# Rane Professional Audio Reference

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## A

**AAC (*Advanced Audio Coding*)** Shortened name for the MPEG-2 Advanced Audio Coding specification, declared an international standard by MPEG in April 1997; however, now the term is used also to refer to MPEG-4 advanced audio coding.

**A&R (*artists and repertory*)** Historically the record industry term for the department or person that acts as the go-between the artist and the record label. Their job is to select and sign the performers to the label, decide what songs they will record, and select who will work with the artists in the production arranging and performance of the material for the recording of master tapes. These details vary a lot from label to label. For a good discussion on how the A&R world is changing see What Are Record Labels Looking For? by Wendy Day.

**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 distortion** Term coined by Dr. Peter D'Antonio, founder of RPG Diffusor Systems, for the interaction between the room, the loudspeaker, and the listener.

**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

**acoustics** Hearing; from the Greek *akouein*: to hear.

**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 stimulated *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 it is also called). Thus *modem* [*mo*dulator-*dem*odulator] is an acronym, and *AES* [*A*udio *E*ngineering *S*ociety] is an abbreviation or initialism. Compare with portmanteau word [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 crossfader** A device found in dj mixers used to crossfade between two music sources. An active design uses the potentiometer to send a control voltage to some type of voltage-controlled device that controls the audio, while in a passive design the audio appears on the potentiometer itself. Active

designs are more robust and offer greater reliability over passive ones. See "Evolution of the DJ Mixer Crossfader" by Rane's ace dj mixer designer, Rick Jeffs, for addional details.

**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 (***biamped***). A *mono 3-way* unit is a single channel device with three outputs, labeled *Low*, *Mid* and *High* (***triamped***). In this case, the user sets two frequencies: the Low-to-Mid, and the Mid-to-High crossover points. Up to *stereo 5-way* configurations exist for very elaborate systems. 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

**administratium** See: "Administratium" [*No technical glossary is complete without this term. Supporting evidence is given by the fact that it took the Google™ search engine just 0.25 second to return 2,150 versions located on the Web. Among these, it is credited to more than six authors, a hundred different research centers, universities and corporations, and is dated from the 1980s to the 1990s, but there is compelling evidence that it dates back to the '60s. Anyone who can prove who really wrote this classic piece of humor and when, please write me. Thanks.* ]

**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.

**Advanced Audio Coding** See: AAC

**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*. 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. Also see aliasing for a great example created by Prof. Wendy Walker at University of Arizona (requires Netscape web browser)

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

**ambience** 1. *Acoustics.* A perceptual sense of space [Blesser]. The acoustic qualities of a listening space [White]. 2. *Psychoacoustics* The special atmosphere or mood created by a particular environment [American Heritage Dictionary]. Contrast with reverberation

**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.

**AMI-C (*Automotive Multimedia Interface Collaboration*)** "An organization of motor vehicle manufacturers worldwide created to facilitate the development, promotion and standardization of electronic gateways to connect automotive multimedia, telematics and other electronic devices to their motor vehicles."

**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 .

**ARM (*advanced RISC machines*)** The name for a microprocessor group formed from Acorn, backed by Apple, VLSI Technology and Nippon Investment and Finance, in 1990. Acorn Computer was the parent company set up by Dr. Hermann Hauser and Dr. Chris Curry in 1979 to make personal computers, but now enjoys its biggest success selling intellectual property around their proprietary RISC computer, called ARM, which originally stood for *Acorn RISC Machines*.

**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).

**A-taper** See potentiometer

**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.

**audion** Dr. Lee De Forest's name for his 1906 invention of the triode (*three-element vacuum tube*), building upon Sir John Ambrose Fleming's thermionic diode, based on the Edison effect.

**audion piano** The first vacuum tube instrument in 1915, invented by Dr. Lee De Forest.

**audio taper** See potentiometer

**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. And for a refreshingly rational voice on hot audio topics check out Rod Elliott's site, particularly his get-rich-quick scheme for exploiting the gullible regarding burning-in audio cables. [*Absolutely brilliant.*]

**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, which became the advanced technology subsidiary of Aureal Semiconductor, *alas, now defunct*. Aureal 3D made 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 claimed to have solved this problem, but didn't stay in business long enough for anyone to find out.

**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.

**AVD (*advanced video disk*)** A Chinese proposed alternative to the DVD standard to avoid paying what they consider exorbitant royalties. This threatened standard would apply to DVD-like players sold only in China. Members of the *China Audio Industry Association (CAIA)* say the spec could be published in late 2001 if the DVD royalty issue remains unsettled.

**average power** See apparent power

**A-weighting** See weighting filters

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