Rane Professional Audio Reference

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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*. 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.

AC-3 (*audio coding 3*) Dolby's digital audio data compression algorithm adopted for HDTV transmission and used in DVDs, laserdiscs and CDs for 5.1 multichannel home theater use. See: Dolby Digital. Competes with DTS Consumer. The terms *AC-1* and *AC-2* are other versions developed by Dolby for different applications.

Academy curve The name of the standard mono optical track that has been around since the beginning of sound for film. Standardized in 1938, it has improved (very) slightly over the years. Also known as the *N* (normal) *curve* the response is flat 100 Hz-1.6 kHz, and is down 7 dB at 40 Hz, 10 dB at 5 kHz and 18 dB at 8 kHz. This drastic "dumping" of the high-end was to hide the high-frequency "frying" and "crackling" noise inherent in early film sound production. Compare with X curve

Accelerated-Slope™ A trademark of Rane Corporation used to describe their family of patented tone control technologies that produce steeper slopes than normal, thus allowing boost/cut of high and low frequencies without disturbing the critical midband frequencies.

accommodation The most misspelled word in American writing (*two "c"s and two "m"s*).

accordion "An instrument in harmony with the sentiments of an assassin." -- Ambrose Bierce.

acoustic echo canceller See: echo canceller

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, and so on. 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*.

acoustic lobe See: Linkwitz-Riley crossover

acoustic treatments There are only three classic (physical) tools available for the acoustician to treat a room: absorbers, reflectors and diffusers. Absorbers attenuated sound; reflectors redirect sound, and diffusers (hopefully) uniformly distribute sound. Or put another way, these tools change the temporal, spectra and spatial qualities of the sound. Additionally, with today's advanced digital audio tools, all of these elements can be electronically manipulated.

acquisition time The time required for a sample-and-hold (S/H) circuit to capture an input analog value; specifically, the time for the S/H output to approximately equal its input.

acronym A *word* formed from the first letters of a name, such as *laser* for *l*ight *a*mplification by *s*timulated *e*mission of *r*adiation, or by combining initial letters or parts of a series of words, such as *radar* for *ra*dio *d*etecting *a*nd *r*anging. The requirement of forming a *word* is what distinguishes an acronym from an *abbreviation* (or *initialism* as the Canadian academicians say). Thus *modem* [*mo*dulator-*dem*odulator] is an acronym, and *AES* [*A*udio *E*ngineering *S*ociety] is an abbreviation or initialism. [Unsubstantiated rumor has it that the word "acronym" itself is an *acronym*, created from the phrase "*a*bbreviating by *c*ropping *r*emainders *o*ff *n*ames to *y*ield *m*eaning" -- but it has never been confirmed.] (Thanks MR.)

active crossover A loudspeaker crossover requiring power 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. A *mono 3-way* unit is a single channel device with three outputs, labeled *Low*, *Mid* and *High*. 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. See: passive crossover

active equalizer A variable equalizer requiring power 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.

ActiveX A Microsoft developed software technology released in 1996. ActiveX, formerly called OLE (Object Linking and Embedding), is loosely based on the Component Object Model (COM), but provides substantially different services to developers. An ActiveX *component* is a unit of executable code (such as an *.exe* file) that follows the Active X specification for providing *objects*. This technology allows programmers to assemble reusable software components into applications and services. However, component software development using ActiveX technology should not be confused with Object-Oriented Programming (OOP). OOP is concerned with *creating objects*, while ActiveX is concerned with *making objects work together*. Simply stated, ActiveX is a technology that lets a program (the ActiveX component or control) interact with other programs over a network (e.g., the Internet), regardless of the language in which they were written. ActiveX components can do similar things as Java beans, but they are quite different. Java is a programming language, while ActiveX controls can be written in any language (e.g., Visual Basic, C, C++, even Java), Also ActiveX runs in a variety of applications, while Java beans usually run only in Web browsers. ActiveX controls are of concern to the pro audio community, because this is the technology that allows designers of computer-controlled sound systems to create common front-end software control panels that will operate different manufacturer's units, without having to know anything about their internal code or algorithms. Each ActiveX control is made up of *properties*, values associated with the control which might include such things as level settings and meter readings, and *events*, which tell the computer something significant has happened, such as a switch closer or clip detection. ActiveX allows the manufacturer to create an object that fully describes a device, while hiding the implementation details, such as protocol from the programmer. By hiding the communication details, there is no longer a need for different manufacturer's devices *to agree on protocol.* This lack of a protocol standard means that cooperation between manufactures is not required. It allows each manufacturer to choose the best protocol for their devices.

adaptive delta modulation (*ADM*) A variation of delta modulation in which the step size may vary from sample to sample.

ADAT (*Alesis Digital Audio Tape*) Digital tape recording system developed by Alesis, and since licensed to Fostex & Panasonic, putting 8-tracks of 16-bit, 44.1kHz digital audio on S-VHS tape.

ADAT ODI (*optical digital interface*) See **ADAT Optical**.

ADAT Optical Alesis's proprietary multichannel optical (fiber optic) digital interface specification for their family of ADAT modular digital multitrack recorders. This standard describes transmission of 8-channels of digital audio data through a single fiber optic cable.

ADC (or **A/D, *analog-to-digital converter*)** The electronic component which converts the instantaneous value of an analog input signal to a digital word (represented as a binary number) for digital signal processing. The ADC is the first link in the digital chain of signal processing. See data converter bits

ADPCM (*adaptive differential pulse code modulation*) A very fast data compression algorithm based on the differences occurring between two samples.

ADR (*automatic dialog replacement*) Film postproduction term used to indicate the act and location where dialogue that is not taped during production or that needs to be redone is recorded and synchronized to the picture. Usually the name of the room where this occurs, containing a studio with a screen, TV monitors, microphones, control area, console and loudspeakers.

AES (*Audio Engineering Society*) Founded in 1948, the largest professional organization for electronic engineers and all others actively involved in audio engineering. Primarily concerned with education and standardization.

AES24 A developing AES standard for sound systems using computer networks to control audio equipment. Formerly called "SC-10" (after the working group's subcommittee number), the title for AES24-1-1999 (the first part to be published) is *Application Protocol for Controlling and Monitoring Audio Devices via Digital Data Networks -- Part 1: Principles, Formats, and Basic Procedures*. The complete standard is broken down into several parts issued separately. The second part, in the *proposed draft* stage, is titled -- *Part 2: Data Types, Constants, and Class Structure*. The remaining two parts are in process.

AES3 interface (*The interface formerly known as AES/EBU*. Use of *AES3 interface* clarifies that the interface includes the isolation required by AES3 but not required by the EBU specification.) The serial transmission format standardized for professional digital audio signals (AES3-1992 *AES Recommended Practice for Digital Audio Engineering - Serial Transmission Format for Two-Channel Linearly Represented Digital Audio Data*) A specification using time division multiplex for data, and balanced line drivers to transmit two channels of digital audio data on a single twisted-pair cable using 3-pin (XLR) connectors. Issued as ANSI S4.40-1985 by the American National Standards Institute. In addition, information document AES-3id is available describing the transmission of AES3 formatted data by unbalanced coaxial cable. Transmission by fiber optic cable is under discussion. The consumer version is referred to as S/PDIF.

AES/EBU interface See AES3

AFL Abbreviation for *after fade listen*, a term used on recording consoles and mixers, referring to a signal taken after the main channel fader; hence this sampling point tracks the main fader level. Also referred to as *post fade solo*, but since PFL already meant *pre* fade, AFL was adopted to prevent confusion. Got it? Compare with PFL.

algorithm A structured set of instructions and operations tailored to accomplish a signal processing task. For example, a fast Fourier transform (FFT), or a finite impulse response (FIR) filter are common DSP algorithms.

aliasing The problem of unwanted frequencies created when sampling a signal of a frequency higher than half the sampling rate. See: Nyquist frequency.

all-pass filter A filter that provides only phase shift or phase delay without appreciably changing the magnitude characteristic.

Ambisonics A British-developed surround sound system designed to reproduce a true three-dimensional sound field. Based on the late Michael Gerzon's (1945-1996) (Oxford University) famous theoretical foundations, Ambisonics delivers what the ill-fated *quadraphonics* of the '70s promised but could not. Requiring two or more transmission channels (encoded inputs) and four or more decoded output loudspeakers, it is not a simple system; nor is the problem of reproducing 3-dimensional sound. Yet with only an encoded stereo input pair and just four decoded reproducing channels, Ambisonics accurately reproduces a complete 360-degree *horizontal* sound field around the listener. With the addition of more input channels and more reproducing loudspeakers, it can develop a true spherical listening shell. As good as it is, a mass market for Ambisonics has never developed due to several factors. First, the actual recording requires a special tetrahedron array of four microphones: three to measure left-right, front-back and up-down sound pressure levels, while the fourth measures the overall pressure level. All these microphones must occupy the same point in space as much as possible. So far, only one manufacturer (first Calrec, bought by AMS, bought by Siemens, sold, now Soundfield Research) is known to make such an array. Next, a professional Ambisonics encoding unit is required to matrix these four mic signals together to form two or more channels before mastering or broadcast begins. Finally, the consumer must have an Ambisonics decoder, in addition to at least four channels of playback equipment.

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**.]

Ampère, **André Marie** (1775-1836) French physicist and mathematician who formulated Ampère's law, a mathematical description of the magnetic field produced by a current-carrying conductor.

amplifier classes Audio power amplifiers are classified according to the relationship between the output voltage swing and the input voltage swing; thus it is primarily the design of the output stage that defines each class. *Classification is based on the amount of time the output devices operate during one complete cycle of signal swing.* This is also defined in terms of output bias current [the amount of current flowing in the output devices with no applied signal]. For discussion purposes (with the exception of class A), assume a simple output stage consisting of two complementary devices (one positive polarity and one negative polarity) using tubes (valves) or any type of transistor (bipolar, MOSFET, JFET, IGFET, IGBT, etc.).

- **Class A** operation is where both devices conduct continuously for the entire cycle of signal swing, or the bias current flows in the output devices at all times. The key ingredient of class A operation is that both devices are always on. There is no condition where one or the other is turned off. Because of this, class A amplifiers in reality are not complementary designs. They are single-ended designs with only one type polarity output devices. They may have "bottom side" transistors but these are operated as fixed current sources, not amplifying devices. Consequently class A is the most inefficient of all power amplifier designs, averaging only around 20% (meaning you draw about 5 times as much power from the source as you deliver to the load.) Thus class A amplifiers are large, heavy and run very hot. All this is due to the amplifier constantly operating at full power. The positive effect of all this is that class A designs are inherently the most linear, with the least amount of distortion. [Much mystique and confusion surrounds the term *class A*. Many mistakenly think it means circuitry comprised of discrete components (as opposed to integrated circuits). Such is not the case. A great many integrated circuits incorporate class A designs, while just as many discrete component circuits do not use class A designs.]
- **Class B** operation is the opposite of class A. Both output devices are never allowed to be on at the same time, or the bias is set so that current flow in a specific output device is zero when not stimulated with an input signal, i.e., the current in a specific output flows for one half cycle. Thus each output device is on for exactly one half of a complete sinusoidal signal cycle. Due to this operation, class B designs show high efficiency but poor linearity around the crossover region. This is due to the time it takes to turn one device off and the other device on, which translates into extreme crossover distortion. Thus restricting class B designs to power consumption critical applications, e.g., battery operated equipment, such as 2-way radio and other communications audio.
- **Class AB** operation is the intermediate case. Here both devices are allowed to be on at the same time (like in class A), but just barely. The output bias is set so that current flows in a specific output device appreciably more than a half cycle but less than the entire cycle. That is, only a small amount of current is allowed to flow through both devices, unlike the complete load current of class A designs, but enough to keep each device operating so they respond instantly to input voltage demand s. Thus the inherent non-linearity of class B designs is eliminated, without the gross inefficiencies of the class A design. It is this combination of good efficiency (around 50%) with excellent linearity that makes class AB the most popular audio amplifier design.
- **Class AB plus B** design involves two pairs of output devices: one pair operates class AB while the other (slave) pair operates class B.
- **Class C** use is restricted to the broadcast industry for radio frequency (RF) transmission. Its operation is characterized by turning on one device at a time for less than one half cycle. In essence, each output device is pulsed-on for some percentage of the half cycle, instead of operating continuously for the entire half cycle. This makes for an extremely efficient design capable of enormous output power. It is the magic of RF tuned circuits (flywheel effect) that overcomes the distortion create d by class C pulsed operation.
- **Class D** operation is switching, hence the term *switching power amplifier*. Here the output devices are rapidly switched on and off at least twice for each cycle (Sampling Theorem). Theoretically since the output devices are either completely on or completely off they do not dissipate any power. If a device is on there is a large amount of current flowing through it, but all the voltage is across the load, so the power dissipated by the device is zero (found by multiplying the voltage across the device [*zero*] times the current flowing through the device [*big*], so 0 x big = 0); and when the device is off, the voltage is large, but the current is zero so you get the same answer. Consequently class D operation is theoretically 100% efficient, but this requires zero on-impedance switches with infinitely fast switching times -- a product we're still waiting for; meanwhile designs do exist with true efficiencies approaching 90%. [Historical note: the original use of the term "Class D" referred to switching amplifiers that employed a resonant circuit at the output to remove the harmonics of the switching frequency. Today's use is much closer to the original "Class S" designs.
- **Class E** operation involves amplifiers designed for rectangular input pulses, not sinusoidal audio waveforms. The output load is a tuned circuit, with the output voltage resembling a damped single pulse. Normally Class E employs a single transistor driven to act as a switch.

The following terms, while generally agreed upon, are not considered "official" classifications

- **Class F** Also known by such terms as "biharmonic," "polyharmonic," "Class DC," "single-ended Class D," "High-efficiency Class C," and "multiresonator." Another example of a tuned power amplifier, whereby the load is a tuned resonant circuit. One of the differences here is the circuit is tuned for one or more harmonic frequencies as well as the carrier frequency. See References: Krauss, et al. for complete details.
- **Class G** operation involves changing the power supply voltage from a lower level to a higher level when larger output swings are required. There have been several ways to do this. The simplest involves a single class AB output stage that is connected to two power supply rails by a diode, or a transistor switch. The design is such that for most musical program material, the output stage is connected to the lower supply voltage, and automatically switches to the higher rails for large signal peaks [thus the nickname *rail-switcher*]. Another approach uses two class AB output stages, each connected to a different power supply voltage, with the magnitude of the input signal determining the signal path. Using two power supplies improves efficiency enough to allow significantly more power for a given size and weight. Class G is becoming common for pro audio designs. [Historical note: Hitachi is credited with pioneering class G designs with their 1977 *Dynaharmony HMA 8300* power amplifier.]
- **Class H** operation takes the class G design one step further and actually modulates the higher power supply voltage by the input signal. This allows the power supply to track the audio input and provide just enough voltage for optimum operation of the output devices [thus the nickname *rail-tracker* or *tracking power amplifier*]. The efficiency of class H is comparable to class G designs. [Historical note: Soundcraftsmen is credited with pioneering class H designs with their 1977 *Vari-proportional MA5002* power amplifier.]
- **Class S** First invented in 1932, this technique is used for both amplification and amplitude modulation. Similar to Class D except the rectangular PWM voltage waveform is applied to a low-pass filter that allows only the slowly varying dc or average voltage component to appear across the load. Essentially this is what is termed "Class D" today. See References: Krauss for details.

amplitude 1. Greatness of size; magnitude. 2. *Physics.* The maximum absolute value of a periodically varying quantity. 3. *Mathematics.* a. The maximum absolute value of a periodic curve measured along its vertical axis. b. The angle made with the positive horizontal axis by the vector representation of a complex number. 4. *Electronics.* The maximum absolute value reached by a voltage or current waveform.

analog A real world physical quantity or data characterized by being continuously variable (rather than making discrete jumps), and can be as precise as the available measuring technique.

anechoic Literally, *without echo*, used to describe specially designed rooms, *anechoic chambers*, built to emulate a free sound field, by absorbing practically all the sound field.

ANSI (pronounced "ann-see") **(*American National Standards Institute*)** A private organization that develops and publishes standards for voluntary use in the U.S.A.

anti-aliasing filter A low-pass filter used at the input of digital audio converters to attenuate frequencies above the half-sampling frequency to prevent aliasing.

anti-imaging filter A low-pass filter used at the output of digital audio converters to attenuate frequencies above the half-sampling frequency to eliminate image spectra present at multiples of the sampling frequency.

APA (*Audio Publishers Association*) The online resource center designed for audiobook listeners and industry professionals.

apparent power The result of multiplying the rms value of the voltage by the rms value of the current in an electronic circuit. It is expressed in watts (W) for resistive loads and in voltamperes (VA) for reactive loads. It's the amount of power the casual observer *thinks* is available (hence, *apparent*), but because of *power factor* may not be -- the real power is usually less. See power factor .

ASA (*Acoustical Society of America*) Founded in 1929, the oldest organization for scientist and professional acousticians and others engaged in acoustical design, research and education.

ASCII (pronounced "ask-ee") **(*American Standard Code for Information Interchange*)** An ANSI standard data transmission code consisting of seven information bits, used to code 128 letters, numbers, and special characters. Many systems now use an 8-bit binary code, called ASCII-8, in which 256 symbols are represented (for example, IBM's "extended ASCII").

ASIC (*application-specific integrated circuit*) A large-scale integrated circuit whose function is determined by the final mask layer for a particular application or group of applications; for example, an IC that does all the functions of a modem.

ASIO (*audio stream input/output*) A multichannel audio transfer protocol developed by Steinberg North America in 1997, for audio/MIDI sequencing applications, allowing access to the multichannel capabilities of sound cards.

ASPEC (*adaptive spectral perceptual entropy coding*) A bit rate reduction standard for high quality audio. Jointly developed by AT&T Bell Labs, Thomson, the Fraunhofer Society and CNET. Characterized by high degrees of compression to allow audio transmission on ISDN.

asymmetrical (non-reciprocal) response Term used to describe the comparative shapes of the boost/cut curves for variable equalizers. The cut curves do not mirror the boost curves, but instead are quite narrow, intended to act as notch filters.

asynchronous A transmission process where the signal is transmitted without any fixed timing relationship between one word and the next (and the timing relationship is recovered from the data stream).

ATM (*asynchronous transfer mode*) networking An extremely fast networking technology already found on many disk editors (*Avid, Sonic Solutions, Studio Audio, etc.*) and predicted to infiltrate homes within the coming decade. ATM specifies the protocol (i.e., the order and sequence) of the digital information on the network, but not the physical means of transmission (e.g., fiber optic, twisted-pair, etc.). The protocol controls how the entire network is run and maintained.

atmospheric pressure Pressure caused by the weight of the atmosphere. At sea level it has a mean value of one atmosphere but reduces with increasing altitude.

attenuator pad *Electronics*. A passive network that reduces the voltage (or power) level of a signal with negligible distortion, but with insertion loss. Often a purely resistive network, although any combination of inductors, resistors and capacitors are possible, a pad may also provide impedance matching. Pads are referred to by the topology of the network formed, with the two most common being an *L-pad* and a *T-pad*:

- **L-pad** A two-leg network shaped like an inverted, backward letter "L". It usually consists of two resistors that are fixed or adjustable. A true variable L-pad consists of two variable potentiometers that are ganged (tied) together. The ganged sections work to provide either a constant input or a constant output impedance regardless of the attenuation setting. Since modern analog audio electronic circuits consist of stages characterized by very high input and very low output impedances, the term is now broaden to include all L-shaped networks without the requirement of providing constant impedance to the source or load. Volume and level controls are common examples.
- **T-pad** A three-leg network shaped like the letter "T". It usually consists of three resistors that are fixed or adjustable. A true variable T-pad consists of two or three variable potentiometers that are ganged (tied) together. The ganged sections work to provide either a constant input or a constant output impedance regardless of the attenuation setting. Since modern analog audio electronic circuits consist of stages characterized by very high input and very low output impedances, the term is now broaden to include all T-shaped networks without the requirement of providing constant impedance to the source or load.

audio 1. Of or relating to humanly audible sound, i.e., audio is all the sounds that humans hear. 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.]

audio books See: *Pro Audio Reference Books* for books used to create this site.

audio bridge A communications bridge that allows multiple duplex connections over 4-wire telephone connections. Well designed audio bridges, such as Rane's ECB 6 do not connect inputs to their own outputs, thus avoiding feedback. See mix-minus.

audio compression See: digital audio data compression

audio connectors See connectors.

Audio magazine (1947-2000) America's first and longest running audio magazine. Its demise after 53 years of continuous publication leaves a huge void in the consumer audio world. Gone is the last great rational voice, lost amidst the pseudoscientific din dominating high-end audio. An audio warrior is dead and we are lessened.

audio websites A truly astonishing and remarkable list of audio related websites compiled *daily* by Steve Ekblad. Also see Audio & HiFi Page, an equally astonishing and remarkable list of audio related websites compiled by Tomi Engdahl.

auditory filter Term used to describe the concept of critical bands. Analogous to a bandpass filter with a rounded top ("rounded-exponential" after Patterson and Moore, 1986). The filter is slightly asymmetric, being wider on the low-frequency side.

Aureal 3D (*A3D*) Proprietary 3D sound technology first developed by Crystal River Engineering, now the advanced technology subsidiary of Aureal Semiconductor. Aureal 3D makes many claims. At one time their website stated that "since we can hear sounds three dimensionally in the real world by using two ears, it must be possible to create sounds from two speakers that have the same effect." Well, ... *NO* ... it's pretty rhetoric, but flawed logic. Our two ears receive sound coming from sources located in every possible direction, and from *that* information process three-dimensional location -- that is not the problem. The problem is how to make our two ears receive sound from sources located in only two directions, and trick them into hearing three dimensionally -- that is the problem. Aureal claims to have solved this problem.

autoformer *Autoformer* is short for *autotransformer*, or *self-transformer*, from the definition of *auto-*. An autotransformer is one that self-magnetizes to produce the transformer voltage, it does this by not having a true secondary, i.e., there is only one winding with one part acting as the primary and the other part acting as the secondary, but there is no second winding, and no air gap, and thus no true isolation between the primary and secondary. Therefore an autotransformer is a transformer in which part of one winding is common to both the primary and the secondary circuits associated with that winding. For this reason, autotransformers are not the preferred choice for professional audio use because in addition to the transformed voltage (usually 70.7V in the U.S. & 100V elsewhere) you want true isolation. However, they are common because they are cheaper to make since you don't have to wind a separate primary and secondary.

automatic mic mixer A specialized mixer optimized for solving the problems of multiple live microphones operating together as a system, such as found in boardrooms, classrooms, courtrooms, church systems, etc. An automatic mic mixer controls the live microphones by turning up (*on*) mics when someone is talking, and turning down (*off*) mics that are not used, thus it is a voice-activated, real-time process, without an operator, hence, *automatic*. An automatic mic mixer must adapt to changing background noise conditions. Further it must control the additive effect of multiple mics being on at the same time (see NOM). If one mic is on at maximum gain, opening up another one may cause acoustic feedback, so an automatic mixer must also control the system gain to prevent feedback or excessive noise pickup. Dan Dugan patented the first automatic mic mixer and is recognized as the father of this technology. A final problem that automatic mixers solve is maintaining a natural ambience from the room. This is especially critical in recording and broadcasting. A good automatic mixer must make rapid and dramatic changes in the gains of the input channels while maintaining the sonic illusion that nothing is happening at all.

average power See apparent power

A-weighting See weighting filters

B

babbling tributary In LAN technology, a workstation that constantly sends meaningless messages.

back-emf (back-*electromotive force*) Literally, *back-voltage*, is a phenomena found in all moving-coil electromagnetic systems, but for audio is most often used with respect to loudspeaker operation. This term describes the action where, after the signal stops, the speaker cone continues moving, causing the voice coil to move through the magnetic field (now acting like a microphone), creating a *new* voltage that tries to drive the cable back to the power amplifier's output. If the loudspeaker is allowed to do this, the cone flops around like a dying fish. It does *not* sound good. The only way to stop back-emf is to make the loudspeaker "see" a dead short, i.e., zero ohms looking backward, or as close to it as possible. See: damping factor

background music Officially music without lyrics and not performed by the original artist, used as an alternative to silence. Contrast with *foreground music*.

balance control A control found most commonly on professional and consumer stereo preamplifiers, used to change the relative loudness (power) between the left and right channels. One channel is made (apparently) louder by attenuating the opposite channel. This is most often done (in analog designs) with a dual potentiometer with an "M-N taper." An M-N taper consists of a "shorted" output for the first 50% of travel and then a linear taper for the last 50% of travel, operating oppositely for each channel. Therefore, with the control in its center detent position, there is no attenuation of either channel. Rotating it away from the center position causes one channel to be attenuated, while having no effect on the other channel, and vice-versa. Contrast with pan and crossfade controls.

balanced line The IEEE dictionary defines a *balanced circuit* as "a circuit in which two branches are electrically alike and symmetrical with respect to a common reference point, usually ground." This is the essence of a balanced interconnect. Namely, that two lines are driven equally and oppositely with respect to ground. Normally this also implies that the receiving circuits have matching impedances. Exactly matching impedances is preferred for it provides the best common mode rejection. Balances 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, see: *RaneNote 110 Sound System Interconnection*) [*Long Answer:* To understand why balanced lines are so successful, first examine a balanced, or differential (equivalent term) output stage, and then an input stage: A differential output stage simultaneously drives two lines, one positive and one negative. The voltage difference between these two wires is the audio signal. The two signals form an envelope that rides the wires to the balanced input stage. Note that the audio signal exists uniquely between these two lines -- not between them and ground. The complete circuit path travels down on the positive line and back on the negative line. Ground is not needed to transmit the signal -- this is the essence and power of balanced lines. Ground is used only for shielding and safety purposes. Conversely, an **unbalanced line** is one that transmits the audio signal between one wire and ground. The circuit path is down the wire and back through the shield cable connected to ground. Ground is the return path; the circuit does not work without it. A balanced (or differential) input stage extracts the difference between the two input lines, and that, of course, is the desired audio signal. It receives the envelope sent down the cable by the differential output. This circuit's shining virtue is its great noise rejection ability. It has what is called great common-mode rejection. 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. The best balanced line designs have 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.]

balun (*balanced-unbalanced*) A jargon term originally popularized by radio engineers referring to the *bal*anced to *un*balanced transformer used to interface with the radio antenna. Today, expanded to refer to any interface (usually a transformer) between balanced and unbalanced lines or circuitry; may also provide impedance transformation, as 300 ohm balanced to 75 ohm unbalanced, or vice versa. Another popular use is in transitioning between balanced twisted-pair and an unbalanced coaxial cable.

band-limiting filters A low-pass and a high-pass filter in series, acting together to restrict (limit) the overall bandwidth of a system.

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** 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.

banjo "I can see fiddling around with a banjo, but how do you banjo around with a fiddle?" -- Duncan Purney [*from* Barber]

bar A unit of pressure equal to one million dynes per square centimeter. (*Yeah, I know, you expected some wiseacre response, but you ain't gonna get it*.)

Bara, Theda (1890-1955) Anagram of "Arab Death," used as a pseudonym by the Cincinnati-born, Hollywood actress Theodosia Goodman in the 1920s, who became the first woman movie star.

Baroque 1. Music: of, relating to, or characteristic of a style of composition that flourished in Europe from about 1600 to 1750, marked by chromaticism, strict forms, and elaborate ornamentation. 2. When you are out of Monet. (*Thanks JF and I'll never tell.)*

barrier strips Same as *terminal strips*, see connectors.

baseband A transmission medium with capacity for one channel only. Typically found in local area networks (LANs). In baseband LANs, the entire bandwidth, or capacity, of the cable is used to transmit a single digital signal. Everything on that cable (transmitted or received) must use that one channel, which is very fast, so each device needs only to use that high speed channel for only a little of the time. Therefore all attached devices (printers, computers, databases) share by taking turns using the same cable. Baseband as used in videoconferencing means audio and video signals are transmitted over separate cables. Contrast with broadband.

baseband signaling Transmission of a digital or analog signal at its original frequencies; i.e., a signal in its original form, not changed by modulation.

baud rate (pronounced "bawd"; after *Baudot Code* named for the French telegrapher Emile Baudot, 1845-1903) The transmitted signaling speed, or keying rate of a modem. Often confused with bit rate. *Bit rate and baud rate are NOT synonymous and shall not be interchanged in usage*. For example, one baud equals one half dot cycle per second in Morse code, one bit per second in a train of binary signals, and one 3-bit value per second in a train of signals each of which can assume one of 8 different states, and so on - all brought to you by the magic of advanced coding techniques that allow more than one bit per baud. Preferred usage is *bit* rate, with *baud* used only when the details of a modem are specified.

Baxandall tone controls The most common form of active bass and treble tone control circuit based upon British engineer P.J. Baxandall's paper "Negative Feedback Tone Control -- Independent Variation of Bass and Treble Without Switches," *Wireless World*, vol. 58, no. 10, October 1952, p. 402. The Baxandall design is distinguished by having very low harmonic distortion due to the use of negative feedback.

BCD 1. **(*binary-coded decimal*)** Pertains to a number system where each decimal digit is separately represented by a 4-bit binary code; for example, the decimal number 23 is represented as 0010 0011 (2 = 0010 and 3 = 0011, grouped together as shown), while in straight binary notation, 23 is represented as 10111. 2. **(*binary-coded digit*)** A digit of any number system that is represented as a fixed number of binary digits; from the previous example, the decimal digit 23 is represented as 10111.

bel *Abbr*. **b**, **B** Ten decibels. [After **Alexander Graham Bell**.] The Bel was the amount a signal dropped in level over a one-mile distance of telephone wire. See: decibel

Belchfire® Series Term coined by Crown International for their mythical power amplifier, the BF-6000SUX. Based on original research into the first principles of teramagnostriction quasar-quadrature, the BF-6000SUX could have changed the design of all future power amps, but it didn't. In spite of Crown's leap forward into the past of technical declination, the marketplace categorically stated that it did not want 6,000 watts per channel in only one rack space - in spite of its six-foot depth and 206 lbs weight. The only known use of a BF-6000SUX was to drive the experimental Electro-Voice Rearaxial Softspeaker, when Rane demoed their PI 14 Pseudoacoustic Infector using Jensen's JE-EP-ERs Multi-denomial Transpedance Informer for coupling - but many consider that only hearsay.

Bell, **Alexander Graham** (1847-1922) Scottish-born American inventor of the telephone. The first demonstration of electrical transmission of speech by his apparatus took place in 1876. Bell also invented the audiometer, an early hearing aid, and improved the phonograph.

BeOS (*Be operating system*) An operating system (OS) developed by Be Incorporated in 1996, called the first true "media OS," it is becoming very popular for Internet appliances, as well as software designed for live performance venues.

Bessel crossover A type of crossover utilizing low-pass filter design characterized by having a *linear phase response* (or *maximally flat phase response*), but also a monotonically decreasing passband amplitude response (which means it starts rolling off at DC and continues throughout the passband). Linear phase response (e.g., a linear plot of phase shift vs. frequency produces a straight line) results in *constant time-delay* (all frequencies within the passband are delayed the same amount). Consequently the value of linear phase is it reproduces a near-perfect *step response*, i.e., there is no overshoot or ringing resulting from a sudden transition between signal levels. The drawback is a sluggish roll-off rate. For example, for the same circuit complexity a Butterworth response rolls off *nearly three times as fast*. This circuit is based upon *Bessel polynomials*; however, the filters whose network functions use these polynomials are correctly called *Thompson filters* [W.E. Thomson, "Delay Networks Having Maximally Flat Frequency Characteristics," *Proc. IEEE*, part 3, vol. 96. Nov 1949, pp. 487-490]. The fact that we do not refer to these as *Thompson crossovers* demonstrates, once again, that we do not live in a fair world.

bifilar windings A term most often associated (in the pro audio industry) with audio transformer design describing the winding technique of laying two wires side-by-side, providing essentially unity coupling, thus reducing leakage inductance to negligible amounts. Literally *two threads* from Latin *bi-* two, and *filum* thread.

bilinear transform A mathematical method used in the transformation of a continuous time (analog) function into an equivalent discrete time (digital) function. Fundamentally important for the design of digital filters. A bilinear transform ensures that a stable analog filter results in a stable digital filter, and it exactly preserves the frequency-domain characteristics, albeit with frequency compression.

binary A condition in which there are two possible states; for example, the binary number system (base-2) using the digits 0 and 1.

binaural recording or **binaural sound** Believe it or not, the groundwork was laid in the 1920s (*no kidding*, and some claim even earlier) when the idea of placing two microphones in a dummy head was first introduced as a source of loudspeaker stereo (which wouldn't go anywhere until Blumlein's contributions). It was the Germans who first produced a standard artificial listener for evaluating auditorium acoustics, and then played back the results over headphones -- the startling realism launched binaural recording. A binaural recording is made using two microphones placed in the ear canals of an anatomically accurate dummy head, such that all the normal spatial attributes of the human head are present (just as in real listening situations) when the recording is made. Designed to be played back through headphones, the results are nothing but astonishing. First time listeners to binaural recordings often swear someone is there with them, talking and walking around them, such is the realism accomplished.

binky See Larry Blake's Film Sound Glossary

bit *Abbr*. **b** Abbreviation for *binary unit* or *binary digit*. 1. The smallest amount of digital information. A bit can store or represent only two states, 0 and 1. [The original term *binary unit* was coined by John Tukey of Bell Laboratories to represent the basic unit of information as defined by Shannon as a message representing one of two states.] 2. *A little bit* -- from Old English *bita*, meaning a piece bitten off.

bit clock The synchronizing signal that indicates the rate of individual data bits over a digital audio interface.

bit error rate The number of bits processed before an erroneous bit is found (e.g., 10E13), or the frequency of erroneous bits (e.g., 10E-13).

bit rate The rate or frequency at which bits appear in a bit stream. The bit rate of raw data from a CD, for example, is 4.3218 MHz.

bits -- data converter See data converter bits

bit stream A binary signal without regard to grouping.

bit-mapped display A display in which each pixel's color and intensity data are stored in a separate memory location.

blame shifter Shifts the pitch of mistakes down one octave so the audience thinks it was the bass player. [*Thanks to DD at Sound Path Labs*.]

blue moon For half a century, it's been known as the second full moon in the same calendar month; however, recently this definition was corrected by the editors of *Sky & Telescope* magazine. The correct definition, they say, is that a blue moon occurs when a season has *four full moons, rather than the usual three*. Further, they claim the misunderstanding is their fault based on an article they published in 1946. For all the wonderful details, see What's A Blue Moon?

blue whales See SPL

Blumlein, Alan Dower (1903-1942) English engineer who in a short working life span of 15 years wrote or cowrote 128 patents, developed stereophonic sound, designed new uses for microphones, designed a lateral disc-cutting system making modern records possible, developed much of the 405-line high definition television system broadcast in Britain until 1986, and improved radar systems such that they still operated 40 years later. Indeed, a genius by any definition, yet his story had to wait until 1999 to be told completely. Thanks to Robert Charles Alexander, former editor of *AudioMedia* magazine, a definitive biography now exists. Not only that, but Alexander has created a web site dedicated to Blumlein that, by the end of 2000, will have all 128 patents reproduced in their entirety, along with all of his binaural recordings (another of his inventions), downloadable as MP3 files, including binaural film clips (the world's first stereo films).

BNC (*bayonet Neill-Concelman*) A miniature bayonet locking connector for coaxial cable. It was developed in the late '40s by a collaboration of Paul Neill and Carl Concelman. In 1942, while at Bell Labs, Paul Neill developed what became known as the type N connector, named after him, which became a U.S. Navy standard. Carl Concelman, while at Amphenol, developed a bayonet version of the N connector, which became known as the type C connector, after him (*the first true 50-ohm connector*). Then, together, they developed a miniature bayonet locking version of the C connector and it was named the type BNC connector, after both of them. There is even an improved threaded version called the *threaded Neill-Concelman* or *TNC connector* See BNC RF Connectors for additional details, and JCM for examples. [*Thanks to all who wrote me to help clarify this correct meaning. My condolences to all, who with passion, conviction, and great creativity, truly believe differently. It is a sad but true tale that BNC does NOT stand for "baby N connector," or "bayonet connector," or "bayonet Naval connector," or "British Naval Connector" (sorry Microsoft). For further verification search the web for info on Paul Neill and Carl Concelman*.]

Boole, George (1815-1864) British mathematician who devised a new form of algebra that represented logical expressions in a mathematical form now known as **Boolean Algebra**. [See Maxfield]

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.

Boucherot cell See Zobel network

Bps (*always uppercase B*) Abbreviation for bytes per second.

bps (*always lowercase b*) Abbreviation for bits per second.

brewer See zymurgist

bridge 1. In communications networks a bridge is a device that connects two or more different networks and forwards packets between them; specifically a device that (a) links or routes signals from one ring or bus to another, or from one network to another, (b) may extend the distance and capacity of a single LAN system, (c) performs no modification to packets or messages, (d) operates at the data-link layer of the OSI--Reference Model (Layer 2), (e) reads packets, and (f) passes only those with addresses on the same segment of the network as the originating user. 2. A functional unit that interconnects two local area networks that use the same logical link control (LLC) procedure, but may use different medium access control (MAC) procedures. 3. A balanced electrical network, e.g., a Wheatstone bridge. Contrast with hub.

broadband Also *wideband*, a transmission medium having a bandwidth greater than a traditional telephone (speech) channel (4 kHz). [*Some argue that to be "broadband" the medium must support 20 kHz*.] The most common broadband medium is coaxial cable carrying multiple audio, video and data channels simultaneously. Each channel takes up a different frequency on the cable. There will be guardbands, or empty spaces, between the channels to make sure each channel does not interfere with its neighbor. The most common example is the CATV cable. Contrast with baseband.

BSI (*British Standards Institute*) The National Standards organization responsible for coordinating standards preparation for sound equipment in the UK.

buffer In data transmission, a temporary storage location for information being sent or received.

buffer amplifier The IEEE dictionary defines buffer amplifier as "An amplifier in which the reaction of output-load-impedance variation on the input circuit is reduced to a minimum for isolation purposes." This is a bit confusing, but one thing is clear, it says that at the most fundamental level a buffer amplifier isolates (or buffers) the loading effects (impedance) of two stages. It separates them, making them independent. In analog designs, buffer amplifiers are used for just this purpose. If the next circuit stage in a design has impedance characteristics that are detrimental to the preceding stage then a buffer amplifier minimizes this interaction. And its use is not confined to analog design, digital circuits use buffers to minimize similar loading effects.

The term "amplifier" comes about from the fact that most buffer amplifiers also provide either voltage or current gain. In this sense, a normal audio power amplifier can be called a buffer amplifier - it buffers your preamp from your very low impedance loudspeakers. [*Historical Note*: Sometimes a buffer amplifier provides speed as well as isolation. In the mid '70s, National Semiconductor offered in their specialty hybrid circuits line, a product simply named "Fast Buffer," whose purpose was to provide impedance isolation, but could do so at high megawiggle speeds (not a trivial task back then), and if that wasn't good enough, they also offered a "Damn Fast Buffer," that could really get the job done (true story).] As can seen, the term buffer amplifier is a bit vague: it provides isolation, that much is sure, however, it may also offer voltage gain, current gain, or both. And it may even provide an unbalanced-to-balanced function, or vice-versa.

burst error A large number of data bits lost on the medium because of excessive damage to or obstruction on the medium.

bus One or more electrical conductors used for transmitting signals or power from one or more sources to one or more destinations. Often used to distinguish between a single computer system (connected together by a *bus*) and multi-computer systems connected together by a *network*.

buss To kiss.

Butterworth crossover A type of crossover circuit utilizing low-pass filter design characterized by having a *maximally flat magnitude response*, i.e., no amplitude ripple in the passband. This circuit is based upon *Butterworth functions* (or *Butterworth polynomials*). [For the mathematically inclined, these polynomials represent a specialized solution to a general MacLaurin series based upon a Taylor series expansion. Named after S. Butterworth, a British engineer who first described this response in his paper "On the Theory of Filter Amplifiers," *Wireless Engineer*, vol. 7, 1930, pp. 536-541. Eleven years later, V.D. Landon coined the phrase *maximally flat* in his paper "Cascade Amplifiers with Maximal Flatness," *RCA Review*, vol. 5, 1941, pp. 347-362.]

byte *Abbr*. **B** A group of eight bits (a word) operating together. Usually abbreviated in upper-case to distinguish "byte" from "bit" which uses lower-case "b". See: Bps

C

cables Audio systems use many different types of cables as follows (for all the details see Lampen):

- **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** See data cables and Category cables.
- **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-pair*).

Cannon plug See connectors.

Cartesian coordinate system 1. A two-dimensional coordinate system in which the coordinates of a point in a plane are its distances from two perpendicular lines that intersect at an origin, the distance from each line being measured along a straight line parallel to the other. 2. A three-dimensional coordinate system in which the coordinates of a point in space are its distances from each of three perpendicular planes that intersect at an origin. After the Latin form of Descartes, the mathematician who invented it.

CAT 3 (*Category 3 cable*) Unshielded twisted-pair (UTP) data grade cable (usually 24AWG). CAT 3 cables are characterized to 16 MHz and support applications up to 10 Mbps. Typically used for voice telephone and 10Base-T Ethernet systems.

CAT 5 (*Category 5 cable*) Unshielded twisted-pair (UTP) data grade cable (usually 24AWG). CAT 5 cable runs are limited to 100 meters (328 feet) due to signal radiation and attenuation considerations. Longer runs are vulnerable to electromechanical interference. CAT 5 cables are characterized to 100 MHz and support applications up to 100 Mbps. Most common application is 100Base-T Ethernet systems.

Category cables Telecommunications created a *Category of Performance* standard for data cables. This defines standards (mostly tests) for cabling and cabling components (connectors, etc.). Originally there were five categories, however, today, there are effectively only two: *CAT 3* and *CAT 5* (see above). See hyperlink for details.

CATV (*community antenna television* or *cable television*) A broadband transmission medium, most often using 75-ohm coaxial cable carrying many TV channels simultaneously.

CAV (*constant angular velocity*) A disc rotating at a constant number of revolutions per second. The LP is a CAV system at 33 1/3 rpm. Another example is the CAV laser disc that plays two thirty minute sides.

CCIF (*Comité Consultatif International des Téléphonique*, or *International Telephone Consultative Committee*) The CCIF merged with the CCIT becoming the CCITT. In 1992, the CCITT, together with the CCIR, morphed into the ITU

CCIR (*Comité Consultatif International des Radio Communications*, or *International Radio Consultative Committee*) (*International Radio Consultative Committee*) Merged with the ITU and became the ITU-R radiocommunications division.

CCIR ARM See: weighting filters

CCIR-468 See: weighting filters

CCIR 2 kHz See: weighting filters

CCIT (*Comité Consultatif International des Télégraphique*, or *International Telegraph Consultative Committee*) Merged with the CCIF to become the CCITT.

CCITT (*Comité Consultatif International des Téléphonique et Télégraphique*, or *International Telegraph and Telephone Consultative Committee*) Merged with the ITU and became the ITU-T telecommunications division.

CD (*compact disc*) Trademark term for the Sony-Philips digital audio optical disc storage system. The system stores 75 minutes (maximum) of digital audio and subcode information, or other non-audio data, on a 12-centimeter diameter optical disc. The disc is made of plastic, with a top metallized layer, and is read by reflected laser light. Variations (such as the 3" disc) are reserved for special applications.

CD horn EQ See: constant directivity horn

CD-I (*compact disc interactive*) System storing digital audio, video, text, and graphics information interactively, with user control over content and presentation, on a 12-centimeter diameter optical disc.

CD+MIDI A System storing MIDI information in a disc's subcode area.

CD-PROM (*compact disc programmable read-only memory*) A write-once CD-ROM disc.

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

CD-ROM (*compact disc read-only memory*) A method of storing digitally coded information, such as computer information or database, on a 12-centimeter diameter optical disc.

CD-V (*compact disc video*) A system storing five minutes of analog video and digital audio plus twenty minutes of digital audio only on a 12-centimeter diameter optical disc, and longer times on 20- or 30-centimeter diameter optical discs.

CEI (*Commission Electrotechnique Internationale*) See IEC

Celsius *Abbr*. **C** Of or relating to a temperature scale that registers the freezing point of water as 0°C and the boiling point as 100°C, under normal atmospheric pressure. [The term "Celsius" is preferred to "centigrade" in technical contexts.] [After **Anders Celsius**]

Celsius, Anders (1701-1744) Swedish astronomer who devised the centigrade thermometer (1742).

CEMA (*Consumer Electronics Manufacturers Association*) The definitive source for information about the consumer electronics industry.

CE-mark (*Conformité Européenne*) The letter-logo used in marking units certified for distribution within the European Union (EU) that meet the directives as mandated by the European Commission.

center frequency One of the parameters of a *bandpass filter*. The center frequency occurs at the maximum or minimum amplitude response for Butterworth filters, the most common found in audio electronics.

centi- Prefix for one hundredth (10E-2), abbreviated **c.**

centigrade Temperature term generally not used in scientific contexts apart from meteorology. See: Celsius

CERN (*Conseil Européenne pour la Recherche Nucléaire*) European Particle Physics Laboratory. See: World Wide Web

charge *Symbol* **q** 1. *Electricity.* a. To cause formation of a net electric charge on or in (a conductor, for example). b. To energize (a storage battery) by passing current through it in the direction opposite to discharge. 2. *Physics.* a. The intrinsic property of matter responsible for all electric phenomena, in particular for the force of the electromagnetic interaction, occurring in two forms arbitrarily designated *negative* and *positive.* b. A measure of this property. c. The net measure of this property possessed by a body or contained in a bounded region of space.

checksum The sum of a group of data items used for error checking. If the checksum received equals the one sent, all is well. Otherwise, the receiving equipment requests the data be sent again.

chiasmus The term for a reversal in the order of words in two otherwise parallel phrases. For example, the advice from the great sci-fi writer Ray Bradbury to aspiring writers: "*You have to know how to accept rejection and reject acceptance*," or the familiar adage: "*Say what you mean and mean what you say*."

Christie, Samuel Hunter See Wheatstone bridge.

chromatic scale *Music.* A scale consisting of 12 semitones.

chrominance Abbreviated **C**. The color portion of the video signal - includes hue and saturation information but not brightness (see luminance).

CISC (*complex instruction set computing*) See: RISC

class-A An amplifier class

clock A timing device that generates the basic periodic signal used as a source of synchronizing signals in digital equipment.

CLV (*constant linear velocity*) A disc rotating at varying numbers of revolutions per second to maintain a constant relative velocity between pickup and track across the disc radius. The CD is a CLV system rotating from 500 rpm (lead-in track) to 200 rpm (lead-out track). Another example is the CLV laser disc that plays two sixty minute sides.

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. See cables.

CobraNet™ A trademark of Peak Audio identifying their licensed networking technology used for the deterministic and isochronous transmission of digital audio, video, and control signals over 10 Mbit and 100 Mbit Ethernet networks.

codec (*code-decode* also *compression-decompression*) Originally a device for converting voice signals from analog to digital for use in digital transmission schemes, normally telephone based, and then converting them back again. Broaden now to mean an electronic device that converts analog signals, such as video and voice signals, into digital form and compresses them to conserve bandwidth. Most codecs employ proprietary coding algorithms for data compression, common examples being Dolby's AC-2, ADPCM, and MPEG schemes. It is data compression (and direct digital video & audio inputs) that has evolved the newer meaning of ***co**mpression-**dec**ompression*.

combining response See: interpolating response

compander A contraction of compressor-expander. A term referring to dynamic range reduction and expansion performed by first a compressor acting as an encoder, and second by an expander acting as the decoder. Normally used for noise reduction or headroom reasons.

complex frequency variable An AC frequency in complex number form.

complex number *Mathematics.* Any number of the form $a + bj$, where a and b are real numbers and j is an imaginary number whose square equals -1; and a represents the ***real part*** (e.g., the resistive effect of a filter, at zero phase angle) and b represents the ***imaginary part*** (e.g., the reactive effect, at 90 degrees phase angle).

composite video A video signal combining luminance, chrominance, and synchronization data on a single coax cable using RCA connectors and color-coded yellow.

compression 1. An increase in density and pressure in a medium, such as air, caused by the passage of a sound wave. 2. The region in which this occurs.

compression wave A wave propagated by means of the compression of a fluid, such as a sound wave in air.

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 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.

conjobble An English word no longer in print (*except here*) meaning *to settle, arrange; to chat* (late 17th century).

connectors Audio equipment uses many types of connectors as follows:

- **Cannon connector** or **Cannon plug** Alternate reference for XLR.
- **Elco connector** or **Elco plug** AVX Elco 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 (Varicon 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 *pluggable* 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 pluggablility, 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).
- **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 pluggable. Has become known as the *U.S. style terminal blocks*. Contrast with Euroblocks.
- **¼" TRS (*tip-ring-sleeve*)** 1. Stereo ¼" 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*]
- **¼" TS (*tip-sleeve*)** Mono ¼" connector consisting of tip (T) [*signal*] and sleeve (S) [*ground & shield*] for unbalanced wiring.
- **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. Mics used the 3-pin P3 version. Some loudspeakers use the P4 or P8 versions of this connector to this day. 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 directivity (CD) horn A horn-loaded high frequency driver that exhibits more or less constant distribution of high-frequency sound in the horizontal direction. This is done by using one of several special dual shaped horn designs created to solve the traditional problem of horn-loaded driver output varying with frequency. All CD horns exhibit a high frequency roll-off of approximately 6 dB/octave beginning somewhere in the 2 kHz to 4 kHz area. Fixed EQ boost networks that compensate for this are known as **CD horn EQ** circuits.

constant group delay See: group delay

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. See: RaneNotes

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).

contour control *DJ mixers.* A control found on professional DJ performance mixers used to change the shape or taper (*contour*) of the fader action. Thus at, say, 50 % of travel, a fader may allow 50 %, or 10 %, or 90% of the audio signal to pass depending on the taper of the control. The *contour control* (switched, continuous or stepped variable) changes this amount.

control voltage In audio electronic circuits using voltage-controlled amplifiers, or other gain-controllable devices, a DC voltage proportional to the audio input signal amplitude, sometimes frequency dependent, used to set the instantaneous gain of a VCA or other device. It is normally developed in the *side-chain* of the electronic circuit.

convolution A mathematical operation producing a function from a certain kind of summation or integral of two other functions. In the time domain, one function may be the input signal, and the other the impulse response. The convolution than yields the result of applying that input to a system with the given impulse response. In DSP, the convolution of a signal with FIR filter coefficients results in the filtering of that signal.

Corba (*common object request broker architecture*) An ORB (object request broker) standard developed by the OMG (object management group). Corba provides for standard object-oriented interfaces between ORBs, as well as to external applications and application platforms (from *Newton's Telecom Dictionary*; see *Pro Audio Reference Books*). Not to be confused with CobraNet.

correlation A mathematical operation that indicates the degree to which two signals are alike.

CRC (*cyclic redundancy check*) An integrity checking process for block data. A CRC character is generated at the transmission end. Its value depends on the hexadecimal value of the number of ones in the data block. The transmitting device calculates the value and appends it to the data block. The receiving end makes a similar calculation and compares its results with the added character. If there is a difference, the recipient requests retransmission.

crap (*completely ridiculous audio performance*) Favorite acronym used to describe the characteristics of poor sound equipment.(thanks C.D.!)

crest factor The term used to represent the ratio of the peak (crest) value to the rms value of a waveform measured over a specified time interval. For example, a sine wave has a crest factor of 1.4 (or 3 dB), since the peak value equals 1.414 times the rms value. Music has a wide crest factor range of 4-10 (or 12-20 dB). This means that music peaks occur 12-20 dB higher than the rms value, which is why headroom is so important in audio design.

critical band *Physiology of Hearing.* A range of frequencies that is integrated (summed together) by the neural system, equivalent to a bandpass filter (auditory filter) with approximately 10-20% bandwidth (approximately one-third octave wide). [Although the latest research says critical bands are more like 1/6-octave above 500 Hz, and about 100 Hz wide below 500 Hz.] The ear can be said to be a series of overlapping critical bands, each responding to a narrow range of frequencies. Introduced by Fletcher (1940) to deal with the masking of a pure-tone by wideband noise.

cross-coupled A type of balanced line driver loosely based on servo-loop technology. Developed to emulate some of the features of a balanced line output transformer, the circuit employs positive feedback taken from each side of the outputs coupled back (*cross-coupled*) to the opposite input circuitry where it is used to fix the gain of the positive and negative line drivers. Each gain is typically set to unity (one) for normal operation and changes to *two* whenever either of the output lines is shorted to zero. In this manner, it emulates a transformer in that there is no change in output level if one of the lines becomes short-circuited to ground; however, since the gain of the ungrounded side has increased 6 dB then *the headroom of the system has been reduced by 6 dB* due to the short. In this sense this circuit does not act like a transformer, which does not change gain when one side is shorted to ground.

crossfade Within the audio industry, a term most often associated with dj mixers. DJ mixers usually feature a *crossfader* slide-type potentiometer control. This control allows the dj to transition from one stereo program source (located at one travel extreme) to another stereo program source (located at the other travel extreme). It is the crossfader that becomes the main remix tool for turntablists. Contrast with pan and balance controls.

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 (recording) See: print-through

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.

cue 1. A term found throughout various audio fields meaning to monitor, or listen (via headphones) to a specific source. In mixers (particularly dj mixers), the term is used interchangeably with solo or PFL as found on recording consoles. 2. Music. a. A section of music used in film or video ranging from a short piece of background music to a complex score. b. An extract from the music for another part printed, usually in smaller notes, within a performer's part as a signal to enter after a long rest. c. A gesture by a conductor signaling the entrance of a performer or part. 3. A signal, such as a word or an action, used to prompt another event in a performance, such as an actor's speech or entrance, a change in lighting, or a sound effect.

current *Symbol* **i, I** *Electricity.* a. A flow of electric charge. b. The amount of electric charge flowing past a specified circuit point per unit time, or the rate of flow of electrons. [As electrons flow in one direction, the spaces left behind, called *holes*, appear to flow in the opposite direction. Thus, current can be visualized as *electron flow* (negative current flow), or in the opposite direction, *hole flow* (positive current flow, sometimes called *conventional* current flow).]

current loop A data transmission scheme that looks for current flow rather than voltage levels. This systems recognizes no current flow as a binary zero, and having current flow as a binary one. Favored for its low sensitivity to cable impedance, and independence of a common ground reference; hence current loops do not introduce ground loops. MIDI is an example of a current loop interconnect system.

cut-only equalizer Term used to describe graphic equalizers designed only for attenuation. (Also referred to as notch equalizers, or band-reject equalizers). The flat (0 dB) position locates all sliders at the top of the front panel. Comprised only of notch filters (normally spaced at 1/3-octave intervals), all controls start at 0 dB and reduce the signal on a band-by-band basis. Proponents of cut-only philosophy argue that boosting runs the risk of reducing system headroom.

cutoff frequency *Filters.* The frequency at which the signal falls off by 3 dB (the *half power point*) from its maximum value. Also referred to as the *-3 dB points*, or the *corner* frequencies.

D

DA-88 Tascam's model number for their digital multitrack recorder using Sony-developed "Hi8" 8 mm videotape as the storage medium. Becoming a generic term describing this family of recorders. See: DTRS

DAB (*digital audio broadcast*) 1. NRSC (National Radio Systems Committee) term for the next generation of digital radio broadcast. 2. Initials of the compiler of this Pro Audio Reference.

DAC (or **D/A, *digital-to-analog converter*)** The electronic component which converts digital words into analog signals that can then be amplified and used to drive loudspeakers, etc. The DAC is the last link in the digital chain of signal processing. See data converter bits

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. That is about 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.]

DAR (*digital audio radio*) EIA term for the next generation of digital radio broadcasting standards.

DASH (*digital audio stationary head*) A family of formats for ensuring compatibility among digital multitrack studio recorders using stationary (as opposed to rotating) heads. The DASH standard, popularized by Sony and Studer, specifies 2 to 48 tracks, with tape speeds from 12 to 76 cm/sec.

DAT (*digital audio tape recorder*) 1. A digital audio recorder utilizing a magnetic tape cassette system with rotary heads similar to that of a video recorder. 2. A little bit of something as in *dis & dat*.

data cables Analog audio signals require a relatively small bandwidth and are interconnected using standard cables. In contrast to analog audio, digital audio and digital control signals require a very large bandwidth and must be interconnected with specially designed *data* cables. See Category cables.

data compression See: digital audio data compression

data converter bits The number of bits determines the data converter *precision*. The more bits available, the more precise the conversion, i.e., the closer the digital answer will be to the analog original. When an analog signal is sampled (at the sampling frequency), it is being sliced up into vertical pieces. Each vertical piece is then estimated as to its amplitude (*How large is the audio signal at this instant?*). This estimation process is the data converters job. It compares the original signal against its best estimate and chooses the closest answer. The more bits, the more choices the data converter has to choose from. The number of choices is the number "2" raised to the number of bits (*this explanation is simplified for clarity*). For example, 16-bits, creates 2 to the 16th power of choices, or 65,536 possible answers for the converter to choose from. And the higher the sampling rate, the more slices for any given time period. Again, the more slices, the more accurate will be the data conversion. All of which, ultimately determines how well the reproduced signal sounds compared to the original. For example, if a signal is recorded using "16-bits at 48 kHz", then for every one second of the audio signal, it is sliced up into 48,000 pieces. Then each piece is compared against a ruler with 2 to the 16th graduations, or 65, 536 voltage levels. Each sample instant is compared against this ruler and one value is assigned to represent its amplitude. For each second, 48,000 samples are given specific values to represent the original signal. If the same signal is recorded using "24-bits at 96 kHz" then for the same one second period, there will be 96,000 slices, or samples, and each one will be compared against a voltage ruler now divided into 2 to the 24th divisions, or 16,777,216 choices. Obviously this converter can choice an answer that is far closer to the original than before, and it gets to do this for twice as many samples. All of which, in the end, means this converter recorded samples that more closely approximated the original audio signal. [*Where it gets interesting is in trying to answer the question of what is enough? Sure, more bits are more accurate, but can the human ear tell the difference. In most cases, once you go beyond true 16-bits, the answer is no. All benefits above 16-bits/48 kHz are very small refinements, not monumental improvements. What really is going on, is that the advertised "16-bit/48 kHz" recordings of yesterday weren't. They used 16-bit converters but their accuracy was not 16-bits, it was more like 14-bits. Similarly today, the advertised "24-bit" converters are not 24-bit accurate, but they are certainly at least 18-bit accurate, and that makes an audible difference. So, if you can find a true 16-bit system and compare it with a typical 24-bit system of today, they will sound very nearly identical. And the sampling rate getting faster makes even less of an audible difference. For example if you compare a typical 16-bit/96 kHz system against a 24-bit/48 kHz, you will pick the 24-bit system every time. If you have a choice, always choose more bits, over a higher sampling rate*.]

DAW (*digital audio workstation*) Any of several software/hardware systems using a computer as the basis for creating, editing, storing, and playback of digital audio, using the computer's hard disk as the recording medium.

dB (*decibel*) See: decibel

DB-25 connector A 25-pin D-shell connector standardized for RS-232 serial communications.

DB-9 connector A smaller 9-pin version of the connector used for RS-232 communications. First made popular by IBM in their AT personal computer.

DCC (*Digital Compact Cassette*) Philips's digital version of the standard analog cassette tape system. A DCC recorder plays and records digital cassettes, as well as playing analog cassettes. [Now discontinued.]

DCE (*Data Communications Equipment*) Within the RS-232 standard, the equipment that provides the functions required to establish, maintain, and terminate a connection, as well as the signal conversion, and coding required for communication between data terminal equipment and data circuit - e.g., a modem or printer. See: DTE. The main difference between DCE and DTE is the wiring of pins 2 and 3, thus the need for a null modem cable when tying two computers together.

deadly nevergreen An English word no longer in print (except here) meaning *the gallows* (late 18th, early 19th centuries).

Dead Musician Directory "A site about dead musicians ... and how they got that way" Hey! Don't laugh, these guys are dead serious. Dead Rockers, jazz, reggae, bluegrass

Dead Recording Media A great chronicle of obsolete devices compiled by David Morton: "... site devoted to the dead, dying, or very ill technologies of sound recording."

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, for brevity reasons if no other.*]:

> **0 dBu** Preferred informal abbreviation for the official dB (0.775V); 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 (1.0V); 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.*]

decimal digit Everyday normal, base-10 numbers.

DED (pronounced "dead") **(*dark emitting diode*)** A variation of LED technology used exclusively by the CIA for clandestine equipment. Also popular as power-off indicators.

de-emphasis See: pre-emphasis

de-esser A special type of audio signal compressor that operates only at high frequencies (>3 kHz), used to reduce the effect of vocal sibilant sounds.

De Forest, Lee (1873-1961) Known as “the Father of Radio,” he was an American electrical engineer who patented the triode electron tube (1907) that made possible the amplification and detection of radio waves. He originated radio news broadcasts in 1916.

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.

delta modulation A single-bit coding technique in which a constant step size digitizes the input waveform. Past knowledge of the information permits encoding only the differences between consecutive values.

delta-sigma ADC See: delta-sigma modulation

delta-sigma modulation (also **sigma-delta)** An analog-to-digital conversion scheme rooted in a design originally proposed in 1946, but not made practical until 1974 by James C. Candy. The name *delta-sigma modulation* was coined by Inose and Yasuda at the University of Tokyo in 1962, but due to a misunderstanding the words were interchanged and taken to be *sigma-delta*. Both names are still used for describing this modulator. Characterized by oversampling and digital filtering to achieve high performance at low cost, a delta-sigma A/D thus consists of an analog modulator and a digital filter. The fundamental principle behind the modulator is that of a single-bit A/D converter embedded in an analog negative feedback loop with high open loop gain. The modulator loop oversamples and processes the analog input at a rate much higher than the bandwidth of interest (see: Sampling (Nyquist) Theorem). The modulator's output provides 1-bit information at a very high rate and in a format that a digital filter can process to extract higher resolution (such as 20-bits) at a lower rate.

Descartes, **René** (1596-1650) French mathematician and philosopher. Considered the father of analytic geometry, he formulated the Cartesian system of coordinates. [Then there's the story about how Descartes met his ultimate demise: It seems he was in a bar in Paris sipping a glass of Kir when the bartender asked if he would like another. M. Descartes responded "I think not," whereupon he disappeared without a trace.] (Thanks to Glenn D. White for this.)

deserializer A serial-to-parallel data converter; used in buses and networks.

destructive solo See: solo

diatonic 1. *Music*. Of or using only the eight tones of a standard major or minor scale without chromatic deviations. 2. A popular summer drink without the gin or the sugar.

DI (*direct*) box See direct box

dictionary "A malevolent literary device for cramping the growth of a language and making it hard and inelastic. This dictionary, however, is a most useful work." -- Ambrose Bierce.

difference-tone IMD See: IM

diffraction grating A usually glass or polished metal surface having a large number of very fine parallel grooves or slits cut in the surface and used to produce optical spectra by diffraction of reflected or transmitted light.

diffuse Widely spread out or scattered; not concentrated.

diffuser (or **diffusor**, British spelling; in acoustics, the British spelling is seen most often.) A commercial device that diffuses, or scatters sound. First invented by Manfred R. Schroeder ["Diffuse Sound Reflection by Maximum-Length Sequences," *J. Acous. Soc. Am.*, Vol. 57, No. 1, pp 149-150, Jan 1975], and made commercially successful by Dr. Peter D'Antonio and his company RPG Diffusor Systems. Diffusors are the acoustical analog of diffraction grating -- see above.

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.

digital audio data compression Commonly shortened to "audio compression." Any of several algorithms designed to reduce the number of bits (hence, bandwidth and storage requirements) required for accurate digital audio storage and transmission. Characterized by being "lossless" or "lossy." The audio compression is "lossy" if actual data is lost due to the compression scheme, and "lossless" if it is not. Well-designed algorithms ensure "lost" information is inaudible - that's how you win the game.

digital audio watermarking See watermarking

digital clipping See: 0 dBFS

digital filter Any filter accomplished in the digital domain.

digital hybrid See: hybrid

digital overs See: overs

digital signal Any signal which is quantized (i.e., limited to a distinct set of values) into digital words at discrete points in time. The accuracy of a digital value is dependent on the number of bits used to represent it.

digitization Any conversion of analog information into a digital form.

DIN Acronym for *Deutsche Industrie Norm (Deutsches Institut fuer Normung)*, the German standardization body.

dipless crossfader A crossfader design that does not attenuate the first audio signal until the fader is moved past the 50% travel point, while simultaneously increasing the second audio signal to 100% at the center point. With this design there is no attenuation (*dip*) in the center position for either audio signal, hence "dipless."

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.

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.

direct sound Sound first arriving. Sound reaching the listening location without reflections, i.e., sound that travels *directly* to the listener.

disc The term used for any *optical* storage media. Originally popularized to refer to phonograph records. From Latin *discus*, the term refers primarily to *audio* and *video* storage systems, such as compact discs, laser discs, etc., but the advent of CD-ROMs and computer optical storage units blurs this distinction. Compare with disk

discreet Marked by, exercising, or showing prudence and wise self-restraint in speech and behavior. You may want to be discreet when bussing someone.

discrete Constituting a separate thing; distinct, or a set of distinct things.

discrete Fourier transform (DFT) 1. A numerical method of calculating the coefficients of the Fourier series from a sampled periodic signal. 2. A DSP algorithm used to determine the Fourier coefficient corresponding to a set of frequencies, normally linearly spaced. See: Fourier theorem.

disk The term used for any *magnetic* storage media such as computer diskettes or hard disks. From Greek *diskos*, the term refers primarily to *non-audio digital data storage*, but the advent of hard disk digital audio recording systems fogs this up somewhat. Compare with disc

distance learning A specialized form of videoconferencing optimized for educational uses. Distance learning allows students to attend classes in a location distant from where the course is being presented. Two-way audio and video allows student and instructor interaction.

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.

distribution amplifier A splitter with added features. Distribution amplifiers (usually) feature balanced inputs and outputs with high-current line drivers (often cross-coupled) capable of driving very long lines.

dither The noise (analog or digital) added to a signal prior to quantization (or word length reduction) which reduces the distortion and noise modulation resulting from the quantization process. Although there is a slight increase in the noise level, spectrally shaped dither can minimize the apparent increase. The noise is less objectionable than the distortion, and allows low-level signals to be heard more clearly.

DIY Acronym for *do-it-yourself*, usually referring to various hobbies, especially audio-related.

Dolby Digital® Dolby's name for its format for the digital soundtrack system for motion picture playback. Utilizes their AC-3 system of digital compression. The signal is optically printed *between* the sprocket holes. Now being introduced to Home Theater on laser disc and DVD. Dolby Digital may use any number of primary audio delivery and reproduction channels, from 1 to 5, and may include a separate bass-only effects channel. The designation "5.1" describes the complete channel format. Surround decoder systems with Dolby Digital automatically contain Dolby Pro Logic processing to ensure full compatibility with the many existing program soundtracks made with Dolby Surround encoding. *No abbreviations are to be used.*

Doppler effect [After **Christian Johann Doppler**, 1803-1853, Austrian physicist and mathematician who first enunciated this principle in 1842.] 1. For an observer, the apparent change in pitch (frequency) of a sound (or any wave) when there is relative motion between the source and the listener (or observer). The classic example is the train phenomenon where the pitch of the whistle sounds higher approaching and lower leaving. 2. Gave rise to the variation known as the *dope-ler effect*, defined as the phenomenon of stupid ideas that seem smarter when they come at you in rapid succession. [Thanks, JF.]

DOS (pronounced "doss") **(*disk operating system*)** A software program controlling data in memory, disk storage, running programs and I/O management.

double balanced See cables.

double bass A large viola that plays one octave lower than the cello, thereby doubling the bass. Also referred to as a *Baroque doghouse* for its deep tones. See: Slatford for historical details.

downward expander See: expander

dreamt The only English word ending in the letters "mt."

dropout An error condition in which bits are incorrect or lost from a digital medium. Also occurs with analog tape - audio or video.

DSD® (*Direct Stream Digital®*) See: SACD

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.

DTE (*Data Terminal Equipment*) 1. Within the RS-232 standard, the equipment comprising the data source, the receiver, or both - e.g., personal computers or terminals. See: DCE. The main difference between DTE and DCE is the wiring of pins 2 and 3, thus the need for a null modem cable when tying two computers together. 2. *CobraNet* refers to DTEs as the source and sink devices on the network, i.e., they source and sink audio. Rane's *NM 48 & NM 84 Network Mic/Line Preamps* are DTE devices.

DTMF (*dual tone multi-frequency*) Normal everyday pushbutton touch-tone dialing system, where a combination of two tones is used for each button pushed.

DTRS™ (*digital tape recording system*) Tascam's suggested term for describing their DA-88 type digital multitrack recorders.

DTS Coherent Acoustics® (now *DTS Cinema*) A competing digital soundtrack system for motion picture playback developed by Digital Theater Systems Inc. (backed by Stephen Spielberg and Universal Studios). Its novelties are: 1) not requiring a special projector to read digital code off the filmstrip like its competitors; 2) using only very moderate compression (3:1 verses Dolby's 11:1); and 3) offering 20-bit audio. The discrete digital full bandwidth six (6) channel sound is contained on a CD that is played synchronously with the film. The synching time code is printed between the standard optical soundtrack and the picture.

DTS Zeta Digital™ (now *DTS Consumer*) Digital Theater Systems' audio compression scheme applied to laser disc, DVD and CD technology for home theater use. Competing format with Dolby's AC-3 algorithm.

ducker A dynamic processor that lowers (or "ducks") the level of one audio signal based upon the level of a second audio signal. A typical application is paging: A ducker senses the presence of audio from a paging microphone and triggers a reduction in the output level of the main audio signal for the duration of the page signal. It restores the original level once the page message is over.

Dudley, Homer Inventor of the vocoder. [*Dudley, H. (1939) "The Vocoder," Bell Labs Record, 17, pp. 122-126*]

duplex Pertaining to a simultaneous two-way independent transmission in both directions. Often referred to as "full duplex" which is redundant. See also: half-duplex.

DVD (Officially "DVD" does not stand for anything. It used to mean "digital *versatile* disc" - and before that it meant "digital *video* disc" also known as *hdCD* in Europe.) A 12-centimeter (4.72") compact disc (same size as audio CDs and CD-ROMs) that holds 10 times the information. Capable of holding full-length movies *and* a video game based on the movie, or a movie *and* its soundtrack, or two versions of the same movie - all in sophisticated discrete digital audio surround sound. The DVD standard specifies a laminated single-sided, single-layer disc holding 4.7 gigabytes, and 133 minutes of MPEG-2 compressed video and audio. It is backwards compatible, and expandable to two-layers holding 8.5 gigabytes. Ultimately two discs could be bounded together yielding two-sides, each with two-layers, for a total of 17 gigabytes. There are four main versions:

- **DVD-Video** (movies) As outlined above.
- **DVD-Audio** (music-only) The standard is flexible, allowing for many possibilities, leaving the DVD-player to detect which system is used and adapt. Choices include 74 minutes for 2-chs at 24-bits, 192 kHz sampling, or 6-chs at 24-bits, 96 kHz, all utilizing lossless compression (type MLP for *Meridian Lossless Packing*). Quantization can be 16-, 20-, & 24-bits, with sampling frequencies of 44.1, 88.2, and 176.4 kHz, as well as 48, 96, and 192 kHz all supported.
- **DVD-ROM** (read-only, i.e., games and computer use)
- **DVD-RAM** (rewritable, i.e., recording systems). Matsushita is currently the leader in density with 4.7Gb and 9.4Gb claimed for single-sided and double-sided discs respectively, compared with 2.6Gb and 5.2Gb offered by standard DVD-RAM technology. There are several competing formats:
  - **DVD-R** (2nd-generation DVD-RAM; Hitachi & Matsushita) Primiary application is peripheral drive for PCs, but is also of interest for video servers and video-disk cameras. Consumer applications not ruled out.
  - **DVD-R/W** or **DVD-RW** (Pioneer) Aimed at VCR replacement.
  - **DVD+RW** or just **RW** (because it is not sanctioned by the DVD Forum) (Sony, Philips & Hewlett-Packard) A 3-Gbyte system, positioned (for the moment) as a PC peripheral, but considerations for a 4.7Gbyte consumer version are looming.
  - **MMVF** (NEC's 5.2 Gbyte *Multimedia Video File* Disk system) Now officially shifted from a laboratory project to a business project.

dynamic controllers (or **dynamic 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 "dynamic" since the processing is completely program dependent. The two most common dynamic effects are compressors and expanders, with limiters, noise gates (or just "gates"), duckers and levelers being subsets of these. Another dynamic controller category includes exciters, or enhancers. And noise reduction units fall into a final dynamic 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 loud.

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 minimum output signal is determined by the noise floor of the unit, 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).

dyne A unit of force, equal to the force required to impart an acceleration of one centimeter per second per second to a mass of one gram. Old usage for sound pressure.

E

EASE (*Enhanced Acoustic Simulator for Engineers*) A computer modeling tool distributed by Renkus-Heinz for ADA (Acoustic Design Ahnert), who developed the software and introduced it in 1990 at the 88th AES Convention in Montreux.

EBU (*European Broadcasting Union*) An international professional society that, among other things, helps establish audio standards.

echo canceller A technique using DSP (analog circuits exist, but DSP solutions are overwhelmingly superior) that filters unwanted signals caused by echoes from the main audio source. Echoes happen in both voice and data conversation, therefore two types of cancellers are encountered: ***acoustic*** and ***line***. "Acoustic" echo cancellers are used in teleconferencing applications to suppress the acoustic echoes caused by the microphone/loudspeaker combination at one end picking up the signal from the other end and returning it to the original end. It is similar to sound system feedback problems (where the sound reinforcement loudspeaker is picked up by the microphone, re-amplified through the loudspeaker, only to be picked up again by the microphone, to be re-amplified, and so on), only made much worst by the additional time delay introduced by the telecommunication link. "Line" echo cancellers are used to suppress *electrical* echoes caused by the transmission link itself. Such things as non-perfect hybrids, and satellite systems (creating round-trip delays of about 600 ms), contribute to very annoying and disruptive line echoes.

ECS (*Engineered Conference Systems*)™ Rane Corporation trademark for their teleconferencing equipment.

Edison effect In 1883, Thomas Edison noticed that certain materials, when heated by a filament in a vacuum, emitted electrons that could be attracted to an electrode held at a positive potential with respect to the emitter. This became known as the *Edison effect* and according to Edison, was discovered by accident when experimenting with his new invention, the incandescent lamp. Twenty years later, this effect became the basis for inventing the vacuum tube.

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

EEPROM or **E2PROM (*electrically erasable programmable read-only memory*)** A version of read-only memory that can be *electrically* erased and reprogrammed by the designer. Differentiated from standard EPROM (one "E") which requires ultraviolet radiation for erasure.

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. Compare with insert loop

EFP (*electronic field production*) mixer Pretentious equivalent for ENG mixer

EIA (*Electronic Industries Alliance*) Founded in 1924 as the *Radio Manufacturers Association* (RMA), The EIA is a private trade organization made up of manufacturers which sets standards for voluntary use of its member companies (and all other electronic manufacturers), conducts educational programs, and lobbies in Washington for its members' interests.

Elco plug See connectors.

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.

electronic music Glossary of terms.

EMC Directive (*ElectroMagnetic Compatibility*) 1. A directive issued by the European Commission aimed at establishing product compatibility within the EU (European Union). Article 1.4 defines electromagnetic compatibility as the ability of an electrical and electronic appliance, equipment or installation containing electrical and/or electronic components to function satisfactorily in its electromagnetic environment (*immunity* requirement) without introducing intolerable electromagnetic disturbances to anything in that environment (*emission* requirement). 2. Due to the significant increases in development time and product costs imposed by the EMC Directive, many believe the initials really stand for "eliminate minor companies." [*Thanks DC.*]

EMP (*Experience Music Project*) Paul Allen's (co-founder of Microsoft) interactive music museum, located in Seattle, that celebrates and explores creativity and innovation in American popular music as exemplified by rock 'n' roll. [*Very cool place ... come visit sometime.*]

ENG (*electronic news gathering*) mixer Portable battery-powered mixer accommodating at least two or three mic inputs, used in the field to record speech and outdoor sound effects. Some specialized models have built-in telephone line interfacing.

enhancers See: exciters

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.

Eric "Hoss" Cartwright's given name -- "Hoss" was a nickname ... ... this, for all you Jeopardy! fans.

error correction A method using a coding system to correct data errors by use of redundant data within a data block. Often data is interleaved for immunity to burst errors. Corrected data is identical to the original.

ESTA (*Entertainment Services & Technology Association*) A non-profit trade association representing the North American entertainment technology industry.

ether From a Greek word meaning "upper air," a term used in early physics (based on ancient beliefs), a magical medium thought to explain the propagation of electromagnetic waves.

Ethernet A local area network (LAN), originally developed by Xerox (the name was coined by its inventor Bob Metcalfe after the old science term *ether*), used for connecting computers, printers, workstations, terminals, etc., now extended to include audio and video using CobraNet technology. Ethernet operates over twisted-pair, coaxial cable, or fiber optic cable at various speeds designated "**10Base-T**" up to 10 megabits/sec (Mbps), "**100Base-T**", a.k.a. ***Fast Ethernet***, up to 100 Mbps, and based on fiber interconnect: "**1000Base-F**" up to 1 gigabit/sec, or 1000 Mbps. (The number in the front designates the speed in megabits/second. "Base" indicates the network is baseband. The letter following determines the type of cable and its requirements. 10Base-T, for example is unshielded twisted-pair, using a star topology.)

Euroblocks Shortened form for *European style terminal blocks*. See connectors.

exciters (or **enhancers)** A term referring to any of the popular special-effect signal processing products used primarily in recording and performing. All exciters work by adding harmonic distortion of some sort - but harmonic distortion found pleasing by most listeners. Various means of generating and summing frequency-dependent and amplitude-dependent harmonics exist. Both even- and odd-ordered harmonics find favorite applications. Psychoacoustics teaches that even-harmonics tend to make sounds soft, warm and full, while odd-harmonics tend to make things metallic, hollow and bright. Lower-order harmonics control basic timbre, while higher-order harmonics control the "edge" or "bite" of the sound. Used with discrimination, harmonic distortion changes the original sound dramatically, more so than measured performance might predict.

expander A signal processing device used to *increase the dynamic range* of the signal passing through it. Expanders complement compressors. For example, a compressed input dynamic range of 70 dB might pass through a expander and exit with a new *expanded* dynamic range of 110 dB. [*Long answer*: Just like compression, what "expansion" is and does has evolved significantly over the years. Originally expanders were used to give the reciprocal function of a compressor, i.e., it undid compression. Anytime audio was recorded or broadcast it had to be compressed for optimum transfer. Then it required an expander at the other end to restore the audio to its original dynamic range. Operating about the same "hinge" point and using the same ratio setting as the compressor, an expander makes audio *increases* and *decreases bigger*. From this sense came the phrase that "*expanders make the quiet sounds quieter and the loud sounds louder*." Modern expanders usually operate only *below a set threshold point* (as opposed to the center hinge point), i.e., they operate only on low-level audio. The term *downward expander* or *downward expansion* evolved to describe this type of application. The most common use is noise reduction. For example, say, an expander's threshold level is set to be just below the smallest vocal level being recorded, and the ratio control is set for 3:1. What happens is this: when the vocals stop, the "decrease below the set-point" is the change from signal (vocals) to the noise floor (no vocals), i.e., there has been a step decrease from the smallest signal level down to the noise floor. If that step change is, say, -10 dB, then the expander's output will be -30 dB (because of the 3:1 ratio, a 10 dB decrease becomes a 30 dB decrease), thus resulting in a noise reduction improvement of 20 dB.

extensible Of or relating to a programming language or a system that can be modified by changing or adding features. Capable of being extended: *AES-24 is an extensible protocol.*

F

5.1 surround sound The digital audio multichannel format developed by the Moving Picture Experts Group (see: MPEG) for digital soundtrack encoding for film, laserdiscs, videotapes, DVD, and HDTV broadcast. The designation "5.1" (first proposed by Tom Holman of THX fame) refers to the five discrete, full bandwidth (20-20kHz) channels - *left, right, & center fronts*, plus *left & right surrounds* - and the ".1" usually refers to the limited bandwidth (20-120Hz) *subwoofer* channel, but can also refer to a special effects/feature channel. Terminology used by both Dolby Digital and DTS Consumer (the home version of their theater Coherent Acoustics system).

Fahrenheit *Abbr*. **F** Of or relating to a temperature scale that registers the freezing point of water as 32°F and the boiling point as 212°F, under normal atmospheric pressure. [In scientific and technical contexts temperatures are now usually measured in degrees Celsius rather than Fahrenheit.] [After **Gabriel Daniel Fahrenheit**]

Fahrenheit, Gabriel Daniel (1686-1736) German-born physicist who invented the mercury thermometer (1714) and devised the Fahrenheit temperature scale.

FAQ (*frequently asked question*) Acronym commonly seen on bulletin boards, Internet Web sites, and corporate information centers. By compiling FAQ lists (*FAQs*), organizations significantly reduce time spent repeatedly answering the same questions.

far end Teleconferencing term meaning the distant location of transmission; the other end of the telephone line, as opposed to your end (known as the *near end*)..

far-end crosstalk Crosstalk that is propagated in a disturbed channel in the *same direction as the propagation of the signal* in the disturbing channel. The terminals of the disturbed channel, at which the far-end crosstalk is present, and the energized terminals of the disturbing channel, are usually remote from each other.

far field or **far sound field** The sound field distant enough from the sound source so the SPL decreases by 6 dB for each doubling of the distance from the source (*inverse square law*). Contrast with near field.

Fast Ethernet See: Ethernet

fax on demand One of the terms for the process of ordering fax documents from remote machines via telephone, using a combination of voice processing and fax technologies. Also called *fax-back*.

fax-back See: **fax on demand**.

feedback See acoustic feedback.

feedback The word "feedback" is the longest word in the English language that uses all the letters "A" through "F." [*Thanks, Brad, for being so observant while playing Trivial Pursuit*™.]

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 us continuous frequency shifting (a very small amount) to prevent frequency build up and feedback before it happens.

FDDI (*fiber distributed data interface*) An ANSI standard describing a 100 megabytes/sec (MBps) fiber optic LAN; now also specified for twisted-pair use.

femto- Prefix for one thousandth of one trillionth (10E-15), abbreviated **f**.

FFT (*fast Fourier transform*) 1. Similar to a discrete Fourier transform except the algorithm requires the number of sampled points be a power of two. 2. A DSP algorithm that is the computational equivalent to performing a specific number of discrete Fourier transforms, but by taking advantage of computational symmetries and redundancies, significantly reduces the computational burden. [It is believed the FFT was first described by Cornelius Lanczos of the Boeing Co. in the 1940's.]

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. See cables.

film sound glossary See Larry Blake's Film Sound Glossary; find out what a "binky" is.

filter Any of various electric, electronic, acoustic, or optical devices used to reject signals, vibrations, or radiation of certain frequencies while passing others. Think sieve: pass what you want, reject all else. For audio use the most common electronic filter is a *bandpass filter*, characterized by three parameters: *center frequency*, *amplitude* (or magnitude), and *bandwidth*. Bandpass filters form the heart of audio *graphic equalizers* and *parametric equalizers*.

FIR (*finite impulse-response*) filter A commonly used type of digital filter. Digitized samples of the audio signal serve as inputs, and each filtered output is computed from a weighted sum of a finite number of previous inputs. An FIR filter can be designed to have linear phase (i.e., constant time delay, regardless of frequency). FIR filters designed for frequencies much lower that the sample rate and/or with sharp transitions are computationally intensive, with large time delays. Popularly used for adaptive filters.

Firewire See: IEEE 1394

firkytoodle An English word no longer in print (except here) meaning to engage in intimate physical affection, as a prelude to sexual intercourse; foreplay (17th to 19th century).

flanging Originally, "flanging" was achieved using two reel-to-reel tape recorders playing the same program, in synchronization, with their outputs summed together. By alternately slowing one machine, then the other, different phase cancellations occurred in the summation process. The "slowing down" was done simply by pressing against the *flanges* of the tape reels, hence the original term "reel flanging," soon shortened to just "flanging." Since the two identical signals would alternately add and subtract due to the introduced phase (timing) difference, the audible effect was one of a sweeping comb filter. It was described as a "swishing" or "tunneling" sound. Soon electronic means were devised to mimic true "reel flanging" by using delay lines and mixing techniques. Adding a low-frequency oscillator to modulate the audio delay line's clock signal created a sweeping effect, much like a jet airplane taking off. The best flangers used two delay lines. Compare with: phaser

Fleming, Sir John Ambrose (1849-1945) British electrical engineer and inventor known for his work on electric lighting, wireless telegraphy, and the telephone. He invented and patented the first tube, a diode, he used for signal detection.

Fletcher-Munson Curves Fletcher and Munson were researchers in the '30s who first accurately measured and published a set of curves showing the human's ear's sensitivity to loudness verses frequency. They conclusively demonstrated that human hearing is extremely dependent upon loudness. The curves show the ear most sensitive to sounds in the 3 kHz to 4 kHz area. This means sounds above and below 3-4 kHz must be louder in order to be heard *just as loud*. For this reason, the Fletcher-Munson curves are referred to as "equal loudness contours." They represent a family of curves from "just heard," (0 dB SPL) all the way to "harmfully loud" (130 dB SPL), usually plotted in 10 dB loudness increments.

floating point An encoding technique consisting of two parts: a *mantissa* representing a fractional value with magnitude less than one, and an *exponent* providing the position of the decimal point. Floating point arithmetic allows the representation of very large or very small numbers with fewer bits.

floating unbalanced line A quasi-balanced output stage consisting of an unbalanced output connected to the *tip* of a ¼" TRS (tip-ring-sleeve) jack through an output resistor (typically in the 50-300 ohms range). An equal valued resistor is used to tie the *ring* terminal to signal ground. The *sleeve* connection is left open or "floating." Thus, from the receiver's viewpoint, what is "seen" are two lines of equal impedance, used to transfer the signal. In this sense, the line is 'balanced," although only one line is actually being driven. Leaving the sleeve open, guarantees that only one end of the shield (the receiving end) will be grounded. A practice that unbalanced systems often require. For trouble free interconnections, balanced lines are always the preferred choice.

floobydust A contemporary made-up term, one meaning being derived from the archaic Latin *miscellaneus*, whose disputed history springs from Indo-European roots, probably finding Greek origins (influenced, of course, by Egyptian linguists) -- meaning a *mixed bag*, or a *heterogeneous motley mixed varied assortment*. Popularized within the audio community when borrowed and used by the author of this webpage as a chapter title in the National Semiconductor *Audio Handbook* first published in 1976.

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

Foley A term synonymous with film sound effects. A recording studio *Foley stage* is where the sound effects are generated in synch with the moving picture. Named after *Jack Foley*, who invented sound effects for film sound while working for Universal. He simultaneously added music and effects to the previously silent film "Showboat" and the first "Foley" session was born.

foreground music Officially music with (or without) lyrics and performed by the original artist. Used where it is believed people will pay attention to it. Contrast with *background music*.

Fourier analysis *Mathematics.* Most often the approximation of a function through the application of a Fourier series to periodic data, however it is not restricted to periodic data. [The Fourier *series* applies to periodic data only, but the *Fourier integral transform* converts an infinite continuous time function into an infinite continuous frequency function, with perfect reversibility in most cases. In this sense, it is not an approximation. The DFT and FFT are examples of the Fourier series, but are not approximations either unless the time data is an approximation itself, such as for sampled data systems, which introduces sampling errors.]

Fourier, **Baron Jean Baptiste Joseph** (1768-1830) French mathematician and physicist who formulated a method for analyzing periodic functions and studied the conduction of heat.

Fourier series Application of the *Fourier theorem* to a periodic function, resulting in sine and cosine terms which are harmonics of the periodic frequency. [After **Baron Jean Baptiste Joseph Fourier**.]

Fourier theorem A mathematical theorem stating that any function may be resolved into sine and cosine terms with known amplitudes and phases.

FPGA (*field-programmable gate array*) A programmable logic device which is more versatile (i.e., much larger) than traditional programmable devices such as PALs and PLAs.

free field or **free sound field** A sound field without boundaries or where the boundaries are so distant as to cause negligible reflections over the frequency range of interest. Note that if the boundaries exist but completely absorb the sound then a virtual free field is created, thus anechoic chambers are used to measure loudspeakers.

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.

full duplex Redundant term. See: duplex

FX unit Slang for "effects unit."

G

gain suppression See suppression.

GAL® (*generic array logic*) Registered trademark of Lattice Semiconductor for their invention of EEPROM-based low-power programmable logic devices.

gate See: noise gate

gated or **gated-on** *Teleconferencing*. Term referring to microphone inputs on an automatic mic mixer that turn off (*close*) after speech stops. Contrast with last-on.

Generation X The tenth generation of Americans since 1776. [From *Roman numeral X meaning 10*.]

gibi *Symbol* **Gi** New term standardized by the IEC as Amendment 2 to IEC 60027-2 *Letter Symbols to be Used in Electrical Technology* to signify binary multiples of 1,073,741,824 (i.e., 2E30). Meant to distinguish between exact binary and decimal quantities, i.e., 1,073,741,824 verses 1,000,000,000. For example, it is now 16 gibibits, abbreviated 16Gib, not 16 gigabits or 16Gb.

giga- A prefix signifying one billion (10E9), abbreviated **G**.

gigabyte Popular term meaning a billion bytes but should be *gibibyte* meaning 2E30 bytes. See: gibi.

GIGO (*garbage in garbage out*) Popular acronym used by programmers to indicate that incorrect information sent to a system generally results in incorrect information received from it.

glass Popular jargon referring to glass fiber optic interconnection, or fiber optics in general.

GPIB (*general purpose interface bus*) See: IEEE-488.

granulation noise An audible distortion resulting from quantization error.

graphic equalizer A multi-band variable equalizer using 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.

gray code A sequence of binary values where only one bit is allowed to change between successive values. Generally "quieter" (producing less audible interference) than straight binary coding for execution of commands in audio systems.

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. See: SRM 66

group delay The rate of change of phase shift with respect to frequency. *Mathematically*, the first derivative of phase verses frequency. The *rate of change* is just a measure of the slope of the phase shift verses linear (not log) frequency plot. If this plot is a straight line, it is said to have a "constant" (i.e., not changing) phase shift, or a "linear phase" (or "phase linear" -*European*) characteristic. Hence, *constant group delay*, or *linear group delay*, describes circuits or systems exhibiting constant delay for all frequencies, i.e., all frequencies experience the same delay. Note that pure signal delay causes a phase shift proportional to frequency, and is said to be "linear phase," or "phase linear." In *acoustics*, such a system is commonly referred to as a "minimum phase" system. For a circuit example, see: Bessel crossover.

GUI (*graphical user interface*) A generic name for any computer interface that substitutes graphics (like buttons, arrows, switches, sliders, etc.) for characters; usually operated by a mouse or trackball. First mass use was Apple's Macintosh® computers, but is now dominated by Microsoft's Windows® programs.

gyrator filters Term used to describe a class of active filters using gyrator networks. Gyrator is the name given for RC networks that mimic inductors. A gyrator is a form of artificial inductor where an RC filter synthesizes inductive characteristics. Used to replace real inductors in filter design.

H

Haas Effect Also called the *precedence effect*, describes the human psychoacoustic phenomena of correctly identifying the direction of a sound source heard in both ears but arriving at different times. Due to the head's geometry (two ears spaced apart, separated by a barrier) the direct sound from any source first enters the ear closest to the source, then the ear farthest away. The Haas Effect tells us that humans localize a sound source based upon the first arriving sound, if the subsequent arrivals are within 25-35 milliseconds. If the later arrivals are longer than this, then two distinct sounds are heard. The Haas Effect is true even when the second arrival is louder than the first (even by as much as 10 dB.). In essence we do not "hear" the delayed sound. This is the hearing example of human *sensory inhibition* that applies to all our senses. Sensory inhibition describes the phenomena where the response to a first stimulus causes the response to a second stimulus to be inhibited, i.e., sound first entering one ear cause us to "not hear" the delayed sound entering into the other ear (within the 35 milliseconds time window). Sound arriving at both ears simultaneously is heard as coming from straight ahead, or behind, or within the head. The Haas Effect describes how full stereophonic reproduction from only two loudspeakers is possible. (After Helmut Haas's doctorate dissertation presented to the University of Gottingen, Gottingen, Germany as "Über den Einfluss eines Einfachechos auf die Hörsamkeit von Sprache;" translated into English by Dr. Ing. K.P.R. Ehrenberg, Building Research Station, Watford, Herts., England Library Communication no. 363, December, 1949; reproduced in the United States as "The Influence of a Single Echo on the Audibility of Speech," *J. Audio Eng. Soc.*, Vol. 20 (Mar. 1972), pp. 145-159.)

half-duplex Pertaining to a transmission over a circuit capable of transmitting in either direction, but only one direction at a time. See also: duplex

Hamster switch *DJ Mixers.* A control found on professional DJ performance mixers that reverses fader action. For example, if a fader normally is *off* at the bottom of its travel and *on* at the top of its travel, then activating the hamster switch reverses this, so *off* is now at the top and *on* is at the bottom of travel, or alternatively, it swaps left for right in horizontally mounted faders. Used to create the most comfortable (and fastest) fader access when using either turntable, and to accommodate left-handed and right-handed performers. Credited to, and named after, one of the original scratch-style crews named *The BulletProof Scratch Hamsters*.

handshaking The initial exchange between two communications systems prior to and during transmission to ensure proper data transfer.

happiness "An agreeable sensation arising from contemplating the misery of another." -- Ambrose Bierce.

hard disk A sealed mass storage unit used for storing large amounts of digital data.

hard disk recording See: DAW (*digital audio workstation)* and HDR

hardware The physical (mechanical, and electrical) devices that form a system.

harmonic series 1. *Mathematics.* A series whose terms are in harmonic progression, as in 1 + 1/3 + 1/5 + 1/7 + . . . 2. *Music.* A series of tones consisting of a fundamental tone and the overtones produced by it, and whose frequencies are consecutive integral multiples of the frequency of the fundamental.

HAVi (*Home Audio/Video interoperability*) An industry standard for home networks designed to link consumer electronics products. Developed by eight consumer giants -- Grundig, Hitachi, Panasonic, Philips, Sharp, Sony, Thomson Multimedia and Toshiba -- the main aim of this protocol is to ride on IEEE 1394 interface, connecting digital TVs, set-top boxes, DVD players and other digital consumer products.

hdCD (*high density compact disc*) See: DVD

HDCD (*high definition compatible digital*) Pacific Microsonics' trademark for their encode/decode scheme that allows up to 24 bit, 88.2 kHz digital audio mastering process, yet is compatible with normal 16 bit, 44.1 kHz CD and DAT formats. Claimed to sound superior even when not decoded, and to be indistinguishable from the original if decoded.

HDR (*hard-disk recorder*) An audio recording device based on computer hard disk memory technology. Typically, these machines are configured like analog tape recorders offering 24-48 tracks, utilizing 24-bit / 48-96kHz data converters with optional I/O to interface with ADAT, TDIF, or AES3, and file format interchangeability with DAWs.

HDTV (*high definition television*) The standard for digital television in North America, still being revised. When finished will include a definition for picture quality at least that of a movie theater, or 35 mm slide, i.e., at least two million pixels (compared to 336,000 pixels for NTSC).

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.)

HeadWize A non-profit (*i.e., no ads*) site specializing in headphones and headphone listening, featuring articles, essays, projects and technical papers on all things headphone -- very informative.

Helmholtz, Hermann Ludwig Ferdinand von (1821-1894) German physicist and physiologist who formulated the mathematical law of the conservation of energy (1847) and invented an ophthalmoscope (1851) [An instrument for examining the interior structures of the eye, especially the retina, consisting essentially of a mirror that reflects light into the eye and a central hole through which the eye is examined. You aren't a real doctor without one.] Famous for his book, *On the Sensations of Tone* first published in 1862.

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

Hertz, **Heinrich Rudolf** (1857-1894) German physicist who was the first to produce radio waves artificially.

hexadecimal A number system using the base-16, i.e., each number can be any of 16 values. Normally represented by the digits 0-9, plus the alpha characters A-F. Each hexadecimal digit can be represented by a four-bit binary number.

Hi8 See: DA-88

high-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-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***.

Holophonics An acoustical recording and broadcast technology claimed to be the aural equivalent to holography, hence the name. Holophonics is an encode process that occurs during the recording session using a special listening device named "Ringo." It is claimed that "playback or broadcast is possible over headphones or any existing mono or stereo speaker system, with various levels of spatial effect. Optimal effects occurs when two tracks (stereo) are played utilizing digital technology over headphones and minimal effect when played over a single mono speaker (two tracks merged into one and played over a single speaker)."

homicide "The slaying of one human being by another. There are four kinds of homicide: felonious, excusable, justifiable and praiseworthy, but it makes no great difference to the person slain whether he fell by one kind or another -- the classification is for advantage of the lawyers." -- Ambrose Bierce.

hope "Desire and expectation rolled into one." -- Ambrose Bierce.

house mixer See: FOH

howlround What the British call acoustic feedback.

HRRC (*Home Recording Rights Coalition*) An advocacy group that includes consumers, retailers, manufacturers and professional servicers of consumer electronics recording products.

HRTF (*head-related transfer function*) The impulse response from a sound source to the ear drum is called the *head-related impulse response* (HRIR), and its Fourier transform is called the *head-related transfer function* (HRTF). The HRTF captures all of the physical cues to source localization, and is a surprisingly complicated function of four variables: three space coordinates (azimuth, elevation & range) and frequency, and to make matters worst, they change from person to person. Interaural (i.e., *between the ears*) time differences, interaural time delays and the physical effects of diffraction of sound waves by the torso, shoulders, head and pinnae modify the spectrum of the sound that reaches the ear drums. These changes allow us to localize sound images in 3D space and are captured by the HRTFs. HRTFs have been named and studied since at least the early '70s (Blauert)

HTML (*hypertext markup language*) The software language used on the Internet's World Wide Web (WWW). Used primarily to create home pages containing hypertext.

HTTP (*hypertext transfer protocol*) The name for the protocol that moves documents around the Internet/Web. Used by the various servers and browsers to communicate over the net.

hub 1. In broadband LAN use, a central location of a network that connects network nodes through spokes, usually in a star architecture. Think of it as a digital splitter, or distribution amplifier. 2. In complex systems, hubs perform the basic functions of restoring signal amplitude and timing, collision detection and notification, and signal broadcast to lower-level hubs.

hum components The harmonics of the AC mains supply. The Americas (except the southern half of South America), Japan, Taiwan, Korea and the Philippines use a 60-Hz system, placing the most annoying 2nd and 3rd harmonics at 120 Hz and 180 Hz. For Europe, and the rest of the world using 50-Hz mains, these components fall at 100 Hz and 150Hz.

HVAC *Construction.* Term used to stand for the *heating, ventilating, & air conditioning* system of any building. *Electrical engineering.* Term used to mean *high-voltage alternating current*.

hybrid A telecommunication term used to describe an interface box that converts a conversation (or data signal) coming in on two pairs (one pair for each direction of the conversation or signal) onto one pair and vice versa (i.e., a 2-wire to 4-wire converter). This is necessary because all long distance circuits are two pairs, while most local circuits are one pair. The name comes from the original use of a "hybrid coil" in the telephone whose function was to keep the send and receive signals separated. Both analog and digital hybrid designs are found. A fundamental (and unavoidable) problem in any 2-wire to 4-wire design is leakage (crosstalk) between the transmit and receive signals. In analog designs leakage is reduced by modeling the impedance seen by the transmit amplifier as it drives the hybrid coil. Because telephone-line impedance is complex and not well-modeled by a simple passive RLC circuit, only 10 dB to 15 dB of leakage reduction is usually possible. Digital hybrids use DSP technology to model and dynamically adapt to provide much greater reduction than analog designs, typically resulting in reductions of 30 dB to 40 dB. However, the best digital hybrids incorporate acoustic echo cancelling (AEC) circuitry to gain even greater improvements. The AEC works to cancel out any remaining signal coming from the loudspeaker (far-end received signal) from the microphone signal before they can be retransmitted to the far end as acoustic echo. Digital hybrids with AEC achieve total leakage reduction of 50 dB to 65 dB.

hyperlink The protocol that allows connecting two Internet resources via a single word or phrase; allowing the user a simple point-and-click method to create the link.

hypertext Within WWW documents, the linking of words to other sections of text, pictures or sound is called hypertext. Hypertext is created using the HTML software language. Also used frequently in Help files.

I

IC (*integrated circuit*) A solid state device with miniaturized discrete active components on a single semiconductor material.

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.

IEEE (*Institute of Electrical and Electronic Engineers*) The largest professional organization for electrical engineers. Primarily concerned with education and standardization.

IEEE-488 also referred to as the *general purpose interface bus (GPIB).* Most common parallel format computer interface for simultaneous control of up to 15 multiple peripherals.

IEEE-1394 (aka ***Firewire***) A joint Apple and TI implementation of the IEEE P1394 Serial Bus Standard. It is a high-speed (100/200/400 Mbits/sec now, with 1 Gbit/s on the horizon) serial bus for peripheral devices. Supported by Apple, IBM, Intel, Microsoft and Sony, it is intended to replace Apple Desktop Bus (ADB) and SCSI (Microsoft announced Windows 95 support for IEEE 1394). Firewire supports automatic configuration ("plug and play") and hot-plugging (changing peripheral devices while running). It is also isochronous, meaning that a fixed slice of bandwidth can be dedicated to a particular peripheral - video, for instance. IEEE 1394 is on its way to becoming the optimal digital interface for 21st-century applications. Fast, inexpensive and reliable for audio/video as well as computer peripherals, IEEE 1394 carries all forms of digitized video and audio. A single Firewire interface can be used for all entertainment-center interconnections, done in a daisy-chain fashion. New computer peripherals such as digital television, CD-ROM, DVD, digital cameras (Sony was first) and home networks (See Digital Harmony) are the first users. See: USB for complementary low-speed system.

IIR (*infinite impulse-response*) filter A commonly used type of digital filter. This recursive structure accepts as inputs digitized samples of the audio signal, and then each output point is computed on the basis of a weighted sum of past output (feedback) terms, as well as past input values. An IIR filter is more efficient than its FIR counterpart, but poses more challenging design issues. Its strength is in not requiring as much DSP power as FIR, while its weakness is not having linear group delay and possible instabilities.

IM or **IMD (*intermodulation distortion*)** An audio measurement designed to quantify the distortion products produced by nonlinearities in the unit under test that cause complex waves to produce beat frequencies, i.e., sum and difference products not harmonically related to the fundamentals. For example, two frequencies, f1 and f2 produce new frequencies f3 = f1 - f2; f4 = f1 + f2; f5 = f1 - 2f1; f6 = f1 + 2f2, and so on. Numerous tests exist, each designed to "stress" the unit under test differently. The most popular follow:

- **SMPTE/DIN IMD** The most common IMD measurement. SMPTE standard RP120-1994 and DIN standard 45403 are similar. Both specify a two-sine wave test signal consisting of a large amplitude low-frequency tone linearly mixed with a high-frequency tone at ¼ the amplitude of the low frequency tone. SMPTE specifies 60 Hz and 7 kHz mixed 4:1. The DIN specification allows several choices in both frequencies, with 250 Hz and 8 kHz being the most common.
- **ITU-R** (*old* **CCIF**)**, Twin-Tone, or Difference-Tone IMD** All these terms refer to the same test and are used interchangeably. The test specifies two equal-amplitude closely spaced high frequency signals. Common test tones are 19 kHz and 20 kHz for full audio bandwidth units. While all combinations of IM distortion products are possible, this test usually measures only the low-frequency second-order product falling at f2-f1, i.e., at 1 kHz.
- **DIM/TIM (dynamic/transient intermodulation distortion)** A procedure designed to test the dynamic or transient behavior, primarily, of audio power amplifiers. The other IM tests use steady-state sine wave tones, which do not necessarily reveal problems caused by transient operation. In particular, audio power amplifiers with high amounts of negative feedback were suspect due to the inherent time delay of negative feedback loops. The speculation was that when a rapidly-changing signal was fed to such an amplifier, a finite time was required for the correction signal to travel back through the feedback loop to the input stage and that the amplifier could be distorting seriously during this time. The most popular test technique consists of a large amplitude 3 kHz square wave (band-limited to ~20 kHz) [*Historical Note*: This test proved that as long as the amplifier did not slew-limit for any audio signal, then the loop time delay was insignificant compared to the relatively long audio periods. Thus, properly designed negative feedback was proved not a problem. Subsequently, this test has fallen into disuse.]

image impedances The impedances that will simultaneously terminate all of a network's inputs and outputs in such a way that at each of its inputs and outputs the impedances in both directions are equal. In this manner the input and output impedances "see" their own "image."

image parameters Fundamental network functions, namely image impedances and image transfer functions, used to design or describe a filter.

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-dependant. 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.

infrasonic Generating or using waves or vibrations with frequencies below that of audible sound. Compare with subsonic - commonly used (erroneously) to mean infrasonic.

inline mixer Term referring to the normal long narrow vertical strip format common to all medium to large-scale mixing console designs (*mixers*}. Non-inline designs typically refer to rack-mount mixers, i.e., those that are 19" wide, and designed to fit into standard rack cases. These are as small as 1U space (1.75" H). Sometimes these are designed similar to an inline design laying on its side, now having a horizontal control flow instead of a vertical one. In the middle of the pack are rack-mount mixers that still use the inline vertical format, but do rack mount, but normally take up 10 or more spaces.

insertion loss The loss of voltage (or power), as measured in dB, resulting from placing a pad (or other power absorbing network) between a voltage (or power) source and its load impedance. It is the ratio of the voltage (or power) absorbed in the load without the pad (or network) to that when the network is inserted. For example if the voltage across a load is 2 volts without a network and 1 volt with the network, then the insertion loss is stated as 6 dB.

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.

instrument-level See levels.

interlayer-transfer See: print-through

interleaving The process of rearranging data in time. Upon de-interleaving, errors in consecutive bits or words are distributed to a wider area to guard against consecutive errors in the storage media.

Internet To try and define the Internet in a few words is a futile task. Click the hyperlink for a wonderful Internet history timeline. Contrast with WWW.

interpolating response Term adopted by Rane Corporation to describe the summing response of adjacent bands of variable equalizers using buffered summing stages. If two adjacent bands, when summed together, produce a smooth response without a dip in the center, they are said to *interpolate* between the fixed center frequencies, or *combine* well. [*Historical note:* Altec-Lansing first described their buffered equalizer designs as *combining* and the terminology became commonplace. Describing how well adjacent bands combine is good terminology. However, some variations of this term confuse people. The phrase "combining filter" is a misnomer, since what is meant is not a filter at all, but rather whether adjacent bands are buffered before summing. The other side of this misnomer coin finds the phrase "non-combining filter." Again, no filter is involved in what is meant. Dropping the word "filter" helps, but not enough. Referring to an equalizer as "non-combining" is imprecise. All equalizers combine their filter outputs. The issue is how much ripple results. For these reasons, Rane adopted the term "interpolating" as an alternative. Interpolating means to insert between two points, which is what buffering adjacent bands accomplishes. By separating adjacent bands when summing, the midpoints fill in smoothly without ripple.]

inverse square law *Sound Pressure Level.* Sound propagates in all directions to form a *spherical* field, thus sound energy is *inversely* proportional to the *square of the distance*, i.e., doubling the distance quarters the sound energy (the *inverse square law*), so SPL is attenuated 6dB for each doubling.

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.

IP (*intellectual property*) Referring to protected proprietary information, usually in the form of a patent, maskworks (integrated circuits or printed circuit boards), a copyright, a trade secret, or a trademark. Often misused to mean many different things.

IP (*internet protocol*) IP is the most important of the protocols on which the Internet is based. Originally developed by the Department of Defense to support interworking of dissimilar computers across a network, IP is a standard describing software that keeps track of the Internet work addresses for different nodes, routes outgoing messages, and recognizes incoming messages. It was first standardized in 1981. This protocol works in conjunction with TCP and is identified as TCP/IP.

IP address Another name for an *Internet address*. A 32-bit identifier for a specific TCP/IP host computer on a network, written in dotted decimal form, such as 209.238.233.232, with each of the four fields assigned 255 values, organized into hierarchical classes.

IRMA (*International Recording Media Association*) An advocacy group for the growth and development of all recording media and is the industry forum for the exchange of information regarding global trends and innovations.

ISBN (*International Standard Book Number*) In bibliography, a 10-digit number assigned to a book which identifies the work's national, geographic, language, or other convenient group, and its publisher, title, edition and volume number. Its numbers are assigned by publishers and administered by designated national standard book numbering agencies, such as R.R. Bowker Co. in the U.S., Standard Book Numbering Agency Ltd. in the U.K., Staatsbibliothek Preussischer Kulterbesitz (Prussian State Library) in Germany, and the Research Library on African Affairs in Ghana. Each ISBN is identical with the Standard Book Number, originally devised in the U.K., with the addition of a preceding national group identifier. [*Now if that isn't more than you will ever need to know about this subject then I'll eat a book.*]

ISDN (*Integrated Services Digital Network*) A high-capacity digital telecommunication network (mainly fiber optic) based on an international telephone standard for digital transmission of audio, data and signaling - all in addition to standard voice telephone calls. A cost-effective alternative to satellite links.

ISO (*International Standards Organization* or ***International Organization for Standardization*)** Founded in 1947 and consisting of members from over 90 countries, the ISO promotes the development of international standards and related activities to facilitate the exchange of goods and services worldwide. The U.S. member body is ANSI. [*Interesting tidbit*: according to ISO internet info, "ISO" is not an acronym. It is a derived Greek word, from *isos*, equal. For example, *isobar*, equal pressure, or *isometric*, equal length. Take a small jump from "equal" to "standard" and you have the name of the organization. It offers the further advantage of being valid in all the official languages of the organization (English, French & Russian), whereas if it were to be an acronym it would not work for French and Russian.]

isochronous (pronounced "i-sok-ronus") ("iso" *equal* + "chronous" time) A term meaning time sensitive; *isochronous transmission* is time sensitive transmission. For example, voice and video require isochronous transmission since audio/video synchronization is mandated.

ITU (*International Telecommunications Union*) Headquartered in Geneva, Switzerland, ITU is an international organization within which governments and the private sector coordinate global telecommunication networks and services. The ITU is divided into three sectors: radiocommunications (**ITU-R**), telecommunications development (**ITU-D**), and telecommunications standards (**ITU-T**).

ITVA (*International Television Association*) A global community of professionals devoted to the business and art of visual communication.

J

jackfield British term for patchbay

Java™ The trademarked name for a powerful object-oriented programming language developed by Sun Microsystems. Java allows high-speed fully interactive Web pages to be developed for the Internet or any type of platform.

jitter A tendency towards lack of synchronization caused by electrical changes. Technically the unexpected (and unwanted) phase shift of digital pulses over a transmission medium. A discrepancy between when a digital edge transition is supposed to occur and when it actually does occur - think of it as nervous digital, or maybe a digital analogy to wow and flutter.

jitter timing error Short-term deviations of the transitions of a digital signal from their ideal positions in time.

JPEG (*Joint Photographic Experts Group*) A standard for lossy compression of graphic-image files.

juke A roadside drinking establishment that offers inexpensive drinks, food, and music for dancing, especially to the music of a jukebox. [Derivative Note: probably from Gullah *juke*, *joog* disorderly, wicked of West African origin; *Wolof dzug* to live wickedly Mandingo (Bambara) *dzugu* wicked. Gullah, the English-based Creole language spoken by Black people off the coast of Georgia and South Carolina, retains a number of words from the West African languages brought over by slaves. One such word is *juke*, “ bad, wicked, disorderly, ” the probable source of the English word *juke*. Used chiefly in the Southeastern states, *juke* (also appearing in the compound *juke joint*) means a roadside drinking establishment that offers cheap drinks, food, and music for dancing and often doubles as a brothel. “To juke” is to dance, particularly at a juke joint or to the music of a jukebox whose name, no longer regional and having lost the connotation of sleaziness, contains the same word. *And you thought you were smart.*

JSA (*Japanese Standards Association*) The National Standards organization responsible for coordinating standards preparation in Japan. <

justify To shift a numeral so that the most significant digit, or the least significant digit, is placed at a specific position in a row.

K

kelvin *Abbr*. **K** A unit of absolute temperature equal to 1/273.16 of the absolute temperature of the triple point of water. This unit is equal to one Celsius degree. A temperature in kelvin may be converted to Celsius by subtracting 273.16. [After **First Baron Kelvin**]

Kelvin, William Thomson, First Baron (1824-1907) British physicist who developed the Kelvin scale of temperature (1848) and supervised the laying of a transatlantic cable (1866). His pioneering work in thermodynamics and electricity helped develop the law of the conservation of energy.

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

kibi *Symbol* **Ki** New term standardized by the IEC as Amendment 2 to IEC 60027-2 *Letter Symbols to be Used in Electrical Technology* to signify binary multiples of 1024 (i.e., 2E10). Meant to distinguish between exact binary and decimal quantities, i.e., 1024 verses 1000. For example, it is now 16 kibibits, abbreviated 16Kib, not 16 kilobits or 16Kb.

kilo- Abbreviated **k** (*always lower-case*). A prefix signifying one thousand (10E3).

Kilo- Abbreviated **K** (*always upper-case*). A prefix popularly used in computer work to signify multiples of 1024 (i.e., 2E10), but should use kibi. Meant to distinguish base-2 (binary) from base-10 (decimal) magnitudes. For example, a "16K" memory is actually 16,384 bits (i.e., 16 times 1024, or 2E14), but should now read "16Ki".

kludge or **kluge** A system, especially a computer system, that is constituted of poorly matched elements or of elements originally intended for other applications. Or as an article by Jackson Granholme in "Datamation" put it: "An ill-assorted collection of poorly matching parts, forming a distressing whole." [From *The American Heritage Dictionary*: *The word kludge is not "etymologist-friendly," having many possible origins, none of which can be definitively established. This term, found frequently in the jargon of the engineering and computer professions, denotes a usually workable but makeshift system, modification, solution, or repair. Kludge has had a relatively short life (first recorded in 1962 although it is said to have been used as early as 1944 or 1945) for a word with so many possible origins. The proposed sources of the word, German klug, kluge, "intelligent, clever," or a blend of klutz and nudge or klutz and refudge, do not contain all the necessary sounds to give us the word, correctly pronounced at least. The notions that kludge may have been coined by a computer technician or that it might be the last name of a designer of graphics hardware seem belied by the possibility that it is older than such origins would allow. It seems most likely that the word kludge originally was formed during the course of a specific situation in which such a device was called for. The makers of the word, if still alive, are no doubt unaware that etymologists need information so they can stop trying to "kludge" an etymology together.*]

Kodak See: Muzak

kSPS (*kilo samples per second*) One thousand (1,000) samples per second. A measurement of data converter speed.

kVA (*kilovoltamperes*) One thousand (1,000) voltamperes. See voltampere.

L

Lamarr, Hedy (1924-2000) Born Hedy Kiesler in Vienna, this Hollywood actress used her knowledge of musical harmony, along with composer George Antheil, to obtain a patent on technology for military communications in 1942, which established the groundwork for today's spread-spectrum communication technology.

LAN (*local area network*) A combination of at least two computers and peripherals on a common wiring scheme, which allows two-way communication of data between any devices on the network.

Laplace, Marquis Pierre Simon de (1749-1827) French mathematician and astronomer who formulated the theory of probability.

laser (*light amplification by stimulated emission of radiation*) A device that generates coherent, monochromatic light waves. All CD players contain a semiconductor laser in their optical pickup.

last-on *Teleconferencing*. Term referring to microphone inputs on an automatic mic mixer that stay on (*open*) until another mic input turns on. Contrast with gated-on. A last-on mic becomes a master mic if left open long enough.

lawful "Compatible with the will of a judge having jurisdiction." -- Ambrose Bierce.

lawyer "One skilled in circumvention of the law." -- Ambrose Bierce.

LCD (*liquid crystal display*) A display of numerical or graphical information made of material whose reflectance or transmittance changes when an electric field is applied. An LCD requires ambient light or back-lighting for viewing.

LED (*light emitting diode*) 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.

leveler A dynamic processor that maintains (or "levels") the amount of one audio signal based upon the level of a second audio signal. Normally, the second signal is from an ambient noise sensing microphone. For example, a restaurant is a typical application where it is desired to maintain paging and background music a specified loudness above the ambient noise. The leveler monitors the background noise, dynamically increasing and decreasing the main audio signal as necessary to maintain a constant loudness differential between the two. Also called SPL *controller*.

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

- **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 decibels.
- **instrument-level** Nominal signal from musical instruments using electrical pick-ups. Varies widely, from very low *mic-levels* to quite large *line-level*s.

lift/dip Popular European term meaning boost/cut.

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 echo canceller See: echo canceller

linear PCM A pulse code modulation system in which the signal is converted directly to a PCM word without companding, or other processing.

linear phase response Any system which accurately preserves phase relationships between frequencies, i.e., that exhibits pure delay. See: group delay

line-level See levels.

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, 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. See: RaneNotes

Linux A computer Unix-type operating system (OS) invented by *Linus Torvalds* in 1992, who wrote it as a student at the University of Helsinki. He created this OS because he couldn't afford one that could accomplish what he wanted with his available hardware. He then posted it on the network for other students, where it grew and became very stable and powerful. Today, for free, the software, source code, etc., is available off the web.

lossy See: digital audio data compression

loud Having offensively bright colors: *a loud necktie.*

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.]

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

low-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**.

L-pad See attenuator pad.

LSB (*least significant bit*) The bit within a digital word that represents the smallest possible coded value; hence, the LSB is a measure of precision.

luminance Abbreviated **Y**. That part of the video signal that carries the information on how bright the TV signal is to be. The black and white signal.

M

macintosh (also **mackintosh**) *Chiefly British* A raincoat or a lightweight, waterproof fabric that was originally of rubberized cotton. [After Charles *Macintosh* (1766-1843), Scottish inventor]

MADI (*multichannel audio digital interface*) An AES recommended practice document *Digital Audio Engineering - Serial Multichannel Audio Digital Interface (MADI) AES-10-1991 (ANSI S4.43-1991)* specifying and controlling the requirements for digital interconnection between multitrack recorders and mixing consoles. The standard provides for 56 simultaneous digital audio channels which are conveyed point-to-point on a single coaxial cable fitted with BNC connectors along with a separate synchronization signal. Fiber optic implementation is specified in document AES-10id-1995, entitled *AES information document for digital audio engineering - Engineering guidelines for the multichannel audio digital interface (MADI) AES 10*. Basically, the technique takes the standard *AES/EBU interface* and multiplexes 56 of these into one sample period rather than the original two.

magic "An art of converting superstition into coin. There are other arts serving the same high purpose, but the discreet lexicographer does not name them." -- Ambrose Bierce.

magnitude 1. *Mathematics.* a. A number assigned to a quantity so that it may be compared with other quantities. b. A property that can be quantitatively described, such as the volume of a sphere, the length of a vector, or the value of a voltage or current waveform.

Maine The only American state whose name is just one syllable.

master mic *Teleconferencing*. Term referring to the microphone input on an automatic mic mixer that is the last to detect audio. A last-on mic becomes a master mic only if left open long enough.

master port *Teleconferencing*. Term referring to the audio input port that is the last to detect audio.

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 (and as big as a Volkswagen). Also see: mix-minus.

matrix switcher See router.

maximally flat magnitude response See: Butterworth crossover

maximally flat phase response See: Bessel crossover

MAU (*multistation access unit*) See token ring.

Mbps (*million bits per second*) (*always lower-case b*) A popular measure of transmission speed, but should be *Mibps*, or *mebi bits per second*. See: mebi

MBps (*million bytes per second*) (*always upper-case B*) A popular measure of transmission speed, but should be *MiBps*, or *mebi bytes per second*. See: mebi

MD (*MiniDisc*) Trademark term for the Sony digital audio recordable optical storage system utilizing data compression to reduce disc size.

MDM (*modular digital multitrack*) Generic term used to describe any of the families of digital audio multitrack recorders. The most common examples being the Alesis ADAT series and the Tascam DA-88 series.

mebi *Symbol* **Mi** New term standardized by the IEC as Amendment 2 to IEC 60027-2 *Letter Symbols to be Used in Electrical Technology* to signify binary multiples of 1,048,576 (i.e., 2E20). Meant to distinguish between exact binary and decimal quantities, i.e., 1,048,576 verses 1,000,000. For example, it is now 16 mebibits, abbreviated 16Mib, not 16 megabits or 16Mb.

media converter or **media manager** The *ability* to manage and the *process* of managing different media (coaxial cable, twisted-pair cable, and fiber-optics cable) used within the same network. Media management involves cable performance monitoring, cable break detection, planning for cable routes, as while as converting data signals between the various media.

medical conferencing See telemedicine.

medium 1. In telecommunications, the transmission path along which a signal propagates, such as a twisted-pair, coaxial cable, waveguide, fiber optics, or through water, or air. 2. The material on which data are recorded, such as plain paper, paper tapes, punched cards, magnetic tapes, magnetic disks, or optical discs.

mega- 1. A prefix signifying one million (10E6). abbreviated **M**. 2. A prefix popularly used in computer work to signify multiples of 1,048,576 (i.e., 2E20), but should use *mebi*. .

megabyte Popular term meaning a million bytes but should be mebibytes. See mebi

megaflops See: **MFLOPS**

MFLOPS (pronounced "mega-flops") ***(million floating point operations per second)*** A measure of computing power.

MI (*musical instrument*) A broad term used to describe the musical instrument marketplace in general. Reference is made to "the MI market," or to a specific "MI store." If a store sells band instruments, for instance, it is an *MI store*.

mic-level See levels.

micro- Prefix for one millionth (10E-6), abbreviated **µ**.

microbar 1. A unit of pressure equal to one millionth of a bar. 2. A really small place to have a beer.

microcontroller See: **microprocessor**

microprocessor An integrated circuit that performs a variety of operations in accordance with a list of instructions. The core of a microcomputer or personal computer, a one chip computer.

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.

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

MIDI show control A term originally created by Charlie Richmond (Richmond Sound Design) to describe a new form of MIDI control designed for live theater venues. His efforts resulted in the official MIDI Show Control (MSC) specification. This document states: "The purpose of MIDI Show Control is to allow MIDI systems to communicate with and to control dedicated intelligent control equipment in theatrical, live performance, multi-media, audio-visual and similar environments." The magazine TCI has posted a great review article (March 1997) on MSC titled "MIDI Show Control -- Five Years Later."

military music "Military justice is to justice what military music is to music." -- Groucho Marx [*from* Barber]

milli- Prefix for one thousandth (10E-3), abbreviated **m**.

Minifon An early portable dictating machine developed in the 1950s using wire recorder technology. An example of "dead recording media."

minimum-phase filters *Electrical circuits* From an electrical engineering viewpoint, the precise definition of a minimum-phase function is a detailed mathematical concept involving positive real transfer functions, i.e., transfer functions with all *zeros* restricted to the left half *s-plane* (complex frequency plane using the Laplace transform operator *s*). This guarantees unconditional stability in the circuit. For example, all equalizer designs based on 2nd-order bandpass or band-reject networks have minimum-phase characteristics. *Acoustics* A term used to mean a linear phase (or phase linear *European*) system. See: group delay

MIPS (*million instructions processed per second*) A measure of computing power.

mix-minus A specialized matrix-mixer where there is one output associated with each input that includes all other inputs *except* the one it is associated with. (The output is the complete *mix, minus* the one input.) In this manner, the simplest mix-minus designs have an equal number of inputs and outputs (a *square* matrix). For example, if there were 8-inputs, there would be 8-outputs. Each output would consists of a mix of the seven other inputs, but not its own. Therefore Output 1, for instance, would consist of a mix of Inputs 2-8, while Output 2 would consist of a mix of Inputs 1 & 3-7, Output 3 would consist of a mix of Inputs 1,2 & 4-7, and so on. Primary usage is large conference rooms, where it is desirable to have the loudspeaker closest to each microphone *exclude that particular microphone*, so as to reduce the chance of feedback.

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*.

MLP (*Meridian Lossless Packing*) A lossless audio coding scheme developed by Meridian Audio Ltd.. MLP has been selected as the optional coding scheme for use on DVD-Audio, as well as other transmission, storage and archiving applications. It is a true lossless coding technology, in that the recovered audio is bit-for-bit identical to the original. Unlike perceptual or lossy data reduction, MLP does not alter the final decoded signal in any way, but merely "packs" the audio data more efficiently into a smaller data rate for transmission or storage. It is simple to decode and requires relatively low computational power for playback.

MLS (*maximum-length sequences*) A time-domain-based analyzer using a mathematically designed test signal optimized for sound analysis. The test signal (a *maximum-length sequence*) is electronically generated and characterized by having a flat energy-vs-frequency curve over a wide frequency range. Sounding similar to white noise, it is actually periodic, with a long repetition rate. This test signal is most often tailored to be pink noise, as the preferred response for fractional octave analysis. Similar in principle to impulse response testing - think of the *maximum-length sequence* test signal as a series of randomly distributed positive- and negative-going impulses. See: **MLSSA**

MLSSA (pronounced "Melissa") **(*maximum-length sequences system analyzer*)** Trademarked name for the first MLS measurement instrument designed by DRA Laboratories (Sarasota, FL). Maximum-length-sequences methods were used for room impulse response measurement by M.R. Schroeder in 1979 (based on work dating back to the mid-60's); however, it was not until 1987 that the use of MLS became commercially available. The first MLS instrument was developed and made practical by Douglas Rife, who described the principles in his landmark paper (co-authored by John Vanderkooy, University of Waterloo) "Transfer-Function Measurement with Maximum-Length Sequences" (*J. Audio Eng. Soc.*, vol. 37, no. 6, June 1989), and followed up with new applications described in "Modulation Transfer Function Measurement with Maximum-Length Sequences" (*J. Audio Eng. Soc.*, vol. 40, no. 10, October 1992).

MMA (*MIDI Manufacturers Association*) The original source for information on MIDI technology, where companies work together to create the standards upon which MIDI compatibility is built.

MMCD (*multimedia compact disc*) See: DVD

MMVF (*multimedia video file*) See: DVD

modem (*modulator-demodulator)* A peripheral device used to convert digital signals ("1s" and "0s") into analog signals (tones) and vice-versa, necessary for communication using standard telephone lines.

mojo 1. A charm or amulet thought to have magic powers. 2. *Slang*: power, luck, etc., as of magical or supernatural origin. 3. Mojo Series™ Rane Corporation trademark for their series of economical products designed for high quality performance and reliability aimed at the working musician. 4. Abbr. *Mother Jones* magazine, or reference to their Internet news network: The Mojo Wire.

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

mono 3-way, etc. See: active crossover

monotonic *Mathematics.* Designating sequences, the successive members of which either consistently increase or decrease but do not oscillate in relative value. Each member of a monotone increasing sequence is greater than or equal to the preceding member; each member of a monotone decreasing sequence is less than or equal to the preceding member.

month One of the words in the English language without a rhyme -- some others are "orange," "purple" and "silver."

Moore's Law 1. Named after *Gordon Moore*, a cofounder of Intel, who wrote in an *Electronics* magazine article in 1965, that computer chip complexity would double every twelve months for the next ten years. Ten years later his forecast proved to be correct. At that time, he then predicted that the doubling would happen every *two* years for the *next* ten years. Ten years later, he was, once again, proved correct. By combining the two predictions, *Moore's Law* is often stated as a doubling every 18 months. 2. The dictum that requires you to buy a new computer every two years. [Thanks DC.]

MOR (*magneto-optical recording*) An erasable optical disc system using magnetic media and laser reading/writing.

motional feedback See: servo-loop

MP3 (*MPEG-1, layer 3*) A type of digital audio compression popularized for transmitting songs over the Internet. MP3 allows realtime audio streaming for Internet encoding and downloading. MP3 files are identified by the suffix ".MP3" Typically MP3 compresses CD-quality audio down to about one minute per 1MB file size. Also see Wired magazine's MP3 site.

MPEG (*Moving Picture Experts Group*) A working group within SMPTE who set, among other things, specifications for compression schemes for audio and video transmission. A term commonly used to make reference to their image-compression scheme (MPEG-2) for full motion video.

MPGA (*Music Producers Guild of the Americas*) A professional guild for music producers and audio recording engineers.

MS-DOS® (*Microsoft® disk operating system*) Microsoft's registered trademark for their PC operating system.

MSB (*most significant bit*) The bit within a digital word that represents the biggest possible single-bit coded value.

MSPS (*million samples per second*) A measurement of data converter speed.

multi-denomial transpedance informer Term coined by Jensen Transformers for their mythical product, the JE-EP-ERs, first introduced in 1987, which almost changed the whole audio transformer industry. The Jensen JE-EP-ERs pioneered the use of triple electonomic shielding and intrinsic eddy-breeding, until outlawed by Congress in 1988. Voluntarily discontinued when their stock of zeta-metal ran out, preventing any further use of interstage transpedance informance. Considered by many to be the only necessary accessory when coupling a Rane PI 14 Pseudoacoustic Infector to a Crown Belchfire® BF-6000SUX amplifier for playback using an Electro-Voice Rearaxial Softspeaker.

multimedia Generally refers to personal computers capable of multiple forms of communication methods. These constitute a minimum combination of stereo audio, video, text, and graphics, plus the more complex system includes fax and telephony provisions.

multiplex To interleave two or more signals into a single output; a process of selecting one of a number of inputs and switching its information to the output.

multipoint conference Telecommunication term referring to conferencing between three or more sites.

MUSICAM (*masking pattern adapted universal sub-band integrated coding and multiplexing*) A flexible bit rate reduction standard for high quality audio. Jointly developed for digital audio broadcast by CCETT in France, IRT in Germany and Philips in the Netherlands.

music vs. noise 1. "The sensation of a musical tone is due to a rapid periodic motion of the sonorous body; the sensation of a noise to non-periodic motion." from *On the Sensation of Tone* (1862) Hermann Helmholtz. 2. "Of all noises, I think music is the least disagreeable." -- Samuel Johnson [*from* Barber]

mute A control found on recording consoles, some mixers, and certain signal processing units that silences (*mutes*) a signal path, or output. Various uses.

Muzak™ (*music + Kodak*) 1. Trademark of the business music company founded in 1928 by General George Owen Squier who patented the transmission of background music (phonograph records played through the telephone system). He created the name by merging the word "music" with that of his favorite high-tech venture, the *Eastman Kodak Company*. The word "Kodak" was coined by Eastman himself, and in 1888 he first registered it as a trademark. According to Eastman, he invented it out of thin air. He explained: "I devised the name myself. The letter "K" had been a favorite with me - it seems a strong, incisive sort of letter. It became a question of trying out a great number of combinations of letters that made words starting and ending with 'K.' The word 'Kodak' is the result." 2. "I worry that the person who thought up Muzak may be thinking up something else." -- Lily Tomlin [*from* Barber]

N

NAB (***National Association of Broadcasters***) A professional trade organization for people working in the radio and television industry.

NAMM (***National Association of Music Merchants***) A professional trade organization for people working in the music business -- primarily in retailing and manufacturing of music products. "The International Music Products Association."

nano- A prefix for one billionth (10E-9), abbreviated **n**.

NARAS (***National Academy of Recording Arts & Science***) See *The Recording Academy*.

NARM (***National Association of Recording Merchandisers***) An industry organization made up primarily of music retailers acting as an advocate body for the common interests of merchandisers and distributors of music to industry and public policy makers.

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 designs.

N curve (***normal curve***) Same as Academy curve

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 1500Hz. 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.

near end Telecommunication term referring to your end; the local room, as opposed to the far end.

near-end crosstalk Crosstalk that is propagated in a disturbed channel in the *direction opposite to the direction of propagation of the signal* in the disturbing channel. The terminals of the disturbed channel, at which the near-end crosstalk is present, and the energized terminal of the disturbing channel, are usually near each other.

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.

negative feedback The act of comparing a fraction of the output signal to the input signal at the input to an amplifier in such a way that the amplifier will keep this fraction of the output signal always exactly the same as the input signal. Negative feedback is of prime importance in designing with opamps and audio power amplifiers. As applied to audio amplifiers, negative feedback is first attributed to Bell Labs scientist Harold S. Black, as described in the *Bell Labs Technical Review, 1934*.

network Generally used to mean a multi-computer system (as opposed to a single computer bus-type system) where multiple access is allowed from more than one computer at a time. Characterized by full two-way (duplex) communications between all equipment and computers on the network. See CobraNet for an example

network glossary See CobraNet's glossary for many useful terms.

nibble A group of four bits or half a byte (8-bits).

noise criterion (NC) curves See NC curves

noise floor Normally the lowest threshold of useful signal level (although sometimes audible signals below the noise floor may 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.

noise measurement filters See: weighting filters

noise reduction See: expander

noise shaping A technique used in oversampling low-bit converters and other quantizers to shift (*shape*) the frequency range of quantizing error (noise and distortion). The output of a quantizer is fed back through a filter, and summed with its input signal. Dither is sometimes used in the process. Oversampling A/D converters shift much of it out of the audio range completely. In this case, the in-band noise is decreased, which allows low-bit converters (such as delta-sigma) to equal or out-perform high-bit converters (those greater than 16 bits). When oversampling is not involved, the noise still appears to decrease by 12 dB or more because it is redistributed into less audible frequency areas. The benefits of this kind of noise shaping are usually reversed by further digital processing.

NOM (***number of open mics***) An acronym believed first created in 1967, or 1968, by Bill Snow after he retired from Bell Labs and went to work at Altec Lansing Research. It's use was popularized by Dan Dugan, the father of the automatic microphone mixer and Altec Lansing, the manufacturer of his first design. In Dan's original design, the automatic mic mixer, like human operators, turned the gain down on unused mic channels and turned the gain up on active channels, all the while ensuring that the overall level remained roughly constant. As a rough approximation, each doubling of the number of open mics (NOM) cuts the gain by 3 dB, i.e., as more mics are opened up the mic mixer reduces overall gain. If not, as mics open and close, the reverberation and ambient noise fluctuates unacceptably. NOM attenuation techniques work to provide the gain, stability, and low noise qualities of a single open mic with the benifits of multiple mics. [*Historical Note: This concept was first written about by C.P. Boner & R.E. Boner, in their paper "The Gain of a Sound System" April 1969, reproduced in* Sound Reinforcement: An Anthology *(Audio Engineering Society, NY, 1978.*]

nominal This word has several definitions but the one of importance to pro audio is its engineering sense meaning: insignificantly small; trifling: *a nominal amount*. It does ***not*** mean *average* or *typical* as is so often seen.

NOMM (***number of open mics & mixers***) A term created by Rane Corporation extending the concept of NOM (above) to include multiple mixers as well as microphones. As used by Rane, it is NOM-like in that feedback stability is maintained, however, since large systems have mics across multiple mixers, Rane includes these mixers in the NOMM calculation. For example, in audio conferencing, when the chairman is speaking and someone else quickly answers "yes," coughs, or drops a pen, most mic mixers running in NOM mode are annoying because they reduce the level of the chairman mic just because someone else made a noise. The Rane NOMM approach avoids this annoyance by keeping the chairman's mic at the same gain while still allowing the interruption to be heard, yet at a reduced gain from its full gated level.

nonvolatile Refers to a memory device that does not lose its data when power is removed from the system.

normaling jacks See patchbay

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).

NPO 1.*Ceramic capacitors* Temperature coefficient designator meaning *negative-positive-zero*, i.e., the capacitance drifts negative and positive averaging zero. A marking meaning stable with temperature. 2. *Medicine* Abbreviation for *nil per os*, or *nothing by mouth*.

NSCA (***National Systems Contractors Association***) "Founded in 1980 as the National Sound Contractors Association, the NSCA underwent a name change in 1994 to better reflect the diversification found within the hi-tech industry of electronic systems. Rather than focusing solely on the installation of audio systems, today's innovative member companies of the NSCA expanded into other fields, including audio, video, intercom/paging, telecommunications, security/access control, and many others." [from NSCA website]

NSP (***native signal processing***) Intel-designed method of using a powerful microprocessor (like their Pentium CPU) for signal processing functions normally done by separate DSP chips. Not finding many backers.

NSSP (***National Standards Systems Network***) A Web-based service launched by ANSI, along with government and industry partners. A full search & sales service provides for locating and buying virtually any standard. More than 100,000 global standards are available. Over 25 standards groups provide technical specs for this database, including ISO. The EIA endorsed the project..

null modem cable Special wiring of an RS-232 cable such that a computer can talk to another computer without a modem (thus "null" modem). As a minimum, a null modem cable reverses pins 2 and 3 on a standard RS-232 cable - but other pins may also need changing and shorting together.

Nyquist frequency The highest frequency that may be accurately sampled. The Nyquist frequency is one-half the sampling frequency. For example, the theoretical Nyquist frequency of a CD system is 22.05 kHz.

O

¼" TRS or **¼" TS** See connectors.

object-oriented or **object-based programming** (Abbreviated **OOP**) A software technique in which a system program is expressed completely in terms of predefined things (objects), consisting of a set of variables and operations which can be performed on them, and the connections between objects.

octal A number system using the base-8, i.e., each digit can be any of 8 values, represented by the digits 0-7. Each octal digit can also be represented by a three-bit binary number (since 2E3 =8).

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.

ohm *Abbr.* **R**, (Greek upper-case *omega*). 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. [After **Georg Simon Ohm**.]

Ohm, **Georg Simon** (1789-1854) German physicist noted for his contributions to mathematics, acoustics, and the measurement of electrical resistance.

ohmage Misnomer. No such word. *Bad, bad, super bad*. Wrongfully used as a term for loudspeaker resistance. The correct term is impedance -- learn it; use it.

one-bit data converter Loose reference to any of the various data conversion schemes (e.g., delta-sigma, adaptive delta modulation, etc.) that use only one binary bit (i.e., levels 1 and 0) in the conversion and storage process.

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 = 2E1/3 = 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, 2E1/3 equals 10E1/10 (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. See: critical bands. Compare with: third-octave

OOP See: object-oriented

op amp (*operational amplifier*) An analog integrated circuit device characterized as having two opposite polarity inputs and one output, used as the basic building block in analog signal processing.

optical-fiber cable See fiber-optics.

optocoupler Any device that functions as an electrical-to-optical or optical-to-electrical transducer.

orange One of the words in the English language without a rhyme -- some others are "month," "purple," and "silver."

ordinate *Mathematics.* The plane Cartesian coordinate representing the distance from a specified point to the *x*-axis, measured parallel to the *y*-axis.

OSD (*on-screen display*) chip An integrated circuit providing all necessary functions for adding text to television or video monitor display screens.

OSI (*open system interconnection*) The only internationally accepted framework of standards for communication between different systems made by different vendors. The model originally developed by ISO describing computer communication services and protocols without making assumptions concerning language, operating systems or application issues. The main goal is to create an open systems networking environment where any vendor's computer system, connected to any network, can freely share data with any other computer system on that network

OTPROM (*one-time programmable read-only memory*) A redundant term, incorrectly used to mean PROM - a PROM, by definition, is a one-time device.

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

overs A term associated with A/D converters used to describe input signals exceeding the full scale range (0 dBFS). *Overs* indicators vary from simple single LEDs to elaborate calibrated digital meters. To be of genuine value the overs indicator, however displayed, must be based on reading the true digital code associated with the input level. It is important to distinguish between O dBFS and overs; they are not the same. O dBFS is the absolute highest voltage level that any particular A/D can convert. It produces the equivalent of a digital code consisting of all 1s. No digital level can exceed 0 dBFS. A 0 dBFS voltage level *and all levels greater than this* produce the same output code of all 1s. A true overs indicator actually counts the number of times that the 0 dBFS level was exceeded and displays this number. As yet there is no standard as to how many samples exceeding 0 dBFS constitutes an over. Everyone agrees that very brief excursions beyond 0 dBFS (producing digital clipping) cannot be heard; however no such agreement exists as to just how many samples it takes before an over is audible.

oversampling 1. Sampling at a rate higher than the sampling Nyquist theorem. 2. A technique where each sample from the data converter is sampled more than once, i.e., *oversampled*. This multiplication of samples permits digital filtering of the signal, thus reducing the need for sharp analog filters to control aliasing.

Oz From Frank Baum's "The Wizard of Oz," the name was created when he looked at his filing cabinet and saw "A-N," and "O-Z," hence "Oz."

P

p's and q's From old British saying: *pints and quarts* [... and you know pints and quarts of what.] 1. Socially correct behavior; manners. 2. The way one acts; conduct: *was told to watch his p's and q's or he would be fired.*

PA-232 An RS-232-based variant of the PA-422 AES standard.

PA-422 A pro audio implementation of Electronics Industries Association EIA-422 interconnection standard, defined and adopted by the Audio Engineering Society as *AES Recommended practice for sound-reinforcement systems - Communications interface (PA-422) AES 15-1991 (ANSI S4.49-1991)*.

pad See attenuator pad.

PAL® (*programmable array logic*) Original registered trademark of Monolithic Memories Inc. (now owned by Advanced Micro Devices, Inc.) for their fuse-link once-programmable logic parts that have a programmable AND array, but a predefined OR array. See also: PLA, PLD & FPGA.

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.

PAQRAT® A registered trademark of Rane Corporation for their recording converter devices, RC 24T & RC 24A, that convert AES/EBU stereo 18-24 bit digital audio two track data into 16-bit compatible four tracks for recording and playback on 1st-genertion 16-bit modular digital multitrack tape machines such as Alesis ADAT and Tascam DTRS (DA-88) models.

paragraphic See: parametric equalizer

parallel interface The printer port in the PC world. A parallel port conforming to the quasi-standard called the Centronics Parallel Standard (there is no EIA standard). Originally a 36-pin connector, now more often a D-25 type connector. A parallel (as opposed to *serial*) interface transfers all bits in a word simultaneously. See also: serial interface.

parametric equalizer 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.

parity A redundant error detection method in which the total number of binary 1's (or 0's) is always made even or odd by appending one or more bits.

pascal *Abbr.* **Pa** A unit of pressure equal to one newton per square meter. [After **Blaise Pascal**.]

Pascal, **Blaise** (1623-1662) French philosopher and mathematician. Among his achievements are the invention of an adding machine and the development of the modern theory of probability.

passband The range of frequencies passed by an audio low-pass, high-pass or bandpass filter. Normally measured at the *-3 dB point*: the frequency point where the amplitude response is attenuated 3 dB (decibels) relative to the level of the main passband. For a bandpass filter two points are referenced: the *upper* and *lower* -3dB points. The -3dB point represents the frequency where the output *power* has been reduced by one-half. [Technical details: -3dB represents a multiplier of 0.707. If the voltage is reduced by 0.707, the current is also reduced by 0.707 (ohms law), and since power equals voltage-times-current, 0.707 times 0.707 equals 0.5, or *half-power.*]

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

passive equalizer A variable equalizer requiring no power 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.

PC (*personal computer*) Original term coined by IBM to describe their first personal computers; now used to mean all IBM-compatible personal computers, or any personal computer.

PC-Card See: PCMCIA

PC-DOS® (*personal computer disk operating system*) IBM's trademarked acronym for their PC operating system. If PC-DOS runs on an IBM *compatible,* it is then called *MS-DOS*.

PCI (*peripheral component interconnect*) Intel-designed high performance CPU interconnect strategy for "glueless" I/O subsystems. A 32- or 64-bit local-bus specification, characterized by being self-configuring, open, high-bandwidth and processor-independent - allowing for modular hardware design.

PCM (*pulse code modulation*) A conversion method in which digital words in a bit stream represent samples of analog information. The basis of most digital audio systems.

PCMCIA (*Personal Computer Memory Card International Association*) 1. The association and first name given to the standardized credit-card size packages (aka *smart cards*) for memory and I/O (modems, LAN cards, etc.) for computers, laptops, palmtops, etc. Nicknamed *PC-Card*, which is now the preferred term. 2. Popularly believed to stand for *People Can't Memorize Computer Interface Acronyms*.

PDA (*personal digital assistant*) A small palmtop-like computer designed for specific tasks such as a pocket calculator. Other examples include personal electronic diaries, memo takers, communicators, web browsers, dictionary-translators, etc. Apple's *Newton* is a PDA. IBM refers to theirs as *personal communicators.*

peaking response Term used to describe a bandpass shape when applied to program equalization.

peak program meter See: PPM

peer-to-peer A network term popularly used to mean an equal access network where every node can send/receive data at any time without waiting for permission, i.e., each node can act as a client or server. An example would be a group of computers that communicate directly with each other, rather than through a central server.

period *Abbr.* **T, t** 1. The period of a periodic function is the smallest time interval over which the function repeats itself. [For example, the *period* of a sine wave is the amount of time, T, it takes for the waveform to pass through 360 degrees. Also, it is the reciprocal of the frequency itself: i.e., T = 1/f.] 2. *Mathematics.* a. The least interval in the range of the independent variable of a periodic function of a real variable in which all possible values of the dependent variable are assumed. b. A group of digits separated by commas in a written number. c. The number of digits that repeat in a repeating decimal. For example, 1/7 = 0.142857142857... has a six-digit period.

peripheral Equipment physically independent of, but which may interface to a computer or a controller.

PFC (*power-factor-corrected*) See power-factor-corrected.

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*. Compare: AFL

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.

phase delay A phase-shifted sine wave appears displaced in *time* from the input waveform. This displacement is called *phase delay*.

phase linear Chiefly European phrase meaning "linear phase." Any system which accurately preserves phase relationships between frequencies, i.e., that exhibits pure delay. See: group delay

phase lock loop A circuit for synchronizing a variable local oscillator with the phase of a transmitted signal. The circuit acts as a phase detector by comparing the frequency of a known oscillator with an incoming signal and then feeds back the output of the detector to keep the oscillator in phase with the incoming frequency. Commonly used for bit-synchronization.

phaser Also called a "phase shifter," this is an electronic device creating an effect similar to flanging, but not as pronounced. Based on phase shift (*frequency dependent*), rather than true signal delay (*frequency independent*), the phaser is much easier and cheaper to construct. Using a relatively simple narrow notch filter (all-pass filters also were used) and sweeping it up and down through some frequency range, then summing this output with the original input, creates the desired effect. Narrow notch filters are characterized by having sudden and extreme phase shifts just before and just after the deep notch. This generates the needed phase shifts for the ever-changing magnitude cancellations.

phase shift The fraction of a complete cycle elapsed as measured from a specified reference point and expressed as an angle. **out of phase**. In an un-synchronized or un-correlated way. See: polarity

phasor 1. A complex number expressing the magnitude and phase of a time-varying quantity. It is math shorthand for complex numbers. Unless otherwise specified, it is used only within the context of steady-state alternating linear systems. [Example: 1.5 /27° is a phasor representing a vector with a magnitude of 1.5 and a phase angle of 27 degrees.] 2. For some unknown reason, used a lot by Star Fleet personnel.

phlogiston A hypothetical substance formerly thought to be a volatile constituent of all combustible substances released as flame in combustion. See: smoke

Phoenix-blocks (or **-connectors** or **-strips)** A term, becoming generic, meaning disconnectable, or pluggable terminal blocks, after Phoenix Contact connector company, although dozens of companies make them. Also called *Euroblocks*. See connectors.

phon A unit of apparent loudness, equal in number to the intensity in decibels of a 1,000 Hz tone judged to be as loud as the sound being measured.

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

phonograph "An irritating toy that restores life to dead noises." -- Ambrose Bierce.

phono jack Same as *RCA*, see connectors.

pi *Symbol* (Greek lower-case *pi*) 1. *Mathematics.* A transcendental number, approximately 3.14159, represented by the Greek lower-case pi symbol, that expresses the ratio of the circumference to the diameter of a circle and appears as a constant in many mathematical expressions. 2. *Filters.* Equal to 180 degrees or integral multiples thereof.

PI 14 See: Pseudoacoustic Infector

pico- Prefix for one trillionth (10E-12), abbreviated **p**.

pin jack Same as *RCA*, see connectors.

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. Pink noise is created by passing white noise through a filter having a 3 dB/octave roll-off rate. 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*.

pixel (*picture element*) The smallest element on a display surface, like a video screen, that can be assigned independent characteristics.

PLA (*programmable logic array*) A programmable logic device in which both the AND & OR arrays are programmable.

PLD (*programmable logic device*) The generic name for an integrated circuit offering a vast array of logic function building blocks that the circuit designer defines (programs) to interconnect for specific applications.

plenum 1. A ductwork system in which air is at a pressure greater than that of the outside atmosphere. 2. Such a system located in the space above a suspended ceiling, used to circulate air back to a building's HVAC.

plenum cable The type of cable used when smoke retardant properties are required. Plenum cable is specifically designed for use in a plenum area (see above) which is typically used as the distribution system in buildings. Most cities requiring all cable ran through a plenum ceiling to be *plenum cable* which has insulated conductors jacketed with PVDF (*polyvinylidene difloride*) -- a material providing low flame spread and low smoke producing properties. Plenum cables are approved by Underwriters Laboratories for non-conduit applications located in environmental air spaces. This low cost alternative has replaced traditional conduit use in many commercial installations.

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.]

post-echo See: print-through

POTS Acronym for *plain-old telephone system*. The normal single line basic telephone service. Often used in reference to modems associated with regular telephone lines.

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).

power factor *Abbr.* **PF** *Electronics*. The ratio of the total power in watts (resistive load) to the total apparent power in voltamperes (VA) (reactive load). The difference between *watts* and *VA* is due to reactive load impedance. Apparent power equals watts only for a purely resistive load (i.e., zero degrees phase shift between the applied voltage and the resultant current). Power factor is best thought of intuitively as the multiplier (ranging between 0 and 1) that you must use to obtain the real power from the apparent power. For example if you measure the rms voltage and current of a circuit and multiple them together you obtain the apparent power, but you must multiple this value by the power factor to obtain the real power. If the load is purely resistive then the phase difference between the voltage and current will be zero and the power factor will be one, and the apparent power will equal the true power -- but only for a resistive load. For a reactive load (any load with inductive and/or capacitive reactance, i.e., any *real* load) there will be a phase difference between the voltage and the current due to the phase delay introduced by the reactive elements. Simply put, since the maximum voltage and current do not occur at the same instant of time the amount of power developed is less than the measured rms voltage and current multiplied together.

Since power factor is a ratio, and hence unitless, it can be expressed in several ways -- all of them equal. It is the ratio of watts to voltamperes, of resistance to reactance, and if the phase shift in degrees is known (*phase angle>*), it is the cosine of that angle, or ***cos Æ***. If the angle is zero the PF = 1, and if the angle is 90° the PF = 0.

power-factor-corrected (*PFC*) Any system that has a power-factor-correcting device, such as a capacitor, installed to reduce the phase difference between the rms voltage and rms current.

PowerPC A super powerful RISC processor PC jointly developed by IBM, Apple and Motorola, designed to run *any* PC operating system (MS-DOS, UNIX, Windows, OS/2, Mac OS. etc.). Featured in Apple's line of "PowerMac" computers.

PPM (*peak program meter*) An audio meter originally developed in Europe to accurately measure and display *peak* audio signals (as opposed to *average* audio signals; see VU meter). The PPM augments the VU meter and it is normal to find both in modern recording studios. The PPM is particularly valuable for digital audio recording or signal processing due to the critical monitoring required to prevent exceeding 0 dBFS and reducing overs. There are two standards: IEC 60268-10 for analog meters and IEC 60268-18 for digital meters. [*These are available to buy on the IEC website*.] An interesting aspect of PPM design is that rather than respond instantaneously to peaks, they require a finite 5 ms integration time, so that only peaks wide enough to be audible are displayed. IEC 60268-10 translates this into a response that is 1 dB down from steady-state for a 10 ms tone burst, 2 dB down for a 5 ms burst, and 4 dB down for a 3 ms tone burst -- requirements satisfied by an attack time constant of 1.7 ms. The IEC specified decay rate of 1.5 seconds to a -20 dB level can be met with a 650 ms time constant.

precedence effect See: Haas Effect

pre-echo See: print-through

pre-emphasis A high-frequency boost used during recording, followed by de-emphasis during playback, designed to improve signal-to-noise performance.

print-through The name for the magnetic tape recording phenomena where the act of layering, or winding layer upon layer of tape causes the flux from one layer to magnetize the adjacent layer, thus *printing through* from one layer onto another layer. Also called *crosstalk* or *interlayer transfer*. The most vulnerable parts of the magnetic tape are the blank spots, particularly leaders and spaces between material that happen to occur adjacent to loud passages. Two other terms come from print-through: on layers played back *before* loud passages it gives a *pre-echo*, whereas on playback *following* the loud passage it gives a *post-echo*.

Programming, Law of The law states that every program contains at least one bug. The law further states that every program can be shortened by at least one instruction. Therefore, the law concludes, every program can be reduced to one instruction that does not work. The law is not wrong. [*Thanks TP*.]

PROM (*programmable read-only memory*) A memory device whose contents can be electrically programmed (once) by the designer.

proof "Evidence having a shade more of plausibility than of unlikelihood. The testimony of two credible witnesses as opposed to that of only one." -- Ambrose Bierce.

propagation The motion of waves through or along a medium. For electromagnetic waves, propagation may occur in a vacuum as well as in material media.

proportional-Q 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.

protocol A specific set of rules, procedures or conventions relating to format and timing of data transmission between two devices. A standard procedure that two data devices must accept and use to be able to understand each other.

Pseudoacoustic Infector Term coined by Rane Corporation for their mythical product, the PI 14, first introduced in 1988, which *almost* caught the attention of the music industry. An acoustic stimulator designed to add a little bit of *This* and a little bit of *That* to recordings, to give them a sense of *Now* previously unobtainable. Rane's PI 14 introduced a unique *Here-to-There* (*and-Back-Again*) pan control. Transformer operation required the Jensen JE-EP-ERs when coupling directly into a Crown Belchfire® BF-6000SUX for playback through an Electro-Voice Rearaxial Softspeaker. Today, PI 14s are considered quite scarce and highly collectable.

psophometric See: weighting filters

psychoacoustics The scientific study of the perception of sound. Called "the music of science" by Roederer.

punch-in/punch-out *Recording studio*: To engage/disengage record mode on a track previously recorded, usually for purposes of correcting unwanted segments.

purple One of the few words in the English language without a rhyme -- some others are "month," "orange" & "silver."

PVC cable (*polyvinyl chloride*) The most common type of cable used when smoke retardant properties are not required, i.e., when a building's HVAC system is run through metal ducts - not open ceilings. This cable is sheathed in PVC, the standard jacketing of most electrical cable. PVC is a tough water and flame retardant material, but is not smoke retardant. If PVC catches fire, it emits noxious gases, and if the cable is run in a plenum area, the deadly gases can be dispersed throughout the building.

PWM (*pulse width modulation*) A conversion method in which the widths of pulses in a pulse train represent the analog information.

Q

q (lower-case) *Physics.* The symbol for charge.

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

QSound The name of a Canadian company and its proprietary and patented 3D sound technology. Designed for two channel playback systems, QSound finds success in the computer and arcade game markets, as well as movie theaters. Using advanced signal processing techniques, QSound adds localization cues to the original material. Since loudspeakers and headphones create quite different playback environments, different algorithms exist for each. QSound allows the music producer to locate specific sound events in virtual positions outside the physical locations of the two loudspeakers. The effect is primarily one of widening the sound field. QSound works best when the listener is positioned in the sweet spot located equidistant between the speakers.

quad flat pack The most commonly used package in surface mount technology to achieve a high lead count in a small area. Leads are brought out on all four sides of a thin square package.

quad mic cable See cables.

quantization error Error resulting from quantizing an analog waveform to a discrete level. In general the longer the word length, the less the error.

quantization The process of converting, or digitizing, the almost infinitely variable amplitude of an analog waveform to one of a finite series of discrete levels. Performed by the A/D converter.

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

quasi-balanced line See: floating unbalanced line

quasi-parametric See: parametric equalizer

R

radian *Mathematics.* A unit of angular measure equal to the angle subtended at the center of a circle by an arc equal in length to the radius of the circle, approximately 57°17'44.6 *Filters.* Frequency is measured in radians/second. One cycle (360°) equals 2 pi radians.

radicalism "The conservatism of tomorrow injected into the affairs of today." -- Ambrose Bierce.

rail-switcher A term used to describe audio power amplifier designs utilizing more than one power supply for the output, and a means of switching between them based upon the input signal. This scheme improves efficiency. See: Class G Amplifiers and compare with tracking power amplifiers.

RAM (*random access memory*) A memory device in which data may be read out and new data written into any address or location.

RaneNotes A series of technical notes written by Rane's technical staff.

RaneWare® A registered trademark of Rane Corporation used to identify Rane software products -- not something to keep you dry.

RAQ (*rarely asked questions*) The really important questions that should be asked, but never are. The answers to RAQs are kept hidden within government and corporate walls.

rarefaction 1. A decrease in density and pressure in a medium, such as air, caused by the passage of a sound wave. 2. The region in which this occurs.

RCA jack See connectors.

R-DAT or DAT (*rotary head digital audio tape recorder*) A digital audio recorder utilizing a magnetic tape cassette system similar to that of a video recorder.

reactance The imaginary part of an impedance (see).

real-time operation What is perceived to be instantaneous to a user (or more technically, processing that completes in a specific time allotment).

rearaxial softspeaker Term coined by Electro-Voice for their mythical loudspeaker, the SP13.5TRBXWK. Claimed by many to be the speaker that couldn't be made, it might have changed all future loudspeaker design, but it didn't. Characterized by being undirectional, the designer's claimed it produced silken highs and woolen lows. The only loudspeaker known to incorporate both "presence" *and* "absence" controls. Based on a ridiculously simple principle that still cannot be explained, the SP13.5TRBXWK was only heard once, during the Rane demo of their PI 14 Pseudoacoustic Infector, coupled by a Jensen JE-EP-ERs Multi-denomial Transpedance Informer to a Crown Belchfire BF-6000SUX amplifier. No one survived.

reconstruction filter A low-pass filter used at the output of digital audio processors (following the DAC) to remove (or at least greatly attenuate) any aliasing products (image spectra present at multiples of the sampling frequency) produced by the use of real-world (non-brickwall) input filters.

Recording Academy, The The organization formerly known as NARAS. Think *Grammy*; often confused with SPARS.

recording console See: mixer

recording terminology See: Recording Institute of Detroit, who claims to have posted the largest glossary of recording terms on the web.

recursive A data structure that is defined in terms of itself. For example, in mathematics, an expression, such as a polynomial, each term of which is determined by application of a formula to preceding terms. Pertaining to a process that is defined or generated in terms of itself, i.e., its immediate past history.

reflectors In acoustics, an object or surface that reflects, or bounces back the original signal. A perfect reflector would reflect with no loss of energy. A diffuser is a special kind of reflector.

resistance See impedance

reverberation time also **RT_{60}** *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 "RT_{60}."

rhythm The only English language word containing two syllables with no natural vowels. [*Thanks GS.*]

RIAA (*Recording Industry Association of America*) A professional trade organization representing the U.S. recording industry. RIAA® members create, manufacture and/or distribute approximately 90% of all sound recordings produced and sold in the United States.

RIAA equalization curve The standard first proposed by the RIAA (see above) and adopted by the disc recording industry in 1953, reaffirmed in 1964 by both the RIAA and NAB and issued as international standard IEC 60098 (old IEC 98) by the IEC, which remains in effect today. The curve is used in cutting vinyl records and its inverse is required in phono playback preamplifiers. The curve attenuates low frequencies and amplifies high frequencies (relative to a 1 kHz reference point) in order to achieve the maximum dynamic range for a lateral cut vinyl disc (as opposed to the older method of vertical cutting). The grooves in a stereo phonograph disc are cut by a chisel shaped cutting stylus driven by two vibrating systems arranged at right angles to each other. The cutting stylus vibrates mechanically from side to side in accordance with the signal impressed on the cutter. The resultant movement of the groove back and forth about its center is known as groove modulation. The amplitude of this modulation cannot exceed a fixed amount or "cutover" occurs. Cutover, or overmodulation, describes the breaking through the wall of one groove into the wall of the previous groove. Since low frequencies cause wide undulations in the groove, they must be attenuated to prevent overmodulation. At the other end of the audio spectrum, high frequencies must be amplified to overcome the granular nature of the disc surface acting as a noise generator, thus improving signal-to-noise ratio.

"Ring it up ... !" Phrase coined from the first cash registers that ran a bell for emphasis when the drawer opened, signifying the end of the calculation.

ring topology A network topology where all nodes are daisy chained together (connected) in a closed loop.

RISC (*reduced instruction set computer*) A computer design that achieves high performance by doing the most common computer operations very quickly, utilizing a high speed processing technology that uses a far simpler set of operating commands. Primarily found in workstations and PowerPCs. The alternative to *CISC (complex instruction set computing)*, the original way of doing computing.

RJ (*Registered Jacks*) As in red *RJ-12* modular telephone jacks used by Rane Corporation for external power supply connection.

RMS or **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.

ROM (*read-only memory*) A memory from which data, after initial storage, may only be read out, but new data cannot be written in. The normal audio CD is an example of a read-only system.

root mean square *Abbr.* **rms, RMS** *Mathematics.* The square root of the average of the squares of a group of numbers. A useful and more meaningful way of averaging a group of numbers.

rotary equalizer A multi-band variable equalizer using rotary controls as the amplitude adjustable elements. Both active and passive designs exist with rotary controls. Center frequency and bandwidth are fixed for each band.

router An audio device used to selectively assign any input to any output, including the ability to add inputs together. In this way, one input could go to all outputs, or all inputs could go to just one output, or any combination thereof. An *n x m matrix* forms the core of any router, where there are *n* inputs and *m* outputs. Typically, level controls are provided on all inputs and outputs; balanced and unbalanced designs exist. More elaborate designs are called matrix-mixers.

RPM (*Remote Programmable Multiprocessor*)™ Rane Corporation's trademark for their line of DSP multiprocessor-based digital audio signal processing devices.

RS (*Recommended Standard*) As in RS-232 serial interface standard, et al.

RS-232 The standard serial interface (*EIA/TIA-232-E*) used on most personal computers. A format widely supported for bidirectional data transfer at low to moderate rates. The most common interface method used to connect personal computers with peripheral hardware and instruments. Use is restricted to one peripheral at a time and short distances. The standard originally called for DB-25 connectors, but now allows the smaller DB-9 version.

RS-422 The standard adopted in 1978 by the Electronics Industry Association as *EIA-422-A, Electrical characteristics of balanced voltage digital interface circuits*. A universal balanced line twisted-pair standard for all long distance (~1000 m, or ~3300 ft) computer interconnections, daisy-chain style.

RS-485 The standard describing the electrical characteristics of a balanced interface used as a bus for master/slave operation. Allows up to 32 users to *bridge* onto the line (as opposed to RS-422's need to *daisy chain* the interconnections).

RS-490 The standard adopted in 1981 by the EIA entitled *Standard Test Methods of Measurement for Audio Amplifiers*. The power amp testing standard for consumer products.

RT_{60} See: reverberation time

RW 232™ (also RaneWare) A trademark of Rane Corporation used to identify Rane's RS-232-based variant of the PA-422 AES standard.

S

70 volt line See: constant-voltage

sabin A non-metric unit of sound absorption used in acoustical engineering. One sabin is the sound absorption of one square foot (or one square meter -- a *metric sabin*) of a perfectly absorbing surface--such as an open window. The sound absorption of a wall or some other surface is the area of the surface, in square feet, multiplied by a coefficient that depends on the material of the surface and on the frequency of the sound. These coefficients are carefully measured and tabulated. The unit honors Wallace Sabine (see below). Sabine used this unit, which he called the *open window unit (owu)*, as early as 1911. [*From Rowlett's How Many? A Dictionary of Units of Measurement*]

Sabine, Wallace Clement Ware (1868-1919) American physicist and Harvard University professor who founded the systematic study of acoustics around 1895. Regarded as the father of the science of architectural acoustics.

SACD® (*Super Audio CD®*) Also known as *DSD®* or *Direct Stream Digital®*, joint trademark of Sony and Philips for their proposal for the next generation CD-standard. Sony and Philips have split from the DVD ranks to jointly propose their own solution comprised of a 1-bit, 64-times oversampled direct-stream digital SACD format. The original SACD proposal was for a hybrid disc comprising two layers: a high density (HD) DSD layer in the middle, and a standard density CD layer at the bottom. The two layers are read from the same side of the disc; the CD laser reads the bottom reflective layer through the semi-transmissive HD layer, while the middle layer is read by the HD laser delivering high-quality, multichannel sound without sacrificing backward compatibility. The HD layer has three tracks: the innermost is for two-channel stereo; the middle is a six-channel mix; and the outer is for such additional information as liner notes, still images and video clips. Maximum playing time is 74 minutes. This proposal turned out to be too expensive, so the SACD first release is a single-layer SACD-only disc.

SAE (*Society of Automotive Engineers*) The international trade organization comprised of 80,000 engineers, business executives, educators, and students representing 100 countries that functions as *the* resource for technical information and expertise used in designing, building, maintaining, and operating self-propelled vehicles for use on land or sea, in air or space.

sample rate conversion The process of converting one sample rate to another, e.g. 44.1 kHz to 48 kHz. Necessary for the communication and synchronization of dissimilar digital audio devices, e.g., digital tape machines to CD mastering machines.

sample-and-hold (S/H) A circuit that captures and holds an analog signal for a finite period. The input S/H proceeds the A/D converter, allowing time for conversion. The output S/H follows the D/A converter, smoothing glitches.

Sampling (Nyquist)Theorem A theorem stating that a bandlimited continuous waveform may be represented by a series of discrete samples if the sampling frequency is at least twice the highest frequency contained in the waveform.

sampling frequency or **sampling rate** The frequency or rate at which an analog signal is sampled or converted into digital data. Expressed in Hertz (cycles per second). For example, compact disc sampling rate is 44,100 samples per second or 44.1 kHz, however in pro audio other rates exist: common examples being 32kHz, 48kHz, and 50kHz. [*Historical note re 44.1kHz vs. 44.056kHz: Since the first commercial digital audio recorders used a standard helical scan video recorder for storage, there had to be a fixed relationship between sampling frequency and horizontal video frequency, so these frequencies could be derived from the same master clock by frequency division. For the NTSC 525-line TV system, a sampling frequency of 44,055.94 Hz was selected, whereas for the PAL 625-line system, a frequency of 44,100 Hz was chosen. The 0.1% difference shows up as an imperceptible pitch shift.*]

sampling The process of representing the amplitude of a signal at a particular point in time.

SAR (*successive approximation register*) A type of analog-to-digital converter using a digital-to-analog converter to determine the output word successively, bit by bit.

SCMS (pronounced "scums") **(*serial copy management system*)** The copy protection scheme applied to consumer digital recording equipment - it does not apply to professional machines. This standard allows unlimited analog-to-digital copies, but only one digital-to-digital copy. This is done by two control bits (the *C* and *L* bits) contained within the digital audio data.

screeched The longest one-syllable word in the English language.

SCSI port (pronounced "scuzzy") **(*small computer system interface*)** A standard 8-bit parallel interface used to connect up to seven peripherals, such as connecting a CD-ROM player or document scanner to a microcomputer.

SD (*super density compact disc*) See: DVD

SDDS® (*Sony Dynamic Digital Sound*) Sony's competing format for the digital soundtrack system for motion picture playback. The signal is optically printed *outside* the sprocket holes, along both sides of the print. Sony recently developed a single camera system that records all three digital formats (Dolby Digital, DTS & SDDS) on a single inventory print, thus setting the stage for long term coexistence of all formats.

SDIF (*Sony digital interface format*) Sony's professional digital audio interface utilizing two BNC-type connectors, one for each audio channel, and a separate BNC-type connector for word synchronization, common to both channels. All interconnection is done using unbalanced 75 ohm coaxial cable of the exact same length (to preserve synchronization), and is not intended for long distances.

SDMI (*Secure Digital Music Initiative*) A multi-industry group defining a specification to protect digital music distribution.

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.

semitone *Music.* An interval equal to a half tone in the standard diatonic scale. Also called half step, half tone.

serial interface A connection which allows transmission of only one bit at a time. An example in the PC world is a RS-232 port, primarily used for modems and mice. A serial interface transmits each bit in a word in sequence over one communication link. See also: parallel interface.

serializer A parallel-to-serial data converter; used in buses and networks.

servo-loop; -locked loop; -mechanism A self-regulating feedback system or mechanism. Typically a feedback system consisting of a sensing element, an amplifier, and a (servo)motor, used in the automatic control of a mechanical device (such as a loudspeaker). In audio, usually the name applies to a class of electronic control circuits comprised of an amplifier and a feedback path from the output signal that is compared with a reference signal. This topology creates an error signal that is the difference between the reference and the output signal. The error signal causes the output to do whatever is necessary to reduce the error to zero. A loudspeaker system with *motional feedback* is such a system. A sensor is attached to the speaker cone and provides a feedback signal that is compared against the driving signal to create more accurate control of the loudspeaker. Another example is Rane's *servo-locked limiter™* which is an audio peak limiter circuit where the output is compared against a reference signal (the *threshold* setting) creating an error signal that reduces the gain of the circuit until the error is zero.

servo-locked limiter™ Rane Corporation trademark for their proprietary limiter circuit. See: servo-loop

Shannon, Claude E. (1916-) American mathematician and physicist who is credited as the father of information theory. In his master's thesis Shannon showed how an algebra invented by the British mathematician, George Boole in the mid-1800s could represent the workings of switches and relays in electronic circuits. His paper has been called "possibly the most important master's thesis in the century."

shelving response Term used to describe a flat (or shelf) end-band shape when applied to program equalization. Also known as *bass* and *treble* tone control responses.

show control See MIDI show control.

SI (*International System of Units*) The International System of Units, universally abbreviated SI (from the French *Le Système International d'Unités*), is the modern metric system of measurement. SI is the dominant measurement system not only in science, but also in international commerce. See link for a downloadable copy of Barry N. Taylor's *Guide for the Use of the International System of Units (SI)*. This free 86 page document is *the* definitive source of SI info.

sibilant *Linguistics. adj.* Of, characterized by, or producing a hissing sound like that of (s) or (sh): *the sibilant consonants; a sibilant bird call.* A sibilant speech sound, such as English (s), (sh), (z), or (zh).

SID (*slew-induced distortion*) See: DIM/TIM

side-chain In a signal processing circuit, such as one employing a VCA, a secondary signal path in parallel with the main signal path in which the condition or parameter of an audio signal that will cause a processor to begin working is sensed or detected. Typical applications use the side-chain information to control the gain of a VCA. The circuit may detect level or frequency or both. Devices utilizing side-chains for control generally fall into the classification of dynamic controllers.

sidetone *Telephony*. The feature of a telephone handset that allows you to hear yourself talk, acting as feedback that the phone is really working. Sidetones are actually short line echoes bled back into the earpiece. Too much sidetone sounds like an echo and too little sounds so quiet that people think the phone is broken. Sidetones are good for people but can cause acoustic feedback in teleconferencing systems if not treated properly.

sigma-delta See: delta-sigma modulation

signal levels Audio signal levels: see decibels.

signal-to-noise ratio See: S/N

silicon dust Nickname for microchips. Believed first coined by National Semiconductor to describe the world's smallest op amp (*as of May 5, 1999*), the LMV921. Used in surface mount technology (SMT), they are about the size of a single letter on this page.

silver One of the English language words without a rhyme -- others are "month," "orange" & "purple."

SINAD (pronounced "sin-add") or **S/N+D (*signal-to-noise and distortion*)** Acronym for the ratio: (signal + noise + distortion) / (noise + distortion). Or, as Metzler explains, it is the reciprocal of THD+N stated in decibels (dB). Originally developed for measuring FM receivers, it now also appears on A/D data sheets. Generally, the term "SINAD" is favored by the communication industry, while "S/N+D" is used by the audio industry, but they both mean the same thing. It is the preferred way to specify the dynamic range, or maximum S/N, since the noise and distortion products are measured in the presence of a signal. [*A signal is applied to the input, the output is passed through a notch filter to remove the signal and what remains is measured. Then the ratio of the rms value of the measured output signal to the rms value of everything else coming out (i.e., noise + distortion) is expressed in decibels.*] This gives a more accurate picture of real dynamic performance. Sometimes the measurement is stated for three reference levels of 0 dBFS, -20 dBFS, and -60 dBFS.

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.

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

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

sinusoid *Mathematics.* See: sine curve

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).

smoke From the *phlogiston* theory of electronics, it is smoke that makes ICs and transistors work. The proof of this is self-evident because everytime you let the smoke out of an IC or transistor it stops working -- elementary. This has been verified through exhaustive testing, particularly regarding power amplifier ICs and transistors. (Incidentally, wires carry smoke from one device to another.) [Origin unknown but classic.]

smoothing filter See: anti-imaging filter

SMPTE (pronounced "simty") **(*Society of Motion Picture and Television Engineers*)** A professional engineering society that establishes standards, including a time code standard used for synchronization.

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.

S/N+D or **S/(N+D)** See SINAD

SNMP (*Simple Network Management Protocol*) The most common method by which network management applications can query a management agent using a supported MIB (Management Information Base). SNMP operates at the OSI Application layer. The IP (Internet Protocol)-based SNMP is the basis of most network management software, to the extent that today the phrase "managed device" implies SNMP compliance.

snollygoster Defined in 1895 as "a fellow who wants office, regardless of party, platform or principles and who ... gets there by the sheer force of monumental talknophical assumancy". [McQuain, *Never Enough Words*]

solo A term used in recording and live-sound mixing to describe monitoring (via headphones) a single channel without affecting the main outputs (see: PFL) -- same as *cueing*; however, it can also refer to certain console designs where it replaces the main mix with the soloed channel (called *destructive solo*).

sone A subjective unit of loudness, as perceived by a person with normal hearing, equal to the loudness of a pure tone having a frequency of 1,000 hertz at 40 decibels sound pressure level.

sonorous 1. Having or producing sound. 2. Having or producing a full, deep, or rich sound.

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. See sound.

sound absorption See absorption.

sound off To express one's views vigorously: *He was always sounding off about his boss.*

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®.

Sound Recording History Fantastic site put together by David Morton.

SPARS (*Society of Professional Audio Recording Services*) Founded in 1979, a professional trade organization that unites the manufacturers of audio recording equipment and providers of services, with the users. Their goal is worldwide promotion of communication, education and service among all those who make and use recording equipment. Often confused with NARAS.

spatial Of, relating to, involving, or having the nature of space.

Spatializer A single-ended spatial enhancement technique developed by Desper Products, Inc., a subsidiary of Spatializer Audio Labs, Inc. Widely licensed in both the consumer audio and multimedia computing markets, the Desper, or *Spatializer* process is normally used as a postprocessor. The *Spatializer* technology manipulates the original signal in a way that causes the listener to perceive a stereo image beyond the boundaries of the two loudspeakers. It claims to place sounds in front of the listener in an arc of 180 degrees, with excellent imaging and fidelity.

S/PDIF (*Sony/Philips digital interface format*, also seen w/o slash as **SPDIF)** A consumer version of the AES3 (old AES/EBU) digital audio interconnection standard based on coaxial cable and RCA connectors.

spectra A plural of *spectrum*. In pro audio use, the distribution of frequency of a sound signal, especially: the distribution of sound energy, arranged in order of frequency wavelengths.

spell checker A software program used by word processors to tell you that the following truism has no spelling errors: "*Dew knot trussed yore spell chequer two fined awl mistakes.*"

spiff 1. To make attractive, stylish, or up-to-date: *spiffed up the the old storefront*. 2. Attractiveness or charm in appearance, dress, or manners: *"He may need more than spiff to get him through the bad patches ahead" James Wolcott* [Possibly from dialectal *spiff* well-dressed] 3. Giveaways (usually in the form of money) by manufacturers as added incentive ("make attractive") to personnel selling their goods. Compare with swag

spiral quad Same as *star quad*; see cables.

SPL controller See: leveler

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. See: distribution amplifier

spooler Comes from the acronym SPOOL derived from Simultaneous Peripheral Operation On-Line. A program or piece of hardware that controls a buffer of data going to some output device, including a printer or a screen. Spooling temporarily stores programs or program outputs on magnetic tape, RAM or disks for output or processing. *And you thought you were done learning for the day -- HA.*

SRS (*Sound Retrieval System*) A stereo image enhancement scheme invented by Arnold Klayman in the early '80s while working for Hughes Aircraft, and since 1993, marketed by SRS Labs, Inc. A standalone spatial enhancement scheme, SRS benefits from not requiring encoding of the signal, but thus prevents the audio producer from determining the location of individual sound effects. The results vary, being heavily dependent upon the original stereo mix. The goal is to extend the sound field well beyond the limitations of the loudspeakers, and make the overall sound seem more expansive. The elimination of the sweet spot is claimed.

star quad mic cable See cables.

star topology 1. A set of three or more branches with one terminal of each connected at a common node. 2. A communications network based on a star pattern where all equipment is connected to a central location with a single path.

star-wired ring See token ring.

steganography The science of communicating in a way that hides the existence of the actual communication. The practice of hiding information in a wider bandwidth carrier. This field covers the techniques used in digital watermarking schemes.

stereo or **stereophonic sound** 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.

stereo 2-way or **stereo 3-way**, etc. See: active crossover

STP (*shielded twisted-pair*) See cables; also *Scientifically Treated Petroleum*, but that's another time and another story.

subcode Non-audio digital data encoded on a CD that contains definable information such as track number, times, copy inhibit, copyright, etc.

subgroups See: groups

submix See: groups

subsonic Having a *speed* less than that of sound in a designated medium. [Use infrasonic if referring to frequencies below human hearing range.]

subtend 1. *Mathematics.* To be opposite to and delimit: *The side of a triangle subtends the opposite angle.* 2. To underlie so as to enclose or surround: *flowers subtended by leafy bracts.*

supersonic Having, caused by, or relating to a *speed* greater than the speed of sound in a given medium, especially air. [Use ultrasonic if referring to frequencies above human hearing range.]

suppression also **gain suppression** In teleconferencing the term used to describe the technique of instantaneous reduction of a sound system's overall gain to control acoustic feedback, and thus reduce echoes.

S-video Also called ***Y/C video***, a two-channel video channel that transmits black and white, or luminance (Y), and color portions, or chrominance (C), separately using multiple wires. This avoids composite video encoding, such as NTSC, thus providing better picture quality. Found mostly on S-VHS, Laserdisc, DVD and Hi8 products.

swag 1. *Slang* Stolen property; loot. 2. *Slang* Herbal tea in a plastic sandwich bag sold as marijuana to an unsuspecting customer. 3. *Australian* To travel about with a pack or *swag*. 4. *Slang* Giveaways (usually in the form of merchandise "loot") by manufacturers as added incentive to personnel either selling or buying their goods. Compare with spiff

sweet spot Any location in a two loudspeaker stereo playback system where the listener is positioned equidistant from each loudspeaker. The apex of all possible isosceles (two equal sides) triangles formed by the loudspeakers and the listener. In this sense, the sweet *spot* lies anywhere on the sweet *plane* extending forward from the midpoint between the speakers.

symmetrical (reciprocal) response Term used to describe the comparative shapes of the boost/cut curves for variable equalizers. The cut curve exactly mirrors the boost curve.

synchronous A transmission process where the bit rate of the signal is fixed and synchronized to a master clock.

Syn-Aud-Con (*Synergetic Audio Concepts*) A private organization conducting audio seminars and workshops, sponsored by several pro audio companies.

T

3-dB down point See: passband

3D sound A term used to describe a three-dimensional sound field. A true 3D sound field positions sound anywhere in a semi-spherical shell surrounding the listener. Sound must come from anywhere directly behind to directly overhead to directly in front of the listener and all points left and right. It if does not, it is not 3D sound. The term is popularly misused by multimedia companies to describe systems, effects and techniques purported to create 3D sound from two sources and designed for two loudspeaker playback; however, the result is not 3D sound. It is enhanced two-dimensional sound. Strictly speaking a broadening, widening, enhancing, or spreading of the left/right sound stage is not 3D. No two loudspeaker system is capable of locating sounds directly to the rear of the listener; nevertheless, some of these systems truly impress. The best enhancement schemes come very close to recreating a quarter-spherical sound shell, extending to nearly 180 degrees left-to-right, approaching 90 degrees overhead, with greatly improved depth of field. For further information see the Ultimate Spatial Audio Index, and Links to the World of Spatial Sound.

10Base-T, **100Base-T**, **1000Base-F** See: Ethernet

T-1 A digital transmission scheme utilizing two twisted-pair capable of handling a minimum of 24 voice channels. Used for connecting networks across remote distances.

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. 2. A proposed Rane product line aimed at the coffin market, since abandoned.

talk box A poor man's vocoder. Popularized by Peter Frampton and Joe Walsh in the '70s. See Heil Talk Box for a demo of the original and most popular model.

taste test or **tongue test** An actual voltage testing method recommended by Terrell Croft in his book *The American Electricians' Handbook*, published by McGraw-Hill in 1913. Here's the passage found on page 48:

> "The presence of low voltages can be determined by 'tasting.' The method is feasible only where the pressure is but a few volts and hence is used only in bell and signal work. Where the voltage is very low, the bared ends of the conductors constituting the two sides of the circuit are held a short distance apart on the tongue. If voltage is present a peculiar mildly burning sensation results which will never be forgotten after one has experienced it. The 'taste' is due to the electrolytic decomposition of the liquids on the tongue which produces a salt having a taste. With relatively high voltages, possible 4 or 5 volts, due to as many cells of battery, it is best to first test for the presence of voltage by holding one of the bared conductors in the hand and touching the other to the tongue. Where a terminal of the battery is grounded, often a taste can be detected by standing on moist ground and touching a conductor from the other terminal to the tongue. Care should be exercised to prevent the two conductor ends from touching each other at the tongue, for if they do a spark can result that may burn."

And from the same book comes these words of wisdom for testing for the presence of electricity by touching the two conductors:

> "Electricians often test circuits for the presence of voltage by touching the conductors with the fingers. This method is safe where the voltage does not exceed 250 and is often very convenient for locating a blown-out fuse or for ascertaining whether or not a circuit is alive. Some men can endure the electric shock that results without discomfort whereas others cannot. Therefore, the method is not feasible in some cases."

[*We don't know how Mr. Croft died, but perhaps we could hazard a guess*. Thanks RH.]

TCP/IP (*transmission control protocol/internet protocol*) A set of protocols developed by the Department of Defense in the '70s to link dissimilar computers across many kinds of networks and LANs. Popular with Ethernet users.

TDIF (*Teac digital interface format*) Tascam's (Teac) 8-channel digital audio interface to their DA-88 digital multitrack recorder, using unbalanced signal transmission and a DB-25 type connector.

TDS (*time-delay spectrometry*) A sound measurement theory and technique developed in 1967 by Richard C. Heyser at the Jet Propulsion Laboratories of the California Institute of Technology.

TEF (*time-energy-frequency*) The term adopted to describe the entire spectrum of *TDS* measurements, including energy-time curves. Popularized by Richard Heyser through his participation in *Synergetic Audio Concepts* seminars. Made practical in 1979 by the Techron division of Crown International - Cal Tech's first TDS licensee, and introduced as the TEF System 10.

tele- Distance; distant: *telescope*. [Greek *tele*- meaning far off.]

telecommunication Communicating over a distance by wire, fiber or wireless means.

teleconferencing An *audio* conference held by three or more persons over a distance. Normal usage refers to voice conferencing, also termed *audioconferencing* that includes all forms of audio. The term is sometimes extended to include video and document, or data, conferencing. Note that the term does not mean *telephone* conferencing, but rather *distance* conferencing, although telephone lines are often used. [*Thanks to RG at Q Factor for pointing out this important distinction.*] Contrast with videoconferencing.

telemedicine A specialized form of videoconferencing optimized for medical uses. Also referred to as *medical conferencing*, it allows distance learning in medical education and delivers health care (including assisted medical operations) to patients and providers at a distance.

temporal Of, relating to, or limited by time.

terminal strips See connectors.

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 References: Metzler)]

THD+N (*total harmonic distortion plus noise*) The most common audio measurement. A single sine wave frequency of know 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 20-20kHz). 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 20kHz, 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 that "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.

theremin Considered the first electronic musical instrument, invented in 1919 by Russian born *Lev Sergeivitch Termen*, which he anglicized to *Leon Theremin*. The theremin is unique in that it is the only musical instrument played without being touched. Interestingly, when granted a US Patent in 1928, there were 32 prior patents referenced, going all the way back to Lee De Forest. A theremin works by causing two oscillators to "beat" together. The *beat frequency* equals the difference in frequency between the two signals. Beats are a physical phenomenon occurring in the air when sounds are mixed. A theremin uses one oscillator operating well above the upper limit of human hearing as a reference tone, and another oscillator whose frequency is varied by the proximity of a human hand, for instance, to a capacitive sensing element shaped like an antenna. A typical machine has two antennas and you play it by moving your hands nearer to and farther from the antennas. One antenna controls the volume of the sound, while the other controls the frequency, or pitch, of the sound. Used together you can creates sounds that can range from being very sci-fi-ish -- a sort of quivering sound -- as heard in early sci-fi movies like *The Day the Earth Stood Still*, to very complex jazz licks. The theremin even appears as Dr. Hannibal Lecter's favorite instrument in Thomas Harris' bestseller *Hannibal* (Delacorte, 1999).

It was the theremin that got Bob Moog (inventor of the *Moog Synthesizer* and considered the father of modern electronic music) interested in electronic music. His latest company Big Briar now makes some of the world's best theremins.

See the Theremin web ring for additional info; and to view the fascinating, bizarre, and stranger-than-fiction true-life story of Leon Theremin, check out the film (available on video), *Theremin: An Electronic Odyssey*, by Steven M. Martin (1994), including several performances by *Clara Rockmore*, perhaps the best theremin player ever.

thermionic valve See vacuum tube

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 spit into three equal parts, it is potentially too confusing. The preferred term is one-third octave.

Thompson filters See: Bessel crossover

THX® Lucasfilm, Ltd. term meaning several things: 1) Their audio playback design and certification program for commercial cinema theaters; 2) Their audio playback specification for home cinema systems; 3) Approved audio/video playback equipment meeting their standards of quality and performance; and 4) DVDs, Laserdiscs and VHS tapes mastered by them to meet their quality and performance standards. The term comes from two sources: George Lucas's first film *THX-1138*, and a somewhat tongue-in-cheek reference to *Tomlinson Holman's eXperiment*, after their original technical director, patentee and creative force behind all the above (who now runs TMH Corporation).

TIA (*Telecommunications Industry Association*) Created in 1988 by a merger of the US Telecommunications Suppliers Association (USTSA) and the EIA's Information and Telecommunications Technologies Group (EIA/ITG). This organization works with the EIA in developing technical standards and collecting market data for the telecommunication industry.

tiger A great cat whose skin is striped, not just his fur.

TIM (*transient intermodulation distortion*) See: IM

timbre (pronounced "tambur") 1. The quality of a sound that distinguishes it from other sounds of the same pitch and volume. 2. *Music*. The distinctive tone of an instrument or a singing voice.

time 1.a. A nonspatial continuum in which events occur in apparently irreversible succession from the past through the present to the future. b. An interval separating two points on this continuum. c. A number, as of years, days, or minutes, representing such an interval. d. A similar number representing a specific point on this continuum, reckoned in hours and minutes. 2. *Music*. a. The characteristic beat of musical rhythm: *three-quarter time*. b. The rate of speed at which a piece of music is played; the tempo. ["Time is what keeps everything from happening all at once." unknown source]

time delay No such thing; a misnomer. You cannot delay *time* (see above). Misused to mean *signal delay* or just *delay*.

token ring A LAN baseband network access mechanism and topology in which a supervisory "*token*" (a continuously repeating frame [group of data bits] transmitted onto the network by the controlling computer; it polls for network transmissions) is passed from station to station in sequential order. Stations wishing to gain access to the network must wait for the token to arrive before transmitting data. In a token ring topology, the next logical station receiving the token is also the nest physical station on the ring. This mechanism prevents collisions on this type of network. Normally connected as a ***star-wired ring*** where each station is wired back to a central point known as the ***multistation access unit (MAU)***. The MAU forms a ring of the devices and performs the back-up function of restoring the ring should one of the devices crash or lose its cable connection.

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.

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. See also: Baxandall tone controls

toroid The name for any doughnut-shaped body. [*Mathematics*: a surface generated by a closed curve rotating about, but not intersecting or containing, an axis in its own plane.] The shortened popular name for the doughnut-shaped (toroidal) transformers common to audio equipment; favored for their low hum fields.

TOSLINK (*Toshiba link*) A popular consumer equipment fiber optic interface based upon the S/PDIF protocol, using an implementation first developed by Toshiba.

T-pad See attenuator pad.

tracking power amplifiers A term used to describe audio power amplifier designs utilizing a variable power supply for the output, and a means of controlling the power supply based upon the input signal. This scheme improves efficiency. See: Class H Amplifiers and compare with rail-switchers.

transcendental number *Mathematics*. 1. Not capable of being determined by any combination of a finite number of equations with rational integral coefficients. 2. Not expressible as an integer or as the root or quotient of integers. Used of numbers, especially nonrepeating infinite decimals.

transform switch *Turntablist mixers*. This switch selects either *phono* or *line* as the channel source, but is commonly used for *transforming*, or quickly gating the source on and off.

transversal equalizer A multi-band variable equalizer using a tapped audio delay line as the frequency selective element, as opposed to bandpass filters built from inductors (real or synthetic) and capacitors. The term "transversal filter" does not mean "digital filter." It is the entire family of filter functions done by means of a tapped delay line. There exists a class of digital filters realized as transversal filters, using a shift register rather than an analog delay line, with the inputs being numbers rather than analog functions.

tree topology A LAN topology that recognizes only one route between two nodes on the network. The map resembles a tree or the letter T.

triple point of water A system is at the "triple point" when ice (solid), water (liquid), and vapor coexist in equilibrium. This point is the freezing point of water and is set by international agreement to equal 273.16 degrees kelvin (0 degrees Celsius; 32 degrees Fahrenheit)

truncate To eliminate without round-off some low-order bits, often after performing an arithmetic computation.

TTL (*transistor transistor logic*) The workhorse digital logic integrated circuit family introduced as a standard product line in 1964.

turntablist A performing artist who uses two or more turntables as music sources from which he/she creates original results by quickly cutting and mixing the sounds of each using specially designed performance mixers such as Rane's TTM 54i.

twin-tone IMD See: IM

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. See cables.

two-bit Costing or worth 25 cents: *a two-bit cigar*.

U

UART *(universal asynchronous receiver-transmitter*) The device that performs the bidirectional parallel-to-serial data conversions necessary for the serial transmission of data into and out of a computer.

UDP (*user datagram protocol*) A TCP/IP protocol describing how messages reach application programs within a destination computer. This protocol is normally bundled with IP-layer software. UDP is a transport layer, connectionless mode protocol, providing a (potentially unreliable, unsequenced, and/or duplicated) datagram mode of communication for delivery of packets to a remote or local user.

UDP/IP (*user datagram protocol/internet protocol*) See: UDP above.

ULSI (*ultra-large-scale integration*) A logic device containing a million or more gates.

ultrasonic Of or relating to acoustic frequencies above the range audible to the human ear, or above approximately 20,000 hertz. Compare with supersonic.

unbalanced line See: balanced line

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.

UPS (*uninterruptible power supply*) A back-up power supply (commonly used with computers) that automatically continues to supply power when the main AC source fails.

URL (*uniform resource locator*) A Web address. A consistent method for specifying Internet resources in a way that all Web browsers understand. For example, "http://www.rane.com," is the URL for Rane's home page on the web. The "http" part tells the Web browser what protocol to use, and the remainder of the URL, "www.rane.com," is the Internet address.

USB (*universal serial bus*) A new low-speed (12 Mbits/sec) serial bus that acts like a special purpose local area network. Proposed by a consortium of Compaq, Digital, IBM, Intel, Microsoft, NEC and Northern Telecom in March of 1995, the intended purpose is to replace the typical cable ghetto found on most PCs. A USB equipped machine would have only three ports: USB, monitor, and LAN. The USB port would support 63 devices, and eliminate the need for all specialized parallel, serial, graphics, modem, sound/game or mouse ports. USB is completely "plug and play," i.e., it detects and configures all devices automatically, and allows "hot swapping" of devices. See: IEEE-1394 for complementary high-speed system.

UTP (*unshielded twisted-pair*) See cables.

UV (*ultraviolet*) Electromagnetic radiation at frequencies higher than visible light yet lower than those of x-rays. Commonly used to erase EPROMs and in wireless and fiber optic data transmission.

uxoriousness 1. Excessively submissive or devoted to one's wife. 2. "A perverted affection that has strayed to one's own wife." -- Ambrose Bierce.

V

VA (*voltampere*) See voltampere.

vaporware Refers to either hardware or software that exist only in the minds of the marketeers.

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. 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.

valve British term for vacuum tube, popularized because the first tube was known as the *Fleming valve* named for its inventor Ambrose Fleming.

VCA (*voltage-controlled amplifier*) 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.

variable-Q equalizer See: proportional-Q equalizer

vector *Mathematics.* A quantity, such as velocity, completely specified by a magnitude and a direction.

videoconferencing Video *and* audio communication held by two or more people over a distance using a codec at either end and linked by digital networks (T-1, ISDN, etc.). Contrast with teleconferencing.

virus A self-replicating program released into a computer system for mischievous reasons. Once triggered by some preprogrammed event (often time or date related), the results vary from humorous or annoying messages, to the destruction of data or whole operating systems. *Bad bad*.

VLSI (*very-large-scale integration*) Refers to the number of logic gates in an integrated circuit. By today's standards, a VLSI device could contain up to one million gates.

vocoder (*voice coder*) 1. Invented by Homer Dudley (no fooling) in 1936 at Bell Labs, and called a "phase vocoder." It was an electronic device for analyzing and synthesizing, or generating artificial speech. Homer Dudley was the first person to recognized that the basic information rate of speech is low and that if you broke it down into its basic components, these could be transmitted over a quite narrow bandwidth, and then reconstructed at the receiving end. Thus was born the speech synthesizer. The vocoder principal is based on determining the *formants*, or vowel sounds, of the speech signal, along with its fundamental frequency and any *noise* components such as *plosive* sounds (a speech sound produced by complete closure of the oral passage and subsequent release accompanied by a burst of air, as in the sound (p) in *pit*, or (d) in *dog*), hisses, or buzzes. Typically this is done by using two sets of filter banks -- one for analysis and one for synthesis -- and an "excitation analysis" block. The analysis filter bank is much like those used in real-time analyzers. The audio is presented to a bank of parallel connected bandpass filters, whose output levels are converted into DC voltage levels proportional to the signal passing through each bandpass filter. This captures the formant information. The excitation analysis block determines and codes the fundamental frequency and noise attributes. Reconstruction occurs by using the encoded DC levels, mixed with the excitation block output, to gate each output bandpass filter, which are then summed together to recreate a facsimile of the original speech signal. Early pictures and audio samples (*from* Prof. Edward A. Lee, UC Berkeley). 2. Once vocoder basics were established, they found new uses in electronic music applications. The MI (musical instrument) vocoder uses speech input to modulate another music instrument signal so that it "talks." Use of vocoders peaked in the '70s after being popularized by such notables as Wendy Carlos, Alan Parsons and Stevie Wonder. This vocoder version has two inputs, one for the vocal microphone and one for another instrument. Talking or singing into the microphone modulates or superimposes vocal characteristics onto the other instrument. Compare with talk box

volatile Refers to a memory device that loses any data it contains when power is removed from the device. Examples would include static and dynamic RAMs.

volt *Abbr.* **E**, also **V**. The International System unit of electric potential and electromotive force, equal to the difference of electric potential between two points on a conducting wire carrying a constant current of one ampere when the power dissipated between the points is one watt. [After **Count Alessandro Volta**.]

Volta, **Count Alessandro** (1745-1827) Italian physicist who invented the battery (1800). The volt is named in his honor.

voltampere (*VA*) The product of rms voltage and rms current in an electronic circuit. It is the unit of apparent power in the International System of Units (SI).

vote "The instrument and symbol of a freeman's power to make a fool of himself and a wreck of his country." -- Ambrose Bierce.

VOX (*voice operated exchange*) Also called *voice operated relay*, originally a tape recorder feature where speech starts the recording process and silence stops it. However it is not restricted to tape recorders, for instance, cellular phones use VOX to save battery life, and teleconferencing systems use it to determine the number of active mics. See NOM.

VRML **(*virtual reality modeling language*)** A developing standard for describing interactive 3D scenes delivered across the internet. In short, VRML adds 3D data to the Web. Heavily supported by Silicon Graphics (SGI) workstations, competing with Sun's Java loaded workstations.

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. Contrast with PPM.

W

W3 An abbreviation for World Wide Web.

walla The film industry term for background crowd noises in a movie.

Walla Walla A city of southeast Washington near the Oregon border south-southwest of Spokane. Founded in 1856 near the site of an army fort, it is a manufacturing center in an agricultural region famous for sweet yellow onions. In spite of its name, a quiet community.

watermarking 1. *Paper* The act of adding a translucent design impressed on paper during manufacture and visible when the paper is held to the light. 2. *Audio or video* Embedded data code within the digitized audio or video image that can be recovered but which will not affect the quality of the product. Various methods exist, but all consist of very short (2-5 microseconds long) pieces of code containing all the relevant data about the copyright owner and performance royalties. All make use of the science of steganography.

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

Watt, James (1736-1819) British engineer and inventor who made fundamental improvements in the steam engine, resulting in the modern, high-pressure steam engine (patented 1769).

watts rms *No such thing.* See apparent power and rms power.

wavelength *Symbol* (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.

web ring A group of websites all sharing a common theme. For example, web rings exist for fans of certain bands, movies, TV shows, authors, race car drivers, etc. Soon we will have a web ring for web rings.

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. Four weighting filter designs dominate (See: References: Metzler):

- **A-weighting** 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** 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.***]

Wheatstone bridge 1. An instrument used for measuring resistance. The circuit used is a 4-arm bridge, all arms of which are predominantly resistive. The bridge is a two-port network (i.e., it has two terminal pairs across opposite corners) capable of being operated in such a manner that when voltage is applied to one port, by suitable adjustment of the resistive elements in the network, zero output can be obtained at the signal output port (usually a meter). Under these circumstances the bridges is termed *balanced*. 2. An Omaha based music group. [*Although the circuit used in a Wheatstone bridge was first described by* ***Samuel Hunter Christie*** *(1784-1865) -- the son of James Christie, founder of the well-known auction house -- in his paper "Experimental Determination of the Laws of Magneto-electric Induction" (1833),* ***Sir Charles Wheatstone*** *(1802-1875) received credit for its invention because of his adaptation of the circuit in 1843 for the measurement of resistance. Wheatstone also invented the concertina, the stereoscope and contributed significantly to the development of the telegraph.*]

white noise 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.]

wide-range curve Same as X curve

widget (perhaps alteration of *gadget*) 1. A small mechanical device or control; a gadget. 2. An unnamed or hypothetical manufactured article. 3. As developed by *Guinness*, a small disk with a pinprick-size hole that fits inside their beer cans. As the beer is packaged, a small amount of stout is forced into the widget and held there under pressure. Once the pressure is released by opening the can, the beer is freed from the widget and a stream of bubbles flows upward. Now when the stout is poured, it looks like a pub-poured draught with the characteristic Guinness head (thick collar of foam), without the widget it looks like any other beer. It also reproduces the creamy texture and low carbonation of a draught pint. Now also used by *Murphy's* and *Beamish*.

Wintel A contraction of the words "Windows" and "Intel." Used to describe personal computers made from Intel microprocessors and running Microsoft Windows software. It is reported that this "Wintel standard" accounts for 80% of all PCs.

wire See cables

WOM (*write-only-memory*) Term coined by Signetics in 1972 for their 25000 Series 9046XN Random Access Write-Only-Memory integrated circuits. Based on SEX (Signetics EXtra secret) processes, these devices employ both enhancement and depletion mode P-Channel, N-Channel, and NEU-Channel MOS transistors (devices which simultaneously, randomly, or not at all, enhance or deplete regardless of gate polarity). The world's supply of WOMs was quickly consumed by newly designed airline baggage-handling equipment, where they are still used today to store the exact real-time location of each bag. WOM production was suddenly discontinued when it was discovered that the only copy of the mask code had been accidentally filed into a WOM location.

word An ordered set of bits that is the normal unit in which information may be stored, transmitted, or operated upon within a given computer - commonly 16 or 32 bits.

word clock The synchronizing signal that indicates the sampling frequency or rate of sample words over a digital audio interface.

word length The number of bits in a word.

World Wide Web (*WWW* and/or *W3*) 1. A way to present resources and information over the Internet, or according to its inventor CERN , "The World Wide Web (W3) is the universe of network-accessible information, an embodiment of human knowledge." 2. Satirically called the *World Wide Wait*.

WOROM (*write-once read-only memory*) Systems in which data may be written once, but not erased and rewritten. Usually refers to CD-ROM technology that can be recorded only once.

write To record data on a medium.

WWW (*World Wide Web*) See: World Wide Web.

X

X 1. The electronic symbol for reactance - the imaginary part of impedance.

X curve (*extended curve*) In the film sound industry an *X curve* is also known as the *wide-range curve* and conforms to ISO Bulletin 2969, which specifies for pink noise, at the listening position in a dubbing situation or two-thirds of the way back in a theater, to be flat to 2 kHz, rolling off 3 dB/oct after that. The *small-room X curve* is designed to be used in rooms with less than 150 cubic meters, or 5,300 cubic feet. This standard specifies flat response to 2 kHz, and then rolling off at a 1.5 dB/oct rate. Some people use a modified small-room curve, starting the roll-off at 4 kHz, with a 3 dB/oct rate. Compare with Academy curve

xerography The name created by the Haloid Company in 1946 (from the Greek *xeros* for dry and *graphein* for writing) for the process invented by Chester F. Carlson on October 22, 1938, which he named *electrophotography*. In 1960, Haloid-Xerox introduced the 914 copier, the first pushbutton, plain-paper, xerographic office machine. The company soon became known simply as **Xerox**.

X Generation See Generation X.

Xilinx® (pronounced *zi-links*; after *xi* the 14th letter of the Greek alphabet) Leading manufacturer of field-programmable logic devices.

XLR See connectors.

XOR Acronym for *exclusive OR*, a type of logic gate where a logic 1 output is based upon A or B inputs being present - but not both.

Y

Y 1. The electronic symbol for admittance - the inverse of impedance. 2. Abbreviation for luminance (black & white) video signal.

Y2k (*year two thousand*) Follow link to read the best Y2k statement.

Y/C video See S-video.

Y-connector 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. See RaneNote 109 for details.

Y/N Software program "yes/no" response prompt. A "Y" or "N" keystroke is expected.

yapped *Book jargon* Refers to the edge of the cover of a book bound in paper or other soft material. *Yapped edges* are not flush with the pages but extend beyond the edges of the book making them fragile.

yield The number of devices that work as planned, specified as a percentage of the total number actually fabricated. Normally used to quantify a run of integrated circuits.

Yt Chemical symbol for *yttrium* - my absolute favorite element, next to *ytterbium*.

YUV video The coding process used in CD-I in which the luminance signal (Y) is recorded at full bandwidth on each line and chroma values (U and V) are recorded at half bandwidth on alternate lines.

Z

Z The electronic symbol for impedance.

z-transform A mathematical method used to relate coefficients of a digital filter to its frequency response, and to evaluate stability of the filter. It is equivalent to the Laplace transform of sampled data and is the building block of digital filters.

zap To eradicate all or part of a program or database, sometimes by lightning, sometimes intentionally.

zeal "A certain nervous disorder afflicting the young and inexperienced." -- Ambrose Bierce.

ZIF (*zero insertion force socket*) A standard IC-socket design requiring the user to move a lever to insert or remove the chip -- as opposed to pressing and prying the chip manually -- hence, *zero insertion force*. The lever actuator (hopefully) eliminates damaging the IC pins.

Zobel network or **Zobel filter** [Also called *Boucherot cell* -- if you know the origin of this, please write me.] 1. A filter designed according to image parameter techniques. 2. *Audio amplifiers.* Zobel networks are used in audio amplifiers to dampen out high frequency oscillations that might occur in the absence of loads at high frequencies. It is the commonly seen series resistor-capacitor combination located directly at the output of the driver stage, just before the output inductor. Typical values are 5-10 ohms in series with 0.1 microfarads. The network limits the rising impedance of a loudspeaker due to the speaker coil inductance. This phenomenon is further aggravated by the output inductor found in most power amplifiers used to disconnect the load at high frequencies. See Douglas Self's book for a good discussion of audio amplifier Zobel networks. 3. *Loudspeakers.* Some loudspeaker crossover designs include Zobel networks wired across the tweeter (high frequency) driver to compensate for the rise in impedance at high frequencies due to the inductance of the voice coil. The goal here is to try to keep the load seen by the crossover circuitry as resistive as possible. [After *Dr. Zobel of Bell Labs*.]

zymurgist See brewer

zymurgy The scientific study of the process of fermentation in brewing and distilling.

Zyzyyzyski, Zyzeikkel The last name listed in the *1998 Snohomish County, Washington, U.S.A.* telephone directory - really (*but, alas, no more*).

zyzzyva (The last word in the English dictionary) Any of various tropical American weevils of the genus *Zyzzyva*, often destructive to plants. [New Latin *Zyzzyva* genus name probably from *Zyzza* former genus of leafhoppers]

Compiled by Dennis A. Bohn, v.p. research & development, Rane Corporation. Write me:

Who said, "*The worst I ever had it was wonderful!*"?

This is the end -- for now.

Pro Audio Company Names -- Initialed & Made-up -- What They Mean

Last Updated February 1, 2000

Return to Pro Audio Reference

AEI Music ***Audio Environments Inc.***

AKG Acoustics ***Akustische u. Kino-Geräte*** (Acoustic and Cinematography Equipment)

ALESIS A loose acronym for ***Algorithmic Electronic Systems***, adjusted to make spelling and pronunciation easier. One of two companies born from the ashes of *MXR*.

ALTEC LANSING ***All Technical Products*** plus ***Lansing*** from partner *James B. Lansing.*

AMEK According to founder Graham Langley, there is no particular significance to the name.

AMPEX After the initials of founder *Alexander M. Poniatoff* plus *EXcellence*.

AMS Neve ***Advanced Music Systems***, now merged with *Neve*, after Rupert Neve, father of modern audio console technology.

APHEX ***Aural Perception Heterodyne Exciter***

API ***Automated Processes, Inc.*** also ***Audio Products, Inc.*** "Automated Processes" was chosen out of frustration to avoid the repeated name rejection by NY State when incorporting, according to Lou Lindauer, founder (all of the many names originally picked were rejected). When Paul Wolff bought the company in 1985, he could not continue to use the original name for legal reasons, so he came up with "API Audio Products." When he sold the company to the ATI Group in 1999, they incorporated under the original "Automated Processes" name. Got it?

ART ***Applied Research & Technology***; one of two companies born from the ashes of *MXR*.

ARX ***Audio Research X***; evolved from *Audio Research & Technology* after its initials became confused with those of *Applied Research & Technology*, so they replaced "& Technology" with "X" and registered it worldwide.

ASHLY Audio After ***Larry Ashley***, one of the five founders of the original sound company (which evolved into manufacturing). They chose his name for the alphabetical advantage, then modified it by dropping the "e" to prevent conflict with another company; in addition, it added a little intrigue.

ASPI Digital ***Atlanta Signal Processors, Inc.***

B&W Loudspeakers ***Bowers & Wilkins***

BASF ***Badische Aniline Soda Ash Fabrik***

BBE Sound ***Barcus Berry Electronics*** after founder Barcus & John Berry, inventor of early acoustic piezo pick-ups.

BGW Systems ***Brian Gary Wachner***, founder.

BSS Audio ***Brooke Siren Systems*** after founders Chas Brooke and Stan Gould.

CALREC ***Calder Recordings***; shortened from Calder Valley Sound Recording Group

CEDAR Audio ***Computer Enhanced Digital Audio Restoration***

DAR ***Digital Audio Research***

dbx ***David Blackmer***, founder, whose new company is **Earthworks**.

DDA ***Dearden Davies Associates***, after founders Gareth Davies and David Dearden, whose new company is **Audient**

DIGIGRAM ***Digital Gramaphone*** per Neil Glassman, president of this French company.

DOD ***David O. DiFrancesco***, founder, whose new company is **Rolls**.

DTS ***Digital Theater Systems***

DUKANE ***Dupage*** and ***Kane*** counties in *St. Charles, IL*, the county boundries where they are located; originally named *Operadio* when they first manufactured battery-operated radios.

EAW ***Eastern Acoustic Works***

E-MU Systems ***Electronic Music Systems***

ENSONIQ Name created by founder Bob Yannes from the made-up *ensonic* – replacing the "c" with "q" because *Compaq* was getting lots of publicity with its start-up company in 1983.

FOCUSRITE An off-the-shelf *shell company name* (a pre-registered U.K. legal entity) used initially to quickly establish a company, normally changed after startup, however Rupert Neve liked it, because it proved memorable and many people find it analogous to accurate listening.

FOSTEX Believed a combination of *Foster* (parent company) plus *EXcellence*.

FSR "Just stands for FSR," states Phyllis Gillick, Inside Sales.

HHB Communications ***Half-Human Band***: name of their rock group 23 years ago.

IED ***Innovative Electronic Designs***

IRP Professional Sound Products L.P. ***Industrial Research Products … Limited Partnership***

JBL ***James B. Lansing***; after selling his interests in *Altec-Lansing*, he used his initials for his new company.

JVC ***Japan Victor Company***

KEF ***Kent Engineering & Foundry***, by founder Raymond Cooke in Kent, U.K.

KRK Systems ***Keith R. Klawitter***, founder.

MACKIE After ***Greg Mackie*** who previously founded *TAPCO* and *Audio Control*.

MUZAK ***Music + Kodak***

MXR ***"Mixer"*** Defunct company that spawned *ART* & *ALESIS*.

NHT ***Now Hear This***

NVISION ***n*** for any number, ***Vision*** for visual resolution; put them together ***n+Vision*** and you get any number of scan lines at any data rate, which is HDTV, their business.

NXT ***New Transducers Ltd.***

QSC Audio Products ***Quilter Sound Company*** after founder *Patrick Quilter*.

RANE An *anagram* made from the letters common to the founder's first and last names.

RCF ***Radio Cine Forniture***, Italian company owned by *Mackie*.

ROLLS Founder David DiFrancesco states, "Just a name we pulled out of the blue; it is short and "rolls" off the tongue."

RPG Diffusor Systems ***Reflection-Phase Grating*** Note the "o" rather than an "e" in *Diffusor* -- it is the European spelling favored by Manfred Schroeder.

SADiE ***Studio Audio Distribution in Europe***

SAMSON Technologies ***Sam's Sons***; founders are the sons of Sam Ash (Sam Ash Sound, NY).

SEK'D ***Studio für Elektronische Klangerzeugund der Universität Dresden*** (Studio for electronic Klangerzeugund of the University of Dresden)

SESCOM ***Scientific Electronic Systems Company***

SONY Latin *sonus* (sound) plus English slang nickname *Sunny* (young, bright & cute), drop an "n" and *voilà*. In 1946, the company was founded by Akio Morita and Masaru Ibuka, and named *Tokyo Telecommunications Engineering Corp*. As the company grew and aimed at world markets, Morita changed the name in 1958, claiming that "Sony" was pronounceable in any language and easily remembered.

SPL ***Sound Performance Lab***

SRS Labs ***Sound Retrieval System***

SSL ***Solid State Logic***

SSM ***Solid State Micro Technology for Music*;** originally *Solid State Music Technology*, which changed when ownership changed; subsequently bought out by PMI, who was then acquired by **Analog Devices**, the current owner.

t.c. electronic "Beats me -- any good ideas?" responded Kenneth Ersgard, Marketing Manager, TC Electronic.

TAD Pioneer ***Technical Audio Devices***

TAPCO ***Technical Audio Products Company,*** *Greg Mackie's* first audio company, acquired by *Electro-Voice*, now *EVI Audio*, who retired the brand name, and was subsequently bought by *Telex*.

THAT ***Travaline, Hebert and Tyler*,** or ***Tyler, Hebert and Travaline,*** founders – take your pick, there is no T-name order priority – all previously worked at *dbx.*

TL Audio ***Tony Larking***, founder.

TOA Electronics Exact origin lost, however it is believed to derive from the two Japanese sounds closest to the original kanji characters representing the company. These were "to", possibly shortened from "toyo" meaning the East or the Orient (eastern Asia), or "toi" meaning "far," combined with possibly another contraction of "ajia" meaning Asia, or "wa" shortened form of "wafu" meaning Japanese, together they mean "*Far East*," or "*Eastern Asia*." Today, to emphasis the worldwide nature of the company, the name is spelled out *T-O-A*, not pronounced.

TOLECO Systems ***The Oliver Electric Company*** after the founder's cat. Today the cat's dead and the company's defunct.

UREI ***United Recording Electronics Industries***

XTA The letters do not stand for anything per John Austin, founder.

ZSYS ***z-systems*** from *z-domain* or z-transform, the mathematical space used in designing DSP algorithms -- their primary product -- and coincidentally, works well with the founder's name: Glenn Zelniker.

Return to Top

Return to Pro Audio Reference

Compiled by Dennis A. Bohn, v.p. research & development, Rane Corporation. Write me

Pro Audio Reference Books

Last Updated August 1, 2000

Return to Pro Audio Reference

Acquiring New Books

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Acquiring Out-of-Print Books

Some of the books listed are out-of-print and unavailable through normal book sources, but they are among the most useful. Try to locate them through your local used bookstores -- absolutely worth the trouble. In addition try any/all of the following used and rare book search sites: Advanced Book Exchange - Alibris - Bibliofind - Powell's Technical Books

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Annotated List of Pro Audio Reference Books

The American Heritage Dictionary of the English Language, 3rd ed. (Houghton Mifflin ISBN 0-395-44895-6, Boston, 1992). [*If you only own one dictionary, make it this one.*]

The IEEE Standard Dictionary of Electrical and Electronics Terms, 6th ed., (IEEE Std 100-1996; IEEE ISBN 1-55937-833-6, New York, 1997). [*The authority for electrical & electronic terminology disputes.*]

Alexander, Robert *The Inventor of Stereo, The Life and Works of Alan Dower Blumlein* (Focal Press ISBN 0-240-51577-3 Oxford, England, 1999). [*At last, a magnificent biography of a man finally restored to his rightful place in history.*]

Barber, David W. *Better Than It Sounds* (Sound And Vision ISBN 0-920151-22-1, Toronto, 1998). [*Humorous musical quotations.*]

Benson, K. Blair, ed. *Audio Engineering Handbook* (McGraw-Hill, New York, 1988). [*This is a great compilation of articles written by some of the best in the business. Too bad it is out of print. A lot of the material is done at an advanced level and quite difficult, but valuable. Contains the best article I've found on audio standards, written by Daniel Queen, AES Standards Manager.*]

Beranek, Leo L. *Acoustics* (McGraw-Hill, New York, 1954; Reissued 1986 by Acoustical Society of America ISBN 0-88318-494-X). [*The classic text. Long out-of-print, but now available. Difficult, but definitive.*]

Beranek, Leo L. *Music, Acoustics & Architecture* (Wiley , New York, 1962). [*Another gem from Dr. Beranek -- this time in non-technical language -- the essential meaning of acoustics to the performance and appreciation of music.*]

Beranek, Leo L. *Noise Reduction* (McGraw-Hill, New York, 1960). [*Written in 1960, for a special MIT summer program, this book went on to become the foundation for modern noise control.*]

Bierce, Ambrose, *The Devil's Dictionary* (Oxford University Press, ISBN 0-195-12627-0, Oxford, 1998). [*Originally published in the late 1890s by "The Wickedest Man in San Francisco" -- this is better than therapy.*]

Blauert, Jens, *Spatial Hearing; The Psychophysics of Human Sound Localization; Revised Edition* (MIT Press ISBN 0-262-02413-6 Cambridge, MA, 1997). [*Everything known about human sound localization, between two covers.*]

Bohn, Dennis A., "Operator Adjustable Equalizers: An Overview," *The Proceeding of the AES 6th International Conference: Sound Reinforcement* (Audio Engineering Society ISBN 0-937803-13-8, New York, 1989). [*Hey, you have to promote yourself at least once -- that's the rule.*]

Croft, Terrell, *The American Electrician's Handbook, 2nd ed.* (McGraw-Hill, New York, 1921). [*A gem of a book: leather bound, all edges gilt, wonderful!*]

Davis, Don & Carolyn Davis, *Sound System Engineering, 2nd ed.* (Sams ISBN 0-672-21857-7, Indianapolis, 1987). [*The textbook written by the husband and wife team that founded Synergetic Audio Concepts (Syn-Aud-Con), and pioneered sound system seminars.*]

Davis, Gary & Ralph Jones, *Yamaha Sound Reinforcement, 2nd ed.* (Hal Leonard ISBN 0-88188-900-8, Milwaukee, 1989). [*It's hard to find affordable, useful and accurate pro audio references-- this is all three. The best $/page/information-content value around.*]

Deketh, J. *Fundamentals of Radio-Valve Technique* (Philips Technical Library reprinted by Audio Amateur Press ISBN 1-882580-23-0 Peterborough, NH, 1999). [*First published in English in 1949, this handy book has been newly reprinted by Audio Amateur Press (available from Old Colony). This is one of the classic tube texts, and deserves a place on your shelf if you have any vacuum tube, or valve, interest at all.*]

Frederiksen, Thomas M., *Intuitive Analog Electronics* (McGraw-Hill ISBN 0-07-021962-1, New York, 1989). [*Only a very few technical writers have the real gift -- Bob Pease, John Watkinson, Jim Williams, and more recently, Clive Maxfield, all come to mind, but before any of them, came Tom Frederiksen; Tom has true gift. I was most fortunate to have learned directly from Tom, as were a couple of the other aforementioned. It simply cannot be written or said clearer or more interestingly than the way Tom does it. (FYI: He has a whole "Intuitive" series -- five titles that I know of. Find them, search them out, you won't be disappointed.)*]

Giddings, Philip, *Audio System Design and Installation* (Howard W. Sams ISBN 0-672-22672-3, Indianapolis, 1990). [*Don't let the title fool you, this is the book on interconnection wiring, grounding, shielding and AC power.*]

Helmholtz, Hermann, *On The Sensations of Tone* (Dover ISBN 0-486-60753-4, New York, 1980). [*One of the world's greatest scientific classics -- a genuine treasure. Written in 1885, and still considered one of the best sources for physiological acoustics.*]

Holman, Tomlinson, *Sound For Film And Television* (Focal Press ISBN 0-240-80291-8 Boston 1997) [*From the man who invented and developed Lucasfilm's THX® program, and now a professor at the University of Southern California School of Cinema-Television.*]

Howare, David M. & James Angus, *Acoustics And Psychoacoustics* (Focal Press ISBN 0-240-51428-9 Oxford, England 1998) [*A beautiful blend of acoustics and psychoacoustics from musical and scientific perspectives. Minimal math.*]

Krauss, Herbert L., Charles W. Bostian & Frederick H. Raab, *Solid State Radio Engineering* (John Wiley & Sons ISBN 0-471-03018-X, New York, 1980). [*Written in 1980, and still the acknowledged leader, this is one of the best books ever penned about classic analog radio. With the advent of digital radio, there will never be another text of this caliber on this topic.*]

Lampen, Stephen H. *Wire, Cable & Fiber Optics for Video & Audio Engineers, 3rd ed.* (McGraw-Hill ISBN 0-07-038134-8, New York, 1998) [*Trust me, you need this book -- there is none better on this subject.*]

Langford-Smith, Fritz. *Radiotron Designer's Handbook, 4th ed.* (Originally published by the Wireless Press for Amalgamated Wireless Valve Company Pty. Ltd., Sydney, Australia in 1934; revised three times to produce the 4th (and final) edition in 1952; reproduced and distribute by RCA Electronic Components, Harrison, NJ for the last time in May, 1968; resurrected and reissued as a 1,000 page paperback by Newnes (Butterworth-Heinemann) ISBN 0-7506-3635-1, Oxford, England, 1997) {*Along with Tremaine's book below, this is another of the ancient-audio sacred texts. This tome is a comprehensive reference covering basic audio principles and the practical design of all types of classic radio receivers, audio amplifiers and record-producing equipment up to the invention of the transistor. This is the book you consult to learn how they did compressors in the very old days, for instance.*

Maxfield, Clive. *Bebop to the Boolean Boogie* (HighText Publications ISBN 1-878707-22-1, Solana Beach, CA, 1995). [*Try it; you'll love it.*]

McQuain, Jeffrey *Never Enough Words* (Random House ISBN 0-679-45804-2 New York, 1999). [*Wonderful source for Americanisms. Where else are you going to find out that "teetotaciously exflunctified" means "totally worn out." Sports figures are accused of "talking trash," well, this is what is called "talking tall."*]

Metzler, Bob. *Audio Measurement Handbook* (Audio Precision, Beaverton, OR, 1993; FAX (503) 641-8906). [*Just how do you measure transient intermodulation distortion, anyway? Bob explains.*]

Moore, Brian C. *An Introduction to the Psychology of Hearing, 4th ed.* (Academic Press ISBN 0-12-505627-3, San Diego, CA 1997). [*Uncommonly clear explanations of the most complex hearing issues, from the leading researcher in the field.*]

Morton, David. *Off The Record: The Technology and Culture of Sound Recording in America* (Rutgers University Press ISBN 0-8135-2747-3, New Brunswick, NJ 2000). [*Great history full of surprises especially regarding dictation and answering machines. Be sure and check out his Dead Recording Media website.*]

Moses, Bob. "Asynchronous Serial Communications," *The Proceedings of the AES 13th International Conference: Computer-Controlled Sound Systems* (Audio Engineering Society ISBN 0-937803-25-1, New York, 1994). [*A great primer on all things asynchronous, from the originator of this web site.*]

Newton, Harry. *Newton's Telecom Dictionary, 15th ed.*(Miller Freeman ISBN 1-57820-031-8, New York, 1999). [*Telecommunication's magnum opus -- there is nothing else like it.*]

Olson, Harry F. *Acoustical Engineering* (Van Nostrand, New York 1957; reissued 1991 by Professional Audio Journals, Inc., Philadelphia, PA Available from Old Colony). [*Along with Beranek's Acoustics above, these two form the definitive bookends for all later acoustics books.*]

Olson, Harry F. *Musical Engineering* (McGraw Hill, New York 1952) [*An audio master presents the first unified engineering treatment of all the elements that enter into the production and reproduction of music.*]

Pickles, James O. *An Introduction to the Physiology of Hearing, 2nd ed.* (Academic Press ISBN 0-12-554754-4, San Diego, CA 1988). [*An excellent complement to Moore's book above on the* psychology*, this covers the* physiology *of hearing.*]

Pohlmann, Ken, ed. *Advanced Digital Audio* (Sams ISBN 0-672-22768-1, Carmel, IN, 1991). [*Difficult subjects clearly explained by this acclaimed professor of music engineering at the University of Miami -- an accurate, authoritative digital audio resource.*]

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Slone, G. Randy *High-Power Audio Amplifier Construction Manual* (McGraw-Hill ISBN 0-07-134119-6, New York, 1999). [*A very valuable do-it-yourself book, highly recommended.*]

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Transnational College of LEX *Who Is Fourier? A Mathematical Adventure* (Language Research Foundation ISBN 0-9643504-0-8, Belmont, MA, Third Printing 1998). [*A very different book, deceptively simple in style -- almost comic book-like -- but, I believe the best book on Fourier ever. Buy it; give it a chance -- don't be put off by the style. I bet you learn something and tell all your friends to get a copy. It's that good.*]

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Return to Top

Return to Pro Audio Reference

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