Introduction

This dictionary was born in Scotland in the spring of 1996 and represents my mid-life opus. Yielding to the typical mid-life crisis seemed both unimaginative and self-indulgent. My motivation for writing a dictionary was that I was entering my third score of life and it was time for a career change; I do this every twenty years whether I need to or not. A dictionary appealed as it's one thing to peruse the technical journals, but another to actually understand them. Having done the go-learn-everything-about-the-new-career-so-I-don't-look-like-a-total-idiot thing before, I knew that the hardest thing to do is explain something you don't really understand to someone else. This dictionary was my imaginary audio friend (in the wilds of Scotland, company is hard to come by.)

As with my other successes in life, such that they are, next to my aunts I owe the greatest debt of thanks to Len. Monica Anderson was my first (real) audio friend, although she and I have been friends for years. Monica helped me put together my first very own project studio and has graciously accepted, if not actually read, drafts of this dictionary. If I had known about Godrick Wilkie's book, I probably wouldn't have gone through this exercise, but it's better that I didn't and I did because he's English and it's very true about two cultures separated by a common language. The most awe-inspiring treatise in audio I've come across is the book by Glenn D. White; it's a good complement to the Wilkie book because the first is too simple, and the latter quite rigorous. I have tried to tread midway between the two. Tell me if I've strayed from the path.

I also owe thanks to my favorite musicians: Karen Bentley, Dimitry Cogan, Dennis James, Viviana Guzman, and Jonathan Salzedo et al. for letting me record them and get a practical understanding of what this all means. I found audio engineering to be just like programming; it's great to read all the books, but it really doesn't make any sense until you get your hands dirty and, in this case, trip over the mic cables.

A few notes on the structure of the dictionary: In alphabetizing the words, I have ignored symbols in the middle of words, for example "deemphasis." This is because of the utter lack of standardization in spelling technical words, i.e., optional hyphens, spaces, or other non-alphanumeric characters. If you don't see it, be creative and look around. If you still can't find it, **send me mail**. Also, whether a term is listed completely spelled out or under its acronym is pretty arbitrary. In general, I listed it the way I most often see it used, e.g., DAT is listed under DAT, not digital audio tape. Again, look around. If that doesn't work, complain.

I have chosen to italicize words used in definitions which appear elsewhere in the dictionary. Italics seemed less obtrusive than bold or SMALL CAPS. However, I haven't italicized every word that is defined (frequency, for example, occurs in about every other sentence), but only the first instance in a definition, and then only when useful for understanding the current context. Occasionally, italics are used to just set off a term, such as "This is called the quadratic residue sequence" or, "those wires are called legs." distinctly from the rest of the text. This only happens about six places in the dictionary; I did it because it seemed distracting to put the terms in quotes, like they weren't really words. You'll know you've found one of these when you go to look it up and it's not defined. (If really need to know all about the quadratic residue sequence, (1) you probably already know, and (2) w.l.o.g., you will be happier with the White book; and (3) legs are, well, legs.)

At the end of many of the definitions there will be a "See" or "See also." These additional terms are not listed in any particular order; the dictionary has been an evolutionary process and I just add the new terms as I find them and have never gone back to reorder them alphabetically, by relevancy to the current word, or by any other heuristic. I've tried to make sure that all related terms are included where necessary for context or completeness; please let me know if I'm missed some. Reading a dictionary, even one's own, can get a little tedious. I've read this one completely seven times now. It used to take exactly the same length of time the train took from London to Edinburgh. I'd probably have to go to Istanbul now to do it in one train ride. From Virginia.

Sometimes the definition of one term is so inexplicably intertwined with another that it didn't make sense to separate them, even if the alphabet did, so I didn't. When this happened, I would just cross-reference one term, arbitrarily, and make a combined definition in one place. An example of this is direct sound(1). Sometimes things seem obvious, e.g., opposite of passive is active. But then the opposite of outboard isn't inboard, it's onboard, complete with hyphen; the opposite of boost isn't squash, it's cut. So, if a few things seem to have the "duh syndrome," I tried to err on the side of completeness. Sometimes the choice of main definition is pretty arbitrary, usually where there are two or more terms in common use, such as production sound and location sound. Sometimes the reason for the choice is completely non-obvious as in rolloff frequency vs. cutoff frequency. In this case, I have deliberately chosen a term which may not be the most commonly used. I picked the former as there is almost never a slope parameter that is actually a cutoff, only in the case of a limiter. The slope almost always is, in fact, a rolloff and it seemed better to use that term as it more closely matched what is actually happening, rather than subscribing mindlessly to "Better Tomes and Jargon." There are actually four terms in common use which describe filter slope: the two mentioned above, besides critical frequency and corner frequency. They are all reasonable; pick the one you like--I did. And, the use of any term varies among practitioners, by specific field within audio or by location. I realized early on that there was no way to please all of the noise nerds all of the time...

And, being me, I left out most of the computer terms. I assumed that this dictionary would be (uniquely) useful to those of the digital persuasion, i.e., those of us with weak musical and/or MIDI backgrounds, but who are very strong on acronyms. I hope I didn't get too EE-bound. In my short tenure as a recordist/mixing engineer, I appreciate how important it is to actually understand about balanced lines and ground loops, especially when it becomes appallingly clear that the well-meaning volunteer who cut the ground wire at the service panel to eliminate the hum from the church PA system did not have my $8,000, 24-bit A/D in mind. Again, complain, but gently.

For the more musical, I would appreciate input on what I've left out musically, but also what I've neglected to define in the technical forest that has made things more difficult for you in trying to use the dictionary. One really finds out who one's friends really are when one asks them to read draft copies of one's dictionary. I'm not exactly friendless, but it's getting very close.

In keeping with my anarchistic view of the universe, there is also no particular rule for inclusion of detailed data other than what would be useful in the field. I did not include all MIDI notes and commands as one is likely to be somewhere quiet with a bookshelf when one needs that, and, I know that everyone, but everyone, takes the manuals for their synths and samplers along to every gig, right? The charts on dB, information on AC, pin-outs, etc. I thought might be needed in the field (read this: I've needed it), and it would be handy to have this information in one, portable source. Again, comments welcome.

Finally, in response to almost universal feedback that DD should be published (which is probably more the issue that not everyone wants to sit in the wilds of Scotland, writing dictionaries): I don't want to spend any more time on this dictionary; it's accomplished its purpose, i.e., I can now read the trade press. And, as I said, I'm running out of friends. I welcome comments on alternate definitions, things I forgot, things which are unclear. Be kind: this nanoscopic contribution to better global karma has had a gestation of over two years of my life.

Sandy Lerner,
Sono Luminus
1997, 1999

A

1:1: One to one. In standard usage, a copy of the edited *worktrack* copied onto another roll of striped mag film so that sound editors and mixers will have access to the *worktrack*. In general, however, it denotes any single-track-to-single-track copy, and thus has variants 3:3, 4:4, etc.

3:1 rule: A rule for microphone placement: space microphones at least three times the mic-to-source distance. For example, if two mics are each placed one foot from their sound sources, they should be at least three feet apart. This method prevents the blurred, colored sound caused by *phase cancellation* between microphones.

3-stripe: See *3-track*.

3-track: A mix of all the soundtracks of a film, in which the sounds are divided into the *DME* stems, each stem recorded on a separate stripe along the width of the 35mm magnetic film. Also called *three-stripe*.

4+2: Four Plus Two. Film sound slang for a 6-track element (usually mag film) that contains a 4-track *M&E*, one track of material for a foreign-language mix, and one track of original dialog as a reference.

4:2:4: See *Dolby ProLogic™*.

4-track: A *film soundtrack* format used for overseas markets. Called a *completely filled* mix, the four-track stereo M&E mix is ready for the addition of dubbed languages. The M&E tracks should include background sound effects and room tone for every scene, i.e., all sound except dialog.

50% level: The standard reference level for optical sound recordings that corresponds to the width of the track at 50% modulation, or at 6dB below clipping. In practice, there is about 2dB headroom available, if all of the recording / playback heads are perfectly aligned.

5.1-channel format: A digital, discrete *six-channel mix* of Left / Center / Right / Left Surround / Right Surround / subwoofer mix. 5.1 is not a specific surround format tied to any particular company or *codec*. However, all the hardware is the same for any 5.1-based system except for the codec. It is planned that CDs, laserdiscs, and DVDs will have an ID flag to let the decoder know which codec was used, enabling decoders to recognize all incoming bitstreams and automatically switch modes and process the incoming signal appropriately. It is a listening platform and hardware concept for a surround loudspeaker system. See *DTS*, *Dolby Digital*, *HDTV*, *CDS*, *LFE*.

5-2-5 matrix: See *Logic 7*.

70mm: See *film*.

7.1 Split-Surround: The additional two speakers are employed at the front of the soundstage to deliver more uniform sound in wide-format theaters of screen widths of up to 60’ or more, where there might be seats with *hole-in-the-middle* in between the C-L, and C-R channels. See *SDDS*.

A

85: A common *SPL* level reference in the film audio business, which is found by setting the SPL of pink noise is sent through one speaker (L, C. or R) at 0VU (analog) bus level, which is the equivalent of –20dBFS in digital recording. Measurement is made at the console, with an SPL meter set to *C-weighting* and the meter *ballistics* set to slow response.

88: The *SPL* for *Dolby Stereo SR* films. If a film has been monitored at *85* during the final mix, the stems will be lowered 3dB each when making an SR *Lt-Rt printmaster* to accommodate the increased gain from summing the stems.

A: The left-hand part of a *stereo* signal.

A-2: See *Voice of the Theater*.

A-3: Dolby laboratories low-bit-rate *codec* system used in its *Dolby Digital* format film, in both broadcast and consumer video formats.

A-4: See *Voice of the Theater*.

A-7: See *Voice of the Theater*.

A440: See *concert pitch*.

AAC: Advanced Audio Coding. A flexible *streaming* format that supports multichannel audio including subwoofer and embedded data channels, using a variety of sample rates up to 96kHz. AAC is being developed as a successor to *MPEG-2*.

Aachen Head: A *binaural* microphone developed by Head Acoustics.

AB recording: In the US, this means recording with a *spaced pair*. In Europe, this means recording with a *coincident pair*.

A/B: A comparison between two recordings of the same material; pre- and post-equalization, or pre- and post-effects, or any other comparison between two similar audio devices.

AB-reel: Term for a 23-minute or 2,050' maximum reel of film specially made for theater screening. The AB-reel may originally have been made from two 1,000' edit reels; "Projection reel 1AB" would have been originally been reel #1 and reel #2 during editing and mixing. [In the event that the total footage of the first three editing / mixing reels added up to less than 2,050', there may be a projection reel "1ABC," but this is rare.] It is becoming more commonplace to edit films in AB reel format as the *mag film* units are gradually replaced with DAWs. AB-reels are also known as "big reels" or "2,000-foot reels."

AB-reels are not the same as *A/B-rolls*, in which the camera negative is checkerboarded into two strands, allowing for simple optical effects such as fades and dissolves to be made when making original-negative prints (see *EK Neg*) called *interpositives*. This latter process is not limited to two (A,B) rolls, but can involve as many rolls of film as desired, e.g., a camera negative cut in four strands would have a "D-roll."

ABS: ABSolute time. *Timecode* which is the actual running / recording time in HH:MM:SS, where 00:00:00 is the head of the tape. For example, DATs use ABS timecode. See also *feet/frames*.

A

absorption coefficient: The ability of a material to absorb, rather than reflect, sound waves. A higher absorption coefficient means better acoustical *damping*. See *bass trap*, *boundary effect*, *standing wave*, *Sabins*.

Material			**Frequency (Hz)***			
	125	**250**	**500**	**1,000**	**2,000**	**4,000**
Acoustic Panels	0.15	0.3	0.75	0.85	0.75	0.4
Brick	0.024	0.024	0.03	0.04	0.05	0.07
Carpet	0.05	0.1	0.2	0.25	0.3	0.35
Concrete	0.01	0.01	0.02	0.02	0.02	0.03
Curtains	0.05	0.12	0.15	0.27	0.37	0.5
4” Fiberglass	0.38	0.89	0.96	0.98	0.81	0.87
Wood Floor (joists)	0.15	0.2	0.1	0.1	0.1	0.05
Glass	0.03	0.03	0.03	0.03	0.02	0.02
Seated Person	0.18	0.4	0.46	0.46	0.5	0.46
Plasterboard	0.3	0.3	0.1	0.1	0.04	0.02
Plywood on 2” Batten	0.35	0.25	0.2	0.15	0.05	0.05
3/4” Wood Panel	0.1	0.11	0.1	0.08	0.08	0.11

*Note: A coefficient of 1.0 means 100% absorption, such as an open window, while 0.0 means 100% reflection. All figures are given for one square meter of material.

AC-1: A form of *ADPCM* first used in 1985 for digital radio (sound-only) applications and since adopted for other *DBS* (direct broadcast satellite) services, including soundtrack-with-video, satellite communication networks, and digital cable ratio systems. AC-1 has a data rate between 220 kbps and 325 kbps.

AC-2: A *transform encoding/ decoding* scheme for audio *compression* developed by Dolby labs which uses 256-band transform coding at a data rate of 128 kbps or 192 kbps on two channels. Used in the Dolby Fax System and also *DP5xx encoding*.

AC-3: See *Dolby Digital*.

ACA: Active Combining Amplifier. See *combining amplifier*.

Academy centerline: See *optical track*.

Academy curve / Academy sound: The name of the standard mono *optical track* that has been around since the beginning of sound on film. Standards were codified in 1938, although the standard has changed somewhat through the years. The standard specifies a flat response throughout the range of 100Hz–1.6 kHz and is down 7dB at 40 Hz, 10dB at 5 kHz, and 18 dB at 8 kHz. Also called an *N-Curve*. See also *X-Curve*.

Academy leader: The visual countdown that precedes the first program *frame* of a motion picture. Symbols and numbers on the academy leader are used for aligning the various film reels and the optical track for composite printing, for aligning the *workprint* and edited soundtracks for mixing, and for timing the change-over from one reel of film to another during projection. Academy leader contains one number per foot following the Picture Start, with 11, 10, etc., leader to three. (As projected, these numbers appear upside-down.) Named after the Academy of Motion Picture Arts and Sciences, which sets all film format standards. See also *leader*, *SMPTE Universal leader*, *plastic leader*, *fill leader*, *LFOP*.

A

academy Theater: Specifically, the Samuel Goldwyn Theater at the Academy of Motion Pictures Arts and Sciences on Wilshire Boulevard in Los Angeles, considered the best-sounding theater in the world. Academy members screen films at the Academy Theater prior to voting on them for the Oscar awards.

AC bias: See *bias*.

accelerando: An indication that the *tempo* of a piece of music should gradually be increased.

acceptance angle: The usable working area in front of a microphone is defined by the *polar pattern* and is called the acceptance angle.

accidental: In a musical scale, the accidentals are the extra sharp and flat notes that are not part of the *diatonic* series. For example, in the key of C on the piano, the accidentals are the black keys.

AC coupling: Coupling between electronic circuits that passes only time-varying signals (i.e., alternating current), not direct current.

A-chain: The part of the motion picture reproduction system in a theater that contains the sound transducer (such as an optical analog track reader or digital sound format decoder), preamp, noise reduction and matrix decoding, where applicable. The A-chain equipment decodes the sound in preparation for the *B–chain* and loudspeakers.

AC-M: A newly developed *codec* based on a soft data *compression* ratio of between 2:1 and 3:1. Used in the Dolby Digital Dubber, it is designed specifically to record eight tracks of 20-bit material on removable media, including Iomega Jaz and MO drives. AC-M is said to be optimized for multiple record / replay generations. Initial tests have reported as many as 14 codec processes being possible with no audio degradation.

Acmade: The British manufacturer of *edgecoding* machines.

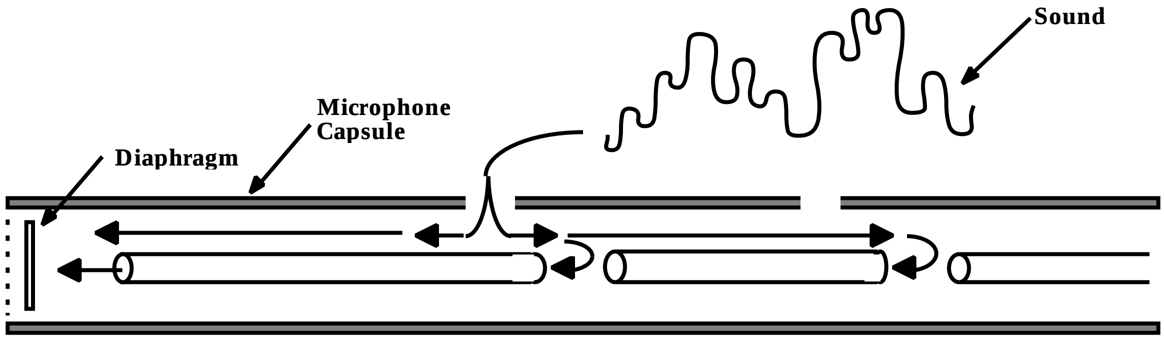
acoustic baffle: See *baffle*.

acoustic feedback: A squealing sound when the output of an audio circuit is fed back in *phase* into the circuit's input. See *feedback*.

acoustic intensity: See *sound pressure level*.

A

acoustic labyrinth: (1) A type of design for the housing of highly *directional microphones* that enhances the *rejection* of *off-axis* sources. Two or more concentric tubes in front of (and sometimes around) the capsule create a compact series of folded pathways through which all sounds approach the diaphragm. Those arriving on-axis reach the capsule via these paths in *phase coherence*. Off-axis sounds, due to the different lengths of the passways, reach the diaphragm and are partially or fully removed due to *phase cancellation*.



Acoustic Labyrinth

Sound entering from sides and rear arrive out-of-phase

(2) A type of speaker enclosure in which sound waves emanating from the rear of the *woofer* cone travel through a long, folded interior path before coupling with the outside. This extends bass response considerably.

acoustic lens: A device placed in front of a high-frequency speaker that disperses or directs the sound in a desired pattern. Normally used to increase the angle of dispersion, either horizontally, vertically, or both.

acoustic suspension: A *loudspeaker* designed for, or used in, a *sealed enclosure*. Typically, a low-frequency loudspeaker *baffle* where most of the damping of the cone is the result of the elasticity of the air in the sealed cabinet.

acoustics: The science or study of sound and its interaction with the human hearing mechanism.

active: (1) An audio device which requires a power source such as from an AC line or battery, as opposed to *passive*. Sometimes amplifying components such as transistors or ICs are called active circuit elements. (2) See *MIDI patchbay*.

active crossover: See *crossover network*.

active equalizer: An equalizer that employs *active* components such as transistors or ICs in its processing circuits. A pre-amplifying circuit generally follows each stage of actual equalization, boosting the signal level to restore *unity gain*. See also *passive equalizer*.

active monitor: A type of loudspeaker which has amplification circuitry built-in. In addition, a true active monitor system utilizes active equalization and active crossovers to precisely contour the system sound. If there is only one amplifier driving all transducers, and / or there is no active equalization or crossover circuitry, the terms *powered speaker* or *powered monitor* are more correct.

A

active sensing: A MIDI *system message* that carries no note data or control instructions, but simply indicates to a receiving device that the MIDI line is in working order.

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

adagio: A slow or leisurely tempo: 66-76 bpm.

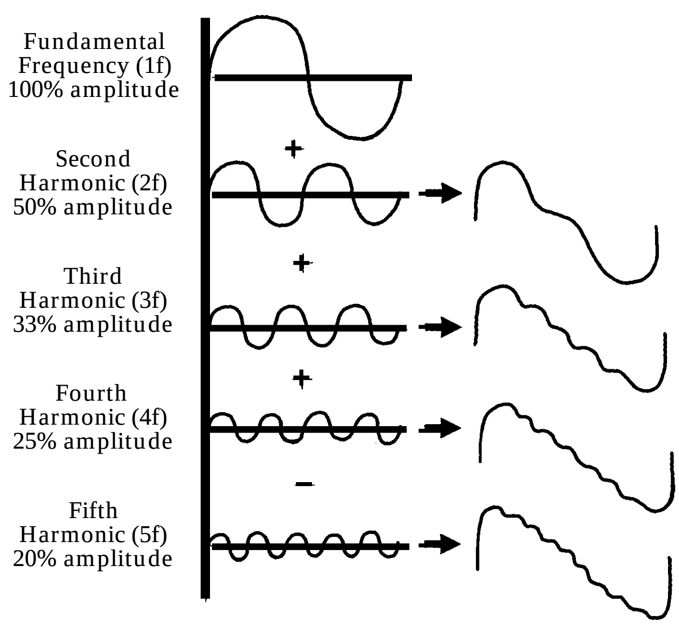
A-DAM: Akai Digital Audio Multitrack. A format developed by Akai in 1987 for recording twelve tracks of digital audio data on a standard *Video-8* cassette and which allows the synchronization of multiple decks for 24- or 36-track recording. The tape runs at four times the normal Video-8 speed and gives about 15 minutes of recording time at 44.1kHz.

ADAT: Alesis Digital Audio Tape. A second-generation (1992) *MDM*. Like the Tascam DA-88, ADAT systems record digital audio on consumer videocassette formats and provide for interlocking up to 16 8-track, rack-mount recorders in sample-accurate (48kHz) sync for up to 128-track recording. ADAT is a 16-bit format, currently supported as well by Panasonic, using T-180 S-VHS tape. ADAT-II is a newly proposed 20-bit S-VHS format used by newer Alesis and Studer 8-track recorders. See also *DTRS*.

ADB: Apple Desktop Bus. The original *serial* interface for the keyboard, mouse, and other "desktop" peripherals on Apple computers. ADB has recently been replaced by *USB.*

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

additive synthesis: The generation of complex musical waveforms in electronic synthesizers by the linear addition of *sine wave* components whose frequency relationship is a *harmonic series*. See *sample synthesis, sound synthesis, subtractive synthesis.*



The Making of a Complex Waveform

A

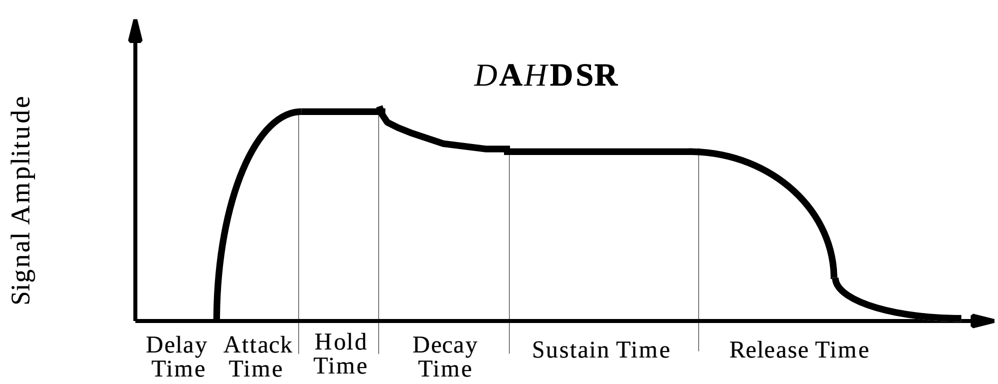
address track: A control / timing track on the edge of videotape (1”, C, and ¾” formats) that contains control data for quick and accurate location of program material, recorded at the same time as the picture. See *control track*.

adjustable turnover: A variable tone control in a preamplifier which allows the adjustment of the boost / cut and the frequency below or above which the gain / attenuation is applied (turnover), but not the *rolloff slope* of the *shelving equalizer*; if it were possible to adjust the rolloff slope, the result would be a fully *parametric* tone control.

ADPCM: Adaptive Delta (Differential) Pulse Code Modulation. A type of *split-band*, time-domain audio *compression* algorithm for 16-bit digital audio based on describing level differences between adjacent samples. Different from conventional linear PCM by coding only level differences between samples, rather than the absolute level of each sample. According to the characteristics of the audio signal, ADPCM adapts the step size represented by each quantizing interval to accommodate rapid changes in level caused by high frequencies or *transients*, thereby providing an overall reduction in bit rate; the compression ratio is 4:1. There are at least two ADPCM standards: Microsoft and *IMA* / ADPCM, the latter popular for multimedia applications. *See delta modulation*, *split-band*, *sub-band*, *transform coding*.

ADR: Automatic Dialogue Replacement. Recording of dialog for a scene after it has been shot, usually to replace *location sound* that is unusable because of street noise, camera noise, etc. The *workprint* and *magnetic film* transfer for the scene are spliced into continuous loops and projected in a sound studio so that the actors can recreate the phrasing and feeling they had on the set. New takes are recorded on a separate mag film loop and / or other synchronous tape until an acceptable performance is obtained. *Virgin looping* is the process of recording onto a blank piece of mag film which would later be manually synced to the picture. Also known as *looping*. See *lip sync*.

ADSR: Attack / Decay / Sustain / Release, the four segments of a common type of sound synthesizer *envelope*. The controls for these four *parameters* determine the duration (or in the case of sustain, the height) of the segments of the envelope. Two additional parameters, D (Delay Time) and H (Hold Time) are available on some synthesizers. See *envelope*, *envelope generator*.



ADT: Auto Double Tracking. An effect produced by taking a track and copying the material onto another track, delayed by a few ms, then mixing it with the original. Like *chorusing*, but with a shorter delay. See also *double-tracking*.

A

AES: Audio Engineering Society. The professional organization whose members report on new technological developments in audio, and bring together designers, manufacturers, and users of various audio equipment to establish international standards.

AES/EBU: Audio Engineering Society / European Broadcast Union. A standard for encoding multiple channels of digital audio along a *serial* cable, officially named AES3-1985. The standard specifies 3-pin *XLR* jacks and *balanced line* cables, usually running at +4dBm. Originally designed as a *self-clocking* system, a subsequent addendum to the specification permitted *master clock* systems. Two channels of digital audio data are *multiplexed* on a single conductor within the cable, with a maximum bit depth of 24 bits. Data are transmitted at 64 times the sample rate, allowing the possibility of sending two channels of 24-bit audio (plus *ECC*) to play the resulting stereo signal in *real-time*. AES / EBU does not carry the *SCMS* copy code. It has been adopted by the *EIAJ*, which calls it the CP-340, Type 1. See also *S/PDIF*.

AES/EBU null clock: See *null clock*.

AF: Audio Frequency. Means having frequencies within the audible range, usually taken to be 20Hz-20kHz. This frequency range is an average; many people hear tones below 20Hz, although most people are virtually deaf above 15kHz or 16kHz.

AFM: (1) Audio Frequency Modulation. A processing scheme used for recording high-quality analog audio in videocassette recorders equipped with “Hi-fi” stereo audio. (2) American Federation of Musicians. The union that represents professional musicians in all their client and employer relations.

after-fade listen: See *post-fade listen*.

aftertouch: A type of MIDI *controller* data, generated by pressing down on one or more keys after they have reached and are resting on the keybed. Also called *pressure*. See *channel pressure*, *poly pressure*.

AFTRA: American Federation of Television and Radio Artists.

AGAC: American Guild of Authors and Composers. A third performing rights organization similar to ASCAP and BMI but much smaller. AGAC primarily represents modern classical composers.

AGC: See *AVC*.

aharmonic: See *inharmonic*.

AIFF: Audio Interchange File Format. A common Macintosh audio file format. It can be mono or stereo, at sampling rates up to 48kHz. AIFF files are QuickTime-compatible and support uncompressed mono, stereo, and multichannel audio at many different resolutions and sampling rates, including the *CD* standard. It was designed to serve as a universal interchange format that allows any program to open a digital recording created by any other program. AIFF is high-quality audio, used in pro-level Mac and PC audio software. As AIFF is a standard for uncompressed audio, Apple introduced AIFF-C which can use *MACE* and IMA / *ADPCM* compression with ratios as high as 6:1, but the audio sound quality suffers.

AIM: Amplitude Intermodulation Distortion. *Intermodulation distortion* where one signal will cause *amplitude modulation* of another signal.

A

A.I.R.: Always In Record. The practice in a recording session to record virtually everything on the off-chance that something which was not formally recorded as a *take* will be useful.

airline version: A remixed and possibly re-edited version of a film that has any objectionable material removed. The airline film standard is more stringent even than those of the broadcast networks, and is often used as a benchmark for TV viewing.

air suspension: An *acoustic suspension loudspeaker*.

algorithmic composition: A type of composition in which the large outlines of the piece, or the procedures to be used in generating it, are determined by the human composer while some details, such as notes or rhythms, are created by a computer program.

alias: A file on a Mac that serves as a pointer to another file. The most common use for an alias is to sit on a desktop or in a top-level folder, where the real document or application file is nested deep within the file system. This is similar to a *shortcut* file on a PC-type computer.

aliasing: *Distortion* that is produced when higher *harmonic* components of the input audio signal sampled by a digital recording device, or generated within a digital sound source, lie above the *Nyquist frequency*. This happens when the sampling rate is less than twice the frequency of the signal being sampled. The effects of aliasing differ from some other types of distortion in that its pitch changes radically when the pitch of the intended sound changes. Also called *foldover*. See *anti-aliasing filter*.

alignment: (1) In tape recording, the process of adjusting all parameters of the position and orientaton of the tape heads and guides with respect to the tape path. See *azimuth*. (2) The adjustment or calibration of any parameter of an electronic circuit or device, e.g., program *level*, *bias* level, to bring this parameter into conformance with an industry standard. (3) The process of matching mixer and recorder meters so that only one meter needs to be watched during recording. When the mixer and (analog) recorder are both peaking about 0VU, this minimizes the noise and distortion in both units. Ideally, both units would be matched with a steady tone (the C or B two octaves above middle-C, or about 2kHz, for example.) See *line-up tone*.

alignment recording: See *biased noise*.

alla breve: A term historically related to mediæval note lengths, in which the breve was one of the shortest notes. In modern usage, the term is usually used to denote $\frac{2}{2}$ (cut-time). In commercial and popular music, it is frequently used to mean *half time*, i.e., play twice as fast. See *time signature*.

allegro: A lively to reasonably fast *tempo*: 116-150 bpm. *Allegretto* is a slightly slower tempo than allegro.

All-Notes-Off: A MIDI command, recognized by some but not all synthesizers and sound modules, that causes any notes that are currently sounding to be shut off. The *panic button* on a synth or sequencer usually transmits All-Notes-Off messages on all 16 MIDI channels.

all-pass filter: See *all-pass network*.

A

all-pass network: An all-pass network, also called an *all-pass filter,* is an electrical circuit with a uniform amplitude response versus frequency response, but with a *phase-shift* which does not vary in a *linear* relationship with frequency. (A pure time-delay device such as a digital *delay line* will have a phase-shift which is directly proportional to frequency, i.e., its phase-shift increases at a constant rate with frequency.) Complex filters often have significant *phase distortion* because they are not *phase linear,* and an all-pass network can be designed to correct phase anomalies without affecting the amplitude response.

alternating current (AC): An electrical current that periodically changes in direction. The rate of alternation is called the *frequency* and is measured in cycles per second, or *Hertz*. Audio signals are always alternating, the frequency corresponding to the pitches of the sounds the signals represent. See Appendix B.

ambience: Ambience refers to the acoustical qualities of a listening space, such as *reverberation*, *echoes,* background noise, etc. On most music recordings, some of the ambience is recorded along with the music and are, to a certain extent, reproduced in the listening environment, e.g., an organ in a cathedral. See *room tone, walla, NC Curve.*

ambience track: An edited roll of magnetic film, or one track of a multitrack tape, assembled by the sound editor in preparation for the final mix of a motion picture or video production, containing the series of *room tones* or *ambient sounds* of the various sets and locations in which a scene was shot.

ambient noise: *Ambient sound* which is environmental in nature, such as traffic noise coming through walls, heating or air conditioning, or other extraneous sounds which cannot be turned off or removed.

ambient sound: Sounds such as *reverberation*, *room tone*, *walla* and *atmospherics* that form a background to the main sound, usually in the context of a *film soundtrack*, taking place at any given moment. The lack of ambient sound is noticeable because the human hearing system expects it. See also *ambient noise.*

ambisonics: A system for the reproduction of a three-dimensional *sound field*, using two or more transmission *channels* and four or more loudspeakers. See *Soundfield microphone.*

AMEI: Association o f Music Electronics Industries. A group that works with *MMA* on MIDI standards, among other things.

ampere (A): The unit of measurement for electrical *current* in coulombs (6.25×10^{18} electrons) per second. There is one ampere in a circuit that has one *ohm resistance* when one volt is applied to the circuit. One should not speak of the “flow of current.” The current exists; the charge flows. This is analogous to the current in a river, which consists of the flow of water.

AMPEX: A former manufacturer of videotape recorders, analog audio tape recorders, and associated magnetic tape media. For the historic trivia fan, AMPEX is an acronym based on the founder’s name, Alexander M. Poniatoff Excellence.

amplifier: An electrical circuit or device designed to increase the current, voltage, or power of an applied signal. An amplifier is an *active* device and, strictly speaking, should always increase the power of a signal; some amplifiers, such as certain distribution amplifiers, may only reduce the *impedance* level of the signal for the purpose of driving long lines.

A

amplifier gain: The amount of amplification that an amplifier provides is called its *gain*. The gain is a ratio of the input signal level to the output signal level and is simply a number. Commonly expressed in dB, one should not express the voltage gain of an amplifier in dB unless the input and output *impedances* are matched as the gain of a typical amplifier is not related to its power output capability. For instance, if an amplifier has a voltage gain of 10, it might be said that it has a gain of 20dB because it actually would raise the power level of a signal by 20dB if the input and output impedances were matched. In practice, however, this is very seldom the case, and the true power gain is usually very much different from what would be predicted by the voltage gain. See *impedance matching*.

amplitude: The relative strength (amount) of a signal, without regard to its frequency content. Amplitude is measured by determining the amount of fluctuation in air pressure (of a sound), voltage (of an electrical signal), or numerical data (in a digital application). When the signal is in the audio range, amplitude is perceived as *loudness*. Amplitude is the measurement of how much energy the sound has, i.e., the total change in air pressure during a single cycle of the sound wave. Amplitude, or sound pressure, is measured in a scale called *decibels* (dB). An increase of 3dB is equal to a doubling of a sound's pressure. Amplitude can be expressed as either a negative or positive number, depending on the signals being compared. See also *magnitude*, *SPL*.

amplitude errors: See *frequency response errors*, *jitter*.

Amplitude Modulation (AM): The instantaneous amplitude modulation of one signal by another. This results in the formation of *sidebands* which contain the same information as the original signals, but translated upwards and downwards in frequency. In AM radio transmission, the audio signal is combined with a very high-frequency sine wave, called a *carrier*, in such a way that the amplitude of the carrier is varied in exact response to the amplitude and frequency of the signal. This is called the amplitude modulation of the carrier. The modulated carrier is transmitted at high power where it is received by radio sets that are tuned to the carrier frequency. The modulated carrier is then demodulated by a process called *detection*, recovering the original signal. In radio, a circuit that does amplitude modulation is also called a *heterodyne*.

amp/speaker simulator: A *filter* circuit that mimics the amplifier and loudspeaker voicing of an electric guitar and amplifier system.

AM suppression: The ability of an FM tuner or receiver to reject *amplitude modulation* of the received signal and be sensitive only to *frequency modulation*. Much of the interference and noise in broadcasting appears as amplitude modulation, so a tuner with good AM suppression will have less *distortion* and *noise* than a tuner with poorer suppression. Also called *AM rejection*.

anacrusis: See *beat*.

A

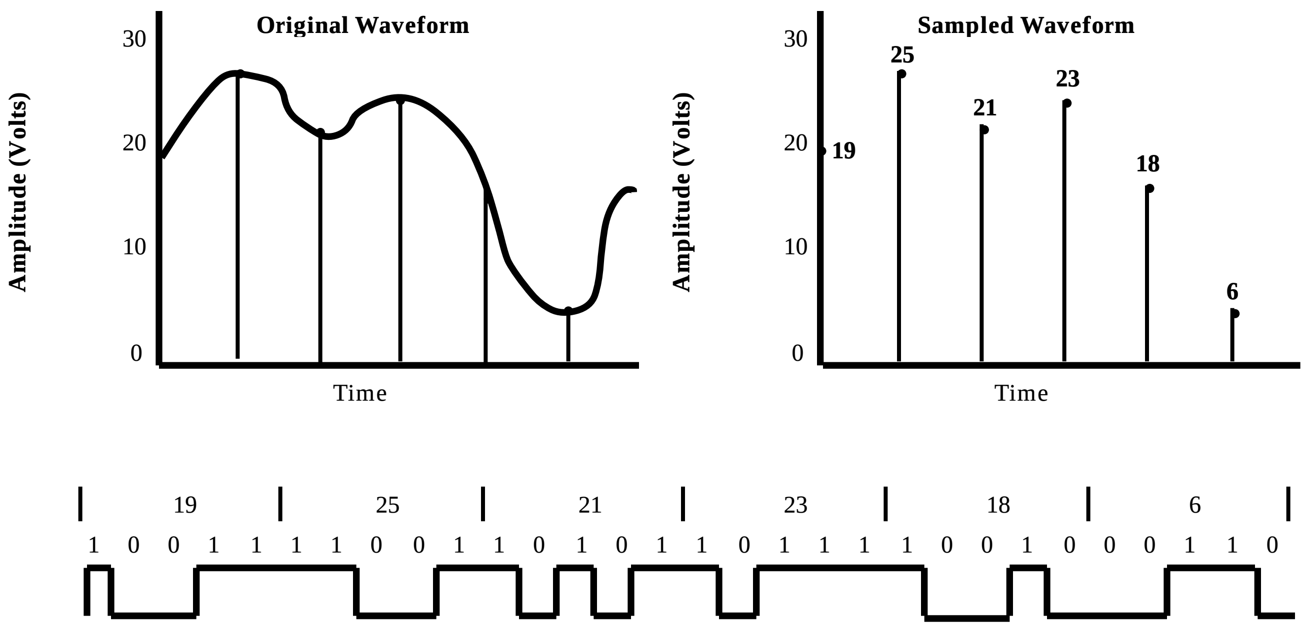
analog: Capable of exhibiting continuous fluctuations. An audio signal is an electrical replica, or analog, of the waveform of the sound it represents. The voltage of the signal varies up and down (negatively and positively, in electrical terminology) the same way as the sound pressure varies up and down at the microphone. In an analog synthesizer, such parameters as *oscillator pitch* and *LFO* speed are typically controlled by analog control voltages rather than by digital data, and the audio signal is also an analog voltage. Compare with *digital*.

analog recording: Any method of recording in which the recorded *waveform* is a continuous representation of the original signal, e.g., conventional magnetic tape recording.

analog sequencer: A *sequencer* into which sounds for storage and playback are fed as analog signals, via analog *potentiometers*.

analog synthesis: See *subtractive synthesis*.

analog-to-digital converter (ADC): Commonly abbreviated *A/D converter* or just *A/D*. A device that changes the continuous fluctuations in voltage from an *analog* device (such as a microphone) into *digital* information that can be stored or processed in a *sampler*, *DSP*, or digital recording device.



Analog-to-Digital Conversion

anamorphic: The camera / projector lens system which squeezes an image, usually originally in a 2:1 *aspect ratio*) onto film during shooting and "unsqueezes" it during projection. The resulting viewed image has an aspect ratio twice as wide as what was originally recorded on the film, e.g., if the image on the print is 2.2:1, the screen image will be 2.4:1. See also *CinemaScope*, *flat*(4), *'scope*.

andante: At a "walking" speed: 76-94 bpm. *Andantino* can mean either a little faster or a little slower than andante, although it more commonly denotes a little faster.

A

anechoic: Without echo. Said of an acoustic which is *free-field*, and specifically of a room which is designed to produce no *reverberation* or other echo effects. This is achieved by giving the walls very irregular surfaces of considerable and varying depths so that, in theory, all sound waves which strike them are completely absorbed and not *reflected*. Anechoic chambers are used to test audio equipment and for other types of acoustic and electromagnetic research.

anharmonic: See *inharmonic*.

anhysteretic: See *hysteresis*.

anode: The anode in any electronic component, such as a silicon diode or a vacuum tube, is the electrode normally connected to the positive voltage.

answer print: The first composite print made from the edited picture negative in 35mm film, or the *A*- and *B-rolls* of a motion picture in 16mm. Each shot is exposed, color-balanced, and otherwise processed. Further changes and corrections can be made in a second or third answer print, if necessary. In many contracts, the delivery of the answer print is specified because it means that *post* has ended and release printing can begin, although the *release print* is usually made from an internegative, not the answer print. The answer print is not the same as a *black-track* answer print which contains no *soundtrack*.

anti-aliasing filter: Before a signal is subjected to the process of *A/D conversion*, it must be passed through a lowpass *brick-wall filter* to remove any components that are higher than the *Nyquist frequency*. This is because it requires at least two samples per cycle to determine the existence and strength of a frequency component or the A/D process will create *aliased* signals. See *reconstruction filter*, *decimation*, *FIR*, *IIR*.

anti-imaging filter: See *reconstruction filter*.

antinode: A place of minimum *sound pressure level* in a standing wave, as opposed to a *node*, which is a maximum level.

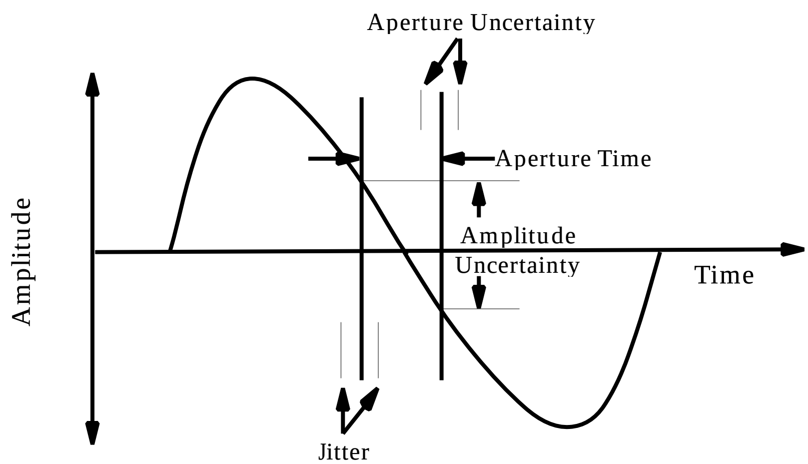
antiphase: See *out-of-phase*, *phase reversal*.

antiphonal: A term used to describe music that is played or sung in alternating sections by two separate groups of performers, widely separated.

AOR: Album-Oriented Rock or Adult Oriented Rock. A tendency of some FM radio stations to play longer album tracks.

A

aperture time errors: In an *A/D converter*, the sample-and-hold circuit would ideally take zero time to determine the *level* of the signal waveform and to hold this level until the next sample is called for. However, it takes a finite time to charge the holding capacitor in the *sample-and-hold* circuit, and this is called the *aperture time*. Because the time required to establish a new value of charge depends on the amount of change in the signal level from one sample to the next, the aperture time will vary with the rate of change in signal level, increasing for high-level, high-frequency signals. The starting time of the sampling aperture is also slightly uncertain, and this is called *jitter*. In other words, lack of precision in the sampling time leads to amplitude errors in rapidly changing signals. The errors involve aperture time, uncertainty in aperture time, and jitter. The result is distortion of the sampled signal which rises with frequency.



Aperture Time Errors

Apple () menu: The main menu on a Mac, used to access system utilities (such as Keycaps), applications, files, and control panels. This is the equivalent of the *Start* menu on a PC-type system.

AppleScript: A system-wide macro facility on Macs which gives operating system-level control for compatible applications.

APPV: Audio Post-Production for Video. The process of preparing the individual soundtracks and the final mix for a videotaped production.

APRS: Association of Professional Recording Studios. An industry body set up to ensure a uniform standard of service and practice in the area of sound recording.

A

APRS Tape-Label System: The *APRS* has decided on a standard color-code for tape labelling:

Type	Color	Details (Note: A clone is digital and a copy is analog.)
Session Tape takes.	blue	A multitrack or two-track work tape that may contain out-
Original Master	red	The first-generation of the final stereo product. Not necessarily suitable for production purposes.
Production Master program indicated. For encoding for the	green	All necessary EQ and treatment has been applied to the material (vinyl, CD, cassette, DAT) for the format example, a CD master would need further PQ- pressing process.
Production Master Copy DAT) copy and should not be	orange	If source and duplicate are digital, then the copy is a (CD, clone. If the duplicate is analog, the tape is a duplicated further without the producer's consent.
CD Tape Master (original, regenerated	grey	Fully prepared and PQ-encoded tape for glass mastering digital clone). Any clones generated must include timecode and relaid PQ-encoding information.
Safety Copy permission.	pink	Strictly for safety. Not to be used without producers'
Not For Production	yellow	Identifies a tape that is not currently suitable for production.
Media Copy or on the box.	yellow	Supplied for a specific medium and not for general production reproduction. May also include timecode as detailed

APT x100: See *ISDN*.

A&R: Artists and Repertoire. The department of a record company that selects the performing groups or artists who will be signed to the label, what songs or compositions each artist will record, and who will work with the artist in the production, arranging, and performance of the material in the production of master tapes.

aria: Italian for air (song). Generally indicates a composition for solo voice with accompaniment, also by extension, a lyrical instrumental piece.

A-roll: Film footage used to introduce or provide backup material for a live video broadcast.

arpeggio: The playing of *chord* patterns by sounding each note in a sequence, rather than simultaneously. An *arpeggiator* is a device which will automatically produce arpeggiation, given the parameters which control Direction (up or down or random), a Hold button which allows note patterns to be triggered which keep playing when the keys are released, and a Range control which sets the group of notes to be played over.

A

arrangement: (1) A version of a piece of music for resources other than those originally intended. This may be an instrumental version of a vocal number, a piano reduction of an orchestral piece, or may involve altering other parameters of the original, such as its harmony, rhythm, or structure. (2) In sequencers, a term sometimes used for the general layout of *tracks*, *channels* and *patches*, rather than a complete song. This template can often be saved as a separate file.

articulation: The way of characterizing notes (usually in a melody) by the precise control of their individual lengths to produce or eliminate gaps between them. The terms *staccato* and *legato* reflect the two extremes of articulation. It is one of the most important ways by which music can be shaped into phrases.

ASCAP: American Society of Composers, Authors, and Publishers. The first performing-rights organization formed in 1908, ASCAP collects fees for broadcast of recorded compositions on radio and television, and for live public performances of music and distributes payments to the copyright holders of these compositions.

ASCII: American Standard Code for Information Interchange. The most common encoding for transmitting text data digitally. The code employs 8-bit binary words, by which each letter of the English language, numeral, and symbol is uniquely designated.

aspect ratio: The width-to-height ratio of an image. Specifically in film, the format that the film image is intended to be shown in, most commonly expressed as width relative to height, where the height parameter has been scaled to represent 1 unit. Standard TV screens are 1.33:1, films shown in U.S. theatres are 1.85:1, *anamorphic* films are 2.4:1. Ratios may also be expressed as integers, e.g., the TV ratio may be expressed as 4x3, and widescreen TVs are 16x9, or an aspect ratio of 1.78:1.

asperity: A small irregularity or imperfection in the surface of a magnetic tape. Low-frequency noise in analog tape recordings caused by asperities produce *asperity noise* in the recording, a type of *modulation noise* in that the noise is manifested in the band immediately above or below the program signal. See *calendering*, *dropout*.

assemble editing: Editing of an audio or video program by making a master copy of the various takes, rather than physically splicing pieces of tape together. Virtually all digital editing is done this way. The opposite of *insert editing*.

assembly: See *copyediting*.

assigns: Push-buttons on the input modules of a control console that connect, or assign, that particular input to any of the output busses of the console. The signal is routed to the desired tape track of the destination device usually by a matrix of switches in each module of the mixing and / or recording console. This routing process is known as *assignment*.

asynchronous: Not according to a fixed rate of repetition. An asynchronous signal can occur at intervals which do not necessarily coincide with a fixed-rate system or master *clock* pulse.

ATA: See *IDE*.

A

ATF: Automatic Track Following. The system used by R-DAT players to ensure that the rotary heads follow the recorded track. This uses a set of signals that is recorded along with the digital data and which are passed to the *servo* controls to ensure that the tape is correctly positioned with respect to the heads.

atmospherics: See *backgrounds*.

ATR: Audio Tape Recorder. This is the analog version. A digital audio tape recorder is called a *DTR*.

ATRAC: Adaptive TRansform Acoustic Coding. A lossy, *5:1*-format*split-band perceptual coding* and *compression* scheme for reducing data to be written on a *MiniDisc*. ATRAC offers a 5:1 data reduction ratio in the case of MiniDisc, employing the equivalent of 52 filter bands for spectral analysis and requantization. Later versions of ATRAC vary the size of the sample blocks dynamically between 11.6ms and 1.45ms according to the input signal to allow for temporal masking, providing extremely good resulting sound.

A-track: The primary dialog track cut by the picture editor. The B- and subsequent tracks would be used for *overdubs*.

attack: The first part of the sound of a note. In a synthesizer *envelope*, the attack segment is the segment during which the envelope rises from its initial value (usually zero) to the attack (peak) level (often the maximum level for the envelope) at a rate determined by the attack time parameter. See *ADSR*.

attack time: (1) The rate of *attack* of a note. (2) The time it takes for a *compressor* or *limiter* to reduce its gain after a strong signal is applied to it. See *release time*.

attack transient: The actual attack *waveform*. See *transient*.

attenuation: The reduction, typically by some controlled amount, of an electrical signal.

attenuator: A potentiometer (*pot* or *pad*) that is used to adjust the *amplitude* of the signal passing through it. The amplitude can usually be set to any value between full (no attenuation) and zero (infinite attenuation). Pots can be either rotary or linear (sliders), may have discrete dentents (more often in older equipment), and can be either hardware or virtual sliders on a computer screen.

A-type: See *Dolby noise reduction*:

AU (.AU): An audio file format developed by Sun Microsystems, supported by some PC and Mac audio programs. This format supports stereo and mono files with either 8-bit or 16-bit resolution. It can encode *linear* files, or use *µLaw* or *ADPCM* compression.

audio: Literally, “I hear” in Latin. The term pertains to any signal, sound, waveform, etc., that can be heard, as opposed to *subsonic* or *ultrasonic* sound, radio-frequency signals or video signals.

audio coding mode: A parameter in *Dolby Digital* surround-sound format which refers to the number of channels and their location in for form F/R, where F is the number of front channels and R is the number of rear channels. For example, 5-channel surround is called 3/2 mode, stereo is 2/0, and mono is designated 1/0.

A

audio enhancer: Any *dynamic signal processing* device that in some sense improves a dull or lifeless sound. It can be a simple as *EQ* or a complex *DSP* algorithm. Examples of exciters are the Aphex Aural Exciter, BBE Sonic Maximizer, or SPL Vitalizer. Enhancers combine *dynamic equalization* with either *harmonic synthesis* or *phase manipulation*.

audio frequency: See *AF*.

audio silence: A type of diagnostic recording made with the recording *set-up* as planned, but with all faders down. Used to make a reference measurement of the *noise floor* and / or a tape of *biased noise*.

audio taper: A type of *potentiometer* designed for use as a volume control in audio equipment where the resistance varies in a *logarithmic*, rather than a *linear*, fashion with rotation of the knob. This gives a better correlation between control rotation and the subjective *loudness* of the signal.

audio-to-MIDI: Software or hardware that takes a *monophonic* instrumental or vocal line, analyzes the pitches, amplitude, and timbre, and converts the line to MIDI notes, complete with *pitch-bend*, MIDI *velocity* and volume, and possibly additional controller data.

AudioX: An open MIDI driver specification / standard being promoted by Cakewalk™.

auditory masking: See *frequency masking*, *masking*.

augmentation: (1) The increase of a major or perfect *interval* by one half-step to make an *augmented interval*. (2) The appearance of a musical idea in note durations which are longer than those used for its first appearance. This technique was often used in the ployphonic music of the middle ages and renaissance, as well as in contrapuntal music (e.g., fugues) of the baroque and later periods.

aural: Of, relating to, or perceived by the ear.

auto-assembly: In *on-line* editing, the process by which the edit-programmer produces the edited video master tape according to the instructions on the *EDL*, without human intervention. This is only possible where footage is consistently lit and exposed.

auto-correct: See *quantization*.

auto-input: One of the electronic operating modes of a multitrack recorder. When auto-input is selected, all channels will remain in *sel-sync* playback mode until the machine is placed in *record mode*. Any channels that are in *ready* status will then begin recording and will automatically pass their input signals direct to their outputs. When recording is stopped, these channels return to sel-sync playback mode. Also called *stand-by* mode.

autolocator: A device for controlling the transport system of a tape recorder, allowing *timecode* referencing such as *SMPTE*. Usually a number of *locate points* can be stored by the device. Some sequencers have an autolocate facility. Also called *zero locate*.

Automatic Volume Control: See *AVC*.

Avid: A brand of *nonlinear* video editing system, which, while not being exactly an industry standard, is the most commonly used digital video editing system.

A

automation: A system where manual control of a process is replaced or enhanced by computer control, such as mixing desk automation where faders, mutes, and equalization can be controlled in part or in whole by a computer. In *write mode*, the automation system produces a continuous record of all the actual fader settings and adjustments made by the engineer during a mix. Most systems allow changes on replay, while remembering and recreating previous manipulations of other tracks. The level changes are recorded and recreated by *VCA*s in each input module of the console. The VCA-produced data can be recorded directly onto a track of the multitrack tape, giving a continuous record of all mixdown fader settings. Or, the VCA outputs can be recorded onto a separate disk. In the latter system, alignment of the fader data with the multitrack master tape is achieve by referring to a common *SMPTE timecode* recorded on the tape and disk systems. See *mute mode*, *mute-write*, *null-point*, *read mode*, *snapshot automation*, *update mode*, *write mode*.

autopanner: A device for processing a signal so that it can be made to appear at various positions in a stereo image via a remote control or MIDI commands.

aux or **auxiliary**: An assignable, *line-level* input with no dedicated input source. Generally refers to an input connector in a preamplifier or integrated amplifier, signal processor, mixer, effects device, etc. The aux input has no *de-emphasis* or other special equalization and accepts *line-level* signals. Tone controls on a preamp usually also affect signals sent to the aux input.

auxiliary: A bus allowing a signal to be sent from a mixing desk prior to the main output, usually to provide an input to effects. See *effects send*.

auxiliary envelope: An extra envelope in a synthesizer that, instead of being hard-wired to a *filter* or amplitude, is intended as a modulation source that can be applied to various destinations.

auxiliary messages: A classification of MIDI messages which includes Active Sensing, All Notes Off, Local On / Off, and Reset, and which describes whether the particular MIDI device responds to any of thsese messages.

aux section: A smaller, independent *mixer* within the main mixing console which has an output consisting of a mix of everything going into the channels on which the appropriate *effects send* been turned up.

auxiliary send: Also called *aux send*. See *effects send*, *insert point*.

AVC: Automatic Volume Control. A circuit which adjusts the *gain* of an audio device in inverse proportion to the incoming signal level. An example is a portable tape recorder which is designed for speech recording; when the speaker is close to the microphone, the gain is reduced so as not to overload the tape. Also, a circuit which increases a TV or radio receiver's gain when it is tuned to weak stations and decreases the gain when it is tuned to strong stations. Called *AGC* (Automatic Gain Control) in TV.

AVI: Audio Video Interleaved. Microsoft's answer to Apple's QuickTime, and not compatible with Macs.

A

A-weighting: An *equalization curve* first applied to sound level meters in an attempt to make their measurements correspond better to the perceived *loudness* of sounds, decreasing the sensitivity of the meter to frequencies below 1kHz. An important note is that the bottom octave (32Hz) is attenuated by almost 40dB; the second octave (63Hz) by 26dB, and the third octave (125Hz) by 16dB. See *B-weighting*, *C-weighting*, *equal loudness curves, SPL*.

axis: In microphones, the direction of maximum sensitivity, generally perpendicular to the surface of the *diaphragm* or ribbon. In loudspeakers, the line projecting through the center of the *voice coil* toward the listening area. This is usually the direction in which the speaker exhibits the best overall *frequency response*. See *acceptance angle*, *off-axi*s, *directional microphone*, *polar pattern*.

azimuth: In a tape recorder, the azimuth is the angle that the gap in the *record* or *playback head* makes with the direction of the tape travel, and it must be precisely 90˚ to ensure proper high-frequency performance.

B

B: The right-hand part of a *stereo* signal.

baby boom: The nickname of the Dolby 70mm process that dedicates two of the six tracks on a 70mm print to low-frequency information (signals below 250 Hz) This term is no longer used as the new digital multichannel film sound formats specify a dedicated subwoofer track.

backbeat: A music term referering to the second and fourth *beats* in a four-beat *bar*, often emphasized by the drummer.

back coating: In *magnetic recording tape*, a thin coating applied to the non-oxide or back surface of the tape to reduce slippage between tape layers, prevent accumulation of static charge, and to minimize curling or wrinkling.

backfill: To edit fill between words so that the whole length of a scene, including sections where the take or angle in question is not being used, is contiguous.

backgrounds: Sound effects that sonically define the time and place of a location. Also called *ambience*, *atmos* or *atmospheres*, backgrounds give a sense of lush sonic effects and placements, more specifically, usually a tasteful use of *pan* controls, *reverbs*, *delays*, and other positioning tools. BGs are considered sound effects and are not the same as *room tone.*

backing track: Pre-recorded music used by a singer or other musician during performance and which augments or entirely replaces other performers. This has become increasingly popular as musicians attempt to recreate the sound of their studio recordings live on stage.

backing vocals: In popular music, extra vocal parts which fill gaps in, or harmonize with, the lead vocal line. Usually sung by specialist session singers. Usually abbreviated *bvox*.

backline: On-stage instrument amplification.

back plate: In a *condenser* microphone, the fixed, rigid *capacitor* element that is charged with an electric *polarity* opposite to that of the *diaphragm*.

backtiming: Subtracting the length, in minutes and seconds, of a recorded segment from the time in a longer program at which the segment is supposed to end. If a three-minute segment is to end a 30-minute program, backtiming will indicate that the end segment needs to roll at 27:00.

backward masking: See *temporal masking.*

BAC&S: British Academy of Compusers and Songwriters. A group being formed among the current Association of Professional Composers, the Composers' Guild of Great Britain and British Academy of Songwriters, Composers and Authors, building a larger and more influential "umbrella" organization.

B

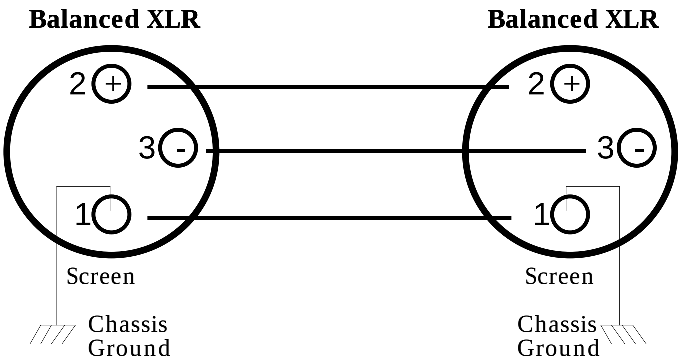
baffle: A partition placed between two sources of sound, or between a sound source and a microphone, to prevent sound from passing through. The baffle, or *screen*, may be made of any material with a high *absorbtion coefficient*. Most baffles are designed as movable partitions, and are used to isolate individual instruments in recording studios.

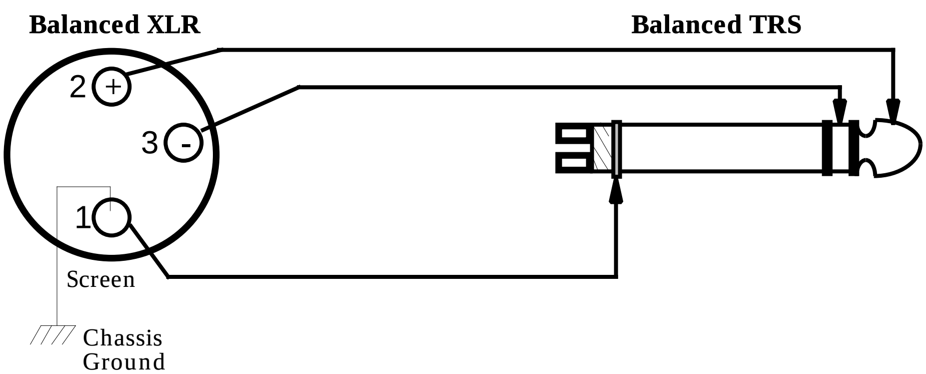
bake off: Hollywood colloquialism for the meeting of the Sound Branch of the Academy of Motion Picture Arts and Sciences in which the members hear ten-minute clips of the seven films that have made the semifinals of the Best Sound Effects Editing award.

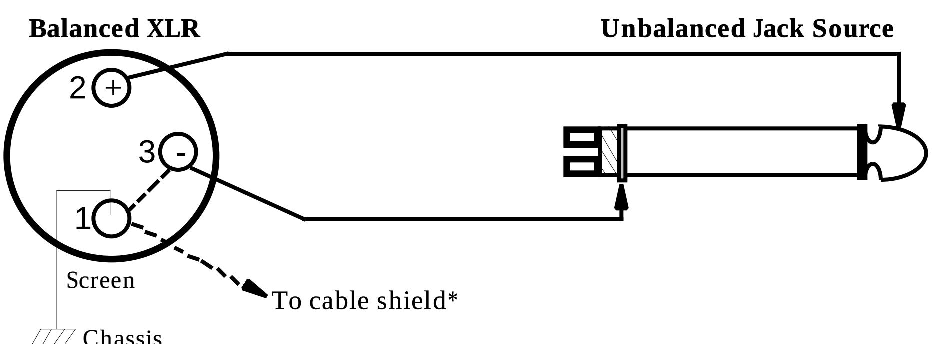
balance: (1) The amount of relative signal provided to each of two (or more) audio channels. (2) A control on a synthesizer which adjusts the relative volumes of two different sounds which it can voice simultaneously. Not to be confused with *pan*.

B

balanced line: Audio lines in which the signal *current* is not carried by the cable shield of a *shielded cable*. This requires two conductors for the signal, enclosed in a shield, with neither conductor connected to the shield. The circuit utilizes two identical conductors operated so that the voltages on each of them are equal in *magnitude*, but opposite in *polarity* with respect to ground. Compare with *unbalanced line*. See *common-mode*.







*If no connection between pins 1 & 3, with short to cable shield, there will be a 6dB loss in signal strength

B

balance stripe: See *mag film*.

ballistics: The dynamic behavior of the needle in a meter, such as a *VU meter*.

balun: BALanced-to-UNbalanced. A transformer device used to convert a singled-ended (unbalanced) signal to a differential (balanced) signal. A balun is a essentially transformer with one leg of the input and output windings hooked together. More complicated devices may also change impedences at the same time. The most common use for a balun is a 75Ω coaxial-300Ω twin-lead converter used in television.

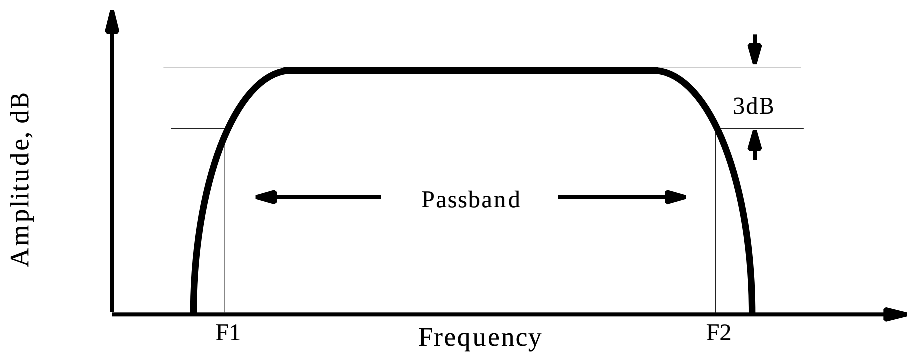
band: (1) An extent along the frequency dimension in which a signal exists is the *band*. For instance, an *octave* band is one octave wide. The *AF* band is 20Hz-20kHz wide. (2) The wider spiraled grooves that separate any two selections on a record. (3) Band is also used to indicate any single selection on a record, cassette, or reel-to-reel tape or CD, i.e., a *track*(3).

band-limited: A signal is said to be band-limited if its frequency content is restricted to a particular frequency range. For instance, the output signal of a CD player is band-limited to 20kHz by the *reconstruction filters* built into the player.

band masking: See *frequency masking*.

band part: A notated form of a piece of music, derived from a *full score*, usually containing only the music for a single instrument or pair of similar instruments.

bandpass filter: A filter which has both a high-frequency and low-frequency *rolloff*, and only frequencies in between are allowed to pass. When applied to sound synthesis, a band-pass filter makes the waveform sound like it is coming down a phone line as telephone lines cannot reproduce lows or highs. The opposite of a *band-reject* filter.



Bandpass Filter Frequency Response

band-reject filter: A filter that discriminates against signals in a specific frequency band. The most common band-reject filters reject a vary narrow frequency band, and they are usually called *notch filters*. The opposite of a *bandpass* filter.

B

bandwidth: (1) The capacity of the channel through which information can pass. In audio, the *rated bandwidth* of a device is the portion of the frequency spectrum it can handle without significant degradation. In digital communications, the bandwidth is the amount of data that can be transmitted in a given period of time. (2) The bandwidth of a *bandpass* filter is the upper *rolloff frequency* minus the lower rolloff frequency, i.e., the frequency range in *Hertz*, or *band*, passed by the filter.

bank: (1) A set of patches. (2) A related set of items, e.g., a filter bank: a set of filters that work together to process a given signal.

Bank Select: A type of MIDI *controller message* which specifies which bank of (receiving) sequencer programs to use; a way to get around the 128 program limit specified by MIDI.

bantam: See *TT connector*.

bar: In written music, a grouping of pulses into a convenient unit which falls between two *barlines*. A barline is the vertical line which crosses the *stave* at regular intervals. The bar begins with the downbeat and ends immediately before the next downbeat, and will contain a constant number of beats of the type determined by the *time signature*, e.g., a bar of $\frac{4}{4}$ will have four quarter-note *beats*.

Barkhausen effect: The tendency of the magnetic elements or *domains* on a magnetic medium to influence one another and to become magnetized in one direction or another as a group rather than individually. This means that a magnetic medium, such as recording tape, has a graininess in its magnetic makeup which is what causes most background noise, or *tape hiss*. *Modulation noise*, which is only present in conjunction with a recorded signal, is also caused by the Barkhausen effect, and is sometimes called *Barkhausen noise*.

barney: See *blimp*.

base: In *magnetic recording tape*, the thin ribbon of polyester or other plastic material to which the oxide and *back coating* are applied, measured in *mils*. For example, the base of most professional recording tape is 1.42 mils thick.

basic channel: In a MIDI device, the channel on which the device receives fundamental messages governing its operation, e.g., Reception Mode changes. In *Mono Mode*, the basic channel is the lowest-numbered channel.

basic track: The group of instruments or vocalists recorded first during a multitrack session. This group, perhaps including bass, drums, and standard rhythm section, will be played back through headphones to other instrumentalists who later *overdub* solos, lead or background vocals, or narration, and other *sweetening* or sound *effects*. See also *backing track*, *bed*.

basket: The metal frame of a *loudspeaker*.

bass: The very low end of the audio spectrum, approximately 20Hz-200Hz or 300Hz.

Bass Intermodulation (BIM): Bass intermodulation is a type of *distortion* caused by the modulation of audible frequencies by *subsonic* noise.

B

bass build-up: An increase in molecular pressure variation, not molecular velocity, which occurs at low frequencies at room boundaries. The pressure *nodes* for all frequencies build up in the corners of rooms, particularly at the intersections of the walls and floor or walls and ceiling. These regions can roughly be thought of as *resonant* spaces, and energy can be trapped quite effectively by placing frictional absorbers at the desired *quarter wavelength* out into the room from the corner. See *bass trap, boundary effect*.

bass reflex enclosure: A type of loudspeaker enclosure with a hole, or *port*, in the surface on which the speakers are mounted, usually the front. Since this allows some of the energy from the rear of the speaker cones to project into the listening area, bass reflex systems have relatively high *efficiency*. Sometimes called a *ported enclosure* or a *vented enclosure*.

bass tip-up: See *proximity effect*.

bass trap: A specially designed low-frequency sound absorber to reduce the effects of *standing waves* in recording studios. It is a tuned absorber and may have a narrow or wide range of frequencies over which it operates. It usually consists of resonant wood panels with absorptive material behind them, or suitably shaped slots in a wall or ceiling. See *bass build-up, boundary effect*.

B-chain: The film industry's term for the sound reproduction system, including *amplifiers*, *crossovers* and *loudspeakers*. See *A-chain*, *chain*.

B-Channel: See *ISDN*.

beat: A regularly occurring *pulse* that can be heard or implied. (1) When two *periodic* signals are less than 20Hz or so apart in frequency, and if they are mixed together, the amplitude of the combined signals will fluctuate as they alternately reinforce and cancel each other. These amplitude fluctuations cause *loudness* fluctuations and are called beats. See also *difference tone*. (2) In music, the sensation of a basic pulse from which all rhythm in the piece is derived. Beats are of three types: *downbeat* is a strongly accented pulse, such as the first in the *bar*; *offbeat* is any pulse other than the downbeat; *upbeat*, also called the *anacrusis*, is a special case of offbeat which immediately precedes the first beat of the bar, and hence the *bar line*.

bed: Background music used underneath a narrator or foreground dialog. Primarily applied to commercial radio or television spots. Also called *basic tracks*.

bel: The logarithm in base 10 of the ratio of two different levels of *power*, acoustic or electric. Since large changes in *loudness* correspond to fractional portions of a bel, the *decibel*, $1/10$ of a bel, is used as the measurement unit of level for sounds and audio signals. See Appendix A.

bell filter: A type of filter that allows the boost or attenuation of a specified set of frequencies around a center frequency. Bell filters often allow user adjustment of the center frequency, Q, and the amount of boost or cut. Bell filters are sometimes also known as *haystack filters*.

bench: In film, the editing table which consists of rewinds handling reels of 35mm picture and *mag film*, a sprocketed *synchronizer* that keeps the reels in sync, in addition to providing a count, and a *squawk box*. See *mut*.

B

bend: To change *pitch* in a continuous sliding manner, usually using a *pitch-bend* wheel or lever. An upward bend is created by pushing away from the front of the modulation controller, and creates an increasing pitch, and vice-versa. See *bend depth*.

bend depth: The amount of *pitch-shift* possible if the *pitch-bend* modulation controller is moved as far as possible. This is usually set to a *whole step*, but for special effects (such as electric guitar), it could be set to an octave or more.

Betacam: A professional analog videotape *format* employing the ½” Beta format, but at an increased tape speed, which gives picture quality comparable with the 1” C format. Betacam also allows separate recording of the red, green, and blue picture information via its RGB mode for computer use. This capability gives much better control of edge-cuts in special effects. Often called *Beta* for short.

Betamax: A system used for ½” color videotape recording, developed by Sony for consumer systems. Generally acknowledged to give higher picture quality than VHS.

B-format: A 1” professional video *format* developed by Bosch. Although generally considered superior to the standard *C-format*, B-format equipment is used only in a few production and post-production facilities. B-format video masters must be transferred to C-format for broadcast.

BG: Background. The *walla* in a commercial or other video production, over which other sound effects, music, and dialog are dubbed.

biamp: Short for biamplification. A two-way *crossover network*.

bias: (1) Bias is the voltage or current that establishes the *intrinsic noise floor* of an *active* device. (2) In an analog tape recorder, bias is an *ultrasonic* signal, usually between 100kHz-200kHz, which is mixed with the audio signal and applied to the recording head, reducing distortion by reducing the *hysteresis* inherent in the tape recording process. This process is known as *AC bias* because the bias current is alternating. The ideal setting of analog bias involves a compromise between the *MOL* of the tape, *noise*, and third-*harmonic distortion*. In general, classical recordings use a bias setting with lower distortion and lower MOL; rock or other recordings prefer a higher distortion in order to get the highest *S/N ratio*. Digital recorders do not require bias as the signal consists only of a bitstream of 0s and 1s, regardless of the audio frequency being recorded. (3) See *electret microphone*.

biased noise: A tape loop of audio silence, processed by a recorder with some kind of noise-reduction enabled, such as *Dolby-SR* (where it is called *SR noise*) or dbx. Used to check equipment for ground loops or other problems in recordings and / or masters. Sometimes called an *alignment recording*. See *Dolby noise*.

bi-directional microphone: A *figure-eight microphone*.

bin: (1) A barrel into which strips of film hang, suspended from a row of pins or small nails above. Also called an *editing bin*. (2) In tape duplication, the container or housing that holds a tape *loop* to be duplicated.

B

binaural: Literally, “having two ears.” Because humans have two ears spaced apart by the width of the head, the human hearing mechanism can make use of *amplitude*, *phase* (arrival time) and spectral (*frequency response*) cues to help determine the direction from which a perceived sound is coming. See *binaural synthesis*.

binaural synthesis: A type of recording-playback system where two microphone inputs are specially processed to simulate the frequency-dependent time delays that would occur between the ears on a human head. The binaural localization cues are preserved, and the listener is able to achieve localization of sounds as if s/he were actually at the site where the recording was made, despite the fact that binaural recording has no ability to accurate *image* the sound. Also called *dummy head* recording.

binder: A liquid or gelatinous medium in which oxide particles are suspended for application to *magnetic recording tape*. Usually consists of a solvent that evaporates, and an adhesive substance which, when dry, permanently bonds the oxide to the base.

binding post: A type of *terminal* which allows wires, such as loudspeaker wires, to be connected to the output of an amplifier with alligator clips, banana plugs or bare wire.

binky: Film sound slang for a mixing “top sheet,” indicating the layout and content of the *premixes*. The layout is usually one column per premix.

bin-loop master: A special tape that is used in cassette duplication equipment. It contains both sides of the tape and is either run at a very high speed or, for higher quality *dubs*, in real-time.

B inputs: (1) An additional set of inputs to a mixer channel that allow either additional (but not simultaneous) tracks to be assigned a mixer; (2) More commonly these days, a different source of the same information that is appearing on the A inputs. This latter technique allows a sound editor to work *offline* on a sequence while the mixer is adjusting the overall EQ and level in automation, while playing back from another copy. The material is recorded to tape, after witching inputs, when the editor is finished.

BIOS: Basic Input-Output System. An operating system which resides on ROM and is used to control disk access, exclusively. Used in some samplers and sequencers to control the internal hard drive.

bi-phase: An electronic reference signal used by mag recorders, editing stations, and film projectors. See *bi-phase modulation*, *pilot tone*, *neo-pilot*, and *control track*.

bi-phase modulation: In *SMPTE timecode* generation, the electronic process that produces the signal containing the SMPTE data itself. A 1.2kHz *square wave* is momentarily modulated to 2.4kHz with each new bit of location information coming from the *master clock*.

bi-phase/tach: An electronic pulse used by some film equipment and other motor-driven devices. Similar to a *bi-phase* signal, but different in the way directional information is provided. See also *tach pulse*.

B

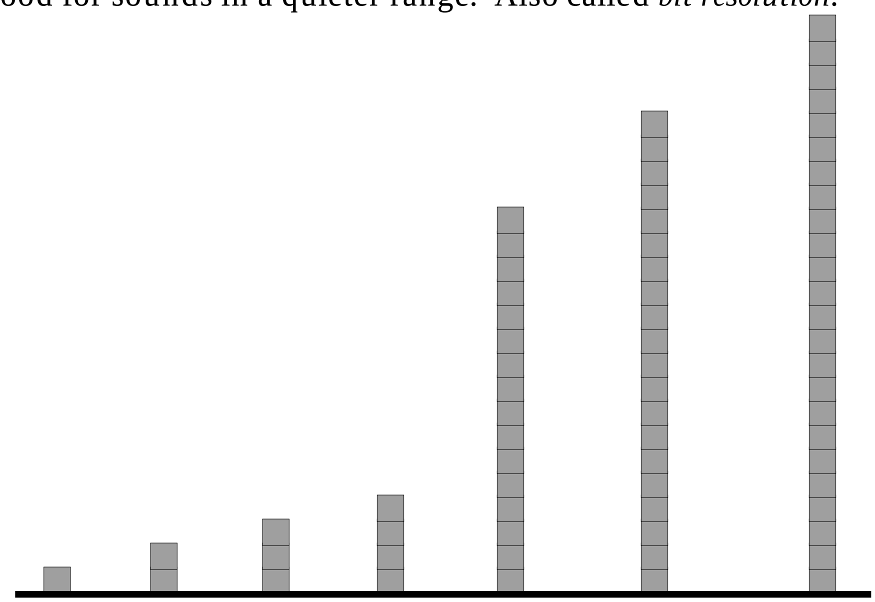
bipolar: A type of loudspeaker design where the sound emanates from the sides of the monitor, specifically designed to be *surround-sound* monitors. These type of speakers work well for *ambience* material, but less well for dialog, soundtrack or main effect sound. This is opposed to a *direct radiator* speaker which distributes the sound in front, or a *tripole* design which is a combination of a direct radiator and a bipole.

birdies: Extraneous high-pitched whistles sometimes present in tape-recorded signals where the high-frequency content of the signal causes *beats* with the *bias* signal. Also used to refer to high-pitched interference in AM radio reception.

bit: Binary digIT. The representation of data using base-2 arithmetic, i.e., a series of ones and zeroes. Digital audio is encoded in *words* that are usually 8, 12, 16, 20 or 24 bits long (the *bit depth*). Each added bit represents a theoretical improvement of about 6dB in the *S/N ratio*.

BITC: Burnt-In Timecode. Video that shows the *SMPTE* time on-screen in a window along with the picture, eliminating the need to watch a time-code reader. Accurate in still-frame. Sometimes called a *window dub*.

bit depth: The number of data bits used to encode each *sample* point. Bit depth determines the accuracy of a sampler, converter, or other digital device in capturing momentary changes in a sound's amplitude. Typical bit depth is 16 bits, which is good for capturing loud sounds, but less good for sounds in a quieter range. Also called *bit resolution*.



Number of Bits	1	2	3	4	16	20	24
Amplitude Steps	2	4	8	16	65,536	1,048,576	16,777,216
Dynamic Range	6dB	12dB	18dB	24dB	96dB	120dB	144dB

Dynamic Range as a Function of Resolution (Bit-Depth)

bite: A subjective term for the sharp onset or *attack* of a musical instrument, especially brass instruments. Excessive bite can result from positioning microphones too close to the instrument or from *distortion* caused by a momentary overload. See *attack transient*.

B

bit resolution: See *bit depth.*

bit shifting: A technique for *lossless compression* which, rather than encoding the entire data word, only bit cells with data (ones) are stored, and the null data (zeroes) are removed. For example, if only 19 bits of a 24-bit word contain data, only those bits are transmitted.

bit-splitting: A feature on some *A/D converters,* digital recorders, DAW or other digital devices to choose *word* lengths to accommodate various output channels, such as a choice between six outputs at 20-bit resolution, or four output channels at 24-bits per sample.

black-burst: A type of clock reference, this is essentially a video signal without any picture and without any positional information. Also known as *house sync* as a black-burst signal is typically distributed throughout a recording / editing facility as the facility *master clock* due to the extremely accurate clock signal provided. See *video black.*

blacking: The recording of a *periodic* signal on a blank video tape which marks the start of each video *frame.* See *video black, video sync, control track.*

black-track print: A version of the *answer print* which has no sound, i.e., it is "silent," made from the original camera negative. The first answer prints are usually black-track in order to proceed with the color timing, even when though post-production sound has not been finalized.

blanking interval: The blanking interval occurs at the end of each video *frame,* during which video information is absent. The interval occurs when the CRT electron gun scanner goes from the bottom-right corner of the screen to the beginning of the next *field*(4) in the top-left corner.

bleeding: See *crosstalk, channel separation*.

blimp: A solid cover for a motion picture camera, designed to completely contain camera noise. A *barney* is a padded cover for a portable camera which attenuates, but does not eliminate, camera noise.

blocking: Plotting actor, camera and microphone placement, and movement in a production.

Blue Book: A *CD* specification for data, as opposed to sound or video.

Blumlein pair: A stereo miking technique which uses two *figure-eight* mics, crossed at a 90˚ angle, set up as closely as possible to one another. This is also sometimes called *coincident figure-eights*. See also *coincident pair.*

BNC: (1) Bayonet-Nut Coupler. A two-conductor, low voltage, locking connector most commonly used for the connection of video and high-frequency *clock* signals. (2) Blimped Newsreel Camera. The 35mm Mitchell Camera model which was the industry standard for over 30 years. See *blimp.*

board: A synonym for a recording console or *mixer*. (2) Short for a film storyboard.

B

boom: (1) In general recording, any sort of microphone stand with extending sections that allows the microphone to tilt and be pointed at a target, such as above a performer or some section of an orchestra, etc. Also called a fishing rod, although the latter usually refers to a lighter-weight rig more suitable for close miking. (2) The *LFE* in a mix. See *baby boom*.

boomerang: To mix a *sample* with a backward version of itself.

boost: Boost refers to an increase in *amplitude*, usually of a specific frequency or within a frequency *band*. *Equalizers*, the most common of which are tone controls, cause boost or cut of selected frequency ranges.

boost/cut control: A single control which has “no change” at its center-point. If the knob is rotated counter-clockwise, the input is attenuated; rotated clock-wise, the input is amplified.

bootstrap: An arrangement where the apparent *impedance* of a circuit element is reduced by applying an appropriate *feedback* voltage to it, improving the *linearity* of a circuit, thus reducing its *distortion*. It is especially useful in circuits that are required to carry a very wide range of power or voltage levels. Used in power amplifier output *stages*.

bounce: In *multitrack* recording, the process of recording several tracks and mixing those sounds down to one or two unused tracks. For example, on an 8-track recorder, you could record six tracks, bounce them down to the two remaining tracks, freeing up the original six tracks for recording use.

boundary effect: A sound reflection effect due to *room modes* (*standing waves*) which accumulate at walls. Sound wave reflections appear to make the localized sound level increase as all of the room modes terminate at the boundary (wall). Essentially as the wavefront approaches the wall, the amounts of molecular motion become smaller and smaller while the pressure differences become greater and greater as the wall resists the motion of the air molecules, the wall becoming a pressure *node*. The rigidity of the wall surface determines how much the pressure rises, i.e., how much of the pressure is reflected versus how much is absorbed. This occurs on a mode-by-mode basis at each *resonant frequency*. At very low frequencies, nothing large is rigid. However, at higher frequencies, boundary effect is more pronounced, e.g., frequencies above 100Hz in a room with typical walls. A related effect is often observed at a control room window, where the window itself will resonate at one or more resonant frequencies so that the window passes the resonant frequencies through to the (recording) space on the other side, somewhat reducing the boundary effect within the control room, but not providing sound isolation from the adjacent space(s). This last effect is worse for lower frequencies as higher frequencies tend to be absorbed by the glass in the window. Also called the *pressure zone effect*. See *absorption coefficient, bass build-up, bass trap*.

boundary microphone: A boundary mic uses a small *condenser microphone* capsule mounted very near a sound reflecting plate, or boundary, so there is no delay in the reflected sound. Direct and reflected sounds add in-phase over the audible range of frequencies, resulting in a flat response, free of phase cancellations, excellent clarity and reach, and the same tone quality anywhere around the mic. Boundary mics have a *directional* response that is either half-*omni*, half-*cardioid* or half-*supercardioid*. An example of a boundary microphone is a PZM (pressure zone microphone.)

B

bpm: Beats Per Minute. The usual measurement of *tempo.*

bps: Bits per second.

Bps: Bytes per second.

break: In a piece of music, a break is a solo or section of reduced instrumentation, or even complete silence. In modern usage the term usually implies an opportunity for an instrumental solo.

breakjack: A type of *jack* socket fitted with switching terminals, so that insertion of a plug breaks an existing connection. Also called a *normalled connection*.

breakpoint: On synths and samplers, the specific value at which the tracking of scalable *parameters*, such as *velocity*, starts to take effect, or at which the nature of the *scaling* changes.

breath controller: A device which a performer blows into, bites, or presses with the lips, allowing the articulated sound to be digitally recorded by a synthesizer or sampler. Breath controllers can control volume, *filter* frequency or amount of *LFO*. They incorporate a device known as a *stress bridge*.

breathing: Audible fluctuations in the noise level of a signal caused by poorly adjusted or unsuitable *noise reduction* systems which produce a variable *program level* and / or *noise floor*. Also called *pumping*, *noise pumping* or *breathing*. Pumping is caused by the action of a *compressor*, occurring when one loud sound source causes severe gain reduction in the compressor. With each loud sound, the level of the other instruments will decrease sharply. Pumping occurs during program material. Breathing, on the other hand, occurs when the program stops long enough for the compressor to cease its gain reduction, suddenly boosting the noise floor of the program. *Quantization noise* can also exhibit breathing. See also *compander*.

brick-wall filter: A very sharp filter which masks any frequency outside the *passband*, for example, the lowpass filter at the input of an *A/D*, used to prevent frequencies above the *Nyquist frequency* from being encoded by the converter. See *aliasing*, *reconstruction filter*, *anti-aliasing filter*, *decimation*, *FIR*, *IIR*.

bridge: (1) Meter bridge. A structure mounted at the rear of a mixing desk, or on other equipment such as a tape recorder, which contains a number of *VU* or *PPM* meters. (2) Bridge mode. A method of driving a single load, such as a loudspeaker, from two similar (ideally identical) amplifiers in order to double the power presented to the load; a stereo amplifier operating at 200W per channel could provide approximately 400W into a single load in bridge mode. Many stereo amplifiers designed for sound reinforcement offer this option. See *bridged mono*. (3) See *bridging*. (4) Originally an eight-bar section of contrasting material in the middle of a song, but later applied to a linking section of any length. Also called a *bridge passage* or *middle-eight*. See also *break*.

bridged mono: A method of combining both channels of stereo power amplifiers to create a doubly powerful single-channel (*monaural*) amplifier. See *bridge*(2).

bridge passage: A section of music which links two musical ideas. A bridge is usually used to connect movements in different *keys* and / or *tempos*. See *bridge*(4).

B

bridging: The opposite of *impedance-matching*. When the input of an audio device is connected to the output of another device, it is a bridging connection if the second device does not appreciably load the first device and essentially no power is transferred. The second device is sensitive to the output voltage of the first device, and this is maximized when the loading is minimized. Most audio connections are bridging, and the load impedance is at least ten times greater than the source impedance. A bridging connection is made by connecting everything in *parallel* (all the plus inputs connect to the plus output, all the minus inputs connect to the minus output.) This not only allows for a number of loads to be connected to the same source before overloading it, but this also gets the maximum voltage swing possible from the source.

brightness: The amount of high-frequency signal present in a sound, which tends to make the sound appear closer. The opposite of *darkness*.

broadband: Including a wide range of frequencies, generally the entire *audio* range. Usually used in terms of referring to the broadband performance of an audio device with respect to some specification such as *noise*, *distortion*, etc.

B roll: See *A-roll*.

broom: To discard recorded sound during a mix. "Site brooming" is when a director rejects a whole group of effects, often the product of several days' work.

BTSC: Broadcast Television Systems Committee. The FCC committee that decided upon the *MTS* standards for stereo television sound in the U.S.

BTX: A brand name of electronic devices that will maintain synchronization between two tape recorders, tape recorder and a projector or video playback machine, etc. Used primarily to *interlock* one or more multitrack recorders to a video playback, for purposes of recording, overdubbing, or mixing music in sync with picture. The device uses *SMPTE timecode* for electronic control of all machines.

bucking: The cancellation of one signal or frequency component of a signal by another signal with equal amplitude but opposite *polarity*. See also *phasing*, *flanging*, *comb filter*.

buffer: An amplifier with a high input *impedance* and approximately *unity gain*. Used, for example, in a *mixer* at the back panel outputs for headphones and control room monitors to prevent the two loads from overloading the fader output and causing A-rolloff in high-frequency response. A sort of internal distribution amplifier.

bug: A small *contact microphone*, designed for stringed and wind instruments which work along similar lines to a *piezo pick-up*.

bulk dump: A *System-Exclusive* description of an actual sound sent over MIDI.

bulk eraser: A tape demagnetizer that can erase an entire cassette, reel of ¼" or multitrack tape without removing the tape from its carrier. Essentially a powerful electromagnet. Some bulk erasers have circuits built in that automatically fade the magnetic field up from and ultimately back down to zero. This eliminates pops and other erasure noise normally left on tape if the eraser is suddenly turned on or off. Also called a *degausser*.

B

bulk tuning message: A System-Exclusive message of the non-real-time type that allows the exchange of tuning data between MIDI devices as well as other devices such as computers, allowing microtuning or different *temperaments* by defining a specific pitch value. The frequency range is from 8.1758Hz to 13,289.73Hz, in steps of one half-step / 2^{14} =0.0061 *cents*, for each of the 128 notes in the MIDI range. Two messages are involved: a bulk tuning dump request message which is transmitted by a device in order to signify that it is ready to receive, and a bulk tuning dump message which contains the data for 128 tuning programs, each containing 128 pitch values.

bumpers: Small segments of music in a television or film score that usually precede a dissolve. In television, usually used before commercial breaks.

burnt-in timecode: See *BITC*.

bus or **buss**: In a *mixer*, a path via which the user can route a signal from one or more inputs to a specified destination. Typical destinations include: groups, mix, *auxiliary send*, *foldback*, etc. For example, "routing inputs 1-8 to the mix bus" means that the eight input signals appear additively at the mix output.

buzz track: Alignment film used to set the lateral alignment of the *optical film* recording areas for replay.

bvox: See *backing vocals*.

B-weighting: Frequency correction approximately corresponding to human hearing at 70dB SPL. See *A-weighting*, *C-weighting*, *equal loudness curves*.

bypass: A facility on an *effects* unit which allows the user to switch the incoming signal directly through to the unit's output, cancelling the effect so that an *A/B* comparison may be made quickly between the *wet* and *dry* signal.

C

cadence: A musical punctuation, indicating the end of an idea, or preparing the ground for transition to a new one; essentially a juxtaposition of two *chords*.

calendering: To reduce the *asperities* on the surface of a magnetic tape, the tape is squeezed between large steel rollers, a manufacturing process called calendering.

Calrec Soundfield microphone: See *Soundfield microphone*.

cancellation: See *phase cancellation*.

canned: Slang for pre-recorded, as opposed to *live*(3), music or visuals.

Cannon connector: *See XLR*.

cans: Headphones.

capacitance: See *impedance*.

capacitor (C): A device made up of two metallic plates separated by a dielectric (insulating material). Used to store electrical energy in the electrostatic field between the plates. It produces an *impedance* to an *alternating current*. Also called a *condenser*.

capo: The beginning of a piece of music. See *D.C.*

capstan: In a tape machine, the tape is moved by the effect of friction between a rotating motor-driven pillar, the capstan, and a *pinchwheel*, also called the *capstan idler*, that holds the recording tape securely against the capstan when a tape transport is in record or play mode. The *capstan motor* directly or indirectly drives the capstan and moves the tape past the heads. The capstan itself may be the extended shaft of the capstan motor.

capsule: In a microphone, the *diaphragm* or actual sound receptor, including, in various types of mics, the *moving coil*, *ribbon*, permanent magnet, or fixed *condenser* plate, and the housing in which these are mounted.

cardioid microphone: A *directional microphone* with an *acceptance angle* that is most sensitive to sounds coming from the front and sides, while rejecting sounds coming from the rear. Called cardioid because the *polar pattern* of the microphone is roughly heart-shaped. All directional mics have a *proximity effect*, whereby sound sources close to the mic will have an exaggerated low-frequency response. *Supercardioids* and *hypercardioids* are cardioids, but with a trade-off in the rear lobe. When using supercardioids and hypercardioids as sound reinforcement mics, it is important to note that the maximum rejection is not directly behind the mic as it is with a cardioid, but is off to the side between 110˚-126˚. However, a pair of hypercardioid microphones used as a stereo *X-Y pair* yields a very clean cardioid response pattern. See *pressure gradient*.

C

carrier: (1) A signal that is constant in amplitude or frequency and can be modulated by some other signal. The carrier itself does not transmit any information; all of the intelligence is in the modulation *sidebands,* which are in a band of frequencies on either side of the carrier frequency. Some signals, such as FM stereo, involve more than one carrier to encode the information, and the lower-frequency carrier is called a *subcarrier*. The subcarrier is mixed with parts of the audio signal and used to modulate the main carrier. In the receiver, the subcarrier is recovered by demodulation of the main carrier and then demodulated to recover its signal. See *amplitude modulation, frequency modulation, frequency modulation.* (2) In *FM synthesis,* the carrier is the operator at the bottom of a stack in an algorithm, through which the composite effect of other modulating operators connected to it is heard.

cartridge: (1) The needle assembly at the end of a phonograph tonearm. (2) In broadcasting, a short, looped tape usually used for recorded messages and / or commercials.

CAS: Cinema Audio Society. A Los Angeles-based organization of film and television recording personnel, founded in 1966.

Cat. 43: The Dolby Laboratories device that turns a Cat. No. 22 Dolby A-Type *noise reduction* card into a 4-band “noise fighter.” The precise frequencies of the bands are optimized for production sound problems and differ from those used in standard noise reduction applications. In 1991, Dolby formally introduced *SR*–type noise reduction, called the Cat. No. 430.

cathode: The cathode in any electronic component, such as a silicon diode or a vacuum tube, is the electrode normally connected to the negative voltage.

CAV: Constant Angular Velocity: In a mass storage device, such as a disk, CAV means that the disk assembly rotates at a constant speed, i.e., the data rate will increase for the tracks near the edge, and decrease for tracks near the center spindle. As opposed to *CLV*.

CCCC: See *LCRS*.

C.C.I.R.: Comité Consultatif International Radio. An international radio standards committee, whose recommended recording *pre-emphasis* and *post-emphasis* curves are standard on all recorders in most European and some other countries. The European analog to the *NAB*.

C

CD: The CD sampling rate is 44.1kHz and there are 32 bits per sample, so the data rate of the encoded analog data is 1.41Mbps, but the inclusion of parity, synch, and subcode bits raises the real data rate to 4.3Mbps. A CD will hold about 650 Mb, or about 74 minutes of stereo, 16–bit audio. The digital portion of the CD audio system is not stereophonic, but *sequential monaural*. See also *Control and Display signals*. Compare with *Direct Stream Digital*. The CD file format is defined by *ISO 9660*. For more information on CD standards, please see Sound on Sound, “Compact Disc Formats” by Mike Collins, January 1998. See also *SACD*. (Super Audio Compact Disc.) The CD specification is specified in “books,” each defining the standard for a particular type of CD:

Blue Book: CD Data (CD Extra)

The latest of the books to appear, this specifies the CD Extra format, designed to include CD-ROM data on a standard (audio, Red Book) CD. A CD Extra is actually a multisession CD, containing the audio tracks in its first session, followed by a data track in the second session, etc.

Red Book: System Description Compact Disc Digital Audio (CD-DA)

CD-DA (Digital Audio) Established in 1980 as the first of the books which defined consumer audio on CD. A Red Book CD may have up to 99 tracks; each track is divided into blocks of data called sectors; each sector contains, in addition to audio data, *EDC/ECC*, and 98 control bytes of *PQ subcodes*.

Yellow Book: System Description Compact Disc Read-Only Memory (CD–ROM)

CD-ROM (Read Only Memory). Yellow Book extends the Red Book specification by adding two new track types:

CD-ROM Mode 1: Storage of computer data
Mode 1 sectors include an improved ECC for Data.

CD-ROM Mode 2: Compressed audio, video, picture data
Mode 2 (Forms 1 & 2) CD-ROM/XA (Extended Architecture) is used to integrate computer data with compressed audio and/or video, including Photo CD and Karaoke CD.

White Book: System Description Compact Disc Bridge (CD-V)

Developed to cover the CD-V (Video) format, and supported by JVC, Matsushita, Philips, and Sony. These are a special kind of CD-ROM/XA bridge disc that allows the play of films and music videos on a dedicated CD-V player, or on a CD-i player equipped with a CD-V cartridge, or a computer with a CD-ROM/XA drive, an MPEG-1 decoder, and host playback application. The CD medium is modified to record video signals as well as digital stereo audio signals. The video information is recorded in analog form rather than digital. CD-V discs contain full-screen, full-motion video and CD-quality audio, and are independent of any broadcast standard, e.g., NTSC, PAL.

Green Book: System Description Compact Disc Read-Only Memory Extended Architecture (CD-ROM/XA)

C

Orange Book: System Description Compact Disc Systems Part II: (CD–WO)

A digital standard for recordable, write-once CD. The specification covers both disk-at-once and track-at-once. Many older CD-ROM drives cannot read multisession discs, however, these discs can be converted to a Red / Yellow / Green Book disc by adding a TOC, allowing the disc to be read by any CD player.

CD Single: A small CD-DA that can record 20 minutes of stereo music; it is 80mm in diameter.

CD-DA: CD-Direct Access. Software for writing audio data on hard disk onto a CD-R, for example, Toast™ or Gear.™ Such packages create an unfinished audio session in *disc-at-once* mode. Digital audio tracks must first be converted to a computer *file format* like .WAV or AIFF.

CD Extra: Formerly called *CD Plus*. A solution to *mixed-mode* CDs, CD Extra inverts the track structure of Mixed Mode by creating two separate sessions: first audio, then data. CD Extra is a part of the Blue Book standard, making Blue Book CDs fully compatible with the Red Book in that Blue Book (data) CDs can be safely used on audio players. The one problem with CD Extra format discs is that its *multisession* format makes it unusable by first-generation CD-ROM players.

CD-I: (CD Interactive) An extension of Yellow book, allowing discs to contain a mix of audio and video, plus data which the user can control interactively. CD-I discs use Mode 2, Form 1 and Mode 2, Form 2 tracks which, like CD-ROM / XA, enable computer data and compressed audio, video, or pictures to be played back at the same time. CD-I tracks cannot be played on normal CD-ROM drives, but specialized CD-I players can play audio CDs, CD+G, Photo CD, and with a CD-V cartridge, Karaoke CD or CD-V discs.

CD + MIDI: A type of CD which includes both audio data and MIDI data, i.e., a recording of both the sound of a musical performance, as well as the MIDI data used to generate it. This allows the user to “play with the performance” by choosing different patches, etc. This requires a MIDI Out socket on the CD player.

CD-R: See *CD*.

CD Single: A small CD that can record 20 minutes of stereo music; it is 80mm in diameter.

CD-V: CD Video. The CD medium modified to record video signals as well as digital stereo audio signals. The video information is recorded in analog form, rather than digital, like a small laser disc.

CEDAR: Computer Enhanced Digital Audio Restoration. A British-developed system for the restoration and preservation of old audio recordings. See also *NoNoise*.

C

cent: The smallest conventional unit of *pitch* deviation. One hundred cents equal one *half-step*. In an instrument, a cent is a term used in discussing pitch resolution; one cent is good, more than six cents is bad. See *half-step*.

center detent: A notched position in the range of a variable control, allowing the user to return the control to precisely that position, such as the midpoint between the left and right channels in a *balance* control. Use to denote the *flat* position on tone controls, etc.

center frequency: The frequency that is boosted or attenuated most by the operation of any *parametric equalizer* or other similar processing device or circuit. See *Q*.

center tap: In a *transformer*, the electrical midpoint of the windings, made accessible for external connection. Used, for example, in delivering power to *balanced line condenser* microphones. See *phantom power*, Appendix B.

C-format: The international standard *format* for professional 1" videotape equipment. Developed by Sony, and sometiines called *S-format* after that company's name. See *B-format*, *Betacam*, *VHS*.

CG: Computer Graphics.

chain: Also called *iron*. An integrated system composed of separate audio and/or video recording, processing, or playback circuits and/or devices which are used in conjunction with one another to produce one output result. See *B-chain*, *program chain*, *signal chain*, *side chain*.

change-over dots: See *projection*.

change-over projection: See *projection*.

channel: An independently processed or recorded signal. (1) An electrical signal path. In analog audio (such as a mixer), each channel consists of separate wired components. In the digital domain, channels may share wiring, kept separate through logical operations. (2) A system for independently addressing up to sixteen separate MIDI devices over a single MIDI cable. MIDI provides definitions for 16 channels which transmit not audio signals, but digital control signals for triggering synthesizers and other devices. MIDI data are associated with a particular channel by virtue of a *Channel ID Number* that is interwoven with other MIDI data being recorded. A *track* holds data that (depending on the sequencer) may or may not be restricted to one MIDI channel. MIDI's 16-channel limitation has been overcome by employing multiple independent MIDI ports that each route sixteen channels, offering the possibility of hundreds of channels. (3) The left or right signals of a *stereo* audio system, or the left, right, center, surround and/or subwoofer signals of a multichannel system, such as *LCRS* or *5.1*. (4) In film, A complete, self-sufficient recording setup. A *production channel* would include a recorder, mixer, microphones, headsets, etc. A *transfer channel* would include a 1/4" tape deck, a 35mm *mag recorder*, a *resolver*, and a monitoring system.

channel assignment matrix: In a recording console, the group of buttons or switches by which the signal from any input channel can be assigned to one or more *busses*, and thereby be sent to one or more tracks of the multitrack recorder.

C

channel bit rate: The actual bits being read from a digital medium are greater than the number strictly required to encode the audio signal. This is because of *ECC* and synchronization bits, etc. For example, with a CD, the audio bit rate is 1.41Mbps, but the channel bit rate is actually three times as high, 4.32Mbps.

channel insert: An *insertion point* in a mixer channel which opens up the signal path and allows an *outboard* device to be inserted *in-line*. The *output point* (the place where the signal is routed to the outboard device) is called the *channel insert send*, and the place where the effected comes back into the mixer is called the *channel insert return*. The actual point at which the channel signal path is broken with the insert connection is not standard among all consoles. Some are between the preamp and equalizer sections, some after the equalizer, but before the fader, and some are post-fader. Some are switchable with an internal jumper or other modification. If, for example, the channel insert send is post-fader, the fader setting will affect the action of a compressor that is inserted into the channel's signal path. On the other hand, a post-fader insert is good when it is desirable to send a single channel's signal direct to a tape track, making the fader into a convenient record-level control. See *normalled connection*.

channel message: A class of *MIDI messages* which only affect devices on a *MIDI network* set to a particular channel, i.e., all non-*system messages*. Channel messages may be of either Channel Mode or Channel Voice type. See *MIDI*.

channel mode: See *MIDI mode*.

channel path: The record section of the signal chain in a *mixer*. See also *monitor path*.

channel pressure: A type of MIDI *channel message* that is applied equally to all of the notes on a given channel; the opposite of *poly pressure*, in which each MIDI note has its own pressure value. Also called *aftertouch*, channel pressure is generated on keyboard instruments by pressing down on a key or keys while they are resting on the keybed. Also called *channel key pressure*.

channel separation: Channel separation refers to the amount of *crosstalk* between the channels of a stereo system. It is the inverse of interchannel crosstalk, as measured in decibels. A small amount of crosstalk is equivalent to a large channel separation.

channel strip: One of multiple identical sections in a *mixer* from the *mic preamp* and *phantom power* (if present) to the *bus* outputs, and typically includes the input *pad*, EQ, and signal routing, including *pan*, *effect sends* and *effect returns*, and main channel *fader*, and optionally an automation interface. There is one channel strip per mixer input.

Channel Voice: A classification of MIDI *channel message* relating specifically to a musical performance, where features of the performance (notes, articulation, etc.) are individually described by a unique message. Channel Voice messages include Note On, Note Off, Polyphonic Key Pressure, Channel Pressure, Program Change, Pitch-bend, and Controller Change. These messages all include a specific channel number, allowing similar messages to address different devices on the same *MIDI network*. The message will only be implemented by a receiving device whose channel number matches that of the message.

channelize: See *MIDI mapping*.

C

characteristic impedance: See *termination*.

charge (C): Charge is a measure of the quantity of electricity and its unit is the *coulomb*. In an electrical circuit, charge consists of negative charges, or electrons. A positive charge can be thought of as simply an absence, or deficiency of electrons. Charge is what is moving in an electric *current*. See *ampere*.

chart: (1) A musical score or arrangement. The term is used both to designate the conductor's *full score*, or any *band part*. (2) A list of current hit singles or albums.

chase: (1) The process whereby a slave device attempts to *sync* to a *master clock*. (2) In MIDI parlance, to chase means, upon playback, to look backward to earlier MIDI events to see if there were any program or channel change messages prior to the playback point which would affect playback. See *controller chasing*.

chase-lock: A type of controller for a video or audio recorder that will listen to the *SMPTE timecode* signal from the *master clock* device and will adjust its own speed to find the correct time and then will lock into synchronization with the external timecode. Unlike *sync-lock*, chase-lock controllers respond to changes in timecode sequence.

chasing: See *controller chasing*.

chassis ground: The practice of connecting the signal ground of a device to the rack rails or other common grounding location on a multi-component electronic system.

chatter: When the input signal level to a noise gate hovers near the threshold level, the gate may be unsure if it should be open or closed. It may rapidly open and close, resulting in the audio cutting in and out; this is known as the gate "chattering." To correct this problem, adjust the threshold setting to be slightly lower or higher.

checksum: A number derived from arithmetical actions on data, used to check that data has not been corrupted after transmission or recording and replay.

chip: (1) In vinyl record production, the thin thread of acetate lacquer that is carved out of the master disc by the cutting stylus. Also called *swarf*. (2) A slang term for *integrated circuit*.

chirping: An effect caused by the overuse of *single-ended noise reduction* systems whereby the low-level signals take on an electronic, "ringing" character, known as chirping. If the signal is very noisy, the *noise floor* itself begins to sound chirpy, which can be more annoying than the original, broad-spectrum, noise.

chord: The playing of multiple notes simultaneously. The opposite of an *arpeggio*. See *inversion*.

chorus: (1) A regularly repeated section of a song or other musical composition. (2) A group of singers, also called a *choir*.

C

chorusing: A type of audio effect in which a *delayed* (30-40ms) or *detuned* copy of a signal is mixed with the original signal. The mixing process changes the relative strengths and *phase* relationships of the *overtones* to create a more complex sound. See *ADT, double-tracking*. The mixture becomes extremely complex as the relative phases of the signals cause partial cancellation and reinforcement over a broad frequency spectrum. The simplest way to achieve chorusing is to detune one synthesizer *oscillator* from another to produce a slow *beating* between them. See *comb filter*.

chromatic: Pertaining to the full twelve-note *scale*, as opposed to the eight-note *diatonic* scale.

Cinema Digital Sound (CDS): A new system of digitally recording motion picture sound format introduced by the Optical Radiation Corporation, a division of Kodak, in 1990, for the film "Dick Tracy" for digital sound on 35mm or 70mm film *formats* via a laser beam, which reportedly combines the dynamic and frequency ranges and low distortion of the CD on six discrete channels. Five channels encompass the full audio bandwidth and the sixth is designated a subwoofer channel, containing only the lowest frequencies. The CDS-encoded film is capable of being shown with conventional stereo optical sound, but requires a special sound system to reproduce the six channels digitally. First used in 1990, this format lasted only two years and is now obsolete. See *AC-3, 5.1*.

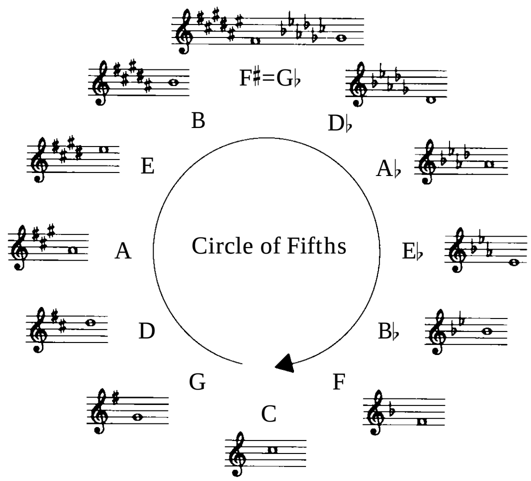
CinemaScope: The trademark of a widescreen camera system developed by Twentieth Century Fox, the first true stereophonic motion picture sound system which had the soundtracks on the same film with the picture. First used in 1953, CinemaScope was responsible for popularizing the *anamorphic* film *format*.

Cinerama: A widescreen system comprising three 35mm cameras / projectors running in *interlock* with 7-track *mag film*.

CIRC: Cross Interleaving Reed-Solomon Code. The combined error detection and correction scheme used in CDs. See *interleaving*.

C

Circle of Fifths: Also known as the *Cycle of Fifths*. A way of thinking of the twelve major and minor keys as a circle, arranged in steps of a fifth, which can be read in either direction. Starting from Cmaj and proceeding clockwise, the key signature of each new key gains one sharp until Fmaj is reached. At that point, F♯ becomes G♭maj and the cycle continues, removing a flat at each step until back to C. If one goes counter-clockwise, the circle is a series of perfect cadences, with each new tonic key becoming the dominant of the next. For this reason, the Circle of Fifths is often used for *modulation*(3), especially to or back from a remote key, i.e., a key on the far side of the circle.



circuit: A complete path that allows electrical *current* from one terminal of a voltage source to the other terminal.

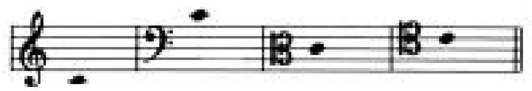
circumaural: A headset with a large cushion which surrounds the ear to exclude external noise, unlike *supraaural* or *intraaural* designs.

CIT: See *SDMI*.

clangorous: Containing *partials* that are not part of the natural *harmonic series*, i.e., partials which are not whole-number multiples of the *fundamental frequency*. Clangorous tones often sound bell-like.

clef: In written music, a symbol placed at the beginning of the *stave* which assigns a *pitch* to a specific line on the stave, and by inference, to all of the other lines and spaces. Three clef symbols are commonly used, derived from the medieval forms of the letters G (𝄞), F(𝄢), and C(𝄡).

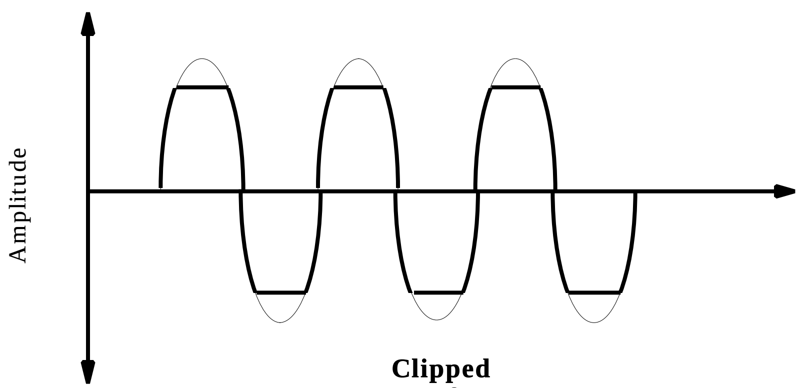
C



Four Clefs Showing the Position of Middle C

click track: A click track records a series of clicks, like a metronome, on one channel of a multitrack tape recorder or one channel on a MIDI sequencer. The click track is used to synchronize the recording of subsequent tracks by playing it back via headphones to the musicians while they are *overdubbing* the added tracks.

clipping: A *distortion* caused by cutting off the peaks of audio signals. Clipping usually occurs in an amplifier when its input signal is too high or when the volume control is turned up too high. A clipped waveform contains a great deal of *harmonic distortion* and sounds very rough and harsh. *Hard clipping* results in very sharp edges on the waveform, producing the maximum amount of high-harmonic content. *Soft clipping* produces rounded edges of the clipped waveform and is much less grating on the ears and tweeters than hard clipping as it contains much less very high-frequency energy. Different amplifiers produce different clipping effects; tube amps often produce soft clipping. See *full code, digital black.*



Clipped Waveform

clock: (1) Any of several types of timing control devices, or the periodic signals that they generate. Clock pulses are usually derived from crystal-controlled oscillators. See also *MIDI Clock, master clock*. (2) In recording sessions for jingles or film scores, a stopwatch.

clock noise: An artifact of *digital-to-analog conversion* that creates staircase-like changes in voltage produced by the converter. Most clock noise is caused by shifts in the *zero-crossing* times. See *quantization noise, reconstruction filter*.

clock reference: See *master clock.*

clock resolution: The precision (measured in *ppq*) with which a sequencer can encode time-based information. A sequencer's internal clock is always set to some ppq value, and this setting is one of the main factors that determine how precisely the sequencer can record time-dependent information. The actual clock speed is usually determined by the *bpm* setting. See *MIDI clock.*

C

closed loop: A closed loop system is one which modifies its behavior based on the difference behavior between an output variable and a set point, i.e., it relies on *feedback* to determine its output. The opposite of *open loop*, an example of a closed loop system is a household thermostat. In audio, the main use for closed loop systems is in *power amplifier* output stages. See *bootstrap.*

cloth track: See *Foley*.

CLV: Constant Linear Velocity. As opposed to *CAV*, a mass storage system that has a disk whose speed varies to keep the data rate at the read / write head constant, regardless of the location of the data on the disk. The *CD* standard specifies CLV format.

CMRR: Common Mode Rejection Ratio. In a *balanced line* connection, the common mode is the noise whose phase is common on both lines. The degree of attenuation of the common mode by a *differential amplifier* is called the *rejection ratio*, and like other level-modulating devices, the output change is measured in dB.

cocktail party effect: The phenomenon of human aural discrimination among sounds of equal *loudness*, e.g., the ability to hear one conversation out of many at a party. Related to auditory *masking*.

coda: Musical symbol (𝄌) which references a section of music which is to be repeated from or repeated to, as in "D.C. al Coda," meaning "play from the beginning to the Coda." A coda is a musical passage which gives a sense of completion to a movement or work, possibly an extended *cadence* or a substantial passage. A *codetta* is a short musical passage which links the subject and answer in a fugue, or at the end of the first passage of a sonata.

codec: COderDECoder. A device that digitizes an input waveform, eliminating redundant information, reducing the number of bits needed to carry the same data, then decoding the data at the receiving end, hopefully with a high degree of sonic transparency. See *PCM*, *PWM*.

code window: A display of the *SMPTE timecode* numbers, usually corresponding to each frame of picture viewed on a monitor. These numbers appear in a window that replaces a portion of the program image, usually at the bottom right of the screen. Depending on the equipment used, the timecode data in the window can be either generated in *real-time* from the source of the SMPTE timecode data, or have been recorded as a permanent part of the picture on a particular copy of a pre-recorded program. *BITC* is generally recorded onto copies of footage that will be used for *off-line* editing. The source of the SMPTE timecode data in this latter case is the *sync track* or *control track* of the original video footage.

coding: The process of altering the form of a signal, such as from analog to digital. Or, the coding may be used to allow a transformation of the signal not possible in its original form such as in *noise reduction*. Or, coding may be used to take advantage of effects inherent in the coding process itself, such as in *PCM* which allows recording and playback with low noise. The complementary stage to coding is *decoding*, a process which attempts to reconstruct the original signal, i.e., from digital back to analog for use with loudspeakers. See *codec.*

coercivity: The magnetic field strength required to bring any specific type of recording tape, when fully saturated, to complete erasure. Measured in Oersteds, and abbreviated H_c.

C

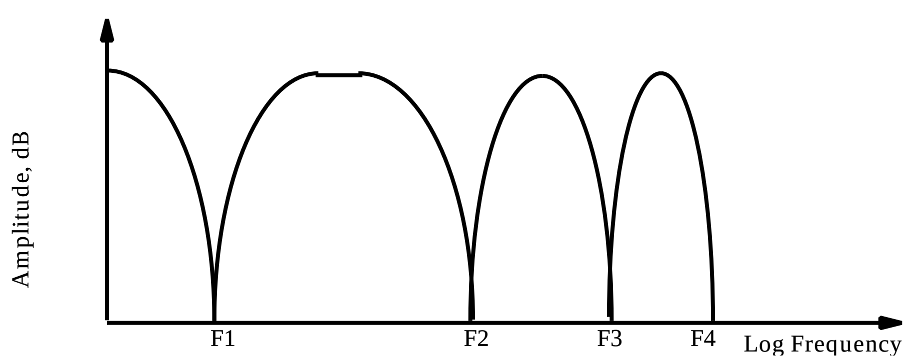
coherence: The *polarity* relationship between two complex sounds or signals being combined, measured at any instant. Total *phase* coherence indicates complete phase alignment, or full signal reinforcement. Incoherence, to any degree, designates a partial to complete *phase difference*, producing partial to total *phase cancellation*. The opposite of *incoherent*. See also *isochronous*.

coincident pair: Also known as an *X-Y* or *XY* pair, this is a microphone configuration which commonly uses two *cardioid* or *figure-eight* microphones mounted at right angles to one another, the latter preferred only for special applications. This is called a coincident pair because the two microphones are mounted as closely as possible to each other so that the sound being captured arrives at both microphones at exactly the same time, regardless of the direction of the source. All coincident configurations have to use *directional* microphones in order to create the necessary level differences between the two channels of the stereo system; *omnidirectional* microphones do not produce level differences proportional to the angle of *incident sound*. This technique is favored by many broadcast applications because of good *mono compatibility*. Recording with a coincident pair is called *XY recording* in the US and the UK, *AB recording* throughout Europe, and also *crossed pairs*, or *normal stereo*. In the US, "AB recording" means a *spaced pair*. See also *Blumlein pair*, *near-coincident pair*. Contrast with *spaced pair*.

coloration: Coloration is the term for subtle *distortion* which results in a change in the *timbre* of a sound without that sound being otherwise noticeably distorted, such as a smearing type of distortion produced by *intermodulation distortion*. More prevalent at high audio frequencies.

color burst: See *video black*.

comb filter: A type of *notch filter* which produces a series of very deep notches, or dips, in its frequency response. The spacing of the notches along the frequency axis is at multiples of the lowest frequency notch. A comb filter is produced when a signal is time-delayed and added to itself. Frequencies where the time delay is one-half the period and multiples of these frequencies are cancelled when the signals are combined because they have opposite *polarity*, usually used to filter out 60Hz hum and its associated harmonics. If the signals are of equal strength, the cancellation is perfect and the notches are infinitely deep on a decibel scale. See *common mode*. Also called *timbral interference cues*.



Comb Filter

C

combining amplifier: An amplifier, also called a *summing amplifier*, that combines two or more signals prior to sending them to a single audio bus, signal processor, tape recorder track, or other destination. For example, on a *mixer* the *aux send* controls on all channels feed a combining amplifier whose output can be routed to a reverb system, cue or headphone amp, the monitor amplifier, etc. There are also devices which are *active* combining amplifiers, called an *ACA*, as well as passive *combining networks*,

combining network: A typically *passive* network in which two or more signals are combined before being sent to a single bus, signal processor, or other destination.

combo: A combination of loudspeaker(s) and amplifier in one unit, usually portable. Used by guitarists, keyboard players, etc. for stage amplification.

Comma of Pythagoras: See *diatonic comma*.

commag: A technical term for *composite* magnetic print.

common mode: Common mode refers to equal voltages induced in the two wires of a signal-carrying pair. In a *balanced line* circuit, the signal voltages are of opposite *polarity* in the two signal wires. Any voltage which appears with the same polarity on each wire is called a common-mode voltage. Usually noise, such as a 60Hz *hum*, is induced in audio cables equally and in the same direction, and so is a common-mode voltage. If the signal is connected to a *differential amplifier* input, the common-mode voltages will cancel, while the signal voltages, being of opposite polarity on each input terminal, will add together. This is the reason why balanced lines are less prone to induced noise from external influences. See *CMRR*.

common mode rejection: The measurement of how well a *balanced* circuit rejects a *common mode* signal. See also *CMRR*.

comopt: A technical term for *composite* optical print.

compander: Short for compressor/expander. A compander is a device for *noise reduction* in audio devices such as tape recorders. The compander will reduce the *dynamic range* of the signal before sending it to be recorded. The compression makes the softer passages louder so the dynamic range recorded on the tape is less than it would be if it were not compressed. Then, on playback of the tape, the signal is expanded; that is, the softer passages, which are too loud on the tape, are reduced in volume to match the original signal, restoring its dynamics. In the expansion, which is similar to a fast-acting *AVC*, the noise introduced by the tape recording process is effectively reduced because the music, when loud, masks the noise, and during the soft passages, the volume is turned down, making the noise comparatively softer. Digital companding allows a device to achieve greater apparent dynamic range with a lower *bit depth*. See *dbx, Dolby noise reduction*.

companding converter: An *A/D-D/A* pair which uses a non-linear scale, i.e., one that has larger steps towards peak amplitude and smaller steps towards minimum amplitude. This scale increases the ability of the converter to resolve small changes in low amplitude signals, reducing *distortion*, but with the penalty of increased *noise*. The overall effect is that of a compressed analog input signal and a resulting expanded digital output. See *compander*.

C

compatibility: (1) The degree to which different pieces of equipment can be used together or are interchangeable, e.g., whether a tape recorded with one type of *NR* can be replayed on a tape player equipped with another type of NR. (2) See *mono compatibility*.

complementary: Any pair of audio signal processing procedures which perform two equal and opposite processes on the signal, one before recording, the other after playback. *Noise reduction* and tape recorder *pre*- and *post-emphasis* are examples. See *encoding*.

completely filled: See *4-track*.

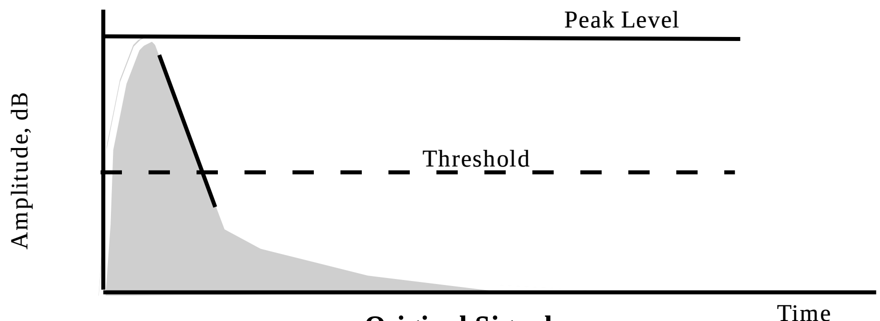
composite equalization: The overall *frequency response* modification produced when a signal passes through more than one equalizing circuit in the same device, or through several equalizers in a series.

composite print: Film print that contains a *sountrack*.

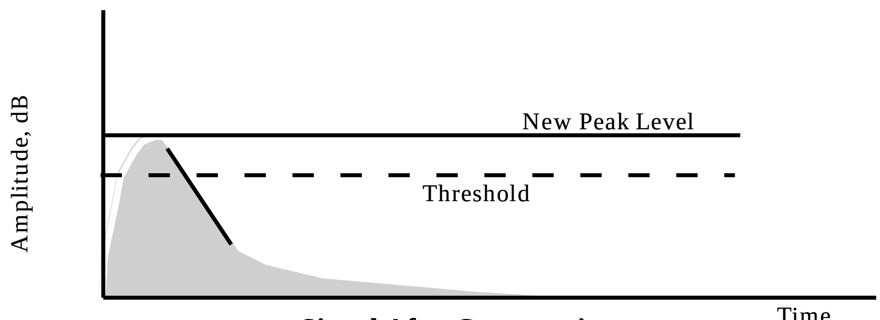
compound time: See *time signature*.

C

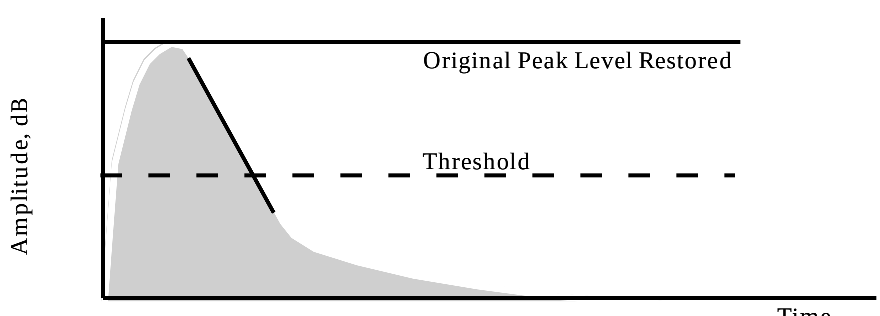
compression: (1) The process of reducing the *dynamic range* of an audio signal by reducing the peaks so as to be able to boost the low levels. For every dB of compression applied, the *S/N ratio* is worsened by 1dB, assuming that the make-up gain is set so that the maximum levels of the compressed and uncompressed signals are the same, as the quieter parts of the original signal, plus any noise contained in these regions, will be raised in level. (2) A *dynamic-range* problem in loudspeakers caused by nonlinearity under conditions of high input power levels. At very high levels, the acoustic output increases more slowly or ceases to increase altogether as the input power increases, producing *nonlinear distortion*, i.e., a *frequency response* curve very different for very high levels. (3) Data compression used on digital audio files is a process *ADPCM*, *MACE*, for example. (4) The opposite of *rarefaction* whereby a quantity of data is reduced in order to occupy less storage space. See *ATRAC*.



Original Signal



Signal After Compression



Signal After Make-Up Gain

C

compression driver: A specialized mid- or high-frequency speaker consisting of a small *diaphragm* and *voice coil* coupled to a large magnet structure. The unit is mounted to a horn which acoustically matches the *impedance* of the driver to the impedance of the air and shapes the signal. Expensive due to the precise tolerances required, compression drivers are substantially more efficient than traditional *direct-radiating* cone speakers.

compression ratio: (1) The ratio of the dB change from input level to output level effected by a *compressor*, once the threshold has been exceeded. (2) In data *compression*, the ratio of the number of bytes of uncompressed to compressed data, an indication of the space-saving efficiency of the compression algorithm.

compressor/limiter: A device for reducing the effective *dynamic range* of an input signal by preventing it from rapidly exceeding or falling below a selected amplitude threshold. The first part of a *compander*, it is used to make loud parts of a signal softer and soft parts louder. Beyond the threshold, the ratio of the signal's input level to its output level (e.g., 2:1, 4:1) is user-selectable. A compressor is commonly used to keep mic levels within an acceptable range, but because it can slow a signal's rate of decay below the threshold, compressors are also used to add sustain to instruments such as electric guitar and bass. The *limiter* acts like a compressor, but operates only at the top end of the dynamic range. The limiter has a faster attack time (1µs to 1ms) than the compressor alone (1ms to 10ms). A compressor/limiter is inserted between the outputs of a MIDI soundcard, synthesizer, or mixer and the inputs of the mixdown deck. See *hard knee compression*, *soft knee compression*.

Comtek: (1) A Salt Lake City-based company that makes portable wireless transmitters and receivers. (2) The generic name for wireless headphone feeds to directors and for wireless timecode feeds to slates.

concert pitch: Established by *ISO* in 1955, the agreed reference frequency of 440Hz, for the note A above middle-C, notated A=440.

condenser microphone: A condenser, or *capacitor*, mic capsule has a conductive *diaphragm* and a metal backplate placed very close to the diaphragm. They are charged with static electricity to form two plates of a capacitor. When sound waves strike the diaphragm, it vibrates, varying the spacing between the plates. In turn, this varies the capacitance and makes a signal analogous to the incoming sound waves. There are two types of condenser mics: the true condenser and the *electret* condenser. In the former, the diaphragm and backplate are charged with a voltage from a circuit. In the latter, the diaphragm and backplate are charged by an electret material, which is in the diaphragm or on the backplate. All true condenser mics need a power supply to operate, such as a battery or *phantom power*. In general, condensers have a smooth, detailed sound with a wide, flat *frequency response*--usually up to 15kHz-20kHz, useful for cymbals or instruments that need a detailed sound, such as acoustic guitar, strings, piano, or voice. Condenser mics tend to be more expensive and fragile than *dynamic* microphones. Note that *omnidirectional* condenser mics have deeper lows than *cardioid* condensers, making the former a good choice for pipe organs and bass drum. See also *boundary microphone*.

conductance: The reciprocal of *resistance*, or electric *current* divided by *voltage*. The traditional unit of conductance is the *mho* (*ohm* spelled backwards). See also *impedance*.

C

conductivity: A material which exhibits efficient thermal or electrical transference through itself is said to have a high conductivity. Conductivity in a material is rated as its *resistivity*, the inverse of conductivity, in *ohms* per meter. Metals have a high conductivity owing to the large number of free electrons in metal atoms which efficiently transfer the current or heat from one part of the material to another. An *insulator*, on the other hand, is a material with few free electrons, and hence does not readily pass heat or current.

cone: The vibrating *diaphragm* of a *dynamic* or *moving coil* loudspeaker, usually made of paper and shaped roughly like a cone.

conform: (1) To re-edit sound *stems* to match a new version of the picture edit, which is the final matching of all music, dialog and / or special effects to the video image. This may involve synchronization, editing, and re-recording one or more of the components of the final sound / video mix. (2) To assemble sound elements from their original sources to match their location in a picture edit, often with the assistance of an *EDL*.

console: See *mixer*.

consonant: Literally “sounding together.” Musical tones that are consonant sound harmonious or in tune when sounded together rather than discordant or harsh. Musical intervals composed of tones that have relatively simple frequency ratios are more consonant than ones with more complex ratios. The most consonant interval is considered to be the *octave*, which has a frequency ratio of 2:1.

contact: See *wrap*.

contact enhancer: A chemical compound which, when applied to plugs, sockets, or other metallic electrical connection, improves the electrical conductivity between the metal surfaces, making a better, less noisy contact.

contact microphone: A mic that is physically attached to the body of an instrument or other sound source. It is primarily the vibration of the contact microphone’s body itself that is the *transducer*. By comparison, other microphones contain an internal *diaphragm* or membrane that vibrates in response to sound carried to it through the air, while the *capsule* of the microphone itself remains motionless. See also *bug*, *piezo pick-up*.

container: Film sound slang for Dolby Laboratories’ peak *limiter* designed specifically for controlling the dynamics of program material during *SVA* printmastering.

Continue: A MIDI Real-Time *system message*, correctly written Song Continue.

continuous controller: A type of MIDI *channel message* that allows dynamic, *real-time* control changes to be made in notes that are currently sounding. There are 128 possible continuous controllers on each of 16 MIDI channels, and each of these controller types can have any data value between 0 and 127. Modulation (such as pan or volume) is an example of a true MIDI continuous controller. Continuous controller 1 is always the modulation wheel; controller 7 is the instrument’s main volume. See *controller*.

C

continuous sync: A software feature where the *DAW* will create a new clock based on incoming *SMPTE timecode* to enable recording to the DAW from an *ATR*. The result is that the sample rate of the DAW will vary continuously, effectively speeding up and slowing down to track the timecode variations. Continuous sync requires dedicated hardware, and may not be available on all DAWS. For example, ProTools™ has a feature for continuous sync which is necessary when syncing continuously to an ATR while recording on any digital machine as the SMPTE timecode-based clock is not guaranteed to be at the precise sampling rate. The ProTools Slave Driver™ does the sample rate conversion in the ProTools hardware so that the audio quality of the digital data isn't compromised.

contour generator: See *envelope generator*.

Control and Display signals (C&D): Also called *PQ codes*. In the *CD* format, eight additional bits are added to each *frame* of audio data; this means that a byte of information is available from the disc every 136 µs. Each bit in the added byte is given a one-letter name, P-W. Thus, eight separate subcodes can be recorded on and recovered from the CD. So far only P and Q are used: the P-code is used for the pause signal between musical tracks and at the end of the last track, and the Q-code tells the player if the recording is two- or four-channel (no quadraphonic CD player is yet available). The Q-code also contains timing information about the tracks and identifies the country of origin and date of the recording. No standard has been defined for the use of the other six subcodes.

control module: The part of a synthesizer that tells the sound generators and *controllers* what to do to make a given note. These modules include *envelope generators*, *LFO*s, the keyboard itself, and the *modulation* and *pitch-bend* wheels. These allow control of some aspects of a synthesizer's sound by sending signals to the sound generators and modifiers telling them now to behave. For instance, the keyboard sends a signal to the oscillator telling it what frequency to play. Also called *modulation modules*.

control panel: A file which becomes a part of the Mac's system software, giving the user either control over or adding functionality to various aspects of the operating system, peripherals, or applications. See also *extension*.

control track: (1) One track of a multitrack magnetic tape recorder used for recording special signals that provide control information to the recording console during automated mixdown. (2) A dedicated track prerecorded with a *pilot tone*, used on video tape which marks the start of each video frame in order to resolve playback speed by controlling and synchronizing the video frames. Can be used to count time for editing, but is prone to slip and lose count during winding. In 1" and ¾" formats, *SMPTE timecode* information is sent to a separate *address track* which creates confusion about the names of both of the tracks. See *blacking*, *sync-lock*.

control voltage: A voltage, usually varying, used in synthesizers to control various parameters of the signal being produced. Control voltages are used for *envelope* control, *pitch* control, and *filter bandpass* and *rolloff frequency* control, etc. Suitable control voltages can be generated in various ways, one of the most straightforward of which is by a standard keyboard. See *VCA*, *VCF*, *VCO*.

controller: (1) Any device, for example, a keyboard, wind synth controller, or *pitch-bend* lever, capable of modulating a sound by altering the action of some other device. (2) Any of the defined MIDI data types used for controlling the on-going quality of a sustaining tone via a *controller message*. In many synthesizers, the controller data category is more loosely defined to include pitch-bend and *aftertouch* data. See *continuous controller*.

controller change: A Channel Voice message which allows for musical effects such as *vibrato* or *sustain* on currently active voices.

controller chasing: A sequencer feature whereby whenever playback is requested, the sequencer looks back for the most recent controller, *pitch-bend*, *aftertouch*, and similar parameters and sets everything accordingly so that playback started in the middle of a song replays correctly.

convolution: (1) In any *linear* system or device, the output signal is a function of the input signal and the characteristics of the device. The interaction between the input and the device is described by a mathematical infinite integral called convolution. The output is the input convolved with the *impulse response* of the device. The *spectrum* of the output of a device is simply the spectrum of the input multiplied by the *frequency response* of the device via the *FFT*. (2) The *modulation* of one audio file by another. For example, the use of a hand-clap echo sample could be convolved with a guitar chord sample to produce an echo effect which sounds like it was produced by the guitar.

copy editing: The process of re-recording or copying selected extracts from original sound or video recordings and rearranging their order as they are copied, so that the copy will have all the desired segments in the correct order. This copy is called an *assembly*, and will generally need fine editing in order to meet timing or other production requirements.

corner frequency: See *rolloff frequency*.

correlated noise: See *distortion*.

cosine microphone: See *figure-eight microphone*.

cottage loaf microphone: See *supercardioid* or *hypercardioid microphone*. UK usage.

coulomb: The coulomb is the unit of electric *charge* (C), and is the quantity of electricity transferred in one second by a *current* of one *ampere*.

counts: A slang term for footage numbers or *cues* for specific events in a film or videotape. Also called *footage counts*. See *feet/frames*.

coupling: The process of or means by which energy is transferred from one system or medium to another. For example, the coupling of acoustic energy from a loudspeaker to the surrounding air.

cps: (1) centimeters per second. The speed of movement of tape past a tape read / record head, also denominated in *ips*, inches per second. (2) cycles per second, or *Hertz*.

C.R.: Con Repeats. As in, “Play from the beginning with repeats” is written, “D.C. (C.R.).”

C

CRC: Cyclic Redundancy Check. A system of recording a *checksum* number along with data in order to detect, and in some cases, correct any corruption of the data. See *ECC*.

crescendo: A gradual increase in *loudness* of a musical sound.

crest factor: The ratio between the average amplitude as shown on a *VU meter* and the instantaneous amplitude as shown on a *peak meter*. The human ear is very sensitive to this difference.

critical distance: The distance from a loudspeaker where the *direct sound* is equal in intensity to the *reverberant* sound. See also *free-field*.

critical frequency: See *rolloff frequency*.

crossed pairs: See *coincident pair*.

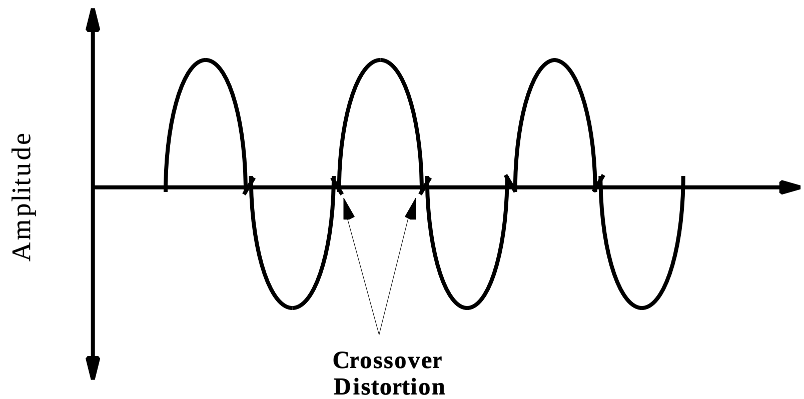
cross-fade: A *velocity* threshold effect in a synthesizer in which one sound is triggered at low velocities and another at high velocities, with a fade-out/fade-in transition between the two. If the transition is abrupt rather than gradual, the effect is called *cross-switching* rather than cross-fading. Cross-fading can also be initiated from a footswitch, *LFO*, or some other controller.

cross-fade looping: A sample-editing feature found in many samplers and most sample-editing software, in which some portion of the data at the beginning of a *loop* is mixed with some portion of the data at the end of the same loop, so as to produce a smoother transition between the end of the loop and the beginning of the loop replay.

cross-interleaving: See *interleaving*.

cross-mod: Cross-modulation test. A means of determining correct exposure on a track negative to result in minimum distortion on a positive print. Tests are conducted to determine the relationship of specific optical cameras to specific laboratories.

crossover distortion: A type of distortion present in some amplifiers which increases for low-level signals. In many amplifiers, the output devices are connected so that one of them is active during the positive half of the waveform, an the other one is active for the negative half. There is a region near zero current where the signal is transferred from one to the other. If this is not done smoothly, there will result a small discontinuity in the output waveform. This discontinuity causes higher-order *harmonic distortion*, and being constant in value, is more noticeable with low-level signals than with stronger ones. See *crossover frequency*.



crossover frequency: The frequency above and below which an audio signal is divided into two bands, each of which is directed to a separate destination. Precisely, the frequency at which each of the two bands is attenuated 3dB by the *crossover network*.

crossover network: A dividing network that splits a full-spectrum signal into two or more frequency bands and routes them to feed the various components of a speaker system. *Passive crossover* is a speaker design whereby multiple speakers are in a single enclosure and sound frequencies are separated and sent to be driven by the appropriate speaker. A two-way *biamp* monitoring system has a high-frequency loudspeaker (tweeter) and a low-frequency speaker (woofer). A three-way *triamp* system has a third speaker to reproduce midrange frequencies. *Active crossovers* divide a *line-level* output signal from a mixer or other sound source and route the resulting signals to individual amplifiers optimized for the different speaker components.

cross-switching: A *velocity* threshold effect in a synthesizer in which one sound is triggered at low velocities and another at high velocities, with an abrupt transition between the two. If the transition is smooth rather than abrupt, the effect is called *cross-fading* rather than cross-switching. Cross-switching can also be initiated from a footswitch, *LFO*, or some other controller. Also called *velocity-switching.*

crosstalk: In *multichannel* audio transmission systems, such as tape recorders, record players, or telephone lines, a signal leaking from one channel to one or more of the others is called crosstalk. Also known as *bleeding*. See *channel separation*.

C

crystal sync: A system for generating a sync signal that will ensure proper synchronization of film footage and its corresponding *sync sound* without using any *sync reference*, such as 60Hz AC line frequency. Two piezoelectric crystals, each tuned to the same high frequency, are installed in the camera and recorder. The crystal in the camera precisely controls the motor speed during shooting. The crystal in the recorder produces a *pilot tone* that is recorded on tape in the same way which the camera motor speed would be recorded through the conventional sync cable. Because the two crystals are tuned identically, the *dailies*, when synched with the magnetic film copy of the original sound takes, will maintain perfect sync once the *slate* marks are aligned. Crystal sync generators are also installed in portable video cameras and *VTRs*. See *Nagra*, *neo-pilot*.

C-Type: See *Spectral Recording*, *noise reduction*.

cue: (1) (*noun*) A piece of music for a specific scene or event in a film. A specific part of a *film soundtrack* which correlates to a visual event in the film is called a *cue point* or *hit point*. (2) (*verb*) To position a sound source to activate at a specific time. See *Real-Time MTC Cueing*. *spot*.

cue box: A wall-mounted or movable box that receives one or more *monitor mixes* from the recording console, and that has jacks to plug in several sets of headphones used to send the *cue mix* to singers or instrumentalists in overdubbing, a narrator, or other studio talent. Also called a *headphone box*. See *cue system*.

cue line: A line drawn on the *workprint*, meant to be seen during projection in post-dubbing or scoring, which gives the actor or conductor a visual cue to begin.

cue list/sheet: A list of the footages and frames, beginning with 00:00, at which specific shots begin and end. Used by the *re-recording* mixer who needs to know which sounds or music must be played as the final mix proceeds. See also *edit decision list*, *feet/frames*.

cue mix: The blend of live inputs and/or previously recorded tracks sent by the mixing engineer to the headphones of performers playing or singing in the studio. Also called the *headphone mix*.

cue mode: A tape machine operating mode in which the tape lifters are defeated while the playback electronics remain operative. Used most often during editing, thus also called *edit mode*.

cue point: See *Real-Time MTC Cueing*.

cue sheet: A track sheet for mixing that gives locations of edited sounds on a track-by-track bases, either in film footages or in timecode numbers. See *binky*.

cue system: The entire electronic circuitry contained within the recording console that allows the engineer to adjustably feed sound from any input module to the *cue mix*, then out to the musicians' or singers' headphones via the cue amp and *cue boxes*.

cue track: A track of recorded music and/or clicks which are sent over headphones to the musicians and/or singers to assist them in *overdubbing* additional music/vocals. If the track is simply tempo clicks, it is known as a *click track*.

C

cue-up: To locate a desired point, at which a specific sound event happens on a reel of tape, and to position that point just ahead of the *playback head* on a tape recorder. When playback begins, the desired sound will be heard immediately.

current (I): The flow of electrical *charge* measured in *amperes*.

cut: (1) (*verb*) To attenuate amplitude of a signal or particular frequency *band*; the opposite of *boost*. (2) To produce the master for a vinyl LP. (3) In film, the instruction used to terminate filming. (4) (*noun*) A musical selection on a record, tape, or CD, or a particular edited version of film.

cut-and-paste: On a hard-disk audio editing system, a term used to denote the ability of an audio editing program to move and / or copy sections of the recorded audio to a new location in the *track* or to other tracks.

cut effects: Sound effects that are taken from a sound library and edited, usually as opposed to recorded *Foley* effects. See *pull*, *M&E*.

cutoff frequency: See *rolloff frequency*.

cut switch: A switch or button that mutes an audio signal on a *mixer*.

cutter: Sound editor.

cut track: An edited track of a film soundtrack which is ready to be used either as a track in a *premix* (music or effects), or as a (dialog) track in the final mix.

C-weighting: Unlike *A-weighting*, C-weighting measures frequencies uniformly over the audio spectrum. An *SPL* meter will allow the choice of either (or neither) weighting function. See *B-weighting*, *equal loudness curves*.

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*.

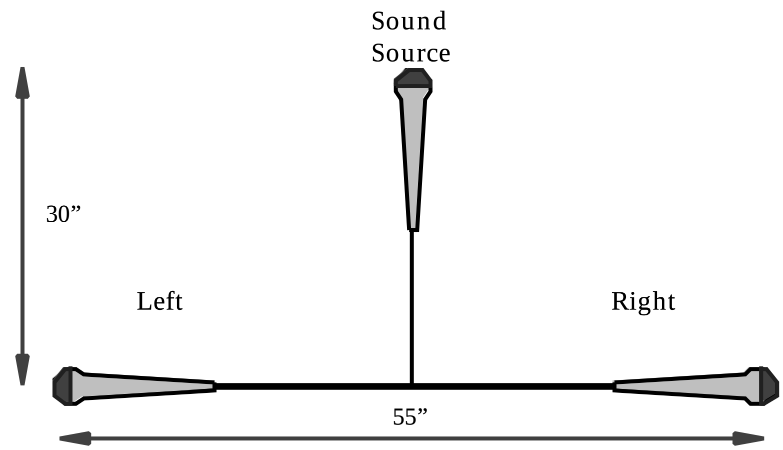
decay: (1) The time it takes for a sound to reach minimum *loudness*; the end of a sound. (2) The second of the four segments of a typical *ADSR* envelope. The decay control determines the amount of time it takes for the envelope to fall from the peak reached at the end of the *attack* segment to the *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:	1VRMS=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

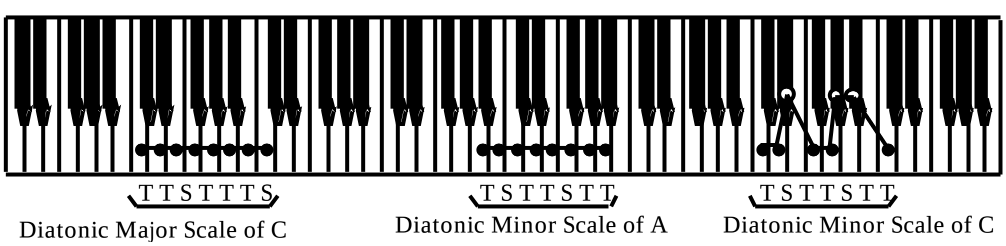
DI: Direct Injection. Also called a *direct box*. (1) The use of some form of mechanical or electrical pick-up mechanism on an instrument for the purpose recording or amplification. DI also refers to the connection of an electronic keyboard or power amplifier feeds to a mixer. A DI consists of (usually) 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*.

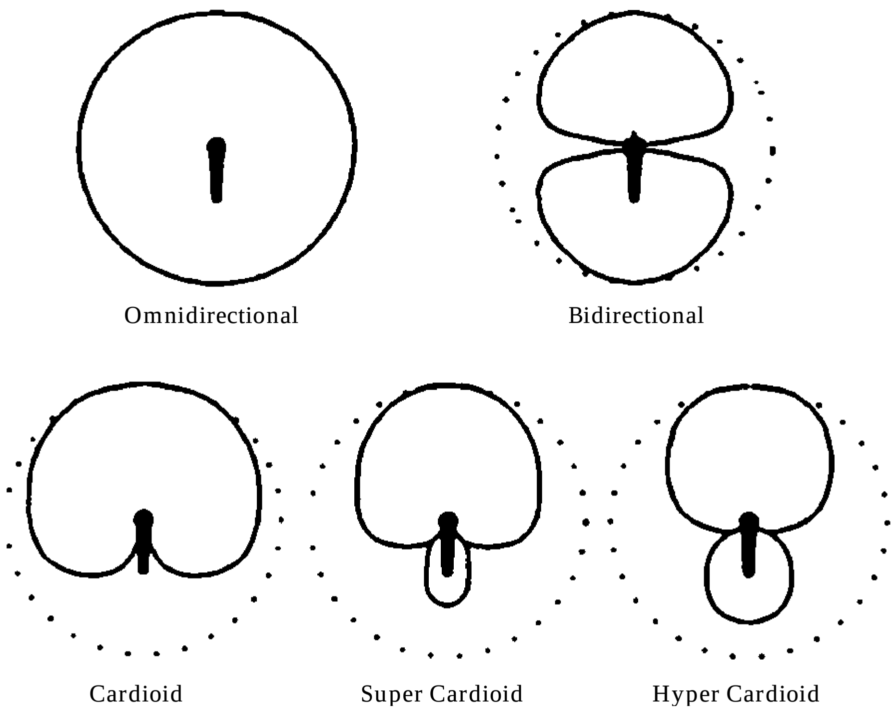
direct coupling: A connection between two devices that allows both DC and AC between them.

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*.

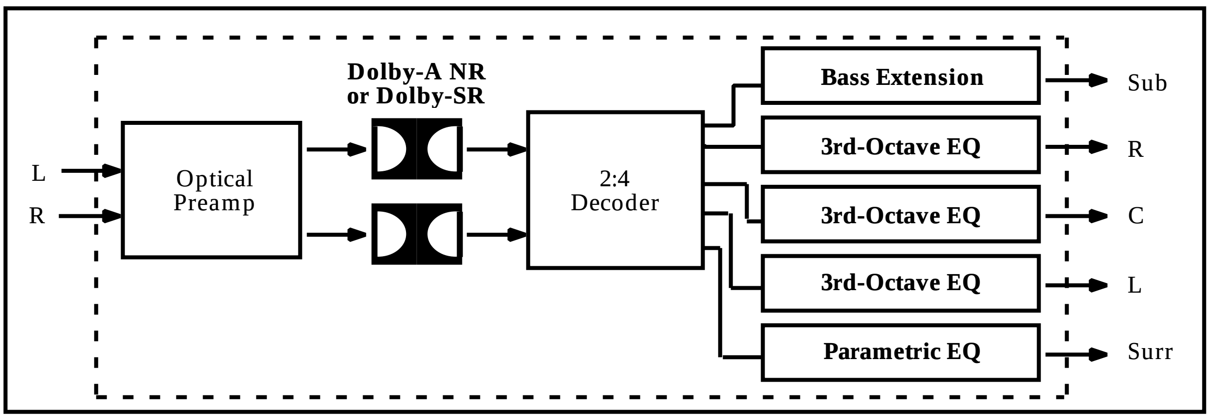
Dolby ProLogic™: A four-channel *perceptual coding* scheme developed by Dolby Labs where an *LCRS* audio signal is converted into two channels of analog audio, then recovered to yield discrete left, center, right, and mono surround channels. This is a hardware version of their surround decoder originally 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.

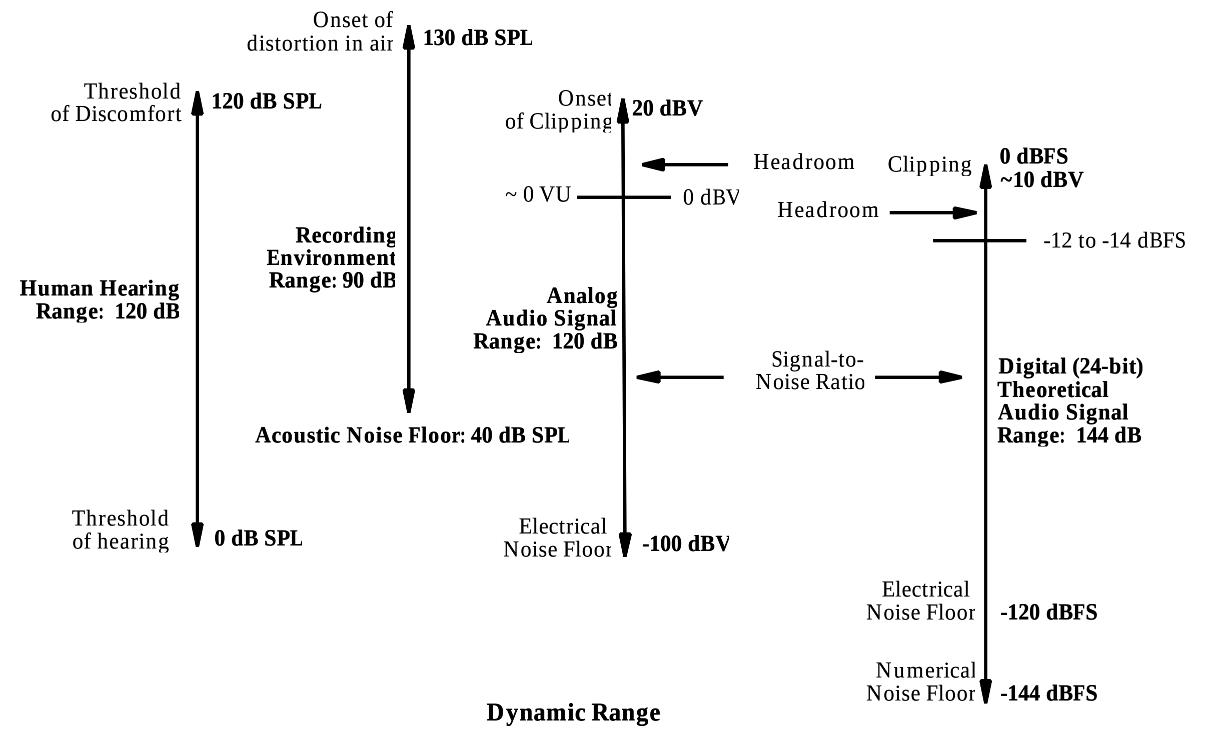
D

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

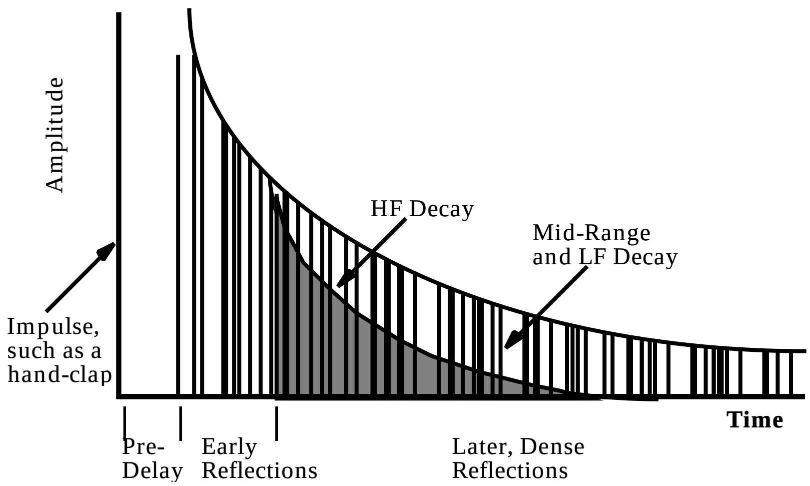
D

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*.

E

early reflections (ER): (1) The first and following reflections from adjacent room boundaries, as opposed to later reflections which are produced by farther surfaces or which have taken a longer path to reach the listener. (2) A reverb algorithm whose output consists of a number of closely spaced discrete *echoes*, designed to mimic the bouncing of sound off of nearby walls in an acoustic space. See *ESS*, *reflections*, *reverberation*.



Early Reflections

earwig: A small earpiece micropnone used to give actors an audio reference (frequently a guide music track) so that their live audio can be recorded live. See also *thumper*, *IEM*.

ECC: Error Correction Code. See *error correction*.

echo: An audio effect which is a discrete (where the onset of the repeated sound is distinct) repetition of a sound arriving at least 50ms after the *incident* sound, as opposed to *reverberation*, which is a continuous wash of closely spaced, non-discrete, echoing sound. See *delay*(3).

EDC: Error Detection Code. See *error correction*.

edge: A subjective impression of a certain roughness in the reproduced sound of a musical instrument. It is usually caused by non-uniform, high-frequency response in the loudspeaker or other audio device.

edgecode: Inked numbers applied outside the sprocket holes on file prints and mag film, used for synchronization reference. See *Acmade*, *preview codes*.

edge track: (1) In multitrack recording, either of the recorded tracks located along the edge of the tape. (2) The U.S. standard position of the recorded track on 16mm magnetic film, i.e., the position along the edge opposite the sprocket holes. See *film soundtrack*.

edit controller: See *edit programmer*.

E

edit decision list (EDL): Prior to editing a master recording or motion picture, the various takes are auditioned and a list of the desired ones is created, along with notes telling exactly where the cuts are to be made. The resulting document is the EDL. This consists of the list of *SMPTE timecodes*--in *feet/frames*, including instructions for fades, dissolves, and other special effects--corresponding to all the segments that the editor of a videotape production has decided to use in the final cut. The EDL is usually computer-generated. See also *playlist*.

editing: Intercutting of several analog tape or digital data recordings of an audio or film take in order to make an improved performance.

editing block: A cast metal block with a channel that holds *magnetic tape* firmly and in a straight line. Diagonal slits through this channel allow a razor blade to make precisely angled cuts in pieces of tape, so that two separate pieces aligned in the channel may be spliced together. The resulting splice, if properly made, will be inaudible as it passes over the playback head of the recorder.

edit master: Video industry term for the tape containing the finished (edited) program.

edit mode: See *cue mode*.

editorial sync: Alignment of picture and soundtracks such that their start marks are equal numbers of frames prior to the first frames of picture and sound, respectively. See *projection sync*.

editor/librarian: A piece of computer software that allows the user to load and store patches and banks of patches (the librarian) and edit patch parameters (the editor) by patch name.

edit programmer: A computer used to perform on-line edits and auto-assemblies. The video editor enters the *EDL*, a sequence of *SMPTE timecodes* corresponding to the shots and specific frames to be connected. The edit programmer then controls the video playback and *re-recording* decks to produce the edited video master tape according to the editor's instructions. Depending on the sophistication of the specific unit used, the editor may have to perform some special effects manually, on prompts given by the edit programmer. Also called an *edit controller*.

edit switch: On a tape recorder, a switch that engages the play mode but not the take-up motor. Tape is driven past the playback head and reproduced, but then spills off the machine and may be edited out. This process is called a *dump edit*. On some machines, the edit switch merely defeats the tape lifters, allowing the editor to *scrub* the tape past the playback head.

effect send level: The amount of effect to be added, such as *reverb, chorusing*, or other enhancements, to each *channel*.

effects: Abbreviated *FX*. Any form of audio signal processing or a device to produce: *reverb, delay, chorusing, echo, flanging*, and *phasing*, rotary (Leslie) speaker simulation, *distortion*, and *tremolo*, etc. See *processor*.

effects bus: The mixing bus in a recording console used to mix the signal to be sent to the various effects devices. Also called the *effects send bus*.

E

effects control: Two classes of Controller Change messages which are used to introduce and adjust some kind of effect such as reverb.

effects control 1 & 2: Controller Change messages which are intended to be assignable to parameters (other than depth) which appear in a synthesizer or effects unit and which control some aspect of an effect such as *reverb* time or *pitch-shift*. They operate in conjunction with Effects Depth messages; the two message types taken together are called Effects Control.

effects depth: (1) A parameter on a synthesizer, effects unit, etc. which can be adjusted by the user to alter the mount of a particular effect, such as *reverb*, *delay*, or *chorus*. (2) Effects Depth controllers. Controller Change messages which are used to implement the function described in Effects Depth. These were initially assigned to specific effects, but are now generalized and operate in conjunction with Effects Controls 1 & 2 messages; the two message types taken together are called Effects Control.

effects loop: A mixing console circuit that is used to add an effect to a signal or a group of signals. When the effect unit is plugged into the *effects bus* circuit (via the *effects send* and *effects return* jacks), it literally functions as a loop, splitting the signal off from the mixer and sending it to the effect, then returning it to the mixer, where it is combined with the original signal.

effects master: See *effects send*.

effects return: An input on a mixing console that receives the *wet* signal from the effects devices. The effects return inputs usually have volume controls (faders) to control the intensity of the particular effect in use.

effects send: An output from a *mixer* that is connected to the input of an effects device. The effects send outputs usually have volume controls to set the *effect send level*, and the overall level of all the effects send outputs may be controlled by an *effects master* control operating from the effects bus. Effects sends (usually referred to in this case as *aux sends*) are typically used to feed effects processors such as reverbs, or are used to feed monitor systems, either speakers on stage or headphones in the studio. Whereas the main outputs of a mixer have a mix of everything that has a main fader turned up, the effects sends, with their own mix controls, have an independent mix. Effects sends are also used to feed the *house mix* to the PA system, when they are usually called *post-fader sends*. Also called an *aux (auxiliary) send*. See *insert send*.

effects track: (1) An edited track of magnetic film containing sounds other than dialog or music. There can be many *effects* prepared for a film mix. (2) In videotape productions whose sound is assembled on a multitrack tape, the track or tracks on which sound effects are recorded. (3) In the 35mm *three-track mix* of a motion picture, the recorded track that contains sounds mixed from all the effects tracks. See *film soundtrack*.

efficiency: A measure, usually applied to loudspeakers, of how much of the input electrical energy is converted to sound energy, expressed in percent. The remaining energy is converted into heat.

E

EFM: Eight-to-Fourteen Modulation. The data *encoding* scheme used in CDs in order to optimize the process of reading off the disk. Groups of eight data bits are regrouped into fourteen-bit blocks by *EFM modulator* during cutting of the CD master, permitting about 25% greater data density to be laser-inscribed on the disc and allowing easier error recognition. An *EFM demodulator* in the CD player decodes the data.

EIAJ: Electronic Industries Association of Japan.

EIDE: See *IDE*.

eigentone: See *standing wave*.

EK neg: Eastman Kodak negative. Film laboratory colloquialism for “original camera negative.” Used in film production to describe a *release print* made from the original negative, whether or not any of the film involved was actually made by Kodak, Inc. Also called an OCN, probably for Original Color Negative.

electret: If two metal plates have molten wax poured between them and a high DC voltage is sent across the two plates, this assembly yields a permanent electric field, in the same way that a magnet produces a permanent magnetic field. It is hypothesized that the polar molecules in the wax align, producing the electric field. An assembly of this type is used to provide a polarization voltage for small *condenser* microphones so that they do not require *phantom power* at 48V, but operate instead at a small pre-amplified voltage of 5V-12V. Microphones constructed in this way are called *electret microphones*.

electroacoustic: The name for interactions between electrical and acoustic phenomena. The science of electroacoustics deals with the application of electrical principles and apparatus to acoustical phenomena. *Transducers*, such as microphones and loudspeakers, are electroacoustic devices.

electroacoustic transducer: A device which converts sound waves to electrical signals. Transducers such as *microphones*, *loudspeakers*, and phonograph cartridges are electroacoustical devices. The primary problem with electroacoustic transducers is that they do not exhibit a linear *frequency response* except for a relatively small range of signal frequency and amplitude. See *DI*.

electromagnetic (EM): The name for interactions between electrical and magnetic phenomena. The science of electromagnetics deals with the application of electrical principles and apparatus to magnetic phenomena. Transformers, antennæ, and phonograph cartridges are electromagnetic devices. Longer explanation: There are four known forces operating in the universe: strong and weak nuclear forces, gravity, and electromagnetism, the latter two being the two manifestations of electromagnetic force. These are mutually affective, i.e., a magnetic field can influence an electric field and vice versa as an electromagnetic wave consists of both an electric field and a related perpendicular magnetic field. The electromagnetic spectrum consists of (in order of increasing frequency) radio waves, microwaves, infrared light (heat), visible light, UV light, X-rays and gamma rays. All electromagnetic waves propagate at the same speed, the speed of light.

electromagnetic compatibility (EMC): Audio equipment that is designed to be immune to *EMI* is said to be electromagnetically compatible. Shielding is one EMI technique, as is line-filtering, etc.

E

electromagnetic pick-up: See *piezo pick-up*, *DI*.

electronic feedback: See *feedback*.

electrostatic loudspeaker: A *dipole* speaker with a *transducer* that uses the audio signal to vary the strength of an electric field which, in turn, induces vibration in a metallic or metalized membrane. In principle, it is the reverse of an *electrostatic microphone*, and very different from the more common electromagnetic *voice coil* arrangement. Used for consumer equipment as the power output is low. Electrostatic speakers are usually quite large, such as 6' high by 2' or 3' wide. They are always *direct radiators*, and they must be large to attain reasonable efficiency at low frequencies. The radiation pattern of an electrostatic speaker in a free-field is similar to that of a *figure-eight* microphone. Because of their large size, electrostatic loudspeakers tend to become very directional in the high-frequency range. They are also characterized by a low *impedance*, and this is problematic for some amplifiers. See also *planar loudspeaker*.

electrostatic microphone: A class of microphone, of which *condenser* and *electret* are types, in which air pressure changes cause changes in the *capacitance* of a condenser. The capacitor is normally *biased* by a voltage which is supplied from batteries or via *phantom power* from the signal cable. The electret is an exception, as this requires such a small biasing voltage that it is possible to charge it permanently at the time of manufacture; Sennheiser mics use a proprietary biasing scheme which utilizes RF instead of a DC voltage.

electrostatic noise: A field of random electrical charges that can affect an audio line. Electrostatic noise can be generated by neon or fluorescent lighting, electrical motors, and other broad-spectrum emissions sources. Electrostatic noise is the electrical field which is generated by *EMI*.

elliptical equalizer: A special equalizer which causes the two channels of a stereo signal to be more nearly in *phase* at low frequencies, making the signal easier to cut into a record (an LP stylus has an elliptical cross-section).

elliptical filter: A multiple-element, *lowpass* or *bandpass filter* which has the steepest possible *rolloff slope* and a small amount of *ripple* in the passband, with one or more *notch filters* added to it. Elliptical filters are used as *anti-aliasing filters* in digital audio devices.

EMI: Electromagnetic Interference. Stray *electromagnetic* fields generated from any current-carrying conductor such as nearby motors, switching controllers, high-power contactors, etc. which cause a brief, intense *pulse* that often couples into low-level signal circuits causing *noise*. This interference can enter either directly into the signal path, or indirectly via the power or ground connection. High-power RF transmitters can cause similar effects, called *RFI*. See *induction*.

emphasis: See *resonance*.

encoding: (1) The process of converting the already sampled, numerical voltage of the analog input into binary numbers and assembling these with any location and error-related data generated elsewhere into complete digital words, usually of 16 bits. See *analog-to-digital converter*. (2) The application of any type of processing to a signal before recording which will later be removed by *complementary* processing during playback. Most *NR* systems are good examples of the encoding / decoding process. See *stretched*.

E

end-addressed: A microphone that is aimed at the sound source, as opposed to *side-addressed*, which is aimed with the side of the mic at the sound source.

Enhanced CD: A *multisession* CD format which allows Red Book, Yellow Book and Blue Book CD data to be stored on one disc. See *CD Extra*.

envelope: The shape of the amplitude vs. time graph of a musical sound; the shape of a sound as it changes over time. The shape of a synthesizer's envelope is controlled by a set of rate (or time) and level parameters. The envelope is a control signal that can be applied to various aspects of a synthesizer's sound, such as *pitch*, *filter rolloff* frequency, and overall *amplitude*. Usually, each note has its own envelope(s).

envelope follower: A device used in electronic music synthesis that converts the *envelope* of a musical signal into a *control voltage*. That is, the output voltage will be low when the signal is soft and high when the signal is loud. The control voltage can then be used to control any number of parameters in the synthesizer.

envelope generator: A hardware device or software routine that generates a sound *envelope*. Also known as a *contour generator* or *transient generator* because the envelope is a contour (shape) that is used to create some of the transient (changing) characteristics of the sound over time. The purpose of an envelope generator is to give a shape to each note. By itself, an envelope generator makes no sound. Its output is used as a control source that tells some other part of the synthesizer what to do. Typical synths have three envelope generators for each oscillator: one to control *pitch*, one to control the *filter rolloff*, and one to control *amplitude*. See *ADSR*.

envelope tracking: A function that changes the length of one or more *envelope* segments depending on which key on the keyboard is being played. Envelope tracking is most often used to give higher notes shorter envelopes and the lower notes longer envelopes, mimicking the response characteristics of percussion-activated acoustic instruments. Also called *keyboard tracking*, *key follow*, and *keyboard rate scaling*. See *ADSR*.

EOX: End Of eXclusive. A System-Common MIDI message used as a flag in a MIDI datastream to indicate the end of a *System-Exclusive* message transmission.

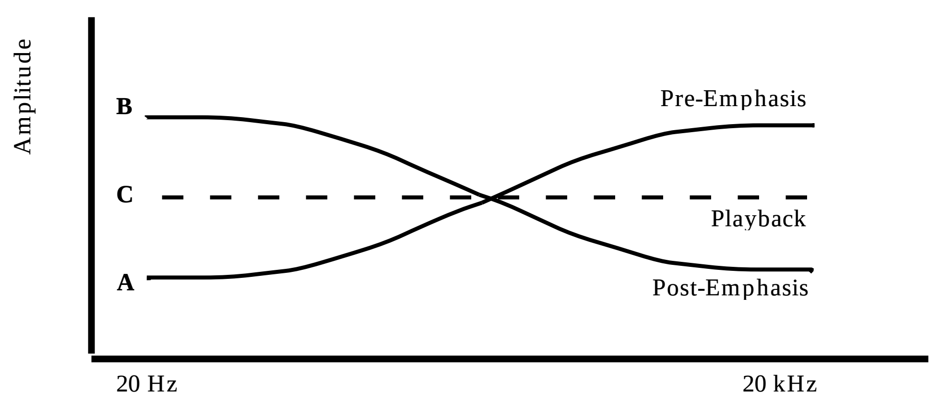
EP: Extended Play. A type of 7" phonograph record, usually played at 33⅓ rpm, allowing two songs to be cut on each side for a total recording time of up to 7 minutes. Also a 10" or 12" album with between three and seven songs. EPs are intended to sell at less than album prices, and are a way of establishing new artists without requiring record buyers to pay full LP prices. (2) Also, Executive Producer.

E

equalizer: An adjustable audio *filter* inserted in a circuit to divide and adjust *its frequency response*, altering or distorting the relative amplitude of certain frequency ranges of an audio signal. The *effects processor* used for equalization. Equalizers come in two varieties, graphic and parametric. A *graphic equalizer* typically has a number of fixed-frequency bands (5-10 in consumer equipment, 31 in professional equipment), each wired to its own front-panel slider. The control is over the amount of cut or boost (in dB) at each band. A *parametric equalizer* goes two steps further: the center frequency of each band can be selected by the user, as can the bandwidth. This affords more precise control over which frequencies will be affected by the boost or cut in amplitude. Because *EQ* circuitry with these controls is more expensive to build, a parametric equalizer will typically provide fewer bands than a graphic equalizer. A *semi-parametric* equalizer, sometimes found in multieffects devices, provides control over the center frequency of each band, but not over the bandwidth. See also *active equalizer*, *passive equalizer*, *shelving equalizer*, *Q*.

equalization (EQ): An *effect* that allows the frequency-selective manipulation of a signal's amplitude. The simplest *equalizers* are *shelving* types, offering the ability to cut or boost *gain* above or below a given frequency. Equalization doesn't only change the level of specific parts of the audio spectrum, it also changes the phase of the affected frequencies relative to those that aren't being EQ'd, i.e., EQ affects both the frequency response and phase relationships of the signal. See *composite equalization*, *pre-emphasis*, *room equalization*.

equalization curve: In tape recording and playback, a standardized equalization effect applied to an audio signal. *Pre-emphasis* and the complementary *de-emphasis* curve is applied to the recorded and reproduced signal, respectively. Pre- and post- equalization curves are different for each standard tape speed, and standards are given by the various organizations such as NAB, CCIR, and IES for their respective countries. Also described as a *pre-emphasis curve* and *de-emphasis* or *post-emphasis curve*. See also *RIAA curve*.

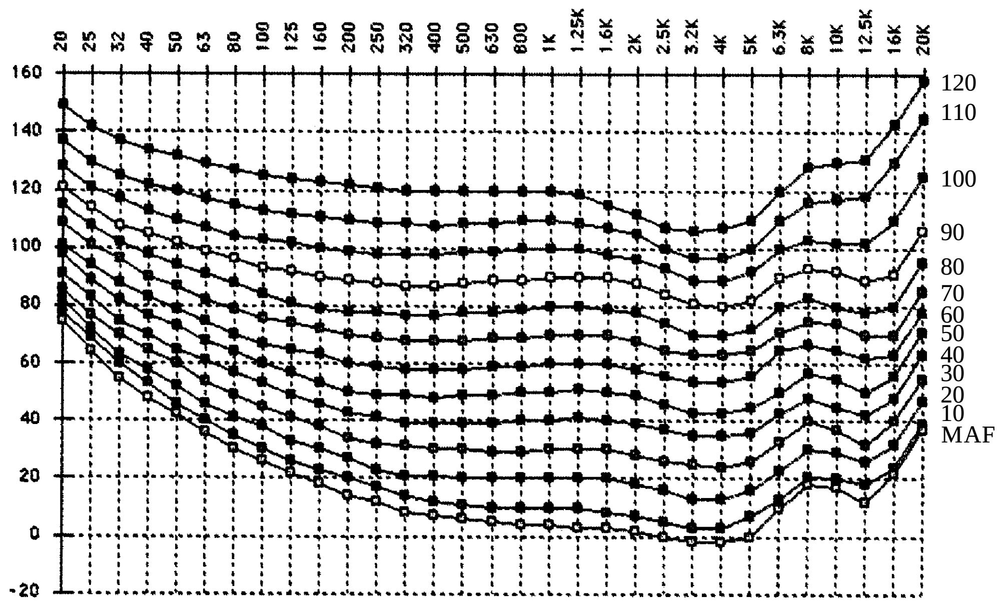


Pre-Emphasis and Post-Emphasis Curves

E

equal loudness curves: Also known as *Fletcher-Munson curves* or *phon lines*. Equal loudness curves are the inverse of *frequency response curves* and reflect the phenomenon that humans do not hear all frequencies as having equal *loudness*. In other words, human hearing is not liner in frequency. This is particularly problematic in recording as a mixed master will be perceived differently depending on the playback level. Specifically, there is a marked drop-off in aural sensitivity at low frequencies. At the opposite extreme, humans have high sensitivity to sounds in the 1kHz-8kHz range, with sound again dropping away above 12kHz. Also called *equal loudness contours*.

In the graph below, note that at 60dB SPL, a 1kHz tone is perceived as of equal loudness as a 20Hz tone at over 100dB SPL. At low levels, these differences are accentuated: the same 1kHz tone at 10dB SPL requires 80dB SPL at 20Hz.



Robinson-Dadson Equal Loudness Curves
20 Hz - 20 kHz
Right-Hand Legend Reference 1kHz

equal-tempered: A system of tuning in which the *diatonic comma* is divided equally between the twelve *half-steps* of the *octave*. All the half-steps are equal in size and are exactly one-twelfth of an octave, spanning a frequency ratio of $\sqrt[12]{2}$, or about 6%, or 1:1.059. In equal temperament, all the *intervals* are the same regardless of the *key* in which one is playing, and none except the octave is perfectly tuned. This makes it very easy to modulate from one key to another, although the keys lose their individuality because they all have equal *intervals*. The result is that the fourth and fifths are within 0.001% of just intervals, however the thirds are about 0.01% away from pure thirds which produce audible beats, the thirds in all keys being equally bad. See *temperament*, *syntonic comma*, *diatonic comma*.

equivalent input noise (EIN): EIN is becomming a common method for specifying noise in audio equipment. This is a derived figure equal to the noise measued at some gain setting, minus the gain. For example, if a microphone preamplifier puts out -85dBV noise when set for 40dB of gain, the EIN is -125dBV. Note that, while -125dBV seems better than -85dB, both figures represent the same amount of noise.

E

ER: See *early reflections*.

erase head: The *head* on a tape recorder that erases magnetic information on the tape, located just before the record head in the tape path. A high-level, high-frequency (150-300kHz) tone, called an *erase frequency* which, when fed through the erase head, re-randomizes the orientation of the tape's magnetic *domains* so that the signal to be recorded will have no *hysteresis*. See *erase oscillator*.

erase frequency: See *erase head*.

erase oscillator: A very high-frequency oscillator built into a tape recorder to supply current to the erase head. In most machines, the same oscillator supplies the *bias* and *erase frequency*.

error concealment: A technique to reduce the audible effect of a digital error in a digital audio system when the error cannot be corrected by the techniques of digital error correction. Error concealment usually consists of making a smooth transition from the last good data block before the error to the first good data block after the error, usually in some form of interpolation, i.e., crossfading. Error concealment is the reason that a digital copy from one source is often <u>not</u> exact an exact clone of the digital master. When duplicating a digital master, *error correction* and error concealment algorithms must be thoroughly understood and the dubs checked for reproduction quality. See *error protection*.

error correction (ECC): Error Correcting Code. In digital audio systems, the sampled amplitudes of the signal waveform are expressed by digital encoding. If, in the transmission of the digital words, some bits are missing or incorrect due to tape dropouts, etc., the result will be gross distortion of that portion of the signal when it is reconstructed. Error correction is made possible through the use of a parity check bit added to each data word, as well as more complex schemes. See *error concealment*, *error protection*, *CRC*.

error detection: In digital playback, the use of error bits and data derived from the audio samples to check the completeness and accuracy of the audio data before passing it on to the *D/A*. See *error protection*.

error protection: All of the circuits and data handling procedures that together accomplish *error detection*, *error concealment*, and / or *error correction* functions in any digital recording and playback format.

ESS: Early Sound Scattering. A design for control rooms where the characteristic reflections are so uniformly random that they have no character to impose on the listening space. An ESS control room is one which features a highly diffusive front end (including the monitor walls) which scatters the early sound using *Schroeder*-type *diffusers*. The body of the room is absorbent, with most of the lows damped by membrane panels. These rooms can be made fairly *live* compared to older control rooms, with a flat frequency response and good stereo imaging, both of which remain stable right to the rear corners of the room. As compared with *LEDE* and *RFZ* designs.

event editing: See *step input*.

exciter: A device for artificially enhancing a signal by adding new *partials* to it. These devices are said to compensate for loss of high frequencies in analog tape recordings. Also called an *aural exciter*.

E

expander: (1) A signal processing device which is the inverse of a *compressor*, providing the gradual attenuation of signals that fall below a user-defined threshold. This process, known as *expansion*, reduces background noise and at the same time increases the *dynamic range* of the input signal. (2) A synth, with out a keyboard or other master controller, often rack-mounted. Also called a *tone module*.

expansion: See *expander*(1).

expansion ratio: In an expander that is working below its *threshold*, the ratio given by the number of dB change in input over the number of dBs change in output. Typical ratios are in the range 1:2 or even 1:20. Expansion ratio is the opposite and complement of *compression ratio*.

expression: One of the defined MIDI Controller Change messages, usually assignable to some *parameter* in a synthesizer, such as Volume or Filter Cut Off.

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

extinction frequency: In *magnetic tape recording*, the high frequency beyond which significant cancellation occurs because its wavelength on tape, at the specified tape speed, approaches the width of the *head gap*.

F

F1: See *PCM-F1*.

fade: (1) Slow alteration of the level of a signal, usually using a *potentiometer*. See *fade-in/fade-out*. (2) Of a piece of music, usually commercial music, the repeated section at the end of the song which is subjected to a gradual fade-out. See also *outro*. (3) Short for fade-in / fade-out. Optical effects in which a scene is printed with exposure increasing or decreasing to blackness for fade-in and fade-out, respectively.

fader: (1) A variable *attenuator*, or volume control. (2) A variable control used to change the distribution of power between front and rear speakers.

fade-in / fade-out: A feature of most audio editing software that allows the user to apply gradual *amplitude* increase or decrease over some segment of the sound. Fade-in starts with no signal and gradually increases the level. Fade-out starts with a signal present and gradually decreases the level, normally to silence. See *crossfade*.

far-field: If a sound source is operating in an enclosed space, the SPL will vary with the distance that the measuring microphone is from the source. At certain close ranges, the levels will obey the *inverse square law* and at these distances, there will exist approximately a *free-field*. At greater distances, the reduction in measured level with increased distance will be less than predicted by the inverse square, and finally a region will be reached where the level is almost constant regardless of the distance, and this is called the *reverberant field*. The area between the free-field and the reverberant field is called the far-field. Its extent is a characteristic of the directionality of the sound source as well as of the acoustics of the room.

FASA: Frequency, Amplitude, Spectrum, and Ambience. An audio production method which is based on the criteria that can be changed in sonic terms to enhance a recording:

> **Frequency**: Pitch, transposing parts, chord inversions, layering with other octaves.
>
> **Amplitude**: Level, use of dynamic range to cut and boost sections, and relative volumes among parts.
>
> **Spectrum**: Textures and the range of frequencies present, layering sounds with others, introducing new textures from other parts, changing the sounds for a part, like playing a percussion line as a bass part, the contrast in frequency and textures used.
>
> **Ambience**: Space, reverb, and image information such as panning, depth, height, forward or recessed, for each part playing.

Faulkner array: A *near-coincident* microphone configuration which uses a pair of *figure-eight* microphones, both facing directly forward, but separated by about 8”.

feed: In signal routing, an output from one device that is sent into another.

feedback: There are two types of audio feedback: acoustic and electronic. (1) Acoustic feedback is where a gain control is set too high in a sound reinforcement system and the amplified sound enters the microphone and is reamplified until a steady howl or whistle is heard. This is also called *regeneration*. (2) Electronic feedback (or *negative feedback*) involves the application of a small portion of the output voltage of an amplifier to the input so as to cancel part of the input signal, reducing the gain of the amplifier, but also reducing the distortion and noise introduced by the amplifier. See *bootstrap*. (3) A specific application of feedback in *FM synthesis*, where at least one operator in each algorithm is equipped with a feedback loop.

F

feed reel: The input reel on a tape recorder, from which audio or video tape is fed to the *head stack* and onto the *take-up reel*.

feet/frames: Footage numbers for film, either separated by a colon or by a “plus” sign. For example, 101:16 and 0101+16 both indicate a point 101 feet and 16 *frames* into the film. There are 16 frames per foot of 35mm film, and 40 frames per foot of 16mm film. See *SMPTE time-code*, *LFOP*, *ABS*.

FFT analyzer: A digital device which performs the transformation from the time domain to the frequency domain of a sound *spectrum* over a wide frequency range and dynamic range. It is used to measure *distortion*, *S/N ratio*, *flutter* and *wow*, as well as the *phase* response and *frequency response* of audio devices. See *Fourier analysis*.

fidelity: The accuracy with which a music reproduction system will recreate the sound of the original music.

field: (1) The subjective environment which a listener perceives while listening to sound, such as a stereo field. See *stereophonic*, *ambisonic*. (2) The area around one or more microphones; the *acceptance angle* of the microphone. (3) The spatial area of *electromagnetic* force. (4) In video, a subgroup of visual data consisting of either the odd- or even-numbered lines of any *frame*. In NTSC, for example, each field is displayed separately for $\frac{1}{60}$ of a second within the total frame duration of $\frac{1}{30}$ second. For each frame, field number one contains line #1, #3...#525; field number two contains lines #2, #4...#524. PAL television broadcasts use an analogous scheme, but has a different frame rate and number of lines per frame. See *blanking interval*.

field rate: Frequency at which video fields occur: 59.94Hz in NTSC, 50Hz in PAL.

fifth: The interval between a note and the one seven half-steps above or below it. See *interval*.

figure-eight microphone: A *directional microphone* whose pick-up pattern resembles the figure 8, meaning that it is insensitive to the sides but has full sensitivity at the front and back. As the polar pattern resembles the shape of a cosine curve, the figure-eight microphone is sometimes also called a *cosine microphone*. Figure-eight mics were traditionally *ribbon mics*, but now they can also be *condenser mics*. Also called a *bi-directional microphone*.

file format: The data in a computer file has a particular order and length. The specification which determines the structure of the file is called the file format and is software- and/or hardware-specific. Files, such as MIDI files, may contain data, instructions to other software programs and/or hardware devices, and/or programs. The file may also contain ECC data, network information, and other non-user overhead data. Some file formats are made publicly available to allow the implementation of *plug-ins*; others are proprietary to the vendor. The file format usually begins with a file header, followed by data, followed by *ECC* data, followed by some kind of *stop bit*, if the file format is variable-length. See *AIFF*, *RIFF*, *.AU*, *MPEG*, *.MOD*, *.RA*, *SFI*, *SMF*, *.SMP*, SND, *.WAV*, *HFS*, *ISO 9660*, *.VOC*.

fill: The sound between words in a production track that is used both to replace undesirable noise on the track and to create *handles* for use in extending the track at the beginning and the end.

F

filled: Filled effects is a version of the effects *stem*(s) of a *soundtrack* which includes all effects, including *cut effects* and *Foley*. See *M&E*.

fill leader: The film that is inserted into units of mag film in order to keep synchronization during silent sections. Fill leader is usually made up of recycled *release prints*. See also *leader*, *plastic leader*.

film: Now, 35mm film accommodates the 6-track digital sound, but previously almost all films released in 70mm from 1971-1992 were originally photographed in 35mm and then blown up to the 70mm format specifically for playback with 6-track sound. The motion picture exhibition format from 1955-1971, 70mm, contained 6-track magnetic sound, using camera equipment manufactured by *Todd-AO* and Panavision. The camera negative was 65mm wide, with the additional 5mm outside the sprocket holes used for the magnetic stripes on *release prints*. Almost all modern 70mm prints in the U.S. have no magnetic track, but instead use *DTS* in conjunction with a wide *timecode* track outside of the perforations.

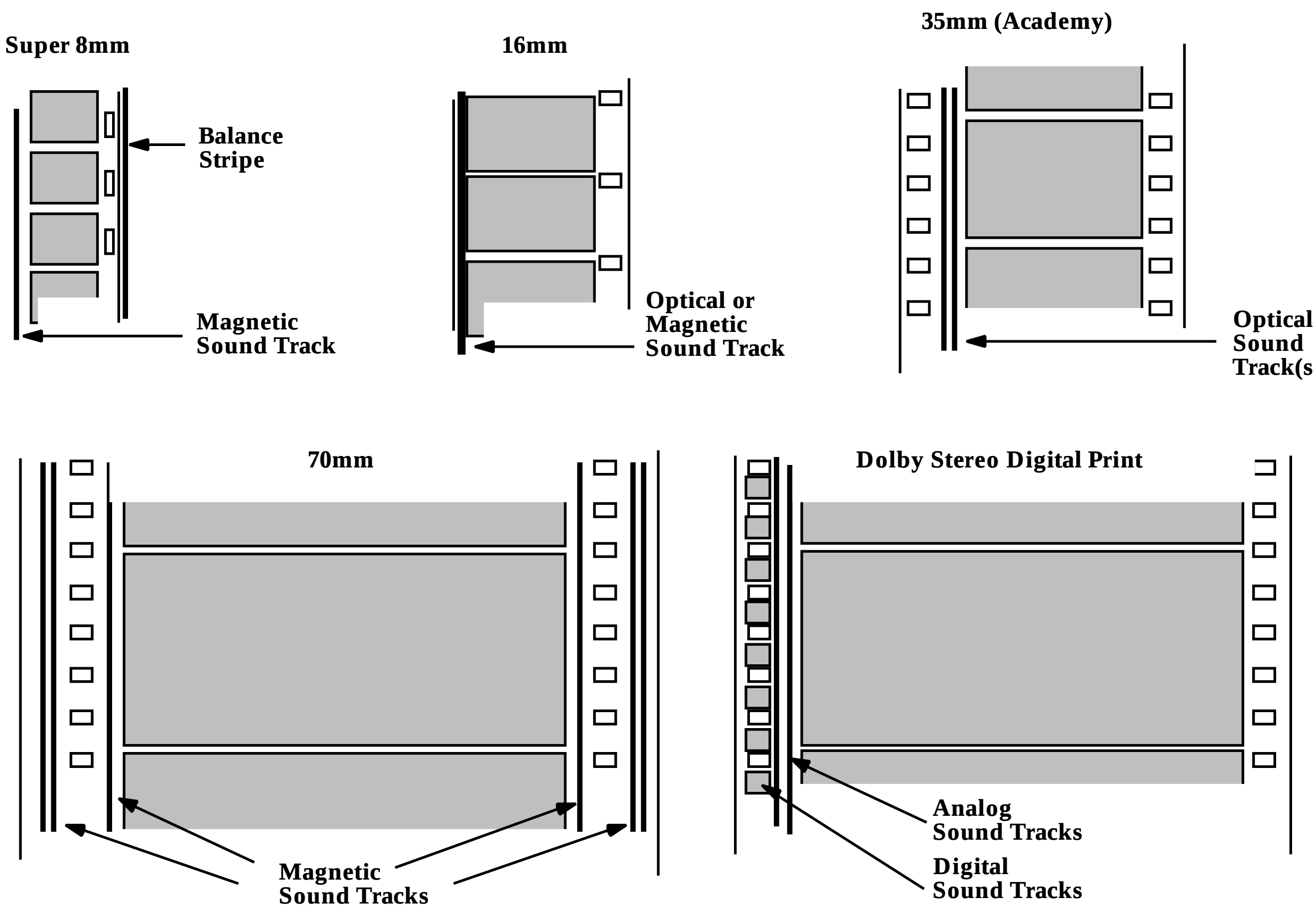
The image, in its widest and standard form has an *aspect ratio* of 2.2:1, which is narrower than the 2.4:1 *anamorphic* 35mm format that is the source of many 70mm prints. However, when *flat* 1.85:1 films are blown up to 70mm, they usually retain their original aspect ratio, with black borders on the side. the IMAX/OMNIMAX special venue format also uses 70mm film, although it runs horizontally through the camera/projector, and each frame is 15 *perfs* long, as opposed to the standard five perfs. Sound is always *double-system*, utilizing *mag film* or custom digital formats.

film chain: A device consisting of a motion picture projector and video camera, used to copy films onto videotape or to broadcast them directly. To adapt the 24 fps U.S. *frame rate* to the 30 fps NTSC video frame-rate, some chains use a projector with a five-bladed shutter, which shows each frame of film five times onto the vidicon tube of the video camera. The resulting 120 fps are regrouped four-at-a-time into 30 video images per second.

film footage: There are 16 *fps* per foot of a standard 35mm film image, each lasting four sprocket holes. At the standard rate of 24 fps, 35mm film runs at 90 feet per minute, or 18 inches per second. See *frame*.

F

film soundtrack: The audio component, including *DME*, of a film composition. There is usually a requirement for sound to be synchronized to the video image. This has been achieved by a variety of means, including the recording of sound on *optical tracks* etched into the film emulsion alongside the frames, fixing magnetic tracks on the film surface, synchronizing the film with a separate tape machine by means of mechanical sprockets, and electronic sync using systems such as *SMPTE*. See also *Dolby Stereo*, *LC Concept*, *optical track*, *SR.D*, *pilot tone*, *layback recorder*, *source track*.



Film Soundtrack Formats

filter: (1) A type of equalizing device for subtractively eliminating selected frequencies from the sound spectrum of a signal and perhaps, in the case of a *resonant filter*, increasing the level of other frequencies. See *VCF*. For example, a *lowpass filter* passes lower frequencies and removes the higher frequencies. By raising or lowering the filter *rolloff frequency* parameter, a sound will be made brighter or darker. (2) A device or MIDI software *filter* that eliminates selected messages from the MIDI data stream, usually called *MIDI filtering* by *data thinning*. See also *running status*.

F

filter resonance: The greater the *resonance* on a filter, the greater the effect of the filter: as resonance control is turned up, a little peak appears at the *rolloff* frequency. The *harmonics* that fall within that peak are accentuated. The greater the resonance, the higher the peak and the more pronounced is this effect. The effect of the swept resonant peak does not occur in real instruments. See also *Q*.

filter scaling: See *keyboard scaling*.

final mix: The mixing of the final *stems*, which, when mixed and replayed represent the film's finished *soundtrack*. In a stereo film (or surround-encoded TV programming) it is most common to record the *DME* stems on three pieces of 4- or 6-track magnetic film, *with Dolby-SR noise reduction*, or on analog or digital multitrack tape, or onto a *digital dubber*. These stems, also known as *dub masters*, are then used to create the *print masters*, the *M&E* mix, a mono mix, and possibly an airline version. For a non-surround-encoded stereo mix, then the stems might be in standard stereo format, but this precludes the subsequent production of a *5.1* mix, say for *DTV*.

finder: The user interface of the Mac operating system, allowing access to the file system, peripherals, and other components of the system hardware and software.

fine: The End.

fine cut: A stage in the editing of a film or video production at which the *workprint* or *EDL* is completed, denoting that the production is ready for final cut approval.

FIR: Finite Impulse Response: A class of *filter* designs whose impulse response falls to zero in a predictable, finite length of time, as opposed to an *IIR* (Infinite Impulse Response) design whose impulse response may never fully attenuate the impulse. A common use of FIR is to produce filters with a *linear phase response*. Filters such as analog *anti-aliasing filters* which have a very sharp *rolloff slope* produce unacceptable amounts of *phase distortion* in the output signal; FIR filters are used at the end of an oversampling *A/D* chain to eliminate any signal that would represent *aliasing* in the slower output datastream, without causing significant phase distortion.

F

FireWire: A digital audio transmission medium developed by Apple computer, designed to support dozens, possibly hundreds, of high-bandwidth audio streams. IEEE 1394 is designed to replace point-to-point AES/EBU connections, will support multiple data formats so that audio, video, MIDI, and control signals may all be send over a single cable. FireWire also distributes power as well as data, permitting hot-plugging of devices. IEEE 1394 is designed to be a fully specified bus, bi-directional and with the ability to broadcast from a single source to multiple receivers. Currently (late 1998) there is a 4.5–limit between any two adjacent nodes, designed to support a simple, low-cost clocking mechanism to be built into the standard to support isochronous data transfers for audio and video. The isochronous clock embedded within the IEEE 1394 standard runs at 8MHz, or one "tick" every 125µs. This is problematic for audio signals which require upwards of 44MHz clock rates, so the FireWire standard is being modified to address the problem of high-resolution synchronization. The signal can traverse up to a maximum of 16 hops, effectively extending the distance to about 70 meters. This was originally developed to support the transfer of high-bandwidth signals between computer peripherals; the multi-layer IEEE 1394 standard also allows the use of many other cabling technologies, including Category 5 twisted pair copper wire and 50mm multi-mode optical fibre, the later permitting distances between devices of hundreds of meters. See also *mLan*.

Comparison between SCSI and FireWire

	SCSI	**Firewire**
Number of devices supported	6	63 per segment
Connector size	Large-Medium	Small
Requires termination	Yes	No
Sets device ID automatically	No	Yes
More than one computer per segment	No	Possible
Device-sharing	No	Possible
Hot-pluggable	No	Yes
Topology requirements	Serial	None
Transmission method	Parallel	Isochronous
Computer required for data transfer	Yes	No
Standard Transfer rate	40 Mbps	400 Mbps
Maximum transfer rate	640 Mbps	800 Mbps min.
Maximum network length	12 meters	72 meters

fishing rod: See *boom*.

fixate: The process whereby a CD's overall lead-in, program data and lead-out areas are written. This is done during the *track-at-once* recording process when the final session is written, allowing all of the data contained on the disc to be read by any CD or CD-ROM drive.

fixed formant: A frequency characteristic of a sound, e.g., human vowel sounds are formants which are relatively fixed in frequency, even though the *pitch* of the voice may be changing as in singing. It is the fixed formant frequencies in the presence of the varying pitch of a musical instrument that shapes the instrument's *timbre* and makes the instrument recognizable.

F

flamming: An undesirable audio occurrence in which one of the instruments used on a rhythm track strikes slightly behind the others. It is caused primarily by the improper application of delay or tempo change.

flange: The round metal sides of a tape reel that keep the tape aligned as it winds onto the hub.

flanging: Named for the original *effects* technique where a second audio tape playback deck was slowed by a thumb on the reel flange, flanging is a special audio effect where a delayed version of a signal is mixed with the original signal, creating a swooshing sound. This is caused by the fact that when the time delay is different for the two combined signals, there will be frequencies where the phase-shift is 180˚ and the signals will cancel, causing deep dips or holes in the frequency response curve called the *comb filter* effect. As the speed is varied, the frequency of the dips is swept across the frequency range, giving the swooshing sound. Electronic flangers contain an adjustable electronic *delay line*. If the time delay is very short, the effect is called *phasing*. See also *notch filter*.

flat: (1) The condition of a note which is, either deliberately or accidentally, lowered in pitch. This might be only a few cents or as much as a tone (double flat, 𝄫). In music notation, a flat is indicated by the flat symbol (♭), meaning lower the tone by a *half-step*. (2) The same as *dry*. Unequalized, uncompressed, and otherwise unprocessed, describing a signal from any source: mic, instrument, or tape playback. (3) The neutral position on a *tone control*, effecting no change to the signal. See *center detent*. (4) Film projection using non-anamorphic lenses. In the U.S. the term flat is synonymous with an aspect ratio of 1.85:1 widescreen. See *anamorphic*.

flat response: The faithful reproduction of the *amplitude* of an audio signal, specifically, variations in output level of less than one decibel above or below a median level over the entire audio spectrum. A system which has the same gain at all frequencies of interest will yield a graph of the gain versus frequency that will be *linear*.

flattening: A general term for the process of moving the final stems, tracks, etc. to audio/video tape, usually involving a substantial reduction in the number of tracks on which the sound is carried and merged with the time-coded video. Also specifically refers to the process in ProTools™ whereby stereo audio and video is exported to a QuickTime movie or other format.

flat wind: To employ a slower-than-normal fast forward or rewind mode on a tape transport in order to wind the tape smoothly and evenly onto the reel or hub, usually for storage. Flat winding prevents edge curling and other types of deformation damage. See also *heads-out*.

Fletcher-Munson effect: Fletcher and Munson measured the sensitivity of human hearing at various volumes and frequencies with the finding that humans hear best in the range of 3kHz-4kHz; the sensitivity falls off rapidly at lower frequencies and somewhat more slowly at higher frequencies. In other words, very soft sounds must be more powerful at frequencies lower and higher than 3kHz-4kHz in order to be heard. The *loudness* control on music reproduction systems was designed to compensate for the Fletcher-Munson effect. See *equal loudness curves.*

F

flicker noise: At high frequencies, *intrinsic noise* is dominated by *Johnson noise*. At lower frequencies, there exists a critical frequency, *f*, at which noise rises proportional to $\frac{1}{f}$. Below this critical frequency, the noise is called flicker noise. Sometimes referred to as *flicker effect*.

floating unbalanced output: a ¼" phone-type output where the sleeve of the output stage is not connected inside the unit, and the ring is connected, usually through a small resister, to the audio signal ground. This allows the tip and ring to "appear" as equal *impedance*, not-quite-balanced output stage, even though the output circuitry is unbalanced. Floating unbalanced often works to drive either a balanced or unbalanced input, depending if a TS or TRS standard cable is plugged into the output. If a floating unbalanced connection hums, a *ground-lift* cable is required. Also known as a *pseudo-balanced* output, or *quasi-balanced*.

floor: In a *noise gate*, the amount of attenuation applied to an input signal whenever it is below the threshold level. For example, in a gate whose threshold is set to -10dBm and whose floor is set to -30dB, 30dB of attenuation is applied whenever the incoming signal drops below -10dBm.

flutter: (1) In a tape recorder, if the tape speed varies, the pitch of the recorded music will vary. If the rate of variation is fairly high, typically above 5Hz or so, it is called flutter. If the speed varies at rates below several *Hertz*, it is called *wow*. Flutter is actually a type of *frequency modulation distortion* and imparts a *tremolo*- or *vibrato*-like character to the music. (2) See *chatter*.

flutter echo: An acoustic effect where sound is reflected back and forth between two parallel surfaces such as walls. The same effect as *standing waves*, but at lower frequencies, flutter echo is created when reflections are lower than 15Hz, or when walls are greater than 25' apart.

flux: Lines of force surrounding a magnet. In measuring the strength of a magnetic field at a particular location, the number of lines of force per unit area in a plane perpendicular to the direction of the lines. The standard reference unit in magnetic tape for recording is the nanoweber per meter, and is called *flux density*.

fluxivity: The numerical measure of maximum *flux density* a specific type of recording tape can hold. *Reference fluxivity* is a standardized amount of flux, specified in nanowebers per meter, which is laboratory-recorded on a test tape at various frequencies. Such *reference tapes* are used to calibrate tape recorder 0dBVU playback levels.

fly in: (1) To mix sounds from a non-sync source into the live sound for a TV show, or into the mix for a videotape production or spot. One may fly in a narrator, Foley, etc. (2) To record sections from one or more tracks of a multitrack tape onto a second recorder (generally a two-track), then copy them back onto the multitrack in another section of the performance. For example, one might take the background vocals from one chorus and fly them into another section of the song. Short fly-ins can be done without *SMPTE* sync, although it is somewhat difficult.

FM sync: The 13.5kHz frequency-modulated *sync pulse* recorded on *Nagra* IV-S recorders.

F

FM synthesis: A *sound synthesis* technique which multiplies *sine waves* together in an attempt to generate complex waveforms more quickly (than *additive synthesis*), usually adding several of these products together in an attempt to get its more effective results, which is why a 6-operator FM sounds better than a 4-operator FM as more products are being summed. See *sound synthesis.*

foldback: The general term for the part of a *sound reinforcement* system in an auditorium which supplies amplified sound to the performers so they can hear themselves. See *monitor mix*.

foldback send: See *monitor send.*

foldover: See *aliasing.*

Foley: Creating sound effects by watching the picture and mimicking the action, often with props. Foley artists, also known as *Foley walkers*, make use of a variety of objects and/or surfaces to elicit realistic sound effects; most commonly used in the recording of on-screen footsteps, hence the term "walkers." Foley effects are named after Hack Foley, who was the head of the sound effects department at Universal Studios. The audio track which contains Foley sounds is known either as a *cloth track* (west coast) or *rustle* (east coast.)

footage counts: See *counts.*

formant: A frequency band in the spectrum of a voice or musical instrument that contains more energy or amplitude than the adjacent area, i.e., the partials are quite closely spaced in the region, giving the sound its *timbre*. For example, the formants produced by the human vocal tract are what give vowels their characteristic sound. See *fixed formant*.

format: (1) In vinyl records, the size of the disc and its rpm rating. (2) The physical specifications of a specific film or print, e.g., 35mm, as will as the type of soundtrack (optical or magnetic, stereo or mono, with or without *NR*), whether it is color or b/w. (3) The width of a videotape, and the designation of the electronic system by which it is recorded. (4) In radio and TV, the type of programming featured. (5) *Magnetic tape* format. For any tape recorder or recorded tape, the number of tracks, their width and position with respect to the tape, and the overall width of the tape itself. Tape speed is not always included, e.g., 8-track 1" and 24-track. This is usually called *track format*. (6) See *file format*.

forward masking: See *temporal masking.*

Fourier analysis: A technique, usually performed using a *DSP* algorithm, that allows complex, dynamically changing audio waveforms to be described mathematically as sums of sine waves at various frequencies, amplitudes, and phases. The Fourier transform allows a function that represents an audio signal (signals are in the time domain because they exist in time) to be transformed to another function which represents the same signal in the frequency domain. The signal in the frequency domain is called a *spectrum*, and the same signal in the time domain is called a *waveform*.

F

four-stage envelope: The Yamaha DX7 synthesizer introduced a new type of envelope generator, one which had four *rate control* and four *level control* parameters, for a total of eight *parameters*. Each rate parameter controls how long it takes the envelope to move from one level to the next. This is referred to as a four-stage envelope because each rate/level pair is considered a stage. Technically, an ADSR envelope only has one true stage, the decay/sustain stage and two more partially controllable stages because the attack and release levels are fixed at 100% and zero, respectively.

fourth: The interval between a note and the one five half-steps above or below it. See *interval*.

four-track: See *quarter track*.

fox holes: Small perforations on 35mm *release prints* that allowed for the addition of *mag stripe* for the *CinemaScope* process. This process has now largely been replaced by *Dolby Stereo*.

fractal music: Music created by the use of fractal equations. By assigning musical parameters such as pitch and volume to the x and y axes, it is possible to produce music as the Mandelbrot set is calculated.

frame: (1) The internal structural support of a *loudspeaker* when holds the *voice coil* and the *diaphragm*. (2) The basic unit of *SMPTE timecode*, corresponding to one frame of a film or video image. Depending on the format used, SMPTE time can be defined with 24, 25 30 or 29.97 frames per second (fps). (3) In digital audio, a frame is a unit of digital information. In the CD, a frame covers six sampling periods, or 136μs.

Usage and Country	Frame Rate per Second	Time per Frame
UK and European Film industry	24	41.66ms
UK and European video and TV	25	40.00ms
USA B/W TV and video	30	33.33ms
USA Color TV and video	30 drop-frame (~29.97fps)	33.37ms

frame lock: Frame lock maintains synchronization between master and slave transports, using the positional information available in the timecode address. Also called *frame sync*. See *SMPTE timecode*.

frame rate: See *frame*.

free encoding: An extension of *spaced microphone recording techniques*, systems which create pseudo-stereo from a mono source will also generate a strong surround signal, and stereo-width controls can be used to manage the balance between frontal and surround channels. Increasing stereo width also increases the level of the surround channel, whereas decreasing width reduces the surround content. This is because a surround decoder will automatically send anything which is of a similar level, but opposite polarity between left and right channels, straight to the surround output. Artificial reverberation, for example, is automatically spread across L, S, and R. See *LCRS*.

F

free-field: A sound source radiating into three-dimensional space where there are no reflecting surfaces is said to be radiating under free-field conditions. The *SPL* as measured at various distances from the source would obey the *inverse square law* precisely. There is no such thing as a true free-field, but it is approximated in an anechoic chamber. Because all rooms have at least a small amount of *reverberation*, the sound field from a source is always contaminated with reflected sound. See also *far-field, near-field, reverberant field.*

FreeMIDI: A Macintosh operating system extension developed by MOTU that enables different programs to share MIDI data. For example, a sequencer could communicate with a *librarian* program to display synthesizer patch names, rather than just numbers, in the sequencer's editing windows.

freewheeling: A condition in which a clock *synchronizer* continues to generate *timecode* even when it encounters *dropouts* in the timecode source, or in which a digital audio playback device continues to generate audio in the absence of, or while ignoring, a timecode input. See *jam sync*.

frequency: The number of waves (or cycles) arriving at or passing a point in one second, expressed in *Hertz*. See *pitch*, Appendix D.

frequency distortion: Frequency distortion results when the amplitude of the output of a system or a device varies as the frequency of the input varies, while the amplitude of the input is held constant.

frequency doubling: See *doubling.*

frequency masking: An audio artifact which occurs when several sounds are mixed, all which occur in the same frequency range. This happens because human ears tend to blend simultaneous sounds into a single, composite sound. When several instruments or other sounds emphasize similar frequencies, those frequencies accumulate and can either become too dominant or can cause one sound to mask another. Also called *band masking* or *auditory masking*.

frequency modulation (FM): (1) A change in the frequency (pitch) of a signal. At low modulation rates, FM is perceived as *vibrato* or some type of *trill*. When the modulation wave is in the audio range, FM is perceived as a change in *timbre*. FM synthesizers, commonly found on soundcards, create sounds using audio-range frequency modulation. See *FM synthesis*. (2) Frequency modulation is the instantaneous changing of the frequency of a carrier in response to a modulation signal, usually an audio waveform. As the signal voltage varies up and down as it follows the waveform, the frequency of the carrier varies up and down from its nominal unmodulated value. The FM receiver is tuned to the carrier frequency, and the received signal, after suitable conditioning, is applied to a special circuit called an FM *detector*, also called a demodulator or discriminator, which recovers the audio signal. See *amplitude modulation.*

frequency modulation distortion: Examples of frequency modulation distortion are *flutter* and *wow*, and *Doppler distortion* caused by the motion of rotary (Leslie) loudspeaker cones.

frequency ratio: The ratio of the higher *pitch* in an i*nterval* to the lower pitch. See *consonant*, *harmonic*.

F

frequency response: The amplitude response of a system or device as a function of the input frequency characteristic. It is a complex function which describes the way in which the *gain* and *phase* of a system or device vary with the frequency of the stimulus. Frequency response is a characteristic of a system or device, not a characteristic of a signal. See *linear*.

frequency response curve: A graph of the *frequency response* of a device, i.e., the graph of its output amplitude response vs. the input frequency. See *linear*. For example, the frequency response curve for a microphone is a graph of the mic's output level in dB at various frequencies. The output level at 1kHz is placed on the 0dB line and the levels at other frequencies are placed above or below that reference level. The shape of the response curve suggests how the mic sounds; a wide, flat response tends to sound accurate and natural. A rising high end or a *presence peak* around 5-10kHz sounds more crisp and articulate. Note that the response curve is measured at a specified distance from the mic, usually 2-3 feet; the curve reflects the performance of the microphone only for that particular distance.

frequency response errors: Any deviation from a *linear* output response in an audio device.

frequency shifter: A device that linearly shifts all the frequencies of a complex input signal. Also called a *spectrum shifter*. All frequency components are shifted linearly, i.e., by the same number of *Hertz*, in contrast to a *pitch-shift*. In a pitch-shift, all the frequency components are shifted by a constant percentage, and therefore, high frequencies are shifted proportionally more than lower ones. A pitch-shift-by-speed change thus preserves all the musical intervals between components. A true frequency shifter, in contrast, destroys the *harmonic* relationships between the components. The sound of a consonant musical tone becomes disconsonant or *clangorous* or harsh, depending on the amount of shift. Frequency shifters are used in electronic music synthesizers for special effects.

fringing: A rise in the level of low frequencies when a recording is replayed by a tape head with a narrower track width than the one used to record the tape. Low frequencies from the recorded areas adjacent to those actually being played back *bleed* into the playback signal.

FSK: Frequency Shift Keying. FSK is a sequence of two alternating audio tones, typically generated by a sequencer, drum machine, or computer MIDI interface, that is recorded on one track of an audio or video tape for synchronization to MIDI sequencers and drum machines. See *pilot tone*.

full-coat: Magnetic film coated with oxide across its entire width, available in 16mm and 35mm widths. Contrast with *full stripe*. See *mag film*.

full code: A term meaning that a sample word is set to all ones, i.e., that it is representing the largest number possible at that word length. This is important in the representation of digital audio amplitude, where a full code word is equivalent to -0dB *headroom*, or the amplitude that is the loudest sound which can be encoded without *clipping*. See also *digital black*.

full score: A notated form of a piece of music which contains the complete music for all instruments or vocal parts, aligned vertically, i.e., the full complement of *band parts*.

full-stripe: Magnetic film with oxide coating in just the area where the recording takes place, allowing the transparent material to be written on. Contrast with *full coat*.

F

fullth: A subjective term applied to a recorded musical program with many voices in the lower mid-range of frequencies, e.g., cellos and violas, background vocals, rhythm piano, etc., giving the mix a lush or heavy richness. Also called *gush*.

full-track: A ¼" tape *format* in which a single, mono track is recorded across the entire tape width. Loosely used to refer to wider tape formats in which each of two or four tracks is ¼" wide.

fundamental: The perceived *pitch* of a sound; the lowest frequency vibration component in a complex sound which also carries a set of higher-pitched vibration called *overtones*. The fundamental is always the first *harmonic* and / or *partial* component of a sound. See *DCO*.

fusion: In *stereophonic* music reproduction, fusion is the perception that sounds from two or more loudspeakers are being produced by a single sonic *image*.

fuzz box: An effects device designed to produce *clipping* distortion, most frequently used with electric guitars. See *DI*.

fx: See *effects*.

G

G722: An international telecommunications standard for data reduction used for limited-bandwidth speech over digital telephone networks; used only in basic ISDN applications.

gaffer: (1) On a film set, the head electrician, now more commonly called the "Chief Lighting Technician." (2) In film, the head of a crew, e.g., the "gaffing mixer" would be the re-recording mixer-in-charge, formerly known as the *gunner*.

gain: The output voltage of a device divided by its input voltage. Most *passive* devices have a negative voltage gain, and most *active* devices, especially amplifiers, have a positive voltage gain. Usually expressed in dB, this is correct only if the input and output *impedances* are the same, a condition not usually met. The square of a voltage ratio is a *power ratio* if the condition of matched impedances is met. See *amplifier gain*, *impedance-matching*.

gain-before-threshold: In a *compressor* or *limiter*, the decibel gain applied to signals below the threshold level, i.e., before the compression circuit.

gain control: The fader that controls the strength of the output signal of an amplifier. This term is misused on many amplifiers, since the gain remains constant, while the gain control actually adjusts the signal input level. Also erroneously called *volume control* on consumer equipment.

gain riding: Manual, real-time volume control during recording to prevent overload and distortion at loud levels, and to avoid noise problems at low levels.

gain stages: Electronic components (or sets of components) whose purpose is to provide signal amplification in an *active* device.

gallows arm: A type of mic stand which consists of a vertical section, to the top of which is fitted an adjustable rod which carries the microphone.

galvanic isolation: In transformer, galvanic isolation means that no electrical current can flow directly from one winding to the other as they have no direct electrical contact. However, a signal can flow between the windings via electromagnetic coupling.

gang: To mechanically or electrically couple two or more controls, such as post-effects processors, faders, etc. The combined assembly is then called a *ganged* fader, etc. See *grouping*.

gap: The distance between the pole pieces of a magnetic *tape head*. The *gap width* is the dimension of head gap measured along the tape path, typically 90 mils for a professional playback head, 150 mils for a record head, and 400 mils for an erase head. Sometimes specified in *microns*.

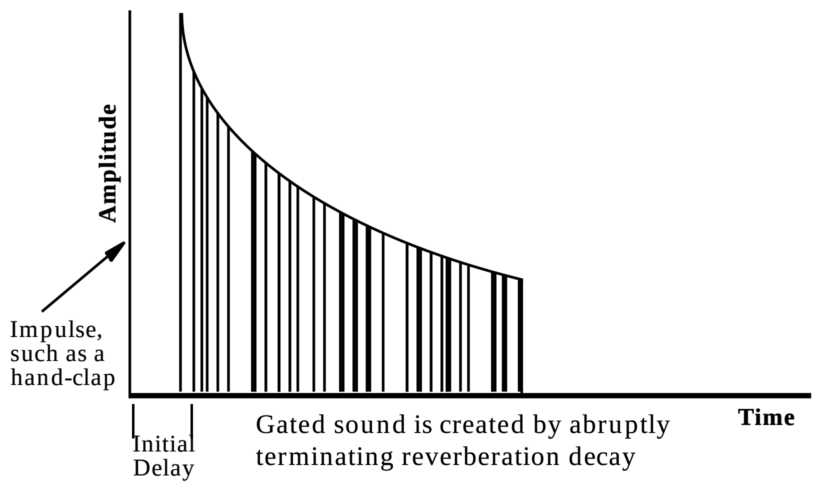
gap scatter: Any deviation from perfect *head gap*-track alignment on a multitrack tape recording.

gap width: See *gap*.

gate: (1) See *noise gate*. (2) A *control voltage* generated by any key on a synthesizer keyboard that instructs signal generators and other devices to begin operating. (3) The part of a movie camera that has an opening to allow light from the lens to expose the film, and that holds the film steady during that exposure. In a projector, the light source illuminates the frame held steady in the gate. The lens then projects the image onto a screen.

G

gated reverb: The use of a *noise gate* to cause a sudden termination of a reverberation effect, without allowing the normal *decay* segment to complete. This gives the sound an unnatural, industrial timbre.



Gated Reverb from a Percussive Sound

gauge: The width of a particular film stock, e.g., 8mm, 16mm, 35mm, 65mm, or 70mm. See *format*.

gauss: (1) The unit of magnetic field strength, reflecting maximum *flux density*. (2) The unit of measurement for *remanent* magnetization on recording tape.

General MIDI (GM): A subset of the MIDI specification which is a minimum set of requirements for MIDI devices aimed at ensuring consistent playback performance on all instruments bearing the GM logo. Some of the requirements include 24-voice *polyphony*, a standardized group (and location) of 128 sounds, that the synthesizer be 16-part *multitimbral*, and provides for a standard *pitch* encoding. Some manufacturers have proposed supersets to GM, e.g., Roland's GS and Yamaha's XG.

generation: A term for the number of successive times a sound has been copied on analog magnetic tape. The original recording is the first generation, a copy from that is a second generation, etc. Thought to be less relevant in digital recording, but that's thoughtless.

generator module: A synthesizer module that generates sound, usually through an *oscillator*.

ghost: The slight *pre-echo* heard on a record one revolution before the beginning of a loud band, or just after the loud band stops. The waveform carved by the cutting stylus in the modulated groove deforms the adjacent blank groove, resulting in a faint repeat of whatever the modulated groove contains. Analogous to *print-through* on magnetic recording tape.

gink: In film, to screw up.

G

glass master: A glass disc with a light-sensitive coating, whose surface can be etched with pits by a laser beam as modulated by an audio signal. This surface is then sealed with a coating of silver. Used as a master for the dies from which CDs are eventually pressed. The laser device that burns the pits in the coating on the glass surface of a CD is called a *glass master lathe*.

glide: A function, also called *portamento*, in which the *pitch* moves continuously from one note to the next, such as is possible on a violin or trombone, instead of jumping over the intervening pitches, such as is necessary on a keyboard instrument or woodwind.

glissando: A direction for piano or harp by sliding the fingers over the keys or strings. Only the C_{maj} and pentatonic *scales* can thus be played (on the white and black keys, respectively). Compare with *arpeggio, portamento*.

global: Pertaining to or governing all of the operations of a digital synthesizer, module, or other instrument.

GM: See *General MIDI*.

gnat's nut: See *RCH*.

gobo: Any kind of moveable sound-absorbing surface or panel used in recording sessions to acoustically separate sound sources. See *baffle*.

grabbing: The process of importing digital audio from an audio CD directly onto a computer's hard disk. Also called, by the verbose, *digital audio extraction*.

grain: (1) A subtle type of distortion found in some audio devices, mostly digital devices but sometimes also power amplifiers, possibly due to *crossover distortion*. (2) The ferrous particles on a tape which determine the amount of distortion caused by the *Barkhausen effect*. Digital media have grain only if the *reconstruction filters* are badly designed. See *granulation*.

granular synthesis: A sophisticated form of additive synthesis, combining sound elements called *grains*, which have a specific duration (typically 1-50ms), waveform, peak amplitude, and bell-curve amplitude envelopes. Hundreds or thousands of grains are combined per second to form an *event*. An event has such attributes as start time, duration, initial waveform, waveform slope, initial center frequency, frequency slope, bandwidth, bandwidth slope, initial grain density (number of grains per second), slope, initial amplitude, and amplitude slope. Essentially, a sound event is sliced into time *screens* that contain the amplitude and frequency dimensions of hundreds of events. These screens are assembled into *books* that define a complete sound object.

granulation: An *aliasing* type of distortion in digital audio systems due to the uncertainty in the level of the samples is known as granulation, also called *quantization distortion*. If the sampling rate is an exact multiple of the input tone frequency, granulation results in *harmonic distortion*, i.e., the distortion components are at multiples of the input frequency. If not, the granulation resembles random noise, in which case it may properly be called *quantization noise*. See *modulation noise*.

graphic editing: A method of editing parameter values using graphic representation (for example, of envelope shapes) displayed on a computer screen.

G

graphic equalizer: A graphic equalizer can be recognized by the row of faders across the front panel, each fader controlling its own narrow section of the audio spectrum. Other than the highest and lowest faders, which control *shelving filters*, each of the filters in a graphic equalizer is a fixed-frequency *bandpass filter*, where the range of each fader is fixed, and the width of each individual band of a third-octave equalizer is actually wider than a third-octave. See *equalizer*, *parametric equalizer*.

ground: Refers to a point of, usually, 0V, and can pertain to a power circuit or a signal circuit.

ground lift/lifter: The practice of or a device for disconnecting the shield on one end of a *balanced* cable to eliminate a *ground loop*. Sometimes in the form of a switch found on some audio adapter boxes or *DI* boxes, this switch or cable modification disconnects the chassis ground of a device.

ground loop: The situation which arises when two pieces of equipment, each having an established chassis ground internally connected to signal ground, are then connected via a *shielded cable*. This forms a relatively large loop from chassis ground to signal ground (shield), shield ground to signal ground, and chassis ground back to chassis ground. Because the electrical pathway formed in this manner has a finite impedance, a difference in potential may occur from one end of the loop to the other, allowing an AC-frequency signal to form in the circuit. This signal will manifest itself as a *hum* which can, in extreme cases, be louder than the audio signal. The solution is to break the screen connection between the two devices, ideally at the end of the cable that is plugged into a receiver, such as a mixer or amplifier with a *ground lifter*.

group delay: The rate of change of *phase* of the response of a device or a system as a function of frequency. A pure time delay, equal at all frequencies, gives a constant slope of *phase* versus frequency. If, in an audio component (frequently a *passive network*), this slope is not constant but varies with frequency, the component is said to produce *group delay distortion*. This is equivalent to a time delay that varies with frequency, called a group delay because the distortion occurs within a group of adjacent frequencies, but not over the entire spectrum. The audible result is a loss of precision in musical transients; they are spread out or smeared in time and a more diffuse stereo *image* results.

group fader: A control which sums and adjusts the output of several other faders which have been routed to that group. See *gang*, *grouping*.

grouping: A feature of some sequencing programs or mixers which allows for the assignment of several faders to a *group master fader* that controls the overall level for the group. The software analog of a hardware *gang*.

group master: See *submaster*.

grunge: See *mid-range smear*.

guard band: A narrow, unrecorded area between the recorded tracks of a magnetic tape in order to reduce crosstalk between the channels of the tape recorder, resulting in each track of the tape using slightly less than $1/n$ tracks-width of the tape.

guide vocal: In multitrack recording, a preparatory vocal track to serve as a template for the later recording of instrumental tracks, eventually replaced by a final version.

G

Guillotine splicer: A type of splicer for motion picture film and magnetic film that is generally used to assemble the workprint and edited soundtracks. For picture cutting, it slices along the frame line between images. A second blade can slice magnetic film diagonally to avoid pops on playback. While holding the two ends of picture or mag film to be joined in a sprocketed channel, non-stretching tape is applied, completing the splice. The editor can undo the splice if the result is not satisfactory, and may also reassemble the pieces in their original, or any other, order. Also called a *tape splicer*.

gun microphone: A highly directional type of microphone used for long-distance recording, e.g., for wildlife or surveillance. Also called a *rifle microphone*, *shotgun microphone*, or *interference microphone*.

H

Haas effect: The Haas Effect refers to the brain's ability to integrate *incident sound* and *early reflections* into a single sound. Those early arrivals which occur within the first 5-35ms that are not more than about 10dB louder than the direct sound will be combined and added to the first arrival and localized to its source. If the delayed sound is more than 10dB louder or the delay is greater than 35ms, the listener will perceive distinct echoes. This is a type of sensory inhibition which causes the response to the direct sound source to inhibit response to the reflections. Haas further noted the *Precedence Effect*: the position of a perceived sonic image created by two sound sources depends on both the arrival times and the relative levels. If two sound sources arrive at the same time and at the same level, human aural perception will image the sound toward the centerpoint between the two sounds. If two sounds are equally loud, the *image* will shift toward the earliest-arriving signal. Further, if one source is louder, the image can be moved back to the center between the two sources by adding delay, provided that the delay time is less than 25ms. Beyond 25-30ms, the ear begins to hear the delayed sound as a discrete echo and the image shift effect no longer works. This phenomenon is usually used to simulate a stereo image in a monophonic recording where the original signal is panned hard left and a copy is delayed (1-50ms delay) and panned hard right. The Precedence Effect may be mitigated by slightly attenuating the volume of the *dry* sound.

half-space loading: The placement of a loudspeaker against a wall or other hard, reflective surface. Such a placement typically improves the performance of a loudspeaker, enabling a wider dynamic range, as well as improving the amplitude response of the speaker. An alternative to placing the loudspeaker in a *free-field*.

half track: (1) A recording format in which two parallel tracks are recorded in a single pass on a $\frac{1}{4}$" tape, each track using slightly less than half of the tape width. On some machines, a very narrow track with *SMPTE* or other synchronizing information is recorded and reproduced in the *guardband*. In this case, the data is *frequency-modulated* onto a very high-frequency tone in order to minimize *crosstalk* or *bleeding* into the audio. (2) A tape machine which records on half of the tape width only. This allows the tape to be inverted at the end of its play time, doubling the recording time for a given length of tape. See also *two-track*, *four-track*.

half-step: The musical interval of a minor second in a *diatonic* scale, equal to 100 *cents*. In *equal temperament*, there are twelve semitones in each octave, so in the equal tempered scale, the minor second has a frequency ratio of the $\sqrt[12]{2}$, or about 6%. In *just intonation*, the minor second has a frequency ratio of $\frac{15}{16}$. Also called a *semitone*. See *scale*.

half-time: See *alla breve*.

handles: Sound sections between works in a *production track* that enable the re-recording mixer to cross-fade smoothly between shots with different *backgrounds* and / or *room tones*.

handshaking: In data transmission, the process of checking that a receiving device is ready to receive, or that a transmitting device is ready to transmit. Also, the method whereby such checking takes place. In MIDI, handshaking occurs in System-Exclusive, where messages are sent between two devices to ensure that both are present and that both have received or transmitted blocks of data.

hang: In film, the act of playing back a given element during a mix for the purposes of adding the track(s) to the mix. "We won't premix the Foley cloth but will hang it at the final mix instead."

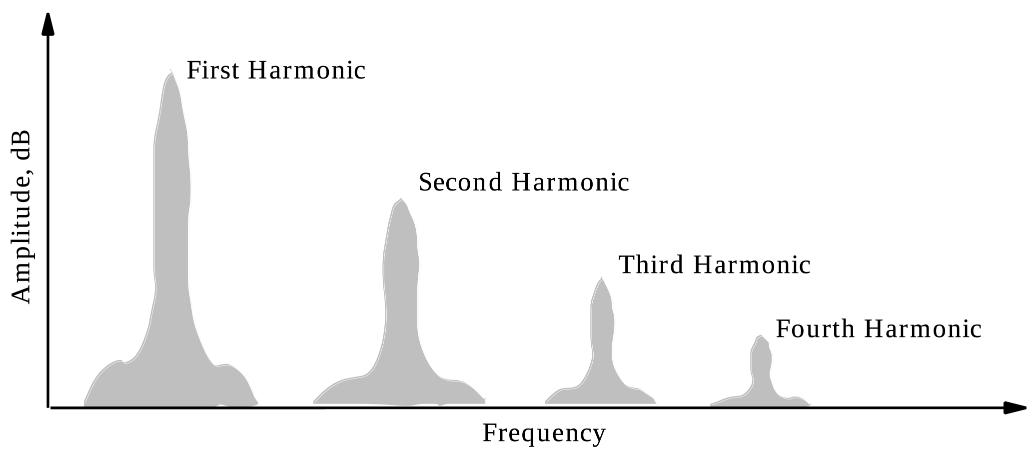
H

hangover: The *resonance* which continues in a loudspeaker cone after the input signal has stopped. See also *damping factor*, *impulse response*, *ringing*.

hard disk recording: A computer-based form of tapeless recording in which incoming audio is converted into digital data and stored on a hard disk. Sort of the digital counterpart to *direct-to-disc* analog (vinyl) recording, but being digital, the recording can be edited.

hard knee compression: A characteristic of certain designs of a *compressor* wherein nothing happens to an input signal until the signal reaches the threshold limit, but as soon as it does, the full level of gain reduction is applied, as determined by the ratio control setting. A graph of the input gain against the output gain will show a sharp change in *slope* at the threshold level. Compare with *soft knee compression*.

harmonic: A frequency that is a whole-number multiple of the *fundamental frequency*. For example, if the fundamental frequency of the sound is 440Hz, then the first two harmonics are 880Hz and 1.32kHz. A harmonic is the same as a *partial* where the partials exhibit the property that the *overtones* are mathematical multiples of the fundamental frequency. See *harmonic series*, Appendix C.



Harmonic Spectrum

harmonic distortion: The onset of harmonic distortion is the displacement of energy from a single frequency to its *harmonic*s. The presence of harmonic frequencies added to an output signal by an electrical circuit or speaker, generally undesirable, caused by the system not being perfectly *linear*, such as when an amplifier is operated in a nonlinear portion of its *transfer curve*. It is expressed as a percentage of the original signal:

$$\textbf{THD} = \frac{\textbf{(Total Signal - Fundamental Signal)}}{\textbf{Total Signal}}$$

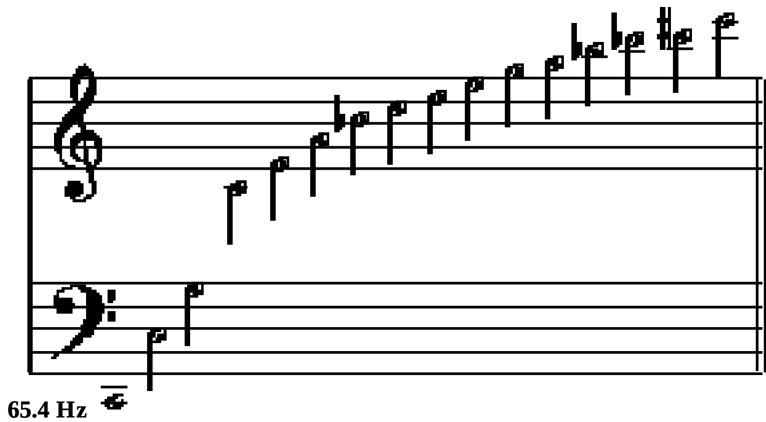
In a perfect audio device, such as an amplifier or tape recorder, the output signal would be a replica of the input signal with no changes except possibly the amplitude of the signal. See also *doubling*.

harmonic enhancement: A technique used by *aural enhancers*. See *harmonic synthesis*.

harmonic envelope: The natural *decay* in the *harmonics* of a natural instrument over time.

H

harmonic series: A set of all of the frequencies which are an integral multiple of the frequency of the lowest tone, or *fundamental*. See *harmonic*, *partial*. Humans perceive a harmonic series as a single *pitch* whose tonal quality is determined by the exact mix of related harmonics present. Below are illustrated the first sixteen harmonics in the harmonic series for the fundamental, C=65.4Hz. The notes indicating the 7th, 13th, 14th and 15th harmonics occur slightly flat or sharp of the notated pitch.



The first sixteen harmonics for the fundamental C=65.4 Hz

harmonic series tuning: A tuning system which is based on the first sixty *harmonics* of the *tonic*, resulting in a tuning not based on the usual diatonic scale. There are more notes per octave as the tuning progresses up the harmonic series; the top 32 keys of a keyboard cover one octave in pitch.

harmonic structure: The sequence of *chords* used in a piece of music.

harmonic synthesis: A technique used by *aural enhancers* which creates new high-frequency harmonics not present in the original recording. Adding a small amount of carefully controlled distortion can make a sound quality appear cleaner and more detailed. This happens by sending some dry sound to a side-chain *highpass filter*. The output of the filter is processed dynamically to add *phase-shift* and create synthesized HF (only) harmonics related to the dry signal.

harmonizer: See *pitch-shifter*.

hat: See *top hat*.

haystack filter: See *bell filter*.

H

HDCD: High Definition-Compatible CD. A trademark *dithering* process by Pacific Microsonics. The "HDCD process effectively cancels the additive distortions and simultaneously provides additional data to reduce the subtractive distortions" and is compatible with existing consumer digital playback equipment, claiming that there is a clear improvement in the fidelity of the conventional CD. The process works by converting an analog signal into a digital signal with a word length of "longer than 16 bits" and at a sampling frequency of "greater than 100kHz." These data can then be encoded into the standard CD format, or used with 20- or 24-bit recording/editing hardware/software. When used with an HDCD decoder, the reconstructed signal is output at the appropriate > 16-bit, > 44.1kHz format.

HDTV: High Definition TeleVision. A term designating any television system using many more than the standard number of lines per *frame* specified in the NTSC, PAL, or SECAM systems. Experimental HDTV systems have been developed to provide high-resolution computer animation for motion pictures, flight simulators, etc., but are unlikely to be used for broadcast any time soon due to their inherent incompatibility with existing broadcast standards. The HDTV standard includes *5.1* audio, using *AC-3* encoding.

head: (1) On a tape recorder, an electromagnetic *transducer* that (i) converts electrical energy in the signal into a magnetic field that induces magnetization in the tape, or (ii) produces an electrical signal in response to the varying *remanent* magnetism stored along a passing length of tape. See *erase head, playback head, record head, sync head*. (2) In general, the transducing mechanism used in recording or playing back signals on various media, e.g., the cutting head of a record mastering lathe, the optical head of a motion picture projector, etc.

head gap: See *gap*.

head losses: Limitations in the *frequency response* of the signal a tape head can transfer to or read from tape due to its inherent design or construction.

headphone box: See *cue box*.

headphone mix: See *cue mix*.

headroom: The amount of additional signal above the nominal *input level* that can be sent to a module before *clipping* distortion occurs. On a digital tape, input levels are set very low, -15VU to -12VU, to allow adequate headroom for occasional input peaks that might exceed -12VU. See *dynamic headroom, dynamic range, overs*.

head shield: A metal shield installed around as much of the *playback head* as is possible, in order to minimize distortion due to *EMI*.

heads-out: A tape recording which has been rewound and is ready to play. It is generally considered best for long-term storage to leave recordings *tails-out* for minimum *print-through*.

head stack: The assembly of tape heads in a magnetic recorder. The head stack normally consists of an erase head, a record head, and a playback head. Also called a *head block*.

H

helical scan: A type of videotape, data, or audio recorder in which the tape is wrapped around a large rotating drum, on which the actual record and playback heads are mounted. Since the heads rotate quickly and write parallel tracks at a very small angle with respect to the tape path, the signal written on the tape is may times the actual length of the tape itself. Thus, helical scan recording offers very high resolution at low tape speeds. Almost all consumer and professional videotape formats employ the helical scan principle, largely replacing the *quadruplex recorder*.

Helmholz resonator: A structure used in *loudspeaker* systems which is designed to *resonate* at a particular frequency. Because of the particular design of the resonator, the sound at the tuned frequency is dampened, and two bands, one each of higher and lower frequency, are produced, extending the *bass* response of the loudspeaker.

Henry: (1) See *Jecklin disk*. (2) A measurement for *inductance*.

Hertz (Hz): The unit measurement of *frequency* which equals one cycle per second, named after German physicist H.R. Hertz. The frequency range of human hearing is from about 20Hz-20kHz.

heterodyne: To mix two frequencies together producing the sum and difference of the two input frequencies; any information contained on either original frequency is continued in the sum and difference frequencies. Heterodynes are used as the basic design for all AM, FM, amateur radio, CB, TV, radar, and satellite systems. See *amplitude modulation*.

HFS: Hierarchical File System. A Mac-specific logical file format for CDs. CDs written in HFS cannot be read on PCs. Compare with *ISO 9660*.

high-fidelity: Refers to the reproduction of sound with little or no *distortion*. At least 15kHz of audio *bandwidth* is required for stereo high-fidelity.

hi-fi video sound: The result of encoding the stereo soundtracks input to hi-fi type VHS or Beta format videotape recorders on an *frequency modulated* carrier wave. This information is recorded along with picture data via the video record heads. Reproduction of hi-fi sound approaches digital quality audio.

high band: A type of video system in which the picture information is encoded on a much higher carrier frequency than early color video systems; the broadcast standard currently in use.

high-frequency compression: See *HX/HX pro*.

high-output low-noise (HOLN): A type of *magnetic recording tape* with very high sensitivity to applied magnetic fields, and with a very high *S/N ratio*, commonly used in professional audio applications.

highpass filter: A filter that attenuates the frequencies below its *rolloff frequency*.

hiss: Audio noise that sounds like air escaping from a small aperture. See *Barkhausen effect*.

hit point: See *cue*.

hold time: An *envelope* parameter that specifies how long the *attack* segment of an envelope is to be held at full level.

H

hole-in-the-middle: An undesirable effect due to an extreme angle used with a *coincident pair* where the stereo *image* is all left and right, with very little sound in the center. Or, a similar phenomenon created by a surround system where the loudspeakers are too far apart to deliver balanced sound adequately to all seats in a theater.

hook: In popular music, the short melodic idea designed to be instantly memorable. It is often used for the chorus of a song, as well as for a *fade*.

horn: A type of loudspeaker enclosure named for its characteristic shape, with the speaker itself mounted in the narrow end of its tapered interior surface. Because the sound waves emanating from the speaker itself are internally enlarged before they exit from the larger end of the tapered surface, horn enclosures are highly *efficient*. Also, any horn-shaped device placed in front of a speaker to disperse sound.

horn tweeter: A high-frequency loudspeaker which has a horn-shaped flare fixed to the front in order to increase acoustic efficiency and better control the *directivity*.

hose: Slang term for an audio cable, e.g., a microphone cable or a *snake*.

hot: In a *balanced line* system, the conductor which carries the in-phase component of a signal. For example, pin 2 of an *XLR* connector.

hot hole: In film, slang for the projector gate itself, where the picture start mark is threaded up at the beginning of a session.

house mix: An output on a *sound reinforcement* control console that is used to feed the power amplifiers for the loudspeakers in the venue, usually highly equalized to correct for *house modes*.

house mode: The unique acoustic profile of a particular performance or recording venue. It is necessary to know the *reverberation* patterns within the space so that microphones, speakers, acoustic *damping*, etc. can be appropriately placed to produce or simulate the spatial and *ambience* effects desired, or to correct for *room modes*.

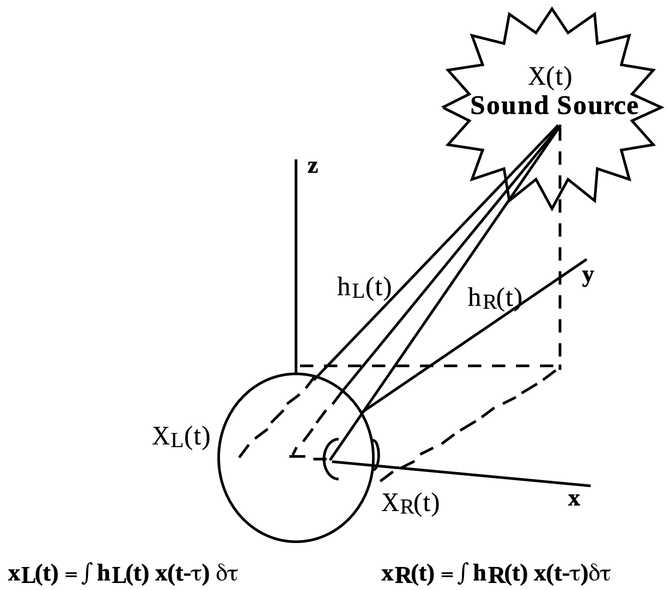
house sync: Also called *black-burst*.

HPF: See *highpass filter*.

HRIR: Head-Related Impulse Response. See *HRTF*.

H

HRTF: Head-Related Transfer Function. A function used to find the sound pressure that an audio source produces at the ear-drum. This is described by the impulse response from the source to the ear drum, called the Head-Related Impulse Response (HRIR), and its *Fourier transform* is the HRTF. The HRTF captures all of the physical cues to the source localization; once the HRTF for the left and right ears is known, accurate *binaural* signals may be synthesized from a *monaural* source.



Head-Related Transfer Function

HSS: HyperSonic Sound.™ A *loudspeaker* technology developed by American Technology Corporation. This design produces audio by mixing an *ultrasonic* carrier with audio *sidebands*, in much the same manner of heterodyning. The mixing takes place in the air, relying on the nonlinearity of the atmosphere. The resultant sound is actually not generated at the *transducer*, but all along a projected column of ultrasonically vibrated air in front of the transducer as a conversion by-product of the interaction of the ultrasonic waves. Inaudible ultrasound energy is projected, which in turn emerges as *AF* sound from adjacent reflective surfaces. An acoustical sound wave is created in the air molecules by down-converting ultrasonic energy into the AF. This process supposedly is free of the problems of conventional speaker *voice coils*, *cones*, *crossover networks*, or enclosures. In addition, because sound is generated along the entire length of the projected column, there is minimal (1dB) amplitude loss as a function of distance from the transducer, across an average-sized room.

hub: The cylindrical plastic or metal center of a tape reel to which the tape is attached and around which it is wound.

hum: Audio noise that has a steady low-frequency pitch, typically caused by the effects of induction of nearby AC lines or leakage of AC line frequency into an amplifier's signal circuits, usually at 60Hz or 120Hz.

humanize: To introduce slight, random variations to the timing, velocity, duration and possibly other parameters of a track to make *quantized* tracks sound more natural. See *percentage quantization.*

H

hum switch: A switch found on some audio equipment, such as amplifiers for musical instruments, which reverses the neutral and hot leads of the power cord in order to reduce *hum*. The ground lead is unaffected.

HX/HX Pro: Headroom eXtension. A special circuit developed by Dolby Labs to reduce the tendency in cassette recorders toward *self-erasure*. In magnetic tape recording, loud, high frequencies in the signal look like *bias* to the tape which will tend to erase the signal as it is being recorded. The effect is called *high frequency compression*. The HX Pro system senses the level of high frequencies and reduces the level of bias dynamically.

hybrid amplifier: An amplifier that uses a combination of transistors and tubes, supposedly combining the best characteristics of each.

hypercardioid microphone: The narrowest of the *unidirectional* patterns, the hypercardioid is a variation on the *cardioid* microphone pick-up pattern which is most sensitive at the front and sides, while rejecting sounds entering 110°-250˚ to the rear, with a small lobe of sensitivity at 180˚ to the rear. The pick-up pattern of a hypercardioid is narrower than that of a *supercardioid* and is somewhat similar to that of a *figure-eight* mic, but the response is asymmetrical in that the hypercardioid has greater sensitivity to sound arriving at the front of the capsule than to sound arriving a the rear. See *acceptance angle*. Also called a *cottage loaf mic* in the UK, for reasons related to bread.

hyperinstrument: An instrument which has had its sound-producing capability enhanced electronically.

hysteresis: In magnetic tape recording, the hysteresis inherent in the process of magnetizing the tape represents a large nonlinearity, and this causes *harmonic distortion*. The use of *bias* in the recording process reduces the effect of the hysteresis, and hence, reduces the distortion. Compare with *Barkhausen effect*.

hysteresis loop: The graph of applied magnetic force vs. *remanent* magnetism. One measure of a specific recording tape's performance.

I

IDE: Integrated Disk Electronics. A standard interface *bus* in PCs, most commonly used for hard disks. EIDE is Enhanced IDE, somewhat faster than the original IDE specification. This later evolved into ATA (Advanced Technology (AT) Attachment) and UltraATA. This evolution is fairly parallel to the *SCSI* bus technology used by Macs.

IEC characteristic: The European *pre-emphasis* and *de-emphasis* equalization standard for magnetic tape recording.

IEM: In-Ear Monitor. Earphones used by musicians when recording to hear a special *cue mix*, *overdubbing*, or during a performance to better hear other musicians. Sometimes used instead of *stage monitors* to reduce problems of feedback or to provide each musician with a separate *monitor mix*. See *earwig*.

IFPI: International Fédération Phonographique Industrie. The European equivalent of the *RIAA*.

IIR: Infinite Impulse Response. See *FIR*.

IMA: Interactive Multimedia Association.

image: (1) The apparent relative placement of individual sound sources, as imagined by a listener of recorded audio, created during the recording and mixing processes, as well as by the final format of the media, e.g., *stereo*, *surround-sound*. See *imaging*(1), *Haas effect*. (2) See *imaging*(2).

image shift: In *multichannel* sound reproduction, a change in the apparent left-to-right position from which a particular sound seems to emanate.

imaging: (1) The ability to localize the individual instruments, voices, or other sound sources when listening to a *stereophonic* recording is called imaging. Accurate imaging with two channels is almost impossible, requiring both channels to have identical *gain* and *frequency response*, the two loudspeakers to be within 1dB of each other in frequency response and the *phase* must be identical. In addition, the listener must be precisely between the two speakers. The lack of accurate imaging with traditional, two-channel stereo has lead to three-channel (LCR) and higher-channel audio recording and reproduction in an attempt to improve the listening experience. Contrast with *stereo spread*. (2) The resulting output of a *D/A converter* is a stair-step waveform which contains a great deal of high-frequency distortion called *images*. To reconstruct a smooth replica of the original signal, the stair-step is passed through a steep lowpass filter called an *anti-imaging*, or *reconstruction filter*. See *quantization error*.

J

J-card: The printed card inserted into a cassette tape box, so named because it resembles a "J" when viewed from its end.

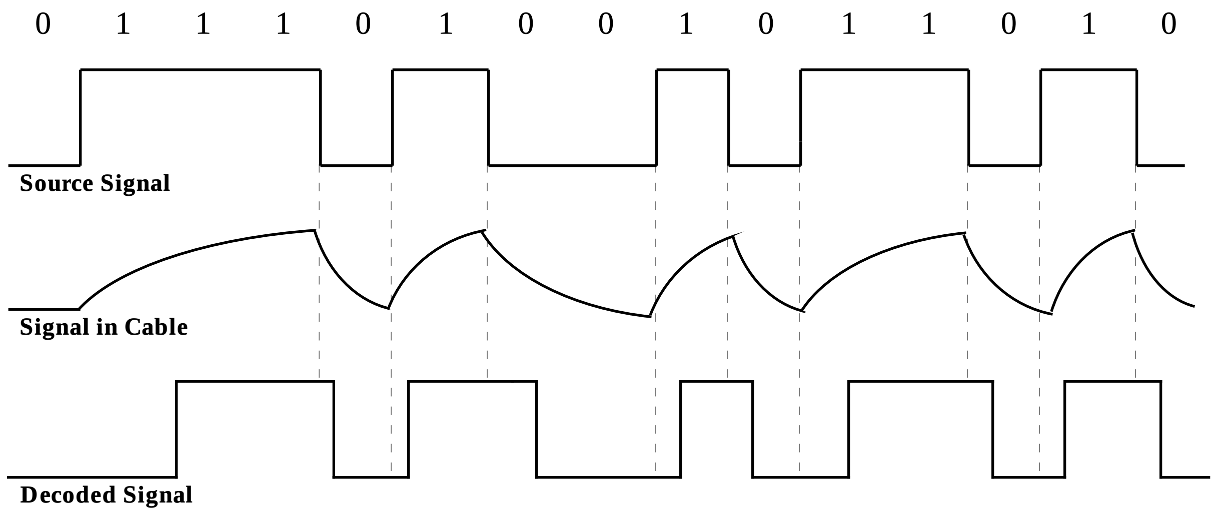
jack: A female connector, frequently mounted on the chassis of an audio device, which serves as a receptacle for the male connector, called a *plug*, on the end of an audio cable.

jam sync: A family of techniques in which a *synchronization* device reads *timecode* and regenerates new timecode that may not have the same address as the original timecode. Usually, the transferring of a timecode and *user bits* from an external reference source to a *SMPTE timecode* generator, either once, called *one-time jam sync*, which will align two codes at one *frame* only, allowing each to proceed at its own internal rate from that moment forward, or continuously, which will force the generator to mimic the timecode numbers of the reference source continuously. Timecode is read up to the last good address, then the generator uses the next consecutive address to generate a new timecode, called *Jam Timecode*, or *JTC*. The process of regenerating SMPTE timecode to a previous reference: the source timecode goes to the timecode synchronizer which reads it and regenerates a new copy. If there is a *dropout* in the timecode, the synchronizer will *freewheel*, continuing to create timecode to cover the dropout. Used to recover from dropouts or non-continuous timecode caused by editing.

Jecklin Disc: Also called a *Henry*. A disc usually made of plywood, typically 10"-12" in diameter. There is a mounting for the microphone on one edge, and the disc is covered in an absorbent material. The concept is that the microphone spacing matches that of human ears and the disc provides the sound-shadowing effects of the head, so the ensemble should be able to capture sounds in the microphones which will most closely match what a person would hear, effectively transporting the listener to the recording venue. Results are variable. See also *ambisonic.*

J

jitter: (1) The lack of precision in digital sampling times, leading to *amplitude errors* in signals with rapidly changing amplitudes, resulting in distortion of the sampled signal which rises with frequency. The starting time of the sampling aperture is the non-zero time that it takes for *the sample-and-hold* circuitry to determine the level of the signal waveform and to hold this level until the next sample is called for. Because the time required to establish a new value of charge depends on the amount of change in the signal level from one sample to the next, the aperture time will vary with the rate of change in the signal level, increasing for high-level, high-frequency signals. The starting time of the sampling aperture is also slightly uncertain, and this is called jitter. Also called *sampling offset uncertainty*. See *aperture time errors*. (2) Timing errors introduced into channel-coded interfaces such as AES / EBU and S / PDIF where the word clock is embedded within the data stream. Cable capacitance reduces HF signal, resulting in founded corners and sloping edges, as opposed to a sharply divided on / off pulse wave. As the word clock timing is defined by the midpoint of the pulse wave, any strictly nonvertical slope creates timing uncertainty and, therefore, jitter.



Jitter Caused by Cable Impedence

Johnson noise: See *noise floor*. Johnson noise is the broadband *white noise* power associated with electrical *resistance* at temperatures above absolute zero. The Johnson noise level is the limiting minimum noise any circuit can attain. Also called *thermal noise*.

joint: In tape editing, the point of connection between two pieces of tape spliced together.

Joule's Law: A formula for converting *watts* into *amperes*:

$$\mathbf{P} = \mathbf{V} \times \mathbf{I}, \text{ or alternately, } \mathbf{I} = \mathbf{P} / \mathbf{V}.$$

JPEG: Joint Picture Expert Group. A body that defined a standard for data compression originally for still images. This has been extended to M-JPEG for use in random-access (non-linear) editing systems. Lossy, but generally acceptable. See *MPEG*.

JTC: See *jam sync*.

J

just intonation: A family of tuning systems that can have fewer than twelve tones per *octave* or many more than twelve. Just intonation is based on any pure, natural *scale* determined by the frequencies of the *harmonic series* of the *tonic*, not one that is artificially fixed by keyboard instruments. Purely tuned just intervals are almost never used in music because of their incompatibility with the octave. Scales in just intonation are never *equal tempered*, and vice-versa. See *syntonic comma*, *diatonic comma*, and *temperament.*

just noticeable difference: A *psychoacoustic* term which refers to the smallest timing difference the human aural system is capable of detecting between two sound sources, approximately 6μs. This is just an artifact of human hearing and is not related to the *Haas effect*.

K

k: Kilo. One Thousand. For example, 20kHz=20,000Hz.

K: (1) Often used in place of "k" for one thousand, but precisely means 1,024 bytes, or 2^{10} bytes, abbreviated KB. (2) Temperature measured by the Kelvin scale. The various film and TV lighting sources are always measured by the Kelvin scale, and most color film stocks are balanced to give proper color reproduction at either 3,200˚K (tungsten-halogen lamps) or 5,400˚K (noonday sun).

key: In music, the key is the pitch of the *tonic* of the musical *scale* used. Tonal music gravitates toward a home key, the tonic. Key is established by the use of a fixed *scale* of notes based on this tonic note and can be emphasized by other, related notes and by *cadences*. The tonic note or chord assumes greater importance than the others and leads, by extension, to a hierarchy of chords with the *dominant* (based on the fifth note of the scale) of particular significance. See *temperament*.

keyboard control voltage: The *control voltage* parameter that tells the signal-generating circuit exactly which key has been depressed.

keyboard rate scaling: See *envelope tracking*.

keyboard scaling: A function with which sound can be altered smoothly across the range of the keyboard by using key number as a modulation source. *Level scaling* changes the loudness of the sound, while *filter scaling* changes its brightness.

key code: See *keyboard control voltage*.

key follow: See *envelope tracking*.

keying input: In a signal processing or generating device, an input for a control signal that determines the type and amount of processing applied to the audio signal, or of the sound produced, respectively.

keying signal: The signal sent to the *keying input* of a signal-producing or signal processing device, which then activates the device.

key map: A keymap assigns a *sample* to each MIDI note or key on a keyboard. The map is set to respond to a specific MIDI *channel* so incoming MIDI notes on that channel trigger the samples assigned to them.

key numbers: Numbers on the side of film stock created during film manufacture that are visible on the developed negative and positive prints made therefrom.

key pressure: See *poly pressure*.

key signature: The group of *sharp* or *flat* symbols placed immediately after the *clef* symbol on the *stave* at the beginning of a piece of music, and at the beginning of every subsequent stave, to indicate the *key*. These sharps or flats are presumed to be active for the duration of the piece or section, unless cancelled either temporarily (for one bar or part of the current bar) by an *accidental*, or more permanently, by the placing of a new key signature.

keyboard tracking: See *envelope tracking*.

K

kirsch: In film, means when a director has requested a change in the sound and then gives his or her approval to what was, in fact, no net change, either deliberately or accidentally on the part of the mixers. Kirches can be self-inflicted as when a mixer adjusts a control when it is not in the signal path, or listens for a change while the *PEC/direct paddles* are in *playback mode*, as opposed to *input mode*.

KSHRFOO: The traditional first seven microphone input channels on a mixing console in a rock recording / SR set-up. By convention, these are for the Kick-drum, Snare, Hi-hat, Rim, Overhead left, and Overhead right mics on the drum kit.

L

L/A synthesis: Linear Arithmetic synthesis. A *sound synthesis* method developed by Roland that creates new sounds by attaching the *attack* portion of a sampled waveform to a simpler waveform. Human sound recognition is heavily influenced by hearing the *attack transient* part of a sound, but simple waveforms require less storage than samples. By combining the two, L/A synthesis is capable of relatively sophisticated sounds with modest data storage requirements.

labels: Special non-audio information encoded along with the audio in digital recording systems, used to encode information about the recording session, number of microphones used, dates, etc.

lacquer master: The disc produced from a master recording tape which is used to press vinyl copies.

land: (1) The flat area of vinyl between the grooves of a record. (2) The flat area between the laser-carved pits of a CD.

largo: Italian for "broad." A slow or stately tempo, 48-60 bpm.

later reflections: See *early reflections*.

Lavalier microphone: A small microphone, either *condenser* or *dynamic*, which can be easily hidden in a piece of clothing so as not to be seen by the camera. Also called a *peanut*.

layback: Transfer of the finished audio mix back onto the video *edit master*. See *layoff*.

layback recorder: A videotape recorder, usually 1" format, on which a mixed soundtrack with all *DME* stems can be *re-recorded* in *sync* with the edited video master. Because of its special purpose, a layback machine should have less *flutter* and higher quality audio heads and electronics than standard 1" video decks. Some layback machines designed especially for that purpose have no video reproduction capability at all. They merely read timecode and do an extremely high-quality job of recording audio, and nothing else. The layback process is also called *re-laying*. See *layback*, *layoff*.

layer: See *split point*.

layering: Sounding two or more voices, each of which typically has its own *timbre*, from each key depression. Layering can be accomplished within a single synthesizer, or by linking two synths together via MIDI and assigning both to the same MIDI channel.

layoff: Transfer of audio and timecode from the video *edit master* to an audio tape. See *layback*.

layover/layup: Transfer of audio onto hard disk or multitrack tape.

LBR: Laser Beam Recorder. The device used to create a *CD* master for duplication.

LC Concept: A system, developed by a French company of the same name, for implementing digital audio for cinemas. The system relies on the presence of an optical timecode on the film which is used to synchronize the digital audio soundtrack stored on a separate magneto-optical disc reader, i.e., the film carries no sound at all, allowing for multilingual presentation from the same film print. This also solves the problem of getting high-quality audio onto film.

L

LCRC: See *LCRS*.

LCRS: Left, Center, Right, Surround. The four playback channels used in 35mm motion pictures, now available on home hi-fi systems. L, C, and R speakers are located behind the screen. The S channel surrounds the audience and may be mono or encoded stereo. See *matrix*, *surround-sound*. Variants include LCRC, when the fourth track is to be assigned to the center, or even CCCC, as in a center-channel dialog *premix*.

lead sheet: An abbreviated musical *score*, consisting of a melody line with chord names or symbols, and sometimes including lyrics.

leader: Blank (unexposed) motion picture film attached to the beginning or end of a reel of film, usually used for threading a playback machine, and which contains information about the reel's content such as film title, reel number, etc. as well as the count-down section. Opaque leader is used in *A* and *B Rolls*, in editing *workprints* and *film soundtracks*, to fill spaces between specific sound *effects* or musical segments, or to fill in for picture or sound segments to be added later. See also *Academy leader, SMPTE Universal leader, plastic leader, fill leader*.

leadering: The process of removing the out-takes, count-offs, and noises between *takes* in a magnetic tape (and by extension, digital) recording. In analog magnetic tape recording, this process also involves inserting *leader tape* between songs.

leader tape: Nonmagnetic plastic or special paper tape that is spliced onto magnetic tape between musical selections and at the beginning and end of the magnetic tape, protecting the tape and delimiting the selections. Some leader is timed and has marks every 7½" or 15" to allow the tape editor to insert the desired time between selections.

lead-in: See *spiral*.

leakage: The pick-up of unwanted, *off-axis* sounds by a *directional microphone* due to the fact that its directional pattern is not ideal or that the microphones and/or instruments are not sufficiently isolated from one another, as in a multitrack studio recording. Also called *spill*.

learning curve: In mechanical or electronic systems controlled by computers, the computer's ability to learn the hardware/software, input/output environment and use this information to control the system's state.

LEDE: Live End Dead End. A commercial trademark used to indicate a particular acoustical design of a recording studio control room. In this design, the area around the monitors is made acoustically absorbent, or *dead*, while the area behind the listener's position is made reflective, or *live*, in an attempt to increase the accuracy of the reproduction. See also *ESS*, *RFZ*.

LFE: Low-Frequency Effects. The equivalent of the subwoofer designation for audio-for-video, where the low-frequency band between about 20Hz-120Hz is *matrixed* or channeled for replay. In home audio systems, the subwoofer will frequently contain LF information from the main channels in addition to the original LFE track. See also *in-band gain*.

legato: A musical effect whereby the *decay* of one note overlaps the *attack* of the next.

leger line: See *stave*.

L

Lemo: A Swiss company which makes high-quality, very dense connectors. Rarely used, Lemo connectors are found on some specialty audio equipment, such as *Soundfield* microphones (because of the large number of capsules) or compact mics which require a high density of pins in a small space. There is no standard for the pin-outs in Lemo connectors, a fact which contributes to their scarcity.

lento: Italian for "slowly."

Leq: Equivalent sound Level. The Leq of a sonic event is that constant *SPL* which has the same amount of energy as the actual event. Thus, the Leq is a long-term average, or integration, of an SPL. It is approximately the average of the powers of instantaneous levels taken at equal intervals over time during the measurement period. Leq is a convenient way of accurately measuring the level of a fluctuating sound over a range of a few seconds to several hours.

Leslie cabinet: A type of loudspeaker cabinet, developed by Don Leslie in the 1930's and used in electronic (especially Hammond) organs. The sound from fixed *transducers* is dispersed via a rotating horn or (for bass speakers) an aperture in a rotating chute. This causes a continuously varying *Doppler* shift of the pitches in the audio signal, which mixes, with some *phase cancellation*, to give a swirling, chorus-like effect.

Leslie simulator: An *effects* unit which is intended to create the effect produced by a *Leslie cabinet*. It is similar to a *chorus* unit, but produces a richer effect.

level: Loosely used when the *magnitude* of a signal is meant, usually *voltage*. Strictly speaking, the term should be reserved for the value of a *power* in dB. The measured level of an audio signal is the *amplitude* that is caused by the sum of the *powers* of all of the components of the sound.

level control: An *envelope* parameter which controls the level of certain synthesizer actions, such as the *sustain* portion of an *ADSR* envelope. Compare with *rate control*.

leveling: The use of a *compressor* set to high ratios and very slow *attack* and *release* times. With a digital recorder, it may be beneficial to have some kind of leveler followed by a processor that does peak-limiting.

level scaling: See *keyboard scaling*.

level-sensing circuit: An electronic circuit that generates a *control voltage* in proportion to signal level. This control voltage can then be used to affect the amount or type of signal processing done by a separate device. Also called a *detector*.

LFE: Low Frequency Effect (film) or Low Frequency Enhancement (audio). The *subwoofer* channel signal in a *5.1* surround mix. See *in-band gain*.

LFO: Low Frequency Oscillator. An oscillator whose output is *infrasonic*, typically used as a control source for modulating the sound to create *vibrato*, *tremolo*, trills, and so on. Unlike a normal oscillator which produces audio signals, an LFO is a generator module that produces a modulation / control signal. The LFO's signal output is in the form of a slow, periodic waveform, usually less than 20Hz. The most common parameters found in the LFO are depth, frequency (rate control) and *waveform selection*. See Appendix C.

L

LFOA: See *LFOP*.

LFOP: Last Frame of Picture. Film acronym for the length of a given reel of film, usually connoting the head *leader* up to and including the last frame of the reel. Because it is standard to start counting with the *Picture Start* from of the leader as 0000+00 (zero feet, zero frames), the actual running time of a reel can be calculated by subtracting 11+15 (eleven feet, fifteen frames) to account for the 12-foot, 8-second leader. The *two-pop* is at 0009+00, and the first frame of picture of a reel is at 0012+00, sometimes referred to as *LFOA*.

librarian (software): Allows for computerized storage and organization of MIDI information for large numbers of synthesized or sampled sounds. Information is organized to be specific to synthesizer manufacturers' protocols. Librarian software sends *patch parameter* instructions to the synth via a MIDI cable. See *editor/librarian*.

lift: A section of a longer piece of music which may be edited out and used independently. For example, a musical phrase which is part of a longer piece of commercial music which may be used for use for another purpose than which it was originally written.

lifter: A tape transport's head-lifter mechanism. Tape machines normally lift the tape off the heads when in fast-forward or rewind mode. The synchronizer intelligently controls the machine's lifter operation to read *timecode* when required.

light metronome: A metronome which silently marks beats by flashing a light on and off, as opposed to audible clicks, to mark the tempo.

Lightpipe: A *serial*, *multiplexing*, eight-channel interface for digital audio on a single fiber-optic cable, terminating in a proprietary connector. The Lightpipe was invented by Alesis to connect its ADAT *MDM*s. The data rate is 256 times the *sample rate*, or four times the data rate of *AES/EBU* or *S/PDIF*. See also *TDIF*.

light valve: The mechanism which controls the intensity of light or the area on which light falls in the making of an *optical track* for a *film soundtrack* from the finished mix. For variable-density tracks, it consists of a narrow slit whose width is varied by the waveform reproduced from the mix, and which in turn modulates the width a beam of light that is focused on a continuously moving strip of photographic film.

Lightworks: A particular brand of *nonlinear* picture editing system. See *digital dubber*.

LIMDOW: Light Intensity Modulation Direct OverWrite. A format for *MO* disks where the direct-overwrite technology eliminates the need for an erase cycle and allows for the writing of new data directly over existing data, with the result that the burst transfer time is cut in half.

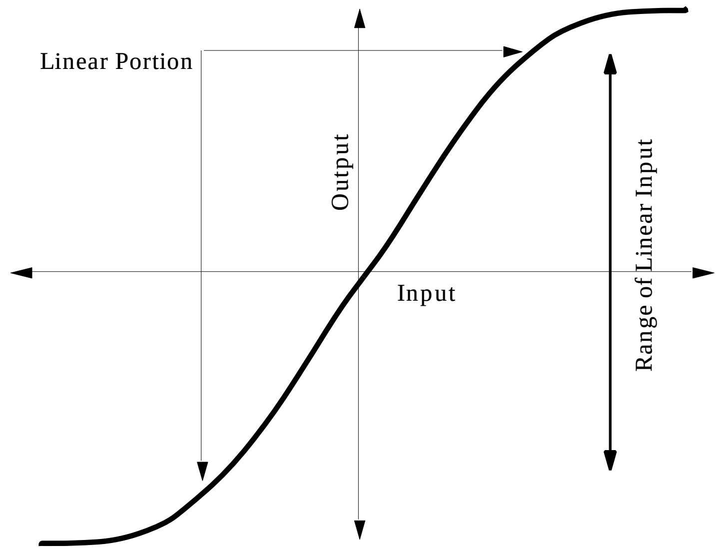
limiter: A special type of *compressor* which prevents the signal from exceeding a certain preset threshold setting, no matter what the input signal level may be, by using compression ratios of 20:1 or greater. Limiters are sometimes used in front of power amplifiers to prevent high-level signals from causing distortion. See *compressor/limiter*. Called a *clipper* in Europe.

line: (1) A signal path or actual cable through which a signal passes. (2) One horizontal scan of the raster in NTSC, PAL, or SECAM video signals. See *field*(4).

L

line amplifier: Now, any amplifier with a *line-level* output and an output *impedance* of approximately 600Ω.

linear: (1) A system is said to be linear if it meets the conditions of proportionality and additivity: if its output level changes smoothly in proportion to input level changes, and if input x causes output X and input y causes output Y, then $x + y$ at the input must cause $X + Y$ at the output. Most tests in audio including *frequency response, gain, phase, impulse response,* etc. assume linearity.



Linear

(2) Uncompressed, i.e., an audio file that has not been processed by some kind of *compression* algorithm, such as *ADPCM*. (3) A process which works in a sequential fashion, such as magnetic tape recording, playback the or editing tape media, etc., as opposed to a sequence of steps which can be taken in any order and / or in any location, such as the *random-access* editing and playback processes which are made possible by digital storage technology.

linear distortion: Any type of distortion that a linear system is capable of producing, as opposed to *nonlinear distortion*. Some types of linear distortion are *frequency response errors* and time-delay errors such as *phase-shift*.

line input: Any set of input terminals of an audio device designed to accept *line-level* signals, or signals above about 25mV RMS. Normally high *impedance* and, therefore, not suitable for most microphones.

line-level: The average audio voltage level of a signal at a particular point in an audio system above 25mV *RMS*. The output level of a preamp is typically line-level, and the input level of a power amplifier is line-level. In home or semi-pro equipment, the input or output operating level is usually -10dBV. In commercial audio systems, line-level is metered with a *VU meter*, where 0VU corresponds to 0.775V RMS of a signal. The line-level in pro audio systems may be +4dBm (1.23V RMS) or (archaic) +8dBm (1.95V RMS) or even +20dBm (~9V). Typical line-level audio signals include *synthesizer* outputs, *mixer* outputs, and *effects* outputs. As opposed to *mic-level*.

L

line-matching transformer: An electronic component that matches the output *impedance* of one device with the input impedance of the next device in a signal path.

line pad: A *passive* attenuation *network* that can be inserted in a *line*.

line-up: The procedure carried out to ensure that recording, editing, playback, amplification, etc. equipment works to the highest possible standard. It consists of systematic adjustment of the equipment according to a schedule and may involve specialized calibration and test apparatus such as a multimeter, tone generator, oscilloscope, etc.

line-up tone: (1) Also called a *reference tone* or *reference frequency*, a *sine wave* used for *servo* control, such as on a *sync tone*. See *vari-speed*. (2) A sine wave tone at one of a range of standard frequencies (usually 100, 1,000 and 10,000Hz) It is set to zero-level and is intended to be used for calibration, such as during a *line-up* procedure. The *APRS*-specified line-up tones for *magnetic tape recording* are:

Length	Tone	Calibration Purpose
20”	1kHz at 0VU (0dB)	Maximum level check
20”	1kHz at -10dB	Calibrate the -10dB level
20”	10kHz at -10dB	Azimuth line-up check
20”	100kHz at -10dB	EQ alignment

link: See *track-at-once*.

lip ribbon: A ribbon microphone with a guard which is placed on the upper lip. The proximity of mouth and microphone makes it useful in situations with high background noise, e.g., battlefields or boxing matches.

lip sync: The process of matching dialog sound to the picture. See *ADR*.

Lissajous: See *X/Y function*.

little dipper: Nickname for a popular *dip filter* previously manufactured by UREI.

Little Old Ladies with Umbrellas: Film sound expression for how loud a film can be before the movie patrons will complain. The effect is, therefore, that the top end of the dynamic range available to mixers is not necessarily defined with regard to a theater’s ability to reproduce a mix. See also *popcorn noise*.

live: (1) Acoustically reflective, as opposed to *dead*. See *LEDE*, *reflections*. (2) In electrical systems, a conductor which carries current. (3) A broadcast which is transmitted as it happens, i.e., in *real-time*.

live side: The side of a microphone which is most sensitive to sound. See *acceptance angle*.

live-to-two-track: See *direct-to-two-track*.

Lmax/Lmin: Lmax/Lmin are measurements of the *dynamic range* of a recording, Lmax obviously representing the maximum measured *level* of the recording, and Lmin, its minimum-level counterpart. The dynamic range of an audio signal is Lmax-Lmin.

L

load: (1) Any component or device that consumes power produced by a separate source. Or, to connect such a device to a power source. (2) To copy the contents of a file, database or program from disk or other storage medium into memory.

loading: Placing a *resistive* load across a line, and generally one that is of lower *impedance* than the line or device to which it is connected. This draws additional *current* from the preceding device, and can cause electrical power capacity problems.

load resistor: (1) A simple resistor placed across a transmission line in order to decrease the *impedance*, generally for *impedance-matching* purposes. (2) A resistor wired across the outputs of a power amplifier, simulating the impedance of a speaker.

lobes: In a mic's *polar pattern*, the expanding curves represent the maximum value for each direction of highest *sensitivity*. For example, the bi-directional polar diagram of a *figure-eight* microphone shows two equal-sized lobes 180˚ apart.

local control: With Local Control on, playing a synthesizer or sampler does two things: it triggers built-in sound generators and sends data to the MIDI Out. With Local Control off, the keyboard still sends data to the MIDI Out but does not drive the internal sound generators, which now respond solely to data appearing at the MIDI In. In other words, Local Control off disconnects a synthesizer's keyboard from its sound generator, while leaving them both active for MIDI purposes. See *MIDI mode*.

local/remote switch: The switch on a synthesizer that selects whether tones will be generated in response to its own keyboard, or from a remote device via MIDI.

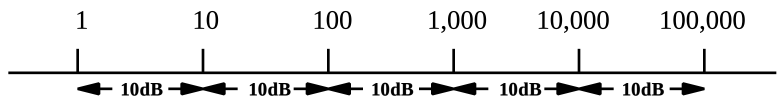
locate point: See *autolocator*.

location sound: Sound recorded and / or mixed on location during the film or video shoot; also known as *production sound*, *live sound*, *location recording*, and *live recording*.

L

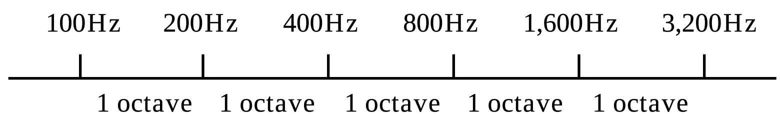
logarithmic: Having to do with the logarithms of numbers rather than the numbers themselves. In graphs of audio phenomena, frequently the log of amplitude is plotted versus the log of frequency. The common log of a number is the power to which the number 10 must be raised to obtain the number. A log scale is a scale where distances are proportional to the logs of the represented numbers, while a linear scale has distances proportional to the numbers themselves. See Appendix A.

Linear Amplitude Scale



Logarithmic Amplitude Scale

Linear Frequency Scale



Logarithmic Frequency Scale

Logic 7: Differing significantly from the discrete *5.1 surround-sound* formats, Logic 7 is a *matrix*-surround format with full-bandwidth channels. Logic 7 uses a proprietary *decoder* to combine data from a discrete five-channel digital mix into two channels, thus Logic 7 is known as a *5-2-5* matrix. Additionally, the matrix can decode to seven channels instead of five, in which case the matrix creates two side loudspeaker channels, moving the rear channels completely to the rear. Logic 7-encoded material can be played through conventional two-channel systems, as well as ProLogic- and Dolby Surround-encoded systems, although the encoded material will sound best when replayed through a Logic 7 decoder.

logical editing: To set up note criteria (such as pitch range, velocity range, duration range, placement within a measure, etc.), to which digital editing operations (e.g., cut, transpose, quantize, etc.), will apply. Also called *conditional editing*, *change filtering*, *selection filtering*, *split notes*.

longitudinal timecode (LTC): Refers to *SMPTE timecode* recorded on one of the audio tracks of a video tape. Usually the highest-number edge track at -3dB.

L

loop: (1) A piece of material that plays over and over. In a sampler, loops are used to allow samples of finite length to be sustained indefinitely. See also *sustain loop, release loop.* (2) A section of tape with the two free ends joined, used for creating repeated sounds. Tape loops were used in the first *delay* units, where a short tape circulated around a system consisting of a record head followed by a series of replay heads to pick up the increasingly delayed signal (as well as an increasing proportion of noise.)

loop (cont'd): (3) In tape recorders equipped with *zero-locators*, a transport operating mode in which the engineer has designated a starting and ending point, either in tape time or *SMPTE timecode*, and instructed the locator and machine to play the enclosed tape segment repeatedly, rewinding to the starting point each time the end point is reached. Most video *interlock* devices can be programmed to cause both video and any synchronized audio decks to repeatedly reproduce a loop of picture and its corresponding sound. The engineer may place the audio or video deck into record mode during a section of each repeat of the loop in order to replace dialog or other *sync sound*, or to perform *insert edits*.

loop (cont'd): (4) In cameras and projectors, a slack section of film located just before and after the *gate*. The loop prevents tearing of the film as it passes from continuously turning sprockets to the intermittent movement of the supply reel. (5) A wire or cable system which has at least two ends joined together, usually creating *ground loops*. (6) An electronic connection where a device has a circuit from its output back to its input. See *feedback*.

looped recording: A sequencer option whereby a saved sample is played over and over again. The new data can either replace previously played data in real-time, or add to what was played previously.

looping: See *ADR*.

looping modes: A loop can play (1) forward from start to end, (2) in reverse from end to start, or (3) alternating between forward and reverse. Also called *loop type*. See also *crossfade looping*.

Loop Points Request: A Universal System-Exclusive message of the non-real-time type, within the *SDS*, which allows a receiving device (e.g., a sampler) to request that a transmitter (e.g., a computer) send information about the two sample numbers between which a loop will occur.

Loop Points Transmit: A Universal System-Exclusive message of the non-real-time type, within the *SDS*, which allows a transmitting device (e.g., a computer) to request that a receiver (e.g., a sampler) send information about the two sample numbers between which a loop will occur.

L

loop tempo: To find the exact *tempo* of a loop when you know the *sampling rate* that was used to make the sample (assuming you are using the sample at its original pitch), set the start point and the loop point at the desired points, and subtract the start point's value from the loop point's to find the length of the sample:

$$\frac{\textbf{beats} \times \textbf{rate} \times \textbf{60}}{\textbf{length}} = \textbf{bpm}$$

For example, assume a two-bar, $\frac{4}{4}$ loop=eight beats, sampled at 32kHz. The loop (according to the sampler) is 135,500 sample words: (8 x 32,000 x 60) / 135,500 = 113.35 bpm.

For each *half-step* that the sample has been transposed downward, multiply the length parameter by 1.0595. For each half-step upward, divide the length by 1.0595, i.e., if the loop is being played two *keys* higher, divide by 1.0595 twice. Then use the same formula, substituting the new length figure for the original one.

Lo-Ro: Left only-Right only. Indicates a standard left-right stereo signal that has been *downmixed* from a larger format mix, such as *5.1*. Because the surround information has been incorporated into the stereo signal without matrix encoding, a Lo-Ro mix cannot be subsequently *decoded* back into the larger format. See also *Lt-Rt*.

loss: The opposite of *gain*. When a signal passes through a circuit or audio device, if the output power is less than the input power, the circuit or device is said to have loss, usually expressed in dB. See *insertion loss*, *passive*.

lossy / lossless: If, upon decoding by a *codec*, an audio file *compression* algorithm restores the sound to its original fidelity, it is said to be lossless. To the extent that the exact sound quality of the uncompressed signal cannot be reconstructed, the algorithm is said to be lossy.

loudness: Loudness is a subjective attribute of sound and cannot be quantified. If a large group of listeners is asked to adjust the strength of two signals so that one is twice as loud as the other, the average *power* difference will be about 10dB, and this will be almost independent of the absolute levels of the two sounds. The loudness of a sound, especially a complex sound containing many frequencies, has no simple relation to its *SPL*.

loudness control: An addition to some amplifiers or preamplifiers which attempt to correct for the reduced aural sensitivity to low-frequency, low-level sounds. The loudness control is simply a bass-boost circuit which has a relatively greater effect as the volume is turned down so that the perceived *loudness* of each frequency is the same as the loudness of a 1kHz tone.

loudspeaker: A *transducer* which converts electrical energy into acoustical energy. The most common type of loudspeaker today is the *dynamic loudspeaker* which has a *resonant frequency*, the frequency at which it will vibrate naturally if perturbed. The resonant frequency, also called the *natural frequency*, will be near the lowest frequency that the speaker will reproduce well, and is that frequency at which it is easiest to move the *cone* (the output from the speaker will be at a maximum). Damping must be added to a speaker system in order to reduce this peak in response.

low-frequency oscillator: See *LFO*.

lowpass filter: A filter that attenuates the frequencies above its *rolloff frequency*.

L

L-pad: A type of *potentiometer* that maintains constant *impedance* at its input while varying the signal level at its output. L-pads are most often used as an external *balance* control or variable attenuator (volume control).

LPF: See *lowpass filter*. In other circles, a Liquidity Preference Function. See *TLA*.

LSB: Least Significant Bit. The smallest change in signal voltage level which an *A/D converter* can encode. The value of the LSB is also equal to the amplitude resolution of a digital system, in other words, the minimum nonzero difference in level between two successive samples is 1 LSB.

Lt: See *stereo optical print*.

Lt-Rt: Left total-Right total. Indicates the presence of *matrix encoding* of four channels on a 2-track stereo master. Compare with *Lo-Ro*. See *downmix*, *stereo optical print*.

M

μ: Micro (Greek “mu”). One millionth.

μ-law or **μ law**: The international standard telephony-encoding format, used to compress audio before posting it to the WWW. It’s a fast 2:1 compressor for 16-bit audio used mainly by Sun and NeXT computers. This scheme compresses 16-bit files to 8-bit, nonlinear resolution that offers better dynamic range than standard, linear 8-bit audio files. However, the sampling rate is low-fidelity, only 8kHz, about the sound quality of a telephone receiver. It has strong cross-platform support with playback software for Macs and Windows systems.

μs: Microsecond. One millionth of a second.

m: Milli. Premix meaning one-thousandth of the quantity that follows, e.g., milliseconds (ms), and millimeter (mm).

M: Mega. One million times the quantity that follows, e.g., megabit (Mb), megabyte (MB), and megawatt (MW).

M&E: Music & Effects. The “M” and “E” in *DME*. A *three-track* film soundtrack mix, less dialogue, used for foreign voice dubbing. A foreign language film version requires that all sound effects that are otherwise included in the dialog stem are copied across to the effects stem. If these production effects are not clear of dialog, then they must be replaced either by Foley tracks or by cut effects. Once the effects are complete, the track is said to be filled, thus contracts specify “music and filled effects.” Also known as the international version, or *mufex*. See *stem*, *final mix*.

MACE: Macintosh Audio Compression/Expansion. Lossy audio *compression* algorithm, included in the Mac’s system software. It works with 8-bit digital audio files, and supports compression ratios of 3:1 (music) and 6:1 (speech). Resulting audio quality is not the best.

MADI: Multichannel Audio Digital Interface. Sometimes erroneously called the Musical Audio Digital Interface. A professional multichannel version of *AES/EBU* for transmitting up to 56 channels of digital audio data over a single coaxial cable terminated with BNC connectors. MADI uses a second cable for *word clock*, with a fixed data rate of 100Mbps used on large, open-reel digital multitracks. Optical MADI implementations are available.

MAF: Minimum Audible Frequency. The lowest line on an *equal loudness curve*, representing 0dB SPL.

mag: Shorthand for sprocketed *magnetic film*. Film that contains only sound, but no picture.

mag dubber: A type of sprocketed tape recorder/playback machine that reproduces one or more audio tracks onto the magnetic area of *magnetic film.* Some mag *dubbers* which are equipped with dual sets of sprockets can reproduce more than one size of magnetic film, e.g., 16mm and 35mm. See *mag-optical print*.

Magnasync/Magnatech: Two brands of *mag dubber*. They can be used to transfer a sound source onto *magnetic film*. These brand names are also used generically to indicate any sprocketed tape recorder or playback unit. See *dubber*.

magnetic distortion: A type of distortion in *dynamic loudspeakers* caused by nonlinearities in the interaction between the magnetic field in the *gap* and the *voice coil.*

M

magnetic film: Audio recording tape manufactured using a base of the same physical film stocks, e.g., 16mm, 35mm, etc., and which contains a magnetic area running longitudinally down the film for the recording of an audio track or tracks. Magnetic film is 3-5 mils thick, so that the same length of film and magnetic film will be of equal diameters when wound on reels. *Full-coat* magnetic film has magnetic oxide applied across its entire width. *Striped* magnetic film can have one or more thin stripes of oxide applied longitudinally on the film base. There is usually one (wide) stripe containing a single track of audio (in the same size and location as track-one of a 3-track), while another (smaller) stripe is placed on the opposite side to make the film pack evenly when wound together, usually known as a *balance stripe*. The balance stripe is sometimes used to record timecode from ¼–inch or DAT timecoded *production masters*. Also called *mag*. Not used since the advent of synchronized audio multi-track recording. See *film soundtrack*.

magnetic recording tape: Most magnetic tapes have a mylar or polyester *base* with a thin coat of magnetic material, usually gamma ferric oxide or chromium dioxide, but newer tapes are double-layered which combine the good low-frequency response of ferric oxide and good high-frequency response and low noise of chromium dioxide; the oxide is cured onto the base and the tape is *calandered*. The metal particles have a random orientation in unmagnetized tape, but they are aligned into definite magnetic patterns by the magnetic field produced by the recording head. If all other factors are the same, the wider the track, the greater the *S/N ratio*: doubling the track width improves the S/N ratio by 3dB. Professional analog tape recorders are available with tape widths up to 2” and up to 24 tracks. There is a thin *guardband* of uncoated base tape between the tracks to yield improved *channel separation*, reduce *crosstalk*, and provide some tolerance for differences in head/track alignment among machines. See *Barkhausen effect*, *back coating*, *MOL*, *bias*, *domain*, *extinction frequency*, *scrape-flutter filter*.

Magnetic tape comes in a number of widths and formats (all denominated in inches):

Tracks	Historical Width	Next Standard	Most Recent
24	2	1	
16	2	1	½
8	1	½	¼
4	½	¼	⅛ (cassette)
2	¼	½	(DAT)
4	¼		Quarter-track
2	¼		Stereo (Half-track)
1	¼		Mono

magnetometer: A device for measuring magnetism and magnetic fields. Useful for testing whether or not tape heads need to be degaussed, and also for verifying the magnetic fields generated by unshielded speakers.

magneto-optical disk: See *MO*.

magnitude: The portion of the *frequency response* or *impedance* of a device that represents the amplitude is called the magnitude, as distinguished from the *phase*, which is the other part. Precisely, the term magnitude only applies to complex quantities, i.e., quantities characterized by both a magnitude and a phase. For noncomplex quantities, the term *amplitude* is used.

M

mag-optical print: A motion picture film that has both an *optical soundtrack* and magnetic soundtracks so it can be reproduced in conventional theaters with optical sound equipment and also in houses equipped with stereo magnetic sound.

mag stripe print: A 35mm or 70mm print with magnetic oxide stripes painted lengthwise down both sides of film, on either side of the perforations. These formats are now obsolete. See *print master*.

manual: The keyboard(s) on an organ or harpsichord played by the hands, as opposed to the pedalboard, which is a keyboard played with the feet.

map: A table in which input values are assigned to outputs arbitrarily by the user on an item-by-item basis, used as input to a *mapper*.

mapper: A device that translates MIDI data from one form to another in real-time. See *MIDI mapper*.

mark-in/mark-out points: In video synchronization and *post-production*, the *timecode* addresses selected by the editor as the beginning and end points of a *loop*. The synchronizer will stop *master* and *slave* transports at the mark-out point after each insert take, then automatically return all machines to the mark-in point in preparation for another take. Various synchronizers automatically add *pre-roll* and *post-roll* times to the mark-ins and -outs, so it is important to understand how each unit internally defines all of these locations.

mark/space ratio: See *duty cycle*.

marry: To print sound and picture onto the same strip of film, as on an *answer* or *release print*.

masking: A subjective phenomenon wherein the presence of one sound will inhibit the ability to hear another sound. See *frequency masking*.

master: (1) A *gain* control on a *sound reinforcement* or recording console that controls the level of a mixture of signals whose levels have been set by the individual channel *pots*. A console will have a master gain control for each output signal. (2) The final version of a performance which will be used for the production of copies in a film production, this is the *master dub*. See *APRS label system*, *bin-loop master*, *cut*(3), *direct metal mastering*, *direct-to-disc*, *edit master*, *glass master*, *lacquer master*, *post-production*, *stamper*, *transfer*, *two-track*, *master tape*, *mastering lathe*. (3) One device within a *MIDI network* or recording / dubbing *chain* which provides the *master clock*.

master balance: A message of the Universal System-Exclusive type used for controlling the balance between the left and right outputs of a multitimbral synthesizer, in preference to adjusting individual channel balance.

M

master clock: A clock signal which is sent from a master device to all slaves to maintain tape position synchronization. In this process, the master clock device sends a signal from a dedicated word-clock output to a dedicated word-clock input on all slave devices over a separate cable, typically terminated with a *BNC* connector. Master clock generators are available at various accuracies, measured in ppm (parts per million): The AES defines Grade 1 clocks as having a long-term accuracy of ±1ppm and Grade 2 clocks as ±10ppm. The IEC specifications are Level I ±50ppm for "pro" equipment, Level II (consumer) accuracy at ±1,000ppm, and Level III has inaccuracy measured in days. It is important to be aware of the master clock spec on digital mixers, as some are specified with IEC Level II clocks, making attachment to high-resolution recorders problematic, and in this case, an external clock would be necessary to provide sufficient timing accuracy.

See *house sync*, *self-clocking*, *reference source*, *sync reference*.

master controller: In a MIDI network, the device which a musician plays in order to control other devices in the network. Typically a keyboard, but a master controller could also be drum pads or some other MIDI generator such as a string or wind controller.

master dub: See *master*(2).

master fader: A fader to which the groups or channels in a mixing desk are connected. It normally controls the level of the stereo output from the desk.

mastering: The stage between mixing and the pressing plant, where cuts are assembled in the final order and the *master*(2) prepared for duplication: song-to-song levels are equalized, the stereo image is properly balanced, fade-ins and -outs and any crossfades are added, any last-minute compression / limiting is added to even out the dynamic range of the compilation as a whole, if necessary, additional effects such as some reverb to smooth any abrupt transitions, and clean-up of all remaining noise: hum, pops, clicks, crackles, etc. (Old) The common term for the process of transferring the musical signal from a magnetic tape, usually called a *master tape*, to an acetate master disc, being the first step in the manufacture of phonograph records from tapes.

mastering lathe: A lathe bed and carriage mechanism (the actual cutting stylus or head, arm, and armature). Using a high-wattage amplifier to drive the cutting stylus, with a pitch / depth control computer that controls the depth, width, and spacing of grooves being cut, the mastering lathe makes the master lacquer disk from which metal parts and then vinyl records are ultimately made.

master tape: (Old) Records are usually made from tape recordings, and the edited tape from which the acetate is cut is called the master tape. It could be an original recording, but more often it is a copy of original tapes.

matching: See *impedance-matching.*

matching transformer: Short for an *impedance-matching* transformer. Used to interconnect devices or cables of different *impedances*. Necessary, for example, when using a low-impedance mic with a guitar amp, which has high-impedance inputs. The transformer increases the energy transfer between the mic and the input, preserving the high-frequency response in the signal.

M

matrix: (1) A term used to describe any system which allows devices to be connected as though they were arranged along the two axes of a grid, i.e., a structured form of *patching*. For example, the VCS range of modular synths, where the outputs of the various modules are connected to the left edge of a grid of holes, while their inputs are ranged along the top edge. Electronic versions of a matrix are implemented in software. (2) See *matrixing*.

matrixing: Matrixing is the linear mixing of two or more signal channels at specific amplitudes and phases to form two or more new signals. These new signals can be combined in similar ways to recover the original signals. The circuit topology used for matrixing is called a matrix. Matrixing is a linear addition of signals used to encode directional information, e.g., *Scheiber matrixing* used in Dolby Surround-sound, and it is not the same as *modulation*.

matrix modulation: A method of connecting modulation sources to destinations in such a way that any source can be sent to any combination of destinations.

MAX: A real-time MIDI processing and graphic programming environment. Max software was first developed at *IRCAM* and later commercialized by Opcode Systems.

maximum output level (MOL): For an audio device such as a tape recorder, the MOL is generally taken to mean the output signal level that results in 3% *harmonic distortion* at low frequencies and usually 3% *intermodulation distortion* at high frequencies. Any higher signal output than the MOL will result in rapidly increasing distortion, and is a function of both input signal frequency and of the device itself. MOL also applies to a specification for analog magnetic tape. The MOL for a tape is frequency-dependent; all magnetic tape saturates faster at higher frequencies; it is also speed-sensitive: as the recording speed is slowed, the distortion point is lowered. MOL may be referenced to an absolute flux level, or to a test tape. See *third harmonic distortion*.

MCPS: Mechanical Copyright Protection Society. The UK equivalent of *BMI/ASCAP*. See also *PRS*.

MD: See *MiniDisc*.

MDM: See *Modular Digital Multitrack*.

meantone: In tuning, the meantone system was a common technique used for keyboard instruments before *equal temperament* came into general use. Meantone temperament is a tuning in which the thirds are turned pure, and all of the fifths are one-fourth of a *diatonic comma* too narrow. All the whole tones are equal and are precisely half a major third, hence the name meantone. It provides for the pure intonation of the key of Cmaj and those lying near it at the expense of the more extreme sharp and flat keys, which is the reason why remote keys were rarely used in keyboard works before the adoption of equal temperament. There was, for example, a pure F♯ and B♭, but these notes were out of tune when used as G♭ or A♯.

measure: Also called a *bar*. The space between two bar lines in music notation, suggesting a unit of time. Measured music has sections with a well-defined *meter*, and which can be easily notated within bar lines. Almost all western music of the past five hundred years is of this type.

mega: Prefix meaning “one million times the unit that follows,” e.g., a megawatt is one million watts. See *M*.

M

Melco: See *ProDigi*.

memory requirements: One minute of MIDI information takes up approximately 0.035MB of memory, compared to about 10MB for CD-quality stereo digital audio (16-bits at 44.1kHz), and over 34MB for 24-bit, 96kHz sampling precision.

merger: A MIDI accessory that allows two incoming MIDI signals to be combined into one MIDI output.

Meridian Lossless Packing: See *MLP*.

metadata: Generically, parameters which apply globally to a particular data transmission, as opposed to the data actually carried in the transmission. These data usually include type of codec, number of channels, channel format, originating node information (in a network context), type of data encryption, etc. In Digital Dolby, metadata specifically refers to the parameters which travel alongside the audio in the Dolby Digital stream as auxiliary data. The metadata here provides scalable decoding information about the audio which can be interpreted in different ways by different receivers, allowing a producer to tailor a program's mix to the playback environment without requiring the medium to store multiple versions, e.g., a 5.1 mix and a stereo mix.

meta event: See *SMF*.

meter: The *time signature* of the music, i.e., how many *beats* are in each *measure*.

MCI: Media Control Interface. A multimedia specification designed to provide control of onscreen movies and peripherals like CD-ROM drives.

mic-level: The nominal output level of a microphone, usually -50dBv to -40dBv, as opposed to *line-level*. This corresponds to a few millivolts of power.

mic/line switch: On a recording console, the two-position switch that allows the engineer to select whether each module will control the *mic-level* signal from a microphone in the studio, or the previously recorded track of a tape or other signal coming into the console at *line-level*.

micron: A standard unit of length, equal to one millionth of a meter. Used to specify very small measurements, such as tape recorder *head gap* widths.

M

microphone: An *electroacoustic* device which delivers an electrical signal when actuated by a sound. A microphone consists of an acoustic system that supplies mechanical (acoustic) energy to a *transducer*, which converts the energy into electrical energy. Microphones are classified by their acoustical parameters, by their method of transduction, and by their *directional* characteristics. See *capsule*, *cardioid*, *supercardioid*, *hypercardioid*, *ribbon microphone*, *moving coil microphone*, *condenser microphone*, *dynamic microphone*, *boundary microphone*, *Soundfield microphone*, *Lavalier microphone*, *contact microphone*, and *omnidirectional microphone*.

Attribute		**Construction**	
	Moving Coil	**Ribbon**	**Condenser**
Normal Polar Response	cardioid	figure-8	cardioid / switchable
Robustness	high	low-average	average
Cost	low	average	high
Examples	Shure SM58 Electrovoice RE20	Beyer M88	AKG C451 Neumann U87
Transient/HF Response	good	very good	excellent
Diaphragm Weight	high	low	average
Output	average	low	high
Sensitivity/Efficiency	average	low	high
Application	general purpose vocal, brass combos, kick drums	strings, vocal overheads	acoustic instruments, piano, vocals, snare, hi-hats
Side Effects	average sound	handling and rumble slightly fragile	crackles when wet needs phantom power
Sound Characteristic	solid	smooth	crisp

microphone preamplifier: See *preamplifier*.

microphonic noise: Noise generated within an audio cable, caused by changes in *capacitance* between the inner conductors in the cable and / or its shield. Microphonic noise can result from unstable dielectric (insulating) material that allows the conductors and / or shield to move in relation to one another.

microtuning: A system that uses different *intervals* between notes in a *scale*. A number of microtuning systems attempt to reduce *beat* frequencies introduced by the simultaneous playing of the notes of a chord. Some systems use different numbers of notes in an octave (up to 53). It is necessary to decide on the *key* before a microtuning system can be selected. Instruments which support microtuning are called *microtonal*. See *equal temperament*, *just intonation*, *temperament*.

middle-eight: See *bridge*(4).

M

MIDI: Musical Instrument Digital Interface. MIDI is a specification for the types of control signals that can be sent from one electronic music device to another. MIDI is a *serial* protocol, with a word length of 30 bits and a transmission speed of 32kbps. *MIDI messages* are either *channel messages* or *system messages*, the first of which describes the actual musical content of the sound, and all other synthesizer actions affecting that sound are controlled by the latter.

MIDI analyzer: A device that gives a visual display of MIDI activity when inserted between two pieces of MIDI equipment.

MIDI Bank Change: A type of *MIDI controller* message which is used to select alternate banks of MIDI patches where access to more than 128 patches is required.

MIDI choke: See *MIDI delay*(2).

MIDI Clock: A timing reference signal sent over a MIDI cable a the rate of 24ppq; a System Real-Time message used to communicate timing information among instruments in a MIDI system. Also known as *MIDI Sync*. See also MTC.

M

MIDI Controllers: (1) Devices which generate *MIDI messages* and which typically resemble musical instruments such as keyboards, guitars, drums, etc. Although originally conceived with a keyboard paradigm, MIDI controllers are now available as guitars, wind valves, drum kits, xylophone, piano, accordion, the violin family, as well as keyboards of all types. (2) MIDI Controller messages. These are a type of Channel Voice message designed for adjusting individual controls, such as pan position or channel volume, on equipment in the MIDI network. While not a part of the MIDI specification, certain conventions exist. The table below gives some of the more common controller numbers. See also *controller change*, *continuous controllers*, *switched controllers*.

Controller messages can be switched, i.e., their value is either On or Off, or they can be continuous. Controller messages 0-31 take one additional data byte and can therefore carry values in the range 0-127. However, these can be paired with controllers 32-63 to provide two bytes of resolution, e.g., Controller 4 is paired with Controller 36. When this is done, the controller in the range 0-31 takes the MSB and its pair in the range 32-63 takes the LSB, for a range of 16,384 possible values. Most continuous controllers carry values ranging upward from 0, although physical controllers that center around zero, such as balance, pan, and pitch-bend may be implemented so that their associated controller message carries values centered on the midpoint.

MIDI Controllers

Controller #	**Function**	
0	bank select MSB	**Continuous Controllers**
1	modulation wheel	
2	breath controller	
4	foot controller	
5	portamento time	
6	data entry MSB	
7	channel volume	
8	balance	
10	pan	
11	expression controller	
12	effects control 1	
13	effects control 2	
16-19	general purpose 1-4	
32-63	LSB for controllers 0-31	**Switched Controllers**
64	sustain pedal	
65	portamento	
66	sostenuto	
67	soft pedal	
68	legato footswitch	
69	hold 2	

Controller #	**Function**	
70-79	sound controllers 1-10*	

	synths	**effects units**
70	sound variation	exciter
71	harmonic content	compressor
72	release time	distortion
73	attack time	equalizer
74	brightness	expander

Controller #	**Function**	
80-83	general purpose 5-8	
91-95	effects depth 1-5**	
96-101	data controllers	
120	all sounds off	**Channel Mode Messages**
121	reset all controllers	
122	local control	
123	all notes off	
124	omni mode off	
126	omni mode on	
126	mono mode on (poly mode off)	
127	poly mode on (mono mode off)	

* Manufacturers may implement these as desired. The first 5 default as indicated.
** Originally assigned to specific effects such as chorus, phaser, tremolo, etc.

M

MIDI Delay: (1) A facility provided on some sequencers to allow a track to be fractionally delayed or advanced relative to others. Particularly useful for synthesizer voices which speak late, or to give a part a sense of urgency by being played very slightly ahead of the beat. Also called *MIDI offset*. (2) Noticeable delay in the transmission caused by MIDI Choke. This usually happens when too many MIDI devices try to send *bulk dumps* or unthinned *continuous controller* data over the same MIDI port.

MIDI Echo: A feature that routes MIDI messages appearing at a device's MIDI In port through its processor, unaltered, to the MIDI Out port. This allows control of a MIDI sound module simultaneously from a sequencer and a keyboard. MIDI Echo differs from *MIDI Thru* in that there is a direct, hard-wired connection between the MIDI In and Thru jacks, so the datastream doesn't pass through the device's processor.

MIDI filter: See *filter*(2).

MIDI interface/adapter: A device that converts data from a MIDI device to a format that a computer can recognize.

MIDI loop: A (mistakenly) hard-wired *loop*. See *note-doubling.*

MIDI Machine Control (MMC): A protocol for using MIDI commands, usually from a sequencer, to control the transport functions (stop / play / record / locate / rewind / fast forward) of a tape recorder or other mechanical device. MMC is intended to link MIDI equipment with more traditional equipment such as audio and video tape machines and multimedia computer devices.

MIDI Mapper: An applet that automatically maps channel, program change, and note number data. For example, a map could cause all notes coming in on MIDI channel 3 to go out on MIDI channel 7.

MIDI Merge: The process of combining MIDI messages transmitted from two or more MIDI devices into one coherent MIDI data stream so that the messages appear to have been generated by only one device. This is not just connecting MIDI cables as MIDI messages are structured and this structure needs to be preserved. For example, if two Note On messages arrive simultaneously at the two inputs, the merge device will have to store one of them in a buffer until the first is sent. Because MIDI messages are variable in length, and because *real-time* massages have to take a priority, the merge device must be able to identify and distinguish between different data types. This generally requires a separate microprocessor, making a MIDI merge unit more expensive than a MIDI Thru device.

M

MIDI message: A full instruction consisting of at least one status byte and frequently with one or more data bytes, which causes a MIDI device to perform one of the functions defined in the MIDI specification. See also entries for each message type.

MIDI Messages

(Message examples)

Channel Messages (apply to an individual channel)	**System Messages** (apply to the whole system)

Channel Voice	**Channel Mode**	**System Common**	**System Exclusive**	**System Real-Time**
(Note On)	(Local On)	(Song Position)	(Data for specific	(Timing Clock)
(Program Change)	(Reception Mode)	(Tune Request)	items of equipment)	(System Reset)

MIDI module: A device for generating sound which does not have an integral keyboard.

MIDI network: A collection of MIDI devices connected together in such a way that MIDI messages can pass between them. The most common network is a *daisy chain* (each device connected to the previous device, i.e., in a linear arrangement) or in a *star topology*, where each device is connected to a central point, such as a multi-port *MIDI interface*.

MIDI Mode: Also called *channel mode* or *reception mode*. A setting that determines how a particular MIDI device or instrument reacts to transmitted voice and channel data. Four modes are created by different combinations of the messages Omni On / Off (defining the ability to react to data on all MIDI channels) and Poly / Mono (playing notes *polyphonically* or *monophonically*):

Omni On/Poly	Mode 1	The receiving instrument reacts to data on all MIDI channels (Omni) while playing polyphonically.
Omni On/Mono	Mode 2	Similar to Mode 1, but the instrument plays monophonically (rarely used).
Omni Off/Poly	Mode 3 “Multi”	Each synth (or multitimbral part) plays polyphonically on its own MIDI channel.
Omni Off/Mono	Mode 4 “Mono”	Used for MIDI guitar as it allows each string to play monophonically on its own MIDI channel.

MIDI Note Number: The decimal number, from 0-128, which represents the *equal temperament* scale of about eight octaves, where 60 represents Middle-C, having a frequency of 261.63Hz. The MIDI note number 36, for example, corresponds to the 4th key on a piano, referred to as C1, with a frequency of 32.7Hz. Middle-C is sometimes called C3 or C4, depending on the author. Commonly, modern instruments are tuned to *A440*, that is A3 / A4, MIDI note number 69.

MIDI Offset: See *MIDI Delay*.

MIDI Out/Thru: A MIDI output port that can be configured either to transmit MIDI messages generated within the unit (Out) or to retransmit messages received at the MIDI In (Thru). See *MIDI echo*.

M

MIDI patchbay: Essentially a patchbay for MIDI signals. Passive patchbays simply present MIDI In and MIDI Out sockets conveniently on a front panel to facilitate patching via patch-cords. Active patchbays will generally have MIDI sockets on the rear panel, switches and displays on the front panel, and will also have a memory which allows commonly used patches to be stored. These, in turn, may be triggered by specific MIDI Program Change messages sent to the unit. As all of this requires some processing power, an active patchbay will often perform other functions such as *MIDI Merge* or *MIDI mapping*.

MIDI port: A means of bypassing the 16-channel limit of the MIDI specification by using a MIDI interface with multiple MIDI sockets that carry totally independent signals and effectively provide separate *MIDI networks* that function in parallel. Typically, there may be four such sets of sockets, each socket representing one MIDI port that can be used for up to sixteen channels. Such interfaces, which are generally the hub of a large MIDI network, are occasionally built into master keyboards. More typically, they are add-on hardware devices attached to a computer and will usually only operate in conjunction with sequencing software from the same manufacturer.

MIDI Show Control (MSC): A protocol in the MIDI specification designed to integrate and control stage equipment such as lighting, hydraulics, rigging, video machines, pyrotechnics, and fog machines. MSC is intended to control dedicated equipment in theater, live performance, multimedia and audio visual applications.

MIDI slop: The timing variations which occur within a multitimbral tone generator. As several different instrument sounds are generated on multiple channels, the machine must generate and output the notes in a short space of time, causing timing variations among the various notes. This is not the same as *MIDI delay*.

MIDI Splitter: See *MIDI Thru*.

MIDI Sync: One of the synchronization protocols supported by MIDI, either *MIDI Clock* or *MTC*.

MIDI Thru: There are two types of MIDI Thru. One, a simple hardware connection, is found on the back panels of many synthesizers. The Thru *jack*, in this case, simply duplicates whatever data is arriving at the MIDI In jack, being a hard-wired connection between the two. Compare with *MIDI Echo*. Sequencers have a second type, called a *Software Thru*. In this case, data arriving at the In-jack is merged with data being played by the sequencer, and both sets of data appear in a single stream at the Out- (not the Thru-) jack. A Software Thru is useful because it allows a master keyboard to be hooked up to the sequencer's MIDI input and a tone module to its output. The keyboard can they play and produce sound through the tone module, and the sequencer can also send its messages directly to the tone module. Also called a *MIDI Splitter*. See also *MIDI Out/Thru*.

midrange: A loudspeaker designed to reproduce the middle frequencies of the sound spectrum, generally most efficient between about 1kHz-4kHz.

midrange smear: A type of sonic degradation in sound reproduction systems that is the result of nonlinear *frequency response* in the playback *chain* which interacts with the broad band of frequencies present in music, producing a veil of *distortion* products throughout the midrange frequencies, difficult to define and test. Also called *grunge*.

M

mil: Short for millinch, i.e., one-thousandth of an inch. Tape thicknesses and sometimes widths are usually given in mils.

MiniDisc (MD): A digital format that uses Sony's proprietary ATRAC (Adaptive TRansform Acoustic Coding) data compression to fit up to 74 minutes of digital audio on a 64mm, recordable, erasable magneto-optical disk. Commercial prerecorded MD releases use an optical-only playback format and are not recordable. MD audio is compressed during recording, using lossy compression, so that the audio quality has lower fidelity than the same data written to a CD. See also *Digital Compact Cassette, DAT.*

mixdown: The process of bringing together separate tracks into a unified whole. The tracks can be audio tracks (from hard disk or tape) or MIDI tracks (from a soundcard, tone generator, or digital keyboard) or a combination of both. Usually, at least three or more such *source tracks* are mixed down to the left and right audio tracks of a stereo mastering deck (analog tape deck, DAT recorder, or even a hi-fi VCR.)

mix: (1) (*noun*) The composite blend of live sounds and / or recorded *virtual tracks* to a new program, usually of fewer tracks and often to a different recording / storage medium. (2) (*verb*) To produce the new composite program, adding the live sounds, recorded virtual tracks, effects and other processing such as equalization, compression, reverb, etc., ready for the production of a recording of the desired number of tracks and on the desired recording and / or storage medium, possibly for use as a *master*.

mixdown session: A recording session during which the separate tracks of a multitrack tape are processed and combined or blended into two or more channels, then rerecorded onto a multitrack recorder. The resulting tape is called the *master* tape and is the finished studio product. The master is ready, at this stage, for *dubbing*.

mixed-mode: A mixed-mode *CD* includes more than one type of track format, usually *Red Book* audio and *Yellow Book* Mode 1 (ECC) information. Mixed-mode consists of a ROM track followed by any number of Red Book audio tracks. Mixed-mode has two problems: first, the CD-ROM track can be accidentally read by some audio players, causing an ugly static sound and potential speaker damage. Second, retailers don't know whether to market the discs as music or data. See *pregap, CD Extra*.

mixer: A device that adds two or more audio signals together. Also called a *mixing console*, *console*, *mixing desk*, *desk*, *mixing board*, or just *board*. See also *multiple mixer*, *split console*, *in-line*, *channel path*, *monitor path*.

mixing: (1) Using a *mixer*. (2) See *slip cue*.

M

mLan: A specification proposed as a part of the *FireWire* protocol standard which provides a definition of how to send multiple sample-accurate AES3 signals, raw audio, MIDI and other control information over a single 1394 cable. The specification reduces the clock jitter inherent in the IEEE 1394 8kHz isochronous clock to about 40ns. Implementations of mLan go farther, reducing the clock jitter among connected devices to less than 1ns. IEEE 1394 has adopted a portion of mLan as an official supplemental standard for handing IEEE 1394 audio and music control data; AES is also considering mLan, but is concerned that it is not sufficiently accurate for professional studio use as, while mLan ensures that multiple 1394 nodes are synchronized, the specification does not provide accuracy necessary to synchronize devices using the IEEE 1394 network to carry different types of signals, such as noncoherent audio, video, and / or musical control data.

MLP: Meridian Lossless Packing. A lossless file compression scheme developed by Meridian Audio of Cambridge, England adopted for use by the *DVD-A* standard. MLP is not a *perceptual coding* scheme, but instead, reduces the size of audio files without altering their contents. MLP may be used on sampling rates up to and including 192 kHz of 24-bit words, on up to 63 channels. The MLP compression process does not depend on assumptions made about incoming data; instead, it works on the audio waveform, using three proprietary technologies to reduce file size: first, non-data is removed from the audio file; second, matrixing is used to reduce the correlation among channels using a large palette of special filters. Lastly, Hufman coding is used to reduce the data rate by efficiently encoding the most likely occurring successive values in the serial stream. MLP coding produces variable-rate output data, and a proprietary data buffering scheme smoothes the output rate to keep peaks from exceeding carrier bandwidth limits.

MMA: MIDI Manufacturers' Association. See also *AMEI.*

MO: Magneto-optical. A type of removable digital storage medium used by several digital audio recorders. MO drives are now typically in the 2-4Gb range. See *LIMDOW*.

mod: Short for (1) modulation or (2) modification.

MOD (.MOD): Short for module. A compact type of computer file, originally developed on the Amiga, that plays back audio files when loaded into a *MOD player*. MOD files are a cross between MIDI and digital audio files, containing both digital samples and playback instructions that tell the player which notes and samples to play, i.e., MOD files include a user-defined group of digitized instrument sounds that are used to play the music. Basic MOD files contain just four tracks of 8-bit mono audio data: various *sampling rates* are used; the data can be compressed; and, each track can trigger any one of up to 31 samples from the MOD file for four-voice *polyphony*. Each instrument can have its own volume, and a few simple effects are included, such as echo and pseudo-reverb (achieved by repeating notes), tremolo, and pitch-bend. Most MOD files are in the `.STrK` format, but there are other file formats such as `.S3M`. There is a recent trend toward 8-channel MOD players, threatening compatibility.

mode: (1) See *room mode*; (2) *Scales* which became established in the Middle ages; (3) rhythmic modes of mediæval music which were classified into six patterns which corresponded to poetic rhythms.

modem: A device (MOdulator / DEModulator) that allows digital information to be sent over an (analog) telephone line.

M

moderato: A moderately fast *tempo*, about 95-115 bpm.

modifier module: A synthesizer module which takes raw sound and modifies its *timbre* (tone) or its *amplitude* (volume) in some way. In most synths, these modules include filters and amplifiers. Also called a *modulator*. See also *VCA*, *VCF*.

modular digital multitrack (MDM): *Multitrack* recording systems which record digital audio data on a videocasette, using a rotating drum in the same manner as a DAT recorder. MDMs are typically expandable by locking together multiple MDM modules. There are two MDM standards: the Alesis ADAT and compatibles which record on S-VHS tape, and the Tascam DA-88 and compatibles, which record on Hi-8mm tape. Examples of MDMs include the Akai A-DAM, Alesis ADAT, Tascam DA-88 and Yamaha DMR8/DRU8. Also called a *modular recorder*.

modular recorder: See *modular digital multitrack*.

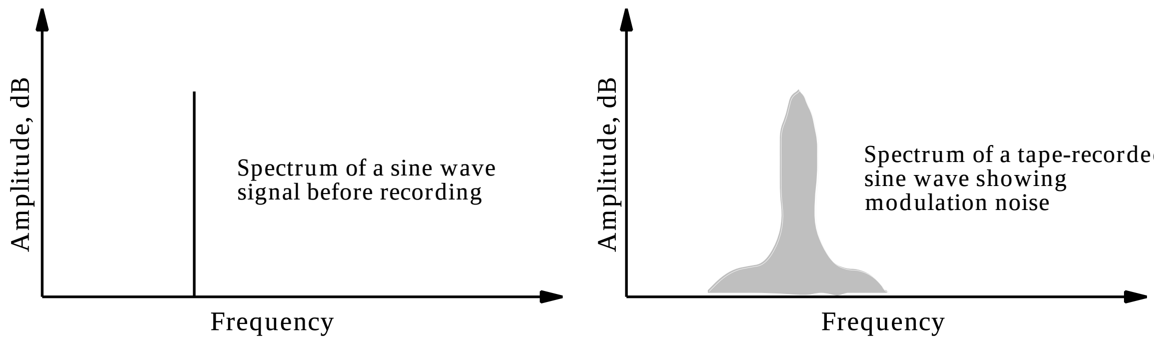
modular synthesizer: A type of synthesizer developed in the 1950's and popularized in the 1970's in which the components, such as *controllers*, *oscillators*, *filters* and *amplifiers* are designed as separate devices and interconnected by patch cords. Every module has input and output sockets that are used for interconnecting with the others. They don't have MIDI capabilities, memories, or presets, and they very rarely have hard-wired internal connections, all connections being "modular" via patch cords. The underlying principle of modular synthesis is voltage control, used to control VCOs, trigger an ADSR envelope. Also, in modular synthesis, there is little or no difference between audio and modulation signals, e.g., the audio output of a VCO can be used to modulate the control input of a second VCO, and a mixer can mix control voltage signals just as an audio mixer would.

modulation: (1) The process of sending a control signal to a sound source so as to change the character of the sound while the sound is playing. The modulation signal tells the receiving module to do something to the sound by changing one of its parameters in a predictable way. The place where a modulation signal originates is referred to as a *modulation source*. The place where the receiving module accepts the control signal is called a *modulation input*. For example, in a synthesizer, MIDI Velocity messages can be used to modulate the *rolloff frequency* of a filter. An *envelope generator* modulating an amplifier's amplitude setting causes the signal's level to change over time. (2) Changing a carrier signal in such a way as to carry information, e.g., *amplitude* or *frequency modulation*. (3) In music, a change of *key*. (4) The variation in the normally geometric groove on a record, which encodes the audio signal. (5) Electrically, another term for signal level, a usage typically encountered in England and Europe. See *overmodulation*, *undermodulation*.

modulation module: See *control module*.

M

modulation noise: *Noise* which is present only in conjunction with a signal is called modulation noise. In analog tape recorders, the recording process has a certain granularity due to the fact that the magnetic characteristics of the tape are not completely uniform as the magnetic domains are of finite size. A recorded signal has an irregularity which sounds like the addition of noise. In digital audio systems, there is also an uncertainty in the level of the signal because of *quantization error* in the *A/D converter*. This uncertainty also sounds like added noise and is not present if the signal is not present. Compare with *distortion*. See *Barkhausen effect*, *granularity*.



modulation routing: The routing of a *control voltage*, either via hardware or software, from one module source to another.

modulation synthesis. See *phase distortion synthesis*.

modulation wheel: One of the defined MIDI Controller Change messages. Physically, the mod wheel most often appears as a wheel at the left side of a keyboard. When operated, it induces some effect such as *vibrato*, although its precise function varies from device to device and can often be programmed by the user. See *real-time controller*.

modulator: See *modifier module*.

module: A hardware sound generator with no attached keyboard. A module can be either physically separate or integrated into a modular synthesizer, and is designed to make some particular contribution to the process of generating electronic sound.

modulometer: (Old) A meter, similar to the *VU meter* in appearance, which responds to the peak signal rather than the average level, accounting for *pre-emphasis*. Archaic, used in radio broadcasting to prevent *overmodulation*.

mod wheel: See *modulation wheel*.

MOL: See *maximum output level*.

molto: Italian for “a lot,” “much,” or “very,” e.g., *molto vivace*, very lively.

monaural: Literally, “one hearing.” Monaural refers to a sound system with only one channel, regardless of the number of loudspeakers used, as opposed to *stereophonic*, which must employ more than one independent channel. See *mono*.

M

monitor: (1) (*noun*) Originally, a loudspeaker in the control booth of a recording studio. More recently, this means the speakers placed on the stage to allow performers to better hear themselves, more often known as *foldback*. (2) (*verb*) To listen to or measure a signal at some point in the recording or synthesis chain. (3) The section on a mixing desk for adjusting aspects of the monitoring process. See *monitor module*.

monitor mix: A mixed signal in a *sound reinforcement* console that is sent to the monitor speakers on the stage. Because the monitors are near the microphones, this signal is usually highly equalized by notching out the *house modes* to reduce the tendency for *acoustic feedback*. There may be more than one monitor mix, each routed to a different musician or section.

monitor module: On a recording console, the module with the switching and master volume controls for studio and control-room monitors. This module often contains master *cue mix* selector switches and cue mix volume level controls, and sometimes the master reverb sends and returns.

monitor path: The replay portion of the signal chain in a *mixer*. See also *channel path*.

monitor send: This refers to a mixer's *auxiliary send* that is connected before a channel's output fader, usually called a *pre-fader send*. For example, if a pre-fader send is used to route part of a vocal track to a *stage monitor*, when the main vocal channel's level is raised, the amount sent to the monitor would be unchanged. Monitor sends are usually used for *foldback* or *cue mixes*. See *effects* (post-fader) *send*. Also called a *foldback send*.

mono: Abbreviation of *monophonic*. (1) Capable of producing only one note at a time. A clarinet is monophonic. A bagpipe or an organ is *polyphonic*. (2) An audio signal that is carried by a single channel. The number of microphones or loudspeakers that are used to generate and replay the sound is irrelevant; what counts is that the various signals are mixed to one channel. See *monaural*, *stereo*.

mono compatibility: The ability of a *stereo* musical signal to be mixed down to a single *monaural* channel without violence to the *timbre* of the signal. The result of the addition of two stereo channels would ideally sound like a signal that was recorded with a single microphone. Usually recorded using a *coincident pair*, mono-compatible stereo signals can be added together without creating *phase cancellations* which damage audio quality. See *ORTF*.

Mono Mode: One of the basic reception modes of MIDI devices. In Mono Mode, an instrument responds *monophonically* to all notes arriving over a single MIDI *channel*. In a guitar-to-MIDI converter, each string sends data over a separate MIDI channel. See *MIDI Mode*.

monophonic: See *mono*.

monotic: Literally, "with one ear." Generally refers to a sound presentation where only one ear hears the sound, e.g., through a headphone. See also *dichotic*, *diotic*.

MOR: Middle Of the Road. A very broad spectrum of popular music that is selected to be offensive to few people.

MOS: Mit Out Sound. Or, "without sound," in all-English. Any shot photographed without *sync sound*, also called *wild picture* (the analog of *wild sound*) or *non-sync picture*.

M

mother: In vinyl record production, a positive impression made by a two-step plating process, reproducing the exact shape of the grooves on a *lacquer master* on a metal disk. In *direct metal mastering*, the master itself is a mother. From it, stamper and pressings will be made directly.

motif: Also called *motive*. In music, a short but memorable melodic or rhythmic idea. Often used as building blocks for longer melodies or even complete movements. Motifs tend to remain recognizable as a binding force in the structure even when transposed, inverted or otherwise altered. A famous motif is the first four notes of Beethoven's *Fifth Symphony*.

motional feedback: A type of mechanical *negative feedback* where the actual motion of the *cone* of a low-frequency *loudspeaker* is used to generate a signal that is fed back to the amplifier. The motion of the cone itself is then inside the feedback loop, and distortion can be significantly reduced. In all motional feedback schemes, great care must be exercised to ensure that the system is stable. This means that *phase-shift* between the drive signal and the feedback signal must be accurately controlled.

motion controls: The controls on a video recorder / editor analogous to transport controls on an audio tape recorder. There is a set of six basic transport controls: Play, Stop, Record, Rehearse, Rewind, Fast Forward) and the six additional transport control functions: Location, Cue, Allstop, *Rollback*, Replay, and Edit.

motion sensing: In a tape transport, an *electromechanical* system designed to prevent damage to the tape either when the operator presses the Play button while the transport is in a fast-wind mode, or when two conflicting motion buttons are pressed simultaneously. The system brings the tape to a stop.

motorboating: A low-frequency oscillation caused by certain types of instability, usually in a power amplifier.

motor cue: See *projection*.

moving coil microphone: In a moving coil mic, a coil of wire is attached to a diaphragm and is suspended in a magnetic field. When sound waves vibrate the diaphragm, the coil vibrates in the magnetic field and generates an electrical signal similar to the incoming sound wave. For some reason, moving coil mics are called *dynamic microphones*, but not *ribbon microphones*.

Moviola: Trade name (used generically) for an upright film editing machine common in the U.S., containing a mono speaker. The term is often applied to any type of viewing and / or editing machine that runs picture and sound interlocked in sync. It is used by the film editor to decide which parts of specific takes of specific scenes will be used in the *workprint*. The film and soundtracks may run vertically on an upright Moviola (old-style) or horizontally on a flatbed version. All of these have been gradually replaced by digital audio workstations.

MP3: An audio file *codec* standard, allowing an entire CD of audio data to be compressed for *streaming* transmission over the internet. (MP3 stands for MPEG 1, Layer III.) MP3 is increasingly widespread in its use to transmit internet audio, and is currently the *bête noire* of the music industry, owing to the recent marketing of MP3 players (which have withstood an injunction attempt by *RIAA*.)

M

MPC: Multimedia Personal Computer. A specification stating the minimum hardware requirements a computer must meet to display the MPC logo. They include 2MB of RAM, a 16MHz 386SX processor, and 8-bit sound capabilities. This specification was established in 1990, and has since been superseded by the *MPC2* specification.

MPC2: Multimedia PC Level 2. This specification requires the same types of hardware as *MPC* Level 1, but with increased power and capacity: 4MB of RAM, a 25MHz 486SX processor, and 16-bit sound capability.

MPEG/MPEG-1: Moving Picture Expert Group. A body that defined a standard for *data compression* specifically for moving images: animation, audio, and video. This compression scheme includes its own file format. It is a type of *delta modulation* (differential) compression and hence very efficient, but not useful for nonlinear editing applications. Compression ratios are up to 200:1, lossy but acceptable. There are three layers of MPEG1 encoding of increasing complexity, each layer with its own format. Layer I is fast with typically a 4:1 reduction in data rate with a 32-band filter bank, but offers less compression for comparable quality. Layer II (`.MP2` files) is a popular compromise for use with audio files with ratios of 5:1 to 12:1, retaining much of the original sound file's quality by employing more complex spectral analysis. At a ratio of 8:1, CD-rate audio exhibits little audible loss. Layer III takes longer to compress, but offers higher ratios while retaining much of the audio quality by varying filter bank bandwidths to better simulate the critical bands in human hearing as well as some nonlinear quantizing to increase the efficiency of the data reduction. See *MPEG-2*, *MPEG-3*, *MPEG-4*, *DLS-2*.

MPEG-2: A professional standard for *MPEG* requiring specialized hardware and software; not used in consumer systems. MPEG-2 is an extension of MPEG-1 providing multichannel surround-sound capabilities such as 5.1, although other mixes are also supported. The original MPEG-2 was designed to be backward compatible with MPEG-1, although an incompatible MPEG-2 NBC standard has gained approval for DVDs and broadcast applications.

MPEG-3: A later *MPEG* standard which uses a *compression ratio* of approximately 13:1. The format is fast enough and good enough that this format is widely used to send digital music over data lines and is supported by Mac, PC, and UNIX platforms. In addition, the MPEG-3 compressed *file format* is compatible with all computers.

MPEG-4: Heavily influenced by Apple computer's QuickTime™, MPEG-4 is aimed primarily at game applications. In addition to streaming video and digital audio, it allows for the transmission of MIDI, provides a GM-compatible synthesizer, facilitates the transfer and playback of DLS-2 and MIDI files, and incorporates a user-configurable synthesis language. This latter feature is called the Structured Audio Orchestra (SAOL) allowing the user to specify almost any existing synthesis method and create algorithms at one end, and have the sound played back identically at the end-user's player. The MPEG-4 specification includes: DLS-2, MIDI/DLS sync, SAOL, digital audio transmission, audio spatialization, and text-to-speech.

MPSE: Motion Picture Sound Editors. A Los Angeles-based honorary organization of film and television sound editors founded in 1965 which give out its annual "Golden Reel" award.

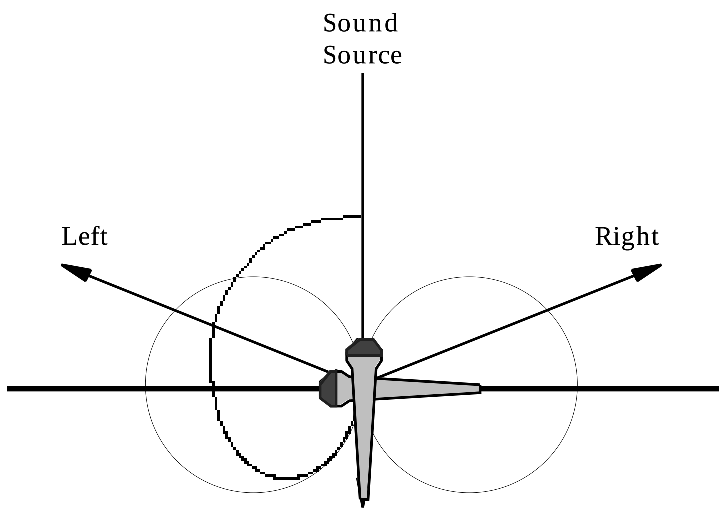
M

MPU-401: A peripheral developed by Roland for interfacing IBM PC-compatibles with MIDI devices, eventually accepted as an interface standard by many hardware and software companies.

MPX: Short for multiplex. The letters “MPX” when found on a button on a cassette deck mean the button activates a 19kHz *notch filter* to eliminate the FM stereo *pilot* so it won’t cause gain mistracking in *Dolby noise reduction* systems. The Dolby playback system would sense the pilot and mistake it for high-frequency content of the music. The pilot can also cause audible *birdies* by *beating* with the AC *bias* used in tape recorders.

ms: Millisecond. One thousandth of a second.

MS or **M-S** or **M&S**: Mit Seit (or Mid-Side). A miking technique which uses a *cardioid* microphone facing directly into the sound source and a figure-8 microphone facing sideways. The figure-8 picks up the left half of the source with one phase and the right half with the inverted phase. By changing the *matrixing* of the combined patterns of the two microphones, the width of the apparent stereo *image* can be manipulated. In M-S matrixing, the output of the side mic is added to the main mic channel to get the left channel, and subtracted to get the right channel. By adjusting the level of the side mics, you can adjust the width of the stereo image. MS techniques are most effective in larger, live rooms where it is possible to get the mics at least 10-15 feet away from the source and side walls.



Mid-Side Microphone Placement and Response

To decode MS signals to normal left and right, pan the M mic to the center and split the S mics to feed a pair of adjacent channels (or a single stereo channel.) Gang the two S-faders together, panned hard left and right. Switch the *phase reverse* on the right channel. Listening with the monitoring switched to mono, balance the levels of the two S-channels for minimal output (make sure there is no EQ patched into either channel.) Once the two S-channels have been aligned, revert to stereo monitoring, fade up the M-channel and adjust the balance between the M and S signals for the desired image spread. Putting a phase reverse in the M-channel will swap the stereo image; the image width can be varied from mono to stereo to extra-wide by moving the S-fader.

M

MSB: Most Significant Bit. The leftmost (highest) value in any numbering system, but specifically the highest value (binary) bit, although sometimes it can mean Most Significant Byte. As opposed to the *LSB*.

MTC: MIDI Time Code. One of two protocols for *MIDI Sync*. MTC is a way of transmitting *SMPTE timecode* or other time-reference data over a MIDI cable. SMPTE and MTC don't provide any start or stop commands, nor do they change with tempo. They provide an absolute (positional) timing reference in minutes and seconds, rather than a music-related timing reference in bars and beats. Unlike SMPTE, MTC must share a MIDI cable with sequence information, so it is not as fast or accurate. This is not usually a problem, but if the MIDI channel is close to capacity, MTC data may be delayed, producing a small amount of timing *jitter*. An additional part of the MTC specification is the Setup message, which allows the specification of a list of events that should occur at specific times. See *MIDI Delay*(2), *MIDI Clock*.

MTS: See *BTSC*.

MU: Musicians' Union. (UK)

muddy: A subjective term that describes a type of *intermodulation distortion* which reduces the clarity or transparency of the sound of a musical instrument, particularly transients, by adding *partials* to the natural *harmonics* of the instrument. Also used to describe the effect of *group delay distortion*.

mufex: Se *M&E*.

mult: Short for multiple. A connection, usually in the form of an *outboard* box, that shorts two or more signals together. Often found in use at a *patch bay* where a group of *jacks* are connected in parallel. If one or other or both of the signals are at a low impedance, the mult will cause distortion and a reduction in level; not a substitute for a *mixer*.

multiband audio processor: A type of *compressor,* used by FM radio stations, which breaks up the audio frequency spectrum into from three to five bands, runs them through individual compression components, and then add them back together, resulting in a kind of re-equalization. This process tends to even out the bands, reducing the boominess of mixes with heavy bass, and the tinniness of mixes with a lot of high-frequency signal. See *split-band compression.*

multichannel: (1) In film, used to refer to a final mix that includes more than stereo information, i.e., *LCRS* or six-channel *surround-sound* formats. (2) In audio, any recording or playback system with at least two tracks.

Multichannel TV Sound (MTS): A standard for transmitting stereo audio signals to home television sets.

multicore: See *snake.*

multieffects processor: An *effects processor* which is capable of producing several types of effects at once.

Multi Mode: A MIDI reception mode in which a *multitimbral* module responds to MIDI input on two or more channels, typically playing a different patch on each channel. See *MIDI Mode*.

M

multimode filter: A type of filter which has a switch that allows a choice among *lowpass*, *highpass*, and *bandpass* modes.

multipath distortion: Multipath distortion is a type of *distortion* afflicting FM and television broadcasting. It is the receipt of the transmitted signal over more than one path due to *reflections* of the audio / video waves off of hills, buildings, etc. Because the path lengths are different, there is a delay between the various signal arrival times. In TV, this causes the familiar "ghosts," or multiple images, on the screen. In radio, this is the "caught between stations" effect.

Multiple Loop Points: A category of message in the *SDS* which allows loop points to be determined or changed within a sampler independently of the sample itself, i.e., without having to retransmit the entire sample. Loop Points Request and Loop Points Transmit are two such messages.

multiple loops: The ability of a sampler to handle more than one *loop* in any given sample. Some machines allow only two loops per sample, while others allow as many as eight.

multiple mixer: A *mixer* which provides more than one combined output signal.

multiplex: When signals are combined in such a way that they can later be separated, they are said to be multiplexed together. A multiplexing device is called a *multiplexer*, abbreviated *mux*. One use of a mux is, in digital recording, a device that converts *parallel* data to *serial* format for output onto a MIDI network. Or, in an *A/D*, each sample is a binary number of bits equal to the *bit depth* of the word, e.g., a 16-bit sample for CD. All 16 bits cannot be stored to tape simultaneously, so a mux is used to sequence the bits for recording. A demultiplexer reassembles the sequenced data into complete word-samples again.

multisample: The distribution of several related samples at different pitches across the keyboard. Multisampling can provide greater realism in sample *wavetable synthesis*, since the individual samples don't have to be *pitch-shifted* over a large frequency range, with the result that the full range of *pitches*, *timbres*, and dynamics of the instrument are more accurately represented. The point on the keyboard at which one sample meets another is called the *multisample split point*.

multisession: Allows the *track-at-once* recording of independent sessions to be written to blank sections of a CD. In multisession mode, the disc is finalized after the last session. Compare with *disk-at-once*, whereby the entire disc is written as one session.

multistage: See *envelope generator*.

multitap: In digital *delays*, the ability to obtain delayed output signals, usually via patch points, at more than one point as the signal passes through a series of delay circuits. The signals derived at each of these taps, each with a different delay time, can be routed to separate destinations.

multitimbral: Capable of making more than one tone family (also called a *tone color* or *tone timbre*) at the same time, i.e., a device which can respond on multiple MIDI channels at once. A typical multitimbral tone generator can play, for example, the brass, piano, and violin parts all at once. *GM* requires 16-part multitimbral synthesis capability. Each multitimbral part can play *polyphonically*, that is play chords, up to the polyphonic voice limit of the module.

M

multitrack: An audio tape recorder capable of handling more than two tracks of information separately, but generally applied to recorders which handle eight or more tracks. Otherwise, the recorder is called a *two-track* or *four-track* recorder. Also, the actual recording tape on which eight or more tracks are recorded. The tape is nonsprocketed and the recorder may be analog or digital. The most common analog format is 24-track, 2” tape; the digital market is shared between the *DASH* format, using either 24- or 48-track ½” tape or the ProDigital format which records 32 tracks on 1” tape. MEMs use video cassettes to record 8-12 tracks of audio, but up to 128 tracks can be simultaneously recorded by locking together multiple transports.

multitracking: (1) The use of wide-format audio recording tapes on which parallel tracks are recorded, each containing a performance by one or more instrumentalists, vocalists, or *virtual tracks*. (2) See *overdubbing*.

multi-WAV: See *WAV/multi-WAV drivers.*

munchkin effect: The effect produced by *pitch-shifting* a sample sufficiently to produce distortion in the shifted tones.

MUSICAM: Masking pattern adapted Universal Subband Integrated Coding And Multiplexing. The first name given to what is now known as *ISO / MPEG* Layer II audio encoding.

music cue sheet: A list of music used in a film *soundtrack,* along with its type of usage (source, *BG* instrumental, visual vocal, etc.), i.e., not a conventional *cue sheet*.

music house: A company with staff composers, arrangers, and producers, and which has its own studio, engineering staff, etc. Clients in need of custom music can contract with the music house for a finished piece, instead of having to hire and manage all of the various creative and technical personnel and required equipment and facilities.

music track: (1) An edited track of *magnetic film* containing music. There may be more than one music track for a film, especially if the editor calls for a dissolve between two scenes, each with its own music track. (2) In a 35mm *three-track mix*, the “M” in *DME*.

mut: Make-Up Table. The motor-driven *bench* designed to load an rewind film. Mut usually refers to the setup that drives a large reel of *mag film* during a *double-system* preview screening.

mute: (1) A control on a *mixer* which allows the channel to be removed from the mix without touching the channel fader, which may be preset to the precise level called for by the composition. (2) A device placed on or in a musical instrument with the intention of reducing its volume or altering its *timbre*. (3) (*verb*) To cut off a sound, input, or track suddenly.

mute mode: In mixing console *automation* systems, an operational *write mode* in which the engineer enters off / on commands for various channels, turning these off and on to avoid noise, mistakes, and any unwanted sounds.

mute-write: The operating mode of *mixer automation* systems in which the engineer’s *real-time* mute commands are written to tape or disk to be saved for subsequent passes and reuse.

M

mutual angle: The angle between microphones in a *coincident pair*. It is possible to change the mutual angle over a small range to adjust the precise relationship between the physical sound source positions in front of the microphones and their perceived positions in the stereo image.

mux: See *multiplexing*.

mylar: A strong polyester plastic used as a base for most recording tape. Mylar is stable, easily coated, and retains its elasticity in extended storage. The word is a Dupont Co. trademark, but the term is used generically for any polyester-base recording tape.

N

NAB: National Association of Broadcasters. A body involved in the specification and development of technical standards in the U.S. audio and broadcasting industry.

NAB characteristic: The *pre-emphasis* and *de-emphasis* equalization standard for magnetic tape recording adopted in the U.S. and Japan.

NAB spool: The 10” metal tape spools used on professional tape recorders.

Nagra: A Swiss brand of professional recorder originally designed to record on-location synchronous sound for motion pictures, the industry standard for thirty years, owing to the quality and ruggedness of the recorders. It uses ¼” tape and is generally equipped with a *crystal sync* generator. Originally Nagras were portable, designed for used while being carried, and are equal in quality to high-end studio recorders. There are now various models and track formats, including some made just for studio use for making *transfers,* film mixes, etc. Nagra makes both digital and analog recorders, mono and stereo; use of a stereo Nagra on-location is almost always to record two separate mono tracks simultaneously, and does not usually mean a stereophonic recording. Nagra means “recorded” in Polish, founder Stefan Kudelski’s native language, and the recorders are manufactured by Kudelski S.A. See *neo-pilot*.

Nagra-D: The digital version of the *Nagra* recorder, using ¼” tape to record up to four tracks of 20-bit audio.

Nagramaster: An *equalization curve* developed for use by Nagra recorders that uses an HF boost during recording and de-emphasis during playback to increase the *SNR* at 15ips.

NAMM: National Association of Music Merchants. A trade association of musical instrument retailers.

nano: Prefix meaning, “one billionth of the quantity that follows.”

nanoweber: A unit of magnetic *flux*, one-billionth of a *weber*. On recording tape, specific or reference *flux density* is measured in nanowebers per meter.

narrowband: A relatively short frequency span, defining a signal or filter which encompasses a small spectrum bandwidth, as opposed to *wideband* or *broadband*. Filters sharper than one-third *octave* are generally considered narrowband filters.

natural: A symbol (♮) which is used to cancel the effect of a *sharp* or a *flat*.

natural frequency: See *resonance.*

NC Curve, NC Contour: Noise Criterion. The *ambient* or background noise in an auditorium or room. See *walla, ambience, room tone.*

N-Curve: Same as *Academy curve*. See *X-Curve.*

N

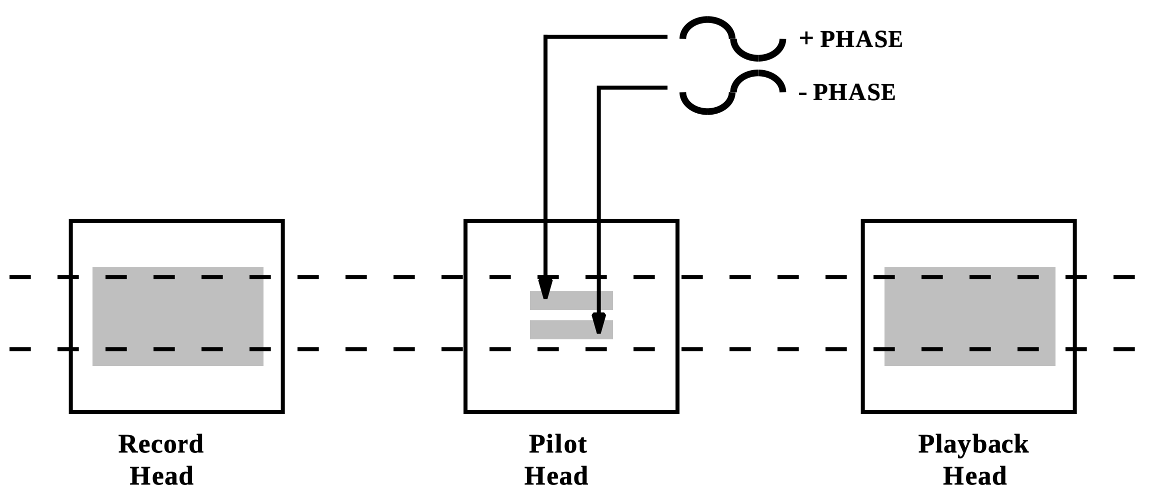
near-coincident pair: A *spaced-pair* microphone technique which uses *directional* microphones, placed approximately 7” apart, or the average spacing between ears on a human head. This allows for some amount of phase difference in the two signals, but not enough to lose mono compatibility, combining the level difference recording characteristics of directional *coincident-pair* microphones in a spaced array. *ORTF* is the most common near-coincident arrangement, but others include *NOS*, and the *Faulkner array*.

near-field: The sound field very close to a sound source is called the near-field. By “very close” is meant less than one *wavelength* at the frequency of interest. A near-field speaker is a compact studio *monitor* designed for listening at close distances (3’-5’) so, in theory, the effects of poor room acoustics are greatly reduced. See also *far-field*, *reverberant field*.

needle-drop: Any single section of music, no matter how short, copied from a music library for use in a film or video soundtrack, or a commercial spot. The producer or client must pay the specified fee to the library or copyright owner for each needle-drop, even when the same section is used more than once in the production.

negative feedback: See *feedback*.

neo-pilot: A system for the synchronization of a motion picture camera with a ¼” tape recorder recording the sound, using a superimposed *crystal sync pilot tone* generated by the camera or an oscillator, usually at 50Hz or 60Hz, on the *full-track* tape in such a way that it is not sensed by the normal full-track playback head, and so is not heard with the recorded sounds. The recovered signal is used to control the speed of the tape recorded in playback so the sound remains in sync with the picture. This process is called *resolving*. Developed for use in analog, mono *Nagra* recorders.



Neo-Pilot Tone

The *sync head* records two thin tracks of *sync tone* near the center of the tape. These thin tracks are read by the resolving circuits that control the motor speed of the player to maintain correct synchronization. As the sync signals are in *antiphase*, when both tracks pass the playback head simultaneously, they are *phase-cancelled*.

N

Neutrik: A German company which makes high-quality audio connectors, including the "Speakon" series, a range of very rugged, locking connectors used in larger loudspeakers and amplifiers.

NICAM: Near Instantaneous Companded Audio Multiplex. A television broadcasting standard that allows transmission of digital audio data alongside video, giving improved audio quality and stereo operation.

no: Used politely and defensively, the most powerful word in any language. It can protect one from naïve or unintended commitments of time, money, and emotion. Use liberally until satisfied with the entirety of any proposed deal, production, etc., which has the acceptable terms codified in writing.

node: See *anti-node.*

noise: (1) a sound which contains all of the frequencies in the audible range. (2) An unwanted sound which is not related to the wanted sound; if it is, it is called *distortion.* Noise is comprised of all audio frequencies at constantly varying amplitudes, therefore, it has no definable *pitch* or *timbre.* See *white noise, pink noise, residual noise, ambient noise, quantization noise, modulation noise, NC-Curve, noise floor.*

noise figure: Most simply, the noise figure of an electronic device is the measurement of how much worse the *S/N ratio* is at the output of the device than it was at the input, expressed in dB. The noise figure is usually important only for low signal-level devices, such as a mic preamp, where there is a very low input signal level which approaches the intrinsic *noise floor* of the environment. Also called the *noise factor*, but only if the measurement is expressed as a linear quantity.

noise filter: Either a *narrowband* or *notch filter* used to eliminate pitched noise, or a *broadband* filter used to attenuate the entire high or low frequency range.

noise floor: The noise floor is the *intrinsic* noise of any audio device or other electronic system, generally measured in dBm. Sometimes the noise floor is measured in terms of *RMS* voltage rather than power, and this makes sense in the case of devices such as voltage amplifiers or tape recorders. Includes *Johnson noise* and *flicker noise.* See *quiescent noise.*

To calculate the intrinsic *noise level* of a device, expressed in watts: if one took one 600Ω resistor on the input of a (noiseless) microphone preamp with a 60dB gain, the output would be about -100dBm. This is the lowest possible noise floor:

$\mathbf{4KTR_{\Omega}B_{Hz}}$ where

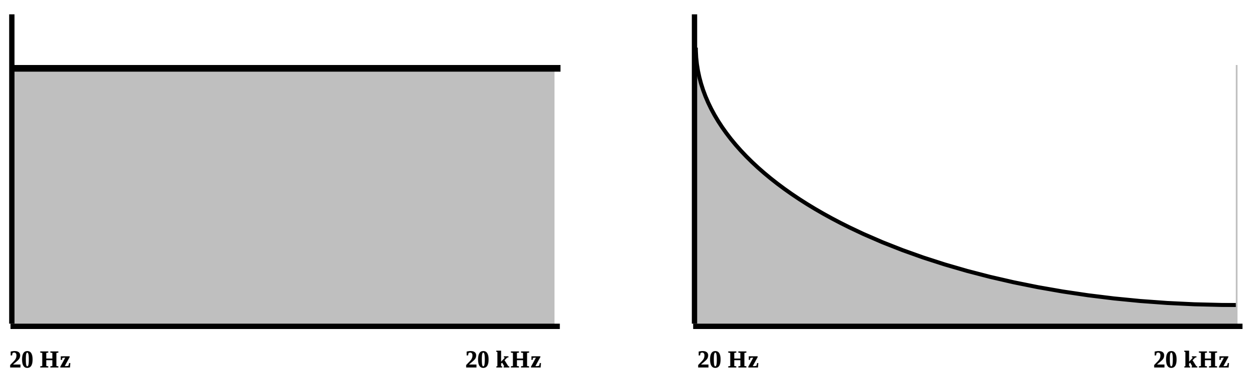
K = Boltzman's constant = 1.38×10^{-23}
T = Temperature in Kelvin
R = Resistance in Ohms
B = Bandwidth in Hz

$$\mathbf{10\left[\log\left(4\left(1.38\times 10^{-23}\right)(298)(600)(20,000)\right)\right]-30 = -157dBm}$$

N

noise gate: A *noise reduction* device through which an audio signal is passed. When the signal level is very small, the noise gate will close, eliminating any residual noise that may be riding on the signal. In the presence of a signal, the noise gate will open, allowing both signal and noise to pass through, as under these conditions, the noise is masked by the signal. A noise gate is a special type of *expander* with an infinite expansion ratio below a preset threshold. The effectiveness is determined by the time constants associated with the gain reduction, and often the background noise can be heard switching on and off with the signal. See *breathing*, *gate*, *floor*.

noise generator: A device to generate *white noise* or *pink noise*, a random signal that contains all frequencies at the same time.



White Noise **Pink Noise**

noise level: (1) The *noise floor* of an electronic device. (2) The steady-state decibel level of *ambient noise* in an acoustical environment. See *NC Curve*. (3) The *residual noise* of a magnetic tape.

noise modulation: See *dither*.

noise pumping: See *breathing*, *noise gate*.

N

noise reduction (NR): Two technologies for noise reduction have become standard in the consumer and professional recording industry: *dbx* and *Dolby*. All *two-ended* noise reduction systems are a type of *compander*, i.e., they operate by encoding the signal at the record end, and decoding the signal, restoring the *dynamic range* and frequency spectrum, upon playback.

Type	Application	CR*	S/N**	Headroom	Emphasis	Split-Band	Range
dbx-I	open-reel	2:1	30dB	+10dB	12dB	No	Linear
dbx-II	cassette	2:1	30dB	+10dB	12dB	No	Linear
Dolby-A	open-reel		< 9kHz <10dB < 15kHz <15dB			4 fixed	-10VU
Dolby-B	cassette		> 4kHz <10dB		(mod. Dolby-A)	No	HF
Dolby-C	cassette		> 1kHz <20dB		(mod. Dolby-A)	No	HF
Dolby-HX	cassette	(adds control of recording bias and equalization by the HF component of the signal)					
D**olby-S**	cassette		> HF <24dB < 1kHz <10dB		(mod. Dolby-SR)	3 fixed, 2 sliding	HF, LF
Dolby-SR	open-reel		> 25dB			5 fixed, 5 sliding	All

*CR is an abbreviation for compression ratio. **S/N means improvement of S/N, in dB.

Single-ended noise reduction systems need no encoding or decoding. The NR is applied to noisy instruments or microphones and works either by *dynamic filtering* or *downward expansion*. Dynamic filtering works will with noisy synthesizer sounds, but can cause *pumping* and *breathing*. A downward expander attenuates any signal below the threshold. It works when applied to cut the buzz on a noisy guitar amplifier, but it can also cut off quiet signals like reverb tails if the threshold is set too high. See *dynamic filter*, *spectral recording*.

noise shaping: Signal processing used in *DAC*s to shift the frequency of noise in a digital signal so that, on conversion, the noise will be outside (usually above) the audio range. This is done by reducing the number of *parallel* bits used to carry the data, increasing the number of *serial* bits and so the frequency of the digital signal. In oversampling digital systems, digital filtering is used to reduce *quantization error*. By changing the shape of the *spectrum* of the *quantization noise*, increasing its high-frequency content by lowering its low-frequency content, an *anti-imaging filter* can then be used to reduce the increased high-frequency noise. See *dither*.

nominal level: The optimum level at which a signal is processed in a particular piece of audio equipment. For instance, if the unit has a VU meter, this level would be represented by the 0VU mark, past which the meter goes into the red. Normally expressed in dBV for professional recording equipment, the two output levels are commonly called *mic-level* and *line-level*. Also called *output level*.

nonlinear distortion: *Linearity* would predict an invariant ratio of gain to input voltage in an audio device such as an amplifier. Manifested as *harmonic distortion* and *intermodulation distortion*, there is a point at which the amplifier gain is reduced as the input gain is increased. This is the nonlinear region of the amplifier. All audio devices have local peculiarities or deviations from linearity at specific signal levels. See also *distortion, linear distortion*.

N

nonlinear recording: Use of a recording medium which is random access, i.e., not recorded to conventional tape. Digital recording systems allow for playback in any order, while linear systems require that playback occur in the order in which the material was recorded.

NoNoise: Sonic Solutions' digital signal processing system that analyzes the digitized signal and senses transient noises, such as clicks and pops, and continuous noises, such as tape *hiss* and AC *hum*. It removes the transients and makes a substitute signal by interpolation. Used to restore old recordings. A competitive program is called *CEDAR*, developed at Cambridge.

non-real-time: (1) The situation where events can occur at any time, independently of other events and without the need for human input or synchronization. See *real-time*. (2) See *Universal System-Exclusive*.

normalize: To boost the highest level of a *waveform* or *sample* in a digital system to 0dB and then raising all other samples by the same proportion. This maximizes resolution and minimizes certain types of *noise*.

normalled connection: A connection, typically on a mixer or patch panel, where the signal path is continuous in the absence of a plug inserted into the signal *chain*. Usually normalled connections are made via *TRS* jacks. In a solidly grounded system, inserting a stereo plug half-way into the jack, where the tip makes a connection, but not the ring, would yield an additional channel output, as opposed to a *channel insert*. In a poorly grounded system, this yields *hum*. See also *breakjack*. Called *normalized* connections in the UK. (*verb*) Normalling.

normal stereo: See *coincident pair*.

norvalizing: Film slang for playing a sound effect at a low level in an attempt to hide the fact that it is not in sync with the picture.

NOS: See *ORTF*.

notch filter: See *band-reject filter*.

note-doubling: An (undesirable) effect caused by a *MIDI loop*, i.e., when data fed to a sequencer's MIDI In appearing at the MIDI Out to be fed back into the keyboard. This sounds like *flanging*, and cuts the *polyphony* of the synthesizer in half.

note number: The value which appears in the first data byte of a MIDI Note On or Note Off message. It determines which note will be turned on or off. 128 notes (more than 10 *octaves*) can be described, with note number 60 being middle-C.

Note Off: A Channel Voice message which causes a device to stop playing the note defined in the message. True Note Offs are seldom used, except on those devices that implement release velocity sensing, the preferred method being to send a Note On with zero velocity. This allows the use of *Running Status*, reducing the amount of data transmitted.

Note On: A Channel Voice message which causes a device to sound the note defined in the message. See *Note Off*.

NR: See *noise reduction*.

N

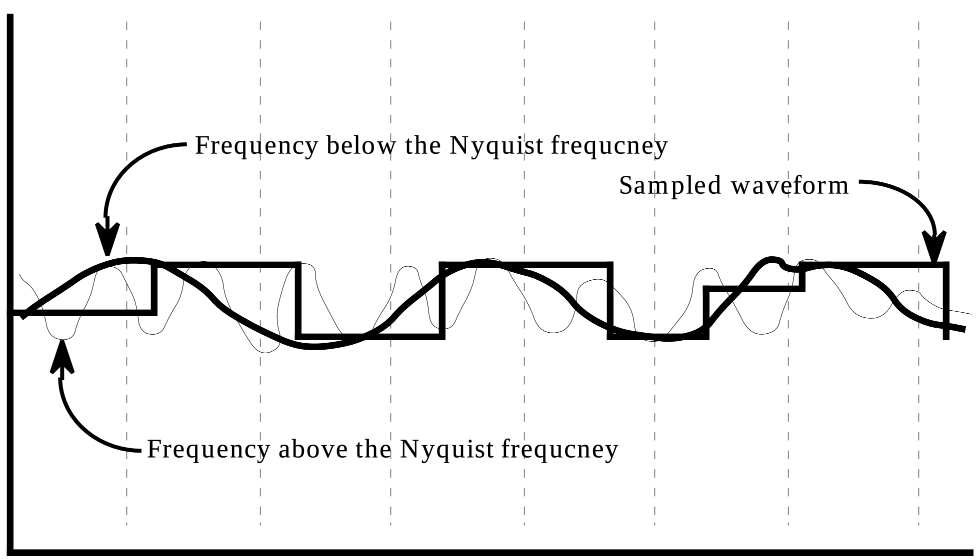
N.T.S.C.: National Television Standards Committee. (1) The American group which defines the format of U.S. color television. (2) The standard this organization has developed for color television transmission, also known as Never The Same Color. See *frame.*

null clock: The *word clock* data imbedded in an audio signal. For example, all *AES/EBU* digital audio signals carry word clock data, but if this clock information is passed without any actual digital audio, the signal is known as “null clock.”

null-point: In the *update mode* of mixing console *automation*, the positions at which all the faders are set at the beginning of an update pass through the mix.

nut: The slotted plastic piece at the headstock end of a guitar neck which is used to guide the strings over the fingerboard, and to space the strings above the frets.

Nyquist frequency: The highest frequency that can be reproduced accurately when a signal is digitally encoded at a given *sampling rate*. Theoretically, the Nyquist frequency is half the sampling rate. For example, when a digital recording uses a sampling rate of 11kHz, the Nyquist frequency is 5.5kHz. Conversely, if one wishes to produce an audio bandwidth of 20kHz, a sampling rates of 44.1kHz is used as the brick-wall filter on the A / D converter starts to roll-off at 20kHz so that the level has dropped to zero at the Nyquist frequency of 22..05kHz, yielding full level throughout the 20kHz AF band. If a signal being sampled contains frequency components that are above the Nyquist limit, *aliasing* will be introduced in the digital representation of the signal unless those frequencies are filtered out prior to digital encoding by means of an *anti-aliasing filter*. The Nyquist Theorem is also called the *Sampling Theorem*. See *brick-wall filter*.



How frequencies higher than the Nyquist frequency create aliased frequencies in the audible range: the sampled waveform is identical to the period of the waveform below the Nyquist frequency due to the way the sampling clock rate intersects the waveforms.

O

Ω: See *Ohm*.

OBU: Outside Broadcast Unit. A team of technicians responsible for recording or broadcast away from a studio.

OCN: See *EK neg*.

octave: The *logarithmic* relation of sound frequencies used in most modern Western music. The frequency of each higher octave is twice the preceding one, i.e., an octave is a frequency ratio of 2:1. An *octave band* consists of all the frequencies within an octave. There is one octave between 100Hz and 200Hz, also between 1kHz and 2kHz. Octaves are perceived as equal pitch intervals, even though the true bandwidth in Hertz varies with the frequency level of the octave. The name arises from the musical practice of defining the eight notes of the scale within a doubling of the frequency. To ears, two frequencies an octave apart sound like the same note.

OE: Operator Error. A failure in any mechanical or electronic system caused by inappropriate action on the part of the humans setting up or operating the system.

off-axis: The opposite of *on-axis*. (1) Not directly in front of a loudspeaker. (2) Not within the optimal *acceptance angle* of a microphone, and therefore not recorded at full level. See *directional microphone*.

off-axis coloration: A dull or colored effect on sound sources that are not placed within the *acceptance angle* of the microphone. To avoid off-axis coloration, place mics so that they are aimed at sound sources that put out high frequencies, such as cymbals, when miking a large source. And, use a microphone that has a flat *frequency response* over the recording field, i.e., has similar polar patterns at midrange and high frequencies. Most large-diaphragm mics have more off-axis coloration than smaller mics (¾” diaphragm or under).

offbeat: See *beat*.

offlay: To separate individual sound effects, pieces of dialog or other sounds originally on one roll of *magnetic film*, placing each on a separate roll to allow for individual equalization or other effects treatment.

off-line: See *on-line*.

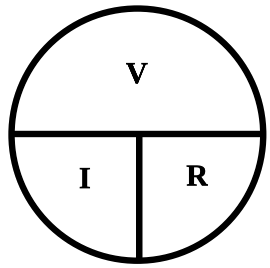
off-mic: See *off-axis*.

offset: (1) A time-difference correction made between two or more devices to achieve proper synchronization. For example, if a VCR and multitrack are 1.5 seconds out of sync, instructing the synchronizer to calculate an offset for that amount could resync the sound and picture. (2) A correction that affects the onset of an event. For example, a *velocity curve* offset defines a threshold below which no velocity data is sent. When the velocity value exceeds the threshold, velocity response follows the selected curve.

ohm (Ω): A unit of electrical resistance or *impedance*, that which opposes an electric *current* in a conductor.

O

Ohm's Law: A basic law of electrical circuits, the mathematical relationship between electrical *voltage*, *current* and *resistance*: the current in an electric conductor is directly proportional to the voltage across it and inversely proportional to its resistance, i.e., the voltage and current in a conductor exhibit a linear relationship. It states that the current, I, in *amperes* in a circuit is equal to the voltage, V, in volts divided by the resistance, R, in ohms: thus, I=V/R. Ohm's law works for DC, and for AC if the resistance is a pure resistance, but if the resistance has any reactive components, inductance or *capacitance*, the current depends on the frequency as well as the voltage.



OMFI: A *file format* first proposed by Avid to allow for digital audio data interchange among *digital dubbers*, editorial *workstations*, and hard disk editors.

omnidirectional microphone: A *pressure operation* microphone with a non-directional *acceptance angle*, i.e., one that is spherical, usually called an *omni*. See *directional microphone*.

Omni Mode: See *MIDI Mode*.

OMS: Open Music System, formerly Opcode MIDI System. A *real-time* MIDI operating system for Macintosh and PC audio applications. OMS allows communication between different MIDI programs and hardware, so that a *sequencer* could interface with a *librarian* program to display synthesizer patch names (rather than just numbers) in the sequencer's editing windows.

on-axis: See *off-axis*.

on-board effects processor: This can be used in a synthesizer to add *reverb*, *chorusing*, or other *effects*. On most synthesizers, it is possible to set the *effect send level* separately for each of the *multitimbral* parts. As opposed to *outboard*.

one-legged: A term to describe a broken electrical connection. In a *balanced line* connection, a symptom is the loss of gain and low frequency content in the signal. In an unbalanced line connection, the signal will probably disappear altogether. See also *open circuit*.

one-shot sampling: A sound which is sampled once and then triggered as necessary.

one to one: See *1:1*.

O

on-line / off-line: (1) The opposite of real-time, i.e., processing to an audio or other file which is not done at the same time as the human actions which initiate the processing. (2) In the videotape editing process, off-line is when the final edit list is compiled on a less expensive machine, i.e., where the *EDL* is created, but not *conformed*, in preparation for the final edit. On-line is where the video tape master is created from the EDL, including all effects using high-quality equipment, usually a 1” video deck. (3) In a network, any device which is able / unable to receive or transmit a signal. (4) In a system of synchronized devices, a slave device which is waiting for a particular *timecode* value to be reached before it will play or record, etc., is said to be on-line.

opamp: Operational amplifier. A *differential amplifier* with extremely high input *impedance* and high gain. Its characteristics can be tailored to various amplification tasks by the application of proper *feedback* to produce various effects, but is characterized by a low, ground-referenced output impedance.

open air acoustic: In a studio, the simulation of open air acoustic is achieved by the use of screens to surround the sound source and a microphone. This ensures that much of the sound energy which travels away from the microphone is absorbed and is not reflected back to it. This absence of *reflection* makes the sound appear to be located outside.

open circuit: A circuit through which an electrical current cannot flow, perhaps because a component has failed or a connection has been broken. See also *one-legged, short circuit*.

open-circuit voltage rating: The output voltage of a microphone with no load, i.e., with infinite *resistance* such as in an open circuit, or when driving a resistive load at least twenty times the microphone’s internal *impedance*. One of the standard specifications of microphones.

open-loop: An amplifier without *feedback* is said to be in an open-loop mode, or to be an *open-loop amplifier*. The feedback around the amplifier closes the loop.

open-reel: A type of tape machine which uses tape wound on spools, rather than tape which is sealed in a cassette.

open track: On a multitrack tape, any track that has not yet been used, or that may be erased and reused for overdubs.

operating level: The voltage level defined for any audio system at its nominal, 100% modulation level, not including any *headroom*. Usually defined as 0dBVU for a steady *sine wave*.

operator: (1) A term used in Yamaha’s FM synthesizers to refer to the software equivalent of an *oscillator, envelope generator*, or envelope-controlled amplifier. (2) The general term for a structural component of *FM synthesis*, analogous to an oscillator / envelope / amplifier in synthesizer parlance.

optical disc (OD): A very dense type of digital data storage medium. The data are *encoded* in a spiraling pattern by a laser that carves tiny pits into the surface of the OD master, every change from *land* to a pit indicating a change from a 1 to a 0, or vice versa. Where no change is recorded, the last-read digit is indicated for each increment of *groove length*. In the reproduction device, a small laser scans the groove, reading the changes from land to pits, converting this information back into data. The data can be text, video, or digital audio, such as CDs and laserdiscs. See *LIMDOW*.

O

optical recording: Sound recording on film. The photographically printed *film soundtrack* is known as the *optical track*. See *SVA*, *variable area*.

opticals: (1) Effects on 35mm film that are made in an optical printer, such as dissolves, *fades*, super-impositions, freeze-frames, matte shots, etc. (2) Loosely, any effect used in a film or video production.

optical sound: The type of sound reproduction on film that employs a photographic printing process of the *optical track*. As opposed to *magnetic film*. See *optical recording*.

optical track: A method of recording an audio signal in the emulsion of a film alongside, and in sync with, the picture frames. The photographically printed *film soundtrack* that appears either between one row of perforations and the picture in an *answer* or release print (in 35mm), or along the edge of the print opposite that with the perforations (in 16mm). The track area on a 35mm print takes up a total width of 100 mils, which, being one-tenth of the space between the sprocket holes, displaces the centerline of the image on the film by 50 mils, hence called the *Academy centerline*. See *DES*. (2) The master or original photographic soundtrack made directly from the mono mix or 35mm *three-track mix*. This strip of film has no video image, but is printed along with A- and *B-rolls* onto the composite answer print or release print. The track can be a negative or positive image, depending on the type of camera and print stock. See also *stereo optical print*, *50% level*.

opto-electric: A device which uses a variation in light intensity to cause a change in electrical current. Variable photoresistors are sometimes used as gain control elements in compressors where the side-chain signal modulates the light intensity.

opto-isolator: An electronic component that can pass a signal via a light path, avoiding a direct electrical connection between two separate circuits. This will prevent voltage spikes generated in one piece of equipment from damaging another unit in the network, as well as breaking *ground loops*. The unit consists of a light source (an LED) and a light detector (a phototransistor) enclosed in a sealed box, the whole package looking IC-like. The part of the MIDI specification that deals with hardware requires that all *MIDI In* connections use an opto-isolator.

ORC: Optical Radiation Corporation. See *Cinema Digital Sound*.

order: In discussing *filters*, the number of *poles* a certain filter possesses is called the order of the filter. Thus a T-section is a third-order filter and an L-section is a second-order filter, etc. The *slope*, in dB/octave, of the filter response in its *stopband* is equal to six times the order.

ORTF: Office de Radiodiffusion-Television Française. A stereo microphone configuration designed by the French national broadcasting system. This method calls for two *cardioid* microphones to be spaced 17cm (6.7”) apart, at an angle of 110˚. The 17cm represents normal ear spacing, and the 110˚ is to simulate the directional pattern of the ears. Recordings made using the ORTF method sound more open and spacious than those produced by the *X-Y* miking method. ORTF works well with headphone applications, but tends to sound somewhat dry and lacking in warmth due to the directional patterns of the cardioids, which pick up little *ambient* room sound. However, because of the close spacing of the microphones and the resultant similarity in phase, ORTF does provide *mono compatibility*, desirable in broadcasting. The Swedish equivalent is *NOS*, which is the same, except that the angle is 90˚ with a spacing of 30cm (11.8”).

O

oscillator: (1) An electronic device which generates a *periodic* signal of a particular frequency, usually a *sine wave*, but sometimes a square wave or other *waveform*. In an analog synthesizer, oscillators typically produce regularly repeating fluctuations in voltage--that is, they oscillate. (2) In a digital synthesis, an oscillator more typically plays back a complex waveform by reading the numbers in a *wavetable*. An oscillator allows a choice of *pitch* and waveform, the first affects the perceived musical pitch and the second affects the *timbre*. Additional parameters that are almost always found in the oscillator section of a synthesizer are those that deal with *vibrato* and *pitch-bend*.

oscillator sync: A *sound synthesis* technique whereby one *oscillator's* cycle is synchronized to that of a second. This forces the *waveform* of the slave oscillator to restart its cycle each time the master crosses the *zero-point*. As a result, the *fundamental* of the slave is the same as the master, but the waveform is radically changed. The pitch of the controlling oscillator is not normally added into the audio mix, but can be shifted by *pitch-bend*, *envelope*, *aftertouch* or an *LFO*, producing substantial changes to the *harmonic* content of the slave oscillator, but without changing the fundamental pitch as does *ring modulation*. Instead, the higher harmonics around the pitch of the slaved oscillator are emphasized, producing a very hard edge to the tone.

ossia: Italian for "or." Usually used to indicate an alternative version of a musical passage.

ostinato: Italian for "obstinate." A short melodic and/or rhythmic idea which is continually repeated, often in the bass. Known as a *riff* in popular music.

outboard: In a recording studio, special equipment such as *effects* devices and power amplifiers that are not included within the recording console are called outboard devices. Compare with *on-board*.

out-of-phase: A condition where two signals have a *phase* difference of 180˚, or one-half cycle. It should be called out of *polarity*, phase being a continuous variable rather than discrete. The same as *antiphase*. See *phase reversal*.

output: The point of exit of a signal from a system, e.g., a section in a *mixer* or other device where the signal is transmitted to a device external to the mixer, such as an *effects processor*, headphones, or monitors.

output impedance: The output impedance of a device is the actual impedance at the source output terminals. See *impedance-matching*.

output level: See *nominal level*.

output point: See *channel insert*.

output power: The lower level that a system outputs under a specified load. Expressed in VA or *RMS* (watts). See also *power bandwidth*, *rated load*.

outro: A term derived from *intro* and which refers to a section at the end of a piece of music; used in popular music in preference to the classical term *coda*. It leads to or replaces, in the case of a *fade-out*, a definite ending. See *vamp*.

out-take: Any take produced in a recording session which is not used in the final master.

O

overbias: The use of more *bias* current in an analog magnetic recorder than is required for maximum sensitivity. Overbias will reduce the *distortion* and the sensitivity to *dropouts*, but will also reduce the high-frequency response, so compensatory equalization must be applied.

overdrive: To input a signal to an audio device in such magnitude that an *overload* occurs. Specifically, a characteristic electric guitar sound in which a guitar amplifier (preferably a tube-type) is overloaded to the point of *clipping*. A similar result can be provided by an *effects* unit of the same name. Not to be confused with *fuzz*, in which the distortion is introduced before the amplification stage.

overdub: To record additional parts alongside (or merged with) previous recorded material, either by a mixing and/or re-recording process or by adding a new track in multitrack recording. Overdubbing enables one-man band productions, as multiple synchronized performances are recorded sequentially. Also called *tracking* or *multitracking*.

overflow: MIDI devices invariably have a limited amount of polyphony; any attempt to exceed this by sending too many simultaneous notes will result in an overflow of MIDI data. If a device has an overflow facility, Note On messages beyond its total polyphonic capacity are passed out to a second device for it to voice. Otherwise, the condition results in *voice-stealing*.

overlap: The film dialog equivalent of an *overdub*. (check)

overload: An audio device is overloaded when the input signal level is so high that it drives the device out if its *linear* range and into distortion or *clipping*. Overload may be continuous or may affect only *transients* in musical waveforms.

overmodulation: A situation which occurs when the *amplitude* of a signal exceeds the limits of the recording or broadcasting system. This causes distortion and can, in exceptional circumstances, damage equipment through which the signal passes. The opposite of *undermodulation*.

overs: Input peaks recorded onto a digital medium in excess of 0VU, causing a crackling, ripping type of distortion. See *clipping*, *headroom*.

oversampling: The principle of sampling a signal at an integer multiple of the normal sampling rate. The factor can be as little as two times, or much more. The effect is to distribute a fixed level of *quantization noise* over an ultrasonic frequency range, diluting the noise in the audio bandwidth and improving the *S/N ratio*. See also *quantization error*, *Shannon's channel capacity theorem*.

overshoot: (1) (*noun*) If a *compressor* or *limiter* is subjected to a sudden large input signal level, its *attack time* may not be fast enough to prevent the output from being momentarily too high. This initial excessive level is called overshoot, and its severity depends on the speed of the device. (2) (*verb*) Imperfect t*ransient response* in an audio device will result in the waveform going past the desired value on fast signal transitions, as is frequently seen on *square waves* and the *impulse responses* of CD players.

O

overtone: Overtones are produced by a musical instrument and are higher in frequency than the *fundamental*. They may or may not coincide with the frequencies of a *harmonic* series. The overtones define the harmonic spectrum of a sound. The fundamental is the first *partial*, so the second partial is the first overtone. Overtones are defined as the harmonics above the fundamental, but in common usage they are taken to mean any partials above the fundamental. In most instruments, the higher overtones are lower in volume than the lower overtones. See *subharmonic*.

P

ϕ: An abbreviation for phase. Some control consoles have a small switch on their input modules labeled "φ" that inverts the *polarity* of the signal in that channel. This is to allow all the signals being mixed together to have the same polarity regardless of wiring errors in patch panels and microphone cables and in the microphones themselves. See *out-of-phase*, *phase cancellation*.

PA: (1) Public Address. The distribution of audio signals to loudspeakers for an audience. (2) A system of equipment, including a mixing desk, amplifiers, loudspeakers, etc., assembled to provide PA sound.

pad: (1) A switch or knob that lowers the level of an incoming signal before it reaches the rest of the circuitry. A short name for an *attenuator*, usually with a fixed amount of *insertion loss*. Pads are used between audio devices if there is a danger of the output of one device causing a signal overload in the input of the other device. An input pad is also called a *trim pot*. See *pot*. (2) A sustained chord part which provides harmonic padding in a piece of popular music. (3) Drum pads.

paddles: (PEC / Direct)

PAF:

PAL: Phase-Alternating Line. The UK television broadcast standard. Uses 625 lines at a *frame* rate of 25fps.

PAM: Pulse Amplitude Modulation. The first stage in digital sampling, in which pulses of fixed-width have their amplitude modulated by an analog signal, i.e., the height of the pulse is determined by the amplitude of the signal. It is followed by an *encoding* stage known as *PCM*.

pan: Short for panorama. (1) (*noun*) Refers to the left-right placement of a sound. (2) (*verb*) Moving a sound from stereo-center to one side or the other, either on a mixer or on a synthesizer, sequencer, etc. Usually this is not completely effective as panning alters only the relative amplitude of the sound left to right, and not the crucial aspect of delay. See *Haas effect*. (3) One of the defined MIDI Controller Change messages assigned to the parameter in a synthesizer which determines the stereo image of the sound, effectively making that controller a *panpot*.

PAN: A dedicated on-line internet service provider for the music industry.

pancake: A 10½" reel of recording tape without the reel *flanges*. Recording studios often buy tape in this form and spool it onto smaller reels as it is used.

panic button: A hardware or software feature that simultaneously sends All Notes Off and Reset All Controllers commands to a MIDI system. Because some instruments don't respond to All Notes Off commands, some panic buttons are designed to send Note Off messages (0-127) on all channels as well.

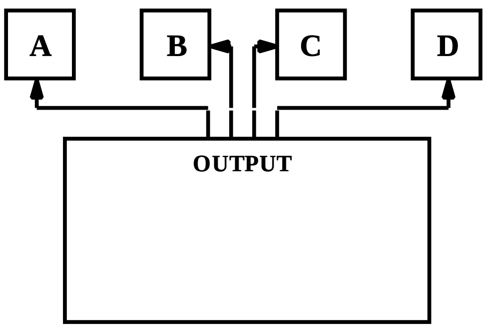
panpot: Short for panoramic potentiometer. An audio *mixer* control which is used for positioning the channel's signal somewhere between the right and left speakers. See *pan*.

paper leader: See *plastic leader*.

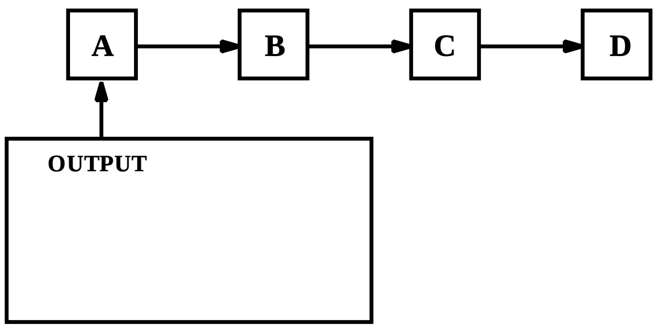
P

parabolic reflector: A dish-shaped structure used to focus sound waves onto a microphone. The reflector faces the sound source and the microphone is mounted backwards, i.e., pointing at the center of the parabola. Used mainly to gather sound in outdoor locations, such as in wildlife recording. A satellite receiving dish for television uses a similar principle to focus incoming *electromagnetic* waves.

parallel: In the connection of one signal or power source to more than one device or destination, the wiring configuration in which the input leads of all the devices meet at a common point. Signal or power routed to this point flows directly to each device. The opposite of *serial* wiring, in which the source is wired to the input of one device, whose output becomes the input for the next, etc.



Parallel Wiring Topology



Serial Wiring Topology

parallel connection: See *bridging*.

parallel interface: A protocol for transmitting data whereby all bits in a word (typically two or more bytes) are sent simultaneously. This method is generally more expensive to implement than *serial* protocols as the connectors and cable must have a pin / wire for each bit, plus a few extra for ground, handshaking, etc. This cost is offset by the faster transmission times.

parallel port: A connector on a PC-type computer which is used to connect devices which use a *parallel interface*, such as some MIDI interfaces and parallel printers.

parity check: In digital recording, one or more bits of data derived from the audio sample and appended to it as a part of the data *word*. The parity bit allows *error-detection* circuitry to determine whether the audio bits are correct, and therefore, whether they should be sent on for D / A conversion, discarded, or repaired. Also called a *parity code*.

parameter: A user-adjustable quantity that governs some aspect of a device's performance. Normally, the settings (values) for all of the parameters that make up a synthesizer *patch* can be changed by the user and stored in memory, but the parameters themselves are defined by the operating system and cannot be altered.

parametric equalizer: A parametric equalizer is a *sweep bandpass* EQ, except that a third control is added to allow the *Q* to be adjusted. Because the filter response is curved, the actual frequency width is measured in increments of 3dB. Also called a *peaking equalizer*. See *equalizer, graphic equalizer*.

P

parasitic oscillation: A malfunction occurring in some audio devices, especially power amplifiers, in which the device will generate an *ultrasonic* signal during a part of the audio signal, only when the input signal is present and only during part of the audio waveform. Although the ultrasonic component is not directly audible, its existence modulates the signal, causing audible distortion and is potentially damaging to tweeters.

partial: The *fundamental*, *harmonic*, *subharmonic*, *overtone*, or a tone at some other frequency which forms part of a complex tone. Any component of a sound, whether or not it is an integer multiple of the *fundamental frequency* is a partial. The fundamental frequency is the first partial of a tone.

PASC: See *Digital Compact Cassette*.

passband: The passband of a filter is the frequency span that the filter passes, or the range of frequencies not attenuated by the filter. The passband is usually measured between the points where the response is 3dB down in amplitude relative to the maximum level. See *rolloff frequency*.

passive: A device is called passive if it contains no amplification circuitry and a signal suffers *insertion loss* in passing through it, i.e., more energy goes into the device than is available at its output. This is opposed to an amplifier or other device which has the potential for at least *unity gain*. Many audio equalizers are passive, as are most *crossover networks*. Passive devices in general do not add any appreciable nonlinear distortion or noise to the signal, but they have so much attendant insertion loss that additional amplification is needed, and this always contributes some noise and distortion. Passive devices can and do cause *phase distortion*, however. As opposed to *active*.

passive crossover: See *crossover network*.

passive equalizer: An equalizer that employs only *passive* electronic components, i.e., resistors, capacitors, and / or inductors. Since these components require voltage, passive equalizers can only cut each operating band, the output signal level is necessarily lower than the input level. See *active equalizer*.

patch: (1) (*verb*) To connect together, as the inputs and outputs of various modules, generally with *patch cords*. (2) (*noun*) The configuration of hookups and settings that results from the process of patching, and by extension, the sound that such a configuration creates. Patch is most often used to denote a single sound or the contents of a memory location that contains parameter settings for the sound, even on an instrument that requires no physical patching. A synonym for *sample* or *program*.

patch bay: A group of similar receptacles, or *jacks*, in an audio system. The act of plugging and unplugging the patch cords is called *patching*. Also called a *router*, *jack field*, or *jack bay*. Increasingly, physical patching is being replaced by digital routing.

patch cord: A short cable, typically fitted with a *phone plug* or *TT connector* at each end, used to make a connection between two points on a patch bay.

patch map: A map with which any incoming MIDI program change message can be assigned to call up any of an instrument's *patches*. This is a table set up by the user with entries such as 1=3, 2=2, 3=984, etc. See *patch mapping*, *MIDI Mapper*.

P

patch mapping: A Program Change message is limited to only 128 values, while some synthesizers can store many thousands of separate patches. This would mean that only 128 of the programs could be accessed via MIDI. Patch mapping is a process whereby a given program change number received by a MIDI device can be linked to any one of the available patches, as determined by a *patch map*. To get beyond the 128 limit, the MIDI *Continuous Controller* message, Bank Select, has been defined for selecting different banks of sounds prior to sending a program change number.

patch point: A location in an electronic circuit at which access to the circuit is provided by a *jack* in the patch bay or console *channel strip*. See *normalled connection*, *output*.

path length: The distance between a sound source and the listener or microphone. See *near-field*, *far-field*.

pattern looping: A digital composition technique whereby long, looped samples are mapped into a sampler along with other samples such as bass riffs, drum variations, and solo samples (vocal sounds, *effects*, etc.) The different loops and solo sounds are brought in and out via a keyboard to create a finished composition.

PA version: Public Appearance version. A prerecorded tape or just the instrumental backing of a song, with which a solo vocal artist can sing during public appearances to promote a record.

PCI: Peripheral Component Interface. An internal bus architecture for PCs and the Mac commonly used for digital audio cards.

PCM: Pulse Code Modulation. A technical term for *sampling*. Any digital method of encoding and decoding the amplitude of an audio signal. For example, an 8-bit PCM yields amplitude values of 0-255, and produces attendant *sampling errors* and *quantization errors*. PCM cards are always ROM, and contain only sampled *waveforms*, contained in a *wavetable*. See *PWM*, *split-band*. See also *PAM*.

PCM-F1: A reference to the (discontinued) Sony digital recording system which used an EIAJ-format, 16-bit PCM processor to convert audio into a digital form that can be stored on consumer videotape. The first attempt at digital audio.

PD: See *ProDigital*.

PDL: Projectionist Dummy Loader. Union designation for the person in a film recording facility who functions both as projectionist and as a machine room operator.

peak: Peak value is the maximum instantaneous excursion from zero of an audio waveform, as measured by a peak meter (*PPM*). The peak value of a sound is also the maximum instantaneous pressure excursion of the sound. See *crest factor*, *VU meter*.

peak expansion: The adjustment of an *expander's* threshold so that most program material passes through unaffected, but peaks or *transients* are heavily expanded. Used to restore peaks to program material that has been overly *compressed*.

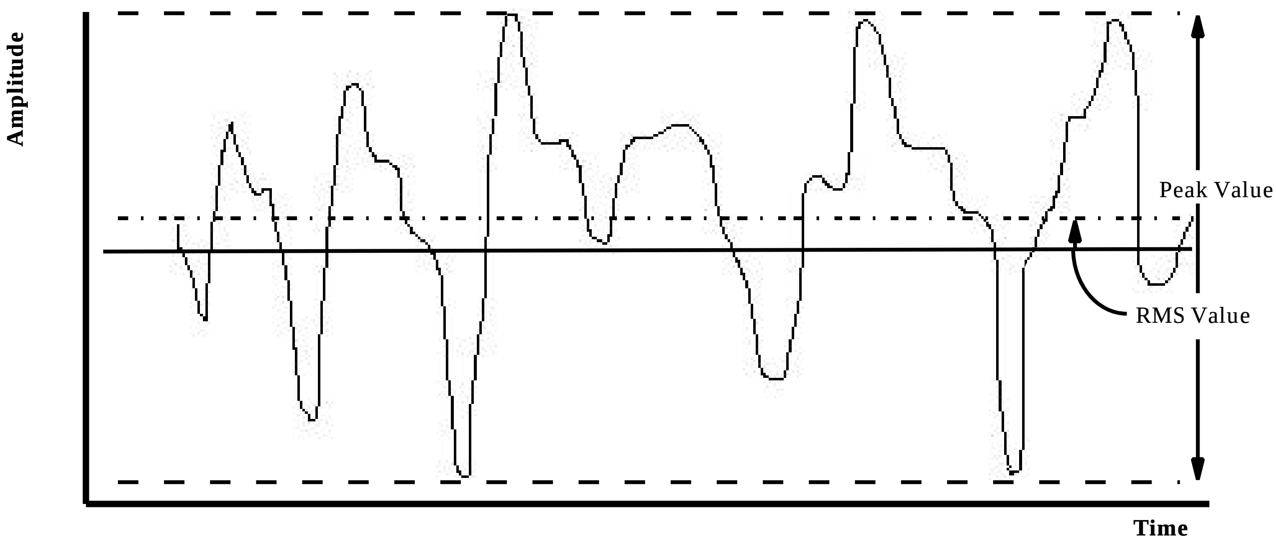
peak hold: A function of some volume indicators that indicates the *peak* level of the signal and holds that level until it is either exceeded by a higher peak or the indicator is reset by a time delay or manual reset.

P

peaking equalizer: Another name for a *parametric equalizer*.

peak level: See *peak*.

peak-to-peak value: A measure of the highest positive-to-negative voltage swing in any specified segment of the program signal. Twice the absolute value of the greater voltage reached by an adjacent positive or negative transient peak. See *PPM*.



Peak-to-Peak Value

peanut microphone: See *Lavalier microphone*.

PEC/direct: Photo-Electric Cell. In film re-recording, the act of switching between playback from the recorder either in playback mode or

pedal: (1) A foot-operated lever on a musical instrument. On a piano, the indication, means to use the sustain pedal. (2) Strictly, pedal point. A sustained or continually repeated note that occurs throughout a passage of changing harmonies. Commonly on the dominant note, it creates a feeling of tension; on the *tonic*, it creates a sense of repose. It often occurs in the bass; if played above the other parts, it is known as an *inverted pedal*. Also known as a *drone* in folk music.

pedalboard: A large and widely spaced keyboard designed to be played by the feet, commonly found on organs, although also available as a type of MIDI controller. Generally used to play bass notes.

perfs: Perforations. Sprocket holes in motion picture film.

pentatonic: A *scale* in which the *octave* is divided into five notes.

percentage quantization: A method of *quantization* in which notes recorded into a sequencer with uneven rhythms are not shifted all the way to their theoretically perfect timings, but instead are *humanized*, with the amount of shift being dependent on the user-selected percentage, called *quantization strength*.

P

perceptual coding: Any audio compression scheme that deletes audio frequencies that are masked by other, more dominant frequencies, and thus are not perceived by the listener. See *split-band coding, transform coding, temporal masking, MPEG, PASC, ATRAC,* and *AC3.*

percussion waveform: A percussion waveform is generated through the *random noise* generator of a synthesizer. If a second-order (two-*pole*) filter that has a high *Q* (low loss) is used after the noise generator, it simulates a drum. If repeated at a predetermined rate, it is called a *repeat-percussion waveform*, which constitutes the basic rhythm section of a synthesizer.

period: In a waveform that repeats a particular pattern over and over, the time required for one repetition of one wavelength is called the period. The waveform is then called *periodic*, and can be expressed as the summation of a series of *sine waves* called *harmonics*. The length of the period of the waveform varies inversely with frequency.

perspective: The effect of front-to-back depth in an audio signal. This may be inherent in the stereo image or may be artificially achieved by varying the amount of *direct sound* and *reflected sound*. The indirect signal may be obtained from mics places away from the source, or may be generated by a reverberation unit.

PFL: See *pre-fade listen*.

PFX: Production sound effects. Tracks of sound effects, as opposed to *Foley*, music, or dialog, which get mixed into the final *DME* tracks of a *film soundtrack.* Sometimes Foley gets mixed into the dialog stem, and *PFX* into the effects stems to separate the effects which accompany the dialog from *BG / walla* effects.

phantom image: A *monophonic* sound panned equally to both speakers.

phantom power: DC power (usually 9V to 52V) supplied to *condenser microphones* by a preamplifier close to the microphone, necessary due to the extremely high *impedance* of the microphone, via the signal wiring of the microphone.

phase: Phase is defined as the time relationship between two corresponding points on a continuous wave, or the angular, or time, displacement between the voltage and current in an AC circuit. A *sine wave* signal is the simplest possible *waveform* and goes through 360˚ in one cycle, whereby it returns to its starting point. Another way of saying the same thing is that the signal has gone through 360˚ of phase angle, or phase change. The phase is actually a measure of time, 360˚ equaling one period of the signal. The time represented by a phase change of a certain number of degrees is thus dependent on frequency.

Phase-Alternating Line: See *PAL*.

phase cancellation: An attenuation of signal components resulting from combining *out-of-phase* waveforms. When two waveforms are mixed, their *harmonics* are added. If these signals are out-of-phase with each other, the amplitudes of the harmonic components differ at various times, as determined by the phase relationship. If the added harmonics have the same polarity, the signal is reinforced at those frequencies, and vice-versa. See *phase distortion*.

phase coherent: A condition encountered in the summation of two or more in-phase signals, in which the signals combine constructively, with little or no *phase cancellation*.

P

phase compensation: In some tape recorders, there is a special *equalizer* whose purpose is to minimize *phase distortion*.

phase difference: The measure of time delay with which two identical signals reach a common electrical or acoustic point. In the case of a *sine wave*, for a pure tone of FHz, 360˚ of phase difference, D, represents one full cycle. If F = 1kHz, D = 360˚ takes exactly 0.001 sec. For any F, the number of degrees phase difference, D, between two sources whose identical outputs arrive at a common point with a time difference of T seconds is:

$$\mathbf{D = F \times T \times 360}$$

phase distortion: An effect caused when *phase-shift* in an audio device is not a *linear* function of frequency. In other words, different frequencies experience different time delays. This changes the waveform of the signal and is especially injurious to *transients*. Most *transducers* produce significant phase distortion. As low frequencies travel slightly faster than high frequencies and as air absorbs high frequencies more readily than low ones, the more delay there is between low frequencies and the higher harmonics of a sound, the sound becomes progressively more smeared and is perceived as more distant.

phase distortion synthesis: A form of modulation synthesis in which the spectrum of a *DCO*'s output signal is altered by modulating the DCO's *clock* frequency within each cycle, while the over-all frequency is kept constant. The oscillator's clock frequency speeds up and slows down, producing rapid *phase* changes as the waveshape is alternately compressed and expanded (distorted) to fit within the regulated period. Popularized by the Casio CZ-series synthesizers.

phase inversion: See *phase reversal*.

phase invert: See ϕ.

phase linear: The ability of an audio device to pass a signal without causing *phase-shift*.

phase-lock: See *sync-lock*.

phase-locked loop (PLL): A *closed-loop* electronic circuit that automatically adjusts and locks the frequency of an oscillator to the correct frequency for receiving a signal. The PLL is the preferred FM detector circuit in commercial systems today as it requires no tuned circuits, hence does not require alignment. It normally has high amplification which produces a strong output audio signal. Since it does not respond to amplitude variations, it also provides limiting action.

phase manipulation: A technique used by *aural enhancers* which realigns the relative phase of existing *harmonics*.

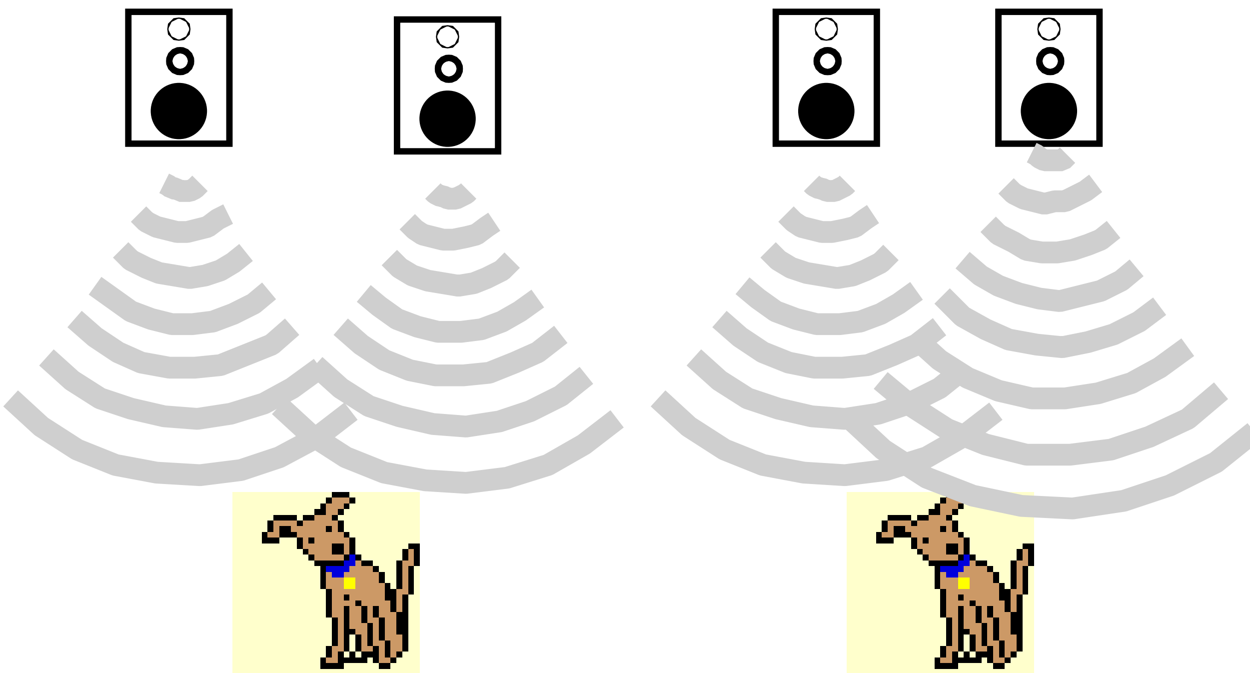
phase meter: An electronic circuit and display that compares two incoming signals and shows the *phase difference* between them.

phase modulation: See *phase-shift*.

P

phase reinforcement: The opposite of *phase cancellation*. An reinforcement of signal components resulting from combining *phase coherent* waveforms. When two waveforms are mixed, their *harmonics* are added. If these signals are not perfectly in-phase with each other, the amplitudes of the harmonic components differ at various times, as determined by the phase relationship. If the added harmonics have the same *polarity*, the signal is reinforced at those frequencies.

phase reversal: (1) The condition where the connection in one channel of a stereo signal are reversed. This is most likely to happen at the loudspeaker, and results in *phase cancellation*, particularly apparent in the bass. (2) In electronic signals, changing the *polarity* of the signal from positive to negative or vice versa, thereby causing a reversal in polarity of the signal. When viewed on an oscilloscope, the waveform flips with respect to the time axis. Also called *polarity reversal* or *phase inversion*. See *common mode, out-of-phase*, ϕ.



The pair of loudspeakers on the left are in phase--the speaker cones cause compression and rarefaction the surrounding air in unison. Those on the right are *out-of-phase*, causing the air compression generated by one speaker to be cancelled by the rarefaction generated by the other.

phase reversal switch: A switch, usually found in a *balanced line*, that allows the user to interchange the two conductors, causing a 180˚ shift in phase of the signal. This is often a feature of recording consoles to allow the engineer to optimize the phase relationships of multiple microphones placed in close proximity to each other, i.e., mics likely to pick up substantially identical signals, such as on a drum kit. See *phase cancellation*, ϕ.

P

phase-shift: An alteration of the phase in the partials of a tone. Virtually all signal processing devices will cause a certain amount of phase-shift, also called *phase modulation*, as none of them are completely phase linear. Phase-shift is a characteristic of a device and is the change in phase impressed on a signal that passes through the device. An electronic device will always add a time delay to an applied signal. If the time delay is constant on all frequencies, the phase-shift between the input and output of the device will be a *linear* function of frequency, and the device is called *phase linear*. Deviations from phase linearity are called phase-shift. Equalizers, in particular, exhibit large amounts of phase-shift. In a complex waveform, phase-shift will cause a distortion of the waveform, even though the *frequency response curve* may be perfectly flat. There is considerable controversy over whether the ear can detect this type of *phase* distortion. See also *PIM*.

phase-shifter/phaser: (1) Phasing is an effect on higher frequencies which make a whispering or ocean-like sound, produced by a device called a *phaser*, also called a *phase-shifter*. See *flanging*. (2) Originally one of the defined MIDI Controller Change messages. It was assigned to the parameter in a synthesizer which alters the depth of the effect described as *phasing*. More recently, this message has been reassigned as one of five generalized Effects Depth messages. See *effects control*.

phase sync: In *SMPTE timecode* synchronization, an option by which the slave machine is speed-controlled in such a way that the phase of its *bi-modulated sync tone* is held in phase with the sync tone on the *master* machine. This provides much closer alignment of the two than just *frame lock*. However, because the synchronizer must make continuous adjustments to the slave's speed, phase sync can introduce noticeable *flutter* when the audio machine is slaved to video. In some synchronizers, only sub-frame information is used to achieve interlock, yielding a 1/100 frame accuracy between machines.

phasing: See *phase-shifter/phaser*.

phon: A unit which takes account of the ear's nonlinear response to the *loudness* and frequency of a sound. The phon uses a *decibel* , i.e., *logarithmic*, scale which is based on the level of intensity of a given sound that corresponds to the dB rating of a pure tone at 1kHz, subjectively judged to be of the same loudness. An increase of one phon is about the smallest increment in loudness that can normally be perceived. The scale practically ranges from 0dB to 130dB, and its logarithmic nature means that a rise of three phons approximates a doubling in intensity. See *equal loudness curves*, *SPL*.

phon lines: See *equal loudness curves*.

phone connector: A ¼" plug connector, called such as it was originally developed by Bell Telephone. Used as audio connectors on electric guitars, synthesizers, and some signal processors and mixers. See also *TRS*, *TT*.

phono connector: Also called an *RCA connector*, these are generically known as *pin-jack connectors* as they contain both the pin and jack portions of a connector. Commonly used on home stereo equipment, the *phono* designation comes from the fact that they are almost universally used for the outputs on phonographs.

P

physical modeling synthesis: A type of *sound synthesis* done by programming a computer to mathematically model the physics of a particular instrument. These models are sets of complex equations which describe the physical properties of the instrument (such as the shape of the bell, or type and density of the material) and the way in which a musician interacts with the instrument, such as plucking, bowing, strumming, blowing, etc.

pick-up: See *DI*, *piezo pick-up*.

Picture Start: See *LFOP*, *sync pop*.

piezo pick-up: A device, often fitted into the bridges of acoustic guitars, where the mechanical vibrations in the bridge cause microscopic distortions to the shape of a piezo-electrical crystal, generating a small voltage in the process. The other common type of pick-up is the *electromagnetic* pick-up used in electric guitars. EM pick-ups use a coil of wire which senses any changes in the *magnetic* field created by a small permanent magnet. As the guitar string above the coil vibrates, it disturbs the magnetic field, and the coil generates a small electrical current which is passed onto an amplifier and loudspeaker. This is the pick-up part of the *DI*.

pigtail: The end of an audio cable which simply has bare wires rather than any type of connector, used to connect cables to binding posts or screw terminals.

pilot: A 19kHz tone transmitted along with stereo FM broadcasts in order to synchronize the local oscillator in the receiver to 38kHz for the detection of the stereo subcarrier. If not filtered out of the receiver output, it can cause problems with *Dolby noise reduction*. See *MPX*.

pilot tone: A 60Hz *sine wave* is recorded on one track of a tape which is used for motion picture sound recording, generated when the film is being shot, thus the frequency is an accurate measure of the camera speed. The pilot is then used later to synchronize the tape playback to the picture action, allowing movie sound to be recorded independently of the film, as *double system sound*. However, with a pilot tone, although the slave can *sync lock* with the master, the slave has no way of knowing where in the program the material the master tape is, and so is severely limited as a synchronization tool. The same is true for speed-only sync codes such as *FSK* and DIN sync. See *neo-pilot*, *control track*, *reference frequency*.

PIM: Phase Intermodulation Distortion. PIM arises in amplifiers that have a nonlinearity such that one signal will cause *phase modulation* of another signal. Phase modulation is the same as *frequency modulation* but to a lesser degree. For PIM to occur, a high-amplitude signal must modulate the *power bandwidth* of the amplifier, and this varying bandwidth varies the phase of another signal, also being amplified. See *intermodulation distortion*.

pin-jack connector: See *phono connector*.

pinch wheel: In a tape recorder, a free-wheeling rubber roller which presses the magnetic tape against the *capstan*, ensuring enough friction to drive the tape past the heads. Also called a *pinch roller*.

ping-pong: (1) A stereo effect generated by an autopanner or some multieffects units, whereby a sound is made to appear at the extreme right and left of the stereo field in rapid alternation. (2) See *bounce*.

P

pink noise: A type of *random noise* which has a constant amount of energy in each *octave* band, as opposed to *white noise*, which has equal energy at all frequencies. Pink noise can be made from white noise by passing it through a filter with a 3dB per octave *rolloff*. Pink noise is used to align the frequency response of tape recorders and loudspeaker systems.

ping-pong: To *bounce* tracks in a multitrack recording.

pinning: Referring to audio level meters, a condition in which the signal is too high, causing the indicator to hit the top of its scale. This can damage ballistic meters, such as *VU meters*, in addition to producing distortion in the program.

pirate ship: Film term with means to make a copy of material for one's own use. Commonly used to refer to making a copy of good sound effects recorded in production, thus the order to "pull up the pirate ship" and to make sure that those recordings will be available after the film is finished and masters are sent away.

pitch: (1) A sound characteristic of repeating vibration at a specific *frequency*. Unpitched sound is called *noise*. Pitch is measured in units called *Hertz* (Hz) which is equivalent to "cycles per second." For practical purposes, pitch and frequency are interchangeable terms. (2) The number of grooves per inch on the surface of a phonograph record. (3) The subjective impression of the frequency, or musical tone of a sound, expressed in the latter case by its name-number, e.g., A2. Also the frequency of that musical note, e.g., for this example, 440Hz. (4) The distance between two perforations or sprocket holes along a strip of film. Camera-original film is generally short-pitch, and print film is generally long-pitch, the difference in these lengths being on the order of .0006" per frame. The two different pitches are necessary to prevent slippage between original and print film as they wind around various sprocket wheels in the contact printers used to make most prints.

pitch-bend: A shift in a note's pitch, usually in small increments, caused by the movement of a pitch-bend wheel or lever; also, the MIDI data used to create such a shift. MIDI Pitch-Bend messages are a type of MIDI *channel message*, but not a MIDI *continuous controller* message. See *bend*.

pitch-shift: To change the pitch of a sound without changing its duration, as opposed to *pitch-transpose*, which changes both. Some people use the two terms interchangeably. Also called *time-stretching*. See also *frequency shifter*.

pitch-to-MIDI-converter: This translates a *monophonic* musical line, such as singing or a reed instrument, into a stream of MIDI data.

pitch tracking: A misleading term meaning frequency-to-voltage conversion. A pitch tracker will accept a complex *periodic* signal and extract from this the *fundamental frequency*. It will then convert this frequency into a direct voltage output that can be used as a *control voltage* in a synthesizer.

pitch-transpose: See *pitch-shift*.

P

pits: On a *CD*, *MD*, or *OD*, microscopic depressions laser-burned into the surface on which the digital data is stored. Each pit-edge encodes a 1 in the datastream. Incremental lengths of flat disc surface (either *land* between pits or the bottom of an extended pit) designate zeros in the data. Audio samples, location and synchronizing information, bands and indexing, visual information, etc. are all encoded in the pits.

Pit Signal Processing (PSP): See *digital watermark*.

pixel: PIXure ELement. The smallest visible element of a picture or image, corresponding in video to the brightness and color information for one location on a single *line* of the video image.

planar loudspeaker: A type of *dipole* loudspeaker design which combines aspects of both *dynamic* and *electrostatic* designs. The planar speaker consists of a large plastic sheet with conducting wires imbedded in it, these wires functioning as the *voice coil*. Many small magnets in front and behind the sheet set up a magnetic field so current in the wires causes a force on it and it moves as a unit, similar to an electrostatic speaker. Planar speakers suffer from the same directional problems as electrostatic loudspeakers, but their impedance is more similar to dynamic designs.

plastic leader: Usually, white or yellow *leader tape* in between songs on a master tape. Plastic leader can pick up static electricity, which can create clicks or pops on the master tape. For that reason, *paper leader* is generally used on masters. There are special types of plastic leaders made of an anti-static base that are used for archival storage where paper leader, which changes shape in varying humidity and deteriorates with age, will not suffice. See also *leader*.

plate: See *back plate*.

plate reverb: An electromechanical substitute for an acoustic reverb chamber, where electronically generated reverb was unavailable, whereby a metal plate was suspended behind the sound source, fitted with a *transducer* and microphone pick-up. The plate was typically 4' by 6', suspended on springs within a sound-deadening case as a reverberant space. A vibrating transducer feeds the *direct sound* into the metal plate, and a pair of pick-ups extract the reverberation as vibrations bounce off the plate's edges. A motorized damping plate parallel to the main plate can be remotely positioned at varying distances to control the duration of the reverb. The plate has a characteristic metallic, bright sound. Other substitutes were *spring reverbs* and *slap echo* devices.

platter projection: See *projection*.

playback: (1) The amplified reproduction of any type of sound recording. (2) The reproduction of a recorded take immediately after it is recorded, done to make æsthetic and technical judgments about the performance and recording quality. (3) On a motion picture set, the reproduction of music or other sounds recorded previously under studio conditions, to which the actors, singers, and dancers in a scene mime and move in exact synchronism. Called *shooting to playback*.

playback-equalization: See *record-equalization*.

P

playback head: The *head* on a tape recorder that is used to detect the varying remanent magnetism present on the tape. The output of this head is then amplified and heard as the recorded program.

playlist: (1) A list giving the chronological order in which a number of pieces of music or sound effects are to be played. The list will often describe the start time, duration, and finish time of each item. (2) In editing, particularly digital audio editing, a list similar to (1) above which gives the order in which sections from various recording takes will be used. It will usually include timing information which may be locked to timecode, as well as information about type and duration of *crossfades*, etc. Both also known as an *EDL*.

plosive: See *pop filter*.

plug-and-play: An oft touted feature of PC-type devices, making the promise that the device, when attached to your system, will simply work as advertised, without making you deal with arcane hardware or software settings and/or subtle issues of compatibility. The typical real situation is “plug-and-pray,” the more common epithet.

plug-in: A third-party software program sold to add additional function to an existing, larger software suite. An example of a plug-in is the Waves™ *TDM*-based plug-in audio diagnostic suite which runs under ProTools.™ There are two types of plug-in: file-based and *real-time*. The former are usually less expensive, less powerful and require the user to wait while the effect is calculated by the computer. The plug-in modifies the data on the disk (*destructive editing*), but no additional hardware is required to use the plug-in. Real-time plug-ins allow the user to hear the effect while the music is playing. The real-time plug-ins require dedicated hardware to process the sound in, of course, real-time. Changes in the original file are not saved unless requested by the user (*non-destructive editing*.)

poco a poco: Italian for “little by little,” or gradually.

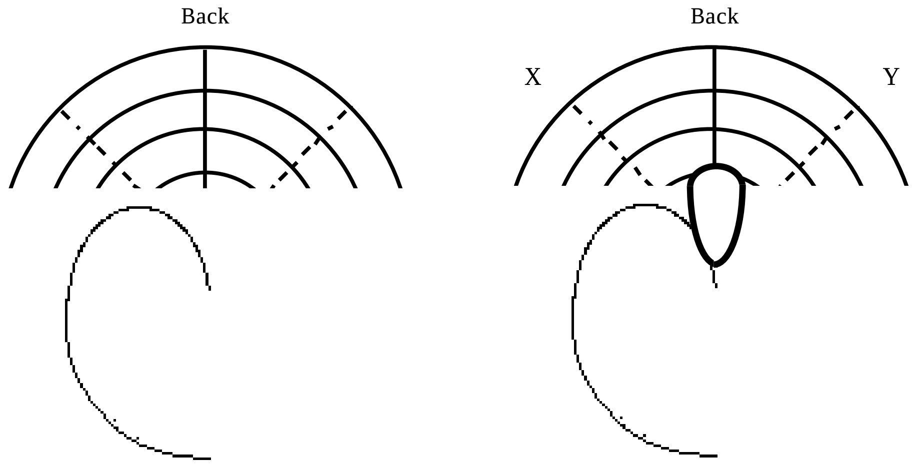
point source: A hypothetical sound source which is very small compared to the wavelengths of the sound it is radiating, and which is radiating into a *free-field*.

polarity: The orientation of magnetic or electric fields. The polarity of the incoming audio signal determines the direction of movement of the loudspeaker *cone* or microphone *diaphragm*, i.e., the sign, + or -, of the transducer’s output voltage when a positive sound pressure strikes the microphone, or when the speaker cone is pushing away from the cabinet. Note that on a microphone, by convention, positive sound pressure, which pushes the diaphragm in, makes a positive voltage on mic pin 2 with respect to pin 3. A sound is perceived as being louder if the largest peaks are in positive polarity, that is when the *compression* portion of the sound wave is pushing the speaker cone outward, toward the listener. Polarity reversal is the same as *phase-reversal*, or 180˚ of *phase-shift*.

polarizing voltage: The DC voltage supplied in opposite polarities to the *plate* and *diaphragm* of a *condenser microphone* (via *phantom power*) or *electrostatic loudspeaker*.

P

polar pattern: A circular, two-dimensional plot that indicates the directional response of a transducer, such as a *directional microphone*. Whole polar patterns are commonly used to show microphone response patterns, they also can indicate the dispersion of a speaker. In the diagram below, points X and Y indicate response nulls in the *hypercardioid* microphone, which represent optimal angles for the placement of stage monitors. A good microphone should have a similar polar pattern from 200Hz to 10kHz, otherwise the mic will produce noticeable *off-axis coloration*. Also called a *polar diagram* or *response pattern*. The *cardioid* pattern microphone is most sensitive to sounds coming directly from the front and least sensitive to sounds coming directly from the rear. The hypercardioid is most sensitive to sounds coming directly from the front and least sensitive to sounds coming in the direction of X and Y, a good place for stage monitors.



Patte

pole: A portion of a *filter* circuit. The more poles a filter has, the more abrupt its *rolloff slope* will be. Each pole causes a slope of 6dB per octave; typical filter configurations are two-pole and four-pole (12dB / octave and 24dB / octave, respectively). See *order*.

Poly Mode: A MIDI reception mode in which a module responds to note messages on only one channel, and plays as many of these notes at a time (*polyphonically*) as it can. In a guitar-to-MIDI converter, Poly Mode allows multiple strings to share the same channel. Compare with *Mono Mode*.

polyphonic: (1) Music which simultaneously has more than one independent melodic line. (2) Capable of producing more than one note at a time. On most electronic organs, all of the notes can be sounded polyphonically at once, but all synthesizers place a limit on how many voices of polyphony are available. *General MIDI*-compliant synthesizers are required to provide 24 voices of polyphony. Compare with *multitimbral*.

poly pressure: Polyphonic pressure. A type of MIDI Channel Voice message in which each key senses and transmits *aftertouch* data independently. With poly pressure, if a chord is played pressing into the top note, only this note will be modulated without affecting the others. Expensive to implement, so rarely seen. Also called *poly aftertouch* or *key pressure*. Compare with *channel pressure*.

P

pop a track: The act of aligning a *two-pop* exactly nine feet from the *start mark*, either on *mag film* or on a *bench*, or in a digital audio workstation.

popcorn noise: Film expression for *ambient noise* in a theater environment that influence the low end of the *dynamic range*, and how soft a sound will be heard (or "read") in an actual theater. See *Little Old Ladies with Umbrellas*.

pop filter: A device that is used to reduce the *popping*, distortion caused by *overmodulation* of a microphone, usually as a result of placing it too close to the sound source. It commonly occurs with the vocal *plosives*: b, p, and t. Also called a *pop screen*, *windscreen*, or *windshield*.

popping: See *pop filter*.

port: (1) An opening in the front surface or baffle of a loudspeaker enclosure, called a *ported enclosure*. (2) One or more openings on the body of a *unidirectional* microphone, either rear-entry or side-entry. Passing through the port(s), sound waves reach the rear side of the *diaphragm* and, through *phase cancellation* or reinforcement, contribute to its *directional* response pattern. See *acoustic labyrinth*.

portamento: (1) A continuous movement in *pitch* from one note to another without step. Instruments with notes of fixed pitch, such as the piano, are unable to do this, but the human voice, fretless instruments, e.g., violins, and instruments fitted with slides, e.g., trombones, can. It is a different effect from *glissando*. (2) See *glide*. (3) One of the defined MIDI Controller Change messages used to switch a synthesizer's portamento on or off.

Portamento Time: One of the defined MIDI Controller Change messages assigned to the parameter in a synthesizer which determines the time taken for *portamento* to occur.

positional reference: A signal that provides location information that various devices can use to establish *synchronization* during playback and recording.

post: (1) Means "after," as opposed to *pre*. In recording studio parlance, it is used to indicate that the signal has already had the designated effect added, such as post-equalization, post-effects, or post-fader. Also used as an abbreviation for *post-production*. (2) Binding post. A connector, consisting of a threaded shaft and nut, used to terminate bare wire. Usually found on loudspeakers and occasionally amplifiers. See *pigtail*.

post de-emphasis: The same as *de-emphasis*.

post fade listen: Also called *after-fade listen AFL*. Signal routing within a *mixing console* to allow audio signals to be monitored at the level set by the fader on that input, rather than monitoring the level coming to the input, as in *pre-fade listen*.. *Aux sends* are usually monitored post-fader. Because of the ambiguity of the abbreviation PFL, be sure to use AFL (after-fade listen) for post-fade listen as PFL is generally taken to denote pre-fade listen.

post-fader send: See *effects send*.

P

post-production: (1) In audio, a term used for tasks which have to be done after mixing is complete and the *master tape* has been made. These include further editing, grouping of individual tracks into an album, and for CD, the addition of *subcode*. (2) Any work on a recording, film, or video that is done after the main recording session or filming is completed. Typical post-production work can include editing, mixdown, looping, sound cutting, making of titles, etc.

post-roll: (1) The number of beats / amount of time a *sample* ends after the playback-end marker. (2) In *SMPTE timecode* synchronization for videotape *post-production*, the number of seconds and / or frames specified which are automatically added to any timecode address subsequently specified as a *record-out* point. Thus, when inserting dialog or music, on each take master and slave machines will continue to roll for the post-roll duration after recording is stopped. This prevents the type of *wow* that occurs when a transport is stopped while the electronics are still recording. Some brands of synchronizer add the post-roll to the *mark-out* rather than the record-out location.

post-score: To compose and / or produce a musical score or jingle after the film or videotape for the production has been shot, and usually after a *fine cut* is completed. The composer can then spot *cues*, take counts, locate hits and really tailor the music to the visuals.

post-stripe: To *re-lay* the mixed soundtrack onto the edited video master, or onto a copy made from it.

post-sync: Post-synchronization. (1) *ADR* recording with *M&E* made to synchronize with an existing film or video tape, usually for translation of a foreign-language film. This term is used mainly in Europe, while in the U.S., the term ADR is used as a synonym. (2) The recording of sound to be added to the *sync sound*.

pot: Potentiometer. A device (commonly attached to a knob or slider) used to adjust some aspect of the signal being passed through it, or to send out a control signal corresponding to its position. A variable resistor, usually controlled by a rotary knob, is used extensively as a volume control, tone control, etc. The term is also sometimes used for a step-type *attenuator*.

power: (1) Electric power: The time-rate of doing work or the rate at which energy is used. A *watt* of electrical power is the use of one *joule* of energy per second. *Watts* of electrical energy equals *volts* times *amperes*. See Appendix B. (2) Acoustic power: The number of watts of energy produced by any sound source or *transducer*. See *acoustic intensity*.

power amplifier: A device that accepts a low-level audio signal and strengthens, or amplifies, it to a suitable *voltage* and *current* level adequate to drive a loudspeaker or similar load. See *preamplifier*, *active*, *differential amplifier*, *integrated amplifier*, *combining amplifier*.

power bandwidth: The bandwidth of an audio device such as a power amplifier, measured while delivering its full *rated load*. This is limited by the *slew rate* of the device and is always narrower than the *small signal bandwidth*.

power factor: See also *watt*, *VA*.

power level: See *level*.

power ratio: See *decibel*, *voltage gain*.

P

PPM: Peak Program Meter. Similar to a *VU meter* in appearance, but which responds to the *peak* level of the signal, rather than the average level. PPMs read the peak-to-peak value, i.e., the voltage swing between the negative and positive peaks. However, they still don't read absolute peak values, because clipped peaks shorter than a ms or so are generally inaudible. Peak detection usually is most effective with percussive sounds. See also *modulometer*.

ppq: Pulses Per Quarter-note. The usual measure of a sequencer's clock *resolution*. Sometimes written as *ppqn*.

PQ subcode: These control bytes contain the timing information which allows the CD player to cue instantly to the beginning of each selection, display the selection's number and running time, and provide a continuous display of elapsed time. See *Control and Display signals*.

preamplifier: (1) In an audio system, the first amplifier to accept the signal from the *transducer* is generally called a preamplifier. Preamps must usually accept very low-level, e.g., *mic-level*, signals and amplify them to *line-level* without adding appreciable noise. (2) A small amplifier built into a *condenser microphone* to boost the very low output level of the capsule before transmission over the mic cable.

Precedence Effect: See *Haas Effect*.

pre-delay: In *reverberation*, the time between the *incident* sound and the first sound reflection is heard.

pre-echo: (1) Depending upon the width of each frequency band in a *transform coding* scheme, it is possible for the decoder to produce variable amounts of pre-echo. For example, a *transient* that has its rising edge contained within a *frequency band*: the *codec* quite correctly detects the transient's presence and codes the information within the appropriate band, and also uses that information to re-use bits from surrounding masked envelopes. Upon decoding, the output level is turned on the for complete duration of the coded frequency band, which means that the signal will be heard before the arrival of the actual transient. Such pre-echoes sound very much like analog tape *print-through* and may produce unacceptable results, particularly on material containing large numbers of sharp level excursions. (2) In room acoustics, any *early reflections* of sound that occur within about 40 ms, the shortest time for which the ear can distinguish two non-simultaneous sounds. Some digital reverb devices include simulated pre-echoes as a part of their hall simulation program.

pre-emphasis: (1) Generally, the process of equalizing a signal to increase the content of a desired frequency band before the signal is sent to another device. (2) A type of high-frequency boost applied to signals about to be broadcast on FM stations or recorded on tape to reduce the apparent noise level. Pre-emphasis brings the high-frequency content of the music up to a level further above the *noise level* of the recording medium. In order to restore the proper balance of high and low frequencies to the reproduced music signal, the boost added by pre-emphasis must be removed by a complementary cut, called *de-emphasis*. See *equalization curve*.

pre-fade: A *fade-out* starting at a predetermined time so that it finishes precisely at the end of a recording.

P

pre-fade listen: Abbreviated PFL. A monitoring point placed before a fader on a mixing desk so that the signal can be listened to before being *boost* or *cut* for recording or broadcasting, i.e., the incoming signal can be heard regardless of the position of the fader on that particular input. Note that *aux sends* are generally monitored *post-fade listen*.

pre-fader send: See *monitor send*.

pregap: A *mixed-mode* CD encoding format designed to hide the computer data at the front of the audio portion of the CD, between index 0 and index 1 of track one. A CD player still thinks it's an audio-format CD, while a computer thinks it's a CD-ROM. Pregap improves upon mixed-mode by not allowing the audio user to access the *Enhanced CD* track directly, but there are other problems: patent disputes and a bug in Windows'95 makes the pregap track inaccessible to PC users. See *CD Extra*.

pre-lay: Usually stands for the act of editing sound onto a *multitrack*, i.e., multitrack editing.

premix: (1) (*noun*) If many tracks of effects or music are required for a specific scene, the mixing engineer may elect to mix all of these effects together onto a single strip of magnetic film or onto a single track (or at least fewer tracks) of the multitrack *master*, then use this one track during the final mix rather than the individual effects tracks. Because this effects mix is done before the *final mix*, it is called the pre-mix. Dialog premixing often does not actually reduce the number of tracks that will go into the final mix, but instead just copies a *cut track* across with careful equalization and fader moves. See *binky*. (2) (*verb*) To mix and bounce two or more tracks of a multitrack tape before making the final mix of all tracks. Done to free up tracks for additional instruments or voices, or to save time in the final mix by having sections already mixed.

pre-production: A term used for those tasks which can, or need to, be done before the actual recording or filming is made. A pre-production suite may be provided for the programming of synths and preparation of samples and sequences for use during the session.

pre-roll: (1) The number of beats / amount of time a sample starts before the playback marker. (2) In *SMPTE timecode* synchronization for videotape *post-production*, a number of seconds and / or *frames* which are to be automatically subtracted from any *record-in* point (some brands of synchronizer subtract the pre-roll from the *mark-in* point, rather than the record-in point.) The synchronizer will then return master and slave machines to a point located ahead of the record-in by the duration of the specified pre-roll. This assures that all machines will be up to speed, properly interlocked, and that the talent can find their place and be ready to perform by the actual record-in point. For example, if the record-in point is 20:10:00, and the pre-roll is 5:00, the deck will actually stop at 20:05:00, precisely five seconds before the specified record-in. See *post-roll*.

pre-score: To compose and / or produce a musical score or jingle before the film or video has been shot. This can be done in anticipation of filming live actors to playback, or in animation to provide the animator with exact scene lengths, *cue* points, and other information necessary to match the animated scene to the music. Almost all animated films are pre-scored, including dialog and effects, after which frame counts are taken from this in the music so that the animators can time actions to fit the soundtrack. The opposite of *post-score*.

P

presence: The intelligibility of a track; a subjective term describing the amount of mid- to upper-midrange frequencies in the sound source. Boosting a track's presence helps to bring the track forward in the *image*, and is effected by increasing the amplitude of frequencies in the range of approximately 800Hz-6kHz, typically in the more narrow band of 2kHz-4kHz.

presence peak: A characteristic of some transducers, typically *dynamic microphones*, whereby the *frequency response* in the 5kHz range is naturally boosted, giving an edgier, punchier sound.

preset: (1) A factory-programmed synthesizer *patch* that cannot be altered by the user. (2) Any patch. Some synthesizer manufacturers make distinctions between presets, programs, and/or patches, each of which contains a different set of *parameters*.

pressure: See *aftertouch*.

pressure gradient: A type of *microphone* construction which supports the *transducer diaphragm* on the top and bottom, leaving it open to the air on both sides. This produces *a figure-eight* response pattern. Mics of this type are more complex and delicate than *pressure operation*-type mics, and have many more mechanical and physical problems such as extreme susceptibility to handling noise, *rumble*, sensitivity to wind, and *proximity effect*. In addition, the diaphragm assembly has to compensate for the inadequacies of the pressure gradient by making the diaphragm resonate at very low frequencies, generally restricting the smoothness and extension of the very lowest part of the audio spectrum. These disadvantages are, however, overlooked because of the *directivity* of these microphones. *Cardioid microphones* are a combination of pressure gradient and pressure operation transduction within a single unit.

pressure operation: A type of *microphone* construction, using a sealed box with a *diaphragm* on one end (much like a drum), producing an omnidirectional *polar response*. This design usually provides a very smooth, extended, and flat *frequency response*. See *pressure gradient*.

pressure sensitivity: See *aftertouch*, *channel pressure*, *poly pressure*.

pressure zone effect: See *boundary effect*.

prestriping: See *stripe*.

preview codes: *Edgecodes* on an edited *workprint* or its copies and sound elements to create a new reference for a given version of the film. When the film is subsequently re-edited, the process of *conforming* multiple tracks can be sped up greatly.

preview head: A supplementary *playback head* on a tape recorder designed for reproducing the master tapes for the manufacture of phonograph records. The signal from the preview head tells the variable-pitch circuitry on the cutting lathe the program amplitude level that it will cut one groove later. This allows the unit to space the grooves so that they do not intersect due to amplitude *transients*. In quieter passages, it allows the circuit to pack the grooves more closely, minimizing *land*.

P

print master: The final edit of a *film soundtrack* that can be transferred directly to a *track negative* or a *mag stripe print* with no further changes in level or EQ. If *noise reduction* is used on a print master, it most often matches that of the final print format, and thus can be transferred stretched to the mag stripe print or track negative. In the case of a *stereo optical film*, the print master contains two tracks, *Lt-Rt*, that are transferred directly to an optical sound negative. The soundtrack of a *discrete* 35mm 4-track or 70mm 6-track mag print will be recorded from a 4- or 6-track printing master in a *real-time* transfer. Also called a *running master*.

print-through: When tape recordings are wound tightly on the reel, the adjacent layers of tape sometimes influence one another so that the signal from one layer will bleed into the next layer of tape. This causes a faint echo of the signal which may be heard as a *pre-echo*, audible before the main signal. Print-through is worse at recording levels which approach the *tape saturation level*. See *heads-out*.

processor: Sometimes used synonymously with the term *effects* device, a processor circuit modifies a signal passing through it, whereas an effects circuit leaves the original signal intact and adds something to it. Processors include *EQ*, *compressor / limiters*, *expanders / gates*, *panners*, and *single-ended noise reduction* units.

ProDigital (PD): (1) A digital audio format used in stationary-head *multitrack* digital tape recorders. In competition with the *DASH* format, ProDigital (also called ProDigi) supports two formats: ¼” 8-channel audio tape and 1” 32-channel audio tape, with various sampling rates, and allows editing of digital tapes by mechanical splicing and *punch-in* recording. (2) A *parallel, master clock* format for up to 32 channels of 16-bit audio. Also called *Melco*. There are three versions of Melco / ProDigital: Dub-A, Dub-B, and Dub-C.

production channel: See *channel*.

production master: See *master tape*.

production mixer: The person who records sound during filming. See *recordist*.

production sound: See *location sound*.

program: (1) (*verb*) To create a synthesizer *patch*. (*noun*) A patch. Also called a *preset*. (2) (*noun*) The desired audio or video signal passing through any system or stored on any medium such as tape, as opposed to *noise*.

program chain: The series of components or devices which is used to process a signal. It usually starts with a microphone and ends with loudspeakers, incorporating a mixing desk, tape recorder, etc. See *chain*. Also called a *set-up*.

Program Change: A MIDI message that causes a synthesizer or other device to switch to a new program / sound / patch contained in its memory. MIDI defines a range of 128 Program Change messages, numbered 0-127. GM goes further, assigning a specific type of sound (e.g., hi-hat) to a specific number.

programmable: Equipped with software that enables the user to create new sounds or other assignments by altering *parameter* settings and storing the new settings in memory. An individual control parameter is said to be programmable if its setting can be stored separately with each individual *patch*.

P

projection: In most commercial movie theaters, all reels are joined together on a platter to form one continuous strip of film through one projector. In screening rooms equipped with two projectors, each reel is kept separate and the projectionist will manually start the incoming projector with s/he sees *change-over dots* on the upper right-hand corner of the screen. This first set of dots is the *motor cue*, with a second set of dots (a second before the end of the current reel), indicating the time to switch to the next reel.

projection sync: The relative location of picture and sound in a motion picture print that produces proper synchronization during projection. In 16mm, *answer* and *release prints* are made with the soundtrack advanced 26 frames ahead of the picture. At 24fps, this distance represents over one second of footage. In 35mm, prints are made with the sound advanced 20 frames. Since each frame of picture must be held still while light shines through it, illuminating the screen, and since quality sound can only be read from film moving continuously past an optical or magnetic playback head, the picture frame and sound frame corresponding to the same event on film must be separated by a distance on film that will match the image projection with the sound reproduction. See *editorial sync*.

ProLogic: See *Dolby ProLogic™*.

propagation delay: The time taken for a signal to move through a circuit, system, or device.

proximity effect: A boost in the low-frequency response of a *directional microphone* that occurs when the sound source is relatively close to the microphone. The phenomenon begins when the source is about two feet away from the mic capsule and becomes more noticeable as the subject gets closer to the mic. A singer can use the proximity effect as a means of adding fullness to a voice; however, the effect can also emphasize low-frequency noises such as breath sounds and *plosives*. See *pop filter*.

PRS: Performing Rights Society. The UK equivalent of *BMI/ASCAP*. See also *MCPS*.

pseudo-balanced: See *floating unbalanced output*.

PSP: Pit Signal Processing. See *digital watermark*.

psychoacoustics: The study of the way in which audio information is processed by the brain. Humans have developed a number of techniques for processing sound. These techniques allow information to be recovered even when obscured by considerable noise and allow the brain to disregard unwanted information. See *cocktail party effect*, *auditory masking*, *perceptual coding*.

psychoacoustic surround-sound: See *transaural audio*.

pull: (1) See *cut effects*. Also, a film term which connotes the act of deciding which sound effects from a library will be used in a scene. See also *spot*. (2) Film term for adding another recorder to a chain.

pull up/pull down: The deliberate miscalibration of an audio sample rate clock in order to compensate for a speed change in the device, such as an analog tape deck, to which the audio is being synced. For example, used in cases where film footage running at 24 fps is translated to an NTSC video tape. See *frame*, *sampling rate*.

pulse: See *beat*, *clock*, *difference tone*, *ppq*, *pulse wave*, *sync pulse*, *tach pulse*, *tempo*, *trigger*.

P

pulse-code modulation: See *PCM*.

pulse wave: A generic term for a variable rectangular waveform that varies between high (+) and low (-). The *square wave* is a pulse wave with a ½ (50%) *duty cycle*, therefore the value of every even-numbered harmonic is zero. A pulse wave with a duty cycle of greater than ½ has the same spectrum as a pulse wave whose duty cycle has the same denominator (e.g., a ⅓ has the same spectrum as a ⅔ duty cycle.) See *PWM*, Appendix C.

pulse width: See *duty cycle*.

pulse-width modulation: See *PWM*.

pumping: See *breathing*.

punch-in recording: A feature that allows a user to enter (punch-in) or exit (punch-out) the recording function while a MIDI sequencer or audio recorder is playing. *Punching* often is used to replace certain sections of otherwise usable recorded art without having to redo an entire track.

pure tone: A sound whose *waveform* is a *sine wave*, or a signal with a single frequency and no *harmonics*.

PWM: Pulse Width Modulation. An analog synthesis technique in which an *LFO* or some other modulation source is applied to vary the length of time a *pulse wave* remains in its high state (i.e., its width). This varies the amplitudes of the *fundamental frequency* and lower *harmonic*s, with an effect similar to sweeping a *lowpass filter*. Used by video laser disc systems, and sometimes as an intermediate stage between sampling and A/D conversion. Better than *PCM* in that it only uses one bit and produces no *quantization noise*. It does, however, have attendant *sampling errors*. See Appendix C.

Pythagoras' comma: See *diatonic comma*.

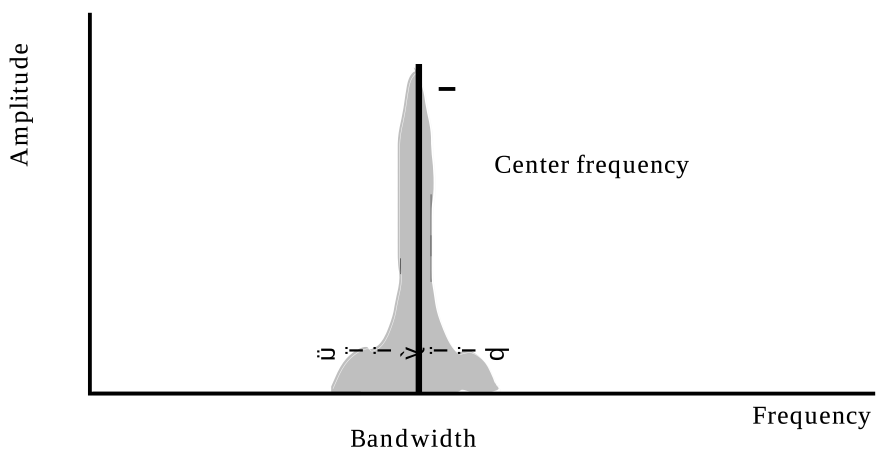
PZM: Pressure Zone Microphone. See *boundary microphone*.

Q

Q: In reference to a *resonant* mechanical or electrical circuit or a *capacitor*, Q stands for "quality factor." (1) In the case of a resonant system, Q is a measure of the sharpness of the *resonant peak* in the *frequency response* of the system and is inversely proportional to the *damping* in the system:

$$\mathbf{Q} = \frac{\textbf{center frequency in Hz}}{\textbf{bandwidth}}$$

Equalizers that contain resonant circuits are rated by their Q-value: the higher the Q, the higher and more well-defined the peak in the response. In filters, the ratio of a *bandpass* or *band-reject* filter's center frequency to its bandwidth defines Q. Thus, assuming a constant center frequency, Q is inversely proportional to bandwidth, i.e., a higher Q indicates a narrower bandwidth. Also called *Q-factor*. See also resonance. (2) In I systems, Q is a measure of the *directivity* of the sound output: a Q=1 means that the system radiates energy equally in all directions, or into 360˚ of solid angle. A Q=2 means that the speaker radiates into a hemisphere; higher values of Q mean that the speaker radiates into increasingly smaller angles, or in other words, has greater directivity. (3) Also a measure of *inductor* or *capacitor* efficiency.



Q

Q-factor: See *Q*.

Q-Lock: A brand of electronic synchronizer used for interlocking various audio and videotape recorders. The name is used generically for any such synchronizer. See *BTX*.

quadraphonic: An sound system which attempts to model a live acoustic using four audio channels to give the effect of sound arriving from different parts of the listening environment. See also *stereophonic*, *LCRS*, *surround-sound*.

quadrature: Two signals which are 90˚ out-of-phase with each other are said to be in quadrature. Also, a signal or function such as *impedance* will have a phase angle that varies with frequency or with time. The phase angle can be resolved into two components, real and imaginary, which have a 90˚ phase difference, where the imaginary part is called the quadrature part.

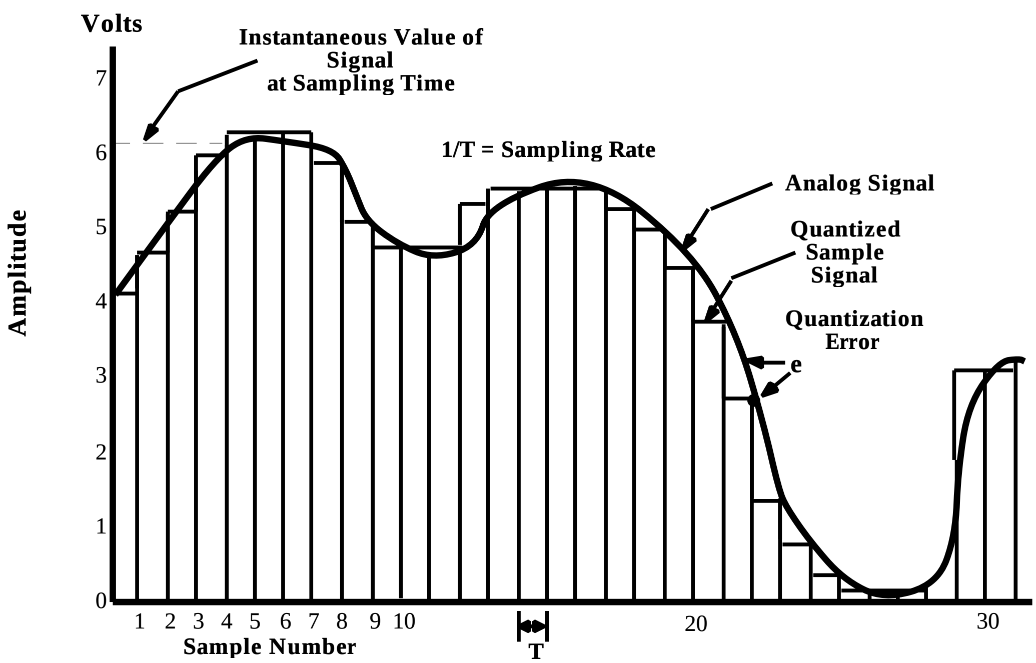
Q

quad track: A *track negative*, and any *release prints* made from the track negative, that contains all three digital sound formats: *Dolby Digital*, *DTS*, and *SDDS*, plus a standard *SVA* analog track.

quality factor: See *Q*.

quadruplex recorder: A *VTR* recording configuration in which four heads are mounted around a wheel that turns in contact with 2” tape. This system has largely been replaced by *helical scan* formats.

quantization: (1) The representation of an analog signal by a vector of discrete values. The signal, after quantization, has a stepped shape rather than its original continuous curve, and the difference between this and the original signal is *quantization error*. See *granulation*, *PCM*, *quantization noise*. (2) A function found on sequencers and drum machines that causes notes whose start time does not correspond to the beginning of a beat to be rounded off to the nearest rhythmic value. See *percentage quantization*.



Quantization

quantization distortion: See *granulation*.

quantization error: The difference between the actual *analog* value at the sample time and the nearest quantized (digitally encoded) value is called quantization error. At worst, the quantized value encoded will be no greater than one-half increment away from the actual analog value. Quantization error is related to the *S/N ratio* and the maximum number of quantization increments is related to *dynamic range*. See *bit depth*.

Q

quantization noise: One of the types of error introduced into an analog audio signal by encoding it in digital form. The digital equivalent of tape hiss, quantization noise is caused by the small differences between the actual amplitudes of the points being sampled and the *bit depth* of the *A/D converter*. In the quantization of a sine wave whose frequency is a submultiple of the sampling frequency, the error will have a definite pattern which repeats at the frequency of the signal, having a frequency content consisting of multiples of this frequency, where it can be considered as *harmonic distortion* rather than *noise*. In music, however, the signal is constantly changing and no such regularity exists, resulting in *quantization error*, producing wide-band noise, called quantization noise. See *granulation*.

quantization strength: See *percentage quantization*, *quantization*(2).

quantize: To produce an output in discrete steps. See *quantization*(2).

quantizing increments: (1) The total number of stepped levels, from *noise floor* to *saturation*, that an *A/D* has available for assignment of the continuously varying analog input voltage with each sample taken. For example, if each sample has a *bit depth* of 10, there will be 2^{10}, or 1,024, quantizing increments. (2) The voltage or decibel difference between any particular quantizing step and the next step higher or lower. In a system with 2^{10}, or 1,024 discrete steps, if signal voltage from noise level to saturation varies from 0.0V to 1.0V, each quantizing increment will correspond to about 0.001V. See *dynamic range*, *sound pressure level*.

quarter track: Sometimes called a *four-track*, refers to most home-type, reel-to-reel tape recorders which use one-fourth the width of the tape for each recorded track, allowing stereo signals to be recorded in both directions, doubling the recording time. Professional stereo tape recorders use one-half the tape for each track, resulting in better fidelity and reduced noise level. See *magnetic tape*. See also *half track*, *two-track*.

quarter-wavelength rule: If a wall is a *node*, then the nearest other node at any frequency will be ½(*wavelength*) away from the wall. Given this, the *antinode* is midway between those two points, or ¼(*wavelength*) away from the wall, for any given frequency. So, for example, if you want to filter out 60Hz, divide 18’ (the wavelength of 60Hz) by 4 = 4’6” and hang a thin layer of frictional material (such as fiberglass or acoustic foam) at that distance from the wall(s) and floor and ceiling.

quasi-balanced: See *floating unbalanced output*.

QUERTY: The usual typewriter keyboard, the same design used on computer keyboards, named after the characters which comprise the left-most letters of the top row, just under the numerals. While this arrangement of letterson the keyboard was designed a century ago to slow down the typing rate of agile typists so that the mechanical keys did not collide and interlock, the format is so ubiquitious that it has made the transition to the electronic computer keyboard, despite attempts to rearrange the keys to allow entry by the fastest possible fingers.

Q

QuickTime: A software multimedia environment developed by Apple Computer, running on the Mac or under Windows™. QuickTime enables the creation and playback of QuickTime movies featuring full-motion video, MIDI tracks and 16-bit *ADPCM* audio. However, QuickTime Movies (documents in QuickTime format) do not need to include pictures, and are a good way to distribute audio files. It is possible to import an AIFF, SND, or SoundEdit file into a QuickTime editing program, such as Adobe Premiere, and save the file as a QuickTime Movie, supported by a number of audio applications, including SoundEdit 16, Deck II, Peak, and Audioshop.

quiescent noise: (1) See *noise floor*. (2) The combined *intrinsic noise* produced by all sound reinforcement devices, plus all extrinsic noise present in the listening space, such as HVAC equipment, traffic noise, measured when the listening/recording space is empty. Sometimes used as a synonym for *noise floor*, confusing everyone.

R

rack: A type of shelving, usually with enclosed sides and back, to which audio components can be attached vertically, one on top of the other. Components are normally screwed into front-mounted, tapped metal strips with holes which are spaced so as to accommodate the height of devices of various *U*-sizes. Racks are usually 19” wide and have their height denominated in U-units.

radiation impedance: The acoustic *impedance* that acts as a load on a loudspeaker, opposing the motion of the cone.

radiation pattern: (1) The *polar pattern* that graphs a loudspeaker’s directional characteristics for a group of specific test frequencies. (2) The three-dimensional graph of the intensity with which any sound source emits various frequencies at all angles around itself.

radio frequency (RF): An *alternating current* or *voltage* having a frequency above about 100kHz, so-called because these frequencies are radiated as electromagnetic waves by radio and television, and as high as 30GHz (30,000 MHz). The constant frequency of the *carrier* wave (the frequency which you tune into) falls within this range. This is then *modulated* by the audio (or other) signal, according to some process such as *AM* or *FM*.

radio microphone: A mic with a built-in RF transmitter used instead of a cable connection to give a performer increased mobility. A receiver system picks up the transmitted signal for distribution to a PA, etc.

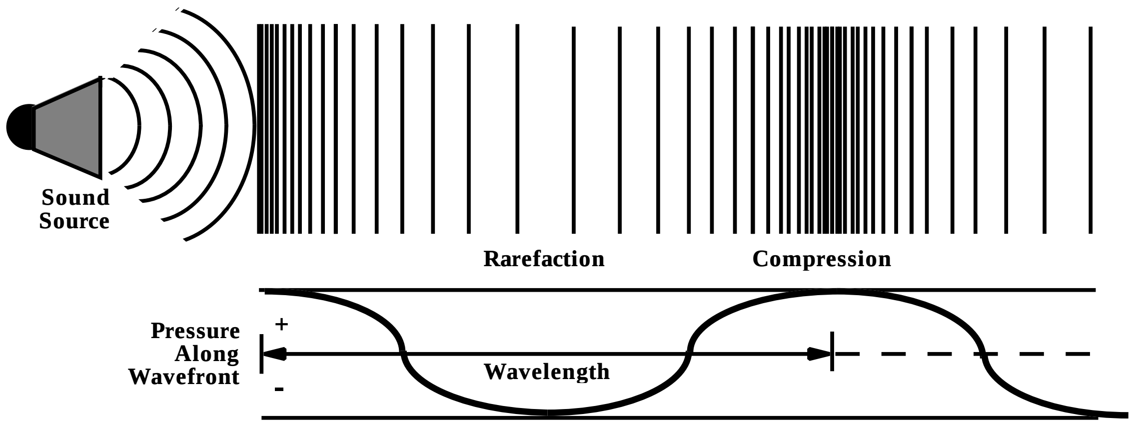
ramp wave: See *sawtooth wave*.

random access: Storage systems where data may be stored and accessed in any order, independent of the ordinal position of the data when it was originally recorded. This is the opposite of *linear*(3) access, or linear recording media such as magnetic tape which necessarily preserves the sequential relation of the data as it is recorded, and depends on this sequential relation for accurate playback. See *non-linear recording*.

random noise: Sound where there is no predictable relationship between the frequency or amplitude of the waveform over time. See *white noise, pink noise*.

R

rarefaction: The spreading apart of air molecules which lowers the local air pressure, during the second half of each complete cycle of a sound wave. This corresponds to the portion of the wave that appears below the x-axis when graphed. The opposite of *compression*.



Rarefaction and Compression of a Sound Wave

raster: The characteristic patterns of horizontal lines formed by the scanning beam of the TV picture tube. Also, the actual electronic circuit that creates the scanning spot that traces these *lines* on the TV screen.

rate: In a digital *delay* or *flanger*, a circuit and control that enable varying of the length of time during which the *depth* circuit completes one full increase or decrease cycle of the nominal delay time. To the ear, the rate control varies the speed of the apparent *vibrato* added to the input signal by the depth circuitry.

rate control: An *envelope* parameter which controls the rate or timing of certain synthesizer actions, such as the attack, decay and release portions of an *ADSR* envelope. Compare with *level control*.

rated bandwidth: The frequency range, normally 20Hz-2kHz, over which the performance of an audio device is rated with respect to specification characteristics such as *power output bandwidth*, *harmonic distortion*, etc.

rated load: The load *impedance* into which a power amplifier or loudspeaker is designed to operate safely, and upon which other rated characteristics are based. See *power bandwidth*.

RCA connector: See *phono connector*.

RCH:

R

R-DAT: Rotary-head Digital Audio Tape: A standard for digital audio tape recording which employs a rotating head mechanism similar to that of video recorders. Two channels of 16-bit digital audio information, plus *subcode* and track information, can be recorded onto a tape approximately 3mm wide, traveling at a very low speed of about 50 cpm. The tape is housed in a case similar to a video cassette, though much smaller. R-DAT machines offer three sampling rates: 32kHz, 44.1kHz, and 48kHz. Maximum continuous record time for a cassette is two hours at standard play and four hours at long play, with reduced quality. See also *DAT*.

reach: The clear pick-up of quiet, distant sounds by a microphone due to a high *S/N ratio*. See *self-noise*. The higher the *SPL* of the sound source at the mic, the higher the S / N ratio. Given an SPL of 94dB, a S / N spec of 74dB is excellent, 64dB is good. The higher the S / N ratio, the cleaner (more noise-free) is the signal, and the greater the reach of the microphone.

reactance: The complex component of *impedance*.

read: The quality of a sound in the context of its placement within a film, theater, or other audio situation is described as how the sound "reads." See *popcorn noise*.

read mode: In console *automation*, the operational mode in which automation data concerning the fader level or other parameter for each channel is read back from data storage and used to reproduce those settings in *real-time*, actually controlling each parameter exactly as it was done on the recording pass.

ready: An operational mode of tape recorder electronics. For any tracks placed in ready mode, the record circuits are enabled. When the master record button is pushed and the tape begins moving, the ready tracks begin recording. The opposite of *safe* mode.

RealAudio (.RA): RealAudio files use a proprietary format designed specifically for playing audio-on-demand in real-time over the internet, introduced by Progressive Networks in 1995, consisting of a server application, and encoder, and a player which works within a Web browser. Normally, the RealAudio player delivers 16-bit sound, although an 8-bit option is available. Data rates range from 14.4kBps (approximately the sound quality of a mono AM station) to dual ISDN Stereo at 16kHz, nearly CD-quality. See *RTSP*.

real-time: Occurring at the same time as other, usually human, activities. In real-time sequence recording, timing information is encoded along with the note data by analyzing the timing of the input. In real-time editing, changes in parameter settings can be heard immediately, without the need to play a new note or wait for computational processes to be completed. (1) The ability of a computer or other device to carry out a process without noticeable delay, such as real-time editing on a sequencer where changes are made to the music as it plays. The opposite of *off-line*(1). (2) In a sequencer, the ability to record MIDI messages as they are played on a keyboard or other controller, i.e., to behave like a tape recorder. This is generally the method preferred by musicians. The opposite of *step-time*. (3) Events which have to occur at particular times to ensure synchronization between devices such as a timecode message, as opposed to those (non-real-time) events which can occur at any time. (4) See *System-Exclusive*.

Real-Time Analysis: See *RTA*.

R

real-time control: A non-preprogrammed control signal generated by the player via a *controller* such as a *pitch-bend* wheel, *mod wheel*, *aftertouch* (pressure) sensor, footpedals, etc. It is common to have one of these real-time controllers affecting the depth (amount) of modulation signal being sent from some other (nonreal-time) source to the modulation destination. See *continuous controller*.

real-time dubbing: Duplicating a tape at its normal playing speed rather than at a higher speed, resulting in better quality than high-speed dubbing.

real-time input: MIDI input generated in *real-time*, such as during a performance or studio recording session. See *real-time control*.

Real-Time MTC Cueing: MIDI messages similar to Set-Up which contain information such as *cue points*, *punch-in/-out points*, event start and stop points, and event names. Unlike conventional *MTC* Set-Up messages which include details of the absolute times in the future at which events should occur, Real Time MTC Cueing messages are of the Universal System-Exclusive Real-Time type and so are to be acted on when received.

reassign: An output bus designed for internal re-routing and combining within a mixing console. See *insert point*.

recapitulation: In a musical structure, the final return to the theme from the main opening section, usually modified to occur in the home *key*.

reception mode: See *MIDI mode*.

reclock: To align bars and beats in a digital editor to music recorded without a tempo reference.

reconstruction filter: In a digital audio system, in order to recover the analog signal from the digital words, a *D/A converter* is used. The output of the converter is a stair-step waveform which contains a great deal of high-frequency artifacts called *images*. To reconstruct a smooth replica of the original signal, the stair-step is passed through a steep *lowpass* filter, also called an *anti-imaging filter*. It is similar, or even identical, to the *anti-aliasing filter* at the input of the *A/D converter*, but its purpose is very different. Also called an *anti-imaging filter*. See *quantization*, decimation, *FIR*, *IIR*.

record-equalization: Also the same for *playback-equalization*. In tape recording, the internal and *complementary* alteration of the *frequency response* of input signals prior to recording and output signals after playback. By boosting highs prior to recording and reducing them after playback, some tape noise is eliminated. In addition, the equalization curve can compensate for nonlinear response of the specific type of recording tape in use. There are a number of standard record / playback curves. See also *pre-emphasis*, *RIAA*.

record head: The *head* on a tape recorder that applies a varying magnetic force to the tape so that the audio signal will be recorded on the tape for later playback. A very high-frequency signal is mixed with the audio program before it reaches the record head. This *bias* signal helps to linearize the over-all *frequency response* of the tape itself, reducing distortion.

R

record-in/record-out: In *SMPTE timecode* synchronization for videotape post-production, the user-specified SMPTE timecode addresses at which the synchronizer will automatically place the audio or video recorder in record mode (*punch-in*) and subsequently cancel the record mode (*punch-out*). See *pre-roll*, *post-roll*, *mark-in/mark-out*.

recordist: (1) The person who operates the recording device during a recording session, or in film, the person who is in charge of aligning and loading the recorders and playback *dubbers*. (2) The person who records sound during film shooting (Europe). In the U.S., this person is called the *production mixer*.

rectifier: A device for converting *AC* to *DC*. Rectification is done through a network of *diodes* in a power supply to convert power line voltage to DC to power *active* devices. Rectification is also used to recover the signal in an *amplitude modulated* wave form.

Red Book: Published by Philips and Sony to set out the complete standard for audio *CD*s so that all CD players will be compatible: uncompressed, 16-bit, 44kHz audio data. The *Yellow Book* publishes standards loose enough to allow computer manufacturers to make CD-ROM players, thus all CD-ROMs are not compatible with all players.

reduction: (1) The mixing of a number of tracks of a multitrack recording to produce a *mono* or *stereo* master, also called *mixdown*. (2) In music, an arrangement of a *full score* for performance by a smaller group of instruments or, more typically, just for piano.

redundancy: The digital transmission of more bits than strictly necessary in order to improve the reliability of the transmission, such as ECC encoding. See *error correction*.

reel motor: In a tape recorder, the motor that controls the motion of either the *feed* and / or *take-up reels*.

reel size control: On a tape transport, the control that maintains proper tape tension by accommodating for various sized feed and / or take-up reels which have different amounts of angular momentum. Newer machines use an infrared or other beam to automatically measure and set the tensions for the various reel sizes.

reference frequency: See *line-up tone.*

reference level: (1) The reference level on an audio device is a signal level near the maximum possible for the device but low enough to ensure low *distortion*. (2) The reference level of a *power ratio* is the unit of power (i.e., watts or volts) being compared. See *decibel.*

reference source: The *clock* signal used to determine the rate at which a *timecode* generator and synchronizer will run; the *master clock* generator. The reference source can be thought of as the system master clock. The reference source can be an internal *crystal sync*, *video sync*, AC power *pulse*, external *pilot tone*, or the timecode reader.

reference tape: A laboratory-recorded test tape which contains a series of *line-up tones* all recorded at a known, standard level or *fluxivity*. The tape is used to verify the performance of the recorder's playback system. Once this is done, the record system is adjusted to produce a signal that, when played back on the playback system, are identical to the input signal. Also called a *test tape*.

reference tone: See *line-up tone*.

R

reflections: Sounds that do not take a direct route to a person's ears, but which bounce first off of a stage, balcony, wall, or other non-absorptive boundary before arriving at the ears. The combined audio effect of all of the reflections of a sound is called *reverberation*. Early reflections are the first reflections to reach the ears and sometimes sound distinct, like little echoes. Closer instruments will generally have a longer delay between the initial *dry* signal and the first early reflections. See also *ESS*.

refraction: The splitting-up of a complex sound wave into separate frequency *bands* as the original wave passes from one elastic medium into another, e.g., from air into water. Similarly, a prism disperses white light into the familiar rainbow of colors, bending the shortest wavelengths (violet) the least, and longest (red) most. See *diffraction*.

regenerating timecode: In copying a video or audio tape with *SMPTE timecode*, the process of reading the code from the master tape and creating a perfect electronic duplicate of it for recording on the copy. The new code is created by a separate device and is necessary to ensure that the audio or video copy is free of timing errors and *dropouts*.

regeneration: (1) See *feedback* and *resonance*. (2) See *timecode regeneration*.

register: A specific part of the *pitch* range of an instrument, voice, or melody, e.g., a cello in its *tenor* register. See also *tessitura*.

Registered Parameter Number (RPN): An extension to MIDI which allows for additional Control Change messages. RPNs have a Control Change number of 100 (LSB) and 101 (MSB). These are defined in the following way:

Registered Parameter Numbers

MSB (CC 101)	LSB (CC 100)	Parameter
00	00	Pitch-bend Sensitivity
00	01	Fine Tuning
00	02	Coarse Tuning
00	03	Tuning Program Select
00	04	Tuning Bank Select

Unregistered Parameter Numbers (NRPNs) are vendor-defined and may vary among systems. NRPNs have a Control Change number of 98 (LSB) and 99 (MSB).

registration: The choice of *stops*, i.e., *timbres*, in organ music.

regroup: The transfer procedure in which material is copied from one medium to another, e.g., from multiple units of *mag film* to a *multitrack*, to facilitate re-recording. For example, a facility might have only five playback dubbers on a *re-recording stage*, and they might transfer twenty units of mag film to 24-track tape in four passes. See also *transfer*.

rehearse mode: In *SMPTE timecode* synchronization for videotape post-production, a synchronizer mode that simulates the engineer-specified operations, including *record-in* and *record-out* commands) at the designated code addresses, but does not actually *punch-in* and *-out*.

reinforcement: An increase in acoustic or electric amplitude that occurs when two or more waves are, to some extent, *phase coherent*. The opposite of *phase cancellation*.

R

relay: An *electromechanical* device, essentially a solenoid-operated switch. Largely superseded by the transistor except in certain high-current applications. One possible such use is in amplifiers where a relay may break the connection to the loudspeakers before very high amplitude sounds can damage the loudspeakers.

re-laying: To *post-stripe* a video master tape, recording the mixed *film soundtrack* back onto the tape after it has been striped for post-dubbing or the addition of narration, music, and / or effects. See *layback recorder*.

relay station: In broadcasting, a remote site with equipment to receive a signal, either through a telephone line (sometimes called a *music line*) or a microwave link, and retransmit it for improved local reception.

release: (1) The portion of an *envelope* that comes after the *sustain* portion and is the amount of time it takes for the sound to go from the sustain level back to silence. The release segment begins after the key is lifted, i.e., the release time to zero-level after a MIDI Note-Off message arrives or the key is released. See *ADSR*. (2) The time it takes for a *compressor's* gain to come back up to normal once the input signal has fallen back below the *threshold*.

release loop: A set of *loop* points that define a portion of the sound to be played repeatedly during an *envelope's* release phase. A release loop starts playing back after a key is released when sample playback will finish the current pass through the *sustain loop* and then move on to the remainder of the sound, which may or may not contain a release loop. The release loop will be heard for the length of time determined by the *release time* parameter setting.

release print: A composite print of a film made for general exhibition purposes. Release prints are generally made using the same exposure, color balance, and effects employed in the final *answer print*. See *internegative, EK neg*.

release time: The length of time it takes for a signal processing device, generally a *compressor*, *limiter*, or *expander*, to return to its nominal gain-before-*threshold*, once the input signal level no longer meets threshold conditions. See *attack time*.

release velocity: The speed with which a *controller* key is raised or otherwise released, and the type of MIDI data used to encode that speed. Release velocity sensing is rare but found on some instruments. It is usually used as a *rate control* for the *release* segment of the sound *envelope*.

reluctance: The opposition to a magnetic force or field, exhibited by another field and its source, such as an *electromagnet*, or by unmagnetized but potentially magnetic objects such as the oxide domains on *magnetic recording tape*.

remanence: Also called *remanent flux*. The amount of magnetization left on *magnetic recording tape* when an applied magnetic force is removed. Measured in lines of force per quarter-inch of tape width. See *flux density*, *gauss*, *hysteresis*, *residual magnetization*.

R

remote: (1) A recording session which occurs at a location other than a recording studio, such as a concert hall or church, i.e., *on location*. (2) As in a remote control, used to activate a device without using the device's control panel. (3) A device controlled from a distance either through a remote control transmitter or keypad, or by using the controls of another device. For example, playing a remote synthesizer through on a *MIDI network*, or operating certain sync tape machines as remotes by transmitter control.

remote keyboard: A keyboard which can be slung around the neck, like a guitar. It will transmit the performance through a MIDI cable or via a radio / infra-red transmitter matched to an appropriate receiver and MIDI module. Essentially, a type of *MIDI controller*.

repertoire: (1) Music which a soloist or group already knows, i.e., is performance-ready. (2) The catalog of songs or performers that a publishing or record company has signed to it.

replay head: In a tape recorder, an *electromagnetic* device for converting the magnetic patterns previously recorded on tape into a voltage whose amplitude is proportional to the pattern. On some systems, the replay head is also used as a record head, i.e., the replay head is driven rather than listened to, and it may also be used as a *sync head*. The replay head is sometimes called a *repro head*.

repro: One of the operating modes of tape recorder electronics. Tracks placed in repro mode will play the tape via the normal playback head, i.e., not via *sel-sync*. Alternative to *sync mode*.

reproducing characteristic: In audio tape playback, the standardized *equalization curve* introduced by de-emphasis. See *pre-emphasis*.

repro head: See *replay head*.

re-recording: The process of mixing all edited *DME stems,* sync and non-sync, of a film or video production to *mono*, *stereo*, *multichannel* or whatever audio format is desired for the final *print master* with the picture, also known as *dubbing*. Usually done at a *re-recording stage*.

re-recording stage: The facility where *re-recording* is done.   Contrary to what one might think, there is rarely a stage / platform / dais involved. See *dubbing theater*.

resampling: (1) The process of sampling a previously mixed sample to create a new sample. (2) The changing of a signal encoded at one sampling rate to a different sampling rate via a *re-sampling converter*.

Reset All Controllers: A Channel Voice message which instructs a MIDI device to set all controllers to their inactive condition, effectively doing for controllers what an All Notes Off message does for notes. The message is actually a type of Controller Change message.

residual magnetization: Similar to *remanence*, but generalized to designate the magnetism remaining in any magnetic material once the applied magnetic field is removed.

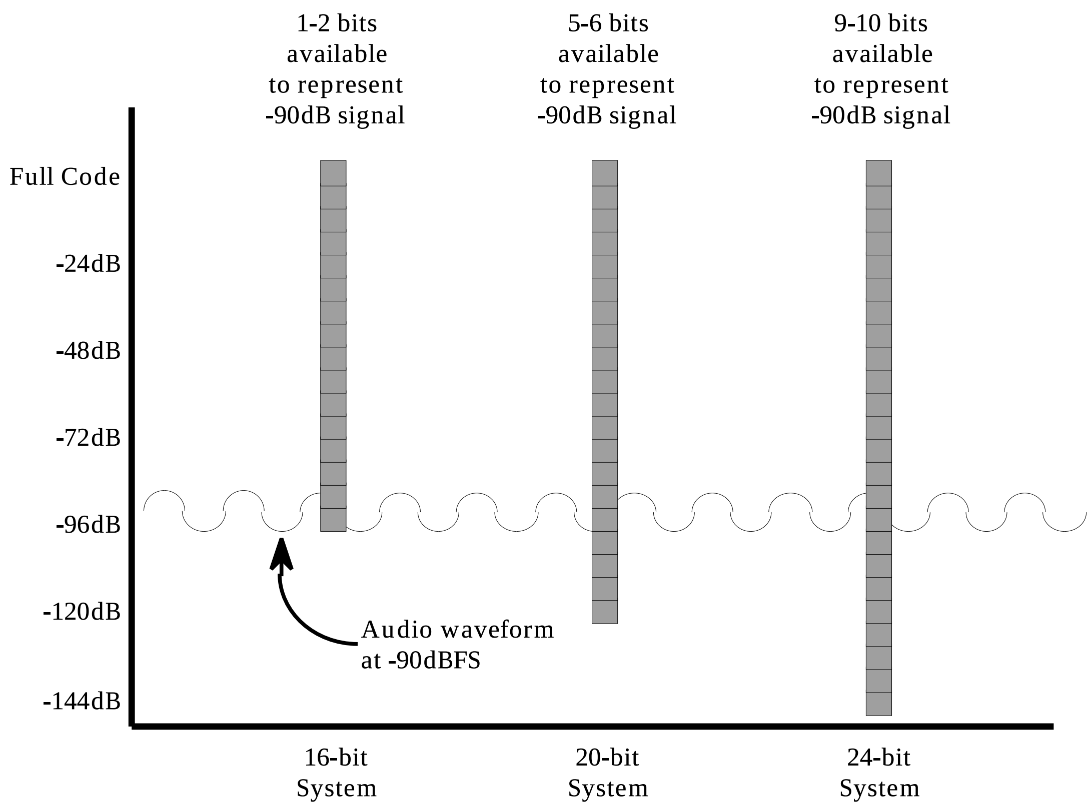
residual noise: The noise that remains on *magnetic recording tape* after full erasure.

resistance: In electrical or electronic *circuits,* a characteristic of a material that opposes the flow of electrons, measured in ohms (Ω). Calculated by the formula **R=E/I**, where *E* is the voltage, and *I* is the *current* flow in *amperes*, through the circuit. Resistance results in loss of energy in a circuit dissipated as heat. *Conductance* is the reciprocal of resistance. See *impedance*.

R

resistive network: A *circuit* composed of *resistors*, commonly used in the construction of *attenuators*.

resolution: (1) The fineness of the divisions into which a sensing or encoding system is divided. The higher the resolution, the more accurate the digital representation of the original signal will be. See *bit depth*. In MIDI sequencing, the recording resolution, expressed in *ppq*, measures the timing accuracy with which a device or program can record or reproduce MIDI data. See *sampling rate*. (2) In video, the number of *pixels* which comprise a horizontal *line* of image.



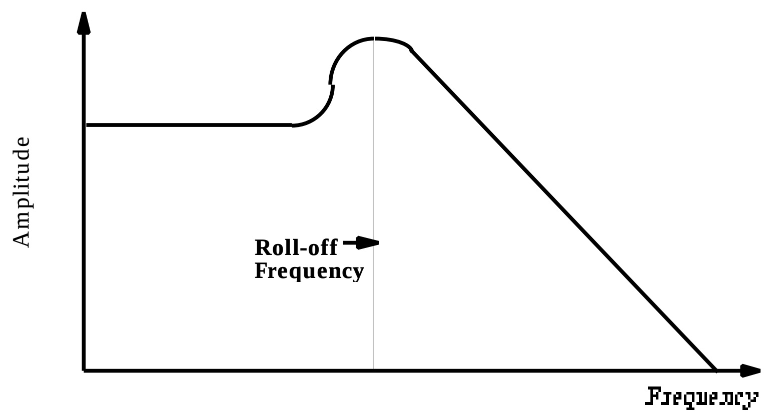
Low-Level Signal Resolution as a Function of System Bit-Depth

resolver: (1) (*noun*) The electronic circuit within or connected to a *Nagra* or other synchronous tape recorder that acts as a *servo* to control motor speed on playback to maintain proper synchronization of sound with picture. It keeps the recorded *sync tone* in phase with the 60Hz AC powering the recorder and projector, or can sync the playback with *crystal sync*, or other *reference source*. (2) (*verb*) The process of regulating audio and/or video tape speed by comparing a *reference tone* on the tape with an external *master* and adjusting the speed of the first device so that the two stay in sync is called *resolving*. This can be done striping the tapes with a *pilot tone*, or by a *timecode* such as *SMPTE*. The master deck reads the location from all slave machines and fast-forwards or reverses them until it reads the same timecode from every machine. It then monitors the timecode adjusts the slaves' transports continually to maintain *frame lock*. See also *resolver*, *reference source*.

R

resolving: The process of *synchronizing* the internal clock(s) on one or more devices to an external *master clock*.

resonance: The sympathetic or induced vibration of a system (a solid, a membrane, or an air space) in response to the presence of vibration in the air; the tendency of a mechanical or electrical system to vibrate at a certain frequency when excited by an external force and to keep vibrating after the excitation is removed, e.g., a bell. When a vibrating object (such as a guitar body) is stimulated by a second *oscillator* (such as a vibrating string), its pattern of vibration may be altered. If the two vibrate at the same or a *harmonically* related frequency, they tend to *phase-lock* together, reinforcing the sympathetic vibration at this common *resonant frequency*. Oscillations at non-harmonic frequencies have far less effect due to *phase cancellation*. (1) A function of a filter in which a narrow band of frequencies, the *resonant peak*, becomes relatively more prominent. If the resonant peak is high enough, the filter will begin to oscillate, producing an audio output even in the absence of input. Filter resonance is also known as *emphasis* and *Q*. It is also referred to in some older instruments as *regeneration* or *feedback*, because feedback was used in the circuit to produce a resonant peak. (2) The tendency of a speaker to vibrate most at a particular frequency; sometimes referred to as the *natural frequency*.



Low-Pass Filter with Resonance

resonant filter: Nearly all musical instruments are a type of resonant filter, i.e., they pass certain particular frequencies and attenuate others, producing the tuning of the instrument. Each instrument, from a simple electrical resonator to a complex resonant system such as a guitar or organ pipe take an initial *impulse* and impress upon it the particular *transient* profile which gives the resonator its unique *timbre* and *pitch*.

resonant frequency: See *resonance*, *loudspeakers*.

resonator: An acoustic device that has a *resonance*. Virtually all musical instruments have some sort of resonator as part of their tone-producing mechanisms.

response pattern: See *polar pattern*.

restored timecode: In *SMPTE timecode* synchronization, a newly generated, continuous timecode that will maintain sync with an external reference code. Used to replace a discontinuous timecode, such as one that includes unrelated segments of code, perhaps copied from unrelated scenes and *takes*. See *jam sync*.

R

resultant tone: See *difference tone*.

resynthesis: A form of synthesis / sampling which is based on the analysis of sound data which is later used to reconstruct the sound, usually hundreds of sine waves which collectively build up the harmonic content of the sound, sometimes with the imposition of additional parameters and / or constraints, but all in *real-time*. This means that only small amounts of data need to be stored, unlike traditional sampling, and that any parameter of the sound can be infinitely changed. This new technology should allow the faithful recreation of any existing instrument and the creation of truly original sounds and textures. Material construction, acoustic response, perspective, morphing between sounds and transformations, such as blowing a piano, in any dimension will be possible.

retentivity: The *flux density* present on a specific type of magnetic recording tape after a magnetic field of *saturation* strength is removed. The maximum flux density the tape can store. See also *remanence*.

return: See *send*.

reverb: A type of signal processing *effect* which produces a continuous wash of echoing sound, simulating an acoustic space such as a concert hall. See *darkness*, *reflection*, *echo*, *DSP*.

reverb plate: See *plate reverb*.

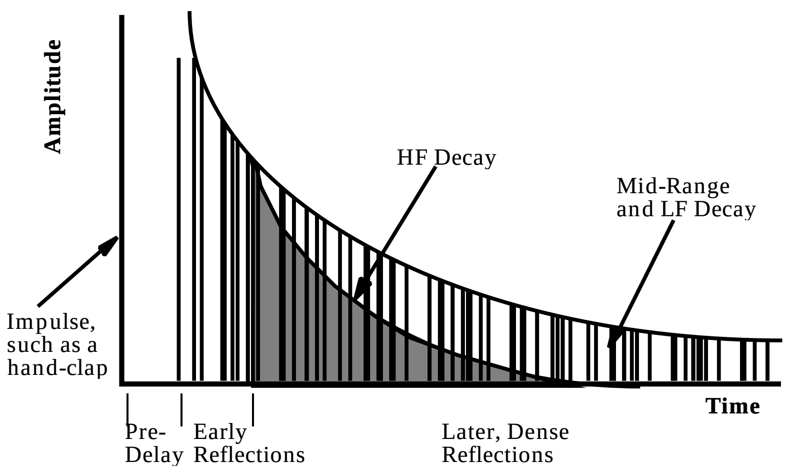
reverb spring: A spring which is used to produce *reverberations*. One *transducer* causes it to vibrate and the reflected wave motions are picked up by other transducers.

reverberant field: In a room with *reverberation*, if a listener is close to source of sound, the direct sound will predominate, and the listener is said to be in the *direct field* of the source. At greater distances, the reverberant energy will predominate, and this region is called the reverberant field. See also *free-field*, *near-field*, *far-field*, *decay*(3).

R

reverberation: The decaying residual signal that remains after a sound occurs, created by multiple reflections as the original sound wave bounces off walls, furniture, and other nonabsorbtive barriers within a room or other acoustical environment. Reverberation contains the same frequency components as the sound being processed, but no discrete *echoes*. An average club has a natural *reverberation time* of about a half-second; many concert halls and auditoriums have a natural reverberation time of two seconds or more. A room with very little reverberation is called a *dead* room, which is the opposite of a *live* acoustic space which is very reflective. Reverberation is composed of *early reflections* and later reflections.

High-frequency sound waves have to cause the surrounding air molecules to vibrate quickly enough to pass the sound energy onwards, consequently high-frequency reflections die out faster than mid-frequency or bass reflections. Also, high-frequency sound is more readily absorbed by soft furnishings. Low-frequency sounds are only reflected by large and heavy objects, so there may be very little low-frequency reverberant sound. However, in larger rooms, there can be substantial *bass build-up*.



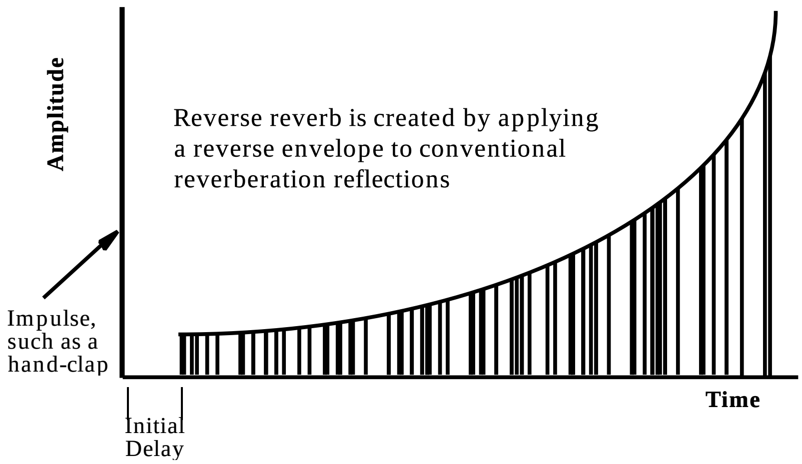
Reverberation

reverberation time: The time of reverberation is defined as the time it takes for the *SPL* to *decay* to one-millionth of its former value, a 60dB reduction, hence called the *RT-60* of the space. Also called *decay time*.

reverberator: A device for the generation of synthetic *reverberation*, either analog, such as a *plate* or *spring reverb*, or a digital *effects processor* which simulates the reverberation according to various *parameters* such as room size (small, medium, large) and room type (club, cathedral, studio).

R

reverse reverberation: A digitally simulated effect whereby a sound envelope is created by the usual *attack, release*, and *sustain* stages, but the *decay* portion of the envelope is purposely reversed so that the reverberant sound increases in amplitude, rather than naturally decreasing.



Reverse Reverb from a Percussive Sound

rewind motor: See *supply motor.*

RFI: Radio Frequency Interference. Caused by radio stations, cellular phones, and other sources of radio energy that is transmitted through the air, RFI is a common source of induced noise. See *induction, EMI.*

RFZ: Reflection Free Zone. For example, hopefully the mix position in a studio. As with the *LEDE* school of control room design, both acoustic designs seek to create a room which imposes none of its own character upon the audio. As opposed to early sound scattering (*ESS*) control room designs.

rhythm: The *beat, tempo, measure* and *meter* of the music, plus the variation of beats as they are superimposed over the main pulse.

rhythm track: Basic tracks usually recorded first in a multitrack session. These are generally played back for the musicians as a *cue mix* for dubbing purposes.

RIAA: Recording Industries Association of America. An industry body set up to define standards and practice in the recording industry. In particular, its internationally accepted standard for the recording characteristic involving *emphasis* and de-emphasis for vinyl records. See also *SMDI, RIAJ.*

R

RIAA Curve: An *equalization curve* established by the RIAA and applied to all music as it is transferred from *master tape* to master disc. This *pre-emphasis* curve introduces a 20dB skew in the frequency content of the program, decreasing the bass content and increasing treble over a 500Hz threshold. The complementary *de-emphasis* is applied by the phonograph preamplifier during reproduction. Since groove width increases in proportion to the low-frequency content of the program, and since vinyl disc surface noise is most noticeable in the high frequencies, the RIAA curve is designed to enable more dense program recording while minimizing the effect of surface noise.

RIAJ: Record Industry Association of Japan. The Japanese equivalent of the *RIAA*.

ribbon microphone: A type of *dynamic microphone* which has a thin metal foil, or ribbon, suspended in a magnetic field. Sound waves vibrate the ribbon in the field and generate an electrical signal. Ribbon mics are usually quite fragile, but are used for their warm, smooth tone quality. They work well with digital recording and on brass instruments to mellow the tone. Ribbon mics are either *figure-eight* or *cardioid*. See also *condenser microphone***.**

ribbon tweeter: A high-frequency *loudspeaker*. The audio signal is connected to the ribbon, which is a very thin (usually aluminum) strip suspended in a magnetic field. The *current* in the ribbon establishes another magnetic field, causing the ribbon to move in synchrony with the input signal waveform, hence, is a *direct radiator* of sound. Ribbon tweeters are effective at very high frequencies, and usually require a step-down transformer because of the very low *impedance* of the ribbon.

riff: A short, catchy musical phrase usually played between lyric lines in a song, often repetitively, thereby acting as a *hook*.

RIFF: Resource Interchange File Format. A file specification adopted jointly by Microsoft and IBM for multimedia sound applications.

rifle microphone: See *gun microphone*.

ringing: Any device, electronic or mechanical, is said to ring if it continues to produce a signal or to move after its input is stopped. Ringing, a type of *transient distortion*, is caused by too little *damping*, and is particularly prevalent in audio *transducers*. Low-frequency ringing is called *hangover*.

ring modulator: A special type of *timbre* modifier module that accepts two signals as audio inputs and produces their sum and difference tones at its output, but does not pass on the frequencies found in the original signals themselves. This greatly increases the number of *harmonics* contained in the two sounds and introduces a gross nonlinearity, causing huge amounts of *harmonic* and *intermodulation distortion*. The ring modulator is used in the generation of electronic music, usually in conjunction with a *lowpass filter* which reduces the high-frequency roughness of the resulting metallic sound. It gets its name from its circuit configuration which is a circle, or ring, of four *diodes*. See also *oscillator sync*.

ripping: The process of extracting samples from *MOD* files for use in digital compositions.

ripple: (1) Irregularities in the *frequency response* of a filter which has a nominally flat response in its *passband*, or (2) irregularity in the value of *DC* voltage in a power supply which manifests itself as *hum* in the loudspeakers.

R

RMS: Root Mean Square. A formula for describing the *level* of a signal. RMS is derived by squaring all of the instantaneous voltages along a waveform, averaging the squared values, and taking the square root of that number. For sine-like signals,

average power = RMS(voltage) × RMS(current)

The voltage of an audio signal is usually measured in terms of the RMS value of the signal. The *RMS value* of an alternating current produces the same heating effect in a circuit as the same value of a direct current.

Robinson-Dadson curves: See *equal loudness curves*.

rock and roll: A system used in *dubbing* or mixing by which the projector, dubbers, and recorder can run in synchronization in reverse. Thus, if a mistake is made in mixing a particular section, all sources can be rolled back past the mistake, and a new take can be *punched-in* before the mistake. Also called *rollback*. *Selsyn motors* on all machines involved are required to make rock and roll possible.

rollback: (1) See *rock and roll*. (2) On video recording / editing systems, the rollback, or *RLB*, function is used to rewind machines by a predetermined amount from the current position. The default rollback time is 15 seconds.

rolloff: (1) The difference between the input amplitude and the output amplitude in a filter over a specified frequency band, expressed in dB. (2) See *rolloff filter*, *rolloff slope*.

rolloff filter: A filter which has a reduced output as the frequency is increased is called a rolloff filter. A tone control is a rolloff filter when turned down. Sometimes the attenuated portion of the frequency content of the signal itself is called *rolloff*. A circuit that attenuates a signal that is above (*lowpass*) or below (*highpass*) at specified frequencies. For example, microphones usually have a bass rolloff filter to remove wind noise and / or excessive breath pops.

rolloff frequency: The frequency above or below which a filter begins to filter out the *harmonics* of the *waveform*. As the rolloff frequency is raised or lowered, more of the harmonics of the sound will be filtered out. Specifically, the frequency at which the response of an equalizer or other audio device is reduced by 3dB, and can refer to both lowpass and highpass response curves. The rolloff frequencies of an amplifier are the frequencies where the output voltage drops to 0.707 of the middle range output. A decrease of the voltage by a factor of 0.707 is equivalent to -3dB, so these critical frequencies are often referred to as the *3dB down points*. Also called *cutoff frequency*, *critical frequency*, or the *half-power point*. Moving the rolloff frequency in real-time will produce a *wow* effect, which can be accentuated by increasing the *filter resonance* level.

rolloff slope: The acuity of a filter's *rolloff frequency*. Rolloff is generally measured in dB / octave. A shallow slope, such as 6dB per octave, allows some frequency components beyond the rolloff frequency to be heard, but at a reduced volume. When the rolloff slope is steep (on the order of 24dB / octave), frequency components very close to the rolloff frequency are reduced in volume so much that they fall below the threshold of audibility. Rolloff slopes are sometimes called *skirts*. See *filter, pole, sharp*(2).

R

room equalization: The alteration of the *frequency response* of signals that will be sent to speakers or monitors, done in order to compensate for *room modes*. These problems are generally identified by sending *pink noise* to the speakers, then adjusting the frequency response of the resulting sound that arrives at the listening position. Most room EQ references use ⅓-octave 31-band graphic equalizer, with additional over-all bass and treble adjustments. See also *RTA*. Also called *voicing*.

room mode: Also called *standing waves*, room modes come in three types: axial, along the axes of a room (front to back, side to side, floor to ceiling); tangential, or any two pairs of opposite surfaces; and, oblique modes which are the product of the reflections of all six surfaces. To calculate the frequency of a particular mode,

$$f = \frac{c}{2}\sqrt{\left(\frac{n_x}{L_x}\right)^2 + \left(\frac{n_y}{L_y}\right)^2 + \left(\frac{n_z}{L_z}\right)^2}$$

Where *f* is in *Hertz*, *c* is the speed of sound (1130 ft./sec. or 344 meters/sec.), *L* is the room dimension, and *n* is the order of the mode. It's not important to know the frequencies of the room mode, just how evenly spaced they are. If they are not uniformly distributed, the room will show a response peak where they are nearly coincident. The most problematic frequency range is 50Hz-150Hz. The golden ratios of height to width to length are: 1.14:1.39:1 or 1.28:1.54:1 or 1.60:2.33:1 (Bolt's golden ratios), then the modes will be perfectly spaced and low-frequency response is smooth by design. See also *standing wave*.

room sound: The characteristic *ambient* sound of a concert hall or other listening space.

room tone: The *ambient noise* of a room, set or location where dialog is recorded for the production shoot. Usually recorded as wild sound, room tone is used by film and dialog editors as a *bed* to form a continuous tone through a particular scene. Not to be confused with *ambience*, which can be sound effects and/or reverberation added when the dialog is mixed. See also *NC Curve*, *walla*.

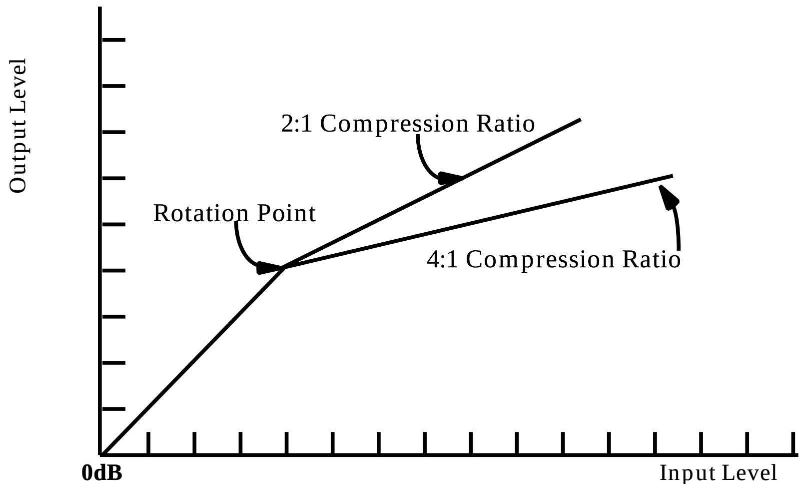
root: The lowest note of an uninverted *chord*, and therefore the note which usually gives the chord its name. For example, in a triad, it is the lowest note when the chord is arranged as two thirds on top of one another: C is the root of the chord of C (C, E, and G), an arrangement known as *root position*. If the root is not the lowest note, the triad is said to be *inverted*.

root mean square: See *RMS*.

rotary head: The video recording system that uses a rotating drum carrying two or more heads which sweep across the tape at a small angle, typically 5˚-8˚. This allows a high rate of scanning to be combined with a low tape speed.

R

rotation point: In a *compressor* or *expander*, the input signal level at which the graph of the device's *transfer characteristic* intersects its *unity gain* curve. At this level, there is no net change from input to output level.



Rotation Point

rough cut: A stage in the editing of the *workprint* or videotape at which the scenes and shots are in order and cut approximately to their correct length, between an assembly and a *fine cut*. See *copy editing*.

rough mix: A mix of a recording in progress, either as it occurs or, in the particular case of a stereo mixdown of a multitrack recording, one made at the end of a day's work for overnight audition.

routing: (1) (*verb*) The process of directing a signal from one point to another. (2) (*noun*) The pathway a signal takes, e.g., through which busses it passes on a mixing desk.

Rt: See *stereo optical print*.

RT-60: The *reverberation time* of a space. Suggested times for various room volumes are:

Room Vol. (10^3 cu.ft.)	Range of RT-60 Times (seconds)
1-2	0.3-0.4
2-3	0.3-0.55
3-5	0.4-0.65
5-7	0.45-0.7
7-10	0.5-0.75
10-15	0.55-0.8
15-20	0.65-0.9

R

RT-60 meter: A hardware device for measuring the acoustical characteristics of a listening / recording space. Gated *pink noise* is played and analyzed by the meter. The meter then provides a read-out of the locations reverb profiles for the different frequencies, usually 125Hz, 250Hz, 500Hz, 1kHz, 2kHz, and 4kHz.

RTA: Real-Time Analyzer. A piece of hardware that measures the *loudness* of an audio signal, either through a flat-response microphone or a line input, and categorizes the loudness into different frequency bands, used to flatten out the *frequency response* of a monitoring system. An RTA usually displays frequencies at one *third-octave* or octave *bands*. The decibel level of each frequency band is displayed graphically.

RTSP: Reat-Time Streaming Protocol. An internet protocol for real-time transmission of high-fidelity audio, suitable for live concert feeds as well as professional audio applications. See *RealAudio*.

rubato: A musical term indicating that a conductor or musician may take expressive liberties with the *tempo* during the designated section of a score.

rumble: A low-frequency mechanical vibration in a turntable, microphone, or tape transport. Rumble is specified as a *S/N ratio* in decibels, with -50dB being common.

rumble track: Stereo *white noise* lowpass filtered at about 40Hz so as to be felt and not heard.

run: A command used by DOS to launch an application program. The RUN command may be used with various switches (subcommands) and is uniquely powerful and straightforward. This is no doubt owing to the sacred–cow status of the command conferred by its origin in the numerous progenitors of DOS, decades before anyone from Microsoft could confuse it.

running master: The same as a *print master*.

running status: A *data compression* scheme for MIDI data whereby status bytes can be skipped if they would be the same as the most recent status byte. For example, a four-note chord can be transmitted using only nine bytes rather than the usual twelve. See also *MIDI delay*(2).

rushes: See *dailies*.

rustle: See *Foley*.

S

S: Side. The difference component of a stereo signal, i.e., the components which come from the side of the stereo field. See *M & S*.

S & S: Sampling and Synthesis. A term used to describe synthesizers which combine elements of these processes to produce their sound.

Sabine equation: The *reverberation time* of a room is found using the Sabine equation:

$$\mathbf{RT60 = 0.049 \times (V/Sa)}$$

where *V* is the volume of the room in cubic feet, *Sa* is the total number of sabins present, and *RT-60* is the reverberation time.

sabin: The efficiency with which materials in a room absorb sound and damp *reverberation* is measured in sabins, abbreviated *Sa*. The number of sabins of absorption is found by multiplying the number of square feet of a particular material by the *absorption coefficient* of that material. Also spelled *sabine*.

SACD: Super Audio Compact Disc. Philips/Sony's proposal for a next-generation CD which combines *DVD* technology to produce a hybrid disc that will play in conventional CD players, but offering better audio quality than currently available CDs when played in a DVD player. The upper later is the "conventional (Red Book) layer, while the lower layer provides around 4.7 Gb of high-density storage, increasing the audio capacity to 4.7Gb, the same as a first-generation DVD, and allow text, graphics, and video alongside audio. Audio will be encoded via either standard 16-bit PCM at 44.1kHz, or "Super Audio" using Sony's *DSD* data format and Philips' *DST* data compression technologies, yielding 74 minutes of six-channel audio with purportedly a frequency response of DC–1MHz and the same dynamic range as conventional digital recordings of 24–bit/96kHz resolution. SACD employs*SBM* noise-shaping, and *PSP* copy protection. The intention of the SACD standard is to ultimately combine a stereo DSD track and a six-channel DSD surround mix, plus optional data, text, graphics, and video data.

Here is a comparison between conventional (Red Book) CD and (the current (3/99) prototype specification for) SACD:

	Conventional Compact Disc	**Super Audio Compact Disc**
Diameter	4-3/4" (120mm)	4-3/4" (120mm)
Thickness	1/20" (1.2mm)	1/20" (1.2 mm)
Signal Sides	One	One
Signal Layers	One	Two
Data Capacity		
Reflective Layer	780 Mb	780 Mb
Semi-Transmissive Layer		4.7 Gb
Audio Coding"		
Standard Audio	16-bit PCM, 44.1 kHz	16-bit PCM, 44.1 kHz
Super Audio		1-bit DSD, 2.8224 MHz
Multichannel	2-channel	6 channels of DSD
Frequency Response	5-20,000 Hz	DC-1MHz
Dynamic Range	< 96 dB	< 120 dB
Playback Time	74 minutes	74 minutes
Enhancements	CD Text	Text, Graphics, Video

S

SACEM: Société des Auteurs, Compositeurs et Éditeurs de Musique. The French equivalent of *BMI* / *ASCAP*.

safe mode: One of the operating modes of tape recorder electronics. Any tracks placed in safe mode are prevented from entering record mode, even if the engineer accidentally depresses the record-ready button or master-record button. Opposite of *ready mode*.

safety: Short for *safety copy* or *safety master*, a duplicate of any audio or video tape made in case the master itself is lost or damaged. Also called a *protection copy*.

sample: (1) A digitally recorded representation of a sound. Also, a single *word* of the data that makes up such a recording. Also called a *patch* or a *program*. (2) To make a digital recording by taking regular measurements of the instantaneous voltage of an analog signal. See *sampling*. (3) Any single measurement of such a voltage.

sample-and-hold: The part of the *A/D converter* which actually does the job of sampling the signal. It measures the instantaneous signal voltage at a particular time and holds this level constant for the duration of the sampling interval as determined by the *sampling rate*. This level is meanwhile converted into a digital *word* before the sample-and-hold moves to the next sample. See *aperture time errors*.

sample (playback) synthesis: The production of a sound where a digital *oscillator* plays back a digitally sampled recording of an actual sound, such as a note played on a trumpet or guitar.

sampler: Essentially a digital recorder. A device that digitally records and plays back external sound sources, usually by allowing them to be distributed across a keyboard or other controller and played back at various pitches. Compare with *synthesizer*.

sampling: The process of encoding an analog signal in digital form by reading (sampling) its level at precisely spaced intervals of time. See *sample*, *sampling rate*.

sampling error: See *jitter* and *aliasing*.

sampling rate: The rate at which *PAM* samples of an analog signal are encoded by a digital device. The higher the sampling rate during the encoding process, the greater the spectral bandwidth of signal it is able to record accurately. Typical sampling rates vary from 11kHz to 96kHz. The sampling rate for CDs is 44.1kHz. See *sampling*, *Nyquist frequency*. Sampling rates can be changed via a *sampling rate converter,* when the process is known as *resampling*.

SAOL: Structured Audio Orchestra Language. See *MPEG-4*.

SAP: Second Audio Program. A sub-channel used in multichannel television sound, often for second language programming.

SATB: Soprano, Alto, Tenor and Bass.

S

saturation: In analog magnetic tape recording, saturation is the maximum magnetization that a tape can attain. Actual recorded levels are less than saturation because *saturation distortion* is introduced if saturation is approached, especially at low frequencies. At high frequencies, it is not possible to reach tape saturation because the signal itself acts to partially erase itself as it is being recorded because the high-frequency signal causes the record head to act like a tape degausser. This is called *self-erasure*, and limits the maximum level attainable in a tape recorder at high frequencies. Distortion occurs in an analog tape recording caused by input levels set in excess of 0VU. Setting a record input level too high on an analog medium is more forgiving than on a digital medium, and levels up to +3VU can sometimes be tolerated, but with a corresponding loss of high frequencies. See *overs*, *headroom*, *retentivity*.

saturation distortion: The distortion that results on magnetic recording tape when the applied audio signal is greater than its *retentivity*.

saturation point: The input signal level to a tape recorder that will cause the record head to produce saturation on the magnetic tape.

sawtooth wave: A geometrical waveform, typically generated by an oscillator, resembling a series of ramps. Sawtooth waves sound the same whether they rise left to right or fall left to right or a are series of alternating patterns (where the wave is sometimes called a *ramp wave*.) The sawtooth waveform contains all possible *harmonics*, both odd and even, giving a powerful and brassy timbre. The magnitude of each harmonic is the reciprocal of its number in the series, e.g., the fifth harmonic is $\frac{1}{5}$ the amplitude of the first. See Appendix C.

SBM: Super Bit-Mapping. See *dither*.

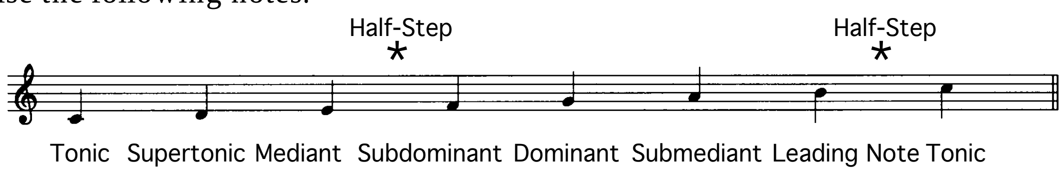
S

scale: Music is made up of sounds *pitched* at relative *intervals*. The spacing of these intervals makes a scale. There are several principal *modes* each of which can be found by starting on the notes A-G and playing up the white notes (only) on a piano to the corresponding note an *octave* higher. They are Aeolian (A-A), Locrian (B-B, rare), Ionian (C-C), Dorian (D-D), Phrygian (E-E), Lydian (F-F), and Mixolydian (G-G). These can be transposed to start on different pitches as the interval pattern between notes is the essential feature.

The modern major and minor scales correspond to the Ionian and Aeolian modes, respectively, and form the basis of most western music: (F=full-step, H=half-step, T=three half-steps)

Scale Name	**View**	**Intervals**	**Example**
Ionian (Major)		FFHFFFH	CDEFGABC
Aeolian (Minor)*	Tones are a minor third above tonic	FHFFHFF	CDE♭FGA♭B♭C
Dorian	Tones are major scale, one step below the tonic	FHFFFHF	CDE♭FGAB♭C
Phrygian	Pure minor with a lowered second degree	HFFFHFF	CD♭E♭FGA♭B♭C
Lydian	Minor scale with a raised fourth tone	FFFHFFH	CDEF♯GABC
Mixolydian	Tones are a major scale fifth below tonic	FFHFFHF	CDEFGAB♭C
Locrian	Tones are a major scale, half-step above tonic	HFFHFFF	CD♭E♭FG♭A♭B♭C
Harmonic Minor		FHFFHTH	CDE♭FGA♭BC
Whole tone		FFFFFF	CDEF♯G♯A♯C
Chromatic	Uses all 12 half-steps of the diatonic scale	12*H	CD♭DE♭EFG♭GA♭AB♭BC
Diminished		FHFHFHFH	CDE♭FG♭A♭ABC
Pentatonic	Five notes, omitting two of the seven	FFTF	CDEGA

*The Aeolian minor scale is also called the natural minor scale.

Within a scale, there is an ascending or descending series of notes that subdivide an octave into various and usually unequal pitch steps. The scales were collectively known as modes, and before about 1600 all were in common use. Between about 1600-1900 western music was centered on just two of the above scale patterns, the major and minor scales, which form the basis of *diatonic* harmony. Major and minor scales can begin on any note: those starting on C comprise the following notes:

All major scales preserve this same pattern of steps. Minor scales have their Mediant a half-step lower and may also lower the Submediant and Leading Note. See *Circle of Fifths*. Many non-Western cultures employ scales with different patterns of tone and semitone, and different tunings. Closer-pitched divisions called *microtonal* intervals are used in Middle Eastern and Indian music. Other scale types include the Pentatonic (5-note), Whole Tone (6-note) and other variants of the major and minor scales.

S

scale construction: See *tonic* and *whole-step*. For example, the relative minor of a major scale starts at either a sixth up or a third down on the major *scale*. To find the third, fourth, fifth, etc. of a tonic, count up that number of scale steps. For example, to find the sixth of a tonic, the major sixth would be nine *half-steps* above the tonic; the minor is diminished, i.e., it is eight half-steps above the tonic. The fifth is seven half-steps above the tonic or below it. A fourth is five half-steps above or below the tonic. The seventh is ten half-steps (minor) or eleven half-steps (major) above or below the tonic.

scale distortion: Because the human ear has a sensitivity which varies with *frequency* and with *loudness* level, a musical ensemble must be reproduced at the same loudness as the listener would experience at the actual event if *frequency distortion* is not to occur. This happens because of the apparent amplitudes of the different frequencies will differ, with accentuation of the extreme high and low frequencies. Also called *volume distortion*. See *equal loudness curves*.

scaling: In a synthesizer or sampler, a method of relating a *parameter* to a control so that the degree to which the control effects the parameter can be varied. For example, a *pitch-bend* wheel might be scaled to produce a *half-step* bend for a given amount of movement, or it might be scaled to give a whole *octave*. Nonlinear scaling between control and parameter will produce an output corresponding to some sort of curve. If there is a point on the curve where the output changes radically in response to input, this is often called a *breakpoint*. The scaling of keyboard parameters is called *keyboard tracking.*

scan: The way melody and lyrics are phrased together. A good scan means words and music fit well together and are easily sung and understood. Bad scan may occur when words such as “a” or “the” are sung on high notes or emphasized notes of the melody, sounding awkward or misphrased.

Scheiber matrix encoding/decoding: The algorithm used in *Dolby Stereo* optical process to produce quadraphonic *LCRS* sound from two channels. In this process, common in-phase information would bleed into the center channel, while the surround channel would receive out-of-phase material. See *SVA*.

Schroeder diffusers: Used to construct *ESS*-type acoustic environments, a structure comprising a number of wells of different, carefully-chosen depths. As a ray of sound strikes the irregular surface, instead of bounding off it like a mirror, it bounces out of each well at a slightly different time, resulting in many small reflections, spread out in both time and space. The operating range of a single diffuser is limited to about four octaves because, if the deepest well is deeper than about fifteen times its width, it begins to behave as a diaphragmatic absorber. The well depths are most commonly given by:

$$\mathbf{d_h = {}^{L}\!/\!_{2N} \times \left[h^2\right] mod N}$$

where *d* is the depth of the diffuser, *h* is the well number, *N* is the prime number on which the sequence is based, and *L* is the wavelength of the lowest operating frequency. This is called the *quadratic residue sequence.*

S

SCMS: Serial Copy Management System. A *DAT* format *subcode* which prevents direct digital copies by inserting a copy-protect message when a digital-to-digital copy of a recording is made. Once the flag is in the subcode of a tape, no subsequent copies can be made from that tape. SCMS is designed into most home digital recorders, and is a problem when transferring material from a consumer-type deck to a digital workstation or when trying to make safety copies of recording sessions. SCMS does not, however, prevent copies made using analog inputs. Some pro-level DAT decks include an SCMS defeat switch or use *AES/EBU* digital interfaces which are unaffected by the SCMS flags. Pronounced "scams" or "scums."

'scope: See *anamorphic*. Originally an abbreviation for *CinemaScope*™.

score: (1) (*noun*) The original-music composition for a motion picture or television production, recorded after the picture has been edited. (2) The conductor's chart, containing all *band parts* of a musical arrangement, and the individual band parts. (3) (*verb*) To write the music for a motion picture or television production soundtrack.

scoring paper: Music paper with several (usually five) lines printed above each *stave* for other information needed by the composer while writing a *score* or jingle. These lines may contain elapsed time counts or *SMPTE timecode* addresses, summaries of on-screen action, dialogue and / or narration, required effects, etc.

scoring stage: A large recording studio equipped with synchronous multitrack or other recording equipment, interlocked film and video playback, and large-screen projection in the studio itself. Motion pictures and video productions are scored here, the conductor watching the image and footage or *SMPTE timecode* data, and conducting performers so that the finished recording aligns properly with the footage.

scoring wild: The recording of a motion picture or television score using non-synchronous recorders. The conductor defines timings for each part of the score from footage counts of the edited film, and seeks a performance that approximates the required timing to a close tolerance, perhaps one-half of a second. The wild score can be made to fit exactly if its playback speed can be adjusted during transfer, by editing in / out short pauses, etc. See also *wild sound*.

scrape-flutter filter: In tape transports, a smooth or low-friction, non-magnetic, low-mass flywheel installed in the tape path in the order to minimize the pressure with which the tape meets guides, rollers, or other potential sources of scrape-flutter.

scratch demo: A quick and inexpensive demo, usually done in a home studio, to give a client a rough idea of what type of music is being composed or produced under contract.

scratching: A technique employed by some DJs, consisting of the rapid back and forth movement of a record turntable to cause the pick-up to produce the rhythmic scratching sound that is characteristic of rap and hip hop. This is done manually, with the turntable drive disengaged, or on a special turntable made for the purpose. Many records now feature scratching as an integral part of the recording process, and some CD players are now available with facilities for scratching and other effects.

screen: (1) A panel with surfaces designed to reflect or absorb sound, used to alter the acoustic behavior of a recording space, or to isolate one performer from another. (2) The shield part of a *shielded cable*. (3) A computer monitor.

S

scrub: To move backward and forward through an audio *waveform* under manual control in order to find a precise point in the wave for editing purposes.

SCSI: Small Computer System Interface. A bus specification standard used by personal computers to attach high-volume peripherals, such as mass storage devices, scanners, hard-disk recorders, etc. SCSI is a *parallel* data bus, and comes in several versions:

Type	Data Path Width (bits)	Max. Data Rate (Mbps)	Ave. Data Rate (Mbps)	Max. Total Cable Length
SCSI-1	8	5	2	9’9”
SCSI-2	8	5	2	19’6”
Fast	8	10	6	9’9”
Ultra (Fast-20)	8	20	8	4’11”
Ultra2 (Fast-40)	8	40	10	9’9”
Fast Wide	16	20	10	9’9”
Ultra Wide	16	40	12	4’11”
Ultra2 Wide	16	80	14	9’9”

S-DAT: Stationary-head Digital Audio Tape. A bi-directional cassette tape designed for domestic digital recording. The head is stationary, as opposed to the rotating head of *R-DAT*. The tape speed is very low compared to professional stationary-head systems, such as *DASH*. The required data rate for stereo operation is achieved by distributing the 16-bit data over 20 data tracks in each direction. The sample rates are the same as for R-DAT, but 4-channel recording is possible at 32kHz with 12-bit *nonlinear* operation. S-DAT is not compatible with DCC as the latter does not conform to the S-DAT standard. See *DAT*.

SDDS: Sony Dynamic Digital Sound. Developed by Sony, the SDDS split-surround format uses the usual *5.1* stems, plus additional left-center and right-center channels. The additional two speakers are employed at the front of the soundstage to deliver more uniform sound in wide-format theaters of screen widths of up to 60’ or more, where there might be seats with a *hole-in-the-middle* in between the C-L, and C-R channels. An important aspect of the SDDS format is that it can be decoded into four, six, or eight channels for playback on a wide variety of audio systems. Proponents of this format claim that the extra front channels make a significant difference in the amount of depth, fullness and natural *image* of the audio. See *7.1*.

SDII: Sound Designer II. The audio format native to Digidesign’s Sound Designer™ II (Macintosh) audio editing program. The original SoundDesigner format supports uncompressed, 16-bit mono sound files at several sampling rates. SDII supports 16-bit stereo files with sampling rates up to 48kHz; *ADPCM* compression is available at 2:1 and 4:1 ratios.

SDMI: Secure Digital Music Initiative. A working group formed by the *RIAA* to develop a “voluntary” method for protecting music copyright protection on the internet. SDMI is backed by the major labels, the aim of which is to simultaneously legitimize the distribution of music, while protecting copyright holders. There is an approximate analog in the UK with the name (Government’s) Creative Industries Taskforce (CIT).

SDS: Sample Dump Standard. The MIDI standard used to transfer digital audio samples from one instrument to another over a MIDI cable. See *SMDI*.

SDU4: See *DS4*.

S

sealed enclosure: The opposite of a *ported enclosure*. A loudspeaker cabinet with no vents or ports, e.g., an *acoustic suspension* system or an *infinite baffle*.

search-to-cue: A feature in *zero-locators* that allows the engineer to instruct the recorder to find a designated time location on tape and stop there and await further instructions.

second: The *interval* between one note and another, one *half-step* (minor second) or two half-steps (major second) above or below it.

segue: (1) An instantaneous switch from one musical selection to another without a gap; the absence of *cross-fade*. (2) Musical term meaning, "continue on without stopping." (3) A piece of music written to fill a gap, particularly in a musical or to link scenes in a film or TV program.

SEL: Sound Exposure Level. The SEL of a noise event is the *A-weighted SPL* lasting one second that would have the same acoustic energy as the event itself. SEL is a way of comparing the noisiness of events that have different durations, such as airplane fly-over noise.

selectivity: The characteristic that describes the ability of a tuned circuit or a receiver to select the signal frequencies desired and reject all others.

self-clocking: A device synchronization system whereby the clock information is embedded within the datastream, and the receiving device locks to it, as opposed to a *master clock* system. This system is not as stable as an internal clock as cables and network problems may introduce jitter. On professional systems, self-clocking protocols will be augmented with a master clock interface which sends *word clock* data.

self-erasure: See *saturation*, *HX/HX Pro*.

self-noise: The *intrinsic noise* or *hiss* produced by a microphone, measured in the absence of any input signal. Usually the self-noise specification is *A-weighted*: a self-noise figure of 18dB SPL or less is excellent, 28dB SPL is good, and over 35dB SPL is not good enough for quality recording. *Dynamic mics* have very low self-noise. The *S/N ratio* of a microphone is the difference in dB between the microphone's *sensitivity* and its self-noise. See also *reach*.

sel-sync or **sel synch**: Selective synchronization. (1) In a multitrack tape recorder, the use of the *record head* to replay material from other tracks to be heard by the musicians while they *overdub* a new track. This is essential for accurate synchronization as the extra few milliseconds afforded by its position will compensate for the inevitable delay if the signal was taken from the main *replay head* which is some millimeters farther down in the tape stream. Also called *simul-sync*. Ampex has trademarked Sel-Sync, but other products have similar features. The replay quality is less good when using the record head for a function for which it was not designed, thus it is important that the recorder returns to monitoring from the main replay head during *mixdown*. See *auto-input*. (2) On a recorder used for synchronization of sound with motion pictures or videotape, a separate *sync head* that records and plays the *sync tone* or other sync signal.

Selsyn motor: The trade name for a type of synchronous motor used to drive the projector, *dubbers*, and recorder in the dubbing, mixing, or *re-recording* theater. These motors, when connected to the generators and drive motors, will run in rigid interlock with one another, starting, running, slowing down, and running in reverse without loss of sync. See *rock and roll*.

S

semi-parametric EQ: A semi-parametric *equalizer* has two controls: frequency and boost/cut. A fully parametric band would allow the adjustment not only of the center frequency and the amount of cut or boost, but also the *Q* of the frequency band that is being affected.

semitone: See *half-step*.

sempre: Italian for "always," e.g., "sempre legato," always smoothly.

send: An output on a recording or sound reinforcement console for a signal to be sent to another device, such as an *equalizer* or *reverberator*. The signal rejoins the *chain* at the console via the *return* connector. Typical consoles will have several sends and returns. See *effects send*.

send level: A term used for both hardware audio *mixers* and software *synthesizers*, send level is the amount of a sound that is sent to an *effects processor* via the *effects send*.

sensitivity: (1) The minimum required signal at the input of an audio device in order to produce the rated output is generally called the sensitivity of the device. The higher the sensitivity, the lower the signal required at the input. A device with high sensitivity can process very small signals, but may be distorted by large ones, whereas one with low sensitivity can process large signals without distortion, but may add an unacceptable level of noise to a small signal. (2) In general, all information about a transducer's *response characteristics* to incoming sound waves. With respect to microphones, a standard performance specification that indicates the output voltage generated when a sound of known *SPL* and frequency arrives at the diaphragm. Given in mV by most manufacturers and generally specified for *broadband* response to *pink noise*. See *efficiency*, *transfer characteristic*. (3) See *velocity sensitivity*.

sensurround: A now-obsolete motion picture sound system which uses very strong, very low frequencies to simulate the effects of explosions, earthquakes, etc. It uses a special soundtrack on the film and separate amplifiers and high-power subwoofers to produce the effect. Originally, the first sensurround films simply triggered a noise generator during the scenes to have augmented LFE. Later versions recorded the very low-frequency on the print.

separation: (1) In a multiple microphone set-up, the extent to which any one microphone is able to reject unwanted sounds intended to be picked up by other microphones. Separation is desirable if *phase cancellation* is to be avoided, and can be increased by careful microphone choice an placement. See *directional microphone*. (2) The degree of isolation between signals flowing in two paths. Specified in decibels, indicating the level of a signal induced by one signal path in the other. See *crosstalk*.

s**ep mag**: Separate magnetic film. Terminology for a print whose audio track is on a separate roll of mag film to be run in interlock with the picture; the same as *double-system*.

sequence: (1) A set of music performance commands (notes and controller data) stored by a *sequencer*. (2) To edit the master tapes of an album, putting the songs in the desired order, prior to cutting acetates and *lacquer masters*, etc.

sequencer: A device or program that records and plays back user-determined sets of music performance commands, usually in the form of *MIDI* data. Most sequencers also allow the data to be edited in various ways and stored on disk.

S

sequential monaural: A single bit stream read from a CD is a really one-channel serial data-stream, with time-division *multiplexing* used to interlace the left and right audio channels. See *CD-ROM*.

serial: (1) Data transmission whereby all bits are sent one after the other, as opposed to using a *parallel* interface where bits are sent in groups. Parallel protocols are more complex than serial and the connectors are more complicated and more expensive, all of which makes it harder to get the specification implemented correctly among vendors. For this reason, MIDI uses a serial transmission protocol and specifies a *serial interface*, even though the transmission time required is substantially higher than for a parallel protocol. As MIDI files are quite small, compared with files which contain actual audio or video content, the relative lack of transmission speed was felt to be a good trade-off for the simplicity of its implementation. (2) Network connections where devices in the network are hooked up in a line, i.e., one to the next. This is called a *daisy-chain*, and is the opposite of a *parallel* topology.

serial interface: A hardware interface which transmits computer data using a *serial* protocol. The connector requires only one transmit pin and one receive pin, plus ground. However, the time taken to transmit each word is greater than for a *parallel* interface.

serial port: A connector on a Mac which is used to connect devices which use a *serial interface*, such as some MIDI interfaces, and printers.

servo: Short for servomechanism. A control system that uses *feedback* from an output signal to compare to a *reference signal*. The difference between the two is an error signal, which is used to change the output signal in such as way as to reduce the error. Used to control speeds in tape recorders and turntables and to position the laser beam in CD players. *Negative feedback* in an amplifier is an example of an electronic servo.

session tape: The original recording made during a live playing session.

set-up: (1) The part of a recording session in which the engineer places various instrumentalists or vocalists, sets up the microphones, and gets basic sounds through the recording console. (2) The *program chain*.

Set-Up: A Universal System-Exclusive message of the non-real-time type which defines a list of events which are to be carried out a given *MTC* times by the receiving device, such as event name, start and stop times, plus *effect* parameters.

SEU4: See *DS4*.

seventh: The *interval* between a note and another that is seven *scale* steps above or below it. This will be either ten *half-steps* (minor seventh) or eleven half-steps (major seventh).

SFI: A file extension specifying Turtle Beach's SoundStage™ audio format. Typically encountered as `FILENAME.SFI`.

S-format: See *C-format*.

sfx: Sound effects.

Shannon's channel capacity theorem: The formula,

S

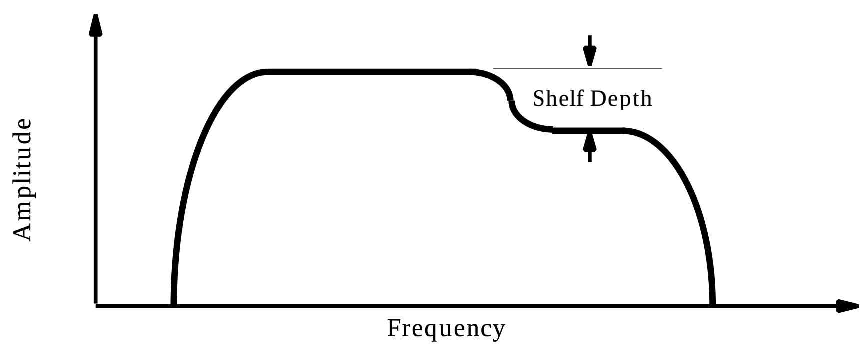
$$\mathbf{DR_{max} = W \log_2 (1 + SN)} \,,$$

where DR is data rate, W is channel *bandwidth*, and SN is the *S/N ratio*. It expresses the relationship between these three variables in digital audio. See also *oversampling*.

sharp: (1) Higher in pitch, as opposed to *flat*. In reference to musical *scales*, sharp (♯) indicates one *half-step* higher. (2) In *filters*, sharp refers to the rapidity with which the response of the filter falls off in the filter stopband, synonymous with *rolloff slope*. In general, the sharper the filter, the greater its *phase-shift*. See also *FIR*.

shielded cable: One or more insulated wires covered by a wrapped or braided metal shield. The wires carry the electricity, while the shield acts as an *EMI* barrier to the audio signals. See *balanced line*.

shelving EQ: A typical layout for semi-pro mixers provides low- and high-shelving equalization from a pair of knobs labeled “bass” and “treble” and one *semi-parametric* midrange band. The low and high EQ are called *shelving* because a graph of their response curve looks somewhat like a drawing of a shelf. Any portion of the tone above (in the case of treble EQ) or below (in the case of bass EQ) a factory-preset is boosted or attenuated.



Shelving

shock mount: A microphone suspension system that prevents mechanical vibrations of the stand from reaching the mic. Usually made of elastic bands mounted on a metal frame, which together hold the mic in position without rigid mechanical contact with the stand.

shoot: In film, slang for recording. It is derived from the previous use of *optical sound* in all film sound recording, i.e., sound recorded on film.

short circuit: A circuit in which some unplanned event, such as the failure of a component or a stray piece of metal bridging two *live* contacts, has reduced the *resistance* in the circuit, resulting in increased *current* flow through the circuit. It can be said that a complete short circuit has zero resistance and consequently infinite current flow, however, AC line voltage will limit this to either 120V or 240V, which is plenty of current by which to get dead. In anticipation of these accidents, most devices have fuses or other componentry integral to the device and / or its power supply which are designed to fail, breaking the circuit, in the presence of abnormally high voltage.

shortcut: A file on a PC-type computer that is similar to the Mac’s *alias* feature.

S

shotgun microphone: See *gun microphone*.

shuttle: To wind the tape on a tape recorder back and forth in order to locate a specific audio event.

SI: Système Internationale. The international measurement standard known as "metric," as in SI Units. See *watt*.

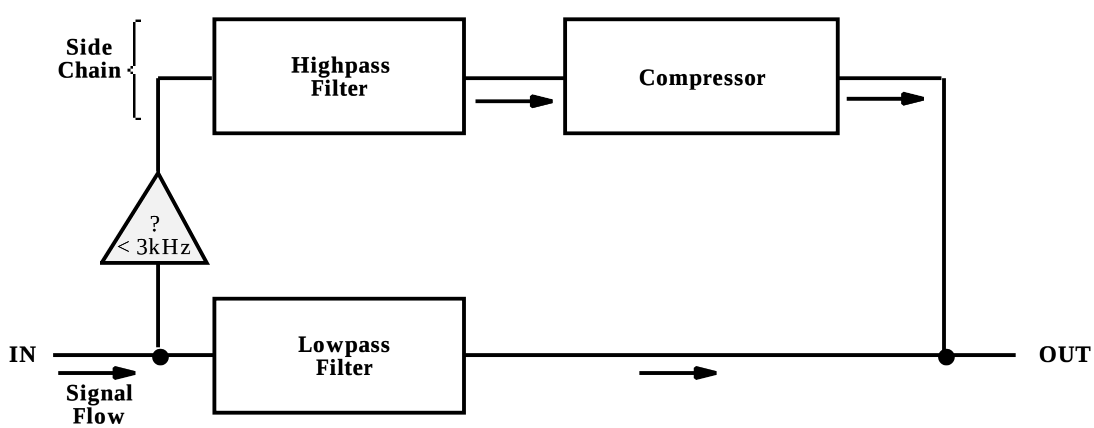
sibilance: The high-pitched whistling caused by air passing around the teeth, such as is produced by saying certain letters: f, s, t, x, or soft-c. Most sibilance occurs in the 5-10kHz region. See *de-esser*.

Side: (1) Any master tape of one song, whether recorded for inclusion in an album, or as the A- or B-Side of a single. Always capitalized, Side is used in recording contracts, so precise definition is important. (2) In the *AFTRA* Code of Fair Practice, a Side is a master tape of no more than 3½ minutes. If the total length of the song is longer than that, AFTRA singers receive additional payment.

side-addressed: See *end-addressed*.

sidebands: (1) Frequency components outside the natural *harmonic series*, generally introduced to the tone by using an audio-range wave for *modulation*. See *AM*, *FM*. (2) Elements of a high-frequency carrier signal created when the data / voice signal is modulated with the *carrier* signal, as in *FM synthesis*. The new sets of *partials* generated by the modulated carrier give the tones a *timbre* other than that of the original *sine wave*.

side chain: A circuit which measures how strong the input signal is which is being modulated by a *compressor*. This information is then used to control the *gain* of the circuit output. The compressor will behave differently, depending on whether the side chain responds to average signal levels or to absolute signal *peaks*. In the example below, for *Dolby B*-type *companding*, the side chain feeds any signals above 3kHz to the compressor:



Side Chain

sign: A musical symbol (𝄋) indicating a particular place in the music, e.g., D.S. (Dal Signo), meaning "play from the sign." D.S. (S.R.) means "play from the sign without repeats," and D.S. (C.R.) means "play from the sign with repeats."

S

signal: (1) The desired portion of electrical information, i.e., the information content of any transmission medium, i.e., the part of the *waveform* that is not *noise*. (2) A generic name for any one of a number of forms (magnetic orientation, voltage) which audio may take in the *program chain*.

signal entropy: A condition caused by audio data which is characterized by frequent *transients*, causing a *compression* algorithm to poorly encode data. If the signal is compressed before encoding, the data compression will be much more effective, with less effect on the audio. Signal entropy is not a problem for *ADPCM*-type compression algorithms.

signal generator: A test instrument that produces one or more of the following types of *waveforms* through a wide range of frequencies: sine wave, square wave, sawtooth, etc. See Appendix C.

signal ground: Each component of a sound system produces its own internal ground. This is called the audio signal ground of the device. Connecting devices together with cables can tie the signal grounds of the units together in one place through the conductors in the cable. See *ground loops*, *chassis ground*.

signal processing: The *modulation* of an audio signal in a generally desirable way by any device inserted into the audio *chain*.

signal processor: See *processor*.

signal-to-noise (S/N) ratio: Also abbreviated *SNR*. The ratio, expressed in dB, between the *signal* and the *noise*. To have good *dynamic range*, there must be a good S/N ratio so that the softest signals are not overshadowed by noise. There is approximately a 6.03dB increase in dynamic range per word-bit increase in a digital system; the S/N ratio can be expressed by:

S/N ratio = 20 log (number of quantization increments)

Bits	Increments	S/N ratio (dB)
8	256	54
14	16,384	90
16	65,536	102
20	1,048,576	120

A live musical performance needs a dynamic range of at least 70dB, or a minimum of 14 significant bits. See *noise*, *bit depth*.

simple time: See *time signature*.

simulDAT: A DAT recording made during a *Telecine* transfer in which the production audio is transferred to a DAT whose *timecode* matches that of the videotape.

simul-sync: See *sel-sync*.

sine wave: The *waveform* of a pure *alternating current* or voltage. It deviates about a zero point to a positive value and a negative value, consists of a single frequency and has a musical *pitch*, but a neutral *timbre*. Audio-range sine waves contain only the *fundamental frequency*, with no *overtones* and form the building-blocks for more complex sounds which are combinations of sine waves. Sub-audio sine waves are used by an *LFO* to *modulate* other waveforms to produce *vibrato* and *tremolo*. Also called a *sinusoid*. See *Fourier analysis*.

S

single-ended noise reduction: Single-ended NR systems work on playback only. Analog systems work by combining *dynamic filters* with low-level *expanders* so that the signal level decays, high frequencies are progressively filtered out. At very low levels, the expander acts as a soft gate to clean up pauses. Digital systems are more complicated and use a split-band expander. See *noise reduction*.

Single Note Retuning Message: A Universal System-Exclusive message of the real-time type which is intended to provide a performance control over the tuning of an individual note.

single-stripe: Magnetic film which is coated with oxide, containing a single audio track.

single-system sound: A way of producing sound for motion pictures where the sound is recorded directly onto the film in the camera at the time of shooting. Sound and picture are thus on a single strip of film from the start, with image and sound in projection sync with each other. Used primarily for news and sports where the relatively limited-fidelity of single system cameras can be tolerated. Cheaper than *double-system sound,* but also less flexible. Video is, by definition, single-system sound, but may be used as a double-system if a separate audio recorder is used during filming.

six-channel mix: A nomenclature used for an audio-only *surround-sound* mix, more generally referred to as *5.1* which can refer to either audio-only or audio-for-video.

six-mix: See *six-channel mix,5.1*.

sixth: The *interval* between a note and the one six *scale* steps above or below: either eight *half-steps* (minor sixth) or nine half-steps (major sixth).

skew: A deflection or dislocation of the proper path for magnetic tape as it passes over a misaligned tape head, guide, or roller.

slapback: (1) An unwanted and distracting *echo* caused by a reflective surface in any environment, e.g., a large window in a recording studio, a cement wall at a concert venue. (2) A tape delay and the *slap echo* it creates. See *delay line.*

slap echo: The single repetition of a signal at a fixed time delay to simulate an *echo* from a single reflecting surface, as opposed to a multiple echo from a *time delay*, where the delayed signal is repeatedly fed back into the delay input. The sound is repeated once or twice only, with a short delay of 40-60ms.

slate: To identify the various takes in a recording session by announcing the take numbers and recording them on one track of the tape. Slating and the notes taken at the time of recording are important once tape editing begins, otherwise it would be almost impossible to find any particular take.

slave: An audiotape or videotape transport, motion picture projector, magnetic film recorder or *dubber* whose movements follow the movement of a single *master* transport, accomplished electronically by using, for example, *SMPTE timecode*, or electromechanically by linkage of sprocketed machine motors to provide identical movement of all sprocket drives.

slew limiting: The effect which occurs when the input signal amplitude would require the output to change faster than the maximum *slew rate* of the device.

S

slew rate: Slew rate is a common specification for *transient* behavior of electronic devices. It is expressed in V/µs and tells how fast the output of the device rises, given a rapidly rising input. A slew rate of 40V/µs means that it would take one µs for the output to change by 40V. Since studio equipment typically operates on +15V and -15V internally, its output can never really swing 40V. In reality, it might go 5V in ⅛ µs. An audio device will have a maximum slew rate above which it cannot operate. This maximum limits the high-frequency power output of amplifiers and limits the high-level, high-frequency handling capacity in all audio devices. Too low a maximum slew rate results in slew-induced *distortion*, called *transient intermodulation distortion* (TIM).

slip cue: In playing records, the process of locating the first note of music in any band, holding the disc motionless while the turntable rotates beneath it, then releasing the record so that music begins precisely when the disc is released. This is done either to effect precise *segues*, or to align the rhythm of one record with another one already playing, called *mixing*.

slope: See *rolloff slope*.

slop print: See *workprint*. Also called a *dirty dupe* or *scratch print*.

small-room X-Curve: See *X-Curve*.

small signal bandwidth: The bandwidth an audio device will exhibit at relatively low signal levels. Most audio devices will have a wider frequency range at low signal levels than high signal levels because of such effects as *slew limiting*. See *power bandwidth*.

smart slate: Timecode *slate* that contains a timecode generator. A “dumb” slate must be fed timecode continuously.

smart sync: The ability of some A/D-D/A converters, digital recorders, or other digital devices to automatically sync to *word clock*.

SMDI: SCSI Musical Data Interchange. A specification for sending MIDI sample dumps over the SCSI bus. This requires that the computer be connected to a sampler via both SCSI and MIDI interfaces. See *SDS*.

SMF: Standard MIDI File. A *file format* for transferring data between sequencers and between Macs and PCs. Sequence data is stored on disk in one of three formats: Type 0 packages all data into a single track, regardless of the number of original tracks. Type 1 retains the track layout, although the tempo and time signature of all tracks will be those of track one. Type 2 is the same as Type 1, except that each track keeps its own tempo and time signature, but is seldom implemented. The files include information called *meta event*s about such things as track and instrument names, copyright notice, tempo change, *SMPTE timecode* offset, lyrics and *key* and *time signatures*. The time between sequence events is encoded in *delta time* messages of between one and four bytes.

SMP: Turtle Beach’s SampleVision™ audio file format. Typically encountered as `FILENAME.SMP`.

SMPTE: Society of Motion Picture and Television Engineers. Also used as an abbreviation for *SMPTE timecode*.

SMPTE curve: An *equalization curve* standard in the U.S. for 35mm *mag film*.

S

SMPTE leader: See *SMPTE Universal leader*.

SMPTE sync track: A square wave recorded onto an audio track. Note that this is only possible with digital audio, so for the most reliable results when transferring audio, use *jam sync*. See also *sync track*.

SMPTE timecode: A high-frequency timing reference signal developed by *SMPTE* and used for synchronizing film and videotape to audio tape and software-based playback systems. SMPTE is a tempo-independent code which comprises a continuous stream of absolute positional data, so if a short section of code gets lost or corrupted, the system knows exactly where it's supposed to be the next time a piece of valid code is read. Usually generated at the picture source, i.e., the SMPTE *master clock* generator that drives the film or television camera system, the signal is recorded onto the videotape or along the edge of the motion picture film, and sent simultaneously to the audio recorder. The signal contains encoded numerical information, allowing the same point in film or tape time to be located on the separate strips of film / videotape and audio tape, for proper alignment or *ADR*. The playback operator can select a SMPTE timecode number that instructs videotape and audiotape machines to locate a certain point and begin playing in sync from that absolute location.

Timecode data are in the form of a timecode address (*TCA*), which make up the HH:MM:SS:FF part of the timecode word, where HH is a two-byte number for absolute time hour, MM is minutes, SS is seconds, and FF denotes the absolute *frame* number. See *frame*, *jam sync*, *LTC*, *MTC*, *BITC*, *VITC*, and *bi-phase modulation*. Also called *longitudinal timecode*. As opposed to speed-only sync codes such as *pilot tone*, *FSK*, and *DIN sync*.

SMPTE Universal leader: Designed primarily for video, the *leader*, after Picture Start, features a sweep hand counting down from eight seconds. See *Academy leader*.

S/N: See *signal-to-noise ratio*.

snake: An assembly of cables or wires often used to carry several channels of audio signals between two points, such as from the stage to a mixer. The most commonly used snakes consist of several sets of *balanced line* connections, with a *snake box* at the mic end, fitted with the appropriate number of *XLR* -type microphone sockets, the long, large, multiwire cable on which the individual channels are carried, terminating in a large modular connector for connection to a distribution panel which is, in turn, connected to a power amplifier or preamp. Also called *multicore* or *rope*. A snake box is also called a *stage box*.

snake box: See *snake*.

snake track: A special *optical soundtrack* on a test film for adjusting the alignment of the optical sound system in a motion picture projector equipped to reproduce *SVA* soundtracks.

snapshot automation: A form of MIDI-controlled mixer *automation* in which the controlling device records the instantaneous settings (the snapshot) for all levels and panpots, and recalls these settings on cue.

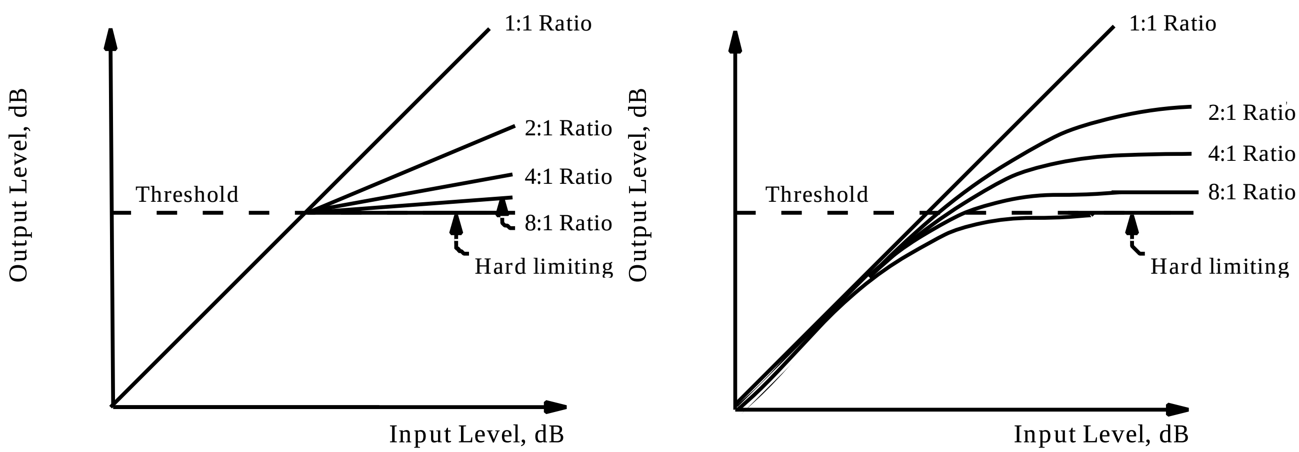
SND: See *SouND resource*.

SNR: See *signal-to-noise ratio*.

S

soft clipping: A form of *clipping* where the edges of the clipped waveform are rounded rather than sharp. Soft clipping is much easier on the ears and tweeters than hard, as it contains much less very high frequency energy. Compare with *hard clipping.*

soft knee compression: The output of a *compressor* whose gain reduction is brought in progressively over a range of input signal values, such as over 10db or so, starting a few dBs below the threshold. When the input signal amplitude comes within the range of the threshold, the compressor starts to apply again reduction, but with a very low ratio setting. As the input level increases, the compression is automatically increased until, at the threshold level, the ratio becomes infinite. Compare with *hard knee compression*.



Hard-Knee Compression **Soft-Knee Compression**

soft pedal: (1) On a piano, the pedal for producing a softer sound, accomplished by shifting the action sideways so that fewer strings per note are struck. On less expensive instruments, the hammers are simply moved closer to the strings. (2) A MIDI Controller Change message assigned to a *parameter* in a synthesizer which reproduces the function of a soft pedal.

Software Thru: When Software Thru is On, MIDI data at the sequencer's MIDI input passes through to the output. When it is Off, MIDI data at the input does not appear at the output. See also *MIDI Thru*, *Local Control.*

> **Local Control On, Software Through On**: Not recommended as this setting can cause *note-doubling.*
>
> **Local Control On, Software Thru Off**: Recommended mode. The sequencer records data, and you hear what you play on the keys.
>
> **Local Control Off, Software Thru On**: Alternate recommended mode. The keyboard acts like a master controller, but you'll only hear what you play if Software Thru is On and directing the input signal to a specific channel, i.e., the one that feeds the sound you want to hear. Most sequencers let you choose whether Thru echoes data on the incoming channel or directs this data to a different channel.
>
> **Local Control Off, Software Thru Off**: Not recommended. This allows the playback of tracks already recorded into the sequencer, but it is not possible to monitor what is being played or recorded.

solid-state: A term which indicates that a device uses semiconductor (IC) devices, instead of vacuum tubes. Note that tubes in the UK are called *valves.*

S

solo: (1) A feature on a mixing console that automatically routes one or more selected *channels* to the recording monitors or headphones without disturbing the main audio mix. Some studio consoles use *destructive solo*, where the soloed instruments replace the mix in the main stereo bus. On sound reinforcement mixing consoles, solo functions are normally routed to headphones (non-destructive solo), allowing the engineer to check console channels while the concert is in progress. *In-place solo* is a function that permits the user to hear individual channels, but also in the correct stereo perspective, as defined by that channel's *pan* control. (2) A section of music which gives particular prominence to one musician.

solo-in-place: In a recording console, the function that allows all input signals to be *muted* except one, permitting that one to pass through to the monitor or mix buses exactly as heard in the composite mix, preserving its level, equalization and other processing and stereo position.

sone: A standardized unit of perceived *loudness*. A graph of sones vs. *SPL* shows, for any constant SPL, the relative loudness the ear perceives at each frequency, i.e., the inverse of the *equal loudness curves*.

song: (1) Strictly, a piece of solo vocal music, with or without accompaniment, and of short duration. In popular music, this has come to mean any music which is sung. (2) A synonym for a sequence, i.e., a selection of MIDI performance data recorded on a sequencer.

Song Position Pointer (SPP): A System Common MIDI message that tells a device how many sixteenth-notes have passed since the beginning of the MIDI sequence file. An SPP message is generally sent in conjunction with a Continue message in order to start playback from the middle of a song. Alternately, by reading a sync signal (such as *FSK* or *SMPTE timecode*) recorded on a tape track and converting it to an SPP message, a *synchronizer* could instruct a MIDI sequencer to start playing a sequence at a given location and keep it in sync with the tape.

Song Select: A System Common MIDI message which outputs the number of the song (sequence) which a sequencer or drum machine should play.

soprano: A high human or instrumental voice above the *alto* in pitch, ranging from about middle-C upwards. The very highest instruments in a given family are sometimes called sopranino, such as sopranino recorder. Soprano parts are notated with the treble *clef*.

sostenuto pedal: A pedal found on a grand piano and mimicked on some synthesizers, with which notes are sustained only if they are already being held on the keyboard at the moment when the pedal is pressed. Compare with the *sustain pedal*.

sound: The vibration of some physical object, resulting in a pressure front moving through air molecules at 1130 feet per second, a zone of high pressure followed by a zone of low pressure. All sounds can be analyzed as having a number of specific characteristics: *pitch* (frequency), *timbre* (tone), *amplitude* (volume) and *envelope* (the shape of the sound as it changes over time.) What distinguishes one sound from another is the combination of *harmonics* and other *partials* that are present in the sound and how the amplitudes of the various partials change over time.

sound blanket: A think, sound-absorbent blanket, often a mover's quilt, that can be spread on the floor of a set or hung just outside camera range. Its purpose is to damp unwanted *reflections*, *echoes*, or *reverberation* which would affect *location sound*, or to keep unwanted outside sounds from reaching a microphone during shooting of film or videotape.

S

sound board: A separate, resonant piece of wood in a piano, located under the string assembly. The sound board reflects the sounds back from the bottom of the piano case and out away from the piano. The quality and shape of the sound board give each piano a distinct *timbre*.

soundcard: A circuit board that installs inside a computer, adding new sound capabilities. These capabilities can include an *FM* or *wavetable synthesizer* and audio inputs and outputs. MIDI inputs and outputs are also normally included.

sound check: A diagnostic procedure carried out before a performance or recording, particularly when amplified instruments and / or a *PA* system is used to ensure that all of the equipment is in working order, and that sound levels are set correctly.

Sound Controller Message: One of ten defined MIDI Controller Change messages which are assigned to general-purpose parameters in a *synthesizer* or *effects* unit and allow *real-time* editing of sound quality and effects from a *sequencer*.

sound cutting: See *track laying*.

sound designer: (1) Most commonly, the person who creates special sound effects for film; originally, the person assigned responsibility for the over sound of the film, and who supervises the sound editing and *re-recording*. (2) SoundDesigner™, a popular digital synthesis program.

SoundEdit: A format used on the MacRecorder which became a standard for exchanging 8-bit mono or stereo sound files at four sampling rates from 5.5kHz to 22.255kHz; compression is available for mono files at 3:1, 4:1. 6:1, and 8:1. Newer SoundEdit software supports 16-bit sounds with sampling rates up to 64kHz.

sound field: The area and / or pattern of air pressure disturbance caused by the *compression* and *rarefaction* of energy in the *AF* band.

Soundfield microphone: An extension of *MS* recording for *ambisonic* recording. This technique captures and reproduces true surround-sound, with height information as well as 360˚ horizontal imaging, as opposed to the artificial spatial position of various cinema surround systems. This is accomplished by six separate capsules which allow separate recording of left-right, front-rear, and up-down sound sources. Most use of these microphones is for stunningly accurate stereo recording.

sounding: The act of recording sound on a mag *release print*.

sound intensity: Defined as a measure of the net flow of acoustic energy in a *sound field*. The units are *watts* per square meter, and because the energy moves in a particular direction, sound intensity is a vector quantity, i.e., it has magnitude and direction. Sound intensity is not able to be measured directly, and it should not be confused with *SPL* which is what a sound level meter measures.

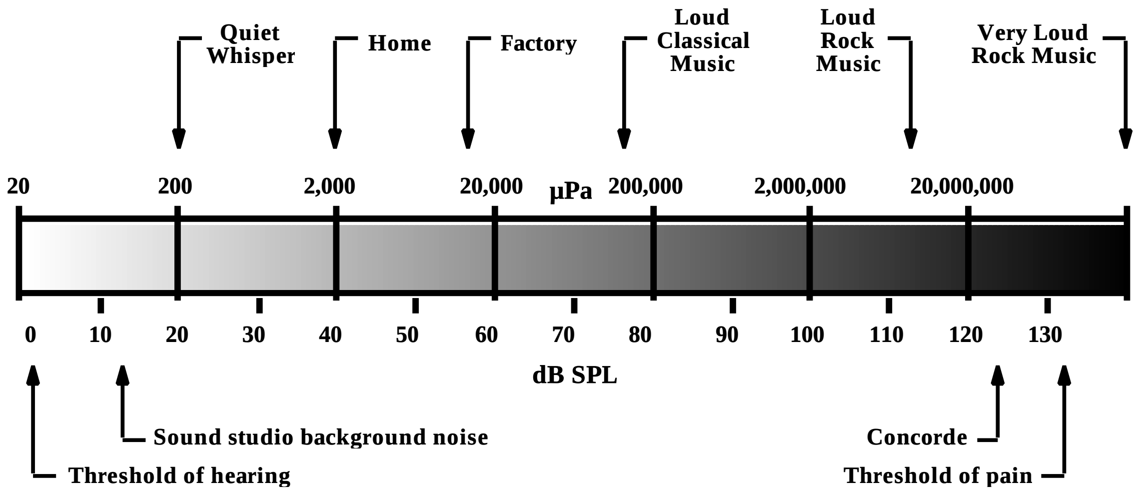
Sound Manager: A part of the Mac operating system that handles audio functionality such as input, mixing, and playback. Currently the Sound Manager supports only mono or stereo recording and playback at 16-bit, 48kHz resolution. Most Mac audio cards do not use Sound Manager for this reason, achieving both increased *bit-depth* and *sampling resolution*. See also *WAV/multi-WAV drivers*.

S

sound module: See *tone module*.

sound-on-sound: The same as *overdubbing*.

sound pressure level (SPL): The *amplitude* of an acoustic wave stated in dB that is proportional to the logarithm of its *acoustic intensity*. A sound wave progressing through air causes the instantaneous air pressure at any given point to vary above and below the barometric pressure in accordance with the *waveform* of the sound. This variation in pressure is used as a quantitative measure of the strength of the sound, and is called *sound pressure*. This is the quantity which a pressure microphone measures, and if it is expressed on a dB scale and referenced to a pressure of 20 µpascals, it is called the sound pressure level. The amplitude dynamic range of human hearing goes from 0dB, or 10^{-16} watt/sq.cm., the upper threshold of human hearing to 130dB or 10^{-4} watt/sq.cm., the threshold of pain, a factor of 10^{13} in range.



sound quality: The *timbre* of the sound, modified by the following parameters:

Sound Quality	Technical Term	Parameter
Overall Timbre	Harmonic Content	Oscillator Waveform
Brightness	Amplitude of HF Harmonics	Filter Rolloff
Tone Changes	Dynamic Filtering	VCF Envelope
Volume Changes	Dynamic Amplitude	VCA Envelope
Vibrato	Pitch Modulation	LFO to Pitch
Tremolo	Amplitude Modulation	LFO to VCA
Pitch	Frequency	Oscillator Tuning
Note Start Speed	Attack Time	VCA Envelope: Attack
Note Stop Speed	Release Time	VCA Envelope: Release
Percussiveness	Attack Treatments	VCA Envelope: Decay

sound reader: A magnetic *playback head* mounted either directly on a *sync block*, or on a free-standing metal base with alignment rollers. Used by film editors to check synchronization of all effects and music, they slowly hand-crank the *workprint* and various rolls of magnetic film through a sync block on the editing bench. Signal from the sound reader is amplified by a *squawk box*.

S

sound reinforcement: The general term for a sound system designed to amplify the voice and/or music to improve its intelligibility to an audience. A sound reinforcement system always consists of at least one microphone and associated preamps, control console, amplifiers and loudspeakers. Often abbreviated *SR*.

Sound Resource Format: A Macintosh file format, usually abbreviated *SND*, which tends to be used for small, short sound effects, such as beeps and clicks. They were originally used for system sounds. *AIFF* and *QuickTime* formats are more commonly used to record narration or music, especially on larger files.

sound stage: A theatrical stage for filming that is specially treated for the simultaneous recording of dialogue and/or music. A sound stage must have a low *NC curve* and a low *reverberation time*, and the cameras and other equipment used must be specially designed for quiet operation. This is not the same as a *re-recording stage*.

sound stripe: A narrow strip of magnetic material applied to one edge of motion picture film for recording of the *film soundtrack*, in the manner of a tape recorder. 70mm release prints of motion pictures exclusively use magnetic soundtracks, as do some 35mm prints. Compare with *SVA*. See *stripe*.

sound synthesis: The process of electronically creating a sound with an *oscillator*. Types of sound synthesis include *additive synthesis*, *subtractive synthesis*, *sample (playback) synthesis*, *FM synthesis*, and *physical modeling synthesis*, *VPM*.

sound system equalization: The equalization of a *sound reinforcement* system, either to increase its amount of gain before *feedback*, or to make its overall *frequency response* more *linear*.

soundtrack: See *film soundtrack*.

source track: Audio input into the *mixdown* process. (1) Generically refers to the music contained in a film, although it literally means the entire audio portion of a film, video or television production, including effects and dialog. (2) Also refers to the physical space on film that contains the audio information.

SOX: Start Of eXclusive. A MIDI message used to indicate the start of a System-Exclusive message.

spaced microphone recording: *Stereo* recording techniques including *Decca trees*, *binaural*, *binaural synthesis*, *spaced pair*, etc. which produce a large difference signal between the left and right channels. Spaced pairs should be placed directly in front of the sound source with a spacing of between one-half and one-third the width of the actual sound stage. Contrast with *coincident pair*.

S

spaced pair: A *stereo* microphone technique where two or more microphones are pointed directly at the source, separated by two feet or more, according to the width of the *sound stage*. Depending on the mic-to-source distance, this method can provide an extremely wide (occasionally exaggerated) stereo perspective. (It is possible to move one mic forward a bit to equalize balance without sacrificing phase or stereo *image*.) While spaced mics produce the widest stereo image, they can suffer poor *mono compatibility*. The use of more than two microphones is called a *spaced array*. Spaced microphone techniques can use either *omnidirectional* microphones or *directional microphones*. See *coincident pair, near-coincident pair, spaced microphone recording, Faulkner array*.

spacer: Plastic tape in different colors used to separate sections of film or magnetic tape, used for the purposes of identification or protection. See also *leader*.

S/PDIF: Sony / Philips Digital Interface Format. A consumer standard, similar to the professional *AES/EBU* standard, for encoding digital audio. Given the international standard number IEC-958, S / PDIF was originally designed to be the standard for transmitting audio data between CD players and DAT players, at a time when DAT was thought to be the next consumer audio recording medium. The data transmission is the same as AES / EBU: four times the sample rate. S / PDIF uses either standard *unbalanced* coaxial cable and *phono jacks*, or fibre optic cable and a connector called a *Toslink*, usually at -10dB. Note that the coax is not an RCA audio cable, but a video-grade cable: the impedance on this cable is 75Ω, and that S / PDIF carries the *SCMS* copy code. S / PDIF is a *self-clocking* scheme. The EIAJ has adopted S / PDIF as CP-340 Type II.

speaker simulator: The sound produced by guitar and, to a lesser extent, bass amplifiers is dramatically affected by the actual loudspeaker cabinet through which the amplifier is played. This is due to the speaker itself which does not reproduce high-end frequencies well. A speaker simulator emulates the frequency response of a conventional loudspeaker cabinet and is typically built in as a part of a *DI* box, combining the elements of the DI with the simulator.

Speakon: See *Neutrik*.

spectral gain intermodulation: The misuse of a (manual) *compressor*, set to high ratios and fast attack and release constants, producing loss of *transients*, loss of high-end, and other undesirable effects such as a kick drum *ducking* the vocals. The remedy for spectral gain intermodulation is to set a low threshold to bring low signals up, but also to be less extreme in the ratio, attack, and release settings. See also *breathing*.

spectral recording (SR): See Dolby SR™.

spectrum: The range of *frequencies* present in a *waveform*, such as an audio signal. When a time-varying signal is subjected to frequency analysis, it is transformed from the time domain to the frequency domain. The frequency-domain representation of the signal is called the *spectrum*, and the time domain representation is called the *waveform*. The two quantities contain the same information, and one can be converted into the other by a mathematical operation of the Fourier transform. *Spectrum analysis*, *frequency analysis*, and *Fourier analysis* are synonymous.

spectrum shifter: See *frequency shifter*.

S

speed: Called out by the production sound mixer on-location, indicating to the camera crew and others that the audio recording chain is now recording.

spill: See *leakage*.

spiral: The widely separated grooves that follow the last selection on a record, leading the stylus quickly toward the center groove. Also called the *tail-out* on mastering lathe controls. The less-widely separated grooves preceding the first selection on a record are called the *lead-in*.

splice: The joint made between two pieces of magnetic tape or motion picture film in the process of editing. Or, the point in audio or screen time at which this joint occurs.

split-band: The analysis or processing of a signal by separating its frequency spectrum into discrete regions, or *bands*. See *split-band coding*, *split-band compression*.

split-band coding: A *transform* encoding / decoding technique where the signal is split into a number of independent bands, used to take advantage of the spectral redundancies within the audio spectrum. A type of split-band encoding is *sub-band coding.*

spilt-band compression: Compression where different sections of the audio *spectrum* are compressed separately. With a wideband compressor, if there is a dominant portion of the spectrum, no matter how small, it affects the whole waveform. With split-band devices, the greatest effect is with high ratios, where the effect is more like localized limiting. Split-band compressors are similar to *multiband audio processors* in the first stage of audio processing.

split console: A *mixer* where the monitor section is actually another complete mixer; the recording input and monitoring functions are entirely separate. The structure of a split console is: input channels, groups, tape monitor channels, and stereo output. This type of design quickly becomes unwieldy as the number of tracks increases, and performing simple functions such as *bounce-downs* often requires external signal patching to re-route monitor returns through input channels and on to the *group sends*. The opposite of an *in-line* mixer.

split feed: (1) The process of routing the same signal to two or more separate outputs, modules, or devices. (2) The actual device that allows the signal to be routed to more than one destination. Also called a *Y-connector*.

split keyboard: A single keyboard is divided electronically to act as if it were two or more separate ones, separated into *zones* at a *split point* or points. The output of each note range is routed into a separate signal path in the keyboard's internal sound producing circuitry or transmitted over one or more separate MIDI channels. Applications include playing a bass sound with the left hand while playing a piano sound with the right.

split point: Where two keyboard zones adjoin, but do not overlap, on a *split keyboard*, the division between them is called the split point. Where the *zones* overlap, they are called *layers*.

split reel: A film reel that can be separated into two separate flanges, so that the film itself, wound on a plastic core, can be stored without reels. Used constantly in editing pictures and magnetic film.

S

split surrounds: Also known as *stereo surrounds*. A term used to refer to the Dolby 70mm sound format that gives two surround channels (LR, RR) on a mono-surround-compatible print. Also indicates the use of separate surround speakers in any sound format which have two (or more) discrete surround tracks.

SPP: See *Song Position Pointer*.

spot-erase: To erase a very small segment of recorded tape, on one track or all. Most accurately done by slowing down the tape by *varispeed*, enabling the engineer to *punch-in* and *-out* at exact points marked on the tape itself.

spotting: Used in film scoring, the process of identifying the specific scenes on film where music *cues* will take place, including information on length and style. Also, the act of reviewing the film with the director to determine work that will be needed on the *soundtrack*.

spring reverb: An electromechanical *effects* device that uses springs to simulate natural room or hall reverberation. Spring reverbs work basically like *plate reverb* systems, but are much smaller and produce an inferior effect. A *transducer* sets up vibrations in the spring which rattles back and forth, extracted by a pick-up at the other end. The character of a particular spring reverb unit is fixed (other than wet / dry balance), but can be optimized for the sound source at the design stage by careful choice of the number, length, diameter, and compliance of the spring(s). Also called a *spring-line reverb.*

square wave: A square wave is a periodic *waveform* which consists of a *fundamental* and all odd-numbered *harmonics*. The harmonics gradually decrease in amplitude as frequency rises, and they are in *phase* with the fundamental. The square wave is a good test signal because it exercises the device at many frequencies at the same time, as does music. Also called a *pulse wave*. See *duty cycle*, Appendix C.

squawk box: A small table-top amplifier and speaker used for playback of magnetic film passing through a *sound reader*. The poor sound quality is reflected in the name of the device. See *bench*.

SR: Used as an abbreviation for (1) *sound reinforcement* or (2) *spectral recording*. See *Dolby-SR, Dolby SR.D.*

S.R.: Senza Repeats, as in “play from the sign without repeats,” written D.S. (S.R.).

SR noise: See *biased noise*.

SR.D: See Dolby SR.D™.

SSL: (1) Solid-State Logic, circuitry composed of only *solid-state* components, as opposed to vacuum-tube devices. (2) A company which makes large, expensive mixers.

staccato: Italian for “short.” Used to indicate that notes should have a short duration and be detached as widely as possible from one another. The opposite of *legato.*

staff: See *stave*.

S

stage: (1) In an electronic device, when a signal is repeatedly amplified, any portion of the circuit in which an increase in amplification is achieved. In a mixer, for instance, there is generally a stage or *buffer* after each *fader* or *pot*, providing a light load to the fader and a constant *impedance* to the next *stage* or device. (2) A *dubbing stage*.

stage box: See *snake*.

stage monitor: A wedge-shaped *loudspeaker* placed on the stage close to the musicians in a live or broadcast performance. It allows them to hear the *monitor mix*, e.g., the bass player may particularly need to hear the drum part.

stage sync: How well in sync the *Foley* or *ADR* is when it is recorded.

stampers: Used to press plastic *CDs* , the stamper is a negative copy mold made from the *glass master* where the pits are reversed into spikes.

standard operating level: A *reference level* by which various pieces of equipment can be adjusted to produce identical output levels or meter readings. In professional recording, standard operating level is defined as 0VU=+4dBm. In broadcasting, 0VU=+8dBm. See *line-level*.

stand-by mode: See *auto-input*.

standing wave: An acoustical *resonance* in a room caused by parallel wall surfaces, also called a *room mode* or an *eigentone*. Any set of parallel walls will establish a series of standing waves, the lowest one of which has the wall spacing as a half-wavelength. The sound waves interfere with one another to produce a series of places where the *SPL* is high and another series of places between then where the SPL is very low, as if the sound wave were stationary between the two surfaces. See also *room mode, quarter-wavelength rule*.

star network: A type of *MIDI network* in which a master unit is directly connected to all devices in the system, instead of linking them in a *serial* topology. This obviates the need for MIDI Thrus, but requires the use of a device that has a number of *parallel* MIDI sockets.

Start: A MIDI System real-time message which causes a device to start playing the current *song*. Usually only implemented by sequencers and drum machines. See *Stop*.

start bit: In *serial* data transmission, the use of an extra bit to precede the data bits which serves to alert the receiving device, preparing it for data to follow. See *stop bit*.

Start menu: The main menu in a Windows™ system, used to access applications, files, and control panels. This is the equivalent of the Apple () menu on a Mac.

static: Any dynamic, high-frequency, intermittent *noise* is called static. Static is distinct from continuous noises, such as *hum* or buzz.

static signal processing: Signal processing in which the nature of the input signal has no effect on the type or amount of processing applied to it, as opposed to *dynamic signal processing*.

status byte: A MIDI message byte that defines the meaning of the data bytes that follow it. MIDI status bytes always begin with a 1 (hex 8-F) and are numbered 128 or greater, while data bytes always begin with a 0 (hex 0-7). The status byte determines what type of instruction is described by the message, and is usually followed by one or more data bytes.

S

stave: Also called a *staff*. In notated music, the group of five horizontal lines upon and in-between which notes are written. The lines and spaces represent increasing *pitch* from bottom to top. The precise correspondence between line and pitch is determined by the *clef* which is found at the beginning of every stave. The stave will occasionally have short *leger lines* added above or below when needed for notes that fall outside of the range of the stave. It is common to group staves of simultaneous parts into a *score* by joining them with a vertical line at the left edge of the page and often by linking their bar lines.

steering: See *Dolby Motion Picture 4:2:4*.

stem: Usually used to indicate the inputs to the final mixdown, being comprised of the *DME* tracks. By inference, the many individual tracks which create each of the dialog, music, and effects portions of the soundtrack would be leaves(?), and the final, mixed and edited DME inputs for each soundtrack are the stems.

step input: A method of loading events (such as notes) into a sequencer's memory one event at a time. Common step values are sixteenth- and eighth-notes. After each entry, the *SPP* will advance one step then stop, awaiting new input, as opposed to recording while the clock is running, called *real-time input*. Also called *event editing*, *step recording*, *step-mode*, and *step-time*.

step recording: See *step input*.

step-time: See *step input*.

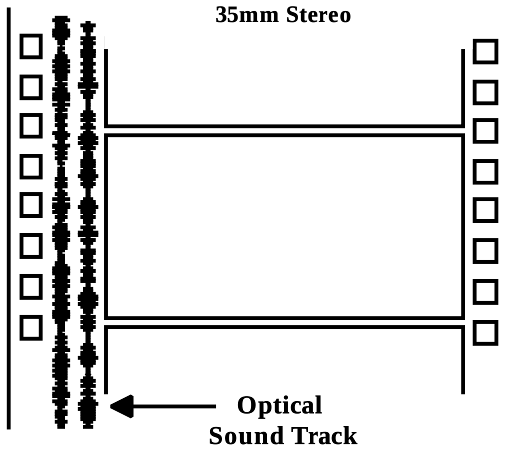
stereo bar: A mounting for a pair of microphones on a single microphone stand.

stereoizing: Audio signal processing to turn a track recorded in *mono* into a realistic *stereo* field. In an analog world, stereoizing is accomplished by remixing the mono track with a copy of itself, and panning the output channels hard left and hard right, with slightly different EQ on each channel to widen the image. In a digital environment, it is typically accomplished by overdubbing the mono channel with a copy of itself, and running each channel through a reverb, each set with a slightly different delay. In the analog case, this results in only a slight improvement in spreading the *stereo image*; with *DSP*, it is possible to use complicated combination of reverb and other *psychoacoustic* effects to get a fairly realistic stereo image. See also *double-tracking*.

stereo link switch: A control on a dual-channel *compressor* or other effects device that sums the *side-chain* inputs together, then controls both channels from the same side-chain. This is to ensure that both channels of a stereo signal are subjected to exactly the same amount of gain reduction or other processing so as not to shift the stereo *image*.

stereo optical print: There are two tracks on 35mm stereo optical prints, referred to as *Lt –Rt*, on which are *matrixed* four channels of information. The 4:2 *encoding* is done during the print mastering, with the 2:4 decoding occurring at playback (at the theater). All of the stereo optical prints: Dolby Stereo (A-Type), Dolby-SR, DTS Stereo, and Ultra-Stereo occupy the same area as standard mono optical prints. The degree of *mono compatibility* is mix-dependent.

S



stereophonic: From the Greek meaning "solid sound," referring to the construction of believable, solid, stable sound images. This has come to mean any sound system with two loudspeakers. However, it precisely refers to a sound system which provides the listener with an illusion of directional realism, regardless of how many *channels* are used. For example, the use of a *reverberation* effect with strong *early reflections* and a fairly fast *decay* are an effective way of creating a wide *stereo spread* from a *mono* source, such as a voice or instrument which has been recorded with a single microphone.

stereo spread: Most two-channel stereo recordings are recorded and mixed to produce an impression of diffusion and spread of the sound between the loudspeakers, where the use of artificial *pan* and reverberation produces a pleasing, spacious sound. However, this is not the same as accurate, multichannel *imaging*.

stereo variable area: See *SVA*.

stereo X-Y pair: See *X-Y pair*.

sticky shed syndrome: A problem affecting some types of analog tape where, after a long time in storage, the binder breaks down, causing the oxide to shed. The tape will then start to adhere to the tape heads and guides when played. The usual short-term cure for this is to bake the tape for several hours at 120˚F (50˚C). The long-term cure is to make a new copy as the tape is replayed.

stinger: In a musical score, a short, accented *chord* played to underscore a specific dramatic event on-screen. See *hits*.

STL: Studio-to-Transmitter Link. A broadcast link used to carry audio / video back to the station from a remote site, while the studio personnel is listening off-air. Problematic because of the transmission delays between source and return. See also *OBU*.

stock arrangement: A commercially published arrangement of a song, not a custom-written piece of music or mix.

S

stomp box: A floor unit *effects processor* with built-in footswitches, meant to be patched between a guitar an amplifier.

stop: A lever or similar control that brings into play a particular *timbre* on organs and harpsichords. See *drawbar*, *registration*.

Stop: A MIDI System *real-time* message which will cause a MIDI device to stop playing the current *song*. Usually only implemented by drum machines and sequencers. See *Start*.

stopband: The frequency *band* which is not passed by a *filter,* as opposed to the *passband*. A filter can have more than one stopband, for example a *bandpass filter*.

stop bit: In *serial* data transmission, the use of an extra bit to follow the data bits which serves to alert the receiving device that the flow of data has stopped. See *start bit*.

straight transfer: See *transfer*.

streaming: Transfer of audio or video data in a way that enables multimedia content to be experienced in *real-time* rather than requiring the entire file to download first.

stress bridge: A small capsule within a *breath controller* that functions as a tiny *potentiometer* and produces a varying voltage that is then converted into a *MIDI* data stream.

stretched: A descriptive term for an *encoded* signal, as in a *NR* system. Sometimes used to describe the signal after subsequent decoding. In film, a stretched *transfer* involved making a new recording of an encoded recording without decoding, and then re-encoding the material. In this way, a stretched transfer retains the original noise reduction encoding level. It is not recommended to do the transfer on a decoded version because any response error is multiplied by the *compression ratio*, typically 2:1, of the noise reduction system.

strike: To take down or disassemble a set used in a motion picture, video, or musical performance.

stringendo: Italian for "gradually becoming faster and with increasing tension."

stripe: (1) (*verb*) To record *timecode* on a blank track of audio tape or video tape or *magnetic film*. (2) (*noun*) A timecode and *video black* written to a video tape before any audio or video information is written. (3) *SMPTE timecode* written to unexposed *mag film* where a *frame* in the *leader* is arbitrarily chosen to be 00:00:00:00. Information about reel number, etc., is contained in the *user bits* data area. If the timecode is recorded before any other signals go on the tape, the procedure is known as *prestriping*. Short for *single-stripe*.

striped: *Magnetic film* with one or more stripes of oxide applied longitudinally on the film base. Typically, a "striped" tape refers to a tape which contains only a timecode track. See *striping*; also *film soundtrack*, *single-stripe*.

striping: (1) The process of recording timecode onto a blank tape track. (2) Recording *sync sound* from the edited *video master* onto an external sync recorder, usually to build music and effects tracks, add narration, loop or post-dub some or all of the dialogue, etc. When all the resulting tracks have been re-mixed, the mix is then *re-layed* or *post-striped* onto the *video master*, hopefully still in sync.

S

striping off: Copying a track from a multitrack master, usually to *single-stripe* 35mm mag film to facilitate editing. See *regroup* and *layback*.

stylus: The needle mounted in a *cartridge* on the tone arm of a turntable. It tracks the groove of a vinyl record and its vibrations are converted, by the *transducer* in the cartridge, into an audio signal.

sub-band coding: A type of the *split-band coding* process, where the incoming audio signal is split into a number of independent frequency sub-*bands*, and the accuracy of the *quantization* is varied in each band according to the input signal energy. Critical regions of the audio spectrum can, therefore, be coded more accurately, with quantizing energy being biased toward the high-sensitivity, low-frequency regions. High-energy regions are also coded more accurately than in *PCM* techniques, yielding a lower coding noise platform. See *transform coding*.

sub bass: Frequencies which are lower than the range of the monitors in use, or below the range that a typical monitor can reproduce with fidelity. Sub base is different from infrasonic in that the former applies to the reproduction of low-frequency audio, while the latter refers to the *AF* range itself. In practice, the two terms are often used interchangeably.

subcode: In digital audio systems, particularly *CD* and *R-DAT*, additional data interleaved with the audio information which carries synchronization and user information, such as tags and comments which is independent from the audio data. The three main types of DAT subcodes are Start IDs which mark the beginning of each song, Program Numbers which are the ordinal numbers assigned to each Start ID, and Skip IDs which causes a player to not play the selection whose number is the current Start ID and go to the next Start ID. See also *PQ Codes*.

subframe: In *SMPTE timecode* synchronization, a unit of time smaller than a *frame*, generally $\frac{1}{100}$th of a frame in length. *Synchronizers* are designed to maintain synchronization to within a small number of subframes. See *frame sync*, *phase sync*.

subharmonic: A *harmonic* lower in frequency than the *fundamental*. Sometimes subharmonics are produced by loudspeakers that have poorly controlled cone resonances. The audible effect is a *distortion* component one octave lower than the input signal frequency.

Sub ID#1, **Sub ID#2**: In MIDI, two data bytes which follow a Universal System-Exclusive data byte in *a system-exclusive* message which define the type of message. A Sub ID#1 value of 1 indicates MTC and bay be followed by a Sub ID#2 value of 1 for a Full Message, or 2 (User Bits), or 15 (Set-Up). A Universal non-real-time Sub ID#1 value of 5, on the other hand, indicates a *sample dump* and may be followed by a Sub ID#2 value of 1 for a Multiple Loop Points message or 2 for a Loop Points Request.

subito: Italian for “suddenly,” as in *subito* ***f***, suddenly loud.

submaster: (1) A control on a *mixer* that controls the level of a group of signals, sometimes called a *group master*. Several submasters may be fed into a master control for final level control of the console output signal. The use of submasters makes it easier to handle a large number of input signals. See *gang*. (2) Any tape used in the making of a master tape, but representing an earlier generation number of some or all of the material included in the final master.

S

submix: The mixture of signals fed into a *submaster* control in a recording or sound reinforcement console. The submix is usually a mix of signals that remains constant over a period of time, and it is convenient to control it as a single signal. Also called *submix bus*, *subgroup*, or *mix group*.

subsonic: Literally, "under sound." Subsonic actually means "slower than sound." The proper term for a signal having frequencies below the human hearing range is *infrasonic*.

subtractive synthesis: The technique of arriving at a desired *timbre* by filtering *waveforms* rich in harmonics. Subtractive synthesis is the type generally used on analog synthesizers. This works well on good analogue synthesizers, but when used on samples, reducing the number of harmonics usually just makes the sound flat and lifeless. Also called *analog synthesis*. See *additive synthesis, sound synthesis*.

subwoofer: A special type of loudspeaker designed to reproduce low frequency signals only, usually a band approximately 10-150Hz.

sum: The addition of several audio signals or variants of the same audio signal. This is normally achieved at the inputs of an amplifier in a *mixer*.

summing amplifier: See *combining amplifier*.

Super 35: A widescreen film format that makes use of the full width of the 35mm film frame, including the area normally occupied by the optical soundtrack. Therefore, there can never be any 35mm *EK neg* prints made from a Super 35 negative, requiring an *interpositive* from the full-aperture original negative to be enlarged to an *anamorphic internegative*, where the resulting *aspect ratio* is 2.4:1.

Super-Bit Mapping™ (Direct™): A type of *dithering* method for encoding *DSD*-format audio information for playback on standard (16-bit) CDs. Developed by Sony, the new version of SBM, SBM Direct accepts a DSD input signal (16-bit *PCM*) and produce an output encoding with a (claimed) 20-bit resolution through a one-pass *noise-shaping / decimating* process which uses a very complex *FIR* reconstruction filter. SBM Direct is being developed by Sony in an effort to make their *SACD* format successful against *DVD-A*, as SACD players will offer playback of conventionally recorded CDs with enhanced audio quality. See also *HDCD*.

supercardioid microphone: A variation on the *cardioid* microphone pick-up pattern which is most sensitive at the front and sides, while rejecting sounds entering 135°-225˚ to the rear. The supercardioid has a *unidirectional* pattern narrower than a cardioid, but wider than a *hypercardioid*. Also called a *cottage loaf mic* in the UK. See *directional microphone*.

Super Clock: Digidesign's proprietary version of *word clock*, used as a *master clock* for all Digidesign interfaces. Super Clock runs at 256 times the *sample rate*; word clock runs at the sample rate.

supervising sound editor: The person responsible for the sound editorial process on a film, including dialog, *Foley*, and sound effects.

supply motor: In a tape transport, the motor that establishes proper tape tension on the supply side of the capstan and drives the fast rewind mode. Also called the *rewind motor*.

supraaural: Literally, “on top of the ear.” This term refers to headsets that rest on the outer ear. They are necessarily light in weight and do not exclude external sounds very well. The opposite of *circumaural.*

surround channel: A single audio channel that feeds multiple speakers, either on the walls of a theater, or in a home surround system. In practice, any format which does not have *spilt surrounds* will be used for *ambient* sound only.

surround-sound: Multichannel sound that is reproduced through three or more loudspeakers above or behind, as well as in front of, the listener. See *5.1*, *7.1*, *Dolby Surround-sound*, *Logic 7*, *AC-3*, *THX*, *transaural audio*.

surround tracks: In 35mm and 70mm projection from *release prints* with magnetic sound, one or two of these tracks is amplified through speakers mounted at the sides, rear or on the ceiling of the auditorium in order to produce *surround-sound*.

sustain: (1) The third of the four segments in an *ADSR* envelope. The sustain portion of the envelope begins when the *attack* and *decay* portions have run their course, and continues until the key is released. The sustain control is used to determine the *level* at which the envelope will remain until the note is released. While the attack, decay, and release controls are *rate controls*, the sustain control is a *level control*. (2) A pedal fitted to pianos and some percussion instruments, such as the vibraphone, that is used to prevent notes from being damped, thus lengthening their decay time, and therefore, not a true (infinite) sustain. A pedal offering a similar function is often available for synthesizers where, of course, an infinite sustain can be achieved on some voices. (3) A MIDI Controller Change message assigned to the parameter in a synthesizer which reproduces the function described in *sustain*(2).

sustain loop: A loop which cycles during the sustain segment of an *envelope* before the rest of the sample plays. See also *release loop, loop*.

sustain pedal: The electronic equivalent of a piano’s damper pedal. In most synthesizers, the sustain pedal latches the envelopes of any currently playing or subsequently played notes at their sustain levels, even if the keys are lifted. Compare with *sostenuto pedal.*

SVA: Stereo Variable Area. A motion picture sound system consisting of two closely spaced optical *variable-area* soundtracks on 35mm film. Dolby improved this format with their *Dolby-A compander* system to improve the *SNR,* as well as employing *Scheiber matrixing* to synthesize a third (center) and fourth (surround) channel for surround-sound applications. Most modern movies with stereo soundtracks employ the Dolby SVA system. See *snake track.*

swarf: See *chip*(1).

sweep: To vary a parameter or signal in *real-time,* such a smoothly varied sine wave from low to high frequency to measure the *frequency response* of a device.

sweep EQ: See *parametric equalizer*.

sweetening session: A recording session in which a previously made recording is augmented by the addition of other instruments. See also *overdub.*

sweet spot: The optimum position for a microphone to be placed in front of an instrument or for a listener to sit, relative to near-field, or other area-focused, monitors.

S

swell pedal: A foot-operated control for synthesizers and electric organs that adjusts volume, hence allowing the sound to "swell." Originally, a mechanical device on a pipe organ with a similar purpose.

Swelltone™: A brand of *dubbing theater* speaker system.

swept filter: A type of filter where the frequency of the filter can be controlled electronically, rather than being manipulated by the user. An example of a swept filter is where a *LFO* could be used to sweep the filter frequency up and down, or an *envelope* could be generated to shape the filter sweep.

switched controller: A type of Controller Change message which is used to induce some kind of change between only two conditions, e.g., *sustain pedal* On or Off. As opposed to a *continuous controller*.

sync: Synchronization. (1) The process of time-locking a number of normally independent and free-running systems. Where one of these is a tape recorder, the term *tape sync* is used: one track of the tape is used to carry a *sync track* which is used to provide timing for the other devices. Two devices are said to be "in sync" when they are locked together with respect to time, so that the events generated by each of them will always fall into predictable time relationships. See also *SMPTE timecode*, *crystal sync*, *sel-sync*, *jam sync*. (2) The operating mode of tape recorders that uses the *sel-sync* process of playback. (3) In a synthesizer with two or more oscillators, the ability to lock the frequency of the *slave*, to a *master*. This has the effect of eliminating the *beating* which can occur if one is slightly out of tune with respect to the other. If the slave's frequency is adjusted higher than the master, there will be no actual change in frequency, but an alteration in the *harmonics*, giving the composite sound a different *timbre*. This is most effective if the frequency of the slave is modulated constantly, perhaps with an *envelope generator*, to produce a distinctive kind of wailing sound.

sync block: A device used in editing to keep film and sprocketed soundtracks in sync. It can have two or more gangs of sprocketed wheels that rotate in unison and a counter that displays a *feet/ frame* count from a selected starting point. Also called a *synchronizer*.

sync head: See *sel-sync*(2).

synchronization: (1) The precise alignment of picture and sound during projection and playback such that visual and audio events will be reproduced with the same time relationships they had when originally captured to tape. (2) The process by which this is achieved.

synchronizer: A separate device that reads *timecodes* recorded on two or more devices, compares the timecodes, and adjusts the machines' read / replay / write positions and speeds based on the results of that comparison.

Synclavier: The trade name of one of the earliest commercially available computer music instruments, implementing a type of *FM synthesis*. Sequencing and *monophonic* sampling facilities (at a sampling rate of 50kHz) were soon added. The sampler was expanded into a full hard-disk recording unit which coined the name *Direct-to-Disk*. Towards the end of its development, a Synclavier was an integrated system with ninety-six *voices*, sequencing, and multitrack hard-disk recording, aimed at post-production and other professional users.

S

sync-lock: Sync-lock, also called *phase-lock*, emulates the old *control track* or *pilot tone* method of synchronization. The system reads the timecodes, synchronizes the transports, and takes any deliberate offsets into account. Once the system is locked, the *slaves* only use the speed information that is derived from the timecode. Specific timecode addresses are ignored. This method allows the tape machines to stay locked even if the timecode relationships change. See *chase-lock*.

sync mark: A mark that the editor places on the head *leaders* for the *workprint* and each track of edited *magnetic film*. These generally put all tracks in *editorial sync* with one another so that they will maintain synchronization when reproduced on *dubbers* equipped with synchronous motors. The editor may also place a similar set of markings on the tail leaders of all reels of picture and magnetic film, to check that there has been no slippage in projection or playback.

sync master: See *sync reference*.

sync mode: See *sel-sync*(1).

syncopation: In music, the placing of accents on normally unaccented parts of the bar. The effect is usually to anticipate the main beats. See *upbeat*, *downbeat*.

syncope: A term used to refer to the unnecessary proliferation of tied notes that can occur when a notation module in a sequencer program reflects every rhythmic nuance of the performance data. The problem can often be ameliorated by an option which quantizes only the appearance of the notated music.

sync pop: A single film frame of 1kHz used a guide to synchronize the sound and picture. The pop on the track negative creates a visual guide for the negative cutter who uses it to make a printing start mark. The pop occurs two seconds before the first frame of the picture, and thus corresponds to the “2” frame on the SMPTE Universal leader. On standard film *leader*, the number at the pop is “3” because it counts down in actual film footage.

sync pulse: (1) The output of a *clock* used to keep synchronous devices, such as tape and video recorders, in synchronization. (2) The signal recorded by the *sync head* of a *Nagra*, derived from the camera motor through a sync cable, or from the Nagra’s own *crystal sync* generator. See *FM sync*.

sync reference: The *sync pulse* output by the sync master device, used by the *synchronizer* as the reference signal for all *interlocked* devices in the recording and / or playback *chain*. The sync master may come from *house sync* or some other source which generates the *master clock*.

sync sound: A soundtrack recorded for motion picture where the audio elements are recorded synchronously with the picture. The opposite of *wild sound*.

sync tone: (1) In electronic drum machines and sequencers, a *bi-phase modulated square wave* generated in the machine. This can be recorded onto tape, and when played back into the machine, will sync its signal output with the tape. Analogous to *MTC* and *SMPTE timecode*, but differing in the frequencies and word lengths that various manufacturers use in their machines. (2) The 60Hz *pilot tone* that motion picture cameras send to the *Nagra* or other sync recorder, and that controls the playback speed during transfer so that the *magnetic film* will sync with the *workprint*.

S

sync track: A *sync tone* reference signal recorded onto magnetic tape or *magnetic film*.

sync word: At the end of every 80-bit *SMPTE timecode* word is a 16-bit sync word. The sync word provides direction and *sync-lock* speed information, and marks the end of each timecode word. The bits are: 0011 1111 1111 1101.

synthesis: The process of electronically creating sounds on a synthesizer. The word synthesis has the implication of combining separate elements into a new whole, and is therefore, not necessarily to denote "synthetic" as in artificial. See *sound synthesis*.

synthesizer: A musical instrument that generates sound electronically and is designed according to certain principles developed by Robert Moog and others in the 1960s. A synthesizer is distinguished from an electronic piano or electronic organ by the fact that its sounds can be programmed by the user, and from a *sampler* by the fact that the sampler allows the user to make digital recordings of external sound sources. Synthesizers, particularly if provided with *MIDI* software and hardware, need not have a keyboard, and most need to be connected to amplification equipment if they are to be heard. See also *master keyboard, tone module*.

syntonic comma: The error arising in any *just intonation*, due to the fact that the *octave* is incompatible with the simple frequency ratios of the *intervals* of the *diatonic scale*. If f is the frequency of the *tonic* C, the first *sixth* produces A, with a frequency of $\frac{5}{3}f$. In going to D, an interval of a *fourth* is required, and this is a frequency ratio of $\frac{4}{3}$, so its frequency will be $\frac{4}{3}$ of A, which is $\frac{5}{3}$ of f, or $\frac{4}{3}\times\frac{5}{3}f$, which is $\frac{20}{3}f$. The descending *fifth* gives G, at a frequency of$\frac{2}{3}\times\frac{20}{9}f = \frac{40}{27}f$. The last fifth results in the tonic, C, with a frquency of $\frac{2}{3}\times\frac{40}{27}f = \frac{80}{81}f$. This discrepancy is called the syntonic comma, and is equal to about one-fourth of a *half-step*. It results in the fact that, after the above simple five-*chord* progression, the tonic is no longer at the same frequency at which it started. Intervals of major *thirds* are not commensurate with a perfect fifth, the difference being the syntonic comma. The following integer equation must, therefore, be false for all integers:

$$\frac{X^n}{Y}f = 2^m f$$

where X, Y, n, and m are integers, and f is the frequency of the tonic. The left-hand side represents successive steps of musical intervals, and the right-hand side represents *octave* transpositions. It can be shown that this equation can never be satisfied. See also the *diatonic comma*.

System-Common: A type of MIDI data used to control certain aspects of the operation of the entire MIDI system. System-Common messages include *Song Position Pointer* (SPP), Song Select, Tune Request, and End-of-System-Exclusive (*EOX*).

System-Exclusive (sys-ex or SysEx): System-Exclusive is designed to be an open-ended part of the MIDI specification, defined to enable the transmission of the entire memory contents of a synthesizer over a MIDI cable. This transmission is often called a *bulk dump*. Sys-ex commands are strings of numbers, usually written in hexadecimal format, that carry information to a MIDI device, specific to the manufacturer and particular model. In other words, that device, exclusively, will respond to the commands, and all other MIDI devices ignore them. Sys-ex data is used most commonly for sending patch *parameter* data to and from an *editor/librarian* program, but other types of data such as the contents of tuning tables are transferred in this way as well. See *SDS, SMDI*.

S

system message: A general term for MIDI messages with status byte values of 240 and above, intended for all devices on a *MIDI network*, irrespective of any *channel* assignments. It includes System-Common, System-Exclusive, and System Real-Time message types. See *MIDI*.

System Real-Time: A type of MIDI data that is used for timing reference. Because of its timing-critical nature, a System Real-Time byte can be inserted into the middle of any multi-byte MIDI message. System Real-Time messages include *MIDI Clock*, *Start*, *Stop*, *Continue*, *Active Sensing*, and *System Reset*.

System Reset: A System Real-Time message which causes a MIDI device to reset to its default condition, i.e., its settings when first powered-up. This will usually involve silencing all voices, resetting the display to the opening page, etc.

T

tablature: Pictograms which represent fingering positions on a string fretboard.

tach: See *tach pulse*.

tach pulse: A signal generated by the *tachometer roller* of an audio or video transport, one or more times per rotation. Because the tach roller is in contact with the tape in fast-wind modes as well as play or record mode, it can be used to get approximate tape location data when *SMPTE timecode* data cannot be read. Tach pulses sent by various decks to the *synchronizer* allow it to stop each deck near the SMPTE timecode designated by the engineer. Once in play or record modes, the decks will again interlock via the SMPTE timecode data. Tach pulses do not include location information, only speed and direction. Tach pulse is the mechanical, or analog, equivalent of *word clock*. See also *bi-phase/tach*.

tail: (1) Additional information which follows a block of data either on disk or for the purpose of data transfer between devices. The tail usually serves as a full stop to the data that precedes it. Commonly encountered in MIDI System-Exclusive transfers. For example, in Yamaha System-Exclusive transfers, the last two bytes are tail information in the form of an ECC and an EOX status byte. (2) The end of a reel of tape or film.

tail-out: See *spiral*.

tails-out: See *heads-out*.

take: A term meaning a single, continuous recording. This may be of a complete work, but is more often a short section. The *EDL* is comprised of selected takes in their final order.

take sheet: A sheet of paper on which the engineer makes notes about each take as it is recorded, such as complete or incomplete, good or n.g., which sections of the take are usable, etc.

take-up motor: On a tape recorder, the motor that applies take-up tension to the tape. This motor also powers the fast-forward mode.

take-up reel: The reel onto which tape is wound after it passes from the *feed reel* over the audio or video *heads*.

talkback: A facility on a *mixer* which sends signal from the control room to the recording area, allowing an engineer or other studio personnel to talk over headphones or a loudspeaker to give instructions, identify takes, etc. A microphone on the console is normally routed directly into the studio amp and monitors, while a relay mutes or attenuates the volume of the control room monitors to prevent *feedback*.

tangency: The parameter of tape-head alignment that determines the geometric relationship between the plane passing through the head gap and the tape passing over the gap. Ideally this plane would be exactly perpendicular to the plane that is tangential to the tape passing the gap. Misadjustment of tangency will cause uneven head wear on both sides of the gap.

tap: In a *digital delay*, a point in the circuit after one or more delay stages at which the delayed input signal can be patched-out and routed to another destination.

T

tape: (1) (*noun*) Recording medium consisting of a magnetic coating applied to a plastic substrate. See *magnetic recording tape*. (2) (*verb*) To record music or other program material, whether or not the recording is actually written to magnetic tape, for storage, editing, and/or playback. (3) Another name for the tape recorder operating mode normally called *repro*.

tape bias: See *bias*.

tape counter: A mechanical or electronic display of the relative amount of tape that has passed through the transport system of a tape recorder. In professional systems using some kind of *timecode*, it may also show the absolute time value recorded at any given point on the tape.

tape delay: The original *delay lines* were made by using a three-head tape recorder to record a signal while playing it back on the same machine. The distance between the record and replay heads causes a time delay which varies with tape speed; this technique is called tape delay. If some of the replay signal was mixed with the direct signal, a pseudo-*reverb* could be created. The record-replay idea was further developed with specific machines which used tape loops and multiple replay heads and the ability to adjust the contribution and feedback of each head. These machines were replaced with *DDL*s.

tape echo: An early method of producing echo effects by means of a *tape loop*. See *loop*(2).

tape head: The *transducer* used in a *magnetic recording tape* machine to create patterns in the magnetic surface of the tape during the recording process, or conversely, to read the patterns on playback. See *magnetic recording tape*.

tape hiss: Noise that is characteristic of analog *magnetic recording tape*, produced by the random fluctuations in the positioning of magnetic particles along the tape, and heard as a low-level *hiss* during playback. The noise, although *broadband*, is most noticeable in the high frequencies. See *Barkhausen effect*, *noise reduction*, *grain*(2).

tape loop: See *loop*(2).

tape pack: The smoothness with which magnetic recording tape is wound onto the hub of a reel, or simply, the *pack*.

tape returns: Mixer inputs that typically come from a *multitrack* tape recorder. They are different from common line inputs in that they are typically switchable between *monitor* and *mixdown* functions, depending on whether or not the recorder is recording, or the mixer is mixing.

tape speed: Professional equipment records at standard speeds of 30ips (76cps), 15ips (38cps), or 7 ½ ips (19cps). Consumer recorders run at 1⅞ ips (4.75cps). Faster tape speeds mean higher-quality recording as the signal has more tape area onto which to be recorded.

tape splicer: See *Guillotine splicer*.

tapeless studio: A recording set-up which uses exclusively digital recording, storage, and playback equipment.

T

tape sync: A method for synchronizing a pair or group of independent devices which uses one track of a tape recorder attached to the other device(s) on which is striped the *master* timecode signal which provides timing for the other device(s). See *FSK*, *pilot tone*, *SMPTE sync track*, *sync track*.

tape type: Categories for recording tape are based on the magnetic coating of the tape. Usually applied to cassette tapes, they include: Type I (ferric oxide), Type II (chromium dioxide), and the now-discontinued Type III (metal).

tape weave: An improper gradual up and down motion of magnetic tape as it passes over the tape heads in a tape recorder, causing *tape skew*. It is usually caused by the tape not having been properly slit, but can also result from worn tape guides.

Tartini tone: See *difference tone*.

taskbar: The menu on a Windows™ system which displays open applications.

TCA: Timecode Address. The HH:MM:SS:FF bits of the *SMPTE timecode* word.

TDF: Triangular probability Density Function. See *dither*.

TDIF: TASCAM Digital Interface Format: An eight-channel digital interface used to connect TASCAM *MDM*s to one another and to compatible external gear. The TDIF format uses *master clock* sync and carries eight channels of digital audio on a electrical cable with 25-pin *D-sub* connectors. Each wire in the cable carries two multiplexed channels, which closely resembles AES/EBU. The entire cable can handle eight channels to and from any compatible device, and so is bi-directional. The maximum *bit depth* is 24 bits, and the data rate is four times the sample rate. See also *Lightpipe*.

TDM: Time-Division Multiplexing. Digidesign's proprietary 24-bit *DSP* environment, providing *real-time* digital audio processing and mixing on ProTools™ hardware. Many third-party manufacturers make software *plug-ins* to add audio functions to TDM-based-systems. TDM itself refers to the division of each sample period into 256 different addresses, each available to a plug-in.

Telecine: (1) A machine that transfers film to video signal for broadcast or storage onto videotape. Also generically refers to the process of film-to-videotape transfers. It consists of a film projector with a special optical system that connects to a television camera which records the projected film image. Better quality than a video-video dub, but the Telecine transfer must happen in real-time. Telecine occurs at three points in the filmmaking process: first, when the film is transferred to video in preparation for editing on a nonlinear system; when an edited *workprint* is transferred to video to give sound editors a guide with which to edit sound; and, when an *interpositive* is transferred to a videotape to create a master for home video release. (2) The UK name for a film *chain*.

Telharmonium: Also called the Dynamophone. An early electric keyboard invented around 1900, weighting over 200 tons, it was the feature of the world's first music broadcasting service in New York in 1906. The Telharmonium used the *tone-wheel* principle and so was the predecessor of the later, smaller Hammond organ.

temp dub: A quick and temporary mix of a film soundtrack made during post-production for screening and evaluation in *double-system*.

T

temperament: In the tuning of a musical instrument to a *scale*, temperament is the compromise, or deliberate mistuning, of pure or *just* intervals so the various frequency ratios between notes of the scale are compatible with *octaves*. This compromise is called a temperament, of which there are theoretically an infinite number. Also called *tempered tuning*. See *meantone, equal temperament, just intonation, syntonic comma, diatonic comma.*

temp score: Music placed by the director or music editor to get an impression of how a scene will work once it's *scored*.

tempo: The speed of the pulse, or *beat*, of the music. See *rhythm*.

tempo-dependent: A clock, such as the *MIDI clock*, which is dependent on the tempo of the music for tracking, i.e., the MIDI clock will transmit more MIDI clocks per second if the master *sequencer* increases the tempo, as compared with other synchronization signals which encode information about absolute time, such as *SMPTE timecode.*

tempo map: Data containing the initial tempo of a composition and the *SMPTE timecode* location of the song start, plus the degree and location of any subsequent tempo changes are stored in a tempo map. Usually the tempo map is built by the sequencer and stored along with sequence data.

temporal masking: A data reduction technique which takes advantage of the fact that a loud sound affects the perception of quieter signals both before and after it. If, for example, a relatively quiet signal occurs 10-20ms before a louder one, it may still be masked by the louder signal. This is called *backwards masking*. The hearing mechanism also takes time to recover from a relatively louder sound, and this creates a masking effect which extends up to 100-200ms after the masking signal has ceased, called *forward masking.* The length of the masking is related to the relative amplitude of the masking signal. To reduce data by using this technique, the input signal is divided up into blocks of samples usually around 10ms in length, and each block is analyzed for transients which act as temporal maskers. Most systems vary the length of the block to take advantage of both backwards and forwards masking. Also called *time-masking.* See *perceptual coding*.

tenor: From *tenere,* the Latin "to hold." Originally, a vocal part with sustained notes in sacred music. Now used to refer to a male voice which has a range from about C-below-middle-C upwards about two octaves, and by extension, instruments which have a similar *tessitura*. Music written for the tenor register is notated with a tenor *clef*, which has middle-C on the fourth line of the stave. The term *counter tenor* is used to specify a male voice singing in the alto range. The other registers are *alto, baritone*, *bass*, and *soprano*. See also *treble* and *voice.*

tent: An area of a recording studio enclosed in absorbtive panels, constructed to provide a *dead* acoustic space.

tenuto: A notation indicating that a note should be held for longer than its nominal value, shown by a short, heavy line on the note.

terminal: The point at which the signal either enters or leaves an audio device, including wire nuts, and by extension, other audio-type connectors.

terminal strip: A series of connections, usually screw terminals, arranged in a line to permanently connect multiple audio lines to such devices as recording or broadcast consoles.

T

termination: A transmission line is said to be terminated if it is connected to an *impedance* equal to its characteristic impedance, i.e., the end of the line is connected to a load that matches the impedance of the line. Microphones should not be terminated because the termination load reduces the signal voltage level. Also misused to indicate the proper *resistance* to which an audio device is intended to be connected. See *impedance-matching.*

tessitura: The range of “comfortable” sounds of an instrument.

test tape: See *biased noise*.

THD: Total Harmonic Distortion. An audio measurement specification used to determine the accuracy with which a device can reproduce an input signal at its output. THD describes the cumulative level of the *harmonic overtones* that the device being tested adds to an input *sine wave*. THD+n is a specification that includes both *harmonic distortion* of the sine wave and nonharmonic *noise*.

theme: A musical idea or phrase that becomes an important motif throughout a *score*.

Thérémin: An early electronic instrument which used *heterodyning* by one fixed RF oscillator which was modulated by a hand and an antenna. Audio-range frequencies were produced by the sum and difference signals from this modulation, making the Thérémin a predecessor to modern *FM synthesis*. The instrument was *monophonic* and almost uncontrollably *microtonal.*

thermal capacity: An unstandardized performance specification for an amplifier which describes its *thermal headroom.* This is not particularly meaningful as amplifier thermal performance varies with the input test signal. Also used as a synonym for thermal headroom.

thermal equilibrium: The length of time it takes electronic circuitry to reach its steady-state operating temperature. For example, it takes about ten minutes for most audio equipment to reach thermal equilibrium, and all components are stabilized as far as voltage, current and temperature swings which occur at start-up having been attenuated.

thermal headroom: A term which denotes the difference between the nominal operating temperature of a power amplifier and the maximum temperature at which the amplifier will continue operating, i.e., before its thermal protection circuitry will shut it down.

thermal noise: See *quiescent noise floor*.

third: The *interval* between a note and the one three *scale* steps above or below it: three *half-steps* (minor third) or four half-steps (major third).

third harmonic distortion: That part of *harmonic distortion* which represents only the third *harmonic* (three times the *fundamental* frequency) of a *sine wave* input to an electronic device. The third harmonic of any tone is musically an *octave* and a *fifth* above the original tone, and is easily noticeable in the output. For this reason, the *MOL* of analog tape recorders, for example, is often specified as that level at which third harmonic distortion reaches 3%.

third-octave: Short for one-third of an *octave.* The usual *bandwidth* on the individual faders on a *graphic equalizer* and graphic displays on most *RTA*s. Also, the Q on *narrowband filters* is usually one-third octave.

three-head: A tape recorder with separate record, replay, and erase heads.

T

three-stripe: See *3-track.*

three-track mix: See *3-track.*

threshold: The specific point in a range at which some process or effect occurs. In an audio signal, it is the lower limit in the dynamic range above which a device, such as a *compressor*, *limiter,* or *noise gate,* will affect the incoming signal. See also *expander*, *gain-before-threshold*.

threshold of hearing: The lowest-level sound detectable by a listener with good hearing. Defined as 0dB SPL and usually specified for a 3kHz tone since the human ear is most sensitive in this frequency range. See *loudness*, *SPL*, *equal loudness curves*.

throat: The opening at which a *driver* attaches to a *horn*. Or, the narrow end of the horn itself.

thumper: A low-frequency sine wave at around 30Hz, triggered by a *noise gate* keyed to a *click track.* A thumper is used to give filmed dancers the beat of a song while recording *sync sound*; the thumper is subsequently filtered out.

THX™: A trademark for a special motion picture *surround-sound* specification developed by Lucasfilm in collaboration with audio engineer Tomlinson Holman. The acronym comes from both the name of George Lucas' first feature film, THX-1138 and also from the initials of Tomlinson Holman eXperiment. Only the loudspeaker crossover network is manufactured by Lucasfilm; other components of the B-chain, including their placement and installation must be approved before the theater can be designated as "THX approved." THX is a theater sound system specification, and as such does not involve any type of film audio format, as does Dolby Digital™, nor does it indicate that any of the film sound was produced by Lucasfilm Ltd.

tie line: A permanent, but undedicated, connection between two points some distance apart, e.g., between a terminal box in the studio and the *patchbay* in the control room.

tie-tack microphone: See *Lavalier microphone.*

tight: (1) Low-frequency performance of a loudspeaker when relatively free of *hangover*. Poorly damped systems are said to sound *loose.* (2) A recording made using *close miking.* Also called *tight miking*.

timbral interference cues: The same as *comb filtering.*

timbre: (1) The perceived quality of a sound, unrelated to *pitch* or *amplitude.* Timbre is determined by how many *harmonics* are present in the sound, their frequencies, and how loud each is in relation to the *fundamental frequency*, and how the relative amplitudes of those harmonics change over time. The initial and ending *transients* of a tone have more to do with the timbre of a tone than its harmonic content. Also referred to as *tone color.* (2) One of the building blocks of a *patch* in a Roland synthesizer.

timbre envelope: The change in the tone of an instrument over time.

timbre modulation: A technique whereby a sample *loop point* length or position is changed via modulation, such as sweeping through a *wavetable* to produce an extreme stuttering effect.

T

time alignment: In a multiple-driver loudspeaker system, it is important that the time delay of each driver and its associated *crossover network* be the same to preserve accurate *transient* response, i.e., the high frequencies and low frequencies reach the listener's ear at the same time. Crossover networks have different delays depending on the frequency range they cover, as do speakers; woofers have more delay than tweeters. One can place tweeters farther from the listener, or build time-delay into the high-frequency signal path. *See transient response*.

time-base corrector (TBC): An electronic device that repairs *dropouts* and other defects in the synchronizing signals of a video picture, then acts as an outside *sync reference* to which playback machines can conform.

timecode: A type of non-audio signal that contains information about elapsed time on a film, tape, or disk recording. Used for a synchronization reference when synchronizing two or more machines such as sequencers, drum machines, and tape decks. See *SMPTE timecode, MTC, FSK*.

timecode generator: An electronic device that produces *SMPTE timecode* signals, which can be used to synchronize the *frame rate* of motion picture cameras and recorders, television cameras, VTRs, VCRs, etc.

timecode regeneration: The process of creating a new timecode based on an incoming timecode signal or *positional reference*. See *dropout, freewheeling*.

time-coherent: A device is said to be time-coherent if it exhibits an essentially linear *phase-shift* over frequency, i.e., the characteristics of a pure delay. It is not necessary that the frequency response be flat, just that the phase-shift vary linearly with frequency. Apparently, humans can hear time differences in the 5µs-10µs range. A lack of time-coherency is usually problematic in multi-way speaker systems and some microphones. The *crossover* point in multi-way loudspeakers, without careful control and compensation, causes the speaker to output a multi-lobed pattern, i.e., the main energy output is not in front of the speaker, and the *polar pattern* is asymmetric. In a microphone, a perfect characteristic *impulse response* would be where the *diaphragm* would cease moving as soon as the sound ended. However, bodies in motion tend to remain in motion; in addition, all of the microphone resonates, not only the diaphragm. All of this additional motion contributes to the coloration of the sound by smearing the various arrival times of the various frequencies. By using a very small, lightweight diaphragm housed in a capsule that has a minimum of resonance and reflective surfaces, coloration is reduced.

time masking: See *temporal masking.*

time signature: The *meter* of a work is indicated at the beginning by two numbers, one above the other: the lower indicates the chosen unit of measurement (half-note, quarter-note, etc.), while the upper indicates the number of such units which make up a measure. In *simple* times, such as $\frac{2}{4}$, $\frac{3}{4}$, and $\frac{4}{4}$, each *beat* is normally halved to form eighth-notes, etc. In *compound* times, such as $\frac{6}{8}$, $\frac{9}{8}$, or $\frac{12}{8}$, the basic beat is grouped into threes to form dotted notes. In faster tempos, musicians can regard these dotted notes as the actual beat. The time signature, 𝄴, is called *common time* and indicates $\frac{4}{4}$ time, while 𝄵, *cut time*, indicates the generally faster $\frac{2}{2}$ time.

T

time-stretch: To change the length of a sample without altering the pitch. Compare with *pitch-shift.*

timing clock: See *MIDI clock.*

tip, ring, and sleeve: See *TRS*.

TLA: (1) Three Letter Acronym. From NASA, a long time ago. Modern usage is somewhat disparaging, as the world is now speaking exclusively in TLAs. (2) Transformer-Like Amplifier. A design for microphone *preamplifier* design by Rupert Neve which uses *active* components to simulate the operational characteristics of conventional transformer coupling, but with lower crosstalk.

TOC: Table Of Contents. The inner track of a CD, containing information about the disc such as number of tracks, their position on the disc, timing, and disc ID number.

Todd-AO: Todd-American Optical. The 70mm widescreen process developed jointly by Mike Todd and the American Optical Company. Also the name of a film sound company in Hollywood.

tonal: Music that is written with a primary pitch, i.e., is written in a specific *key*, such as C, is called tonal. See *tonic.*

tone: (1) Refers to a signal which has a particular and usually steady *pitch.* The simplest of tones are *sine waves*. (2) Tone also refers to *timbre,* or quality of a musical sound, such as “bright,” “mellow,” etc. (3) A synonym for *whole-step*, an *interval* of two *half-steps*, e.g., from C to D. (4) A synonym for note, i.e., any particular sound.

tone color: See *timbre*.

tone control: One or more knobs on an audio amplifier or preamp which modify the relative balance between the treble and bass tones. Tone controls are actually *equalizers* and they change the *frequency response* curve of the device implementing them.

tone module: See *expander*(2).

tonic: The reference (lowest) pitch on which a musical *scale* is built, i.e., the “do” of do-re-mi. It is defined in terms of the musical note rather than in terms of absolute frequency. For example, if one changes the *key* of a musical selection from C to A♭ this is a change of the tonic from the note C to the note A♭.

top hat: A plastic disc and a handle designed for placing over a tape *pancake,* taking the place of the upper reel *flange* to allow smooth winding of the tape. Also called a *hat.*

topping and tailing: The process of removing extraneous sound at the beginning and end of a *song,* any fades are added or adjusted, and running times are calculated for a master tape or digital copy.

top sheet: See *binky*.

Toslink: See *S/PDIF*.

T

touch-sensitive: Equipped with a sensing mechanism that responds to variations in *key velocity* or *key pressure* by sending out a corresponding control signal. See *aftertouch*.

track: (1) (*verb*) To be controlled by or follow in some proportional relationship, as when a filter's *rolloff frequency* tracks the keyboard, moving up or down depending on what note is played. (2) (*noun*) One of a number of independent memory areas in a sequencer. By analogy with tape tracks, sequencer tracks are normally longitudinal with respect to time and play back in sync with other tracks. See *channel*. (3) One of the longitudinal areas of a magnetic tape which is formed by recording an audio signal along it, separated by *guardbands*. (4) A section on a recording, containing a discrete song or other selection.

track-at-once: A *CD* production method whereby one or more tracks is burnt at a time; a link is written between the tracks. This method is often used to create *multisession* CDs. A disadvantage of track-at-once recording is that gaps between tracks must be at least two seconds in length. Compare with *disk-at-once.*

tracking: (1) The process of following the *envelope* of a signal's waveform, as in a level meter of any type. (2) In describing the performance of a phonograph stylus or other *transducer*, the measure of it s ability to precisely follow the instantaneous *waveform* of the applied signal, be it mechanical, e.g., a record groove, acoustic, or electrical. (3) A synonym for *over-dubbing*. (4) See *double-tracking*(1).

tracking error: (1) Because the tonearm of a record player is pivoted at the stationary end, the stylus moves across the record in an arc and meets the groove at an ever-changing angle. The cutting stylus, however, moves in a straight line, always 90˚ to the groove direction. This condition is called tracking error. (2) An error which occurs in disk drives when the read / write head cannot be placed accurately over the data blocks due to a misalignment in the head *servo*, a defect in the recording medium, or a misalignment in the recording process. Also, a similar error in CD player mechanisms.

track laying: (1) The process of recording audio signals to a track or tracks, either simultaneously or sequentially, prior to *mixdown*. (2) The editing and assembly of the various tracks of *magnetic film* containing dialogue, narration, sound effects, music, etc., in preparation for re-recording, mixing, or dubbing. Also called *sound cutting*.

track negative: Film terminology for a soundtrack negative.

track select: The process of enabling specific tape machine tracks for recording.

transaural audio: The use of *psychoacoustics* to give the listener the illusion of sound coming from all around, even though there are only two loudspeakers. The listener must be positioned on the center line of the two speakers to correctly perceive the *phantom images*. An example is the QSound™ system. Also called *psychoacoustic surround-sound, or fifty-yard line surround*.

transducer: A device which converts mechanical, magnetic, or acoustic energy into electrical energy, or vice versa. See *DI, electroacoustic, electroacoustical transducer*.

T

transfer: (1) The same as *regroup*. The process of copying audio and / or video from one medium to another, possibly combining audio and video or audio and audio (such as from a sampler and sequencer to multitrack tape). For example, the final audio mix for a film is transferred (typically these days) from a digital workstation's hard disk and written to the audio tracks on the *print master*, which is then copied for distribution. Implicit in the term transfer is a change of medium. Transfer from one like device to another, such as cassette duplication, is called *copying* or *dubbing*. (2) The *magnetic film* itself that results from this copying process. A *straight transfer* is made without any equalization or compression of the audio signal as it goes from the source to the magnetic film.

transfer channel: See *channel*.

transfer characteristic: The graph of the output amplitude of an audio device vs. the input amplitude. If the system were perfect, the transfer curve would be a straight line and the slope of that line would indicate the system's *gain* or loss. Also called the system's *transfer curve*. For example, see the diagram under *rotation point*.

transfer curve: See *X-curve, transfer characteristic*.

transfer suite: A facility where film *location sound* recordings are transferred to sprocketed magnetic tape. The soundtrack and the film can then be mechanically synchronized. See *Telecine*.

transform coding: Used in Dolby *AC-2* and *MPEG codecs*, whereby an input signal is analyzed within the frequency domain as a series of narrow *bands*, e.g., an AC-2 encoder / decoder uses 256 narrow bands. Frequency-masked bands whose information is inaudible because of sustained tones or *transients* will be ignored by such systems, thereby enabling the compression algorithm to concentrate its bit allocation on those bands which contain subjectively relevant information. In addition, transform-based systems use a system of predefined waveform patterns from an established wavetable of sound models and send only the identity code to the decoder to resynthesize the closest-fit library model. As opposed to, for example, *ADPCM*.

transformer: A device consisting of two or more coils of wire wound on a common core of soft iron or other magnetically permeable material. The number of turns in one coil divided by the number of turns in the other one is called the *turns ratio*. An alternating voltage across one coil will appear across the other coil multiplied by the turns ratio.

transformerless input: On an audio device, an input which does not use a transformer for *impedance-matching*. Such inputs will make use of semiconductors such as transistors, and may be described as *active* inputs. Their main advantage is a saving in size, although some argue that they give better performance than transformers, the truth of the assertion probably depending most on the quality of the device, rather than its type.

transient: A non-*periodic* sound *waveform* or electrical signal. Any of the non-sustaining, non-periodic frequency components of a sound, usually of brief duration and higher amplitude than the sustaining components and contribute most to a sound's *timbre*. Transients usually occur near the onset of the sound such as the *attacks* of musical instruments, percussive sound, or speech consonants, which are called *attack transients*.

T

transient distortion: This includes *transient intermodulation distortion* (TIM). The beginning and ending *transients* of musical sounds are largely what determine their *timbre*, rather than their *harmonic* content. An audio device which passes steady-state signals perfectly well may distort the loud, high-frequency transients, causing audible coloration of the music. Low transient distortion means a device must have a wide, linear *frequency response*, no *phase distortion*, and no *hangover*. TIM is caused by amplifier *slew-limiting* and is principally a problem in *solid-state* amplifiers that use large amounts of negative *feedback*.

transient generator: See *envelope generator*.

transient response: (1) The instantaneous change in an electronic circuit's output response when input circuit conditions suddenly change from one steady-state condition to another. (2) Accurate transient response is the reproduction of sound such that high-frequency and low-frequency sounds reach the listener's ear at the same time. See *time alignment*.

transistor: A type of *solid-state* device generally having three terminals called emitter, collector, and base, which has the property of letting current flow in one direction only, under the control of a biasing voltage present at the base. It is really an electronically operated switch, principally used where a gain increase is required, such as in amplifiers.

transmitter: A device for converting an audio and/or video signal into a modulated *carrier* wave which can be radiated by an antenna. By extension, any device generating output which is sent electronically to another location.

transondent: Possessing the ability to freely transmit sound, analogous to transparent in reference to light, e.g., the grill cloth on a speaker must be transondent.

transport: The mechanical portions of a tape recorder, including all parts that handle and guide the tape from the *feed reel*, past the *head stack*, and onto the *take-up reel*.

transport control: The buttons on a tape recorder, i.e., Play, Rewind, FF, Stop, etc. which control the movement of the tape within the unit. These controls may be duplicated on a remote unit. More elaborate controls may include an *autolocator*.

transposition: See *pitch-shift*.

trap: A *band-reject* filter designed to eliminate a particular frequency from a desired signal, e.g., a *bass trap*.

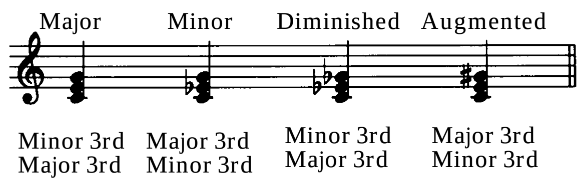
treble: (1) The upper end of the audio spectrum, usually given to be about 2kHz-20kHz. (2) A high voice or musical part equivalent to *soprano*, particularly a part sung by a boy. By extension, instruments which operate in a similar range may be prefixed by the term, e.g., treble recorder. The treble, or G, *clef* is the one with middle-C one *leger line* below the *stave*. See also *alto*, *baritone*, *bass*, and *tenor*.

tre corde: Italian for "three strings." Usually used in piano music to indicate the release of the *soft pedal*. This has the effect of returning the action to its normal position so that all of the strings available are struck. As opposed to *una corda*.

T

tremolo: (1) A *periodic* change in *amplitude*, usually controlled by an *LFO*, with a periodicity of less than 20Hz. Compare with *vibrato*. (2) A MIDI Controller Change message which is assigned to the parameter in a *synthesizer* which alters the depth of the effect described in tremolo. More recently this message has been reassigned as one of the five generalized Effects Depth messages. See *effects control*. (3) Italian for “trembling.” A rapid reiteration of one or more notes, especially on string instruments.

triad: A three-note *chord* that is formed from any note of the *scale*, plus the note a *third* above it and the note a third above that. The lowest note is called the *root* (or root position), and the letter name of the root is used to identify the entire triad, e.g., the triad of C. In practice, the notes may be arranged in any order, although if the lowest note is no longer the root, the triad is said to be *inverted*. As either of the thirds can be either major or minor, four different types of triad are possible, as shown below. The most common are major triads (lower interval a major third) and minor triads (lower interval a minor third). It is important to note that these terms refer to the structure of the *interval* and not to *key*; major triads occur in minor keys, and vice-versa.



Four Types of Triads

triamp: Short for “triamplification.” A three-way *crossover network*.

triangle wave: A *periodic waveform* with a linear increase or decrease in amplitude, followed by a linear change in amplitude at the same rate but in the opposite direction. Triangle waves exhibit a strong *fundamental*, with weak, odd-numbered *harmonics*. Usually used for *vibrato* synthesis. See Appendix C.

trigger: An instantaneous voltage *pulse* generated from a *synthesizer* keyboard when any key is depressed. The trigger is used to initiate the action of some other device, such as an *envelope generator*. The most commonly encountered form of trigger is that generated by drum pads for electronic percussion devices.

trigger sync: A method for slaving a *DAW* to the *SMPTE timecode* from an *ATR*. The DAW initially will sync up to the timecode when starting play or record, but once started will use its internal clock. This feature is available on most DAWs. If you use a trigger sync when recording to the DAW, no two passes will correspond to the ATR exactly because of its speed variations. Tracks from separate passes will start together, but drift in and out of sync over time. This is acceptable for short segments, and in the DAW, a longer audio segment can be sliced up into smaller segments so that trigger sync is appropriate. As opposed to *continuous sync*.

trill: A rapid alternation between one note and the note one *half-step* above it, played in this manner for the notated duration of the note. Used as ornamentation to a musical idea. See *diminution*(3).

T

trim: (1) Usually refers to a small adjustment, synonymous with *tweak*. (2) The attenuation control associated with the first stage of amplification in each module of a console, and by which the incoming level of a mic or line input can be lowered. Usually refers to the *trimpot* on each channel of a mixer used to adjust the incoming signal to set recording levels. (3) A short section of exposed film, *workprint*, or *magnetic film* track that has been cut out of a *take* and included in an *assembly* or workprint.

trimpot: A small pot whose setting is usually adjusted by a small screwdriver, designed to be adjusted only rarely and are used in sensitive circuits such as *equalizers* where they can be finely adjusted and left for long periods.

triplet: Three notes (or rests) executed in the time normally taken by two of the same value. See *tuplet*.

tripole: A type of *loudspeaker* design developed by M&K which is a combination of a *direct radiator* and a *bipole* speaker. These have been specifically designed for use with *surround-sound* systems.

tritone: An *interval* of three *whole-steps*, e.g., F to B.

trombone gobble: A classic sound effect developed by Warner Bros. to accompany a cartoon character being hit on the head.

TRS: Tip, Ring, and Sleeve. A type of *phone connector* found on patch cords for *balanced* audio connection and headphones. See also *TS*, *TT*.

truss rod: A metal bar within a guitar neck which is tensioned so as to counteract the tendency for the neck to bend under the tension of the strings.

TS: Tip and Sleeve. A type of *phone connector* found on patch cords for *unbalanced* audio connection and headphones. See also *TRS*, *TT*.

truck: The vehicle in which video recorders, audio mixers, and other equipment is installed for mobile productions. Normally, sound or video trucks are permanently wired with several record and playback decks, and have the capacity for adding extra machines when necessary. Lighting, mics, stands, and other equipment may also be carried in luggage compartments, so that one vehicle is a complete mobile video and / or audio recording studio.

truncation: Trimming extraneous material for a sample's *head* or *tail*.

TT connector: Tiny Telephone. TT connectors use miniature *phone connector* plugs with a 0.173" diameter. Due to their compactness and reliability, TTs are often used for professional mixer and outboard *patch bays* where a single patch bay may require hundreds of patch points in a limited space. The *TRS* versions of TT connectors are capable of handling *balanced line* signals and are preferred in pro audio installations. Also called a *bantam*. See *phone connector*.

tube: An electronic device which consists of various types of electrodes (anode, cathode, etc.) and a heating element, all contained in a vacuum. Its simplest form, the *diode*, is used as a *rectifier*. The triode is functionally similar to a *transistor*. Tubes are called *valves* in the UK.

Tuchel connector: Another name for a connector which is a *DIN*-standard connector.

T

Tune Request: A System Common message which instructs a MIDI device to re-tune its *oscillators*. This was useful with analog oscillators which are prone to pitch drift, but is less useful with solid-state devices which generate signal based on an electronically generated *clock* pulse, and hence do not drift.

tuning: (1) The process of adjusting the frequency of a radio or television receiver to lock onto a transmitted signal. See *carrier*. (2) The process of adjusting the pitch of all or some notes on a musical instrument in order to confirm with the pitch of other instruments and/or *concert pitch*. See also *beating*, *temperament*, *microtuning*.

tuplet: A generic term used by some sequencers for any non-standard subdivision of a *beat* or beats, derived from the suffix of quintuplet, septuplet, etc., but also encompassing *duplet* and *triplet*. See *time signature*.

turd polishing: General term, including film sound nomenclature, which describes either the process or the futility of making poor quality work acceptable.

turnaround: In a song or arrangement, the *measure* or measures in which one verse or chorus ends, leading into the next section. Also, the short instrumental or vocal lines that accomplish this transition.

turnover: See *adjustable turnover*.

TVA: Time-Variant Amplifier. The *amplitude* section of a synthesizer. Same as a *VCA*.

tweak: To make a fine adjustment.

tweeter: A *loudspeaker* designed to reproduce high frequencies.

two-ended: See *noise reduction*.

two-pop: See *sync pop*, *LFOP*.

two-track: A tape machine which records onto two tape tracks, used primarily for stereo mastering. See also *half track*, *four-track*.

U

U: A term used to define the unit of height in a rack-mounted assembly. A one-U device is 1¾” high (44mm). A two-U device is twice as tall, etc.

UART: Universal Asynchronous Receiver Transmitter. An integrated circuit that carries out the function of asynchronous, bi-directional communication between a microprocessor and a *serial* interface. In MIDI, it is a part of the interface that forms the link between the processor and the MIDI sockets.

UHF: Ultra High Frequency. The band of television broadcast frequencies reserved by the FCC for local and community-access stations, local weather services, stock reports, etc.; channels 14 and higher. Compare with *VHF*.

ultrasonic: Having frequencies above the normal range of human hearing, i.e., higher than about 20kHz.

Ultra ATA: See *IDE.*

Ultra-Stereo: See *stereo optical print*.

U-Matic: A video tape recorder format developed by Sony for professional use. A ¾” *helical scan* video cassette, it has become the world standard for industrial and semi-pro videotape productions. This machine is also used in conjunction with a Sony PCM 1610 or PCM 1630 encoder to record digital audio instead of video.

una corda: One string. Used in piano music to indicate the use of the *soft pedal.* See *tre corda*.

unadvertised specials: In film, sounds that appear on a track, but whose presence is not noted on the *cue sheet*.

unbalanced lines: Any transmission line in which the two conductors are at different potentials with respect to ground. In an unbalanced connection, the ground conductor does double-duty, completing the electrical circuit and serving as a *shield*. Compare with *balanced line*.

undermodulation: A situation which occurs when the *amplitude* of a signal falls well below the optimum level in a recording or broadcasting system, causing it to be masked by *noise*. In digital systems, undermodulation can lead to *distortion*. See *overmodulation.*

underscore: Music that provides atmospheric or emotional background to the primary dialog or narration onscreen to emphasize the action.

unidirectional microphone: Any microphone in which the pick-up pattern exhibits more sensitivity to sounds approaching on-axis, i.e., any variation of the *cardioid* pattern, such as a *supercardioid* or *hypercardioid.* See also *directional microphone, omnidirectional microphone, polar pattern*.

unison: Two notes, or musical lines, which have the same *pitch.* Also used less precisely to include a musical line played or sung in *octaves*.

unit: A single reel of edited *mag film*, corresponding to a given picture reel. The unit can be either *single-stripe* or *fullcoat,* and will usually contain *fill leader* in order to maintain sync with the picture reel.

U

unity gain: A device which neither attenuates nor amplifies a signal. Most signal processing devices have unity gain, which means that they neither amplify sound nor cause *insertion loss*, and therefore can be added into an audio system at various places without upsetting the overall gain of the system.

Universal System Exclusive: A part of System-Exclusive which is intended for all equipment, irrespective of manufacturer. The message uses one of three particular ID numbers after the *SOX* status byte: 125 is for non-commercial or academic use; 126 is for non-real-time use and includes messages for MTC Set-Up, SDS, MIDI File Dump, Bulk Tuning, and GM; 127 is for real-time use and includes MTC Full Message, User Bits, Show Control, Notation, Device Control, Machine Control, Single-Note Retuning messages, Master Balance, Master Volume, and Real-Time MTC Cueing.

Unregistered Parameter Number (NRPN): See *Registered Parameter Number*.

upbeat: See *beat*.

update mode: In console *automation*, the operating mode in which previously written automation data is read back to its respective input faders for alteration by the engineer. The position of each fader at the beginning of an *update take* is defined by the system as its *null-point*. If the engineer leaves these faders at their initial levels throughout the take, no data will be changed. Any fader movement above or below a null point is read in dB, and that track's data changed to reflect the same dB change from the previously written signal level. This updated data is then stored for later use.

USB: Universal Serial Bus. The new slow-peripheral (i.e., keyboard, mouse) interface technology from Apple Computer.

user bits: (1) A group of 32 bits within the 80-bit *SMPTE timecode* message which is available to users for their own purposes, such as recording tape identity numbers, dates, etc. (2) Also, a System-Exclusive message of the real-time type which implements the SMPTE *timecode* message described in *user bits*.

UV-22: See *dither*.

V

VA: An expression of the work which can be performed by an electrical device, but which ignores the *inductance* of the load. It is related to the *watt* in that it is also the product of potential difference in *voltage*, (V) and *current* in *amperes* (A), but it ignores the *power factor* inherent in that unit.

valve: (1) See *tube*. (2) On musical instruments, a device for lengthening or shortening the air path through the instrument, resulting in lower or higher pitched tones, respectively. Valves are typically depressed with the fingers, such as on a trumpet or French horn.

vamp: In a performance of a song, the ending. Either a continuing repetition of the last *chorus* intended to be faded out during the mix, or a *coda* section with solo lines, leading to a hard ending. See *outro*.

variable area: The most common type of *optical soundtrack* used in motion pictures. The variable area track is a transparent line in a black background. The relative width of the transparent part is varied in accordance with the sound waveform. Most movie soundtracks today consist of two parallel variable area tracks, making possible the recording of stereophonic sound. See *SVA*.

variable-rate converter: A new type of digital recording, marketed by Kinetix, which attempts to provide very high resolution sound without resorting to a static increase in *bit depth* and *sampling rate*. Typical linear systems divide audio signals into equal quantums of amplitude and equal quantums of time, producing serially correlated sampling errors, or, errors which are necessarily related to one another and are, therefore, not only not random, and therefore not mutually canceling, but potentially mutually reinforcing at specific bands within the audio frequency, making them more noticeable. Briefly, a variable-rate converter wobbles at random between 44.1kHz and 48kHz to distribute the sampling errors over a wide frequency range, with the general result that they are inaudible. The Kinetix converter also randomizes the quantization steps so that each successive sample is quantized differently. This renders quantization distortion is redistributed as (Gaussian) noise where it is shifted into the 15-18kHz range on output where it is unlikely to be audible. Variable-rate converters provide low bit-rates, do not require dither or compression, produce less background noise and distortion, with increased the audibility of low-level signals and enhanced stereo imaging. The downside is that the variable clock frequency causes problems for interfacing to other digital systems which is why the current Kinetix product includes its own recorder.

variation: A musical form in which a theme is presented and then repeated in a succession of different guises. These may include retaining the *harmonic structure*, while elaborating or otherwise varying the melody, changing the *time signature* and / or rhythmic framework, or using new harmonies, such as substituting the minor for the major of the harmonic sequence.

varispeed: A means of changing the speed at which a tape recorder runs in order to change the *pitch* or duration of a tape recording. Recently, this involves the use of a *servo*-controlled *capstan*, and the speed may be varied by changing the *reference frequency* of the servo.

varispeed oscillator: In tape recorders, an oscillator used to vary the frequency of the AC driving an AC *capstan motor* and, hence, the tape speed. Recorders whose capstans are driven by DC motors often have DC voltage controls for varispeed operation. These are often mistakenly called varispeed oscillators.

V

VCA: Voltage-Controlled Amplifier. A device that responds to a change in *voltage* at its control input by altering the *gain* of a signal being passed through it. Also, the digital equivalent of a VCA which is more properly called a *DCA*.

VCF: Voltage-Controlled Filter. A filter whose *rolloff frequency* can be changed by altering the amount of voltage being sent to its control input. Also, the digital equivalent of a VCF.

VCO: Voltage-Controlled Oscillator. An electronic oscillator whose output frequency is controlled by the application of an external direct voltage. VCOs are used extensively to generate musical signals in *synthesizers*. The ease by which their frequency can be controlled makes them very suitable for *frequency modulation* and complex *sound synthesis*. See *DCO*.

VCR: Video Cassette Recorder. A device for recording and replaying video signals on cassette tape. Cassette *formats* in current use are ¼”, 8mm, ½”, Beta or VHS (consumer formats), and ¾” *U-Matic*. See also *VTR*.

VDP: Video Disc Player.

velocity: (1) A type of MIDI data (range 1 to 127) usually used to indicate how quickly a key was pushed down (*attack velocity*) or allowed to rise (*release velocity*). A Note-On message with a velocity value of 0 is equivalent to a Note-Off message. (2) The velocity of sound: The speed at which sound waves propagate. The precise speed will depend on the density of the medium through which the sound waves travel: in air, at sea level and at 0˚C with 30% relative humidity, this is approximately 1088 ft / sec. (331.7 m / sec.) At average room temperatures it is slightly faster.

velocity compression: Each MIDI Note-On message has a velocity value between 1-127. The velocity corresponds to how hard the key was struck. In *velocity scaling*, (more accurately called a *velocity offset*), a group of notes is selected for editing and then their velocities are all cut or boosted in a linear manner: e.g., with a scaling value of -20, three notes originally recorded with velocities of 65, 91, and 37 would be set to play back with velocities of 45, 71, and 17, respectively. In velocity compression (sometimes called *velocity scaling*), the velocities are multiplied or divided by some factor so that the differences between them get larger or smaller. With a compression value of 75%, for example, the same three notes would be played back with velocities of 49, 68, and 28. This means that the note with the largest starting velocity is reduced the most, while soft notes play back closer to their original velocity, helping to keep them audible. Thus, compression is a better way to smooth out the *transients* in a passage that were played too loudly, without changing the musical dynamics of the piece.

velocity crossfade: The blending of multiple *samples* in varying proportions depending on *key velocity*. Sounds on samplers that are often programmed to use velocity crossfades include pianos and other tuned percussion instruments whose *timbre* changes character markedly depending on how hard a note is played.

velocity curve: A *map* that translates incoming velocity values into other velocities in order to alter the feel or response of a keyboard or tone module. Some devices have a preset range of velocity curves, and some allow users to program their own.

velocity offset: See *velocity compression*.

velocity of sound: See *velocity*(2).

V

velocity scaling: See *velocity compression*.

velocity sensitivity: A type of *touch sensitivity* in which the keyboard measures how fast each key is descending. Compare with *pressure sensitivity*.

velocity switching: See *cross-switching*.

vestigial sideband: In *AM*, whereby a portion of one *sideband* is suppressed.

VFO: Variable Frequency Oscillator. An oscillator used in the generation of electronic music and for audio equipment test signals. Often the VFO is a simple knob or wheel. *VCO*s are a subtype of VFO which are voltage-controllable.

VHF: Very High Frequency. *Electromagnetic* waves with frequencies between 30MHz-300MHz. The subrange of 88.1-108MHz is used for *FM* radio broadcasting in most countries. In the U.S., channels 2-13 on most television sets. Compare with *UHF*.

VHS: Video Home System. A domestic standard ½" cassette *format* for making analog video recordings. The commercially more successful of two such systems which first appeared in the early 1980's, the other being Sony's Betamax. Both formats have been used as a medium for recording digital audio data: Betamax in Sony's *PCM-F1* system, while the successor to VHS, S-VHS is used in the *ADAT* digital multitrack system.

vibrato: A slow *pitch* oscillation; a *periodic* change in frequency, often controlled by an *LFO*, with a periodicity of less than 20Hz, most commonly about 5Hz-8Hz. Vibrato is created in the wave oscillator when a modulation signal is sent from an LFO into the *frequency modulation* input of the oscillator. In synthesizers, *vibrato depth* is the intensity of the effect, and *vibrato delay* is the amount of time a note is set to sustain before the vibrato effect enters. Compare with *tremolo*.

Video-8: A consumer video recording format which uses 8mm tape in a small cassette. It was developed for use in hand-held cameras and other portable equipment, but has also been adopted for use in some digital audio systems, such as *ADAM*.

video black: A video signal composed of all black, containing horizontal sync, vertical sync, and color burst information. Video black is used to *pre-stripe* video tape as the *master clock* (35.8MHz for NTSC) for sound synchronization. *SMPTE timecode* may be derived from the color burst data. Also called *black-burst color signal*, but video black is actually black in color.

video sync: A subset of *video black* which does not necessarily contain color burst signal. It uses horizontal sync or vertical sync to provide a very stable video *reference source*. The signal is used to control the speed of the video machines, digital audio machines, and is used as a timing reference to ensure accurate synchronization. See *master clock, word clock.*

virtual track: A synthesizer part, as opposed to a recording of an acoustical instrument.

vision mixer: A device for mixing video signals from a number of sources. The relative amounts of the signals are controlled by *faders*, primarily for mixing the output of television cameras.

V

VITC: Vertical Interval Timecode. A means of recording *SMPTE timecode* or other timecode into the vertical blanking interval on a high-end, pro video recorder, offering the advantage of being readable even when the tape is not moving. As the VITC signal is recorded by the rotary heads, it must be recorded at the same time as the video signal, whereas timecode signals recorded on a VTR's linear audio tracks can be *striped* either before or after the video picture is recorded.

VMAx: Virtual Multi Axis. A 3D audio technology developed by the Harman Interactive Group which purports to deliver *ambisonic* sound through two channels. VMAx 3D Interactive is a version aimed at multimedia computer applications, while VMAx 3D VirtualTheater™ is the edition of VMAx used for decoding Dolby *ProLogic* and *AC-3* programs for two-channel playback. There is also a VMAx Stereo Enhancement for adding width and depth to stereo *imaging*.

V/O: See *voice-over*.

VOC (.VOC): A file extension specifying the Creative Labs' Sound Blaster™ audio format. Typically encountered as `FILENAME.VOC`. Originally 8-bit, newer cards accommodate 16-bit stereo sound. VOC files support compressed or uncompressed data at a range of sampling rates up to 44.1kHz. Compression options are from 2:1 to 4:1. Not as popular as the *.WAV* format.

vocal booth: An acoustically isolated booth in which one or more singers can perform while rhythm instruments are playing in the studio, but that keeps *leakage* of these instruments from reaching the mics located inside the booth. The same booth can be used to house acoustic instruments being played at the same time as amplified ones in the studio, or vice versa.

vocal score: A notated form of vocal music in which any orchestral parts are expressed as a piano (*treble* and *bass clef*) score, i.e., there is no *full score*.

vocoder: Voice Operated reCOrDER. A signal processor (considered one of a family of *analysis synthesizers*) that imposes the *amplitude* envelopes of one input (control) signal upon a second input (program) signal. In the most common application, someone speaks into a microphone to provide a control signal; the amplitude characteristics of the speech elements are superimposed on an input instrument, giving the latter a "talking" quality. Vocoders use a bank of *bandpass* filters to dynamically analyze the *frequency spectrum* of the control signal and thus continuously derive the amplitudes of the component frequency bands. The resulting amplitude *envelopes* are used to continuously control the operation of another, identically tuned bank of filters. Any program signal applied to the input of this second bank will be shaped by the amplitude envelopes of the control signal, yielding the same spectral characteristics as the control input.

voice: (1) The simplest, individual sound-producing circuitry (*generator module*) an instrument possesses; an element of synthesizer circuitry capable of producing a note, which typically consists of the combination of *oscillator* / *filter* / amplifier with associated *envelopes* for the filter and amplifier. The *polyphonic* capability of a synthesizer is defined by now many voices it has. See voice channel. (2) In Yamaha synthesizers, a *patch*. (3) The instrument for human speech or singing.

V

voice channel: A signal path containing, at a minimum, an *oscillator* and a *VCA*, and capable of producing a note. On a typical synthesizer, two or more voice channels, each with its own *waveform* and *parameter* settings, can be combined to form a single note.

voice coil: The *transducer* part of a loudspeaker, consisting of a coil of wire mounted on a *frame.* Variations in the electrical signal passing through the coil cause it to be repulsed from, or attracted to, a fixed magnet in proportion to the *frequency* and *amplitude* of the signal. The resulting motion is mechanically transmitted to an attached cone, whose piston effect excites the surrounding air to reproduce recorded sound.

Voice of the Theater™: A motion picture theater speaker system developed in the 1940s by the Altec Lancing Corporation, and the industry standard for forty years until the introduction of *direct radiator* speakers. The Altec Lansing loudspeakers included the single-cabinet A-7 and A-4, and the dual-cabinet A-2 for larger theaters.

voice-over: An audio track containing an announcement or narration to accompany a TV or film advertisement, recorded by a person who does not appear on camera. Often abbreviated as V/O. See also *post-sync*.

voice stealing: A process in which a synthesizer that is being required to play more notes than it has *polyphony* switches off some currently sounding voices (typically those that have been sounding the longest or are at the lowest amplitude) in order to assign them to play new voices. See *dynamic voice allocation*.

voiced: An amplifier which is designed to have a *frequency response* which is shaped to a particular instrument, rather than being completely flat. An example are the loudspeakers systems used for electric guitars and basses. The amplification has a very limited frequency response which filters out the worst of the *distortion* due to the amplifier.

voicing: (1) The process of creating a sound on a piano, organ, or synthesizer; voicing means much the same thing as *programming.* (2) *Room equalization*.

volt: Unit of electrical potential between two points in space where V=Resistance × Current (V=IR), *resistance* is in Ω, *current* in *amperes*, equal to the force required to produce a *current* of one *ampere* through an element having a resistance of one *ohm.* See *Ohm's Law*. The voltage of an audio signal is usually measured in terms of the *RMS* value of the signal, but sometimes the peak or average voltages are measured. The RMS voltage squared is proportional to the amount of *power* the signal carries. See also *VA*.

voltage: The difference in the electrical potential of two points in a circuit.

voltage-controlled amplifier: See *VCA*.

voltage controlled attenuator: An amplifier or *resistive network* whose gain is *unity gain* at maximum, and whose attenuation of an audio signal varies in proportion to an externally supplied DC voltage.

voltage-controlled filter: See *VCF*.

voltage-controlled oscillator: See *VCO*.

voltage gain: See *gain.*

V

volume: (1) The degree of *loudness* or *amplitude*. (2) One of the MIDI Controller Change messages assigned to the parameter in a synthesizer which determines output volume, sometimes called the Main Volume, although it only operates on the channel to which it is assigned.

volume control: A voltage divider that adjusts the percentage of input signal without regard to the input frequency that is applied from one amplifier stage to the next. Compare with *loudness control, gain control.*

volume expander: A device for increasing the *dynamic range* and reducing the apparent *noise* of a signal. The second half of a *compander*. A volume expander decreases the system gain as the signal level decreases, making soft signals softer still. This results in an apparent noise decrease because the relative level between the softest and loudest sounds is greater. If the noise level is already low enough that the signal will mask it in the loud passages, the expansion will put the low end of the dynamic range at a point where the ear has reduced sensitivity, making the noise less audible.

VPM: Variable Phase Modulation. A feature of the only modern FM synthesizer, the Korg Prophesy monosynth and XY polysynth. Both synthesizers use the carrier and modulator, and have only two oscillators. However, these can be set to produce all of the audible waveforms.

V.S.: Volte Subito, meaning “turn the page.”

VTR: Video Tape Recorder. A device for recording and replaying video signals on tape, usually open-reel as opposed to cassette. See also *VCR*.

VU: Volume Unit. VU meters show how loud various signals are and measure the *RMS* value of the input voltage. For example, a sine wave that alternates between +1V and -1V will produce a reading on a voltmeter of 0.707V, the average of the voltage in DC. VU meters may be the speedometer type or electrical LEDs, and meter scales in dB. See also *crest factor*, *peak, PPM.*

W

wait: One of the characters transmitted for the purposes of handshaking in a data transfer. The character is sent back to the transmitting device by the receiving device to indicate that the receiver wants the transmitter to pause.

walla: Background *ambient sound* added to a *film soundtrack* to give the affect of an external environment, e.g., street noise, background conversation. See also *ambience*, *room tone*, *NC Curve*.

watt: A unit of electrical *power*. (1) The *SI* Unit of power: 1 Joule per second. 745.7W=1HP. (2) A unit of electrical power, indicating the amount of work deployable into a given load by an electrical device such as an amplifier or motor. It is strictly the product of the potential difference in *voltage*, (V), *current* in *amperes*, (A), and *power factor*. In practice, the power factor is often ignored and the term is reduced to *VA*. See *output power*.

WAV (.WAV): The Windows audio file format. Typically encountered as `FILENAME.WAV`, developed as the standard format for multimedia sound applications. WAV is the most common of several file types that conform to the *RIFF* specification, and so may be referred to as the RIFF WAV format. WAV files can include mono or multichannel audio at 8-bit or 16-bit resolution at a variety of sampling rates up to 44.1kHz. The format supports different compression schemes, but the most common is *IMA / ADPCM* at 4:1 for 16-bit sounds.

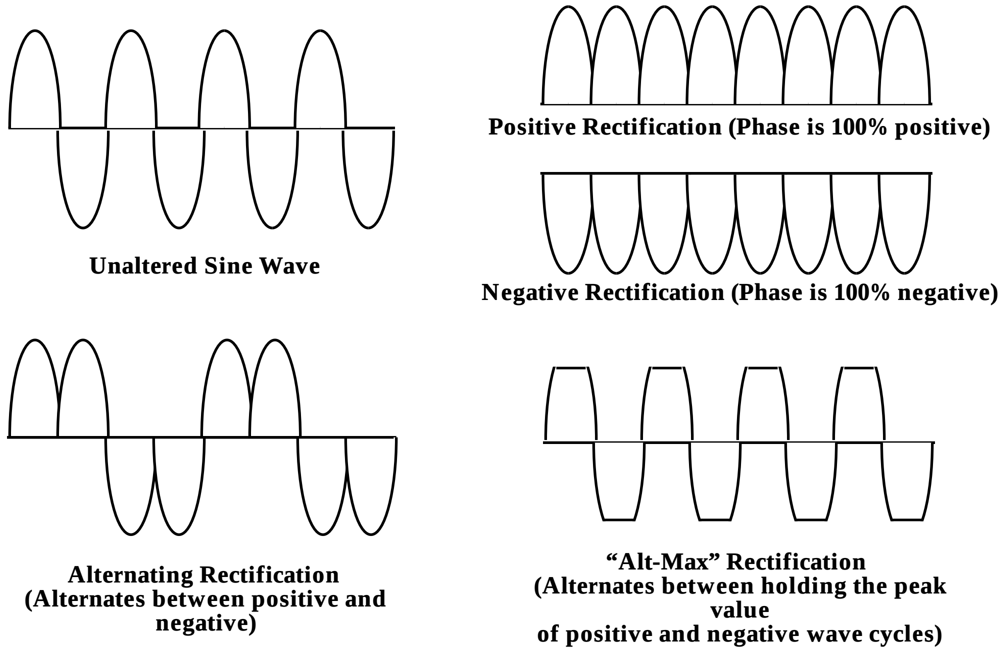
WAV/multi-WAV driver: Used by PC-type system application programs to play sounds on audio cards. Unlike the Mac's Sound Manager, WAV drivers can support multiple channels (i.e., more than stereo 2-channel), and depending on the hardware, resolutions above 16-bit, 48kHz.

waveform: The waveform of a signal is a two-dimensional graph of the instantaneous *amplitude* versus time. See *spectrum*. (1) A signal, either sampled (digitally recorded) or periodic, being generated by an oscillator. Each waveform has its own distinctive timbre. Also called a sample. (2) The graphic representation of this signal, as on a computer screen. See sound synthesis, Appendix C.

waveform modulation: A voltage-controlled change in the *timbre* of a note or entire samplepatch, independent of the pitch or frequency normally designated by the keystrokes. See *sound synthesis*, Appendix C.

W

waveform rectification: A Digidesign™ plug-in that creates new, hitherto-unexplored waveforms:



Waveform Rectification

waveform selection: The waveform selection parameter allows the choice of the shape of the signal generated by the *LFO*. Typical shapes include sine, triangle (tri), square, and sawtooth (saw). The two most common waveforms used for *vibrato* or *tremolo* are the sine and triangle. The square wave is used for trills, and saw for special effects. Some LFOs offer a random (also called *sample-and-hold*) waveform, useful for "computer" sounds. See *sound synthesis*, Appendix C.

wavelength: In a sound wave, the distance between two successive pressure maxima is called the wavelength, and it is equal to the speed of sound divided by the frequency. See *compression*(4), *rarefaction*.

Frequency (Hz)	Wavelength feet/inches	Wavelength meters/cm
20	50'	16.5m
50	20'	6.6m
100	10'	3.3m
250	4'	1.32m
500	2'	66cm
1,000	1'	33cm
2,000	6"	16.5cm
5,000	2.4"	6.6cm
10,000	1.2"	3.3cm
20,0000	.5"	1.65cm

W

wavelength response: The distance spanned by one complete cycle of a sine wave, or the *fundamental frequency* of any complex musical tone, as it travels through an elastic medium, e.g., air, or as it is recorded on tape, film, or vinyl disc.

wavetable: The contents of the *waveform* ROM. A set of numbers stored in memory (ROM) and used to generate a waveform. The wavetable synthesizer on a soundcard typically plays sounds whose digital representations have been stored in a wavetable. See *sound synthesis, wavetable synthesis*.

wavetable lookup: The process of reading the numbers in a wavetable (not necessarily in linear order from beginning to end) and sending them to a *voice channel*.

wavetable synthesis: A method of generating waveforms through lookup tables. Digitized waveforms are organized in a bank, the *wavetable*, where they can be randomly accessed. In many wavetable synths, the resulting waveform is used in *subtractive synthesis*.

weave job: A type of musical track for a spot in which short segments of lead vocal or instrumental lines are interspersed between lines of narration or dialogue. The music weaves in and out, taking the foreground whenever there is no spoken copy.

weber: The basic unit of *magnetic flux*, defined as one volt-second.

weighting: In measuring *frequency response*, introducing a predetermined *equalization* to the signal before taking the measurement. See *A-Weighting, B-Weighting, C-Weighting.*

weighting network: A *filtering network* or *active equalizer* precisely designed or calibrated for use in weighting.

Westrex: The film sound company that, along with RCA, had a monopoly for over forty years, including the whole recording / playback chain. This equipment was leased to studios for royalty fees. By the mid-1970s, this equipment was being replaced by manufacturers of more highly specialized devices such as consoles by Quad-Eight, mag machines by Magna-Tech, and sound format processes by Dolby.

wet: Consisting entirely of processed sound. The output of an effects device is 100% wet when only the output of the processor itself is being heard, with none of the *dry* signal.

white label: A small pressing of a record of CD with an anonymous blank ("white") label, used for market-testing a track on a limited audience before release. Also used by artists to issue a track rejected by the record company to which s / he is under contract.

white noise: A special type of *random noise* where the energy content is the same at each frequency. Strictly speaking, true white noise would have energy extending from DC, or zero frequency. In practice, we see only *band-limited* white noise, e.g., the noise heard when an FM receiver is tuned between stations is quite close to white noise over the AF range. Because of the ears' peculiar method of determining *loudness* of sounds, white noise sounds as if it has more energy at high frequencies than low. Also (imprecisely) called *thermal noise* and *resistance noise*. See *pink noise, noise floor.*

whole-step: The musical interval of a major *second* in a *diatonic scale*. The frequency ratio between the notes of a major second in *just intonation* is $8/9$, and in *equal temperament* it is the $\sqrt[6]{2}$, or about 12%. See *half-step.*

W

wide-range curve: See *X-Curve*.

wide-range monitoring: See *X-Curve*.

wig-wag: Film slang for the warning lights outside *sound stages* to indicate when shooting is taking place.

wild score: See *scoring wild.*

wild sound / wild track: A *film soundtrack* recorded for motion picture where the audio elements are not recorded synchronously with the picture. Wild tracks are frequently used to get a clean recording of dialog that was otherwise unobtainable because of noise on the set. The opposite of *sync sound*. See also *scoring wild.*

window: (1) The frequency *band* in which a device is operational, e.g., the *passband* of a *band-pass filter*. (2) Short for *timecode* window.

window dub: See *BITC.*

windshield/windscreen: See *pop filter.*

wiper: The movable contact in a *pot* is called the wiper, or sometimes, the *arm*.

wireless microphone: A microphone, typically a *Lavalier* microphone, attached to a miniature FM transmitter that broadcasts to an FM receiver. Because the signal is transmitted, no cable is necessary between the microphone and the rest of the audio *chain*.

wobble: To vary between recording or sampling rates. See *variable-rate digital converter*.

wolf note: A term used to describe the inaccuracy in *pitch* which can occur when an instrument such as a harpsichord, which has not been tuned to *equal temperament*, is played on a remote *key*. Also used to denote the undue stridency of a note on an instrument such as a cello, where the pitch of that note is related to the *resonant frequency* of the instrument.

WOM: Write-Only Memory. Memory that can be overwritten by the user, but cannot be read by the user. An example is *WOM/MA* (Manufacturer-Accessible) is used by some hardware manufacturers as a sort of “black box” for units returned for service as a means of recording some of the history of the device.

woofer: A *loudspeaker* designed to reproduce low frequencies only.

word: The smallest possible unit of digital audio; a single number (sample word) that represents the instantaneous *amplitude* of a sampled sound at a particular moment in time. See *bit depth*.

word clock: A signal internally by all digital audio devices, and externally for locking together high-end devices. Word clock is accurate to a single sample word, usually 44.1kHz or 48kHz, i.e., the *sampling rate.* Specialized hardware is required for syncing a digital audio recorder’s word clock to *SMPTE timecode* or some other timing reference, usually through a *BNC*-type connector, rather than the XLR connector used by an AES / EBU *null clock* signal. See *self-clocking*, *master clock*, *Super Clock*.

word length: See *bit depth*.

W

workprint: Copies of the original film or video used as a reference during the sound and/or picture editing process or a *linear* editing system. Also used during *sweetening*. The workprint is usually the first print made of the camera negative with all or selected takes included in their entirety, *slates* and all, also called *dailies*. (2) The edited dailies of a film in an *assembly*, *rough cut* or fine cut. Workprint is often loosely used to refer to all the edited materials, including *magnetic film* tracks, etc. A film copy, usually black-and-white, is sometimes referred to as a *slop print*, and is used to save stress of repeated *rollback* during the mix on the much-spliced workprint. Workprints include *Acmade* edge numbers and *key numbers*. Video copies with *BITC* are known as *window dub*s.

workstation: A *synthesizer* or *sampler* with which the tasks usually associated with electronic music production such as sequencing, effects processing, rhythm programming, data storage on disk, and the editing of stored sequencer data can all be performed by components found within a single physical device, sometimes called a *DAW* by the acronym-friendly.

worktrack: The sound analog of the *workprint*.

worldize: To re-record a track or tracks, usually music, in the space where it would naturally occur in a film.

WORM: Write Once Read Many. Any form of memory which allows data to be stored once only, but the information can be read an unlimited number of times. Recordable CDs are an example of this, although the term generally refers to programmable ICs.

wow: A type of *frequency modulation* distortion which manifests itself as a relatively slow variation in frequency of reproduced sound caused by slow speed variations in the transports of turntables, tape recorders, etc. Pitch fluctuations of 1Hz-2Hz are classed as wow, while faster variations are called *flutter*.

WPC: Watts Per Channel. A unit intended to give an indication of the output of a *power amplifier*. It should be qualified by a *load* condition such as the *impedance* of a loudspeaker, e.g., 200WPC/4Ω, otherwise it is meaningless.

wrap: The parameter of *tape-head* alignment that determines how large an area of tape oxide is in contact with the front surface of the tape head, i.e., the surface containing the *gap*. Also called contact. The correct wrap is different for heads of different frontal design. The combination of wrap and tape tension determines whether the tape makes adequate contact at the gap itself.

write mode: In mixer *automation*, the operational mode in which the system scans channel *fader* levels and perhaps other parameters, making continuous note of the engineer-specified initial conditions for each channel and all changes made by the engineer in *real-time* during a *mixdown*. The scanned data is continuously written to storage, either on one track of the multitrack tape itself, or onto a floppy or hard disk. If stored to disk, the multitrack tape and disk must carry identical *timecodes* for later reproduction of the mix by the system when operating in its *read* or *update* mode.

X-Y-Z

X-copy: An exact, 1:1 copy of audio and / or video material.

X-Curve: eXtended-Curve. As opposed to the *Academy curve*, the X-Curve is also known as the wide-range curve and is codified in ISO Bulletin 2969. Specifications call for *pink noise*, at listening position in a *re-recording studio*, or two-thirds of the way back in a theater, to be flat to 2kHz, rolling off at 3dB per octave after that. The small-room X-Curve is designed to be used in rooms with less than 150 cubic meters, or 5,300 square feet. This standard specifies flat response to 2kHz, rolling off 1.5dB per octave after that. Sometimes a modified small-room curve is used, starting the roll-off at 4kHz, rolling down 3dB thereafter.

XG: A *GM* specification for MIDI instruments by Yamaha. XG instruments support reverb, chorus, and variation effects sends. Over 20 variation effects are available, including *delay*, *flanging*, *phasing*, rotary speaker simulation, *distortion*, and *tremolo*.

XLR: Xtended Locking Round. Developed by ITT / Cannon, XLRs are rugged, locking, multi-pin connectors frequently used in pro audio equipment. While 3-pin XLRs are most commonly seen on microphones and console inputs, other configurations also exist, such as 4-pin XLRs (a standard for stage intercom systems) and 5-pin XLRs (often used on stereo microphones). A designation of “M” after the pin number, such as “XLR-3M” indicates a male connector. Sometimes also called a *Cannon connector*.

X-track: Portions of *production track* that are split off into a separate *unit* or separate track on a workstation because they will be replaced by *ADR*.

X/Y function: Also called a *Lissajous*. Used to determine the *phase* of microphone set-ups, an X / Y function shows phase patterns on an oscilloscope. A *mixer’s* left output channel is connected to the X (vertical) input on the scope; the right output channel is connected to the Y (horizontal) input. If these two channels are *panned* center and are in-phase, they will show up on the scope as a diagonal line that moves up and to the right. If they are *out-of-phase*, the diagonal line will move down and to the right.

X-Y or **XY pair**: See *coincident pair*.

XYZ controller: A 3-axis touch pad that transmits voltages based on finger position along the horizontal and vertical axes (X and Y) and *aftertouch*-type pressure (Z).

Y-connector: A Y-connector, also called a Y-cord, consists of two short audio-type cables with the same type of connector at one end, the other ends of both cables being joined to a single connector of the same type, but of different gender. Used to create *split feeds*, where the same signal is sent to two different places. Sometimes a Y-connector is used to short two channels of a stereo signal together as a sort of mixer, to make a *mono* signal. This can be problematic in low-impedance circuits because each half of the y-connector loads the other one, necessitating a series resistor in each signal path.

Y-cord: See *Y-connector*.

Yellow Book: See *Red Book*, *CD*.

yes: When used to indicate agreement to a verbal contract, potentially the most dangerous word in any language. Use wisely and sparingly. See *no*.

Z: The symbol for *impedance*.

X-Y-Z

zenith: Another word for *tangency*.

zero-crossing: The point in a *sample* where the *waveform* reaches zero *amplitude*. When editing a *loop*, it is usual that both the beginning and end points of a loop are at a zero crossing. However, zero crossings are not important if the beginning and end points of the loop are at the same signal level and have about the same waveform slope. The *harmonic* content of the wave in the region around the loop point has much more effect on the quality of the loop than whether the loop point is at a zero-crossing.

zero frame: The first frame of a roll of film or video tape, appropriately numbered 00:00:00:00.

zero-level: A level of 0dBV. All measurements are made relative to this level as it represents (in properly calibrated equipment) the optimal recording or broadcast level. Higher signal levels than this indicate the possibility of *overmodulation*. Significantly lower levels than this indicate the possibility of *undermodulation*. See *decibel*, *operating level*.

zero-locate: See *auto locator*.

zero return: A control on a tape recorder which will automatically stop the tape on rewind when the tape counter reaches zero.

zone: See *split point*.

Appendix A

Logarithms

Exponents

A logarithm (log) is the exponent to which a given base must be raised to equal the quantity. For example:

Since $\text{Log}^2 = 100$, then the log of 100 to the base 10 is equal to 2, or, $\text{Log}_{10}100 = 2$

Bases

There are three popular bases in use: 10, 2, and $\boldsymbol{e}$. Logarithms to the base 10 are called common logarithms (log). Logarithms in base $\boldsymbol{e}$ are called natural logarithms, where $\boldsymbol{e} \approx 2.71828$, abbreviated Ln:

Base 10

$\text{Log}_{10}2 = 0.301$ is $10^{0.301} = 2$
$\text{Log}_{10}200 = 2.301$ is $10^{2.301} = 200$

Base 2

$\text{Log}_2 8 = 3$ is $2^3 = 8$
$\text{Log}_2 256 = 8$ is $2^8 = 256$

Base e

$\text{Ln}_{\boldsymbol{e}} 2.71828 = 1$ is $\boldsymbol{e}^1 = 2.71828$

$\text{Ln}_{\boldsymbol{e}} 7.38905 = 2$ is $\boldsymbol{e}^2 = 7.38905$

Rules of Exponents

Since a logarithm is an exponent, the rules of exponents apply to logarithms:

$\log (M \times N) = (\log M) + (\log N)$
$\log (M / N) = (\log M) - (\log N)$
$\log M^N = N \log M$

Decibels

The bel is a logarithmic unit used to indicate a ratio of two *power levels* (sound, noise, or signal voltage, or microwaves). It is named in honor of Alexander Graham Bell (1847-1922) whose research accomplishments in sound were fundamental. A 1-bel change in strength represents a change of ten times the *power ratio*. In normal practice, the bel is a rather large unit, so the decibel (dB), which is **1⁄10** of a bel, is commonly used.

$$\text{Number of dB} = 10 \log P2/P1$$

A 1dB increase is an increase of 1.258 times the power ratio: 1dB = 10 log 1.258. A 10dB increase is an increase of 10 times the power ratio, or 10dB = 10 log 10.

Other examples are:

3dB = 2 times the power ratio
20dB = 100 times the power ratio
-30dB = 0.001 times the power ratio

Note that the decibel is *not* an absolute quantity. It merely represents a change in power level relative to the level at some different time or place. It is meaningless to say that a given amplifier has an output of *x*dB unless that output is referenced to a specific power level. If we know the value of the input power, then the *ratio* of the output power to the specific input power (the power *gain*) may be expressed in dB.

If a standard *reference level* is used, then absolute power may be expressed in dB relative to that standard reference, commonly 1mW. Power refrenced to this level is expressed in dBm. Here are a few power ratios and associated dBms:

Power Ratio	dBm
1.258	1
2	3
10	10
100	20
0.001	-30

Appendix B

Alternating Current

To calculate power requirements:

The power required by an electrical device will be expressed in either *watts* (W), *amperes* (A), or as *VA* (Volt Amperes). For these purposes, VA=W. To convert from A or VA to W:

$$P = E \times I \quad \text{or,} \quad P/E = I$$

Where P=power in Watts, E is the voltage (in Volts, either 120 or 240), and I is amps. Ultimately, the total power should be expressed in amperes because electrical circuits are rated in amperes. A standard outlet is 15A. For example, if a mixer is 120V and 70W, the formula is 70 / 120 = .58A. If a limiter is 120V, 30VA, the power required is 30 / 120 = .25A.

Wiring practice: 220VAC One-Phase

For single-phase, 220VAC split into two 110VAC halves, at the transformer, the high-voltage inputs come in via three wires: the 220VAC output winding is divided into two equal parts and the third wire is the *center tap*. If the center tap is used as a reference, the voltage between it and either of the other two wires will be equal, or about 110VAC; these wires are called *legs*.

The three wires from the transformer are twisted together and run as a bundle to the service entrance. The non-insulated wire is the center tap and the electrical code requires that this wire be connected to ground at the service entrance.

Inside the main circuit breaker box are two large, insulated (usually black) wires that go to the two large screw terminals at the top of the banks of the circuit breakers. A volt meter will read 220VAC across the two wires, and 110VAC between either of the wires and the metal case of the box. Circuit breaker boxes are set up so that the breakers in a column alternate legs so as to load-share the current draw from the transformer.

The third wire, color-coded white, goes to a separate terminal block away from the circuit breakers. This is the neutral, the center tap of the transformer. This wire is connected to ground at this point. The terminal block has both the white neutral wires from all the circuits and all the ground wires from these circuits. This is the only place where ground and neutral should be connected. Ground wire is green.

Wiring practice: 3-Phase Wye 120/208

There are four wires used in this connection. A, B, and C are the three hot leads and the fourth wire is neutral. If you measure the voltage between each of the leads and neutral, you will find 120VAC. If you measure between any two of the hot leads, you will find 208VAC (because the two 120VAC circuits are 120˚ apart in phase.) There is no ground connection because the ground is made at the service entrance.

Because of Ohm's law and the resistance in the neutral wire, long runs will develop a voltage in the neutral wire. It can get upwards of 80VAC between neutral and ground in a long power run;

Wiring practice: 3-Phase Delta 120/208

Delta wiring has three wires, the main configuration used in major high-tension lines because there is no need for a fourth wire. Four-wire Delta has the fourth wire coming form the center tap of only one of three transformer windings, requiring a more complicated transformer. Because of this, you only have two connections that yield 120V; connecting to the third hot lead will give you 208V, between the wild leg* and neutral. If you measure between any two of the hot leads, you will find 240V. Note that Wye and Delta look the same; it takes a volt meter to tell the difference. (Ø = phase)

Wires	Wye	Delta
ØA to Neutral	120VAC	120VAC
ØB to Neutral	120VAC	208VAC*
ØB to Neutral	120VAC	120VAC
ØA to ØB	208VAC	208VAC
ØB to ØC	208VAC	208VAC
ØC to ØA	208VAC	208VAC

Appendix B

Troubleshooting

First, find the main circuit breaker box. This will yield the power available and its type. Optimally, there should be a master label which tells you if the panel is one-phase or three-phase. If there is a large master breaker at the top of the box, look at the number of sections operated: if there are two, there is most likely single-phase. If there are three, then it is probably three-phase.

Underneath the master breaker (if there is one), there will usually be two columns of breakers. In the case of a single-phase box, every other breaker in a column is connected to the same leg, and so also for rows. With three-phase, it is every third breaker in each column. To connect to power on the same leg, look for outlets with circuit breakers that are separated. Adjacent breakers are on different legs.

Next, look for grounded outlets close to where you need power. Make a visual inspection, looking for any damage to outlets or covers, and for evidence of major wear, such as a loose fit when a plug is inserted. If possible, check to make sure that the circuit breaker in question actually controls the outlet in question.

Use a voltmeter to check the outlets. There should be 120VAC between hot and neutral and also between hot and ground. Then make sure there is minimal voltage between neutral and ground. Between outlets (using an extension cord), measuring from hot to hot will yield 0V if the outlets are on the same leg; 208V if they are on different legs of a three-phase circuit; 240V for different single-phase circuit.

Impedance

Impedance and signal level are two different things. Level is the voltage swing in a circuit; the higher the level, the higher the voltage swings. Impedance is the resistance to signal flow in a circuit. It is the amount of power needed to move a signal around a circuit.

Connecting two devices and sending a signal between the two uses power. The sending device has to supply power in the form of a signal that is sent to the receiving device.

To determine the impedance of cables and patch cords, first, look at the spec for the cable. On it will be a value for capacitance per foot and inductance per foot. Using

$$\mathbf{Z = \sqrt{L/C}}$$

where Z = the cable impedance in Ω, L = the cable inductance in Henrys and C = the cable capacitance in Farads. For example, for

$$C = 34\text{pF},\ \ L = .17\mu\text{H},\ \ Z \approx 70\Omega$$

Professional line-level equipment designed to drive low source impedances, e.g., 600Ω loads, tends to have source output impedances in the range of 50Ω-600Ω. There is no relationship between impedance and balanced circuits. In practice, there are very few low source load impedances: telephone lines and tube circuits.

Semi-professional equipment, designed to drive loads of 10KΩ, tend to have source output impedances in the range of 600Ω-2KΩ.

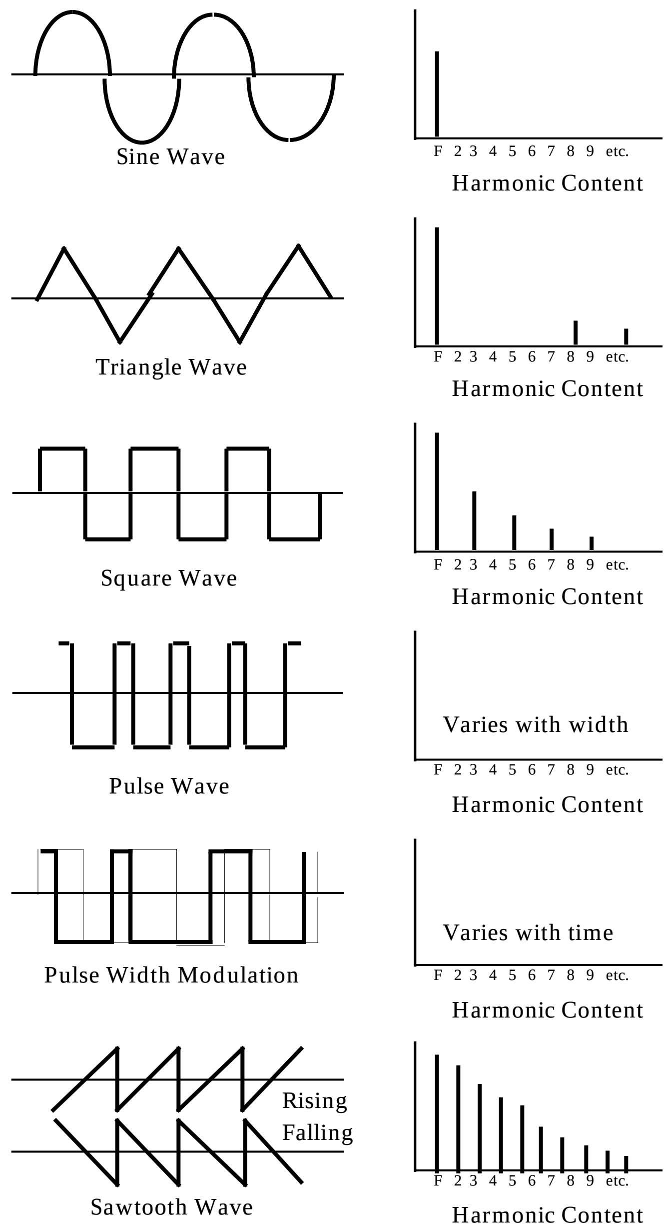
Devices that are designed to be connected as bridging a 600Ω source load impedance typically have impedances of 5KΩ-10KΩ or greater.

Microphone inputs typically have a load termination impedance of about 1.4KΩ for low impedance microphones (mics with a source output impedance of 50Ω-600Ω), mid-range mics 1kΩ-4kΩ, and high impedance mics above 25kΩ. Low impedance microphones are better as they allow for long mic cables without noticeable hum or high-frequency loss.

Home stereo equipment has source output impedances between 1KΩ-10KΩ. This is the actual value inside the box at the circuit level; their load termination impedance ranges from 50KΩ-200KΩ.

Appendix C

Each waveform contains a unique profile of *harmonics* which determines its characteristic *timbre.*



The sine wave is a "pure" tone, i.e., one which contains no harmonics. It is used in additive synthesis to build up more complex sounds, in analog synthesis for *LFO* modulation (such as *vibrato* and *tremolo*), and also as an audio test signal.

The sawtooth waveform, in contrast, contains all the harmonics in the audio range, although not in the same quantities. The loudness of each harmonic is inversely proportional to its frequency, so the harmonic with double the frequency of the fundamental is at half the volume, three times the frequency is at a third the volume, etc. This makes an ideal waveform for producing fuller sounds as it contains all of the frequencies related to the fundamental. The sawtooth may be either rising or falling.

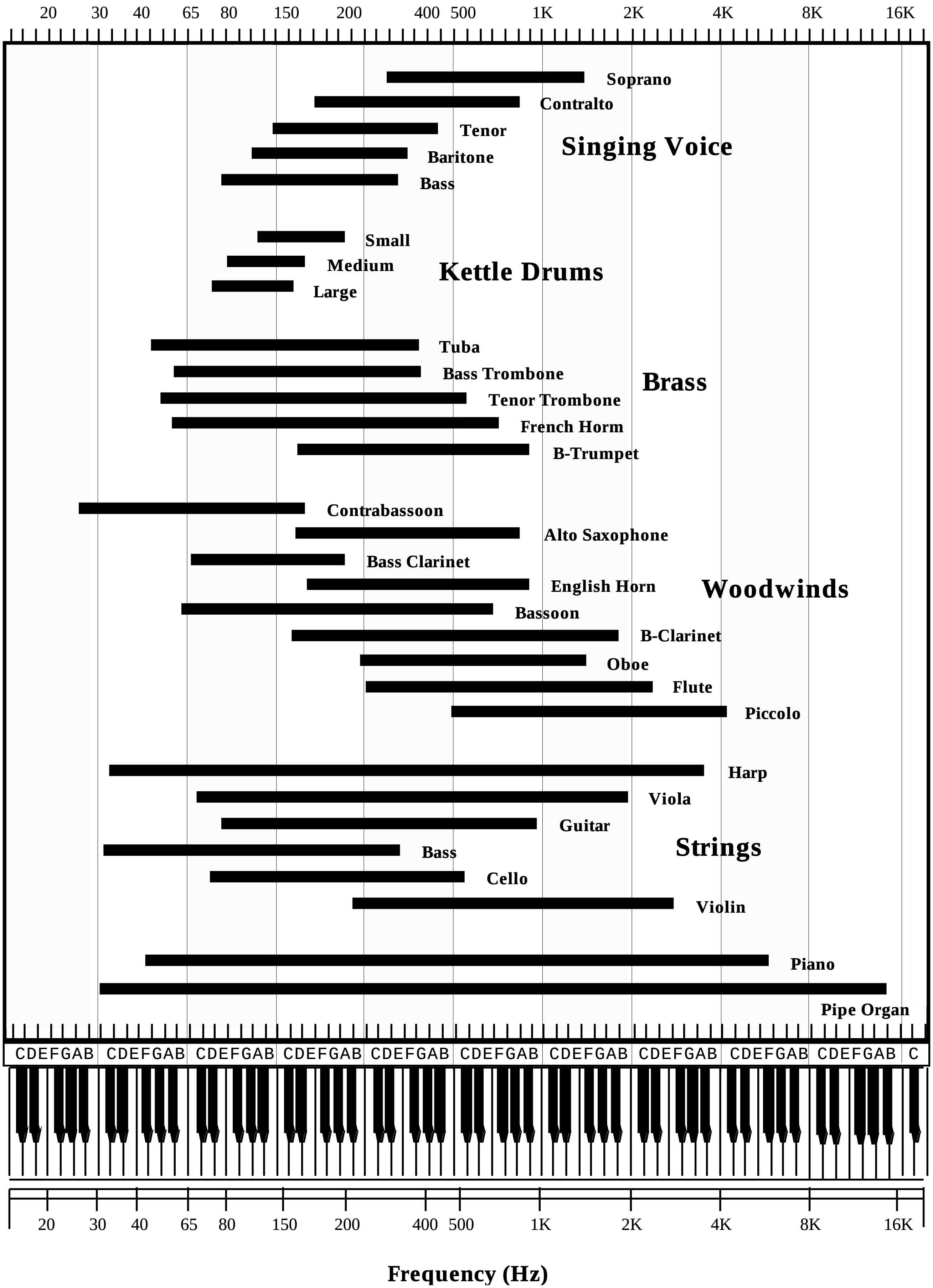
The square wave contains only odd-numbered harmonics, again in inverse proportion to their frequency. The hollow sound produced by the square wave sounds much like the clarinet.

The square wave is actually a special case of the pulse wave, where the negative and positive sections of the cycle are of equal duration. It is the variable nature of the pulse wave which makes it interesting in sound synthesis. Not only are there infinite variations on the harmonic spectra available, but the pulse width may be swept. This is known as *pulse width modulation*. See *PWM*. The width parameter refers to the duration of the positive component in proportion to that of the complete cycle.

The square, sawtooth, and pulse waves contain whole families of frequencies in harmonic series. This means that the human ear perceives these sounds as a single pitch whose tonal quality is determined by the exact mix of harmonics present.

The triangle wave contains the fundamental and a few high harmonics. Normally found only in analog synthesis as a variant on sine wave modulation, but can be used as an alternative to the sine wave for LFO modulation, producing exponential, rather than linear variations.

Appendix D



20 30 40 65 80 150 200 400 500 1K 2K 4K 8K 16K
Soprano
Contralto
Tenor
Baritone
Bass
Singing Voice
Small
Medium
Large
Kettle Drums
Tuba
Bass Trombone
Tenor Trombone
French Horm
B-Trumpet
Brass
Contrabassoon
Alto Saxophone
Bass Clarinet
English Horn
Bassoon
B-Clarinet
Oboe
Flute
Piccolo
Woodwinds
Harp
Viola
Guitar
Bass
Cello
Violin
Strings
Piano
Pipe Organ
CDEFGAB CDEFGAB CDEFGAB CDEFGAB CDEFGAB CDEFGAB CDEFGAB CDEFGAB CDEFGAB CDEFGAB C
20 30 40 65 80 150 200 400 500 1K 2K 4K 8K 16K
Frequency (Hz)

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