

decibel *Abbr.* **dB** Equal to one-tenth of a bel. [After **Alexander Graham Bell**.] 1. A measuring system first used in telephony (Martin, W.H., "DeciBel -- the new name for the transmission unit. *Bell System Tech. J.* January, 1929), where signal loss is a *logarithmic* function of the cable length. 2. The preferred method and term for representing the *ratio* of different audio levels. It is a mathematical shorthand that uses *logarithms* (a shortcut using the powers of 10 to represent the actual number) to reduce the size of the number. For example, instead of saying the dynamic range is 32,000 to 1, we say it is 90 dB [*the answer in dB equals 20 log x/y, where x and y are the different signal levels*]. Being a ratio, *decibels have no units*. Everything is relative. Since it is relative, then it must be relative to some *0 dB reference point*. To distinguish between reference points a suffix letter is added as follows [*The officially correct way per AES-R2, IEC 60027-3 & IEC 60268-2 documents is to enclose the reference value in parenthesis separated by a space from "dB"; however this never caught on, probably for brevity reasons if no other.*]:

0 dBu Preferred informal abbreviation for the official dB (0.775 V); a voltage reference point equal to 0.775 Vrms. [This reference originally was labeled dB*v* (lower-case) but was too often confused with dBV (upper-case), so it was changed to dBu (for unterminated).]

+4 dBu Standard pro audio voltage reference level equal to 1.23 Vrms.

0 dBV Preferred informal abbreviation for the official dB (1.0 V); a voltage reference point equal to 1.0 Vrms.

-10 dBV Standard voltage reference level for consumer and some pro audio use (e.g. TASCAM), equal to 0.316 Vrms. (Tip: RCA connectors are a good indicator of units operating at -10 dBV levels.)

0 dBm Preferred informal abbreviation of the official dB (mW); a *power* reference point equal to 1 milliwatt. To convert into an equivalent voltage level, *the impedance must be specified*. For example, 0 dBm into 600 ohms gives an equivalent voltage level of 0.775 V, or 0 dBu (see above); however, 0 dBm into 50 ohms, for instance, yields an equivalent voltage of 0.224 V -- something quite different. Since modern audio engineering is concerned with voltage levels, as opposed to power levels of yore, the convention of using a reference level of 0 dBm is obsolete. The reference levels of +4 dBu, or -10 dBV are the preferred units.

0 dBr An arbitrary reference level (r = *re*; or *reference*) that must be specified. For example, a signal-to-noise graph may be calibrated in dBr, where 0 dBr is specified to be equal to 1.23 Vrms (+4 dBu); commonly stated as "dB re +4," that is, "0 dBr is defined to be equal to +4 dBu."

0 dBFS A digital audio reference level equal to "Full Scale." Used in specifying A/D and D/A audio data converters. Full scale refers to the maximum *peak* voltage level possible before "digital clipping," or digital overload (see overs) of the data converter. The Full Scale value is fixed by the internal data converter design, and varies from model to model. [*According to standards people, there's supposed to be a space between "dB" and "FS" -- yeah, right, like that's gonna happen.*]

0 dBf Preferred informal abbreviation of the official dB (fW); a *power* reference point equal to 1 femtowatt, i.e., 10^{-15} watts.

0 dB-SPL The reference point for the threshold of hearing, equal to 20 microPA (micro Pascals rms).

Since 1 PA = 1 newton/m2 = .000145 PSI (pounds per square inch)

Then 0 dB-SPL = 2.9 nano PSI (rms) -- *an unbelievably small value.*

This means that since 1 atm = 14.7 PSI, it is equivalent to a loudness level of 194 dB-SPL! [*Thanks to Bob Pease for pointing out these enlightening facts!*]

dBA Unofficial but popular way of stating loudness measurements made using an A-weighting curve.

dBC Unofficial but popular way of stating loudness measurements made using an C-weighting curve.

delay 1. *Crossovers.* A signal processing device or circuit used to delay one or more of the output signals by a controllable amount. This feature is used to correct for loudspeaker drivers that are mounted such that their *points of apparent sound origin* (not necessarily their voice coils) are not physically aligned. Good delay circuits are frequency independent, meaning the specified delay is equal for all audio frequencies (*constant group delay*). Delay circuits based on digital sampling techniques are inherently frequency independent and thus preferred. 2. *MI*. Digital audio delay circuits comprise the heart of most all "effects" boxes sold in the MI world. Reverb, flanging, chorusing, phasers, echoing, looping, etc., all use delay in one form or another. 3. *Sound Reinforcement*. Acousticians and sound contractors use signal delay units to "aim" loudspeaker arrays. Introducing small amounts of delay between identical, closely-mounted drivers, fed from the same source, controls the direction of the combined response.

digital audio The use of sampling and quantization techniques to store or transmit audio information in binary form. The use of numbers (typically binary) to represent audio signals.

direct box Also known as a **DI box**, a phrase first coined by Franklin J. Miller, founder of Sescom, to describe a device that enables a musical instrument (guitar, etc.) to be connected *directly* to a mic- or line-level mixer input. The box provides the very high input impedance required by the instrument and puts out the correct level for the mixer.

directional microphone One whose response is more sensitive to sound arriving from one direction than another.

direct out Term for auxiliary outputs found on some mic preamps, mixing consoles, and teleconferencing equipment. Direct outputs are taken before any signal processing (other than normal mic preamp functions like gain, buffering, phantom power, bandlimiting filters, etc.), or mixing with other channels is done, hence, normally at line-level.

distortion *Audio distortion*: By its name you know it is a measure of unwanted signals. Distortion is the name given to anything that alters a pure input signal in any way other than changing its size. The most common forms of distortion are unwanted components or artifacts added to the original signal, including random and hum-related noise. Distortion measures a system's linearity - or nonlinearity, whichever way you want to look at it. Anything unwanted added to the input signal changes its shape (skews, flattens, spikes, alters symmetry or asymmetry, even if these changes are microscopic, they are there). A spectral analysis of the output shows these unwanted components. If a piece of gear is perfect, it does not add distortion of any sort. The spectrum of the output shows only the original signal - nothing else - no added components, no added noise - nothing but the original signal.

DSP (*digital signal processing*) A technology for signal processing that combines algorithms and fast number-crunching digital hardware, and is capable of high-performance and flexibility.

dynamics controllers (or **dynamics processors)** A class of signal processing devices used to alter an audio signal based solely upon its *frequency content* and *amplitude level*, thus the term "dynamics" since the processing is completely program dependent. The two most common dynamics effects are compressors and expanders, with limiters, noise gates (or just "gates"), duckers and levelers being subsets of these. Another dynamics controller category includes exciters, or enhancers. And noise reduction units fall into a final dynamics processor category.

dynamic microphone A microphone design where a wire coil (the *voice coil*) is attached to a small diaphragm such that sound pressure causes the coil to move in a magnetic field, thus creating an electrical voltage proportional to the sound pressure. Works in almost the exact opposite of a dynamic loudspeaker where an electrical voltage is applied to the voice coil attached to a large cone (diaphragm) causing it to move in a magnetic field, thus creating a change in the immediate sound pressure. In fact, under the right circumstances, both elements will operate as the other, i.e., a dynamic loudspeaker will act as a microphone and a dynamic microphone will act as a loudspeaker -- although not too well.

dynamic range The ratio of the loudest (undistorted) signal to that of the quietest (discernible) signal in a unit or system as expressed in decibels (dB). Dynamic range is another way of stating the *maximum S/N ratio.* With reference to signal processing equipment, the maximum output signal is restricted by the size of the power supplies, i.e., it cannot swing more voltage than is available. While the noise floor of the unit determines the minimum output signal, i.e., it cannot put out a discernible signal smaller than the noise. Professional-grade analog signal processing equipment can output maximum levels of +26 dBu, with the best noise floors being down around -94 dBu. This gives a maximum *dynamic range* of 120 dB - pretty impressive numbers, which coincide nicely with the 120 dB dynamic range of normal human hearing (from just audible to uncomfortably loud).

E:

ear *Anatomy.* The vertebrate organ of hearing, responsible for maintaining equilibrium as well as sensing sound and divided in mammals into the external ear, the middle ear, and the inner ear.

echo 1. *Acoustics* A discrete sound reflection arriving at least 50 milliseconds after the direct sound, and significantly louder than the background reverberant sound field. Contrast with reverberation. 2. *Psychoacoustics* A perceptually distinct copy of the original sound; a delayed duplicate. A single echo may be the result of multiple surface reflections. [Blesser]

Edison plug An ordinary household plug with two flat blades and a ground pin

effects loop A mixer term used to describe the signal path location where an external (*outboard*) signal processor is connected. The loop consists of an output *Send* jack connecting to the effects box *input*, and an input *Return* or *Receive* jack that comes from the effects box *output*. This is the preferred term when *two* separate 1/4", or other connectors are provided to patch in an outboard processor using separate cables for send and receive. These jacks are usually unbalanced, but could be balanced. A stereo effects loop requires four jacks.

electret microphone A microphone design similar to that of condenser mics except utilizing a permanent electrical charge, thus eliminating the need for an external polarizing voltage. This is done by using a material call an *electret* [acronym for *electr*icity + magn*et*] that holds a permanent charge (similar to a permanent magnet, i.e., a solid dielectric that exhibits persistent dielectric polarization). Because electret elements exhibit extremely high output impedance, they often employ an integral built-in impedance converter (usually a single JFET) that requires external power to operate. This low voltage power is often supplied single-ended over an unbalanced connection, or it may operate from standard phantom power. Electret technology was co-pioneered by Jim West and Gerhard Sessler in the 1960s at Bell Labs. Their original research into polymers (an electrical analogy of a permanent magnet) led to electret transducers.

EMI (*electromagnetic interference*) A measure of electromagnetic radiation from equipment

EQ (*equalizer*) A class of electronic filters designed to augment or adjust electronic or acoustic systems. Equalizers can be fixed or adjustable, active or passive. Indeed, in the early years of telephony and cinema, the first equalizers were fixed units designed to correct for losses in the transmission and recording of audio signals. Hence, the term *equalizer* described electronic circuits that corrected for these losses and made the output *equal* to the input. Equalizers commonly modify the frequency response of the signal passing through them; that is, they modify the amplitude versus frequency characteristics. There are also fixed equalizers that modify the phase response of the transmitted signals without disturbing the frequency content. These are referred to as all-pass, phase-delay, or signal-delay equalizers.

F:

fader A control used to *fade* out one input source and *fade* in another. The fading of a single source is called *attenuation* and uses an attenuator.

feedback suppressor An audio signal processing device that uses automatic detection to determine acoustic feedback frequencies and then positions notch filters to cancel the offending frequencies. Other methods use continuous frequency shifting (a very small amount) to prevent frequency build up and feedback before it happens.

FOH Abbreviation for *front of house*, used to describe the main mixer usually located in the audience for sound reinforcement systems. Meant to differentiate the main house mixer from the *monitor mixer* normally located to the side of the stage.

foldback The original term for *monitors*, or monitor loudspeakers, used by stage musicians to hear themselves and/or the rest of the band. The term "monitors" has replaced "foldback" in common practice

frequency 1. The property or condition of occurring at frequent intervals. 2. *Mathematics. Physics.* The number of times a specified phenomenon occurs within a specified interval, as: a. The number of repetitions of a complete sequence of values of a periodic function per unit variation of an independent variable. b. The number of complete cycles of a periodic process occurring per unit time. c. The number of repetitions per unit time of a complete waveform, as of an electric current. (AHD)

frequency response *Audio electronics.* It connotes **amplitude-frequency response** and quantifies a device's maximum and minimum frequency for full-output response. The electrical passband of an audio device. The measure of any audio device's ability to respond to a sine wave program, and therefore is a complex function measuring gain and phase shift (see phasor). It is used to express variation of gain, loss, amplification, or attenuation as a function of frequency, normally referred to a standard 1 kHz reference point.

FX unit Slang for "effects unit."

G:

gain The amount of amplification (*voltage*, *current* or *power*) of an audio signal, usually express in units of dB (i.e., the ratio of the output level to the input level). For example, amplifying a voltage signal by a factor of two is stated as a voltage gain increase of 6 dB. [*Historical Usage Note: originally the terms 'gain/loss' were restricted to power use only, and 'amplify/attenuate' were used for voltage and current -- although I can find no historical explanation for this arbitrary split, and no existing standards can be found that continue to make such a distinction. It is interesting to add that conformity with such a narrow definition of 'amplification' says that the original manufacturers misnamed their products: they should have been called a 'gainifier' -- not an 'amplifier.' According to the true believers a 'power amplifier' is a contradiction since you cannot 'amplify' power, only 'gainify' it. 'Power gainifier' is the correct term according to them.*]

graphic equalizer A multi-band variable equalizer using front panel mechanical slide controls as the amplitude adjustable elements. Named for the positions of the sliders "graphing" the resulting frequency response of the equalizer. Only found on active designs. Center frequency and bandwidth are fixed for each band.

ground *Electronics*. The common reference point for electrical circuits; the return path; the point of zero potential.

ground lift switch 1. Found on the rear of many pro audio products, used to separate (*lift*) the signal ground and the chassis ground connection. 2. Common three-pin to two-pin AC plug adapter used to reduce ground loops. [NOTE: This is unsafe and illegal. DO NOT USE.]

ground loop 1. *Electronics*. Within a single circuit, or an audio system, the condition resulting from multiple ground paths of different lengths and impedances producing voltage drops between paths or units. A voltage difference developed between separate grounding paths due to unequal impedance such that two "ground points" actually measure distinct and different voltage potentials relative to the power supply ground reference point. 2. *Aviation*. The tendency of a tailwheel aircraft (vs. tricycle gear) to pivot around its vertical axis during runway operations in the presence of a high crosswind.

groups (aka ***subgroup*** or ***submix*)** A combination of two or more signal channels gathered together and treated as a set that can be varied in overall level from a single control or set of controls. Mixing consoles often provide a group function mode, where the level of any group of incoming singles may be adjusted by a single slide fader, which is designated as the *group fader*. Likewise in certain signal processing equipment with splitting and routing capabilities, you will have the ability to group together, or assign, outputs allowing control of the overall level by a single external controller.

H:

harmonic 1. Any of a series of musical tones whose frequencies are integral multiples of the frequency of a fundamental tone. 2. A tone produced on a stringed instrument by lightly touching an open or stopped vibrating string at a given fraction of its length so that both segments vibrate. Also called **overtone**, **partial**, **partial tone**. A marching step of 15 inches (38 centimeters) at quick time and 18 inches (46 centimeters) at double time.

headphones An electromagnetic transducer usually based on the principle of electromagnetic induction used to convert the electrical energy output of a headphone amplifier into acoustic energy. Popular nickname is "**cans**."

headroom A term related to dynamic range, used to express in dB, the level between the *typical* operating level and the *maximum* operating level (onset of clipping). For example, a nominal +4 dBu system that clips at +20 dBu has 16 dB of headroom. Because it is a pure ratio, there are no units or reference-level associated with headroom -- just "dB." Therefore (and a point of confusion for many) headroom expressed in dB accurately refers to *both* voltage *and* power. Which means our example has 16 dB of *voltage* headroom, as well as 16 dB of *power* headroom. It's not obvious, but it's true. (The math is left to the reader.)

hertz *Abbr.* **Hz**. A unit of frequency equal to one cycle per second (cps). [After **Heinrich Rudolf Hertz**.]

high-cut filter also **hi-cut filter** See low-pass filter [*In audio electronics, we define things like this just to make sure you're paying attention.*] Contrast with high-*pass* filter below.

high impedance *Abbr.* **Hi-Z** *Electronics.* A device having an electrical impedance of at least 2,000 ohms. [*Note: This value is arbitrary as there is no standard defining exactly what constitutes a 'high impedance*.'] Examples include headphones rated 600 ohms and up (*headphone division between hi-Z and lo-Z is lower than other devices*); microphones rated 10k - 100k ohms; and most circuit *inputs* are high-impedance, rated at 2k-100k ohms. Contrast with: low impedance.

high-pass filter also **hi-pass filter** A filter having a passband extending from some finite cutoff frequency (not zero) up to infinite frequency. An infrasonic filter is a high-pass filter. Also known as a ***low-cut filter***.

hiss Random high frequency noise with a sibilant quality, most often associated with tape recordings.

Hi-Z See: high impedance.

house curve *Sound Reinforcement.* The name given to the weighting, or alteration, of the sound equalization for a room. It is a rule-of-thumb for what to do *after* achieving the flattest possible response. Different venues require different house curves with wide variation between many favorites. The most common one is for speech reinforcement in large auditoriums (*only*) and measures 10 dB down at 10 kHz with respect to 1 kHz (this is a 3 dB/octave slope). This contrasts to the 2 -3 dB used in many small control rooms. The proper choice is heavily dependent on the source material (speech vs. music) and the venue (large vs small; reverberant or dry); there is no one standard.

house mixer See: FOH

I:

IC (*integrated circuit*) A solid-state device with miniaturized discrete active components on a single semiconductor material, invented by Jack Kilby

IEC (*International Electrotechnical Commission*) A European organization (headquartered in Geneva, Switzerland) involved in international standardization within the electrical and electronics fields. The U.S. National Committee for the IEC operates within ANSI

impedance A measure of the complex resistive and reactive attributes of a component in an alternating-current (AC) circuit. Impedance is what restricts current flow in an AC electrical circuit; impedance is not relevant to DC circuits. In DC circuits, resistors limit current flow (because of their resistance). In AC circuits, inductors and capacitors similarly limit the AC current flow, but this is now because of their ***inductive*** or ***capacitive*** **reactance**. Impedance is like resistance but it is more. Impedance is the sum of a circuit, or device's resistance AND reactance. Reactance is measured in ohms (like resistance and impedance) but is frequency-dependent. Think of impedance as the complete or total current limiting ohms of the circuit -- the whole banana. Since AC circuits involve phase shift -- i.e., the voltage and current are rarely in phase due to the storage effects (*think time; it takes time to charge and discharge*) of capacitors and inductors, the reactance is termed "complex," that is there is a "real" part (resistive) and an "imaginary" part (bad terminology, but it means the phase shifting resistance part). To summarize: *resistance* has no phase shift; *reactance* (capacitors & inductors in AC circuits) includes phase shift; and *impedance*, is the sum of resistance and reactance. Just that simple.

in-phase In a synchronized or correlated way

input impedance *Electronics.* The input impedance of a device, usually high in the 2k - 100k ohm range. Input impedance can be frequency dependent and may vary with circuit feedback, therefore the value given should state the frequency range it covers.

insert loop The preferred term for a specialized I/O point found on mixers utilizing a single 1/4" TRS jack following the convention of *tip = send*, *ring = return*, & *sleeve = signal ground*. Used to patch in an outboard processor using only *one* cable, with unbalanced wiring. A stereo insert loop requires two jacks. Compare with effects loop.

I/O (*input/output*) Equipment, data, or connectors used to communicate from a circuit or system to other circuits or systems, or the outside world

isolation *Acoustics.* The isolation of sound is the process by which sound energy is contained or blocked as opposed to being converted into heat (see: absorption). For a good discussion of the differences read the excellent short article by Kurt Graffy, "More Or Less: The Difference Between Absorption And Isolation," *System Contractor News*, April 2003, p. 96, who also provides this wonderful quote attributed to Ted Schultz of Bolt, Beranek and Newman: "Mistaking sound absorption for sound isolation is like mistaking a diaper for an umbrella." [... *that clears up all confusion now doesn't it?*]

J:

jacket *Wire & Cable.* The insulating layer of material that surrounds a wire or cable offering protection; also called **sheath**

jacks and plugs Common name for audio connectors, where **jack = female** and **plug = male** is the standard convention for 1/4" and RCA -- ***only*** -- not followed for other types of connectors. If a connector is on the end of a cable -- XLR and others -- then either sex is a ***plug***.

K:

kHz (*kilohertz*) One thousand (1,000) cycles per second

kVA (*kilovoltamperes*) One thousand (1,000) voltamperes

L:

latency Used to describe the inherent delay in signal processing as well as software processing. The time it takes for a system or device to respond to an instruction, or the time it takes for a signal to pass through a device. It is how long it takes for a result to happen from a command. In telecommunications it is the length of time it takes packets to traverse the media.

lavaliere or **lavaliere microphone** A small electret microphone designed to be worn on a person. The first lavaliere mics were worn around the neck on a lanyard, hence the French name *lavallière, a type of necktie,* used to describe a pendant worn on a chain around the neck [after the Duchess de La Vallière who started the fashion (AHD)]. Today most lavaliere (the final "e" is commonly dropped) mics are attached by clips rather than hung from a cord.

LED (*light emitting diode*) Invented by Nick Holonyak, Jr. in 1962, a self-lighting semiconductor display of numerical or graphical information based on the light emitting characteristics of a solid-state device that emits incoherent (i.e., random direction) light when conducting a forward current.

levels Terms used to describe relative audio signal levels: (Also see decibel).

- **mic-level** Nominal signal coming directly from a microphone. Very low, in the microvolts, and requires a preamp with at least 60 dB gain before using with any *line-level* equipment.
- **line-level** Standard *+4 dBu* or *-10 dBV* audio levels. See decibel.
- **instrument-level** Nominal signal from musical instruments using electrical pick-ups. Varies widely, from very low *mic-levels* to quite large *line-levels*.

limiter A compressor with a fixed *ratio* of 10:1 or greater. The dynamic action effectively prevents the audio signal from becoming any larger than the *threshold setting*. For example, if the threshold is set for, say, +16 dBu and the input signal increases by 10 dB to +26 dB, the output only increases by 1 dB to +17 dBu, essentially remaining constant. Used primarily for preventing equipment, media, and transmitter overloads. A limiter is to a compressor what a noise gate is to an expander.

line driver A balanced output stage designed to interface and drive long lines. Long output lines tax output stages in terms of stability and current demands. Designs vary from direct-drive differential (sometimes using cross-coupled techniques) to transformer drive.

Linkwitz-Riley crossover The de facto standard for professional audio active crossovers is the 4th-order (24 dB/octave slopes) Linkwitz-Riley (LR-4) design. Consisting of cascaded 2nd-order Butterworth low-pass filters, the LR-4 represents a vast improvement over the previous 3rd-order (18 dB/octave) Butterworth standard. Named after S. Linkwitz, a Hewlett-Packard engineer at that time, who first described the problems and solution in his paper "Active Crossover Networks for Non-coincident Drivers," *J. Audio Eng. Soc.*, vol. 24, Jan/Feb 1976, pp. 2-8. In this paper, he credited his co-worker Russ Riley for the idea that cascaded Butterworth filters met all his crossover requirements. Their effort became known as the Linkwitz-Riley alignment. Linkwitz showed that a significant weakness of the Butterworth design was the behavior of the combined acoustic lobe along the vertical axis. An acoustic lobe results when both drivers operate together reproducing the crossover frequency band, and in the Butterworth case it exhibits severe peaking and is not on-axis (it tilts toward the lagging driver). Linkwitz showed that this results from the Butterworth outputs not being in-phase. Riley demonstrated an elegant solution by cascading two 2nd-order (any *even-ordered* pair works) Butterworth filters, which produced outputs that were always in-phase and summed to a constant-voltage response. Thus was created a better crossover.

lobing error *Electronic crossovers*. The amount of on-axis deviation in amplitude from zero (i.e., perfect combined radiation pattern) resulting from phase deviations at the crossover point. Term coined by Lipshitz (Lipshitz, Stanley P. and John Vanderkooy, "A Family of Linear-Phase Crossover Networks of High Slope Derived by Time Delay," *J. Audio Eng. Soc.*, Vol. 31, No. 1/2, January/February 1983, pp. 2-20).

loudness The SPL of a standard sound which appears to be as loud as the unknown. Loudness level is measured in phons and equals the equivalent SPL in dB of the standard. [For example, a sound judged as loud as a 40 dB-SPL 1 kHz tone has a loudness level of 40 phons. Also, it takes 10 phons (an increase of 10 dB-SPL) to be judged *twice* as loud.]

loudspeaker ***Dynamic.*** An electromagnetic transducer based on the principle of electromagnetic induction used to convert the electrical energy output of a power amplifier into acoustic energy. The heart of a dynamic loudspeaker is a coil of wire (the ***voice coil***), a magnet, and a cone. The amplifier applies voltage to the voice coil causing a current to flow that produces a magnetic field that reacts with the stationary magnet

making the cone move proportional to the applied audio signal.
Other loudspeaker technologies exist, among these are ***electrostatic*** (a thin sheet of plastic film suspended between two wire grids or screens; the film is conductive and charged with a high voltage; the film is alternately attracted to one grid and then the other resulting in motion that radiates sound), but for pro audio applications, dynamic loudspeakers dominate.

low impedance *Abbr.* **Lo-Z** *Electronics.* A device having an electrical impedance of at less than 1,000 ohms. [*Note: This value is arbitrary as there is no standard defining exactly what constitutes a 'low impedance*.'] Examples include loudspeakers in the 4-16 ohms range; headphones from 32-150 ohms; microphones rated 50-600 ohms; and electronic circuit *outputs* are low-impedance, rated at 50-300 ohms. Contrast with: high impedance.

low-pass filter also **lo-pass filter** A filter having a passband extending from DC (zero Hz) to some finite cutoff frequency (not infinite). A filter with a characteristic that allows all frequencies below a specified rolloff frequency to pass and attenuate all frequencies above. Anti-aliasing and anti-imaging filters are low-pass filters. Also known as a **high-cut filter**.

Lo-Z See: low impedance

M:

matrix-encoding *Audio.* A technique of storing more than two audio channels on a two-channel medium or transmission format. Dolby Surround is an example, where the center and surround channels are electronically encoded into the left and right channels of a stereo signal (usually by broadband 90° phase shifting and summing). On playback, the center and surround channel are decoded from the left and right signals. The problem inherent with matrix-encoding is the mathematical dilemma of trying to solve for four unknowns (*left, right, center & surround*) when you only have two equations (*the stereo signal*); you can get close but you cannot get the exact right answer (*so you always have crosstalk*). This contrasts with today's discrete digital channels.

matrix-mixer Similar to the matrix switcher (or router) below, but with additional signal processing features on all the inputs and outputs. With a matrix-mixer, not only can you assign any input to any output but you may add EQ, compression, change level, etc. Very elaborate models exist with as many as 32-channels in and 8 or more output channels .

mic splitter A phrase first coined by Franklin J. Miller, founder of Sescom, to describe a box fitted with female (inputs) and male (outputs) XLR mic connectors that allowed mic inputs to be routed to two, or more outputs. Usually passive, either hard-wired, or transformer connected. One common usage is for on-stage mic splitting, where one output goes to the monitor mixer and one to the FOH mixer.

microphone An electroacoustic transducer used to convert the input acoustic energy into an electrical energy output. Many methods exist; see, for example, electret microphone, condenser microphone, and dynamic microphone. The inventor of the carbon microphone is Emil Berliner.

MIDI (*musical instrument digital interface*) Industry standard bus and protocol for interconnection and control of musical instruments. First launched in 1983, now generalized and expanded to include signal processing and lighting control. See MMA [*Historical Note: MIDI began with Dave Smith, President of Sequential Circuits, who delivered an AES paper in the Fall of 1981* (The "USI," or Universal Synthesizer Interface by Dave Smith and Chet Wood, 70th AES Convention, 1981, preprint 1845). *He is credited with the original concept, but he admits that the initial idea of needing some sort of standard interface originated by both Oberheim and Roland.* Sources: *Polyphony*, February 1983, pp. 36-40 and an official IMA (International MIDI Association) publication, *Exploring MIDI* by David Droman in 1984.]

mixer At its simplest level, an audio device used to add (combine or sum) multiple inputs into one or two outputs, complete with level controls on all inputs. From here signal processing is added to each of the inputs and outputs until behemoth monsters with as many as 64 inputs are created -- at a cost of around 10-20 kilobucks per input for fully digitized and automated boards. At these price points a mixer becomes a *recording console*.

Monitor World *Live sound.* Area of the live sound stage where the monitor engineer mixes his/her magic and attempts to decipher cryptic hand signals from the performers. Not to be confused with "Guitar-Tech-Land", where all the babes hang out.

monitor mixer A mixer used to create the proper signals to drive the individual musician stage loudspeaker monitors. Also called *foldback speakers*. Compare: FOH

MOSFET (*metal-oxide semiconductor field-effect transistor*)

moving coil *Transducers.* A type of electromagnetic transducer that operates by having a mechanical device move a coil of wire in a magnetic filed to convert the mechanical movement into an electrical current. Contrast with moving magnet.

moving magnet *Transducers.* A type of electromagnetic transducer that operates by having a mechanical device move a magnet in a coil of wire to convert the mechanical movement into an electrical current. Contrast with moving coil.

mult *Recording.* Slang shortened form for "multiplex" or "multiple." Refers to routing or splitting signals to multiple destinations. Found on patchbays where several "mult" jacks make a signal available to many devices.

N:

narrow-band filter Term popularized by equalizer pioneer C.P. Boner to describe his patented (tapped toroidal inductor) passive notch filters. Boner's filters were very high Q (around 200) and extremely narrow (5 Hz at the -3 dB points). Boner used 100-150 of these sections in series to reduce feedback modes. Today's usage extends this terminology to include all filters narrower than 1/3-octave. This includes parametrics, notch filter sets, and certain cut-only variable equalizer

NC (*noise criterion*) curves A unit of measurement for the ambient or background noise level of occupied indoor spaces, i.e., a measure of its *noisiness* -- true story; real word. The measured noise spectrum (done in octave bands using an SPL meter) is compared against a series of standard noise criteria (NC) curves to determine the "NC level" of the space. The standard NC curves take into account the equal loudness contours of Fletcher-Munson to accurately reflect the listening experience. Each NC curve is assigned a number (in 5 dB increments) corresponding to the octave band SPL measured over the octave centered at approximately 1500 Hz. A space is then said to have a background noise level of "NC-20," for instance, which would be very quiet, comparable to a quality recording studio. Compare with RC rating.

near field or **near sound field** The sound field very close to the sound source, between the source and the far field. Technically, a distance less than one wavelength at the frequency of interest.

near-field monitor A loudspeaker used at a distance of 3-4 feet (1-1½ meters) in recording studios.

NEC (*National Electrical Code*) The name for the United States electrical safety standard (NFPA-70).

noise cancelling headphones Special headphones incorporating a microphone built into the headset that samples the ambient sound and adds it back out-of-phase to the headphone signal. This method actively cancels or nulls out background noise -- works best with high frequencies.

noise color People working in pro audio know the terms *white noise* and *pink noise*, but few recognize the terms "azure noise" or "red noise," but they are real terms. Noise that is not white is called *colored noise* and will have more energy at some frequencies than others, analogous to colored light.
White noise and pink noise are well defined and known; much less so are the others. White noise is so named because it is analogous to white light in that it contains all audible frequencies distributed uniformly throughout the spectrum. Passing white light through a prism (a form of filtering) breaks it down into a range of colors. Examination shows that red light is characterized by the longer wavelengths of light, i.e., the lower frequency region. Similarly, "pink noise" has higher energy in the low frequencies, hence the somewhat tongue-in-cheek term.
The Federal Standard 1037C *Telecommunications: Glossary of Telecommunication Terms* defines four noise colors (white, pink, blue & black) and is considered the official source. No official standard could be found for the others.
The following list of noise colors is loosely based on a rainbow-prism light analogy, where a prism creates a rainbow effect by separating white light passed through it into a visible spectrum labeled red, orange, yellow, green, blue, indigo, and violet from lowest to highest frequencies. Also shown is the approximate slope of the power density spectrum relative to white noise used as the reference:

- **red noise** also called **brown noise**: -6 dB/oct decreasing density (most amount of low frequency energy or power; used in oceanography; power proportional to 1/frequency-squared); *popcorn noise*.
- **pink noise**: -3 dB/oct decreasing noise density (but, equal power per octave; *1/f noise* or *flicker noise*; power proportional to 1/frequency).
- **white noise**: 0 dB/oct reference noise with equal power density (equal power per hertz; *Johnson noise*).
- **blue** (or **azure**) **noise**: +3 dB/oct increasing noise density (power proportional to frequency).
- **purple** (or **violet**) **noise**: +6 dB/oct increasing noise density (power proportional to frequency-squared; most amount of high frequency energy or power).
- **black noise**: silence (zero power density with a few random spikes allowed).

Other noise colors exist for specialized fields like video/photographic/image processing, communications, mathematical chaos theory, etc., but are not found in pro audio circles. Definitions for the noise colors **orange**, **green**, **gray**, and **brown** are found many times on the Web, but all appear to be from the same document.

noise floor Normally the lowest threshold of useful signal level, i.e., the residual noise with no signal present (although sometimes audible signals below the noise floor can be recovered).

noise gate An expander with a fixed "infinite" downward expansion ratio. Used extensively for controlling unwanted noise, such as preventing "open" microphones and "hot" instrument pick-ups from introducing extraneous sounds into the system. When the incoming audio signal drops below the user set-point (the *threshold* point) the expander prevents any further output by reducing the gain to "zero." The actual gain reduction is typically on the order of -80 dB, thus once audio falls below the threshold, effectively the output level becomes the residual noise of the gate. Common terminology refers to the gate "opening" and "closing." Another popular application uses noise gates to enhance musical instrument sounds, especially percussion instruments. Judicious setting of a noise gate's *attack* (turn-on) and *release* (turn-off) times adds "punch," or "tightens" the percussive sound, making it more pronounced. A noise gate is to an expander as a limiter is to a compressor.

notch filter A special type of cut-only equalizer used to attenuate (*only*, no boosting provisions exist) a narrow band of frequencies. Three controls: *frequency*, *bandwidth* and *depth*, determine the notch. Simplified units provide only a frequency control, with bandwidth and depth fixed internally. Used most often in acoustic feedback control to eliminate a small band of frequencies where the system wants to howl (feedback)

O:

octave 1. *Audio.* The interval between any two frequencies having a ratio of 2 to 1. 2. *Music* a. The interval of eight diatonic degrees between two tones, one of which has *twice* as many vibrations per second as the other. b. A tone that is eight full tones above or below another given tone. c. An organ stop that produces tones an octave above those usually produced by the keys played. (AHD)

off-axis response Any direction other than the **on-axis response**, i.e., the response measured along the imaginary straight line drawn through the geometric center of an object. In pro audio most often used in measurements of loudspeakers, microphones and humans.

ohm *Abbr.* **R** or Greek upper-case *omega*, **Ω** A unit of electrical resistance equal to that of a conductor in which a current of one ampere is produced by a potential of one volt across its terminals.

Ohm's Law The law stating that the direct current flowing in a conductor is directly proportional to the potential difference between its ends. It is usually formulated as V = IR, where V is the potential difference, or voltage, I is the current, and R is the resistance of the conductor.

omnidirectional microphone One with a response pattern that is as close to a perfect sphere as possible. Receives sound from all directions equally well.

one-third octave 1. Term referring to frequencies spaced every one-third of an octave apart. One-third of an octave represents a frequency 1.26-times above a reference, or 0.794-times below the same reference. The math goes like this: 1/3-octave = $2^{1/3}$ = 1.260; and the reciprocal, 1/1.260 = 0.794. Therefore, for example, a frequency 1/3-octave above a 1 kHz reference equals 1.26 kHz (which is rounded-off to the ANSI-ISO preferred frequency of "1.25 kHz" for equalizers and analyzers), while a frequency 1/3-octave below 1 kHz equals 794 Hz (labeled "800 Hz"). Mathematically it is significant to note that, to a very close degree, $2^{1/3}$ equals $10^{1/10}$ (1.2599 vs. 1.2589). This bit of natural niceness allows the *same frequency divisions* to be used to divide and mark an *octave into one-thirds* and a *decade into one-tenths*. 2. Term used to express the bandwidth of equalizers and other filters that are 1/3-octave wide at their -3 dB (half-power) points. 3. Approximates the smallest region (*bandwidth*) humans reliably detect change.

oscillator *Electronics & Synthesizers*. A circuit that continuously alternates between two (or more) states (IEEE); the period between alternations defines the frequency of oscillation. A common said complaint of electronic engineering students is that they build "amplifiers that oscillate and oscillators that amplify."

oscilloscope *Electronic Test Equipment.* An instrument primarily for making visible the instantaneous value of one or more rapidly varying electrical quantities (typically voltage) as a function of time or another electrical or mechanical quantity. (IEEE)

outboard unit *External*, usually referring to a separate piece of signal processing gear located remote to a mixer that connects in the effects loop.

out-of-phase In an un-synchronized or un-correlated way. See polarity and phase et al.

output impedance *Electronics.* The output driving impedance of a device, usually low in the 50-300 ohm range. Output impedance is frequency dependent and varies as a function of circuit feedback, therefore the value given must state the frequency range it covers.

overload light or **OL light** An indicator found on pro audio signal processing units that lights once the signal level exceeds a preset point. There is no standard specifying when an OL light should illuminate, although common practice makes it 3-4 dB below actual clipping. Good signal processing design ensures that the OL light illuminates anytime the signal exceeds the set point, anywhere in the signal path, not just the input or output level.

P:

P.A. (*public address*)

pan (*panoramic*) control A control found on mixers, used to "move," or *pan* the apparent position of a *single sound channel* between two outputs, usually "left," and "right," for stereo outputs. At one extreme of travel the sound source is heard from only one output; at the other extreme it is heard from the other output. In the middle, the sound is heard equally from each output, but is reduced in level by 3 dB relative to its original value. This guarantees that as the sound is panned from one side to the other, it maintains equal loudness (power) for all positions. Contrast with balance and crossfade controls.

parametric equalizer First named by George Massenburg, a multi-band variable equalizer offering control of all the "parameters" of the internal bandpass filter sections. These parameters being *amplitude*, *center frequency* and *bandwidth*. This allows the user not only to control the amplitude of each band, but also to shift the center frequency and to widen or narrow the affected area. Available with rotary and slide controls. Subcategories of parametric equalizers exist which allow control of center frequency but not bandwidth. For rotary control units the most used term is *quasi-parametric*. For units with slide controls the popular term is *paragraphic*. The frequency control may be continuously variable or switch selectable in steps. Cut-only parametric equalizers (with adjustable bandwidth or not) are called notch equalizers, or band-reject equalizers.

passive crossover A loudspeaker crossover not requiring a power supply for operation. Normally built into the loudspeaker cabinet. Passive crossovers do not require separate power amplifiers for each driver.

passive equalizer A variable equalizer requiring no power supply to operate. Consisting only of passive components (inductors, capacitors and resistors) passive equalizers have no AC line cord. Favored for their low noise performance (no active components to generate noise), high dynamic range (no active power supplies to limit voltage swing), extremely good reliability (passive components rarely break), and lack of RFI interference (no semiconductors to detect radio frequencies). Disliked for their cost (inductors are expensive), size (and bulky), weight (and heavy), hum susceptibility (and need careful shielding), and signal loss characteristic (passive equalizers always reduce the signal). Also inductors saturate easily with large low frequency signals, causing distortion. Rarely seen today, but historically they were used primarily for notching in permanent sound systems.

patchbay or **patch panel** A flat panel, or enclosure, usually rack-mounted, that contains at least two rows of 1/4" TRS connectors used to "patch in" or insert into the signal path a piece of external equipment (really dense configurations use 4.4 mm miniature or "bantam" jacks). The two rows consists of "send" (top row) and "receive" (bottom row) jacks wired for true balanced interconnection, i.e., tip = positive signal, ring = negative signal, sleeve = shield ground (*unbalanced patchbays exist but should not so no further discussion*). The two rows are tied together by shorting contacts such that the *normal* operation (hence, **"normaling" jacks**) is to short the send and receive tip-to-tip & ring-to-ring (the sleeves are always connected) maintaining the signal path until something is plugged in (or *jacked in* as cyberpunks love to say). Popular in recording studios where it is common to change the units in the signal path for each new session or client.

PFL (*pre-fade listen*) A term used on recording consoles and mixers, referring to a signal taken before the main channel fader. The significance is this signal is not affected by the fader position. Normally used to monitor (via headphones) to an individual input (or a small group of inputs) without affecting the main outputs, particularly useful in that it allows listening to an input with its fader all the way down (off). In broadcast this function is often called *cueing*, while recording or live-sound users may also refer to it as *soloing*.

phantom power The term given to the standardized scheme of providing power supply voltage to certain microphones using the same two lines as the balanced audio path. The international standard is IEC 60268-15, derived from the original German standard DIN 45 596. It specifies three DC voltage levels of 48 volts, 24 volts and 12 volts, delivered through 6.8 k ohms, 1.2 k ohms, and 680 ohms matched resistors respectively, capable of delivering 10-15 ma. The design calls for both signal conductors to have the same DC potential. This allows the use of microphone connections either for microphones without built-in preamps, such as dynamic types, or for microphones with built-in preamps such as condenser and electret types.
[*Phantom Power Mini-tutorial:* Much confusion surrounds phantom power. This is an area where you need to make informed decisions: Is it provided? Do you need it? Is it the

correct voltage, and does it source enough current for your microphone? There is a huge myth circulating that microphones sound better running from 48 volts, as opposed to, say, 12 volts, or that you can increase the dynamic range of a microphone by using higher phantom power. For the overwhelming majority of microphones both of these beliefs are false. Most condenser microphones require phantom power in the range of 12-48 VDC, with many extending the range to 9-52 VDC, leaving only a very few that actually require just 48 VDC. The reason is that internally most designs use some form of current source to drive a low voltage zener (usually 5 volts; sometimes higher) which determines the polarization voltage and powers the electronics. The significance is that neither runs off the raw phantom power, they both are powered from a fixed and regulated low voltage source inside the mic. Increasing the phantom power voltage is never seen by the microphone element or electronics, it only increases the voltage across the current source. But there are exceptions, so check the manufacturer, and don't make assumptions based on hearsay

phase Audio signals are complex AC (alternating current) periodic phenomena expressed mathematically as phasors, or vectors. *Phase* refers to a particular value of *t* (time) for any periodic function, i.e. it is the relationship between a reference point and the fractional part of the period through which the signal has advanced relative to an arbitrary origin. [*The origin is usually taken at the last previous passage through zero from the negative to the positive direction* -- IEEE.]

phase cancellation When two signals have the same exact time relationship to each other, they are said to be "in-phase;" if they do not, they are said to be "out-of-phase." (Compare with polarity) If two out-of-phase signals add together, since this is vector arithmetic (see phasor), they will, in fact, subtract from one another. This is called *phase cancellation*. Another type of phase cancellation occurs when water waves interact. One wave's energy becomes stronger when two waves collide in-phase (*summing*) and becomes weaker when they collide out-of-phase (*cancelling*).

phone jack Same as *¼" TRS*, see connectors

phono jack Same as *RCA*, see connectors

pickup *Transducers.* A device that "picks up" sound and converts it into an electrical signal. Many technologies exist, from the most popular electromagnetic models (**magnetic pickups**) found on electric guitars to piezoelectric models seen on acoustic instruments and used in early phonograph cartridges (soon replaced with electromagnetic models using either moving magnet or moving coil technologies).

piezo *Transducers.* Greek, *to press tight, squeeze*. Shortened form for **piezoelectric,** the name given to a class of materials (***dielectric crystals***) that produce electricity or become polarized when mechanically strained or stressed. In pro audio used to create pickups, microphones and loudspeakers or buzzers, and in digital circuits quartz crystals for stable timing references.

pink noise Pink noise is a random noise source characterized by a flat amplitude response per octave band of frequency (or any *constant percentage* bandwidth), i.e., it has equal energy, or constant power, per octave. Passing white noise through a filter having a 3 dB/octave roll-off rate creates pink noise. See white noise discussion for details. Due to this roll-off, pink noise sounds less bright and richer in low frequencies than white noise. Since pink noise has the same energy in each 1/3-octave band, it is the preferred sound source for many acoustical measurements due to the critical band concept of human hearing. The name comes from the filtering of white noise. White noise is analogous to white light in that it contains all audible frequencies distributed uniformly throughout the spectrum. Passing white light through a prism (a form of filter) breaks it down into a range of colors. Examination shows that red light is characterized by the longer wavelengths of light, i.e., light in the lower frequency region. Similarly, pink noise has higher energy in the low frequencies, hence the somewhat tongue-in-cheek term *pink*.

pitch Frequency or tone of a sound

polarity A signal's electromechanical potential with respect to a reference potential. For example, if a loudspeaker cone moves *forward* when a *positive* voltage is applied between its red and black terminals, then it is said to have a *positive polarity*. A microphone has *positive polarity* if a positive pressure on its diaphragm results in a positive output voltage. [Usage Note: polarity vs. phase shift: *polarity* refers to a signal's *reference* NOT to its *phase shift*. Being 180° *out-of-phase* and having *inverse polarity* are DIFFERENT things. We wrongly say something is *out-of-phase* when we mean it is *inverted*. One takes *time*; the other does not.]

pot (*lowercase*) Shorten form of potentiometer

potentiometer A three-terminal variable resistor. Two terminals connect to the ends of the resistor, while the third terminal is attached to a movable device that makes contact with the resistive element. The movable terminal, or *slider*, is capable of being positioned from one end of the element to the other. Many physical arrangements exist, with the rotary design being the most common, followed by linear motion (used in graphic equalizers, for example), all the way to tiny SMT devices. Often used as voltage dividers in electronic circuits, the input voltage is applied to the top of the resistive element, while the other end is tied to ground or a common reference and the output is taken from the slider. When the slider is positioned to the top extreme, the output equals the input, or the entire voltage; moving it to the bottom extreme gives an output of zero volts; and every possible level between is available as the slider is moved from one end to the other. The most common application uses this arrangement to control the volume of an audio device. In this manner the voltage, or electrical potential is varied, hence, a *potentiometer*. The *taper* of the pot controls the rate at which the voltage changes as the slider is moved. The taper defines the amount of resistive change as a function of travel. Several popular examples follow:

audio taper (aka ***A-taper***): Usually 15% resistance at the 50% rotation point.
linear taper (aka ***B-taper***): Always 50% resistance at the 50% travel point.
log taper (aka ***D-taper***): Often used as an audio taper since its 50% rotation point has 10% resistance.
MN taper (aka ***balance pot***) Special taper developed for home stereo "Balance" controls. Consists of two sections (one for each channel) operating opposite each other. Exactly one-half of each section is a zero resistance surface (i.e., solid-copper or equivalent), the next 50% of travel is linear taper. Therefore for one channel rotating the slider through the first 50% of travel does not change the level at all, while the other channel is reduce from full to zero, and vice versa, with the middle position (usually featuring a center-detent) always passing full signal to each channel.

power 1. *Electricity* a. The product of applied voltage (potential difference) and current in a direct-current circuit (or the voltage squared divided by the resistance, or the current squared times the resistance). b. The product of the effective values of the voltage and current with the cosine of the phase angle (between current and voltage) in an alternating-current circuit. See: apparent power and rms power 2. *Physics* The rate at which work is done, expressed as the amount of work per unit time, and measured in units such as the watt (1 joule per second, which equals the power dissipated (as heat) by 1 ohm of resistance when 1 ampere of current passes through it) and horsepower (equal to 745.7 watts). (AHD)

power amplifier See: amplifier

pressure gradient microphone *Microphone.* If both the front and rear of a diaphragm are exposed to a sound field, then the force that vibrates the diaphragm results from the difference between the sound pressures in front and to the rear of the diaphragm (called the pressure gradient). The magnitude of the driving force depends on the distance between the front and rear sound entries, the frequency, and the angle of incidence and is therefore a directional variable which can be utilized to design directional microphones. Cardioid, figure 8, or hypercardioid polar patterns can be achieved by incorporating appropriate sound paths

proportional-Q graphic equalizer (also **variable-Q**) Term applied to graphic and rotary equalizers describing bandwidth behavior as a function of boost/cut levels. The term "proportional-Q" is preferred as being more accurate and less ambiguous than "variable-Q." If nothing else, "variable-Q" suggests the unit allows the user to vary (set) the Q, when no such controls exist. The bandwidth varies inversely proportional to boost (or cut) amounts, being very wide for small boost/cut levels and becoming very narrow for large boost/cut levels. The skirts, however, remain constant for all boost/cut levels.

PZM (***pressure zone microphone***) Patented by Ed Long & Ron Wickersham in 1982, a technique and design where the microphone is mounted on a flat plate which acts as a reflective surface directing sound into the mic capsule. The PZM principle uses the compression and decompression of air between the plate and the membrane in parallel with the plate (the gap is very narrow, typically only a millimeter or less. This arrangement gives about 6 dB extra amplification of the signal, which means 6 dB less inherent electronic noise.

Q:

Q (upper-case) Quality factor. *Filters.* The selectivity factor defined to be the ratio of the center frequency *f* divided by the bandwidth

quarter-inch jack Same as ¼" TRS or ¼" TS, see connectors.

R:

ack unit See "U"

RCA jack See connectors

real-time analyzer See RTA

resistance See impedance

resonance 1. *Electronics.* In an LRC circuit, it is the condition where the ***inductive*** and ***capacitive*** **reactances** are equal; this is called the resonant frequency. 2. *Physics.* The increase in amplitude of oscillation of an electric or mechanical system exposed to a periodic force whose frequency is equal or very close to the natural undamped frequency of the system. (AHD) A dynamic condition which occurs when any input frequency of vibration coincides with one of the natural frequencies of the structure. That is, the inclination of any mechanical or electrical system to vibrate (*resonate*) at a certain frequency when excited by an external force, and to keep vibrating after the excitation is removed. 3. *Acoustics.* Intensification and prolongation of sound, especially of a musical tone, produced by sympathetic vibration. 4. *Linguistics.* Intensification of vocal tones during articulation, as by the air cavities of the mouth and nasal passages.

reverb *Recording.* Shortened form of ***reverberator*, or *reverberation unit***. Any electronic or acoustical device designed to simulate, or capture, the natural reverberation of a large hard-surfaced (*echoic*) room, and mix it back with the original recorded sound. Reverb today is accomplished by digital devices using complex DSP algorithms; previously done using a chamber, a plate, or springs.

reverberation The total sound field remaining in a room after the original source is silenced. The length of time of this collapsing sound field is called the reverberation time and is defined below. Contrast with echo and ambience. "Reverberation represents the energy decay process after the initial echoes" [Blesser].

reverberation time also **RT60** *Reverberation* is all sound remaining after the source stops. The time it takes for this sound to decay is called the *reverberation time*, and it is quantified by measuring how long it takes the sound pressure level to decay to one-millionth of its original value. Since one-millionth equals a 60 dB reduction, reverberation time is abbreviated "RT60."

RF (*radio frequency*) *Broadcast.* General term for radio waves.

RFI (*radio frequency interference*) A measure of radio frequency (RF) radiation from equipment. An RF disturbance is an electromagnetic disturbance having components in the RF range.

RFID (*radio frequency identification*) Technology using RF signals to ID individuals. It uses silicon chip tags with radio frequency functions and on-board memory that holds unique ID numbers, allowing it to ID and track just about anything.

rms See: root mean square

rms power *No such thing.* A misnomer, or application of a wrong name. There is no such thing as "rms power." *Average* or *apparent* power is calculated using rms values but that does not equal "rms power;" it equals continuous sine wave power output into a resistive load.

root mean square *Abbr.* **rms** (lowercase) *Mathematics.* The square root of the average of the squares of a group of numbers. (AHD) A useful and more meaningful way of averaging a group of numbers.

RTA (*real-time analyzer*) A constant percentage bandwidth spectrum analyzer.

S:

self-noise *Microphones*. Residual noise, or the inherent noise level of a microphone when no signal is present. Microphone inherent self-noise is usually specified as the equivalent SPL level which would give the same output voltage, with typical values being 15-20 dB SPL.

sensitivity 1. *Audio electromechanics.* The standard way to rate audio devices like microphones, headphones and loudspeakers. A standard input value is applied and the resultant output is measured and stated.

- **loudspeaker sensitivity**: the standard is to apply one watt and measure the sound pressure level (SPL) at a distance of one meter. [IEC 60268-5]
- **headphone sensitivity**: the standard is to apply one milliwatt and then measure the sound pressure level at the earpiece (using a dummy head with built-in microphones). [IEC 60268-7] See RaneNote: *Understanding Headphone Power Requirements*.
- **microphone sensitivity**: the standard is to apply a 1 kHz sound source equal to 94 dB SPL (one *pascal*) and then measure the output level and express it in mV/PA (millivolts per pascal). [IEC 60268-4] 2. *Audio electronics*. The minimum input signal required to produce a standard output level.
- **power amplifier sensitivity**: The input level required to produce one watt output into a specified load impedance, usually 4 or 8 ohms. [EIA-490]
- **radio receiver sensitivity**: The input level required to produce a specified signal-to-noise ratio.

sine *Abbr.* **sin** *Mathematics.* 1. The ordinate of the endpoint of an arc of a unit circle centered at the origin of a Cartesian coordinate system, the arc being of length *x* and measured counterclockwise from the point (1, 0) if *x* is positive or clockwise if *x* is negative. 2. In a right triangle, the ratio of the length of the side opposite an acute angle to the length of the hypotenuse. (AHD)

sine curve *Mathematics.* The graph of the equation $y = \sin x$. Also called sinusoid. (AHD)

sine wave *Physics.* A waveform with deviation that can be graphically expressed as the sine curve. (AHD)

slapback See slap echo below.

slap echo also called **slapback** 1. *Acoustics*. A single echo resulting from parallel non-absorbing (i.e., reflective) walls, characterized by lots of high frequency content. So-called because you can test for slap echo by sharply clapping your hands and listening for the characteristic sound of the echo in the mid-range. Slap echo smears a stereo sound field by destroying the critical phase relationships necessary to form an accurate sound stage. 2. *Recording*. Devices that simulate slap echo are popular in recording. One distinct repeat echo is added to an instrument sound resulting in a very live sound similar to what you would hear in an auditorium.

slew rate 1. The term used to define the maximum rate of change of an amplifier's output voltage with respect to its input voltage. In essence, *slew rate* is a measure of an amplifier's ability to follow its input signal. It is measured by applying a large amplitude *step function* (a signal starting at 0 volts and "instantaneously" jumping to some large level [without overshoot or ringing], creating a step-like look on an oscilloscope) to the amplifier under test and measuring the slope of the output waveform. For a "perfect" step input (i.e., one with a rise time at least 100 times faster than the amplifier under test), the output will not be vertical; it will exhibit a pronounced slope. The slope is caused by the amplifier having a finite amount of current available to charge and discharge its internal compensation capacitor. 2. *Mathematics.* Slew rate is defined to be the maximum derivative of the output voltage with respect to time. That is, it is a measure of the worst case delta change of voltage over a delta change in time, or the rate-of-change of the voltage vs. time. For sinusoidal signals (*audio*), this equals 2 pi times the maximum frequency, times the maximum peak output voltage: SR = (2 pi) (Fmax) (Vpeak).

S/N or **SNR (*signal-to-noise ratio*)** An audio measurement of the residual noise of a unit, stated as the ratio of signal level (or power) to noise level (or power), normally expressed in decibels. The "signal" reference level must be stated. Typically this is either the expected nominal operating level, say, +4 dBu for professional audio, or the maximum output level, usually around +20 dBu. The noise is measured using a true rms type voltmeter over a *specified bandwidth*, and sometimes using weighting filters. All these thing must be stated for a S/N spec to have meaning. Simply saying a unit has a SNR of 90 dB means nothing, without giving the reference level, measurement bandwidth, and any weighting filers. A system's *maximum* S/N is called the dynamic range. See RaneNote: *Audio Specifications*.

snake or **audio snake** *Live Sound.* The nickname for the cable running from the stage of a live performance to the main mixing console, which is usually set-up in the middle or rear of the audience (in spite of being called FOH). It typically contains one shielded pair (STP) of wires for each of the stage microphones. The name comes from the multiconductor cable looking sort of snake-like.

sound 1.a. Vibrations transmitted through an elastic material or a solid, liquid, or gas, with frequencies in the approximate range of 20 to 20,000 hertz, capable of being detected by human ears. Sound (in air) at a particular point is a rapid variation in the air pressure around a steady-state value (atmospheric pressure) - that is, sound is a *disturbance* in the surrounding medium. b. Transmitted vibrations of any frequency. c. The sensation stimulated in the ears by such vibrations in the air or other medium. d. Such sensations considered as a group. 2. Auditory material that is recorded, as for a movie. 3. Meaningless noise. 4. *Music.* A distinctive style, as of an orchestra or a singer. (AHD)

sound pressure The value of the rapid variation in air pressure due to a sound wave, measured in pascals, microbars, or dynes - all used interchangeable, but *pascals* is now the preferred term. *Instantaneous* sound pressure is the peak value of the air pressure, often used in noise control measurements. *Effective* sound pressure is the rms value of the instantaneous sound pressure taken at a point over a period of time.

sound pressure level or **SPL** 1. The rms sound pressure expressed in dB re 20 microPa (the lowest threshold of hearing for 1 kHz). [As points of reference, *0 dB-SPL* equals the threshold of hearing, while *140 dB-SPL* equals irreparable hearing damage.] See: inverse square law 2. **Blue whales**, the largest living animals, also make the loudest sounds by any living source. Their low-frequency pulses have been measured at 188 dB-SPL and detected 530 miles away according to The Guinness Book of World Records®.

Speakon® See connectors

spectrum analyzer *Audio Test Equipment.* A type of electronic measurement device used to display the amplitude/frequency components of a continuous signal, as opposed to the amplitude/time domain oscilloscope. The formal IEEE definitions are "(1) An instrument generally used to display the power distribution of an incoming signal as a function of frequency. (2) An instrument that measures the power of a complex signal in many bands. The frequency bands can be either constant absolute bandwidth (e.g., FFT analyzer), or constant percentage bandwidth (e.g., RTA analyzer)."

SPL See: sound pressure level

splitter An audio device used to divide one input signal into two or more outputs. Typically this type of unit has one input with 6-16 (or more) outputs, each with a level control and often is unbalanced.

SR (*sound reinforcement*)

square wave A periodic waveform characterized by a 50% duty cycle and a Fourier series consisting of odd-ordered, equal phase, sinusoidal harmonic components of its fundamental frequency with amplitudes (coefficients multiplying the magnitude of the fundamental sine wave) equal to $1/n$, where n equals the harmonic number. Therefore the first few harmonic amplitudes are 1/3, 1/5, 1/7, 1/9, etc.

stereo or **stereophonic sound** 1. "The word *stereophonics* was derived by combining two Greek words: *stereo*, which means solid and implicates the three spatial dimensions (depth, breadth, and height), and *phonics*, which means the science of sound. Thus, stereophonics denotes the science of 3-dimensional sound" [Streicher & Everest]. 2. Term applied to any system of recording (or transmission) using multiple microphones for capturing and multiple loudspeakers for reproduction the sound. *Stereo* as the term has become popularly used restricts the number of playback loudspeakers to two, but strictly speaking the term can apply to any number of loudspeakers. Although stereo was first demonstrated at the Paris Opera in 1881 (*really*) using carbon microphones and earphones, it would not become widespread until the work of Blumlein in the 1930s. Also see William B. Snow.

subgroups See: groups

submix See: groups

subwoofer A large woofer loudspeaker designed to reproduce audio's very bottom-end, i.e., approximately the last one or two octaves, from 20 Hz to 80-100 Hz. [*Actually misnamed since subsonic means slower than audio, while infrasonic means lower than audio, it should be called an "infrawoofer."*]

T:

talkback 1. A recording console feature where a microphone mounted on the console allows the engineer to speak with the musicians during sessions -- a very useful feature when the console is located in a soundproof control room, or out in the audience for sound reinforcement systems.

THD (*third-harmonic distortion*) See third-harmonic distortion

THD (*total harmonic distortion*) A measurement technique rarely used, but often confused with the THD+N technique described below. Many people mistakenly refer to a "THD" measurement when they really mean the "THD+N" technique. [*For completeness and the abnormally curious:* a true THD measurement consists of a computation from a series of individual harmonic amplitude measurements, rather than a single measurement. "THD" is the square root of the sum of the squares of the individual harmonic amplitudes. And the answer must specify the highest order harmonic included in the computations; for example, "THD through 8th harmonic." (from Metzler)]

THD+N (*total harmonic distortion plus noise*) The most common audio measurement. A single sine wave frequency of known harmonic purity is passed through the unit under test, and then patched back into the distortion measuring instrument. A measurement level is set; the instrument notches out the frequency used for the test, and passes the result through a set of band-limiting filters, adjusted for the bandwidth of interest (usually 20-20 kHz). What remains is noise (including any AC line [mains] *hum* or interference *buzzes*, etc.) and all harmonics generated by the unit. This composite signal is measured using a true rms detector voltmeter, and the results displayed. Often a resultant curve is created by stepping through each frequency from 20 Hz to 20 kHz, at some specified level (often +4 dBu), and bandwidth (usually 20 kHz; sometimes 80 kHz, which allows measurement of any 20 kHz early harmonics). [Note that the often-seen statement: "THD+N is x%," is meaningless. For a THD+N spec to be complete, it must state the *frequency*, *level*, and measurement *bandwidth*.] While THD+N is the most common audio test measurement, it is not the most useful indicator of a unit's performance. What it tells the user about *hum, noise and interference* is useful; however that information is better conveyed by the *signal-to-noise (S/N) ratio* specification. What it tells the user about harmonic distortion is not terribly relevant simply because it *is harmonically related to the fundamental*, thus the distortion products tend to get masked by the complex audio material. The various *intermodulation (IM) distortion* tests are better indicators of sonic purity.

third-harmonic distortion The standard test used on analog magnetic tape recorders to determine the **maximum output level (MOL)**, which was defined to occur at the magnetization level at which a recorded 1 kHz sine wave reached "3 percent third-harmonic distortion." Of course, third-harmonic distortion is nothing more than a measurement of the amplitude of the third harmonic of the input frequency and is the most prominent distortion component in analog magnetic recording systems. The third-harmonic level was used as a convenient figure-of-merit because the 2nd harmonic is difficult to hear, since it tends to reinforce the pitch of the fundamental. The 3rd harmonic is easy to detect on pure tones (although less so on music), thus it makes a good benchmark for comparing sound "off tape" with the original. The distorted tone has an edge to it, containing a component one octave and a quint (interval of a fifth in music) above the fundamental. For this reason the third-harmonic is also called a musical twelfth. Here's the interesting twist. This test was commonly abbreviated and listed on the specification sheet as "THD". Which, of course, was mistaken to mean "total harmonic distortion" instead of "third harmonic distortion." This led to it being mistakenly shortened to just "distortion," so you still find old analog tape date sheets, and many text books defining MOL as the point at which there exist "3% distortion," instead of the correct reference to "3% third-harmonic distortion" -- quite different things.

third-octave Term referring to frequencies spaced every *three* octaves apart. For example, the third-octave above 1 kHz is 8 kHz. Commonly misused to mean one-third octave. While it can be argued that "third" can also mean one of three equal parts, and as such might be used to correctly describe one part of an octave split into three equal parts, it is potentially too confusing. The preferred term is one-third octave.

tone 1. *Music.* a. A sound of distinct pitch, quality, and duration; a note. b. The interval of a major second in the diatonic scale; a whole step. c. A recitational melody in a Gregorian chant. 2.a. The quality or character of sound. b. The characteristic quality or timbre of a particular instrument or voice. (AHD)

tone controls The term most often referring to a two-band shelving equalizer offering amplitude control only over the highest (*treble*, from music, meaning the highest part, voice, instrument, or range) frequencies, and the lowest (*bass*, from music, meaning the lowest musical part) frequencies. Sometimes a third band is provided for boost/cut control of the midband frequencies.

total harmonic distortion See THD and THD+N

transformer *Electronics.* A passive component that uses electromagnetic induction to increase or decrease alternating electric energy (voltage and current), usually consisting of two wirewound coils (windings) inductively coupled. A **step-up transformer** raises voltage and a **step-down transformer** lowers voltage. See Rod Elliott's articles for a thorough and clear exposition of transformer details (*highly recommended*).

transient response The reaction of an electronic circuit, or electromechanical device, or acoustic space to a non-repetitive stimulus such as a step or impulse response. It is the result to a sudden change in the input that is nonperiodic. For example, percussive instruments produce primarily transient sounds. The transient stimulus and resulting response are characterized by the amplitude and the rise time (and fall time if it is an impulse), overshoot, and settling time. The standard reference is to note the maximum amplitude and the time required to reach within 10% of the steady-state value. For a real world example of the comparative transient responses for a full-range and a 3-way loudspeaker system.

triamp, **triamplified**, or **triamplification** Term used to refer to a 3-way active crossover where the audio signal is split into three paths, and using separate power amplifier channels for each driver.

U:

"U" Abbreviation for the "modular unit" on which rack panel heights are based. Per the EIA and ANSI standard *ANSI/EIA-310-D-1992 Cabinets, Racks, Panels, and Associated Equipment*, the modular unit is equal to 44.45 millimeters (1.75"). Panel heights are referred to as "*nU*" where *n* is equal to the number of modular units. Examples are 1U (1.75" high), 2U (3.5" high), 3U (5.25" high), etc. Popularly called ***rack units*** and often abbreviated "RU," which is technically incorrect but not misleading.

unbalanced line See: balanced line

unidirectional microphone or just **directional microphone** One that is most sensitive to sound arriving directly at its front. Compare: omnidirectional mic and cardioid microphone

unity gain A gain setting of one, or a device having a gain of one, i.e., it does not amplify or attenuate the audio signal. The output equals the input.

V:

vacuum tube An electron tube where virtually all the air has been removed (creating a *vacuum*), thus permitting electrons to move freely, with low interaction with any remaining air molecules. (AHD) The first tube was a two-element diode, invented and patented by Ambrose Fleming in 1904, based on the Edison effect. Three years later, in 1907, Lee de Forest developed the first triode (known as the *Audion*) by adding a grid between the cathode (*emitter*) and the anode (*collector*), thus creating the first amplifier since a change of voltage at the grid produced a corresponding (but greater) change of voltage at the anode

variable-Q graphic equalizer See: proportional-Q graphic equalizer.

VCA (*voltage-controlled amplifier* or ***voltage-controlled attenuator*)** An electronic circuit comprised of three terminals: input, output and control. The output voltage is a function of the input voltage and the control port. The gain of the stage is determined by the control signal, which is usually a DC voltage, but could be a current signal or even a digital code. Usually found as the main element in dynamic controllers, such as compressors, expanders, limiters, and gates.

VU meter (*volume unit*) The term *volume unit* was adopted to refer to a special meter whose response closely related to the perceived loudness of the audio signal. It is a voltmeter with standardized dB calibration for measuring audio signal levels, and with attack and overshoot (needle ballistics) optimized for broadcast and sound recording. Jointly developed by Bell Labs, CBS and NBC, and put into use in May, 1939, VU meter characteristics are defined by ANSI specification "Volume Measurements of Electrical

Speech and Program waves, " C16.5-1942 (which is know incorporated into IEC 60268-17). 0 VU is defined to be a level of +4 dBu for an applied sine wave. The VU meter has relatively slow response. It is driven from a full-wave averaging circuit defined to reach 99% full-scale deflection in 300 ms and overshoot not less than 1% and not more than 1.5%. Since a VU meter is optimized for perceived loudness it is not a good indicator of peak performance.

W:

W Abbreviation for watt.

watt *Abbr.* **W** *Electricity* An International System unit of power equal to one joule per second. [After **James Watt**.]

wavelength *Symbol* **λ** (Greek lower-case *lambda*) The distance between one peak or crest of a sine wave and the next corresponding peak or crest. The *wavelength* of any frequency may be found by dividing the speed of sound by the frequency.

weighting filters Special filters used in measuring loudness levels, and consequently carried over into audio noise measurements of equipment. The filter design "weights" or gives more attention to certain frequency bands than others. The goal is to obtain measurements that correlate well with the subjective perception of noise. [Technically termed *psophometric* (pronounced "so-fo-metric") filters, after the *psophometer*, a device used to measure noise in telephone circuits, broadcast, and other audio communication equipment. A psophometer was a voltmeter with a set of weighting filters.] Weighting filters are a special type of band-limiting filters designed to compliment the way we hear. Since the ear's loudness vs. frequency response is not flat, it is argued, we should not try to correlate flat frequency vs. loudness measurements with what we hear. Fair enough. Five weighting filter designs dominate (See: References: Metzler):

- **A-weighting** (not official but commonly written as **dBA**) The A-curve is a wide bandpass filter centered at 2.5 kHz, with ~20 dB attenuation at 100 Hz, and ~10 dB attenuation at 20 kHz, therefore it tends to heavily roll-off the low end, with a more modest effect on high frequencies. It is the inverse of the 30-phon (or 30 dB-SPL) equal-loudness curve of Fletcher-Munson. [*Editorial Note:* Low-cost audio equipment often list an A-weighted noise spec -- not because it correlates well with our hearing -- but because it helps "hide" nasty low-frequency hum components that make for bad noise specs. Sometimes A-weighting can "improve" a noise spec by 10 dB. Words to the wise: always wonder what a manufacturer is hiding when they use A-weighting.]
- **C-weighting** (not official but commonly written as **dBC**) The C-curve is "flat," but with limited bandwidth, with -3 dB corners of 31.5 Hz and 8 kHz, respectively.

- **ITU-R 468-weighting** (*was CCIR, but since the CCIR became the ITU-R, the correct terminology today is ITU-R*) This filter was designed to maximize its response to the types of impulsive noise often coupled into audio cables as they pass through telephone switching facilities. Additionally it turned out to correlate particularly well with noise perception, since modern research has shown that frequencies between 1 kHz and 9 kHz are more "annoying" than indicated by A-weighting curve testing. The ITU-R 468-curve peaks at 6.3 kHz, where it has 12 dB of gain (relative to 1 kHz). From here, it gently rolls off low frequencies at a 6 dB/octave rate, but it quickly attenuates high frequencies at ~30 dB/octave (it is down -22.5 dB at 20 kHz, relative to +12 dB at 6.3 kHz).
- **ITU-R (CCIR) ARM-weighting** or **ITU-R (CCIR) 2 kHz-weighting** This curves derives from the ITU-R 468-curve above. Dolby Laboratories proposed using an average-response meter with the ITU-R 468-curve instead of the costly true quasi-peak meters used by the Europeans in specifying their equipment. They further proposed shifting the 0-dB reference point from 1 kHz to 2 kHz (in essence, sliding the curve down 6 dB). This became known as the ITU-R ARM (*average response meter*), as well as the ITU-R 2 kHz-weighting curve. (See: R. Dolby, D. Robinson, and K. Gundry, "A Practical Noise Measurement Method," *J. Audio Eng. Soc.*, Vol. 27, No. 3, 1979) **[*Before using these terms be aware that the ITU-R, even after 20 years, takes strong exception to having its name used by a private company to promote its own methodologies.*]**

- **Z-weighting** A new term defined in IEC 61672-1, the latest international standard for sound pressure level measurements. It stand for *zero-weighting*, or no weighting; i.e., a flat measurement with equal emphasis of all frequencies.

white noise 1. *Physics.* Analogous to white *light* containing equal amounts of all *visible* frequencies, white *noise* contains equal amounts of all *audible* frequencies (technically the bandwidth of noise is infinite, but for audio purposes it is limited to just the audio frequencies). From an *energy* standpoint white noise has constant power *per hertz* (also referred to as *unit bandwidth*), i.e., at every frequency there is the same amount of power (while pink noise, for instance, has constant power *per octave band* of frequency). A plot of white noise power vs. frequency is flat if the measuring device uses the same width filter for all measurements. This is known as a ***fixed bandwidth*** filter. For instance, a fixed bandwidth of 5 Hz is common, i.e., the test equipment measures the amplitude at each frequency using a filter that is 5 Hz wide. It is 5 Hz wide when measuring 50 Hz or 2 kHz or 9.4 kHz, etc. A plot of white noise power vs. frequency change is *not* flat if the measuring device uses a variable width filter. This is known as a ***fixed percentage bandwidth*** filter. A common example of which is 1/3-octave wide, which equals a bandwidth of 23%. This means that for every frequency measured the bandwidth of the measuring filter changes to 23% of that new center frequency. For example the measuring bandwidth at 100 Hz is 23 Hz wide, then changes to 230 Hz wide when measuring 1 kHz, and so on. Therefore the plot of noise power vs. frequency is not flat, but shows a 3 dB rise in amplitude per octave of frequency change. Due to this rising frequency characteristic, white noise sounds very bright and lacking in low frequencies. [Here's the technical details: noise *power* is actually its *power density spectrum* - a measure of how the noise power contributed by individual frequency components is distributed over the frequency spectrum. It should be measured in *watts/Hz*; however it isn't. The accepted practice in noise theory is to use *amplitude-squared* as the unit of power (purists justify this by assuming a one-ohm resistor load). For electrical signals this gives units of *volts-squared/Hz*, or more commonly expressed as *volts/root-Hertz*. Note that the denominator gets bigger by the *square root of the increase in frequency.* Therefore, for an octave increase (doubling) of frequency, the denominator increases by the square root of two, which equals 1.414, or 3 dB. In order for the energy to remain constant (as it must if it is to remain white noise) there has to be an offsetting increase in amplitude (the numerator term) of 3 dB to exactly cancel the 3 dB increase in the denominator term. Thus the upward 3 dB/octave sloping characteristic of white noise amplitude when measured in constant percentage increments like 1/3-octave.] See noise color. 2. *Music.* Slang term for music that is discordant with no melody; disagreeable, harsh or dissonant

wire See cables

wiring classes *U.S. National Electrical Code (NEC)* defines three classes of wiring according to their fire and shock hazard potential:

- **Class 1** Where *both* fire and shock hazards exist, i.e., the wiring can deliver enough current for a fire hazard and enough voltage for a shock hazard. The most common example is AC power running to equipment. This class requires prevention of all touching and barriers against fire.
- **Class 2** Where *neither* fire or shock hazard exists, i.e. the wiring cannot deliver enough current (internal limiting) for a fire hazard and not enough voltage (less than 120 Vrms) for a shock hazard. Examples here are all normal audio interconnect plus most power amplifier output wiring.
- **Class 3** Where there is not a fire hazard *but there is a shock hazard*, i.e., the wiring cannot deliver enough current (internal limiting) for a fire hazard, but can deliver enough voltage (120 - 300 Vrms) for a shock hazard. Requires touch-proof terminals; seen in audio for very high-output power amplifiers.

X:

XLR See connectors

X-Y microphone technique A stereo recording technique where two cardioid microphones are placed facing each other, at an angle of 90 degrees, with the center of the source aimed at the center between them. Sometimes this technique is incorporated internally in a single microphone using two capsules. Also called the ***coincident-microphone technique*** Compare with ORTF

Y:

Y-connector or **Y-cord** A three-wire circuit that is star connected. Also spelled *wye-connector*. It is okay to use a Y-connector to *split* an audio signal from an output to drive two inputs; it is not okay to use a Y-connector to try and *sum* or *mix* two signals together to drive one input.

Z:

Z The electronic symbol for impedance