D

D/A: See *digital-to-analog converter*.

DAC: See *digital-to-analog converter*.

DAE: (1) Digidesign Audio Engine.™ A Macintosh application that can run behind other applications, such as sequencers or ProTools,™ handling the transfer of audio data to/from the hard disk. DAE is licensed by many sequencer developers to avoid writing their own low-level I/O code. (2) Digital Audio Extraction. The process of capturing CD-Audio tracks digitally from a CD-ROM drive to hard drive, using software such as Astarte's CD Copy™ or OMI's Disc-to-Disk.™

dailies: Uncut footage shot each day during production. If picture editing is on film, with picture and synchronized *mag film*, those elements when edited together become the *workprint* and *worktrack*. Used the chart the progress of the film and for preliminary music cuts. Also called *rushes*.

daisy chain: See *serial*(2).

damping: (1) Damping is the addition of friction to a *resonance* in order to remove energy from a mechanical system, reducing the magnitude of vibration at *resonant frequencies*. For example, the reduction of movement of a speaker cone, due either to the electromechanical characteristics of the speaker driver and suspension, or the effects of pressure inside a speaker enclosure. The electrical analog of friction is *resistance*, and it is used to damp resonating electrical circuits, such as *crossover networks* and *filters*. See also *Q*. (2) Acoustic fiberglass material used inside speaker enclosures.

damping factor: (1) A factor defined as the *rated load* divided by the amplifier output *impedance*. (2) The ability of an amplifier to control the motion of a loudspeaker cone after a signal disappears, i.e., its ability to defeat the natural *ringing* tendency of the body (*cone*) in motion. An amplifier with a high damping factor looks more like a kind of short circuit to the speaker, reducing its vibration when the signal stops. The damping factor of an amplifier will vary with frequency, and sometimes a manufacturer will publish a curve of damping factor vs. frequency. The effect of high damping factors is most audible at low frequencies, where the primary *resonance* of the woofer cone, called *hangover*, is reduced in level.

DAR: See *Digital Audio Recorder*.

darkness: The amount of low-frequency, or corresponding lack of high-frequency, components of a sound. *Reverberation* from distant objects usually has fewer high frequencies and sounds darker than reverb from close objects. The opposite of *brightness*.

D

DASH: Digital Audio Stationary Head. A standard format for ensuring compatibility between Sony PCM-model digital *multitrack* recorders which use stationary, rather than rotating, heads. Originally, the DASH format was designed to support 2/8/16/24-track recorders using reel-to-reel tape. The 8-track and 16-track machines were never marketed, and a ¼" two-track model is no longer in production. The DASH specification now includes double-density, thin-film heads that allow 48-track recording on the same ½" tape originally used by the 24-track devices. DASH-format machines are backward compatible: 24-track machines can be used with newer models, and a project can be started on a 24-track machine and completed on a 48-track recorder, if needed, as the data from tracks 25-48 are written into the spacing between the original 24 tracks. DASH tapes run 30ips at up to 48kHz, with 44.1kHz and other sampling rates supported. In addition to 24-track or 48-track recording, DASH format provides two analog *cue tracks* and one track each for control and *timecode* signals. DASH format recorders are currently manufactured by Sony and Studer. See also *S-DAT*, *ProDigi*.

DAT: Digital Audio Tape. There are two formats: *R-DAT* which uses a rotating head assembly similar to a VCR, records diagonally across the tape, and includes a four-channel format which would permit recording of *ambisonics*; and the S-DAT which uses a stationary head and records several linear, parallel tracks of digital signals. There are no known commercial S-DAT products. The DAT standard format specifies a small cassette that provides up to two hours of 16-bit, *linear*, *sequential monaural*, *PCM* digital recording at a sampling rate of 32kHz, 44.1kHz, or 48kHz. Also called a *DCAC*, Digital Compact Audio Cassette. See also *Digital Compact Cassette*.

data compression: See *compression*(3).

data controller: A *controller change* message which is used to set some *parameter* in the receiving device, for example, the data increment and decrement switches on a synthesizer.

data dump: A packet of memory contents being transmitted from a sending device to a receiving device, usually in the form of MIDI *System-Exclusive* data, or stored in RAM. Also called a *bulk dump* or *block transfer*.

datafiler: A portable device for the replay of previously recorded MIDI data, used in live performances.

data slider: A *pot* fitted to a device such as a synthesizer which allows parameters within the device to be adjusted for programming, etc.

data thinning: A sequencer software feature which allows programs and/or devices to reduce the amount of MIDI data produced by *continuous controllers* such as *pitch-bend*, *after-touch*, etc. This is accomplished by only keeping the continuous controller data when the parameter changes, as opposed to sending all bytes of data, all of the time.

DAW: Digital Audio Workstation. See *workstation*.

dB: See *decibel*.

DB-9 connector: An industry-standard connector for *serial* machine control of professional audio and video transports. Developed by Sony, also called the *Sony 9-pin*.

D

DBS: Direct Broadcast Satellite. See *AC-1*.

dbx™: The dbx *noise reduction* system is a *broadband compander* system that is connected into a recording system in the same way that a Dolby system is. It provides up to 30dB of noise reduction, but unlike Dolby noise reduction, the dbx system works over the entire audio frequency range, using a 2:1 / 1:2 compression / expansion ratio. Dolby-encoded and dbx-encoded tapes are incompatible. Systems using dbx noise reduction are typically more expensive than systems using Dolby.

D.C.: Da Capo. Italian for “Head,” meaning “Play from the beginning.”

DCA: Digitally Controlled Amplifier. Sometimes short for Digitally Controlled Attenuator. The DCA of a digital synthesizer modifies the amplitude of the signal generated by the *DCO*. It is the digital analog of a *VCA*.

DCAC: See *DAT*.

DCC: See *Digital Compact Cassette*.

DCO: Digitally Controlled Oscillator. The microprocessor-controlled sound generator used in a digital synthesizer. The DCO directly generates the original signal that is used as the *fundamental* for the sounds created by the synthesizer. The keyboard tells the DCOs what pitch to produce; the audio signal may then be altered by sound modifiers, including a *DCW*, *DCA*, *differentiators* and *integrators*, and various *modulators* and *limiters*. The digital equivalent of the analog *VCO*.

D-connector: See *D-sub(miniature) connector*.

DDL: Digital Delay Line. See *delay line*.

dead: Acoustically absorptive. The opposite of *live*. See also *LEDE*.

decade: The interval between two quantities plotted along an axis where the second quantity is ten times the first. A frequency ratio or interval of 10:1, as opposed to an *octave*, which is a 2:1 ratio. Sometimes the *rolloff* of a filter or equalizer is expressed in dB / decade, rather than in dB / octave. A-rolloff of 20dB / decade is equal to A-rolloff of 6dB / octave. The decade interval has no musical significance, but is used in the discussion of *logarithmic* quantities such as *decibels*.

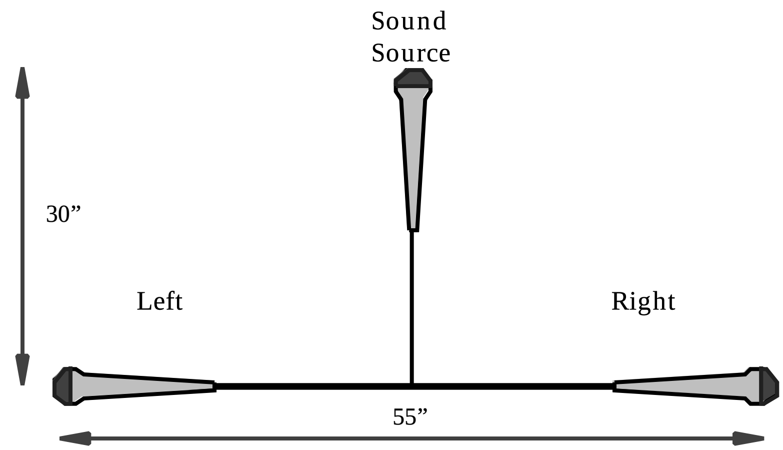
decay: (1) The time it takes for a sound to reach minimum *loudness*; the end of a sound. (2) The second of the four segments of a typical *ADSR* envelope. The decay control determines the amount of time it takes for the envelope to fall from the peak reached at the end of the *attack* segment to the *sustain* level. If no additional energy is put into the sound source (e.g., a cymbal), then the decay is the time during which the sound falls from the loudest point back to silence. (3) The time taken for *reverberation* to die away. See *decay time*, *RT-60*.

decay rate: The number of decibels per second by which *echoes* or *reverberation* of a sound diminish once the sound has stopped. Depending on the sound source and environment, the decay rate may be linear, i.e., a constantly decreasing number of dB per second, or it may begin to decay slowly and then fall off rapidly, or the reverse. Also, various frequencies of the sound may decay at different rates.

D

decay time: See *reverberation time*.

Decca trees: A triangular array of *omnidirectional* microphones, a type of true *spaced-microphone recording* technique, where the central channel is distributed equally to left and right. This yields a very stable central *image*, avoiding the *hole-in-the-middle* which is problematic with many space-pair arrangements. A variant on the Decca tree places three mics (L,C,R) in a triangle configuration, all set to *cardioid*. In all cases, the width of the tree is typically one-half to one-third the sound field width, and the center microphone is slightly closer to the performers. See *binaural recording*.



Decca Tree Microphone Placement

D

decibel (dB): A unit of measurement used to indicate audio *power* level, literally one-tenth of a bel, where the bel is a *power ratio* of 10:1. Technically, a decibel is a logarithmic ratio of two power measurements, which means that there is no such thing as a dB measurement in isolation. In order to measure a signal in dB, you need to know what power (watts, volts) it is referenced to and the *impedance* of the reference system.:

Number of dBs = 10 log (P1/P2),

where P1 and P2 are the two *powers* being compared, and where the log is base-10. Imprecisely, 1dB is the smallest increment in loudness detectable by a careful listener. An increase of about 3dB is a doubling of electrical (or signal) power; an increase of 10dB is ten times more power, but is only a doubling of perceived *loudness*. Some commonly used power ratios, expressed in dB:

Power Ratio	Voltage Ratio	Decibel Value
1	1	0dB
2	1.4	3dB
4	2	6dB
10	3.16	10dB
100	10	20dB
1,000	31.6	30dB
10,000	100	40dB
100,000	316	50dB
1,000,000	1,000	60dB
10,000,000	10,000	80dB
100,000,000	100,000	100dB

However,

Number of dBs = 20 log (V1/V2),

where V1 and V2 are the two *voltages* being compared, and where the log is base-10. This means that the answer is twice what it would be for a ratio of powers. In other words, double the voltage and the level goes up by 6dB; halve the voltage and the level goes down by 6dB. See Appendix A.

Amplifier Power (Watts)	Decibel Level (1W=0dB)
1	0dB
10	10dB
100	20dB
200	23dB
400	26dB
1,000	30dB
2,000	33dB

Commonly used *reference levels* are indicated by such symbols as:

dBm:	1mW=0dBm, 600Ω, a measure of *power*
dBV:	$1V_{RMS}$=0dBV, where V-0,775V, 600Ω
dBu:	0.775V=0dBu, 600Ω, 0dBu=0dBm, a measure of *voltage*
dBv:	Synonymous with dBu, but rarely used
dBA:	With reference to the *A-weighting* scale, at 40 *phons*
dBB:	With reference to the *B-weighting* scale, at 70 phons
dBC:	With reference to the *C-weighting* scale, at 100 phons
dBFS:	The reference signal is the device's Full Scale (peak signal limit)

D

decimation: A form of digital filtering whereby audio data is oversampled and then decimated to the required 44.1kHz. In practice, the sampling rate is 64 or 128 times 44.1kHz. A *digital brick-wall filter* is then applied to the data, resulting in a perfectly *phase*-linear transformation. [This type of filter is impossible in the analog domain due to the phase-shift caused by very steep roll-off filters. See *FIR*, *IIR*.] After the data have been filtered below the *Nyquist frequency*, the next step is decimation where the data are resampled to produce an output stream of 44.1kHz, with the attractive result that the excess data thereby provides increased bit-resolution. See *anti-aliasing filter*, *reconstruction filter*, *DSD*.

deck plate: In a tape recorder transport, the heavy metal plate on which the *headstack*, rollers, and other transport components are located.

decoding: (1) In signal processing, restoring a signal to its original state by reprocessing the signal in a *complementary* manner, e.g., a *NR* system's re-expansion of the signal during playback. (2) In digital recording, the entire process converting the encoded data stream back into an analog signal, including the process of error correction, i.e., *digital-to-analog conversion*.

deconvolution: A mathematical process for separating two signals that have been convolved. See *convolution*.

decrescendo: A musical term indicating a gradual reduction in *loudness*.

de-emphasis: The *complementary* equalization which follows *pre-emphasis*. Sometimes redundantly called *post de-emphasis*.

de-esser: A special type of *compressor* that operates only at high frequencies, usually above 3kHz-4kHz. It is used to reduce the effect of vocal *sibilant* sounds. De-essers are usually used only for vocal music.

defeat switch: A control that can be used to mute a signal on a mixer.

definition: A qualitative term that denotes the clarity of a sound. A sound with poor definition may, like some woodwinds in their middle ranges, be easily mistaken for a similar sound. In recording, the apparent definition of a sound can be increased by boosting the frequency *band* characteristic to the specific sound of the instrument, and cutting other frequencies it has in common with other sounds in the mix.

delay: (1) The first stage of a five-stage D(elay)AD(ecay)SR envelope, which delays the beginning of the envelope's *attack* segment. See *ADSR*. (2) An audio effect which temporarily suppresses the beginning of a sound, producing *echo, chorusing*, *phasing*, and *flanging* effects. A modulated digital delay effect which varies the time and/or intensity of the delay effect over time. See *double tracking*. (3) A signal processor used for flanging, chorusing, and echo, that holds its input for some period of time before passing it to the output, or the algorithm within a signal processor that creates delay. Also used in artificial *reverberation* systems and to provide delayed sound to certain loudspeakers in *time-coherent* sound reinforcement systems. (4) See *MIDI delay*.

delay line: Used to simulate an acoustic *echo* or *reverberation*. There exist both digital delay lines (*DDL*) and analog delay lines as well. The original delay lines were made by using tape recorders to record a signal while playing it back on the same machine. See *tape delay*.

D

delay line feedback: A type of *modulation* which creates a series of *echoes* when the modulation source is boosted. The greater the amount of feedback, the more repetitions of each echoed event.

delta modulation: In the UK, often, and more properly called *delta-sigma modulation*. A type of *PCM* which differs from most other digital *encoding* schemes in that the signal, after being sampled at a fast rate, is encoded as the difference between successive levels, rather than as the absolute level of each sample. Delta modulation requires a very high *sampling rate*, usually around 700kHz, but the digital *words* need for each step contain one bit, whereas conventional PCM samples at only about 45kHz but requires 14-16 bit words. The “delta” phase of delta modulation involves taking the difference of the reconstructed signal and the incoming signal to adjust the output to minimize the quantization error; the “sigma” part involves the summation of the differences to reconstruct the original signal, although there are a number of variant algorithms based on this basic theme. The reason for the popularity of delta modulation-type converters is the inherent *linearity* of the process. See also *ADPCM*.

delta-sigma modulation: See *delta modulation*.

delta time: See *SMF*.

demodulator: A device which recovers the audio signal from a modulated *carrier* waveform. Also called a *detector*. See *amplitude modulation* and *frequency modulation*.

depth: (1) In stereophonic reproduction of music, depth refers to the perceived relative distance between the listener and the various instruments in the sonic *image*. (2) In a digital *delay* or *flanger,* a parameter which modulates the length of delay around the specified delay time. Because this happens in real-time, the pitch of the input signal is varied, causing the output signal to have an apparent *vibrato* effect. The speed of this vibrato is set by a *rate control*.

depth perception: See *depth*(1).

DES: Dolby-Encoded Stereo. A *noise reduction* system employed in the reproduction of stereo *optical tracks* in movie theaters.

desk: See *mixer*.

detector: (1) See *frequency modulation*. (2) See *level-sensing circuit*.

detune: (1) (*noun*) A control that allows one *oscillator* to sound a slightly different *pitch* than another. (2) (*verb*) To slightly change the pitch of one oscillator relative to another, producing a fuller sound.

D

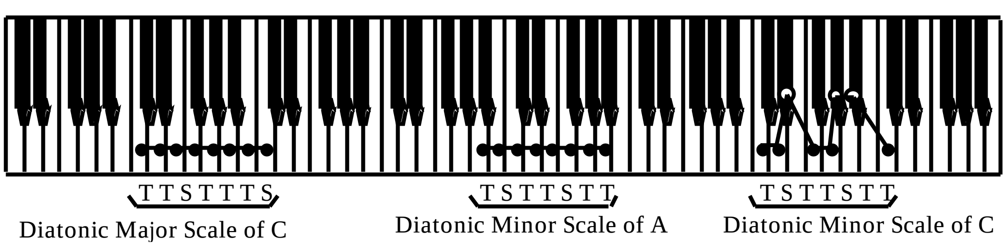
DI: Direct Injection. Also called a *direct box*. (1) The use of some form of mechanical or electrical pick-up mechanism on an instrument for the purpose recording or amplification. DI also refers to the connection of an electronic keyboard or power amplifier feeds to a mixer. A DI consists of (usually) a small electronic box into which an instrument is plugged and the *electroacoustic* pick-up attached to the instrument itself. Pick-ups can be *electromagnetic*, as on electric guitars, *piezo* devices, and also *contact mics*, also called *bugs*. All types of pick-up have unbalanced outputs at mic-level (~-50dBu), so the DI box has to balance the signal and drive it to the mixing desk. DIs can be *passive* or *active* in the typical sense. Active DIs have some form of electronic amplification built-in; this is only a buffering amplifier, separating the instrument pick-up from the rest of the DI, yielding no significant gain. Active DIs offer better sound and playability over passive devices, but require batteries, phantom power, or some other means of powering the internal amplifier. (2) Any device used to convert unbalanced lines to *balanced lines*.

dialog normalization (DN): There is a wide difference in the apparent loudness between different TV programs' audio content. In *DTV*, with by standard is *AC-3* encoded, a program producer chooses one of 31 different dialog normalization (abbreviated DN or "Dialnorm") values and this parameter is carried within the AC-3 datastream, where each step represents a 1dB change in level. The DN value is the difference in dB between the maximum level possible (0 dBFS) and the average loudness level of the program material. The smaller the difference between the maximum and program average levels, the lower the DN value is assigned. The lower the DN value, the lower the output volume of the AC-3 decoder is set in direct proportion, meaning that subjectively louder programs will be played back at lower volumes than those in which the average program level is less loud. This supposedly will obviate the user having to adjust the volume control between programs, once the audio listening level is set by the user.

dialog track: The edited track on magnetic film containing the dialog portion of a film's sound. Sometimes there may be a separate track for each actor in a scene, requiring the tracks to be mixed down to a single track. The "D "part of *DME*.

diaphragm: The membrane part of a microphone's *capsule* or cone of a loudspeaker that moves in response to sound waves or an incoming signal, respectively.

diatonic: A musical *scale* of eight notes spanning one *octave*, consisting of an ascending pattern of two *whole-steps*, a *half-step*, three whole-steps and another half-step. There are two types of diatonic scale in common use in western music: the diatonic major *scale* and the diatonic minor scale. Music which includes notes outside of the diatonic in which the piece is written is said to be *chromatic*.



Diatonic Major and Minor Scales

D

diatonic comma: After playing the *Circle of Fifths*, i.e., twelve ascending perfect *fifths*, followed by seven descending *octaves*, the *pitch* discrepancy between the ending note and the starting note is called the diatonic comma, or the *comma of Pythagoras*. This discrepancy amounts to a little over 1%, or about one-sixth of a half-step and gives rise to various *temperaments* in an attempt to distribute the error as harmoniously as possible. See *scale construction, syntonic comma.*

dichotic: Dichotic generally refers to headphone listening where each ear hears a different signal, as opposed to *diotic*, where both ears hear the same signal. See also *monotic*.

difference tone: A tone produced by combining two tones which are not part of *a harmonic series*, having a frequency difference of 20Hz or greater. Any slower than 20Hz, and the difference of the two notes will be perceived as a pulse, called *beating*. Also called a *resultant tone*. Also called a *Tartini tone.*

differential amplifier: Usually one of the signal input terminals of an amplifier is connected to the chassis of the amplifier, i.e., it is grounded. The amplifier is then sensitive to the voltage difference between the input terminal and ground. However, in a differential amplifier, neither input terminal is grounded. Instead, the amplifier is sensitive to the voltage difference between the two inputs. Used in professional mic preamps where a low-level signal has to go some distance, a differential amplifier cancels the *hum* induced by the proximity of the two input wires to a source of interference. In the UK, a differential amplifier is called an *inverting amplifier*. See *differential input*, *common mode.*

differential input: Signal input response to amplitude differences between two out-of-phase signals. Used in a *balanced* wiring system where the two wires carry signals that are identical, but 180° out-of-phase. The phase difference means that as a signal increases in voltage along one line, its mirror image on the other line decreases. This is useful because signals, such as *hum* and *noise* which have accumulated along a cable acting as an antenna, that are in phase are cancelled. See *common mode, differential amplifier*.

differential output: The output of an amplifier designed to provide two signals that are completely identical, but of opposite *phase*.

differentiator module: A *highpass* filter which can accentuate the higher-frequency *harmonics* and *transients* of a sound *envelope*. Compare with an *integrator module*.

diffraction: The bending of a sound wave around an obstacle and the *reflection* of a sound wave from an obstacle in its path are called diffraction. It is *frequency* dependent. Where the wavelength is short compared to the obstacle, reflection will occur as well as bending of the wave front. When the wavelength is long with respect to the obstacle, little reflection will occur and the bending will be more pronounced. See also *refraction*.

digital: In audio, the opposite of *analog*. The representation of audio or video as a series of encoded binary *amplitude* values, rather than as a continuous *waveform*.

Digital Acoustics Processor (DAP): A consumer audio device that attempts to simulate the acoustics of an auditorium or other room by adding suitable time *delays* and synthetic *reverberation* to recorded signals.

D

digital audio: The application of digital technology to the recording, processing, and reproduction of music is somewhat loosely called digital audio, as opposed to *analog*.

Digital Audio Broadcasting (DAB): An alternative to AM and FM broadcasting with audio quality comparable to that of the CD, it does not suffer from fringe area fading or *multipath distortion*, and requires less radiated power than conventional broadcasting (1kW versus 50kW for AM and up to 100kW for FM.)

digital audio extraction: See *grabbing*.

Digital Audio Mastering System: See *digital multitrack*.

Digital Audio Recorder (DAR): Any type of audio recording system which records upon a digital medium, such as *DAT* or hard disk. DAT or *DCC* recorders, *digital dubbers, digital multitracks*, and hard-disk recording systems are all example of digital audio recorders. These recorders are an alternative to *analog* recorders, such as traditional cassette or reel-to-reel formats which do not convert the *waveform* to a digital representation prior to writing it to the recording medium.

digital black: In digital audio, a term which means complete silence. Digital black is calculated by taking the sample *word* length (e.g., 16, 20, or 24 bits) and multiplying this *bit depth* by 6dB, a number which represents the *dynamic range* represented by one bit. In a 16-bit system, for example, *full code* represents 96dB, the maximum amplitude that the system is capable of encoding without *clipping*. Digital black is at the opposite end of that dynamic range, or 96dB down from full code amplitude.

Digital Compact Cassette: A type of recording format announced by Philips in 1990, designed to compete with the *R-DAT* format. The system allows for the recording and playback of analog cassettes as well as DCCs on the same machine. Uses PASC (Precision Adaptive Subband Coding), derived from the *MPEG-1*, Layer 1 data reduction system to provide data compression (lossy) for the recording of digital audio on ⅛" wide magnetic tape at 1⅞ ips. This format has not been widely adopted. Sometimes called *DCAC* for Digital Audio Compact Cassette. See also *MiniDisc*, *DAT*, and *CD*.

digital delay line (DDL): See *delay line*.

digital dubbers: Film industry term for a multitrack digital recorder, usually having eight tracks per unit, that use removable hard drives or magneto-optical drives as the recording medium. The term is partly a misnomer because previous film sound terminology had used *dubber* to describe a copying device as opposed to a recording device.

digitally controlled amplifier: See *DCA*.

digitally controlled oscillator: See *DCO*.

digitally controlled waveshaper (DCW): A DCW varies the *timbre* of synthesized sound by modifying the *harmonic* content of the tone produced by a *DCO*. See *waveform*.

D

digital multitrack: A device for recording multiple channels of digital audio data at various sampling rates. Two formats have survived: the Sony / Studer *DASH* format and *MDM* machines of either *ADAT* or *DTRS* type. The first digital multitrack recorder was introduced in the late 1970s by 3M, a 32-track recorder called the Digital Audio Mastering System.

Digital Signal Processing (DSP): The manipulation and modification of signals in the digital domain, possibly after having undergone analog-to-digital conversion.

digital time delay: See *delay*.

digital-to-analog converter: Commonly abbreviated *D/A, D/A converter* or *DAC*. A device that changes the sample words put out by a digital audio device into analog fluctuations in voltage that can be sent to a mixer, amplifier, or speaker. All digital synthesizers, samplers, and effects devices have DACs at their outputs to create audio signals, as the *transducers* in loudspeakers are inherently analog devices.

Digital TeleVision (DTV): See *DTV*.

digital watermark: The solution for a piracy and duplication protection scheme developed jointly by Sony and Philips which writes copyright data encrypted within the CD / DVD etc. disc itself. This scheme would, for example, encode discs with a country code so that these discs would only play on players from the same country. This is presumably better than older forms of digital copy protection which tried various *pilot tones* or random number generators, failing ultimately because the results were either too audible or too easy to circumvent. In a digital watermark, the copyright data are stored as a modulation of the width of the injection-molded *pits*. Duplicating the watermark would require the same equipment as that which produced the disc *stamper*, the distribution of which is presumably tightly controlled. It is also possible to synchronize the modulation of the pit widths so that there is a visible pattern formed on the disc pit substrate itself, making an “analog” watermark (without the need for water, of course.) In addition to the watermark and country codes, identifiers for the mastering house and pressing plant, glass master number, ISRC catalog numbers, etc. can be stored.

The digital watermarking technology has been called Pit Signal Processing (PSP) which works by modulating the strength of the laser used to record the digital data onto the *glass master*. One by-product of the watermarking process is that the *EFM* used to encode audio data onto the CD *master* allows the pits to vary in length between 3-11 units. These slight errors in length, or “jitter” result in slight timing errors which can cause a *smearing* of the stereo image as well as an increase in HF noise. The more rigid requirements of pit length control in watermarking should result in a significant reduction of pressing-induced jitter, just generally improving the CD production process.

diminuendo: Synonym for *decrescendo.*

diminution: (1) The reduction of a major or perfect *interval* by one *half-step* to make a *diminished interval*. (2) The appearance of a musical idea in note durations which are shorter than those used for its first appearance. The opposite of *augmentation*. (3) A method ornamentation where notes of long duration are broken into a number of shorter notes, often at different pitches, e.g., a trill.

D

DIN: Deutsche Institut für Normung. A German standards organization that proposed a set of connector configurations in the early 1960s. The standard MIDI connector is the 5-pin DIN where:

Pin 1:	No connection
Pin 2:	Ground
Pin 3:	No connection
Pin 4:	+5V
Pin 5:	MIDI datastream

DIN sync: See *pilot tone*.

diode: A diode is a circuit element which will pass *current* in one direction only, from the anode (positive) to the cathode (negative). Used to make *DC* from *AC*.

diotic: Literally, "with two ears." Diotic generally refers to headphone listening whereby the two ears hear the same signal, as opposed to *monotic*, where only one ear hears the signal. See also *dichotic*.

dip filter: A *parametric equalizer* with an extremely narrow *Q*, designed to remove noise in a small band such as that from a camera or light.

dipole: In *loudspeaker* design, a dipole radiator is a system which radiates forwards and rearwards with equal energy, but with opposite *polarity*. Examples of dipole radiators are *electrostatic* loudspeakers and *planar* speakers. Some cone-type speakers have dipole radiators. For a dipole radiator to have adequate low-frequency response, it must be very large to prevent the rear wave from canceling the front wave. Also, the dipole radiator must not be placed close to and parallel to a wall, working best when not near reflective surfaces.

direct box: See *DI*.

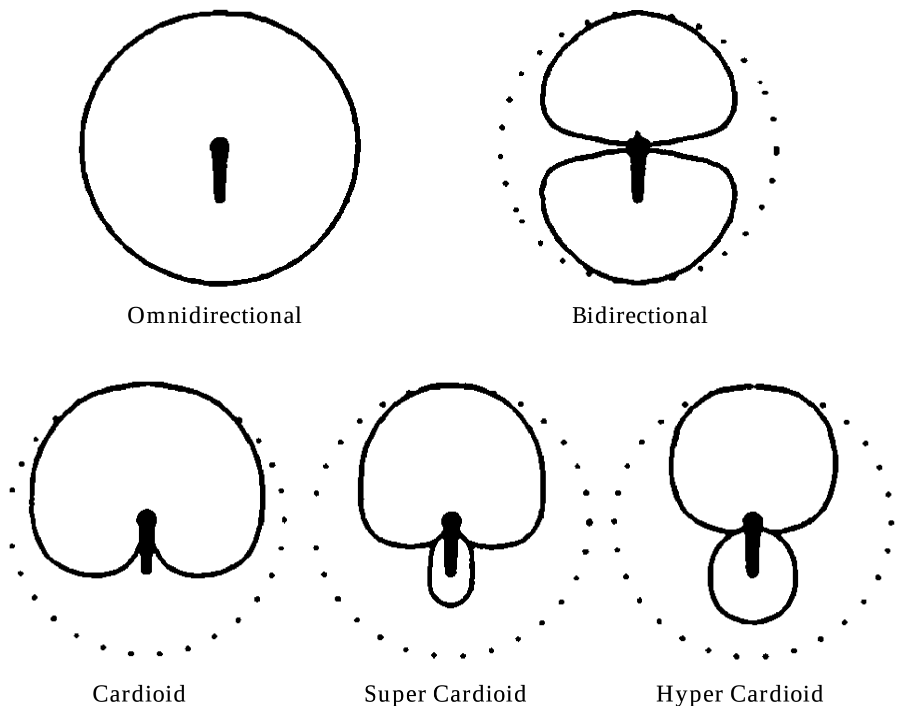
direct coupling: A connection between two devices that allows both DC and AC between them.

direct current (DC): Current in only one direction. DC always has the same direction, from the positive to the negative terminal. Compare with *AC*.

direct field: See *reverberant field*.

D

directional microphone: A microphone which does not have a spherical *polar pattern*, i.e., a microphone which is not *omnidirectional*, such as a *cardioid*, *figure-eight*, etc. having an *acceptance angle* of less than 360˚. See *directivity*.



Directional Microphones

directivity: Describes the angle of coverage of a loudspeaker system or microphone *acceptance angle*, both in the vertical and horizontal planes. High directivity equates to a narrow angle of coverage. The *directivity factor* is a measure of the directionality of the sound output of a loudspeaker. See also *Q*.

direct metal mastering (DMM): A system for cutting a metal mother on a record mastering lathe, eliminating the lacquer master and metal master steps. Release pressings made from a stamper are thus only two stems from the DMM and thus have less noise and distortion than those made by the older, five-step process. The DMM process is also used in CD mastering.

direct output: A recording console output taken directly after the input module and main channel fader, but before the *panpot* and output *bus* assignment switches. This output is sometimes used to avoid crosstalk that may be introduced if the signal is allowed to flow through the complete circuit.

direct positive: A optical (photographic) sound recording that, when processed, results in a track that can be played and edited; now obsolete.

D

direct radiator: A *loudspeaker* which does not have a horn between the moving element and the air is called a direct radiator. Most direct radiator-type speakers are for home use, while horn-type speakers are preferred for *sound reinforcement* applications. Direct radiators generally provide smoother, more uniform response, while horns are much more efficient, providing a greater output level for a given power input. Also, horns have greater *directivity*, which is desirable in sound reinforcement systems. See *compression driver*.

direct sound: (1) See *reverberant field*. See also *free-field*, *critical distance*. (2) The sound received at the recording console from an electronic instrument when using a *direct box*.

Direct Stream Digital™ (DSD): A proprietary CD/DVD data *format* proposed by Sony and Philips for use in the *SACD* . DSD bandwidth is normally 2.844Mb per channel (64 times 44.1kHz), with optional sampling rates of 32 or 128 times 44.1kHz, yielding a slightly higher data rate than that required by 24-bit/96 kHz resolution conventional A/D-D/A systems. DSD uses a *delta modulated* ADC to generate a 2.8224MHz, 1-bit signal, a rate chosen as a simple multiple of the lowest common high-fidelity *PCM* sampling rate, 44.1kHz. The 1-bit datastream is recorded directly to disk, avoiding the *decimation* and oversampling stages, inherently improving the resultant audio quality, simplified error protection, and there is no need to frame the data into words. Sony claims that the sampling rate is so high that it more nearly approximates the original analog signal, allowing equalizers and other effects processors to better simulate analog effects. It is claimed that DSD can have *frequency response* up to 1MHz, or up to a *dynamic range* (within the audio bandwidth) of 120dB, equivalent to about a 20-bit resolution. A number of *DSP* algorithms are available which allow the optimization of either bandwidth or dynamic range. [Note that these benefits apply only to those players which support the DSD standard, as replay on any conventional CD, DVD or other digital format would require the decimation and framing steps.]

Direct Stream Transfer™ (DST): Philips' proprietary technology for *lossless* 2:1 data reduction in digital recordings on *SACD*. DST is optimized for audio-type signals, allowing sufficient storage capacity for double the old CD standard of 74 stereo minutes. By incorporating DST into the SACD standard, it is possible to store two complete 74-minute versions of audio material so as to combine a stereo *DSD* track, and a 6–channel *surround* mix, plus other data, text, graphics, and video, all on the single, high-density DASD layer.

direct-to-disc: A type of analog LP mastering in which a master tape is not used. The signal directly from the control console is used to cut the original acetate disc. This means a direct-to-disc recording cannot be edited, and is made *live*.

direct-to-disk: Recording digital audio data onto a hard disk for replay or editing.

direct-to-two-track: A method of recording in which the instruments and vocals are mixed and recorded directly onto a stereo *half-track* or *DAT*. If analog, no further changes or remixing is possible. The fidelity, edibility, and relatively low-cost of direct-to-two-track digital recording has revived the popularity of this medium for making master tapes, especially those intended for release on CDs. Also called *live-to-two-track*.

DirectX: The most common audio effects plug-in format used by Windows™ software.

discrete: Refers to a 1:1 relationship of recorded tracks on an audio medium or film print and the resulting number of speaker channels. Contrast with *matrixed* sound.

D

discrete 6-track: Traditionally means the five-speakers-behind-the-screen system made popular by the Todd-AO 70mm process (although first used for *Cinerama*). Today the term sometimes means six nonmatrixed tracks, assigned to L,C,R,LS,RS,Subwoofer. See also *5.1*.

discrete output: A direct output from a mixer channel, which services only that one channel.

disk-at-once: A CD production process where the entire disc is written in one burn; the laser is never turned off. Ideal for audio, disk-at-once mode allows gaps between tracks of any length (except the first track, which must have a 2-3 second gap.) Compare with *track-at-once*.

dispersion: (1) The spreading of sound waves as they leave a *loudspeaker*. (2) Another term for *refraction*.

displacement: The distance between some measured position of a moving object, e.g., a speaker cone, and its static position. Also applies to the position of air molecules in a sound wave. See *rarefaction*.

distant miking: The opposite of *close miking*. In recording, the placement of one or more microphones relatively far away from the sound source. This technique picks up a substantial portion of *reverberant* sound, and is therefore used to make most classical or orchestral recordings to capture the sound as closely as possible to that experienced by an audience. In the studio, close and distant mics on any instrument or group may be blended at the mixer to achieve the desired sonic *image*. See *depth*.

distortion: Also called *correlated noise*. Any (usually) unwanted sound which varies with the input signal. (1) Any undesirable change in the characteristics of an audio signal of six types: The two types of (i) *nonlinear distortion* are *intermodulation distortion* and *harmonic distortion*. Other types of distortion are (ii) *frequency distortion* (pitch-shift), (iii) *phase distortion* (time shift), (iv) *transient distortion*, (v) *scale* (*volume*) *distortion*, and (vi) *frequency modulation distortion*. There are other factors which cause music reproduction to be untrue to the original but which are not considered distortion, such as background noise, and a lack of directional realism and proper ambience due to the use of too few channels of reproduction. See *noise*. (2) A sound *modulation* technique whereby the original waveform is distorted intentionally.

D

dither: A noise-based rounding method used to add a tiny amount of controlled noise to a digital audio file to make other, more objectionable errors less obvious and / or to convert from one word size down to a smaller word, e.g., from 24-bit resolution to 16-bits. *Redithering* is a dithering process used in digital-to-digital signal processing to distinguish it from the dithering process used during the original *A/D* conversion. There are several ways to dither:

(1) Add *white noise* at about half or one-third the value of the *LSB*, or half the level that a system can transmit. Thus, for a 16-bit converter encoding a range of 2V, the LSB is equivalent to a difference of 30.5µV (2÷65,536V), yielding dithered noise of about 10-15µV;

(2) *Second-Order* dither in which the dither signal (white noise) is processed by *a high-pass filter* to remove low-frequency components. This makes the noise less apparent to the ear, since humans are less sensitive to high-frequency noise than other levels;

(3) *Noise-shaping* in which the dither signal is run through a set of filters to provide the most energy in regions where the ear is the least sensitive;

(4) (Triangular Probability Density Function) where before the noise is shaped, it has a different spectral content than ordinary white noise. It has better *noise modulation* performance, which is how the noise affects the signal itself;

(5) Sony's *SBM* (Super Bit-Mapping) where the audio is run through a processor with an algorithm that maps a series of higher-resolution samples to a series of lower-resolution samples;

(6) The Apogee *UV-22*, which is not really dither. Instead, it uses a periodic signal centered around 22kHz that has good performance in terms of audibility and noise modulation. Placing the signal this high in the audio spectrum makes it very difficult to hear and produces fewer effects on the character of the signal.

DLL: Dynamic Link Library. Files used by PC-type computer application programs to provide additional functionality to the computer's operating system. The equivalent of an *extension* file on a Mac.

DLS: DownLoadable Samples. A standard which allows multimedia games or other programs to contain samples which would be downloaded to the playback hardware. This is a hybrid between *streaming* digital audio (direct audio from CD-ROM, for example) and MIDI, which alters predefined sounds. DLS is an attempt to solve the problems inherent in the ambiguity of the *GM* specification and the somewhat random sounds produced by the playback hardware. DLS-1 specifies: 16-zone multisampled soundbanks with 128 zones for drum banks, *pitch-shifting*, *ADSR* for amplitude and pitch, and *LFO* for amplitude or pitch. DLS-1 was ratified by the *MMA* in 1997. See *DLS-2*.

D

DLS-2: Downloadable Sounds Level 2. An improvement over DLS which moves closer to enabling multitrack audio and MIDI programs to be used by audio and MIDI hardware and software interchangeably. This is accomplished by transferring entire multisampled instruments, along with the MIDI data, to the end-user's platform, resulting in playback as the author intended. In addition to the specification of *DLS-1*, DLS-2 specifies: resonant filter control, adds delay and hold to envelope segments, effects routing for chorus and reverb, no limit to the number of regions in any soundbank, each reason can have independent envelope and filter data, and is extensible to include other forms of synthesis. Ratified by the *MMA* in early 1999. See also *MPEG-4*.

DLT: Digital Linear Tape. A tape-based computer backup format developed by Quantum Laboratories.

DMA: Direct Memory Access. A digital logic design which allows peripheral devices to communicate directly with the system memory, rather than requiring the central processor to stop whatever processing it was doing to control communications between an attached device (usually having some kind of I/O function) and the computer memory.

DME: Dialog, Music, and Effects. The three basic stems of film soundtracks, originally meant to denote the 35mm 3-track master mix of *academy mono* films**.**

DN: See *dialog normalization*.

Dolby Digital™: The *5.1*–channel digital format created by Dolby Laboratories, first used in 1992 for "Batman Returns." In current usage, the term applies to both the Dolby 35mm theatrical format, which contains the data printed optically between the sprocket holes, and for video formats, such as DVD, laserdisc, and DTV. *AC-3*, as Dolby Digital was first called, used *RF modulation* of the digital signal onto one of the analog tracks, making it possible to fit an entire movie, along with the already existing digital tracks, onto a conventional laserdisc; a *demodulator* was needed to recover the audio back into a digital bitstream.

The Dolby Digital format is a *surround-sound*, *split-band*, *perceptual coding* scheme. AC-3 was designed as a *5.1* multichannel format, using approximately 13:1 *lossy compression*, and is specified as the *matrixing* format for *DVD* and *DTV*. Also used in HDTV broadcasts, *SR-D*, and *DSD* cinema productions. Versatile, in that parameters such as bit-rate and number of channels can be tailored to particular applications, unique in that the data bits are distributed dynamically among the filter *bands* as needed by the particular frequency spectrum or dynamic nature of the program. Data rates vary from 32kbps for a single mono channel to as high as 640 kbps for 5.1 format. The data rate is 320kbps for film, 384kbps for laserdisc, and 384kbps or 448kbps for DVD, although the maximum throughput for the specification is 640kbps. Dolby's current decoder can accept incoming data at 32kHz, 44.1kHz, or 48kHz sample rates, with bit depths of 16, 18, or 20 bits. The commercial competitor to the Dolby Digital format is *DTS*. See *metadata*, *audio coding mode.*

Dolby Fax: See **ISDN**.

D

Dolby Motion Picture 4:2:4: A *matrixed surround-sound* system which combines multichannel *LCRS* audio in such a way that the encoded signal forms a stereo-compatible, two-channel format for recording and broadcasting. Originally developed in 1977 for "Star Wars," and now in wide use. As with any matrix system, it is impossible to completely recover the original multichannel signals with perfect isolation. The decoder disguises this problem through a *steering* process which emphasizes the signal emanating from its appropriate loudspeaker by canceling out a portion of the *crosstalk* in adjacent channels. See also *free encoding, Pro Logic.*

Dolby noise: The *Dolby-SR* analog allows the comparison of the recorded Dolby noise on a tape to that generated by the decoder, with four continuous seconds of noise to identify the generator, and two 2-second sections of noise indicating that the monitoring is off-tape. This allows for confirmation of correct EQ settings as well as playback verification. The broadband *reference signal* used to correctly calibrate the different Dolby *codecs* is called Dolby noise. See *Dolby tone.*

Dolby noise reduction: A type of *two-ended*, dynamic *noise reduction* for *magnetic tape recording* and playback. The essential difference between a *compander* and the Dolby system is that the Dolby system is frequency-dependent. The compander was developed to reduce *distortion*. Dolby applies companding to frequency variations in addition to signal amplitude variations, adjusting gain as frequency changes. The Dolby-A and Dolby-SR systems are used for professional recording in studios. Consumer tape decks use either a Dolby-B or Dolby-C system. Dolby-B operates only at high frequencies and reduces tape hiss by about 10dB. Dolby-C works over a slightly wider frequency range, providing a noise reduction of up to 20dB. All of the Dolby systems operate on quiet passages, below levels of about -10VU. Very strong signals, such as over 60dB or at frequencies below 500Hz are not affected by the Dolby system because these signals are not degraded by tape noise. When the recorded signal is played back, the Dolby circuit reduces the accentuated high-frequency signals so that the *frequency response* of the record / playback system is flat, hence reducing also the high-frequency tape hiss, improving the *S/N ratio* of the taped music. See *asperity, Barkhausen effect, compander, dbx, spectral recording.*

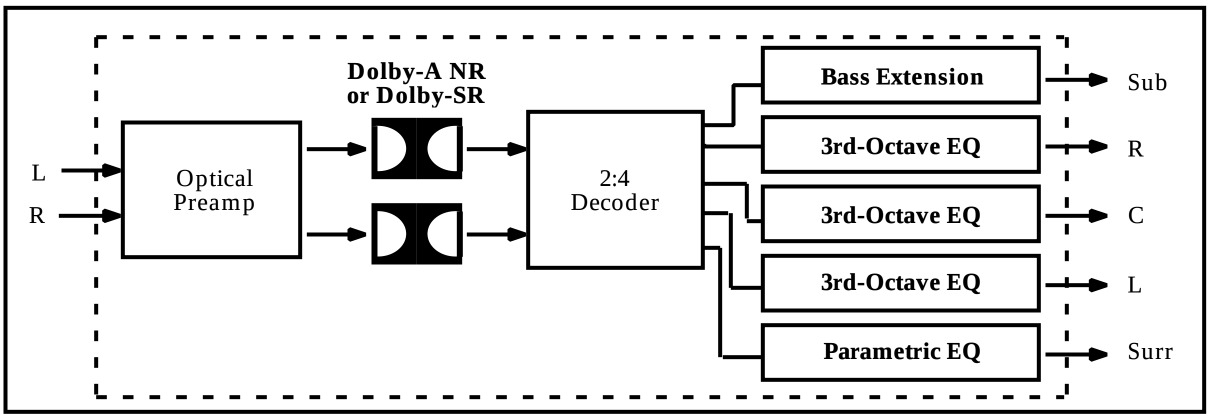
Dolby ProLogic™: A four-channel *perceptual coding* scheme developed by Dolby Labs where an *LCRS* audio signal is converted into two channels of analog audio, then recovered to yield discrete left, center, right, and mono surround channels. This is a hardware version of their surround decoder originally developed for the *Dolby Motion Picture 4:2:4* matrix *surround-sound* system, developed for the Star Wars picture in 1977. If a *subwoofer* is used, it is generally fed by lowpass-filtering a mix of the three front channels at the receiver.

Dolby SR™: Dolby-SR is a complex type of *two-ended*, dynamic *split-band, noise reduction compander* system that outperforms Dolby-A, -B, or -C systems and also results in reduced distortion in most cases. This was developed as an upgrade to the professional Dolby-A, featuring an improvement in tracking accuracy and sliding bands, and closest to C-type noise reduction. S-Type noise reduction is the consumer analog. See *SR.D.*

Dolby SR.D™: A system developed by the Dolby company for placing a digital audio soundtrack onto 35mm film, first used in 1992. The soundtrack includes a *Dolby Digital* mix, as well as an *SR* analog stereo optical track. The data are compressed and printed onto the film between the sprocket holes. See *surround-sound, perceptual coding, Dolby surround-sound, spectral recording.*

D

Dolby Stereo™: The original Dolby Surround system which used four audio channels carried on a stereo *optical track* on 35mm film, using Dolby *perceptual encoding*. On 70mm film, six audio channels are recorded on discrete magnetic tracks laid onto the film. In the broadest and most common usage, the trademark that appears on movie prints, advertisements, and posters which means that a given film has been released in prints that employ Dolby *A-Type noise reduction encoding*. Beginning in 1987, *Dolby-SR* has been available on 35mm *stereo optical prints*. Dolby Stereo on 70mm usually means four discrete primary channels (LCRS) with the left-center and right-center tracks dedicated to low-frequency information (below 250 Hz). The four tracks are normally use A-Type encoding, although selected 70mm films, since 1987, have utilized Dolby-SR encoding. See *film soundtrack*.



Dolby Stereo

Dolby Surround™: The Dolby Laboratories trademark used for surround-encoded material on non-film media, such as videocassettes, videodiscs, and television broadcasts, as well as for home surround decoding devices that do not have *matrixed* center-speaker output. See *AC-3, Dolby Stereo, ProLogic, Dolby Motion Picture 4:2:4, SR.D, matrixing, surround-sound.*

Dolby Surround EX™: The digital release format developed by Dolby Laboratories and THX for use in “Star Wars: Episode One--The Phantom Menace.” Three surround tracks are derived by matrix-encoding them in the two previously existing surround tracks. This should not be referred to as a 6.1-channel format because the additional surround channel is not a *discrete* channel.

Dolby tone: A *reference tone*, usually recorded at the head of a Dolby-A recorded tape, by which the threshold levels of the Dolby noise reduction system are adjusted for proper encoding and decoding of the *companded* signal.

domain: In *magnetic recording tape*, the smallest ferric oxide particle that can be considered as a separate magnet. Defined as 10^{18} molecules of ferric oxide, or, less than one billionth of a gram of material. See *Barkhausen effect*.

dominant: See *Circle of Fifths, key*.

D

Doppler effect: The apparent change in the pitch of a sound when the source of the sound is moving with respect to the listener. Also called *Doppler distortion*.

DOS: Disk Operating System. The original operating system for PC-type computers. Much of the Windows™ operating system is written in DOS; NT is not.

double-system sound: (1) A method of producing sound motion pictures where the soundtrack is recorded on a magnetic tape recorder which is separate from the video recorder and which is synchronized with the movement of the film in the camera, projecting a film with the picture on 35mm film, in interlock with the soundtrack, most commonly on *mag film*. The synchronization was originally done by recording a special tachometer signal on one track of the tape, but is now done with *timecode*. Examples of double-system sound are film and *Nagra*, film and mag *dubber*, videotape and audio tape. (2) A film or video production that utilizes sound recorded on a separate tape recorder, such as a DAT or Nagra. This term is still used even if the video recorder is also simultaneously recording the sound. Compare with *single system*. See *mut*. Also called *sep mag*.

double-tracking: Originally, double-tracking meant the recording of a vocal track on one tape recorder track, then listening to this while recording another similar track. The two tracks are combined and rerecorded into a single track, which will sound more diffuse due to slight differences in the two original tracks as double-tracking produces a slight *chorus* effect to voices. In this case, it is also called *re-tracking*. Double-tracking can be done with *DSP*s which introduce a small randomly varying time delays to one signal and then combine it with the original signal. See *delay*(3), *stereoizing*.

doubling: If a loudspeaker is driven too hard in its low-frequency range, it will produce second-*harmonic distortion*, sometimes with greater amplitude than the fundamental. The doubled frequency sounds one *octave* higher than the fundamental, and is often not musically annoying. This is called *frequency doubling*, or simply doubling.

downbeat: See *beat*.

downmix: A mix derived from a multichannel (usually *5.1* format) source to create a compatible stereo, mono, or other version of fewer channels. The common use of downmixing today occurs in consumer Dolby Digital products to play back a 5.1-channel DVD mix either via Dolby Pro-Logic decoding or in standard two-channel stereo for headphones. In those instances, an *Lt-Rt* or an *Lo-Ro* respectively, are the result.

downward expander: See *noise reduction*.

DP5xx encoding: A family (DP521 / DP522 / DP523 / DP524) of 2-channel *codecs* used for point-to-point and point-to-multipoint signal distribution: ISDN, Switched-56, T1 or DS-3 networks; recording / post-production studio interconnection with or without video; and voice-over and other applications. *AC-2* and *AC-3* perceptual coding algorithms are supported to provide audio transfers at a total data rate between 56 kBps and 384 kBps. With AC-3, single-channel, two-channel, and composite stereo algorithms are supported.

DRAW: Digital Read After Write, an erasable CD that can be re-recorded.

D

drawbar: On a Hammond organ with tonewheels, a slider that shortens the distance between the axle bearing the wheels and the *transducer* which converts their spinning patterns into an audio signal. This has the effect of introducing a particular *harmonic* into the sound to alter its *timbre.* While similar in purpose to a stop on a pipe organ, it has the advantage of being variable in intensity as opposed to a stop's simple on / off action. Drawbars have been retained on more recent electronic organs of the Hammond type, but their function is now to act as simple faders that adjust the gain of different *oscillators.*

drift: In magnetic tape recording, any extended deviation from the nominal tape speed. Drift can be due to excessive take-up tension, improper *capstan motor* control, etc.

driver: (1) A power amplifier which increases the *amplitude* of a *voltage*, *current*, or *power* signal, (2) any *direct radiator* speaker, or (3) the term used to describe the chassis loudspeaker, mid-range unit, or tweeter elements of a loudspeaker system (as opposed to *speaker system* which covers both cabinet and drivers.) (4) A software program which enables communication between a particular make and model of hardware device and the computer's operating system, usually necessary for some kind of I / O device such as a soundcard, printer, or scanner. The problem of outdated hard-disk and soundcard drivers is particularly problematic.

drop-frame timecode: A version of the *SMPTE timecode* used for color video recording where a two *frames* are dropped at the beginning of each minute, except at the beginning of every tenth minute, devised to compensate for the difference between the NTSC (US) standard of 29.97fps and a real-time counter. The difference equates to 108 frames per hour. To avoid this confusion, most audio-only synchronization applications specify a non-drop timecode.

drop-in: See *punch-in*.

drop-out: See *punch-out*.

dropout: (1) In analog *magnetic tape recording*, the quality of the recorded signal depends on the uniformity of the magnetic coating of the tape. If its sensitivity varies on the tape, the signal level will be reduced periodically, and these reductions in level are called dropouts, their combined effects resulting in an increased noise level in the reproduced signal. See *asperity*, *calendering*. In a digital recording, a dropout is caused by an irrecoverable data error. (2) In *timecode,* a loss of sequence in the linear timecode count.

drum booth: An acoustic isolation booth or small room primarily intended as an enclosure for the recording of drums, traps or other percussion instruments and their players. Acoustically sealed off from the main recording space, drum booths have *bass traps* to prevent loud percussive transients from being heard. Some drum booths are not fully enclosed. This type of booth does not provide complete isolation, but does avoid the small-room problem of *standing waves* and lower midrange *resonances* that can give enclosed booths an unnatural, closety sound.

drum pads: A set of pads which have a similar response to the heads of acoustic drums when struck with sticks. They are made for two purposes: to quiet drum practice, and, when fitted with suitable *transducers,* to play electronic (usually sampled) drum sounds. If equipped with MIDI, drum pads can also act as a *controller*, allowing drummers to trigger any type of synthesized sound across a MIDI network.

D

dry: Consisting entirely of the original, unprocessed sound. The output of an *effects* divide is 100% dry when only the input signal is being heard, with none of the effects created by the processor itself. Lacking in *reverberation*. Compare with *wet*, *flat*.

dry/wet balance: This refers to the amount of *dry* signal relative to the amount of *reverb* or other *effect*-processed *(wet*) sound.

D.S.: Dal Signo. "Play from the *sign*(𝄋)."

DS4: The name of the original Dolby Laboratories recording / monitoring unit used by *re-recording* stages during a *Dolby Stereo* mix. Prior to the 2-track *print master*, the unit is used for 4:2:4 monitoring purposes, encoding a 4-channel composite mix into two tracks and then decoding it back into four channels. Later versions of these units include the *SEU4* and *SDU4* units which offer, respectively, the ability to encode and decode *print masters*, although without either the *container* of the optical track simulation featured in the DS4. The *DS10* contains a magneto-optical recorder for theatrical Dolby Digital mixes and also records the *Lt–Rt SR*-encoded print master. None of the above units can be purchased; their use is free for films that have paid for the appropriate license fee and / or trademark agreement.

DS10: See *DS4*.

DSD: See *Direct Stream Digital*.

DSP: Digital Signal Processor. Broadly speaking, all changes in sound that are produced within a digital audio device, other than changes caused by simple cutting and pasting of sections of a waveform, are created through DSP. A digital *reverb* is a typical DSP device.

DST: See *Direct Stream Transfer*.

D-sub(miniature) connector: Also called a *D-connector*, or a *D-type connector*. A type of connector commonly found on computers and data transmission devices, including SCSI devices and computer monitors. D-type connectors have a "D-shaped" angled housing, and have 9-, 15-, and 25-pin configurations, designated DE-9, DA-15, and DB-25, respectively.

DTL: Direct Time Lock. An early MIDI synchronization system developed by MOTU. See *MTC*.

DTR: Digital Tape Recorder. This is the analog version. An analog audio tape recorder is called an *ATR*.

DTRS: A 16-bit format used on Tascam and Sony *MDM*s, providing up to 108 minutes of 16-bit, 8-track record time on an NTSC-120 Hi-8mm videocasette. See *ADAT*.

D

DTS: Digital Theater Systems. A *5.1*-format theater *surround-sound* system which uses six discrete analog channels and *perceptual encoding scheme* for surround-sound on a CD-ROM interlocked to either a 35mm or a 70mm print with timecode. Lossless. The DTS *codec* provides for data rates from 256kbps to 1536kbps, focusing on 1141kbps as the optimum for transparent sound quality. DTS was originally developed for the film industry, however, there are a number of CD titles currently released in DTS format. A DTS CD carries six channels of digital audio in 5.1 format in 20-bit words at a 44.1kHz sample rate, with a compression ratio of about 3:1. An additional *decoder* is needed to play a DTS CD on a standard CD player. . First used in 1993 for the film, "Jurassic Park." See also *Dolby Digital*, *Dolby Stereo*.

DTS Stereo: See *stereo optical print*.

DTV: Digital TeleVision. DTV's audio specification provides up to six discrete channels of *5.1*–format audio, where the LFE channel is band-limited to 25Hz-120Hz. DTV has been developed specifically for the home theater market, as an improvement to the *ProLogic* system.

D-type (connector): See *D-sub(miniature) connector.*

dub: (1) (*verb*) In the most general sense, to dub is "to copy," although in film sound it has many similar meanings. Dub can refer to the act of replacing dialog, usually via *ADR*, either in the original language or in a foreign language. (2) Dubbing is also the common name for *re-recording*.

Dub-A, **Dub-B**, **Dub-C**: See *ProDigital*.

dubber: Film sound term for a playback-only *mag* machine. These were previously known as *dummies*. See *digital dubber*.

dubbing: (1) The act of re-recording *sound effects*, *location sound*, music, dialog, and / or *Foley*. Usually used to refer to the substitution of a foreign language or other replacement for the original dialog track in a film or TV production. (2) The process of making a copy or copies of a recorded analog or digital original. (3) To mix together onto a single track all of the separate edited soundtracks of a film or television production. See *re-recording*, *dubbing theater*. See *transfer*.

dub masters: See *final mix*.

dub stage: See *dubbing theater*.

D

dubbing theater: Also called a *dub stage*. A special studio where music is blended with dialog and sound effects for the final soundtrack. A dubbing theater is actually a small movie theater, with a large screen and full theater surround system. A row of seats is removed from the middle of the theater and a large mixing console specially designed for film sound is put in place. There is also a machine room which houses *dubbers* and projectors, isolating the noise from the recording studio in which the *DME* stems can be recorded in sync with film projected on a screen visible through a window. The screen hangs in a theater equipped with the mixing console which controls the sounds played back by all the dubbers, other prerecorded sources, and the sounds being recorded in the studio. The theater itself is designed to approximate the acoustics of a public cinema. Also called *mixing studio, re-recording studio, re-recording stage*, or *theater*.

duck: (*verb*) To lower the level of music to accommodate dialogue or other sound effects.

dummy load: A high-power resistor that is connected to the output of a power amplifier to make the amplifier function as though it had a loudspeaker connected to it. A dummy load circuit can be used to test amplifier performance as it would perform when connected to loudspeakers, or in a device such as a *speaker simulator* so that the amplifier always sees a high-*impedance* load at the output stage, even if no loudspeaker is connected.

dump edit: See *edit switch*.

duo-bilateral: The technical term for the *variable area* optical soundtrack format used on all 35mm mono and stereo soundtracks. See *SVA*, *Dolby Stereo*, *DTS Stereo*.

duplet: A pair of notes (or rests) executed in the time normally taken by three of the same value, most commonly occurring in compound-time music. The inverse of a *triplet*. See *time signature*.

duration value: The duration of a note is strictly a result of the time difference between when a given MIDI Note-On message is recorded and when a Note-Off message with the same note number is recorded. If a duration value is changed, it will result in the change of the time when the Note-Off message is transmitted. If the note start time is modified, both the Note-On and Note-Off times will be moved forward or backward by the same amount.

duty cycle: In a *pulse wave*, where immediate transitions occur between the high and low levels, *mark* is the time in one cycle occupied by the high level, while *space* is the time in the same cycle occupied by the low level. The ratio of mark to the whole determines the timbre of the sound represented by the saveform. Also called the *mark/space* ratio. See also *pulse wave*, *square wave*, Appendix C.

D

DVD: Digital Video (Versatile) Disc. A new multiple media format agreed upon by Sony, Philips, Toshiba, and others. DVDs are the same size as a CD, only with a higher track and pit density. The first, single-sided DVDs will hold 4.7Gb; as double-sided, multi-layer discs are available, the capacity will be 8.5Gb and 17Gb, respectively. Transfer rates are about 1.35Mbps, or about as fast as an 8X CD-ROM drive. The data format is 24-bit 96kHz. One DVD is sufficient to store a typical movie, eight tracks of Dolby *AC-3* surround audio, and numerous subtitle tracks, which is why the film industry is pushing it: an improvement in the potential of both audio and video quality with the profit profile of a CD. One DVD-5 disc will allow 318 minutes of 48kHz, 20-bit, two-channel audio, or about 144 minutes of 88.2kHz, 24-bit, two-channel audio. At 88.2kHz, 24-bit, LCR plus two channels of 44.1kHz, 20-bit surround-sound, a DVD-5 will hold 75 minutes of audio.

DVD players will be able to play back CDs, but not those written using the *CD-R* standard. As with CD-ROM formats, DVD specifications are referred to in terms of books, A-E:

A:	DVD-Video
	DVD-A: Audio Only
	DVD-AV: Audio-Video, contains a subset of DVD-V
B:	DVD-Data
C:	DVD-Audio
D:	**DVD-R** (write-once)
E:	**DVD-RAM** (rewritable)

Additional DVD subdesignations are:

V:	**DVD-V**ideo
VAN:	**DVD-VAN** (video disc, audio-enabled)
5:	Single-layer DVD (4.7Gb)
9:	Dual-layer low-density (8.5Gb)
18:	Dual-layer high-density (17Gb)

The new DVD-A standard allows for up to six channels of audio. With DVD-V, audio must be recorded at a 96kHz sampling rate, while the DVD-A standard supports 44.1kHz, 88.2kHz, 176.4 and 192kHz sampling rates. The DVD-A specification has two parts: DVD-A (audio only) and DVD-AV. DVD-AV contains a subset of the DVD-V specification to include real-time text, full-screen video, and other MPEG options. DVD-A has an Audio Manager, while, not surprisingly, DVD-V has a Video Manager. If a DVD-A has video titles, it is an –AV disc. The multiple viewing angle feature of DVD-V is not supported in the –AV specification. A working specification for DVD-VAN, a bridge format between DVD-V and DVD-A formats which will enable a DVD-V disc to be playable on a DVD-A player, if so authored.

D

The DVD-A specification includes optional formats for *DTS*, *Dolby Digital*, lossless compression, MPEG-2 BC (Backward Compatible), and *DSD*, and provides for two “channel groups” which is a method for signal partitioning, roughly equivalent to front and rear (six-channel maximum unless otherwise noted):

	Channel Group 1	**Channel Group 2**
Number of Channels	1 to 4	0 to 3
Sample Rate (kHz)	44.1	44.1
	48	48
	88.2	88.2 or 44.1
	96	96 or 48
	176.4 (2 channels max.)	
	192 (2 channels max.)	
Word Length (bits)	16	16
	20	16 or 20
	24	16, 20, or 24

dynamic allocation: See *dynamic voice allocation*.

dynamic effect: (1) An effect which alters the *loudness* characteristics of a signal without introducing any *timbre* changes. The most common dynamic effects are *compression* and *limiting*. (2) Some companies use the term *dynamic effects* to refer to effects devices whose processing parameters can be controlled in *real-time* via MIDI.

dynamic equalization: Equalization where the amount of boost or cut varies according to the dynamics and spectral content of the signal being processed. Dynamic equalization is most often used in *audio enhancers*.

dynamic filter: (1) An early type of *single-ended noise reduction* system that uses one or two filters whose *rolloff* frequencies are controlled by the level of the signal. As the high-frequency signal level falls during soft passages, the high-frequency response is reduced; when the signal level is high, the full bandwidth is restored. (2) A circuit used in aural exciters where a side-chain signal is combined with some *dry* signal in such a way that the original signal is modified both additively and subtractively to create the impression of an increase in both bass and brightness and the mid-range appears more focused. This type of spectral shaping is designed to be closely related to the way the human hearing system changes at different listening levels. See *equal loudness curves*.

dynamic headroom: The ability of a power amplifier to handle short bursts of power without overload.

dynamic loudspeaker: See *loudspeaker*.

dynamic microphone: In a dynamic microphone, a *moving coil* in a magnetic field to generate electricity. Two types of dynamic microphones are the moving coil and *ribbon*. (Moving coil mics are typically referred to as *dynamic mics*, while ribbon mics seem to be called ribbon mics.) Dynamic mics have a rougher response than condensers or ribbons, and can be used to soften fine detail in the recorded sound. A well-designed moving-coil dynamic mic can handle very loud sound without distortion, and so is preferred for miking guitar amps and drums. Dynamic mics also have a pronounced *presence peak* that gives the sound an edge or punch.

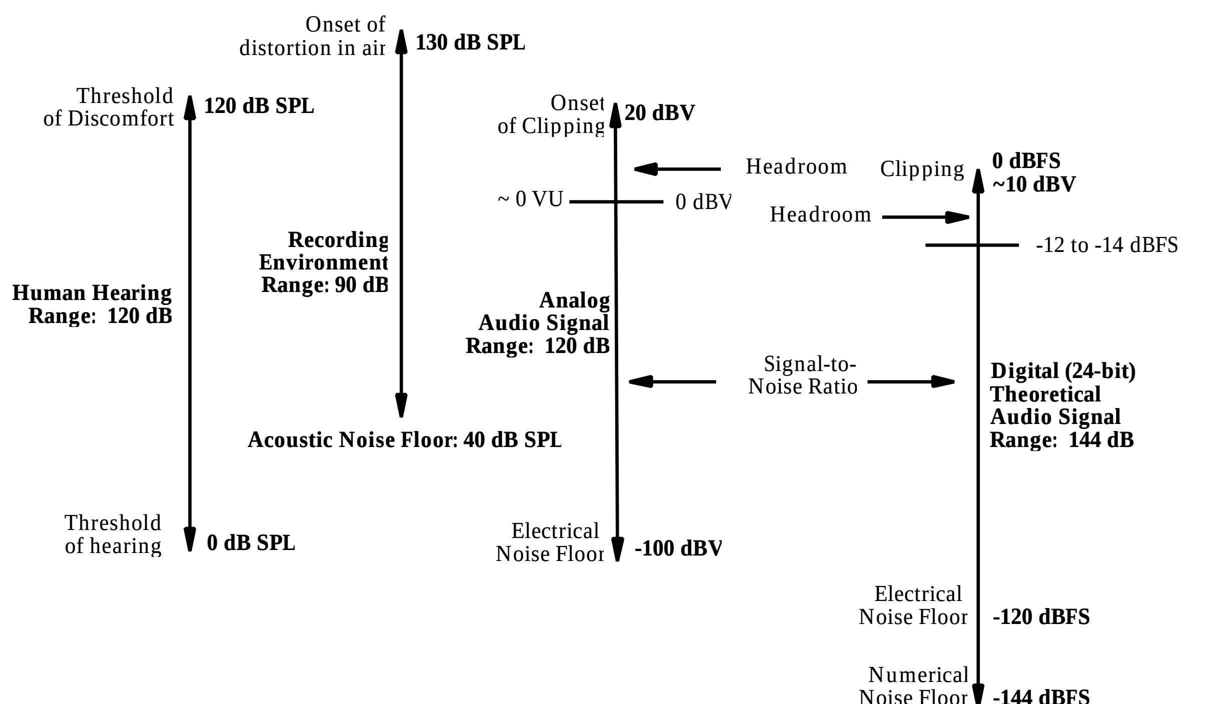
D

dynamic range: (1) The amplitude range of a sound from its softest to its loudest. (2) Also called *dynamic-range*, the range of sound levels which a system can reproduce without distortion, i.e., the peak signal-to-average noise, or the difference between loudest level the system can reproduce without distortion and the *noise floor* of the system. See *Lmax/Lmin*.

Bits	Levels	Theoretical best D.R./noise (dB) (20 log 1 bit error/no. of bits)	Typical Distortion
8	256	48.16	0.5%
12	4,096	72.25	0.1%
16	65,536	96.33	0.002%
24	16.772Mb	144.49	-
32	4.2949Gb	192.66	-

Desired dynamic range can be defined as the range of signal resolution plus the range of amplitudes of the signals in the program material. For example, if there is a 12-bit signal (72dB) and a range between Lmin and Lmax of 30db, the desired production dynamic range would be 102dB.

In terms of recording, headroom plus the S/N ratio equals the dynamic range of the medium. For acoustic spaces, the dynamic range is the range of SPLs between the acoustical noise floor (about 30dB SPL for a quiet recording space) and the onset of nonlinearity in the air (about 130dB SPL). This is about 100dB SPL, approximately the dynamic range of a digital audio recorder, if you count all 16 bits as significant, which, of course, they're not.



Dynamic Range

D

dynamic signal processor: Any electronic device whose type or degree of operation changes with response to level or other characteristic of the input signal, i.e., with *feedback*, for example *compressors, downward* and *upward expanders, gates, limiters, NR systems, flangers*, etc. The opposite of *static signal processing*.

dynamic voice allocation: A feature of *multitimbral* synthesizers where a voice always is made available to sound new notes when all the synth's *polyphonic* voices are in use. In the most common scheme, the most recently played note steals the voice from the oldest note currently sounding, or sometimes the lowest amplitude sound. The usual alternative to dynamic allocation is to assign an inflexible, predetermined number of voices per sound. See *voice stealing*.