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the MUSIC TECH DICTIONARY

– A Glossary of Audio-Related Terms and Technologies –

by Mitch Gallagher

The Music Tech Dictionary: A Glossary of Audio-Related Terms and Technologies

Mitch Gallagher

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About the Author

Mitch Gallagher originally hails from Jamestown, North Dakota, the home of the World's Largest Buffalo. He was introduced to recording music when the manager of the rock band he was in loaned the group a 4-track cassette recorder. His background includes studies in electrical engineering and computer science, and a Bachelor of Arts degree in music from Moorhead State University in Moorhead, Minnesota (now known as Minnesota State University, Moorhead). Graduate studies in composition, electronic music, and classical guitar took him to the University of Missouri, Kansas City.

As a guitarist, he toured the Midwest playing rock and country music. He has performed with big bands, jazz-rock fusion groups, and experimental music groups; in small ensembles; and as a classical and steel-string guitar soloist. He has taught countless guitar lessons, both private and in university classrooms.

As a composer, he has worked in both the commercial and classical realms. *Prophecy #1: At First Glance*, his experimental work for percussion ensemble and synthesizers, received a NARAS (Grammy) award.

He began building his first project studio with a Commodore 64 computer, primitive MIDI software, a low-end drum machine, and a small RadioShack PA for monitoring. Eventually, his studio evolved into MAG Media Productions, which provides a full range of recording, mixing, editing, mastering, and production services, as well as freelance writing and editing services.

In addition to years spent in pro audio retail and as a freelance recording and live sound engineer, Gallagher has taught university-level recording and electronic music classes and labs, seminars on recording, MIDI, and live sound, and has lectured on music technology topics throughout the United States and in Europe.

Mitch was named senior technical editor of *Keyboard* magazine in 1998. In January 2000, he assumed the editor in chief's chair at *EQ* magazine. He has published nearly 1,000 articles on music technology and recording in magazines such as *Performing Songwriter, EQ, Keyboard, Pro Sound News, Guitar Player, Government Video, Extreme Groove, Music Technology Buyer's Guide, Videography, Acoustic Guitar,* and *Microphones & Monitors,* as well as in magazines in Japan, Australia, and throughout Europe. He has released five previous books: *Make Music Now!* (Backbeat Books) was released in 2002. *Pro Tools Clinic: Demystifying LE for Macintosh and PC* (Schirmer Trade Books, 2005) was named the top-selling instructional book by *Music & Sound Retailer* magazine. *The Studio Business Book, Third Edition* (Course Technology PTR, 2006), *Acoustic Design for the Home Studio* (Course Technology PTR, 2006), and *Mastering Music at Home* (Course Technology PTR, 2007) are his most recent titles. His first instructional DVD, *Project Studio Mastering* (Multi-Platinum Pro Tools) was recently released.

In addition to freelance writing and recording and producing a variety of projects in his studio, Mitch is the editorial director for Sweetwater in Fort Wayne, Indiana. Visit him online at www.mitchgallagher.com.

Contents

Introduction

Welcome to *The Music Tech Dictionary*! I hope you find this book to be a useful reference to you in your musical endeavors. This dictionary is intended to provide you with definitions for the sorts of technical terms that you will encounter as you're making music, whether live on stage or in a studio. It isn't intended as the sort of thing that you'd sit down and read from cover to cover (though more power to you, if you choose to!); rather, keep it handy near your music gear so you can consult it when you're hit with an unfamiliar term.

Special thanks must be given to Cathleen Small, who edited every word in this tome, and to Barry Wood, who, as technical editor, waded through and verified each entry. And, no one deserves my thanks more than my lovely wife, Felicia, who is the most understanding woman on the planet! (Imagine putting up with someone writing a *dictionary*…who does that?)

Finally, as you're using this dictionary, if you spot any errors or omissions, or if there are terms you feel should be added, please drop an email to magman@mitchgallagher.com. You just may see your suggestion in the next edition of this book!

Mitch Gallagher

A

a cappella. 1. Vocal music without instrumental accompaniment. 2. In remixing, the raw lead vocal track from the original multitrack recording that is used as the basis for the new arrangement.

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

A/D/A. Analog-to-digital-to-analog conversion.

A-440. The note A, tuned to 440 Hz. A-440 is the standard tuning pitch for most Western music.

AAC (a.k.a. MPEG-2 Part 7 and MPEG-4 Part 3). Advanced Audio Coding, a lossy digital audio data compression and encoding system developed by the MPEG group, as the basis of MPEG-4 and other Internet, wireless, and digital broadcast systems, and intended as the successor to MP3. Claimed improvements over MP3 include more (and higher) sample frequencies, up to 48 channels, increased coding efficiency, improved high-frequency response over 16 kHz, and more. A wide variety of hardware and software devices support AAC, most notably Apple's iTunes and iPods.

A-B comparison. Switching between two audio sources or pieces of equipment to compare a parameter setting, component, or sound quality. The best A-B comparisons are "blind," where the listener does not know the identity of the sources until after the test is concluded. The test results are better if the two sources can be switched instantly.

A-B repeat. Function on hardware or software audio recorders/players that cycles or loops a section of a track.

A-B stereo. A stereo microphone technique that uses two identical omnidirectional microphones placed some distance apart. A-B stereo miking is often used when the sound source is large and/or when the distance from the microphones to the source is great. (Omnidirectional microphones are used to maintain consistent low-frequency response regardless of distance.) Careful mic placement will result in a wide stereo field with a good balance of room ambience and direct source signal. ☞ **See also** *spaced omni*.

ABS. ☞ See *absolute time code*.

absolute phase. More accurately termed *absolute polarity*, absolute phase describes a situation in which the phase remains constant throughout a signal path compared to a reference. In practice, absolute phase means that the + and – polarity of a system are maintained from beginning to end (see Figure A.1). For example, the leading edge of a sound wave usually causes a positive voltage in a microphone, and this voltage remains positive through all the connections and components in the audio system until it reaches the loudspeaker, where it produces a positive (forward) motion in the speaker diaphragm. Some listeners feel that absolute phase is essential for the best sound quality.

A-B-X test. A method of comparing a change in a component or sound quality. The listener is given three sources. A and B are the original source and the original source with some modification to the signal path or sound quality. The third source, X, is the same as either A or B. An assistant selects among the sources while the listener attempts to determine which two are the same and which one is different. The test is repeated enough times to ensure non-random results.

A-weighting. Using a filter to reduce certain frequencies when measuring sound in order to obtain results that match better with the frequency response of our ears. Though A-weighting has many legitimate uses, some manufacturers use A-weighting when making measurements for the specifications for their gear in order to disguise poor performance.

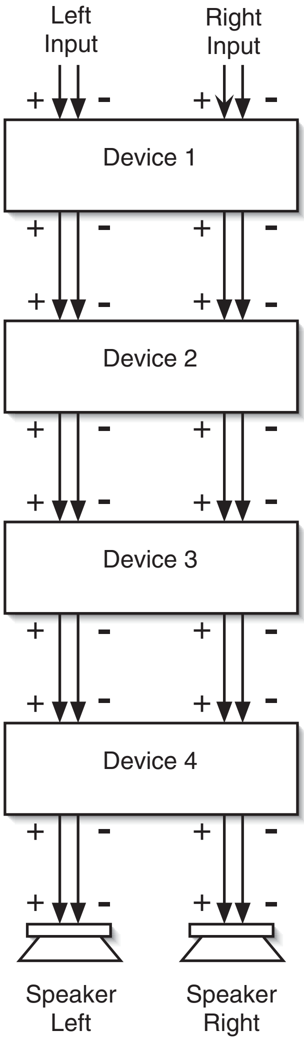

Figure A.1 Positive and negative polarity are maintained throughout a system with absolute phase.

absolute time code (a.k.a ATC). A specific type of time code that is automatically encoded into the subcode area of a digital tape machine. The machine uses this time code for auto-locating and for inter-machine synchronization. Many digital tape machines allow the ATC to be converted to SMPTE for other applications. Unlike SMPTE and other types of time code, ATC always starts at zero at the beginning of the tape.

absorber. Acoustical device that reduces the level of sound waves by converting sound energy into heat.

absorption. In acoustics, using "soft" material to reduce the intensity of sound waves. Absorptive materials convert sonic energy into heat in order to reduce sound energy and level.

accent. Emphasis placed on a specific musical note. Repeated accents at regular rhythmic intervals are a component of "groove" or "feel."

access time. The amount of time it takes for a hard disk or optical disc to reach and read a requested data sector. There are several factors that determine access time, including rotational delay, transfer time, and seek time. Lower access times are always better, especially for audio and video applications in which a large amount of data must be accessed rapidly.

accurate. Uncolored and undistorted reproduction, meaning the output signal for a piece of gear is a faithful version of the input.

acetate. A test disc pressing used to determine how a mastered recording will sound when transferred to a vinyl phonograph record.

acoustic coefficient. Rating for the performance of an acoustical material at a particular frequency. Values range from 0 (totally reflective) to 1 (totally absorptive).

acoustic foam. A special type of open-cell foam designed to absorb sound waves.

acoustic suspension. In a sealed-cabinet loudspeaker design, the air trapped in the enclosure serves as acoustic suspension—as a "spring" that returns the speaker driver to its neutral position.

acoustic treatment. Acoustic devices and materials installed in a space to control the behavior of sound. There are three types: absorbers, bass traps, and diffusors.

acoustics. The study of sound, or the behavior of sound, within an enclosed space.

action. The playability of a keyboard or other musical instrument. For keyboards, *action* refers to how a keyboard works mechanically and how much pressure it takes to press a key. Some types include:

- **piano action.** Key action designed to simulate and respond like a real piano's keys. Usually found on 88-note keyboards.

- **weighted action.** Keys with some resistance, though not necessarily as much as a real piano. Often found on 76- and 88-note keyboards.
- **hammer action.** Key action that incorporates the same type of mechanical hammers as are found on a real piano. Generally found on 88-note keyboards.
- **graded hammer action (a.k.a. progressive hammer action).** Piano-style hammer key action where the response varies depending on the location on the keyboard, to more accurately simulate the action of a grand piano. Generally found on 88-note keyboards.
- **semi-weighted action.** Key action with some resistance or weight that falls between unweighted synth action and fully weighted piano-style action. Often found on 76-note keyboards.
- **synth action.** Unweighted plastic keys, often found on 61-key and shorter keyboards.

active. The opposite of passive. An active device has its own power source and can amplify the signal. The advantage is that signal loss due to processing or other factors can be corrected. The disadvantage is that the amplifier components will add noise, distortion, and coloration to the signal to some degree compared to a passive device.

active monitor. Type of self-contained studio speaker with an amplifier and other electronics built into the speaker cabinet itself (see Figure A.2). Active monitors have some advantages over passive designs: The electronics and amplification can be specifically tailored to the speakers and cabinet used. The speaker-level cables are of minimal length. Because no external amps, crossovers, and other electronics are required, active monitors can be compact, portable, and easy to place, and they generally are more convenient. However, some listeners prefer to match their own electronics to the speaker, and others feel that performance may be compromised in order to fit everything within the space constraints of the cabinet. Others feel that the amplifier and its performance may suffer from the constant vibration due to proximity to the speaker drivers.

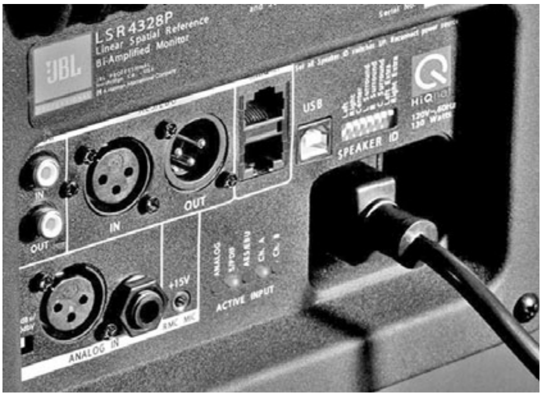

Figure A.2 Active monitors feature built-in electronics including amplifiers, crossovers, and sometimes tone controls.

active sensing. A MIDI system "safety" message automatically sent to indicate that a device is online. Active sensing messages are sent by a piece of gear when it has been sitting idle for 300 ms or so, to let connected devices know that it is still there and that there is no danger of stuck notes due to a broken MIDI connection.

adaptive noise reduction. Technology designed to "intelligently" detect and reduce or remove noise from a signal or environment.

ADAT. Alesis Digital Audio Tape. A groundbreaking modular multitrack digital audio tape/recorder format developed by Alesis in the early 1990s. Arguably responsible in large part for the price versus technology revolution that led to the rise of affordable high-quality home and project studios. ADAT tape machines could record eight tracks with 16-bit resolution (later generations supported 20-bit resolution) to S-VHS videotapes. The native sample rate for ADATs was 48 kHz, though varispeed could be used to achieve a 44.1-kHz sample rate. Multiple ADAT machines could be linked together for increased track count, resulting in the MDM (*modular digital multitrack*) system.

ADAT Optical. Format developed by Alesis for transferring eight tracks of digital audio over a single optical cable using TOSLink connections that has become an industry-standard I/O connection. Companies such as Apogee have developed extensions to the basic ADAT optical format that support higher resolutions and sample rates (though at the expense of reduced track counts). ☞ **See also** *lightpipe*.

ADB. Apple Desktop Bus, an older format and connection type used for connecting peripherals, such as a mouse, keyboard, and trackball, to Macintosh computers. ADB used a four-pin cable that was similar to an S-Video cable. ADB has been superseded by USB.

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

additive synthesis. Type of synthesis based on the work of French mathematician Joseph Fourier, whose efforts showed that a sound wave could be broken down into a series of component partials or overtones. Additive synthesis creates a sound by combining sine waves with different frequencies, envelopes, and amplitudes to construct a more complex waveform. Additive synths are typically digital, though pipe organs and tonewheel organs, which have different stops to represent harmonics, could also be considered additive.

ADR. Automatic Dialog Replacement or Additional Dialog Recording. A process where an actor overdubs new lines to replace the voice recorded during filming or videotaping.

ADSR. An abbreviation for the attack/decay/sustain/release parameters used in synthesizer or sampler envelope generators (see Figure A.3). An ADSR module is a four-stage processor that can be used to control the volume envelope of a note or to modulate another processor, such as a filter. The attack, decay, and release parameters are rate or time controls, while sustain sets volume level.

ADT. ☞ See *automatic double tracking*.

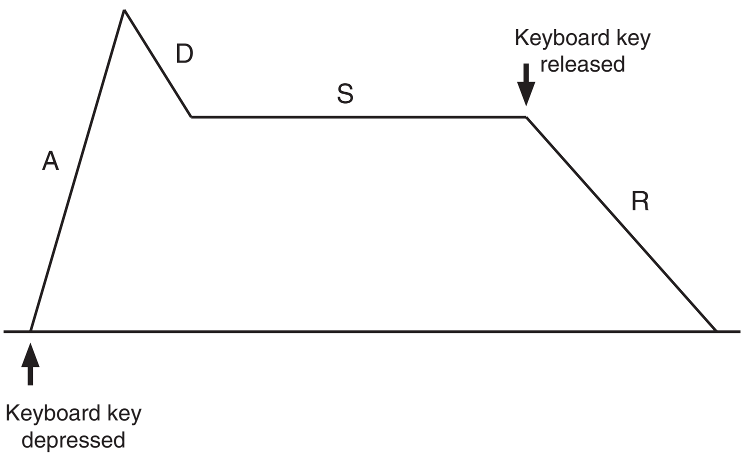

Figure A.3 An ADSR has four stages—attack, decay, sustain, and release—that can be used to control synthesizer or sampler parameters, such as a sound's volume envelope.

AES. Audio Engineering Society. An international professional society dedicated to audio technology. Its committees have been responsible for developing several important industry standards. www.aes.org.

AES/EBU. An abbreviation for Audio Engineering Society/European Broadcast Union. AES/EBU is a protocol jointly developed by the two organizations for routing two-channel digital audio signals between various pieces of equipment. Most typically, AES/EBU signals are carried on 110-ohm shielded twisted-pair balanced cables with XLR connectors that resemble standard microphone cables. (Mic cables may be used to carry AES/EBU, but purists prefer true digital cables.) AES/EBU can be run over distances up to 100 meters with resolution up to 24 bits.

AES3. ☞ See *AES/EBU*.

AFL. After Fade Listen. A switchable function on mixing consoles used to monitor a channel's signal after the volume fader. (The pan control is also sometimes included.) This allows the engineer to listen to the signal by itself, without hearing any of the other channels. ☞ **See also** *solo*, *solo in place*.

AFM. American Federation of Musicians. A union representing professional musicians. www.afm.org.

aftertouch. Pressure changes applied to a keyboard's key after the note has been struck, while the note it plays is sustaining. Aftertouch is carried as a MIDI channel message and has a value ranging from 0 (no aftertouch) to 127 (full aftertouch). Aftertouch messages may be routed to control any parameter that a receiving device allows. Common destination parameters include vibrato amount and volume level (allowing, for example, horn or string patches to swell in volume as a note is held). ☞ **See also** *monophonic aftertouch*, *polyphonic aftertouch*.

AFTRA. American Federation of Television and Radio Artists. A union representing artists involved in news and broadcast, entertainment programming, recording, and other media. www.aftra.org.

AGC. Automatic Gain Control. A circuit that automatically adjusts the level of audio at the input of a device. AGC functions in similar fashion to a compressor or limiter (and may be known as *limiting* in certain devices). AGC is most often included on inexpensive recorders or on devices used for easy remote recording.

aggregate device. A feature of Apple's Mac OS X that allows multiple audio interfaces connected to the computer to appear as one large audio interface to software applications. The interfaces can include any combination of USB, FireWire, and internal devices.

AHDSFR. Attack/hold/decay/sustain/fade/release. An envelope generator that adds a hold time stage between the attack and decay stages and a fade time stage that occurs between the sustain and release stages (see Figure A.4). ☞ **See also** *ADSR*, *AHDSR*, *envelope generator*.

AHDSR. Attack/hold/decay/sustain/release. An envelope generator that adds a hold stage between the attack and decay stages that maintains the peak attack level for a specified amount of time (see Figure A.5). ☞ **See also** *ADSR*, *envelope generator*.

AIFF. Audio Interchange File Format. A digital audio file specification created in 1985 that allows different applications and platforms to share audio files. Most digital audio software, as will some hardware recorders and synthesizers, will create, save, export, and import AIFF files.

AIT. Advanced Intelligent Tape. A special eight-millimeter tape format developed by Sony for large-capacity, high-speed data storage and backup.

algorithm. A mathematical procedure for calculating a specified result. Algorithms are the basis for all processing and operations in audio software.

alias. 1. In the Mac operating system, an icon that points to another file. (Known as a "shortcut" in the Windows operating system.) Double-clicking an alias finds the file the alias points at and launches it. Aliases are convenient for organizing files and folders. 2. A false frequency created when the

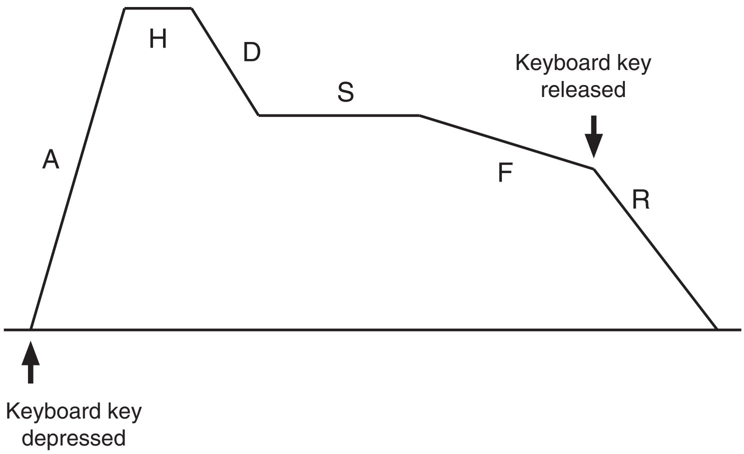

Figure A.4 An AHDSFR adds hold and fade stages to the standard ADSR envelope.

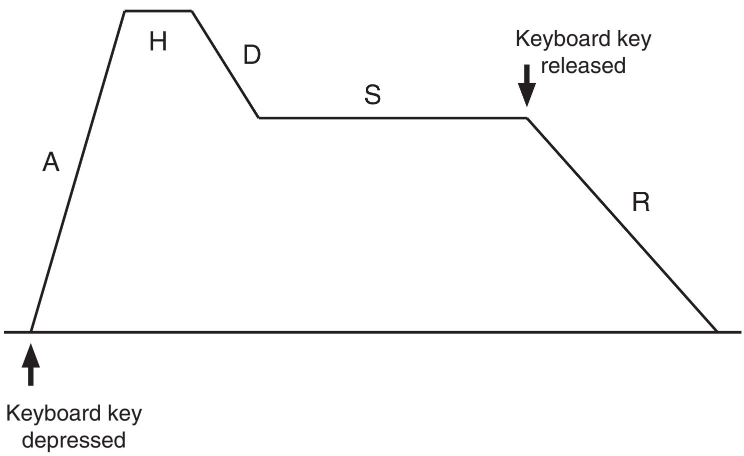

Figure A.5 An AHDSR adds a fifth hold stage to the standard ADSR envelope.

frequency of audio being sampled exceeds the Nyquist frequency. ☞ **See also** *aliasing*.

aliasing. Distortion that results from audio frequencies exceeding the Nyquist frequency during sampling (see Figure A.6). A frequency higher than the Nyquist frequency is folded over into the audible range. For example, if a frequency of 25 kHz is sampled by a system with a Nyquist frequency of 20 kHz, the 25-kHz signal will "fold over" around the Nyquist frequency, resulting in a 15-kHz alias frequency. Most samplers have anti-aliasing filters to prevent this problem, though aliasing can also be a problem when a sample is stretched too far from its base pitch. (For a visual equivalent, think of how a wagon or car's wheels seem to spin backward in a movie. The film's frame rate isn't high enough to capture the speed at which the wheels spin.)

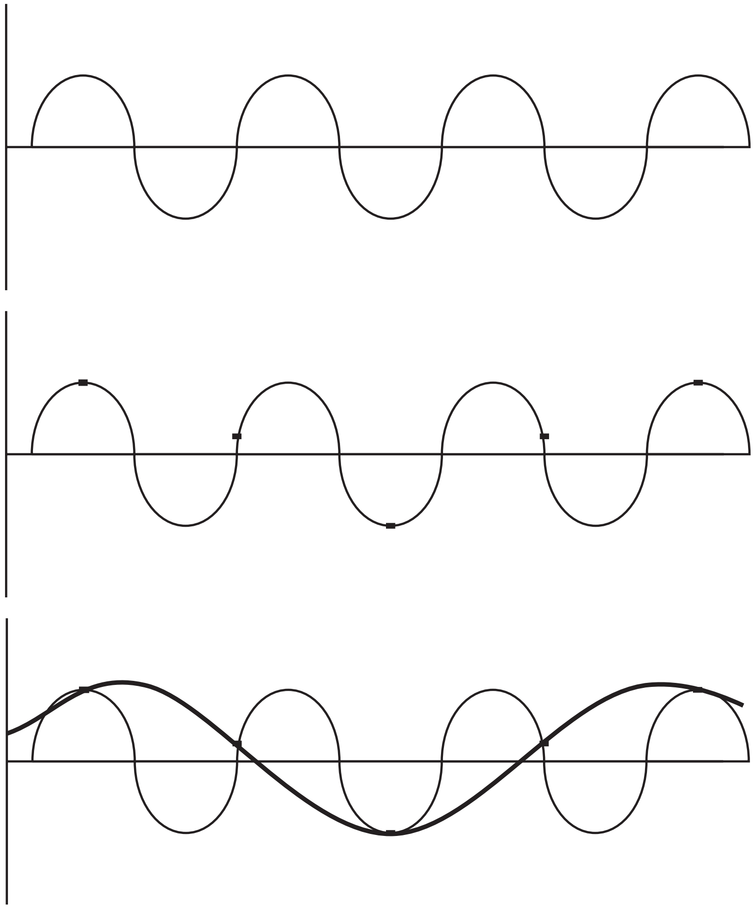

Figure A.6 Aliasing results when frequencies higher than the Nyquist frequency are sampled digitally. The frequency above the Nyquist frequency "folds over" back into the audio range. When the original frequency (top) is sampled at too low of a rate (middle), the result will be an "artificial" lower frequency that does not exist in the original signal.

all-button mode. An undocumented "feature" in UREI (later Universal Audio) 1176 compressor/limiters. The 1176 has four preset buttons selecting ratios at 4:1, 8:1, 12:1, and 20:1. When all four buttons are pressed simultaneously, the ratio changes to between 12:1 and 20:1, and ultra-fast attack and release times create a punchy, overdriven tone that results in intense, in-your-face sounds (see Figure A.7).

all notes off. 1. A MIDI system exclusive command that tells a receiving device to turn off all notes. 2. A "panic button" command found in some synths and MIDI software that sends a note-off message for each of the 128 notes on the 16 MIDI channels.

all tube. A circuit that uses tubes as signal path components wherever possible.

alnico. A magnetic alloy created by combining aluminum, nickel, and cobalt. Various formulations of alnico are used by electric guitar manufacturers for pickups, and alnico has been used in speaker construction for many decades.

alternate MIDI controller. Any MIDI controller that isn't a keyboard. Examples include MIDI guitar, electronic drums, wind controllers, and others.

Figure A.7 Pressing all four ratio preset buttons on an 1176-style compressor/limiter, such as this plug-in re-creation, results in a unique, punchy, overdriven sound.

alternating current. Electric current that switches directions. In the United States, our homes run on AC that changes polarity 60 times per second (60 Hz). Audio signals are also a form of alternating current, with the positive/negative alternation corresponding to the frequency of the sound waves.

AM. ☞ See *amplitude modulation*.

ambiance. ☞ See *ambience*.

ambience. 1. Sense of space. 2. The acoustical character of a space.

Amen break. An oft-sampled four-bar drum loop taken from the song "Amen, Brother," recorded by The Winstons in 1965.

amp. 1. Short for *ampere*, named for André-Marie Ampère, a French physicist and mathematician who explored the link between electricity and magnetism. A unit of electrical current flow. One amp equals 6.24×10^{18} electrons per second. 2. Short for *amplifier*.

amperage. A measure of the number of electrons flowing past a certain point in a second.

amplifier. 1. Electronic device used to increase input signals to a higher amplitude for driving a speaker or other application. Although the most recognizable amplifiers are the common power amplifier and instrument (guitar, bass, keyboard) amplifier, there are amplifiers contained within many other devices, such as synthesizers and samplers. ☞ **See also** *Class A*, *Class A/B*, *Class D*. 2. An electronic component that increases the gain of a signal—for example, "op amp" (operational amplifier).

amplitude. 1. Size or magnitude. 2. The strength or sound pressure level of a signal measured in decibels.

amplitude modulation. A synthesis technique where a carrier wave with fixed frequency has its level (amplitude) modulated by a second wave. If the modulating wave has its frequency in the lower audio range, the modulation will be heard as tremolo (pulsating volume level). At higher modulator frequencies, the result will be sum and difference sidebands in addition to the carrier wave. If only the sum and difference tones are output, the process is called *ring modulation*, and it produces nonharmonic, metallic sounds, or when used to process a vocal as the carrier, it produces "robotic" un-pitched voices. (A great example is the "robot" voices in *Battlestar Galactica*.)

analog modeling. Using digital algorithms to re-create the characteristics of analog devices.

analog sequencer. A module in a synthesizer that plays back a series of voltages, usually set manually using sliders or controlled knobs. This sequence of voltage settings can be used to control the pitch of a VCO (*voltage-controlled oscillator*), the cutoff frequency of a VCF (*voltage-controlled filter*), or any other voltage-controllable parameter.

analog synthesizer. A type of synthesizer that creates and controls sounds using electrical voltages. An analog oscillator is used to generate one of several available waveforms; sine, triangle, sawtooth, and square waves are common options. Filters are used to remove various frequencies from these waveforms. There are a variety of types of modules or stages used in analog synthesizers for creating, processing, and controlling voltages: voltage-controlled oscillators, noise generators, voltage-controlled filters, voltage-controlled amplifiers, low-frequency oscillators, envelope and transient generators, analog sequencers, mixers, ring modulators, and more. Later generations of analog synths added digital/microprocessor control capability, which allowed for saving and recalling presets and MIDI control.

analog. A representation of a signal or information with another continuously varying medium. For example, in audio, using voltages to represent changes in sound pressure. Other analogs include the grooves in a vinyl record and the changes in magnetism on a magnetic tape.

analog-to-digital converter. An electronic device that measures the analog voltage representing an audio signal and converts it into a digital representation of the signal.

anechoic. 1. Totally dead acoustically, literally without echoes. 2. A space that supports no reflection of sound waves. The closest thing to an anechoic situation occurring in earthbound nature is a large, open meadow, but even there, the ground below could create some reflection.

anechoic chamber. A specially treated room designed to be totally absorptive at all frequencies. Anechoic chambers are used for testing and measurement of specifications, not for recording or listening to music.

ANSI. American National Standards Institute. A nonprofit organization that promotes standards and facilitates development of new standards.

anti-aliasing filter. A filter in an analog-to-digital converter that removes frequencies above the Nyquist frequency to prevent aliasing. In some devices, this may be a two-stage process, with an analog filter operating on the oversampled signal, then a digital filter operating after the signal has been downsampled to the actual sample rate.

anti-imaging filter (a.k.a. reconstruction filter). A filter in a digital-to-analog converter used to remove unwanted high frequencies and noise generated by the stair-step waveform resulting from converting a digital representation of a signal into an analog representation.

anti-node. A position along a sound wave at which there is maximum motion, or in a standing wave where there is maximum amplitude. ☞ **See also** *node*.

aperiodic. A non-repetitive event happening at irregular intervals. An aperiodic waveform would not have a pitch, since it doesn't vibrate in a periodic fashion.

API. Application Programming Interface. An application or tools or templates that help programmers create programs that work in a particular operating system. There are also APIs for creating software that runs within another application—for example, the VST plug-in API published by Steinberg.

application. A computer program.

APRS. Association of Professional Recording Services. A British trade association for the sound recording, sound for picture, and music industries. www.aprs.co.uk.

architecture. The components and configuration that make up a piece of hardware or software. The architecture determines what a device is capable of doing, how signals get in and out, and so on.

archive. 1. To back up data for long-term storage. 2. A collection of data that is no longer actively used, but is being stored in case of future need. 3. A data-compressed version of a file, such as a ZIP file.

arpeggiator. A module or control processor provided in some synthesizers that creates an arpeggio based on the chord the user is playing, using either digital control or analog voltages. Arpeggiators can create arpeggios ranging in complexity from basic ascending or descending arpeggiated chord notes to very complex note patterns.

articulation. A musical nuance or expression, such as a slur or legato or staccato performance. Sample libraries attempt to provide examples of as many articulations as possible. For example, a library might include string articulations, such as marcato, spiccato, tremolo bowing, and pizzicato.

artifacts. Extraneous frequencies, noises, or distortions added to a sound by recording, editing, or processing.

ASCAP. American Society of Composers, Authors, and Publishers. A nonprofit organization that licenses copyrighted works and collects and distributes royalties for composers, songwriters, and music publishers. www.ascap.com.

ASCII. American Standard Code for Information Interchange. A standard developed by ANSI that defines how computers deal with text data.

ASIC. Application-Specific Integrated Circuit. A custom integrated circuit chip designed to provide a function not available using "generic" ICs.

ASIO. Audio Stream Input/Output. A driver protocol developed by Steinberg that provides audio software applications with low-latency multichannel access to audio interface I/O, as well as synchronization between audio, MIDI, and video. ASIO supports both the Mac (OS 9 or earlier) and Windows platforms. ASIO bypasses the operating system audio functions to provide direct, high-speed communication.

asperity. An imperfection or roughness in the surface of a magnetic tape. Enough asperities will result in *asperity noise*, a low-frequency rumble.

asynchronous. Not synchronized, or not happening at regular intervals. In computer communications, this means that a start and stop bit are used between messages, rather than specific synchronized timing.

ATA. Advanced Technology Attachment (a.k.a. PATA, Parallel Advanced Technology Attachment). A computer protocol introduced in 1986 that specifies how the motherboard interfaces with the disk controller and hard drives. ATAPI (*Advanced Technology Attachment with Packet Interface*) was later developed, which allowed ATA to support other types of devices. The ATA bus is a parallel protocol that supports master and slave devices. There are several different variations on the ATA protocol:

- **ATA.** A 16-bit, parallel interface supporting up to 8.3 MB per second transfer rates using a 40-pin connector.
- **ATA-2 (a.k.a. Advanced Technology Attachment Interface with Extensions, EIDE, Fast ATA, and Fast ATA-2).** An extension of the ATA disk communication protocol introduced in 1994 that supports DMA and up to 16.5 MB per second transfer rates using a 40-pin connector.
- **ATA-4 (a.k.a. Ultra ATA/33, Ultra DMA, UDMA, Ultra ATA, and Ultra DMA/33).** Up to 33 MB per second transfer rates using a 40-pin connector.
- **ATA-5 (a.k.a. Ultra ATA/66).** Up to 66 MB per second transfer rates using an 80-pin connector.
- **ATA-6 (a.k.a. Ultra ATA/100).** Up to 100 MB per second transfer rates using an 80-pin connector.
- **ATA-7 (a.k.a. Ultra ATA/133).** Up to 133 MB per second transfer rates using an 80-pin connector.

ATAPI. Advanced Technology Attachment Packet Interface. A computer protocol similar to IDE that provides control over optical and tape drives.

ATC. ☞ See *absolute time code*.

ATRAC. Adaptive Transform Acoustic Coding. An audio data compression system developed by Sony for reducing the size of audio files, which works on the premise that low-level audio frequency components will be masked by louder frequencies, and therefore can be removed from the signal.

attack. 1. The beginning portion of a sound. 2. In a compressor, the amount of time between the instant the signal crosses the threshold and when compression begins to occur. With the proper attack setting, the compressor can ignore attack transients while still compressing the rest of the signal. 3. In a noise gate, the amount of time it takes for the gate to open after the signal has crossed the threshold. Very fast settings (10–50 microseconds) allow transients to pass; slow settings (up to one second) can be used to change the attack portion of a signal's envelope. 4. In a synthesizer or sampler, the time it takes for a note to reach full volume. An instrument such as a piano, snare drum, or guitar has a fast attack, where a string section might have a slow, swelling attack. See Figure A.8.

attack time. The time it takes for a signal or sound to go from silence to maximum level. The character and amount of any subsequent room ambience or reverb can be influenced by the attack time of a signal.

attenuate. To reduce the level of a signal.

attenuation. Reduction in signal level.

AU. ☞ See *Audio Units*.

audio. 1. Sound. 2. Electrical signal representing sound.

audio interface. A device that provides audio input and output for a computer. Its features may include analog inputs and/or outputs (with or without analog-to-digital and/or digital-to-analog conversion), digital inputs and outputs, monitor connections

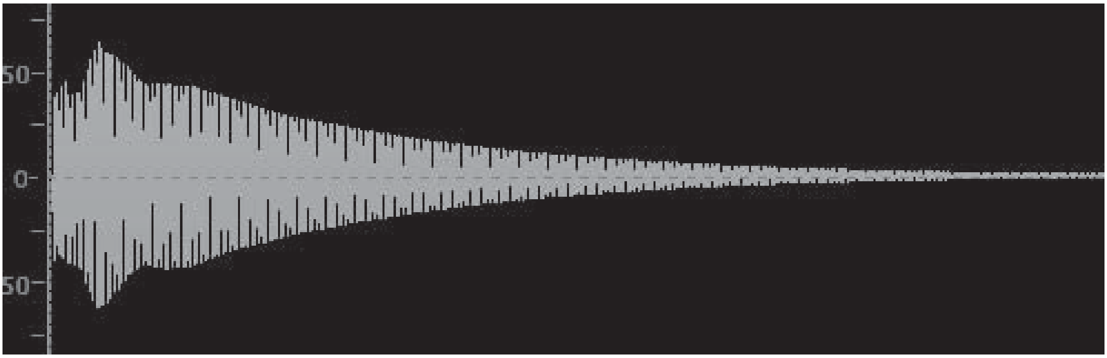

Figure A.8 Attack refers to the beginning of an event, such as the start of a sound wave or synthesizer note, or where a change in a processor's response occurs.

and control, built-in effects or dynamics processing, built-in DSP for reducing or eliminating monitor latency, and clock and/or synchronization inputs and outputs. Audio interfaces are available in a variety of formats, including expansion cards that mount in internal computer slots, external boxes that connect to the computer via USB or FireWire, or combinations of expansion cards mounted in the computer that connect to external input/output boxes.

audio restoration. The process of removing hiss, hum, clicks, pops, and other unwanted noises from an audio recording to restore it to its original sound quality or to enhance it to modern standards.

audio taper pot. A potentiometer that uses a logarithmic curve when attenuating signal. The logarithmic curve more accurately corresponds to how our ears respond to sound level changes, so the volume change using an audio taper pot is perceived as linear by our ears.

Audio Units (a.k.a. AU). A real-time native plug-in format developed by Apple Computer and used in Mac-based Core Audio–compatible host audio software operating under OS X, such as Apple Logic Studio, Ableton Live, BIAS Peak, MOTU Digital Performer, GarageBand, and others. The Audio Units format supports both processing and virtual instrument plug-ins.

audiology. The study of human hearing and perception.

AudioSuite. A non-real-time native plug-in format developed for Pro Tools by Digidesign. Since AudioSuite plug-ins don't operate in real time, they place no load on the computer during playback or recording.

Audiowire. A proprietary protocol developed by MOTU for connecting the company's external audio interfaces to an expansion card mounted in a computer using FireWire-style cable.

author. 1. To assemble and program a CD or DVD. 2. The person who writes a book, such as a music technology glossary.

authorization. A code that allows for the legal use of a piece of software.

authorize. To enable a piece of software for legal use.

auto accompaniment. Software or hardware that can automatically play backing tracks for a song or piece of music. Auto accompaniment can range from basic drum beats in a home organ to full arrangements played by a keyboard, standalone device, or piece of software. In some cases, auto accompaniment can even respond "intelligently" to what the user is playing.

auto input monitoring. A feature on recorders and DAW software that automatically switches a record-enabled track to monitor the audio coming into the input when the recorder is put into record mode, a punch-in takes place, or the transport is stopped. When the software or hardware is playing back, the audio recorded on the track is heard.

auto-locate. To recall a specific time point in a recording and automatically rewind or fast-forward the recorder to that point.

auto-locator. A hardware device that provides remote control over various functions of a recorder, including storing time locations within the recording. These locations can be recalled later, causing the recording to rewind or fast-forward to that spot.

auto punch. A feature of some recorders and DAW software that automatically drops record-enabled tracks in and out of record mode when certain time locations are reached. The user sets in and out points that cause the system to begin and stop recording. This is much easier to do than on older systems, where the recording engineer had to manually "punch" the Record button at the appropriate time, then punch the Play button again at the appropriate time to take the recorder out of record mode.

auto save. A feature of some software that automatically saves the open file to disk at regular time intervals.

automatic double tracking. (**a.k.a. artificial double tracking**). A system, said to be developed for The Beatles for the *Revolver* album, that was used to duplicate the sound of double tracking a vocal—overdubbing a second pass of a vocal or instrument part. In modern studios, an analog or digital processor is used to delay the original signal by a few milliseconds; the delayed version is then mixed with the dry signal. Since the delay is so short, the effect resembles the sound of a vocalist singing along with himself in near unison.

automatic mixer. A type of audio mixer that can detect signal and automatically turn off channels when no signal is passing through them. Automatic mixers are used to help prevent feedback and background noise pickup in churches, audio for video, and broadcast applications.

automation. Using computer control to record changes to switch positions, fader moves, and other control statuses, and to play them back as an audio recording is played. Automation allows accurate, repeatable, and recallable control over a mixer, processor, or performance.

automation curve. A string of automation data entered into a track to control a parameter, such as volume, pan, and others (see Figure A.9).

automation modes. Different ways that automation data can be entered or modified in real time, using a control surface or moving fader system. Typical automation modes include:

- **off.** Automation data is ignored.
- **write.** The first automation pass, in which initial automation data is recorded for the track.
- **read.** Automation data plays back and controls the channel's parameters.

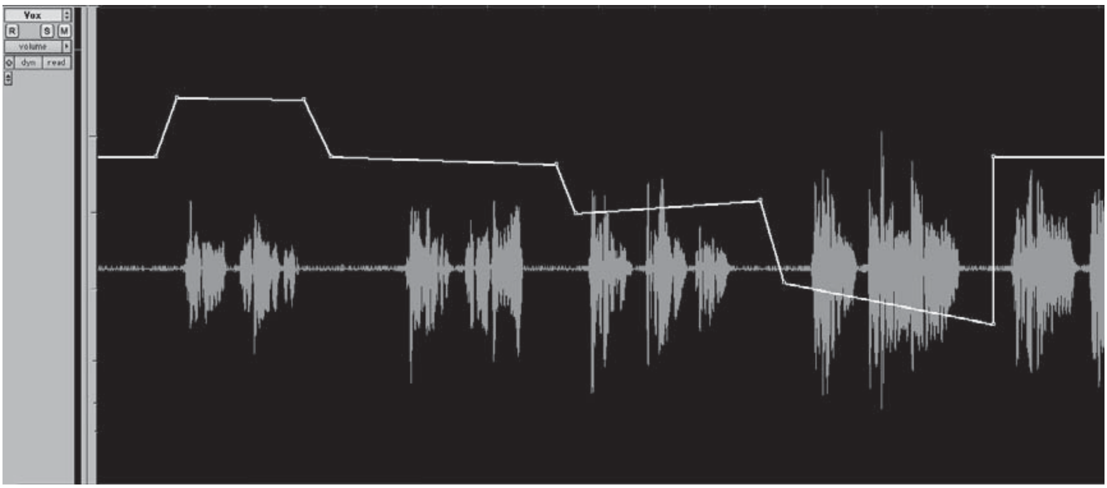

Figure A.9 An automation curve represents the automation data recorded or entered into a track. Here we see a curve that will control the track's volume fader.

- **trim.** The entire automation curve is raised or lowered by a given amount.
- **latch.** Automation recording starts when a touch-sensitive fader or control is touched. When the fader or control is released, automation remains at the current value until recording stops.
- **touch.** New automation data is written over any existing data for as long as a touch-sensitive fader or control is touched. When the fader or control is released, the automation returns to the previously existing value.
- **overwrite.** New automation data writes over and replaces existing data.

autopan. A type of effect that automatically moves a signal back and forth across the stereo field under control of an LFO.

auto-wah (a.k.a. envelope filter). A type of filter effect used with guitars, basses, and other instruments that resembles a wah-wah pedal, except that the cutoff frequency of the bandpass filter is controlled by the level of the incoming signal rather than a control pedal. In some cases, the filter cutoff frequency may be controlled by an LFO.

aux. ☞ See *aux send*.

aux bus. ☞ See *aux send*.

aux send. Short for *auxiliary send*. A bus in a mixer that creates a separate mix, independent of the main signal path and mix. Aux sends are often used to send a mix to an effect such as reverb or to headphones or a monitor system. See Figure A.10.

aux track. A track that is used as a destination for an aux send or as an aux return for internal or external effects processors in a DAW. Aux tracks are also used as virtual instrument tracks in some DAWs.

auxiliary send. ☞ See *aux send*.

average level. The overall average volume level of a signal at a given point in time. The average will fluctuate in response to peaks and dips in the level. One function of compressors and other dynamics processors is to reduce the peaks in a signal so that the average level can be increased.

AVI. Audio Video Interleaved. A Microsoft-developed format for computer multimedia files containing interleaved audio and video data.

AWG. American Wire Gauge. A standard for indicating wire diameter.

AWM. Advanced Wave Memory. A type of synthesis developed by Yamaha that uses sampled sounds as

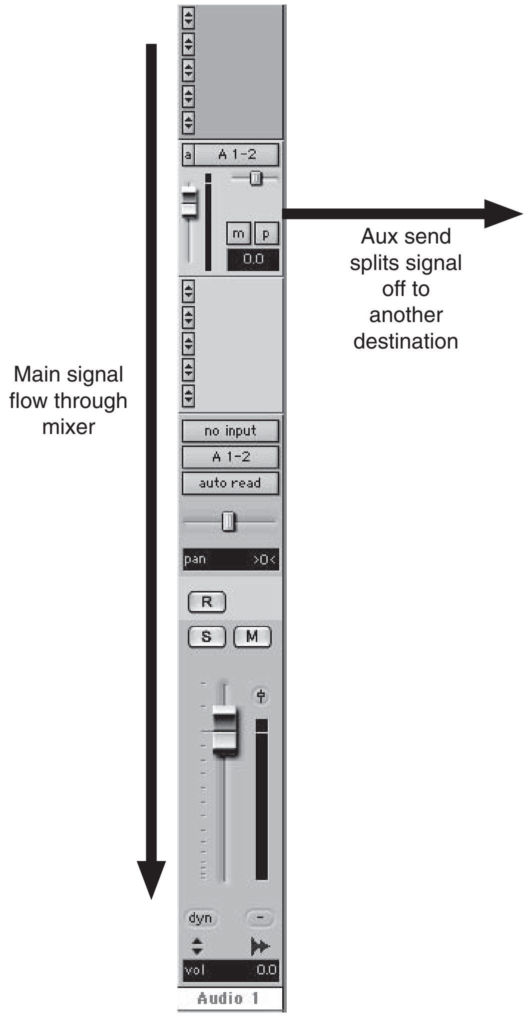

Figure A.10 An aux send splits off signal from a mixer's main signal path and routes it to another destination, such as headphones or an effect processor.

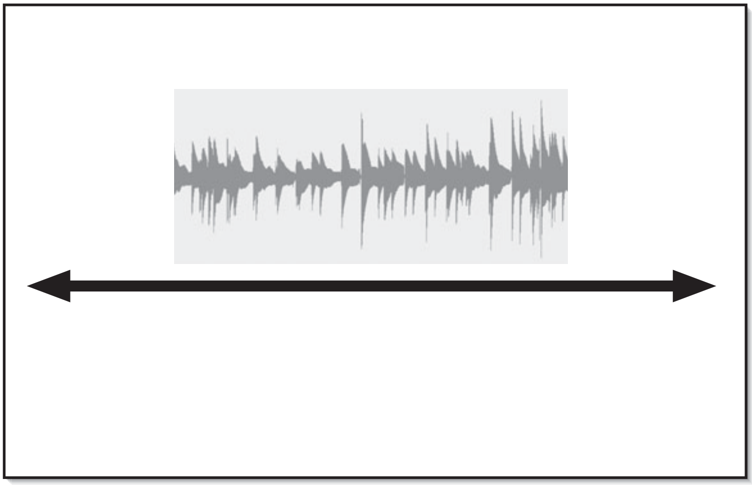

Figure A.11 A sound reflecting between two parallel room surfaces creates an axial mode.

the raw material, which is then processed by filters, modulation, envelopes, and other tools.

ax/axe. Slang for a musical instrument.

axial mode. A room mode caused by sound reflecting between two parallel surfaces (see Figure A.11).

axial room mode. ☞ See *axial mode*.

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B

backbeat. A rhythm/drum pattern in which beats 2 and 4 in a four-beat bar are emphasized. Common in jazz, rock, pop, blues, country, and other styles of popular music. This contrasts with traditional classical and other forms of music that tend to emphasize the "strong" beats—beats 1 and 3.

back-electret. ☞ See *electret*.

background noise. Ambient or environmental noise, hissing, hum, or rumble in a room, space, or recording.

backline. Live-sound jargon for the band's equipment. The backline usually includes instrument amplifiers and drums, and sometimes includes keyboards.

backlit. A type of electronic display that is illuminated from behind for easier visibility under dim lighting conditions.

backplate. In a condenser microphone, a plate placed very closely behind the diaphragm. The backplate and diaphragm form a capacitor, also called a *condenser*. The distance between the backplate and the diaphragm varies with the vibration of the diaphragm in response to sound waves, which results in a change in the voltage between the two. This varying voltage represents the output signal of the microphone. See Figure B.1.

backup. 1. A copy of important data made in case of computer or drive failure. ☞ **See also** *back up*. 2. A spare piece of equipment for use in the event of gear failure.

back up. To make a copy of valuable data on a separate disk or other medium for safekeeping. ☞ **See also** *backup*.

backward-compatible. Data or media that will work with earlier versions of a system or piece of software.

backward masking. Audio content on a recording that can be perceived or understood only when the recording is played in reverse. The idea is for the backward-masked material to sound like part of the music when played forward, but for the message or other content to pop out and become audible when heard in reverse.

baffle. 1. The panel or surface of a speaker cabinet on which the driver is mounted. The baffle is designed to separate the sound coming from the front of the speaker from the sound coming from the rear to prevent interference and phase cancellation. 2. An acoustic divider, usually absorptive, placed to increase the isolation between two sound sources during recording. ☞ **See also** *gobo*.

baffled stereo. A stereo miking technique in which a baffle is placed between the two microphones to enhance the separation between the left and right channels. ☞ **See** *Jecklin Disc* for one example.

balance. 1. A control that adjusts the relative volume level of the two tracks that make up a stereo signal. 2. MIDI Continuous Controller #8, which is defined as control over stereo balance.

balanced. The opposite of unbalanced. A balanced connection uses three conductors—positive, negative (which carries a polarity-inverted version of the positive signal), and ground—within one cable to carry a signal. When the balanced signal arrives at its destination, the positive and negative signals are inverted to be in phase and then are added together to cancel any common noise picked up by both the positive and negative conductors in the cable (known as *common mode rejection*). For this reason, balanced cables are quieter than unbalanced connections and can be run for longer distances. (See Figure B.2.) Balancing is also used with AC power lines to ensure clean power.

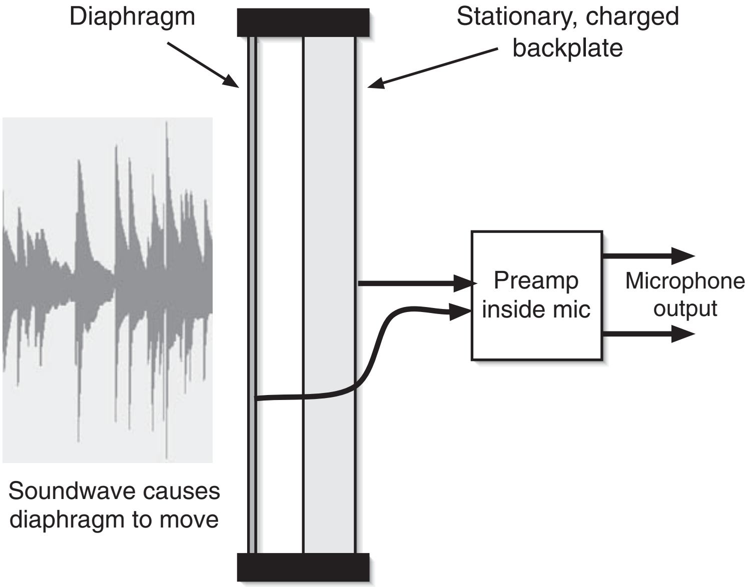

Figure B.1 The output signal in a condenser microphone results from the motion of the diaphragm in relation to the backplate.

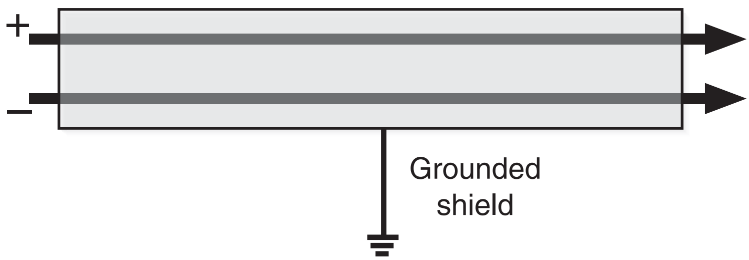

Figure B.2 A balanced audio signal has positive and negative signals that are added together, canceling any noise picked up by the cable. A grounded shield surrounding the two conductors also helps to protect against noise and interference.

balanced power. A type of 120-volt AC power circuit that is created using a transformer and functions in similar fashion to a balanced audio signal. There are positive and negative conductors carrying 60 volts each, as well as a ground conductor. When the electricity reaches its destination, any connected equipment sees a total 120-volt difference between the positive and negative conductors. As the voltage between the positive and negative conductors adds, any noise in the lines is cancelled, resulting in "clean" power and reduced line noise. ☞ **See also** *balanced*.

banana plug. A type of audio connector commonly used for speaker-level signals. Banana plugs can be single or dual. In the dual version, the two plugs are spaced 3/4 inch apart to ensure compatibility with the binding posts found on many power amplifiers and speaker cabinets. (See Figure B.3.) Banana plugs are often designed to "stack" together, making it easy to daisy-chain speaker cabinets together. ☞ **See also** *binding post*.

band. 1. An assemblage of musicians. 2. A range of frequencies.

band limit. The use of a filter to restrict the range of frequencies that is allowed to pass through a system.

band reject filter. A filter that attenuates or restricts the passage of a certain range or band of frequencies, while allowing the frequencies that are outside that range to pass unimpeded. (See Figure B.4.) The opposite of a bandpass filter.

bandpass filter. A filter that allows a certain range or band of frequencies to pass, while attenuating or rejecting frequencies that are outside that range. (See Figure B.5.) The opposite of a band reject filter.

bandwidth. 1. A range of frequencies. 2. The amount of space required or available to carry a signal.

bank. A group of presets or patches within the patch memory of a synthesizer or processor. Banks may be organized by number (Bank 1 contains presets 1 to

Figure B.3 A dual banana plug is compatible with the binding-post connectors found on many power amplifiers and speaker cabinets.

100, and Bank 2 contains presets 101 to 200), type (one bank contains presets for processing vocals, another for guitar, and another for drums), presets created at the manufacturer (Factory Bank) and those created by the user (User Bank), or some other scheme.

bank select. MIDI allows 128 presets to be called up when using MIDI program change messages. The problem is that many devices have more than 128 presets, and they are often organized into multiple banks. To allow access to more than 128 presets, the MMA (*MIDI Manufacturers Association*) specified "bank select," MIDI controller messages (Continuous Controllers #0 and #32) that allow a bank

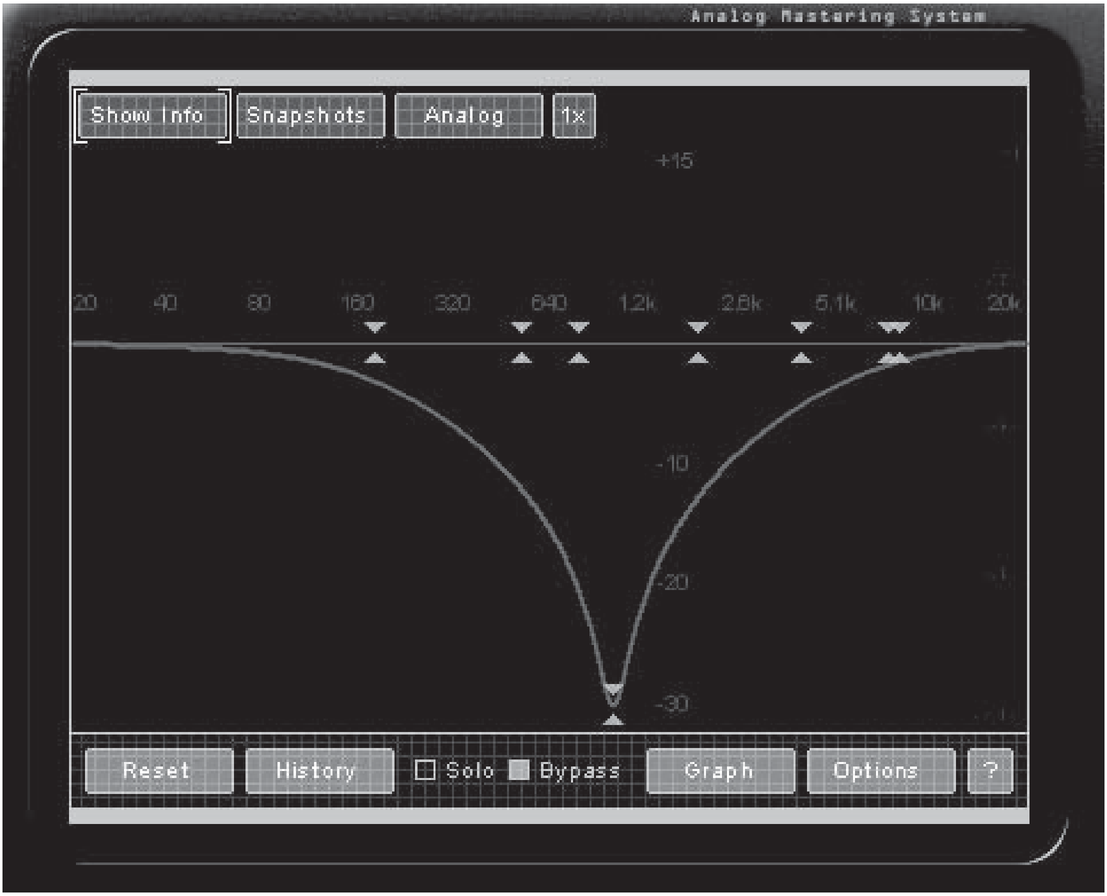

Figure B.4 A band reject filter attenuates a central band of frequency, while allowing frequencies above and below the band to pass unaffected.

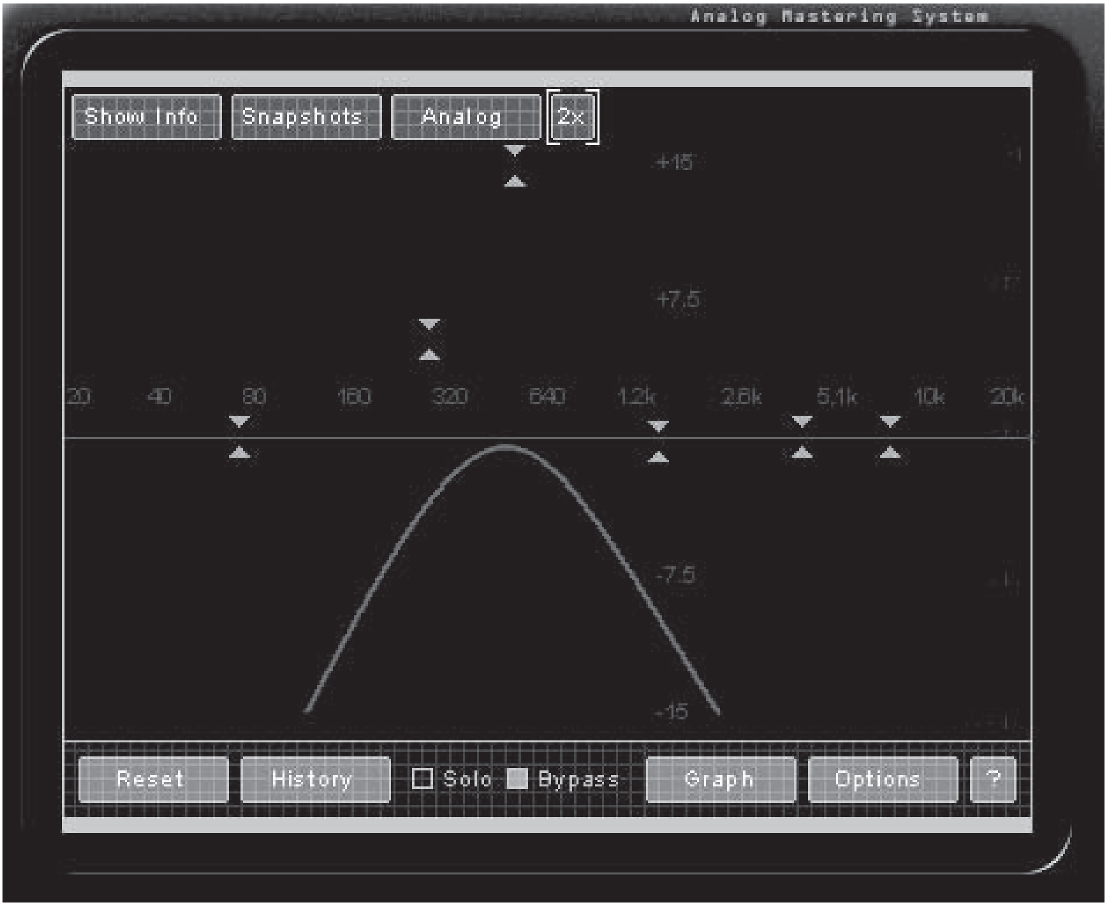

Figure B.5 Bandpass filters allow only a certain range of frequencies to pass, while attenuating or rejecting all other frequencies.

of presets within a synthesizer or other piece of gear to be selected. An individual preset can then be called up from within the selected bank using a program change message. Depending on the piece of gear, either controller or both controllers may be required to call up the bank. Using both controllers, up to 16,384 banks can be accessed, each of which can hold up to 128 MIDI-accessible presets. This would allow access to a total of 2,097,512 presets via MIDI, which is more than enough for producing or performing most songs.

bantam. ☞ See *tiny telephone*.

barrier miking. A microphone technique in which the mic is placed very close to a reflective surface to reduce interaction between direct and reflected sound, thereby preventing phase cancellation. ☞ **See also** *boundary microphone*.

barrier strip (a.k.a. terminal block or terminal strip). A type of connector that uses screw posts to affix the cables or wires instead of jacks and plugs. Found on certain pieces of audio equipment that are intended for permanent or semi-permanent installation.

bass. 1. Low frequencies or pitches. 2. An instrument with four, five, six, or even more strings that (usually) provides the low notes in a musical arrangement. 3. The lowest male voice section in a choir. 4. A type of fish.

Figure B.6 The port in a bass reflex speaker cabinet increases low-frequency output by routing sound from the rear of the speaker out the front of the cabinet.

bass management. A system used to route the low frequencies from a stereo or surround audio signal to feed a subwoofer. According to the Dolby specification, the crossover point for bass management is 80 Hz.

bass reflex (a.k.a. ported). A speaker cabinet design that uses a "port" or hole to enhance the bass response. The sound from the rear of the speaker driver inside the cabinet is routed out through the port so that it combines with the sound coming from the front of the speaker driver. This increases the low-frequency efficiency of the cabinet compared to a sealed or open-back cabinet. See Figure B.6.

bass trap. An acoustic device designed to absorb low-frequency sound waves.

baud rate. A measure of the data transfer speed of a device such as a modem. Named for French telegrapher Emile Baudot. For current technologies, bits per second (bps) is a more accurate measurement than baud rate.

beaming. The tendency of high frequencies to project in a straight line. With speaker drivers, beaming generally begins to occur when the wavelength becomes smaller than the diameter of the speaker or throat of the driver. Horn-based designs help to improve sound dispersion and reduce beaming.

beat. 1. The primary time unit or pulse in music. 2. A drum pattern or loop.

beatboxing. The use of the human voice or body to create percussion sounds.

beating. Cyclical cancellations and reinforcements that can occur when two sound waves of nearly equal frequency (within 30 Hz or so of one another) are combined. Many guitar players tune by striking the same harmonics on adjacent strings and adjusting the pitch of one to match the other, at which point any beating stops. If complex sounds are combined, beating can occur between similar partials within the two sounds.

beat juggling. A DJ technique that is used to create a new piece of music by combining the drums from one record and a riff or vocal part from another record.

beat matching. A DJ technique used in dance clubs to create a smooth transition from the end of one song to the beginning of another without a break between them. Beat matching involves using the pitch control of a turntable or other means to ensure that the tempos of the two songs are the same and that the beats line up rhythmically.

beats per minute. ☞ See *BPM*.

Beige Book. One of a set of "Rainbow Books" with colored covers containing the Sony/Philips-developed specifications for different optical compact disc formats. The Beige Book contains the specification for Photo CD.

bel. A unit of measure expressing the amount a signal drops in level over one mile of telephone wire. Named for Alexander Graham Bell, the inventor of the telephone and researcher into sound, speech, and hearing.

bell filter (a.k.a. haystack filter). A filter that boosts or cuts a band of frequencies around a center frequency, resulting in a basic bell-curve shape to the boost or cut response. See Figure B.7.

beta. A final pre-production version of a product released to a limited number of users for real-world testing in order to determine the stability and quality of the product before it is offered to the general public.

Bezier curve. A mathematically generated curve that requires three or more points—two "anchor" or start and end points, and other points, called *splines*, that determine the shape of the curve. (See Figure B.8.) The more splines in the curve, the more complex the curve can be. Some audio programs provide Bezier curve functions for drawing and reshaping automation data for volume, pan, and other parameters. Named for French mathematician Pierre Bezier. ☞ **See also** *breakpoint automation*.

bi-amp. Using a crossover to split a signal into two frequency ranges (high and low) and sending those ranges to separate amplifiers and speakers optimized for those ranges.

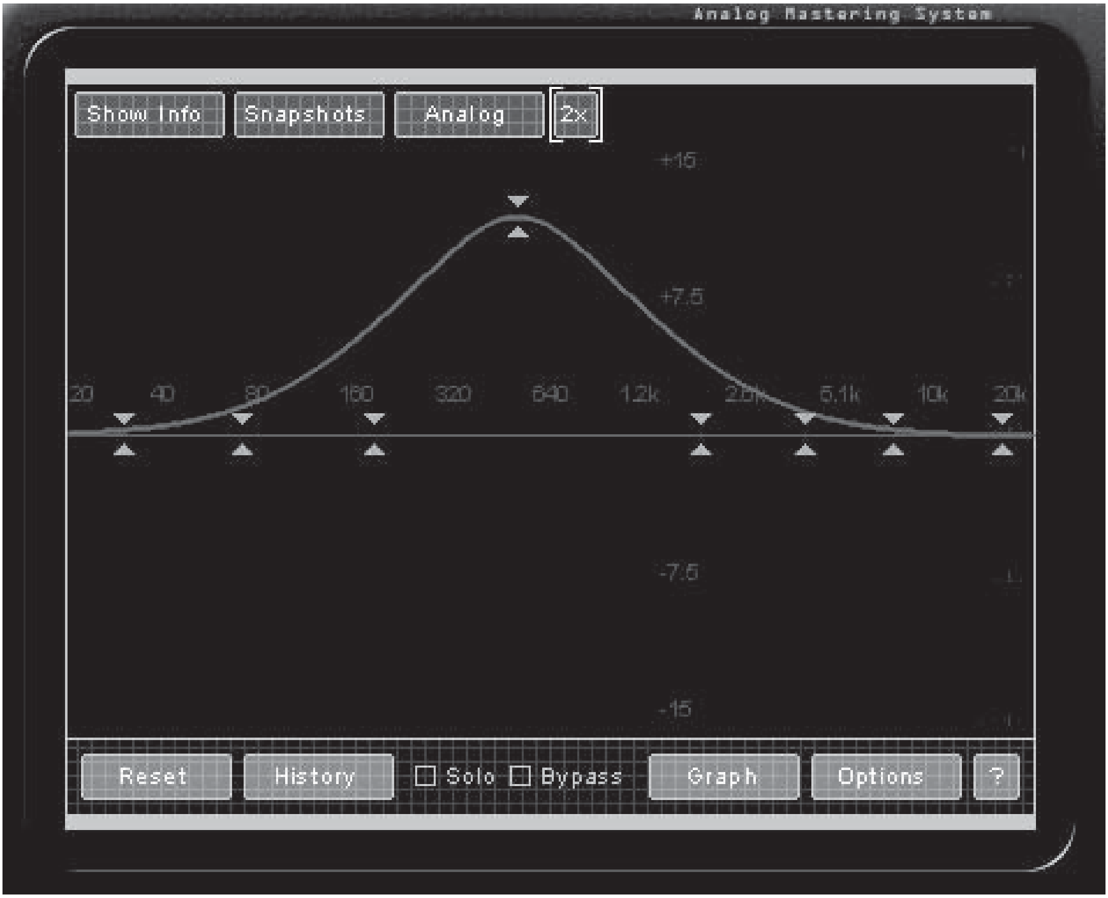

Figure B.7 Bell filters have a response curve that resembles the basic bell curve when boosting or cutting.

bias. 1. A 100-kHz or higher signal applied to magnetic tape during recording to reduce distortion and increase linearity. 2. A small voltage applied to a tube's grid to reduce distortion and increase linearity. 3. A small voltage applied to an FET's gate to reduce distortion and increase linearity. 4. A small current applied to a bipolar transistor's base to reduce distortion and increase linearity. For all four, bias must be set properly, or distortion will increase and/or level and high frequencies will be reduced. 5. The tendency for a musician to always want his or her own part to be the loudest in a mix.

bias beat. A tone or artifact that results when the bias frequencies on two synchronized analog recorders are slightly off.

bi-directional/bidirectional. 1. Carrying signal in two directions. 2. A microphone polar pattern. ☞ **See also** *Figure 8*.

binary. 1. Composed of two parts or things. 2. A number system that uses two as its base, rather than 10, as in our standard system. The binary numbering system uses two values, 0 and 1. This system is the standard for electronic devices, as these two values are easy to represent by high or low voltages or as bit values. As you move to the left in a binary number, each digit doubles in value. See Table B.1.

binaural. A recording technique that attempts to duplicate how human hearing responds in terms of phase, directionality, and physical separation. This

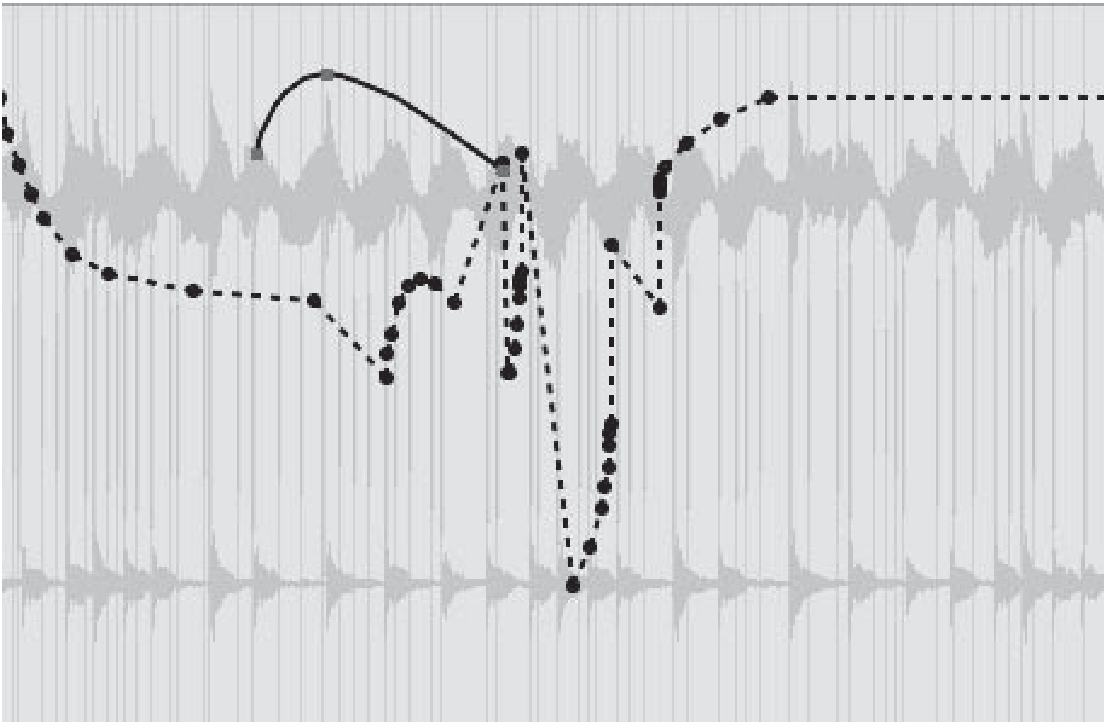

Figure B.8 Bezier curves are mathematically generated and shaped based on splines, or data points. In this example, the handle, or spline, can be used to shape the curve.

is generally accomplished using two microphones mounted in either a dummy head or a sphere, which keeps the sound entering each mic separate, simulating how our ears work on the sides of our heads. The binaural effect is generally best heard through headphones.

binding post. A type of connector often found on power amplifiers and speaker cabinets. Binding posts can accept banana plugs, bare wire, alligator clips, and other cable terminations. This type of connector is convenient and fast, and it offers good connectivity for carrying speaker-level signals. ☞ **See also** *banana plug*.

BinHex. Short for Binary-to-Hexadecimal. A method for converting eight-bit binary files into seven-bit ASCII files so they can be sent through email as text. BinHex files usually have the .hqx extension.

BIOS. Basic Input/Output System. Software that controls how a PC interfaces with its keyboard, mouse, monitor, hard drives, and other hardware, as well as instructions for booting up and other useful utilities.

Bit. Contraction of binary digit. Each bit can have only two possible values, zero or one. ☞ **See also** *binary*, *bit resolution*.

bit depth. ☞ **See** *bit resolution*.

bit mapping. A scheme somewhat similar to dither and noise shaping for optimizing audio quality when reducing digital word length. Sony's Super Bit Mapping is a common example, which claims the sound quality of 20- or 24-bit resolution using 16 bits. ☞ **See also** *dither*, *noise shaping*.

bit rate. ☞ **See** *bps*.

bit resolution. The number of bits available to represent an audio signal in the digital domain. Each bit of resolution provides roughly six decibels of dynamic range. Though it may seem counterintuitive, increased bit resolution improves the ability

Table B.1 Examples of Binary Numbering

Base 10	Binary
0	0000
1	0001
2	0010
3	0011
4	0100
5	0101
6	0110
7	0111
8	1000
9	1001
10	1010
11	1011
12	1100
13	1101
14	1110
15	1111

of a digital device to represent low-level signals (not high-level signals). As the audio level is decreased, the number of bits available to describe the audio signal decreases, thus reducing the integrity of the signal and introducing distortion. With higher bit resolutions, there are more bits available to represent these low-level signals. See Figure B.9 and Table B.2.

bit splitting. A method for recording high-resolution digital signals to a low-resolution recorder. For example, by splitting the digital words in a 24-bit/48-kHz signal, it can be recorded to two 16-bit/48-kHz tracks on an ADAT or other recorder, with 16 bits of data stored to one track and the remaining eight bits stored to the subsequent track. Typically, whatever device does the bit splitting will also perform the reverse operation, rebuilding the 24-bit/signal from the two low-resolution tracks.

blackburst (a.k.a. video sync or house sync). A video signal with no color information, only black. Since the black-only signal contains video frame information, a blackburst generator can be used as a master clock to synchronize video equipment for editing or other purposes.

bleed (a.k.a. spill). Sound waves "leaking" from one space to another or into a microphone not intended for the sound source that creates them.

BLER. Short for Block Error Rate. Data is written in a series 512- or 1,024-byte blocks on CD and hard drives. The BLER is a measure of how many blocks have errors or can't be read. Too many error blocks will make a disk difficult or impossible to read.

block. A 512- or 1,024-byte (or other size) chunk of data on a compact disc or hard drive. Smaller blocks allow for less wasted space on the disk; three 512-byte blocks would be required to store a 1,100-byte piece of information. (The unused space in the third block cannot be used for storing other information.) The same 1,100-byte piece of information would require two 1,024-byte blocks, but more space would go unused in the second block. See Figure B.10.

Blue Book (a.k.a. CD-Extra). A CD format specification jointly developed by Sony and Philips. The Blue Book spec is in the Mixed Mode family and supports placing both audio and data on the same optical disc. Different variations include CD-Plus, Enhanced CD, CD+G, and more. The Blue Book standard fixed a problem with some older CD players that sometimes had difficulty playing Mixed Mode discs, as the data portion on Audio Track 1 would be played as a loud burst of noise.

Bluetooth. A wireless technology that operates in the 2.4-GHz range and is intended as a low-power, low-cost solution for interconnecting computers, PDAs, cell phones, printers, keyboards, mice, and other equipment. The first version of Bluetooth was intended for short-range use, with a range of 10 meters. But later power classes upped the maximum range to 100 meters.

Blumlein pair. A stereo miking technique developed by EMI chief engineer Alan Blumlein during the

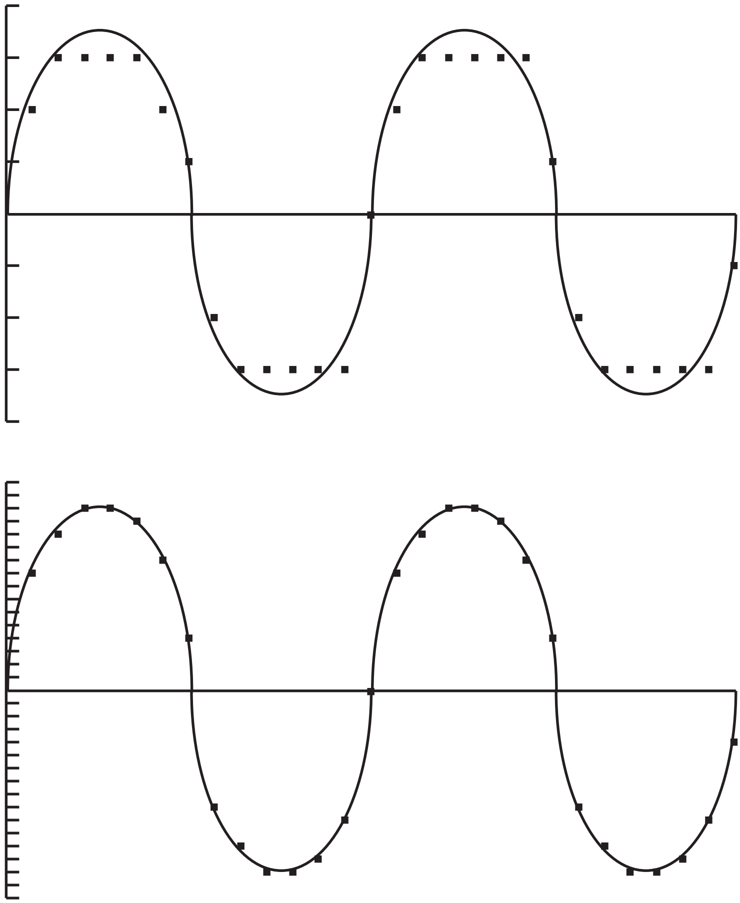

Figure B.9 Adding more bits to a digital audio system increases the resolution from low (top) to high (bottom), resulting in a more accurate picture of the audio waveform.

1930s. Two coincident bidirectional (figure-8) microphones are angled at 90 degrees to one another. In addition to a strong center image, a great deal of ambience will be captured. For this reason, a good-sounding room is required for best results with Blumlein pairs. See Figure B.11.

BMI. Broadcast Music, Inc. A nonprofit organization that licenses copyrighted works and collects and distributes royalties for songwriters, composers, and music publishers. www.bmi.com.

BNC (a.k.a. bayonet connector). A coaxial connector found on test equipment and used to carry digital clock synchronization information for audio gear. Named for its inventors, Paul Neill and Carl Concelman, and its "bayonet" connection type, which uses two small pins to latch the plug to the jack.

boost. To increase. In audio, this usually refers to amplitude or gain.

boot. To start up a computer.

Boot Camp. Apple software that allows an Intel-based Macintosh to run the Windows XP and Vista operating systems.

booth. ☞ See *isolation booth*.

bounce. To combine tracks together to mono or stereo and re-record the result to a new mono or stereo track. The original tracks are then freed up to record more signals. Bouncing was common in the days of analog and digital tape machines, which had a limited number of tracks. It is less common today, when most computer-based DAWs offer unlimited tracks. ☞ See also *bounce to disk*.

bounce to disk. A process found in many DAWs that records the mono, stereo, or surround output of the mixer to the hard drive. This eliminates the step of having to record the final mix to an external machine or back into new tracks. All edits and plug-ins, fades, and other processing are included in the bounce. Some systems bounce in real time—the same amount of time it would take to play back the project. Other systems can bounce faster than real time—in less time than it would take to play back the project.

Table B.2 Increasing the Number of Bits Also Increases the Available Resolution and Dynamic Range

Bit Depth	Resolution	Dynamic Range
1 bit	2 steps	6 dB
2 bits	4 steps	12 dB
3 bits	8 steps	18 dB
4 bits	16 steps	36 dB
8 bits	256 steps	48 dB
12 bits	4,096 steps	72 dB
16 bits	65,536 steps	96 dB
20 bits	1,048,576 steps	120 dB
24 bits	16,777,216 steps	144 dB

boundary microphone. A microphone designed to be placed against or near a reflective surface to reduce the interference between direct and reflected sound. Sound waves in close proximity to a surface—called the *pressure field* or *pressure zone*—are in phase with those reflecting from the surface, resulting in a 6-dB increase in level. The benefits are increased microphone sensitivity and tonal consistency regardless of the distance of the source from the mic. Boundary microphones are often placed on boardroom conference tables and on the stage floor at theaters to pick up sound at plays and musicals, and they are employed in studios to record pianos, drums, and other sound sources. Boundary mics are sometimes referred to as *pressure-zone microphones* or PZMs, though this is a trademarked term.

BPM. Beats Per Minute. An indication of the tempo of a piece of music.

bps. Bits Per Second. A measurement of digital data flow. Sometimes mistakenly called *baud rate*. ☞ **See also** *Kbps*, *Mbps*.

Bps. Bytes Per Second. A measurement of digital data flow. ☞ **See also** *KBps*, *MBps*.

braided shield. A type of cable shielding created by braiding wire strands (usually copper) around insulated conductors. ☞ **See also** *shield*.

breaker. ☞ **See** *circuit breaker*.

breakout box. A box that provides individual access to multiple channels or signals carried over a single multi-wire cable. The most common example is the stage box end of an audio snake used in live sound applications. See Figure B.12.

breakpoint automation. A type of automation system that uses data points inserted into a track to define the automation "curve" (see Figure B.13). By adding, deleting, or moving the breakpoints, the shape of the curve can be changed. Breakpoint curves are based on straight lines, rather than curves as in Bezier curves. ☞ **See also** *Bezier curve*.

breath controller. 1. A hardware device that connects to a keyboard or sound module and senses airflow. The player blows in the breath controller, and the air velocity is measured. The resulting data is used as control data. Breath controllers are commonly used by keyboard players to add expression to performances. 2. MIDI Continuous Controller #2, which is defined to carry breath controller information.

breathing. The audible rise and fall of the noise floor between sounds in a compressed signal. Breathing results from a large amount of compression being applied, which allows the average level to be increased, simultaneously raising the level of noise in the "silences" between sounds. Adjusting the compressor's release time can help reduce breathing. ☞ **See also** *pumping*.

brickwall filter. A filter with an extremely steep cutoff slope. Brickwall filters are useful for applications such as anti-aliasing filters, but they often have unacceptable audible side effects when used in other applications.

brickwall limiter. A limiter with an extremely high ratio (such as infinity:1) that prevents audio levels from exceeding a certain threshold. Most commonly used to prevent overs in digital systems and for overload protection in amplifiers and sound systems.

bridged (a.k.a. mono bridged). A power mode found on some amplifiers that allows two channels of an amp to drive a single speaker for increased power output. The same signal is sent to the inputs of two amplifier channels, with one channel flipped

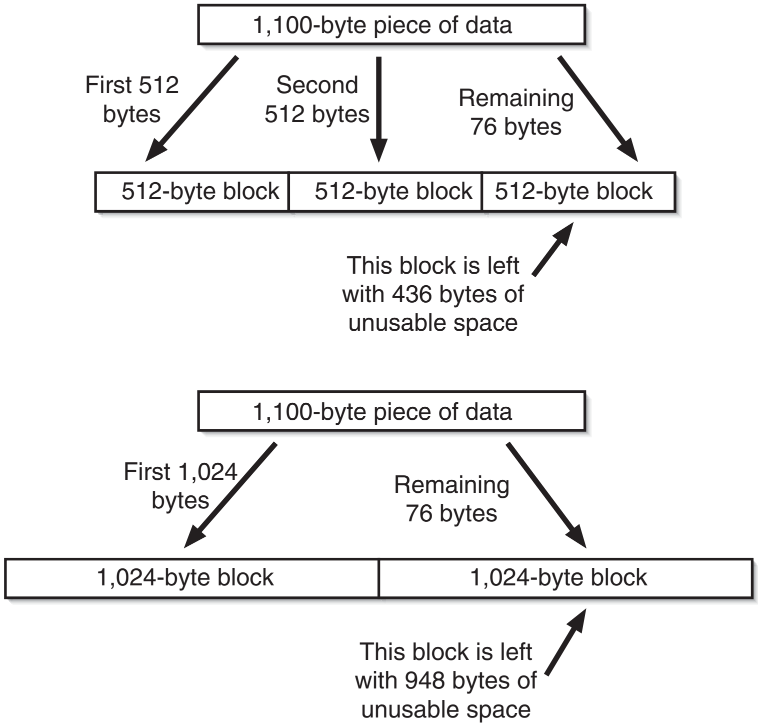

Figure B.10 A comparison of how data block size can impact how much data can be stored on a disc or drive—the larger the block size, the more space is wasted.

to reverse polarity. When the speaker is connected across the positive terminals of the amp's channel outputs, it sees much higher power than it would from one of the channels alone. It is important to ensure that the speaker being used has the correct impedance when bridging an amplifier (usually twice the minimum impedance for the amp). Check your amp's manual first!

British EQ. An equalizer circuit or algorithm that attempts to re-create characteristics of the EQ found in certain UK-built mixing consoles, such as Neves and Tridents (among others), that were in production in the 1950s, '60s, and '70s.

broadband. 1. Effective over a wide range of frequencies. 2. A high-speed transmission method.

broadband absorber. Acoustic device designed to absorb sound waves across a wide range of frequencies.

broadband noise. Noise that comprises a wide range of frequencies.

Broadcast WAV file (a.k.a. BWF). An extension of the common Microsoft WAV file format. Broadcast WAV files include metadata that allows for easy transfer of audio files between different software programs and computer platforms. Among the most useful information in the Broadcast WAV file is time stamping, which specifies where the file is located in a track. Time stamping allows the file to be quickly and easily realigned when it is imported into another program or session.

BTD. ☞ See *bounce to disk*.

buffer. 1. A small amount of temporary memory that's used to manage and even out real-time transfer of data from, to, or within a computer. The size of the buffer is important to the success of many tasks, such as CD burning, audio recording and playback, and more. 2. A type of amplifier or preamplifier that interfaces pieces of equipment together.

buffer size. The amount of RAM contained in a buffer.

buffer under-run. A situation that can arise during data transfers where there is not enough data in a buffer to feed the receiving device. Buffer underruns were common problems in the early days of CD burning, but are less common today.

bug. Computer jargon that refers to a problem in the software code that causes an error.

bulk dump. A method of transferring a large quantity of data (for example, the patch memory of a synthesizer) out of a device using MIDI system exclusive messages. Useful for transferring information into a computer for editing or backup/archival purposes.

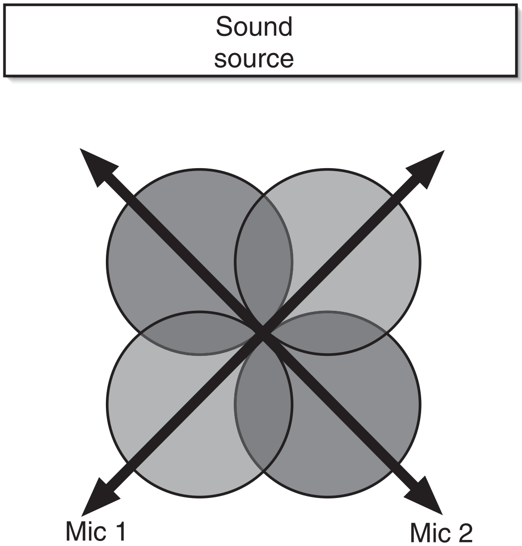

Figure B.11 In a Blumlein pair, two figure-8 microphones are placed at 90 degrees to one another in front of a sound source. The result is a strong center image and good pickup of room ambience.

bulk load. Transferring data into a device using MIDI system exclusive messages. ☞ **See also** *bulk dump*.

burn. To write or create a CD-R/RW or DVD-R/RW.

burn in. 1. Powering or operating a piece of gear to uncover infant failure. 2. Operating a piece of equipment, such as a speaker, for a period of time to break it in. 3. An image permanently etched on a video screen after it has been displayed for a long period.

burst. 1. A short blast of audio used for calibrating equipment. A burst can be either a *tone burst*, which contains sine waves, or a *noise burst*, which contains white or pink noise. 2. A video signal that contains no information. ☞ **See also** *blackburst*.

burst transfer rate (a.k.a. maximum transfer rate). The maximum data transfer rate a drive can attain. Sustained transfer rate is a better measure of drive performance with audio and video applications.

bus. A circuit or virtual signal path that collects audio signals and routes them to a specific destination. Examples include a mix bus, aux bus or send, or effects bus.

bus bar. A common feature of mixing consoles, in which a metal rod is used to bring together all the grounds in the unit and route them to a single earth location.

Figure B.12 A breakout box provides access to the channels or signals within a multi-wire cable.

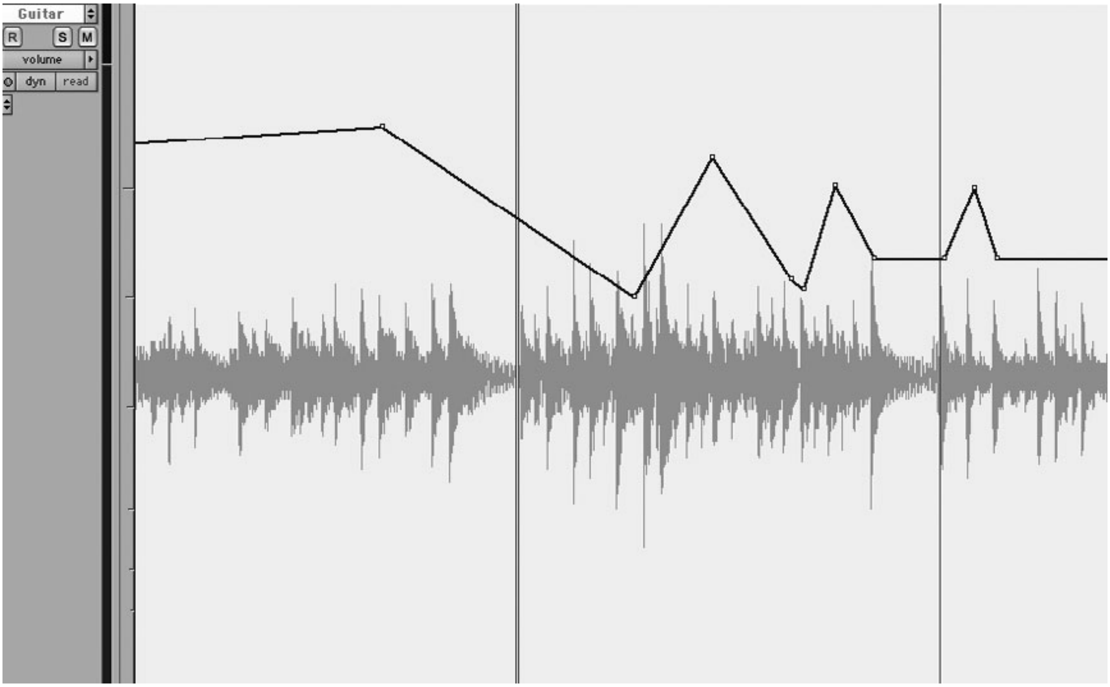

Figure B.13 Linear automation "curves" are created using data points (called breakpoints) inserted in a track.

buss. A kiss. ☞ See *bus*.

butt splice. 1. With analog tape, joining together two pieces of tape that are cut at 90-degree angles. Because a butt splice may pop or click when it crosses the playback head, angled splices are preferred by many engineers. 2. Two regions of digital audio in a track that are joined together to play continuously. A crossfade may be required for smooth playback.

BWF. ☞ See *Broadcast WAV file*.

bypass. A setting that routes a signal through a device without being processed or effected. In hardware, there are two types: regular bypass, which generally includes a buffer amp in the signal path, even when the device is bypassed, and "true" bypass, in which the input of the device is connected directly to the output.

Byte. An 8-bit binary "word" or grouping of data. ☞ **See also** *KB* and *MB*.

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C

cabinet. A speaker enclosure.

cache. ☞ **See** *cache RAM*.

cache RAM. 1. Also known as L1 cache. A type of RAM (*Random Access Memory*) that is built into a computer's microprocessor chip and operates at high speeds. Cache RAM is used to store frequently accessed data. 2. Also known as Level 2 cache, L2 cache. High-speed RAM that is often mounted to a computer's motherboard, close to the CPU, and is used to speed the access to frequently used data and to transmit data from the processor chip to the main memory. 3. Also known as disk cache or hard drive cache. A type of RAM in a hard disk that is used to buffer data transfers. ☞ **See also** *buffer*.

calibration microphone. A microphone that is intended for measurement purposes rather than recording or sound reinforcement applications. Calibration microphones typically have extremely flat response and are designed to provide an accurate, uncolored picture of the audio source.

cancellation. ☞ **See** *phase cancellation*.

cans. ☞ **See** *headphones*.

capacitance. The capability of an electronic component to store a charge.

capacitor. An electronic component that stores an electrical charge. In audio, capacitors are also used as filters to separate high and low frequencies.

capstan. The motor-driven rotating shaft in a tape recorder that pulls the tape past the heads.

capsule (a.k.a. element). The capsule is the part of a microphone where sound is converted into electrical signals. The capsule contains the diaphragm, shock-mounting, windscreens and other protective elements, electronics, and other items.

capture. To record a sound source.

CARAS. Canadian Academy of Recording Arts and Sciences. A not-for-profit organization focused on the promotion of Canadian music and artists. CARAS puts on the Juno awards, the Canadian equivalent of the Grammys. www.carasonline.ca.

CardBus. A 32-bit version of the PC card standard. ☞ **See also** *PC card*.

cardioid. A microphone polar pattern that is vaguely heart-shaped (see Figure C.1). Cardioid microphones are quite directional, picking up sound best from the front, less well from the sides, and not at all directly from the rear. Cardioid-patterned microphones are by far the most commonly used for studio and live applications because they can be positioned to reject sound from the rear, providing good isolation and feedback rejection. Cardioid microphones, like all directional microphones, are subject to proximity effect.

carpet. A poor acoustical absorber best employed on the floor of a room to provide podiatric comfort.

carrier. In AM and FM synthesizers, the audible sound wave. The carrier is modulated by one or more other oscillators to create the desired waveform for a sound.

Cat 5. Short for Category 5. A type of twisted-pair cable used for Ethernet networks and other communication applications.

Cat 5e. Short for Enhanced Category 5. Cat 5e is an improved version of Cat 5 cable, with better crosstalk specs.

Cat 6. Short for Category 6. A very high-quality twisted-pair cable used for gigabit Ethernet and other high-speed communication applications.

CBR. ☞ **See** *constant bit rate*.

CC. ☞ **See** *continuous controller*.

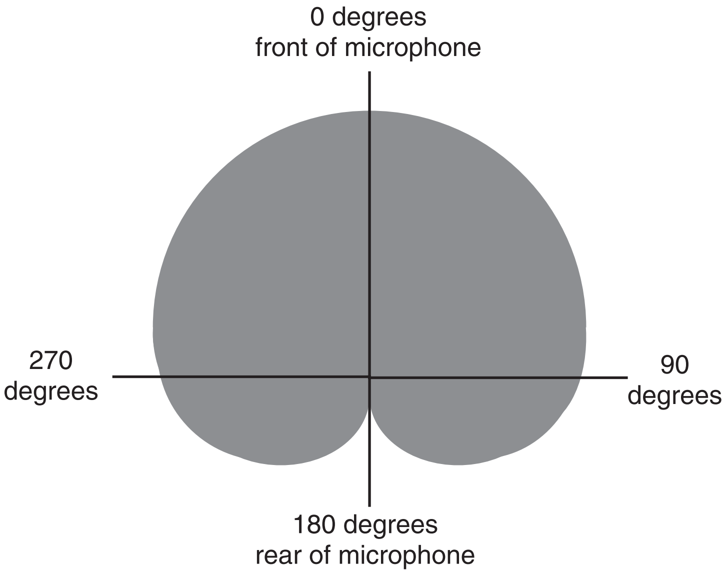


Figure C.1 A cardioid polar pattern is heart-shaped, picking up sound best from the front and rejecting sound from the rear.

CCIR. Consultative Committee for International Radio. A standards organization that created many of the measurement standards used for audio equipment. The CCIR eventually became part of the United Nations' ITU (*International Telecommunication Union*).

CC number. ☞ **See** *continuous controller*.

CD. ☞ **See** *compact disc*.

CD Extra. (a.k.a. Enhanced CD, CD Plus). A compact disc format that combines both Red Book audio and Yellow Book computer data on the same disc. ☞ **See also** *Blue Book*.

CD Plus. ☞ **See** *CD Extra*.

CD Text. An extended version of the Red Book CD spec that allows text data to be stored in the lead-in area of an audio disc. CD Text is used by some car CD players and other devices to display CD track names.

CD24. A 24-bit audio CD format developed by Alesis for use in the company's MasterLink CD/hard drive recorder and intended for recording and playing high-resolution master recordings from compact disc media.

CDDA. Compact Disc Digital Audio. The "formal" name for standard audio compact discs. ☞ **See also** *Red Book*.

CDDB. Compact Disc Database. A trademark of Gracenote, Inc., CDDB is an online database that can be accessed by media players and CD ripping software for artist name, CD title, track list, and other information. CDDB was intended to solve the lack of track names and other information on standard audio CDs. The database identifies a CD using calculations based on the track duration information in the disc's TOC.

CD+G. An extension of the audio compact disc specification that adds graphics content. CD+G discs will play in a standard audio CD player, or, when used in a CD+G-compatible player, will provide a graphics signal. CD+G discs are used for karaoke and other applications. ☞ **See also** *Blue Book*.

CD-quality. Digital audio at 16-bit, 44.1-kHz resolution. The term *CD-quality* does not have any meaning regarding the actual quality of the audio; it simply specifies the sample rate and bit resolution.

CD-R. Compact Disc-Recordable. Orange Book optical media that can be recorded by end users. CD-R media uses a dye layer into which "pits" are burned by a laser to represent data. Once the disc has been written, it cannot be modified. Once burned, a CD-R becomes compatible with Red Book, Yellow Book, and other format optical media players.

CD-ROM. Compact Disc Read-Only Memory. Compact disc optical media that conforms to the Yellow Book standard. CD-ROMs are used to store computer data.

CD-RW. Compact Disc-Rewritable, a.k.a. CD Erasable. A version of CD-R media that uses an alloy recording layer that can be melted and rewritten. CD-RW discs are not as reflective as CD-R or CD media and do not meet Red Book or Orange Book specs, so some older players may not be able to read them.

ceiling. The highest level that a limiter will allow a signal to reach. In digital recording and mixdown, the ceiling is often set just below digital 0 to provide protection from "overs" and clipping.

cent. 1/100 of a semitone. Pitch and frequency do not have a linear relationship, but cents and pitch do; for this reason, it is often clearer to use cents to describe tuning and other pitch relationships.

center detent. A notch that indicates the center position in a potentiometer's travel. Center detent pots are used for pan pots, boost/cut controls, and other applications.

center frequency. With a bell filter, the frequency to which the filter is set. Frequencies around the center frequency are also boosted or cut, creating the characteristic bell shape. See Figure C.2.

center section (a.k.a. master section). The part of a mixing console that contains all the "master" controls—subgroups, monitor controls, master aux sends and returns, master faders, and more. In larger consoles, the center section is in the middle of the channels, allowing easy access. On smaller consoles, the center section is often on the right side.

challenge/response. A type of copy protection in which the software is unlocked when the user inputs the correct answer to a question, or "challenge." The response is typically a code that is keyed to one specific install of the software. The user obtains the response from the manufacturer after registering the software. The advantage to challenge/response systems is that no copy protection hardware (such as a dongle) is required. The disadvantage is that the software is useless if the user tries to install it and there is a problem getting the proper response—such as if the manufacturer has gone out of business.

channel. A signal path that deals with a single mono or stereo audio signal.

channel aftertouch. ☞ See *monophonic aftertouch*.

channel masking. In SP-MIDI, the process of a composer establishing the MIDI channel priorities for playback of a particular piece of music.

channel message. A category of MIDI messages that are sent over a specific MIDI channel and that are only intended for and only received by devices set to that MIDI channel.

channel pressure. ☞ See *monophonic aftertouch*.

channel separation. The amount of crosstalk or signal bleed between two channels. The higher the crosstalk, the lower the separation, and vice versa.

channel strip. 1. A single input channel of a mixing board. 2. A piece of outboard gear containing a microphone/line/instrument preamp, equalization, and sometimes compression, limiting, gating, de-essing, and/or other processing. 3. A plug-in containing equalization and dynamics processing, such as compression, limiting, and gating. See Figure C.3.

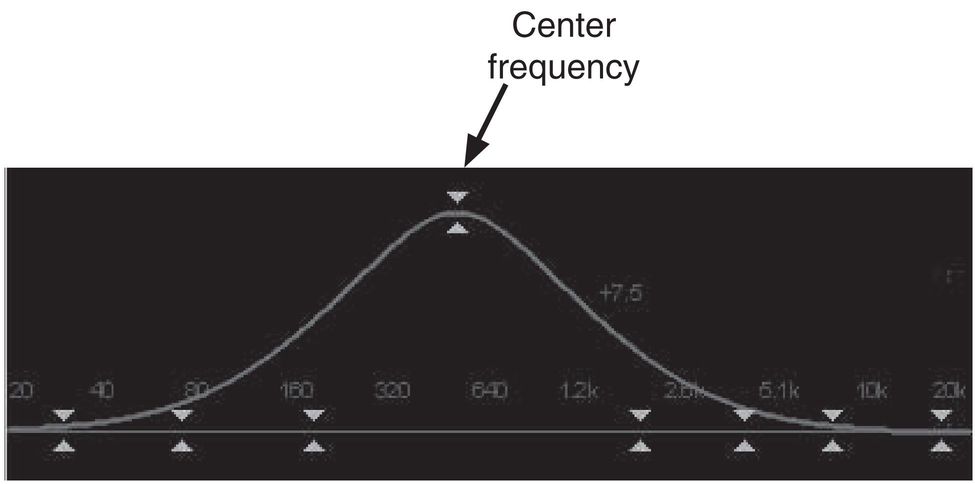


Figure C.2 Bell filters boost or cut frequencies around a center frequency.

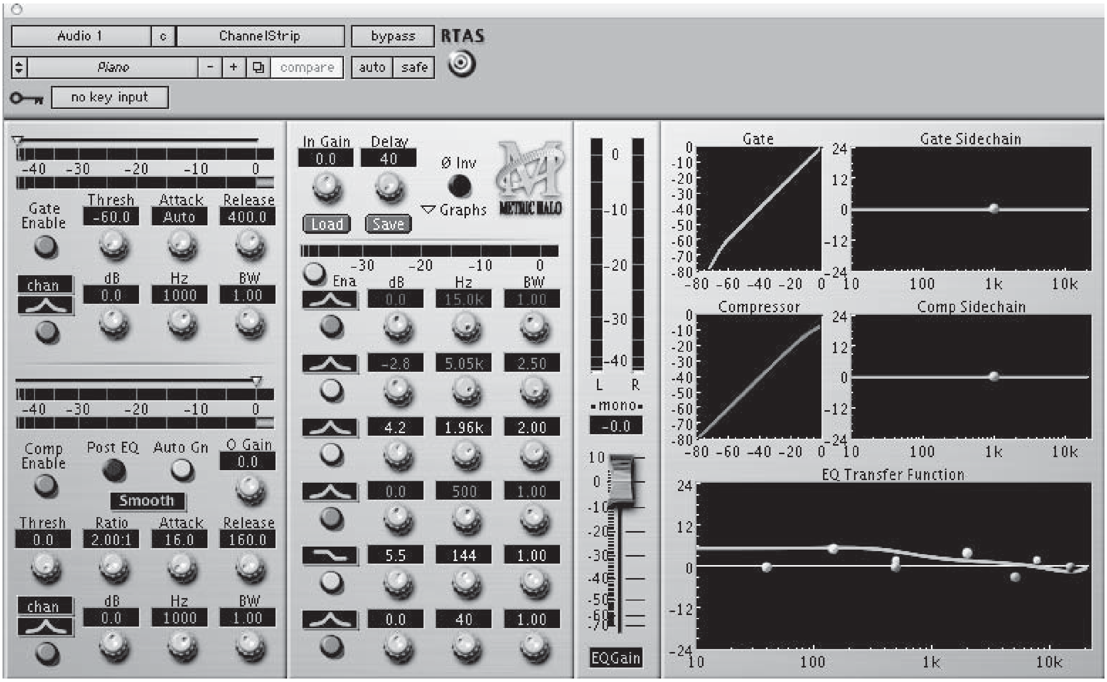


Figure C.3 A plug-in channel strip contains a variety of processing in a single unit, such as EQ and dynamics processing—in this case, gating, compression, and equalization. A hardware channel strip would add a microphone preamp to the list of capabilities.

chase. To synchronize or "slave" a device so that it locks to or follows a time-code master.

chatter. When a noise gate is presented with a signal level slightly above or below its threshold, the gate may rapidly jump back and forth between open and closed states, causing the audio to cut in and out, or chatter.

chip. Short for microchip. An integrated circuit made from a semiconductor material.

chipset. Literally, a group of chips that are designed to work together. In a computer, the chipset is a group of chips that defines the functions of the CPU and controls communications with external devices.

chorus. 1. A "refrain" or repeat of an important section of a song. 2. An audio effect created by processing a signal through a number of short modulated delays, then mixing the delayed versions with the original dry signal. The result is a thick, rich sound with a lot of motion that simulates a number of voices or instruments simultaneously performing a part. A chorus-style effect can also be created by slightly pitch-shifting a signal, then combining the shifted version or versions with the dry signal.

CIRC. Cross-Interleaved Reed-Solomon Code. An error-correction scheme used in compact disc players. CIRC adds one parity bit to every three data bytes. This is sufficient to correct scratches up to 8.5 mm long in a CD surface.

circuit breaker. A reset-able safety device, similar to a fuse, used to protect electrical circuits from overload. A circuit breaker automatically flips open when the current in a circuit exceeds a specific maximum amount.

circumaural. Headphone ear cups that circle around the ears and rest on the sides of the head of the wearer. Most circumaural headphones are closed designs that provide good isolation from external sound getting into, and internal sound getting out of, the headphones.

class. A type of amplifier design that determines how the sections of an amp handle current flow. There are several types:

- **Class A.** A type of amp in which current flows all the time, whether it is a single-ended or push-pull design. Because current is always flowing, the amp responds very quickly to input signals, and crossover distortion is reduced. True Class A amps are expensive and inefficient to operate.
- **Class AB.** A type of amp that operates in Class A for part of the time and Class B for the remainder. Class AB is the most common class of amplifier; it offers good efficiency and response time that falls between Class A and Class B.
- **Class B.** A type of amp in which current only flows when there is input signal present. Because current must start to flow when signal appears, the amp has a slower slew rate and increased crossover distortion. However, Class B amps are inexpensive and efficient to operate.
- **Class D.** A type of amp in which the input signal is used to modulate an ultrasonic square wave, which is then low-pass filtered to create the amplifier's output signal, which is driven using a push-pull, or switching, output design. Class D amps are highly efficient and lightweight, making them ideal for battery-powered and portable applications. (Note: "D" does not stand for digital. Class D amplifiers are analog designs, with no digital encoding of the signal.)
- **Class G.** A type of Class AB amplifier that uses multiple power rails with different voltages on each. Whichever rail is closest in voltage to the output signal is used, making the amp very efficient and lightweight. The disadvantage is increased distortion when the output signal falls between the rails.
- **Class H.** A Class G amplifier with modulated power-rail voltages, so that the rail is always within a few volts of the output signal, making the amp extremely efficient.

class compliant. A USB device that uses the "generic" MIDI or other drivers that are built into a computer's operating system and does not require proprietary drivers in order to function.

clean. The opposite of dirty. A signal that does not contain noise, distortion, or other undesirable artifacts.

click. ☞ See *tick*.

click and hold. A computer mouse technique in which the mouse pointer is positioned over an icon or file name, and the mouse button is pressed (clicked) and held down. Typically, this action is followed by dragging the icon or file while keeping the mouse button held down.

click track. A MIDI or audio track that provides a metronome-style "click" or percussion sound that serves as a tempo guide for recording the other parts in a piece of music.

clip. 1. To cut or delete data from a document or track. 2. Term used by some manufacturers for a region or segment of audio or MIDI data within a track.

clipboard. A memory space in a computer where data that has been cut or copied out of a document or file is temporarily stored. Typically, a clipboard only holds information from a particular program. This information can be retrieved and placed back into a document or file using the Paste command. When that program is shut down, the clipboard memory is emptied. Clipboard contents are lost when the computer's power is turned off as well.

clipper. A module in a synthesizer that clips or distorts the signal, producing a great deal of additional harmonic content.

clipping. A type of distortion that results from a signal's level or voltage being higher than an analog circuit's electronics can reproduce accurately or a digital converter can represent. The signal above the circuit's maximum level is literally clipped, or squared off, resulting in a great deal of harmonic distortion. In extreme cases, the clipped waveform will resemble a square wave. At very low levels, carefully controlled clipping distortion can be used to enhance signals, adding brightness and intelligibility. See Figure C.4.

clock. A timing reference used in digital circuits to specify a given number of cycles or operations in a period of time.

clock speed. The rate at which a clock operates, which drives how fast operations occur in a digital device.

close field. ☞ See *near field*.

close miking. Recording with a microphone placed very close to the source—from a fraction of an inch to a foot or so away. The advantage to close miking is that the microphone picks up only a very small

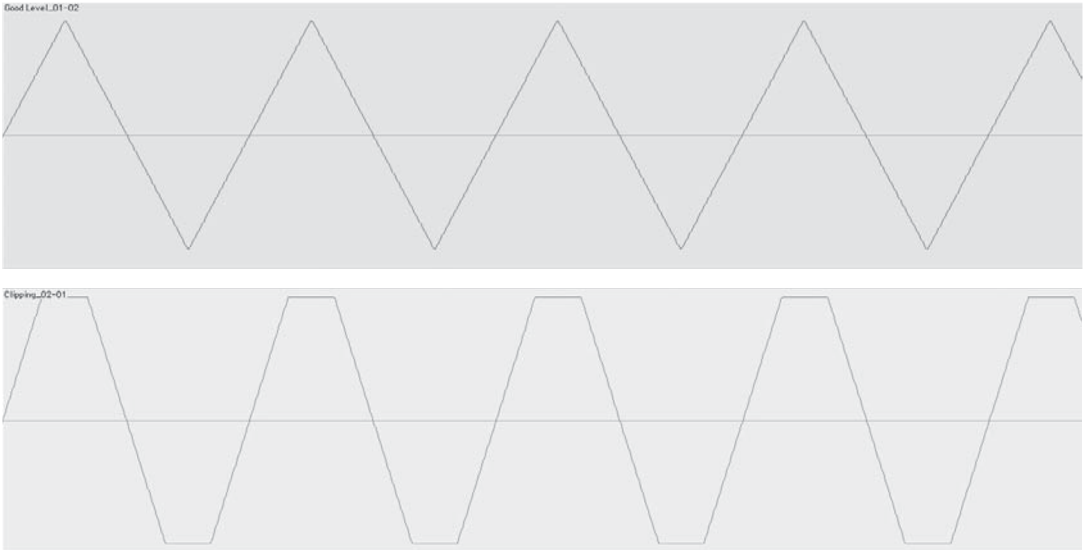


Figure C.4 Clipping occurs when a signal exceeds the maximum level that a circuit can pass without overloading. Higher levels will result in the top of the waveform being squared off and distorted (bottom).

amount of room ambience. (If the source is loud enough, essentially zero ambience will be captured.) Through careful placement, the proximity effect of directional mics can also be put to use for thickening a sound.

closed ear. A type of headphone with ear cups that are sealed to increase isolation from external sound entering the phones or internal sound escaping into the room.

cloud. Acoustical device suspended from the ceiling over the listening position to reduce first reflections. A cloud may be either an absorber or, in cases where the ceiling height is tall enough, a diffusor.

CLV. Constant Linear Velocity. A rotating disc moves at a different speed at the inside of the disc than at the outside. A CLV system speeds up or slows down the rotating speed of the disc so that surface velocity at any given point on the disc is moving at the proper rate as it passes a read or write head. All types of CD and DVD drives and players use the CLV approach.

CMOS. Complementary Metal Oxide Semiconductor or Complementary-Symmetry Metal Oxide Semiconductor. A type of low-power, low-heat integrated circuit commonly found in computers and other devices. CMOS chips are especially useful for battery-powered applications. For example, battery-powered CMOS chips are used in many computers to store the date, time, and system configuration.

CMR. ☞ **See** *Common Mode Rejection.*

CMRR. Common Mode Rejection Ratio. A specification that indicates the amount of cancellation that will occur at the input of a balanced system. ☞ **See** *Common Mode Rejection.*

coarse. The opposite of fine. Coarse refers to large increments of change or measurement, usually in reference to a control or parameter movement.

coaxial. A cable, speaker, or other item where two or more parts share the same center point. In a coaxial speaker, the high-frequency driver is mounted in the center of the low-frequency driver. In a coaxial cable, one insulated conductor runs through the center of another conductor. (Typically, the outer conductor is a grounded shield, and the inner conductor carries the signal.)

code. 1. A representation of information that must be translated or decoded to be understood. 2. Computer instructions or programming.

CODEC. Coder/Decoder or Compressor/Decompressor. A software algorithm or hardware component that encodes and decodes signals or information to, for example, create digital audio from analog audio (and vice versa), or MP3 files or other formats.

coercivity. A specification defining a particle's tendency to stay magnetized. Coercivity is an important spec for all magnetic media, such as analog recording tape, hard drive platters, and more.

coil. ☞ See *inductor*.

coincident pair. (a.k.a. XY). A stereo miking technique in which two microphones are placed so that their diaphragms are as close to in the same place as possible, usually by mounting one above the other (see Figure C.5). Cardioid-pattern mics are usually used, though other polar patterns could be used as well. The width of the stereo image can be controlled by the angle between the microphone capsules. ☞ **See also** *XY stereo*, *mid-side stereo*, *Blumlein pair*.

collapse. 1. To make a track or lane in a DAW smaller. 2. To close a program window.

coloration. Change in the timbre of a sound or signal.

com port. An input or output connection on a PC that is used for connecting external peripheral devices.

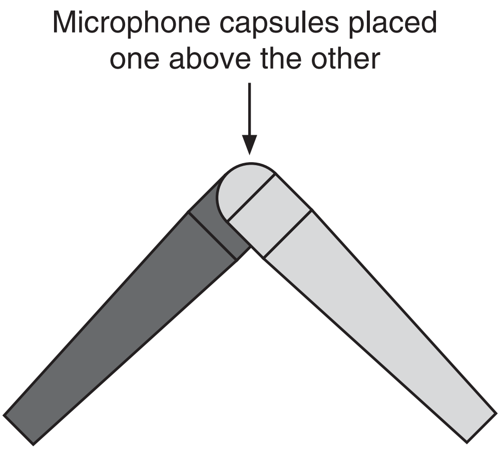


Figure C.5 Coincident microphones are placed so their diaphragms are as close together as possible.

comb filter. A type of device that produces comb-filtering effects.

comb filtering. A type of cancellation or filtering effect that causes a series of deep notches and peaks in frequency response, usually due to phase differences between sound waves. Comb filtering results in significant coloration of the sound. Comb filtering can be used as an effect, as in a phase shifter. Acoustical comb filtering, resulting from sound wave interactions in a room, can be problematic for making accurate recording or mix decisions.

combi. In some synthesizer workstations, a combi is a complete multitimbral setup, with all presets, MIDI channels, effects settings, and more.

combi mode. Term used by some manufacturers for the multitimbral mode in their synthesizers.

combination. ☞ See *combi*.

combination tone. ☞ See *sum tone*, *difference tone*.

combo jack. A type of connector that can accept either XLR or 1/4-inch connectors.

Common Mode Rejection (a.k.a. CMR). A technique used in signal transmission, in which two versions of a signal—one "normal" and the other with reverse polarity—are sent down two conductors in a cable. When the signals reach their destination, the polarity-reversed version is flipped to normal polarity and added to the non-polarity-reversed version. Reversing the polarity at the destination cancels out anything that was common to the two conductors, such as noise that was picked up during transmission. ☞ **See also** *balanced*.

comp. 1. Short for "composite." A technique for creating a track from the best parts of several different performances recorded to other real or virtual tracks. ☞ **See also** *composite track*. 2. Short for "accompaniment," typically playing rhythmic chords behind a solo or vocal.

compact disc (a.k.a. CD). A 120 mm (4.75-inch) optical media. Compact discs are read by lasers that follow a spiral track of pits in the sublayers of the disc. The discs are read from the inside to the outside; the disc rotates at 500 RPM when reading the inside part of the disc, and slows to 200 RPM as the laser reaches the outside of the disc. A variety of standards exist for using and formatting compact disc media, which are gathered into the Rainbow Books, a set of colored volumes, each of which contains the specification for a different standard. ☞ **See also** *CLV*.

CompactFlash (a.k.a. CF). A standardized data storage card type with no moving parts that was first introduced in 1994. CompactFlash cards use flash memory and have capacities ranging from 2 MB to 64 GB or higher. There are two categories, Type I and Type II, which are of different thicknesses (3.3 mm and 5 mm, respectively). There are four speeds: CompactFlash, CompactFlash High Speed, CompactFlash 3.0, and CompactFlash 4.0. CompactFlash cards retain their memory even when they are removed from a card slot or the power is interrupted.

companding. Compression/expansion. A noise-reduction technique in which the input signal is compressed before it is recorded or processed, and then expanded on playback or output. As the signal is expanded, any noise that has been added in recording or processing is pushed down and made less audible.

composite track. A track created by combining sections of several other tracks (see Figure C.6). ☞ **See also** *comp*.

compression. 1. An area of increased pressure caused by a sound wave; the opposite of a rarefaction. 2. Processing that reduces the dynamic range of an audio signal. ☞ **See also** *compressor*. 3. The process of reducing the size of a data file to save storage space or to allow for transmission to another device. ☞ **See also** *data compression*.

compression driver. A small speaker driver that mounts to the "throat" or small end of a horn. The driver produces the sound waves at high sound pressure levels, while the horn provides dispersion.

compressor. A device or plug-in that reduces the dynamic range of a signal. A threshold level is set.

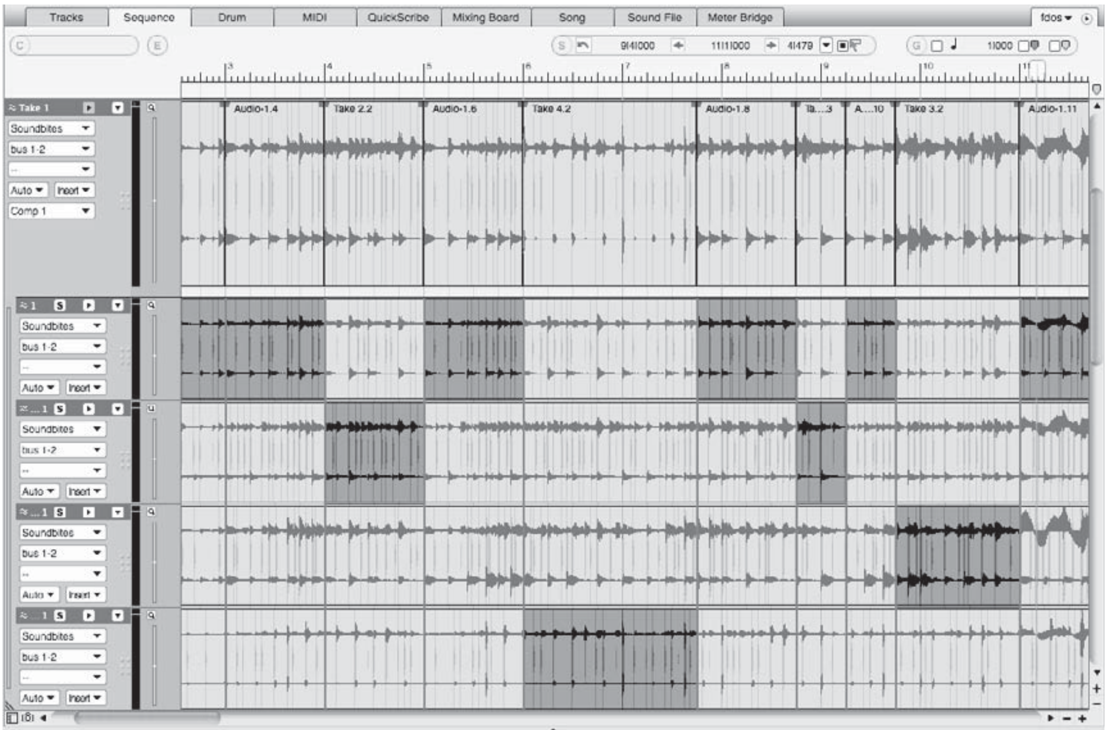


Figure C.6 Comping is the process of combining the best parts of two or more tracks to make a new composite track. In this example, the top track is the "comp," which has been created using the best parts of the four takes below.

When a signal level rises above the threshold, the signal level or gain is reduced by an amount defined by a ratio setting. The ratio determines how much the output signal is allowed to rise in level versus the input level. For example, a 2:1 ratio would allow the output to increase by one decibel for every two decibels the input level increases. A 5:1 ratio would allow the output to increase by one decibel for every five decibels the input level increases. Other controls determine how fast the compressor operates on a signal after it has crossed the threshold (attack) and how fast the compressor lets go of the signal after it drops back below the threshold (release). See Figure C.7.

compressor/limiter. A dynamics processor that can function as either a compressor or a limiter. The difference is how the parameters are set. The most important difference is the ratio, which is 10:1 or higher for limiting and less than 10:1 for compression.

concert pitch. An instrument-tuning standard that is referenced against the note A above middle C equaling 440 Hz.

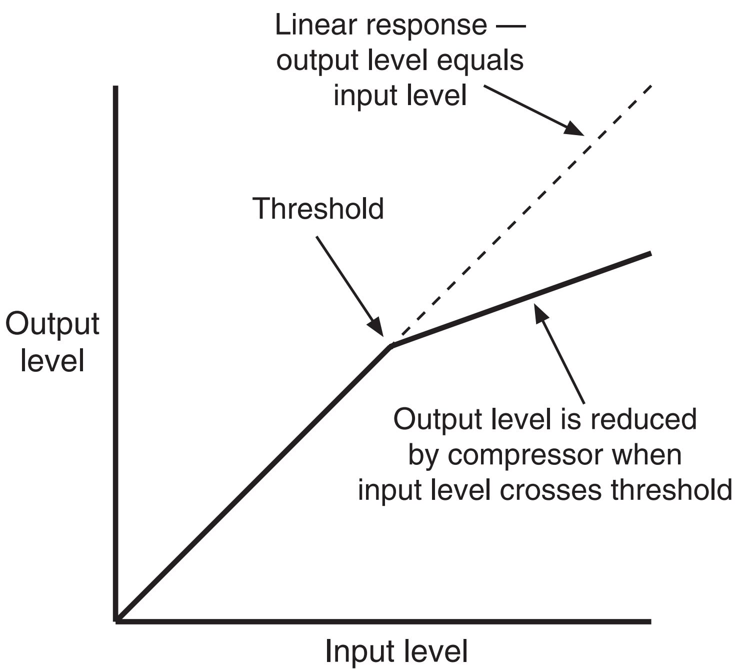


Figure C.7 A compressor reduces the gain or level of a signal that exceeds a threshold.

condenser. ☞ See *capacitor*.

condenser microphone. A type of microphone that uses a thin metal or metal-coated diaphragm that is positioned very close to a backplate. Together, the diaphragm and backplate form a capacitor that is either permanently charged or charged using an external voltage, such as phantom power. The diaphragm moves in response to sound waves, causing the distance to the backplate to vary and changing the capacitance and creating the output signal.

conductor. A material that will carry or transfer electrical current.

conductor track. A track in a DAW or sequencer that is used to hold a tempo map. Some DAWs or sequencers also store time signature and key changes in a conductor track.

cone tweeter. A type of high-frequency driver composed of a voice coil (typically 3/4-inch) suspended in a magnetic field that moves a small, light cone to create sound waves. Various materials are used for the cone, including ceramic, paper, treated fabric, stiffened silk fabric, and more. Cone tweeters are inexpensive but do not disperse sound as well as some other types, such as dome tweeters.

confidence monitoring. A system used in tape recorders (analog and digital) in which a separate playback head is placed after the record head to play back the audio a fraction of a second after it was recorded. This allows the engineer to hear exactly how the audio is being recorded. ☞ See also *repro*.

configuration. An arrangement or setup of components or devices in a system.

conform. To edit a soundtrack to fit a film or video.

console. ☞ See *mixing console*.

constant bit rate (a.k.a. CBR). Audio data compression algorithms that do not change bit rate regardless of the complexity of the input signal. The same bit rate is used for simple audio material as is used for complex material. In some cases, this rate may be too low to adequately represent the audio; in other cases, it may be too high, resulting in wasted space. Constant bit rates are better for transmitting data over a limited-bandwidth connection, as the rate can be set to stay below the maximum. For maximizing compressed file storage efficiency, variable bit rate algorithms (VBR) are often the better choice.

Constant Directivity. A trademarked term for a type of speaker horn that has equal SPL at all frequencies over the range of angles the horn is designed to address.

constant Q. In most equalizers, when a band is pushed up or pulled down, as the center frequency is boosted or cut, other frequencies around the center frequency are also boosted or cut. The more the center frequency is boosted or cut, the wider the band of frequencies that is affected, potentially causing interaction between adjacent bands. A constant Q design maintains the same bandwidth of affected frequencies no matter how much the center frequency is boosted or cut, preventing interaction between bands and allowing finer control. See Figure C.8.

consumer. Products intended for the general public. In audio, "consumer" generally refers to home stereo and home theater equipment, as well as keyboards, digital pianos, and other products aimed at home use. ☞ **See also** *semi-pro*, *professional*.

contact microphone (a.k.a. piezo microphone, transducer microphone). A microphone that creates electrical signals in response to mechanical vibrations carried through physical contact rather than sound waves carried through the air.

contact pressure. The amount of pressure headphone ear cups place on the sides of the wearer's head. Closed ear headphones typically have higher pressure to ensure a tighter seal. There actually is a DIN standard (Standard 45500 Part 10) for how much pressure headphones can exert.

content. The material that makes up an artistic work or production.

contextual menu. A computer menu that pops up or opens when an icon or other item is selected with a right-hand mouse button or by a combination of a modifier key and a mouse click. The options presented in a contextual menu will change depending on what is appropriate or possible for the item selected.

Continue. A MIDI System Real Time message that tells a sequencer to begin or resume play from its current location.

continuity. An unbroken signal path.

continuous controller. ☞ See *MIDI continuous controller*.

control panel. A small piece of software that provides setup options for, and control over certain aspects of, the operation of a peripheral device, program, or computer. See Figure C.9.

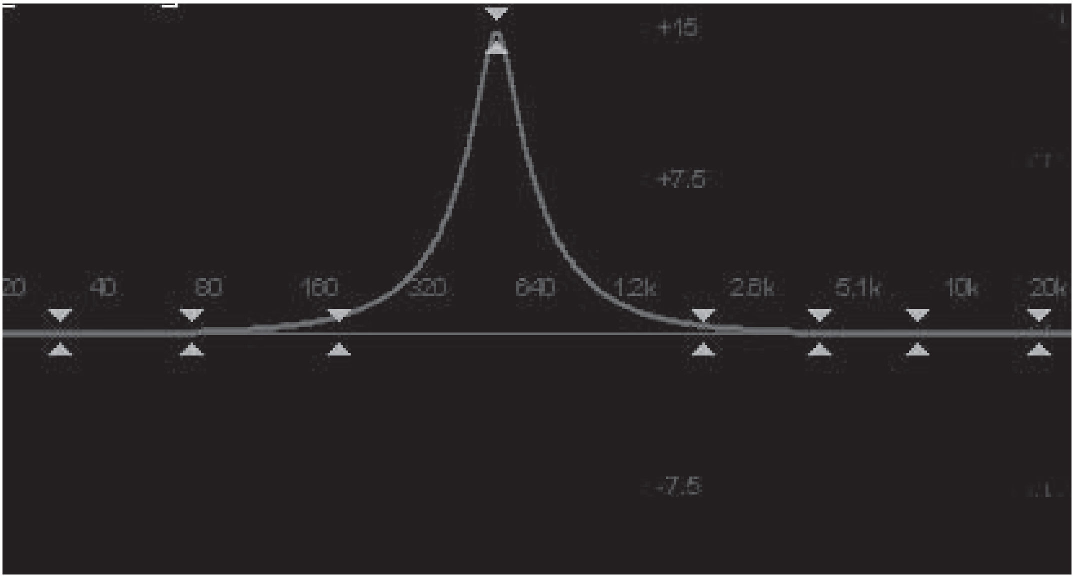
control room. The part of a studio where the recording and processing gear lives and the engineer and producer run sessions. Typically, the control room is acoustically isolated from the actual "studio" or live room or isolation booths where microphones are used to capture performances and sounds, though in many studios, the control room is also used for miking instruments and vocals and capturing sounds.

control surface. At its most basic, a control surface is a hardware "remote control" that provides manual access to functions, parameters, and settings within a piece of software. Control surfaces may resemble mixing consoles, but no audio runs through a control surface. Rather, the faders, knobs, and switches send out MIDI, OSC, or proprietary commands that are received by the computer and software. (In some cases, a control surface may have some monitor control and other audio capabilities built in.)

control voltage (a.k.a. CV). An electrical signal that is used to change the parameters in an analog circuit. Control voltages are commonly used in analog synthesizer modules, such as voltage-controlled oscillators, voltage-controlled filters, and voltage-controlled amplifiers, as well as in some effects pedals and VCA-based mixer automation systems.

Controller. ☞ See *continuous controller*.

controller keyboard. (a.k.a. keyboard controller, master controller, controller) A keyboard that usually does not contain any synthesis, sampling, or sound-generation ability, but is used as a MIDI controller that provides note and control information for MIDI software and hardware instruments. In



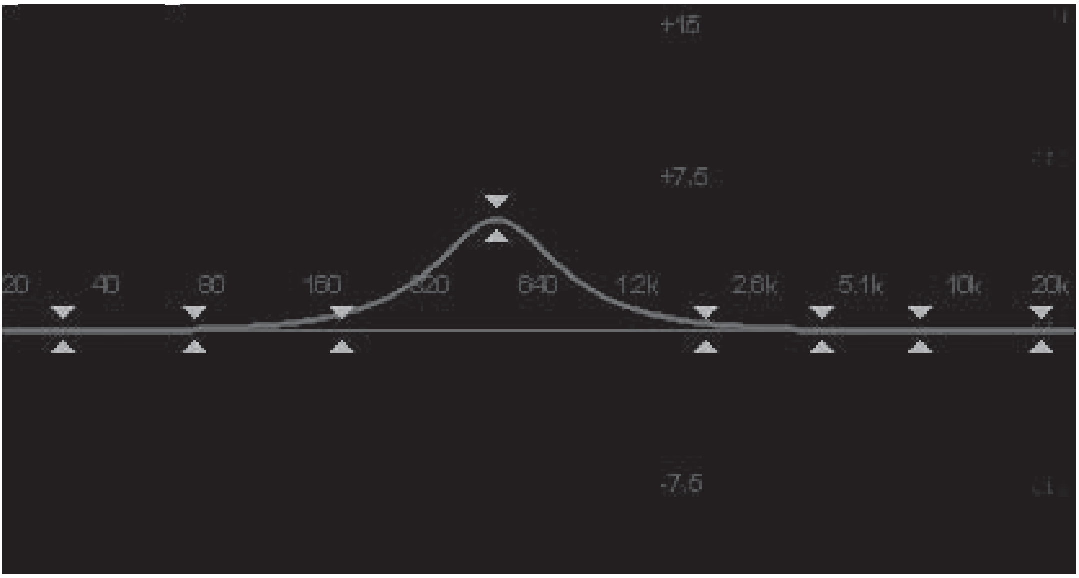


Figure C.8 A constant Q equalizer maintains the same bandwidth for each band, no matter how much boost or cut is applied.

addition to standard keyboard keys, a controller keyboard may provide sliders, knobs, and switches that can be "mapped" or assigned to control certain functions in the receiving devices. Controller keyboards are available in 25-, 37-, 49-, 61-, 76-, and 88-key lengths. In some cases a standard synthesizer or workstation keyboard can be used as a controller keyboard, or a synth expansion card can be added to certain controller keyboards to add sound-generation capabilities. Older controller keyboards

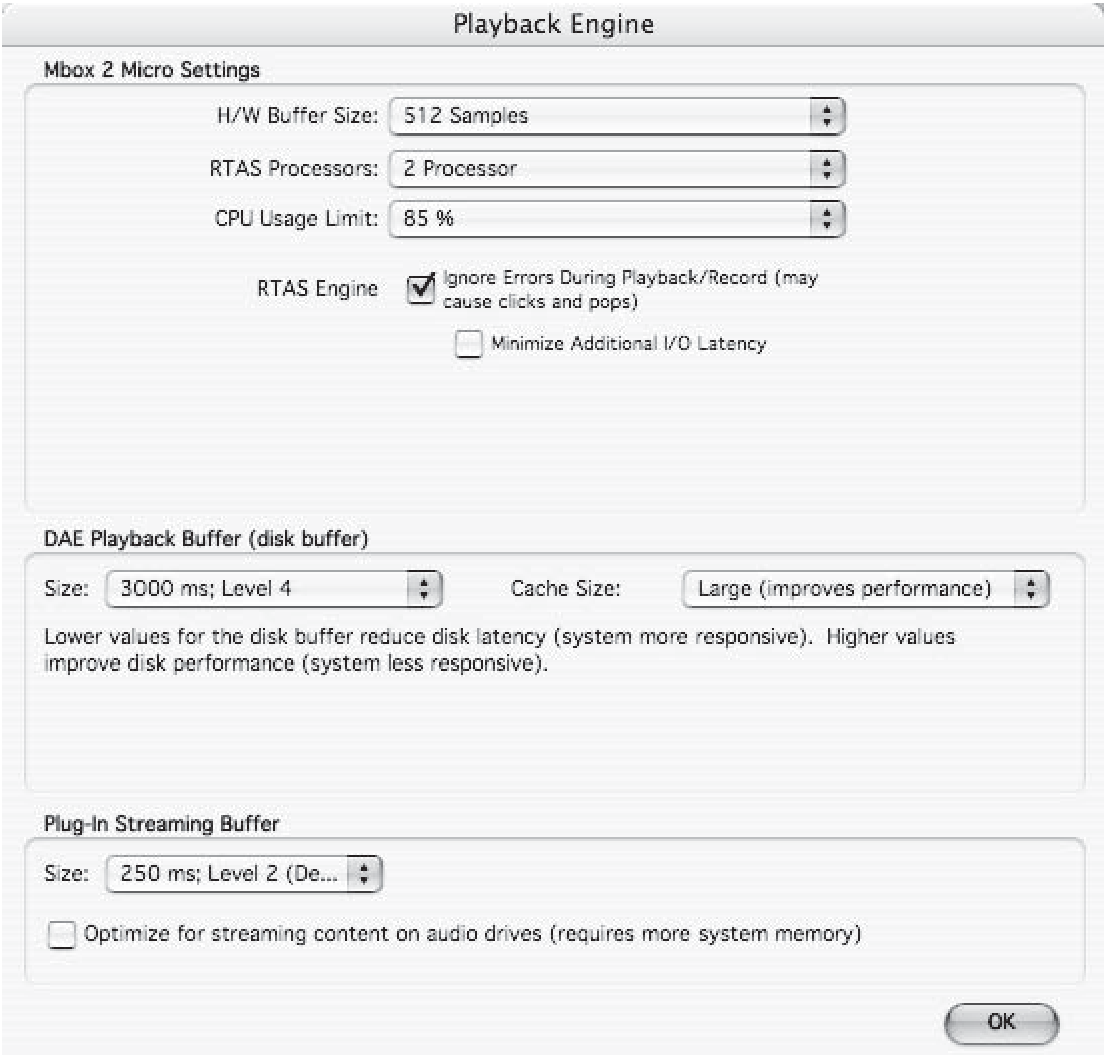


Figure C.9 A control panel for a piece of software or hardware contains setup and control options.

are connected via MIDI cables, while modern units can often be connected directly to a computer using USB.

conversion latency. The amount of time (generally a millisecond or two) it takes for a signal to enter an A/D or D/A converter and to emerge from the output after being converted.

converter. In audio, a device that converts analog signals to digital or digital signals to analog.

convolution. A digital audio process where one signal is used to modify a second signal. Mathematically, convolution is multiplying the first signal by the second signal in the frequency domain, creating a sort of filter. For example, with convolution reverb, an impulse—a sound file created by recording the reverb sound of a room—might be used to process another waveform, making it sound like the second sound was recorded in the ambience of the first room's reverb. Although creating realistic reverb effects is the most common use for convolution, any sound can be used to modify another sound. For example, a guitar speaker cabinet's response might be used to process a direct guitar sound. Or, the response of a specific microphone type might be used to change the tonality of a vocal—the possibilities are extensive.

convolution reverb. A type of reverb effect in which the reverb sound of a real room is used to process other sounds, creating very realistic spaces. ☞ **See also** *convolution*.

copy. 1. To make a duplicate of an item for backup, archival, or to be used for some other purpose. 2. A computer command that makes a duplicate of selected data and stores it to the clipboard.

copy protection. A hardware device or software code that is designed to prevent unauthorized copying or distribution of copyrighted software or other intellectual property.

Core. An Intel brand of computer processors. Multiple Core (multi-core) processors can be combined on a single chip (called a *die*), resulting in Core Duo and other processors. The first Core processors were intended for notebooks; later Core 2 processors were aimed at desktops and laptops. Several types have been created, some with multiple processors:

- **Core Duo.** A die containing dual computational cores and 2 MB of shared Level 2 cache.
- **Core 2.** A later version of Intel's Core processors, with 64-bit processing support.
- **Core 2 Duo.** Core 2 processor with dual computational cores and 4 MB of shared Level 2 cache.

Core Audio. 1. The audio subsystems developed by Apple for the Macintosh OS X operating system. Core Audio is designed to remove the responsibility for creating audio protocols from third-party software developers and provide a standard that those developers can access and build upon. Core Audio handles communication between the system and audio programs, with support for multiple channels and high-resolution audio. 2. Microsoft term for the audio protocols that are used in Windows Vista. ☞ **See** *Vista Core Audio*.

Core MIDI. The built-in MIDI protocols in Apple's Macintosh OS X operating system. Core MIDI is designed to remove the responsibility for creating MIDI protocols from third-party software developers and provide a standard that those developers can access and build upon. Core MIDI handles communication between the system and MIDI programs.

corner. The worst place to put a studio monitor. Placing a speaker in a corner, or half-space, causes a 6-dB boost in low frequencies, making the sound inaccurate.

corner frequency. ☞ **See** *3 dB down point*.

corrupt/corruption. A file, disc, or storage media that contains catastrophic errors.

COSM. Composite Object Sound Modeling. A modeling technology, developed by Roland, designed to emulate microphones, amplifiers, pickups, and a variety of other types of equipment and processing. COSM has been used in Roland's keyboards, guitar and effects processors, standalone digital recorders/mixers, and more.

couch. Marginally effective acoustic absorber well suited to supporting the posterior regions of humans.

count in. A bar or two of metronome, click track, or counting before a song starts that is used to establish the tempo and when the song should start. When used for a studio recording, the count in is muted or edited out of the track after the recording is completed.

count off. ☞ **See** *count in*.

counter. A display on a recorder or in a DAW or other audio software that indicates the current time position in the song. The time may be indicated in musical bars and beats, real time, time code, or samples. See Figure C.10.

coupling. 1. An audio source, such as a studio monitor, transmitting vibrations into another item, such as a desk on which it is resting. Decoupling, or isolating, monitors or other items helps maintain the accuracy of the sound. 2. Various methods for connecting electronic circuits.

Figure C.10 A counter in a recorder or piece of audio software that indicates the time position in a song. The time may be displayed in a variety of formats.

CPU. Central Processing Unit. A CPU is the microprocessor "brain" in a computer that controls all operations and functions.

crash. A condition in which a computer program or part of the operating system suffers an error or catastrophic event and stops functioning.

CRC. Cyclic Redundancy Code or Cyclic Redundancy Check. A type of error correction that uses a code derived from a block of data to verify the accuracy of a transmission.

CRIA. Canadian Recording Industry Association. The CRIA is an organization that develops standards and works to improve the Canadian recording industry. www.cria.ca.

critical band. In psychoacoustics, the band of frequencies that will mask a given signal.

critical distance. In acoustics, the distance at which the level of a sound source equals the level of reflections from surrounding surfaces.

cross modulation. A function of some synthesizers where an oscillator can be used to modulate another oscillator.

cross platform. A piece of hardware or software that is compatible with both Windows PC and Macintosh computers.

cross switch. A synthesis and sample programming technique in which different key velocities are used to switch between synth patches or samples. A low velocity might play one program, a medium velocity a second program, and high velocities a third. In sampling, cross switching is often used to switch between multisamples. A sound is sampled at a number of volume levels, then low velocities are assigned to trigger soft samples, medium velocities play medium volume samples, and high velocities play the loud samples. In some instances, cross switching may be assigned to trigger in response to some other controller, such as a footswitch or mod wheel. See Figure C.11.

crossfade. 1. An audio editing technique in which one region of audio in a track fades out while a subsequent region of audio fades in. Long crossfades are used to gradually transition between audio regions. Very short crossfades—so short as to be virtually inaudible—are used to smooth over click and pop sounds that can result from splicing two audio regions together. 2. A synth/sampler programming technique similar to cross switching, where different key velocities are used to switch between synth patches or samples. The difference is that the crossfade technique has a smooth transition between samples or synth sounds instead of an abrupt switch between sounds or samples. 3. A DJ technique, now used in a variety of applications, in which a slider control is used to smoothly fade one track out while another track is faded in.

crossover. A device that splits full-range audio into two or more independent frequency ranges. In sound reinforcement or other amplification/speaker applications, this allows, for example, the bass frequencies to be routed to their own amp and speaker, the midrange to be sent to its own amp and speaker, and so on. A crossover may also be used to split the audio range so that the separate frequency ranges can be processed independently. Heavy compression

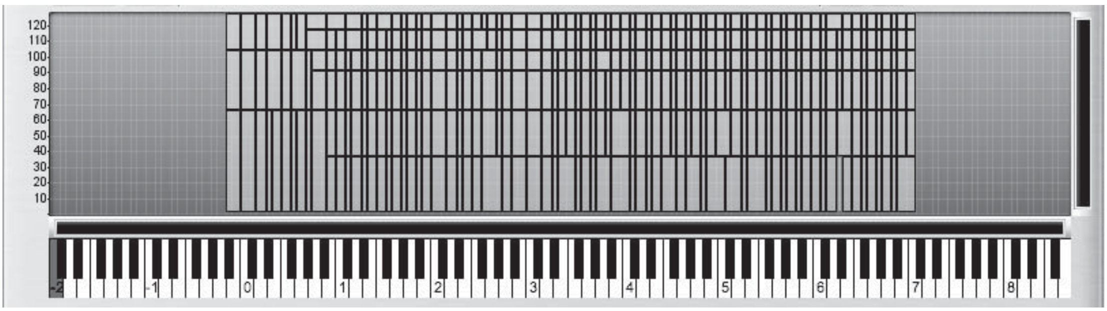


Figure C.11 The cross-switch function in synthesizers and samplers is used to trigger different programs or samples based on velocity values. In this case, a separate sample has been assigned to each of the key zones, and different samples are played for the zones based on the velocity of the key strikes.

might be applied to bass frequencies, light compression to the high frequencies, and no compression on the midrange. See Figure C.12.

crossover cable. Ethernet cables are usually wired so that a hub is required for proper communication. Without a hub, the connections end up reversed at one end. A crossover cable is an Ethernet cable specifically designed to be used to directly connect two devices together without using a hub or switcher.

crossover distortion. A type of distortion that is most prevalent in push-pull amplifier designs, where separate transistors or tubes are used to amplify the positive and negative portions of a waveform. Crossover distortion occurs at the transition point between the positive and negative components.

crossover frequency. The frequency or frequencies at which a crossover divides a full-range audio signal into two or more separate frequency bands.

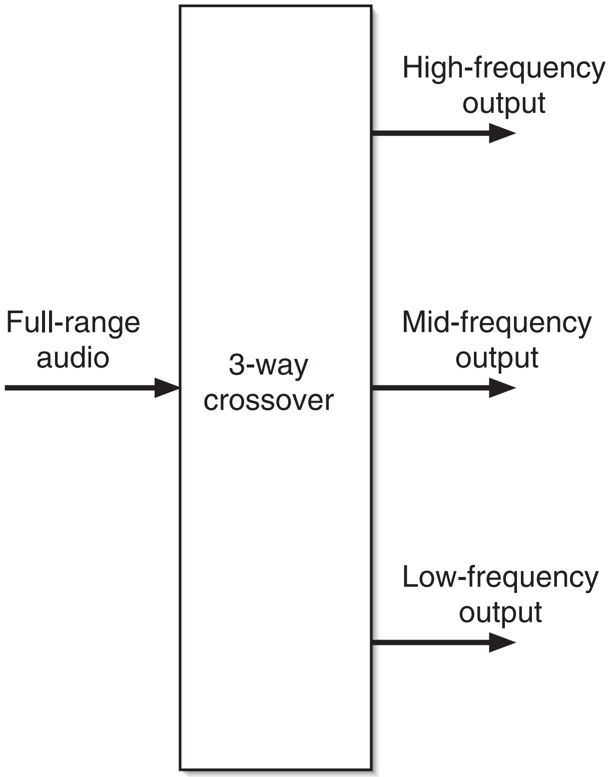


Figure C.12 A crossover splits full-range audio into two or more frequency bands so those bands can be amplified or processed separately.

crossover point. ☞ See *crossover frequency*.

crosstalk. Signal that "bleeds" or leaks through from one signal path to another.

CRT. Cathode Ray Tube. A video display technology used in televisions and computer monitors in which an electron beam "gun" moves across the inside of the screen, lighting up phosphor dots to create the picture. Lighter weight, more compact LCD, LED, and plasma screens have replaced CRT displays for most applications.

CS update. Customer Service Update. An interim software update between numbered versions of a program that addresses bugs and minor problems.

Csound. A free cross-platform audio programming language, based on the C computer programming language, designed for synthesis and sound processing. A piece of music in Csound uses two files—an orchestra file, which defines the sounds used, and a score file, which defines the notes and performance parameters.

cube. The worst possible shape for a recording or monitor space, with all the walls and the ceiling and floor configured as squares of equal dimensions. Since all three dimensions (height, length, width) in a cube are identical, the acoustic mode resulting from that dimension will be substantially worse than if the dimensions were different.

cue. 1. An indication to musicians when a particular musical event or gesture should take place. 2. A piece of music composed for a specific film scene. 3. To position a tape recorder, DAW, or LP to begin play at a specific point.

cue list (a.k.a. cue sheet). A listing of what happens when during a piece of music or a film. A cue list is often referenced against time code, though other time formats may be used in a DAW.

cue mix (a.k.a. monitor mix). A headphone mix sent to musicians during recording so they can monitor other musicians and tracks.

cue sheet. ☞ See *cue list*.

current. The flow of electricity produced by voltage, measured in amps.

current limiting. A protection function in some power amplifiers that prevents too much current from flowing if the load impedance drops too low. Current limiting controls the current without shutting it off completely, as a fuse would.

cursor. A position indicator on a screen usually controlled by a mouse in GUI operating systems. The cursor shows where user input will take place, such as a mouse click, text entry, an edit point, and so on.

cut. 1. A song or track. 2 The process of recording tracks. 3. A computer command that deletes selected data and stores it to the clipboard.

cutoff frequency. The point at which the response of a filter is 3 dB below the passband or un-attenuated level. ☞ **See also** *3 dB down point*.

cut-only equalizer. A graphic equalizer that is only able to attenuate the level of frequencies and is unable to boost frequencies. The signal passes unattenuated when the sliders are all the way at the top of their travel. Many cut-only EQs are passive devices. Some engineers prefer using cut-only equalizers—or using a regular equalizer only for cutting frequencies—in order to preserve headroom.

C-weighting. Using a flat-response, limited-bandwidth filter to obtain measurements that correlate better with how our ears hear.

cycle. One repetition of the positive and negative states or peak and trough regions of a waveform. See Figure C.13.

cycles per second (a.k.a. frequency, Hertz). Number of complete peak/trough cycles in a sound wave that occur in a second.

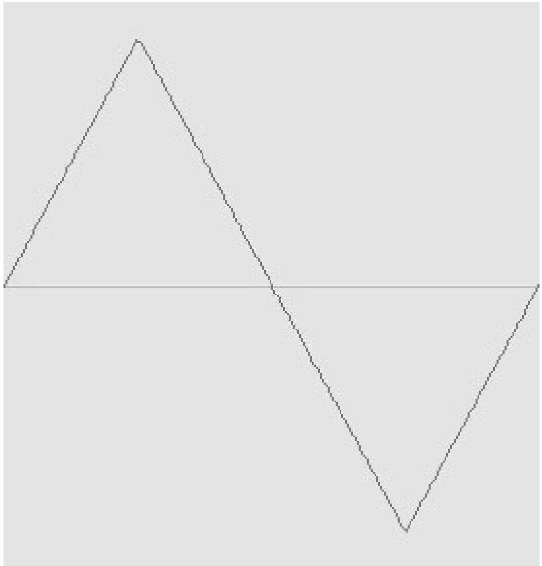


Figure C.13 A cycle is one complete positive/negative repetition of a waveform.

D

DA. ☞ See *distribution amplifier*.

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

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

DAE. Digidesign Audio Engine. An application that runs in the background on a computer and provides communication between audio software, such as Pro Tools, and Digidesign hardware and audio interfaces. DAE manages the transfer and processing of all the digital audio in the system.

daisy chain. A serial wiring arrangement where device one connects to device two, which connects to device three, which connects to device four, and so on. A MIDI system connected by wiring the thru of each device to the input of the next device is an example of a daisy chain. See Figure D.1.

damp. ☞ See *damping*.

damper. A weighted felt pad in an acoustic piano that mutes the string, or stops it from vibrating. When a key is played or the sustain or sostenuto pedal is pressed, the damper is lifted and the string is allowed to ring.

damper pedal. ☞ See *sustain pedal*.

damping. Muting a signal or reducing its vibrations in level. Examples of damping can range from a muffler on a drum head to acoustic absorbers in a studio.

damping factor. The ability of an amplifier to control unwanted speaker movement. Since a speaker's voice coil is a coil of wire in a magnetic field, whenever a speaker moves it actually generates a small amount of current, which feeds back to the amplifier. If the amp's impedance is low (damping factor is high), the motion of the speaker will stop quickly. Low-frequency drivers are most affected by damping factor. Other factors can affect an amp's damping factor, such as speaker cable resistance, a passive crossover between the speaker and the amp, and the resonant frequency of the speaker.

DASH. Digital Audio with Stationary Head. A stationary recording head technology used for early reel-to-reel large-format digital recorders from Sony and Mitsubishi. DASH recorders were available in 2-, 24-, and 48-track formats.

DAT. Digital Audio Tape, a.k.a. R-DAT. A rotating head, helical-scan, stereo digital audio recording format that used tape cassettes somewhat similar to video cassette tapes (though much smaller). The DAT standard supported four resolutions/sample rates: 12-bit/32 kHz, and 32, 44.1, and 48 kHz at 16 bits. Digital audio is recorded to DAT without any data compression or other processing, which made it a popular mixing and mastering format. Rackmount and portable field-recorder units were available for professional use, and a few models were released for consumer use as well (though consumer DAT was a complete flop).

data. Literally, information.

data compression. A wide variety of techniques for reducing the size of data files in order to conserve disk space or transmission bandwidth. There are two broad categories: lossless, which uses statistical algorithms to reduce file size without compromising data quality or accuracy, and lossy, or perceptual coding, which removes unessential data to reduce file size with limited impact on accuracy. There are a variety of types: JPEG (for graphic image files); FLAC, MPEG-2, MPEG-4, and AAC (for audio files); and many others.

data encryption. Encryption is a method of encoding or transforming information so that only those who possess the key can understand it. Data encryption is used to protect the privacy and confidentiality of information stored on computers.

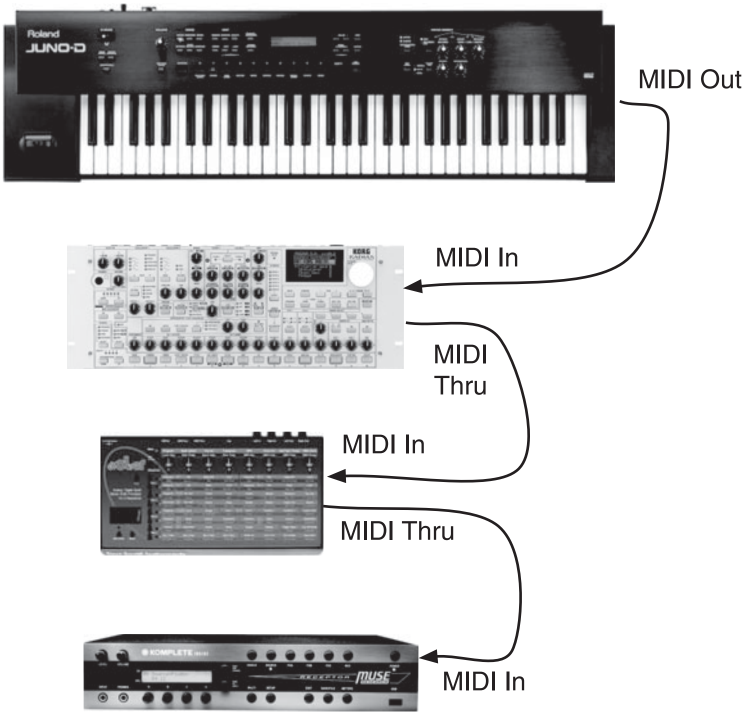


Figure D.1 MIDI equipment is often connected together in daisy-chain fashion, MIDI thru to MIDI input.

data spanning. A feature of some CD- and DVD-burning programs that can spread a data file that is too large to fit on a single disc across two or more discs.

data transfer rate. The average number of bits or blocks of data being transferred in a given amount of time. In other words, how fast data is transmitted between devices. ☞ **See also** *bit rate*.

DAW. Digital Audio Workstation. A digital device or system that can be used to record, edit, mix, and process audio signals (see Figure D.2). There are three basic components to a DAW: digital audio software, analog-to-digital and digital-to-analog converters, and a computer. In some cases, these components are integrated into a single standalone device. In other cases, a computer is used as the base platform for the system. Digital audio workstations were developed as early as the 1970s, but the technology really grew to widespread prominence in the late 1990s. Several of today's computer-based DAWs grew out of MIDI sequencing programs developed in the '80s and '90s, including Opcode Studio Vision, MOTU Digital Performer, Steinberg Cubase and Nuendo, Apple Logic, Cakewalk SONAR, and others. Digidesign Pro Tools, Sony Creative Software ACID, Ableton Live, and others were not based on sequencers, though they now include extensive MIDI features.

dB. ☞ **See** *decibel*.

dB SPL. Also known as dB, though this is an incomplete reference without the "SPL" identifier. A decibel reference for sound pressure level. All decibel "measurements" are actually ratios against a particular reference. In the case of dB SPL, the reference is the accepted threshold of human hearing, or 20 micro-pascals of sound pressure. At the other end of the scale, 130 dB SPL is considered the threshold of pain. The optimal listening level for studio recording and mixing is around 85 dB SPL, a level that allows for safe long-term listening and that also provides the most even frequency response from our ears. Note that decibels and sound pressure levels are related in a logarithmic fashion—this is because the span between the softest and the loudest sounds we can hear is so huge as to be unwieldy. Decibel ratios allow this huge span to be represented in a more compact way. See Table D.1.

DB-9. A 9-pin audio connector. ☞ **See also** *9-pin, D-sub*.

DB-25. A 25-pin audio connector. ☞ **See also** *D-sub*.

dBFS. Decibel Full Scale. A decibel reference that is equal to the maximum voltage level that can be sent into an analog-to-digital converter without clipping. Full scale, or full code, refers to all the bits in a

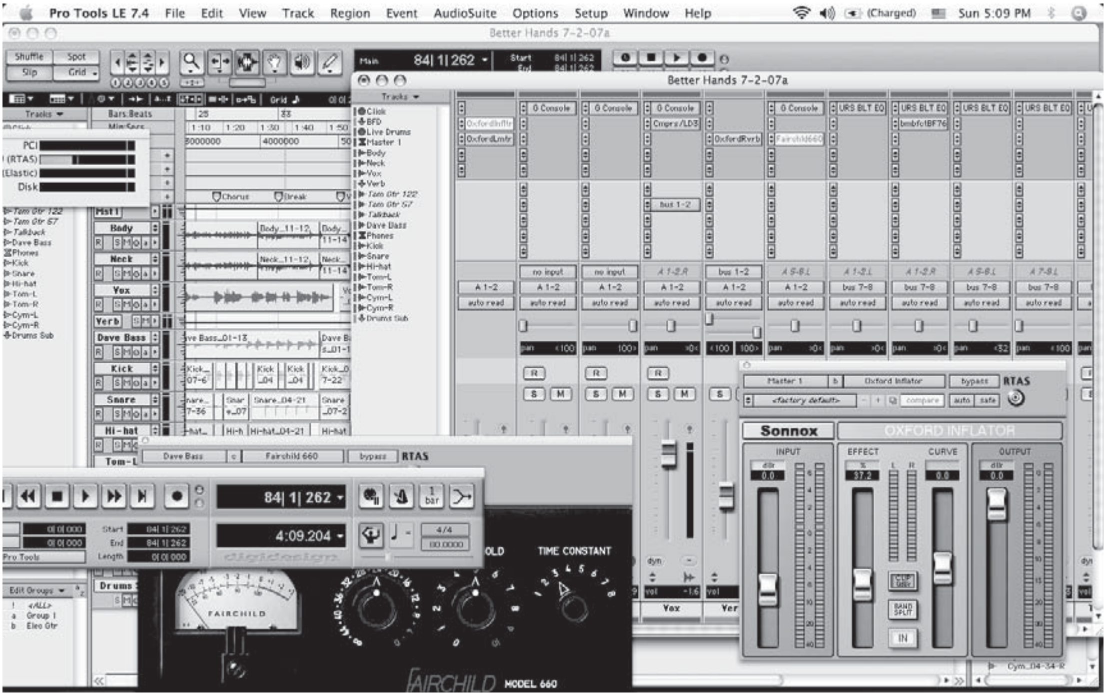


Figure D.2 A DAW can record, process, edit, mix, and play back audio.

digital word being utilized to represent a signal's level; any higher level will result in distortion. In most digital systems, 0 dB is the full-scale reference, though there may or may not be some headroom built in above this reference in the system. Note that 0 dBFS is not the same as 0 VU or 0 dB in an analog system. Different digital systems reference analog 0 dB to various digital levels, such as −18 or −14 dB. See Figure D.3.

dBm. Decibels referenced against one milliwatt of power, usually into a 600-ohm load. A value of 0.775 volts will produce one milliwatt at 600 ohms; this is the 0 dB reference point for power levels in mixers, recorders, and other audio equipment.

dBu (a.k.a. dBv [small "v"]). Decibels referenced against voltage—in this case, 0.775 volts—but without referencing an impedance load. At 600 ohms, 0 dBu is the same as 0 dBm, but many pieces of gear use impedances much higher than 600 ohms, so dBu usually ends up referenced against just voltage. In mathematical terms, the dBu ratio (all decibel ratings are ratios between two things) is calculated using 20log(V2/V1), where V1 is the 0.775 reference voltage, and V2 is the voltage for the dBu being calculated. For example, plugging in 1.23 volts, or 20log(1.23/0.775), gives a result of +4 dBu, the nominal input and output level for professional equipment. ☞ **See also** *dBV*.

dBV (large or capital "V"). A decibel reference similar to dBu, referenced against voltage without an impedance load. The difference between dBu and dBV is the reference voltage level. Where dBu references 0.775 volts, dBV references 1 volt. In mathematical terms, the dBV ratio (all decibel ratings are ratios between two things) is calculated using 20log (V2/V1), where V1 is the 1.0 reference voltage, and V2 is the voltage for the dBV being calculated. With a voltage of 0.316 volts, or 20log (0.316/1), we get −10 dBV, the nominal input and output level for semi-professional equipment. ☞ **See also** *dBu*.

dbx. 1. A manufacturing company that specializes in dynamics and noise reduction processors. 2. A type of analog noise reduction system that compresses signals as they are being recorded to analog tape,

Table D.1 Sound Pressure Levels

dB	Sound Source
0	Silence; threshold of human hearing.
30 dB	Totally quiet.
60 dB	Quiet room, normal conversation.
85 dB	Maximum recommended monitoring level for listening to audio for extended periods of time.
90 dB	Kitchen blender.
110 dB	Chainsaw at three feet.
120 dB	Front row at a concert.
130 dB	Threshold of pain.
150 dB	Jet engine at 100 feet.
190 dB	Point at which a sound wave is considered a shock wave, equal to 0 on the Riechter scale. It's a bad idea to try to get your guitar or bass amp, PA system, stereo, or studio monitors this loud!

then expands the signals as they play back from tape. The expansion process restores the compressed signal while simultaneously pushing down any noise originating on the tape. There are two varieties of dbx noise reduction. Type I is intended for wide-band recordings, while Type II has greater "pre-emphasis" in the high frequencies, making it better for video tracks, telephones, and other applications.

DC. ☞ **See** *direct current*.

DC offset. A constant direct current voltage in an audio signal (which by nature is alternating current, or AC) that results from component imperfections or imbalances in an analog-to-digital converter. DC offset is a problem because it reduces the available headroom for the signal and can result in clicks and pops when an audio file is edited.

DCA. Digitally Controlled Amplifier. An amplifier module in a synthesizer or sampler whose gain is controlled digitally rather than using analog voltages. This allows for much more accurate and fine control, with better repeatability. ☞ **See also** *VCA*.

DC-coupled. An analog circuit in which the components are directly connected together, without capacitors between them. This has advantages; it has better low-frequency response, there is no change in the sound over time since there are no capacitors to degrade, and there is a linear phase relationship between channels in multichannel devices, such as mixers. However, the lack of capacitors means there is nothing to protect the circuit from unwanted DC voltage in the signal, which can cause distortion.

DCF. Digitally Controlled Filter. A filter module in a synthesizer or sampler whose parameters are controlled digitally rather than using analog voltages. This allows for much more accurate and fine control, with better repeatability. ☞ **See also** *VCF*.

DCO. Digitally Controlled Oscillator. An oscillator module in a synthesizer or sampler whose frequency is controlled digitally rather than using analog voltages. This allows for much more accurate and fine control, with better repeatability and tuning stability. ☞ **See also** *VCO*.

DDCD. Double-Density Compact Disc. A CD format specified in the Purple Book. DDCD has a storage capacity of 1.3 GB, double that of a regular CD. This is achieved by increasing the density of tracks and pits used to store data in the optical media. DDCD was conceived as a stopgap to help delay DVDs making CDs obsolete, but it was never successfully launched to the public.

DDL. Digital Delay Line. A digital device that can capture a discrete copy of a signal and replay it after a certain amount of time has elapsed, creating an echo effect.

DDR. Double Data Rate. A type of DRAM computer memory chip that is similar to SDRAM but transfers data twice as fast.

dead. A space in which all or most acoustic reflections are absorbed. The ultimate dead space is an anechoic chamber, which is specially constructed

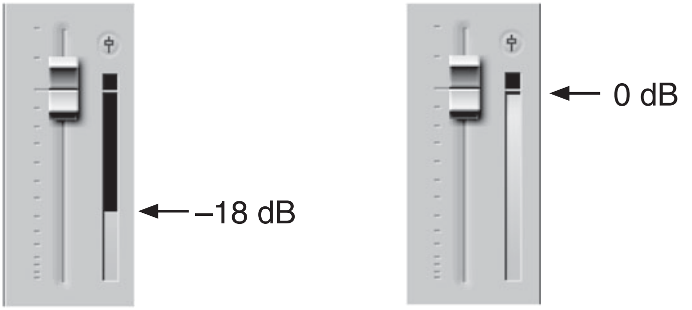


Figure D.3 A digital meter reads 0 dB as the highest possible level before overload (right), unlike a VU meter. A signal that reads zero on a VU meter typically reads −18 dB or −14 dB on a digital meter (left). Many experts recommend −18 as the best level for recording into 24-bit digital audio systems, as it allows plenty of room for clean transients.

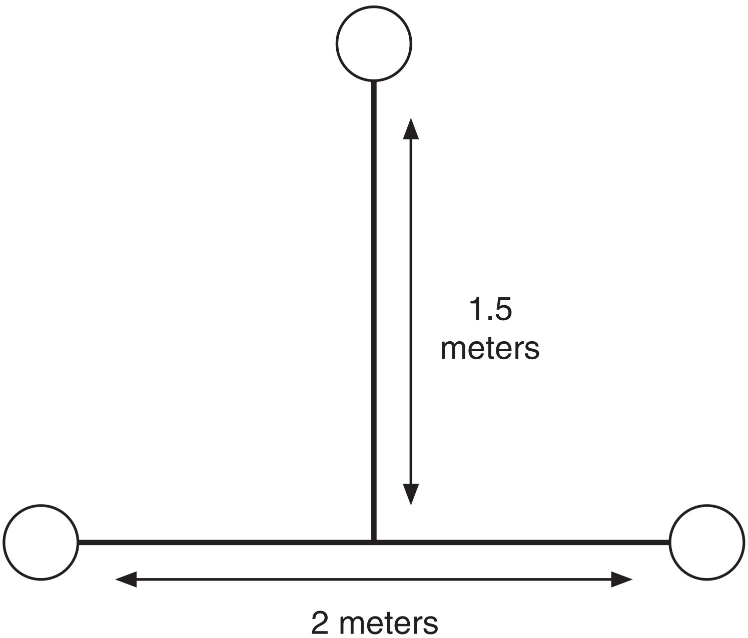


Figure D.4 Decca Records developed the Decca Tree, a three-microphone stereo technique using omnidirectional microphones placed in a T arrangement.

and treated to absorb all reflections at all frequencies.

decay. 1. The manner in which sound falls to silence. 2. A parameter in an envelope generator (for example, ADSR) that controls the drop from the attack's peak level to the note's sustain level.

decay time. ☞ See *reverberant decay*.

Decca Tree. A stereo-miking technique developed at Decca Records that uses three omnidirectional microphones. (Traditionally, Neumann M 50 small-diaphragm tube condenser mics are used.) The microphones are mounted to a T-shaped bar (although three mic stands could also be used), with two microphones spaced two meters apart on the sides of the T, and the third mic placed 1.5 meters in front of the side microphones and centered between them (see Figure D.4). The Decca Tree arrangement is most often used for orchestral and film-score recording. It provides a solid stereo image that holds up well under Dolby surround processing.

decibel. 1. One-tenth of a Bel. 2. The ratio between two audio levels. A decibel is an expression of the ratio between an audio signal and a 0 dB reference, and not actually a measurement of audio level. Because of the way in which our ears respond to volume, these ratios are logarithmic in nature. 3. The smallest volume change the human ear can perceive without a reference to compare against, in isolation. See Table D.2.

Table D.2 Decibels, Power, and Volume

Decibel Increase	Power Increase	Volume Increase
3 dB	2× power	
6 dB	4× power	
9 dB	8× power	
10 dB		2× volume
12 dB	16× power	
15 dB	32× power	
18 dB	64× power	
20 dB		4× volume

decimation (a.k.a. downsampling). A process that converts an oversampled digital signal into a digital signal at a standard sample rate by strategically dropping samples. For example, a 44.1-kHz CD sample rate might be oversampled at 64x, resulting in a sample rate of 2.8224 GHz. Decimation is used to reduce the rate back to a standard 44.1 kHz. This is a two-stage process. First, the signal is low-pass filtered to prevent aliasing, then it is downsampled or reduced in sample rate.

deconvolution. In convolution reverbs, a sweep, gunshot, or other impulse signal is used to excite an acoustic space so that the space's sound can be captured and used to create an impulse response, or IR. The IR is used to calculate reverb effects based on the actual sound of the space (a process called *convolution*). To use the impulse response, the sweep, gunshot, or whatever sound was used to excite the space must be removed from the raw captured sound of the space. The process of removing the sound is called *deconvolution*.

decorrelation. In audio, using delay, reverb, or other processing to convert an audio signal to stereo or multichannel surround. Decorrelation was originally a technique used by older THX home theater systems to stereo-ize the mono rear channel signal to feed both rear surround channel speakers. Stereo reverbs also use decorrelation to create differences between the two channels, resulting in a more spacious effect.

decoupling. Isolating an object, such as a speaker cabinet, monitor, or even a room's floor or wall, from its surroundings to prevent the transfer of vibrations or resonances.

de-esser. A dynamics processor that is set up to respond to high frequencies in order to reduce the levels of sibilants in a signal. (Controlling sibilants can be a creative choice to make a track sound better or a necessity to reduce high-frequency distortion.) A number of dedicated de-esser hardware and software units have been released, or any compressor with a sidechain input can be made into a de-esser by routing the audio signal to both the audio input and the sidechain input and boosting the high frequencies of the sidechain input using an equalizer. This makes the compressor more sensitive to high frequencies, particularly "s," soft "c," and other sibilant sounds. See Figure D.5.

default. The original, out-of-the-box settings for a device as it comes configured from the manufacturer.

de-install. ☞ See *uninstall.*

delay. 1. Also known as echo. A discrete repeat of a signal at some period of time after the original signal is played. 2. A device that captures a signal, holds it for a certain period of time, then plays it back to create an echo effect. 3. A side effect of latency, or the period of time it takes for a process to take place or for a signal to pass through a signal path. Latency-related delay is a problem when monitoring signals during tracking or overdubbing, or when performing live.

delay compensation. A function of some DAWs that coordinates the timing of signals or DAW tracks to compensate for the delays created by plug-in processing or other latencies. The idea is to restore the proper timing between those tracks or signals so they play back in sync with one another.

delay time. In an echo or delay effect, the amount of time that elapses between the original sound and when the delayed sound is heard.

delete. To erase a file from a hard drive, a quantity of text from a document, or an audio segment from a track, or to otherwise remove data from a document, session, or computer.

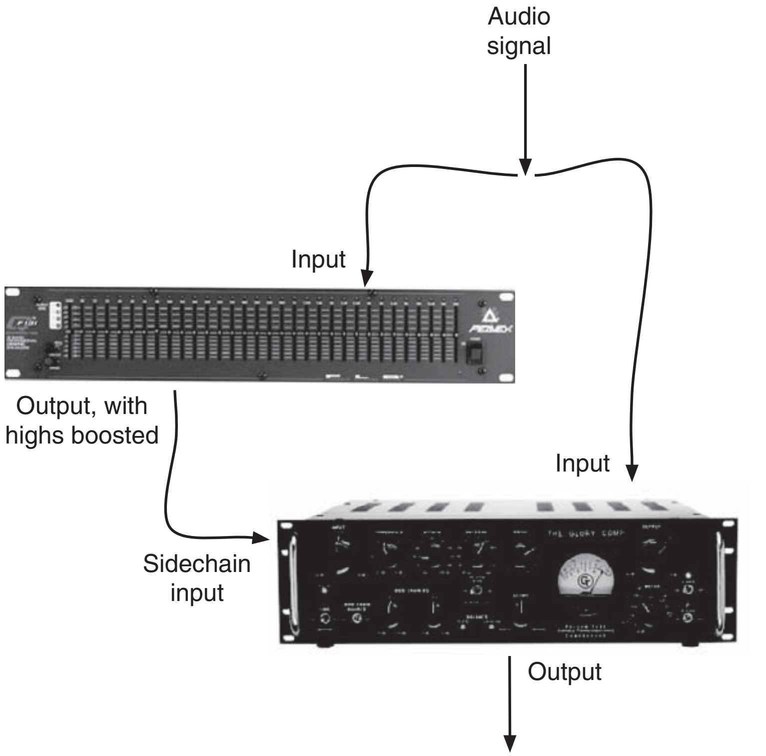


Figure D.5 To create a de-esser from a compressor, route a signal to both the compressor's audio input and its sidechain input. Use an equalizer to boost the high frequencies of the signal feeding the sidechain input.

demagnetizer. As analog tape is recorded or plays back, the magnetism on it tends to slightly magnetize the metal parts of the recorder it passes. Over time, accumulated magnetism on these parts can degrade the audio contained on other tapes as they are recorded or played. A demagnetizer is a device that removes any magnetism that may have built up on the heads and other tape machine components. Demagnetizers must be handled with care to effectively demagnetize the desired items without affecting things that should not be demagnetized (such as tapes and hard drives).

demo. Short for demonstration. 1. In music/studio terms, "demo" refers to a rough version of a recorded musical piece created to test or evaluate the piece or to use as an example of the work to play for others. 2. A product offered for sale that a retailer or manufacturer has used for demonstration purposes and that is therefore not in strictly new condition.

density. ☞ See *diffusion*.

desktop. A part of a computer's operating system graphical user interface analogous to the surface of a physical desk. The desktop is the basic screen "surface" displayed on the monitor screen. It can show background pictures and graphics; icons for hard drives, files, and programs; as well as other items.

destination. The parameter to which a modulation signal or message is routed.

destructive editing. Audio editing or processes that permanently modify or overwrite the original file. In many cases, there is no undo command or function to restore the original file after destructive editing takes place. The only way to get back to the original file is to make a backup copy of it first, before destructive editing takes place. Fortunately, most DAWs and audio programs offer non-destructive editing and processing that does not destroy the original audio data.

destructive recording. Audio recording or overdubbing that permanently modifies or overwrites a preexisting file. In many cases, there is no undo command or function to restore the original file after destructive recording takes place. The only way to get back to the original file is to make a backup copy of it first, before destructive recording takes place. Fortunately, most DAWs and audio programs offer nondestructive recording that does not overwrite older audio data when new data is recorded.

detector. ☞ See *sidechain*.

detent. A notch in the travel of a potentiometer or fader that indicates a certain setting, such as zero, the center position of a pan pot, or a nominal level.

detune. Slightly pitch shifting a signal up or down, usually by five or fewer cycles per second. The detuned, or pitch-shifted, version of the signal is often mixed back in with the original un-pitch-shifted signal, creating a chorus-like effect resulting from beating and very slight playback speed differences between the two versions of the signal. A common production trick is to pan the dry, un-shifted signal in the center, to pan a version pitch-shifted up by a few cents to the left, and to pan a version pitch-shifted down by a few cents to the right. This creates a wide, rich effect useful for thickening signals.

device aggregation. ☞ See *aggregate device*.

device driver. A small piece of software that allows software applications to access peripheral hardware devices connected to a computer.

DFD. Direct From Disk. A proprietary technology developed by Native Instruments for playing large samples directly from a hard drive without having to load them completely into RAM first. A small amount of RAM is used to pre-load or buffer the samples to compensate for hard-drive latency.

DI. Direct Injection. ☞ See *direct box*.

dialog. An alert or window that opens in a software application and that contains a message to, or requests an action or input from, the user.

dialog box. ☞ See *dialog*.

diaphragm. A very thin circular (other shapes are very rarely used) sheet of metal or metal-coated Mylar held under tension in a microphone capsule. The diaphragm moves in response to air-pressure changes resulting from sound waves. The diaphragm motions either move an attached coil of wire in a magnetic field or change the distance relative to a charged plate to create electrical signals analogous to the sound waves.

diaphragmatic trap. ☞ See *membrane trap*.

die. A small piece of semiconductor material containing an integrated circuit.

dielectric (a.k.a. insulator). A material that does not conduct electricity.

difference tone (a.k.a. combination tone). A frequency created under certain circumstances when two other frequencies are sounded together. The new frequency will be at the difference between the two original frequencies. For example, the difference in tone created by two tones at 500 and 300 Hz would be at 200 Hz.

differential. ☞ See *balanced*. The term *differential* is usually used in reference to computer cabling.

diffraction. In acoustics, diffraction is the tendency of sound waves with long wavelengths to bend around objects (to diffract) instead of reflecting off them.

diffuse. Scattered or spread out.

diffusion. 1. In acoustics, the process of breaking a single reflection into many smaller, lower-level reflections and scattering them in different directions. This reduces the intensity of the reflection and helps to reduce acoustic interference problems. 2. With artificial reverbs, a parameter that sets how dense or "thick" the reverb effect sounds. Also known as *density*.

diffusor (a.k.a. diffuser, though the "diffusor" spelling is more commonly used in acoustics). An acoustic device that scatters sound waves (see Figure D.6). ☞ See also *diffusion*.

digital. Signals or information stored and transmitted as series of discrete, non-continuous electrical impulses representing digits (numbers) rather than as a continuous analog voltage.

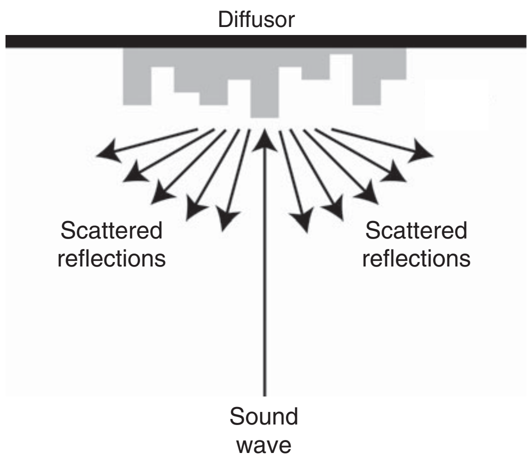


Figure D.6 A diffusor breaks up and scatters reflections to reduce acoustic interference problems.

digital 0. On a digital meter, 0 is the highest level possible, at the top of the meter. At 0, all of the bits for the digital signal are turned on, so any signal higher than that will be clipped or distorted (although some systems do scale digital 0 down a small amount to allow for some headroom). Note that digital 0 does not equal 0 as indicated by an analog VU meter. Various manufacturers calibrate their systems so that 0 VU equals −18 dB, −14 dB, or another level on the digital meters to allow for sufficient headroom when recording transient materials, such as drums. ☞ **See also** *dBFS*.

Digital Audio Workstation. ☞ See *DAW*.

digital black. In audio, a digital signal with no audio present or a volume of zero. In video, a digital signal with a pure black picture. The digital signal still carries all subcode and clock information; there is simply no "content."

digital synthesizer. A synthesizer that uses digital signal processing (DSP) to create sounds, rather than generating and modifying electrical voltages like an analog synthesizer. Digital synthesizers are available as standalone hardware units or as virtual software synthesizers that run on computers or in DAWs.

digital-to-analog converter (a.k.a. D/A, DAC). A device that converts a digital representation of an audio signal into an analog voltage.

dim. A function on many mixers that automatically reduces the monitor output levels when the talkback microphone is used. Some mixers also have a dim control that reduces the monitor level when pressed to allow for easier conversation in the control room. On some mixers, the amount of dim level is fixed; on others, there is a control for setting the dim level.

DIMM. Dual In-line Memory Module. A type of computer SDRAM memory expansion board that contains several RAM chips. A DIMM is essentially a double SIMM that provides a 64-bit data path. There are many versions, ranging from 72-pin to 240-pin, different form factors, and more, all available in a range of total memory capacities. ☞ **See also** *SDRAM*, *SIMM*.

DIN. Deutsche Industrie Normung, Deutsche Institute fur Normung, or Deutsche Industrie Norm. A standards organization in Germany. DIN standards are used for different plugs and connectors (a MIDI connector is a DIN connector), as well as a variety of noise, signal, and rack measurements.

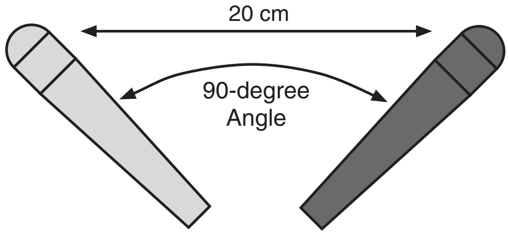


Figure D.7 With the DIN stereo mic technique, two cardioid microphones are positioned 20 cm apart, at a 90-degree angle.

DIN stereo. A stereo microphone technique where two cardioid microphones are placed 20 cm apart, angled at 90 degrees (see Figure D.7). Because cardioid microphones are used, the mics must be carefully placed in relation to the sound source to avoid inaccurate low frequencies due to proximity effect.

DIN sync. ☞ See *Sync 24*.

diode. An electronic component that only passes current in one direction. Diodes are used in power supplies and other applications. A special type, the LED (*light-emitting diode*) produces light when current passes through it.

dip. An area where cancellation of sound waves causes a decrease in level at a particular frequency or range of frequencies.

DIP switch. Dual In-line Package switch. A type of compact slider switch designed to be mounted on a printed circuit board inside a device. Often there are a number of switches in a signal DIP module, which are used to semi-permanently set up rarely accessed parameters that define the operation of the device.

direct box (a.k.a. DI). A device used to convert high-impedance unbalanced signals to low-impedance balanced signals. Direct boxes can be either active (using a powered preamplifier) or passive (using transformers), and may include other features such as ground lift, polarity invert, and others. In the studio, a direct box is used to connect an instrument-level signal, such as an electric guitar or bass, to a mixer or preamp microphone input. Live, direct boxes are used to run the outputs from instruments such as guitars and electronic keyboards through a snake to the front-of-house mix position.

direct field. Speakers or a sound source set up so that the listener primarily hears the direct sound, with few or no reflections and little reverb. ☞ **See also** *near field*.

directional. 1. Moving in a single direction. 2. A type of microphone that is sensitive to sound coming from one or more directions (such as in front of the mic) and not sensitive to sound coming from other directions (such as from the sides or rear of the mic).

direct out. An output connection on many mixers that routes a single channel out of the board. A direct out taps off the signal before it is bused to subgroups or master outputs (see Figure D.8). In some cases the signal is taken directly after the mic preamp, after the EQ, after the channel fader, or other points in the signal path. Direct outs are most commonly used to feed a channel's signal to a recorder or DAW input, although they can be used to send the channel anywhere desired.

direct radiator. A speaker that does not use a horn design for interfacing the driver with the air. Direct radiators can provide smoother sound than horn designs, but they are not as efficient and they lack the directivity and throw. For this reason, direct radiators are often used for studio and home use, while horns are often used for PA systems.

direct recording. Recording without using microphones, by routing signals from electronic instruments or from pickups on acoustic instruments straight into a mixer or recorder.

direct sound. Sound from a source that arrives at the listener's ears without reflecting off any surfaces.

DirectSound. A type of DirectX software developed by Microsoft for managing, capturing, and playing audio through soundcards in the Windows operating system. DirectSound provides a direct connection between audio applications and soundcard and audio interface drivers.

Direct Stream Digital (a.k.a. DSD). A technology jointly developed by Sony and Philips for recording and playing digital audio intended for the companies' jointly developed SACD format. Direct Stream Digital uses a 1-bit oversampling approach with sample rates as high as 2.8224 or 5.6448 GHz. Frequency response for Direct Stream Digital systems is typically 0 Hz to 120 kHz, with 120-dB dynamic range. ☞ **See also** *1-bit*.

DirectX. 1. A family of protocols developed by Microsoft for use in Windows audio, graphics,

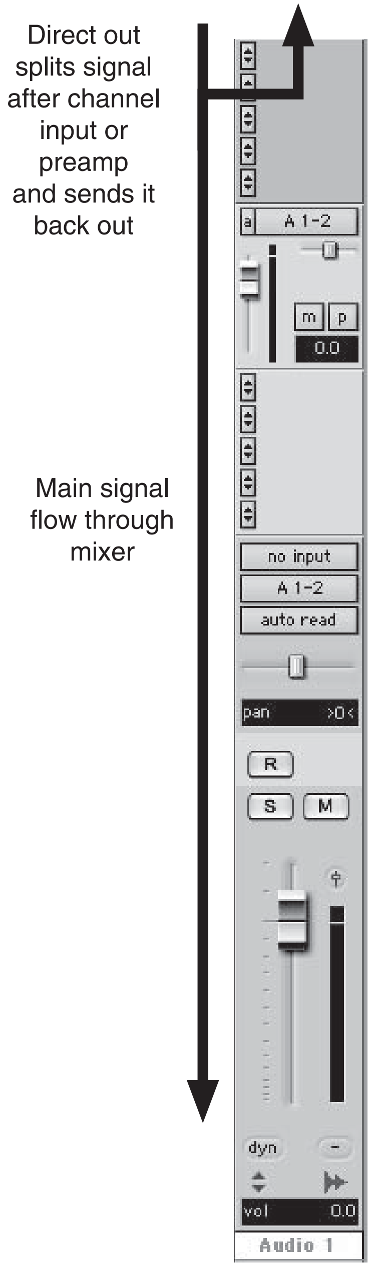


Figure D.8 Direct outs on a mixer tap off the signal before it is bused and send it back out to another destination, such as a recorder or interface input.

game, and multimedia applications. 2. In audio, DirectX is the name given to a format for plug-ins that can be used in real time within Windows DAWs and other audio software, or, depending on the plug-in, may work off-line (in non-real-time).

dirty. A signal containing harmonic distortion (intentional or not!) or noise.

disc. Optical storage media. ☞ **See also** *disk*.

disc-at-once. A Red Book compact disc burning mode in which the entire disc is written at one time, in a single pass, without turning off the laser. Burning a CD with disc-at-once prevents imperfections on the CD caused by turning the laser on and off. These imperfections don't cause problems for most CD players, but they can cause errors for a disc replicator if the CD is being duplicated for distribution. ☞ **See also** *track-at-once*.

discrete. Literally, separate. In electronics, discrete refers to using separate, individual components to create a circuit rather than using integrated circuit chips. With a discrete circuit, the designer can select each component, optimize interactions, and customize the final results, as opposed to using the pre-packaged design in an IC chip. Discrete designs generally are more expensive, require more room, take more time to create, generate more heat, and use more power, but they are considered by many musicians and engineers to have superior sonic quality.

disk. Magnetic storage media. ☞ **See also** *disc*.

disk cache. ☞ **See** *buffer*.

disk image. ☞ **See** *image*.

dispersion. 1. Scattering or distribution of sound waves as they travel from a sound source, such as a speaker. 2. The angle of coverage a speaker can produce. There are two dispersion characteristics for speakers: vertical and horizontal.

distant miking. Placing a microphone more than a few feet from a sound source. Distant miking will capture more of the room ambience along with the source, and with directional microphones generally will have more natural frequency response.

distortion. Literally, any change in a signal other than making it louder or softer. This would include equalization, compression, and other forms of processing. But in practice, distortion tends to be considered a negative or undesirable change in a signal's waveform.

distribution amplifier. (a.k.a. DA). A specialized type of amplifier used to split a signal so that it can properly feed multiple devices.

dither. In audio, dither is low-level noise intentionally added to a digital audio signal. At very low levels, a digital audio system can have difficulty determining whether the last few bits should be on or off, resulting in quantization noise. A small amount of noise—dither—can cause the bits to positively turn on and off, making softer audio signals clearly audible and fade-outs to silence smoother. Dither is usually used in the final stages of production, such as when reducing the bit depth of a signal from 24 bits to 16 bits for creating a final CD.

divergence. A control found in some surround panners that determines how much of a signal sent to the front goes to the left/right speakers and how much goes to the center channel. In practical terms, divergence determines the "width" of a center-panned signal in a surround mix.

DLL. Dynamic Link Library or Dynamically Linked Library. In Windows operating systems, a collection of software resources that is available to other programs. In Mac OS X, DLLs are known as *shared libraries*.

DLS. Downloadable Sounds. A specification developed by the MIDI Manufacturers Association that provides a format for loading samples into wavetable-based synthesizers. Because the sample files are small, DLS works well for multimedia and game audio, as well as for web audio, where fast delivery is required. There are several levels of DLS (Level 1, Level 2, and so on) that add features and different methods of synthesis.

DMA. Direct Memory Access. A feature that allows hardware devices (disk drives, sound cards, video cards, and so on) to directly access memory locations without going through the computer's CPU. In some cases, DMA is also used to transfer data among the memory locations within the computer. In most computers, DMA is arranged into channels, with each channel allocated to a device. ☞ **See also** *UDMA*, an enhanced version.

documentation. The manual and other instructional and information items that come with a piece of software or hardware.

Dolby. Short for Dolby Laboratories. A company founded by physicist Ray Dolby in 1965 that specializes in noise reduction and stereo and surround encoding hardware and software. Some of the formats created by the company include:

- **Dolby A.** The original system developed in the mid-'60s for film and analog tape noise reduction.
- **Dolby B.** The successor to Dolby A, Dolby B provides 10 dB of high-frequency noise reduction. There were versions available for professional use as well as consumer formats, such as analog cassette tape.
- **Dolby C.** Introduced for consumer cassette decks in the early '80s, Dolby C provides around 20 dB of high-frequency noise reduction.
- **Dolby Digital, a.k.a. AC-3.** A system for encoding or data-compressing 5.1 surround audio into less space than a single channel of CD-resolution audio.
- **Dolby E.** A broadcast and post-production-oriented system for encoding eight channels of audio plus Dolby Digital metadata for transfer over AES connections.
- **Dolby HX Pro.** A system introduced in the early '80s intended to increase the high-frequency headroom of analog tape recordings. It is not, as commonly assumed, a noise reduction system.
- **Dolby Pro Logic.** The successor to Dolby Surround, Dolby Pro Logic adds a dedicated center channel speaker and can decode surround information to left and right rear speakers.
- **Dolby S.** The consumer cassette deck version of Dolby SR, with 24 dB of high-frequency noise reduction and 10 dB of low-frequency noise reduction.
- **Dolby SR.** Dolby Spectral Recording. A professional system providing extensive noise reduction as well as enhanced dynamic range for analog recordings.
- **Dolby Stereo.** A professional system for encoding four channels of audio (left, center, right, and surround) on film releases for playback in movie theaters.
- **Dolby Surround.** The consumer/home version of Dolby Stereo.
- **Dolby Virtual Speaker.** A system that uses psychoacoustic processing to simulate a 5.1 surround sound speaker system using a pair of stereo speakers.

dome tweeter. A type of high-frequency driver composed of a voice coil suspended in a magnetic field that moves a dome-shaped diaphragm to create sound waves. Various materials are used for the dome, including titanium, beryllium, stiffened silk fabric, and more. Factors such as stiffness, low mass, and damping are important, as is lack of ringing or resonance. ☞ **See also** *tweeter*.

dongle. A hardware copy-protection key that attaches to a computer, usually using a USB port, and allows use of a piece of software. Without the dongle, the software will not run, or it may run in a limited demo mode.

Doppler effect. (a.k.a. Doppler shift). Named for Christian Doppler, a German physicist. The Doppler effect is the apparent change in the pitch of a sound as the source moves toward or away from the listener. Think of a car blowing its horn, and how the pitch goes up as it drives toward you, then falls as it drives away. The Doppler effect is the basis of how rotating speaker cabinets, such as Leslie speakers, create their effect.

DOS. Disk Operating System. The part of an operating system that controls and manages disk drives.

double. 1. ☞ **See** *doubling*. 2. ☞ **See** *double precision*.

double-busing. A feature found on some recording mixers that essentially sends the output of a subgroup or bus to a pair of duplicate outputs, sort of like a built-in Y cable. For example, on an eight-bus mixer, subgroup 1 would feed outputs 1 and 9, subgroup two would feed outputs 2 and 10, and so on. This allows the outputs of an eight-bus mixer to be permanently connected to 16 inputs on a recorder or interface, reducing the need for a patch bay.

double fast. A standard for using a single AES/EBU cable to carry stereo 96-kHz digital audio. The AES spec provides for stereo (two-channel) audio at sample rates up to 48 kHz on a single cable. The double-fast standard increases the data rate so that a 96-kHz stereo signal can be carried over a single cable. ☞ **See also** *double wide*.

double precision. A computer numbering system that uses twice as many bits to represent a number as a single-precision system. For example, a double-precision system would use 32 bits to store a number that would require 16 bits in a single-precision system. The extra bits increase not just calculation precision and accuracy, but also the resulting magnitudes that can be stored.

double tracking. Double tracking is recording the same part twice, to separate tracks. The two takes will not be exactly identical, so there will be timing and pitch differences between them. When the tracks are mixed, this results in a thicker, richer sound, or if they are panned opposite one another, a wide, full sound. ☞ See also *doubling*.

double wide. A standard for using two AES/EBU cables to carry stereo 96-kHz digital audio. The AES spec provides for stereo (two-channel) audio at sample rates up to 48 kHz on a single cable. The double-wide standard splits a 96-kHz stereo signal so that each AES cable carries half the data. ☞ See also *double fast*.

doubling. 1. ☞ See *double tracking*. 2. Using a short delay effect to simulate the effect of double tracking. Delay times are in the 10 to 50 ms range, short enough to not be heard as a discrete echo. A track is recorded, then during mixdown, sent through the delay. The delayed and original signals are mixed together or panned opposite one another in the stereo field.

downsample. To convert a higher sample rate to a lower sample rate. ☞ See also *decimation*.

downward expansion. Expansion is the opposite of compression, in this case applied as a form of noise reduction. When a signal falls below a certain threshold level, downward expansion pushes the signal down by a certain amount, set as a ratio. For example, with a 1:3 ratio, for every decibel the input signal falls below the threshold, the expander will push the output signal down by 3 dB. By carefully setting the threshold and attack and release parameters so that desired signals are not affected, unwanted noise between sounds, such as hiss and hum below the threshold, can be reduced.

drag and drop. A technique used in computers with GUI (*graphical user interface*) operating systems, where the user positions the mouse pointer over an icon, clicks the mouse button and holds it down, drags the icon to another location (usually a window or an application icon) by moving the mouse while keeping the mouse button pressed, then releases the mouse button.

drain. A bare wire inside a shielded cable that is used to more easily connect the shield to ground or to a component.

DRAM. Dynamic Random Access Memory. A type of RAM that uses capacitors in an integrated circuit chip to store bits of data, one bit per capacitor. DRAM differs from SRAM (*Static RAM*) in that the charge in the capacitors fades and must be refreshed after a certain period of time. The advantage is that DRAM is simpler and more compact per bit than SRAM. DRAM chips are combined onto circuit board modules, such as SIMMs, DIMMs, and other types, to make installation easier.

drawbar. A sliding bar on a Hammond organ that controls the volume of a particular tonewheel. Sliding the drawbar out increases the volume, while pushing it in lowers the volume of its tonewheel. By adjusting the blend of the tonewheels using the drawbars, different tonalities can be achieved.

drive. 1. Short for hard drive. 2. The amount of gain applied to a signal.

driver. 1. The element of a speaker or monitor that creates sound waves. 2. Software for interfacing peripherals with a computer. ☞ See *device driver*.

drop frame. A system for correcting for the fact that color NTSC video has 29.97 time code frames per second. To compensate for the missing 0.03 frames per second and to make synchronization between devices work out properly, two frames are dropped from the time code every minute, except for every tenth minute.

drop-down menu. ☞ See *pull-down menu*.

dropout. An area of an analog tape that contains no signal. Dropouts can occur for a variety of reasons, including tape inconsistency and the shedding of magnetic material from the tape.

drum grid. A graphic editor window in a sequencer or DAW that is designed to make programming drum and percussion rhythms faster and easier. The vertical axis of the grid represents the individual drum instruments, while the horizontal axis represents time. Notes can be inserted and deleted from the grid to create rhythms, and often the velocity and other parameters of the notes can also be adjusted graphically. See Figure D.9.

drum machine. An electronic musical instrument designed to create drum and percussion sounds through synthesis or sample playback, and to sequence or record and play back rhythms using those sounds. A drum machine often includes pads or buttons for manually playing its drum sounds or entering rhythms into the internal sequencer.

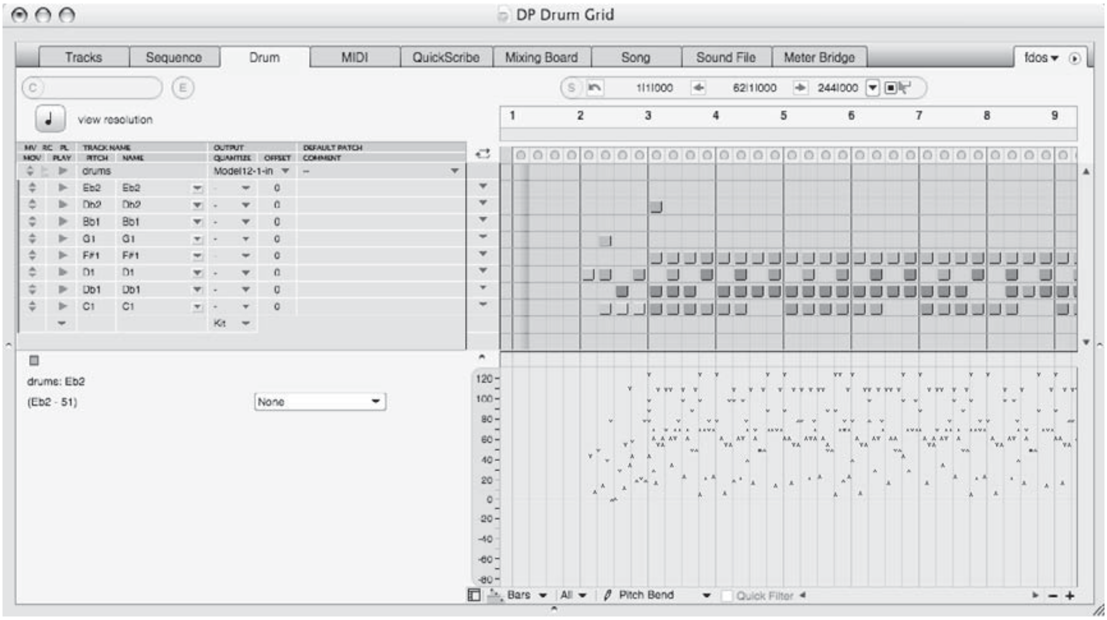


Figure D.9 A drum grid editor in a sequencer or DAW provides an easy way to program and edit MIDI drum parts—simply mark the square in the rhythmic grid where you want a hit to occur.

drum overheads. One or more microphones placed over a drum kit to capture the sound of the cymbals along with the sound of all the drums and a small amount of room ambience (depending on how high the mics are above the cymbals and drums). Typically, the overhead mics are mixed with individual mics on some or all of the drums to create a complete picture of the overall drum sound.

dry. A sound or signal without reverberation or other processing on it.

DSD. ☞ **See** *Direct Stream Digital*.

DSP. Digital Signal Processing. An algorithm designed to perform mathematical operations on various data types. Audio applications use DSP algorithms that have been designed to produce audio signals and to process digital audio signals in a variety of ways, such as with equalization, compression/limiting, mixing, reverb, and more. These algorithms might run on the computer's CPU, or they may be designed to run on DSP chips that have been added to the computer via an expansion card or in an external box.

DSP chip. Digital Signal Processing chip. An integrated circuit chip optimized for running DSP algorithms to process data.

D-sub. Short for D-subminiature, a.k.a. DB connector. A multi-pin connector type used on some professional audio equipment. The connector is roughly D-shaped and can carry a number of channels of signal. For example, a DB25 connector, with its 25 pins, can carry eight balanced signals. Other common sizes include 9-pin (commonly used for synchronization and control signals) and 50-pin. D-sub connectors and multichannel cables are used to save space, make cable routing easier, and help organize and group like signals.

DTRS. Digital Tape Recording System. A family of multitrack digital audio tape recorders developed by TASCAM in the early 1990s, including the DA-88, DA-38, DA-78HR, and DA-98. Along with Alesis ADATs, DTRS digital recorders were arguably responsible in large part for the price versus technology revolution that led to the rise of affordable high-quality home and project studios.

DTRS tape machines could record eight tracks with 16-bit (later models supported 24-bit) resolution to Hi-8 videotapes. The native sample rate for DTRS machines was 48 kHz, though varispeed could be used to achieve a 44.1 kHz sample rate. Multiple DTRS machines could be linked together for increased track count, resulting in the MDM (*modular digital multitrack*) system.

DTS. Digital Theater Systems. A company that developed the DTS digital 5.1-channel surround sound system for commercial and home theater use. DTS has also released a number of audio-only 5.1 surround sound music albums on encoded compact discs.

dual concentric (a.k.a. coaxial, point source). A type of speaker where the tweeter is mounted in the center of the woofer and the sound all radiates from a single point. Using a dual-concentric design causes the sound of all frequencies to arrive at the listener's ears simultaneously, reducing slight colorations sometimes heard with "regular" speakers.

dual core. Two independent computer processors mounted on a single chip. The advantages to doing this over having two separate chips include compact size, reduced power requirements, reduced heat, and the ability of the chip to handle two independent instructions per cycle. ☞ **See also** *core*, *core duo*.

dual diaphragm. A microphone that contains two diaphragms. By electronically combining the two diaphragms in different ways, various polar patterns can be created. Since the polar patterns are created electronically instead of acoustically, in many dual-diaphragm mics, the polar pattern is switchable, either using a switch on the mic or using a remote control on the mic's power supply. In rare cases, a mic will have two completely different diaphragms—for example, one large diaphragm and one small diaphragm that the user can switch between or use simultaneously.

dual-layer DVD. A DVD containing two layers that can each hold data, allowing nearly double the storage capacity of a standard single-layer DVD. One of the layers is semi-transparent to a great enough extent to allow a laser to focus through it onto the second layer.

dual mono. 1. A two-channel amplifier or processor that maintains complete separation between the two channels, essentially designed and constructed as two separate units inside one case. 2. Using a completely separate monophonic amplifier to drive each of the two speakers in a stereo system. 3. Using two microphones to capture different aspects of a sound source, resulting in two decorrelated signals or tracks. 4. Storing the left and right channels of a stereo audio track in two separate monophonic files on a hard drive.

dub. A copy of a tape.

ducker. A compressor that is set up to reduce the level of a signal or track in response to another signal or track. Typical applications include automatically turning down a background music track during a page or other announcement, dropping the background audio level during a radio commercial so the voiceover can be heard, and so on. A compressor is set up as a ducker by routing the signal that needs to heard—the announcer's voice, for example—to the sidechain input of the compressor, and the background music to the audio input or inputs on the compressor. When the announcer speaks, the sidechain will trigger the compressor and reduce the level of the background music.

ducking. Using a compressor to automatically turn down a track or signal so that another track or signal can be heard. ☞ **See** *ducker*.

dummy head. A stereo miking apparatus that uses two microphones mounted in the ears of a fake human head. The idea is to capture the audio as it would be heard by a human listener. The effect is best heard through headphones. ☞ **See also** *baffled stereo*, *binaural*.

duty cycle. In synthesis, the amount of time a square or rectangular wave is above zero (positive) versus the amount time it is below zero (negative). A 1:1 duty cycle indicates that the waveform is positive and negative for equal amounts of time. See Figure D.10.

DV. Digital video.

DVD. Variously, Digital Versatile Disc, Digital Video Disc, or nothing at all! DVDs are the same physical size as compact discs, but with 4.7 GB of storage capacity for a single-sided, single-layer disc and 8.5 GB for a single-sided dual-layer disc. There are three broad categories: DVD Audio (music), DVD Video (movies), and DVD-ROM (computer

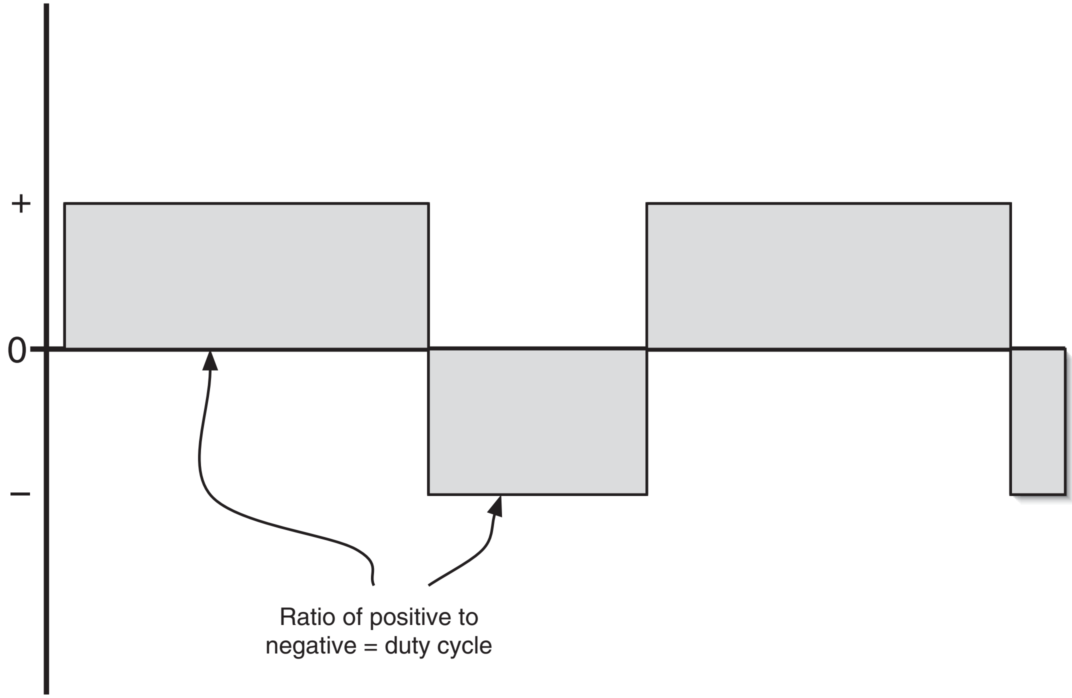


Figure D.10 The duty cycle of square or rectangular wave is how long the wave is positive versus how long it is negative.

software and games). Types and formats of DVDs include:

- **DVD-A.** DVD-Audio. A format developed for high-resolution and surround audio delivery. A DVD-A player is required to play back the disc. DVD-A can accommodate up to six channels of 24-bit/96-kHz audio plus a stereo mix of the six channels, or two channels of 24-bit/192-kHz audio. MLP (*Meridian Lossless Packing*) cuts the disc space required for each channel in half. There is also a video zone that can contain Dolby Digital audio tracks, allowing the disc to play in a standard DVD player.
- **DVD-R.** DVD Recordable. The DVD equivalent to a CD-R; a WORM (*Write Once, Read Many*) format for storing up to 4.7 GB of data.
- **DVD+R.** A similar format to DVD-R created by the DVD+RW Alliance. The only difference between the two formats is how the laser's position on the disc is determined.
- **DVD+RW.** A version of DVD+R that can be written and rewritten many times.
- **DVD-RAM.** The DVD equivalent of CD-RW. DVD-RAM is a high-capacity rewritable optical storage media for storing up to 4.7 GB of data for backup and archiving purposes.
- **DVD-ROM.** DVD-Read-Only Memory. The DVD equivalent of a CD-ROM. The DVD-ROM spec supports capacities up to 17 GB. DVD-ROM discs will not play in DVD video players; a DVD-ROM or DVD-RAM drive is required.

DVI. Digital Video Interface. A connection standard created by the DDWG (*Digital Display Working Group*) for displaying analog and digital video on a digital monitor, such as an LCD display. There are three types: DVI-A for analog video, DVI-D for digital video, and DVI-I for both types on a single connector.

DXi. DirectX Instrument. A DirectX-compatible virtual instrument plug-in format for software synthesizers and samplers.

dynamic. A process or characteristic that is constantly changing.

dynamic allocation (a.k.a. voice stealing). All electronic keyboards, synthesizers, and samplers have a finite amount of polyphony—only so many "voices" can be played simultaneously. Dynamic allocation is a system for "stealing" voices when the polyphony of the instrument has been exceeded. Instead of additional notes not sounding when polyphony limits are exceeded, the instrument cuts off an old note and gives that voice to the new note. Usually the oldest note is "stolen," though there are other schemes, such as stealing the softest note.

dynamic automation. Automation that supports continuous, real-time control over parameters. This allows for smoothly changing fader levels, EQ changes, and other parameter changes over time. ☞ **See also** *snapshot automation*.

dynamic equalization. An equalizer that responds to level changes in the input signal. Very few hardware equalizers offer dynamic EQ. Some plug-ins offer this feature, though they are more commonly referred to as *multiband compressors* or *limiters*. See Figure D.11.

Figure D.11 A multiband compressor is a type of dynamic equalizer that can dynamically change the levels of different frequency bands in response to input levels.

dynamic microphone. A type of microphone in which a thin diaphragm moves in response to sound waves. The diaphragm moves a coil of wire suspended in a magnetic field, creating an electrical signal. Dynamic microphones do not require phantom power. They are generally more durable than other types of mics and can handle high sound pressure levels without distorting, but they may not be as sensitive to high frequencies and transients as some other microphone types are. Dynamic microphones are used live for most applications. In the studio, dynamic microphones are used on electric guitars and basses, drums, some vocals, and other sources.

dynamic range. The ratio (in decibels) of the loudest to the softest signals a system can handle without distortion. In other words, the range of levels a piece of gear is capable of reproducing without distortion. In still other words, the difference between the noise floor and the onset of distortion.

dynamics. 1. Changes in the volume level and intensity of music. 2. Processors that manipulate the dynamics of audio signals, such as compressors, limiters, expanders, de-essers, and duckers.

E

ear buds. Compact transducers somewhat like headphones, but designed to fit inside the ears of the listener. Some ear buds, especially those used for live in-ear monitoring purposes, completely seal out external sound. Others, such as the low-end models included with many portable MP3 players, do not seal out external sound as well.

ear fatigue (a.k.a. tired ears). A condition that occurs after listening at high volume levels or for long periods of time. Ear fatigue may be as simple as lack of focus, or as significant as TTS (*Temporary Threshold Shift*), a change in the ears' response. Ear fatigue can lead to poor tracking and mix decisions. The best way to prevent it is to control listening levels (85 dB is the recommended listening level for accurate ear response and long sessions) and to take frequent "silence" breaks.

early reflections. 1. In acoustics, the first reflections to be heard after the direct sound from a source. Early reflections tell the listener a great deal about the size of a room. 2. In reverb effects, there are often two parts to the effect: the tail, or wash of ambience, and the early reflections, a group of echo-like short delays used to give the impression of being in a real room. See Figure E.1.

earth. ☞ See *ground*.

EASI. Enhanced Audio Streaming Interface. An older driver protocol developed by Emagic (now owned by Apple) that provides audio software applications with low-latency multichannel access to audio interface I/O. EASI bypasses the operating system audio functions to provide direct, high-speed communication.

EBU. European Broadcasting Union. An organization of broadcasters that develops cooperation and standards for the audio-visual community. www.ebu.ch.

echo (a.k.a. delay). 1. In acoustics, a discrete reflection of a sound wave arriving after the listener hears the direct sound. 2. In effects processors, a delayed duplicate of the original signal heard at some time after the direct sound.

Edac. The Canadian-manufactured version of Elco connectors.

Edison effect. A tendency of some materials to emit electrons when heated. This phenomenon was discovered by Thomas Edison when he was experimenting with light bulbs. The discovery of the Edison effect led to the later development of the vacuum tube.

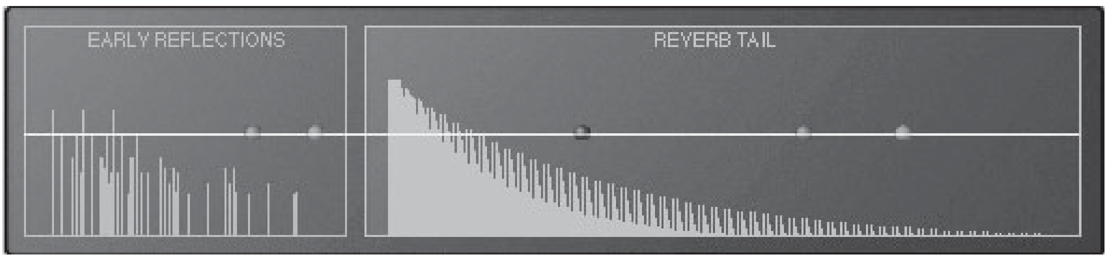


Figure E.1 In a reverb processor, early reflections are very short delay-like repeats that simulate the acoustic reflections heard in a room.

edit. To modify audio or MIDI data in some way. DAWs, MIDI sequencers, and audio editing software offer a wide variety of editing possibilities, including cut, copy, paste, transpose, quantize, trim, crossfade, and many more.

edit buffer. A temporary memory area in a synthesizer, sampler, or effects processor where presets are loaded to be edited. Changes to the preset are saved into the edit buffer. When the editing is finished, the contents of the edit buffer can be saved back to a user preset location.

Edit Decision List. ☞ See *EDL*.

editor. 1. A window in a DAW, sequencer, or audio software optimized for editing a particular kind of data or for viewing and editing data in different ways. 2. Software for editing audio recordings or samples, MIDI data, or synthesizer or processor presets. 3. A person who edits audio or other data.

editor/librarian. Many synthesizers, effects processors, controllers, and other devices offer very limited built-in programming interfaces—small screens, limited numbers of knobs and switches, and many layers and pages that the user must wade through to edit or create a preset. An editor/librarian is a piece of software that provides access to the parameters in a hardware device using MIDI System Exclusive messages. The software can display the parameters on the computer's screen in a visually friendly manner, make edits to presets, and store a library of presets for the device on the computer's hard drive.

EDL. Edit Decision List. A listing of the takes, cuts, edits, and changes that will comprise the master recording. EDLs are most common in video and film work but are also used by DAWs and CD mastering programs.

effect. A hardware device or software processor that modifies a signal. Examples include delay, echo, phase shifting, flanging, chorus, tremolo, and many others.

effects loop. An output and an input on a device, which are intended for routing signal out to an external effects processor then back into the device. Examples include the effects loops commonly found on guitar amplifiers, aux send and return buses on mixers, and insert points on outboard gear. Effects loops may be either serial, where the effects process the entire signal, or parallel, where the signal is split and one half is routed out to the effects and back in. See Figure E.2.

effects return (a.k.a. FX return). An input used to bring the output from an effects device into a mixer. ☞ See also *return*.

effects routing. The signal path in a multi-effects device that configures how a signal will be sent through one or more effects processing modules and what order those processors will be put in.

effects send (a.k.a. FX send). An output used to route signal from a mixer or other device to an effects processor, such as a reverb or delay. In a mixer, sends used for effects are typically positioned post-fader. ☞ See also *aux send*.

efficiency. A speaker specification that indicates how well a driver will convert electrical signals into sound energy—the remainder of the signal is converted to heat and lost. Typical direct radiator efficiency ratings are 1 or 2%. Horns typically fall around 20%, though some can reach 30% efficiency. While more efficient speakers can generate

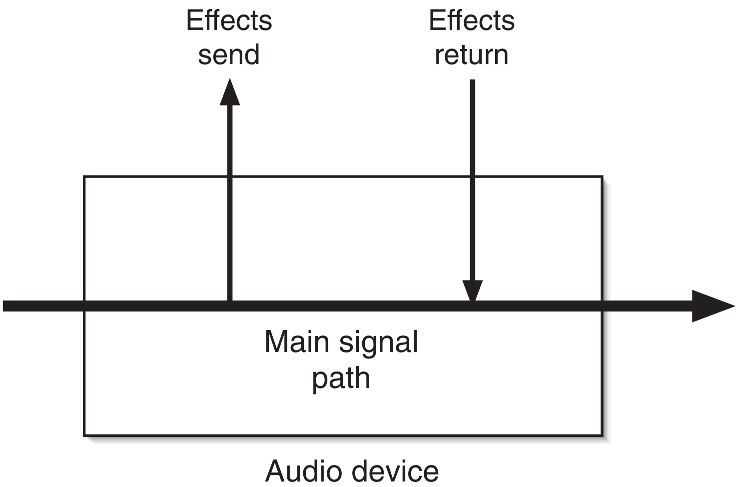


Figure E.2 An effects loop provides a way to insert an effects device or processor into a piece of gear's signal path.

higher volume levels with lower-power amplifiers, lower-efficiency speakers typically have better damping and resistance to resonances.

EFX. Short for effects, a.k.a. FX.

EG. ☞ See *envelope generator*.

egg carton. A type of "home brew" acoustical device, originally designed to transport and store chicken eggs, which provides little acoustical benefit.

EIA. Electronic Industries Alliance. A manufacturers' trade association that sets voluntary standards for its member companies. www.eia.org.

EIDE. Enhanced Integrated Drive Electronics. A version of IDE developed by Western Digital that used DMA, which reduced the load on the CPU during disk operations. DMA was later added to the "official" ATA standard.

eigentone. ☞ See *room mode*.

eighth space. A sound source, such as a speaker, located at the junction of three surfaces, such as the corner of two walls and the ceiling, is said to be in eighth space. Eighth-space placement of a speaker or other sound source results in a 9-dB level increase (mainly in the low frequencies) over free space placement. ☞ **See also** *free space*, *half space*, *quarter space*.

EIN. Equivalent Input Noise. A specification for the self-noise of a microphone preamp, assuming a 200-ohm impedance microphone is connected to the input. The lowest theoretical EIN rating is –129.6 dBu, though EIN is one of those specs that a manufacturer can manipulate or spin in a variety of ways, including using different mic impedances and dBV instead of dBu ratings. EIN is not a significant spec for most applications, except recording very low-level sounds, as self-noise for most preamps is extremely low.

EL34. A common pentode vacuum tube used in guitar and power amplifiers.

Elco. A brand of multi-pin audio connectors. There are several different configurations, including 20-, 38-, 56-, 90-, and 120-pin versions. Elco connectors are designed to only fit together in one direction and have a screw/nut arrangement that locks the male and female connectors together.

electret (a.k.a. back-electret). A type of condenser microphone that has a permanently charged backplate. An electret element has the advantage of not requiring an external voltage to power the plate, though power may still be required for internal preamp circuitry. Electret elements may lose some charge over time, resulting in reduced sensitivity and increased self-noise. Electret elements are generally found in lower-priced microphone models, while true condensers are used in higher-priced models.

electric piano. A keyboard instrument that produces sound when metal strings, tines, reeds, or other vibrating items are struck by hammers in a fashion similar to an acoustic piano. The vibrations are picked up by pickups and sent to an amplifier. Electric pianos are not synthesizers, but rather electromechanical instruments. Some of the most popular electric pianos include the Rhodes, which used vibrating metal tines of stiff wire; the Wurlitzer, which used metal reeds; and the Helpinstill and Yamaha CP-70, which used metal strings.

electro-optical (a.k.a. ELOP). A type of circuit employed in some compressors and limiters that uses photo-resisters or light-dependent resistors with an LED or other light source to reduce gain instead of a VCA. Some engineers believe that electro-optical compressors have smoother response characteristics than other types.

electrode. A positively or negatively charged conductive terminal in a component. A diode has two electrodes, a triode has three electrodes, a pentode has five electrodes, and so on. (Trivia note: Though most people assume an anode is a positive electrode, and a cathode is a negative electrode, this is not necessarily true. The terms *anode* and *cathode* refer to the direction of flow of anions and cations, which can vary depending on the circuit.)

electromagnetism. The physics of particles, magnetic fields, and electrical charges. Dynamic speakers, dynamic microphones, motors, analog and digital tape, hard drives, and other common items depend on electromagnetism for their operation.

electrophone. An instrument that produces sound using electricity. Electrophones include instruments that create sound using electricity, but not those that use amplifiers to make their sound louder. Other categories of instruments include chordophones (strings), aerophones (winds), membranophones (instruments with drum heads), and idiophones (various percussion and other instruments that do not use strings, heads, or wind to produce sound).

ELOP. ☞ See *electro-optical*.

EMI. Electromagnetic Interference. Errant electromagnetic fields that are picked up through the air by circuitry or cabling and that result in hum or buzz in the audio signal. EMI can be prevented or reduced using shielding, proper grounding, balanced lines, isolation transformers, and other methods.

emphasis. Modifying a signal as it is recorded (called *pre-emphasis*), by, for example, boosting the high frequencies. As the signal is played back, the modification is removed (called *de-emphasis*), in this example, by reducing the high frequencies. As the emphasis is removed, any noise added during the recording process is removed. In this example, hiss generated during recording will be removed when the high frequencies are reduced on playback.

emulation. Using modeling or other digital processing to re-create the sound or functions of a device or instrument.

encoder. 1. A device or algorithm that creates a representation of information that must be translated or decoded to be understood. Encoders are used to create versions of data that can be easily and safely stored and transmitted. 2. A knob found on some control surfaces, digital mixers, and other devices used to send digital control messages that are used to control parameter settings in a DAW or other destination.

encrypt. To encode information so that it cannot be understood without a key or other means of translation. Encryption is used to protect the privacy of data for storage and transmission. ☞ **See also** *data encryption*.

end address. A microphone that is physically designed so that its diaphragm is perpendicular to the body of the mic, with the capsule oriented to pick up sound best from the end of the mic, rather than from a side or sides (see Figure E.3). ☞ **See also** *side address*.

Figure E.3 An end-address microphone picks up sound best from the end of the mic.

endian. A system for ordering the bits in a binary word. Some computers use "little endian" systems, in which the bits go up in binary value toward the left of the binary word. Others use "big endian" systems, in which bits go up in binary value toward the right of the binary word.

engine. The algorithms that provide the audio or MIDI processes in a DAW, sequencer, audio editing software, or plug-in.

ensemble. 1. A group of vocalists and/or instrumentalists. 2. An effect similar to chorusing, designed to create the sound of many instruments playing a part at once.

envelope. 1. The volume "shape" of a sound. For example, most drums have a fast attack and a quick decay, with little sustain. A violin has a slower attack (unless it is played pizzicato) and sustains for as long as the player draws the bow across the string. 2. A multistage response curve that takes place over time and can be applied to control a synthesis parameter, such as volume or pitch (see Figure E.4). For example, the stages of a volume envelope might include an attack time, or how fast a sound begins; a decay time, or how fast the sound's initial transient settles to the sustain level; a sustain level, which determines how loud the sound will be while it is being held; and a release time, which determines how long the sound rings on after the sound ends.

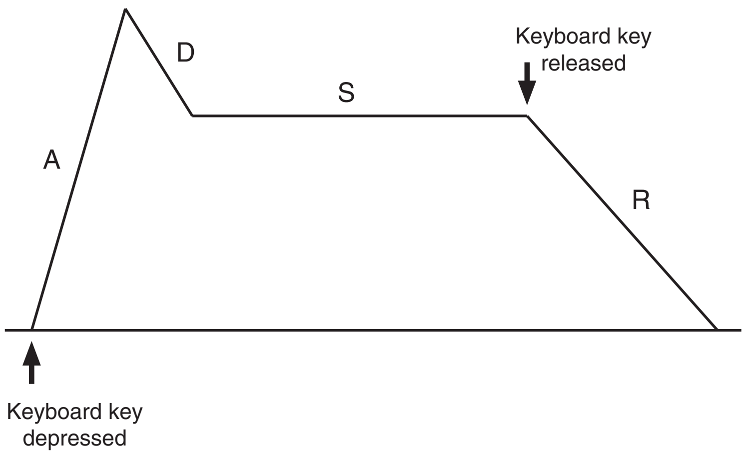


Figure E.4 An envelope is a multistage control signal that can be routed to control the volume, pitch, or other parameter of a synthesizer or sampler sound.

Other stages are also possible, depending on the capabilities of the envelope generator. ☞ **See also** *ADSR*, *AHDSR*, *AHDSFR*.

envelope filter. ☞ See *auto-wah*.

envelope follower. A device or algorithm that converts the envelope characteristics of a sound into a control signal, either digital or a control voltage. The envelope follower's control signal output can be routed to control another parameter in the device or system.

envelope generator (a.k.a. EG). A module or algorithm in a synthesizer or sampler that outputs a multistage control signal, or "envelope," that is used to control a parameter, such as volume, pitch, or others. ☞ **See also** *envelope*.

envelope tracking. ☞ See *rate scaling*.

envelopment. The sense of audio completely surrounding or enveloping the listener.

EOX. End of Exclusive. A MIDI System Common message that indicates the end of a System Exclusive message.

EPP. Enhanced Parallel Port. A computer port that supports two-way communication and high-speed transfers from multiple daisy-chained devices.

EPROM. Erasable Programmable Read-Only Memory. A type of ROM chip that can be reprogrammed as required. The chip retains the contents of its memory until it is exposed to ultraviolet light.

EQ. ☞ See *equalizer*.

equal temperament. A tuning system that divides each octave into 12 equal parts (called *semitones*). Most modern Western musical instruments use equal temperament.

equalizer (a.k.a. EQ). An audio processor that boosts or cuts the level of a particular frequency or range of frequencies. Equalizers are used to modify the frequency response of a system or the tonal shape of a signal. Equalizers were originally developed by the telephone company to enhance the audio quality of phone lines.

equilateral triangle. A triangle in which all three sides have the same length. An equilateral triangle is the ideal positioning for studio monitor speakers, where the monitors are set up at two corners of the triangle, with the listener's head as the final corner.

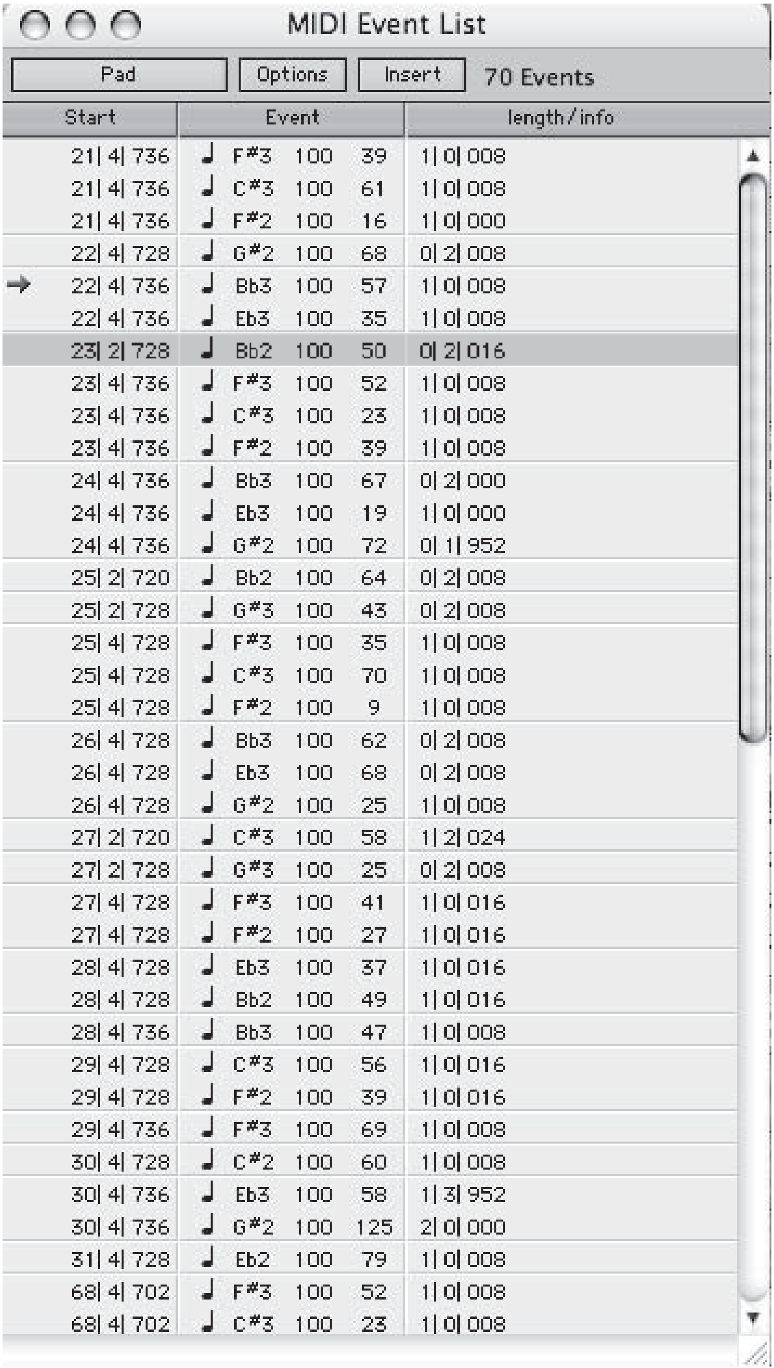
equivalent input noise. ☞ See *EIN*.

erase head. A head on a tape recorder used to randomize the tape's magnetism before it passes the record head. This gives the record head "fresh" tape on which to record and eliminates the need for the record head to record over existing magnetic signals.

error correction. A variety of schemes used to ensure that the data written and read by a digital storage system is accurate. ☞ **See also** *CRC*, *CIRC*, *parity*.

eSATA. External Serial Advanced Technology Attachment. A version of SATA designed for use with external hard disks or optical drives. eSATA is hot-swappable and can be faster than USB or FireWire protocols.

Ethernet. A family of networking technologies developed by Xerox for LAN (*Local Area Network*) applications. Ethernet cables have been adopted by some manufacturers for additional proprietary applications.



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22| 4| 728	♩ G#2 100 68	0| 2| 008
22| 4| 736	♩ Bb3 100 57	1| 0| 008
22| 4| 736	♩ Eb3 100 35	1| 0| 008
23| 2| 728	♩ Bb2 100 50	0| 2| 016
23| 4| 736	♩ F#3 100 52	1| 0| 008
23| 4| 736	♩ C#3 100 23	1| 0| 008
23| 4| 736	♩ F#2 100 39	1| 0| 008
24| 4| 736	♩ Bb3 100 67	0| 2| 000
24| 4| 736	♩ Eb3 100 19	1| 0| 000
24| 4| 736	♩ G#2 100 72	0| 1| 952
25| 2| 720	♩ Bb2 100 64	0| 2| 008
25| 2| 728	♩ G#3 100 43	0| 2| 008
25| 4| 728	♩ F#3 100 35	1| 0| 008
25| 4| 728	♩ C#3 100 70	1| 0| 008
25| 4| 728	♩ F#2 100 9	1| 0| 008
26| 4| 728	♩ Bb3 100 62	0| 2| 008
26| 4| 728	♩ Eb3 100 68	0| 2| 008
26| 4| 728	♩ G#2 100 25	1| 0| 008
27| 2| 720	♩ C#3 100 58	1| 2| 024
27| 2| 728	♩ G#3 100 25	0| 2| 008
27| 4| 728	♩ F#3 100 41	1| 0| 016
27| 4| 728	♩ F#2 100 27	1| 0| 016
28| 4| 728	♩ Eb3 100 37	1| 0| 016
28| 4| 728	♩ Bb2 100 49	1| 0| 016
28| 4| 736	♩ Bb3 100 47	1| 0| 008
29| 4| 728	♩ C#3 100 56	1| 0| 016
29| 4| 728	♩ F#2 100 39	1| 0| 016
29| 4| 736	♩ F#3 100 69	1| 0| 008
30| 4| 728	♩ C#2 100 60	1| 0| 008
30| 4| 736	♩ Eb3 100 58	1| 3| 952
30| 4| 736	♩ G#2 100 125	2| 0| 000
31| 4| 728	♩ Eb2 100 79	1| 0| 008
68| 4| 702	♩ F#3 100 52	1| 0| 008
68| 4| 702	♩ C#3 100 23	1| 0| 008

Figure E.5 An event list editor provides a text list of MIDI events (in this case, a stream of notes) that can be precisely edited and processed.

Euroblock. A type of terminal strip used for audio products intended for permanent installation. With a Euroblock connector, the insulation at the end of the wire is stripped off, and then the wire is inserted into a slot in the block. A screw is tightened down to hold the wire securely.

even-order harmonic distortion. A type of distortion that results primarily in extra even harmonics being added to a signal. Even-order harmonic distortion is often produced when tube devices clip; it tends to be more pleasant sounding to most listeners than other types of distortion (and even desirable in guitar amplifiers).

event. A MIDI Channel Message command. Examples include note on, note off, and program change messages. Continuous controllers, aftertouch, pitch bend, and similar types of messages produce a stream of events.

event list. A text list of the events that are recorded into a MIDI track in a sequencer or DAW. Event lists make certain types of event processing easier and allow precise editing of values and placement of events. See Figure E.5.

excite. In acoustics, to add energy to a room.

exciter. A type of audio processor that increases the brightness in a signal without increasing the signal's level, as would happen when using EQ. Some exciters work by adding a small amount of controlled harmonic distortion to the signal. Others use compression and filtering to create high-frequency content that can be added back into the dry signal to increase brightness.

excursion. The maximum distance a speaker driver can move in and out from its resting position. A larger excursion can move more air, creating high SPLs, but there are tradeoffs, such as poor damping.

expand. To enlarge a DAW or sequencer track so that its data can be viewed more easily.

expander. The opposite of a compressor. An expander is used to increase the dynamic range of a signal. In an expander, when the input signal crosses a threshold setting, the output level of the signal is processed to decrease based on a ratio. For example, with a 1:2 ratio, a drop of 1 dB in signal level at the expander input results in a reduction of 2 dB in signal level at its output. Expanders are often used to reduce the amount of background noise in a signal.

expansion slot (a.k.a. slot). An internal expansion interface in a hardware device, such as a computer, sampler, or other device, used for adding expansion cards or optional hardware. Slots can be used to add RAM to a sampler or computer, an audio interface to a computer, an A/D converter to a preamp, or a variety of other options. Expansion slot formats range from "standards," such as PCI, to proprietary formats.

export. Saving data out of a program in a format that allows it to be opened by another program. For example, an audio program that uses AIFF as its "native" audio file format might export an audio file in WAV file format so that it can be opened by other programs that don't support AIFF.

ExpressCard. A card format from the PCMCIA that is about half the size of a PC card and that supports wireless communication, more memory, and security.

expression. MIDI Continuous Controller #11, with a range from 0 through 127. Expression is often routed to control musically expressive volume changes that do not affect the overall volume of the track. For example, MIDI Controller #7 is used to set the overall level, and Controller #11 might be used for slight crescendos and decrescendos.

expression controller. A foot pedal or other device that is connected to a keyboard or MIDI controller's expression input and used to manipulate parameters. Expression controllers usually have 1/4-inch connectors.

extension. An older Macintosh term for a piece of software similar to a device driver.

external sync. A setting in keyboard workstations, drum machines, hardware DAWs, and other devices that determines whether the unit is using its internal clock or slaving to an external clock.

F

F1. A two-track recording format that stored digital audio on Beta or VHS videotape.

factory preset. A program or patch created for a synthesizer, effects processor, or other device by the manufacturer and included with the device when it ships. In some cases, factory presets are stored in ROM and cannot be changed; in others, they are stored in RAM and can be modified by the user. Factory presets are intended to provide the user with a working library of useful sounds as well as a set of sounds that show off the capabilities of the device to potential purchasers in stores.

fade in. A gradual increase in volume level, starting from silence. See Figure F.1.

fade out. A gradual decrease in audio level, ending at silence. See Figure F.1.

fader. Another name for a potentiometer, though most engineers use the term to refer to a volume control that slides in a linear fashion instead of turning in a rotary fashion. Faders are typically used in mixers.

fader flip. 1. A function of some analog mixers that have two input signal paths and faders per channels.

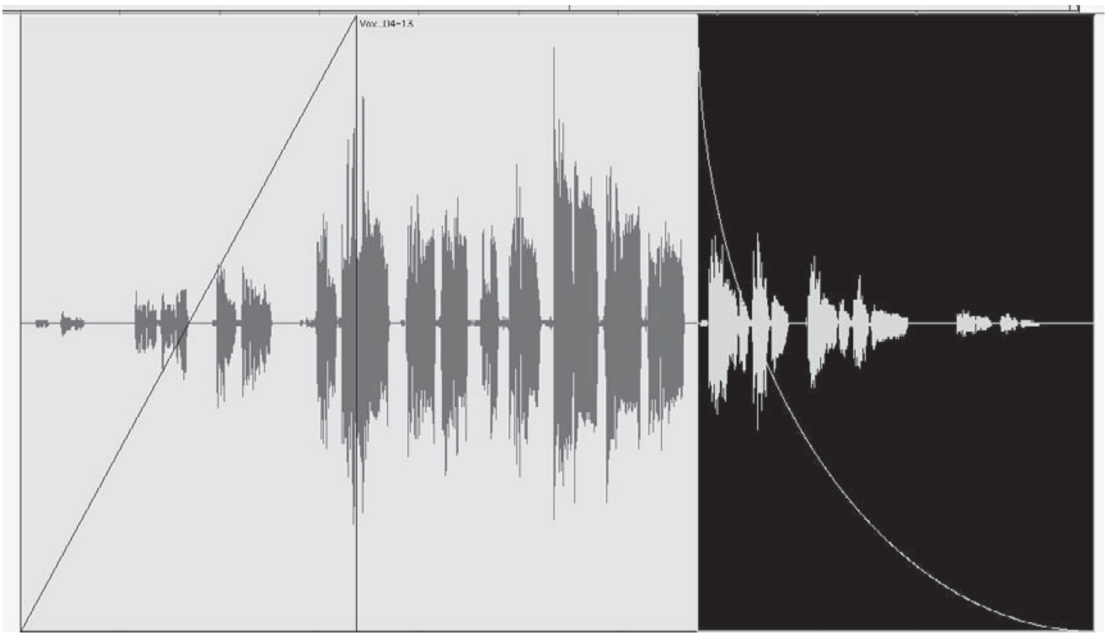

Figure F.1 Most DAWs provide a fade function that can fade an audio file in and out.

(Often one path is the regular input path, and the other path, with a smaller fader, is used as a tape return for bringing recorded signals from a multi-track tape machine into the mixer.) Fader flip reverses which signal path is using the main faders and which is using the "small" secondary tape return faders. 2. A function found in many digital mixers that allows one set of faders to control additional channels or other functions in the mixer.

fader group. A function of some mixers that can link the operation of faders so they move together—when one fader in the group is moved, the rest of the faders follow along, maintaining the relative differences in their settings.

fall time. The time it takes for a signal or voltage to change from a high value to a low value, typically measured as the time it takes to go from 90% to 10% maximum level.

FAQ. Frequently Asked Questions. A feature of the documentation or support web pages for a device that lists and answers common questions from users.

far field. Speakers or a sound source placed beyond the near-field range (more than three to four feet or so from the listener). Far-field speakers typically are substantially larger and more powerful than near-field monitors and are more influenced by room acoustics, reflections, and reverberations.

Faraday shield (a.k.a. Faraday cage). A type of electromagnetic shield named for physicist Michael Faraday and created by placing grounded conductive material, such as copper or aluminum, around the object being shielded. Faraday shields are used in a variety of applications, such as around MRI machines, microwave ovens, and others. In audio, one example of a Faraday shield is the shielded wire used in coaxial cables; shields are also used around a variety of sensitive electronic components and circuits.

fast forward. A transport control that causes a tape to fast-wind toward the end of the tape or a DAW or audio program to quickly scroll toward the end of the song or project.

Fast Fourier transform (a.k.a. FFT). An efficient algorithm for calculating a Fourier transform. ☞ **See also** *Fourier transform*.

FAT. File Allocation Table. A file stored in Sector 0 on a disk or other media containing information about the media, which areas of the media are unused and which contain data, and where each file is stored. ☞ **See also** *sector*.

FAT-32. A type of file allocation table that supports drives larger than two gigabytes in size. ☞ **See also** *FAT*.

feathering. A technique in which equalization is spread out over a range of frequencies rather than focused on a single frequency. For example, instead of a large boost at 500 Hz, a small boost is added at 500 Hz, a smaller boost at 480 and 520 Hz, a still smaller one at 460 and 540 Hz, and so on.

feedback. 1. A problem caused by the sound from a speaker entering a microphone, being amplified, coming out of the speaker, being picked up by the microphone, being amplified, coming out of the speaker…a loop occurs, resulting in a runaway signal at a particular frequency and the familiar howl or squeal of feedback. See Figure F.2. 2. A runaway signal that results when the output of a device is inadvertently connected to its input, resulting in screaming out-of-control frequencies. 3. Also known as regeneration. With an echo or delay device, a portion of the echo output being sent back to the input of the processor, where it is delayed again, producing an additional repeat. The amount of feedback determines the number of echo repeats that are heard. 4. With a modulation device, such as a flanger or phase shifter, a portion of the output signal that is sent back to the input of the device, resulting in a more intense effect. 5. A guitar performance technique where an electric guitar's strings vibrate in response to high volume levels from its amplifier, causing infinite sustain and the emphasis of different harmonic frequencies.

feedback eliminator. A device that combines a spectrum analyzer and notch equalizer that can automatically detect acoustic feedback and adjust the equalizer to reduce the level of the frequency that is feeding back.

ferrofluid. An oil-like substance that contains magnetic material. Ferrofluid is used in speaker drivers to draw heat away from the voice coil, allowing the coil to handle more power.

FET. Field-Effect Transistor, a.k.a. Unipolar Transistor. A type of transistor that uses an electronic field between the substrate layers in the semiconductor material to control voltage. FETs have linear response and high input impedance, making them ideal for microphone electronics and for power

Figure F.2 Feedback results when a signal loops repeatedly through a device or system, increasing in intensity until it is out of control.

amps, among other things. ☞ **See also** *transistor*, *JFET*, *MOSFET*.

FFT. ☞ See *Fast Fourier transform*.

Fibre Channel. A network technology standardized by ANSI and used for storage systems, including SANs (*Storage Area Networks*). Fibre Channel offers high-speed connections for hard drives, RAIDs, and other storage devices. Fibre Channel and SANs are usually used by large production installations in which multiple operators require access to the same information and drives.

FIFO. First In, First Out. A computer data storage and processing paradigm where the first data to enter a buffer or other memory is the first to be read, processed, and output.

figure 8 (a.k.a. bi-directional). A microphone polar pattern resembling a sideways "8" shape (see Figure F.3). Figure-8 microphones are bi-directional, picking up sound nearly equally well from the front and from the rear, and not at all from the sides of the mic. Figure-8-patterned microphones are mainly used for studio applications, where they can be positioned to reject sound from the side, but still pick up ambience from the rear. They are less commonly used for live applications, as the rear lobe makes them susceptible to feedback problems. Figure-8 microphones, like all directional microphones, are subject to proximity effect. ☞ **See also** *pressure-gradient microphone*.

filament. A short tungsten wire in a vacuum tube that is used to heat up the cathode to make it easier for electrons to cross to the anode.

file. A collection of computer data stored on a drive or other media. There are a variety of types of files: data, program, directory, and more.

file path. The location of a file on a computer hard drive. The path describes all the steps that are necessary to navigate through nested folders and directories to find the file. For example, *Macintosh HD/Applications/Digidesign/Pro Tools/Pro Tools LE* would tell the user or computer to look in the hard drive named Macintosh HD, inside the Applications folder, which contains the Digidesign folder, which contains the Pro Tools folder, which contains the Pro Tools LE program file. Paths can be very simple or very long and complex, depending on how many folders are nested together.

file server. ☞ See *server*.

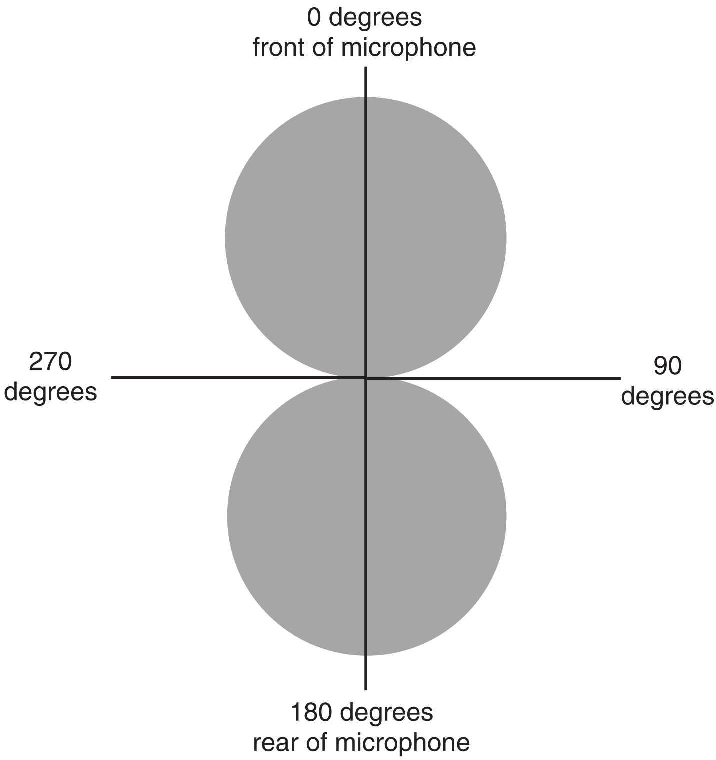

Figure F.3 A figure-8 polar pattern consists of two spherical pickup lobes, one in front of the microphone and the other behind it.

filter. A type of audio processor that reduces or attenuates the level of a range of frequencies. In many cases, a filter is combined with an amplifier so that frequencies can be boosted as well as cut. A number of filters can be combined into one device or processor to create an equalizer. There are a variety of types, including:

- **bandpass.** A filter that attenuates the frequencies above and below a band of frequencies that is allowed to pass unaffected.
- **band reject.** A filter that attenuates a band of frequencies and allows the frequencies above and below the band to pass unaffected.
- **high cut.** A filter that attenuates the frequencies above a cutoff frequency and passes those below unaffected.
- **high pass.** A filter that attenuates the frequencies below a cutoff frequency and passes those above unaffected.
- **high shelf.** A filter that can boost or attenuate the frequencies above a cutoff frequency and passes those below unaffected.
- **low cut.** A filter that attenuates the frequencies below a cutoff frequency and passes those above unaffected.
- **low pass.** A filter that attenuates the frequencies above a cutoff frequency and passes those below unaffected.
- **low shelf.** A filter that can boost or attenuate the frequencies below a cutoff frequency and passes those above unaffected.
- **notch.** A filter that attenuates a very narrow band of frequencies.

filter slope. ☞ See *slope*.

filter sweep. A technique in which the cutoff frequency of a filter is increased or decreased in real time, while the filter is passing signal. The sonic effect depends on the type of filter being swept and the amount of filter resonance applied.

finalize. 1. Also known as fix-up. The process of converting an Orange Book CD into a closed Red Book–compatible CD. 2. A mastering-style process taking its name from the TC Electronic Finalizer family of digital audio devices, which provide equalization, compression/limiting, and other processes used in mastering stereo mixes.

fine. The opposite of coarse. Fine refers to small increments of change or measurement, usually in reference to a control or parameter movement.

fingerprint EQ. A type of digital equalizer that analyzes the frequency spectrum of one signal, then applies that spectrum to another signal.

FIR filter. Finite Impulse Response filter. A type of digital filter that uses the average of several samples to create its output. Because they do not cause phase distortion, FIR filters are used for critical applications, such as anti-aliasing and anti-imaging applications in analog-to-digital and digital-to-analog converters. ☞ **See also** *IIR filter*.

FireWire (a.k.a. IEEE 1394). A serial communication protocol developed by Apple and now available in a wide variety of professional and consumer audio and video equipment. FireWire supports asynchronous and isochronous transfers, allows up to 63 devices per bus, and is hot-swappable. FireWire does not require communication through the CPU or system memory. There are several types, including emerging standards that are not yet in widespread use. FireWire hard drives have become a popular choice for data storage and recording in studios.

- **FireWire 400.** 100, 200, or 400 Mbits/second, half-duplex, up to 4.5-meter (about 15 feet) cable lengths.
- **FireWire 800.** 800 Mbits/second, full-duplex, up to 100-meter cable lengths, backward compatible with FireWire 400.
- **FireWire S1600.** 1.6 Gbits/second, backward compatible with earlier FireWire formats.
- **FireWire S3200.** 3.2 Gbits/second, backward compatible with earlier FireWire formats.
- **FireWire S800T.** 800 Mbits/second over Cat 5e cable.

firmware. Basic software and instructions for a device that are stored in one of several types of ROM or programmable ROM chips or in flash memory. Firmware is used to load the operating system (occasionally the OS may be stored as firmware) and contains other instructions necessary for the device to set up and operate.

first reflection. Sound waves that reach the listener's ears after one bounce from a surface, less than 20 milliseconds after the direct sound from the source. First reflections are often the most destructive in terms of acoustic cancellations and reinforcements in a room. See Figure F.4.

fixed pattern. A microphone that has a single polar pattern.

fixed point. A computer number and math system that has a set number of digits after the decimal point. For example, if the system has two digits after the decimal point, numbers such as 1.15 can be accurately represented. However, a disadvantage is that a number such as 1.0143 would have to be rounded to 1.01. ☞ **See also** *floating point*.

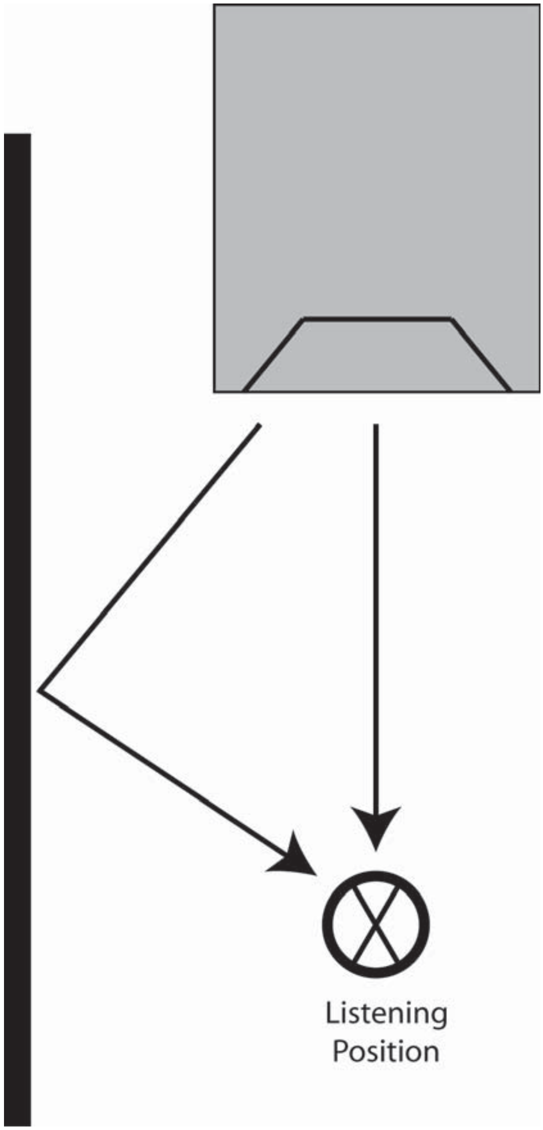

Figure F.4 A first reflection is a sound wave that reaches the listener immediately after the direct sound, after reflecting off of one surface.

FLAC. Free Lossless Audio Codec. A lossless open-source audio data compression/decompression file format. FLAC can result in file size reductions up to 50% without loss of data.

flam. 1. A percussion technique in which a grace note is played immediately before the actual note.

2. Two identical sounds that occur very close together in time. Depending on the time difference between them, this can produce a "doubled" effect or can result in phase cancellation.

flange. The rim, or sides, of a tape reel, which attach to the reel's hub.

flanging. An effect created in the 1960s using two analog reel-to-reel tape decks playing back the same audio slightly out of sync. The signals were mixed together, and the speed of one of the decks was changed by pressing on its tape reel flange, creating a continuously moving comb-filter cancellation and reinforcement. Analog and digital electronic versions of the effect have been created, which allow for greater control over the effect—as well as portability! ☞ **See also** *through-zero flanging*.

flash. ☞ See *flash RAM*.

flash drive. A drive that records to flash memory. ☞ **See also** *jump drive*.

flash memory. ☞ See *flash RAM*.

flash RAM (a.k.a. flash memory, FEPROM [Flash Erasable, Programmable, Read-Only Memory]). A type of RAM (*Random Access Memory*) that retains its contents when power is removed. Flash RAM can be written and read by a computer or other device, just like a hard drive or other storage media. A variety of forms are available, including flash drives and others.

flat. 1. Having an even frequency response, without dips or peaks due to electronic or physical characteristics. 2. A device or room in which all frequencies are outputted at unity gain—that is to say, at the same output level as they came in. Since a flat device or room doesn't emphasize or deemphasize any frequencies, it provides a true picture of the signal that will translate well to other systems and rooms. 3. Setting all the boost/cut controls on an equalizer to zero so there is no change to the signal.

flat response. A device that outputs all the frequencies in a signal at the same level they were at when the signal was inputted.

Fletcher-Munson Curve (a.k.a. equal-loudness contours). A series of response curves developed by American physicists Harvey Fletcher and W. A. Munson to show the human ear's response to frequencies at different volume levels.

float. To suspend the floor of a studio on hard rubber "pucks," springs, or other material to isolate it from the rest of the structure. 2. An unconnected electrical ground. An example is a telescoping shield.

floating floor. A studio floor that is isolated from the surrounding structure.

floating point. A technique used to represent large numbers in computers. There are three parts to a floating-point number: the sign (positive or negative), the mantissa (the value represented as a fraction), and an exponent (indicates the position of the decimal point). For example, 192,000 would be represented as 1.92 times 10 to the power of five. This type of representation allows computers to very efficiently process data, especially for DSP and other operations. Floating-point math and value representation are also used in DAWs to increase headroom for certain operations. ☞ **See also** *fixed point*.

floating window. A window in a computer program that always remains on top of the program's other windows. Examples might include a tool palette, a video playback window, or another window that needs to remain visible and accessible at all times.

floppy disk. A type of data storage medium using a thin Mylar disk encased in a plastic envelope or shell. A variety of sizes and capacities were introduced over the years. Floppy disks have been replaced by flash drives, CD-RWs, and other more durable media with higher storage capacities.

flutter. A fast variation or fluctuation in the playback speed of a tape recorder, resulting in an audio artifact resembling vibrato. ☞ **See also** *wow*.

flutter echo. A fast echo effect cased by sound waves bouncing between two parallel hard surfaces in a room and creating fast, discrete echoes. The resulting effect is a very short "rattling" type of echo often described as fluttering.

flux. 1. Magnetic energy, measured in webers or nanowebers. 2. A gel-like substance used during soldering to lower the melting temperature of the solder.

FM. ☞ See *frequency modulation*.

FM synthesis. Frequency Modulation synthesis. A type of digital synthesis developed by John Chowning and first utilized by New England Digital for the Synclavier and Yamaha for the DX7. FM synthesis uses one or more oscillators to modulate one or more audible oscillators. The audible oscillators are called *carriers*, while the oscillators used to produce modulation are called *modulators*. Early types of FM oscillators (called *operators*) produced only sine waves, but through intricate modulation routings were capable of creating extremely complex waveforms. Later synthesizers built upon the FM concept allowed the use of operators producing additional waveforms. FM synthesizers are quite flexible, but are especially adept at producing bell-like tones.

foil shield. A type of shielding used in some cables, in which a metal foil (usually aluminum) is used in place of braided strands of wire. Foil shielding provides more complete coverage and therefore better shielding than stranded types.

foldback (a.k.a. monitor system). A system that is intended to feed a mix back to the musicians in a studio or onstage so they can hear themselves and each other while recording or playing.

fold down. Reducing a multichannel surround mix to a smaller number of channels, such as stereo.

folder. A graphical representation of a computer directory. A computer folder is analogous to a physical folder that is used to store paper documents, except that virtual folders are used to store computer files.

foot pedal (a.k.a. expression pedal). A foot-operated control device—often a rocking platform that turns a potentiometer. Foot pedals are used to control volume and continuous controllers with keyboards and to control certain parameters in effects processors and other devices.

footswitch. A switch that is designed to be operated by foot. Footswitches are used for a variety of purposes, but are mainly used to bypass and enable effects processors during live performance.

formant. A resonance produced by a vocal tract or other sound generator—the formants in a voice or instrument are what make it identifiable. The formants remain the same regardless of the pitch or frequency the sound generator is creating. Since manufacturers started providing formant processing in devices and plug-ins that transpose or shift pitch, much more realistic-sounding results have been possible.

format. 1. The "type" of a file—the file format—describes the protocol used to write the data. 2. To format a disk or other media is to prepare it for use with a particular operating system. There are two types of formatting, low level (usually handled by the drive manufacturer) and high level (which is done by the end user). 3. The particular host protocol that a software plug-in conforms to; examples include RTAS, VST, MAS, and Audio Units.

formatted capacity. The amount of storage space available on a drive after it has been formatted. The advertised capacity of many hard drives is the unformatted capacity, which looks better since it is bigger. But the formatted capacity is a far more useful spec because it reveals how much actual space is available for storing data.

Fourier analysis. A mathematical method named for French mathematician Jean Baptiste Joseph Fourier for analyzing the frequency spectrum of a waveform. ☞ **See also** *Fast Fourier transform*, *Fourier transform*.

Fourier transform. A mathematical method for analyzing a waveform that allows for transfer between the time and frequency domains. In audio, a Fourier transform is based on the fact that audio waveforms can be represented as the sum of many component single-frequency waves (sine waves). A Fourier transform is both the graph showing the frequency content of a waveform and the mathematical equation that can be used to represent it. An inverse of the Fourier transform can be used to synthesize the sound that was analyzed.

FPU. Floating Point Unit. A "coprocessor" or supplemental processing chip in a computer dedicated to handling and speeding up floating-point math operations.

fragmentation. As a computer writes, erases, and rewrites data onto a hard disk, areas appear on the disk where no data is stored. When new data is written, it will fill in the existing holes first, sometimes breaking up a single file into multiple parts spread across different locations on the drive. This is called *fragmentation*. Fragmentation can be a problem when it becomes so widespread that the drive mechanism is forced to constantly jump around to find all the parts of each file, significantly slowing down the read process. Various disk utilities can be used to defragment, or *defrag*, a drive to restore its performance.

frame. A single image in a video or film. SMPTE time code represents each frame in a video or film with one word of data, indicating an exact time location.

frame accurate. A device or system that synchronizes with timing accurate to the frame level.

frame rate. The speed of the time code, usually expressed in frames per second (FPS). A number of rates are in common use, including 25, 29.97 drop frame, and 30 FPS. Note that the time code frame rate is not related to the tempo of the music. Rather, it is a timing constant for synchronized gear to reference.

free field. An area with no reflective surfaces. The only true free field space is outer space, since even outdoors the earth causes reflections.

free space. ☞ See *free field*.

freeware. Software created by an individual and offered for free download. Freeware is copyrighted, though the author requests no payment or licensing fees. ☞ **See also** *shareware*.

freewheel. A function of some time code readers that can continue to operate as if locked to time code even if the incoming time code signal is interrupted.

freeze. A function of some DAWs that renders a virtual instrument or audio track using real-time plugins to an audio file in order to free up the system's DSP resources. Once a track is rendered or frozen, it plays from hard disk instead of consuming CPU power for real-time processing. In most cases the track can be unfrozen in order to make changes to it, then re-frozen at any time.

frequency. The number of times a sound wave vibrates, or moves through a complete cycle, in a second.

frequency doubling. A phenomenon caused by harmonic distortion that makes a low-frequency signal seem to sound an octave higher than it actually is.

frequency modulation (a.k.a. FM). A technique for modulating an audio waveform's frequency (the carrier) with another audio waveform (the modulator). The most common audio example is vibrato, where a low-frequency signal is used to slightly change the pitch of another signal, creating a wavering pitch. Frequency modulation is used to encode a signal for radio broadcast and is used in various types of synthesis—in particular, FM synthesis.

frequency range. The span of frequencies a device can reproduce.

frequency response. 1. How a device or space responds to a range of frequencies. 2. Maximum and minimum frequencies a device can pass with full level.

frequency response curve. A graph of the difference in the output amplitude of the range of frequencies versus the input amplitude of the range of frequencies when passed through a device. In other words, a graph of how well a device reproduces the range of frequencies. A flat frequency response curve indicates that the output amplitude of a given frequency will be the same as the input amplitude of the same frequency. See Figure F.5.

frequency sweep. ☞ See *sweep*.

front address. ☞ See *end address*.

front loaded. A speaker cabinet in which the driver is mounted on the front of the cabinet, on a baffle. ☞ **See also** *horn loaded*.

front side bus. ☞ See *FSB*.

Fs. Frequency, Sampling, a.k.a. fs, FS, fS. An abbreviation for sampling rate.

FSB. Front Side Bus. A data bus that connects a computer's CPU to its RAM, motherboard, and other items. The FSB is critical to the speed of the system because it is responsible for shuttling virtually all of the data in the system to and from the CPU.

FSK. Frequency Shift Keying. A system that uses an audio frequency modulated by a square wave to represent data as well as for synchronizing some sequencers and drum machines to other gear. Other systems that use FSK include modems and fax machines.

full code. ☞ See *full scale*.

full duplex. A circuit that is able to send and receive data at the same time. Computer audio systems are said to be full duplex if they can record and play audio simultaneously.

full normal. A type of normal connection in a patch bay in which a signal fed into the top back jack of the bay will be automatically connected to the bottom back jack of the bay. Plugging a cable into either the front top or front bottom jack will break the "normal" connection, allowing the signal to be re-routed. ☞ **See also** *normal*.

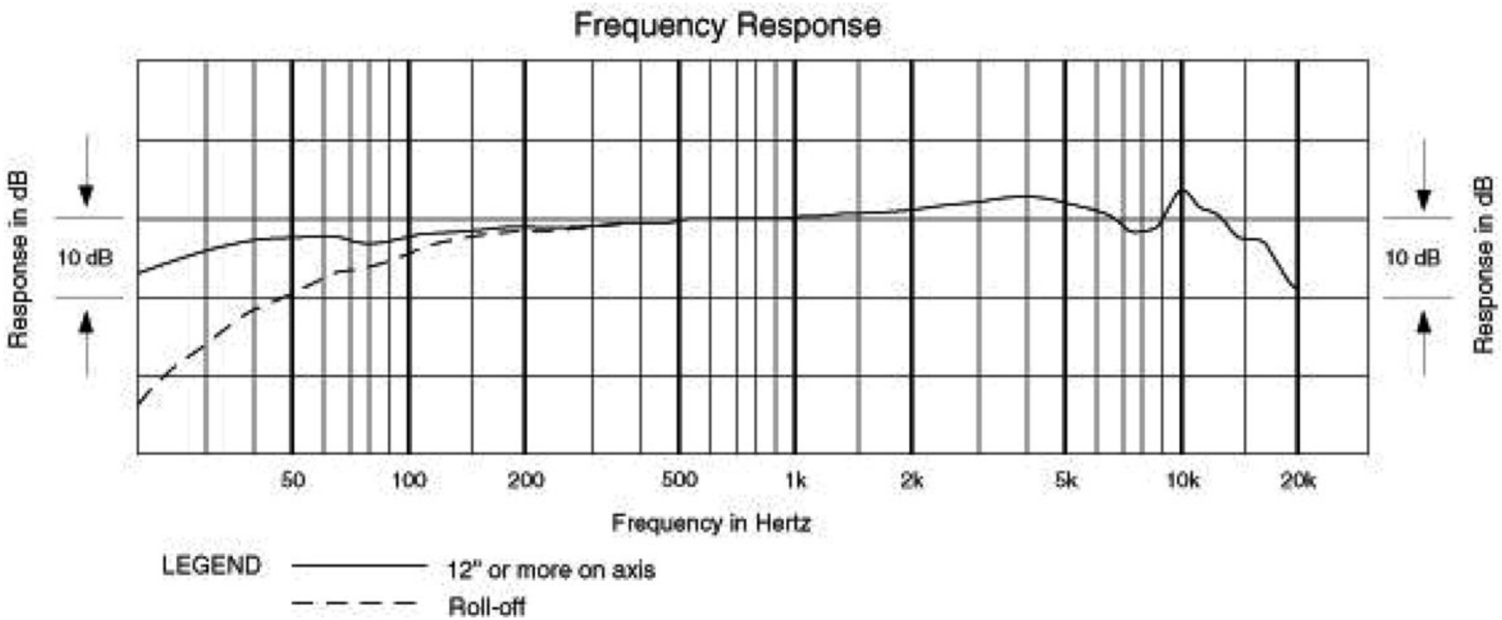

Figure F.5 A frequency response curve is a graph of how well a device reproduces the frequency range. In this case, a microphone manufacturer has provided two curves, one (solid line) for the normal response, and the other (dotted line) for the microphone with low-frequency roll-off engaged.

full range. Technically, a speaker that is able to reproduce the entire audio frequency range, though in practice, the actual range will be somewhat limited at the high and low ends.

full scale. The highest signal level that can be represented by a digital system, where all the bits in a sample word have a value of one. A signal of any higher level will result in clipping.

full track. A tape recorder head format in which a single monophonic channel covers the entire width of the tape.

function generator. An electronic device used to generate waveforms, such as triangle or sine waves, at various frequencies. Function generators are used to provide signals to test and analyze other equipment.

fundamental. The base, core, or primary frequency of a pitched sound. The fundamental is almost always the lowest-frequency component of a given sound.

fuse. An electronic component designed to protect a device from damage or fire by failing if the amount of electrical current exceeds a certain point.

fuzz. A buzzy, highly distorted effect often used with electric guitar.

FX. Short for effects.

FX return. ☞ See *effects return*.

FX send. ☞ See *effects send*.

G

gain. Amplification applied to a signal, expressed in decibels.

gain before feedback. The maximum volume level a sound system can attain before microphones begin feeding back.

gain factor. A measure of the amount of gain a vacuum tube can produce. ☞ **See also** *12ax7*, *12at7*, *12ay7*.

gain range. The amount of gain available in a preamp.

gain reduction. The attenuation of a signal's peak levels using a compressor or limiter.

gain stage. A point in a circuit or device where the gain or level of a signal can be amplified, or where level can be controlled/adjusted.

gain staging. Setting up and managing the gain or level of a signal at various points in a signal path for optimum performance. Gain staging includes setting up each stage so that no single stage is providing all or most of the amplification, managing headroom and signal-to-noise, managing levels so that all faders and controls are at reasonable points (none are turned all the way up, and none are turned all the way down), all meters are showing safe levels, digital devices are receiving enough signal for good resolution, and so on.

gain structure. ☞ See *gain staging*.

GAS. Gear Acquisition Syndrome or Guitar Acquisition Syndrome (depending on who is suffering from the affliction). The burning need to acquire a piece of equipment or to own the latest and greatest models.

gate. ☞ See *noise gate*.

Gauss. A measure of magnetic flux. Named for German mathematician Karl Gauss.

GB. ☞ See *gigabyte*.

General MIDI (a.k.a. GM). A standard developed to make it easy to transfer a MIDI song from one system to another while providing consistent performance. A GM-compatible synthesizer must adhere to a variety of requirements, such as providing a standardized set of 128 presets that are stored in a certain order (for example, Patch 1 is always a piano), 24 notes of polyphony, a number of specifically defined continuous controller assignments, and so on.

General MIDI Lite (a.k.a. GM Lite). A version of the General MIDI standard designed to work with cell phones and handheld devices. The polyphony requirement is reduced to 16 notes, the GM preset set is used, and seven continuous controllers are specified.

General MIDI 2 (a.k.a. GM2). An extension of General MIDI that ups the polyphony requirement to 32 notes. The standard patch set requirement is upped to 256, more controllers are specified, and backward-compatibility with GM is required.

generational loss. An analog tape phenomenon in which audio quality is lost with each additional generation of signal that is recorded or added. (In most cases, high frequencies are lost and noise builds up.) Examples of when generational loss occurs include when one analog tape is copied to another, or when overdubs are accomplished by bouncing between two analog tape decks.

genlock. The process of locking a device to a synchronization generator or clock.

GHz. ☞ See *gigahertz*.

gig. 1. Short for gigabyte. 2. A "job" in musician-speak.

gigabit. One billion bits, or 1,000 Mb. *Gigabit* is also used to refer to a transmission speed of one billion bits per second, or 1,000 Mbps, such as is provided by gigabit Ethernet.

gigabyte (a.k.a. GB). One billion bytes, or 1,000 MB.

gigahertz. A frequency of one billion Hertz. Gigahertz is used as a unit of measure for computer clock speeds, for ultra-high-frequency radio signals (UHF), and for other extremely high-frequency signals.

glass master. A step in manufacturing compact discs, where the audio data is written using a laser onto a glass plate covered with photoresist coating. The plate is then used as a mold for creating the metal data layer for the final compact discs.

glide (a.k.a. glissando, portamento). A synthesizer function that smoothly changes the pitch from one note to the next with no break in the sound.

gliss. ☞ See *glissando*.

glissando. A slide between pitches; how this is accomplished varies from instrument to instrument.

glitch. 1. ☞ See *stutter effect*. 2. A sudden malfunction or error. 3. A short unwanted noise or other audio artifact. 4. An unexpected setback. 5. A genre of experimental electronic music based around short sampled sounds and noises used to create beats.

global. Parameters that affect a device as a whole, as opposed to more specific parameters that may affect only a particular function or preset.

GM. ☞ See *General MIDI*.

GM2. ☞ See *General MIDI 2*.

gobo. Short for "go-between." A movable divider or barrier used in a studio to isolate one instrument from another to help reduce bleed.

gold sputtered. A type of microphone diaphragm made from Mylar covered with an extremely thin—molecule thin—layer of gold. The gold provides conductivity, yet is light enough and thin enough not to impede the motion of the diaphragm in response to sound waves.

Golden Mean. A ratio of 0.618, which, when applied to room dimensions, is believed by some to provide the ideal spacing of room modes across the frequency response of the space. In practice, this works out to the width being 1.6 times the height, and the length being 2.6 times the height.

Golden Ratio. ☞ See *Golden Mean*.

goniometer. ☞ See *jellyfish meter*.

gooseneck. A flexible metal extension that can be added between a mic stand and a microphone adapter. A gooseneck allows the microphone to be easily repositioned in any direction. Some manufacturers coat the gooseneck with rubber or other materials to cut down on noise created when the microphone is moved.

GPI. General Purpose Interface. A hardware device featuring switches or other functions that can be controlled using MIDI or SMPTE and that can be used to control a function in a non-MIDI-compatible piece of hardware.

GPU. Graphics Processing Unit, a.k.a. Graphics Accelerator. A graphics component in a computer that is optimized to assist with processing video data for the computer's display.

graded hammer action (a.k.a. progressive hammer action). A type of weighted keyboard action whose response varies with position on the keyboard. Typically, the higher notes have a lighter feel than the bass notes, simulating an acoustic grand piano key action.

granular synthesis. A type of digital additive synthesis that uses very short sounds called *grains*, which can be combined and processed in a variety of ways. Granular synthesis can create complex sounds, but it requires a powerful computer.

graphic equalizer. A type of audio equalizer with a separate level control (usually a slider) for each of a number of frequency bands. The name "graphic" EQ comes from the fact that the curve or arrangement of the control sliders is visually analogous to the response of the unit. Standard sizes/formats include 15 bands per channel (where each band covers 2/3 octave) and 30 or 31 bands per channel (where each band covers 1/3 octave). Graphics equalizers are easy to use but do not provide the flexibility and precision required for many studio tasks. Graphic EQs are often used to shape the sound of live sound systems and to control feedback in stage monitor systems. See Figure G.1.

graphical user interface (a.k.a. GUI). A computer operating system in which the user interacts with the computer by responding to and manipulating graphic elements, such as icons, menus, switches, knobs, and other items, many of which are analogous to real, physical items. For example, a graphical user interface might allow the user to store data in a folder on the screen, which functions in the same way a real folder does for storing papers. See Figure G.2.

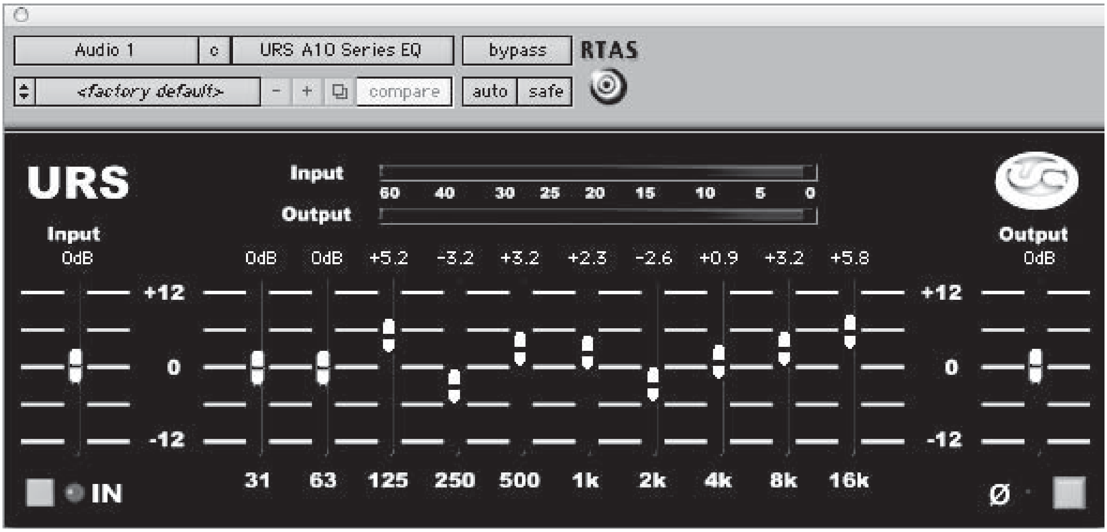

Figure G.1 A type of equalizer that has a slider for the level of each band of frequencies, but that does not allow control over bandwidth or frequency center, is called a graphic equalizer.

Figure G.2 A computer's GUI, or graphical user interface, consists of a variety of elements, such as icons, menus, switches, control panels, knobs and other controls, folders, and more.

Green Book. One of a set of "Rainbow Books" with colored covers containing the Sony/Philips-developed specifications for different optical compact disc formats. The Green Book contains the specification for CD-Interactive or CD-I.

grid. An electrode in a vacuum tube that controls the flow of electrons.

grid mode. A MIDI and audio editing mode found in some DAWs where data is locked to a grid representing rhythmic divisions, such as eighth notes, sixteenth notes, sixteenth-note triplets, and so on; all new data or edited data snaps to the nearest grid location and conforms automatically to the preset rhythmic value of the grid. This can make it very easy to add data or to edit drum parts without messing up the rhythms.

grille. The acoustically transparent front panel of a speaker or guitar or bass combo amplifier. The grille is intended to protect the speaker drivers from damage, as well as to provide a visually attractive look for the front of the unit.

grille cloth. Acoustically transparent fabric used to protect and conceal drivers in a speaker cabinet.

groove. 1. A rhythmic pattern in which the rhythm has been performed or manipulated in such a way as to have a particular "feel." This may involve pushing/pulling certain beats, shortening or lengthening certain time intervals or rhythms within the larger rhythm, or delaying or rushing certain beats to create a specific effect. 2. The continuous spiral on a vinyl LP record that contains a representation of the audio data. There is only one groove on a record, which begins at the outside and spirals toward the center of the LP.

groove quantize. A function found in sequencers and DAWs that quantizes a MIDI track or selected MIDI data on a track to match the rhythmic feel of another MIDI performance—another track, a preprogrammed rhythm, a MIDI loop, or other rhythmic material.

ground (a.k.a. earth). The zero reference for voltage in a circuit or device (typically connected eventually to the actual earth or ground). Voltage on a "hot" conductor is measured against ground, which is at zero potential or volts.

ground lift. 1. A method or device—often a switch on a device—that disconnects a signal's connection to ground in order to prevent ground loops or hum. 2. A 3-to-2 prong AC adapter that removes the connection to an AC outlet's ground conductor. These adapters are *never* a good idea . . . don't use them!

ground loop. A situation in which a system or signal has multiple paths to ground available at once. This results in 60-cycle hum; the intensity of the hum depends on the particular ground loop situation in a system.

group. A number of faders or channels on a mixer (whether hardware or software) that have been linked together so they can be controlled by a movement of any one of the faders or channel controls. Different mixers will allow different controls to be grouped; some only support faders, other add support for pan, still others might allow grouping of other controls, such as solo, mute, and others. The faders and controls in a group will usually move relative to one another so that the relative position of each control stays the same compared to the other controls. ☞ **See also** *fader group*.

group delay. The time it takes for a signal to pass through a circuit, referenced or plotted against frequency. In a filter, some frequencies may be delayed slight more than other frequencies, resulting in phase distortion within the signal being filtered.

GSIF. GigaSampler Interface. A driver that supports streaming of samples from a hard drive so that longer samples can be played than will fit in a sampler's or computer's RAM or that can be efficiently handled from within RAM. Named for GigaSampler, one of the first software samplers, and the first to offer streaming of samples from a hard drive—GigaSampler was developed by NemeSys Music and later acquired and eventually discontinued by TASCAM.

guard band. A blank track on an analog tape between a track used for time code and the next track used for audio. The guard band prevents the time code signal from bleeding into the audio track.

GUI. ☞ **See** *graphical user interface*.

guide track. ☞ **See** *scratch track*.

guide vocal. ☞ **See** *scratch vocal*.

H

Haas effect (a.k.a. precedence effect). A psychoacoustic phenomenon caused by short delays creating an apparent change in the stereo balance in a signal. If a signal is split to come out of both the left and right channels of a sound system, but one side is delayed, the dry (un-delayed) signal will seem louder. A delay of even a few milliseconds will create this effect; the longer the delay, the more intense the effect.

half damper. A piano performance technique, originally for acoustic pianos, in which the sustain pedal is pressed partway down, which partially raises the dampers off the strings. Some digital pianos are able to emulate this effect.

half duplex. A device that supports input and output, but only in one direction at a time.

half normal. A type of normal connection in a patch bay in which a signal fed into the top back jack of the bay will automatically be connected to the bottom back jack of the bay. Plugging a cable into the front top jack does not break the "normal" connection to the bottom back; instead, it splits the signal to feed whatever is plugged into the front and continues to feed the normal connection out of the bottom back row, creating a Y. Plugging into the front bottom jack does break the "normal" connection to the bottom rear jack, allowing the signal to be rerouted. ☞ **See also** *normal*.

half space. A sound source, such as a speaker, located against a surface, such as a wall, is said to be in *half space* (versus being in free space, where sound is able to travel in any direction). Half-space placement of a speaker or other sound source results in a 3-dB level increase (mainly in the low frequencies) over free-space placement. ☞ **See also** *free space*, *quarter space*, *eighth space*.

half-track. An analog tape recorder head format in which two channels cover the entire width of the tape.

hall reverb. A type of digital reverb preset that is intended to re-create the ambience of a large hall.

hammer action. A type of key action that incorporates the same type of mechanical hammers as are found on a real piano, in order to give the player the feel of playing an acoustic piano. Hammer-action keybeds are generally found on 88-note keyboards.

handling noise. Noise created in a microphone by physical handling, movement, vibration, and shock. Some microphones have internal shockmounts to reduce or eliminate this problem, particularly models intended for handheld use onstage.

handshake. A signal that verifies connection and communication between two devices or programs.

hangover. In addition to the obvious definition, hangover is the tendency of a speaker cone to keep moving after signal from its amplifier stops or after a transient passes. Damping factor is used to control hangover.

hard disk. Technically, the spinning oxide-coated Mylar disk inside a hard drive that is used to store data. In common use, *hard disk* refers to the enclosure, drive electronics and mechanism, and the disk itself. ☞ **See also** *hard drive*.

hard drive. A unit containing an enclosure, drive and interfacing electronics, motors and other mechanical elements, and an oxide-coated disk, all used to create a device that can store and read large amounts of computer data at high speed.

hard knee. The mode of operation of most compressors, which immediately start attenuating the signal according to the ratio as soon as the signal crosses the threshold (see Figure H.1). ☞ **See also** *soft knee*.

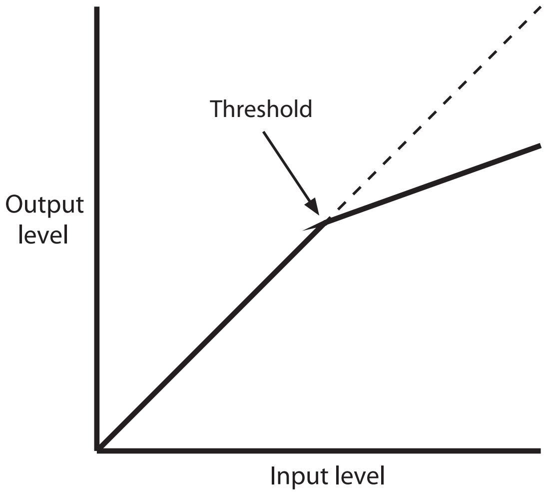

Figure H.1 Most compressors use hard-knee operation, in which gain reduction is applied as soon as the signal crosses the threshold.

hard reset. A command or function that returns a device to its initial power-on state—as it was when the power was first turned on or when the device left the factory. All memory is cleared, and all settings and parameters are returned to their default states.

hardware. A device or item made of actual physical components and parts.

hardware bypass. ☞ **See** *true bypass*.

hardwired. 1. Connected with physical cables and wires. 2. A setting, routing, or parameter that is permanently set and cannot be changed by the user.

harmonic. A component frequency in a waveform occurring at an integer multiple of the fundamental's frequency. (Technically, the fundamental is also a harmonic.) ☞ **See also** *harmonic series*, *fundamental*, *overtone*.

harmonic distortion. Distortion added to a signal in the form of additional harmonics or changes in the relative levels of the original sound's harmonics.

harmonic series. One of a series of frequencies including and related to the fundamental frequency of a tone. The harmonic series consist of integer multiples (1×, 2×, 3×, 4×, and so on) of the fundamental. For example, the harmonics for a 1,000-Hz tone are 1,000 Hz, 2,000 Hz, 3,000 Hz, 4,000 Hz, and so on.

Harmonizer (a.k.a. pitch shifter). Harmonizer is a term trademarked by Eventide and used in association with the company's various models of pitch-changing processors.

harmony generator. A device, plug-in, or program that changes the pitch of a signal to create harmonies that can be blended with the original signal to create chords or harmonized melodic lines.

HDCD. High-Definition Compatible Digital. A type of audio compact disc that supports 20-bit resolution. A compatible drive is required to play back the audio with full resolution. HDCD discs are backward compatible with regular CD players, though only with 16-bit resolution. HDCD has not seen wide acceptance in the marketplace.

HDMI. High-Definition Multimedia Interface. An all-digital connection protocol that carries both audio and video signals, allowing single-cable connections. Some computer monitors offer HDMI connections.

head. 1. A transducer that reads the magnetism on a tape or hard disk and converts it into electrical signals, converts electrical signals to magnetism in order to store them to tape or disk, or uses magnetism to "erase" stored data or signals on a tape or disk. 2. An instrument amplifier that is in a separate enclosure from its complementary speaker cabinet. 3. The beginning of a tape (as opposed to the end or tail).

head crash. A disk failure in which a head contacts the moving platter surface, damaging the magnetic surface and causing permanent data loss.

header. A part of a file that contains information about that file, such as creation and update dates, file size, sample rate, resolution, number of channels, and more.

head gap. The distance between the north and south poles of the magnet in a tape recorder head.

head stack. The part of a tape recorder that contains the tape heads.

headphone amp. An amplifier designed and optimized for raising the level of a signal from line level to sufficient power to drive headphones. Some headphone amps have multiple channels, each with their own input and output so that each musician in a group can be fed his or her own mix with its own volume level.

headphone output. An output, usually a 1/8-inch or 1/4-inch TRS jack, that is intended to drive headphones.

headphones (a.k.a. cans, phones). A type of monitoring device featuring a cup or pad that contains small drivers for each ear. The ear cups are joined by a headband that holds them in position on the listener's head. There are two types: open (lets sound in and out of the headphones) and closed (seals the sound into the headphones and isolates the listener from external sound). There are also two types of physical designs: circumaural, in which the ear cups encircle the ear and the cup rests on the side of the listener's head, and supraaural, which features pads that rest on the listener's ears. Headphones typically have 1/8-inch or 1/4-inch TRS jacks for connecting to a headphone amplifier or headphone output.

headroom. Dynamic range available between the normal or nominal operating level and the onset of clipping or distortion. Headroom is important in a variety of situations, such as when dealing with combining signals together when mixing (which adds level) and for cleanly handling transients and other signals that have wide, fast level changes. See Figure H.2.

heat. The transfer of energy due to a temperature difference. Heat is the enemy of electronic equipment.

heat sink. A device that is designed to dissipate heat from sensitive electronic components that are not able to dissipate heat quickly enough on their own.

Helmholtz absorber. An acoustic device consisting of a resonator that vibrates in response to sound waves at a particular frequency or range of frequencies. In practice, a Helmholtz absorber is a box enclosing a volume of air, with a series of slits or holes in one surface. Air motion due to sound waves causes the absorber to resonate in much the same way as blowing across the opening of a soda bottle creates a tone. The energy required to cause the absorber to resonate reduces the sound energy in the room. Named for Hermann von Helmholtz, a German physicist and physiologist who wrote the book *On the Sensations of Tone*.

Helmholtz device. ☞ See *Helmholtz absorber*.

Helmholtz resonator. ☞ See *Helmholtz absorber*.

hertz (a.k.a. Hz, cycles per second). The number of vibrations or complete cycles of a sound wave occurring within a second. Named for Heinrich Hertz, a late-19th-century physicist who first investigated and artificially produced radio waves.

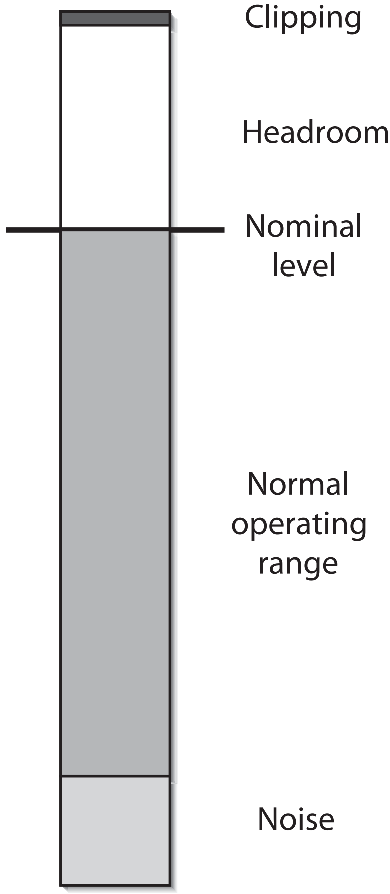

Figure H.2 Headroom is the dynamic range above a device's nominal level and below clipping.

hexadecimal (a.k.a. hex). A number system that uses 16 as its base, rather than 10, as in our standard system. The hexadecimal numbering system uses 16 values: 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, and F, which represent the decimal values 0 through 15. Hexadecimal provides a convenient way to represent binary numbers, as each hexadecimal digit can represent four bits (also know as a *nibble*), or two hex digits can represent a full 8-bit byte. See Table H.1; to represent longer binary numbers, multiple hexadecimal digits are used, one digit for each four bits. For example, the binary number 0010110110001110 would be split into groups of four bits (0010 1101 1000 1110) and

Table H.1 Decimal, Binary, and Hexadecimal Numbering

Decimal	Binary	Hexadecimal
0	0000	0
1	0001	1
2	0010	2
3	0011	3
4	0100	4
5	0101	5
6	0110	6
7	0111	7
8	1000	8
9	1001	9
10	1010	A
11	1011	B
12	1100	C
13	1101	D
14	1110	E
15	1111	F

represented as 2D8E in hexadecimal. Hexadecimal numbers are occasionally encountered by users in the deepest levels of MIDI programming and in MIDI implementation charts.

hexaphonic pickup. A type of guitar pickup made of six separate smaller pickups, one for each string. Hexaphonic pickups are most commonly seen in MIDI guitar conversion systems, where each string's motion can be converted to MIDI data independently and is assigned to its own MIDI channel. In a few cases, hexaphonic pickups are used to produce a separate analog instrument-level output for each string so that each string's signal can be processed and amplified separately, allowing unique panning and sonic possibilities.

HF. ☞ See *high frequency*.

HFS. Hierarchical File System, a.k.a. Mac OS Standard. A hard drive file system developed for Macintosh computers in 1986. Files up to two gigabytes were supported, with file names up to 31 characters long and a maximum of 65,535 files per volume.

HFS+. Hierarchical File System Plus, a.k.a. Mac OS Extended. An updated version of Apple's HFS, which allows file names up to 255 characters long, files sizes over two gigabytes, and unlimited files per volume.

hierarchical menu. A menu structure consisting of directories or menus that branch off from the main menu and continue branching to various levels. See Figure H.3.

high-cut filter (a.k.a. low-pass filter). A filter that reduces the frequencies above the cutoff frequency and allows frequencies below the cutoff frequency to pass through unchanged.

high definition. An increased definition or resolution compared to "normal." In audio terms, this would typically mean 20- or 24-bit resolution versus normal 16-bit resolution. In common use, higher sample rates (above the normal 44.1 or 48 kHz) are included when discussing high definition.

high end. 1. A term used to refer to high frequencies and the treble range of the frequency spectrum. 2. A term used to refer to expensive pieces of gear.

high frequency. A term used to refer to the highest frequencies in the audio spectrum; there is no specific high-frequency range, though in general, the range of 6,000 Hz to 20,000 Hz would qualify.

high impedance. An impedance above 600 ohms, typically several thousand ohms or more. This term is most often encountered with instrument inputs, such as those for guitar or bass, and with direct boxes. ☞ See also *impedance*.

high-pass filter (a.k.a. low-cut filter). A filter that passes frequencies above a certain point and reduces the level of frequencies below that point. High-pass filters are used to reduce rumble, as well as boominess caused by excessive low-frequency levels. See Figure H.4.

High Sierra Format (a.k.a. HSF). A CD-ROM standard created by a group of 12 computer hardware manufacturers known as the *High Sierra Group*

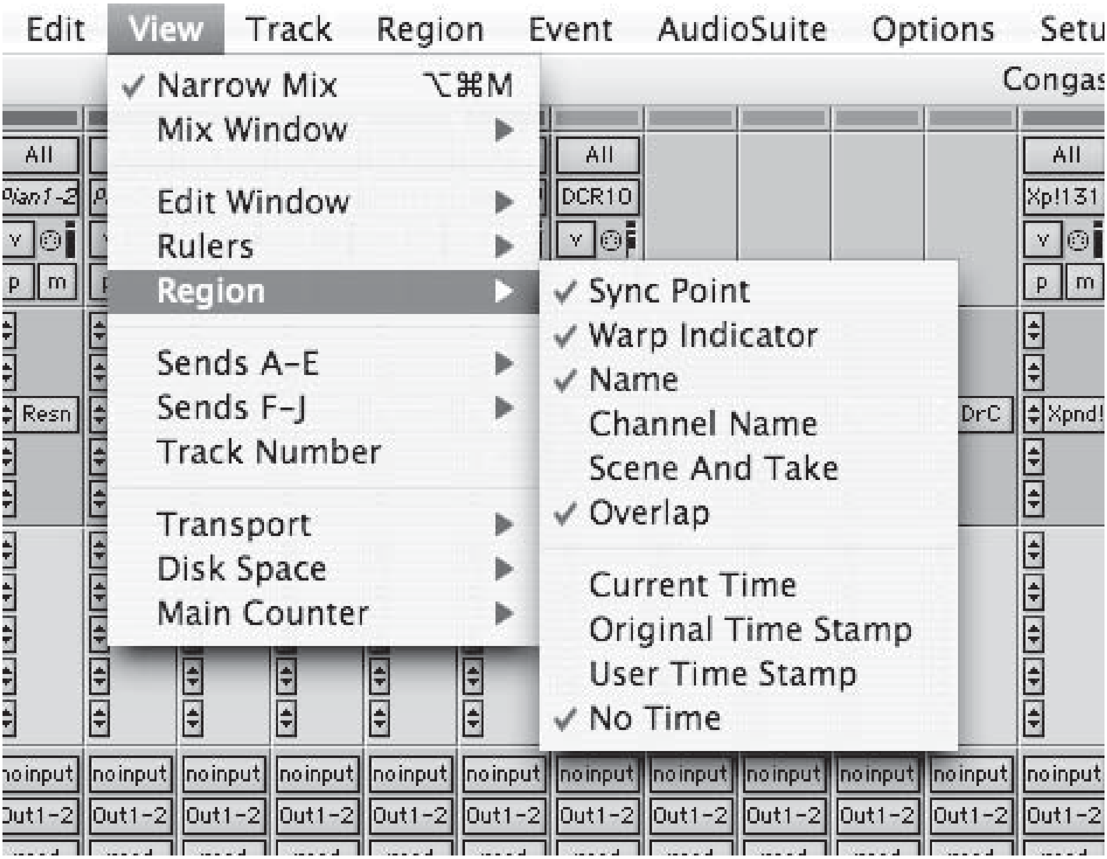

Figure H.3 A hierarchical menu structure has submenus that branch off from the main menu.

because they held their meetings at the High Sierra Hotel and Casino in Lake Tahoe, CA. The High Sierra Format was developed in 1986 based on the Yellow Book specification and later became the standard upon which the ISO 9660 format was based.

High-Z. ☞ See *high impedance*.

hiss. High-frequency noise.

histogram. Technically, a bar graph. This term is sometimes used by manufacturers to refer to LCD or LED meters resembling bar graphs that display frequency or amplitude information over time. See Figure H.5.

Hi-Z. ☞ See *high impedance*.

hold. A parameter on a noise gate that sets the amount of time the gate remains open after the input signal drops below the threshold. Carefully setting the hold parameter prevents the gate from chopping off the quiet decay portion of a sound or reverb tail.

horn loaded. A driver that is attached to a horn or that is mounted into a horn-shaped cabinet in order to control dispersion or to increase throw (projection) of sound over longer distances.

host. 1. The computer on which a program is running or to which a peripheral device is attached. 2. A program, such as a DAW, in which plug-ins are running.

host-based. 1. Software that runs on a computer without the need for any additional DSP or other hardware. ☞ **See also** *native*. 2. A plug-in that operates within another piece of software (the host).

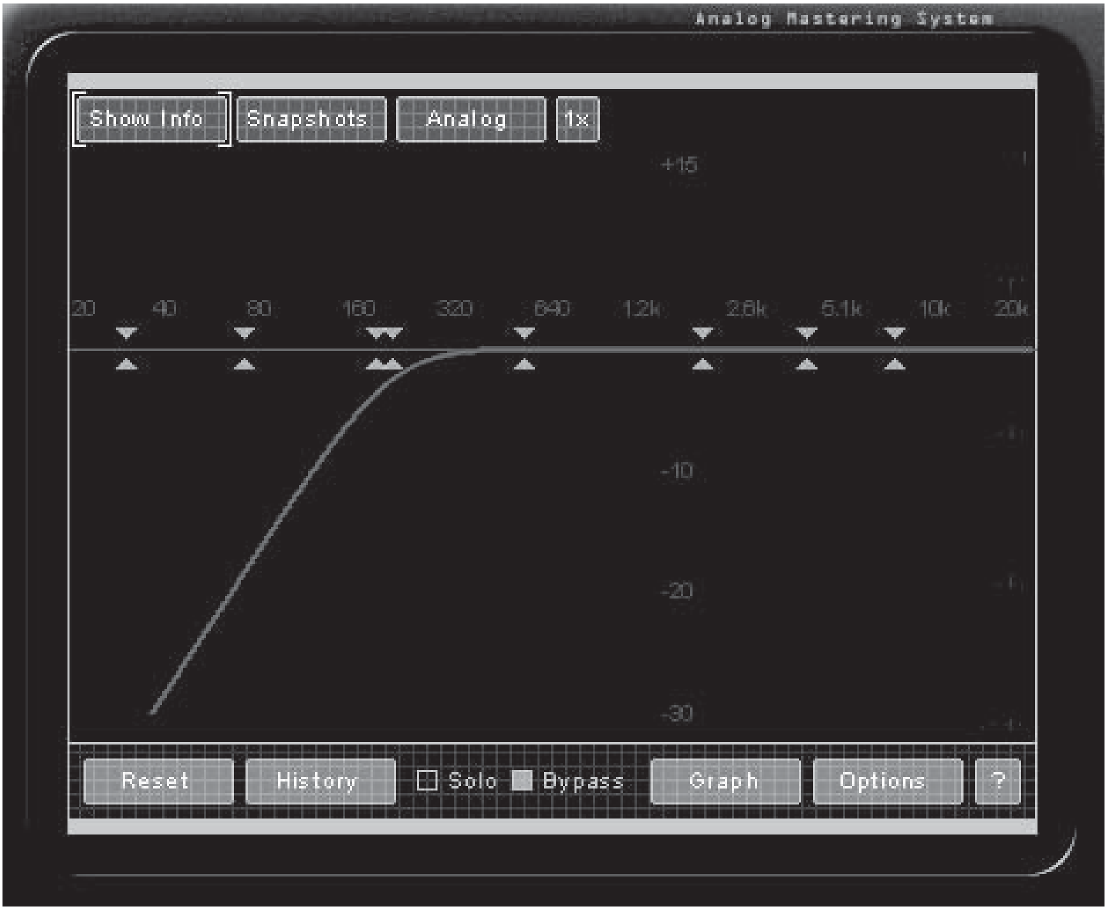

Figure H.4 A high-pass filter allows frequencies over the cutoff frequency to pass unaffected, while reducing the level of frequencies below the cutoff frequency.

host computer. ☞ See *host*.

hot-pluggable. A device that can be safely connected to, or disconnected from, a system while the power is turned on.

hot spot. A position in a room where there is a boost at a particular frequency or range of frequencies.

hot-swappable. 1. ☞ **See** *hot-pluggable*. 2. A type of media that can be safely removed from a drive chassis (such as a removable hard drive) without danger of losing data.

house sync. ☞ See *blackburst*.

HTDM. Host TDM or Host Time Division Multiplexing. A hybrid of native processing (which runs on the host computer's CPU) and DSP-based TDM processing developed by Digidesign that allowed native plug-ins to appear as TDM plug-ins to a Pro Tools system. ☞ **See also** *TDM*.

hub. 1. The center part of a tape reel, which attaches to the tape recorder. The tape is wound around the hub and held in place by flanges, the sides of the reel, which attach to the hub. 2. A device for connecting multiple devices to a single computer port,

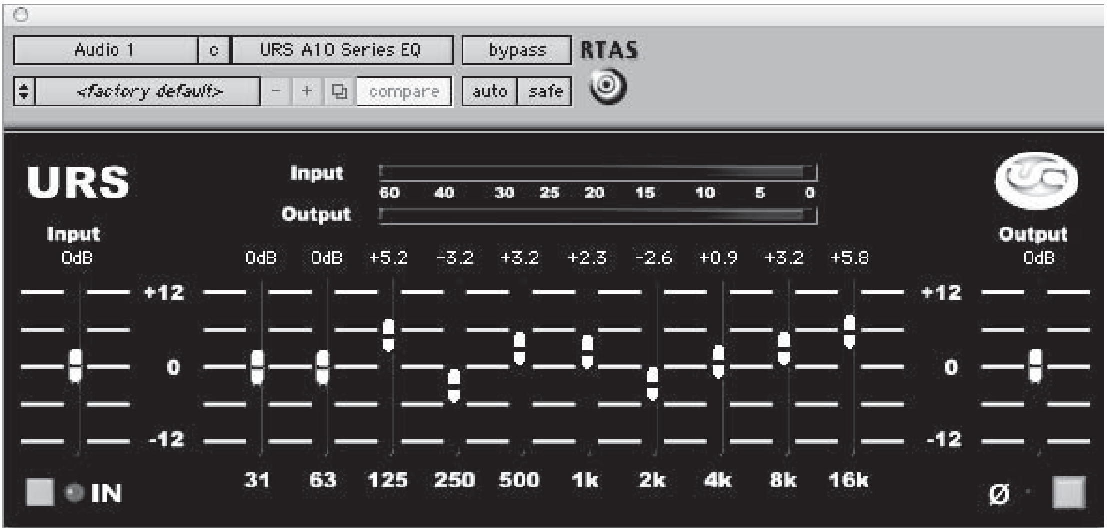

Figure H.5 Because they resemble bar graphs, certain types of meters are referred to as histograms. In this example, there is a histogram (to the right of the left/right input meters) of the waveform peaks in the input signal.

such as a USB hub, which allows several USB devices to connect to a single USB port.

HUI. Human User Interface. The HUI was a control surface with knobs, faders, and switches like a mixer (though it was a control surface and does not pass sound itself) that was developed and manufactured by Mackie Designs. The HUI had a generic protocol that was compatible with a number of DAWs. Subsequent control surfaces from Mackie and other companies have offered support for the HUI protocol, which has become a de facto standard.

hum. A type of low-frequency noise that is typically caused by grounding problems or EMI. Hum of this type generally is centered at 60 Hz based on the AC power frequency—and thus called *60-cycle hum*—with harmonics sometimes audible at 120 Hz, 180 Hz, and so on. In Europe and some other areas, the frequency of AC power is 50 Hz, resulting in hum at 50 Hz with harmonics at 100 Hz, 150 Hz, 200 Hz, and so on.

humanize. A function found in MIDI sequencers that attempts to manipulate MIDI notes (especially those that have been heavily quantized or that have been entered in step time or to a grid) to make them seem more natural and "human." Typically, this is accomplished by adding random variations to note timing, velocity, duration, and other parameters. See Figure H.6.

humbucking coil. A coil of wire that is intended to pick up noise, which can be inverted in polarity, and then used to cancel the noise in a desired signal path.

HX Pro. A system for analog tape developed by Dolby Laboratories that increased high-frequency headroom, reducing the tendency of loud high frequencies to erase their own signal or to distort due to oversaturation.

HVAC. Heating, Ventilation, and Air Conditioning. The ducts and climate control equipment in a building. In recording studios, HVAC must be installed and managed carefully in order to minimize the noise it creates.

hybrid. Something that combines more than one technique or technology. A piece of gear might be a hybrid of tube and solid-state technology, or digital and analog, or various other things.

hybrid synthesizer. A synthesizer that combines two or more types of synthesis to create sounds.

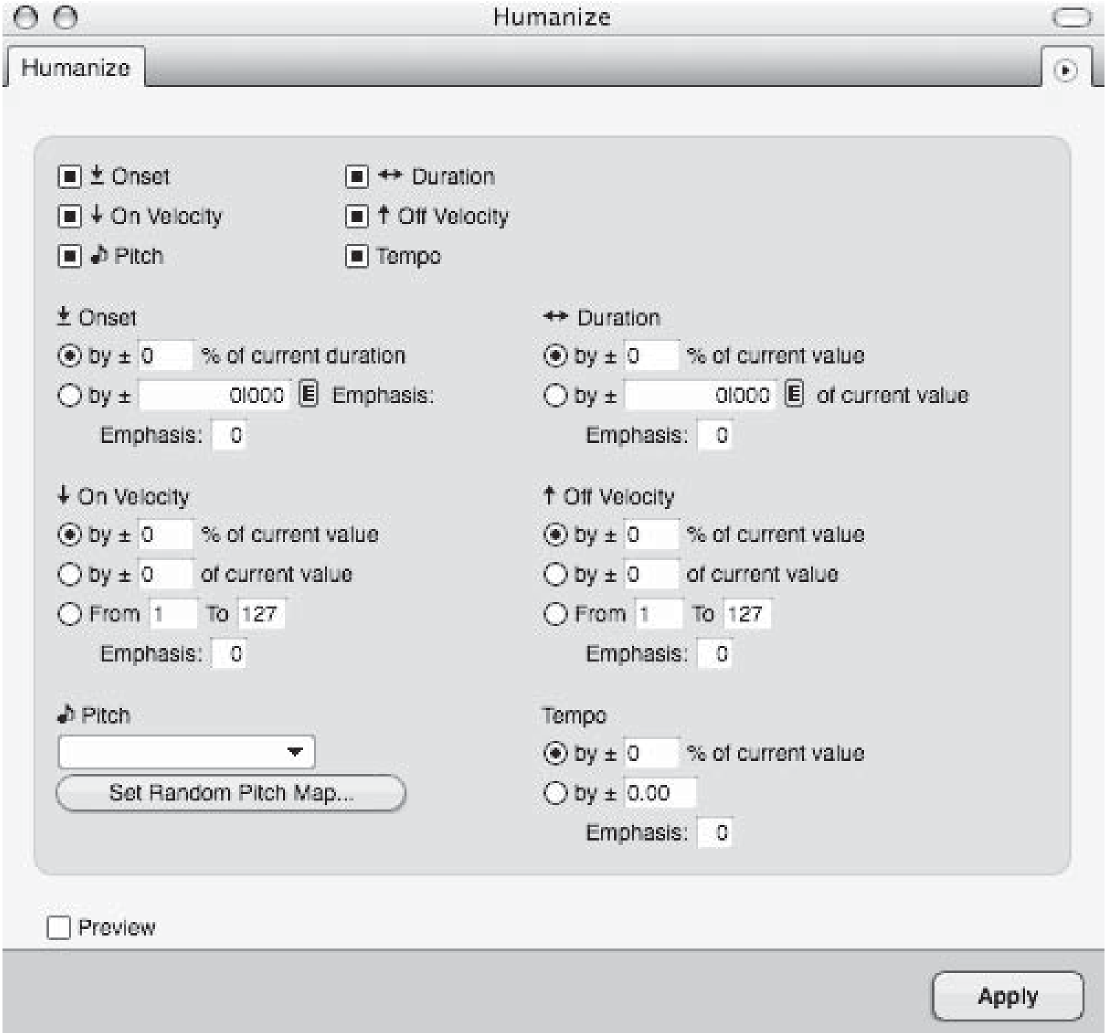

Figure H.6 Many DAWs and sequencers have a "humanize" function that can randomize a number of parameters by small amounts to make a track seem more natural.

hyperacusis. A hearing problem caused by damage to the inner ear, where quiet and moderate-level sounds seem too loud. Around 40% of people with tinnitus also suffer from hyperacusia.

hypercardioid. A microphone polar pattern that is more directional than cardioid and that has a small rear lobe (see Figure H.7). There are two null points, typically around 150 and 210 degrees. ☞ **See also** *cardioid*, *polar pattern*.

hysteresis. 1. Literally, to lag behind. The term is used to describe the tendency of a change in magnetism to lag behind the application and removal of a magnetic field, the response to an input signal being different than the response to an output signal, the

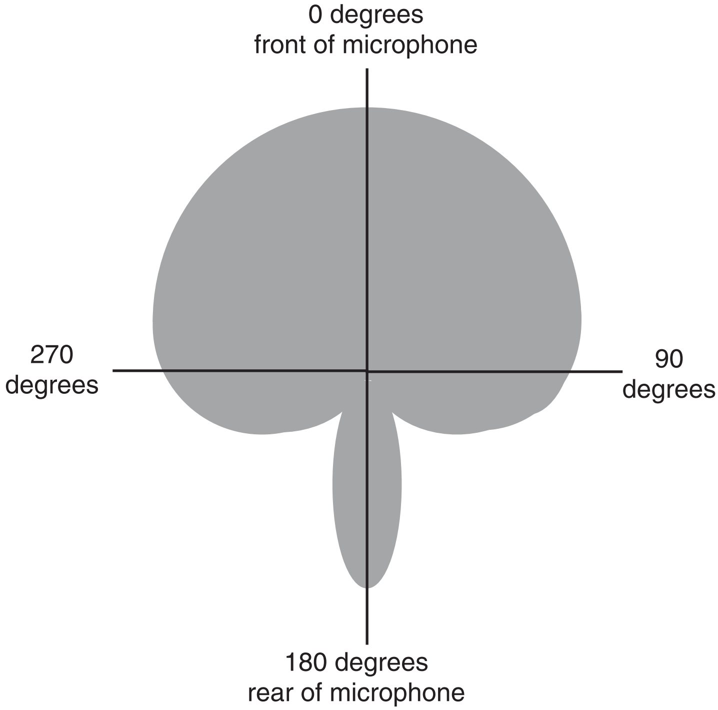

Figure H.7 The hypercardioid microphone polar pattern is more directional than the cardioid pattern and has two nulls, creating a lobe to the rear of the microphone.

action of a gate, and other things. 2. A control found in some noise gates that sets a second threshold for closing the gate. The gate opens when the signal crosses the threshold setting, but does not close until it crosses the hysteresis setting. This allows better control over how the gate operates and helps prevent desired signals from being chopped off when the gate closes.

Hz. ☞ See *hertz*.

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I

IAC (a.k.a. Interapplication Communication). A Macintosh driver that allows different programs running on the same computer to synchronize to one another and transfer data back and forth.

IACC. ☞ See *interaural cross correlation*.

IC. Integrated Circuit, a.k.a. microchip or chip. A semiconductor electronic component that contains thousands of miniaturized components, such as transistors, capacitors, and resistors. A single IC can replace a large circuit board full of discrete components, resulting in the ability to design much more efficient, compact circuits containing the functionality of thousands of components.

icon. A graphic or symbol in a GUI (*graphical user interface*) that represents a file, application, storage device, peripheral device, or other item.

IDE. Integrated Drive Electronics. ☞ See *ATA*.

IEC. International Electrotechnical Commission. An international organization founded in 1906 that develops standards for electrical and electronic safety and performance. www.iec.ch.

IEC 958. The standard developed by the IEC that contains the specifications for AES/EBU (IEC 958 part 4) and S/PDIF (IEC 958 part 3) digital transfer protocols.

IEEE. Institute of Electrical and Electronics Engineers. A trade organization founded in 1884 that develops standards for electronics. www.ieee.org.

IEM. In-Ear Monitor. An earpiece designed to be worn by a musician when recording in a studio or performing onstage. The in-ear monitor is designed to replace headphones and other types of monitors by providing a monitor feed while also isolating the musician from noise in the room.

IID. ☞ See *interaural intensity difference*.

IIR filter. Infinite Impulse Response filter. IIR filters emulate the way capacitors and other analog components function. IIR filters are most often used in digital equalizers (primarily plug-ins) that are designed to function in similar fashion to analog equalizers. ☞ **See also** *FIR filter*.

iLok. A dongle—a hardware copy protection device—developed by PACE that stores authorizations that allow software programs to run. An iLok connects to a computer's USB port and can store authorizations for more than 100 programs, which can be loaded, moved to another iLok, or deleted.

image. A file that contains an exact replica of the data on a floppy or hard disk or a CD or DVD along with the structure of the data on the disk. Image files are often used by CD and DVD burning programs to hold all the data that will be burned to the disc. Image files are also used for backup, archiving, and disk cloning purposes, as well as for data recovery when there is a catastrophic problem with a disk.

image copy. A type of copy that is made without regard for the content being copied. The device or program doesn't even access the data in the file being copied; it simply makes a bit-for-bit copy. Making this type of copy is faster and does not require that the software be compatible with the data file.

image file. ☞ See *image*.

imaging. The ability of a speaker system to provide sufficient directional cues to allow a listener to pinpoint the position of a sound in a stereo or surround mix. Imaging depends on the speakers being precisely matched in level, response, and phase, as well as the acoustic environment of the listening room.

IMD. ☞ See *intermodulation distortion*.

impedance. The resistance of a circuit to alternating current. Impedance is the combination of resistance and reactance (an effect of inductance or capacitance). It is important to match the output impedance of a device to the correct input impedance of the device it is feeding in order to optimize performance or to prevent damage in some cases (such as with speakers and amplifiers). Most modern devices have very high input impedance so they can be driven by the low-impedance outputs of other devices.

import. A software function that allows a program to load, translate, and use data and files created in another program.

impulse. A "spike" of sound of very short duration used to perform acoustical measurements.

impulse response. Literally, how a device or space responds to an impulse. By capturing or recording the response of a room or device to an impulse signal, FFT (*Fast Fourier transform*) analysis can be performed to obtain frequency-, phase-, transient-, and time-related response information that can be used to manipulate other data in a process called *convolution*. See Figure I.1.

in the box. A work flow or production in which all processing and mixing takes place inside a DAW, without using external hardware processors, recorders, or mixers.

inductor (a.k.a. coil). An electronic component consisting of a coil of wire wrapped around a core. Inductors pass direct current but resist the flow of alternating current—higher frequencies are resisted more than lower frequencies.

infant failure. The proven tendency of electronic equipment to fail very early in its life—within the first day or two of use. Some manufacturers attempt

Figure I.1 A convolution reverb such as Trillium Lane Labs, TL Space can use an impulse response created from the sound of a real room to add reverb to a track or signal.

to combat this tendency with an extended "burn in" period before their products are shipped to retailers or end users.

infrasonic. Frequencies below the range of human hearing.

initialize. To restore a device to its default state. Initializing often erases any user data or settings.

inline mixer. A mixer in which each channel has two paths—one for the normal incoming signals from microphones and other sources, and the other for returning signals from a multitrack tape recorder. ☞ See also *split mixer*.

input. A connection or jack used to send signal into a device.

input monitoring. A function on some software and hardware recorders that routes the signal coming in the input connections directly to the outputs so it can be monitored.

input sensitivity. ☞ See *sensitivity*.

input transformer. A transformer used to match the input of a device to the device that is feeding it. ☞ See also *transformer*.

insert. ☞ See *insert point*.

insert point (a.k.a. patch point). A jack or jacks on a mixer that break into the signal path for a channel, allowing a processor, such as a compressor, limiter, or equalizer, to be inserted. Inserts can also often be found on the master outputs of a mixer. See Figure I.2.

insert slot. A point in a virtual mixer in a DAW where a plug-in can be loaded. See Figure I.3.

installer. A piece of software that automatically installs another piece of software. Installers are used to ensure that all the various drivers and other parts associated with an application are installed to the proper locations in the system.

instance. Loading a plug-in into a track in a host DAW creates a single instance of that plug-in. An advantage of software plug-ins over hardware processors or instruments is that multiple instances of the same plug-in can be loaded and active at once, versus a hardware processor, which can only be used on one track at a time.

instantiate. To open or load a plug-in within a host program.

instrument input. A specialized input on a preamp or audio interface intended to accept signals from electric guitars or basses. An instrument input will be set up to accept instrument levels and may have an impedance optimized for instrument pickups.

instrument level. A signal level of around –20 dBu as generated by an instrument, such as an electric guitar or bass. In most cases, a direct box, or a preamp with an instrument-level input, is required to raise the instrument's signal to a level other devices can deal with.

instrument track. A specialized track in a DAW that combines a MIDI track for controlling a virtual

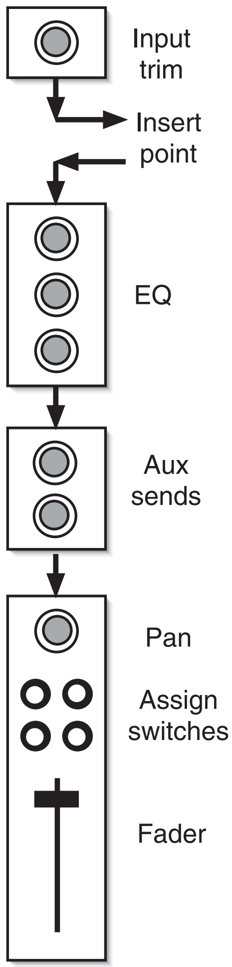

Figure I.2 An insert point can be used to interrupt a mixer channel's signal path so that an external processor can be applied to the signal.

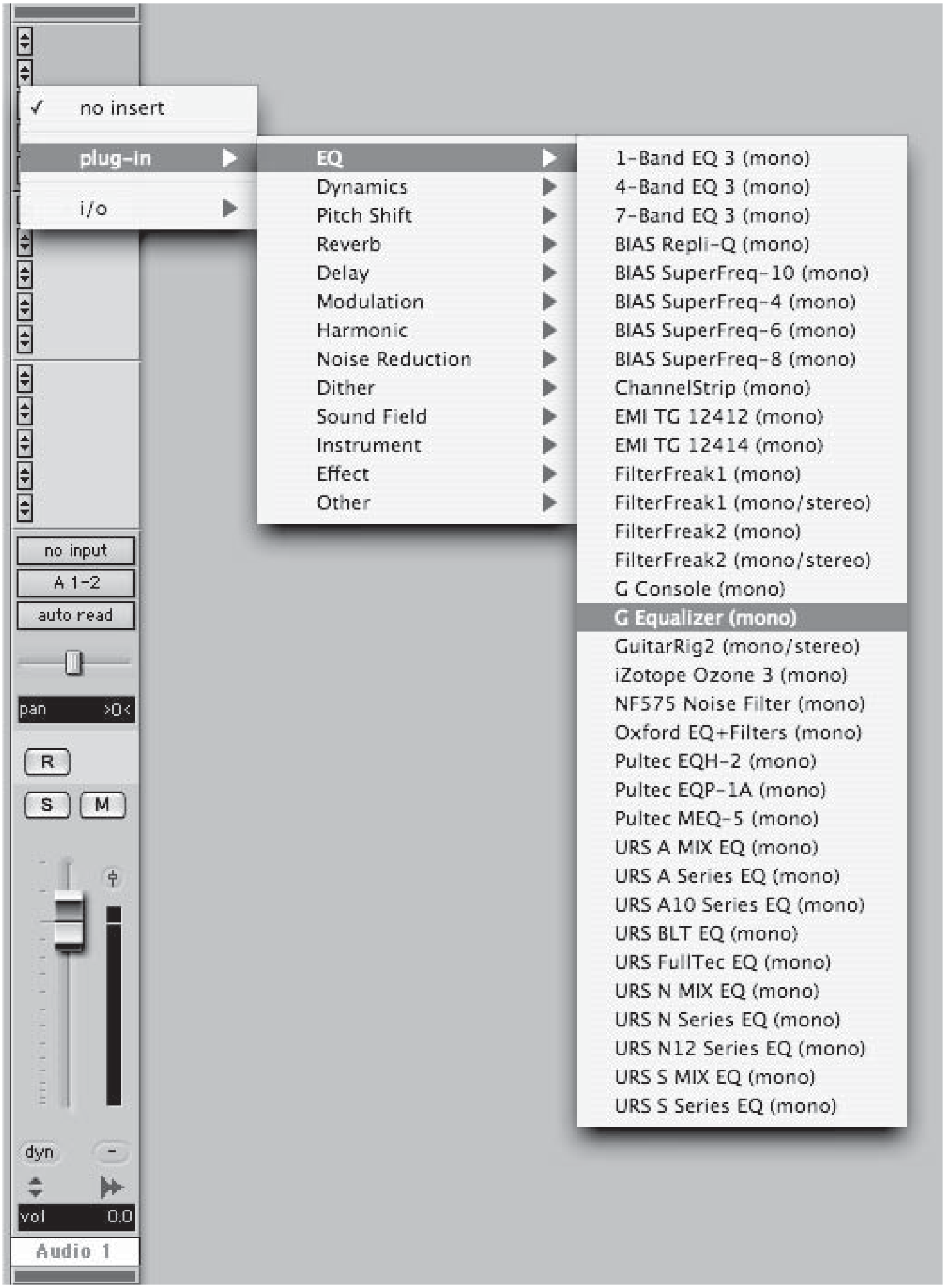

Figure I.3 DAWs feature insert slots—analogous to insert points in hardware mixers—that can accept plug-in processors.

instrument with an audio track that can accept the output signal from the instrument. Before the development of instrument tracks, DAWs used two tracks for each instrument—one for MIDI, and one for the instrument's audio. Instrument tracks provide a more convenient way to deal with virtual instruments.

insulator. A material that does not conduct electricity. Insulators are used to keep signal paths from contacting one another and signals from interacting, and as protection from potentially dangerous voltages.

integrated circuit. ☞ See *IC*.

intelligibility. The ability to discern aural details in a given space.

interapplication communication. ☞ See *IAC*.

interaural cross correlation (a.k.a. IACC). The measure of the similarity of a signal received by the two ears of the listener. Interaural cross correlation indicates how much "envelopment" a listener will feel when listening to a sound.

interaural intensity difference (a.k.a. IID). The difference in the volume in a sound as it reaches the right ear versus the left ear or vice versa, created by "head shadow," or the interference of the listener's head in the sound's path. Interaural intensity difference helps the listener identify the direction of a sound source.

interaural phase difference (a.k.a. IPD). The difference in phase in a waveform as it reaches the right ear versus the left ear or vice versa. Interaural phase difference is affected by the signal's frequency and the interaural time difference. Interaural phase difference helps the listener identify the direction of a sound source and the sound's frequency.

interaural time difference (a.k.a. ITD). The difference in arrival time between a listener's right ear and left ear or vice versa. Though this time is extremely short, it is enough for the human ear to identify the direction of a sound source.

interface. 1. A device that allows other devices to communicate. In the audio/music world, there are two main types: MIDI interfaces, which allow MIDI-compatible devices to communicate with a computer (direct USB connections have made MIDI interfaces obsolete for many users), and audio interfaces, which allow a computer to record and play back high-quality multichannel audio. A variety of other types of interfaces are in use as well: FireWire connects storage devices and other hardware devices with computers, GUIs (*graphical user interfaces*) allow a computer to communicate with users, and so on. 2. To connect or communicate.

interleaved stereo file. An audio file in which two channels of data are combined together for storage as one unit. (The two audio channels remain discrete; only the data is interleaved.)

interleaving. A system of data storage designed to reduce or eliminate errors and to increase read speeds. Interleaving works by breaking up a piece of data and distributing it among other pieces of data so that it is spread across the storage media. Read speed is increased because the data is dispersed in a manner that is efficient for the computer and drive to handle as the disk rotates—the drive spends less time waiting for the correct sector to rotate under the read head.

intermodulation distortion (a.k.a. IMD). New sum and difference frequencies that result from the interaction of two or more other frequencies in a signal.

interpolate. To create a value between other values by estimating, calculating, or averaging.

interrupt. ☞ See *IRQ*.

intersymbol interference (a.k.a. ISI). Smearing of a digital clock pulse that may result in increased jitter. ☞ See also *jitter*.

intonation. Pitch accuracy.

inverse reverb. A reverb with an envelope that produces a slow buildup of reverberation with a quick cutoff.

Inverse Square Law. A physical law that says intensity is inversely proportional to the square of distance. In acoustic terms, this results in a 6-dB drop every time you double the distance from the source; 10 times the distance reduces loudness by 20 dB. Note that as with all decibel-related phenomena, this is all relative: Doubling a one-inch distance to two inches produces the same 6-dB drop that doubling a one-foot distance to two feet does.

I/O. Short for input/output. An abbreviation typically used to refer to the input and output connections on a piece of gear.

IPD. ☞ See *interaural phase difference*.

IPS. Inches Per Second. A measure of tape speed.

IR. ☞ See *impulse response*.

IRQ. Interrupt Request. A signal sent from a piece of software or a hardware peripheral in a PC that tells the operating system to stop or suspend an operation in order to start another operation. Interrupts are given priorities that help the operating system decide how and when to proceed.

ISA. 1. Industry Standard Architecture. An older type of computer expansion slot and bus. 2. Any of a variety of abbreviations and organizations, such as Internet Security and Acceleration, International Society of Arboriculture, International Society of America, International Student Association, Informix Server Administrator, International Summer Award program, and countless others.

ISO. International Organization for Standardization or *Organisation internationale de normalisation*. A non-governmental federation of standards agencies from 130 countries, founded in 1947. www.iso.org.

ISO 9660. A standard developed in 1988 by the International Organization for Standardization based on the older High Sierra file system (HSF). ISO 9660 specifies a file system for CD-ROMs. Various levels of the standard support different file name lengths. Level 1 allows for eight-character file names. Level 2 allows for 31-character file names.

isochronus. A transmission type in which the data must be delivered within certain time limits so that synchronization can be maintained.

isolation. Keeping sound from entering or escaping from a space.

isolation booth. A small room designed to contain, or isolate, the sound of a source so it can be recorded without bleed to or from other sound sources.

isolation cabinet. An enclosure intended to contain the noise from a computer, hard drives, and other equipment. Isolation cabinets are usually equipped with low-speed fans to help keep the equipment inside cool.

isolation transformer. A transformer that is used to electrically isolate one circuit from another circuit to prevent ground loops and other problems. ☞ See also *transformer*.

ISRC. International Standard Recording Code. A code defined by ISO (*International Organization for Standardization*) for identifying a sound recording or music video recording—an ISRC is issued by the RIAA per track, not for the media or collection the track is part of. The ISRC identifies the country of origin, the identity of the person or company registering the recording, the year of recording, and the serial number of the recording. Changes in ownership of the rights to a recording do not change the ISRC. ISRC codes can be embedded in Red Book CDs or encoded into MP3 and other file types. One way ISRC codes are used is for tracking songs on download services such as iTunes.

ITD. 1. Initial Time Delay. The time between the arrival of a direct sound and its first reflection. 2. ☞ See *interaural time difference*.

ITU. International Telecommunication Union. A United Nations agency charged with standardizing and developing telecommunications. www.itu.int.

ITU 775. International Telecommunication Union, Operational Bulletin No. 775. A document containing recommendations for positioning the speakers for a 5.1 surround sound system.

J

jack. The female-gender connector counterpart to the male-gender plug (see Figure J.1). A jack is typically mounted to a panel or piece of gear, whereas a plug is part of an interconnecting cable. There is no defined signal flow for a jack; it can carry either input or output signals. Though the terms *jack* and *plug* are often used interchangeably, there is a distinct difference between the two.

jam sync. A feature of some timecode-generating devices that can generate fresh timecode while receiving timecode from another source. Jam syncing is used to replace old timecode that has been damaged, is unreliable, or is corrupted.

Jecklin Disc (a.k.a. OSS [*Optimal Stereo Sound*] Disc). A stereo miking technique using a pair of omnidirectional microphones separated by a baffle. Jürg Jecklin developed this technique while trying to simulate the qualities of binaural recording using omni mics. The baffle (or Jecklin Disc) is a circular 12-inch disc covered on both sides by absorptive foam. The two microphones are placed on either side of the disc, 17 centimeters apart, and angled outward at 20 degrees from one another (see Figure J.2). The Jecklin Disc technique produces natural sound quality with excellent stereo separation. ☞ **See also** *baffled stereo*.

Figure J.1 A jack is a female connector and is the counterpart to a male plug. Here we see a panel-mount XLR jack on the left and a 1/4-inch jack on the right.

jellyfish meter (a.k.a. goniometer). A type of meter used to display the relationship of stereo or multichannel surround signals (see Figure J.3). A jellyfish meter may show stereo correlation, phase, mono compatibility, and level information.

JFET (*Junction Field Effect Transistor*). A type of transistor often used in audio circuits because it has some response characteristics that are more similar to vacuum

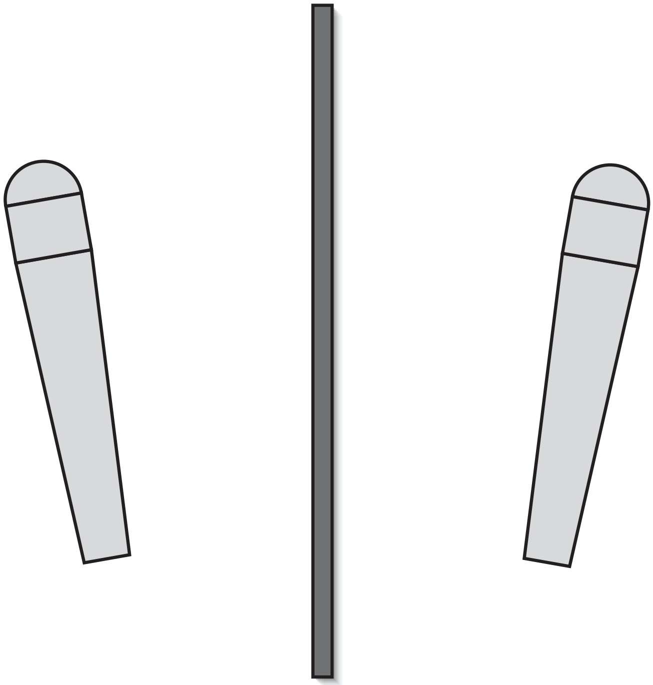

Figure J.2 A Jecklin Disc is used to increase stereo separation when using two omnidirectional microphones. The microphones are placed 17 centimeters apart, angled out at 20 degrees, with the circular, absorptive disc centered between the mics.

tubes than some other types of transistors. JFETs are often used in low-noise, high-impedance applications. ☞ **See also** *FET*, *transistor*.

JIS (*Japanese Industrial Standards*). An organization that defines specifications and standards for a wide range of applications, including audio. www.jisc.go.jp/eng.

Jitter (a.k.a. sample offset uncertainty). Errors in sample timing. Excess jitter can result in distortions and phase problems with digital audio, especially in the higher frequencies. Most digital audio devices use clock buffers to reduce jitter. Another solution preferred by many engineers is to use an external clock source to provide higher-quality clock with less jitter.

jog. A video term that has been adopted by the audio community. To *jog* is to move the transport location in a track by a small amount to find an exact location for editing or playback. Many control surfaces have a jog wheel that allows the user to scroll through the track by small distances. ☞ **See also** *shuttle*.

journaling. A system that records or logs a computer's disk operations in case of power failure or disk problems. Journaling allows the computer to pick up where it left off during a disk operation if there is a problem, as the computer can replay the log and continue from the last point recorded. Journaling is recommended for servers and other applications but may cause access time problems with audio or video applications. Many manufacturers recommend disabling journaling for drives that will be used for audio or video.

joystick. A controller that can transmit data from two axes at once. While joysticks are common for gameplay, they are also used as modulation controllers in keyboards, for controlling the blend of multiple sources, and for positioning sounds in multichannel (for example, 5.1) surround sound fields.

jump drive (a.k.a. flash drive, thumb drive, USB flash drive, USB stick). A compact hardware data storage device based on flash memory. Jump drives generally connect to computers or other devices using USB. They rarely require drivers or other software and are generally plug-and-play or class-compliant.

jumper. A small piece of wire or a small clip that connects two points on a circuit board. Jumpers are often removable so that users can change the configuration or response of a piece of gear.

just intonation. A musical tuning that relies on whole-number ratios. In practice, this relates the pitches to the harmonic series, which comes from whole-number ratios based on a single base tone or frequency. Because the pitches come from ratios based on a specific pitch, it is difficult to change keys in just intonation, as the pitches in each key are tuned slightly differently. However, the intervals within a single just-intonated key will be quite pure. ☞ **See also** *equal temperament*.

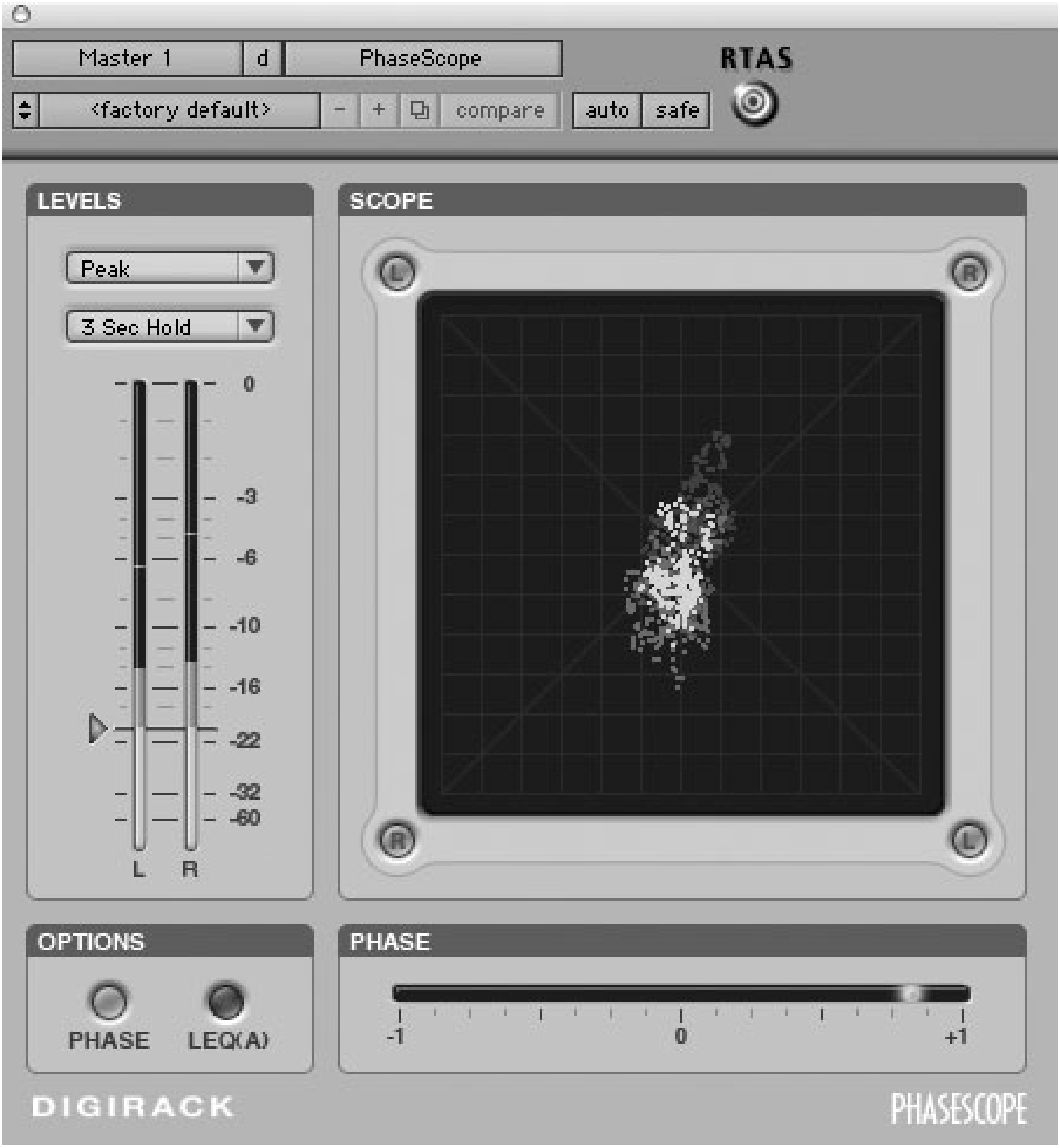

Figure J.3 Jellyfish meters are used to display information about stereo and multichannel surround audio signals.

K

Kb. Short for Kilobit (1,024 bits).

KB. Short for Kilobyte (1,024 bytes).

Kbps. Short for Kilobits per second. A measurement of data flow.

KBps. Short for Kilobytes per second. A measurement of data flow.

kernel. The lowest, most basic level of a computer operating system, containing such services as memory management, hardware interfacing, and so on.

kernel panic. An operating system action resulting from a fatal error in a piece of software that is operating at the kernel level, such as a device driver. Usually the panic action consists of displaying a message and waiting for a reboot. Kernel panic can result from hardware failure, bugs, invalid memory addresses, and other errors.

Kerr effect. The operating principle behind magneto-optical drives, where laser light reflected off a magnetized MO disc changes phase to represent binary data.

key. 1. The pitch center in a musical piece or scale. 2. The main performance component of a keyboard instrument. 3. A control input on a dynamics processor, such as a compressor or noise gate, used to trigger the processor from an external source. ☞ See also *side chain*.

keybed. 1. A rail in a keyboard instrument that stops the downward travel of the keys. 2. The entire key action in a keyboard instrument.

keyboard. 1. An alphanumeric input device used with computers. 2. Also known as a manual. A musical instrument featuring keys that are pressed by the player to sound notes.

keyboard mapping. ☞ See *keymap*.

keyboard scaling (a.k.a. rate scaling, level scaling, envelope scaling). A function of some keyboards that allows the key number to be used to modulate a parameter. For example, keyboard scaling might be assigned to change the filter cutoff as higher notes are played, so higher notes are brighter than lower notes.

keyboard split. ☞ See *split point*.

key click. An attack transient that occurs in some Hammond organs. There are nine contacts under each key, one for each drawbar/tonewheel. As the key is pressed, the contacts close at slightly different times, causing a percussive click at the attack of each note. The random phase of the tonewheel results in the key click constantly changing in timbre, making it difficult to duplicate with a synthesizer.

keygroup. Term used by AKAI samplers for a keymap.

key input. ☞ See *side chain*.

keymap. The assignment of individual samples to a particular key (or keys) on a keyboard. This could be assigning the sampled notes of a piano across the keys to simulate a real piano, or it could be assigning a separate drum sound from a drum kit to each key.

key velocity. ☞ See *velocity*.

kHz. Abbreviation for kilohertz (1,000 hertz).

kick drum tunnel. A technique used by some engineers for recording kick drum. A tunnel is constructed of blankets, empty drum shells, or other materials that extends out from the front of the kick drum. A microphone is placed at the end of the tunnel and picks up a more ambient sound because of its distance from the drum. The tunnel serves to isolate the distant mic from noise bleed from other drums and instruments in the studio.

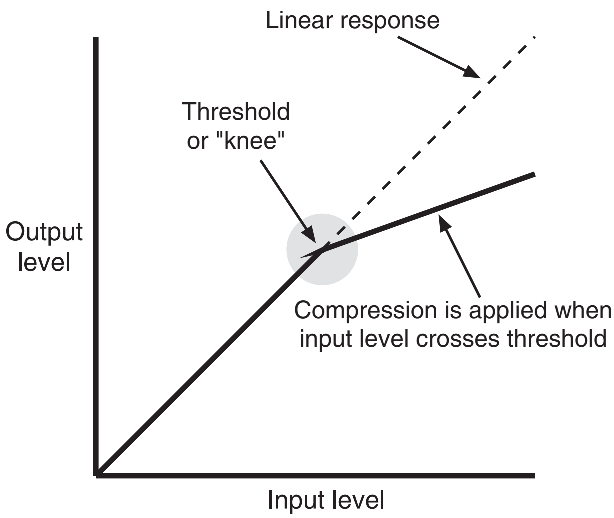

Figure K.1 The knee is the point in a dynamics processor where the response changes—where compression begins to be applied.

kill switch. A switch on some DJ mixers that "kills" or removes certain frequencies. Used as an effect or, when remixing, to remove sounds and instruments so that tracks can be blended together.

kilo. With a lowercase *k*, a prefix meaning 1,000.

Kilo. With an uppercase K, a prefix standing for 1,024, as used in computer and digital systems. Refers to the binary number 2 to the 10th power.

Kilobit. 1,024 bits.

Kilobyte. 1,024 bytes.

kilohertz. 1,000 hertz or cycles per second.

kilowatt. 1,000 watts.

kit. 1. A set of drums or percussion instruments. 2. A British term for gear or equipment.

knee. 1. A joint in a human leg. 2. The point in a dynamics processor where a change in response takes place, where compression begins to occur (see Figure K.1). Hard knee response is a quick change in response once the knee is reached; soft knee is a more gradual, smooth change in response. ☞ **See** *hard knee, soft knee.*

knob. A rotary control or controller.

L

LA synthesis. ☞ See *linear arithmetic synthesis.*

lag processor. An analog synthesizer module somewhat similar to a compressor for control voltages—a lag processor smoothes out peaks in control voltage signals.

LAN. Local Area Network. A number of computers and peripherals in a building or limited area that are interconnected into a network.

land. 1. The area between the pits in a commercially duplicated (which are usually stamped) compact disc. 2. The area between the spirals of the groove in a vinyl LP record.

lane. An area in a DAW window associated with a particular track and dedicated to displaying volume, pan, velocity, or another parameter related to the MIDI notes or audio data in the track. See Figure L.1.

large-diaphragm microphone. A microphone with a diaphragm measuring 3/4-inch or larger in diameter. Most large-diaphragm mics are condensers, though there are a few large-diaphragm dynamic microphones. The large-diaphragm design has more surface area than small-diaphragm designs and is therefore often more sensitive. Large-diaphragms are popular for many studio applications because

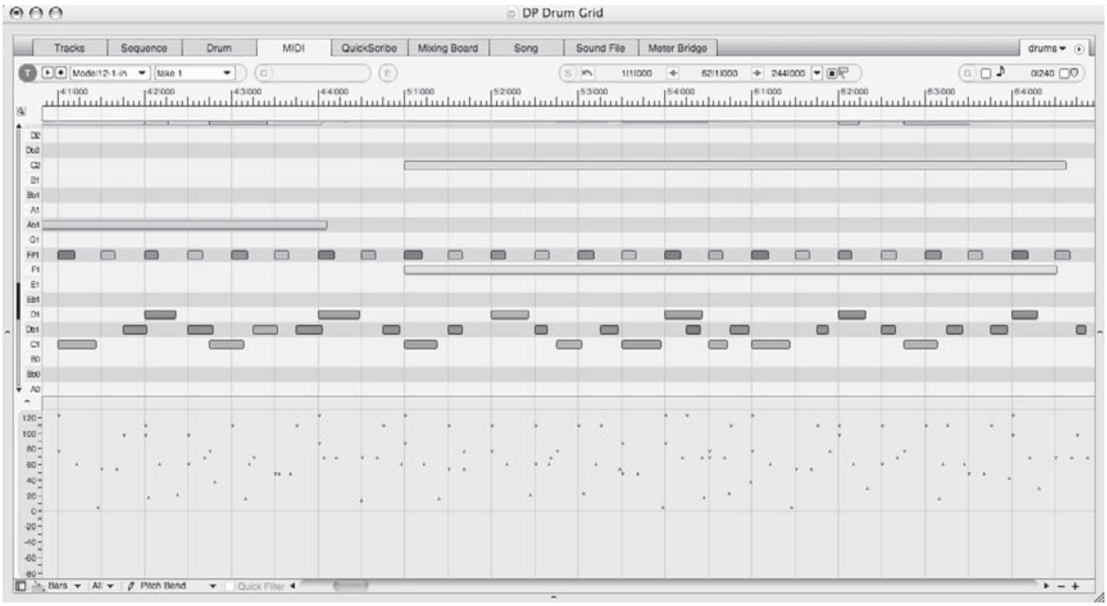

Figure L.1 A lane is used to display control information related to the MIDI or audio data in a track. In this case, MIDI notes are shown in the top section, while velocities are shown in the bottom lane.

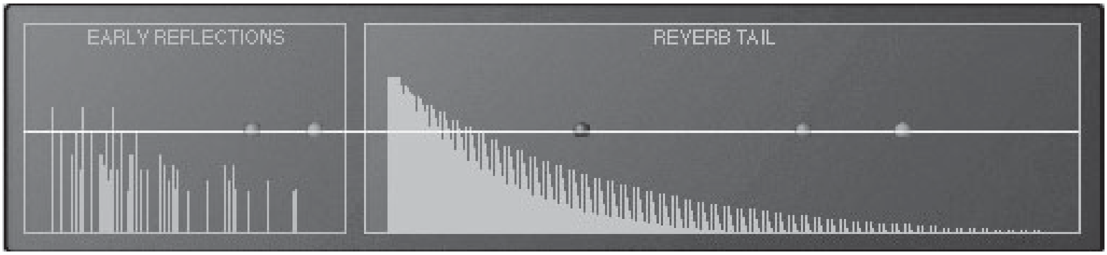

Figure L.2 The late reflections portion of a reverb is the wash of ambience that occurs after the early reflections.

they have a wide frequency range and an associated "big" sound that works well for vocal and many instrumental applications.

laser. Light Amplification by Stimulated Emission of Radiation. A device that produces "coherent" or extremely focused light. In audio, lasers are used for burning and reading optical discs, such as various types of CDs and DVDs.

latency. Any number of time delays, such as: 1. A time delay resulting from an audio signal passing into a DAW when recording audio, then passing back out in order to be monitored by the performer. This type of latency is a critical problem when tracking or when overdubbing on top of existing tracks. 2. The time required for a plug-in to process a signal in a DAW. This type of latency is a critical problem when using plug-ins for parallel processing, and correcting it is essential for maintaining stereo imaging and phase accuracy. 3. The time delay between a MIDI note on message and the note actually beginning to sound. 4. The time it takes for a hard drive to locate and retrieve a piece of data. This type of latency is critical to high-demand applications, such as multitrack audio recording and playback.

latency compensation. A function of most DAWs that corrects for the latency (timing delays) introduced when using plug-ins to process tracks in a multitrack production.

latency-free monitoring. A technique for eliminating the time delay between when a sound enters a DAW to be recorded and when it comes back out in order to be monitored by the musician or performer. Various methods are used, such as monitoring signals through an external mixer before they enter the DAW or using DSP processing power in the computer's audio interface to compensate for signal latency. See Figure L.3.

late reflections. The complex, diffuse "reverberation" or wash of ambience portion of a reverb that occurs after the early reflections. The late reflections give our ears significant cues about the size of a room or space (see Figure L.2). ☞ **See also** *early reflections*.

launch. To start a computer program.

lavalier. A compact microphone designed to be worn attached to clothing. Lavaliers are typically used for speaking applications when a handheld or stand-mounted microphone isn't practical—though creative recording engineers have used them for a variety of interesting applications.

layer. 1. In synthesizers or samplers, a single preset that is triggered by a specific range of values for a MIDI controller. By combining layers in different ways, the instrument can produce composite sounds or can switch or crossfade between layers set to the same MIDI channel using velocity or another MIDI controller. For example, a low velocity might trigger a soft piano sample. A medium velocity might trigger a medium-loud piano sample, and a high velocity might trigger a loud piano sample. Or, completely different sounds might be placed on each layer, or two layers might be set up to sound in response to the same velocity value. ☞ **See also** *crossfade, cross switch*. 2. In recording, to overdub parts to create a fuller sound.

layering. Combining two or more presets or sounds in a synthesizer or sampler to create a new composite sound.

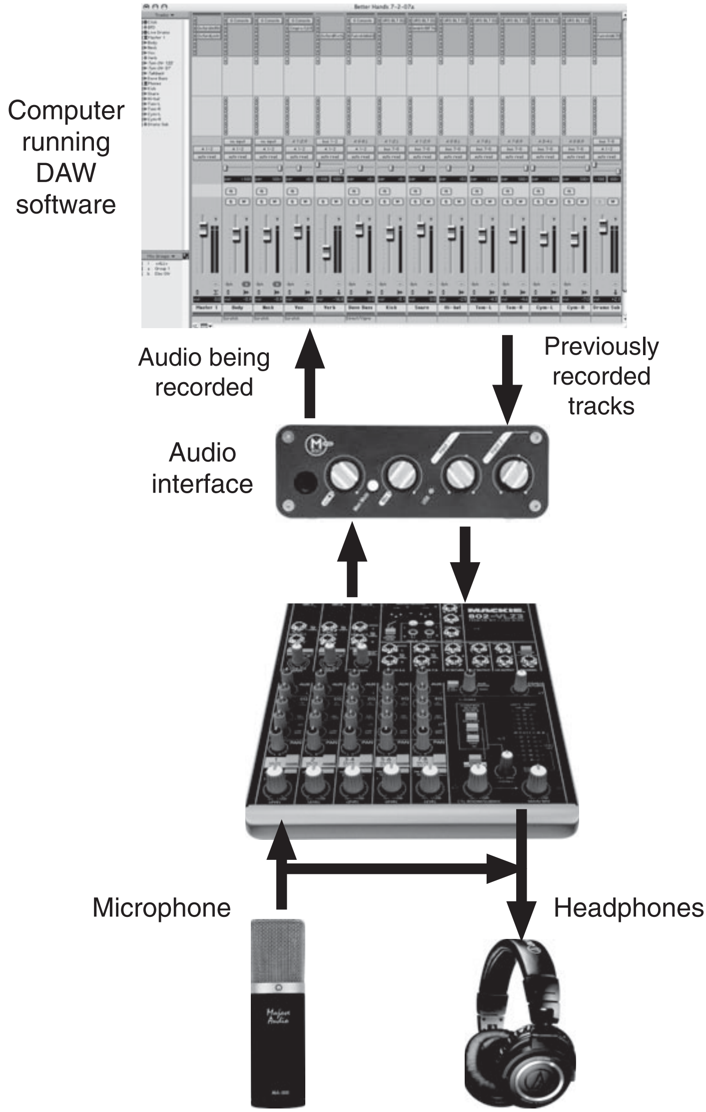

Figure L.3 A mixer can be used to monitor signals that are being recorded, eliminating the problem of latency when tracking or overdubbing.

LCD. Liquid Crystal Display. A material that changes the phase of light. Liquid crystals are placed between polarized plates that only pass light of a certain phase to create a "cell." The liquid crystal material changes the phase of light passing through it under the control of electrical voltage so that it either passes through the polarized plate or is blocked. By combining a large number of cells into a matrix where each cell represents a pixel, a high-resolution, monochrome, or color display can be created.

LE. Limited Edition. 1. A feature-limited version of a piece of software often designed to hit a particular price point, intended for semi-professional users or intended as a demo for the full version. 2. A special version of a product that is manufactured in limited numbers, often with the idea of creating a "collectible" item.

leader tape. A short piece of non-magnetic tape (usually plastic or paper) that is spliced into a reel of analog recording tape to separate one song from another. Leader tape is also used at the beginning and end of a reel of tape to identify the end and beginning and to protect the reel.

lead-in. The area on a CD-R where the table of contents is written. ☞ **See also** *TOC*.

leakage. 1. Ambient sound created by a source that is picked up by a microphone placed on another source. For example, the sound of a drum set may leak into the microphones that are placed on a grand piano. 2. Sound escaping from headphones worn by a musician (usually a vocalist) that is picked up by the musician's microphone. 3. The frequency of one tonewheel in an electronic organ being sensed by the pickup for an adjacent tonewheel.

least significant bit. ☞ **See** *LSB*.

LED. Light Emitting Diode. A semiconductor component that produces light when electric current passes through it. LEDs are widely used as indicator lights in all types of electronic gear. Newer, high-power versions are used for area lighting purposes.

LEDE. ☞ **See** *live end/dead end*.

legacy. 1. An older version of a software program. Most new versions are designed to be compatible with older, legacy versions of themselves. 2. Data that was created using an older version of a program. Most new versions of programs support opening and working with older legacy data.

legato mode. A synthesizer function where, if an existing note is held and a new note is played, the new note will not trigger another note attack. The existing note will remain in the sustain portion of its envelope and will be smoothly re-pitched to whatever the new note's pitch might be.

Leslie. A family of rotary speakers invented by Donald Leslie and first produced in 1941. The original Leslie cabinet was designed to be used with Hammond organs. ☞ **See also** *rotary speaker*.

level. The volume or strength of a signal.

leveler. A dynamics processor that attempts to maintain a constant signal level, usually by applying compression. Most levelers have a long attack time that allows short peaks through; the goal is to maintain a usable dynamic range. The consumer versions of levelers are also known as *automatic gain controls* and are used for applications, such as keeping the level of television commercials the same as the program level.

Lexan overlay. A removable plastic sheet that fits over a control surface and that has labels to indicate the functions of the various controls when the control surface is used with a specific piece of software.

LF. ☞ **See** *low frequency*.

LFE. Low-Frequency Effects. The ".1" of a surround sound system, used to carry low-frequency information. The concept is to provide a separate driver and amplifier for the power-intensive low-frequency components of film and video sound effects, such as explosions.

LFO. Low-Frequency Oscillator. A type of oscillator dedicated to producing low-frequency audio signals that are used for controlling and modulating other oscillators or parameters on processors.

LFO waveform. The shape of the waveform produced by an LFO. The LFO waveform can greatly affect the sound of any modulation produced using the output from the LFO.

librarian. Software designed to store and organize presets for a synthesizer or other MIDI device. Librarians communicate with MIDI hardware using System Exclusive commands.

library. 1. A collection of presets for a synthesizer or other MIDI device. 2. A collection of sounds for a sampler, including program and sample data. Commercial sample libraries are available from a number of manufacturers and can be targeted toward a specific group of sounds (a library of solo strings, for example), or they can be more generic libraries that contain a variety of sound types. 3. A collection of audio or MIDI loops.

light-emitting diode. ☞ **See** *LED*.

Lightpipe (a.k.a. ADAT Lightpipe). A term used by Alesis for their eight-channel optical digital interconnection protocol, first introduced on the company's ADAT eight-track modular digital audio recorders. Lightpipe uses the same connectors and cables as the two-channel TOSLink protocol, though the two formats are not compatible signal-wise. Though the ADAT has waned in popularity, Lightpipe has become an industry-standard format for multichannel optical connections.

limiter. A compressor with a very high ratio—10:1 or even higher—resulting in very little change in output level no matter how much the level of the input signal changes. A limiter is typically used to establish a ceiling level above which signals aren't allowed to pass. Limiters are also used to raise signal levels without danger of pushing them into distortion. This is done by using the limiter to reduce the level of peaks, thus allowing the overall signal level to be safely increased. ☞ **See also** *compressor*.

line amp. A type of amplifier used to balance a signal and raise it to a level that can withstand long cable runs.

line conditioner. ☞ See *power conditioner*.

line input. An input on an audio device that is designed to accept line-level signals. In mixers and some types of preamps, the line-level inputs bypass the high-gain preamps necessary for microphone level signals, resulting in a shorter, purer signal path.

line level. Technically, any signal voltage greater than 25 millivolts RMS, though the range is in practice 100 millivolts up to 12 volts. The two common line levels in use today for professional and semi-professional audio are 0.316 volts (–10 dBV) and 1.23 volts (+4 dBu).

linear. A mathematical function or device response that is close to a straight line. Flat response, for example, is linear frequency response—the output level of each frequency equals its input level. The ear's response to volume changes, however, is nonlinear—it is a logarithmic function. ☞ **See also** *linearity*.

Linear Arithmetic synthesis (a.k.a. LA synthesis). A type of digital synthesis developed by Roland for the D-50 and subsequent keyboards and modules. LA synthesis uses sampled attacks combined with synthesized waveforms for the other portions of each note. This allows efficient use of sample memory, as well as creative combinations of unrelated attacks and waveforms.

linear interpolation (a.k.a. averaging). A technique for correcting errors in a digital signal. The system averages the sample values before and after a corrupt or missing sample to provide an estimation of the value of the missing sample.

linearity. 1. The ability for an analog-to-digital or digital-to-analog converter to capture or reproduce signal levels accurately. 2. The ability of an amplifier to output signal levels in direct proportion to input signal levels. If a device is completely linear, the output-to-input signal ratio is constant, regardless of the input signal level.

linear phase response. A filter in which all the frequencies passing through are delayed by the same amount in terms of phase. A filter with linear phase response will not introduce audible artifacts into the sound that is passing through it. True linear phase response is not possible with analog circuitry; a digital filter is required. ☞ **See also** *group delay*.

linear taper pot. A potentiometer that changes resistance in a linear fashion. At 1/4 of its travel, the resistance is at 25%; at half of its travel, the resistance is at 50%; and so on, so in a technical sense, the resistance change is even. Linear taper pots are typically used to control parameters and settings other than volume, since our ears respond to volume changes in logarithmic fashion, and a linear taper pot sounds as if it is changing level at an uneven rate.

Linear Time Code. ☞ See *LTC*.

Linkwitz-Riley filter. A type of electronic crossover named for its inventors, Siegfried Linkwitz and Russ Riley. A Linkwitz-Riley crossover is created by cascading a low-pass filter and a high-pass filter. The big advantages to Linkwitz-Riley crossovers versus some other types are that the Linkwitz-Riley design results in an in-phase output signal and 0 dB of gain at the crossover point. (A Butterworth crossover design, for example, has a 3-dB peak at the crossover point.) The "order" of the crossover indicates the steepness of the filter slopes. For example, a 2nd-order Linkwitz-Riley crossover cascades two 1-pole filters for a 12-dB/octave slope. A 4th-order Linkwitz-Riley crossover cascades two 2nd-order filters for a 24-dB/octave slope. An 8th-order Linkwitz-Riley crossover cascades two 4th-order filters for a 48-dB/octave slope.

Linux. An open source version of the UNIX operating system, originally created by Finnish programmer Linus Torvalds.

Lissajous phase meter. A type of meter used to display the relative phase and polarity of two or more audio signals. Named for Jules Antoine Lissajous, a French mathematician.

listenback. Speakers set up in the studio or live room so that the musicians can hear the playback of recorded tracks without coming into the control room.

listenback mic. A microphone set up in the studio or live room so that the musicians can communicate with the engineer and producer in the control room. ☞ **See also** *talkback mic*.

listening position. The position where the listener is ideally located when monitoring audio—in stereo this is typically the position that results in the listener forming an equilateral triangle with the two speakers.

live. In acoustics, a reflective space that has no absorption to control the acoustic response or reverb time.

live end/dead end (a.k.a. LEDE). Trademarked term for a type of studio design featuring absorption in the front of the control room and reflective surfaces in the rear of the control room.

live room. 1. A room constructed with all or mostly reflective surfaces, resulting in strong acoustic reflections and long reverb times. 2. A studio room where performers are recorded.

load. 1. In electronics, an impedance or circuit that is connected to the output from another circuit and to which power is delivered. 2. The manner in which a driver is connected to a speaker cabinet, especially a horn (often called a *horn-loaded driver*).

lobar polar pattern. A highly directional polar pattern found only in shotgun microphones. See Figure L.4.

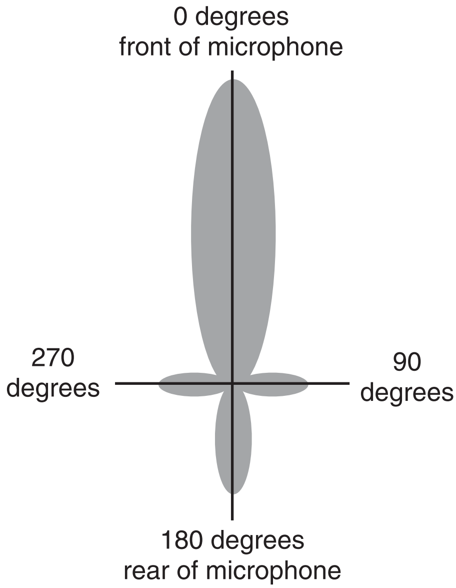

Figure L.4 A shotgun microphone has a lobar polar pattern that can pick up distant sounds without interference from unwanted sounds.

lobe. A portion of a microphone polar pattern that is not omnidirectional (spherical) and that is divided into different parts by null or dead points where no sound is picked up. A common example is a figure-8 pattern, where there is a spherical lobe in front of the microphone and a spherical lobe behind the microphone, which are separated by a null to the side of the mic. The most extreme example of a lobe is a shotgun mic, which has an extremely long frontal lobe in its pattern for long-distance pickup. Wireless systems and loudspeakers can also exhibit lobing. See Figure L.5.

local area network. ☞ See *LAN*.

local control. A function found in most keyboards that connects or disconnects the keyboard's keys from its internal sound generator. When local is on, the instrument's keys will play the internal sounds. When local is off, the instrument's keys will only be transmitted externally over MIDI, and the internal sounds will only respond to MIDI messages arriving at the MIDI in port. Local control is used to prevent MIDI loops, where the messages from a keyboard's keys play the internal sound and are sent out over MIDI, and because of a loop, are also received at the MIDI input. The doubled notes will cut the polyphony of the device in half and will likely also sound "hollow" due to phase cancellation from the two messages arriving at slightly different times.

localization. The ability to pinpoint where a sound source is coming from.

local on/off. ☞ See *local control*.

locate. To navigate to a specific point in time on a tape or in an audio project.

locate point. A saved time location on a tape or in a project. The tape recorder, DAW, or other audio program can jump to the locate point with a single button press or mouse click. This provides easy navigation among the important sections in a song or project. ☞ **See also** *marker*.

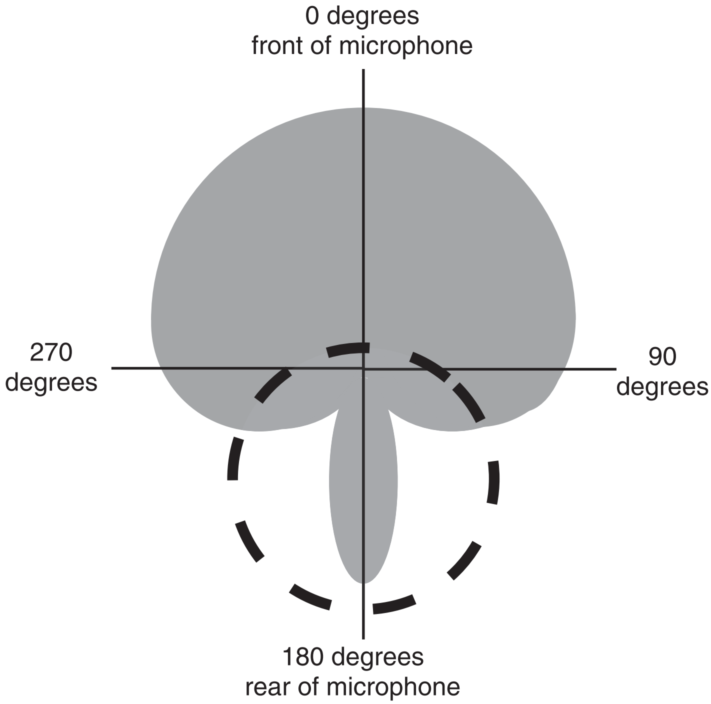

Figure L.5 A section of a non-omnidirectional polar pattern bounded by a null is called a lobe. In this hypercardioid polar pattern, there is a lobe to the rear of the mic (circled in the figure).

lock. To slave to time code or clock signals. A device is "locked" when it is synced to time code or clock signal. Mechanical devices, such as tape recorders and video decks, typically require a short pre-roll period in order to get up to speed and to lock to time code. Non-tape-based digital devices usually lock almost instantly.

lock up. 1. To synchronize two devices. 2. A CPU-based device or piece of software that has suffered a catastrophic error and is frozen—it will not respond to commands or controls. The usual cure for a locked-up device is to reboot it or, in the case of a personal computer, force the problem application to quit or terminate.

look ahead. A feature found in some virtual dynamics processors (compressors and limiters) where the software can examine the approaching audio. This allows the processor enough time to prepare for and accurately respond to fast transients and large peaks.

loop. 1. A short segment of audio, such as a measure of a drum beat, that is set up to repeat. 2. A portion of a sample that is repeated in order to allow the sound to sustain longer than the original sample lasts. 3. A transport control that continuously repeats playback of a selected section of a track or project.

loop point. The beginning or end point of a loop.

loop recording. A function or operating mode in some DAWs and sequencers in which the program is set to repeatedly loop or play a section of a track—for example, from Bar 4 to Bar 8. Every time the program loops over the section, a new take is recorded to a separate audio file. This allows a musician or vocalist to make many passes without having to start and stop playback and recording for each pass.

loop slicing. A technique for slicing apart a piece of audio at transients. The slices can then be played back at faster and slower rates in order to change the tempo of the audio. Or, the slices can be rearranged or otherwise processed to create new sounds, rhythms, and effects. ☞ **See also** *REX file*.

lossless audio compression. A type of audio codec that reduces file size without compromising audio quality or removing data. Two common examples are FLAC (*Free Lossless Audio Codec*), an open source codec, and MLP (*Meridian Lossless Packing*), which is used with DVD-Audio.

loudness. Objectively, the measured SPL (*sound pressure level*) of a sound. Subjectively, loudness depends on the frequency and timbre of the sound and varies from listener to listener.

loudspeaker. ☞ See *speaker*.

low-cut filter (a.k.a. high-pass filter). A filter that reduces the level of frequencies below a certain point. Low-cut filters are used to reduce rumble, as well as boominess caused by excessive low-frequency levels.

low end. 1. A term used to refer to low frequencies and the bass range of the frequency spectrum. 2. A term sometimes used to refer to inexpensive equipment.

low frequency. A term used to refer to the lowest frequencies in the audio spectrum; there is no specific low-frequency range, though in general, the range of 20 to 100 or 150 Hz would qualify.

low-frequency oscillator. ☞ See *LFO*.

low impedance. An impedance of around 600 ohms or less. ☞ See also *impedance*.

low-level format. A type of formatting used to prepare a hard drive to store data. With a low-level format, the entire contents of the disk are erased, not just the directories (as with a high-level format). In most cases, the drive's manufacturer will perform the low-level format; the user will only need to perform high-level formats.

low note priority. An operating mode on some synthesizers and samplers where the unit will only respond to and play the lowest note received over its MIDI channel. This is ideal if a player is using a piano patch on one keyboard and wants a module with a bass patch to follow along on the lowest pitches played, providing a simultaneous bass part.

low-pass filter (a.k.a. high-cut filter, treble control). A filter that reduces the frequencies above the cutoff frequency and allows frequencies below the cutoff frequency to pass through unchanged. See Figure L.6.

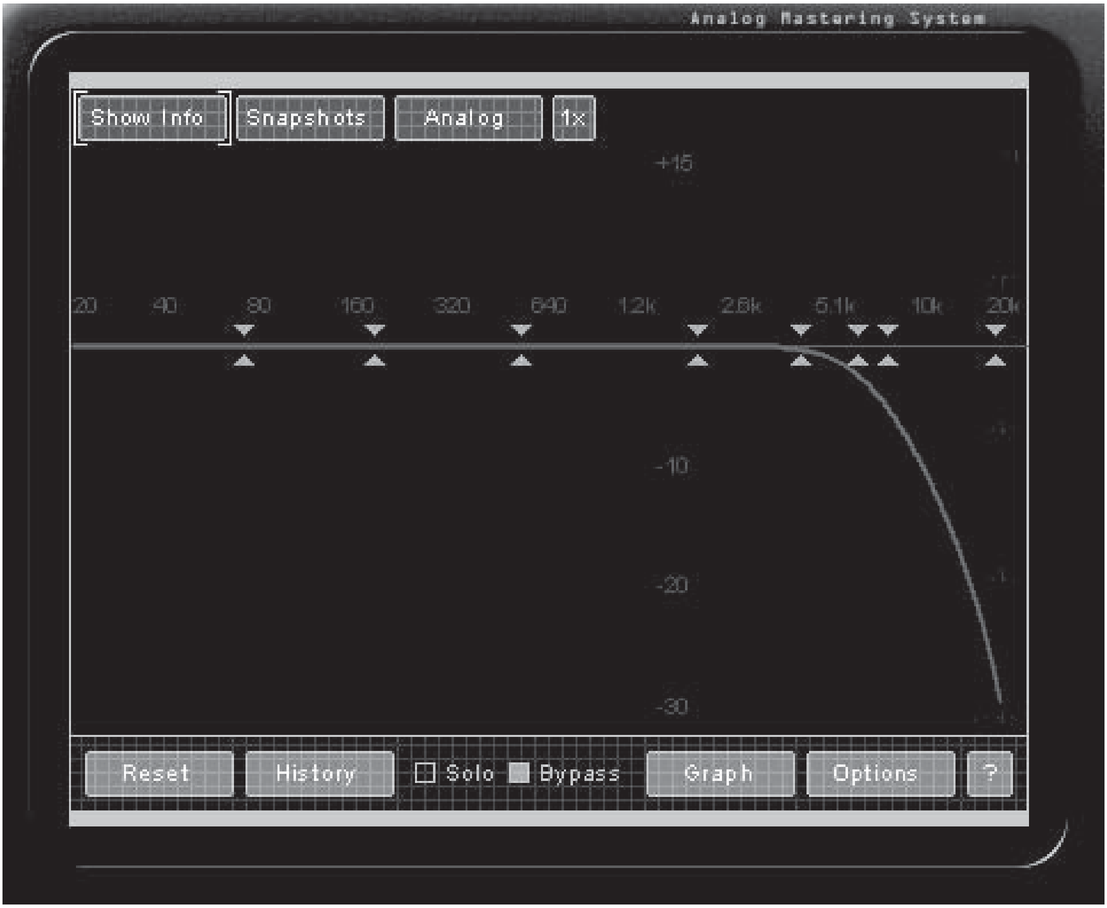

Figure L.6 A low-pass filter reduces the level of frequencies above a certain point, while allowing those below to pass through unchanged.

low-Z/Lo-Z. ☞ **See** *low impedance*.

LP. ☞ **See** *low-pass filter*.

LSB. Least Significant Bit (though technically "LSB" in all caps would refer to "Least Significant Byte"). The bit in a digital word that has the least impact on the mathematical value of the word and that determines whether the value is odd or even—typically the right-most bit.

LTB. Linear Time Base. A high-resolution system developed by Steinberg that provides for transmitting MIDI data with accuracy to 300 microseconds.

LTC. Linear Time Code or Longitudinal Time Code. Time code that is written as a continuous stream along a linear track on an analog audio multitrack or a video recorder. ☞ **See also** *SMPTE Time Code, VITC.*

lump in the line. A type of external AC power supply in which the transformer is in the middle of the cable that connects the device to the wall socket. Many studio owners prefer lump-in-the-line power supplies to wall-wart power supplies because they only require one space on a power strip—a wall wart can cover two or more spaces on a power strip.

lunchbox. A type of compact tabletop enclosure developed by API (*Automated Processors, Inc.*) and used to hold and power API's 500-series audio processing modules (and compatible modules), such as compressors, equalizers, and mic preamps.

LVD. Low Voltage Differential. A technology that allows for higher speed, longer distance SCSI data transfer. Regular SCSI sends a series of voltage spikes and valleys that are interpreted as ones and zeros. With LVD, a signal and an inverse of that signal are sent over two wires, somewhat the same as balanced audio signals. The difference between the two signals is used to represent ones and zeros.

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M

m4p. An .m4p file is an AAC-format digital audio file with added digital rights protection.

machine control. 1. ☞ See *9-pin*. 2. An option for Digidesign's Pro Tools that allows control over, and synchronization with, external players and recorders.

machine room. A dedicated room in a studio designed to isolate devices (such as computers, hard drives, and tape machines) that might contribute to an increased ambient noise floor.

macro. A computer command that combines two or more other commands to accomplish a task. Macros simplify performing complex, multistep operations and allow a single command to automate certain operations.

MADI. Multi-channel Audio Digital Interface. An AES standard for interconnecting digital multitrack recorders and mixers. MADI supports transmitting up to 56 channels of audio information over a single coaxial or optical cable.

magnetic shielding. ☞ See *shield*.

magnetic tape. An analog storage format in which a magnetizable oxide coating is adhered to a thin plastic strip. Record/play heads are used to magnetize the oxide during recording and to "read" the stored magnetism when playing back. Audio information and computer data can be stored on magnetic tape.

magneto-optical. A storage format that combines magnetic and optical technologies. A laser is used to heat the disc medium, which is then polarized by an electromagnet to store the information. The cooled medium retains this polarity, which can then be read by a low-power laser.

main outputs. The primary audio output or outputs on a device, which carry the mono or left-right stereo output signal. On a mixer, the main, or master, outputs may include connections for inserting equalizers or dynamics processors, such as compressors or limiters.

mains. 1. AC power coming from the wall outlet. 2. The front-of-house or audience speakers in a PA system.

makeup gain. Gain that is available to amplify the signal at the final output stage of a device, usually in a dynamics processor, such as a compressor or limiter. Makeup gain allows the device to compensate for peak volume reduced when compression or limiting is applied.

manual. 1. Part of the documentation that comes with a device, which describes its operation, use, and specifications. 2. A keyboard on an organ, harpsichord, clavichord, or other instrument.

mapping. 1. A function that assigns incoming data to a target destination. For example, incoming MIDI Continuous Controller #42 data might be mapped to control modulation. 2. Assigning samples to the keys of a keyboard.

marker. An icon that can be used to store and recall a particular location in the timeline of a DAW or sequencer. Markers might be placed at the various sections of a song, such as the beginning of the verses, choruses, guitar solo, and outro, making it easy to navigate to the different parts during recording, editing, and mixdown. In some DAWs and sequencers, markers can be used to store additional information, such as zoom level, so that when a marker is selected, the song will not only navigate to the proper location, but will also zoom in to the specified level. See Figure M.1.

MAS. MOTU Audio System. A protocol developed by MOTU and licensed to third-party developers

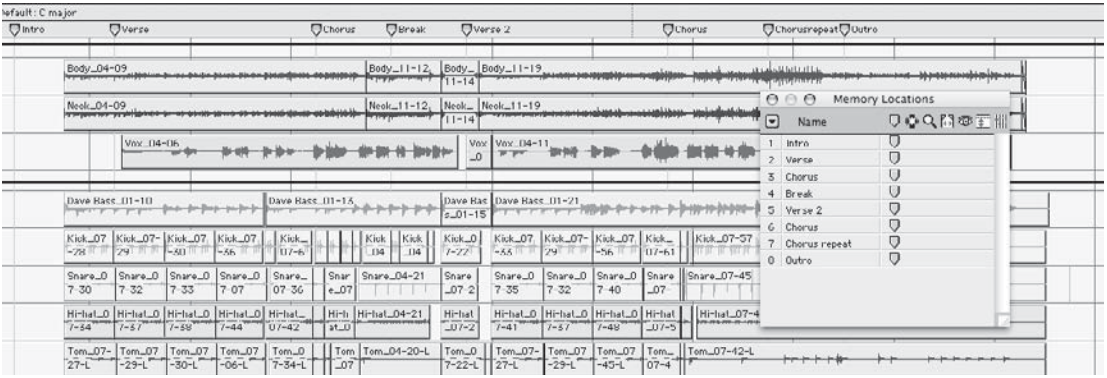

Figure M.1 Markers (a.k.a. memory locations) can be used to store a location in a song in a DAW for quick navigation. The markers in this example are arrayed across the top of the tracks. The menu at the right can be used to locate the session to the selected marker.

and manufacturers for integrating software synths and samplers and effects and processing plug-ins into the company's Digital Performer DAW.

masking. The tendency of a loud sound with overlapping frequency content to a soft sound to cover up those frequencies in the softer sound. Masking is used as the basis for many audio data compression schemes.

master. 1. The device that is providing the synchronization and timing information for the slave devices in a system. 2. The clock that provides the digital clock for synchronizing a digital audio system's sample rates. ☞ **See also** *master clock*. 3. The MIDI keyboard or other controller that is providing the MIDI notes and other information for the slave devices in a MIDI system. ☞ **See also** *controller keyboard*. 4. The final, finished version of an audio production. 5. The process of preparing a recording for duplication or distribution. ☞ **See also** *mastering*.

master clock. The clock that provides the synchronization signal for a digital audio system. A master clock can be either a standalone device or the clock built into one of the devices in the system. The master clock controls and defines the sample rate for the system. ☞ **See also** *word clock*.

master fader. A fader (or faders) in the master section of a mixer that sets the level for the final summed mono, stereo, or multichannel surround output.

mastering. The process of optimizing a recording for distribution in various formats and preparing it for duplication. Mastering can be thought of as the last stage of audio production and the first stage of distribution and manufacturing. Mastering consists of editing, equalizing, and processing the dynamics of a recording, fading it in and out and setting its level so that it will flow well with other songs on an album, maximizing the volume level, and otherwise tweaking the recording to provide the best sound quality and listening experience for consumers.

master keyboard. ☞ **See** *controller keyboard*.

master volume. The final level control in a signal path, which sets the overall output level.

matrix editor. ☞ **See** *piano roll*.

matrix mixer. A type of mixer where any input can be routed to any output, and in some cases the EQ and the level for each input's feed to each output can be controlled. Matrix mixers are often used for creating multiple independent monitor mixes to be sent back to musicians during a live performance onstage.

matrix modulation. A feature of some synthesizers where any modulation source can be routed to control any destination parameter.

matrix surround (a.k.a. Dolby surround). An analog surround format in which the rear left and right channels are encoded into the front left and right signals and are partially decoded on playback.

Because the surround channels can only be partially decoded from the front channels, matrix surround is not as effective as digital systems in which the surround channels are stored as discrete signals. ☞ **See also** *Dolby*.

maximizer. A device or plug-in that uses a compressor or limiter to raise the level of an audio signal as high as possible without distortion (though reduction of peaks will often occur and can be pushed to such an extent that audible distortion will result).

maximum SPL. A microphone specification that indicates the highest sound pressure level a microphone can handle without distorting. The electronics in the mic will distort first, so some microphones contain a pad that can be used to reduce the electronic signal level, allowing higher acoustic sound pressure levels to be handled. The maximum SPL spec is referenced to 0.5% distortion at 1 kHz.

maximum transfer rate. ☞ **See** *burst transfer rate*.

Mb. Megabit (1,048,576 bits).

MB. Megabyte (1,048,576 bytes).

Mbps (a.k.a. Mb/s). Megabits per second or 1,000,000 bits per second.

MBps (a.k.a. MB/s). Megabytes per second or 1,000,000 bytes per second.

MDF. Medium-Density Fiberboard. A wood product made from processed wood fibers combined with resin, and used wherever real wood might be used, such as in monitor and speaker cabinets, guitar and bass amplifiers, PA speakers, and more.

MDM. ☞ **See** *modular digital multitrack*.

mean time before failure (a.k.a. MTBF). Manufacturer spec for how long a drive will last, usually measured in hours of operation.

median plane. The center line between the left and right speakers in a studio monitor setup. The ideal listening position will be on the median plane, equidistant from the two speakers and at the same distance from the speakers as the speakers are spaced apart (forming an equilateral triangle among the two speakers and the listener).

medium-diaphragm microphone. A microphone with a diaphragm measuring somewhere around 5/8- to 3/4-inch in diameter. A medium-diaphragm microphone will deliver response that falls between small-diaphragm and large-diaphragm mics, with good transient detail and full sound.

mega. 1. A prefix meaning one million. This definition is often used for data transfer rates—a megabit transfer rate would mean 1,000,000 bits per second. 2. In computer storage terms, 2 to the 20th power, or 1,048,576, since computer storage media use base 2 or binary mathematics.

megabit. A measure of computer storage indicating 2 to the 20th power bits (1,048,576 bits).

megabyte. A measure of computer storage indicating 2 to the 20th power bytes (1,048,576 bytes).

megahertz. A million hertz or cycles per second.

Mellotron. A keyboard instrument introduced in 1962. The instrument's sounds were contained on racks of short analog tapes with one tape per key. When a key was pressed, its tape played a 6- to 8-second recording of a sound. A spring rewound the tape when the note was released. To play a note longer than six to eight seconds required releasing the key and quickly restriking it. Sounds could be changed by installing new racks of tapes. The Mellotron can be thought of as an early analog sampler.

membrane trap. Type of bass trap that features a thin membrane, panel, diaphragm, or surface that vibrates in response to low-frequency sound waves and absorbs the sound energy.

memory effect (a.k.a. battery memory, lazy battery effect). A tendency of nickel-cadmium batteries to be unable to reach full charge if they are recharged without being fully discharged. In most cases, memory effect can be circumvented by completely discharging and fully recharging a battery several times.

memory location. ☞ **See** *marker*.

menu. Graphical computer programs and operating systems save screen space by organizing commands into sets of related options that can be accessed as necessary by dropping down a hidden part of the window to reveal the available commands. Clicking on a command in a menu or typing in its key shortcut causes it to execute. See Figure M.2.

menu structure. The manner in which a program's or operating system's commands are arranged into menus, including submenus that may branch off of the main menu.

merge (a.k.a. merge file). A DAW command that creates a new audio file by combining two or more existing audio files. The merge function is

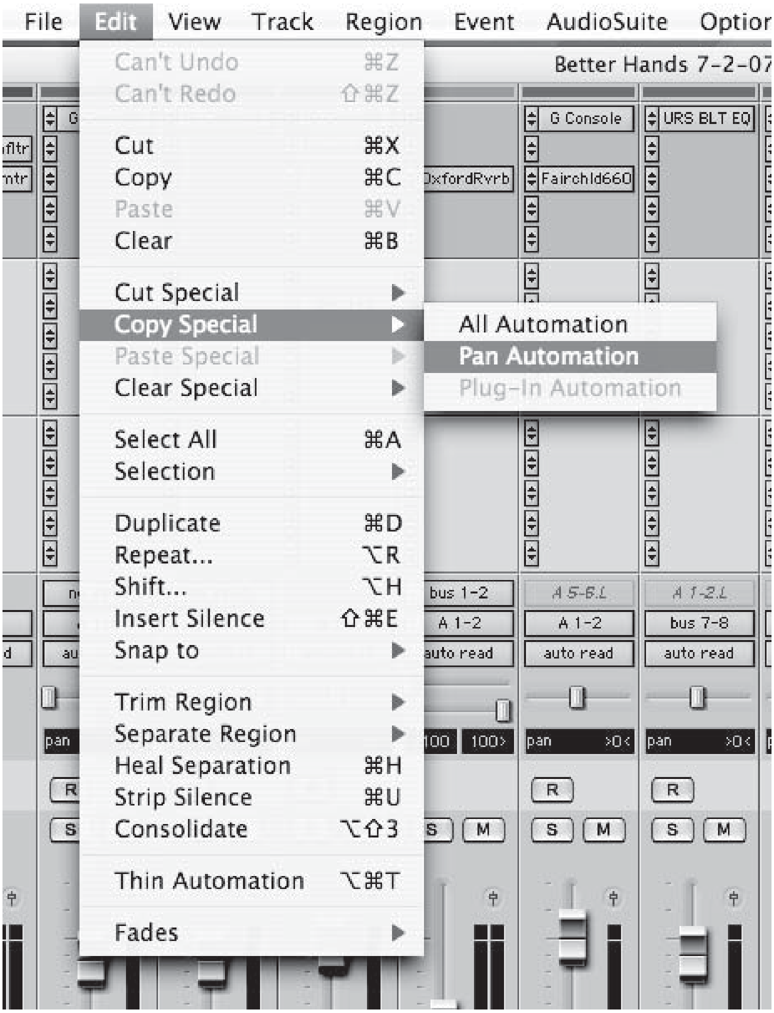

Figure M.2 A program's or operating system's menus organize the available commands into lists. Submenus branch off from the main menu and provide related command options.

especially useful for combining a number of short audio regions resulting from editing a track into a new, single audio file that is easier to manage.

metadata. Literally, data that describes other data. Digital audio metadata may include information such as number of channels, sample rate, bit depth, type of data encryption, and more. Metadata is typically stored in a part of a file called a *header* that an application can access to "learn" about a file.

meter. 1. An indicator on an audio device or in an audio program that displays signal level or other information. 2. The arrangement of stressed and unstressed beats in music.

meter bridge. In a mixing console or multitrack tape recorder, an array of meters that indicate the level of input channels and various outputs. In some cases, a meter bridge is built into the chassis of the unit; in other cases, the meter bridge is a separate option. Some DAWs offer a virtual version of a

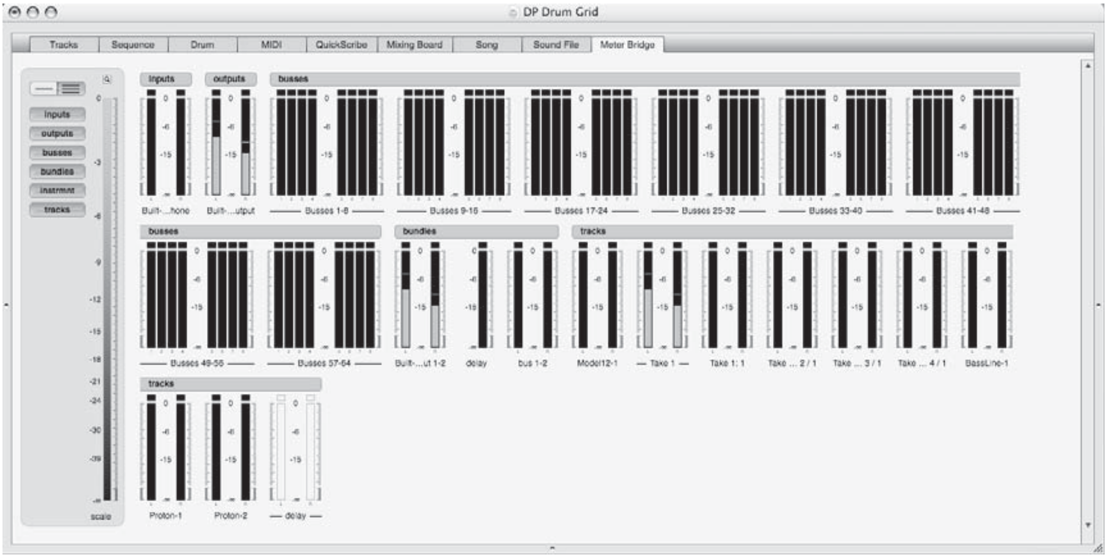

Figure M.3 MOTU's Digital Performer DAW software offers a virtual meter bridge for displaying signal levels.

meter bridge, displaying a number of meters within a window. See Figure M.3.

metronome. A device that indicates musical tempo by audibly clicking and/or flashing a light at a user-specified rate.

MF. Midrange frequency.

MHz. Megahertz.

MI. Music Industry. A term used in reference to manufacturers and retailers of musical instruments.

mic amp. ☞ **See** *mic preamp*.

mic level. The amount of electrical signal level generated by a microphone. Mic levels can range from a few millivolts to 100 millivolts, often falling around –40dBu. Because of these extremely low levels, microphone signals must be preamplified in order to be used with the line-level gear that makes up most signal chains. ☞ **See also** *mic preamp*.

mic preamp (a.k.a. microphone preamp, mic, amp, preamp). At its most basic, a type of amplifier used to raise a mic-level signal to line level (either 0.316 volts or 1.23 volts). Mic preamps can be either standalone pieces of hardware or built into a mixer, audio interface, or other device. Features sometimes found in mic preamps including input impedance switching, pads, polarity reverse, and more. Many mic preamps also include instrument-level inputs.

microchip. ☞ **See** *IC*.

microdrive. A trademarked brand name for a miniature one-inch hard drive designed to fit in a CompactFlash Type II slot.

micron (a.k.a. micrometer). One millionth of a meter, or 0.000039 inches. A human hair measures about 50 microns. A microphone diaphragm can be as thin as just a few microns.

microphone (a.k.a. mic). A transducer that converts sound waves into electrical signals. A variety of types are available, including condenser, dynamic, ribbon, piezo, and others.

microphonic. A component in a signal path that produces audio when it is vibrated. Vacuum tubes are the most common component to become microphonic, though cables and other components are also susceptible.

microsecond. One millionth of a second or one thousandth of a millisecond.

MID. ☞ **See** *Standard MIDI file*.

middle C. A note tuned to 261.1 Hz. The MIDI Specification defines middle C as note number 60.

MIDI. Musical Instrument Digital Interface. A protocol developed jointly in the early 1980s by several manufacturers that allows various pieces of music and audio gear to communicate with and control each other. MIDI is a digital protocol that represents and transmits performance "gestures," such as a key press or control change, as well as various types of synchronization and other data. No audio is carried over MIDI, though samples can be transmitted (very slowly) between devices.

MIDI cable. A three-conductor cable terminated with five-pin DIN connectors. (The outer two pins are not used.) Each MIDI cable carries 16 logical channels in one direction with a 31.25-kb bandwidth. MIDI uses balanced connections and telescoping grounds to enhance stability and to prevent ground loops. The maximum length for a MIDI cable defined by the MIDI Specification is 15 meters, though in practice, most users try to keep MIDI cables as short as possible, and usually less than 30 feet in length.

MIDI channel. A method for organizing MIDI data or messages. The MIDI Specification defines 16 channels, number 1 though 16. Each channel can carry any of an array of possible channelized messages. A device can be set to transmit on a particular channel (similar to a radio or television transmitter). Only devices that are set to receive that channel will respond to those messages (similar to a radio or television receiver).

MIDI channel messages. A type of MIDI message that is coded with a particular channel number and is intended to be received only by devices "tuned" or set to that channel. Examples include MIDI notes, monophonic and polyphonic aftertouch, pitch bend, continuous controllers, program change, and others.

MIDI channel mode. One of four ways in which a device can be configured to respond to incoming channel messages. There are two aspects to channel modes: omni on/off, which sets how the device responds to information on incoming MIDI channels; and poly/mono, which sets how the device responds to multiple note messages received simultaneously (whether it will be polyphonic—where it will sound chords or multiple notes simultaneously—or monophonic, where it will only play one note at a time). See Table M.1.

MIDI clock. A timing reference carried by MIDI that runs at 24 PPQN. The speed of the MIDI clock varies with a song's tempo so that each quarter note receives 24 evenly spaced pulses, regardless of the tempo. MIDI clock is not a form of time

Table M.1 MIDI Channel Modes

Mode	Settings	Description
1	Omni On/Poly	Messages will be received on any MIDI channel, polyphonic. This mode might be used when a performer wants to layer two keyboards together so that both will play any notes he or she plays.
2	Omni On/Mono	Messages will be received on any MIDI channel, monophonic. A performer playing a MIDI wind controller, which can generate only one note at a time, might use Mode 2. In practice, Mode 2 is rarely used.
3	Omni Off/Poly	Messages will be received on only one MIDI channel, polyphonic. This is the default MIDI channel mode for most keyboards and modules. Multitimbral operation or Multi Mode is an expanded version of Mode 3, where a synth or sampler can recognize messages on multiple channels, each responding polyphonically.
4	Omni Off/Mono	Messages will be received on only one MIDI channel, monophonic. Mono Mode is an extended version of Mode 4 often used for MIDI guitars, where each string is sent on a separate channel and responds to just one note at a time.

Table M.2 Common MIDI Controllers

CC #	Assignment
0	Bank Select (coarse)
1	Modulation Wheel
2	Breath Controller
4	Foot Controller
6	Data Entry Slider
7	Channel Volume
10	Channel Pan
32	Bank Select (fine)
64	Sustain Pedal On/Off
124–127	MIDI Channel Mode Select

code, as it does not carry song or time location information and it varies in speed.

MIDI continuous controller (a.k.a. CC). The MIDI Specification provides for 128 different messages (numbered 0 through 127) per MIDI channel that can be used to carry control, switch, and parameter changes. Each continuous controller has a range of values from 0 through 127. In some cases, two continuous controllers can be used together to provide coarse and fine control, though not all devices will respond to fine controls. Continuous controllers can be used to add expressiveness to a sequence or to capture the various performance gestures made by a musician. The MIDI Specification defines a number of continuous controllers for particular applications. Table M.2 shows some of the common continuous controller assignments. (Note that pitch bend is not a continuous controller, as it requires higher resolution. Pitch bend is its own type of MIDI message.)

MIDI control change. ☞ See *MIDI continuous controller*.

MIDI controller. 1. A keyboard or other device that generates MIDI note or control information. 2. ☞ See *MIDI continuous controller*.

MIDI date rate. The rate at which MIDI information is transferred between devices using MIDI cables: 31.25 Kbits per second, with 10 bits transmitted per byte (a start bit, eight data bits, and a stop bit).

MIDI delay. 1. The amount of time it takes for a MIDI message to enter a device through its MIDI in port and to exit through its MIDI thru port. If a receiving device is at the end of a long daisy chain of devices, the delay may accumulate enough to be noticeable. 2. The time it takes for a device to respond to an incoming MIDI message (though there are other factors in this besides MIDI, such as processor speed).

MIDI echo. A function of some devices in which data entering the device's MIDI input is sent out the MIDI output. (Normally, data received by a device's MIDI input is only sent out through its MIDI thru port.)

MIDI implementation chart. A part of a device's documentation typically found in the back of its manual that lists what MIDI messages and data the device can transmit and respond to. This information is organized into a standard chart format that makes it easy to see what MIDI functions the device supports. See Figure M.4.

MIDI in. One of the three possible types of MIDI ports. A MIDI in port receives messages and data from an external sequencer or controller. A MIDI in port can be fed by either a MIDI output or a MIDI thru.

MIDI interface. A peripheral device that allows a computer to receive and transmit MIDI messages and data. MIDI interfaces may be internal expansion cards or external boxes that connect via serial or parallel ports, or using USB or FireWire. Many modern MIDI devices support direct connection to a computer using USB, so MIDI interfaces are not as necessary as they were before the advent of USB. A MIDI interface may have one or more MIDI input and output, each of which can carry 16 channels of information. This means that a two-input/two-output MIDI interface could carry 32 MIDI channels in and out of the computer. An eight-input/eight-output MIDI interface could carry 128 MIDI channels in and out of the computer. Some MIDI interfaces also incorporate extended functions, such as SMPTE time code synchronization, machine control functions, MIDI merging, MIDI filtering and processing, MIDI routing and patch bay functions, and more.

Function		Transmitted	Recognized	Remarks
Basic Channel	Default	1 – 16	1 – 16	Memorized
	Changed	1 – 16	1 – 16	
Mode	Memorized	X	3	
	Messages	X	X	
	Altered	************	X	
Note Number:		*5 – 120 / 21 – 108	0 – 127	*: MS2000 / MS2000R
	True Voice	************	0 – 127	
Velocity	Note On	O 1 – 127	O 1 – 127	
	Note Off	O 64	X	
Aftertouch	Polyphonic (Key)	X	X	
	Monophonic (Channel)	O	O	*1
Pitch Bend		* O / X	O	*: MS2000 / MS2000R *B
Control Change	1	* O / X	O	Modulation wheel *1
	2	O	O	Breath Controller *1, *C
	4	O	O	Foot controller *1, *C
	6	O	O	Data Entry (MSB) *C
	7	O	O	Volume *1, *C
	10	O	O	Panpot *1, *C
	11	O	O	Expression *1, *C
	64	O	O	Damper *1, *C

Figure M.4 A device's MIDI implementation chart shows what MIDI messages the device can transmit and recognize.

MIDI learn. A function of some software, where a virtual control can "learn" to respond to an incoming MIDI continuous controller message from a hardware controller by selecting the desired virtual control, then moving the desired hardware control. This makes it fast and easy to configure a piece of software to work with a control surface or controller keyboard.

MIDI log jam. A situation that occurs when the amount of data to be transmitted by a single MIDI cable exceeds the available bandwidth. Typically, overuse of continuous controllers and pitch bend is to blame. MIDI log jam can be thought of as MIDI overload and can result in timing errors, stuck notes, glitching continuous controllers, and other symptoms.

MIDI Machine Control (a.k.a. MMC). A set of MIDI messages used to control playback and other functions in hardware recorders and players. MIDI Machine Control commands include play, stop, rewind, fast-forward, and others. MIDI Machine Control, when used in conjunction with MIDI Time Code, allows a MIDI sequencer to control a recorder as well as synchronize to it.

MIDI merge. A function of some devices that allows the device to combine two or more streams of MIDI information into a single stream.

MIDI merger. A device that combines two or more streams of MIDI information into a single stream.

MIDI mode. ☞ See *MIDI channel mode*.

MIDI note. A two-part MIDI channel message. The first part of a MIDI note is the note on message, which is sent when a key is struck and tells the receiving device to begin sounding the note. There are three main parts to the note on message: the MIDI channel, the MIDI note number to be played, and the MIDI velocity for that note. The second part of a MIDI note is the note off message, which is sent when the key or sustain pedal is released and tells the receiving device to stop sounding the note. There are three main parts to the note off message: the MIDI channel, the MIDI note number to be stopped, and the MIDI release velocity for that note. The MIDI channel and MIDI note number must match in both the note on and note off messages for the note to be properly stopped.

MIDI note number. A MIDI channel message that specifies which musical note has been played. There are 128 possible MIDI note numbers, ranging from 0 through 127. (Though some manufacturers frustratingly decide to use 1 through 128 for the note range.) Middle C is defined to be MIDI note number 60 by the MIDI Spec.

MIDI out. One of the three possible types of MIDI ports. A MIDI out port carries messages and data generated inside a device by playing keys, hitting pads, playing a sequence, moving a controller, and so on. A MIDI out port is connected to the MIDI in port of another device to allow the first (master) device to control the second (slave) device.

MIDI pan. MIDI Continuous Controller #10, a MIDI channel message that is defined to control the pan setting of a device or a multitimbral part that is set to respond to a particular MIDI channel.

MIDI part. An independent section of a multitimbral synthesizer or sampler that can respond to its own MIDI channel. ☞ **See** *multitimbral*.

MIDI port. A MIDI jack. There are three types: input, output, and thru. See Figure M.5.

MIDI program change. ☞ **See** *program change*.

MIDI through. ☞ **See** *MIDI thru*.

MIDI thru (a.k.a. MIDI through). One of the three possible types of MIDI ports. A MIDI thru port carries a copy of the messages and data that are arriving at the MIDI input of the device. A MIDI thru port can be connected to the MIDI in port of another device to create a daisy chain.

MIDI Time Code (a.k.a. MTC). A version of SMPTE linear time code carried over MIDI and used to synchronize compatible devices, such as sequencers to other pieces of gear.

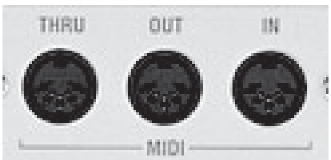

Figure M.5 There are three types of MIDI ports or jacks: MIDI inputs, MIDI outputs, and MIDI thrus.

MIDI time stamping (a.k.a. MTS). A MIDI recording and transmission technique developed by MOTU, where each piece of data is "stamped" or encoded with a specific time location. This provides for extremely accurate playback from a sequencer that is not dependent on a clock or the sequencer's PPQN resolution.

MIDI volume. MIDI Continuous Controller #7, a MIDI channel message that is defined to control the volume of a device or a multitimbral part that is set to respond to a particular MIDI channel.

midrange. Literally, the middle part of a frequency range. There is no exact range of frequencies defined within the range of human hearing as the "midrange"; it falls somewhere between the low frequencies and the high frequencies.

mid-side stereo. ☞ **See** *MS stereo*.

millisecond (a.k.a. ms). 1/1,000 of a second.

milliwatt (a.k.a. mw). 1/1,000 of a watt.

minijack. A 1/8-inch audio jack, either TS (unbalanced) or TRS (either balanced or dual-channel). The female counterpart to a miniplug.

minimum system requirements. The minimum computer power and amount of RAM, along with the minimum OS version, hard drive space, and any peripherals or requirements necessary to run a piece of software.

miniplug. A 1/8-inch audio plug, either TS (unbalanced) or TRS (either balanced or dual-channel). The male counterpart to a minijack.

mini-TOSLink. An optical connector type found on Apple computers and some other audio gear that resembles a minijack, but is slightly longer. ☞ **See also** *TOSLink*.

mix. 1. Creating a blend and proper balance of the sounds that make up a performance or recording. 2. A blended combination of sounds that comprise a performance or recording. 3. A control (sometimes labeled "wet/dry") that is used to set the blend for dry and effected signals in an effects processor.

mixdown. The final stage of producing an audio project in a studio, where individual tracks and signals are combined and processed to create a finished recording, whether mono, stereo, or surround, that will be sent to mastering for preparation for distribution.

mixed-mode disc. An audio CD that also contains computer data. ☞ **See** *Blue Book*.

mixer. At its most basic, a device or software function for combining audio signals. Mixers frequently contain sophisticated audio routing and processing capabilities. Mixers can range from extremely basic units to one containing extended functions and features, including microphone, instrument, and line preamps; equalization; subgrouping; aux and effects sends and busing; monitor control; talkback; input switching; headphone amps; level automation; and more.

mixing board. ☞ See *mixer*.

mixing console. A large-format, desk-style mixing board. ☞ See *mixer*.

mixing desk. U.K. term for mixing board. ☞ See *mixer*.

mix-to-disk (a.k.a. bounce to disk). A function in some DAWs that allows a mix to be recorded directly to a file on a hard disk without routing it to an external recorder or to new tracks. The mix-to-disk function can also be used to bounce submixes to new audio files that can then be imported to new tracks.

mLAN. Music Local Area Network. A Yamaha-developed protocol for networking music and audio gear and computers using FireWire cables and ports. Up to 127 devices can be connected in daisy-chain fashion; devices can be hot-plugged and unplugged. Each cable can carry thousands of MIDI channels of information and more than 150 digital audio channels of information (at 24-bit/48-kHz resolution). Though there are more than 100 manufacturers in the mLAN Alliance, the protocol has yet to be widely adopted.

MLP. Meridian Lossless Packing. A technology, developed by Meridian Audio and licensed to Dolby Laboratories, for lossless audio compression on DVD Audio. MLP can reduce the data rate for 24-bit/96-kHz 5.1 surround audio from 13.8 MB/second to under the 9.6 MB/second maximum DVD-Audio transfer rate and can reduce audio file sizes by 30 to 50 percent.

MMA. MIDI Manufacturers Association. A group of manufacturers of MIDI-compatible equipment that is responsible for developing the MIDI standard. www.midi.org.

MMC. ☞ See *MIDI Machine Control*.

MME. Multimedia Extension. An older type of Microsoft Windows OS driver developed for Windows 3.0 and used for audio and video hardware. MME drivers were replaced by WDM and other lower latency driver types in more current versions of Windows.

MOD. A file format that includes MIDI sequence information and sample playback data. Since the MOD file can contain samples, which can be used either as loops or instrument notes, playback can be consistent no matter what computer the file is played on.

mod wheel. Short for modulation wheel. A wheel, paddle, or stick control found on synthesizers and samplers that is typically internally mapped to control vibrato or some other form of modulation. MIDI Continuous Controller #1 is usually also sent, which is defined to control modulation.

modal. Pertaining to room modes.

modal distribution. How room modes are spaced across the frequency range. ☞ See also *room mode*.

mode. 1. A particular setup in a device that allows for certain types of operation or functions. Examples include edit mode, disk mode, performance mode, and others. 2. An acoustic resonance in a room. ☞ See *room mode*. 3. A musical scale that is derived from another scale.

model. A computer representation of a real item or process.

modeling. In audio, the use of mathematical equations to re-create or represent sounds or processes. Complex computer algorithms are used to imitate the behavior of analog circuits and acoustic properties.

modes of vibration. ☞ See *room mode*.

modifier keys. Keys on a computer keyboard that change the function of other keys when held down. With text, a key may create one character when pressed by itself, a capitalized version of that letter when pressed while the Shift key is held, another character when the Option key is held, and another when the Control key is held. Combinations of modifier and regular keys are also used as shortcuts for accessing commonly used commands, such as Save and Quit.

modular digital multitrack (a.k.a. MDM). One of several types of 8-track digital recorders (tape- and hard disk–based) that could be locked together to create larger systems with more tracks—they were all the rage in the '90s. There are still many thousands of these recorders out there, being used every day by studios around the world. Many pro studios keep a few around for compatibility reasons, or for doing transfers for clients from digital tape to a computer-based DAW.

modular synthesizer. An analog synthesizer that is made up of a number of independent modules, each of which provides a particular function, such as oscillator, filter, envelope generator, amplifier, and so on. The audio and control signals accepted and generated by the various modules are routed using short patch cables.

modular unit. ☞ See *rack unit*.

modulation. Literally, change. 1. A change in a signal or the value of a parameter that occurs in real time, as the signal is passing through the device. 2. MIDI Continuous Controller #1, which is defined as the modulation wheel and is often routed to vibrato (frequency modulation) or another form of modulation. 3. In music, a key change.

modulation noise. Irregularity in a signal that sounds like noise. With analog tape, modulation noise is caused by the non-uniformity of the magnetism. In digital systems, quantization errors result in modulation noise.

modulation wheel. ☞ See *mod wheel*.

module. A tabletop or rackmount synthesizer or sampler that does not have keys of its own. Modules are intended to be controlled and played through MIDI, but otherwise are identical to keyboard synths and samplers.

MOL. Maximum Output Level. A specification for audio equipment, especially analog tape machines, that is standardized as the level a device can put out with 3% harmonic distortion and 3% intermodulation distortion. To truly be useful, MOL needs to be stated or graphed versus frequency.

momentary switch. A switch that does not latch when depressed or flipped. When the switch is released, it automatically reverts back to its default state. Sustain pedals, talkback switches, and other controls typically use momentary switches.

monaural. A single-channel audio signal. ☞ **See also** *monophonic*.

monitor. 1. To listen to audio via speakers. 2. Studio speakers, usually optimized for "flat" frequency response. ☞ **See also** *reference monitor*. 3. A speaker or headphones intended for the musicians to use to hear themselves and each other when recording or performing live. 4. A computer screen or display.

monitor controller. A standalone device that includes the functionality of the monitor section of a mixing console, and is intended for use in mixerless DAW-based studios. Monitor controllers typically offer speaker switching for two or more sets of monitors, monitor level control and calibration, headphone outputs, and input selection for monitoring different sources. Some models add talkback and dim functions, as well as cue functions for musician headphone mixes.

monitor send. A bus used to send a mix to musicians onstage or in a studio so they can hear themselves and each other.

mono. ☞ See *monophonic*.

mono bridged. ☞ See *bridged*.

mono-compatible. A stereo or surround audio signal that maintains its phase and blend integrity when summed to a single channel.

Mono mode. An expanded version of MIDI Channel Mode 4 (Omni off/Mono). Mono mode was originally designed for MIDI guitar applications, where each string was transmitted on a separate MIDI channel. This allows each string/channel to pitch bend independently and to have its own preset. ☞ **See also** *MIDI channel mode*.

monophonic. 1. Also known as mono. An audio signal consisting of a single channel. 2. An instrument capable of playing only one note at a time.

monophonic aftertouch (a.k.a. channel aftertouch). A type of MIDI aftertouch in which a single value is transmitted for each channel. If a chord is played, the highest aftertouch value for all the held keys is the one that is transmitted for all the keys. ☞ **See also** *aftertouch*.

mono summing. Collapsing a stereo or surround signal into a single channel. This is often done to check for phase problems and balances within a mix.

more me. 1. The tendency of musicians to always want to hear themselves loudest in a mix, especially in a monitor or headphone mix when recording. 2. A type of headphone monitor system that allows the musician to add more of himself to his headphone mix.

morphing. 1. A real-time technique for smoothly transitioning from one thing to another. Morphing was originally a video effect that has since been adapted to audio and other media. Morphing is more than simply crossfading from one signal to another. Rather, the beginning sound is processed to gradually convert it into the ending sound, with the intermediary steps audible as the transition occurs. 2. In effects processors, smoothly transitioning from one preset or setting to another, with all the parameters simultaneously changing to create the new setting.

MOSFET. Metal Oxide Semiconductor Field Effect Transistor. A common type of FET often used in audio circuits. MOSFETs are similar to JFETs, though they are made of different materials, and MOSFETs have an insulating oxide that is not present in JFETs. MOSFETs are often used in both digital and analog audio circuits. ☞ **See also** *FET, JFET, transistor*.

motherboard. A printed circuit board containing the primary components in a microprocessor-based device, such as the CPU, RAM, ROM, and more.

motorized fader. A fader with an integrated motor that can move the fader in response to automation data. ☞ See *moving fader, moving fader automation*.

mount. With computers, to "mount" is to a make a hard drive or other storage media accessible by the CPU. Mounting consists of reading the media's file structure and ensuring it is compatible with the computer.

MOV. Metal Oxide Varistor. An electrical component that changes resistance based on incoming voltage. MOVs are often used in surge protectors; as the voltage on the AC line increases, the MOV resistance increases, protecting any connected equipment from excessive voltage.

moving coil. A technology used in microphones, phono cartridges, and other transducers for converting motion to electrical signals or vice versa. In a dynamic microphone, a coil of wire is attached to the diaphragm. When the diaphragm moves in response to sound waves, the coil is also moved within a magnetic field. The motion of the coil generates an alternating-current electrical signal that represents the sound waves. A speaker works in exactly the opposite way; electrical signals cause a coil of wire that is attached to a cone in a magnetic field to move, creating changes in air pressure our ears perceive as sound waves.

moving fader. A motorized fader that moves in response to automation data. ☞ **See also** *moving fader automation*.

moving fader automation. A hybrid digital/analog system found in hardware mixers that uses automation data to control the physical movement of motorized channel faders through which analog audio is passing. Since the levels were being changed by actual faders rather than by VCAs, some engineers felt that moving fader systems sounded better, though modern VCAs have improved to where this isn't an issue. Some control surfaces and digital mixers have motorized faders that move in response to automation, but this isn't the same thing as moving fader automation, as the analog audio doesn't actually pass through the faders.

MP3 (a.k.a. MPEG-1 Audio Layer 3). A part of the MPEG-1 standard that specifies a lossy audio compression format that has been widely adopted by manufacturers and consumers for file storage and playback. MP3 can reduce the CD data rate (1,411 kbits/second for stereo) down as low as 32 kbits/second, while maintaining a 44.1-kHz sample rate.

MPC. AKAI abbreviation for MIDI Production Center or Music Production Center. The original MPC60 was a standalone sampling drum machine/MIDI sequencer combination launched in 1988 and designed by Roger Linn. Subsequent models have added a variety of capabilities, including built-in virtual analog synths, multitrack audio recording, effects, and more.

MPEG. Moving Pictures Experts Group. An organization that develops standards for audio and video digital compression encoders/decoders. www.mpeg.org.

- **MPEG-1.** A standard for lossy audio and video compression approved in 1992 that reduced the data rate for media transmission to 1.5 Mbits/second. MPEG-1 was used with video CD and digital television and audio broadcast. MPEG-1 was also where the MP3 audio standard was introduced.
- **MPEG-2.** An audio and video standard for broadcast digital television over the air, cable, and satellite. MPEG-2 was designed to improve on the weaknesses in the MPEG-1 standard. MPEG-2 also included the AAC standard, designed to be the successor to MP3.

- **MPEG-3.** An abandoned standard designed for HDTV. MPEG-3 is completely different from MP3.
- **MPEG-4.** A still-developing standard introduced in 1998 that included support for web streaming media, voice telephone and videophone, and broadcast television. MPEG-4 Part 3 includes the AAC standard first introduced in MPEG-2 Part 7. MPEG-4 Part 10 supports HD-DVD and Blu-ray discs.
- **MPEG-7.** A standard for multimedia content that uses metadata and time code for content playback. It is intended as a complement to MPEG-4. The combination of MPEG-4 and MPEG-7 is sometimes referred to as MPEG-47.
- **MPEG-21.** A machine-readable format for license and rights information.

MPU-401. Abbreviation for MIDI Processing Unit 401. A type of MIDI interface developed by Roland for PCs around 1984 that evolved into a standard. The original MPU-401 consisted of a computer expansion card that connected to an external box containing MIDI in, out, and thru; tape in and out; and MIDI sync ports. There were a variety of descendants to the original MPU-401 over the years. Support for the MPU-401 standard was finally removed from Windows with the release of Windows Vista (though some other operating systems still support it).

ms. Abbreviation for millisecond (with lowercase "m" and "s"). One-thousandth of a second.

MSB. Most Significant Bit (though technically "MSB" in all caps would refer to "Most Significant Byte"). The bit in a digital word that has the most impact on the mathematical value of the word—typically the leftmost bit.

M-S stereo (a.k.a. mid-side stereo). A stereo miking technique using two coincident microphones—one in cardioid (though omni and other patterns are sometimes used) pattern facing toward the source and one in figure-8 pattern faced at 90 degrees to the source. The figure-8 mic is split to feed two mixer channels, one of which is reversed in polarity. By combining the cardioid mic with the two figure-8 signals and varying the balance of the channels, the stereo width of the resulting sound can be changed. M-S stereo signals are 100% mono-compatible; when the stereo signal is summed to mono, the two out-of-phase figure-8 "side" signals cancel each other, leaving only the mono "mid" mic signal. See Figure M.6.

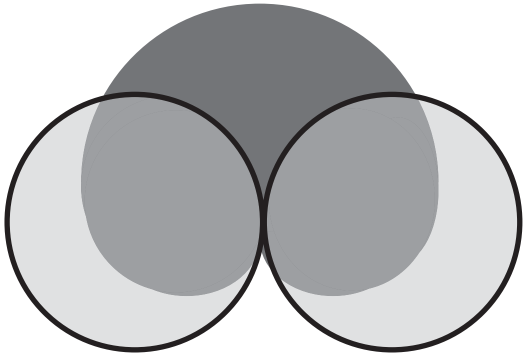

Figure M.6 M-S stereo is a miking technique that uses coincident cardioid and figure-8 mics. It allows the engineer to adjust the stereo width of the recording and provides complete mono-compatibility.

MTBF. ☞ **See** *mean time before failure*.

MTC. ☞ **See** *MIDI Time Code*.

MTC quarter frame. A message that allows subframe accuracy from linear time code sent over MIDI. Four quarter-frame messages are sent per frame, each containing a time-code stamp.

MTS. ☞ **See** *MIDI time stamping*.

mu. 1. ☞ **See** *gain*. 2. A ratio of change in voltages within a vacuum tube that determines its amplification and therefore the gain of a stage in a circuit.

mult. Short for multiple. 1. A passive signal splitter, usually in a patchbay. Several jacks on the bay are wired together so that a signal entering one jack will simultaneously feed the other jacks. ☞ **See also** *normal*. 2. An audio engineering term for splitting an output signal so it can feed multiple inputs simultaneously.

Multi mode. A term used by some manufacturers for the mode in which a synthesizer or sampler operates as a multitimbral device.

multiband compressor. A device, program, or plug-in that separates the audio range into bands of frequencies, then uses independent compressors to process each band independently. For example, a multiband compressor with three bands would separate the audio range into low-, mid-, and high-frequency bands using a crossover. A separate built-in compressor could then be used to process each band, allowing different compression amounts, thresholds, and parameters to be applied to the three bands. A multiband compressor can be thought of as a sort of "dynamic" equalizer. Multiband compressors can be used to shape the overall sound of a track, to tame a troublesome range of frequencies, for de-essing, and for other applications. See Figure M.7.

multiband limiter. A processor that separates the audio range into bands of frequencies, then uses independent limiters to process each band independently. ☞ **See also** *multiband compressor*.

multichannel. Literally, more than one channel, though in practice usually referring to more than two channels of audio. The term is used in a variety of ways; one example would be to describe a processor that can effect two, four, or more channels of audio independently.

multi-client. A function of some software drivers that allows more than one application to access a peripheral device simultaneously.

multi-core. A die or chip containing more than one computational core processor. Additional cores may or may not increase computational speed depending on the application, but they will increase the ability to multitask. ☞ **See also** *core*.

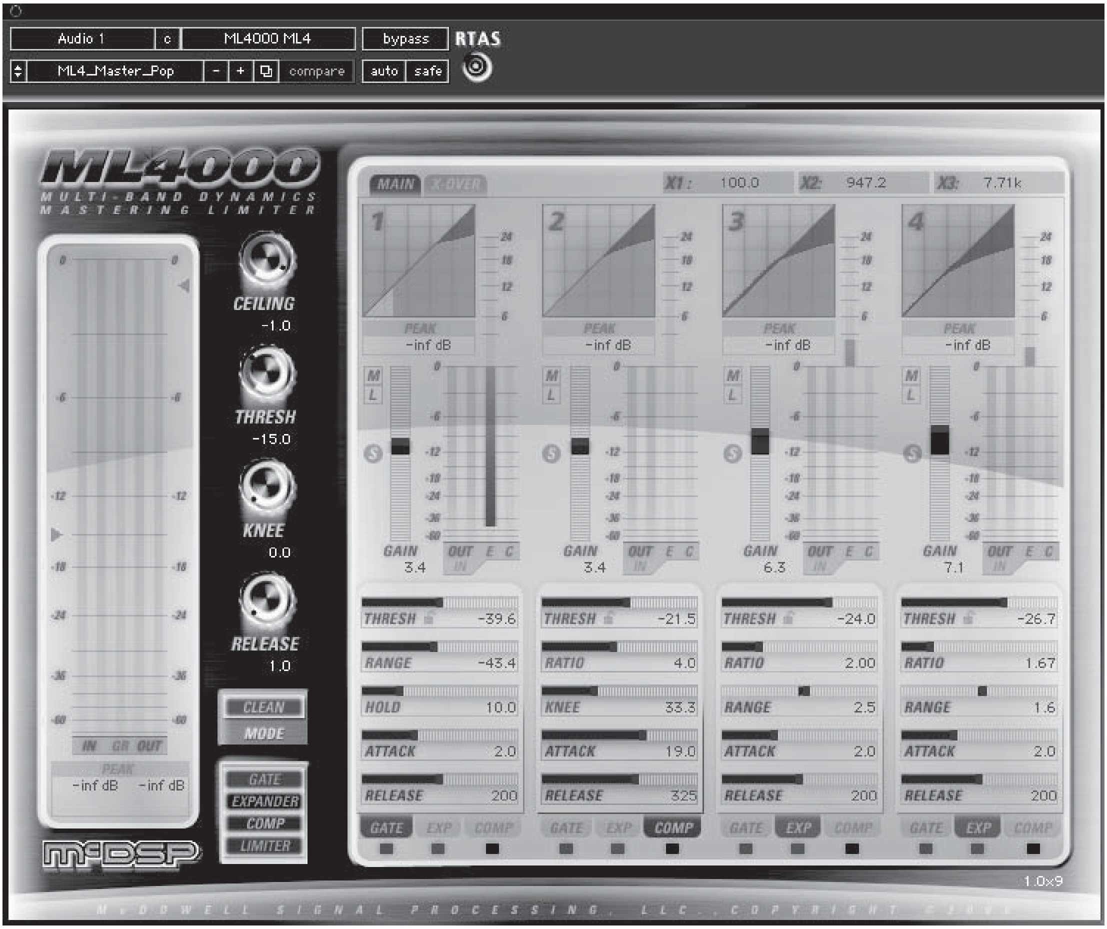

Figure M.7 A multiband compressor splits the audio into separate bands of frequencies, each of which can be processed with an independent compressor.

multi-effects. An effects processor that can produce multiple types of effects simultaneously.

multi-FX. ☞ See *multi-effects*.

multimedia. One of any number of art forms that combines more than one medium or discipline, including combinations of music, graphics, dance, drama, video, and others.

multi-mode filter. Some filters can only produce one type of filtered output: high-pass, low-pass, band reject, and so on. A multi-mode filter can produce several types of filtered outputs simultaneously.

multi-pattern microphone. A microphone that can be switched to use one of several polar patterns. Most multi-pattern microphones utilize dual diaphragms that are electronically combined to create the various patterns.

multiplex. To encode signals so that more than one can be transmitted down a single channel or cable simultaneously.

multi-sample. A collection of related samples that comprise a single preset or sound in a sampler or sample-playback synthesizer. For example, a piano "sample" or sound might consist of a separate sample (or even layer of several samples) for each note on the keyboard. (More commonly, to conserve memory and processing resources, each sample in a multi-sample will cover a range of several musical notes.) Multi-sampling is designed to prevent or at least reduce the problem of samples changing in timbre as they are pitched up or down by limiting how far each sample must be transposed.

multisession. A compact disc that conforms to the Orange Book standard. Multisession CD-Rs are discs that have been burned but are still "open," meaning that more data can be added to whatever data is already stored on the disc. Once the disc is "closed" or "fixed," a TOC (Table of Contents) is added. At that point, the CD-R becomes a Red Book–compatible CD if it contains audio tracks or a Yellow Book–compatible CD if it contains data.

multi-tap delay. A type of stereo delay where the user can specify more than one time interval at which discrete echoes will occur. A multi-tap delay can be thought of as many separate delay lines within a single hardware or software processor, each with its own delay time, output level, panning, and feedback amount.

multitasking. A device that is capable of performing more than one function or process simultaneously. In computers, multitasking refers to the operating system's ability to run more than one program simultaneously. The OS is able to do this by scheduling which task is running, and maintaining a queue of tasks or operations that are waiting to be performed. (The switch from one task to the next is called a *context switch*.) There are three approaches to computer multitasking:

- **multiprogramming.** An operation continues to run until it is finished, it must wait for new data, or scheduling tells the processor to context switch, at which time the next task or operation begins.
- **time-sharing.** The processor performs a context switch in response to an interrupt or other event.
- **real-time.** Some tasks or operations are guaranteed to occur or are given priority in order to maintain the timing of other operations and events.

multi-threading (a.k.a. parallelism). A type of computer program that can nearly instantly switch between multiple streams of operations, or "threads," giving the appearance that more than one stream of instructions is occurring simultaneously. Multithreading also allows the processor to operate more efficiently; for example, if a particular thread is waiting for data in order to continue processing, the OS can switch to working on a thread that has data available. Unlike multi-processing systems, which use multi-core processors, multi-threading takes place on a single processor. Multi-processing systems can also run more than one program at a time, while multi-threading takes place in one program and utilizes one cache and buffer for all the threads. The disadvantage to multi-threading is that multiple threads can conflict with one another, and the execution of any single thread may be slowed (while the overall speed of executing all the threads may be increased). Multi-threading and multi-processing are complementary and are often combined, such as in multi-core computers that are capable of multi-threading. ☞ **See also** *multitasking*.

multitimbral. A function of some synthesizers and samplers that allows the instrument to produce more than one sound (timbre) simultaneously. The device is split into multiple independent virtual parts, each of which can play back notes over a

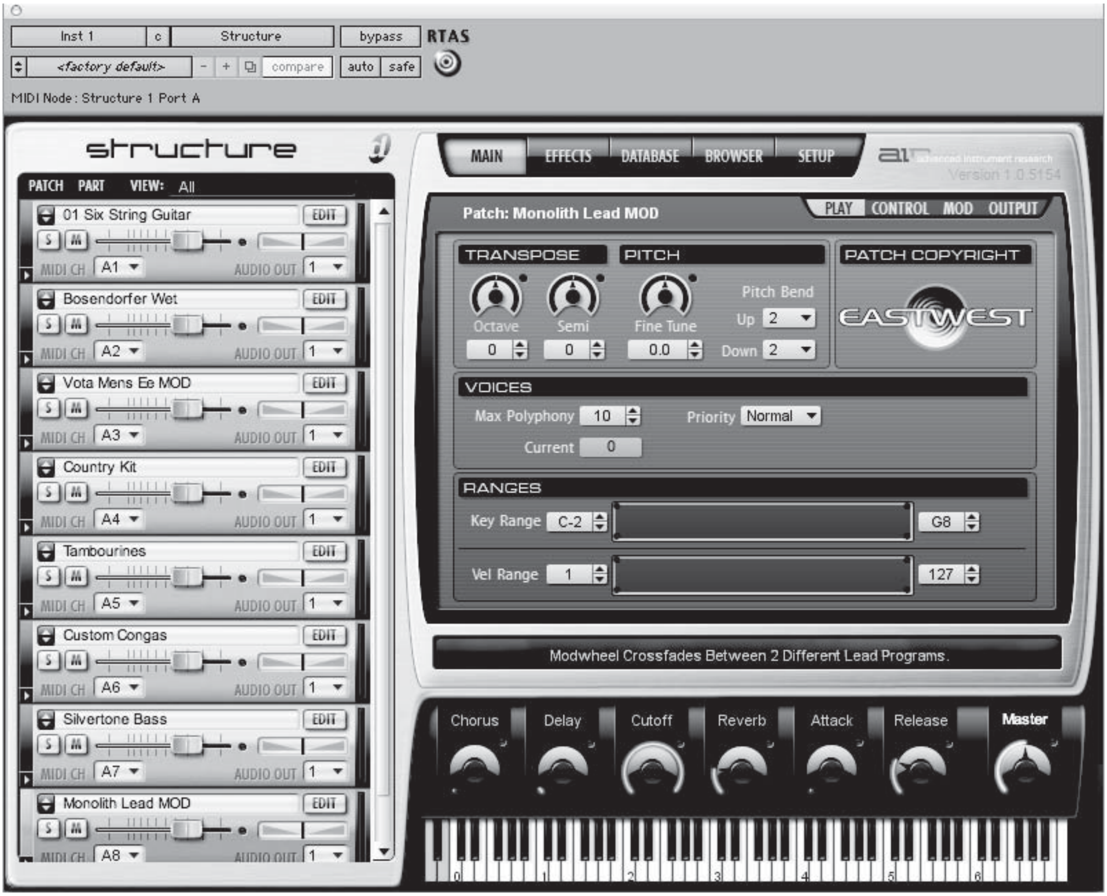

Figure M.8 A multitimbral synth or sampler can be divided into virtual "parts," each of which can have its own sound or preset and can respond to its own MIDI channel.

separate MIDI channel using a different preset or patch. See Figure M.8.

multitrack. 1. Literally, more than one track. 2. An analog or digital recorder that is capable of recording and playing many independent signals simultaneously. 3. A recording consisting of multiple tracks containing independent parts and performances that must be mixed to create a final piece of music or other final audio production. 4. The process of recording a piece of music by using multiple tracks to store independent parts and performances that are later combined at mixdown.

music. Though there are probably as many definitions for "music" as there are musicians and listeners, the most literal definition is likely "organized sound."

mute. 1. To turn off an audio signal or channel. 2. A control on a mixer or other device that turns off or silences a signal or channel.

mute group. A function in some hardware and software mixers that allows a single button to mute and unmute a selected group of channels simultaneously.

Mylar. A brand or trade name for thin polyester film developed by DuPont in the 1950s. Mylar offers good strength and light weight, making it ideal as the base material for microphone diaphragms and headphone drivers.

N

NAB. National Association of Broadcasters. An organization that develops standards for, and works to support, radio and television broadcasting. Certain audio standards from the NAB apply to pro audio applications. www.nab.org.

NAMM. National Association of Music Merchants. The commonly used name for the International Music Products Association (IMPA). NAMM is a trade organization that supports audio and music product manufacturers, distributors, and retailers. NAMM is perhaps best known for its two annual tradeshows, Winter NAMM and Summer NAMM, where manufacturers introduce many of the new products that will be sold by retailers throughout the rest of the year. www.namm.org.

nanoweber. One-billionth of a weber, which was named for German physicist Wilhelm Eduard Weber. A unit used to indicate magnetic strength. Analog tape recorders are calibrated to a certain number of nanowebers per meter of tape in order to achieve maximum headroom and minimum noise and distortion on the tape.

NARAS. National Academy of Recording Arts and Sciences. Best known as the organization that puts on the annual Grammy awards, NARAS is an organization that works to support musicians and the music industry. www.grammy.org.

native (a.k.a. host-based). A term used to indicate that a piece of software runs using only the host computer's built-in processing power and does not depend on an additional DSP card or other hardware for processing power.

navigation wheel. A rotary control on a control surface that allows the user to quickly and accurately locate to a specific point in a project or to a specific track. A navigation wheel may include jog and shuttle functions, up/down and left/right arrows for moving within a window, and other features.

near-coincident pair. A family of stereo miking techniques in which two microphone diaphragms are placed close to one another and angled apart. This increases the stereo width and ambience compared to coincident placement, but may decrease mono compatibility. Examples include:

- **ORTF.** Two cardioid microphones placed 17 cm apart (around 6-3/4 inches), angled at 110 degrees.
- **DIN.** Two cardioid microphones placed 20 cm apart (around 7-3/4 inches), angled at 90 degrees.
- **NOS.** Two cardioid microphones placed 30 cm apart (around 12 inches), angled at 90 degrees.

near field. Positioning a sound source close to the listener, often defined as less than one wavelength, but usually accepted to be within 3 to 4 feet. Though commonly used, "near field" is a trademarked term, so "close field" is often substituted.

near-field monitor. Studio speaker designed to be used in close proximity to the listener. Near-field monitors take advantage of the Inverse Square Law, which says that sound level decreases by the square of the distance. The idea is that the monitors are close to the listener, who will hear primarily direct sound, with any reflections being much lower in level (and therefore much less destructive). "Near field" is a trademarked term, so some manufacturers use "close field" as a synonym.

needle. ☞ See *stylus*.

negative feedback. An amplifier design technique in which some of the amp's output is put out of phase and mixed back into the input so that it "opposes" the incoming signal. Depending on how the design is implemented, negative feedback can reduce distortion, change input and/or output impedance, increase damping factor, and improve frequency response and bandwidth. Poor implementation may result in instability and increased transient intermodulation distortion.

neodymium. A rare-earth metal element (number 60 on the periodic table) used for a variety of purposes, such as coloring glass and constructing magnets often used for microphones and speakers. Neodymium magnets are the strongest permanent magnets known (in addition to being cheap and light), so mics and speakers that use them can be made lighter and with higher outputs than those that use other types of magnets.

neutral. An electrical state of having no positive or negative charge. In power wiring, the neutral line is used as a zero reference for the positive and negative lines (the amount of voltage available is the difference between the "hot" lines and the neutral), as well as the ground.

NiMH. Nickel Metal Hydride. A type of rechargeable battery developed in the late 1980s, using a hydrogen-absorbing alloy instead of cadmium. NiMH batteries can hold up to three times as much charge as a nickel-cadmium type and have no memory effect. NiMH batteries are also recyclable, whereas other types generally are not. NiMH batteries should not be completely discharged if more than one battery is in use. If one battery is empty, and the other batteries do not completely discharge at the same rate, those with remaining charge will damage the empty one. Likewise, a discharged battery or batteries should not be used with a charged battery or batteries.

NLE. Non-Linear Editor. ☞ See *non-linear editing*.

node. 1. A position along a wavelength at which there is no motion. For a given frequency, nodes are spaced 1/2-wavelength apart. 2. A connection point on a ribbon or other cable with connectors at the end and at one or more points in the middle, used to hook up multiple items. 3. A computer or peripheral connected to a network.

noise. Undesired sound that isn't related to any desired sound. (If it is related, it is distortion.) Examples include hiss, hum, rumble, and unwanted background and ambient sounds.

noise cancellation. ☞ See *noise suppression*.

noise floor. 1. The level of ambient noise in a room. 2. The self-generated noise in a device when no signal is passing through the device. Reducing the noise floor increases the dynamic range.

noise gate. A type of dynamics processor that automatically shuts off a channel when no desired signal is present so that any noise floor is muted. A threshold parameter is set that determines the minimum signal level that will open the gate—signals higher than the threshold will open the gate, and audio will pass as normal. When the signal falls below the threshold, the gate closes, muting the audio and silencing any noise. An attack time parameter may be available to determine how fast the gate reacts. A release parameter determines how quickly the gate returns to its closed setting. A hold parameter keeps the gate open for a specific amount of time. The release and hold settings are important for allowing a sustaining signal to decay smoothly without being chopped off by the gate as it drops below the threshold. See Figure N.1.

noise generator. An oscillator that produces a noise signal, often white or pink noise. Noise generators are used in synthesizers and for test and calibration purposes.

noise profile. A capture or "image" of the noise in a track or audio path when no desired signal is present. A noise reduction processor compares the overall sound of the signal against the noise profile to determine what to remove to reduce or eliminate the noise. For example, if the noise profile shows that there is a hum at 60 cycles, when the overall signal is playing, the noise reduction processor can remove that hum, leaving only the desired signal behind to be heard. See Figure N.2.

noise reduction. One of a number of processes and algorithms that are designed to lower the level of noise or to remove it completely.

noise reduction coefficient. ☞ See *NRC*.

noise shaping. 1. The process of filtering certain types of noise in order to make them less audible. In most cases, this means reducing mid- or high-frequency components. 2. A bit-reduction technique similar to dither that is used with digital audio and digital video to reduce quantization error. Noise shaping moves quantization errors to less audible frequencies. Examples include POW-r, which can reduce a 24-bit signal to 16 bits while focusing the noise under 60 Hz and over 12 kHz. 3. A bit-reduction technique similar to dither that uses pre-filtered

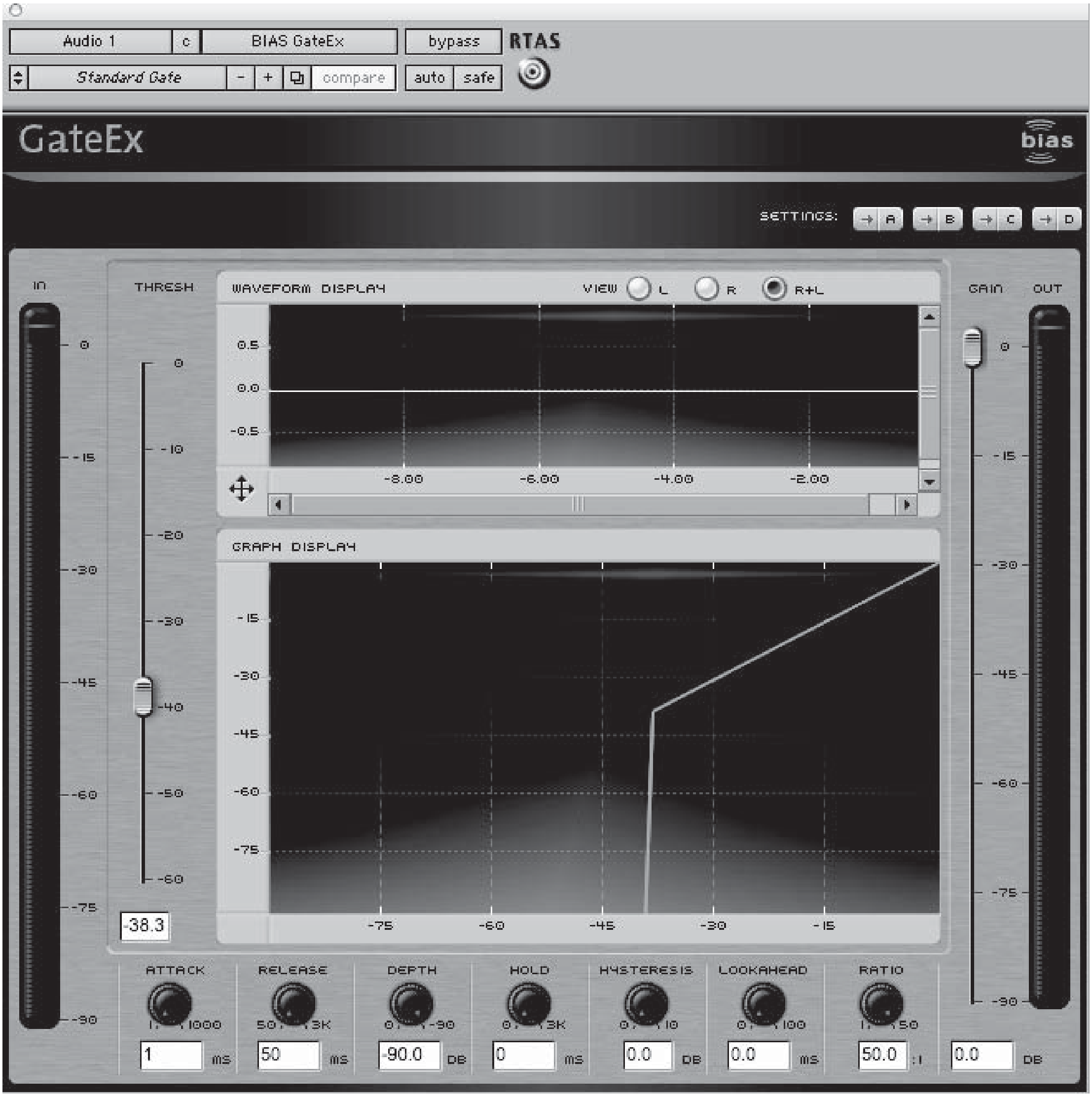

Figure N.1 A noise gate opens when the desired signal crosses the threshold, allowing the signal to flow through. When the signal falls below the threshold, the gate closes, muting the output and silencing background noise.

dither noise. Apogee's UV22 and UV22 HR use this technique.

noise suppression (a.k.a. noise cancellation). A type of sound isolation system that uses active components, such as out-of-phase speakers, rather than relying on soundproofing materials.

NOM. Number of Open Mics. A term used in live sound/sound reinforcement to refer to the number of microphones onstage that are "open" or active—with their gain and volume up and un-muted—and thus picking up background sound and potentially feeding back. In installed systems (such as churches

Figure N.2 Some noise reduction software works by taking a "picture" of the noise, called a noise profile. The noise profile is used to determine what sonic material the software will consider noise in order to remove it from the overall signal.

and some clubs), automated mixers are used that keep the NOM at a minimum by automatically muting unused open mics.

nominal level. The "normal" or typical level at which a device is designed to operate. The nominal level is usually the zero reference level for the device. For example, in a +4 dBu device, +4 dBu will result in a 0 meter reading. Nominal level is usually specified to allow sufficient headroom for peaks without distortion.

non-destructive editing. A type of digital editing in which the raw or original data or file is not permanently changed. Any edits made are simply "pointers" that indicate how the data should be altered and arranged during playback or display in order to achieve the desired result.

non-destructive recording. A type of digital recording in which new takes or punches do not overwrite or erase existing data. The newly recorded data is written to a completely separate file rather than modifying the existing files.

non-drop. Time code in which there are 30 frames per second. ☞ **See also** *drop frame*.

nonlinear. Refers to any format that provides digital access to information, including recorders and workstations that transfer sound directly to and from disk, thus eliminating search time.

nonlinear editing. A feature of hard disk–based DAWs and other digital editing programs that allows random access to source data. This is different from tape and film, which generally require that the engineer edit in a sequential fashion. With nonlinear editing, the editor can instantly jump to any point in the project and make a change. The first nonlinear editing system was developed for video in 1971 by CBS and Memorex.

non-normalled (a.k.a. open). A patch bay configuration in which the top and bottom rows of the bay are not connected. Cables patched into the rear jacks of each row simply pass their signal to the front jacks, where they can be accessed by inserting another cable. ☞ **See also** *normal*.

non–real time. A computer process or event that does not happen live while the listener hears it. Instead, the process is started and the computer performs its computations. When it finishes, the listener is able to play back and hear the result. ☞ **See also** *off line*.

non-registered parameter number (a.k.a. NRPN, sometimes pronounced "NER-pin"). A type of MIDI channel continuous controller message that is not defined by the MIDI specification, but instead is available to be defined by a particular manufacturer for use by that manufacturer's devices. The same non-registered parameter number may be used for different purposes by different manufacturers, which can result in occasional conflicts. A non-registered parameter number is a multipart message. CC #98 and #99 are used to indicate the parameter number, and CC #6 (and optionally #38 for more resolution) indicates the parameter value. Data increment messages (CC #96 and #97) can be used to adjust the current value of a registered parameter number.

non-volatile memory. A type of RAM that does not lose its contents when power is interrupted or shut off.

normal. An internal connection between the top and bottom rows of a patch bay. When a signal is patched in the top row, the signal flows into the patch bay, through the normal, and is outputted through the rear bottom jack. For example, the output from a microphone preamp might be connected to the rear top jack on the bay; a computer audio interface input might be connected to the rear bottom row. When a microphone signal passes through the preamp, it enters the patch bay and exits into the interface input without the need to make any other patch or connection. Depending on how the patch bay is configured, plugging into the front jacks of the bay may interrupt or re-route the signal. There are several possible configurations:

- **full normal.** Plugging into either the front top or bottom jacks of the bay will interrupt the "normal" signal flow from the top rear to the bottom rear jacks. This is useful for having a normal signal path in place, but being able to interrupt that path to access other gear. See Figure N.3.
- **half normal.** Plugging into the front top jack of the bay will *not* interrupt the "normal" signal flow from the top rear to the bottom rear jacks, allowing the signal to "Y," or split. However, plugging into the front bottom jack will interrupt the normal signal flow. See Figure N.4.
- **mult.** A cable can be connected to the front top or bottom jacks to "Y," or split the signal. Signal will also continue to flow from the top rear to the bottom rear patch bay jacks. See Figure N.5.
- **open (a.k.a. non-normal).** There is no connection or normal between the top and bottom rows of the patch bay. See Figure N.6.

normalize. A DSP function that raises the level of an audio file until its loudest point or highest peak is at 0. In some cases, normalizing can be used to raise or lower the highest peak to a user-specified level rather than to 0. Normalizing can be useful, such as when a signal was recorded at an extremely low level. However, normalizing is done at the expense of headroom and will raise any noise that

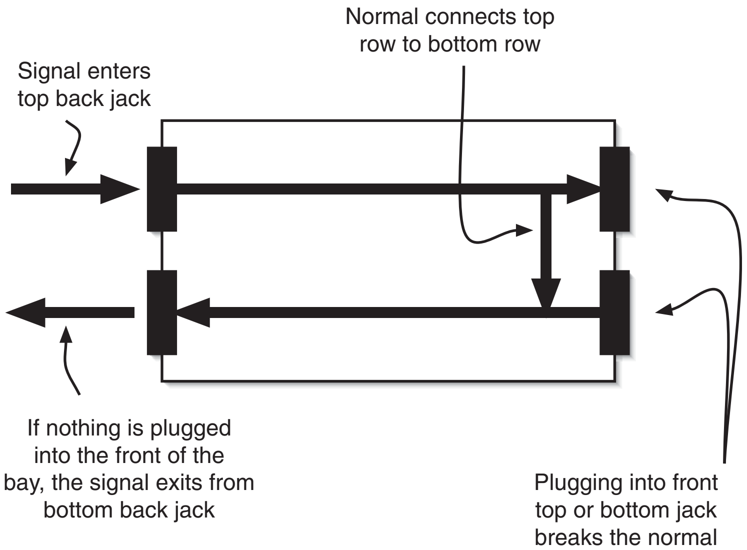

Figure N.3 A full normal is interrupted if a cable is connected to the front top or bottom jacks.

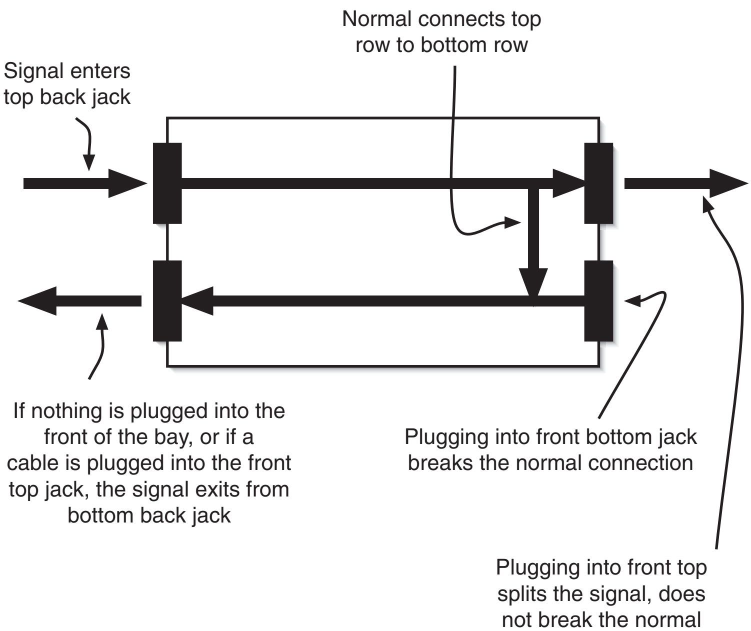

Figure N.4 Plugging a jack into the top front jack on a half-normalled patch bay will result in a signal split. Plugging into the lower front jack interrupts the signal.

was recorded with the file. Normalizing is not the same thing as compression or limiting, which is used to reduce peaks in order to raise the average level. Normalizing does increase the average level, but only to the extent that the peaks can be raised, as there is no dynamics processing taking place.

NOS. New Old Stock. A brand-new component or product that was manufactured some time ago, but was never sold or put into service. Vintage NOS vacuum tubes, in particular, are in high demand.

notation editing. A view or editing window in a DAW or sequencer that displays MIDI data as standard music notation. This allows a composer or engineer who reads music notation to edit the data using a familiar graphic format. ☞ **See also** *score window*.

notation software. A type of software that can transcribe MIDI data to standard music notation. Once the music notation is transcribed, the notation and underlying MIDI information can be edited and processed in a variety of ways to create a graphic layout that can be printed and distributed as sheet music.

notch. A narrow band of frequencies that has been reduced in level compared to the surrounding frequencies, usually by using an equalizer or filter with a very high Q/narrow bandwidth. See Figure N.7.

notch filter. A filter with a very high Q that is used to reduce the level of a narrow band of frequencies. Notch

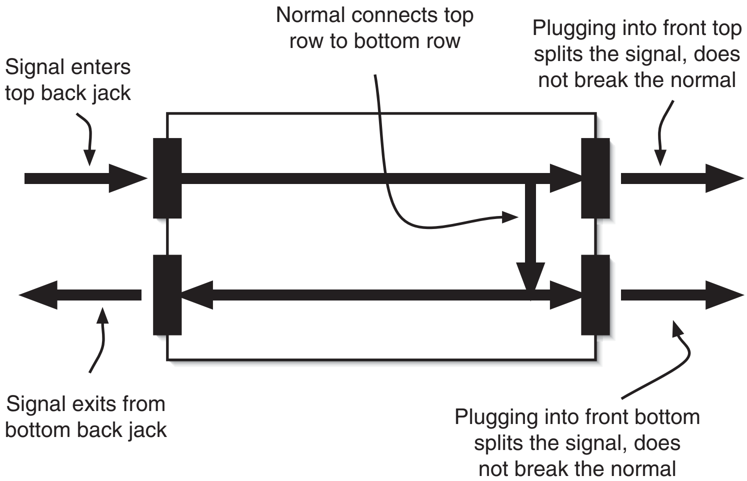

Figure N.5 A mult works like a passive signal splitter.

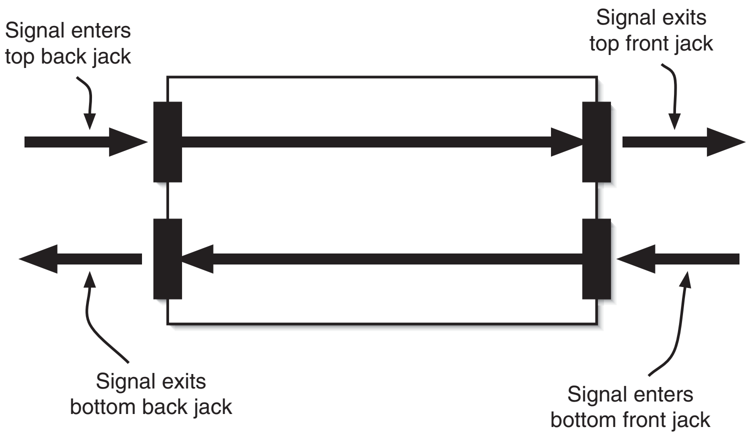

Figure N.6 There is no connection between the top and bottom rows when a patch bay is open or non-normalled.

filters are often used to control feedback in live sound monitor systems, as well as to reduce certain types of noise, such as 60-cycle hum.

note. 1. A single musical event with a specific pitch. 2. A key on a musical keyboard. 3. A MIDI channel message consisting of two parts—a note on and a note off—which is used to initiate and halt playback of a specific sound in a synthesizer or sampler.

note grid. ☞ See *drum grid*.

note number. ☞ See *MIDI note number*.

note off. A MIDI channel message that is used to stop playback of a note on a synthesizer or sampler. A note off message consists of several components, including the MIDI channel for the message, the MIDI note number, and the release velocity for the note. ☞ **See also** *note on*.

note on. A MIDI channel message that is used to trigger or begin an event, such as playback of a note on a synthesizer or sampler. A note on message consists of several components: the MIDI channel for the message, the MIDI note number, and the velocity for the note. A note on message must be followed by a corresponding note off message at some point, or the note will continue sounding, resulting in a stuck note. ☞ **See also** *note off*.

note stealing. A synthesizer or sampler is limited in the number of notes it can sound simultaneously—it has a finite amount of polyphony. When that limit is exceeded, the instrument can refuse to sound another note until an existing note is released, or it can "recycle" the polyphony by stopping one of the existing notes in order to play the new note. Most modern instruments have sophisticated algorithms to determine which note is the least audible, and they will "steal" that note in order to play the new note.

NRC. Noise Reduction Coefficient. An overall performance rating for an acoustical material derived by averaging the absorption or acoustic coefficients across a range of octave bands. Because it is derived from an average, it does not include detailed absorption data for the material, and it is therefore less useful than the absorption coefficients for most comparisons. NRC ratings range from 0 (completely reflective) to 1 (completely absorptive). ☞ **See also** *acoustic coefficient*.

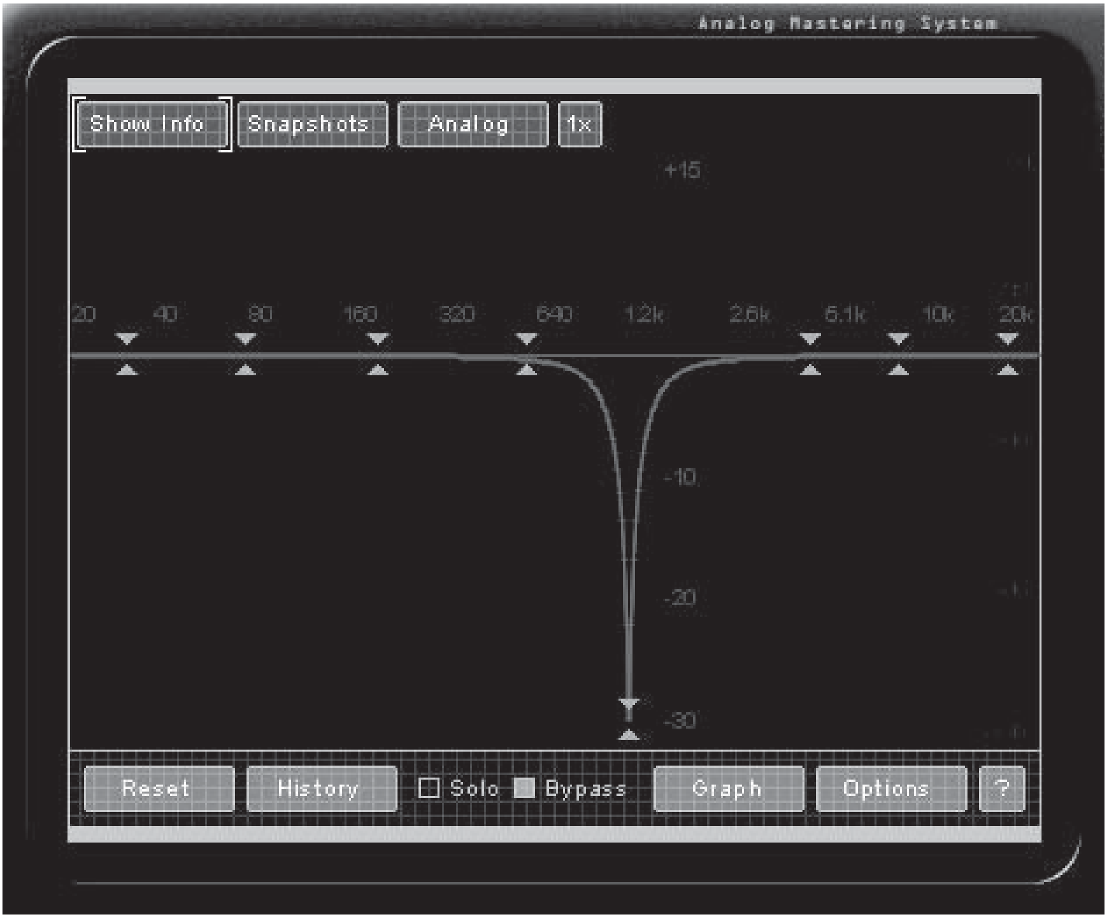

Figure N.7 A notch is a narrow band of frequencies that has been reduced in level by a filter or other means.

NRPN. ☞ See *non-registered parameter number*.

nudge. To move an audio or MIDI segment or region earlier or later in its track by a very small time increment.

null. 1. A position in a room where a dip in the level of a particular frequency or range of frequencies occurs due to phase cancellation or reinforcement. 2. With a mixer, to return a control to its default position. 3. An area in a microphone's polar pattern where no sound is picked up.

Nyquist frequency (a.k.a. folding frequency). According to the Nyquist-Shannon Theory, the highest frequency that may be accurately sampled for a given sample rate. In order to accurately sample a signal's highest component frequency, at least two samples must be taken per cycle, thus the Nyquist frequency is half the sample rate. Frequencies above the Nyquist frequency will be "aliased" or folded over. (See Figure N.8.) For example, if the sample rate is 48,000 Hz, the Nyquist frequency is 24,000 Hz—the highest theoretical frequency that can be sampled (in practice, the rate is somewhat lower). If a frequency of 30,000 Hz is sampled, it will be "folded over," resulting in a "phantom" frequency of 18,000 Hz. Virtually all analog-to-digital converters have anti-aliasing filters or use other technology, which prevent this problem by removing frequencies above

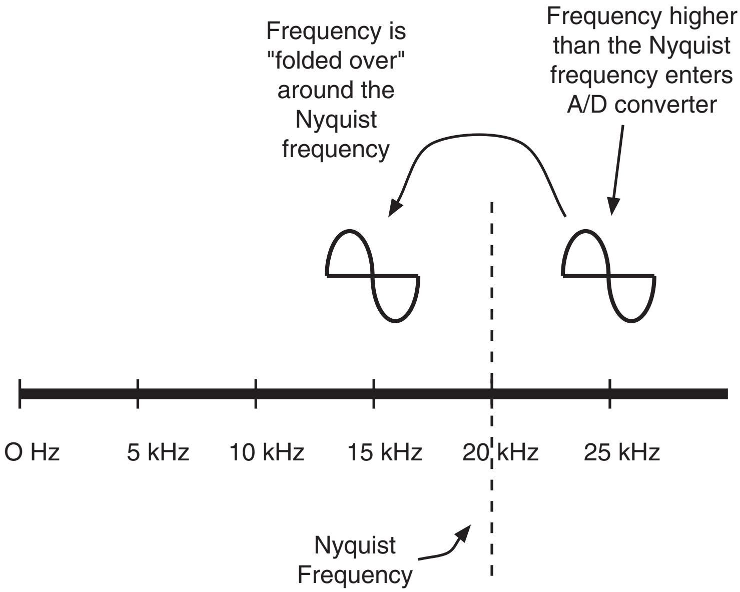

Figure N.8 The Nyquist frequency is half the sample rate. If frequencies above the Nyquist frequency enter an analog-to-digital converter, they will be "folded over" around the Nyquist frequency, resulting in "alias" frequencies in the audible range. For this reason, analog signals are low-pass filtered before they are sampled in order to remove frequencies above the Nyquist frequency.

the Nyquist frequency. The Nyquist frequency was named for Swedish engineer Harry Nyquist.

Nyquist Theorem. ☞ See *Nyquist-Shannon Theory*.

Nyquist-Shannon Theory. A theorem for digitally sampling analog signals that was based on the work of Harry Nyquist in 1928 (though Nyquist did not actually develop the theorem), and later proven by Claude Shannon in 1949. Basically, the theorem states that a signal can be perfectly reproduced if it was sampled at a rate at least twice the highest frequency in the signal. For example, theoretically a signal where the highest frequency is 10 kHz can be perfectly reproduced if sampled with a rate of at least 20 kHz.

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O

oblique mode. A room mode created by a sound wave reflecting across all six surfaces (four walls, floor, and ceiling) in a room. Oblique modes are roughly half as strong as tangential modes, and one-fourth as strong as axial modes.

octave. 1. A doubling or halving of frequency. 2. A musical interval of eight diatonic steps or twelve equal half-steps.

odd-order harmonic distortion. A type of distortion that results primarily in additional third and fifth harmonics being added to a signal. Odd-order harmonic distortion is often produced when digital devices clip; it tends to be unpleasant-sounding to most listeners.

OEM. Original Equipment Manufacturer. A company that sells products made by another manufacturer under its own name or that uses parts or subassemblies from another company to make its own products. The term may also be used to refer to a company that makes products that another company will sell as its own products.

OFC. Oxygen-Free Copper. The more pure, or oxygen-free, copper is, the better it conducts electricity. Some audiophiles and engineers feel this makes OFC a superior choice for audio cables, though it can be substantially more expensive.

off-axis. 1. Positioning a source somewhere other than directly in front of the diaphragm of a directional microphone. Depending on the microphone's polar pattern, an off-axis position may result in attenuated levels and/or coloration. 2. A position that is not directly in front of a speaker. Depending on the position in the speaker's dispersion pattern, off-axis positioning may result in sonic coloration. See Figure O.1.

off line. 1. Inactive. 2. Editing or processing that takes place outside of real time. 3. Not connected to or accessible by a network.

offset. A slight shift in relation to time code that allows devices to synchronize correctly under certain conditions. An offset may move a device's synchronization either earlier or later in time (positive or negative offset).

Figure O.1 A sound entering a directional microphone from off-axis (any direction other than directly in the front) will have some degree of coloration.

Ogg Vorbis. An open source, unpatented audio file format named for the Vorbis data compression scheme used to create the files. (The name "Vorbis" is taken from a name used in Terry Pratchett's *Discworld* series of books, while "Ogg" comes from "ogging," which is computer game jargon.) Ogg Vorbis is part of the Ogg project, whose goal is to create a completely open source multimedia system.

ohm. The unit of measurement for electrical resistance and impedance. The ohm was named for German physicist Georg Ohm.

Ohm's law. An electrical interrelationship, discovered by German physicist Georg Ohm around 1827, between voltage, current, and resistance. Ohm's law states that voltage equals current times resistance ($V = I \times R$). The law can be rearranged to find resistance ($R = V/I$) or current ($I = V/R$). Basically, this means that as, for example, voltage changes, either resistance or current (or both) must also change, and so on.

OLED. Organic Light Emitting Diode, a.k.a. LEP (*Light-Emitting Polymer*) or OEL (*Organic Electroluminescent*). An LED whose luminescent layer is composed of organic material. The big advantage OLEDs have over LCDs (*Liquid Crystal Displays*) is that they don't require a backlight, and therefore do not draw as much power and can be made much thinner. The biggest disadvantage is the limited lifespan of the organic materials—about 25% of the lifespan of LCD or LED technologies.

OMF. Open Media Framework, a.k.a. OMFI (*Open Media Framework Interchange*). A standard file format for exchanging audio and video files between different editing and production programs.

omnidirectional. Literally, in all directions at once. Omnidirectional microphones pick up sound in a spherical pattern, equally well from all directions. Low-frequency speakers tend to be omnidirectional in their dispersion pattern. See Figure O.2.

omni mode. A MIDI channel mode in which a device will respond to messages and data received on any or all MIDI channels.

OMS. Opcode MIDI System, later changed to Open Music System. A Macintosh-based MIDI environment developed by Opcode in 1990. OMS facilitated MIDI communication between hardware devices, computers, and software applications, such as sequencers and editor/librarians. OMS and FreeMIDI, a competing system developed by MOTU, were made obsolete by Core MIDI in Apple's OS X.

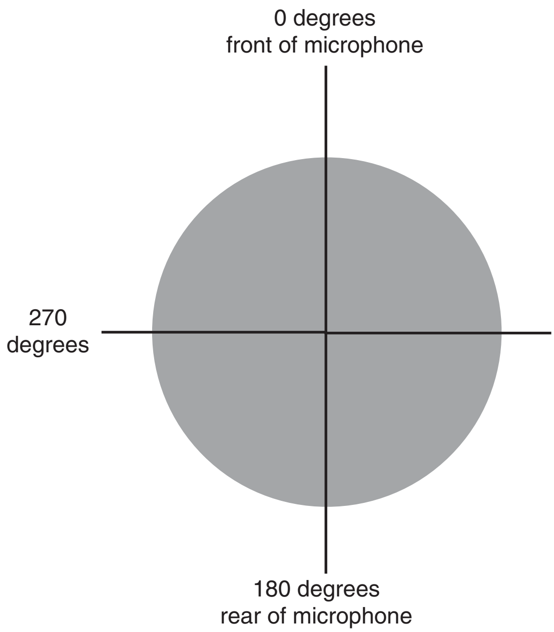

Figure O.2 An omnidirectional microphone picks up sound equally well from all directions.

on-axis. 1. Positioning a source directly in front of the diaphragm of a microphone, at 0 degrees in the polar pattern. 2. A position directly in front of a speaker, in the middle of the dispersion pattern.

onboard. Synonym for "built-in," as in "onboard effects."

one-off. A single copy of something or copies that are made one at a time. For example, a CD-R (or CD-Rs) burned one at a time from an audio project, as opposed to a mass-produced, commercially replicated CD.

one-shot sample. A digital sample that does not loop. A one-shot sample is triggered, plays back, and stops. Examples would include drum hits, orchestral stabs, short musical phrases, sound effects, and others.

op amp. Short for "operational amplifier." A type of electronic integrated circuit chip that amplifies the level of signals. Op amps are used in many types of audio equipment.

opcode. 1. Short for "operation code." A part of machine language that tells the computer what operation is to be performed. 2. Opcode was a MIDI software/hardware company founded in 1985 and acquired by Gibson in 1998. Opcode created popular MIDI software including Vision (sequencer), Studio-Vision (among the first sequencers to add audio recording and editing), Galaxy (universal editor/librarian), OMS (Opcode MIDI System, later Open Music System), and others. By 2000, opcode was defunct.

open. 1. An incomplete electronic circuit with a switch that is not closed or a break at some point. 2. A patch bay that has no normalling between two patch points.

open-air (a.k.a. open back). A type of headphones that do not seal off the wearer's ears from external sound or prevent internal sound from bleeding to the outside world. Open-air headphones can be more comfortable for long periods of time, but they are not suitable for tracking where isolation is required.

open mic. A microphone that is on and capturing, or available to capture, sound.

OpenSound Control (a.k.a. OSC). A network protocol designed for real-time communication among synthesizers, computers, and multimedia devices and software. OpenSound Control was intended to supplant MIDI; it operates at broadband speeds and allows communication via Ethernet connections or over the Internet.

open source software (a.k.a. OSS). Software that has been created and released under an open source license, which allows users to not only use, but modify and redistribute, the software. Open source software is often programmed by a public group of collaborators and distributed for free via download over the Internet.

operating system. A collection of basic instructions that define the operation of all digital devices, including computers, synthesizers, samplers, cell phones, iPods, and so on. The operating system is the underlying program on top of which all other software runs. Examples for computers include Windows, OS X, Linux, and others.

operator. The equivalent of an oscillator—a module that generates an output signal representing a cyclical waveform—in an FM (*frequency modulation*) synthesizer.

operator error (a.k.a. PEBKAC). A common cause of equipment maladies. Generally curable by reading the manual.

optical compressor. A compressor design based around an electro-optical element (a combination light source and photocell light sensor). The intensity of the light source is proportional to the input signal level; the photocell detects the light and controls the level of an amplifier to reduce the output level. Threshold, ratio, and other settings are used to specifically set the response of the compressor.

optical media. A disc media, such as CD or DVD, that stores digital information in a format that can be read (and/or written) by an optical device, such as a laser. Optical media has several advantages over similar removable magnetic devices (such as floppy disks and others), including large storage capacities and durability.

option anxiety. A situation faced by many modern technology-savvy musicians and engineers, where the plethora of available options makes reaching a decision very difficult. Whether it is choosing among buying one of 200 different kinds of microphones, selecting which of a multitude of plug-ins to use for a task, or figuring out which DAW to choose as a platform (among many other similar decisions, small and large), the range of workable options is so large that deciding which is the best solution is a serious problem.

opto-isolator (a.k.a. optocoupler, photocoupler). An electronic component that combines an LED light source and a light sensor to pass signal. When current is present, the LED turns on and light is detected by the sensor. Since opto-isolators use light instead of electrical contact, they can isolate a circuit and help prevent ground loops and damage from voltage spikes. According to the MIDI Specification, MIDI inputs must be opto-isolated.

Orange Book. One of a set of "Rainbow Books" with colored covers containing the Sony/Philips-developed specifications for different optical compact disc formats. The Orange Book contains the specifications for CD-R and CD-RW.

organ stop (a.k.a. stop). Technically, the length of the air column in an organ pipe. The organ stop determines the pitch produced by that pipe; there is one pipe per pitch the organ can produce. Today, "organ stop" is used to refer to the control

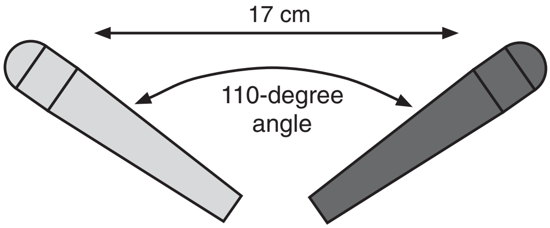

Figure O.3 The ORTF stereo miking technique uses two cardioid microphones placed 17 cm apart at a 110-degree angle.

on the organ that calls up a particular setting. The term carried over to tonewheel and other types of organs as well.

ORTF. *Office de Radiodiffusion Television Français.* A near-coincident stereo microphone technique in which two cardioid microphones are placed 17 centimeters (slightly less than 6-3/4 inches) apart, at an angle of 110 degrees. ORTF provides good mono compatibility and localization and imaging, but does not pick up much room ambience. See Figure O.3.

OS. ☞ See *operating system*.

OSC. ☞ See *OpenSound Control*.

oscillator. An analog circuit or digital algorithm that generates an output signal representing a cyclical waveform.

oscillator sync. A synthesizer function in which a slave oscillator synchronizes the period of its output signal with the period of the output signal from another master oscillator. There are two types: hard sync, in which the slave waveform cycle is retriggered every time the master waveform begins a new cycle, and soft sync, in which the slave waveform is inverted (its direction is reversed) every time the master waveform begins a new cycle.

oscilloscope. A diagnostic device used to display signal waveforms. Oscilloscopes are useful for a variety of measurements as well as for simply viewing and verifying a waveform shape.

OSS disc. Optimal Stereo Sound disc. ☞ See *Jecklin Disc*.

ostinato. A repeating musical pattern. A repeating loop or a repeating pattern played by a synth's arpeggiator could be considered an ostinato.

OS X. Version 10 of the Macintosh operating system. OS X was a complete break from the technologies used for previous Mac operating systems because it was the first Mac system based on UNIX.

outboard gear. Hardware processing equipment used to supplement the capabilities in a mixer or DAW. Outboard gear can include microphone preamps, compressors, limiters, EQs, summing boxes, effects processors, and more.

out of phase. A relationship in time of two sound waves of the same frequency, where the peaks and troughs in the waveforms don't line up with one another perfectly. If two identical signals are 180 degrees out of phase, the highest peaks in one signal exactly line up with the lowest troughs in the other signal, and the two will cancel each other out completely, resulting in silence. More often two signals are less than 180 degrees out of phase, resulting in partial cancellation and a filtered/hollow tonality. See Figure O.4.

output. A connection or jack used to send signal out of a device.

output transformer. A transformer used to match the output of a device to the device it will be feeding. ☞ See also *transformer*.

over. An over is a signal that exceeds 0 dBFS for a certain number of samples in a row. Since a digital system cannot represent a signal higher than 0, the waveform will not be accurately represented. An over is the digital equivalent of clipping an analog signal. A few overs may not be audible or may not create a problem in a track during production, but overs are a problem if they become audible as distortion. Even a single over can be problematic in a final mastered track that is being submitted for duplication to CD.

overdrive. A distortion effect created by increasing the level of a signal until it clips or distorts.

overdub. 1. The process of recording additional tracks over existing basic tracks in order to finish a production. For example, the basic tracks might consist of the rhythm section recorded at one session, all in one pass. Vocals, solos, and other tracks might be overdubbed one at a time at a later session. 2. An additional track that has been added to existing tracks.

over easy. A trademarked term used by dbx for soft-knee compression.

overhead microphones. ☞ See *overheads*.

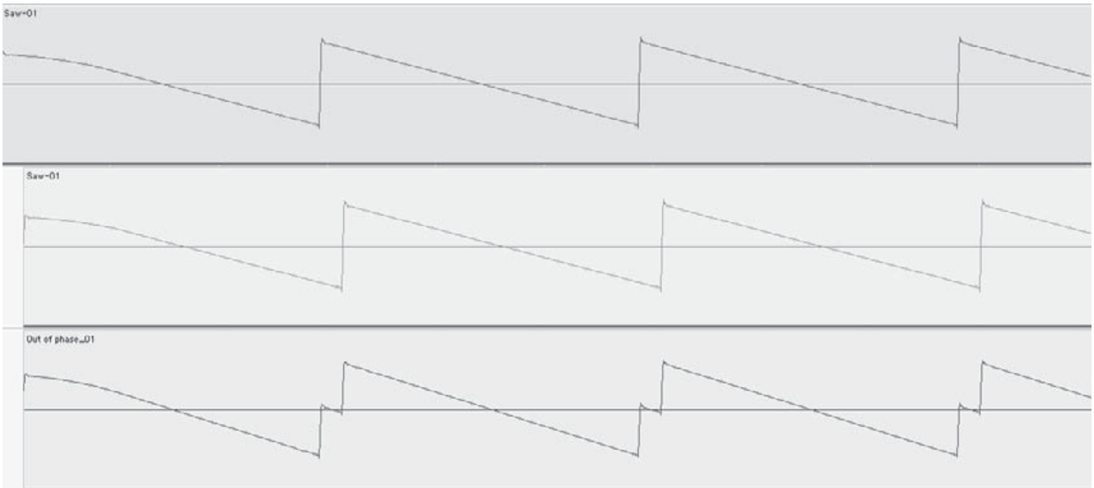

Figure O.4 If two identical signals arrive at slightly different times, there will be phase cancellation problems. Note how different the bottom waveform, which was created by summing the top and middle waveforms, is from the original two waveforms—this is caused by the middle waveform being very slightly out of phase with the top waveform.

overheads. One, two, or more microphones positioned over a drum kit. Overheads are intended to capture cymbals as well as a complete picture of the drum kit along with a certain amount of room ambience. Overheads are also often used with small ensembles, such as string quartets.

overload. A condition in which the maximum level a device can handle has been exceeded. Overload usually results in distortion. ☞ **See also** *clipping*.

oversampling. A sampling and signal-processing technique in which the sample rate is multiplied to a very high rate (anywhere from 4× to 256× or even higher). The biggest reason to oversample is to make it easier to design anti-aliasing filters. Once sampled, the signal is digitally filtered and downsampled to the desired rate. In some cases, certain types of noise may also be reduced.

overtone. Tones within a sound produced by a source that are higher than, and accompany, the fundamental tone. Overtones may or may not be part of the harmonic series of the fundamental, depending on their mathematical relationship with the fundamental. (Overtones must be integer multiples of the fundamental to qualify as harmonics.)

overview. A window or view in a DAW, sequencer, or audio program that displays the entire track or project. This is useful for finding a location within a track.

overwrite. To store new data that erases old data. In destructive recording and editing, new files are created that replace existing files. ☞ **See also** *destructive editing, destructive recording*.

Oxford 911 Bridge (a.k.a. OXFW911). A high-performance FireWire "bridge" chip that allows an IDE drive to communicate with a FireWire 400 bus at high speeds. The Oxford chip is part of the electronics in an external FireWire drive enclosure.

Oxford 922 Bridge (a.k.a. OXFW922). A high-performance FireWire "bridge" chip for use with a FireWire 800 bus. The Oxford chip is part of the electronics in an external FireWire drive enclosure.

oxide. An oxide is a compound created when a metal oxidizes or combines with oxygen—in common language, it rusts. Different formulations of oxide are used as magnetic materials to coat recording tapes and hard disk platters.

P

Pa. ☞ See *pascal*.

PACE. A company (PACE Anti-Piracy) that makes copy protection software and hardware, such as the iLok, used by some music and audio companies.

pad. 1. A control found on microphones, mixers, preamps, and other gear that attenuates the level of the incoming signal so it won't overload subsequent circuitry. 2. A background, string-like, sustaining sound, often produced by a synthesizer, that fills in an arrangement with chords or washes of sound.

page. 1. A screen or window full of related commands, parameters, and functions in a software program or LCD display of a piece of hardware. There might be a page (screen or window) of hardware input and output parameters, another page of digital sample rate and resolution parameters, a page of internal bus routing options, and so on. 2. A contiguous, fixed-length block of computer memory. The computer's processor architecture determines the page size, often 4,096 bytes. The operating system allocates memory to programs in pages. ☞ **See also** *paging*.

paging (a.k.a. swapping). Transferring pages from main memory to a hard drive or other storage device. Paging allows a computer to use hard disk space to hold files or data that are too large to fit in RAM or to use a hard drive or other storage device to temporarily hold data when RAM is full.

palette. A small graphic window in a software program that provides quick access to commonly used commands or tools. See Figure P.1.

pan. A term that comes from the film world, where a camera is turned to capture the panorama of a wide background scene. In audio, a pan control is used to position a sound or track in the left/right stereo field. A surround panner takes this to the next level, positioning the sound in a surround or multichannel field.

pancake. A length of analog recording tape wound on a tape hub, but without the reel flanges (sides of the reel). Pancakes of tape were more economical for studios to purchase than complete reels.

panel trap. ☞ See *membrane trap*.

panic button. A command or button found in most sequencers as well as in some hardware devices, such as MIDI patch bays or interfaces, that is intended to stop stuck notes. A panic button may send a MIDI all-notes-off command or it may send a note off message for each note for every MIDI channel supported by a device.

panner. ☞ See *panoramic potentiometer*.

panoramic potentiometer (a.k.a. pan pot). A control that varies the balance of a signal's level in two or more output channels. ☞ **See also** *pan*.

pan pot. ☞ See *panoramic potentiometer*.

parabolic reflector. Think satellite dish…a parabolic reflector is a dish-shaped device used with directional microphones for picking up distant sounds. The reflector collects and focuses the sound energy onto the microphone, which faces backward, into

Figure P.1 A palette is a small window that provides quick access to commands or tools.

the dish. Parabolic reflectors are commonly seen on the sidelines at football games and other sporting events and are used for capturing wildlife and other sounds where it is difficult to place a mic close to the source.

paragraphic equalizer. An equalizer that combines aspects of parametric and graphic EQs. Usually this means using sliders for parameters such as boost/cut, while also allowing control over bandwidth and frequency settings with rotary controls. Although hardware paragraphic EQs exist, paragraphic software plug-in EQs are far more common.

parallel. Items, events, or processes that occur simultaneously. In audio, an example of parallel might be splitting a signal, then sending it simultaneously through two (or more) processors. In computers, parallel might be multiple simultaneous operations or bits of data sent side by side simultaneously down multiple wires or paths. SCSI is an example of a parallel computer protocol. ☞ **See also** *series*.

parallel port. A type of connection port used on Windows PC computers for connecting hard disks, CD-ROM drives, printers, and other peripherals. Parallel ports are faster than serial ports, which send data sequentially, as multiple pieces of data can be sent simultaneously.

parallel processing. Splitting a signal and routing it through multiple channels and/or processors simultaneously, then recombining the signal. See Figure P.2.

parameter. A factor that defines the conditions of operation for a piece of gear or software or a system. Basically, this means any of the settings that the user makes when operating a device or program.

parametric equalizer. A type of audio equalizer invented by producer/engineer George Massenburg that features separate controls for gain, bandwidth, and frequency for each EQ band. See Figure P.3.

parity. Literally, equality. In computers, parity is a technique for ensuring that data has arrived completely and unaltered when moved or transmitted. In one scheme, the number of bits is counted in a group of data. If the number is even, a parity bit is set to on; if the number is odd, the parity bit is set to off. The computer counts the bits again after transfer, and if the count matches the parity bit for odd or even status, then all is assumed to be well. There are more complex forms of parity, some of which can be used to reconstruct missing or damaged data.

Figure P.2 Parallel processing can be used to make a kick drum sound bigger. In this example, the kick signal is simultaneously being routed through a dry (unprocessed) mixer channel and through a compressor on another mixer channel. The dry and compressed channel outputs are combined, and the levels of the two signals are balanced to achieve the desired result.

part. In a multitimbral keyboard, sampler, or synthesizer, a part is a virtual "slot" that can be assigned a sound and a MIDI channel, and that functions as an independent sound module. The sound generator is

Figure P.3 A parametric EQ allows control over the frequency, boost/cut, and bandwidth of each band.

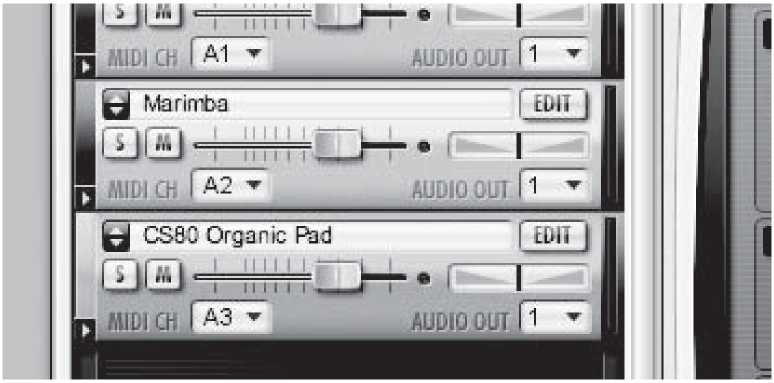

Figure P.4 Each part in a multitimbral synthesizer can be set to produce a separate sound, on its own MIDI channel, with its own volume, pan, and other parameters.

split into multiple independent parts (8, 16, 32, or even more), which can be addressed either from an internal sequencer or using MIDI. (See Figure P.4.) The number of parts is not related to the unit's polyphony, which is the number of notes a sound generator can produce simultaneously. ☞ **See also** *multitimbral.*

partial. ☞ **See** *overtone.*

partition. 1. To divide a hard drive into separate parts. 2. A subsection of a hard drive that has been configured to function as a separate, independent, virtual drive. Partitions appear as separate drives to their host computer. Partitions are usually used to organize data. One partition might hold audio recordings, another might contain sample libraries, a third digital video, and so on. In other cases, partitions are set up to be accessed by different devices.

pascal. The unit used when measuring pressure, or force per area. In audio and sound terms, one pascal (1 Pa) is 94 dB SPL. The pascal is used as a reference when measuring microphone sensitivity and signal-to-noise ratio. For example, the sensitivity rating for a microphone might be –36 dB referenced to 1 V/Pa. Named for Blaise Pascal, a 17th-century mathematician/physicist/ philosopher.

pass. A single attempt at recording a track. A number of passes or attempts may be required for the musician to perform his part correctly.

passband. The range of frequencies that a filter allows to pass through unattenuated (see Figure P.5). The limit of the passband is determined by the filter's 3 dB down points—the frequencies where the

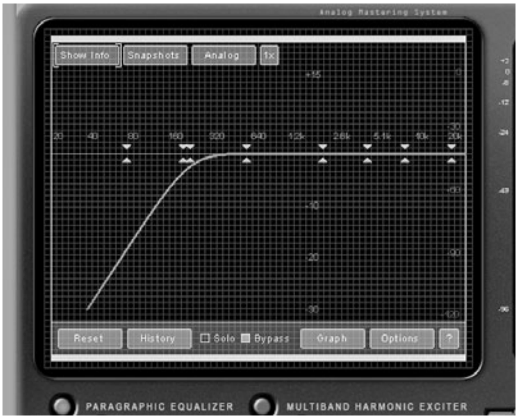

Figure P.5 The range of frequencies in a signal not affected by a filter is called the passband.

response has been attenuated by 3 dB. ☞ **See also** *stopband*.

passive. A device that does not contain any active, or powered, components.

passive monitor. A type of studio speaker that requires an external amplifier and that does not contain any active, or powered, circuitry.

passive radiator. A speaker driver that is not connected to an amplifier, but that moves in response to sound energy inside the speaker's cabinet. In most cases, a passive radiator resembles a low-frequency driver and moves sympathetically with the motion of a low-frequency driver in the same cabinet powered by the amplifier. Passive radiators are used to lower the speaker's resonant frequency and to increase low-frequency response.

password. An alphanumeric code that allows the user to access a piece of software, a computer, a website, or functions of a device.

paste. A command that inserts data from a clipboard or other temporary holding space into an active document, session, track, or file at a selected location.

PATA. Parallel ATA. After SATA (*Serial ATA*) was introduced, "parallel" was added to the Advanced Technology Attachment name to distinguish the two. ☞ **See** *ATA*.

patch. 1. A physical signal connection between two electronic devices. 2. The physical audio signal and control voltage cable connections required on an analog modular synthesizer to create a particular sound. 3. The parameters, configurations, algorithms, samples, and other data required to create a sound in a digital synthesizer. Most digital synths (and many newer analog synths) can store and recall patches (a.k.a. programs) to instantly recall a sound during a performance. 4. A temporary repair. 5. Also known as *software patch*. A piece of software intended to fix problems, add capabilities, or update a computer program.

patch bay. A collection of jacks usually mounted in a rack-mountable frame. Patch bays were first used by the telephone company. The operator would manually patch telephone lines together using banks of jacks that were connected using short cables. In audio, the idea is to bring the inputs and outputs from the gear in a studio or rig to the back of the patch bay. (Not all the connections need to be brought to a bay; only those that regularly are re-patched need to be run through a patch bay.) Then short cables can be used on the front of the bay to connect the inputs to the outputs as needed. Patch bays are basically convenience items—it's easier to re-patch gear using short patch cables on the front of the bay than it is to climb behind gear racks and change the connections. There are a number of different possible patch-bay configurations: Bays can be balanced or unbalanced; can use TT, 1/4-inch, or other connector types; and there are several different types of normalling that determine how the jack points connect inside the bay, some of which automatically connect the top row of jacks to the bottom row. ☞ **See also** *normal*, *open*, *mult*.

patch change. ☞ **See** *program change*.

patch cord. A short cable used to connect patch points on a patch bay. The term has come to be used for almost any cable used to connect two pieces of gear together.

patch dump. ☞ **See** *Sys Ex dump*.

patch list. A list of the names for the presets or programs in a MIDI device stored as a text file that can be loaded into a sequencer. Once the patch list is in the sequencer, the user can call up programs in the device using the names instead of patch numbers.

patch mapping. A table or function in some MIDI devices that can reassign an incoming program change message to call up a different program change number. For example, when the device sees an incoming message with program change #1, it might map the program change to instead call up program change #33. Incoming program change #2 might call up program change #67. Incoming program change #3 might call up program change #16, and so on, throughout the 128 available MIDI program change numbers. There are several common applications for patch mapping. Patch mapping allows a performer or sequencer to call up the programs in a device in a certain order without having to physically re-save the programs in that order. Patch mapping also allows a single program change message to simultaneously call up different programs in devices that are layered together. Patch mapping can also be used to allow MIDI program change messages to access program numbers in a device that are higher than the MIDI spec allows—for example, an incoming MIDI program change #93 might call up patch #287 in a synthesizer with more than 128 internal presets.

patch point. A jack or connection on a patch bay.

path. ☞ See *file path*, *signal path*.

pattern. A series of notes or control changes that is repeatedly played in looping fashion.

PCB. Printed Circuit Board. A flat insulating sheet that is coated on one side with a conductive layer, such as copper. After the circuit is designed, the unnecessary copper coating is etched away, leaving thin traces on the board surface that are use to connect electronic components together.

PC card (a.k.a. PCMCIA card). A type of expansion bus that uses removable cards that slide into an external slot in a Macintosh or Windows computer. Originally designed to add more RAM or ROM to a portable computer, PC cards now offer a variety of capabilities, such as fax/modems, disk drives, networking, and audio interfacing. There are three types, all with the same width and length but with different thicknesses: Type I (3.3-mm thick), Type II (up to 5.5-mm thick), and Type III (up to 10.5-mm thick). A Type II slot can hold one Type II card or two Type I cards. A Type III slot can hold one Type III card or one Type I and one Type II card.

PCI. Peripheral Component Interconnect. A type of computer expansion slot developed by Intel and used in both Windows and Macintosh computers. PCI allows for high-speed, processor-independent communication between the CPU and peripherals. Data transfers up to 132 MB/second are supported. One 64-bit bus runs at a clock speed of 66 MHz, while additional buses are 32-bit at a clock speed of 66 MHz or 64-bit at 33 MHz. The PCI spec allows for two different expansion card lengths, 312 mm and 119 to 167 mm.

PCI Express. ☞ See *PCIe*.

PCIe (a.k.a. PCI-E, PCI Express, Peripheral Component Interconnect Express). An updated version of PCI that addresses many of the shortcomings of the older PCI standard. The original PCI was a shared-bandwidth standard, where devices divided the available data bandwidth as required, which could cause problems if one device used up too much bandwidth. PCIe uses separate bidirectional serial point-to-point data "lanes" to overcome this limitation. The throughput of PCIe has also been increased substantially. There are several PCIe levels, including PCIe 1× (250 MB/s), PCIe 2× (500 MB/s), PCIe 4× (1,000 MB/s), PCIe 8× (2,000 MB/s), PCIe 16× (4,000 MB/s), and PCIe 32× (8,000 MB/s). Because of these high speeds, PCIe has effectively replaced several other internal bus types, including AGP and older versions of PCI. Though the PCIe expansion slot format is physically different from PCI, the two types are software-compatible, making the transition from PCI to PCIe easier for most manufacturers.

PCI-X. Peripheral Component Interconnect Extended. A higher-speed version of PCI that supports one 64-bit bus at a clock speed of 133 MHz, while additional buses are 64-bit at a clock speed of 66 MHz. PCI-X expansion cards are backward-compatible with PCI, though they will operate at 33 MHz speed. PCI and PCI-X cards can also be combined in one computer, but the speed of the slowest card will determine the speed of the rest of the cards.

PCM (a.k.a. pulse code modulation). A family of methods for encoding sampled analog signals into digital form using a series of pulses, and decoding the pulses back to analog signals. PCM data consists of discrete samples of the amplitude of an analog signal, which are represented and encoded using a specific number of bits.

PCMCIA. Personal Computer Memory Card International Association. An agency that sets standards for integrated circuit cards. The PCMCIA is responsible for the PC card, CardBus, and ExpressCard formats. www.pcmcia.org.

PCMCIA card. ☞ See *PC card*.

PDF. Portable Document Format, a.k.a. ISO 32000. A cross-platform file format developed by Adobe Systems for storing text and graphics documents. Many manufacturers now provide documentation for their products as PDF files, saving the expense and paper required for printed documentation. A "reader" program, such as Adobe Reader or Apple Preview, is required to view a PDF file. The PDF format has been adopted as the ISO 32000 standard.

PD synthesis. ☞ See *Phase Distortion Synthesis*.

peak. 1. An area where cancellation and reinforcement of sound waves in a room results in a boost in level at a particular frequency. 2. The maximum amplitude or voltage in a signal. Peaks may occur so quickly that they are essentially inaudible, yet they still consume headroom.

peak hold. A type of non-mechanical (LED or LCD) meter that has an indicator that displays the highest peak signal for a specified period of time or until a higher signal peak occurs. This allows the engineer to see where peaks occur and to see the average level.

peak indicator. A meter or indicator that is intended to show when the highest peaks in a signal reach a certain level.

peaking filter. A type of filter that boosts or cuts a range of frequencies around a center frequency. Peaking filters are found in graphic equalizers, as the midrange bands in parametric and semi-parametric equalizers, in synthesizers, and more. Controls for peaking filters can range from one (boost/cut amount), to two (boost/cut amount and center frequency adjust), to three (boost/cut amount, center frequency adjust, and bandwidth). See Figure P.6.

peak LED. An LED indicator that is intended to light when the highest peaks in a signal reach a certain level.

peak level. The highest level in a file or signal.

peak limiting. A limiter set up to reduce the highest peaks in a signal so that the average level can be raised. ☞ **See also** *limiter*.

peak program meter (a.k.a. PPM). A type of meter optimized for displaying peak levels. Peak program meters are useful for preventing overload distortion, especially in digital systems.

peak SPL. The maximum volume level a sound source or speaker can produce.

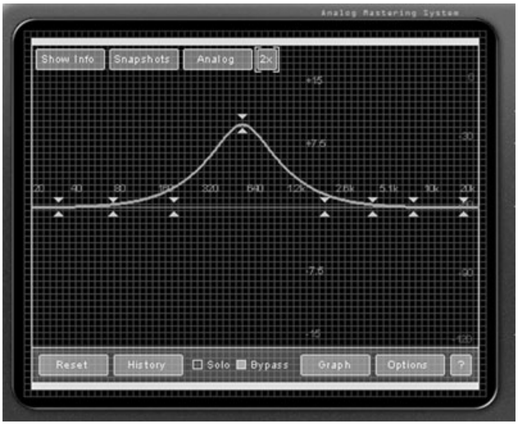

Figure P.6 A peaking filter boosts a range of frequencies around a center frequency.

pedal. 1. A music composition technique in which a note (typically a bass note) is held or repeated underneath changing chords. 2. A floor-based effects processor. ☞ **See** *stompbox*. 3. A foot-operated switch or treadle pedal that is used to send control messages to another device. Examples include sustain pedal, control voltage pedal, volume pedal, and more.

Pencil tool. A mouse tool used in DAW and audio editing programs to "draw" data or edit waveforms. So named because the mouse pointer changes to look like a pencil when the Pencil tool is selected.

perceptual coding. A type of lossy audio data compression method that removes frequencies that are determined to be inaudible because they are "masked" by other audio material.

period. The time it takes for a full cycle of a waveform to occur. The period can be deduced from the frequency of the waveform. For example, a 50-Hz wave goes through a full cycle 50 times per second, so its period is 1/50 of a second.

PFL (a.k.a. Pre-Fade Listen). A mixer function similar to solo that mutes all the channels except the selected channels without disturbing any other mixer settings. The difference is that solo derives the signal after the channel fader/volume control, whereas PFL is not affected by the channel fader. PFL is used to monitor and meter the signal for initial gain setting. Typically, PFL only affects the monitor/headphone bus, so it does not affect the signals feeding any other outputs.

phantom center. A function of some surround playback systems that routes the center channel information to the front left and right speakers, allowing, for example, a 4-channel speaker system to reproduce the fifth- or center-channel audio in a 5.1 signal.

phantom image. A "centered" image perceived between two loudspeakers. For example, in a stereo system, a signal can be panned to the center and clearly positioned between the two speakers even though there is no true center speaker to produce that signal. In a surround system, phantom images can be created between any pair of speakers.

phantom power. A system for delivering voltage, usually 48V DC, "in the background," over a standard balanced microphone cable. The current is carried in balanced fashion over pins 2 and 3 of the cable's XLR connector, while pin 1 serves as ground. Because the phantom power is balanced on the two pins, it cancels out and is invisible to any devices that don't need it. Phantom power is used to power condenser microphones and microphones containing active internal circuitry, as well as some active direct boxes, without the need for an external power supply.

phase. The current position in the periodic cycle of a waveform, given in degrees. Zero degrees is the start of the cycle, while 360 degrees is the end of the cycle. (See Figure P.7.) ☞ **See also** *phase cancellation*.

phase cancellation. Destructive interaction of two identical out-of-phase sound waves. Phase cancellation results in a reduction or increase in level at a particular frequency due to the two sound waves reinforcing or interfering with one another. If waves are out of phase, the phase of their cycles doesn't line up exactly, and cancellation may occur if they are mixed, resulting in what is often described as a "hollow" sound. How much cancellation occurs depends on how far out of phase the two waves are; 180 degrees is completely inverse phase and results in 100-percent cancellation. Conversely, 0 degrees and 360 degrees are completely in phase, resulting in the waves summing and reinforcing one another. See Figure P.8.

phase correlation. A comparison of the phase relationship of two or more signals.

phase correlation meter. A meter that displays the phase relationship between two or more signals. ☞ **See also** *jellyfish meter*.

phase distortion. Changing the phase relationship of the component frequencies within a waveform.

phase distortion synthesis (a.k.a. PD synthesis). A digital synthesis method similar to frequency modulation. Phase distortion was developed by Casio for the CZ family of synthesizers. In phase distortion synthesis, a sine wave, the oscillator's raw waveform, is modified or distorted by changing the phase angle using a distortion algorithm, creating new timbres.

phase invert. ☞ **See** *polarity invert*.

phase lock. A type of synchronization that works by comparing and locking together the zero phase point of SMPTE waveforms. Phase lock can be very accurate, but under certain circumstances it can drift.

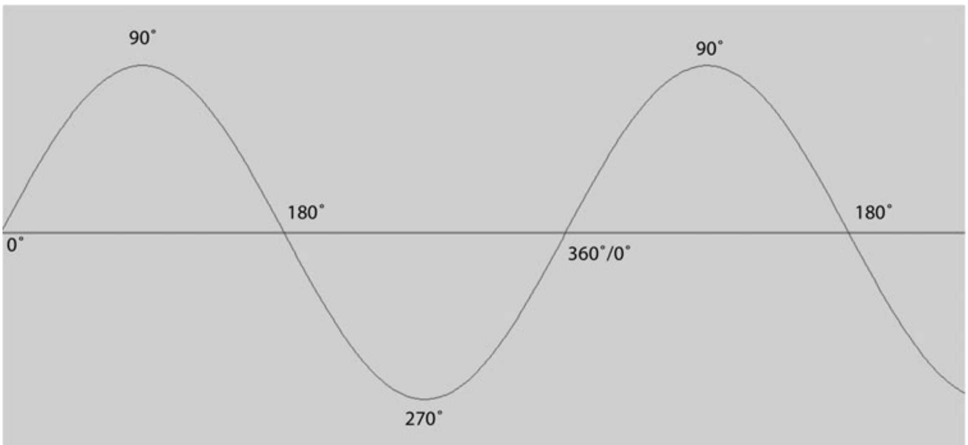

Figure P.7 A wave's phase is a point in its cycle in relation to time.

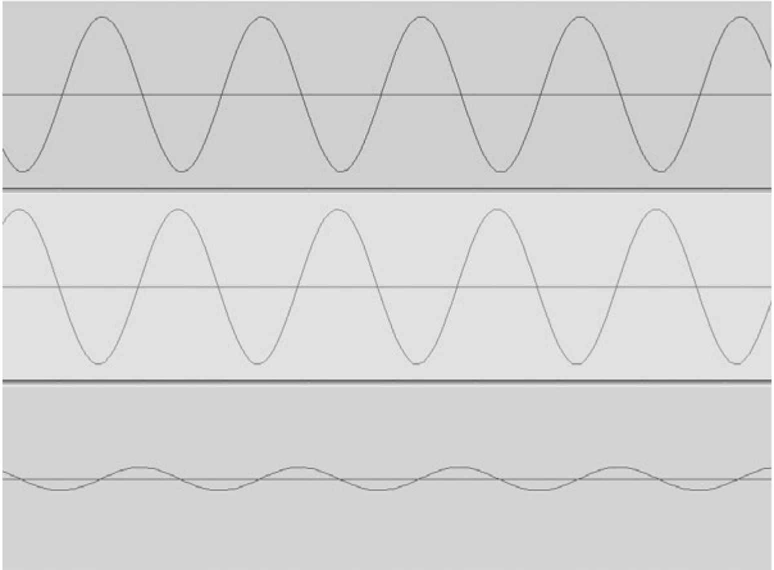

Figure P.8 Two waves that are out of phase will cancel each other to some degree. If they're 180 degrees out of phase, they will cancel completely.

phase-locked loop (a.k.a. PLL). A circuit that compares and locks the phase of an internal oscillator with an input signal. Phase-locked loops are used in wireless and broadcast devices and in various types of clocks, as well as to generate stable signals.

phase modulation. A type of modulation in which the phase of a waveform changes in response to another signal or control. Phase modulation is used in various communication and encoding applications.

phase quadrature. A fancy term for two signals being 90 degrees out of phase. Some effects devices offer a mode in which two LFOs are placed 90 degrees out of phase to enhance a stereo phase shift or chorus effect.

phaser. ☞ See *phase shifter*.

phase reverse. ☞ See *polarity invert*.

phase shifter. An audio effects processor that mixes together a dry signal and a slightly delayed version of that signal to create audible phase cancellations. The delay time is modulated, sweeping the frequencies that are cancelled to create a swooshing, swooping, comb-filtering effect in the output signal. Most phase shifters allow some of the signal to feed back or "regenerate" into the mix to add resonance to the sound.

phase-accurate. The ability of a device to synchronize or operate with phase-level accuracy.

phon. A unit of measurement for loudness. One phon equals 1 dB SPL at 1 kHz. Phons are used to compensate for the varying sensitivity of the human ear at different frequencies.

phon scale. A rarely used logarithmic scale for measuring loudness in which 0 phon equals the threshold of human hearing and 120 phon equals the threshold of pain. ☞ **See also** *phon*.

phone plug. The standard 1/4-inch connector used for many audio and musical signal-carrying applications. Phone plugs were originally developed for telephone switchboards. There are two types: TS, or tip-sleeve, which has two conductors for carrying an unbalanced mono signal; and TRS, or tip-ring-sleeve, which has three conductors that can be used to carry one balanced signal or two unbalanced signals. The "military" version of the phone plug is slightly longer than the standard audio phone plug.

phono plug. ☞ See *RCA plug*.

phono preamp. Short for phonograph preamp. A specialized preamp intended to raise the very weak output signal coming from a turntable cartridge. Most phono preamps also apply a special RIAA equalization curve to the signal, a standard developed for the playback of LPs. The RIAA system uses pre-emphasis on mastering and de-emphasis on playback to manage low-frequency content and high-frequency noise on vinyl records. ☞ **See also** *RIAA equalization*.

physical modeling. Using mathematical equations and algorithms to re-create a particular device, process, or acoustical space. ☞ **See also** *physical modeling synthesis*.

physical modeling synthesis. A type of digital synthesis that uses detailed and complex mathematical algorithms (models) to re-create the properties and qualities of instruments. Physical modeling can create realistic imitations of acoustic sounds, limited only by the accuracy and flexibility of the equations and algorithms used. Generally, physical modeling requires fairly serious computing power for good results, though it does not require much memory or hard drive space compared to using samples.

piano roll. A type of graphic MIDI sequence editor that visually resembles a player piano roll. The sequencer displays the MIDI notes as bars on a grid, which scrolls as the piece plays. The bars may be stretched using the mouse to shorten or lengthen their duration, or they can be moved using the mouse to change the pitches or the rhythmic positions of the notes. Notes can be cut, copied, and pasted, or entered with a Pencil tool. See Figure P.9.

pickup. A type of transducer that converts physical vibrations into electrical signals.

pickup pattern. ☞ See *polar pattern*.

piezo. Short for piezoelectric. A property of some materials, such as quartz and other crystals, that produces electricity when subjected to stress. Piezoelectric materials are used in a variety of transducers, such as microphones, phono cartridges, high-frequency drivers for speakers, guitar pickups, and more. Piezo crystals also work in reverse—when a voltage is applied, the crystal will vibrate at a certain rate. The rate is predictable based on the size and shape of the crystal and is extremely stable, making piezos useful for digital clocks.

piezo tweeter. A type of high-frequency driver composed of a piezoelectric crystal that moves a diaphragm to create sound waves. Piezo tweeters are inexpensive and rugged, but they may not be as accurate or linear in their response as other types of high-frequency drivers.

pigtail. 1. An audio cable with bare wires instead of connectors. 2. A breakout cable or breakout box

Figure P.9 The piano roll editor in a sequencer is a graphic editor in which pitch, rhythmic placement, and note length can be edited, and notes can be inserted or deleted.

that splits the multiple wires inside a snake cable into separate connections.

pinch roller. A rubber roller in a tape deck that presses the tape against the capstan.

ping-pong delay. A delay or echo effect in which the repeats are panned alternately left and right so that the echoes seem to bounce back and forth across the stereo field.

pink noise. A type of random noise signal used for testing purposes and containing equal energy for each octave. Pink noise sounds "bassy" and muffled to our ears. Because of how the energy is distributed in pink noise, it is useful for measuring the frequency response of audio devices as well as rooms and other spaces.

pitch. Musical quality defined by the frequency of a sound wave.

pitch bend. 1. Controlled changing in the pitch an instrument is producing. Some instruments can easily bend the pitch of notes, such as guitars, violins and other stringed instruments, trombones, and others. Other instruments produce only fixed pitches, such as pianos, harps, marimbas, and so on. 2. A wheel or other controller on a synthesizer or sampler that changes the pitch of notes played on the instrument. 3. A standardized MIDI message that changes the pitch of notes. The MIDI pitch bend message is its own type of message, not one of the continuous controllers, because smooth pitch bends require more resolution than controller messages can provide. Continuous controller messages have a resolution of 128 steps; pitch bend messages use two data bytes to increase this to 4,096 steps.

pitch correction. Using a process to "tune" the notes in a performance. This consists of modifying the pitch of a signal or file using a process that does not affect the length of the notes. There are a variety of techniques for doing this. In some cases, the pitch is automatically moved to the nearest half step or scale degree in a process analogous to rhythmic quantization. In other cases, the pitch can be corrected manually or graphically. Some pitch-correction algorithms can maintain the formants of the original sound. Others can be set to allow a certain amount of the original pitch variance (such as slight pitch bends or vibrato) to preserve the "human" aspects of the performance.

pitch envelope. An envelope generator that is routed to control the pitch or frequency parameter of an oscillator in a synthesizer or sampler.

pitch shift. A process or device that changes the pitch or frequency of an audio signal or file. Pitch shift can range from a few cents for chorusing-type effects, to an octave or even more. Some pitch shifters can change the pitch of a note without changing its length. ☞ **See also** *pitch transpose*.

pitch-to-MIDI. A device or process that converts a signal's pitch or frequency to a MIDI note message. Pitch-to-MIDI devices and programs vary in their effectiveness; some are more accurate than others. There is also typically a slight lag or latency, as a full cycle of a waveform, string, or other source must be detected in order for the pitch to be converted to MIDI. This latency is more significant with low frequencies than it is with high frequencies.

pitch transpose. A process that changes the pitch of an audio signal or file, usually by a musical or scalar interval. Some modern pitch transposers are able to do this without changing the length of notes or affecting the formants or harmonic spectrum relationships of the original signal.

pitch transposer. An audio device or plug-in that manipulates or shifts the pitch of incoming signals. ☞ **See also** *pitch transpose*.

plate. The anode or positive component in a vacuum tube. When positive voltage is applied to the plate, electrons are attracted to it from the cathode.

plate reverb. One of the first types of mechanical reverb devices developed for recording studios. An audio signal is sent into one end of a steel plate that is suspended and tensioned by springs. The signal is picked up elsewhere on the plate by a transducer. In addition to the signal transmitted straight through the plate, some signal is reflected and bounces around. The reflections in the plate create a wash of sound, which serves as reverb. Plate reverbs were popular until the advent of digital reverbs, though virtually all digital reverbs offer a preset that emulates the sound of a plate reverb.

platter. A disk coated with magnetic oxide material within a hard drive that actually stores the data. A hard drive may contain one or more platters, depending on its capacity.

play head (a.k.a. playhead). The tape head that reads the information on a magnetic tape and converts it

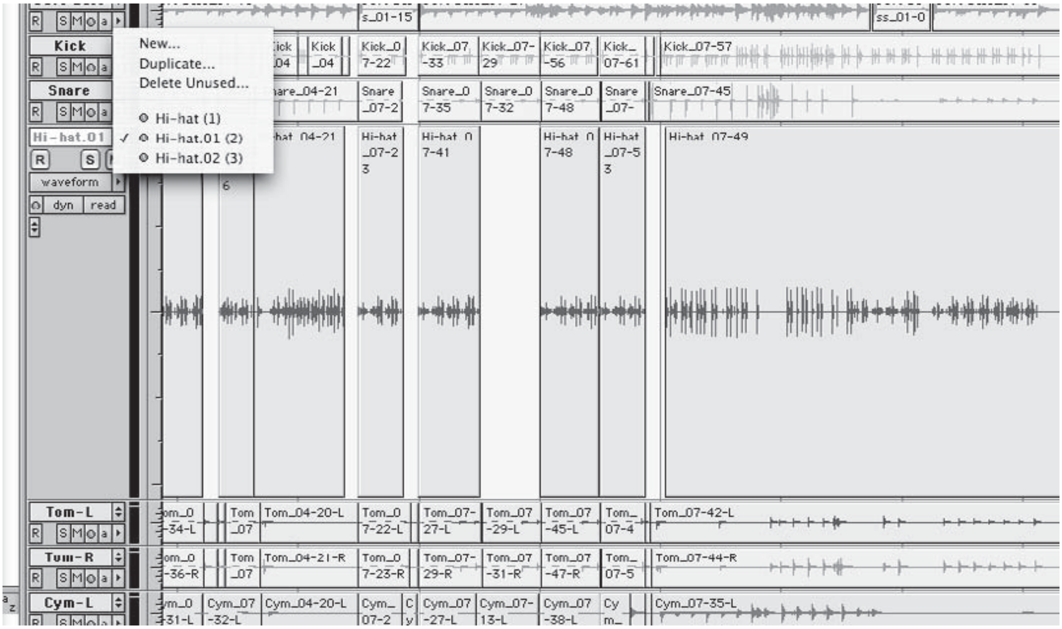

Figure P.10 A play list contains an arrangement of audio files.

to an audio signal. Most DAWs feature a line or cursor that represents a virtual play head, indicating the location in the track. ☞ **See also** *tapehead*.

play list (a.k.a. playlist). A list of files or a graphic arrangement of items that determines the order in which they will play. Play lists can be used for a variety of purposes, including trying different arrangements of audio segments or regions in a track. See Figure P.10.

PLL. ☞ **See** *phase-locked loop*.

plosive. A human vocal sound, such as "p" or "b," that creates a burst of air. When a burst of air from a plosive hits a microphone diaphragm, it creates a loud pop or a bassy thump. There are several techniques for avoiding plosive problems, such as angling the microphone, using a pop filter, or placing a pencil between the singer or speaker and the microphone.

plug and play. A computer feature that allows peripheral devices to work without requiring configuration or installation of a driver or other software.

plug-in. A piece of software that requires a host program to operate and that extends the capabilities of the host. Audio/music plug-ins fall into three very broad categories: audio processors and effects, virtual (software) instruments, and MIDI data processors and effects.

pocket. That place in relation to the beat where an instrumental performance sits most comfortably in the rhythmic flow of a piece or "feels" the best.

pocketing. The process of editing a track to make it fit the rhythmic feel, or pocket, of the rest of the song or performance.

podcast. A method for delivering audio and video over the Internet. A podcast consists of a digital-media file that has been made available via an RSS-syndicated feed or download.

point source monitor. A speaker in which the sound radiates from one point. In most loudspeakers, drivers producing different frequency ranges are distributed across the baffle, resulting in different frequencies coming from slightly different locations. A point source monitor uses either a single, full-range driver or a coaxial/concentric driver in which the high-frequency driver is mounted in the center of the low-frequency driver.

point-to-point wiring. A method of connecting an electronic circuit in which the components are wired directly together rather than mounted to conductive traces on a printed circuit board.

polarity invert. (a.k.a. phase reverse, phase invert, though these are technically incorrect because polarity invert has nothing to do with the phase of a signal in time.) Polarity invert is a function in mixers and DAWs that flips a signal's polarity from positive to negative and vice versa. Polarity invert is used to verify and correct the relationship between two signals, such as two microphones on a drum kit, direct and miked tracks on a bass guitar amp, and so on. Since polarity invert only flips the polarity by 180 degrees and does not allow any other degree of phase adjustment, it will not solve all phase-cancellation problems. However, if two signals are even partially out of phase, typically flipping the polarity one way or the other on one of the signals will result in better sound.

polar pattern (a.k.a. pickup pattern). The sensitivity of a microphone to sound coming from different directions. A graph of this response is called its *polar pattern*. There are a number of different types of polar patterns:

- **cardioid.** A heart-shaped polar pattern that picks up well from the front and rejects sound from the rear.
- **figure-8.** Sound is picked up well from the front and rear, but not from the sides.
- **hypercardioid.** Similar to cardioid, but with rejection at two points—around 150 degrees and 210 degrees from the front.
- **omnidirectional.** Sound is picked up equally in all directions.
- **supercardioid.** Similar to hypercardioid, with more rejection from the rear.
- **wide cardioid.** A pattern that falls between omni and cardioid in its directionality.

pole. In analog audio, a pole is an RC (resistance and capacitance) circuit that contains one capacitor and one resistor. A one-pole circuit functions as a high-pass or low-pass filter with roughly 6 dB of attenuation per octave. Each additional pole in a filter adds another 6 dB of attenuation per octave (12 dB/octave for a two-pole filter, 18 dB/octave for a three-pole filter, and so on). Digital filters emulate this response using mathematical algorithms.

polyphonic. The ability of an instrument to play more than one note simultaneously. ☞ **See also** *polyphony*.

polyphonic aftertouch (a.k.a. polyphonic pressure). Aftertouch refers to pressure changes applied to a keyboard's keys after the notes are struck, while the notes are sustaining. With polyphonic aftertouch, each key on a keyboard can produce its own aftertouch value. Polyphonic aftertouch is carried as a MIDI channel message and has a value ranging from 0 (no aftertouch) to 127 (full aftertouch). Polyphonic aftertouch can be an extremely expressive controller, though not many keyboards are capable of generating it. ☞ **See also** *aftertouch*.

polyphony. The number of notes an instrument can sound at once. For example, a standard guitar can produce a maximum of six notes simultaneously, one per string, An upright bass, cello, viola, or violin can each produce four notes simultaneously. Most grand pianos can produce 88 notes of polyphony. Synthesizers and samplers will also have a set polyphony, ranging from a single note to hundreds of notes. Some software synthesizers are said to have unlimited polyphony, but there is a limit somewhere, set by the performance of the computer system running the synth. ☞ **See also** *note stealing*.

pop. The bassy thump or pop that results from a microphone being hit with a plosive.

pop filter. A screen made from mesh or other materials that is placed between a vocalist and a microphone to help prevent pops. The pop filter must be acoustically transparent so it doesn't color the vocal sound, but substantial enough to break up or disperse plosive air bursts. While most pop filters are made from mesh material (nylon stockings have been used for DIY pop filters), some feature vanes or louvers that are intended to redirect air bursts.

pop-up menu. A menu that only appears when accessed with a command or with a mouse right-click and that opens in a new window. See Figure P.11.

port. An opening in a speaker cabinet intended to increase bass response. ☞ **See also** *bass reflex*.

portable mix. A mix that sounds consistent on a variety of different playback systems. Making a mix portable, or making it translate well from the studio monitor system to consumer speaker systems, requires an accurate monitor system and a well-tuned studio with even frequency response.

portamento (a.k.a. glissando, glide). A function on some keyboards that gradually and smoothly moves the pitch of one note to the pitch of the

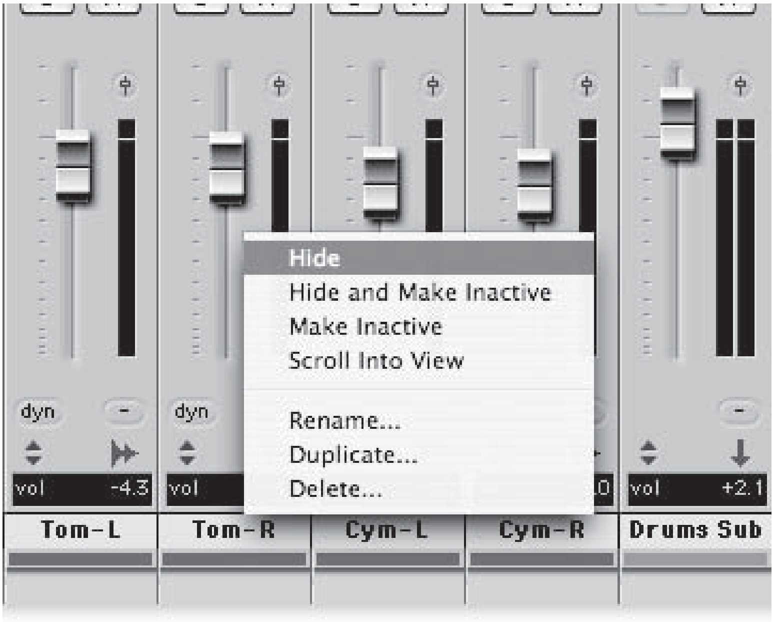

Figure P.11 This pop-up menu was accessed by right-clicking with the mouse on a track name.

next over a user-set period of time, producing a pitch "slide" between notes. Synth portamento was developed to emulate the ability of the human voice and certain instruments, such as violin and trombone, to smoothly slide from pitch to pitch.

ported. ☞ See *bass reflex*.

positional sensing. The ability of some percussion controllers to detect where on the unit's surface a strike occurred. This positional data may be used to trigger an alternate sample or to modulate filter and other parameters to change the sound produced. This allows a more realistic emulation of an acoustic drum or other instrument, in which the sound changes pitch and tone slightly as the instrument is struck in different locations.

post-fader send. An auxiliary bus or send that derives its signal after the channel faders. The levels of signals sent to a post-fader send are affected by the position of the channel fader. Typically, post-fader sends are used as effects sends so that the level of signal sent to the effects follows the level changes created by the faders (and therefore the mix or balance of the dry signal against the effects returns stays constant).

post-production. A term that comes from the film world. In audio, post-production is processing and editing that takes place after the tracking stage of a recording is finished.

pot. ☞ See *potentiometer*.

potentiometer (a.k.a. pot). A variable-resistance electronic component that is used to adjust a voltage level in a circuit. Most potentiometers are rotary controls, though they can also be sliders (usually referred to as faders) or other formats.

power amp. An amplifier intended to drive a loudspeaker. ☞ See **also** *amplifier*.

power compression. Increased voice-coil resistance in a speaker that results from the coil heating up. Power compression can reduce the output of a speaker by as much as 3 to 6 dB. (Remember that losing 3 dB is equal to cutting the power of the amplifier in half.)

power conditioner (a.k.a. line conditioner). An electrical device whose function is to improve the quality of the AC power fed to other devices. Power conditioning includes protecting connected equipment against damage from surges and spikes, and EMI and RFI noise filtering to remove noise from the electrical power.

power cycle. To turn a piece of electrical equipment off and back on. Power cycling can reset memory locations and parameters that are causing instability or crashes. In most digital equipment, power cycling will reboot the software and processors and restore the default or startup state.

power regulator. ☞ See *voltage regulator*.

power spike. A very short peak of high voltage in AC power. Though a power spike may not cause immediate damage, it may weaken components, leading to later failure; it may cause a click or pop; or it may corrupt digital data, among other potential problems. Most modern power supplies are designed to filter out occasional spikes. A power conditioner will also help tame spikes. ☞ **See also** *power surge*.

power supply. A component or circuit in a piece of electrical equipment that converts the incoming AC wall power to the voltage and current the device requires. A power supply may also convert AC voltage to DC voltage as required by the device it is powering. The quality of the power supply is critical to the performance of most electronic equipment.

power surge. A brief increase in the voltage coming over an AC power line. Surges may have a duration anywhere from 15 milliseconds or so up to minutes long, and may be 10 to 35 percent over the normal line voltage. Most power supplies can deal with a certain amount of power surge, but very sensitive devices may have difficulty with strong surges. ☞ **See also** *power spike*.

powered monitor. Though sometimes used as a synonym for *active monitor*, powered monitor implies a speaker that contains a power amp, without any other electronics or crossover circuitry. ☞ **See also** *active monitor*.

POW-r. 1. Psychoacoustically Optimized Word-Length Reduction. An algorithm designed to optimize sound quality when reducing word length from 20 or 24 bits to 16 bits. 2. The consortium of manufacturers that developed the POW-r algorithm. ☞ **See also** *dither*, *noise shaping*.

PPM. ☞ See *peak program meter*.

p-pop. ☞ See *pop*.

PPQN. Pulses Per Quarter Note. The number of rhythmic divisions per quarter note in a MIDI sequencer, which translates to the rhythmic resolution of the sequencer. Older systems used 24 or 96 PPQN, while modern sequencers can have resolution of 960 PPQN or even higher. The higher the resolution, the finer the rhythmic "feel" the sequencer can accurately record and play back. The slower the tempo, the more critical the rhythmic resolution of the sequencer.

PQ subcode. P and Q are two of a number of types of subcode information contained on an audio compact disc and defined by the Red Book audio CD specification. (Other types include R, S, T, U, V, and W, though these are rarely used.) The P subcode is a track separator flag. The Q subcode contains data such as track number and track time. Most audio CD burning software handles PQ information in the background so the user doesn't have to deal with it, though "pro" mastering and disc creation applications may provide access to creating and editing this subcode information directly.

P-RAM. Parameter Random Access Memory, a.k.a. PRAM. P-RAM is memory that is dedicated to storing the parameters or settings for a device. The exact use varies depending on the device. In a synthesizer, P-RAM might hold preset information. In a computer, it might hold configuration and network information. P-RAM is often battery-backed so that it retains the settings for the device even when the power is turned off.

preamp. Short for preamplifier. Electronic device used to raise the level of a very weak signal so that it can be sent to the main stage of amplification or an additional gain stage. A preamp can be a standalone device (for example, a microphone preamp), or it might be integrated into a device (for example, the preamp in a turntable or condenser microphone). Some preamps are intended to be extremely clean and accurate; others are intended to impart a particular "flavor" or signature characteristic to a signal.

precedence effect. ☞ See *Haas effect*.

pre-delay. A parameter in reverb units that sets the length of time between the initial source signal and the appearance of reverb. Setting the pre-delay time properly is essential for maintaining the clarity of a mix. See Figure P.12.

pre-fade listen. ☞ See *PFL*.

pre-fader send. An auxiliary bus or send that derives its signal before the channel faders. The levels of signals sent to pre-fader send are *not* affected by the position of the channel fader. Typically, pre-fader sends are used as monitor sends so that the engineer can create a mix using the faders without affecting what is heard through the monitor or headphone system.

preference file. A file created by a device's user that defines certain aspects of how the device will operate, specifies default settings for global parameters, and customizes the work environment to how the user prefers to work.

preferences. ☞ See *preference file*.

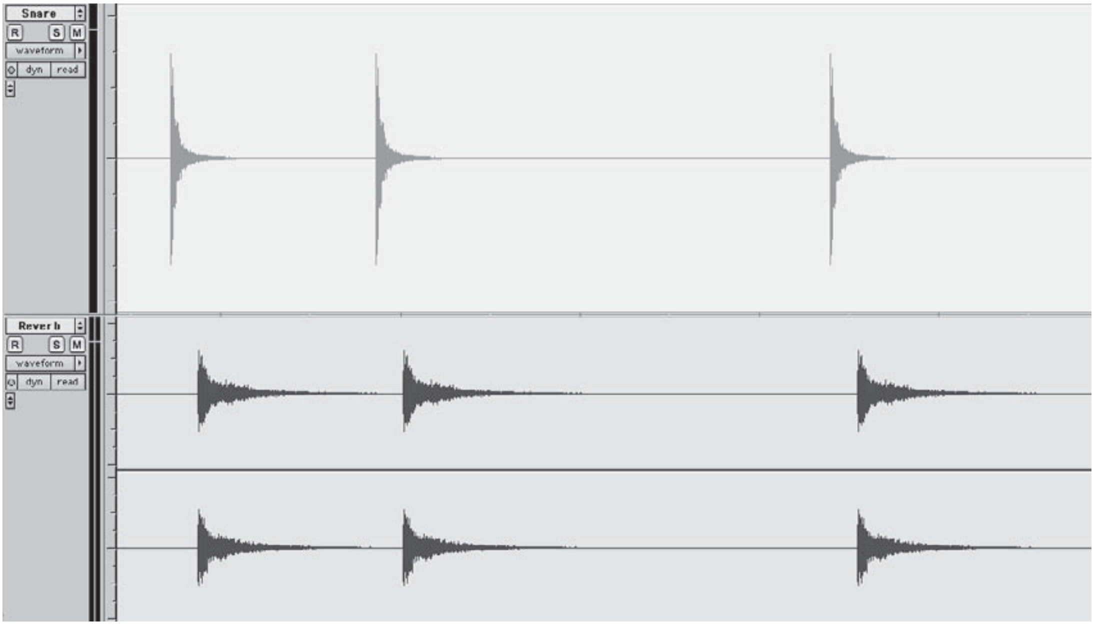

Figure P.12 Pre-delay is the amount of time that elapses after a sound (a snare drum, in this case) and before the onset of reverb.

pre-polarized. A type of condenser microphone capsule that holds a fixed charge and does not require external power. ☞ **See also** *electret*.

pre-production. Work on a piece of music that takes place before any recording sessions take place in the studio. This might include rehearsing, working on arrangements, defining and refining parts, and more. Often the producer of the project will participate and assist with preparing the music for recording.

presence. A frequency range that determines how well a sound will cut through a mix, or how "forward" it will appear for a given volume level. Presence is generally defined to be in the 3- to 4-kHz range. Guitar amplifiers often have a presence control, though any sound can be made more or less present by adjusting this range.

presence peak. A bump or small rise in the upper-mid- and high-frequency response of a microphone. A presence peak will make a signal from the microphone seem louder without increasing the volume, and can add clarity and definition to a signal. Many microphones aimed at vocal use have a presence peak to bring the voice forward and to increase intelligibility.

preset (a.k.a. program, patch). The parameter settings, algorithms, samples, and other data required to create a specific synthesizer sound, processor effect, or mode of operation for a device, saved as a group to a memory location. Presets can be stored and recalled so the user can instantly and exactly re-create the specific setting. Examples of presets would include a piano sound in a synth or sampler, a particular echo setting in an effects processor, a specific zoom setting for a DAW editing window, and so on. Most digital devices and many software programs contain various types of preset saving/recalling abilities. There are two broad categories of presets: RAM and ROM. RAM (a.k.a. user) presets are editable and can be overwritten with new data. ROM (a.k.a. factory) presets can be edited, but cannot be overwritten—the preset location is locked, and new data cannot be written there. (However, a ROM preset could be loaded into the device and modified, and then saved to a RAM location if desired.)

pressure. ☞ See *aftertouch*.

pressure microphone (a.k.a. pressure operative microphone). A type of microphone in which only one side of the diaphragm is exposed to incoming sound waves. Because there is no cancellation to create a directional response, pressure microphones have an omnidirectional polar pattern.

pressure-gradient microphone (a.k.a. velocity microphone). A type of microphone in which both sides of the diaphragm are exposed to incoming sound waves. The microphone responds to pressure differences between the two sides. Since sound waves that arrive parallel to the diaphragm create no pressure difference between the two sides, there is a null in the response from the sides, resulting in the figure-8 polar pattern.

pressure-zone microphone. ☞ See *PZM*.

preview. 1. A feature of some offline or non-real-time processors that allows the user to hear a sample of what a process will sound like after it has been applied to a track or a file. Preview was especially important when most offline processors were destructive (actually changed the data in a file) and there was no undo ability. 2. An application that comes with Apple Macintosh computers and that can open graphic and PDF files.

print. 1. Slang term for recording a signal to analog tape. 2. Converting sequencer-driven synthesizer or sampler parts to audio tracks by recording them. 3. Recording the output of effects devices, such as reverbs and delays, to audio tracks to preserve them as part of a session or so the effects devices can be reused for other tasks. 4. To output text or graphics from a computer onto paper.

print-through (a.k.a. interlayer transfer). A phenomenon that occurs with analog tape reels where the magnetism stored on one layer of the reel magnetizes the adjacent layer of tape. Print-through becomes a problem when loud passages on one layer of tape magnetize a silent area between songs or parts of songs on the next layer and become audible as a "pre-echo." To counter pre-echo, most engineers store analog tape "tails out," so that the end of the tape is on the outside of the reel. This generally results in the print-through being masked by other parts of the song rather than falling in silent areas between songs.

pro. ☞ See *professional*.

processor. 1. The chip on a computer's motherboard that is the primary engine managing and performing operations and actions. ☞ **See also** *CPU*. 2. A device that modifies or enhances audio signals. Examples include compressors, delays, flangers, equalizers, reverbs, and limiters. 3. A device that modifies or enhances MIDI data.

producer. A person responsible for managing the production of a piece of music or an album. A producer is involved with coaching, organizing, and scheduling the production; is in charge of managing the financial and budget aspects of a project; oversees sessions; and supervises the mixing and mastering processes. Some producers become true creative partners with the artists they work with, helping with arrangements, musical decisions, and more.

professional. 1. A musician, producer, or engineer who earns the majority of his living working in the audio or music field. 2. Possessing the skills, ability, and equipment to function in the audio or music professions. 3. Equipment worthy of or intended for use by professionals. ☞ **See also** *consumer*, *semi-pro*.

program. 1. A patch or preset in a device. ☞ **See also** *preset*. 2. To create new sounds, patterns, sequences, loops, or other performance data in a digital music device.

program change. A MIDI channel message that tells the receiving device to recall/load a specific preset or program. There are 128 available program change numbers, ranging from 0 through 127. ☞ **See also** *patch mapping*, *bank select*.

program-dependent compressor. A compressor whose response varies according to the frequency and amplitude content of audio signals. Typically, the attack and release times are made program-dependent, which helps with maintaining and controlling transients and reducing pumping and breathing artifacts. Examples include the Universal Audio LA-2A, the dbx 160SL and 1066, and the Fairchild 670.

progressive hammer action. ☞ See *graded hammer action*.

project studio. A personal recording studio. A project studio may or may not be a home studio, but is typically owned and used by one artist, or is owned and used by an engineer or producer for his projects.

propagation. The motion of waves through or in a medium, progressing from molecule to molecule. The molecules don't move with the wave; the vibration of one molecule is passed along to the next molecule, and so on. A great example is a person starting the wave in a stadium. The next person picks up the wave, and so on, until the wave has moved all the way through the crowd and around the stadium.

property. A preference, parameter, or characteristic that defines and describes how a piece of gear or software operates.

proprietary. A process or piece of hardware or software developed by a manufacturer and limited to use in that manufacturer's product(s). Examples might include hardware that only works with its manufacturer's software, a plug-in that only works in its manufacturer's DAW, a synthesizer that only works with specially formatted floppy disks, and so on. While proprietary is often considered to be a bad thing, it can also result in better compatibility, improved functions, and other benefits.

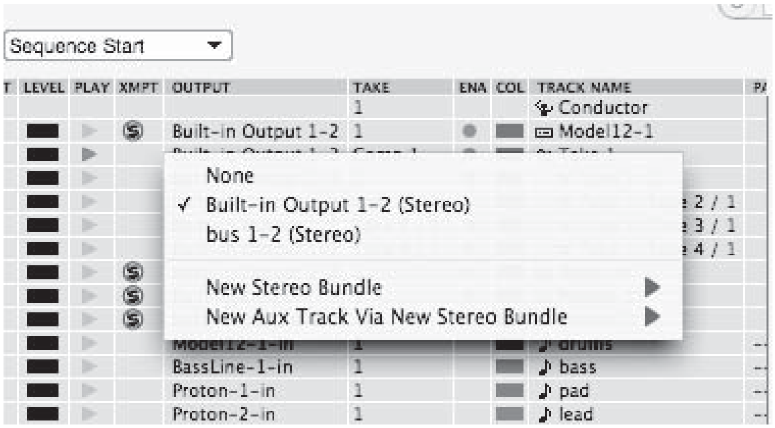

Figure P.13 Pull-down menus are accessed with a mouse click on an item in a program or operating system.

protocol. A standard for transmitting and receiving data between devices or over a network. A protocol may include specifications for transmission, reception, data compression, housekeeping messages, and more. USB, MIDI, SCSI, FireWire, ATA, and others are examples of protocols.

Pro Tools. A family of computer-based DAW hardware and software systems manufactured by Digidesign.

proximity effect (a.k.a. bass tip-up). A distortion in frequency response, usually low-frequency boost, resulting from a directional microphone being placed close to a source. (Omnidirectional microphones do not exhibit proximity effect.) Proximity effect may result in a boost in level up to 16 dB below 100 Hz when a microphone is right up on a source; the bass response will decrease as the mic's distance from the source increases. Radio announcers and some vocalists love proximity effect, as it makes their voices sound fuller. Engineers use careful placement and proximity effect to control the low end when tracking or miking a live performance.

psychoacoustics. 1. A branch of psychology that studies how humans perceive sound and extract information from acoustics. 2. Properties used to trick the brain of a listener into misinterpreting sonic cues. Examples include diffusors that make a room seem larger than it actually is, one sound masking another sound, psychoacoustic models used in lossy compression algorithms, and more.

pulldown. 1. ☞ **See** *pull-down menu*. 2. An audio/video production technique for matching film frame rates to video rates.

pull-down menu (a.k.a. drop-down menu). A menu that appears when selecting a word or item in a program with a mouse click (see Figure P.13). ☞ **See also** *menu*, *pop-up menu, contextual menu*.

pulse. A sharp rise and fall in voltage somewhat similar to a square wave, though a pulse is usually shorter than a square wave. A stream of pulses can be used as an analog clock.

pulse code modulation. ☞ See *PCM*.

pulse wave (a.k.a. rectangular wave). A waveform related to a square wave, but that is not symmetrical. The shape of the wave is determined by its duty cycle (see Figure P.14). Pulse waves have extensive harmonic content and are sometimes described as "hard" and "buzzy" sounding. ☞ **See also** *pulse width modulation*.

pulse width modulation (a.k.a. PWM). A square wave whose pulse width or duty cycle is modulated by a low-frequency signal or a controller, such as velocity or a continuous controller. Pulse width

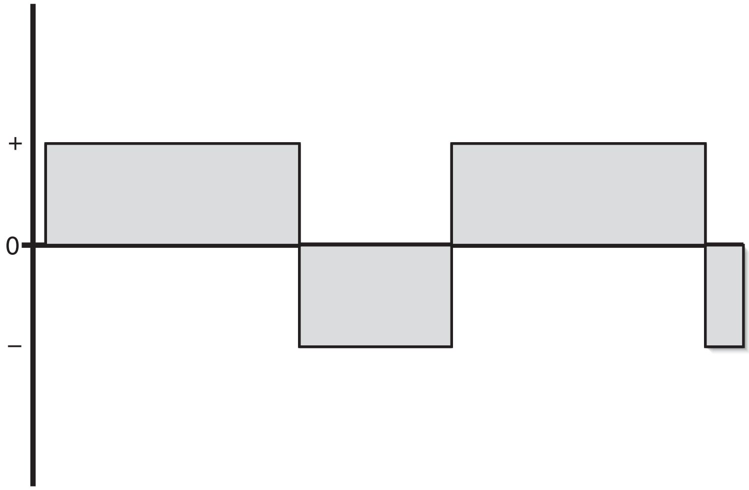

Figure P.14 A pulse wave, or rectangular wave, has a duty cycle greater or lesser than 1:1.

modulation is used in synthesizers to create waveforms whose harmonic content is continuously varying.

pumping. An artifact occurring in a compressor when its release time is set too long. When the release is too long, audio below the threshold that follows audio that has been compressed will also be compressed, pushing it down in the mix. That audio then slowly rises up to its normal level. ☞ **See also** *breathing*.

punch. ☞ See *punch-in*.

punch block. A type of electrical connection strip where wire is pressed into a V-shaped metal terminal using a punch-down tool. The terminal cuts through the wire's insulation and makes electrical contact. No soldering or insulation stripping is required with punch blocks.

punch in. The technique of replacing small portions within an existing track by quickly putting the recorder into record mode at a specific point and recording new audio. Older tape machines required manually dropping the recorder into and out of record mode. Newer machines can automatically drop into and out of record at user-programmed points. Sequencers and DAWs also support punching in and out of MIDI and audio tracks. A punch in may be as short as a single word or note in a track or as long as replacing the majority of a track.

punch out. Switching a recorder out of record mode after punching in.

Purple Book. One of a set of "Rainbow Books" with colored covers containing the Sony/Philips-developed specifications for different optical compact disc formats. The Purple Book contains the specification for double-density CD (DDCD), which can store up to 1.3 GB on a single disc.

push-pull. An amplifier circuit in which two components are used to drive the current—one for the positive state or part of the waveform and one for the negative state or part of the waveform. One switches off while the other is on, and vice versa. ☞ **See also** *single-ended*.

PWM. ☞ **See** *pulse width modulation*.

PZM. Pressure-zone microphone. A term trademarked by Crown. ☞ **See also** *boundary microphone*.

Q

Q. Short for *quality factor*, a.k.a. resonance. The term Q is often used interchangeably with bandwidth, but in fact Q *determines* the bandwidth. The sharper (higher) the Q, the narrower the bandwidth, and vice versa. 1. Q refers to the sharpness of a resonant peak in a synthesizer filter. 2. In equalizers, Q is the center frequency divided by the half-power bandwidth. If an EQ band has a center frequency of 250 Hz and the bandwidth at half power is 100 Hz, the Q is 2.5. 3. In speakers, Q refers to directivity, or how energy is dispersed from the driver. A speaker Q of 1 indicates equal energy dispersed in all directions. A speaker Q of 2 indicates 180-degree dispersion; the higher the Q, the narrower the dispersion.

quad core. A single die or chip containing a four-core computational processor, allowing multiprocessing on a single chip. ☞ **See also** *core*, *Multi-core*.

quadraphonic. Literally, four-channel audio, though most people will think of the 1970s four-channel speaker system (left and right in the front corners of the listening space, and left and right in the back corners). Quadraphonic material was available on eight-track tape, LP records, and reel-to-reel tape in a number of competing formats, including CD-4, Q4, Quad-8, SQ, QS, and Matrix H. The excess of formats combined with lack of interest from consumers in purchasing extra speakers and other equipment killed quadraphonic.

quadratic residue diffusor. Acoustical device designed with a random surface pattern using mathematical formulas. ☞ **See also** *diffusion*, *diffusor*.

quadrature. ☞ See *phase quadrature*.

quality factor. ☞ See *Q*.

quantization. Literally, dividing a continuous signal or event into discrete steps (see Figure Q.1). 1. In digital audio, discrete measurements are taken of a continuous analog signal and represented as discrete binary data using bits for storage. 2. In MIDI sequencers and drum machines, the timeline is divided into a grid at a certain timing resolution (usually measured in PPQN—pulses or parts per quarter note), and each rhythmic event is quantized, or moved to the nearest grid location. The higher the resolution, the finer the grid, and the more "human" the performance can be. Older sequencers might have a resolution of 48 or 96 PPQN, while modern sequencers can have an internal resolution as high as 1,920 PPQN.

quantization error. Errors resulting from attempting to represent a continuous signal or event with discrete steps. With digital audio, in most cases the measurements of the analog signal will not fall exactly on the digital "steps." In this case, the measurement must be moved to the nearest step. The difference between the actual value and the nearest digital step is the quantization error, which appears as extra harmonics in the signal. Because the samples go by so quickly and the signal is constantly varying, these harmonics are perceived as noise. (See Figure Q.2.) ☞ **See also** *quantization noise*.

quantization noise. Wideband noise resulting from the collective quantization errors in a signal. ☞ **See also** *quantization error*.

quantize. To move a rhythmic event to correspond closely or exactly to a timing grid. ☞ **See also** *quantization*, *quantizing*.

quantizing. A feature of MIDI sequencers and drum machines that moves MIDI events to match a timing grid (for example, to the nearest eighth or sixteenth note). Quantizing can be used for correcting or modifying a rhythmic performance or for producing a mechanized feel. See Figure Q.3.

quarter space. A sound source, such as a speaker, located at the junction of two surfaces, such as the corner of two walls, is said to be in quarter space. Quarter-space placement of a speaker or other sound source results in a 6-dB level increase (mainly in the low frequencies) over free-space placement (see Figure Q.4). ☞ **See also** *free space, half space, eighth space*.

quarter track. A configuration of an analog tape or tape recorder head that allows for four tracks, with each track taking one quarter of the tape-head width. In a professional multitrack machine, the four tracks

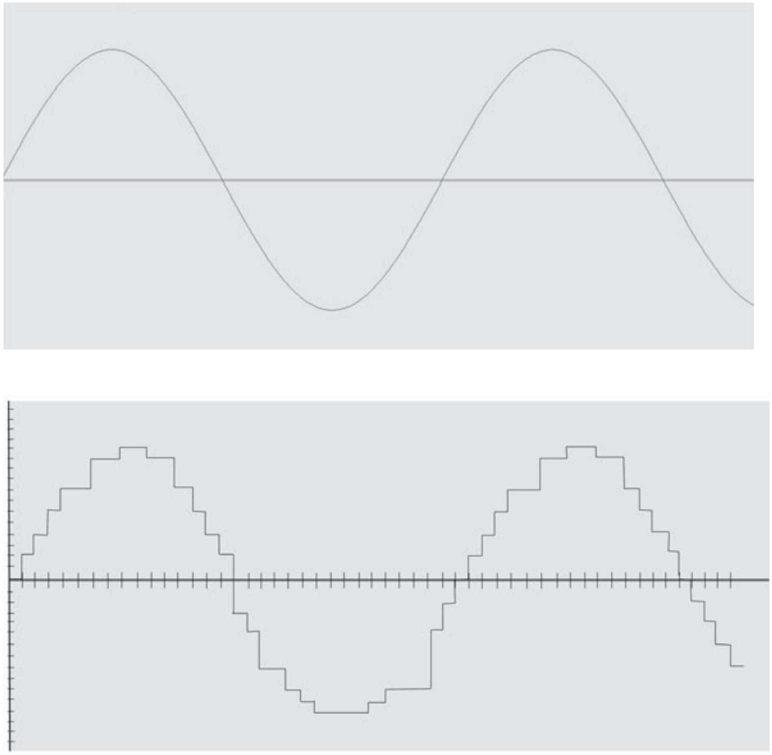

Figure Q.1 Quantization represents a continuous signal with discrete steps.

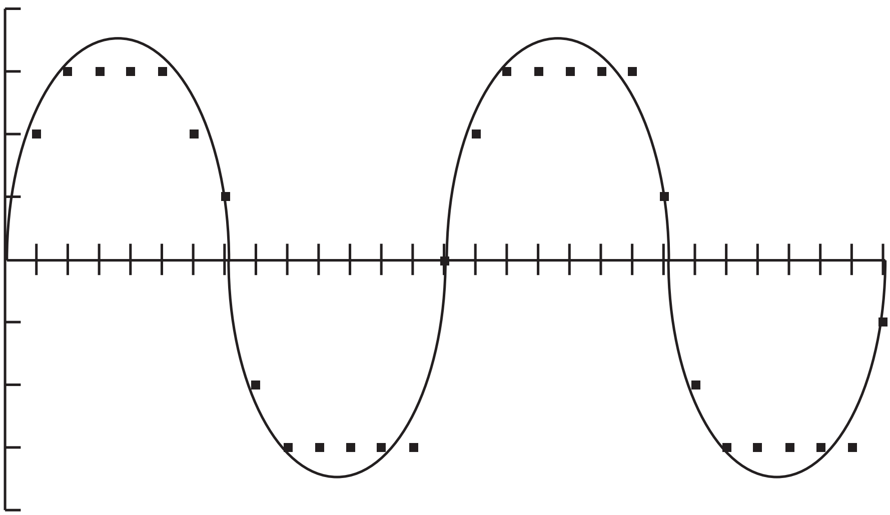

Figure Q.2 Quantization error results from the difference between an actual value being measured and the discrete, quantized steps used to represent that measurement.

Figure Q.3 Moving MIDI events to match a timing grid is called quantizing. The MIDI notes at the top have not been quantized; the bottom shows the same notes after they have been quantized to the grid.

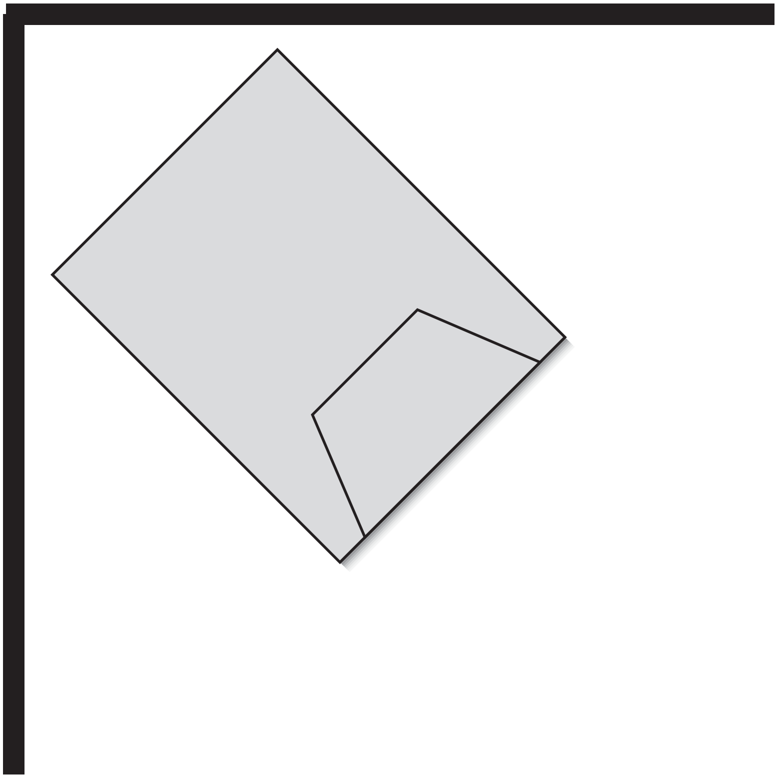

Figure Q.4 A speaker in quarter space will have a substantial bass boost.

will be configured as independent tracks that can be recorded simultaneously. In stereo consumer and mixdown recorders, whether cassette or reel-to-reel, two tracks will be recorded/played in one direction, with a second set of two tracks recorded in the other direction by flipping the tape over.

quasi-parametric (a.k.a. semi-parametric). An equalizer band that allows changing the frequency and cut/boost, but not the bandwidth. Many mixing consoles feature quasi-parametric "sweepable" midrange bands. ☞ **See also** *sweepable mids*.

QuickTime. A multimedia format developed by Apple Inc. for the Macintosh, for storing, playing, and interchanging audio, video, and graphics. Now also available for the Windows platform.

quiescent noise. The operating noise produced internally by audio devices combined with the background noise in a room. This includes hum, hiss, and air-conditioning noise, as well as any noise in the environment.

QWERTY. The standard layout for computer keyboards and typewriters in English. Refers to the first six alphabet/letter keys on the upper-left of the keyboard.

R

rack. An enclosure designed to accept rackmountable equipment. A rack may resemble a piece of furniture for studio use, may be a metal cabinet for commercial installations, or may be a portable unit with handles and removable covers. Some racks have wheels to make them more movable when loaded with heavy equipment. Audio racks are set up to accept gear that is 19 inches wide and are measured vertically by the number of rack units of gear they can hold. ☞ **See also** *rack space*.

rack ear. A bolt-on or integrated flange of metal on the enclosure of a piece of gear that is used to mount the item into a rack.

rackmount. A piece of gear that is physically designed to mount into a standard rack.

rack rail. A wood or metal bar on the inside of a rack that is drilled to accept screws spaced for standard rack spaces.

rack space (a.k.a. rack unit, modular unit). A rack space is a vertical measure of a piece of gear. A single rack space is 1-3/4 inches tall. Equipment can range from a single rack space tall to two, three, four, or even more rack spaces in height.

rack unit. ☞ **See** *rack space*.

RAID. Redundant Array of Inexpensive Disks (the "official" acronym) or occasionally Redundant Array of Independent Disks. A RAID is a computer storage solution that stores and divides or duplicates data among two or more hard drives that appear as a single drive to the computer. There are two reasons to use a RAID system: faster speed, because the data is being stored to and read from multiple drives (that appear as one drive to the computer), or for back up, as the data is being duplicated on separate drives simultaneously (that appear as one drive to the computer). The disks used can be any plain hard drives; a RAID controller, whether hardware or software, is required to accomplish the RAID functionality. There are several types of RAIDs:

- **RAID 0 (a.k.a. striped disks).** Data is divided and distributed across multiple disks to increase speed, throughput, and storage capacity. The failure of any single disk in the array will result in the loss of all data. RAID 0 is ideal for high-throughput situations.
- **RAID 1 (a.k.a. mirrored disk).** Identical data is written to multiple disks simultaneously, creating an instant backup. The failure of a single disk in the array will not result in any loss of data. RAID 1 is good for situations in which it is critical to protect data.
- **RAID 0+1 and 1+0 (or 10) (a.k.a. nested RAID levels).** Hybrid combinations of RAID 0 and RAID 1, where data is distributed across drives, and each of those drives is simultaneously written to a backup set of drives. These are costly solutions because double the number of drives is required, but they do provide excellent throughput and data security.
- **RAID 2.** Intended for use with drives that do not have built-in error correction. RAID 2 is seldom used.
- **RAID 3.** Similar to RAID 0, but stripes data at the byte level. RAID 3 is seldom used.
- **RAID 4.** Similar to RAID 0, except that parity data is written to one of the disks, so a replacement disk can be created if a single disk fails. RAID 4 was largely replaced by RAID 5.
- **RAID 5.** Similar to RAID 4, except parity data is distributed across all the disks. Any two disks will be able to recover data lost on a third. RAID 5 is popular for multiuser installations.

- **RAID 6.** Similar to RAID 5, but data lost on any two disks can be recovered from the remaining disks. RAID 6 provides extra security over RAID 5.

rail. 1. ☞ **See** *rack rail*. 2. Also known as power supply rail. A metal rod or bar used as a bus to distribute or deliver power from the power supply in some audio devices, such as mixers and amplifiers.

Rainbow Books. A set of books with colored covers developed by Sony/Philips that contains the detailed specifications for a number of different optical compact disc formats.

- **Beige Book.** Photo CD.
- **Blue Book.** Enhanced CD, CD+, CD+G, CD+EG, CD+XG.
- **Green Book.** CD Interactive.
- **Orange Book.** CD-R, CD-RW, CD-MO.
- **Purple Book.** Double-density CD.
- **Red Book.** CD digital audio.
- **Scarlet Book.** SACD.
- **White Book.** Video CD, Hybrid CD.
- **Yellow Book.** CD-ROM, CD-ROM XA.

RAM. Random Access Memory. A type of semiconductor chip that the computer CPU can use to store data for later retrieval. Most types of RAM are volatile and lose their contents when power is removed. There are many types of RAM—some general purpose, and others intended for specific applications. Some examples include:

- **Cache RAM.** High-speed RAM used to store frequently accessed data.
- **Dynamic RAM (a.k.a. DRAM).** A type of RAM based on capacitors, which store a charge. DRAM must be refreshed because the capacitors lose their charge over time.
- **P-RAM.** Parameter Random Access Memory. A type of non-volatile RAM used for storing system and device settings in computers and other devices.
- **Static RAM (a.k.a. SRAM).** A type of RAM that does not need to be refreshed; SRAM will even retain its contents for a short time after the power is removed.
- **VRAM.** Video RAM. A type of RAM used as a buffer by video cards.

RAM disk. A computer function that uses part of the machine's RAM to create a virtual disk drive. RAM disks operate just like physical disk drives, but can access data much faster. However, all data in a RAM disk (and the RAM disk itself) will be lost when the program that created it is shut down or the computer is powered down.

random access. A storage system in which the contents of memory chips or a disk can be accessed or retrieved by jumping directly to an item, rather than stepping linearly or consecutively through the data. Random access is much faster than linear access.

randomize. To produce or arrange data or other items in an unpredictable manner.

random noise. A type of noise often used for making acoustic measurements. Random noise features an even, constantly changing distribution of sound energy across the frequency spectrum. ☞ **See also** *pink noise*, *white noise*.

range of human hearing. The range of the highest and lowest frequencies the human ear can perceive. The range of human hearing is generally accepted as 20 Hz to 20,000 Hz, though there are those who feel that we also perceive higher and lower frequencies and that these contribute to the quality of the sound we hear.

rarefaction. An area of decreased air pressure caused by a sound wave. The opposite of a compression.

rate scaling. Rate scaling refers to shortening the length of the envelopes and reducing the volume of the higher notes produced on a synthesizer or sampler. This type of response corresponds more closely to how most acoustic instruments work. ☞ **See** *keyboard scaling*.

ratio. In a dynamics processor, such as a compressor, limiter, or expander, the ratio is the parameter that defines the amount of level reduction applied when an input signal crosses the threshold. The ratio indicates the change in output level for a given input-level change. For example, with a 2:1 ratio, a 2-dB change in input level results in a 1-dB change in output level (or the output signal having its level change reduced by a factor of 50%); with a 4:1 ratio, a 2-dB change in input level results in a 0.5-dB change in output level (or the output signal having its level change reduced by a factor of 75%). See Figure R.1.

rattle. ☞ **See** *flutter echo*.

razorblade. The traditional analog tape editing tool.

RCA plug (a.k.a. phono plug). A type of unbalanced coaxial connector developed by Bell Labs for

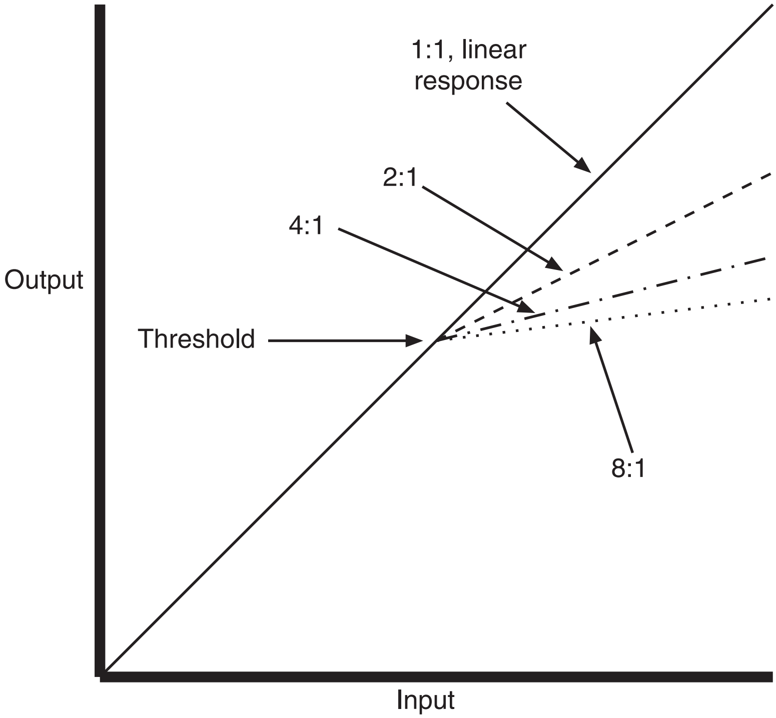

Figure R.1 A dynamics processor's ratio control determines how much the output level will change compared to the input level.

telephone applications, but widely used by RCA (Radio Corporation of America) for phonograph connections. An RCA plug has a center hot pin surrounded by a ring of ground flanges, which grip the corresponding jack, providing a solid connection. RCA plugs are used on consumer and semi-professional analog audio equipment as well as for S/PDIF digital connections on semi-pro and professional equipment.

R-DAT. Rotating Digital Audio Tape. ☞ **See** *DAT*.

RealAudio. A proprietary standard introduced in 1995 by RealNetworks for streaming data-compressed audio over the web. A plug-in decoder is available for most web browsers that support RealAudio streaming. RealVideo is also available for streaming video content.

real time. Typically, this term is applied to computers and software. Literally, "real time" refers to the computer receiving or reading data, processing it, and outputting it without a perceivable time delay. In audio, "real time" indicates a process or event that happens live and continues while the listener hears it, without breaks or pauses to allow the computer to work.

real-time analyzer (a.k.a. RTA, spectrum analyzer). An audio diagnostic and monitoring device that provides a visual display of the amplitude of the frequency spectrum in real time. A real-time analyzer may be either a hardware device or a computer program or plug-in. RTAs are useful for analyzing rooms, setting up and optimizing audio systems, and assisting with equalization in either live or studio situations.

reamp. A trademarked term of the Reamp Company for the process of playing back a recorded signal through a guitar or other amplifier, then recording the resulting sound to another track. Typically a dry, DI track of the desired guitar performance is recorded during tracking or overdubs. This dry track is then reamped through a variety of different amps in order to try different tones and to arrive at the best sound for the track, all without the player having to perform the part over and over. This allows the engineer greater flexibility and more options for getting the best sound. Note that with DAWs, reamping may mean recording the DI guitar track, then playing back through amplifier simulation plug-ins to achieve the desired sound instead of using a miked-up amplifier. See Figure R.2.

rear-ported. A speaker cabinet or enclosure designed with a port on the rear panel. ☞ **See also** *bass reflex*.

recap. To replace the capacitors in a piece of gear, such as a mixer or amplifier. Recapping is done to old or vintage gear because some types of capacitors will deteriorate with age, which can affect sound quality. Recapping is also done to "hot-rod" or

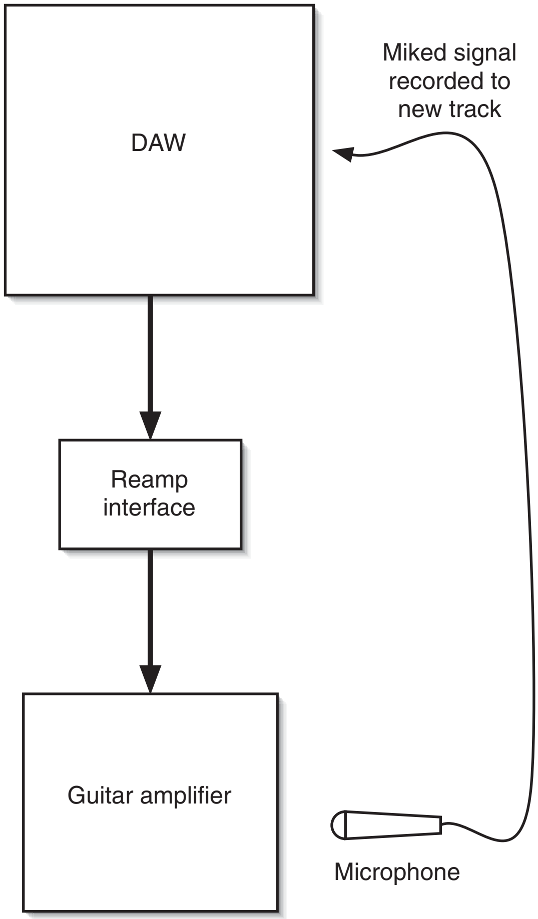

Figure R.2 Reamping allows greater flexibility when deciding on guitar tones. A DI track is recorded, then is sent out through an amplifier and recorded to a new track.

improve sound quality by replacing the stock capacitors in a unit with higher-grade components.

reconstruction filter. ☞ See *anti-imaging filter*.

record. 1. The process of storing sound in magnetic, analog voltage, digital, or other form. 2. A collection of one or more pieces of music intended for distribution on physical media or via download or streaming. 3. A piece of computer data. 4. A vinyl LP; a plastic disk containing sound stored in grooves.

record arm. ☞ See *record enable*.

record enable. A control or function that "arms" a track or makes it available to record. A track must be record enabled in order to record; this helps prevent the engineer from accidentally recording over tracks that have already been recorded.

recording console (a.k.a. recording desk, recording mixer, mixer, board). A mixing console containing features that are intended and optimized for recording studio applications. Typically, this includes additional inputs for monitoring signals from multitrack tape decks, subgroups, talkback, studio and control room monitor control and switching, and in some cases, automation. ☞ **See also** *mixer*.

Recording Industry Association of America. ☞ See *RIAA*.

recording session. ☞ See *session*.

record-ready. A track that is armed and ready to record is said to be record-ready.

rectifier. An electronic component that converts AC electricity to DC electricity. Rectifiers are often found in the power supplies for audio equipment and amplifiers. They are most commonly solid state, though tube amplifiers may also use a tube rectifier.

Red Book. One of a set of "Rainbow Books" with colored covers containing the Sony/Philips-developed specifications for different optical compact disc formats. The Red Book contains the specification for audio CD, a.k.a. CD-DA or Compact Disc Digital Audio.

redo. A computer program command that reverses the undo command. Redo restores the data to its condition before the undo command was employed. Many programs now offer unlimited undo, and therefore offer complementary unlimited redo as well. ☞ **See also** *undo*.

reel. In audio, a "spool" used to wind and store magnetic recording tape. A reel consists of a hub, which is the central axis that the tape is wound around and that mounts to the tape deck, and two flanges, which are the sides of the reel that keep the tape from spilling off.

reel-to-reel. A type of analog or digital tape deck in which the magnetic recording tape is wound and stored on and travels between two open reels, a supply reel and a take-up reel.

reference. 1. To compare the sound quality of two sources. 2. A commercial recording used by an engineer as the basis for sound-quality comparisons.

reference monitor (a.k.a. monitor, studio monitor). A type of speaker optimized to produce flat, uncolored response—this differs from most home stereo, consumer speakers, which are often designed to enhance the sound of music. The idea is for the monitors to deliver true sound to the engineer, so he can depend on them as a reference when making decisions about the sound quality of recordings and mixes. Reference monitors may be either active or passive, and come in a variety of sizes, ranging from models with 3-inch woofers to models with one or more 15-inch woofers. Of course, studio speakers are only part of the monitoring equation; the room, any acoustic treatments, amplifiers driving the speakers, speaker placement, and even cables all affect the sound the engineer hears.

reflect. To bounce off a surface.

Reflection-Free Zone (a.k.a. RFZ™). A trademarked term for an area in a studio control room around the main listening position that has been treated with absorbers to reduce or prevent reflections. See Figure R.3.

reflection phase grating. ☞ See *Schroeder diffusor*.

reflector. An acoustical device used to redirect sound waves.

refraction. 1. The "bending" or speed change of a sound wave as it passes from one medium to another or as air-temperature changes occur. 2. The tendency of low-frequency sound waves to bend around objects rather than reflect off of them.

refurbished. 1. A product that did not pass the manufacturer's quality control and has been remanufactured or repaired. A refurbished product cannot be sold as new—though it may carry a full warranty—and is usually offered at a discount price. 2. A returned product that has been repaired or restored at the factory to meet the manufacturer's specifications.

regeneration (a.k.a. feedback). A parameter on a delay or other processor that sends a portion of the effected signal back through the processor. In a delay, this results in additional repeats. In a processor such as a flanger, this results in a more intense effect.

Figure R.3 A Reflection-Free Zone is an area in a control room where all reflections have been controlled, increasing the accuracy of monitoring. In this case, absorptive acoustic foam has been mounted around the listening position to control reflections.

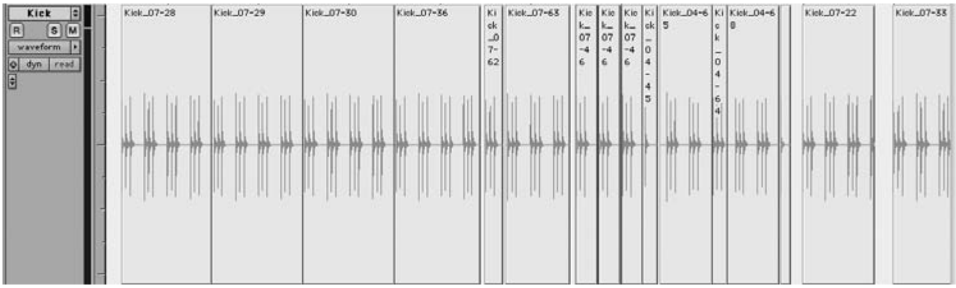

Figure R.4 Each of the blocks on this track containing a waveform is a separate region of audio.

region (a.k.a. segment, clip). A term used in some DAWs, sequencers, and audio programs for a segment or piece of audio or MIDI in a track. A region may be a complete audio or MIDI file, or a part of an audio or MIDI file. See Figure R.4.

register. 1. To notify a manufacturer of the identity of a purchaser of a piece of equipment or software in order to receive support, updates, or other bonuses. 2. A small amount of high-speed memory in a CPU. Registers are used to temporarily hold data that will be needed immediately. The CPU transfers necessary data from RAM into a register, operates on it, then returns the data to RAM. 3. One of six vocal ranges or types (bass, baritone, tenor, contralto, soprano, mezzo-soprano). 4. One of three or more pitch ranges of a woodwind instrument, each produced using different blowing techniques. 5. The relative pitch range of a note, melody, instrument, or group of instruments.

registered parameter number (a.k.a. RPN). A continuous controller that is defined or dedicated to a parameter type by the MIDI specification. A registered parameter number is a multi-part message that uses CC #100 and 101 to indicate a parameter number and CC #6 (and optionally 38 for more resolution) to indicate the parameter value. Data increment messages (CC #96 and 97) can be used to adjust the current value of a registered parameter number. There are currently six RPNs: Pitch Bend Range or Sensitivity, Channel Fine Tuning, Channel Coarse Tuning, Tuning Program Select, Tuning Bank Select, and Modulation Depth Range. A seventh RPN, Null, should be sent to cancel access to the RPN after its data has been sent.

relap. To sand and polish a worn tape-recorder head to restore it to its original shape and performance. Relapping can generally only be done once per head; attempting to relap more than once can result in changes in the head gap.

release. 1. In a compressor, the amount of time between when the input signal drops below the threshold and when the unit is no longer applying compression. 2. In a noise gate, the time it takes for the gate to decay to its closed state after the hold stage is complete. Common settings range from two milliseconds to four seconds. 3. In a synthesizer or sampler, the parameter that sets how long a note continues playing after a key has been let up.

release trigger. A sound or sample that is triggered when a note is released. The most common example application of this is with harpsichord sounds. A real harpsichord sounds notes by plucking a string with a small quill or pick when a key is depressed. When the key is released, the quill returns to its ready position, striking the string for a second time as it returns. With a sampled harpsichord, a release trigger can be used to play a sample of the sound of the quill returning past the string when the key is released.

release velocity. How fast a player releases a key on a keyboard—literally, the time it takes for a key to go from the down position to the up position. Release velocity is sent as part of a MIDI note off message and has a range of values from 0 to 127. Release velocity can be routed to control the reverb amount, release time, or any modulation destination that is supported by the receiving device. Despite its expressive possibilities, release velocity is not widely supported by controller keyboards.

remix. 1. A new version of a piece of music created based on one or more tracks from the original multitrack sessions—which may be substantially processed or edited—and additional loops, melodies, harmonies, effects, and accompaniment material. A remix may be as simple as putting the original tracks over new beats or loops, and as complex as creating a completely original new production at an entirely different tempo, built around just the original lead vocal part. 2. To create a new mix of a track.

remote control. 1. A device used to control another device from a distance. A remote can be wired or wireless. Remotes are useful for being able to operate equipment without leaving the studio sweet spot, or for controlling equipment while recording yourself. 2. The ability of a device to respond to commands from another device.

removable storage device. A device that writes data to and reads from cartridges containing hard disks, rewritable CD and DVD media, or cards containing flash memory; or standalone devices, such as USB thumb drives. Removable storage devices provide several advantages: The media is portable and can be placed in any compatible chassis; when a piece of media is full, a new one can be used to expand the amount of storage; adding a new piece of removable media is usually cheaper than purchasing an additional fixed drive; information can be easily backed up to a new piece of media; and so on. The disadvantage is that removable media tend to be slower than fixed media, making removable devices less useful for audio and video playback and editing.

render. To create and write a new audio file based on the processed, mixed, or edited version of the original file.

repro. Short for reproduction, a.k.a. confidence monitoring. A function in some tape recorders that can play back all tracks, including those currently being recorded, using a repro or play head. This allows the engineer to monitor exactly what is being recorded to tape during the session, though there is a very slight delay because of the time it takes for the tape to move from the record head to the repro head.

resampling. A function of some samplers, where the device can play a sample from RAM through internal DSP and effects and re-record the processed output signal as a new sample.

reset. 1. To return a device to its default, power-on state. 2. To return a setting to its previous position.

Reset All Controllers. A MIDI channel message that tells all receiving devices to return all controllers to their default settings.

resistor. An electronic component that opposes or restricts the flow of electrical current by producing a voltage drop. The amount of resistance provided by the component is measured in ohms.

resolution. ☞ See *bit resolution*.

resolve. 1. To exactly synchronize a slave to a master using time code. 2. To synchronize a digital device to an external word or sample clock.

resonance. 1. A parameter on some filters that emphasizes the cutoff frequency. 2. In acoustics, a boost in a particular frequency due to a room mode or standing wave.

resonant. The tendency to vibrate at and reinforce certain frequencies.

resonant filter. A filter with resonance or emphasis at the cutoff frequency. The filter's Q, or quality factor, determines the bandwidth and sharpness of the resonant peak. In some filters the Q can be increased until the frequencies above and below the cutoff point are no longer heard, and the filter begins to self-oscillate, generating a sine wave tone at the cutoff frequency.

resonant frequency. A frequency at which resonance occurs. Every object and every material has a resonant frequency.

resonant mode. ☞ See *room mode*.

resonator (a.k.a. acoustic resonator). An acoustical device that vibrates sympathetically in response to a sound wave. Because energy at a particular frequency is required to make the resonator vibrate, the energy level at that frequency is reduced in the room. This helps to control room modes and even out the response of the room.

restart (a.k.a. warm boot). A command or process that returns a computer to the state it is in immediately after being powered up, then reloads the operating system. Restarts are often required after installing new software or hardware or updating an application or the operating system. A restart may also be necessary if a device or application locks up or suffers a catastrophic crash.

resultant tone. A sum and/or difference tone that is heard when two loud notes of different pitches are played simultaneously. The sum tone's frequency equals the sum of the two original notes' frequencies. (For example, 500-Hz and 100-Hz tones would result in a 600-Hz sum tone.) The difference tone's frequency equals the difference between the two original notes' frequencies. (For example, 500-Hz and 100-Hz tones would result in a 400-Hz difference tone.)

resynthesis. The process of analyzing a sound, then using additive or other synthesis techniques to re-create that sound.

retro. An item that imitates an older style.

retrofit. To add a component or option to a device after it leaves the factory.

return. An input connection used to bring a signal back into a mixer after it has been processed using an external device, such as a delay or a reverb. Returns are usually paired with sends to create an external loop into which processors can be inserted.

reverb. ☞ See *reverberation*.

reverberant decay. The time it takes for the reverb in a room to stop ringing. ☞ **See also** *RT60*.

reverberation. The sound left ringing in a room after the direct sound from the source is silenced. Sometimes mistakenly called *echo*, reverberation differs in that it is a wash of reflections that typically does not contain discrete, discernible echoes. The reverb quality provides information about the size and shape of the space in which the sound is occurring.

reverb send. A mixer bus or external connection that is dedicated to routing a mix of signals from one or more channels to a reverb processor.

reverb time. A measure of how long reverb lasts in a room. ☞ **See also** *RT60*.

reverse polarity. A signal or connection whose polarity has been flipped so that positive is negative and vice versa. This is not the same as placing a signal 180 degrees out of phase (though the result can be similar)—changing phase implies a time-based change, whereas polarity is an electrical change.

rewind. A transport control that causes a tape to fast-wind in reverse toward the start of the tape or a DAW or audio program to quickly scroll in reverse toward the beginning of the song or project.

ReWire. An industry-standard protocol for Macintosh and Windows developed jointly by Propellerhead Software and Steinberg, and now supported by most audio software manufacturers. ReWire allows real-time MIDI and audio transfer between different applications running on the same computer. ReWire supports up to 256 audio channels, 4,080 MIDI channels, and transport control between applications.

REX file. A file format developed by Propellerhead Software. A REX file consists of a loop or audio file that has been chopped into slices at attack transients, with each slice assigned to a MIDI note number. The REX file is played back by triggering the slices by playing the appropriate MIDI notes. The speed or tempo of the file can be changed without changing its pitch by triggering the slices at a faster or slower rate. The slices can also be played individually or in a different order by choosing certain notes or varying the order of the MIDI notes used to trigger the slices. Most DAWs support the playback of REX files. REX files can be created from audio files and loops using Propellerhead's ReCycle or other compatible software.

RFI. Radio Frequency Interference. A type of EMI consisting of radio and television broadcast signals. RFI can be picked up by electronic and audio equipment and cabling, resulting in buzz, hum, and sometimes even the actual broadcast signal mixed in with the desired signal. Shielding, certain types of filters, and proper grounding all help prevent circuits and cables from picking up RFI.

RFZ. ☞ See *Reflection-Free Zone*.

RIAA. Recording Industry Association of America. An organization that works on behalf of the recording industry. The RIAA presents gold and platinum awards for record sales, fights piracy, and has also developed technical standards for LP and copy protection purposes. www.riaa.com.

RIAA equalization (a.k.a. RIAA curve). A standard developed by the RIAA for vinyl LP playback. The RIAA EQ pre-emphasizes the signal during mastering by reducing low frequencies and boosting high frequencies to allow for easier record pressing and duplication. On playback, the reverse curve is applied to the sound to provide the correct sound, with the low frequencies boosted to restore them to their proper level, and the high frequencies reduced —which also reduces hiss from the LP surface.

ribbon cable. A type of wide, flat cable made up of many side-by-side conductors. Ribbon cables are often used inside computers and digital electronic devices because they are easy to route within a confined space and they provide many conductors for signal routing. Because they are unshielded and not very durable, ribbon cables typically are only used for internal connections within a piece of gear, and not for external signal routing, although some external computer peripherals do use ribbon cables for connecting to expansion cards.

ribbon controller. A control device found on some synthesizers consisting of a long, thin conductive strip or ribbon of touch-sensitive material. As a finger is placed on and slid across the ribbon, control messages are generated and can be used to change various parameters, such as pitch, filter cutoff, and others.

ribbon microphone (a.k.a. velocity microphone). A type of microphone that uses a thin corrugated metal ribbon suspended in a magnetic field as a diaphragm. Air motion created by sound waves moves the ribbon, creating a voltage. Ribbon microphones, by nature, have a figure-8 polar pattern. Ribbon microphones tend to be fragile, produce low output, have strong proximity effect, and are sensitive to mic preamp impedance mismatches. (Some modern designs use active circuitry to boost the low levels and to isolate the microphone from the preamp's impedance.) Ribbon mics are prized by recording engineers for their natural high-frequency response and warm, unhyped sound quality. ☞ **See also** *pressure-gradient microphone*.

ribbon tweeter. A type of high-frequency driver that uses a very thin ribbon of metal or metal-coated plastic suspended in a magnetic field to create sound waves. A drawback of ribbon tweeters has been their low output, though newer designs have surmounted this problem. Ribbon tweeters tend to have good horizontal dispersion with limited vertical dispersion, making them useful for creating stacked arrays for high-power situations.

riding a fader. Manually adjusting a fader's level throughout a recording or mixdown pass in order to manage levels without using compression or limiting.

RIFF. Resource Interchange File Format. A multimedia file format developed by Microsoft that serves as a framework for storing multiple types of data, such as audio, video, graphics, MIDI, and other RIFF files in a single package.

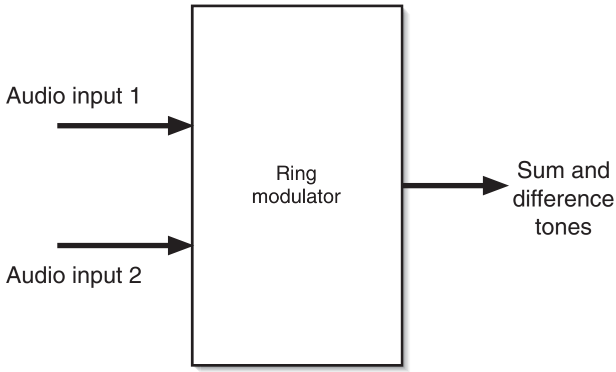

Figure R.5 A ring modulator creates new sounds by generating frequencies that are at the sum and difference of the frequencies in two input signals.

ringing out a room. The process of tuning or optimizing a monitor system to control frequency response peaks in the room and system that are prone to feedback. Ringing out a system is done by increasing the volume level until feedback begins, then using an equalizer to reduce the level of the frequency that is feeding back. The level is then increased again until another frequency feeds back; the equalizer is used to pull that frequency down; and so on.

ring modulator. A type of processor that combines two signals and outputs the sum and difference of their frequencies. No original or dry signal is heard at the output. The simplest example is to combine two sine waves—one of 300 Hz and the other at 200 Hz. The sum of the two signals is 500 Hz; the difference is 100 Hz. With complex waveforms, the output from a ring modulator can be extremely complex, often resulting in clangorous "metallic" sounds that have been used as special effects and to create certain types of "robot" voices. See Figure R.5.

RISC. Reduced Instruction Set Computer. A type of CPU developed by IBM in the 1970s that increases speed by very quickly executing simple, uniform, fixed-length instructions, rather than complex instructions. Because the instructions are simpler, RISC allows faster compiling of assembly language or machine code. The Apple Power Mac computers were RISC-based. ("Power" stands for Performance Optimization with Enhanced RISC.)

rise time. The time it takes for a signal or voltage to change from a low value to a high value, typically measured as the time it takes to go from 10% to 90% maximum level. ☞ **See also** *transient*, *transient response*.

RJ-45. A type of connector used on Cat 5 and Cat 6 cables for Ethernet and other communication applications.

RMID. RIFF .MID or Resource Interchange File Format .MID. A file format originally developed by Microsoft and adopted by the MMA (*MIDI Manufacturers Association*) that combines Standard MIDI Files (SMF) with Downloadable Sounds (DLS) and provides a portable way to transfer files while still allowing accurate and consistent playback on any compatible system.

RMS. Root Mean Square. A method used to calculate the average of values over time. In audio, RMS calculations are used to find continuous power ratings. (There is no such thing as "RMS power"; what is usually meant by RMS power is average power. The confusion may come from the fact that RMS voltage and RMS current can be used to calculate average power.)

RoHS. Restriction of Hazardous Substances. An agreement reached in 2006 by members of the European Union to ban new electronic equipment containing amounts of any of six environmentally damaging hazardous substances that exceed specified levels. The substances include lead, cadmium, mercury, hexavalent chromium, polybrominated biphenyl, and polybrominated diphenyl. Although RoHS has not been adopted in the United States, many U.S. manufacturers have found it necessary to comply so that their products can be exported to Europe.

rolloff. The gradual reduction in level of frequencies by a filter above or below a cutoff frequency, referred to as a *high-frequency rolloff* or *low-frequency rolloff*, respectively. The slope of the rolloff is rated in decibels per octave, with the rolloff slope being steeper with higher numbers. For example, a rolloff of 6 dB per octave would reduce the level 6 dB at a frequency one octave above the cutoff frequency, 12 dB at a frequency two octaves above the cutoff frequency, 18 dB at three octaves above the cutoff, and so on. With a slope of 12 dB per octave, this rate of level reduction would increase twice as fast.

ROM. Read-Only Memory. A type of computer memory chip whose contents can only be read, not modified, erased, or replaced by the user. The contents of ROM chips are sometimes referred to as *firmware* and typically consist of instructions and data required for the device to operate.

ROMpler. A synthesizer based on sampled waveforms stored in ROM. ☞ **See also** *sample-based synthesis*.

room mic. A microphone that is positioned some distance from a sound source in order to primarily capture the room ambience and reflections created by the sound of the source. The room mic signal can be blended in with direct mics placed near the source in order to add depth and realism to the sound.

room mode. 1. A low-frequency standing wave in a room. 2. An acoustic resonance at a particular frequency in a room. Room modes occur when sound reflects between parallel surfaces. They cause anomalies in the room's response that make it very difficult to accurately monitor sound in that room.

room reverb. A type of preset in digital reverbs that attempts to re-create the ambience of a physical space, usually a small to medium-sized room.

room within a room. Type of studio construction in which a floating floor is constructed, then walls and ceiling are built on top of that floor, resulting in a room that is isolated from the surrounding structure.

root directory. In computer file systems, the root directory is the highest-level directory. All the other directories branch off from the root directory.

root key. The note on a keyboard that corresponds to a note being sampled. For example, the note Middle C on a keyboard, to which a sound with the pitch of Middle C will be assigned.

rotary encoder (a.k.a. shaft encoder). A type of control that is turned on a shaft and that resembles a volume knob. Commonly found on control surfaces, a rotary encoder generates MIDI or other data that is used to control parameters in a piece of software or in a hardware device.

rotary speaker. A speaker cabinet in which the driver rotates, or a horn or baffle rotates around the driver. A unique chorusing effect is created in a rotating speaker because of the Doppler Effect; as the driver rotates away from and toward the listener, the pitch of the sound it is producing changes. The most recognizable example of a rotating speaker is the Leslie speaker (actually a family of models), which was invented by Donald Leslie and first produced in 1941. The original Leslie cabinet was designed to be used with Hammond organs, though rotating speakers from several manufacturers are compatible with other instruments, such as electric guitar.

rotational delay. The time it takes for a specific area or location on a hard drive or optical disc to rotate under the read/write head. Rotational delay, together with seek time and transfer time, determines a drive's access time.

rough mix. A quick mix created for the purposes of evaluating a song, mix ideas, an arrangement, processing, or for other reasons. A rough mix may also be used as a guide for overdubbing additional parts. A rough mix is not intended for release to the public, but is a reference for the project's producer, engineer, and musicians.

RPM. Revolutions Per Minute. The number of times a disk or other item that spins around an axis rotates in a minute. This is an important spec for hard drives; the faster the disc revolves, the shorter the time the read or write head has to wait to reach a specific location or piece of data.

RPN. ☞ See *registered parameter number*.

RS-232. Recommended Standard 232. A telecommunication protocol developed in 1969 by the EIA (Electronics Industry Association) and commonly used for personal computer serial ports. RS-232 is intended for short- to medium-distance (up to 15 meters) serial bidirectional data transmission and uses DB-25 or other multi-pin connectors.

RS-422. Recommended Standard 422. A balanced telecommunication protocol developed in 1978 by the EIA (Electronics Industry Association). RS-422 is used for long-distance serial data transmission over twisted-pair balanced lines.

RTA. ☞ See *real-time analyzer*.

RTAS. Real-Time AudioSuite. A real-time native plug-in format developed by Digidesign for use in DAE-compatible host audio software, such as Pro Tools HD, Pro Tools LE, Pro Tools M-Powered, and MOTU Digital Performer. The RTAS format supports both processing and virtual instrument plug-ins.

RT60. Abbreviation for Reverb Time-60 dB. The time it takes for the reverberation in a room to drop in level by 60 decibels.

RTZ. Return To Zero. A transport control that rewinds a recording to the zero point on a time or location counter.

rumble. Low-frequency noise, typically below 50 Hz, often caused by mechanical sources such as traffic, HVAC systems, or trains, or by a playback/recording transport, such as a turntable. In many cases, rumble can be removed using a high-pass filter.

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S

sabin. A measure of how well a surface or material absorbs sound per square foot. One sabin indicates complete absorption, with the usual reference being an open window. Sabin values for various materials range between 0 (perfectly reflective) and 1 (completely absorptive). Named for Wallace Sabin, an American physicist who studied acoustics in the late 1800s and early 1900s. He developed a measurement called the OWU (Open Window Unit) that became known as the sabin.

SACD. Super Audio Compact Disc. A consumer audio optical disc format developed by Sony that uses the Direct Stream Digital (DSD) technology developed by Sony and Philips. SACD uses a CD-sized disc and supports stereo and multichannel audio at sample rates up to 100 kHz, dynamic range of 120 dB, and 1-bit resolution. Though a large number of titles have been released on SACD (some having multiple layers also containing regular CD-compatible audio), the format has not seen large acceptance from the general public. ☞ **See also** *Direct Stream Digital*.

sample. 1. A single digital measurement of an audio signal. These measurements are stored as digital words with a certain bit depth or resolution. 2. The act of digitally recording an analog signal. 3. Digital audio in a sampler, sample-based synthesizer, or other device, such as a single sampled piano note, drum hit, or other sound used as part of a program or preset. 4. A generic reference to a sample-based preset in a sampler, drum machine, or other device, though that preset may in fact contain many individual samples (for example, "That's a great piano sample!").

sample-accurate. The ability of a device to synchronize or operate with sample-to-sample accuracy.

sample and hold. A circuit or module found in analog synthesizers that captures a voltage, then routes that voltage to control some parameter. (Filter cutoff frequency is a common control destination.) The "sampled" voltage remains constant until the next voltage is captured. This results in random voltages modulating a parameter in "step" fashion, jumping from one voltage immediately to the next.

sample CD. A collection of audio tracks, loops, or samples/sample-based programs on compact disc, intended to be used with a sampler, DAW, sampling drum machine, sample-based synthesizer, or other device.

sample clock. A clock that determines and drives the sample rate. ☞ **See** *word clock*.

sample dump. Transmission of the sample memory of one device into another device over MIDI connections. ☞ **See** *Sample Dump Standard*.

Sample Dump Standard (a.k.a. SDS). A standardized protocol for sending the contents of the sample memory of one device to another via MIDI. Sample dumps are commonly used to transfer the memory of a sampler into a computer for editing, or to transfer samples from a computer to a sampler. It can also be used to share samples among any number of samplers.

sample library. A collection of samples, whether for a hardware or software sampler, usually offered on CD-ROM or DVD-ROM by a soundware manufacturer. Sample libraries can range from a collection covering a particular category, such as an orchestral string library, to sound effects, to a general-purpose library that covers instruments, loops, and more.

sample offset uncertainty. ☞ **See** *jitter*.

sample-playback synthesis. A type of synthesis that is based on sampled waveforms or instruments rather than simple analog or digital waves. Because the sampled waveforms are generally stored in the synthesizer's ROM, this type of synthesizer is also known as a "ROMpler." ☞ **See also** *ROMpler*.

sampler. A hardware device that can capture digital audio recordings, distribute them across a key map or keyboard, and play them back at different pitches. Though a number of virtual instruments are referred to as samplers, most of these don't actually "sample" in the traditional sense of taking samples of an analog signal.

sample rate. How many times per second samples are taken, transmitted, or played back by a sampler, processor, digital recorder, DAW, other digital audio device. Sample rate is generally given in kilohertz for PCM devices or in megahertz for DSD devices, though S/s (samples per second) is a more accurate measurement. Sample rate determines frequency response, as defined by the Nyquist Theorem, which says that the highest frequency that can be sampled is one half the sample rate. The standard sample rate for compact discs is 44.1 kHz. DAT and some other digital recorders operate at 48 kHz. High sample rates are two or four times the standard rates: 88.2, 96, 176.4, and 192 kHz.

sample rate conversion. The process of reducing or increasing the sample rate of a signal to another rate. Usually this is done in the digital domain, though it would, of course, be possible to convert the digital signal to analog, then resample at a different rate. The challenge is to make the conversion without changing pitch or sound quality. In some cases, the conversion was considered relatively easy—converting a 96-kHz signal to 48 kHz, for example. But the reality can be more complex, such as when converting from 96 kHz to 44.1 kHz or another non-integer multiple. Historically, sample rate–converted audio was considered to be of lower quality than the original source material, but some current algorithms are capable of extremely high-quality conversions.

sample rate converter. A hardware or software processor that converts digital audio recorded at one sample rate to a higher or lower sample rate. ☞ **See also** *sample rate conversion*.

sample word. A single sample or measurement of an audio signal represented as a group of bits. The number of bits equals the resolution of the sample. In a 16-bit system, a sample word would contain 16 bits, a 24-bit system uses words with a 24-bit length, and so on.

SATA. Serial ATA. A protocol for transmitting data to ATA drives serially at high speed, rather than in parallel fashion as in standard ATA. SATA allows for point-to-point connection of drives, eliminates master/slave drive settings, and simplifies termination issues. SATA supports cables up to one meter long, allowing for external drives. ☞ **See also** *ATA*.

satellite. A small speaker that is intended to be used with a subwoofer. The satellites produce the mid-range and high frequencies, while the larger sub produces the low frequencies.

saturation. Originally, saturation was the maximum amount of magnetism that could be put on an analog tape; any more would result in distortion. Engineers carefully managed the levels going to tape for the compression, warmth, and fatness that came from approaching or reaching tape saturation. Saturation has come to be used as a term for overload distortion, whether in recording, guitar amps, or elsewhere.

sawtooth waveform. A cyclical waveform whose shape resembles a sawtooth. A sawtooth wave consists of a fundamental frequency and all (both odd and even) integer harmonics (see Figure S.1). The sawtooth wave is one of the basic waveforms available in analog synthesis (other common waves in analog synthesis include sine, triangle, and square). Sawtooth waves have a harsh sound, somewhat similar to a bowed stringed instrument.

Scarlet Book. One of a set of "Rainbow Books" with colored covers containing the Sony/Philips-developed specifications for different optical compact disc formats. The Scarlet Book contains the specifications for SACD.

scene (a.k.a. snapshot). A memory location in a digital or digitally controlled mixer or other piece of gear that stores all the settings for the parameters in the device. A scene can be recalled, instantly resetting all the parameters to the values stored with the scene.

Schroeder diffusor (a.k.a. reflection phase grating). A Schroeder diffusor consists of a variety of wells of different depths but equal widths that scatter sound waves. The frequencies at which a Schroeder diffusor will operate depends on the depth of the wells; the wavelength of the lowest frequency the diffusor will affect is four times the depth of the deepest

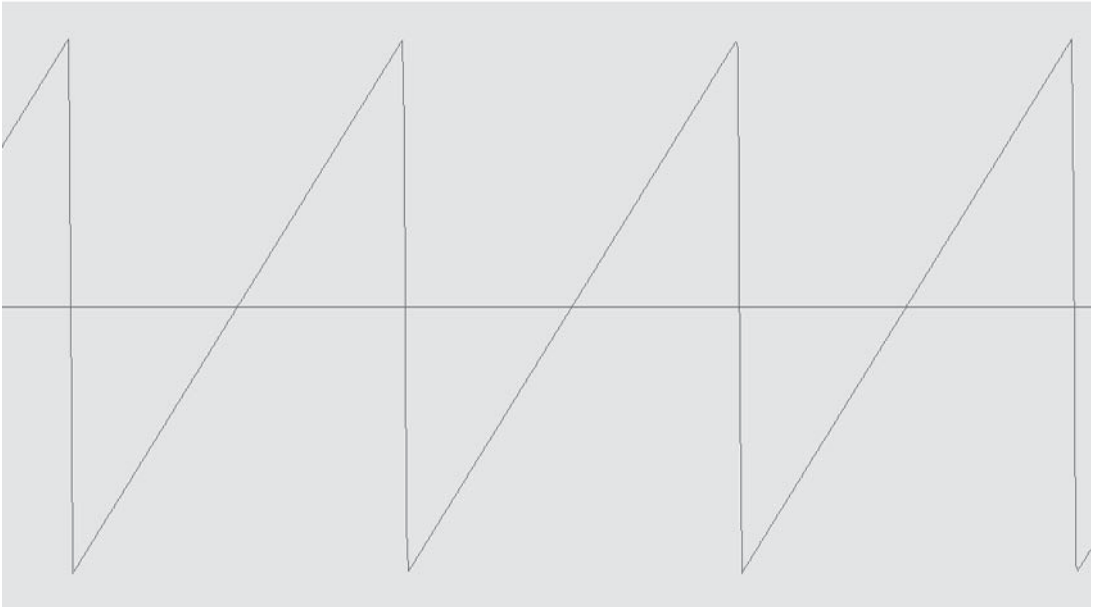

Figure S.1 A sawtooth waveform contains a fundamental frequency plus all odd and even integer harmonics.

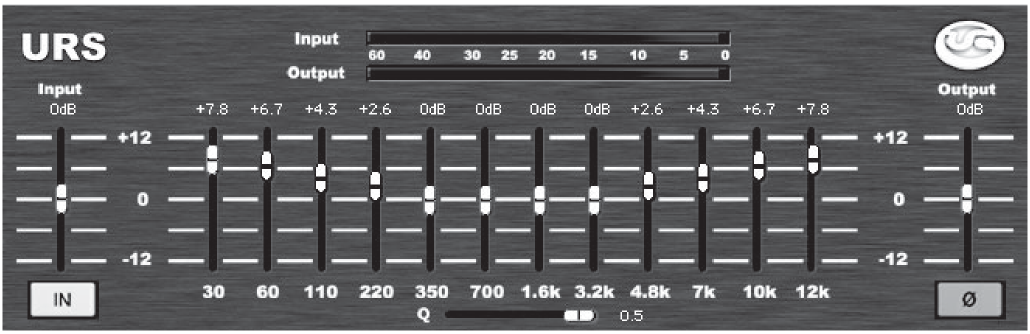

Figure S.2 Cutting the midrange frequencies and either boosting or letting the high and low frequencies pass unchanged results in scooped mids.

well. Named for Manfred Schroeder, a German professor of acoustics who discovered how to use number theory to determine the depth of the diffusor wells. The best known Schroeder diffusor is the QRD (Quadratic Residue Diffusor) type.

SCMS. Serial Copy Management System. A copy protection scheme developed by record labels for DAT recorders. Intended for the S/PDIF ports on consumer and semi-pro DAT machines, SCMS is a bit "flag" that does not allow digital copies to be made of digital copies. So, as many digital copies could be made of an original (first-generation) DAT tape as desired, but no digital copies could be made from any of those copies.

scooped mids (a.k.a. "smiley face" EQ). An equalization curve where the midrange frequencies are attenuated, resulting in emphasized low and high frequencies. Commonly found on consumer stereo systems. The standard EQ setting for certain types of heavy-metal guitar sounds. See Figure S.2.

score. 1. Notated music for a musician or ensemble. 2. The background and feature music in a film (a.k.a. film score) or play. 3. To create the music for a film or play. 4. To arrange or orchestrate a piece of music for an ensemble. 5. In certain software applications, the data required to play a piece of music. ☞ **See also** *score window*.

score window. An alternate way to view MIDI note data in a DAW or MIDI sequencer. A score window displays MIDI notes as standard notation (see Figure S.3). The notation can be edited by inserting, deleting, transposing, moving, or otherwise manipulating the notes. ☞ **See also** *notation editing*.

scratch (a.k.a. scratching). A technique used by DJs, where a vinyl record is manually moved forward and backward during playback to create a scratching sound.

scratch disk. A hard drive or partition used to temporarily hold data.

scratch mix. A rough or quick mix of a piece of music, often created as the tracks are being recorded. A scratch mix is not intended for public release, but is meant to be used by the engineer, producer, and musicians to evaluate performance, arrangement, or mix ideas, or is meant for demo purposes.

scratch track (a.k.a. guide track). A temporary track that is not intended to be used in the final mix. Typically, scratch tracks serve as reference material for overdubbing other parts and are replaced by real tracks later in the production process. An example would be a rough, scratch vocal track recorded so that the instrumentalists have a guide to refer to while recording their parts.

scratch vocal (a.k.a. guide vocal). A rough, temporary vocal track meant to serve as a reference when recording other parts to the song. The scratch vocal will be replaced with a final "keeper" vocal track later in the recording process. ☞ **See also** *scratch track*.

screensaver. A computer utility or part of the operating system intended to prevent an image from burning into a CRT screen when the computer is sitting idle for extended periods of time. Screensavers work by filling the screen with graphic images that randomly move about the screen. Though LCD screens do not suffer as much from burn in, screensavers are still in common use. ☞ **See also** *burn in*.

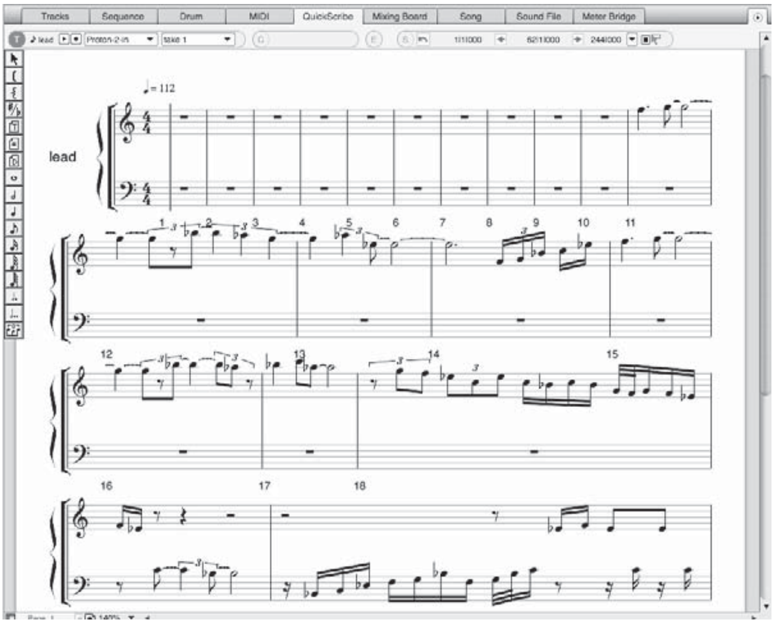

Figure S.3 The score window in a DAW or sequencer displays MIDI notes as standard musical notation. The notes can be easily edited using a mouse.

scribble strip. An area on a mixing console intended for labeling channel strips, aux sends and returns, buses, and other items. On digital and digitally controlled consoles, as well as control surfaces, the scribble strip has become a small LCD screen that displays the channel name and other information.

scroll. To scroll is to move through a window on a computer screen to display text that is too long or graphics that are too wide or long to fit completely within the window.

scroll wheel. A small wheel mounted in a computer mouse that can be rotated to scroll the data, text, or graphics within the active window on the screen. ☞ **See also** *scroll.*

scrub. To manually move backward or forward through a track to find a particular location. The term comes from the manner in which engineers would manually rock reels of analog tape back and forth while listening to the audio, which allowed them to precisely locate a point on the tape for editing.

SCSI. Small Computer System Interface. A hardware interface for connecting hard drives, CD-ROM/CD-R drives, removable drives, and other peripherals to devices such as computers, digital recorders, hardware samplers, and more. SCSI supports up to seven devices in a chain; the chain may be as long as 19 feet. SCSI-1, the original 8-bit-wide spec, has a throughput of 5 MB/second. Fast SCSI increases this to 10 MB/second, and Fast Wide SCSI doubles the bit width to 16 bits, for throughput of 20 MB/second. ☞ **See also** *Ultra SCSI.*

SD card. Secure Digital card. A flash-based memory card. SD cards feature encryption for secure data storage.

SDIF. 1. Sony Digital Interface Format. A digital connection format carrying audio over 75-ohm unbalanced coaxial cables terminated with BNC connectors. The original SDIF was made obsolete by SDIF-2, which used one cable for a single channel of data and another for word clock. For stereo signals, a second data cable is added. SDIF-3 is used for DSD transfers. 2. Sound Description Interchange Format. A standard for exchanging spectral description types developed by IRCAM (*Institut de Recherchet Coordination Acoustique/Musique*), CNMAT (*Center for New Music and Audio Technologies*), and IUA-UPF (*L'institut Universitari de l'audiovisual-Universitat Pompeu Fabra*).

SDII. ☞ **See** *Sound Designer II.*

SDMI. Secure Digital Music Initiative. A voluntary scheme developed by the RIAA for protecting digital music files.

SDRAM. Synchronous Dynamic Random Access Memory. A type of DRAM that can synchronize itself to the bus speed of its host computer. SDRAM is faster than standard DRAM. ☞ **See also** *DRAM.*

SDS. ☞ **See** *Sample Dump Standard.*

SE. Special Edition. A limited, modified, expanded, or extended version of a standard product. A special-edition audio product might contain improved parts, additional capabilities, bundled software or hardware, or other offerings beyond what you would get with the regular version.

sector. A section of a track on a hard drive or other media identified by a specific address. The drive uses the address to locate the sector, which is used to store and read data. Hard drives store from 512 bytes to 4 KB or more of data per sector. The address information is stored in a special sector (Sector 0), in a file known as the FAT (File Allocation Table).

seek time. The amount of time it takes for a device such as a hard disk or optical drive to locate a specific piece of data. With hard disks and optical drives, seek time is the time required for the drive to move its read/write heads from one track to another. Seek time, together with rotational delay and transfer time, comprise a device's access time.

self-noise. The background noise generated by a device's electronic components. Any device or system containing electronics will create some level of background noise. ☞ **See also** *noise floor.*

semi-open. A headphone design that allows some outside sound to pass into the wearer's ears. Although semi-open headphones don't isolate as well as fully closed headphones, many users consider them to be more comfortable to wear and use.

semi-parametric. ☞ **See** *quasi-parametric.*

semi-pro. Short for semi-professional. In audio, products intended for home and lower-level project studio use. The lines between consumer, semi-pro, and pro have blurred, but typically, semi-pro equipment is not as heavy-duty as professional equipment and may have more basic digital connectivity and unbalanced analog connections. Most semi-pro gear also operates at –10 dBv levels, rather than +4 dBu. ☞ **See also** *consumer*, *professional.*

semitone (a.k.a. half step). The smallest interval used in Western music. In the standard Western equal temperament, the octave is divided into 12 equal-sized semitones of 100 cents each. Other temperaments use semitones of varying sizes.

semi-weighted action. A keyboard instrument key action that has some resistance to being pressed—more than an unweighted synth or organ key, but not as much as a weighted or hammer-based piano-style action. ☞ **See also** *action*.

semiconductor. A solid material, such as silicon, germanium, and gallium arsenide, with limited ability to conduct current. Semiconductors are used to create electronic components, such as transistors and diodes. In some cases, such as transistors, the conductivity of the device can be controlled by an external voltage, such as an analog audio signal. ☞ **See also** *diode*, *transistor*.

send. ☞ See *aux send*.

sensitivity. 1. The amount of voltage a microphone produces for a given sound pressure level, usually specified in millivolts per Pascal (mV/Pa). 2. The input level necessary to drive a device to its rated output level. Higher sensitivity makes a device better able to deal with low-level signals, but may result in overload with louder signals.

sequence. 1. The act of recording MIDI data. 2. A recording of MIDI data that can be played back to control MIDI hardware or software synthesizers, samplers, or processors.

sequencer. A hardware device or computer software that can record, edit, process, manipulate, create, and play back MIDI information. Many of the popular commercial sequencers have evolved into DAWs with the addition of support for audio recording, editing, processing, and mixing features.

serial. 1. Data that is transmitted in sequential fashion, in a single stream arranged with one piece of information following another. 2. Events that occur one after another. ☞ **See also** *series*, *parallel*.

serial ATA. ☞ See *SATA*.

serial number. A unique number assigned to a piece of gear or software by the manufacturer for identification purposes.

series. 1. An arrangement of items one after another. In audio, each signal path is often arranged in series format, with the first piece of gear feeding the second, which feeds the third, which feeds the fourth, and so on. 2. A category or line of products that share similar features. 3. A musical tone row or arrangement of pitches. ☞ **See also** *parallel*.

server (a.k.a. file server). A computer or program that functions as a central repository for files and makes them available to other programs or computers on a network.

session. 1. Short for recording session. The time when musicians are working in a studio. 2. In certain audio applications, such as Digidesign Pro Tools, the master project document.

SFX. Short for special effects.

shared library. The Macintosh term for a DLL. A collection of software resources that is available to other programs.

shareware. Copyrighted software created by an individual or company and offered for free download. Typically, the user is granted a trial period. If the user continues to use the software after the end of the trial, the software creator requests a goodwill offering of cash to help defray development and other expenses. ☞ **See also** *freeware*.

shedding. 1. Short for woodshedding. Practicing a musical instrument. 2. A problem exhibited by some recording tape, in which oxide flakes off of the tape backing. A variety of factors affect how much a given piece of tape sheds, including age, quality of the tape, how often the tape is played, how the tape is stored, and more.

shelf. An equalizer or filter band that cuts or boosts all the frequencies above or below the cutoff frequency. See Figure S.4.

shell. 1. A format converter that allows a host program to use unsupported plug-in formats. ☞ **See also** *wrapper*. 2. The layer of an operating system that accepts user input and provides user feedback.

shelving equalizer. An equalizer that uses shelving bands. ☞ See *shelf*.

shelving filter. A filter that uses shelving bands. ☞ See *shelf*.

shield. 1. Conductive material that is used to protect audio conductors and circuits from magnetic and electrostatic fields. For example, the signal conductors (internal wires) in audio cables are typically wrapped in metal foil or braided wire shielding to prevent them from picking up hum or other noise. 2. Magnetic material used to protect a CRT display from interference from other magnets, such as those in speakers. LCD and plasma displays are not affected by magnetic fields and require no shielding.

shielding. ☞ See *shield*.

shockmount (a.k.a. suspension mount, suspension basket). 1. Microphone mount designed to protect a mic from stand-borne vibration or impacts (see

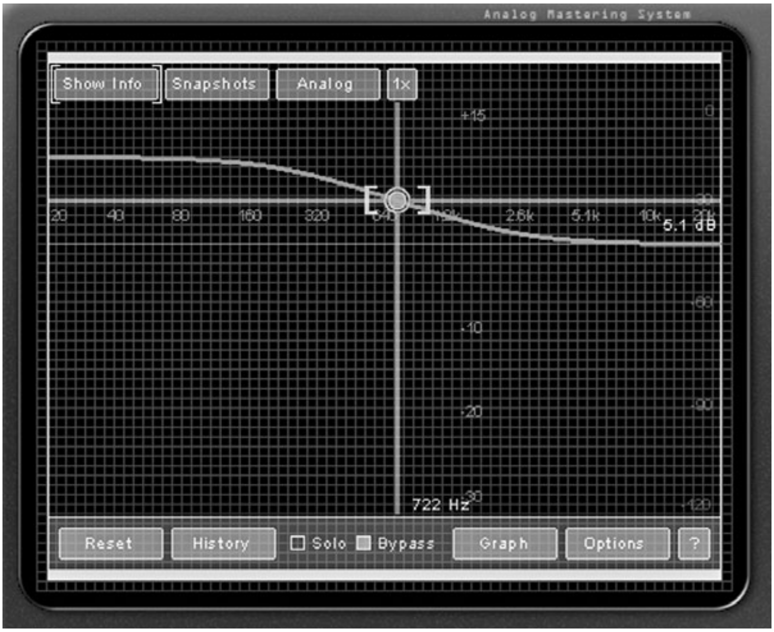

Figure S.4 A shelving EQ boosts or cuts all frequencies above a certain frequency.

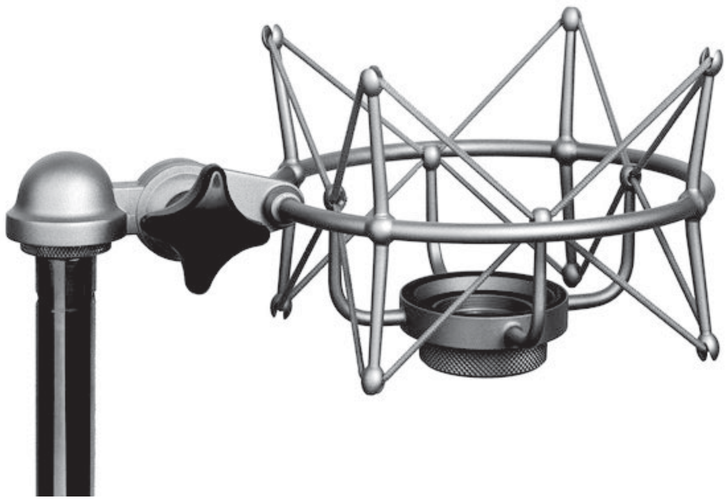

Figure S.5 A microphone shockmount isolates the mic from stand-borne vibrations or impacts.

Figure S.5). A common type, sometimes referred to as a *spider mount*, uses a system of rubber or elastic bands to isolate the microphone from the stand. 2. Isolated rack rails or "case-within-a-case" rack designs that protect rack-mounted devices from vibration and impacts.

shootout. Attempting to compare the quality and performance of a number of similar devices in order to find the best one for a particular application. Shootouts are very difficult to conduct, because it is essential to control the test conditions and parameters precisely for true results.

short. ☞ **See** *short circuit*.

short circuit. A failure in an electronic circuit, usually resulting from unintended contact between components or wiring.

shortcut. In the Windows operating system, an icon that points to another file. Double-clicking a shortcut finds the file the shortcut points at and launches it. Shortcuts are convenient for organizing files and folders. Known as an *alias* in the Macintosh OS.

shotgun microphone. A type of microphone featuring an extremely directional polar pattern. Most shotgun mics use long tubes to create acoustic phase cancellation that allows them to capture distant sounds without picking up surrounding sounds and noises. ☞ **See also** *lobar polar pattern*.

shuffle. A playback mode found on MP3 players, CD players, and other audio devices that plays the tracks back in random order.

shuffler. An electronic circuit used to decode MS signals to stereo, first described by Alan Blumlein and later expanded by Michael Gerzon. The shuffler combines the MS mid and side signals to create two results: the sum of the signals (A+B), which is created by mixing the signals, and the difference between the signals (A–B), which is created by phase reversing the B signal. There are a variety of ways that the sum and difference results can be panned and equalized to change the stereo width (sometimes referred to as *spatial equalization*) or to compensate for problems. ☞ **See also** *MS stereo*.

shuttle. A control on some tape decks, video decks, and control surfaces that serves as a manually controlled fast-forward and rewind function. Shuttle allows the engineer to quickly locate to a point in the track.

sibilance. The term *sibilance* comes from the Latin word for hissing. It refers to high-frequency vocal sounds, such as *ss* and *sh*, that live in the 5- to 10-kHz range. A de-esser is used to control excess sibilance in a vocal without removing high frequencies from the rest of the track.

side address. A microphone physically designed so that its diaphragm is parallel to the body of the mic, with the capsule oriented to pick up sound best from the side or sides of the mic, rather than from the end (see Figure S.6). ☞ **See also** *end address*.

sideband. A sum or difference frequency created when a sound wave (the carrier) is frequency modulated (FM) or amplitude modulated (AM). The upper sidebands result from the carrier and modulator adding, while the lower sidebands result from the difference between the carrier and modulator. For example, a carrier with a frequency of 1,000 Hz and a modulator with a frequency of 300 Hz would result in sidebands at 1,300 Hz (sum) and 700 Hz (difference). FM synthesis uses varying arrangements of carrier and modulator signals to create sidebands, which are shaped using different techniques to generate sounds.

sidechain (a.k.a. key, key input, or detector). An audio input used to trigger a compressor or noise gate's operation. The audio signal coming in the device's sidechain isn't audible (at least not through the device, though it may be heard on another track or input); rather, the sidechain signal tells the compressor to process or the gate to open. A common use for a sidechain is to cause a compressor to reduce the level of background music in an ad when the voiceover comes in (referred to as *ducking*). Other uses include de-essing, dynamic equalization, special gating effects, and more. ☞ **See also** *de-esser*.

Figure S.6 A side-address microphone picks up sound best from the side, rather than the end. In this case, you can see the round diaphragm behind the screen of the microphone.

signal. 1. An electrical impulse, voltage, or current used to represent information. 2. In analog audio, a voltage representing an audio waveform.

signal path. 1. The route a signal takes through a series of pieces of equipment. 2. The devices an audio signal passes through while being recorded, mixed, or processed in a studio.

signal-to-noise ratio (a.k.a. SNR, S/N). 1. The ratio between the level of a desired signal and the self-noise of a piece of equipment. 2. In manufacturer spec terms, the difference between a device's noise floor and its nominal, or normal, operating level. (Manufacturer signal-to-noise ratios are notoriously unreliable, due to different measurement methods and other factors.) The highest possible signal-to-noise ratio a device can have is equal to its dynamic range, though generally the ratio is based on a reference level lower than the maximum undistorted level possible.

silica gel. A desiccant, or moisture-absorbing compound, used to control humidity in packages containing sensitive electronic gear. (Those little white paper packets that are forever falling out of gear boxes. . .)

silk-dome tweeter. A high-frequency driver featuring a central dome made from silk cloth treated to be stiff enough to shape. Some listeners feel that silk-dome tweeters offer smoother sound, better damping, and less resonance, while others hear little or no difference when comparing them to metal-dome tweeters.

SIMM. Single In-line Memory Module. A type of SDRAM circuit board containing multiple memory chips allowing for easy installation. There were several different versions, such as 30-pin and 72-pin, composite and non-composite, different parities, all available in a range of total memory capacities. SIMMs were used in computers and hardware samplers, but have been replaced by DIMMs (*Dual In-line Memory Modules*) for most applications because SIMMs have a 32-bit data path used in older processors, whereas DIMMs have a 64-bit data path that is compatible with modern computers. ☞ **See also** *SDRAM*, *DIMM*.

simplex. A circuit or interface that allows one-way communication or transmission. Note that simplex is not the same as half-duplex, which allows for two-way communication, one direction at a time. A great example of simplex communication is a radio or television broadcast. ☞ **See also** *half-duplex*, *full-duplex*.

sine wave. A type of cyclical waveform containing a single frequency, the fundamental, with no harmonics or overtones. (See Figure S.7.) A flute is sometimes used as a sine wave, though in fact it contains a wide range of ultrasonic harmonics.

single-ended. A type of amplifier in which a single tube or transistor amplifies the entire signal. Single-ended amps are easier to design and use fewer

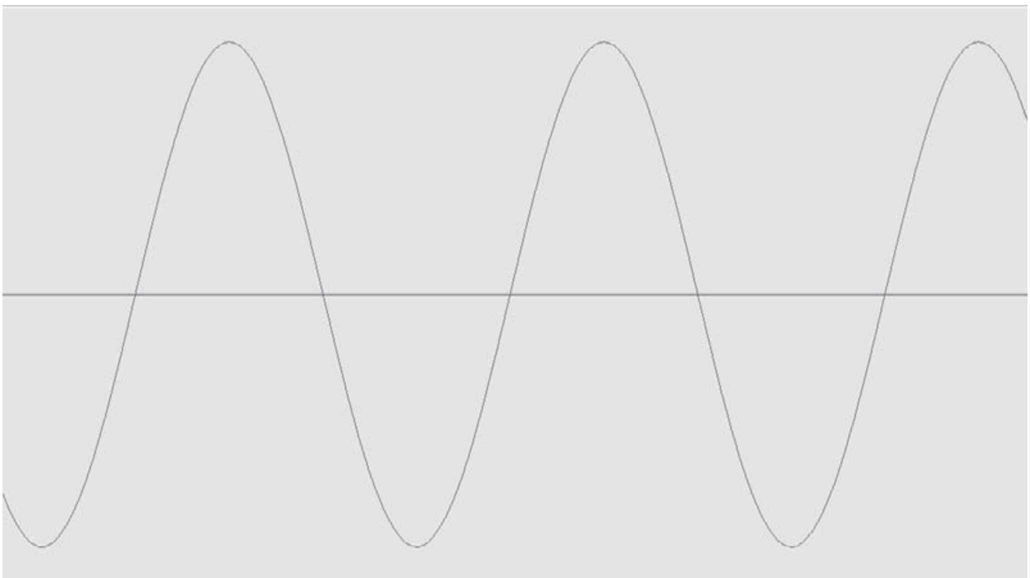

Figure S.7 A sine wave is the simplest waveform, containing just a fundamental frequency without any harmonics.

components, but tend to have low power outputs. ☞ **See also** *amplifier*, *Class A*, *Class A/B*, *Class B*.

skin effect. The tendency of high-frequency signals to travel on the outer surface of a conductor.

skip-back sampling. A feature of some Roland samplers designed to prevent ideas from being lost. With skip-back sampling, audio is always being recorded, so you can go back and capture a certain amount of recently played material.

skirt. Frequencies beyond the selected band processed by an EQ, filter, or other device that are low in level, but still are processed. (See Figure S.8.) For example, in most EQs and filters, the processed band is determined by where the signal is 3 dB down from the rest of the signal. The skirt in this case would be the frequencies that extend past the 3 dB down point. The skirt is still affected by the processor but is not considered part of the bandwidth. ☞ **See also** *3 dB down point*.

slapback. 1. A short echo in a room resulting from a sound wave reflecting between parallel surfaces. You can test for slap echo by clapping your hands and listening for a discrete echo. 2. A popular echo effect on 1950s-era recordings, also popular on rockabilly and other styles of music. Slapback is a single delay repeat in the 40 to 150 ms range, set to be nearly the same volume as the original.

slap echo. ☞ See *slapback*.

slate. From the video term for the clapper used to identify a scene and sync point at the beginning of a shot. Audio slates are recorded at the beginning of tracks as either tones or vocal identification.

slatted absorber. ☞ See *Helmholtz absorber*.

slave. 1. A device that is under the control of another device. 2. To synchronize with, lock to, or chase a master sync device. 3. A device that is locked to and chasing a master device.

slave reel. When recording with analog tape, the only way to add tracks beyond the capacity of a single deck is to add a second slaved deck. To avoid the hassles of locking two decks together, a submix of the existing tracks is created on a new tape, known as a *slave reel*. Overdubs are then added to the slave reel and later mixed with the original reel. In DAW systems, slave reels have been replaced by virtual tracks.

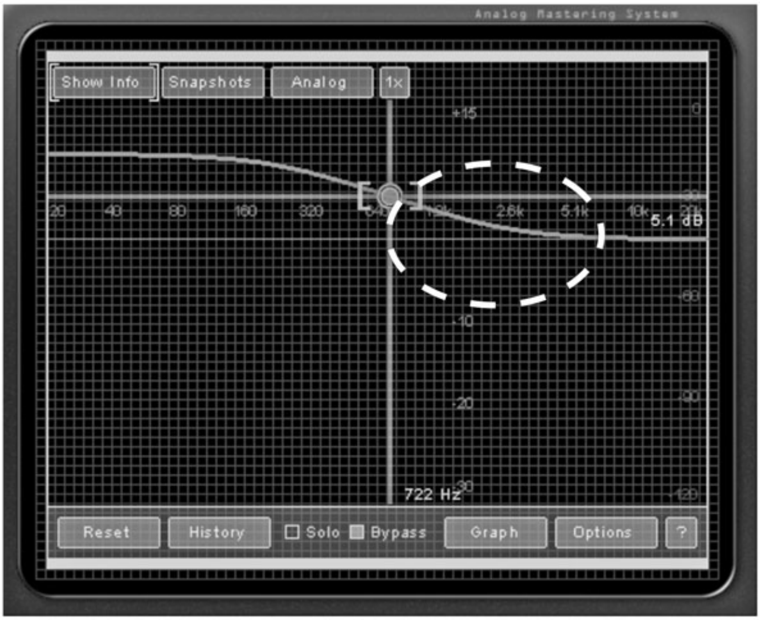

Figure S.8 The skirt contains frequencies that extend beyond the bandwidth selected. In this case, the dashed circle shows the skirt that extends beyond the cutoff frequency of a low-shelf filter.

sleep. A standby mode available on some devices, such as computers. Sleep mode turns off the video display, stops the hard drive, and turns off other active features. Power to the computer's memory is retained so that applications can be left active and ready to go when the computer "wakes up."

slew rate. The speed with which an amplifier or certain other types of gear can respond to changes in the input signal's amplitude. Slew rate is important for accurately reproducing transients, dynamics, and high frequencies. Measured in volts/μs (microsecond).

slice. In the REX files created by Propellerhead

Figure S.9 In a REX file, the audio is chopped into rhythmic slices so that it can be played back faster or slower or otherwise manipulated.

ReCycle and other programs, the audio files aren't actually stretched to change their tempo. Instead, the audio is chopped into small, rhythmic slices that can be played back at a faster or slower rate to change tempo, or certain slices can be deleted, copied, moved, re-ordered, or processed to change pitch or other parameters. See Figure S.9.

slip cue. A DJ technique in which the stylus is placed on a vinyl LP, but the record is held stationary while the turntable continues to spin. The LP is released at the proper time and immediately begins to play.

slope. The rate at which frequencies past the cutoff frequency are attenuated by a filter. Slope is given in dB/octave. The order, or the number of poles in the filter, determines the slope, with each pole providing 6 dB/octave in attenuation. As an example, if the cutoff frequency is 5,000 Hz and the slope is 12 dB/octave (two poles), in the octave from 2,500 Hz to 5,000 Hz, the level would be reduced by 12 dB. The octave from 1,250 Hz to 2,500 Hz would be reduced by an additional 12 dB, from 625 Hz to 1,250 Hz another 12 dB, and so on. See Figure S.10.

slot. ☞ See *expansion slot*.

small-diaphragm microphone. A microphone containing a diaphragm smaller than 5/8-inch or so. Small-diaphragm mics tend to respond well to high frequencies and transients due to the lower mass of the smaller diaphragm compared to a larger diaphragm. Small-diaphragm mics are commonly used for recording acoustic instruments, as drum overheads, for recording small and large ensembles, and other applications.

small-format mixer. A compact, portable mixer typically containing 16 or fewer input channels.

Smart FSK (a.k.a. SFSK, Smart Frequency Shift Key). An FSK signal that has added location information and functions more like true time code. Smart FSK allows the sequencer to locate to a particular time within a sequence. ☞ See *FSK*.

SmartMedia. A type of flash-based memory card developed by Toshiba, originally intended to replace the floppy disk and called SSFDC (Solid State Floppy Disk Card). The SmartMedia format was replaced by SD, xD (a.k.a. XD or Xd), and other card formats.

SMDI. SCSI Musical Data Interchange. A fast method developed by Peavey for transmitting the contents of the sample memory of a sampler to a computer using SCSI. SMDI has the advantage of being *much* faster than the MIDI Sample Dump Standard.

SMF. ☞ See *Standard MIDI file*.

smoothing. Interpolating between discrete digital values (which create a stair-stepped signal) to create a smooth, more analog-like signal. Smoothing can be applied to digital audio signals by a reconstruction filter in a digital-to-analog converter; it can be applied to MIDI controller messages, automation, or to any digital data.

smoothing filter. ☞ See *anti-imaging filter*.

SMPTE. Society of Motion Picture and Television Engineers. A professional organization for television and film audio engineers. Along with providing services to its members, SMPTE has helped develop measurement standards and the SMPTE linear time code system for synchronization. www.smpte.org.

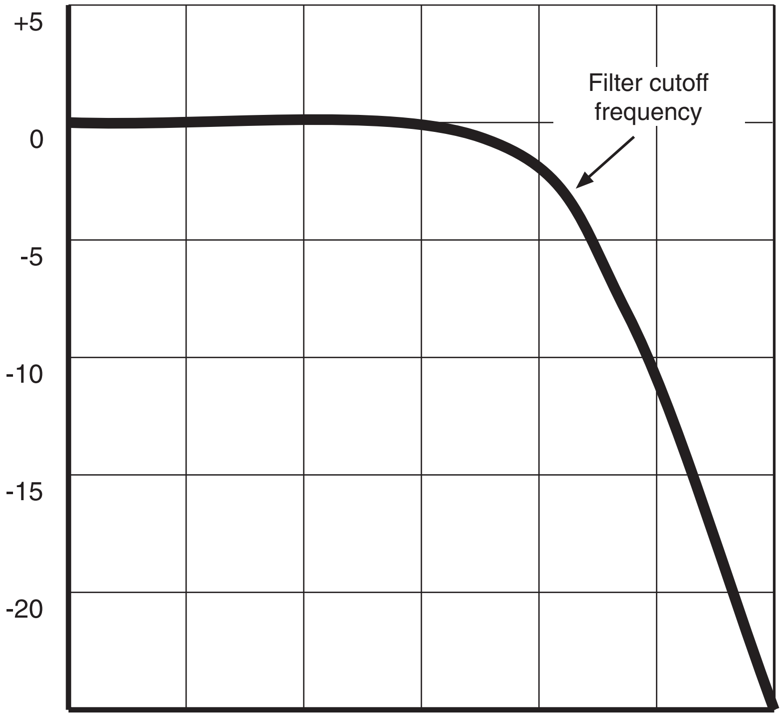

Figure S.10 The slope of a filter is the rate at which the frequencies past the cutoff are reduced in level.

SMPTE time code. An LTC (Linear Time Code) system developed by the SMPTE organization and used for synchronizing audio, video, synths, drum machines, and sequencers. (Trivia note: SMPTE time code was based on a system developed by NASA for tracking space travel.) SMPTE time code is based on video frames and a 24-hour cycle. For example, 01:02:52:27 means 1 hour, 2 minutes, 52 seconds, 27 frames. So, each frame and subframe has a specific address allowing different devices to be synchronized and specific points to be quickly located. Various frame rates are supported, including 24, 25, 29.97, and 30 frames per second.

S/MUX. Sample Multiplexing. A technology developed and licensed by Sonorus for transmitting high-resolution digital audio signals over low-resolution connections. S/MUX is similar to bit splitting, where a high-resolution audio signal is divided into two or more parts so it can be carried over a lower resolution connection. For example, a two-channel 24-bit/96-kHz digital audio signal might be divided so that it could carried or recorded on four 16-bit/44-kHz channels or tracks. At the destination or on playback, the stereo 24-bit/96-kHz signal would be reassembled. The primary difference between bit splitting and S/MUX is that S/MUX transmits each sample intact; it never divides samples. This allows certain types of processing to be applied to the split signals before they are reassembled. ☞ **See also** *bit splitting*.

snake. A multichannel audio cable. The traditional use for snakes has been to transmit a large number of signals from the stage to the mix position in live-sound venues. In this case, there is a stage box containing connection points for cables coming from mic- and line-level sources onstage. A long multi-core cable containing many separate internal cables runs from the stage box to the mix position, where the snake breaks out into separate connectors that plug into the mixer. Snakes are used in studios for connecting multitrack recorders to mixers or patch bays, audio interfaces to mic panels —anywhere a number of connections would ordinarily require separate cables to be run from place to place. It's much more convenient and neat to run one snake cable than to run multiple separate cables.

snap. ☞ See *snap to grid*.

snap to grid. A type of quantization where events such as MIDI notes and audio regions are moved to the nearest location on a rhythmic grid.

snapshot. ☞ See *scene*.

snapshot automation. A type of automation found on some digital mixers, standalone digital audio workstations, and multitrack recorder/mixers, where automation commands recall snapshots. Snapshot automation does not provide continuous linear control over parameters, but it is a memory-efficient way to quickly change the parameter settings for a device. ☞ **See also** *scene*.

soffit. The underside of an architectural feature such as an arch, ceiling, or roof overhang. In studios, a soffit constructed on the front wall is often used to house the main monitor system.

soft clipping. Clipping is distortion that occurs when a signal's level exceeds a device's headroom. The signal can't get any louder and is literally squared or clipped off. Soft clipping smoothes out the sharp "corners" of the clipped waveform, producing a gentler distortion that is less harsh and easier for high-frequency drivers to deal with.

soft knee. A feature of some compressors that gently begins to introduce compression as the signal approaches the threshold. This differs from standard, hard-knee compression, where the compression ratio is fully applied to the signal as soon as the threshold is crossed. Soft-knee compression is generally less audible, especially at high compression ratios. See Figure S.11.

soft limiting. A limiter that uses a soft-knee threshold approach to make the onset of limiting less audible. Soft limiters are often used to raise the average level of an audio signal without clipping or audible limiting. ☞ **See also** *soft knee*.

soft reset. A function of some devices that simulates powering the device off and back on without actually turning off the power. A good example is restarting a computer after a crash or other error. A soft reset may not clear all battery-backed settings and parameters; a hard reset (turning the power off and back on) is required to completely restore a device to its default state.

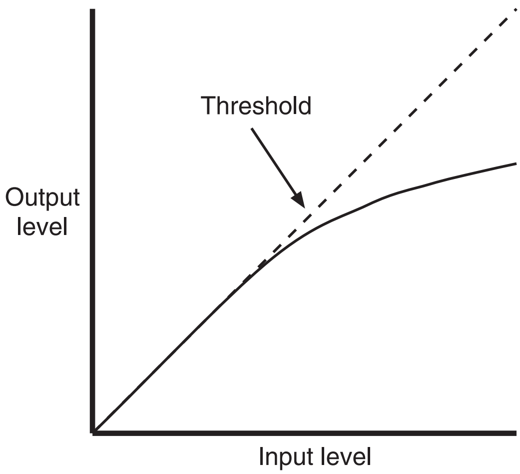

Figure S.11 Soft-knee compression is less audible, especially with higher compression ratios.

soft sampler. Short for software sampler. A virtual instrument that provides the capabilities of a hardware sampler. However, in many cases soft samplers don't actually sample; they're limited to loading and playing back samples and programs that use samples. Soft samplers offer several advantages over their hardware siblings, including more available RAM (depending on how much RAM the host computer contains); more polyphony (limited by the computer's CPU); easy graphic editing of samples, keymaps, and programs; and streaming of huge samples that would be too large to fit into RAM from hard disk. See Figure S.12.

soft synth. Short for software synth, a.k.a. virtual instrument. A computer program that functions as a synthesizer or uses modeling to emulate a hardware synthesizer. As the power available in computers has increased, the quality and capabilities of soft synths have dramatically improved. Software synths may operate as standalone operations in the computer or function as plug-ins within a host program. Advantages of soft synths include high polyphony (limited only by available CPU power), storage and recall of all settings, automation of parameters, no physical space requirements or weight, and lower cost. See Figure S.13.

soft thru (a.k.a. software thru). A function in MIDI sequencers that routes MIDI data coming into a computer's MIDI interface input back out to the interface's MIDI out.

software. A set of specialized commands and operating instructions that tell a computer how to perform various operations and functions.

software patch. A piece of software intended to fix problems, add capabilities, or update a computer program.

solder. A metal alloy that melts at low temperatures and is used to mechanically and electrically connect components.

solid state. An electronic circuit constructed from solid semiconductor components rather than vacuum tubes.

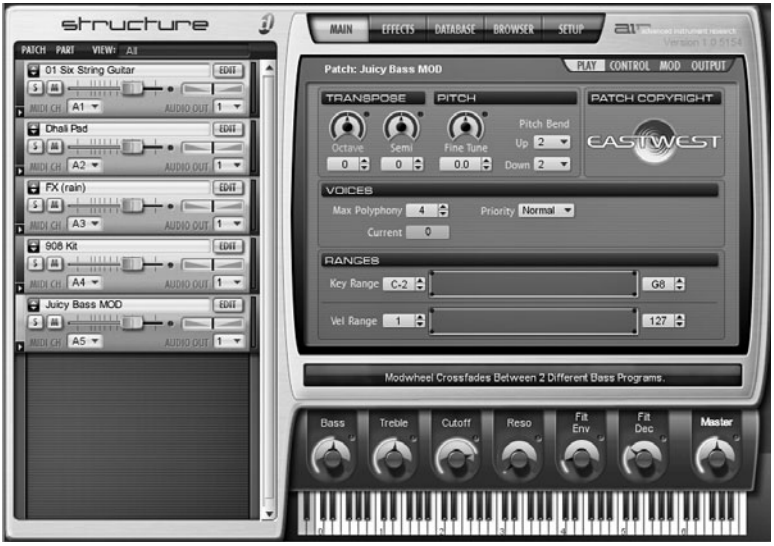

Figure S.12 Structure from Digidesign is a software sampler that offers powerful capabilities to Pro Tools users.

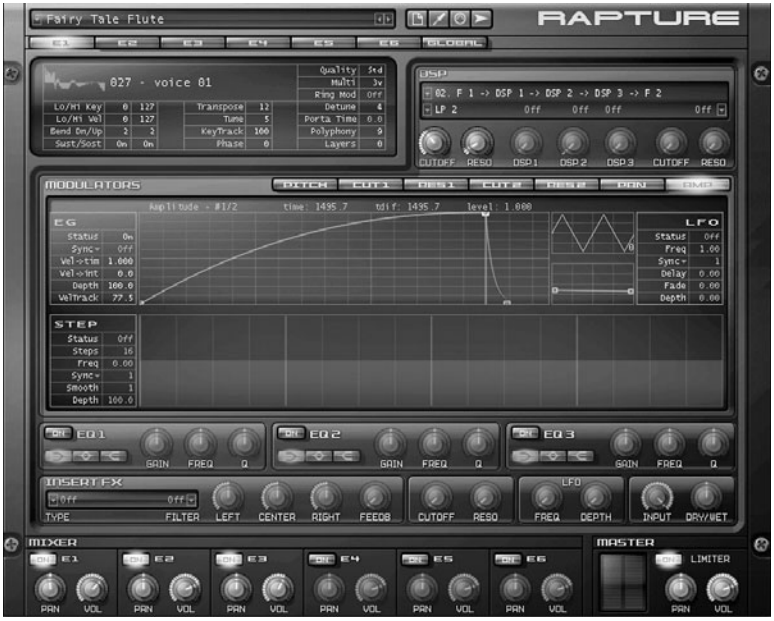

Figure S.13 Cakewalk's Rapture is a software synthesizer that offers an array of sound design and performance options.

solo. A function on a mixer that allows the engineer to monitor only the selected channel(s). Soloing mutes all the channels except the soloed channels without disturbing any other mixer settings.

solo safe. A function on a mixer that prevents a channel from being affected by the solo setting. A channel that is solo safe is not muted when another channel is soloed. This is useful for keeping effects returns and submix inputs from being muted when various channels are soloed.

Song Position Pointer (a.k.a. SPP). A MIDI System Common message that indicates how many sixteenth notes have elapsed since the beginning of a song. (This assumes the original resolution of 96 MIDI clock ticks or pulses per quarter note or six clock ticks per sixteenth note.) Song Position Pointer messages are not used to synchronize devices; rather, they allow a sequencer to locate to a specific point in a sequence. The Song Position Pointer message uses two data bytes that allow it to measure up to 16,384 values, which equals 4,096 quarter notes or 1,024 measures in 4/4 time.

Song Select. A MIDI System Common message that is used to tell a sequencer or drum machine to recall a particular song and prepare it to play.

sonogram. ☞ See *spectrogram*.

sonograph. Instrument that creates a spectrogram or sonogram.

sostenuto. 1. A foot pedal found on a grand piano. A sostenuto pedal causes notes already being held when the pedal is pressed to sustain after the notes are released, but does not cause notes played after the pedal is pressed to sustain. 2. MIDI Continuous Controller #66. The MIDI sostenuto controller turns on with a MIDI value of 127, off with a MIDI value of 0, and (unless remapped by the user) performs the same function as a piano's sostenuto pedal.

sound. Vibrations/air pressure variations with frequencies in the range of human hearing.

sound barrier. 1. Material designed to stop the transmission of sound waves. 2. Also known as Mach 1. The point at which an object moves from transonic (speeds just above and below the speed of sound) to fully supersonic speed.

sound bite. A term used in some DAW programs to refer to a small piece or chunk of digital audio data.

soundcard. A board or card that mounts in an expansion slot inside a computer and provides analog and/or digital audio input and output capabilities. Soundcards may also have mic-level inputs, MIDI inputs and outputs, and other features. The term "soundcard" is more commonly used to refer to Windows-platform consumer audio cards, while semi-pro and professional units for Windows and Macintosh computers are generally labeled as "audio interfaces."

sound check. A period of time before a performance when the setup and operation of a sound system is evaluated. This can range from testing microphones to setting levels to doing a trial performance by the full ensemble for the purposes of setting up EQ, effects, and monitor levels.

Sound Designer. A digital audio application created by Digidesign. Originally developed for sample editing, Sound Designer evolved into the software component of the Sound Tools stereo digital audio editing/recording system. ☞ **See also** *Sound Designer II*.

Sound Designer II (a.k.a. SDII). The digital audio file format used by later versions of Digidesign's Sound Designer software. The SDII file format became popular on the Macintosh platform, though it has largely been supplanted by the Broadcast WAV file format. ☞ **See also** *Sound Designer*.

sound field. The range of panning or left/right stereo width available for positioning sounds. Surround sound formats expand this to include the range of front-to-back panning available.

sound isolation. A more technically correct way of saying "soundproofing."

sound pressure level. The volume or loudness of a sound, expressed in decibels. The reference 0 dB represents the baseline threshold of human hearing. The range most recording engineers recommend working at in the studio is 80 to 90 dB, because our ears have the best response at this volume level.

Sound Transmission Class (a.k.a. STC). A rating that can be used to compare the acoustical isolation provided by different materials. In a series of tests, the sound transmission loss for a series of frequencies is plotted on a graph, the resulting curve is compared to a reference curve, and the STC rating is determined. For example, 1/2-inch gypsum board might have an STC rating of 28. It's important to note that STCs of combined materials don't add, so two layers of gypsum board on a wall do not result in an STC of 56. In this particular case, doubling the gypsum board doubles the mass, which results in an STC increase of 6, from an STC of 28 for a single board to an STC of 34 for a double layer of board. Normal conversation can be heard and understood through a material with an STC rating of 25 to 34. An STC of 65 or higher is considered soundproof by many listeners.

Sound Transmission Loss (a.k.a. STL). A frequency-dependent measurement of the amount of isolation from sound transmission a particular material provides. For example, 1/2-inch gypsum board might have a sound transmission loss rating of 15 decibels at 125 hertz, meaning that a 125-Hz sound wave passing through the gypsum board will be reduced in level by 15 decibels.

sound wave. 1. Wave motion created in air (or other material) as a result of vibration of a material in the human hearing range. 2. A cyclical pressure front consisting of a zone of high pressure (compression) followed by a zone of low pressure (rarefaction), followed by high pressure, and so on, propagating through air or other material.

SoundFont. A standard file format developed by E-MU Systems for storing sound information. SoundFonts contain samples, wavetable synthesis parameters and instructions, sample loop information, and real-time MIDI control response parameters and instructions. SoundFonts require a compatible player; the quality of the output will depend on the quality of the playback device.

soundproof. Impervious to sound waves. This is nearly impossible to achieve without extensive construction, extremely heavy mass, and great expense.

soundset. A group of presets or programs.

soundstage. 1. Also known as studio. A room used for audio production. 2. The sonic space that a recording lives in, or the perceived room or space that listeners hear in the tracks of a recording.

soundware. Sounds developed for synthesizers and samplers. Though any synth, sample patch, or sound is technically soundware, the term is generally applied to sounds developed and offered for commercial sale.

source. 1. In audio, the instrument or object producing a sound. 2. The signal path selected to be monitored by a recorder, audio interface, or VU or other meter, or that is selected to play through a monitor system.

source code. The instructions making up a computer program. Typically, source code must be compiled, or translated to the language the computer's specific process can use.

spaced omni. A stereo microphone technique first used by Harry Fletcher in 1933. The spaced-omni technique uses two identical omnidirectional microphones placed several feet apart (see Figure S.14). With careful placement, spaced omni offers a good balance of room ambience and direct source sound, with tonally natural response. ☞ **See also** *A-B stereo*.

spare bedroom. The most common location for a home studio.

SPARS. Society of Professional Audio Recording Services. An organization made up of audio and multimedia businesses, business owners, engineers, and other industry professionals. www.spars.com.

S/PDIF. Sony/Philips Digital Interface Format, a.k.a. IEC 958 Type II. A two-channel format developed by Sony and Philips for consumer use (though now commonly used for professional and semi-professional applications as well), intended for transmitting digital audio signals between pieces of gear. S/PDIF is very similar to the AES format, and, in

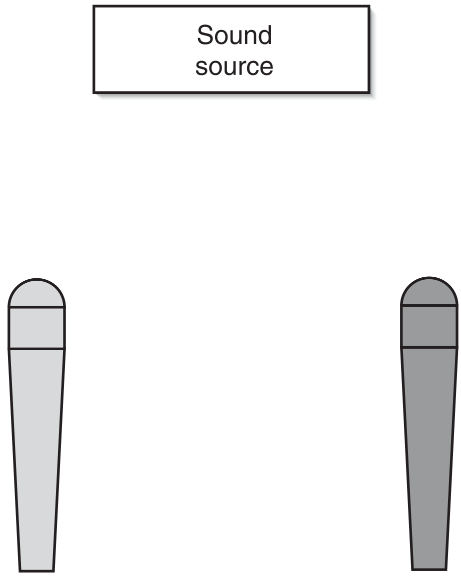

Figure S.14 For the spaced-omni stereo miking technique, two omnidirectional microphones are positioned in parallel in front of the sound source, spaced several feet apart.

general, the two formats are compatible (though cable adapters or a converter box may be required —be sure of what you need before you attempt to interconnect them). S/PDIF is carried over unbalanced coaxial cable, usually using RCA or occasionally BNC connectors, or fiber optic cable with TOSlink connectors. S/PDIF is designed to use 75-ohm cables, though regular audio RCA cables will also work. The S/PDIF specification supports resolution up to 20 bits, though some devices support 24-bit resolution over S/PDIF.

speaker. 1. Also known as driver. A transducer that converts electrical signals into sound waves. 2. Short for speaker cabinet.

speaker cabinet. An enclosure in which a speaker driver is mounted. The speaker cabinet is designed to prevent or manage interaction between the sound coming from the front of the speaker driver and the out-of-phase sound coming from the rear of the speaker driver. A variety of technologies and techniques are used in speaker cabinets to improve the efficiency, frequency response, and sound quality of the speaker, including ports, acoustic suspension designs, horns, and more.

Speakon®. A multi-pin locking connector type developed by Neutrik for high-power applications, such as on power amplifiers and speakers. Speakon connectors are durable, reliable, and affordable; can handle high current levels; and offer the security of only inserting a jack in one way so no mistakes can be made. Two-, four-, and eight-conductor versions are available.

special effects. ☞ See *effects*.

specifications. Measurements of the performance of a product provided by the manufacturer. Because measuring techniques vary, and because measured data can be manipulated using different references and standards, specifications are only somewhat useful for comparing products and rating performance.

spectra. Plural of spectrum. The distribution of sound energy arranged by frequency.

spectral waterfall. ☞ See *spectrogram*.

spectrogram (a.k.a. sonogram, spectral waterfall, voiceprint). A three-dimensional visual representation of a spectrum, displaying changes in sound energy by frequency over time. See Figure S.15.

spectrum. 1. The range of frequencies comprising a waveform. Spectrum refers to the frequency-domain representation of the sound wave; a waveform display is a representation of the same information in the time domain.

spectrum analysis (a.k.a. frequency analysis, Fourier analysis). A mathematical method for analyzing the frequencies making up a sound wave.

speed of sound. The speed that sound waves travel through the air was first analyzed by Sir Isaac Newton. In fact, Newton came up with a figure of 979 feet per second, which is about 15 percent low. Pierre-Simon, Marquis de Laplace, a French mathematician, corrected the errors in Newton's calculations. Through air, the speed of sound is

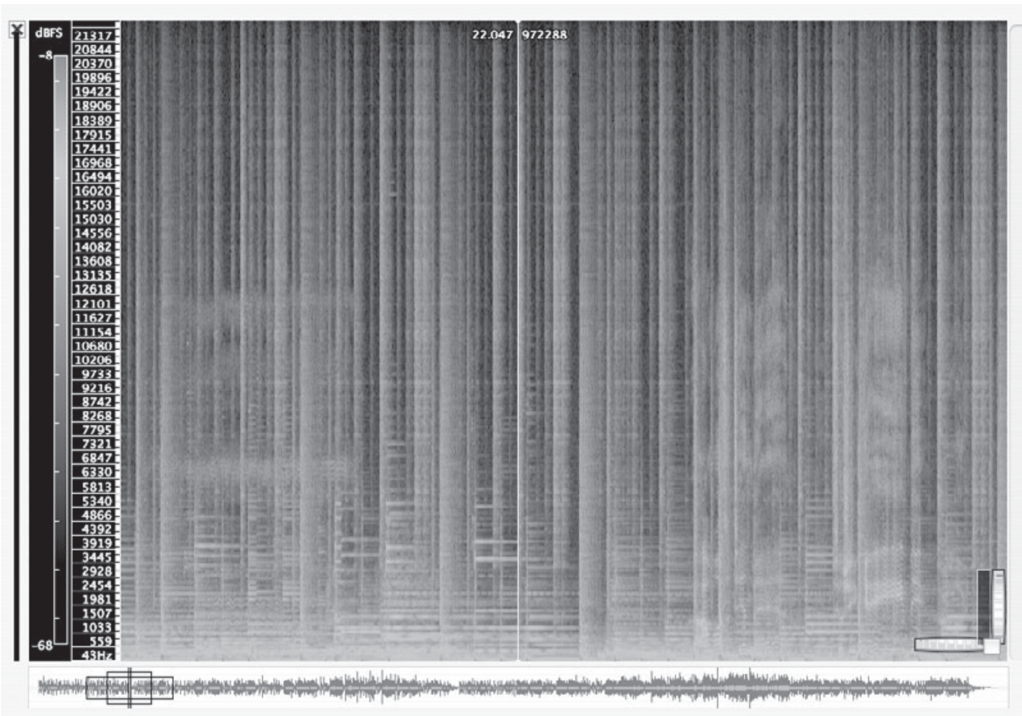

Figure S.15 A spectrogram displays changes in sound energy per frequency over time.

approximately 1,130 feet per second, or 343 meters per second. (The exact figure varies depending on temperature, atmospheric conditions, and altitude.)

sphere stereo. A baffled stereo-miking technique using a pair of omnidirectional microphones mounted flush to, and separated by, an 8-inch solid sphere. The sphere is intended to simulate the acoustic properties of a human head, such as head shadow and ITD. The omnidirectional microphones maintain constant low-frequency response without proximity effect, regardless of distance from the source. Sphere stereo provides good imaging, separation, and depth, and natural sound quality. Sphere stereo differs from the "dummy head" technique, which is intended primarily for headphone listening, as it works well with speakers. ☞ **See also** *baffled stereo*.

spider mount. ☞ See *shockmount*.

spike. ☞ See *power spike*.

spill. ☞ See *bleed*.

SPL. ☞ See *sound pressure level*.

splice. 1. With analog recording, using a razorblade, splicing block, and splicing tape to edit together and join two pieces of tape. Typically, the two pieces of tape are cut at complementary angles so the splice will pass over the recorder's playback head at an angle, helping to prevent a pop or click at the splice point. 2. With DAWs, editing together and joining two pieces of digital audio to create a new single piece of digital audio.

splicing block. A metal block, similar to a miter box used by a carpenter, used to precisely cut analog tape for splicing.

splicing tape. Adhesive tape used to splice together analog recording tape.

split. To divide a keyboard into two or more zones that each can transmit on a separate MIDI channel and/or play a different internal sound. ☞ **See also** *split point*.

split mixer. A type of mixer that has two input sections; one side is used for incoming signals from microphones and other sources, and the other section is used for signals returning from a multitrack tape recorder. ☞ **See also** *in-line mixer*.

split point. The point(s) at which a keyboard is divided into two or more separate zones. The split points define the key range of each zone. ☞ **See also** *zone*.

split stereo file. A stereo audio recording that is stored as two discrete files, one for the left channel and the other for the right channel. Split stereo files are preferred by some applications, such as Digidesign Pro Tools. ☞ **See also** *interleaved stereo audio*.

SP-MIDI. Scalable Polyphony MIDI. An extension of the MIDI Specification for use in 3GPP mobile phones and handheld games, where different devices offer varying polyphony capabilities for playback of MIDI sequences. SP-MIDI allows the creators of sequences to specify MIDI channel priority (called channel masking) and note priority or MIP (Maximum Instantaneous Polyphony). SP-MIDI supports the GM2 soundset and limited MIDI continuous controllers.

SPL meter. A device that measures sound pressure level. An SPL meter is a useful tool for maintaining consistent (and safe) volume levels in the studio and for live performances.

spotting session. A meeting where a film's music editor, director, and composer gather to decide what music will be used at what times in the film.

SPP. ☞ See *Song Position Pointer*.

spread. 1. The width of a stereo signal. 2. The time between two songs on an album. The spread is set during mastering and will vary depending on how the first song ends and the second song begins.

spring reverb. A simple type of mechanical reverb developed by the Hammond Organ company and released under the Accusonics brand name. An audio signal is sent into one end of a spring or springs and is picked up on the other end by a transducer. In addition to the signal transmitted straight through the spring, some signal is reflected and bounces back and forth in the spring. The reflections in the spring create a wash of sound, which serves as reverb. Spring reverbs are still widely used in guitar amplifiers.

sputtering. The process of applying a thin layer of metal to a surface. Sputtering is used to apply the extremely thin layer of gold to the surface of the Mylar diaphragm in microphones.

square wave. A type of cyclical waveform used for synthesis and other applications. Square waves contain a basic frequency, the fundamental, plus all of the odd integer harmonics. A clarinet produces a tone very close to a square wave. Square waves have two alternating states, high and low, making them useful for representing binary digital data. The ratio of the high state period to the total wave period is the duty cycle; a true square wave has a 50-percent duty

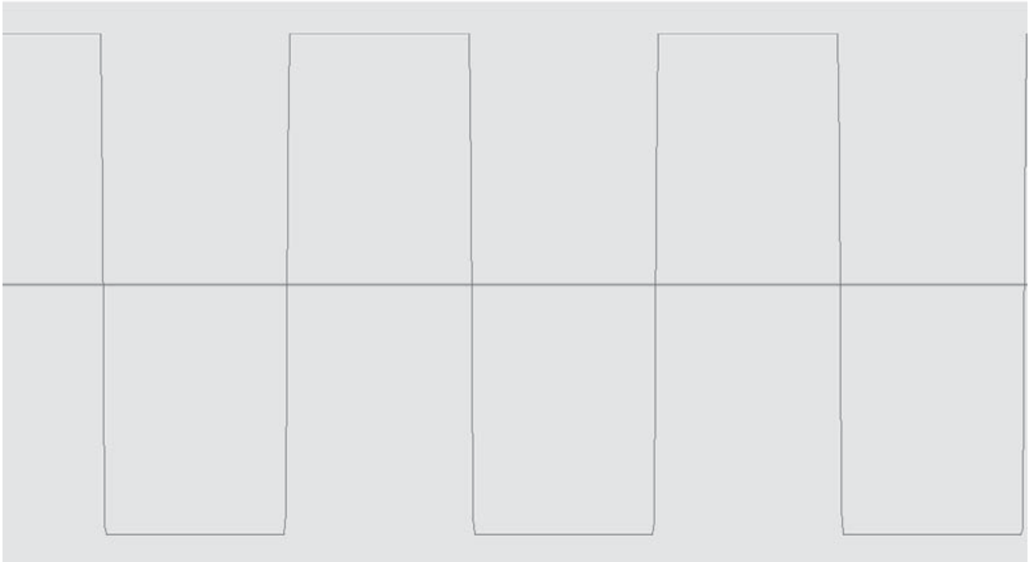

Figure S.16 A square wave contains a fundamental frequency plus all odd integer harmonics. The ratio of high state to overall period is the duty cycle.

cycle. Pulse Width Modulation (PWM) occurs when the wave's duty cycle is modulated. See Figure S.16.

S/s. Samples Per Second. A more accurate way of indicating sample rate. Using S/s reduces the possibility for misunderstanding whether kHz (the usual way of indicating sample rate) is referring to sample rate or a frequency.

stair stepping. The discrete values of a digital representation result in incremental stair-step changes rather than the smooth changes of analog (see Figure S.17). With digital audio signals, stair stepping results in quantization noise. With control or parameter changes, the result is zipper noise or audible steps as the parameter changes.

standalone. 1. A hardware device that can operate by itself and does not depend on other gear to function. 2. A software program that runs directly in the operating system, as opposed to within a host program.

Standard MIDI File (a.k.a. SMF, .MID). A standard, "generic" file format designed to allow musicians on different computer platforms and/or using different sequencing programs to exchange sequence files. There are two types of Standard MIDI Files: Type 0, where all 16 MIDI channels are combined onto one multichannel track, and Type 1, where each of the 16 MIDI channels is stored on its own track. Both types preserve all channel settings and other information, so neither is really better than the other.

standby. A mode available on some devices, particularly guitar amplifiers and other tube-based items. Standby turns off various active features while maintaining power to components such as vacuum tubes.

standing waves. Sound waves reflecting between two parallel surfaces in a room. Standing waves always negatively impact the response of the room and are controlled using acoustic treatments. ☞ **See also** *room mode*.

star ground. A grounding scheme used to prevent ground loops. The ground is lifted from the AC electric power line of each piece of gear, then a new ground wire is run from the chassis of the item to the studio's technical ground.

star network. A network built around a central device or hub. In audio, the most common example is using a master word clock generator with multiple outputs to synchronize all digital devices.

star quad. A type of balanced audio cable using four conductors—two for the positive signal and two for the negative signal—that are twisted into a spiral inside the cable's shield. The doubled and twisted conductors help eliminate EMI and other interference problems.

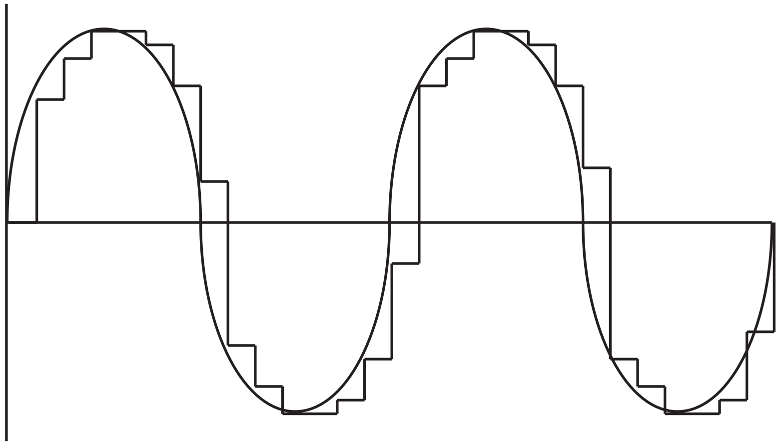

Figure S.17 The discrete values of a digital representation result in incremental stair-step changes rather than the smooth changes of analog.

start time. The time code point at which a sequencer or other device is set to begin playing or functioning. When the sequencer sees the start time arrive in the time code, it starts playing.

startup disk. The hard disk containing a computer's operating system, used to boot up the system. With some computers, such as Macintoshes, you can have different operating systems on separate hard disks and designate which one to use as the startup disk when booting.

status byte. The part of a MIDI message that indicates what type of information the message carries, as well as the MIDI channel over which the message is being sent. There are eight message types: note on, note off, channel aftertouch, polyphonic aftertouch, continuous controller, pitch bend, program change, and system. (There are three types of system messages: System Common, System Real Time, and System Exclusive.)

STC. ☞ See *Sound Transmission Class*.

stem. A mono, stereo, or even surround submix of tracks from an audio production. The term comes from the film world, where separate stems are prepared of Foley, music, sound effects, dialogue, and other tracks. In music production, stems might be prepared of drums, bass, backing instruments, solos, lead vocals, and background vocals. The use of stems simplifies large, complex final mixdowns and allows overall processing to be applied to each stem separately. ☞ See also *submix*.

step. 1. Also known as whole step. In the standard Western equal temperament, the octave is divided into 12 equal-sized semitones or half steps of 100 cents each. A whole step comprises two semitones. 2. A discrete interval or increment.

step entry (a.k.a. step time). A mode, available in many sequencers, in which the MIDI data is entered while the sequencer is stopped. As each note or chord is played in, the sequencer automatically moves forward by a user-specified time increment. Step entry allows a composer or musician to enter much more complex or faster parts than he or she is capable of playing in real time, and to do so with perfect rhythmic accuracy.

stepped attenuator. A rotary control that has discrete switched steps to apply attenuation instead of a continuous potentiometer like a standard volume control. Stepped attenuators allow for accurate settings, as well as precise repeatability of settings.

stepped sine wave (a.k.a. modified sine wave). A digital representation of a sine wave, with stepped values instead of a smooth curve. Many UPS devices use stepped sine waves to drive an inverter to create their AC output power from a DC battery power source.

step sequencer. 1. An analog sequencer that plays back a series of voltages, set using a row of knobs, that can be used to create a note pattern or to change a parameter value, such as filter cutoff point. 2. A simple MIDI sequencer that plays back short note or parameter control patterns that are manually entered into a grid. The grid normally consists of one or two bars subdivided into sixteenth-note or other time increments. Step sequencers are found in drum machines, groove boxes, some DAWs, and in software applications such as Propellerhead Reason, and are most commonly used to create looping ostinato patterns. See Figure S.18.

step time. ☞ See *step entry*.

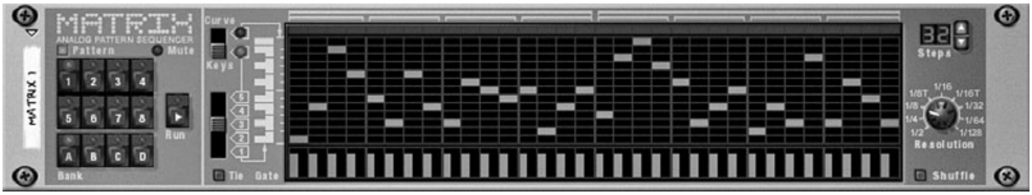

Figure S.18 Step sequencers play back a short looping pattern of notes or control changes.

stereo. Short for stereophonic. 1. A signal comprising two channels of related audio material. 2. A multichannel sound system that provides directional cues. The term has come to mean a two-channel system, but technically refers to any number of channels.

stereo bar. An accessory for mounting two microphones to a single stand. The mic mounts on stereo bars allow the mics to be turned/positioned for a variety of stereo miking techniques, such as XY stereo, ORTF, and others. See Figure S.19.

stereo bus. A two-channel bus. ☞ See *bus*, *aux send*, *stereo mix bus*.

stereo imaging. ☞ See *imaging*.

stereo link. A switchable function that correlates the operation of two independent channels in a processor into true stereo channels. For example, when a two-channel compressor is stereo-linked, a signal crossing the threshold in either channel will trigger compression in both the left and right channels. This maintains stereo imaging as well as the integrity of the stereo field.

Figure S.19 A stereo bar is used to conveniently mount and position a pair of microphones for stereo recording.

stereo microphone techniques. A number of microphone placement and configuration techniques designed to capture a stereo image of the source. Most stereo microphone techniques use two microphones, though some use more. The mics are arranged at particular distances apart and at certain angles relative to one another. Stereo miking offers a number of advantages over using a single mic: better depth, stereo width, more realistic ambience and imaging, and more. There are four basic approaches to stereo miking:

- **coincident pair.** Two mics placed with their capsules as close together as possible, usually one over the other, set at a particular angle relative to one another. Examples include XY stereo, Blumlein pair, and MS stereo.
- **near coincident pair.** Two mics placed within a few inches of one another and set at a particular angle. Examples include ORTF, NOS, and DIN.
- **spaced pair.** Two microphones placed parallel to one another, a few feet apart. Examples include A-B stereo and spaced omni. The Decca Tree technique is related to the spaced pair approach, though a third, center mic is added.
- **baffled pair.** Two microphones placed parallel to one another, but acoustically separated by a solid baffle. Examples include Jecklin Disc, stereo sphere, and dummy head.

stereo mix bus (a.k.a. 2-bus). The main or master stereo output channels of a mixer, to which a mix of all the input channels, tape tracks, and returns is routed for final processing and level management.

stereophonic. ☞ See *stereo*.

stompbox (a.k.a. pedal). An audio processor mounted in a compact, floor-pedal format. Stompboxes have been most commonly used by guitarists and sometime bass players, but in recent years have made more frequent appearances in keyboard rigs and studios. Most stompboxes provide a single, basic processing function, such as distortion, delay, reverb, flanging, EQ, or filtering, though some modern processors are very versatile.

stop. ☞ See *organ stop*.

stopband. The range of frequencies processed (attenuated) by a filter. Some filters, such as bandpass filters, have more than one stopband. ☞ **See also** *passband*.

straight-wire. ☞ See *uncolored*.

streamer. In film scoring, a colored band that moves from left to right across the video screen followed by a circular flash. Streamers are used as cues for composers and conductors for where important musical "hits" should take place.

streaming. 1. A method for transmitting data in a continuous flow. A common example is providing continuously playing, or streaming, audio over the Internet, rather than offering a discrete file of the audio for download. 2. A technique for playing extremely large samples without loading them into the computer or sampler's RAM.

streaming audio. Digital audio that is delivered as a continuous flow of data rather than as a complete file that must be downloaded or transferred. Streaming media will play back while the data is being transferred, rather than requiring that the entire file be transferred or downloaded before it can be played. ☞ **See also** *streaming*.

streaming media. Audio, video, and/or multimedia data that is transmitted over the Internet in streaming fashion by a server, received by the user's computer, and displayed or played back in a player application.

stretch tuning. A technique used by piano tuners that tunes the high notes slightly sharp and the low notes slightly flat. The detuning is very minimal—a few cents at most—and is intended to make the high and low ranges sound more in tune to listener's ears.

stripe. To record time code to analog tape so that multiple tape machines or other devices could be synchronized together. Striping is not necessary with digital systems, as the digital data maintains its own timing and synchronization reference that can be converted to time code if necessary.

striping. 1. ☞ See *stripe*. 2. A technique used in RAID 0 storage systems in which each piece of data is broken up into segments and written in round-robin fashion to the component hard drives of the RAID system. Striping RAIDs offer very fast performance for large files such as those found in video and some audio applications.

strip silence. A type of audio editing function found in DAWs that works similarly to noise gating: A threshold level is set, and any part of the audio in the track that falls below the threshold is silenced.

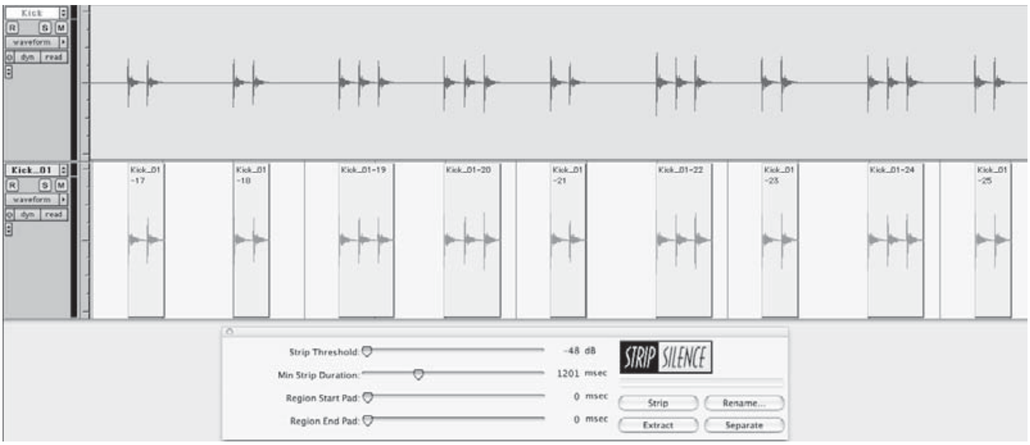

Figure S.20 When a strip silence-type function is used, audio below a threshold level is removed, and the remaining audio is broken up into separate regions.

The advantage of strip silence–type functions over noise gating is that strip silence breaks the track into separate audio regions or segments when it creates the silent areas, which can then be manipulated, processed, and edited separately. See Figure S.20.

stuck note. A MIDI note that never receives a note off message and continues to sound after it should have ended. Stuck notes can occur if a MIDI cable is accidentally pulled out during a performance or playback, or from sequence editing errors or corrupted data, or for other reasons. Cures for stuck notes include sending an all notes off message, using a "panic button" command in a sequencer or keyboard, or turning off the power to the synth or sampler with the stuck note.

studio. 1. A room used for audio production. 2. A room where an artist or musician practices and works.

studio monitor. ☞ See *reference monitor*.

stutter effect (a.k.a. glitch). A DJ and remix effect created by copying a small slice of audio, then repeatedly pasting it into a track to achieve a stuttering effect. See Figure S.21.

stylus (a.k.a. needle). The component of a phonograph cartridge that reads the LP groove. A stylus has two parts: the cantilever and the tip. The cantilever is a short arm that holds the tip and mounts into the cartridge, while the tip is the part that actually sits in the record groove.

sub. ☞ See *subwoofer*.

subcode. Non-audio information that is written to digital tape. Subcode information might include track number, track length, elapsed track time, indexing, time code, and more.

subframe. A subdivision of an LTC frame. Typically, there are 80 or 100 subframes per frame.

subgroup. A number of channels or tracks in a mixer or DAW that are routed to a bus or a dedicated subgroup fader. Usually subgroups are created to make mixing easier. For example, all of the drum tracks or all of the backing vocals in a mix might be routed to a mono or stereo bus so their overall level can be controlled as a group using a single fader.

subharmonic. A frequency below the fundamental in a sound wave, usually at octave ratios of 1/2, 1/4, and so on, to the fundamental frequency. For example, a fundamental at 1,000 Hz might have subharmonics at 500 Hz, 250 Hz, 125 Hz, and so on. Some manufacturers have developed subharmonic "synthesizers" that can output subharmonic frequencies an octave or two below the fundamental based on the inputted sound. Subharmonic synthesizers are used to increase the amount of low end in a signal.

submenu. A menu in a computer program that branches out to open a second menu of related commands or selections. See Figure S.22.

submix. A number of tracks in a song that are mixed together through a bus, then mixed into the main mix. For example, all of the drum tracks or all of

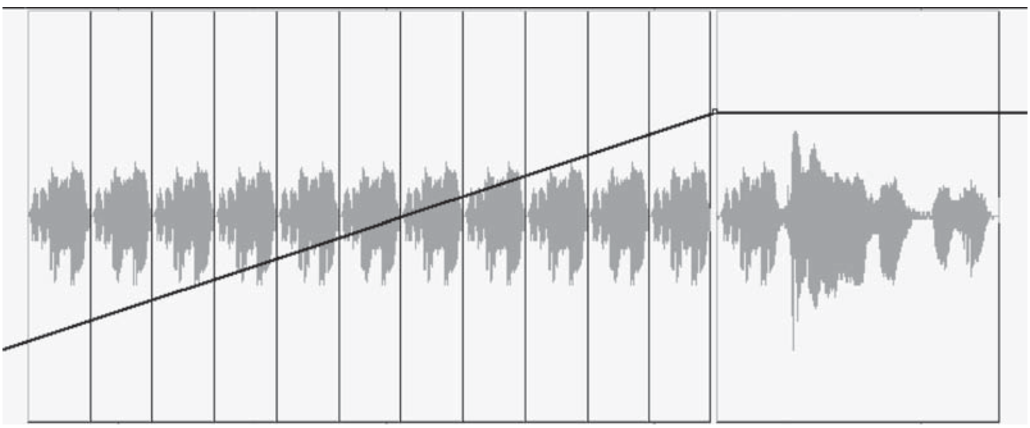

Figure S.21 A stutter effect is created by repeatedly pasting a small slice of audio into a track. In this case, the first part of a word has been repeated, and a volume curve is used to fade it into the full word at the right.

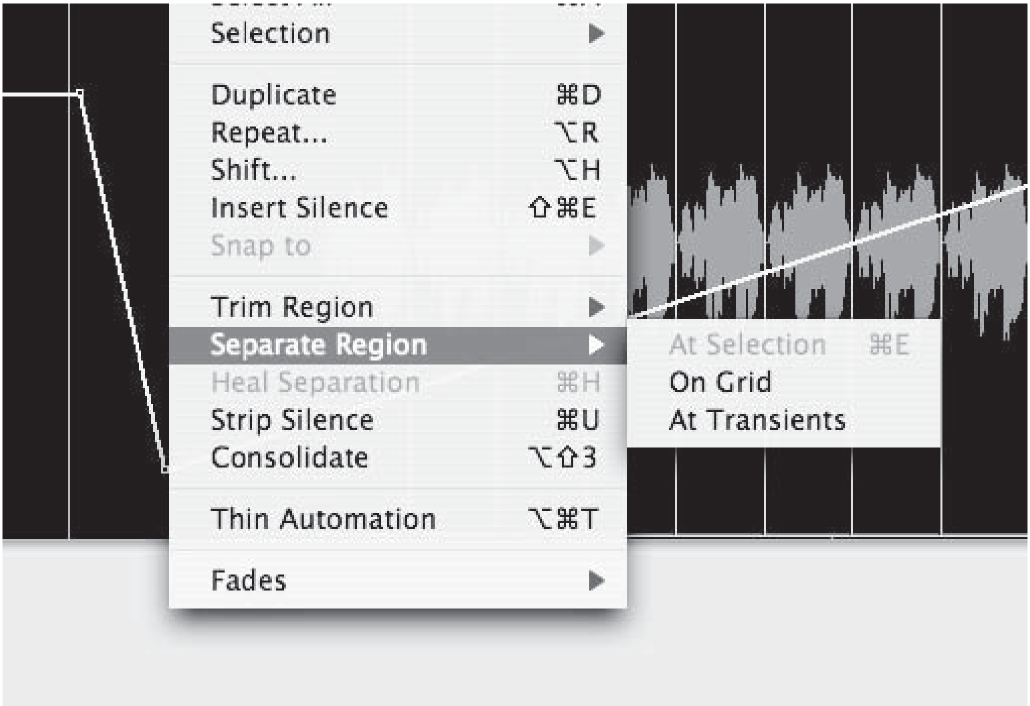

Figure S.22 A submenu is a menu that branches off from a main menu and contains related commands.

the backing vocals in a mix might be mixed to mono or stereo so their overall level can be controlled using a single fader and so they can be processed as a group. Submixes can also be used to organize a mix, to make mixing easier, or if there aren't enough channels in the main mixer to handle all the signals. ☞ **See also** *submixer*.

submixer. A secondary mixer that creates a mono or stereo submix of signals that is then routed into the main mixer.

subsonic. Traveling at speeds slower than the speed of sound. Often incorrectly used in reference to frequencies below the range of human hearing. (☞ **See also** *infrasonic*.)

subtractive synthesis. A type of synthesis that starts with harmonically rich raw sounds, then uses filters to remove unwanted frequencies to achieve the desired final timbre. Most analog synthesizers use subtractive synthesis techniques.

subwoofer. A speaker dedicated to producing low-frequency sound waves, usually below 120 Hz.

sum and difference tones (a.k.a. combination tones). ☞ **See** *difference tone*, *sum tone*.

summing. In audio, mixing or combining signals together. The quality of the summing in a mixer has a great deal of impact on the sound quality of the mixes it creates.

summing box. An external analog hardware mixer that is designed to sum, or combine, the audio outputs from a DAW. The idea is to bypass the software mixer in the DAW and combine the track outputs externally in the analog domain. Summing boxes can range from very simple units with 8, 16, or more analog inputs (with no control over levels; levels are set in the DAW) and a single set of stereo outputs, to full-blown analog mixing consoles. Some engineers prefer to use summing boxes because they feel external analog summing provides superior performance to the built-in software digital mixers found in DAWs.

sum tone (a.k.a. combination tone). A frequency created under certain circumstances when two other

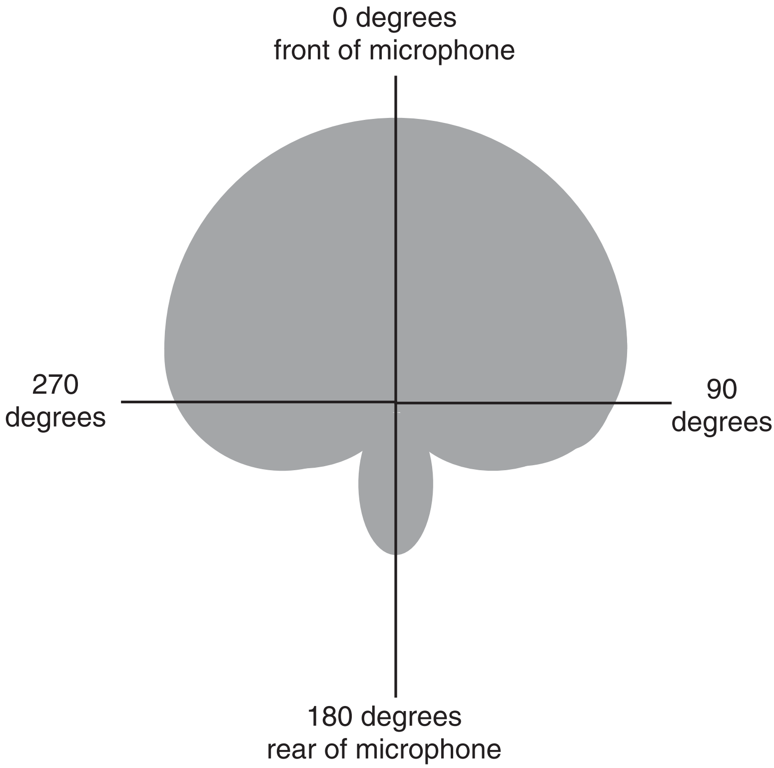

Figure S.23 The supercardioid polar pattern is quite directional for good rejection of unwanted sound.

frequencies are sounded together. The new frequency will be at the sum of the two original frequencies. For example, the sum tone created by two tones at 500 and 300 Hz would be at 800 Hz. ☞ **See also** *difference tone*.

supercardioid. A microphone polar pattern that is more directional than cardioid, and with a smaller rear lobe than hypercardioid. See Figure S.23.

supersonic. Technically, traveling at speeds faster than the speed of sound. Sometimes incorrectly used to refer to frequencies above the range of human hearing. (☞ **See also** *ultrasonic*.)

supraaural. Headphones with cups that rest on the ear rather than on the side of the head surrounding the ear. Because they don't seal around the ear, supraaural headphones generally do not provide good isolation from external sound or sound bleeding out of the headphones.

surface mount. Small electronic components that rest on, and are soldered to, the surface of a circuit board. (As opposed to traditional components that have legs that stick through holes in the circuit board and are soldered to the back of the board, after which the legs are trimmed off.) Surface-mount components save space and are easy for industrial assembly machines to handle, allowing much greater manufacturing efficiency. The tradeoff is that surface-mount boards are difficult to repair should a problem develop.

surge. ☞ **See** *power surge*.

surround. 1. The part of a speaker that connects the cone to the frame or basket. The surround is made from flexible material that allows the cone to move freely in response to the voice coil, but must be stable enough to keep the voice coil centered around the magnet. 2. ☞ **See** *surround sound*.

surround panner. 1. A software control used to position a sound in a surround field. Surround panners can position mono, stereo, or even surround tracks in the surround field and may include additional controls, such as a feed for sending low frequencies from the signal to the system's subwoofer or LFE channel, a "divergence" control for setting the amount of the front left/right portion of the signal that is also fed to the center channel, and more. Some surround panners are modeled after a joystick, where one control is used to set the panning; others use separate controls for setting the side-to-side panning and the front-to-back panning. Most surround panners also allow the channel's feeds to the various speakers to be muted or soloed. 2. A hardware control box that controls a software surround panner.

surround sound. A multichannel sound reproduction system with more than two channels of related audio material. Typical formats include 5.1 (five main speakers with one subwoofer) and 7.1 (seven main speakers plus one subwoofer). Originally intended for film sound, surround sound has also made inroads into music playback with DVD-Audio, SACD, and video sound.

suspension mount. ☞ **See** *shockmount*.

sustain. 1. The stage in an envelope generator that determines how long a note or parameter value will be held. 2. The duration of a musical note.

sustained transfer rate. A specification measuring how fast a drive can send and receive data. Sustained transfer rate is impacted by system processing delays, the time required for head switching, seek time, and more factors. Sustained transfer rate is one of the most useful specs for evaluating how a drive will perform for audio or video applications.

sustain pedal. 1. The rightmost pedal on a piano, which lifts the dampers off all the strings and allows the notes to continue to ring, even if the keys that played them are released. 2. MIDI Continuous Controller #64. The MIDI sustain controller turns on with a MIDI value of 127, off with a MIDI value of 0, and (unless remapped by the user) performs the same function as a piano's sustain pedal by holding any active envelope generators at the sustain stage so the notes continue to sound. (Whether a note continues to sound when held by CC #64 also depends on the polyphony of the sound generator. If there is not enough polyphony to support all the sustained notes, some will be cut off.)

SVGA. Super Video Graphics Array. An enhanced version of the VGA video protocol that can display up to 16.7 million colors with resolutions up to 1,280 × 1,024 pixels.

sweep (a.k.a. frequency sweep). Continuous playback of a tone smoothly increasing or decreasing in frequency. Sweeps are used for a variety of test and analysis purposes.

sweepable mids. An equalizer where the center frequency and amount of boost/cut in a midrange band can be controlled (see Figure S.24). Generally bandwidth can't be controlled in a sweepable-mid EQ, and the high- and low-frequency bands are fixed in frequency. ☞ **See also** *quasi-parametric*.

sweetening (a.k.a. post-production). Sweetening is a video term for adding music, sound effects, and other audio to a film. The term has been adopted in music production for adding effects to an audio track or mix.

sweet spot. The location in a listening room with the best response and imaging. For stereo, this is

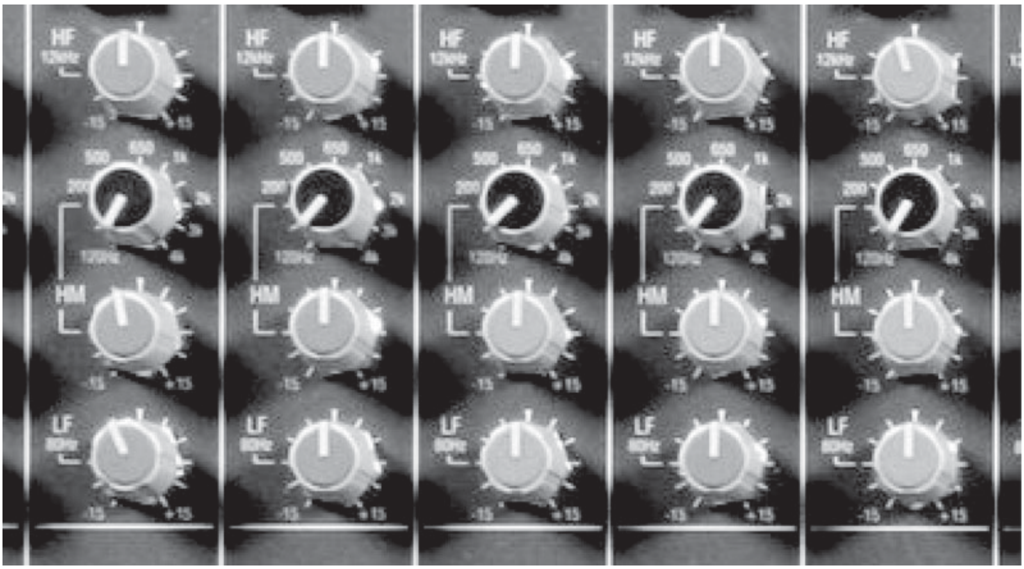

Figure S.24 The channel EQ found in many mixers offers sweepable mids—a midrange band that has control over the boost/cut amount and frequency.

normally the third point in an equilateral triangle with the two monitors. Finding the sweet spot in a studio control room is essential for accurate monitoring and for making solid decisions about the quality of a mix or recording.

swing. 1. A jazz-like rhythmic feel, where the first of a pair of eighth-notes is played longer than the second. 2. A quantization function in sequencers and DAWs that adds an adjustable amount of swing feel to consecutive MIDI eighth notes in an attempt to make the sequence sound and feel more human. Usually the swing parameter is set as a percentage, from 0% (no swing) to 100% (heavy swing).

switching power supply. A type of AC power supply that switches current into a transformer at very high rates, above audio frequencies. The transformers in switching power supplies are lighter, smaller, and less expensive than those in regular "linear" power supplies. Some switching power supplies can also regulate the voltage better than linear supplies.

sympathetic vibration. Vibrations of a particular frequency produced in a material as a result of contact with sound waves of that same frequency. An increase in the volume of the sound caused by the sympathetic vibration is called *resonance*.

sync. Short for *synchronize*.

Sync 24 (a.k.a. DIN Sync). A type of synchronization developed by Roland for older drum machines and other devices, such as the TR-808, TR-909, TB-303, and MC-202. Sync 24 used a 5-pin DIN connector similar to the one later used for MIDI (though the two are *not* directly compatible) and carried start and stop messages as well as a 24 PPQN timing clock stream. Sync 24 was made obsolete by MIDI.

synchronization. ☞ **See** *synchronize*.

synchronize. 1. To connect two or more devices together so that they operate as one unit. 2. To set up one unit to follow the timing of a second unit. 3. To lock together the clocks in digital audio devices.

synchronized delay. An echo effect that is synchronized to the tempo of the song. This allows delays to be set to rhythmic values.

synchronizer. 1. A hardware device that handles the synchronization of two or more analog tape machines. Synchronizers do this by comparing the time code coming from the tape machines with reference time code and adjusting the speed of the tape machines to keep them locked together. 2. A hardware device that reads LTC or VITC and synchronizes other devices to that time code.

synchronous. Events that are coordinated in time, often by a synchronization clock. Some synchronous data transfers, such as those handled by S/PDIF, AES, and other digital formats, do not require handshaking or other confirmation to verify that the transferred information has arrived, while others do require responses in order to stay synchronized.

synth. Short for *synthesizer*.

synth action. A type of unweighted plastic keyboard action that does not attempt to emulate the resistance and feel of an acoustic piano. Typically found on 61-note and shorter keyboards, synth actions use springs instead of weights or hammer mechanisms and are less expensive than other types of actions. Synth action allows for fast playing and may work better for certain sorts of sounds (such as synthesizer leads) than a weighted action. ☞ **See also** *action*.

synthesizer. An electronic musical instrument that can generate waveforms using oscillators. These waveforms can then be combined, manipulated, filtered, and otherwise processed to create musically useful timbres. There are a wide range of different types of synthesis, all of which have their individual strengths and weaknesses when producing certain types of sounds. A few of these include:

- **additive.** Type of synthesis that creates sounds by combining sine waves of different frequencies.
- **AI** (***Advanced Integrated***). Digital sample-playback synthesis method used in the Korg M1 and other synths.
- **analog.** A type of synthesis (typically subtractive, though other types, such as FM and additive, are also possible) based on raw analog waveforms.
- **FM** *(Frequency Modulation)*. Digital synthesis method used by Yamaha in the DX7 and other models that creates sounds by modulating digital oscillators.
- **granular.** Type of additive synthesis that uses tiny "grains" of sound.
- **hybrid.** A synthesizer that combines multiple synthesis techniques.
- **LA** *(Linear Arithmetic)*. Digital synthesis method used by Roland in the D50 and other models that combines sampled attack sounds with waveforms.

- **PD** *(Phase Distortion)*. Digital synthesis method, somewhat similar to FM, used by Casio in the CZ-101 and other models.
- **physical modeling.** Digital synthesis that uses computer-generated models to create sounds.
- **resynthesis.** Digital synthesis method where sounds are analyzed and "reconstructed" in the synth.
- **sample-playback.** Synthesis method based on sampled waveforms and sounds.
- **vector.** Digital synthesis method that allows cross-fading between multiple sounds. Used by Sequential Circuits, Korg, and Yamaha.
- **wavetable.** Digital synthesis where waveforms are built based on parameters stored in a table.
- **Z-plane.** Sample-based digital synthesis that allows for extremely complex modulations as well as morphing.

SyQuest. An obsolete removable disk storage format developed by SyQuest, later purchased by Iomega.

Sys Ex. ☞ **See** *System Exclusive*.

Sys Ex dump. Transferring the RAM contents of a synthesizer or other MIDI-compatible device using System Exclusive messages.

system bus (a.k.a. frontside bus). The data bus in a computer that connects the CPU to the RAM.

System Common message. A type of MIDI system message that is intended for the entire MIDI network. System Common messages include MIDI Time Code Quarter Frame, Song Position Pointer, Song Select, Tuning Request, and EOX (End of System Exclusive).

System Exclusive. A type of MIDI system message compatible with one specific piece of gear. System Exclusive messages are used to carry proprietary control information, to control parameters that can't be accessed with continuous controller messages, and to transfer RAM program data between synthesizers and other MIDI-compatible gear. System Exclusive messages allow editor/librarian software running on computers to display, take control of, and edit the parameters of a MIDI-compatible device.

system preferences. A preference file that contains settings for a computer's operating system. ☞ **See also** *preferences*.

System Real-Time message. A type of MIDI system message that addresses timing, sequence playback, or synchronization. System Real-Time messages include MIDI Timing Clock, Start, Stop, Continue, Active Sensing, and System Reset.

System Reset. A MIDI System Real-Time message that tells any receiving device to reset all its controls to their default position. The exact effect of a System Reset message will depend on how the receiving device is programmed to respond.

T

tab. A feature of a window in a computer program that allows the window to have several views or to display other parameters. By clicking on the tabs, the different views can be accessed. See Figure T.1.

tabletop. A hardware format for synthesizer and processing modules that is designed to be freestanding on a table or desk, rather than rackmounted.

tails out. A method for storing analog reel-to-reel tape "inside out," with the beginning of the reel at the hub and the end of the tape on the outside. This is done both to protect the tape and to reduce the effects of print-through. ☞ **See also** *print-through*.

take. A single recording pass of a track or a complete performance of a song.

talkback. A system that allows the engineer to communicate with talent during a recording session without having to physically walk out into the studio or isolation booths—sort of like an intercom system. Typically, a microphone is set up that feeds into the headphones the musicians are using to monitor the session. A switch allows the engineer to turn the microphone on and off as desired. Talkback may be built into a mixer or may be part of a monitor control box or headphone amp system.

talkback mic. A microphone used by the engineer to communicate with the talent in the studio. ☞ **See also** *talkback*.

tangential mode. A room mode caused by sound waves reflecting across four surfaces in a room, such as four walls or two walls, the ceiling, and the floor. Tangential modes are about half as strong as axial modes and twice as strong as oblique modes. See Figure T.2.

tap. 1. A short delay that is created by tapping some signal off from a longer delay, with its own pan and feedback settings. ☞ **See also** *multi-tap delay*. 2. An output from a transformer. A power transformer may offer taps at different voltage levels, while an audio transformer may offer taps at different signal levels.

tape compression. A characteristic of analog recording tape in which the dynamic response becomes nonlinear as levels near maximum. As the signal exceeds the maximum level the tape can handle, transients and peaks are "squashed" in a gentle way and distortion increases, resulting in a fatter, warmer sound. Many engineers enjoy the sound that tape compression provides, particularly on

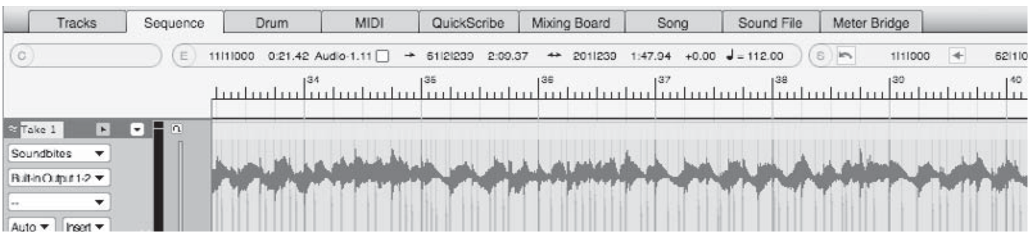

Figure T.1 Tabs on a window in a computer program allow a user to select different views without opening an additional window.

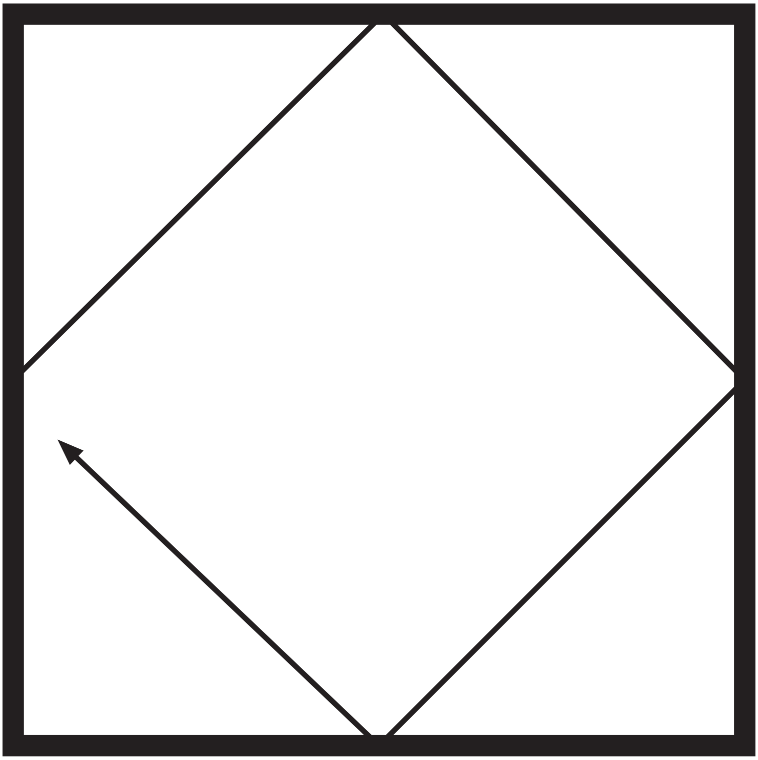

Figure T.2 A tangential mode is caused by a sound wave reflecting across four surfaces in a room.

drums and other transient materials, and use it as a part of creating a sound. A variety of digital and modeling processors are available that attempt to simulate the effects of tape compression.

tape delay. Originally, an echo effect created using a reel-to-reel tape recorder. Tape delay is created by recording a signal onto analog tape using the recorder's record head, then immediately playing back the signal using the playback or repro head. The distance between the record and playback heads and the tape speed determine the delay time. Multiple echoes could be created by feeding part of the delayed signal back into the input of the recorder. Later dedicated tape echo units, such as the Echoplex and others, were introduced to provide echo effects. These had the advantage of being able to physically slide the playback head in relation to the record head to change the delay time. A characteristic of all tape-based delays is that the signal degrades as it is repeated; noise and distortion accumulate, and the high-frequency response is reduced. Today, these effects are created using analog, digital, and modeling delays.

tape head. A transducer in a tape recorder that converts electrical signals into magnetism (record and erase heads) or magnetism into electrical signals (playback or repro head). Tape heads can be either stationary, as in analog and some digital recorders, or rotating, as in R-DAT and other digital recorders.

tape saturation. ☞ See *tape compression*, *saturation*.

taper. The rate at which a potentiometer or fader changes signal levels as it is turned or moved. A linear taper pot changes levels in a linear fashion—at 1/4 of its travel, the signal is at 25%; at half of its travel, the signal is at 50%, and so on, so in a technical sense, the level change is even. An audio taper pot, on the other hand, uses a logarithmic curve that follows the human ear's response to level change, so the volume change *sounds* even. Audio taper pots are typically used for volume controls, while linear taper pots can be used to control other parameters and settings. 2. A person who recorded Grateful Dead concerts on portable cassette or DAT machines.

tap tempo. A function in some time- or tempo-based delays, sequencers, and modulation devices that allows the user to set the tempo or delay time manually by tapping on a switch or footswitch. This allows the device to follow the tempo set by the musician.

TB. ☞ See *terabyte*.

TDIF. TASCAM Digital Interface. A digital audio connection protocol developed by TASCAM for interconnecting their DA-88 and other MDM recorders and digital mixers. TDIF carries eight channels of digital audio over a single shielded cable that uses 25-pin D-sub connectors.

TDM. Time Division Multiplexing or Time Domain Multiplexing. A technique for sending multiple signals down a single cable simultaneously used for early telephone and telegraph communications. The concept was updated and adapted by Digidesign as the basis for how signals are handled in higher-end DSP-based versions of Pro Tools.

technical ground (a.k.a. technical earth). The central ground point of a studio using a star ground scheme. The technical ground is usually a large copper rod driven into the earth.

telescoping shield. A shield in a balanced cable that is only attached at one end and is left "floating" on the other end. This provides all the shielding benefits but reduces problems with ground loops.

temperament. A wide range of methods for adjusting or tempering the pure mathematically correct tuning of the notes and intervals in a scale. This opens up different possibilities for musicians compared to using just intonation. The most common temperament in Western music is equal temperament, in which all half steps have the same size, dividing the octave into 12 equal parts. But a variety of other temperaments are available and in use, including meantone temperament, well temperament, and more exotic scales containing more than 12 notes per octave, such as 19-, 22-, 24-, 31-, 53-, 72-, and 88-tone equal temperament. ☞ **See also** *tuning*.

template. A guide document in a computer program that is used as a basis for creating a new document. For example, a DAW template might include an arrangement of empty audio tracks that are preset to be recorded. This makes it quick and easy to get a session or project going without wasting a lot of time on repetitive preparations.

tempo. The speed of a piece of music, measured in the number of beats that occur in a minute (BPM).

tempo map. A list of the tempo changes that occur during a piece of music. Often a special track in a DAW or sequencer is used to hold the tempo map for a song.

tempo matching. 1. Synchronizing the tempo of two pieces of music using a variety of methods, including time-compression/expansion. 2. Matching a delay or other time-based effect to the tempo of a piece of music.

temporary threshold shift (a.k.a. TTS). A protection mechanism built into the human ear that reduces sensitivity during exposure to high volume levels. Extended exposure to high volume levels can result in temporary threshold shift becoming permanent.

terabyte (a.k.a. TB). One trillion bytes. This may be either 1,000,000,000,000 bytes or 1,099,511,627,776 bytes, depending on whether the application uses base 10 or base 2 (binary) mathematics, respectively. Typically, transmission rates use base 10, while computer storage uses base 2.

terminal strip. ☞ See *barrier strip*.

terminator. A device used to manage source and destination impedance when connecting certain types of gear together. Termination can either be from an external device or built into the devices making up a chain. A primary purpose of termination is to prevent signal reflections, which can corrupt data and reduce performance. SCSI chains are one example of chains that must be terminated for optimal operation.

test tone. A tone of a certain frequency and timbre played back in a room or through a device to help analyze performance, to measure response, or to assist with setup and/or calibration.

test tone generator. A device or program that creates test tones.

TFT. Thin-Film Transistor. A type of technology used to make active-matrix LCD displays used in laptops, monitors, and various other types of electronic gear. An array of TFTs is used to turn the pixels in the display on and off to create images on the screen.

THD. Total Harmonic Distortion. A specification indicating the amount of distortion added to a signal by a device in the form of additional harmonics. THD is the ratio of the power of the fundamental frequency versus the power of all the harmonics in the output signal from a device. The lower the THD, the cleaner or more transparent the device will sound.

THD+N. Total Harmonic Distortion plus Noise. A specification indicating the amount of noise and harmonic distortion added to a signal by a device, referenced to a particular frequency, level, and bandwidth. (Without these references, the spec is meaningless.) THD+N is less useful as a spec than separate THD and signal-to-noise specs.

theremin. A musical instrument developed by Leon Theremin in 1919. A theremin has two antennae; one controls volume, the other controls pitch. The player moves his or her hands in relation to the antennae to play the instrument—to create melodies and to add expression.

thermal noise (a.k.a. Johnson noise, Nyquist noise). A type of random, intrinsic noise caused by the motion of electrons in a wire or component. Thermal noise is the theoretical minimum amount of noise that can be attained by a component or circuit. J.B. Johnson and Harry Nyquist were among the first to study this noise phenomenon.

thin data. A function of some sequencers designed to reduce the density of MIDI data being sent in order to prevent MIDI log jam. Continuous controllers, aftertouch, and pitch bend can all produce a great deal of data when used heavily. The thin data command can delete every other piece, every third piece, or use an algorithm to delete a certain amount of MIDI data for a particular controller or message type.

third-harmonic distortion. The third harmonic of a given fundamental frequency. In analog tape recorders, the third harmonic, at an octave and a fifth above the fundamental, is the most audible distortion component. A signal level with 3% third-harmonic distortion is the reference for determining MOL (Maximum Output Level).

third party. Technically, someone outside the two parties involved in a transaction or dispute. In the audio world, a third-party manufacturer is a company who makes a product that is compatible with a product made by another manufacturer. For example, a third-party manufacturer might make a plug-in that works in another manufacturer's DAW program. Or, a third-party might make an expansion card that works in another company's synthesizer.

threshold. 1. In a compressor or limiter, the level at which gain reduction begins to occur. When the input signal crosses above the threshold level setting, the gain reduction will be triggered and will reduce the signal level by the amount set by the ratio control. When the input signal level drops below the threshold, the compressor or limiter releases the signal and allows it to pass unaltered. 2. In a noise gate or an expander, the threshold is the input level at which the gate opens. The trick is to accurately set the threshold level so that background noise below the threshold is gated or removed, but the desired signal above the threshold is allowed to pass unaltered. In general, the threshold level in a gate should be set just barely high enough to eliminate unwanted noise, allowing the desired signals to pass through without being processed.

threshold of pain. Technically, the SPL or volume level that causes pain in a listener 50% of the time. The threshold of pain varies depending on the listener and the frequency, but is generally given as anywhere from 115 dB to 140 dB SPL.

throughput. The data transmission rate for a hard drive or other storage media, a network, or another means of transmission.

through-zero flanging. A type of tape flanging effect created in the 1960s using two analog reel-to-reel tape decks playing back the same audio slightly out of sync. The signals were mixed together, and the speed of one of the decks was changed by pressing on its tape reel flange, creating a continuously moving comb filter cancellation and reinforcement. At some point as pressure on the flange is increased or decreased, the speed of one deck crosses over from faster than the other to slower than the other or vice versa. The crossover point from faster to slower or slower to faster is called the *zero point*. Modern digital and analog versions of the flanging effect use a short delay to create cancellation and therefore never pass through zero, resulting in a different sound.

thru. ☞ See *MIDI thru*.

thru box. A MIDI signal splitter, with one MIDI input and two or more MIDI thru ports, each carrying an exact copy of the MIDI signal arriving at the input.

thumb drive. ☞ See *jump drive*.

thumbnail. A small image used to represent a large image or video file. In DAWs and audio programs that support video playback, there will often be a video track. Thumbnails are used in this track to provide easier navigation and visual locating.

tick (a.k.a. click). The clock pulses that drive MIDI playback in a sequencer or DAW are sometimes referred to as *ticks*. MIDI itself has a tick, or clock, resolution of 24 pulses per quarter note (PPQN). Most sequencers offer substantially higher tick resolution, providing more accurate playback of subtle rhythmic material.

timbral. (Pronounced *TAM-brul.*) Related to the tonal quality of a sound.

timbre. (Pronounced *TAM-bur.*) The tonal quality or color of a sound.

time aligned. A speaker with more than one driver (such as separate high- and low-frequency drivers), in which the drivers are set up in such a way that the sound from all the drivers arrives at the listener's ear at the same time. This is said to result in better transient response than non-time-aligned designs, where the sound from multiple drivers arrives at the listener's ears at slightly different times.

time alignment. 1. Positioning the drivers in a speaker cabinet so that the sound from each arrives at the listener's ears at the same time. ☞ **See also** *time aligned*. 2. Using delays to correct for differences in distance from the source when using multiple microphones. 3. Using delays to correct for differences in distance from the audience when using PA systems or speakers with remote towers or distant speaker placements. 4. Sliding a track in a DAW in time to compensate for a delay.

time base. The timing reference for a system. In real time, the time base might be seconds. In video or film, it would likely be frames. In a digital audio system, it might be elapsed samples.

time-based effect. A processor that manipulates time in some fashion. Examples include a delay or echo, which produces a discrete time-shifted (delayed) repeat of a sound; flanging and chorus, which mix a delayed version of a sound with the original sound; and reverbs, which create a wash of delayed reflections based on the original sound. Many plug-ins and digital processors allow the timing of a time-based effect to be synchronized or "tempo-matched" to the tempo of the song.

time code. A signal containing timing and location information that is used to synchronize devices and to allow them to navigate to specified locations. Unlike a clock signal, time code runs at a constant rate regardless of the tempo at which a sequencer or DAW is running. There are a variety of types, including SMPTE, MTC, VITC, and others.

time compression/expansion. A type of DSP processing that changes the length (and thus the playback speed or tempo) of an audio file without changing its pitch. A variety of algorithms are available from manufacturers for doing this in standalone software packages, in DAW and other audio software, and in plug-ins. Some are better than others at performing the transformation without affecting the file's audio quality. The quality of the transformation is also dependent on the type of source material and how far the file is to be sped up or slowed down.

time-limited demo. A demonstration version of a piece of software that will only run for a specified time period. After the demo time period expires, the software ceases to work, and the full version must be purchased and installed.

timeline. A list or graphic representation of events in the order they will occur, with timing information. Sequencers, DAWs, and audio editing programs use a timeline approach, where audio files and MIDI data are laid out in tracks in sequential fashion, placed according to the time at which they occur.

time out. Some devices will only wait for a certain period of time before an event happens. If the event doesn't occur, the device will time out and return to its default state. For example, a computer set to receive a data transmission may time out and cease waiting if the transmission does not come through after a certain period of time. An everyday example would be an Internet browser timing out when attempting to load a web page from a server that is not responding.

time signature. A method for indicating the meter for a piece of music. A time signature looks something like a fraction, with the top number indicating the number of beats in a measure, and the bottom number indicating what rhythmic value serves as a full beat. For example, in 4/4 time, there are four beats per measure, and the quarter note gets the beat. In 6/8 time, there are six beats per measure, with the eighth note getting the beat. Time signatures are used in standard musical notation and in sequencers and DAWs.

time stamp. A method for encoding a time location into a piece of audio, MIDI, or other data. Time stamps can be used to place an audio file at an exact location in a track, to accurately play back MIDI data, or to otherwise place or play data accurately.

tinnitus. The phenomenon of hearing sound when no sound is present, often because of hearing damage, because of earwax buildup, or from some medications. Infections; jaw, head, or neck problems; and other medical disorders can also result in the problem. Tinnitus usually manifests as constant or intermittent ringing in the ears, though some may hear hissing, roaring, clicking, or other sounds.

tiny telephone (a.k.a. TT, Bantam). A smaller, more compact version of the standard 1/4-inch phone plug or jack. Tiny telephone jacks are often used for studio patch bays, as more connections will fit into a given amount of space versus using standard phone jacks.

TOC. Table Of Contents. A track on an audio CD that stores information about the disc, including the number of tracks on the disc, their start locations, and more. A CD player reads the TOC in order to locate and navigate to the audio tracks.

tolerance. In electronics, tolerance indicates the accuracy of a standard, specification, or measurement. For example, for an electronic component, the tolerance is how far off a component's value can be from the manufacturer's stated rating and still be acceptable or within spec. Tolerance is usually given as a percentage; tighter-tolerance components are more accurate to their rating but are typically more expensive.

tone. 1. A distinct musical pitch. 2. The timbral or spectral character of a sound. 3. In musical terms, a whole step.

tone burst. A brief signal (usually a sine wave) that is used to excite a room so acoustical measurements can be made.

tone generator. ☞ See *module.*

tone module. ☞ See *module.*

tone sweep. ☞ See *sweep.*

tonewheel. The sound-generating component in a Hammond organ, Telharmonium, or other electromechanical keyboard instrument. In a Hammond organ, a tonewheel is a motor-driven two-inch metal disk with notches or bumps on its edge. The tonewheel spins near a magnetic pickup when a key is pressed, creating an electrical signal that represents the fundamental pitch of the note. The rotation speed and number of bumps/notches on the disc determine the frequency of the note. Additional tonewheels can be added into each note to serve as harmonics in the signal, in the same way that pipes are layered together to create a sound in a pipe organ.

tool palette. A small window in a program that contains a collection of graphical editing tools, with each tool dedicated to a specific task. Typically, the computer mouse is used to select a tool from the palette; the mouse pointer assumes the shape of that tool, and is then used to perform an editing task on data. See Figure T.3.

Figure T.3 A tool palette is a floating collection of editing tools. The tool palette can be positioned where necessary for quick access. This example shows a palette used to select note values in a music notation program.

Figure T.4 A toolbar is an area of a program's GUI that contains buttons providing quick access to frequently used commands.

toolbar. A horizontal or vertical bar or area of a program's window that contains a collection of buttons that can be clicked with the mouse to access frequently used commands. A toolbar may be dedicated to a certain type of buttons, such as a formatting toolbar, or it may contain buttons that access more global or general commands. See Figure T.4.

top end. A generic term used to refer to the high frequencies, high end, or treble portion of a signal.

toroid. A shape resembling a torus—a doughnut is a good example. Toroid or toroidal transformers are used in some electrical equipment because they reduce magnetic-field problems.

TOSLink. Short for Toshiba Link. An optical digital audio interconnect protocol developed and trademarked by Toshiba in 1983. TOSLink carries stereo S/PDIF-format digital audio over a fiber-optic cable. Ten meters is the maximum length for TOSLink cables, though for best reliability, five-meter lengths or less are recommended. Alesis used the same fiber-optic cables and connectors for their eight-channel lightpipe format as are used for TOSLink, though lightpipe and TOSLink are not compatible with one another.

touch sensitive. 1. A keyboard that responds to how quickly a key is pressed in order to simulate the response of an acoustic piano. 2. A fader or other control on a mixer or control surface that becomes active when it is touched by a finger. 3. A type of surface or control that responds to pressure or other physical variation.

track. 1. A piece of recorded music. 2. A discrete recorded signal, such as MIDI or audio. 3. Concentric circular bands of data on a hard drive or compact disc. 4. The process of recording audio.

track-at-once (a.k.a. TAO). A CD-R or CD-RW burning mode in which one track at a time can be written to the disc, and other files/tracks can be added later. Burning a CD with track-at-once results in turning the laser on and off after each track or file. ☞ **See also** *disc-at-once*.

trackball. A control device used as an alternative to a mouse. A trackball has a ball in a socket that is rolled by the user to move the mouse pointer. Trackballs contain one or more buttons that duplicate the click and click-hold functions of the mouse. Because it is stationary, a trackball requires less space than a mouse. Some users find a trackball more ergonomic and comfortable than a mouse.

tracker. A very basic type of sequencer that allows the user to arrange samples on a timeline. Parameters and effects are controlled using hexadecimal codes. The first tracker software emerged in 1987 for the Commodore Amiga computer platform; early software supported four monophonic tracks at eight-bit resolution. Later software supported eight tracks and 16-bit resolution. Tracker music has been featured in numerous computer games, as well as in dance music productions. Tracker software remains available and is still being developed today.

tracking. 1. The process of recording the parts that will make up a piece of music or audio production. 2. The ability of a pitch-to-MIDI converter to accurately follow the notes played on a guitar or other instrument. 3. Using tracker software to create music. 4. On rotating head recorders (a.k.a. helical scan recorders), such as VCRs and DAT machines, the dual head spins at an angle to the tape as the tape moves; the data is written on the tape as an angled stripe. There is also a control track on the tape that contains information about the recording. Tracking is the relationship of the heads' position compared to the data and control tracks on the tape. Some VCRs offer manual tracking adjustment to help compensate for stretched or worn tapes; most helical scan recorder/players use automatic tracking control.

transcription. 1. A representation of a musical performance in standard music notation or tablature. 2. The process of converting audio or MIDI information to standard music notation.

transducer. A device that converts one type of energy into another. Common examples include microphones, pickups, speakers, and tape recorder heads.

transfer rate. ☞ See *data transfer rate*, *bit rate*.

transfer time. The time it takes for data transfer; transfer time is determined by bit rate. Together with seek time and rotational delay, one of the factors that comprise hard drive or optical disc access time.

transformer. An electronic component consisting of two coils of wire wrapped around the same magnetic core. An AC voltage in one coil will transfer through inductance to the other coil, multiplied by the turns ratio, or the ratio of the number of turns in one coil compared to the number of turns in the other coil. Transformers are used in audio devices to step up or step down voltages to the required levels, as well as to isolate circuits—when two signals are "connected" using a transformer, there is no direct physical or electrical connection—to prevent ground loops or other problems.

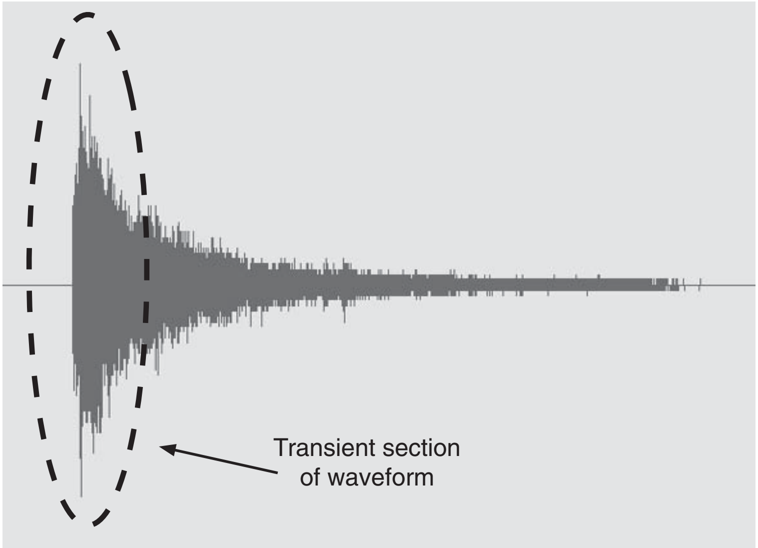

Figure T.5 A transient is a quick spike in level, often found in the attack portion of a sound.

transformerless. A circuit that does not use transformers. Some designers feel that transformers "color" the sound quality of a device in undesirable ways (other designers feel the opposite) and use other components to serve the function of a transformer.

transient. A quick, non-repeating waveform that spikes to a higher level than the surrounding sounds or the average level of the signal. Examples include the pluck or pick portion of a guitar note, the hammer strike portion of a piano note, and the attack portion of most percussion instruments. See Figure T.5.

transient response. The ability of a device to accurately reproduce a transient or quickly changing signal. Good transient response requires the device to respond very rapidly to changing levels. Poor transient response results in dull, flat, dynamically static sound quality.

transistor. A solid-state semiconductor component that passes or conducts electricity under certain conditions and does not conduct well under other conditions. Transistors were originally designed to function in the same way as vacuum tubes, though there are sonic and performance differences between the two technologies. There are several types, but the basic bipolar transistor has three "terminals" (the collector, base, and emitter); when a transistor is used as an amplifier, a large supply voltage is applied to the collector, while a source voltage (such as an audio signal) is applied to the base to control the supply voltage. The output signal on the emitter is a larger, amplified version of the source or control voltage. A bipolar transistor can be used in similar fashion to switch a voltage on and off. ☞ See also *FET*.

transition band. The range of frequencies between the passband and stopband of a filter.

translation. How consistently an audio mix holds up when heard on different playback systems.

transmission loss. ☞ See *sound transmission loss*.

transondent. Sonically or acoustically transparent. The term is typically applied to acoustic materials such as grille cloth, pop filters, and others.

transparent. A term used to describe a device that does not change or "color" the tonal quality of a signal.

transport. The motor and other mechanical components that move the tape in a digital or analog audio or video recorder.

transport controls. Controls that are used to operate the transport mechanism in an audio or video tape recorder or CD or DVD player/recorder. Examples include play, stop, fast-forward, rewind, pause, record, and jog and shuttle. Though hardware and software DAWs; sequencers; and other audio, video, and MIDI programs do not contain mechanical transports, graphical "virtual" transport buttons are provided for controlling playback, record, and locate functions. Some digital mixers, MIDI controllers, and control surfaces also supply transport buttons that send MIDI messages that are used to control software audio, video, and MIDI playback and other functions. See Figure T.6.

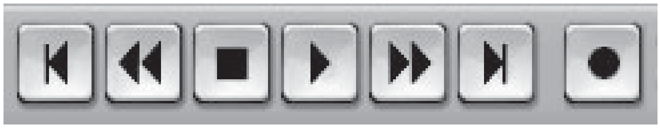

Figure T.6 Even though DAWs and other digital audio devices and programs don't have mechanical transports, familiar transport controls are usually still provided for controlling playback, recording, and location.

transpose. 1. To change the key of a piece of music. 2. To raise or lower the pitch of a note, musical passage, or complete piece by a certain amount.

transwave. A type of wavetable synthesis found in Ensoniq synthesizers and samplers. A transwave was a large sample made of up to 128 smaller waveforms. A program or preset could access and loop one of the smaller waveforms or play through the entire sample, resulting in an evolving sound.

trap. ☞ See *bass trap*.

treble. 1. The high-frequency portion of a sound or signal. There is no exact treble range, but it can be arbitrarily defined as roughly 3 to 4 kHz up through 20 kHz. 2. A control used to adjust the level of high frequencies in a signal.

tremolo. A wavering, pulsating, repeating, cyclical change in the volume level of a note or sound. Tremolo is often confused with vibrato, which is a cyclical change in pitch.

tri-amp. Using a crossover to split a signal into three frequency ranges (high, mid, and low) and sending those ranges to separate amplifiers and speakers optimized for those ranges.

triangle wave. A cyclical waveform whose shape resembles a triangle, with a linear increase/decrease in amplitude. A triangle wave contains the fundamental frequency and odd integer harmonics and is one of the basic waveforms available in analog synthesis. (Other common waves in analog synthesis include sine, sawtooth, and square.) Triangle waves have a smooth sound, somewhat similar to a sine wave, but with additional treble content. See Figure T.7.

trigger. 1. ☞ See *sidechain*. 2. Using a MIDI controller to initiate an event of some sort, whether a note, a control change, a lighting change, or another event. 3. To cause a sample or loop to play back. 4. To automatically replace recorded drum sounds with samples. ☞ **See also** *trigger input*.

trigger input. 1. ☞ See *sidechain*. 2. An input found on some electronic drum modules and sample-playback plug-ins that allows the drum sounds to be triggered by other sounds, such as recorded drums on a track. This allows the recorded drums to be replaced or layered with samples.

trigger pad. An electronic drum pad or a finger pad on a keyboard or other controller that sends a MIDI note or other data when it is struck.

trigger sync. A type of synchronization in which the slave device is not continuously locked to the master device. Instead, the master device is used to start the playback of an event (such as an audio region) at a specific time; the event then plays back using its own clock.

trim. A control found on mixers and other types of gear that determines the input gain for each channel. Trim may affect mic, line, and/or instrument-level inputs. Correctly setting the input trim is essential for proper gain staging and for maintaining a good signal-to-noise ratio.

triode. A vacuum tube with three terminals.

troubleshooting. The process of diagnosing and fixing a problem.

TRS. Tip-ring-sleeve. A type of 1/4-inch, 1/8-inch, or other phone-type audio connector with three

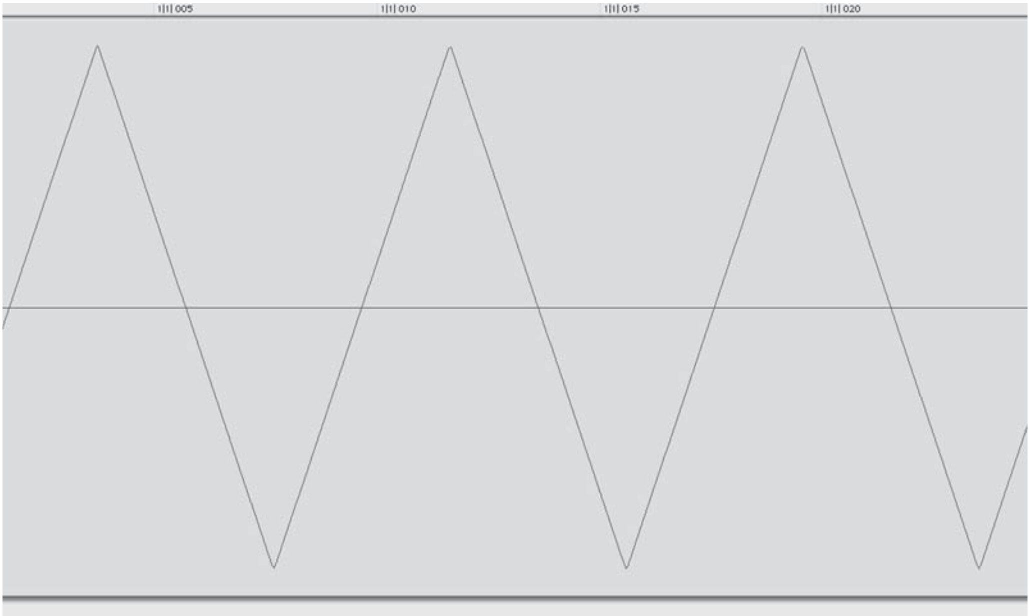

Figure T.7 A triangle wave contains a fundamental frequency and odd harmonics.

sections: the tip of the connector, the barrel (or sleeve) of the connector, and a band or ring between them. TRS connectors can be used for balanced signals (the tip carries positive, the ring carries negative, and the sleeve is the ground) or unbalanced stereo signals (the tip is the hot for one channel, the ring is the hot for the other channel, and the sleeve is the common ground).

true bypass (a.k.a. hardwire bypass). A type of processor, typically a guitar or bass stompbox, in which the signal does not pass through any circuitry when the unit is bypassed; the input connects directly to the output.

true stereo. A type of processor that processes a stereo signal as two completely separate channels. Many early digital processors had a single input or would sum a stereo input to mono before processing and would generate a synthetic stereo effect from the summed or mono input. A true stereo processor maintains a discrete signal path for the two channels and processes each independently.

truncate. 1. To reduce the length of a digital word by dropping or "chopping" off bits representing very low levels. 2. To edit the length of an audio sample or region.

TS. Tip-sleeve. A type of 1/4-inch, 1/8-inch, or other phone-type audio connector with two sections: the tip of the connector and the barrel (or sleeve) of the connector. TS connectors are used for unbalanced signals (the tip carries positive and the sleeve is the ground).

TT. ☞ **See** *tiny telephone*.

tube. ☞ **See** *vacuum tube*.

tuned absorber. Acoustic device designed and optimized for absorbing sound waves at a particular frequency or range of frequencies.

tune request. A MIDI System Common message that tells the receiving device to tune its oscillator(s) to standard (A-440) pitch.

tuner. An electronic device used to detect and indicate the pitch of a note.

tuning. 1. To correct or adjust the pitch of a signal. 2. A mathematical system for determining the pitches of notes and the spacing of intervals in a scale or other musical material. A tuning uses only "just" or pure mathematical intervals; a temperament adjusts those pure intervals for various reasons. ☞ **See also** *temperament*. 3. To adjust the pitch of an instrument to match a standard frequency or to establish typical intervals.

turnkey. A complete system that has been assembled and tested by the manufacturer or retailer and is ready to use right out of the box. A turnkey system includes all components, devices, software, and other parts necessary for operation.

tutorial. A part of a device's or software application's documentation that is intended to instruct the user on how to utilize the device or program or how to use a particular function or feature.

TVA. Time Variant Amplifier. A term used by Roland for a DCA-style amplifier stage.

TVF. Time Variant Filter. A term used by Roland for a DCF-style filter.

tweeter. High-frequency transducer in a multi-driver speaker system. There are many types of tweeters in use, including cone, dome, horn, piezo, ribbon, and others.

twisted pair. A type of cable that uses two insulated wires twisted into a spiral. Twisting the pair of wires reduces crosstalk and induction, as well as EMI problems. Network and other high-speed data cables often use one or more twisted pairs of conductors.

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U

U. Abbreviation for Modular Unit. ☞ **See** *rack space*.

UDMA. Ultra Direct Memory Access. A protocol that offers twice the transfer rate of ATA by allowing devices direct access to RAM. UDMA/33 offers up to 33 MB/second performance, while UDMA/66 provides up to 66 MB/second data transfer.

UI. ☞ **See** *user interface*.

UL. Underwriters Laboratory. A public safety agency dating back to the 1890s. The UL is underwritten, or supported, by insurance companies. Its mission is to test electrical products for safety, to establish standards, and to certify products.

Ultra ATA (a.k.a. ATA-4). A high-speed version of the ATA protocol that provides burst transfer rates up to 33 MB/second.

Ultra DMA. ☞ **See** *UDMA*.

Ultra SCSI. A family of parallel SCSI protocols, some of which use LVD (Low-Voltage Differential) technology to greatly increase performance, increase maximum cable lengths, and increase the maximum number of devices on a SCSI chain. The Ultra Wide SCSI versions double the data path from 8-bit to 16-bit, doubling performance. See Table U.1.

Table U.1 The Ultra SCSI Family

Protocol	Width	Throughput
Ultra SCSI	8-bit	20 MB/second
Ultra Wide SCSI	16-bit	40 MB/second
Ultra2 SCSI	8-bit	80 MB/second
Ultra2 Wide SCSI	16-bit	160 MB/second
Ultra3/Ultra 160 SCSI	16-bit	160 MB/second
Ultra 320 SCSI	16-bit	320 MB/second
Ultra 640 SCSI	16-bit	640 MB/second

ultrasonic. Frequencies above the upper limit of human hearing, which is typically accepted as 20 kHz.

U-Matic. A 3/4-inch videocassette recording format developed by Sony. The Sony PCM-1600 adaptor allowed a U-Matic video machine to record stereo digital audio, and the machines quickly became the standard format for delivering CD masters for duplication. In fact, one story has it that the CD-standard sampling rate of 44.1 kHz is derived from the U-Matic's horizontal-sync rate. The subsequent PCM-1610 and PCM-1630 decks also used U-Matic format videocassettes for storage and were popular for mastering applications. By the 1990s, Sony's 1/2-inch Betacam format had made the U-Matic machines largely obsolete, though U-Matic machines are still in use today.

unbalanced. An audio connection that has a positive wire but that uses the cable shield to serve as both the negative signal conductor and the ground. Unbalanced cables do not offer noise cancellation (there is no common mode rejection) and are susceptible to noise and interference pickup. For this reason it is best to keep unbalanced cable runs as short as possible; generally, 30 feet is given as the upper maximum for unbalanced cable lengths.

uncolored (a.k.a. straight-wire). A characteristic of a device that introduces no coloration or tonal changes to a signal. Just as coloration is desirable for certain applications, lack of coloration is equally desirable for other applications. This term is typically applied to microphones and mic preamps, as well as to other types of gear.

undo. A command/function in most computer programs that allows the last operation to be reversed or undone, returning the data to where it was before the operation. Most modern programs offer many levels or even unlimited undo, allowing the user to revert the data back to an earlier point, before it was edited or other operations were carried out. ☞ **See also** *redo*.

unformatted capacity. The maximum data capacity of a disk drive before it has been formatted to work with a computer. Some portion of the disk space is required for formatting and "housekeeping" information, so the formatted capacity will always be smaller than the unformatted capacity. What really matters to the end user is the formatted capacity, but drives are usually advertised and labeled with their unformatted capacity.

unidirectional. Literally, "in one direction." Unidirectional is generally used as a synonym for *cardioid* with reference to microphone polar patterns, though technically, the term could be applied to any directional microphone. ☞ **See also** *cardioid*.

uninstall. To remove a program from a computer. Usually this involves more than just deleting the program itself from the machine's hard drive. Typically, a program installer puts a variety of support files, drivers, and other software bits into the computer's system and other places. All of these must be removed to completely de-install a program. Most software packages that use an installer application will also uninstall their software, removing all traces from the computer.

unison. Two or more vocal or instrumental parts of the same pitch happening simultaneously.

unison mode. A mode available on many synthesizers that layers detuned oscillators for a fatter sound, usually at the expense of reduced polyphony.

unity. ☞ See *unity gain*.

unity gain. Unity gain is where the gain of a signal going into a device equals the gain coming out. Unity gain is helpful for maintaining gain staging, operating levels, and good signal-to-noise ratios.

Universal Binary. Macintosh-compatible software that contains executable code for both older PowerPC processors as well as the newer Intel processors. When the software is installed onto the computer's hard drive, the operating system detects that the program is Universal Binary–compatible and automatically loads the code that is compatible with the processor type.

Universal Serial Bus. ☞ See *USB*.

UNIX. A trademarked name for the computer operating system developed at Bell Labs in 1969 that is popular for servers and high-end computer workstations at universities, government and research facilities, and major companies. By 1984 there were more than 100,000 UNIX installations in the world, with 750,000 in place by 1987. Over the years, many companies developed their own implementations of the OS. In 1993 Novell, who owned UNIX at the time, transferred the trademark to the X/Open Company (now part of The Open Group), which developed a single specification for APIs. UNIX pioneered, provided, or popularized many features that are now common in other operating systems, such as the hierarchical file system, the TCP/IP networking protocol (important for the development of the Internet), and many more. UNIX is a multitasking, multi-user OS with protected memory (which prevents a program from interfering with another program) that uses many small utilities along with the kernel, or master control program. Current implementations built upon the UNIX standard include Linux, Macintosh OS X, Solaris, and more.

unzip. To extract data-compressed files from a zip archive.

UPS. Uninterruptible Power Supply. A battery backup device that automatically takes over to provide AC power if the normal electric service is interrupted. A UPS typically isn't intended to provide power long-term; rather, the UPS is intended to give the user time to save any critical data and shut down equipment before the power runs completely out.

upsampling. Increasing the sampling rate of digital audio. Typically, digital audio is upsampled so that filtering and other processes can be performed at very high frequencies, reducing artifacts that can be heard in the audible frequency range. Upsampling is sometimes confused with oversampling, though the two are different: Upsampling is a type of sample rate conversion in the digital domain, while oversampling is operating a converter at a very high sample rate. Upsampling is used, for example, by some plug-ins to increase the precision of DSP processing calculations.

USB. Universal Serial Bus. A standard for connecting peripheral devices to a computer. USB was jointly developed by a number of manufacturers called the USB-IF (USB Implementers Forum). USB is a hot-pluggable, plug-and-play protocol that supports simultaneously connecting up to 127 devices in series to a computer. There have been three versions: USB 1.0, USB 1.1 (which fixed a few problems), and USB 2.0, which offers higher speed performance. USB supports three data rates: Low (1.5 Mbps) for keyboards, mice, and other interface peripherals; Full (12 Mbps) for storage devices, audio interfaces, and more; and High (480 Mbps), which is supported in USB 2.0 for high-speed storage, audio interfaces, and other bandwidth-hungry devices. There is a Super Speed mode (4.8 Gbps) that is under development at this writing.

USB flash drive. ☞ See *jump drive*.

USB stick. ☞ See *jump drive*.

USB 1.0. ☞ See *USB*.

USB 1.1. ☞ See *USB*.

USB 2.0. ☞ See *USB*.

user-definable. A parameter or preference that can be set by the user.

user interface. The controls, display, and other items that allow a human to operate a piece of gear or software.

user preset. A program or preset created by the user of a piece of gear, as opposed to a factory preset, which is created by the manufacturer and included with the product.

utility. A piece of software designed to perform a specific function, usually a housekeeping task, such as disk maintenance, various types of setup, and a wide variety of other duties.

UV22HR (a.k.a. UV22). Apogee Digital's patented technology for increasing dynamic range at lower resolutions, similar to dithering or bit-mapping, but using a 22-kHz "bias" tone to modulate the lowest bits in the 16-bit signal. UV-22 is said to provide resolution equal to that of 20 bits using 16-bit digital audio, without increasing background noise.

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V

vacuum tube. An electronic component that features electrodes (anode or plate and filament or cathode) inside a glass tube from which all the air has been removed, creating a vacuum where electrons can easily flow. Additional items inside the tube, including the grid, are used to control the electron flow and therefore the current, or amplification, the tube is providing.

valve. British term for tube.

vaporware. A product that has been announced by a manufacturer but that has not yet shipped or become available to end users.

variable bit rate (a.k.a. VBR). Audio data compression algorithms that vary in response to the input signal. The bit rate is increased when necessary to accurately represent the input signal, and it is reduced when the input can be represented with a lower rate. Variable bit rate algorithms generally provide better quality while using less space than constant bit rate (CBR) algorithms.

variable mu. A type of tube with the ability to decrease gain as the level of the input signal increases. In audio, variable mu designs are used in certain compressors.

variable pattern microphone. A microphone with a switchable or continuously variable polar pattern. In most cases, a variable pattern mic will have dual diaphragms, which are combined electronically to create the various patterns.

varispeed. A feature of analog tape decks (that has been emulated by DAWs) that changes the speed of the tape during recording and playback. Increasing the speed makes the audio play faster and raises the pitch. Conversely, reducing the speed lowers the pitch and plays the audio back more slowly.

VAST. Variable Architecture Synthesis Technology. A synthesis system developed by Kurzweil for the company's K-series synthesizers (K2000, K2500, K2600, and so on). VAST uses algorithms to reconfigure how the oscillators and other synth "modules" are arranged and connected. The VAST engine also provides numerous modulation and processing capabilities for creating expressive and complex sounds.

VBR. ☞ See *variable bit rate*.

VCA. Voltage Controlled Amplifier. An amplifier circuit where the output gain can be controlled using an external voltage. VCAs are common in analog synthesizers, where they can be controlled by envelope generators to shape the volume of sounds. VCAs are also used in some dynamics processors, such as compressors and limiters, and are found in some automated mixers.

VCA Group. Voltage Controlled Amplifier Group. A function of some live mixers that allows the engineer to control the level of a selected group of channels from one fader, without having to route those channels through a subgroup. Essentially, the VCA Group fader serves as a remote control, sending a voltage to a VCA on each of the channels being controlled.

VCF. Voltage Controlled Filter. A filter where the cutoff frequency and sometimes other parameters can be controlled using an external voltage. VCFs are common in analog synthesizers.

VCO. Voltage Controlled Oscillator. An oscillator where the frequency of the waveform can be controlled using an external voltage. In analog synthesizers, each additional volt raised the pitch by one octave (doubled the frequency). So each key in a keyboard controlling an analog synth would send out 1/12-volt more than the previous key in order to raise the pitch by a semitone.

vector synthesis. A type of synthesis developed by Sequential Circuits and later used by Korg, Yamaha, and other manufacturers. A vector synth has four simultaneous sound sources; the user can manually crossfade between the different sounds using a joystick or other control, or the synthesizer can be programmed to automatically crossfade through the sounds in a certain pattern. This creates sonic movement and sounds that can evolve over time.

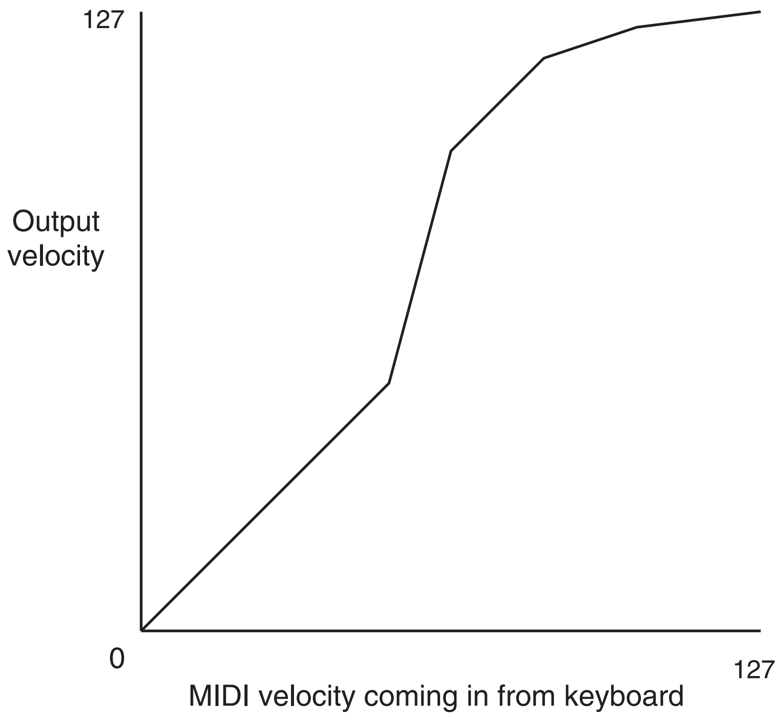

Figure V.1 A velocity curve in a device can be used to customize the response or feel of the keyboard to match the player's touch.

velocity. How fast a player presses a key on a keyboard (not how hard, how fast)—literally, the time it takes for a key to go from the up position to the down position. Velocity is sent as part of a MIDI note on message and has a range of values from 0 to 127. Velocity is typically routed to control the volume of the note played by a key, but it can be used to control any other parameter, such as filter cutoff (higher velocity makes a note brighter), reverb amount, or any modulation destination that is supported by the receiving device.

velocity curve. A function in some keyboards that alters the instrument's response to MIDI velocity messages to customize the "feel" of the keyboard (see Figure V.1). For example, scaling back velocity with a curve will make the player use a heavier touch in order to get the desired response from the keyboard.

velocity microphone (a.k.a. pressure-gradient microphone). A microphone in which the output voltage is related to air particle velocity. Ribbon microphones are velocity or pressure-gradient microphones.

velocity response curve. ☞ See *velocity curve*.

velocity split. ☞ See *velocity switching*.

velocity switching. A function of some synths and samplers that can switch between two or more presets or samples in response to MIDI velocity. Play softly, and one sound is heard. Play hard, and a second sound is heard. See Table V.1.

verb. Short for reverb.

VESA. Video Electronics Standards Association. An organization that develops standards for video and multimedia systems in personal computers. Among the standards developed by VESA are various connectors, the SVGA video standard, the VESA Local Bus high-speed video bus, and more. www.vesa.org.

VGA. Video Graphics Array. A display system developed by IBM that has become standard in personal

Table V.1 Using MIDI Velocity to Switch between Samples in a Piano Patch

Velocity	Sample Played on Middle C
0–20	Very soft piano note sample
21–40	Soft piano note sample
41–60	Medium-soft piano note sample
61–80	Medium piano note sample
81–100	Loud piano note sample
101–110	Very loud piano note sample
111–127	Extremely loud piano note sample

computers. VGA uses a 15-pin connector and offers up to 720×480 resolution and 262,144 colors.

vibrato. A musical effect used by instrumentalists and vocalists to add expression to a performance. Vibrato is characterized by cyclical, repeating variations in pitch, added manually during a performance or programmed into a synthesizer or sampler using an LFO to drive the pitch changes. ☞ **See also** *tremolo*.

video sync. ☞ See *blackburst*.

virtual. Existing only in computer software.

virtual analog synthesizer. A digital synthesizer that emulates or models an analog synthesizer. Physical modeling is used to re-create the characteristics of the components and circuits in an analog synth so that digital signals can be produced that sound as if they were generated and processed by real analog circuitry.

virtual instrument. A computer program that functions as a synthesizer or sampler, often as a plug-in within a DAW. Some virtual instruments attempt to "model" or emulate acoustic instruments or existing digital or analog hardware synthesizers; others strive to create entirely new sonic production and processing capabilities that do not have parallels in the hardware instrument world.

virtual track. 1. Tracks in a DAW that have been recorded into a song or session, but are not active or being played or used. 2. Tracks that have been recorded but cannot play because there are not sufficient system resources to play them. For example, a hardware digital recorder might be able to play up to 16 tracks simultaneously, but each of those tracks might have 16 "virtual" tracks that can be selected to play as alternate takes or different arrangements. 3. MIDI tracks that are driving synthesizers or samplers, but are not recorded to physical audio tracks.

VISE. Visual Installer Setup Environment. A program developed by MindVision that is designed to install other software products. Many software manufacturers use VISE installers to simplify the installation of their software. Just launch the VISE installer, and the software takes care of the rest, including decompressing files and placing all the software elements where required.

Vista Core Audio. Microsoft term for the audio protocols that are used in Windows Vista, including the Multimedia Device API, Windows Audio Session API, DeviceTopology API, and EndpointVolume API, which provide applications with access to a variety of capabilities when recording and playing back audio.

VITC. Vertical Interval Time Code. Time-code information encoded into the frames of a video signal instead of recorded to a linear audio track. The advantage of VITC over LTC is that the time code can be read while playback is stopped, which is not possible with LTC.

VO. ☞ **See** *voiceover*.

vocal booth. A small isolation room in a recording studio designed and treated for vocal recording, though other instruments are also often recorded in vocal booths. ☞ **See also** *isolation booth*.

vocoder. Voice Operated Encoder. An electronic instrument that uses the characteristics of a control signal to configure a number of filters that process the input signal. For example, the signal from a vocal microphone is used to drive filters that are processing a synth sound, effectively making the synthesizer "speak" or "sing."

voice. 1. A note of polyphony in a synthesizer or sampler. 2. Another name for a preset or program.

voice coil. A coil of wire that is suspended in a magnetic field in a speaker. When signal flows through the coil, the assembly moves in the magnetic field, moving the speaker's cone and creating sound waves.

voiceover. An announcer's or commentator's voice that is recorded over a background bed of music for an advertisement or other program material.

voice stealing. ☞ See *dynamic allocation*.

voicing. 1. Programming or creating synthesizer sounds. 2. Setting up and optimizing a studio monitor or live sound system. 3. Adjusting the sound of an acoustic piano to achieve a particular character.

volatile memory. RAM that loses its contents when the power is shut off.

volt. The unit of voltage. Named for Italian physicist Alessandro Volta.

voltage. A difference in electrical charge between two points.

voltage controlled amplifier. ☞ See *VCA*.

voltage controlled filter. ☞ See *VCF*.

voltage controlled oscillator. ☞ See *VCO*.

voltage regulator. A device that maintains a steady AC-line voltage despite fluctuations in the voltage coming from the wall socket. For example, in the United States, a voltage regulator might regulate wall voltage in the range of 80 to 130 volts to the standard 117 volts. Most voltage regulators quickly switch taps on transformers to maintain constant output voltage.

voltage sag (a.k.a. brownout). A temporary reduction in electrical power. In some cases, voltage sags are caused by the power company; in other cases, starting a high-current device will cause a brief sag in power. Voltage sags can wreak havoc on microprocessor-based devices.

voltage spike. ☞ See *power spike*.

volume. 1. The loudness of a signal, subject to the perception of the listener. 2. A computer hard drive or partition on a hard drive. 3. MIDI Continuous Controller #7, which is specified to control the

Figure V.2 The VST standard developed by Steinberg supports both processing and effects plug-ins and virtual instrument plug-ins. In this case, Spectrasonics' Stylus RMX groove instrument is operating as a VST plug-in within Ableton Live software.

volume level of a particular MIDI channel's instrument.

VST. Virtual Studio Technology. A protocol developed by Steinberg and licensed to third-party developers and manufacturers for integrating software synths, samplers, effects, and processing plug-ins into a DAW. VST is a cross-platform standard, though it is not possible to use Windows VST plug-ins with Macintosh hosts and vice versa.

VSTi. ☞ See *VST instrument*.

VST instrument (a.k.a. VSTi). A virtual instrument that runs as a plug-in under the VST standard.

VST plug-in. A plug-in that conforms to the VST plug-in standard. See Figure V.2.

VST2. The second version of VST developed by Steinberg. VST2 added MIDI in and out ports to plug-ins, support for 24-bit/96-kHz sample rates, sample-accurate editing, and more.

VST3. The third version of VST developed by Steinberg. VST3 improved performance and added dynamic I/O (in which the plug-in's I/O will automatically configure itself to the host channel's I/O configuration), activating and deactivating output buses, audio inputs for VST3 instruments, resizable plug-in windows, and more.

VTR. Video Tape Recorder.

VU meter. Volume Unit Meter. A mechanical metering device that displays average signal strength and is optimized to respond to loudness in the same manner the human ear does. Most VU meters don't respond well to transients, so they may be accompanied by peak LEDs to indicate quick peaks.

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W

wall wart. An external power supply consisting of a small box containing a transformer and other components that plugs directly into a wall outlet and then connects to the device it is powering with a thin cable. Wall warts are popular with manufacturers because they are inexpensive and easy to manufacture, and they move the potentially noisy transformer out of the device they are powering. Wall warts also simplify obtaining UL approval for a device. Consumers sometimes have less positive feelings; wall warts can take up more than one space on an outlet strip, are sometimes less than durable, and can fall out of a wall outlet easily. ☞ **See also** *lump in the line*.

warmth. One of those subjective terms that engineers and musicians use for describing audio. "Warmth" is usually defined as a smooth, rich sound quality, without harshness or over-brightness.

waterfall key (a.k.a. square front key). A type of key, often found on B3 organ-style instruments, that doesn't have a lip or overhanging edge. Waterfall keys make it more comfortable for the player to do a "wipe" or gliss with the palm of his hand.

watermark. An identifying mark or code embedded in the data of a document or file. Ideally, a watermark should not be obvious and should not affect the original file. Watermarks are intended to help authors and content creators control distribution of their works.

watt. The unit for power. One watt equals one amp of current flowing through one volt. It was named for James Watt, a Scottish inventor and mechanical engineer who made major improvements to the steam engine, helping to open the door for the Industrial Revolution.

WAV file. A computer audio file format developed by Microsoft for Windows applications, though WAV files are now supported by most Macintosh audio applications as well. WAV files use the .wav file suffix. ☞ **See also** *Broadcast WAV file*.

wave file. ☞ See *WAV file*.

waveform. 1. Technically, a graph of the voltage of a periodic signal plotted versus time. 2. The "shape" of a sound wave; the waveform defines the timbre of a sound.

waveguide. A part of a monitor speaker cabinet designed to physically direct or guide the sound waves that are emerging from a driver, usually a tweeter.

wavelength. 1. Technically, the distance between one peak of a repeating or cyclical sound wave and the next peak. 2. The result of dividing the speed of sound by the frequency of a repeating or cyclical sound wave. 3. The physical length of a sound wave. Wavelength is important in calculating the modes of a room. See Table W.1.

wave scanning. A synthesis technique developed by Native Instruments that uses parallel oscillators that function almost like sequencers drawing from a wave table and fading between waves.

wave sequencing. A function of some oscillators that can crossfade between waveforms while a note is being held.

wavetable. A lookup table containing the parameters and data necessary for a synthesizer to create waveforms. Although the waveforms in a wavetable begin life as samples, in the wavetable itself the sample is "stored" as the instructions necessary to create the waveform using additive synthesis.

wavetable synthesis. A digital synthesizer that creates waveforms using additive synthesis techniques based on the data contained in a stored lookup table. Most wavetable synths are able to crossfade between waveforms while notes are playing to create complex

Table W.1 Frequency versus Wavelength

Frequency	Wavelength
20 Hz	56.3 feet
60 Hz	18.8 feet
100 Hz	11.3 feet
160 Hz	7.0 feet
320 Hz	3.5 feet
500 Hz	2.3 feet
1 kHz	1.1 feet
2.5 kHz	5.4 inches
5 kHz	2.7 inches
10 kHz	1.4 inches
20 kHz	0.7 inches

raw sounds that are then processed using filters, envelopes, and other synthesis techniques.

WDM. Windows Driver Module. A Microsoft-developed driver protocol for the Windows OS platform. The WDM protocol was designed to take advantage of operating system code, DMA, and plug and play to allow greater efficiency.

weighted action. A type of keyboard action that uses weighted keys to simulate the feel of a grand piano.

weighting. When measuring audio, weighting modifies the results to (hopefully) match up better with how our ears hear. An example of weighting might be a curve applied to sound-pressure level measurements to more accurately reflect how our ears perceive loudness. Another weighting might compensate for the frequency response of our ear.

wet. A signal with processing, usually artificial reverb or delay.

wet/dry mix. The parameter that controls the blend of the un-effected (dry) signal and the processed (wet) signal output by an effects device.

White Book. One of a set of “Rainbow Books” with colored covers containing the Sony/Philips-developed specifications for different optical compact disc formats. The White Book contains the specification for Video CD.

white noise. A type of random noise signal used for testing purposes, containing equal energy per frequency. Since the number of frequencies doubles with each octave, white noise sounds somewhat like hiss because the sonic energy increases in the higher octaves.

wide cardioid (a.k.a. subcardioid). A polar pattern that falls between omnidirectional and cardioid patterns. See Figure W.1.

width. The characteristic of stereo spaciousness or the “spread” of a sound across the stereo field. A sound with a narrower width will seem to come from a single location or area in the stereo field, whereas one with a wider width will seem to fill the stereo field.

wild time code. Time code that isn’t resolved or synchronized to a reference source. One example is time code on a video that isn’t synched to the frame rate. Wild time code can make it difficult to sync audio to the video in post-production.

window. A visual interface element of a GUI operating system, somewhat like a screen within a screen that can be “opened” or accessed to display data, an editor, a process, or other information for the user.

Windows. A family of GUI-based operating systems created by Microsoft and first introduced in 1985 for personal computers. There are many different versions, each of which added an extensive array of features, enhancements, and capabilities:

- **Windows 1 through 3.** Early versions of Windows that operated on top of MS-DOS, functioning almost like GUIs for MS-DOS.
- **Windows 95.** Integrated MS-DOS and Windows systems. Introduced in 1995.
- **Windows 98.** Hybrid 16-/32-bit OS introduced in 1998.
- **Windows NT.** The first fully 32-bit Windows OS, introduced in 1993. NT supported multi-processing and multiple users.
- **Windows 2000 (a.k.a. Win2K).** Introduced in 2000 as part of the NT family of Windows operating systems. There were four versions aimed at different markets (Professional, Server, Advanced Server, and Datacenter Server).

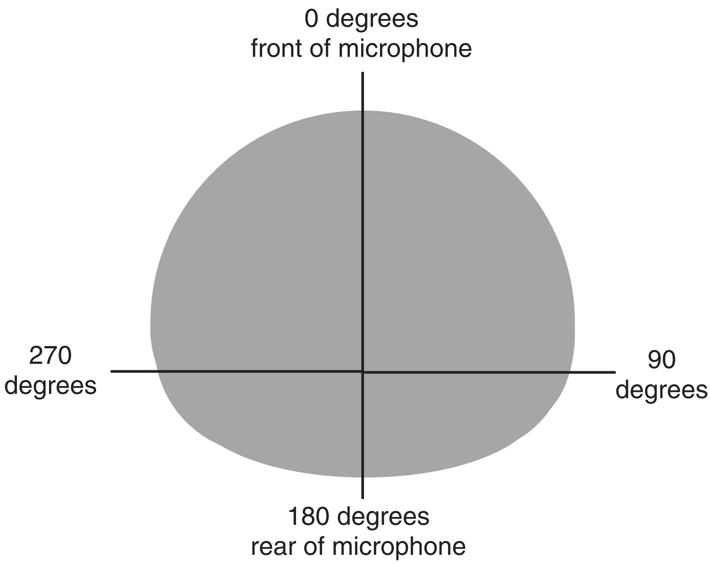

Figure W.1 The wide cardioid polar pattern is more directional than an omnidirectional pattern, but it picks up more sound from the rear than a true cardioid pattern.

- **Windows Me.** Short for Windows Millennium. A hybrid 16-/32-bit OS introduced in 2000 and targeted at home users.
- **Windows XP.** Short for Windows Experience. A 32-/64-bit OS introduced in 2001. There were a number of versions, including Windows XP Home, XP Professional, XP Media Center, XP Tablet Edition, XP 64-bit Edition, XP Professional x64, and XP Embedded.
- **Windows Vista.** A family of six 32-/64-bit operating systems released in 2007, including Vista Starter (32-bit only), Vista Home Basic, Vista Home Premium, Vista Business, Vista Enterprise, and Vista Ultimate.
- **Windows 7.** The successor to Windows Vista, at this writing slated for release in 2009 or 2010.

windscreen. A covering, usually foam, that slips over a microphone and shields it from picking up wind noise.

wireless. A system that transmits audio or data using radio or infrared waves rather than physical cable connections.

wizard. 1. A "helper" program that guides a user through installing or using another piece of software, usually with simple step-by-step instructions and diagrams/graphics. 2. A help function in a hardware device that assists the user in finding the best presets for a particular application.

WMA. Windows Media Audio. An audio data compression format originally developed by Microsoft to compete with MP3. WMA is widely used on portable music players. Later versions, such as WMA 10, added features such as stereo 24-bit/96 kHz support and 5.1 and 7.1 surround support.

woodshedding. Slang term for practicing, usually a musical instrument.

woofer. The low-frequency driver or transducer in a multi-driver speaker system.

word. A grouping of data bits. In digital audio, a word is one complete sample, regardless of the bit resolution.

word clock. 1. A high-resolution digital signal used to control the sample rate of a piece of gear, or the speed at which digital words are transmitted. 2. A type of I/O connection found on some digital audio gear that is used to synchronize the piece of gear or the entire system to an external master clock.

word length. The bit depth, or number of bits, in a word.

workstation. A term that is applied to a wide variety of studio sound creation and manipulation devices, including computer and standalone DAWs, keyboards with built-in sequencers and audio capabilities, MPC-style "groove" devices, desktop computer rigs, and more.

WORM. Write Once, Read Many. A type of CD drive and media that "closes" the disc after it is written or burned, at which point no more data can be added or changes made.

wow. A slow change in pitch or frequency due to speed fluctuations (one or two per second) in a turntable, tape recorder/player, or other mechanical playback device. ☞ **See also** *flutter*.

wrapper. Software that "encloses" another piece of software and allows it to run on another platform. Wrappers are commonly used to make plug-ins of one format compatible with another format without rewriting the actual plug-in code. For example, a wrapper might be used to allow a VST-format plug-in to run under RTAS format in Pro Tools.

write. 1. To "burn" or record data or audio to a recordable CD or DVD. 2. To store data on a hard drive or other media. 3. To create a breakpoint or curve in a dynamic automation system.

write protect. To "lock" a piece of media, a drive, or a file so it cannot be modified or overwritten.

X

XLR. A locking three-pin connector, originally developed by Cannon, often used to carry balanced audio signals. "XLR" comes from Cannon's part designation: "X-series, Latch, Rubber." For balanced audio, pin 1 of an XLR connector is always ground, while pin 2 or 3 can carry the "hot" (normal) signal (the current accepted standard is pin 2 hot), with the remaining pin carrying the "cold" (inverted) signal. (☞ **See also** *balanced*.)

XMF. Extensible Music File. A file specification adopted by the MMA (*MIDI Manufacturers Association*) that allows for playback of a single file within a computer or player. An XMF file wraps and contains all necessary MIDI information (in the form of an SMF—*Standard MIDI File*) along with DLS instrument files and WAV or other digital audio files. The XMF format is designed for easy transportability between platforms as well as fast downloads. There are two types: Type 0 and Type 1. The only difference is that MIDI information can be streamed using Type 0 XMF files.

XY stereo. A stereo microphone technique where two identical coincident cardioid microphones are angled at 90 degrees to one another (see Figure X.1). Because the mic capsules are placed very close together, they are phase coherent and offer good mono compatibility. Since most of the sound is actually picked up off-axis to the microphones, XY stereo miking does not result in an extremely wide stereo field, but it does produce a solid center image without pickup of too much room ambience. The audio quality of the stereo signal will depend on the off-axis response characteristics of the microphones used.

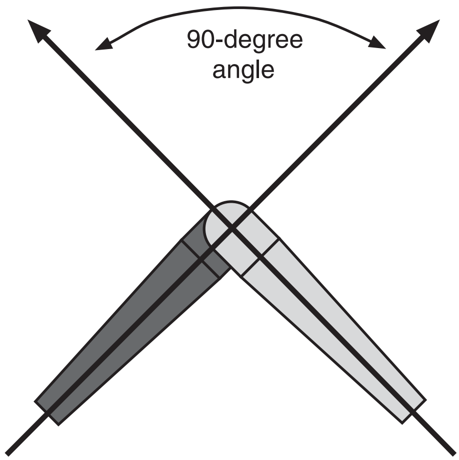

Figure X.1 For XY stereo, two identical cardioid microphones are placed at 90 degrees to one another.

Y

Y cable. A cable that splits a single connection into dual connections. In a few cases, a Y cable can also do the opposite—combine two signals into one. Care must be taken in using Y cables—especially when attempting to combine signals—to avoid impedance and level mismatches, or even potential damage to equipment.

Yellow Book. One of a set of "Rainbow Books" with colored covers containing the Sony/Philips-developed specifications for different optical compact disc formats. The Yellow Book contains the specifications for CD-ROM and CD-ROM XA.

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Z

Z. The standard symbol for electrical impedance. Hi-Z refers to high impedance; lo-Z refers to low impedance. ☞ **See also** *impedance*.

zenith. The angle of a tape recorder head in relation to the recorder's top plate.

zero crossing. The point where an analog voltage changes from positive to negative or negative to positive polarity (see Figure Z.1). Zero crossings are often the best places to loop a sample or to edit two waveforms together, because the levels are zero and will not introduce a click or pop.

zero latency. A feature of some software and hardware DAWs that eliminates delays introduced into the signal. It takes a certain amount of time for an analog signal being recorded to enter an audio interface, pass into the computer, return from the converter, and exit the converter as an analog signal again. This delay can cause timing problems for musicians when overdubbing parts alongside existing tracks. Various schemes have been introduced for reducing latency, such as Steinberg's ASIO protocol and others. Current hardware interfaces have features that support direct monitoring, where the signal is monitored before it passes into the computer, eliminating delays in the signal path (see Figure Z.2).

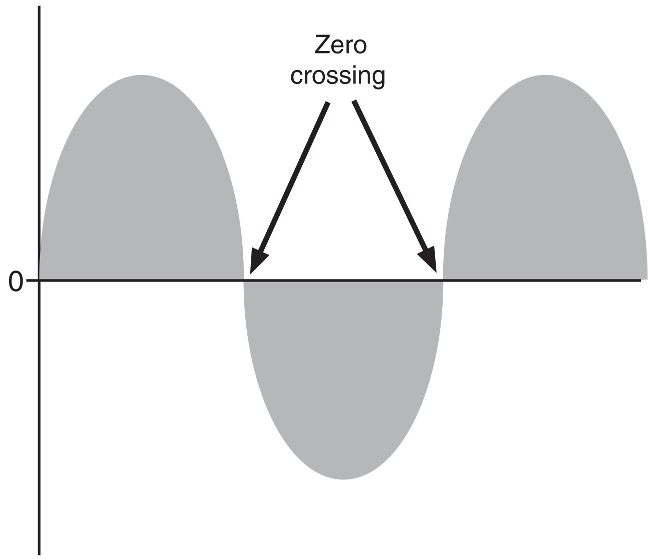

Figure Z.1 A zero crossing is the point when the voltage in an analog waveform switches polarity.

zero reference. A signal level defined as the nominal operating level of a device. This level varies; in a +4 dBu balanced system, the zero reference will be set so that a meter reads 0 VU when a +4 dBu signal is applied. Likewise, in a –10 dBv system, the zero reference will be set so that a meter reads 0 VU when a –10 dBv signal is present. With digital systems, the meters are generally calibrated to allow for sufficient headroom. For example, a 0 VU signal from an analog device might read anywhere from –18 to –12 dB on a digital meter, allowing 2–3 bits of headroom in the digital system. This also means that a 0 dB (full code) output signal from a digital device might be as high as +24 or +28 dB on an analog system.

ZIF socket. Zero Insertion Force socket. A type of mounting socket that allows integrated circuit chips to be quickly and easily changed. This allows the manufacturer or user to easily modify or customize performance.

Zip. 1. A data-compressed computer file format. Zip files, called *archives*, are created using data compression software,

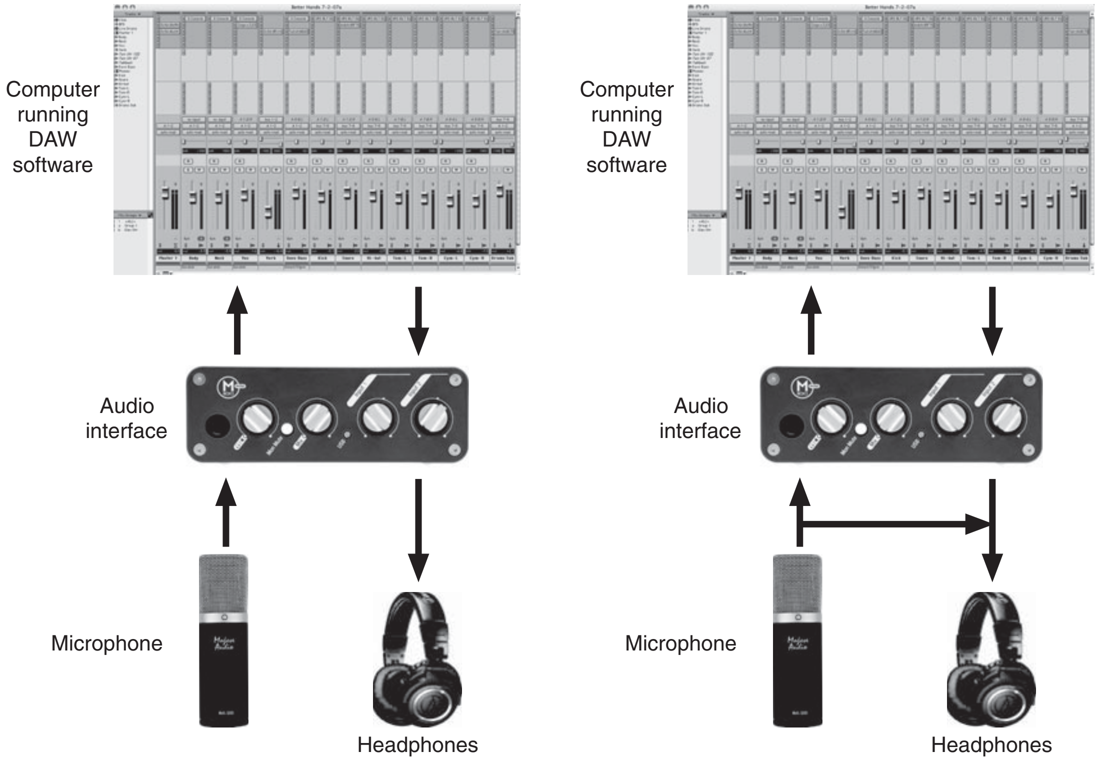

Figure Z.2 When recording with computer-based systems, the signal must enter the audio interface, pass through the computer/software, and come back out through the audio interface—all of which takes a certain amount of time (left). Some audio interfaces, however, have a feature that allows signal from the input to be fed straight to the output, eliminating the delay (right).

and then must be extracted by another piece of software before they can be used. Zipping files can result in a substantial reduction in file size from the original (though often not with audio files) and increased robustness during downloads and when e-mailing files. 2. An obsolete removable disk drive storage system developed by Iomega.

zip cable (a.k.a. lamp cord). Light-gauge, inexpensive speaker wire.

zipper noise. Clicking noise heard when the resolution of a digital control signal is not high enough to smoothly change a parameter (such as filter cutoff, pitchbend, or other settings) in a device. See Figure Z.3.

zone. A portion of a keyboard, or a range of MIDI notes. Zones are used to split a keyboard so that different ranges of notes can have different sounds, such as a bass on the bottom octave, strings in the middle of the keyboard, and a flute in the top octave (see Figure Z.4). ☞ **See also** *layer*.

zoom. Zoom in: Increasing the size of a waveform or other audio or MIDI data on a computer or LCD screen to allow for better visibility and finer editing (see Figure Z.5). Zoom out: Reducing the size of an item being displayed so that it takes up less screen real estate. Various "tools" may be used to zoom, including a magnifying glass, zoom controls, key commands, and menu selections.

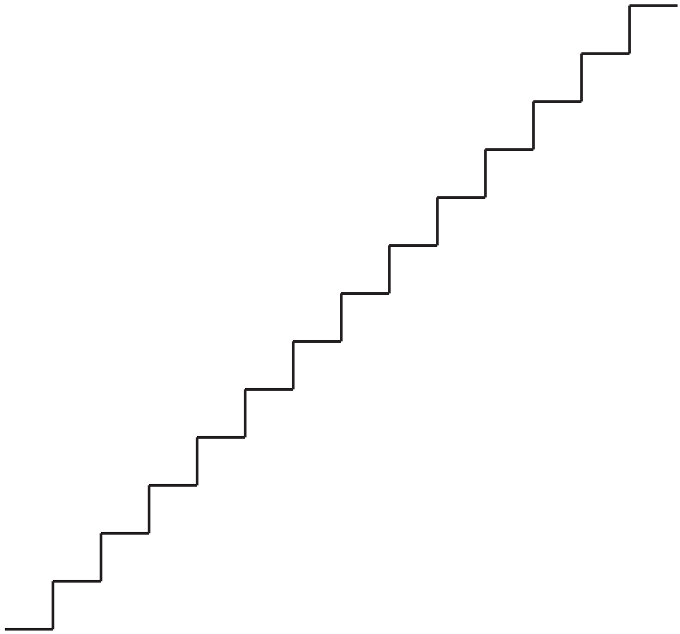

Figure Z.3 An analog signal is smooth and continuous. A digital signal is stepped as the value changes. Zipper noise results when a digital control signal lacks resolution to smoothly change a parameter.

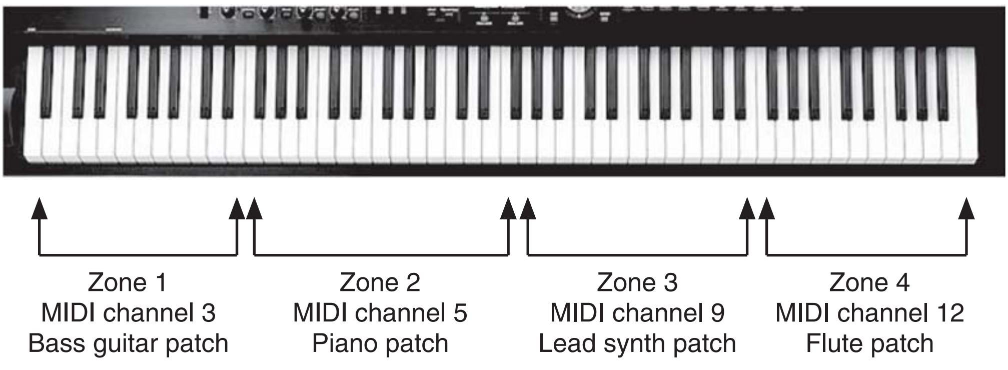

Figure Z.4 Many keyboards can be split into zones, where each area of the keyboard can have its own assigned sound, MIDI channel, and other parameters. In many cases zones can overlap, allowing sounds to be layered.

zoom preset. A feature of some software that allows different zoom in/out levels to be stored so they can be recalled instantly as needed.

Z-plane Synthesis. A proprietary synthesis system developed by E-MU and first introduced in the Morpheus synth module. Z-plane Synthesis uses unique filters to shape the sound of sampled waveforms. Each filter contains seven sections, which together can model almost any resonant characteristic, including vocal formants. In

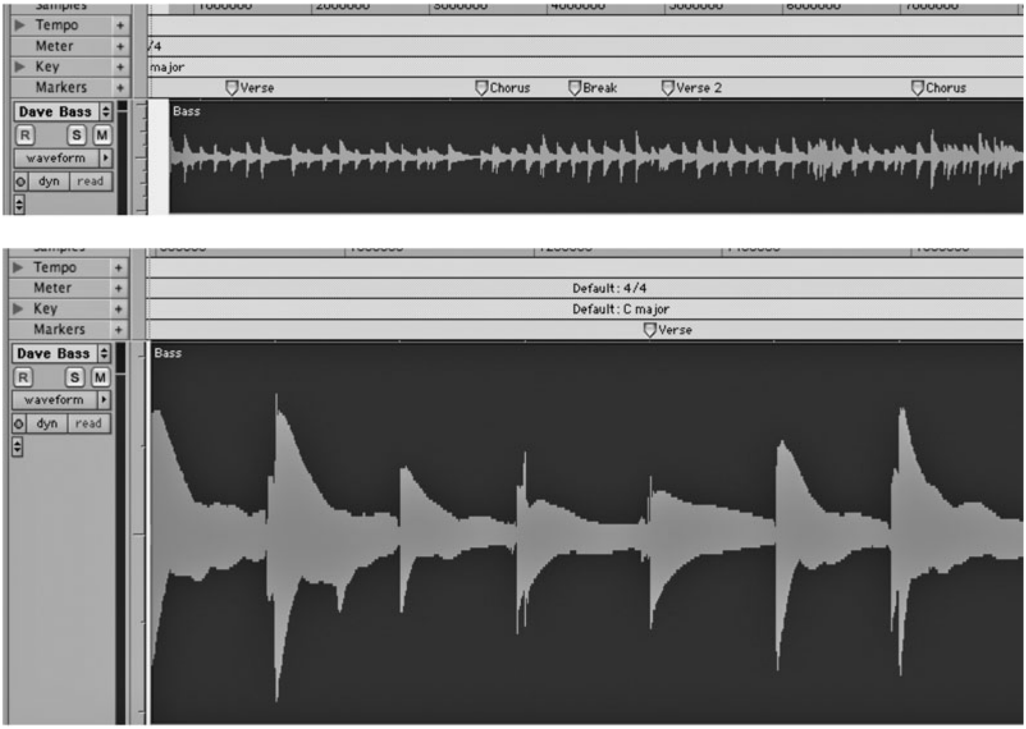

Figure Z.5 DAW and audio editing programs allow the user to increase the size of waveforms and other displays to allow finer editing. A track can be viewed with a low zoom level (top) to show more of the track, or the view can be zoomed in (bottom) to allow a more detailed view.

addition, the filters can interpolate, or "morph," between various resonant characteristics under the control of real-time controllers, such as velocity, aftertouch, and others. Three parameters can be controlled using a three-axis system (X, Y, and Z axes) for additional real-time control over the filters and morphing.

0–9

−10 dBv. The semi-professional audio reference or nominal level. The value –10 dBv equals 0.316 volts, given the 0 dBu reference is 0.775 volts. In practice, –10 dBv gear tends to have unbalanced connections. Note that –10 dBv gear is not directly compatible with +4 dBu gear. A level-matching interface should be used when interconnecting gear of the two levels. Otherwise, +4 dBu outputs will overload –10 dBv inputs, and –10 dBv outputs will not sufficiently drive +4 dBu inputs to prevent noise problems. In reality, –10 dBv gear is not inherently better or worse than +4 dBu gear; however, manufacturers originally applied the –10 dBv level to equipment to save money, so it has become associated with budget, less-than-professional equipment.

1-bit. A high sample rate digital audio recording and playback technology that measures the change in amplitude between samples, which can be represented with one bit, rather than taking a complete 16- or 24-bit resolution measurement of the amplitude for each sample. One-bit samples are taken at rates up to 5.6448 GHz. At these rates, steep filters are not required, wider frequency response is possible, and there is no need for decimation filtering or interpolation. ☞ **See also** *direct stream digital*.

12at7 (a.k.a. ECC81). A dual-triode vacuum tube designed to provide medium gain levels (typical gain factor of 60), and used for certain driver and phase inverter applications in guitar and bass amps and other audio devices.

12au7 (a.k.a. ECC82). A dual-triode vacuum tube designed to provide low gain levels (typical gain factor of 19), and used for certain driver and phase inverter applications in audio devices.

12ax7 (a.k.a. ECC83). A dual-triode vacuum tube designed to provide high gain levels (typical gain factor of 100), and used for preamp applications in guitar and bass amps, as well as microphone preamps and other processing devices.

12ay7 (a.k.a. 6072). A dual-triode vacuum tube designed to provide low gain levels (typical gain factor of 44), and used for certain driver and phase inverter applications in audio devices.

1×12. A speaker cabinet, typically for electric guitar, that contains one 12-inch speaker.

1-inch. A standard width for analog recorder tape. The 1-inch format has been used for high-end 2-track mixdown machines, as well as 8-, 16-, and 24-track multitrack recorders.

1/2-inch. A standard width for analog recorder tape. The 1/2-inch format has been used for 2-track mixdown machines, as well as 8- and 16-track multitrack recorders. The 1/2-inch format was also used on analog 4-track recorders that were used as masters for film and video mixes.

1/2-space. ☞ **See** *half space*.

1/2-track. ☞ **See** *half track*.

1/3-octave. An equalizer that has its frequency bands spaced one-third of an octave apart. This results in the EQ having 30 or 31 bands of equalization.

1/4-inch. A standard width for analog recorder tape. The 1/4-inch format has been used for 2-track mixdown machines, as well as 8-track multitrack recorders.

1/4-space. ☞ **See** *quarter space*.

1/4-track. ☞ **See** *quarter track*.

1/8-space. ☞ **See** *eighth space*.

15-band. ☞ **See** *2/3-octave*.

16-bit. Having a digital resolution of 65,536 possible values to represent an analog audio signal and stored as 16 bits of data. (Technically, there are 32,767 positive and 32,767 negative values to represent the positive and negative polarities of an audio waveform.) Since each bit provides about six decibels of dynamic range, 16-bit systems have a theoretical dynamic range of 96 dB.

16-track. A multitrack tape recorder that can record and play up to 16 tracks of audio. Sixteen-track machines were available that recorded to 1/2-inch, 1-inch, and 2-inch reel-to-reel analog tape. Analog 16-track machines were popular for both home and professional studio use.

2.1. A stereo speaker system consisting of two small "satellite" speakers, which produce mid and high frequencies, and a subwoofer, which produces low frequencies. These systems have gained popularity because the satellites can be compact and unobtrusive, while the subwoofer provides plenty of low-end output that would normally be impossible to achieve from a small speaker system.

2×10. A type of electric bass or electric guitar speaker cabinet that contains two 10-inch speakers.

2×12. A speaker cabinet, typically for electric guitar, that contains two 12-inch speakers.

2-bus. ☞ See *stereo bus*.

2-inch. The standard width for 24-track analog recorder tape.

2-track. A tape recorder that can record and play two tracks of audio. Two-track machines were popular in many studios for mixdown and mastering. Two-track machines were available that recorded to 1/4-inch, 1/2-inch, and 1-inch reel-to-reel analog tape. In some cases, 1/4-track machines are referred to as *2-track* recorders, because even though the tape holds four total tracks, only two are available in each direction.

2-way (a.k.a. bi-amp). A sound system or monitor that divides the full-range frequency range into two smaller ranges, one for low frequencies and one for high frequencies, using a crossover. A separate, dedicated power amplifier and speaker driver are used for each of the ranges. The idea is to optimize the driver and amp for the requirements of each range to improve efficiency and sound quality.

2/3-octave. An equalizer that has its frequency bands spaced two-thirds of an octave apart. This results in the EQ having 15 bands of equalization.

2:1 rule. A rule of thumb for recording that states that, in order to pick up the same amount of room ambience, a cardioid pattern microphone needs to be two times as far from a sound source as an omnidirectional microphone.

20-bit. Having a digital resolution of 1,048,576 possible values to represent an analog signal measurement and stored as 20 bits of data. (Technically, there are 524,287 positive and 524,287 negative values to represent the positive and negative polarities of an audio waveform.) Since each bit provides about six decibels of dynamic range, 20-bit systems have a theoretical dynamic range of 120 dB, though in many cases, 20-bit converters really only had an effective linear capability of 16 bits; the remaining four bits were not linear and were often discarded.

24-bit. Having a digital resolution of 16,777,216 possible values to represent an analog signal measurement and stored as 24 bits of data. (Technically, there are 8,388,607 positive and 8,388,607 negative values to represent the positive and negative polarities of an audio waveform.) Since each bit provides about six decibels of dynamic range, 24-bit systems have a theoretical dynamic range of 144 dB.

24-track. A multitrack tape recorder that can record and play up to 24 tracks of audio. Analog 24-track machines were primarily used in professional recording studios. Twenty-four-track machines were available that recorded to 1-inch and 2-inch reel-to-reel analog tape.

25-key. A compact keyboard with 25 keys, or two octaves, often used for portable performance applications. Most 25-key keyboards have synth actions, as opposed to weighted or hammer actions.

3-way (a.k.a. tri-amp). A sound system or monitor that divides the full-range frequency range into three smaller ranges, one for low frequencies, one for midrange, and one for high frequencies, using a crossover. A separate, dedicated power amplifier and speaker driver are used for each of the three ranges. The idea is to optimize the speaker and amp for the requirements of each range to improve efficiency and sound quality.

3 dB down point (a.k.a. −3 dB point, cutoff frequency, half-power bandwidth). The frequency where the output level of a filter is 3 dB down compared to the input level or the passband (see Figure 1). The

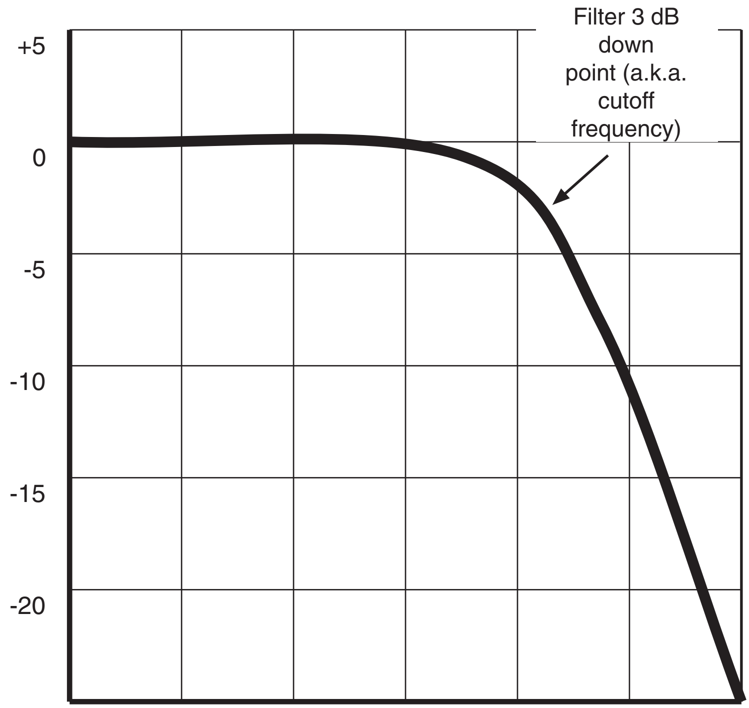

Figure 1 A filter's (in this case, a low-pass filter) –3 dB points are where the response of the filter is 3 dB below the passband.

point where the stopband and the passband of a filter meet. ☞ **See also** *cutoff frequency*.

3.5mm (a.k.a. 1/8-inch, though this is approximate). A small-format version of two- or three-conductor audio connector, often used for headphone jacks and plugs on consumer and some semi-professional audio equipment. A two-conductor (TS or tip-sleeve) 3.5mm connector can carry one unbalanced signal. A three-conductor (TRS or tip-ring-sleeve) 3.5mm connector can carry two unbalanced signals or one balanced signal.

31-band. ☞ See *1/3-octave*.

32-bit floating point. ☞ See *floating point*.

37-key. A compact keyboard with 37 keys, or three octaves, often used for portable performance applications. Most 37-key keyboards have synth actions, as opposed to weighted or hammer actions.

3:1 rule. A rule of thumb developed for placing two or more mics on a single sound source. The 3:1 rule states that when two microphones are picking up the same sound source, the second mic should be placed at least three times as far from the first mic as the first mic is from the source (see Figure 2). So if the first mic is one foot from the source, the second mic should be at least three feet from the first mic. The idea is to minimize phase cancellation problems resulting from short time delays between the two mics. The 3:1 rule works because the signal reaching the second mic will be 9 dB quieter than the signal reaching the first mic, which substantially reduces the effects of any cancellations.

+4 dBu. The professional audio reference or nominal level. The value +4 dBu equals 1.23 volts, given the 0 dBu reference is 0.775 volts. In practice, +4 dBu gear tends to have balanced connections. Note that +4 dBu gear is not directly compatible with –10 dBv gear. A level-matching interface should be used when interconnecting gear of the two levels. Otherwise, +4 dBu outputs will overload –10 dBv

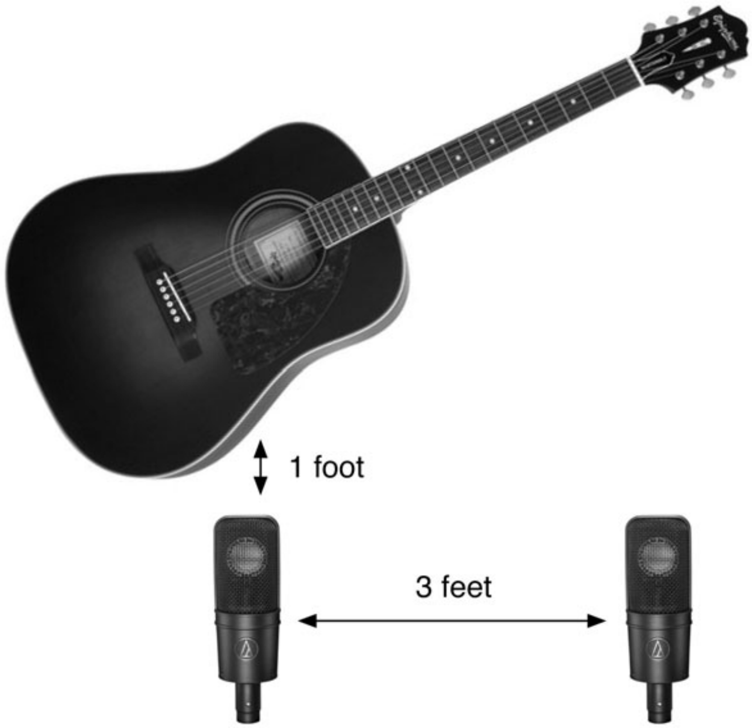

Figure 2 The 3:1 rule for microphone placement says that when two mics are on a source, the second mic needs to be three times as far from the first mic as the first mic is from the source.

inputs, and –10 dBv outputs will not sufficiently drive +4 dBu inputs, which can result in noise problems. +4 dBu gear is not inherently better than –10 dBv gear; however, manufacturers originally applied the –10 dBv level to equipment to save money, so it has become associated with budget and consumer equipment.

4×10. A type of electric bass or electric guitar speaker cabinet that contains four 10-inch speakers.

4×12. A speaker cabinet, typically for electric guitar, that contains four 12-inch speakers.

4-track. A multitrack tape recorder that can record and play up to four tracks of audio. Analog 4-track machines were popular for home and project studios. Analog cassette and 1/4-inch reel-to-reel versions were available.

48-volt. The standard voltage used to power condenser and active microphones via phantom power.

49-key. A compact keyboard with 49 keys, or four octaves, often used for portable performance or studio applications. Most 49-key keyboards have synth actions, as opposed to weighted or hammer actions.

5.1. Surround sound reproduction system consisting of five identical speakers with a dedicated LFE speaker. A 5.1 surround system uses front left, center, and right speakers and left and right surround speakers, along with a subwoofer for the LFE output.

61-key. A keyboard with 61 keys, two octaves shorter than the keyboard found on most grand pianos. Most 61-key keyboards use unweighted synth action, as opposed to weighted or hammer actions.

6L6. A common tetrode vacuum tube used in guitar and power amplifiers.

6V6. A common tetrode vacuum tube used in guitar and power amplifiers, designed for applications where the 6L6 was too powerful.

7.1. A surround sound format consisting of seven identical full-range channels/speakers with one

LFE channel. A 7.1 surround system uses front left, center, and right speakers, left and right surround speakers, and left and right rear surround speakers.

76-key. A keyboard with 76 keys, one octave shorter than the keyboard found on most grand pianos. Most 76-key keyboards have a semi-weighted action, as opposed to true weighted or hammer actions.

8×10. A speaker cabinet, typically for bass guitar, that contains eight 10-inch speakers.

8-bit. Having a digital resolution of 256 possible values to represent an analog signal measurement, which is stored in eight bits of memory. Since each bit provides about six decibels of dynamic range, 8-bit digital devices have a theoretical dynamic range of 48 dB, about the same as a cassette tape.

8-track. 1. A multitrack tape recorder that can record and play up to eight tracks of audio. Both analog and digital 8-track machines were popular for home and project studios. Analog 8-track machines were available that recorded to analog cassette, 1/4-inch reel-to-reel, 1/2-inch reel-to-reel, and 1-inch reel-to-reel. Digital 8-track machines included the Alesis ADAT, which recorded to SVHS tape, and the TASCAM DA-88, which recorded to Hi-8 tapes. 2. A type of consumer tape cartridge system popular in the 1970s.

88-key. A keyboard with 88 keys, the same as is found on most grand pianos. Most 88-key keyboards have weighted or hammer actions.

802.11. One of a family of wireless specifications used for LAN (*Local Area Network*) use. There are several types, which differ in their transmission frequency and speed: 802.11, 802.11a, 802.11b, 802.11g, 802.11n, and 802.11y.

9-pin (a.k.a. Sony 9-pin, though other companies had their own versions). A type of D-sub connector used for synchronization and control protocols by Sony and other companies. The Alesis ADAT also used the same connector for its synchronization and control protocols.